



IPv6/IPv4 Translation for SIP Applications- Socket-Layer Translator and SIPv6 Translator

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Outline

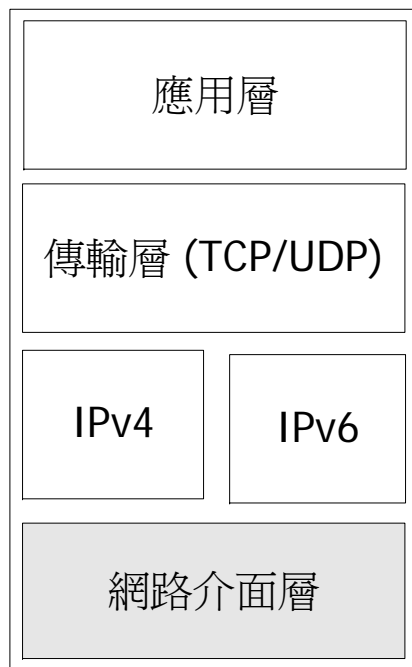
- Introduction to IPv6 Transition Mechanisms
- Socket-layer Translator
- SIPv6 Translator
- Conclusion

IPv4-to-IPv6 Transition Strategy

- Dual Stack
 - Reduce the cost invested in transition by running both IPv4/IPv6 protocols on the same machine .
- Tunneling
 - Reduce the cost in wiring by re-using current IPv4 routing infrastructures as a virtual link.
- Translation
 - Allow IPv6 realm to access the rich contents already developed on IPv4 applications

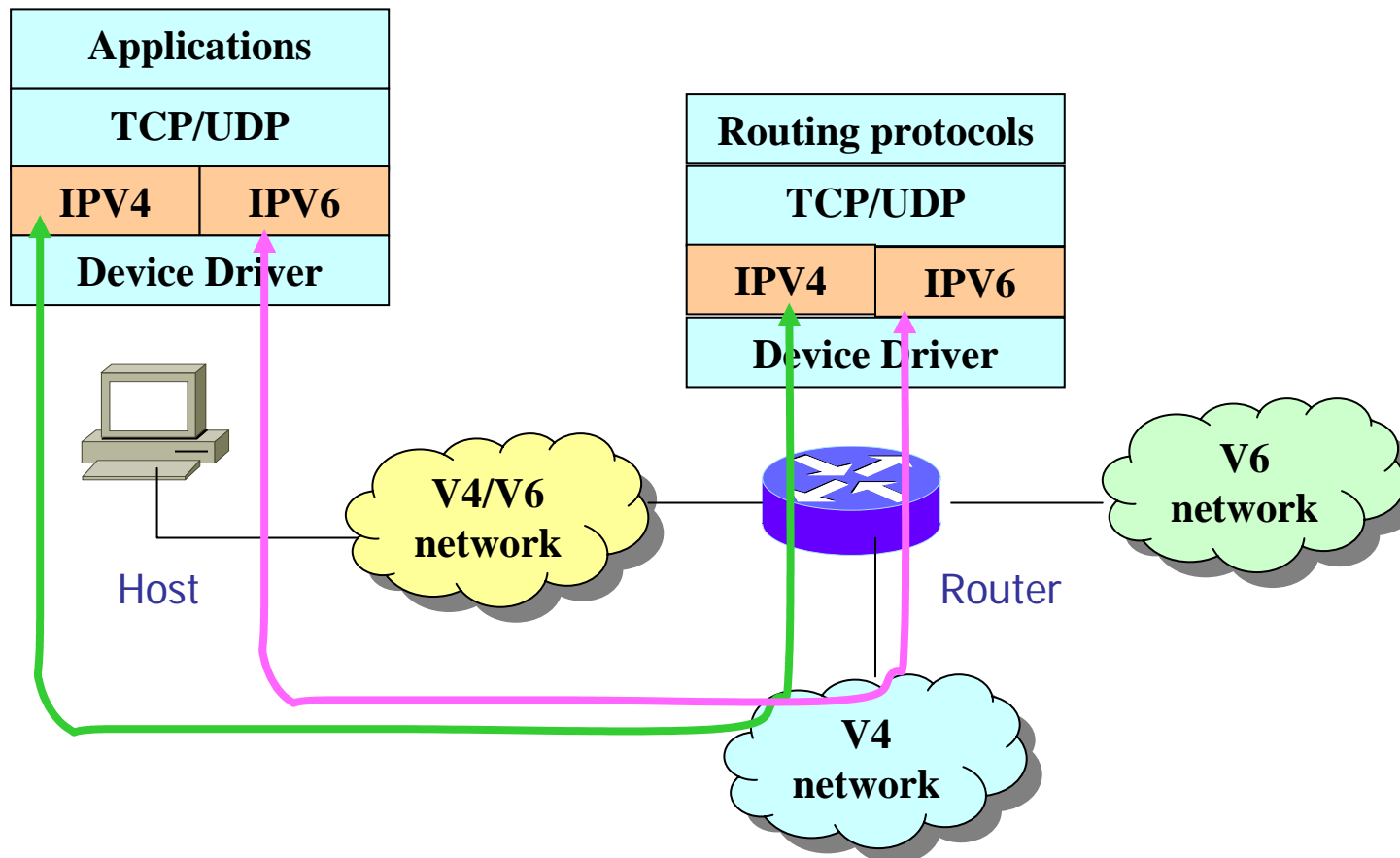
Dual Stack (RFC 2893)

Dual-stack



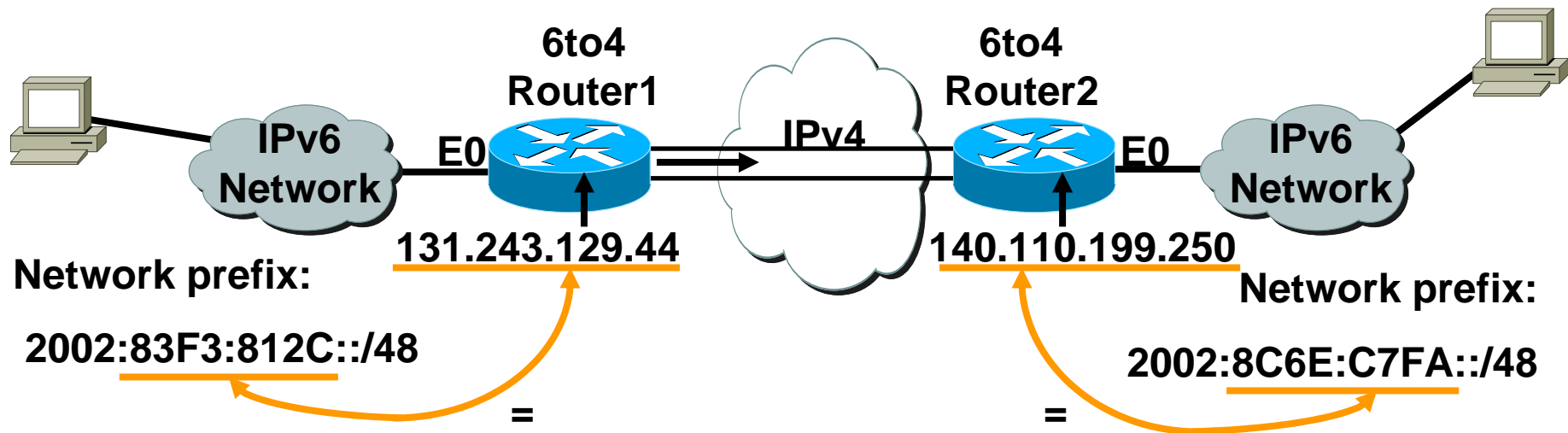
- Dual Stack表示同時裝有IPv4與IPv6兩個通訊協定堆疊。
- 當兩台機器通訊時，可以使用IPv4通訊，或是IPv6通訊。
- 當IPv6已經廣泛地被使用時，就可以移除IPv4通訊協定，而全面使用IPv6協定。

Dual Stack Example



■ 如果所有的機器都是Dual Stack，還需要其他的轉換機制嗎？

6to4 Tunnel (RFC 3056)



6to4 Tunnel:

- Is an automatic tunnel method
- Gives a prefix to the attached IPv6 network
- 2002::/16 assigned to 6to4
- Requires one global IPv4 address on each site

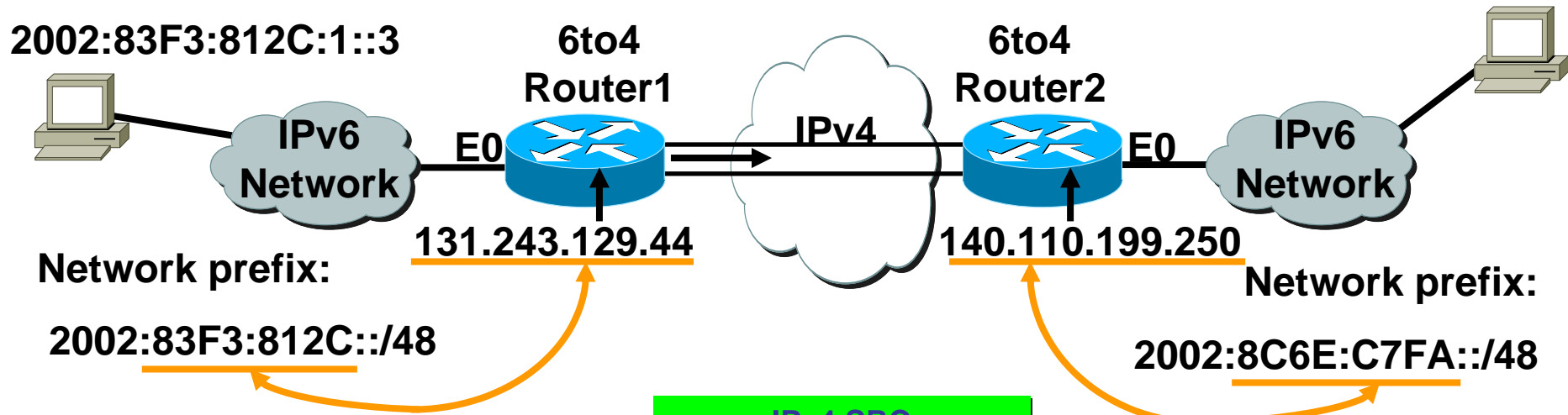
```

router2#
interface Ethernet0
 ip address 140.110.199.250 255.255.255.0
 ipv6 address 2002:8C6E:C7FA:1::/64 eui-64
interface Tunnel0
 no ip address
 ipv6 unnumbered Ethernet0
 tunnel source Ethernet0
 tunnel mode ipv6ip 6to4

ipv6 route 2002::/16 Tunnel0
    
```

6to4 Tunnel Example

2002:8C6E:C7FA:2::5

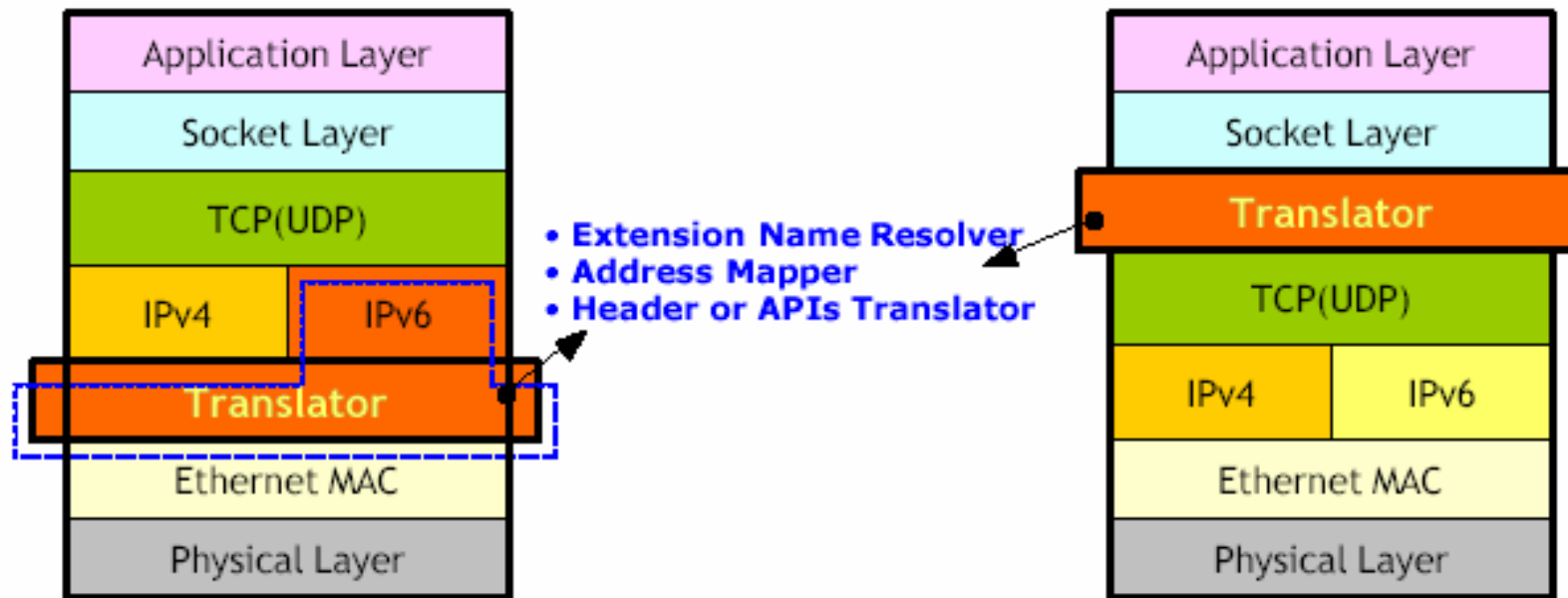


IPv6 SRC 2002:83F3:812C:1::3
IPv6 DEST 2002:8C6E:C7FA:2::5
Data

IPv4 SRC 131.243.129.44
IPv4 DEST 140.110.199.250
IPv6 SRC 2002:83F3:812C:1::3
IPv6 DEST 2002:8C6E:C7FA:2::5
Data

IPv6 SRC 2002:83F3:812C:1::3
IPv6 DEST 2002:8C6E:C7FA:2::5
Data

Address Translations



BIS

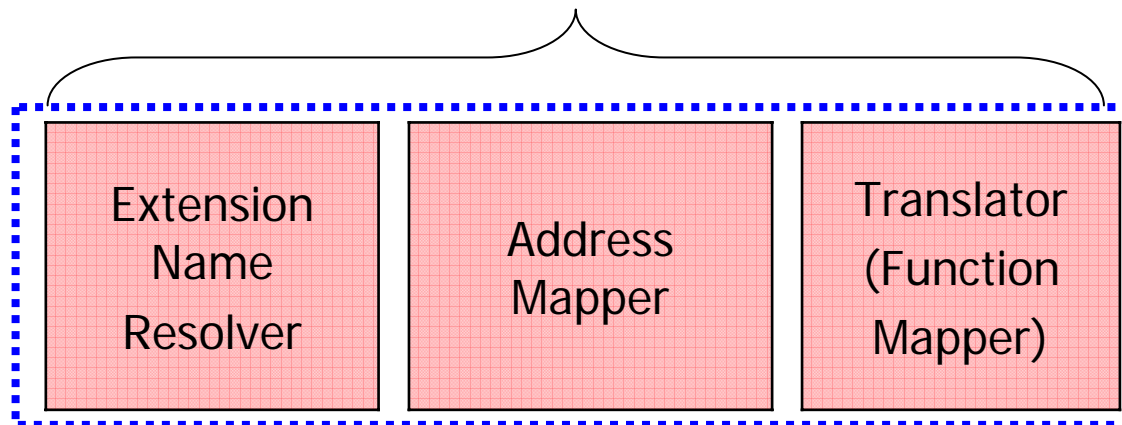
Bump-In-the-Stack

BIA

Bump-In-the-Application

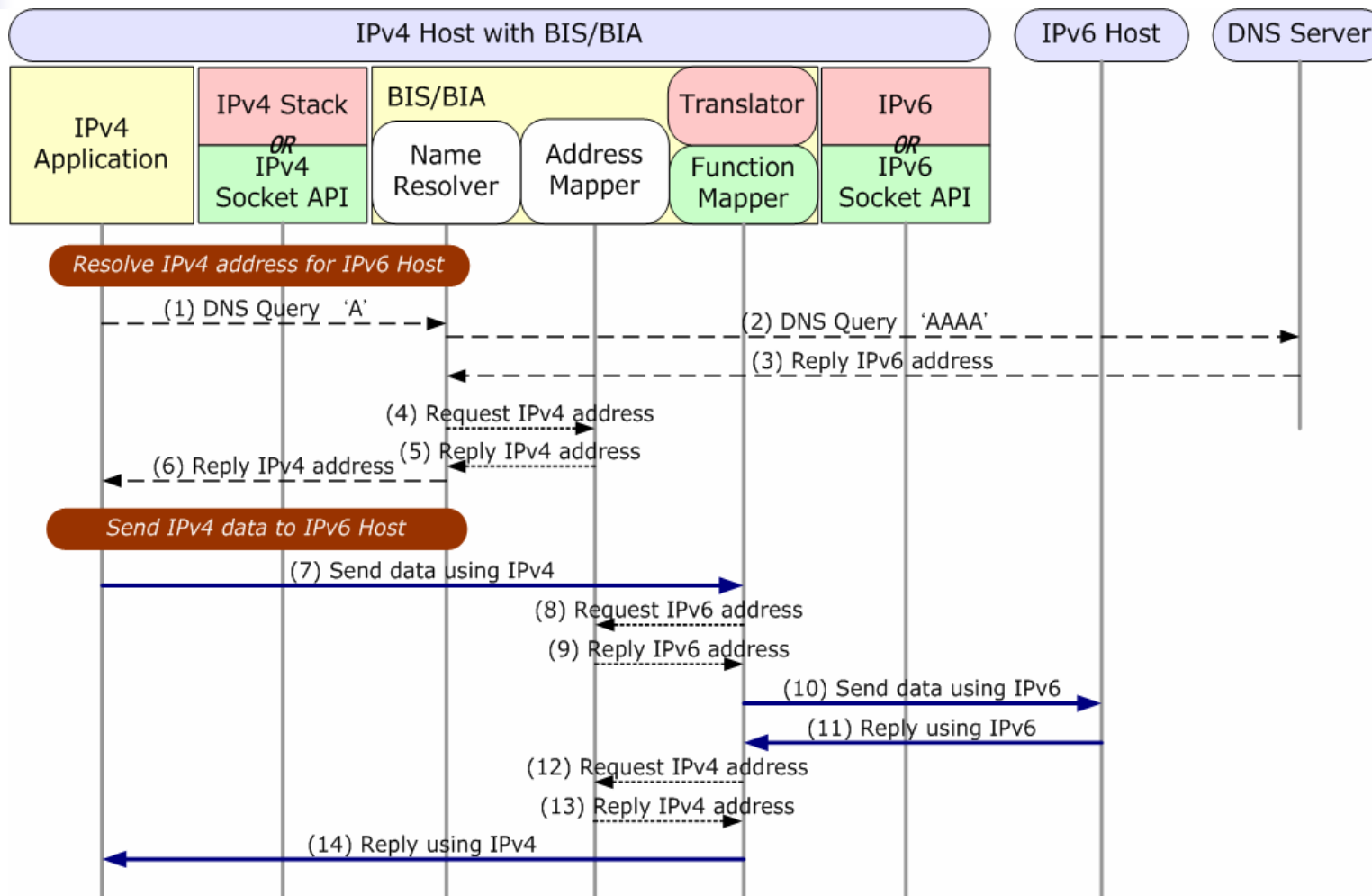
Key Components for BIS/BIA

BIS or BIA Functions

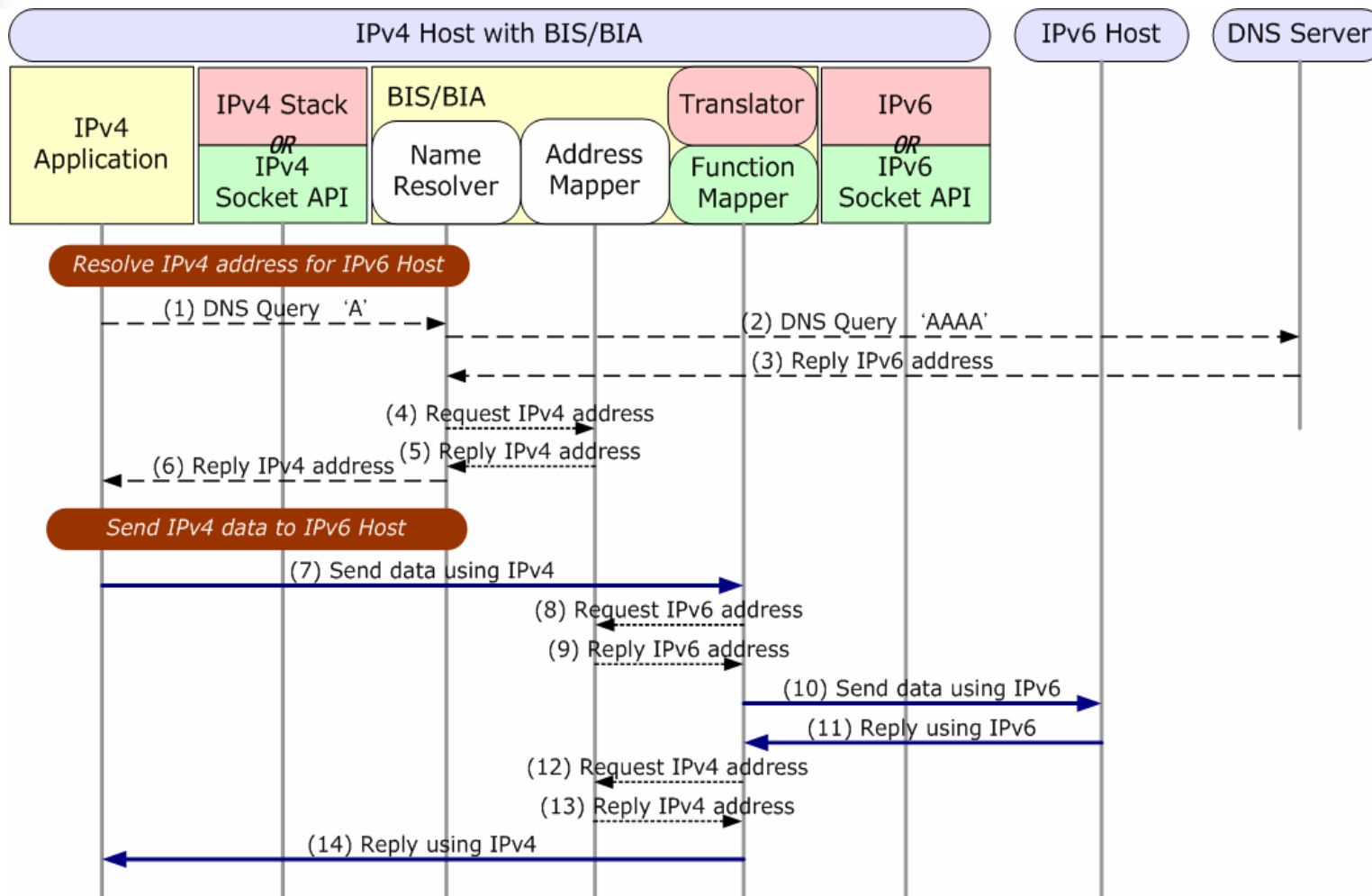


Translation Middleware

BIS/BIA name resolving flow



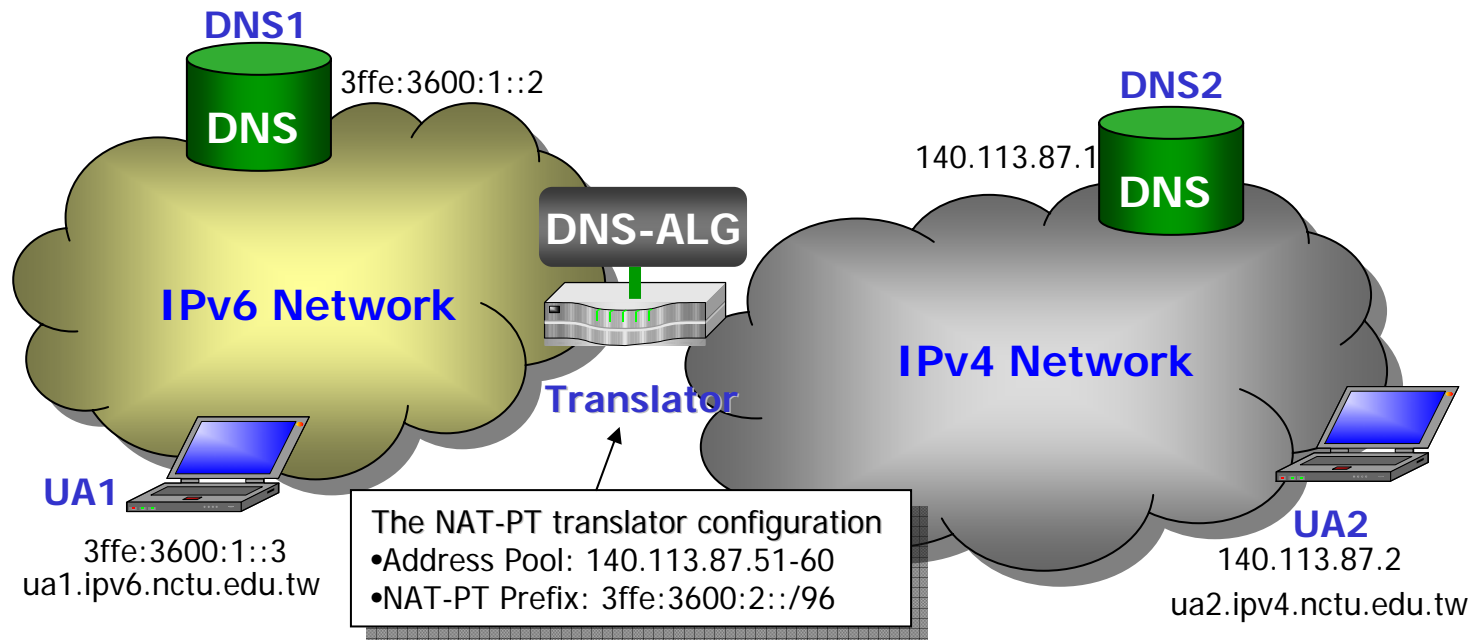
BIS/BIA Translation flow



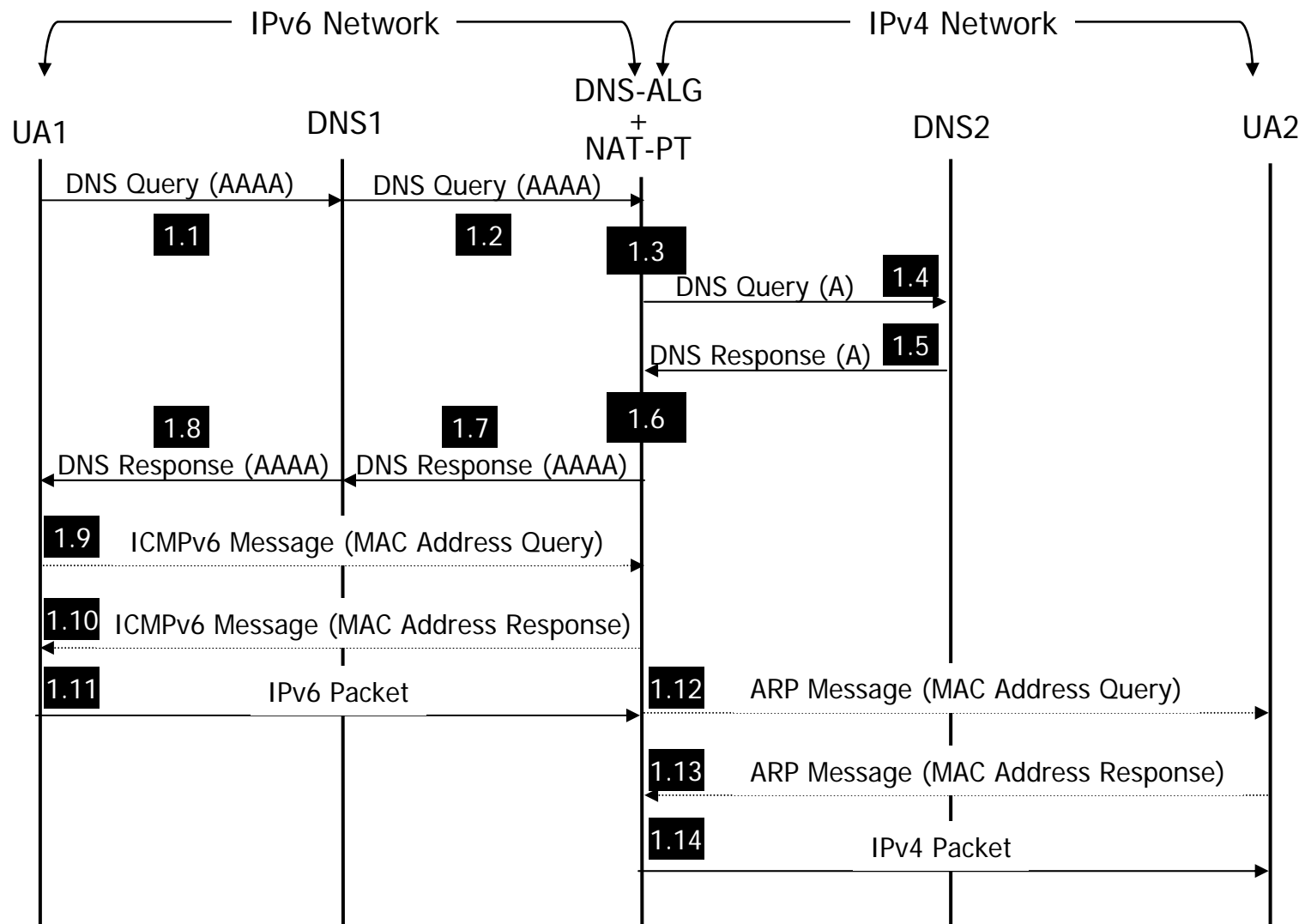
Introduction to NAT-PT

- NATPT (Network Address Translation and Protocol Translation)
 - Translate IPv6 to IPv4 headers, and vice versa.
 - Also includes ICMP headers
- NATPT (NATPT with Port translation)
 - Network Address and Protocol Translation with Port Translation
 - Various IPv6 clients can utilize one IP to connect Internet
- ALG (Application Layer Gateway)
 - ALG assists NAT to deal with upper-layer protocols
 - Translate Upper-layer protocols (such as: DNS, FTP)
 - Also called DNS_ALG and FTP ALG

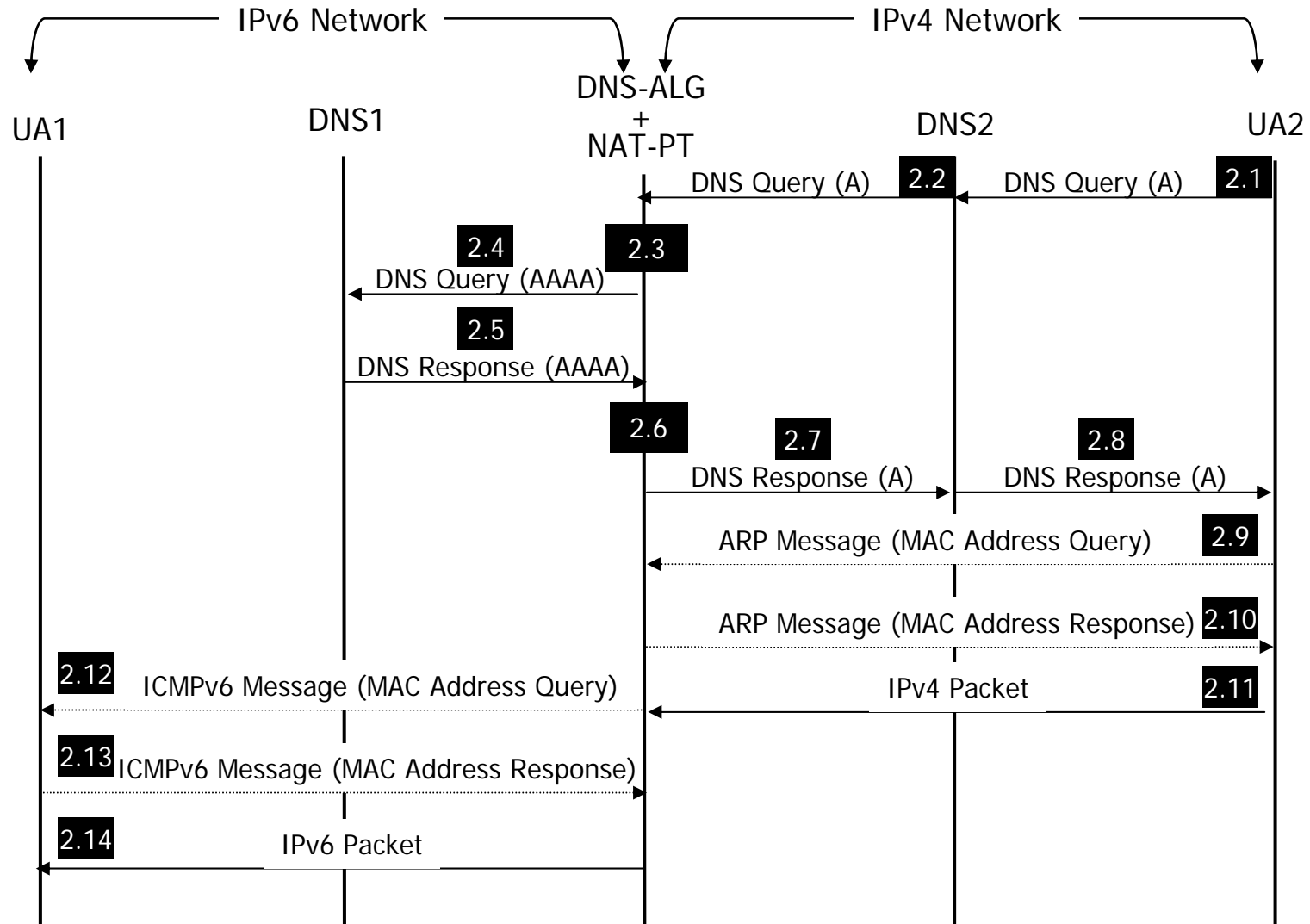
Network Architecture for IPv6 Translation



NAT-PT operations with DNS-ALG (IPv6→IPv4)



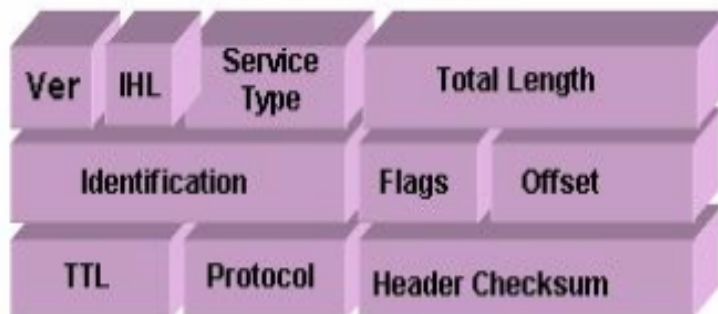
NAT-PT operations with DNS-ALG (IPv4 → IPv6)



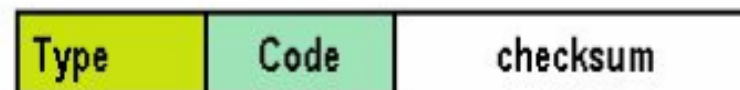


SIIT (Stateless IP/ICMP Translation) RFC 2765

IPv4 header



ICMPv4 header



IPv6 header

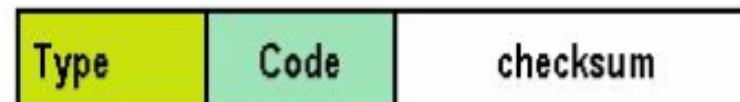


IPv6 fragment header (0)

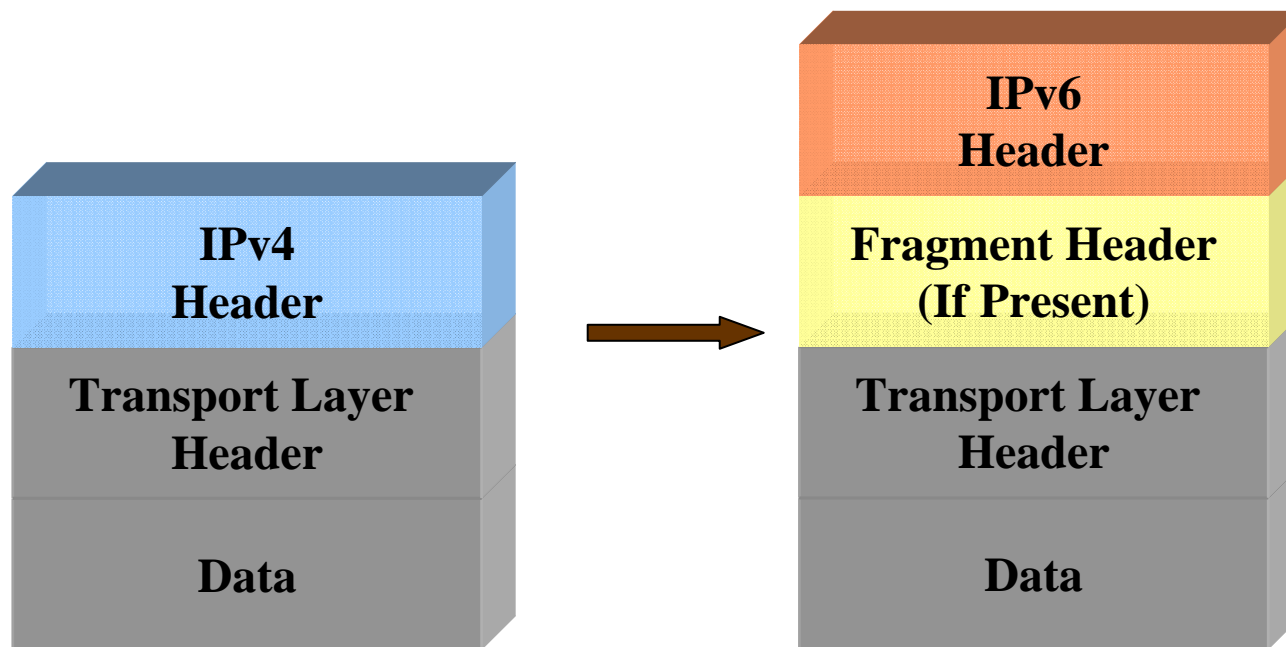


SIIT

ICMPv6 header



Translating from IPv4 to IPv6



Translating IPv4 Headers into IPv6 Headers

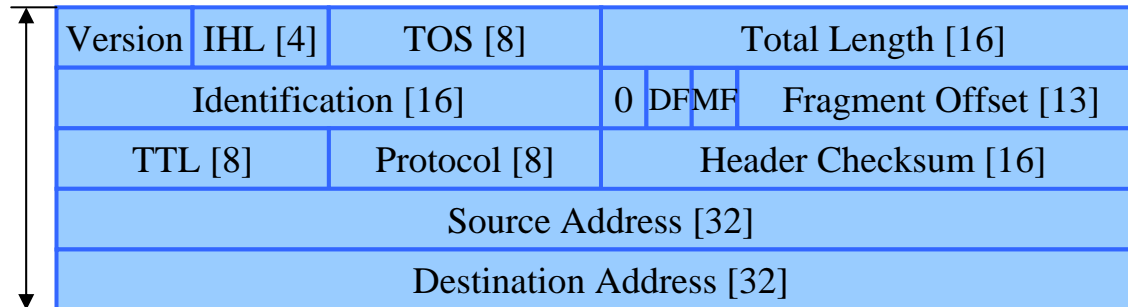
- If IPv4 DF is set.

IPv6 Header Fields	Value
Version	6
Traffic Class	0
Flow Label	0
Payload Length	IPv4 Total length – IPv4 IHL
Next Header	IPv4 Protocol field
Hop Limit	IPv4 TTL - 1
Source Address	High-order 96 bits = IPv6 prefix Low-order 32 bits = IPv4 source address
Destination Address	IPv6 in NAT-PT Mapping table

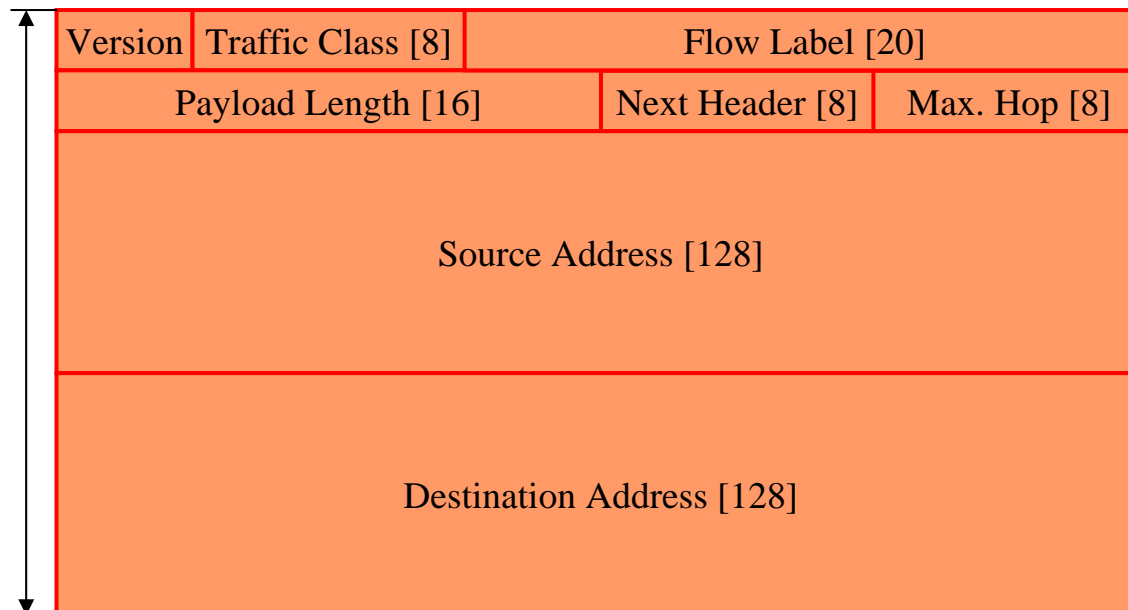
Translating IPv4 Headers into IPv6 Headers

- If IPv4 DF is set.

IPv4
Header



IPv6
Header



Translating IPv4 Headers into IPv6 Headers

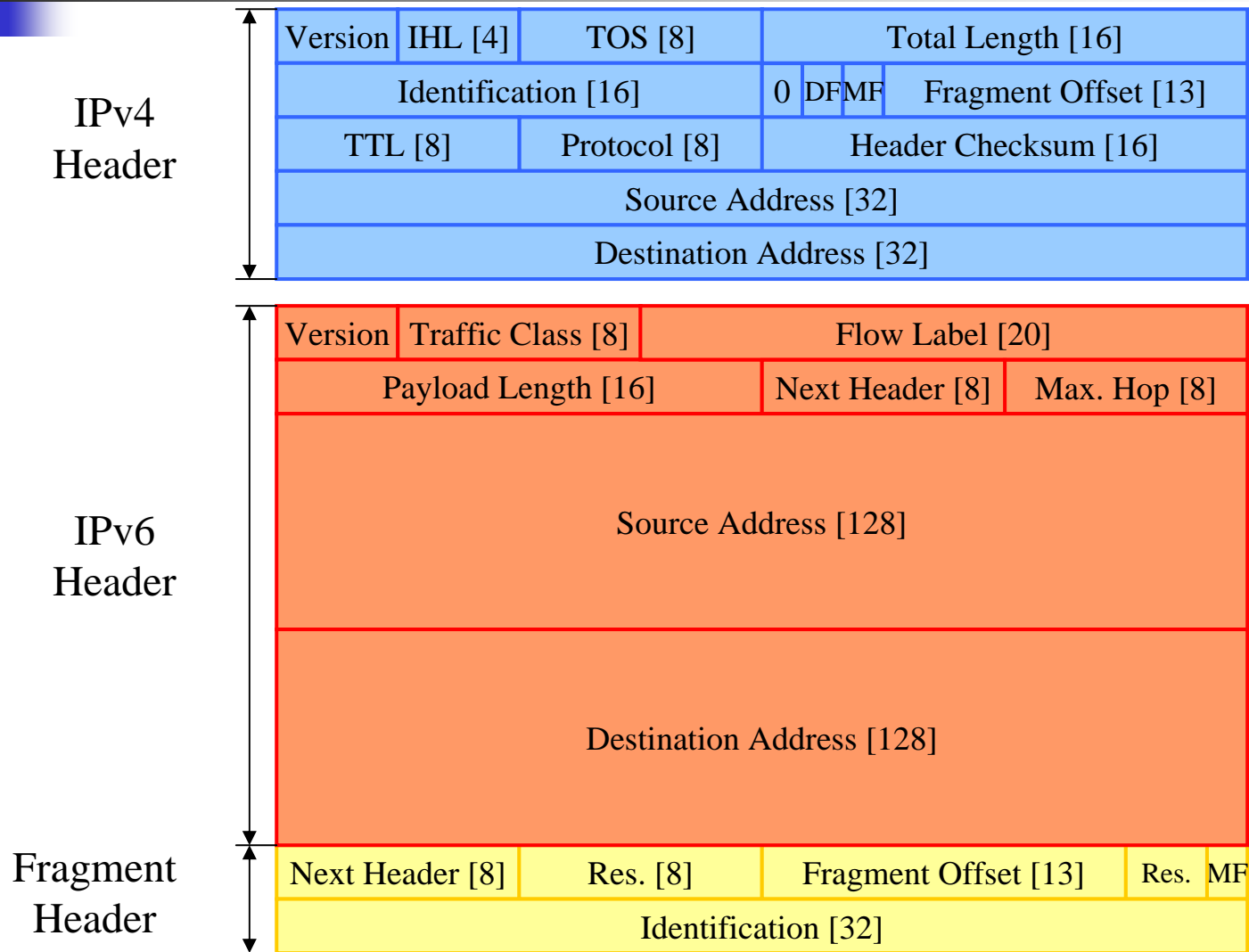
- If IPv4 DF is not set.

IPv6 Header Fields	Value
Payload Length	IPv4 Total length – IPv4 IHL + 8 (Fragment Header)
Next Header	Fragment Header

Fragment Header	Value
Next Header	IPv4 Protocol field
Fragment Offset	IPv4 Fragment Offset field
M Flag	IPv4 More Fragment(MF) bit
Identification	High-order 16 bits = 0 Low-order 16 bits = IPv4 Identification

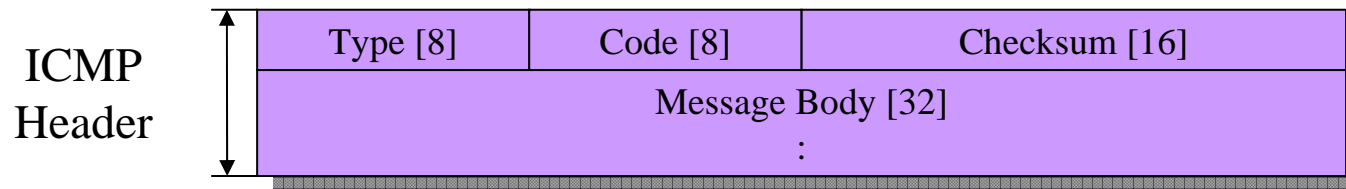
Translating IPv4 Headers into IPv6 Headers

- If IPv4 DF is not set.



Translating ICMPv4 Headers into ICMPv6 Headers

- ICMP Query Message

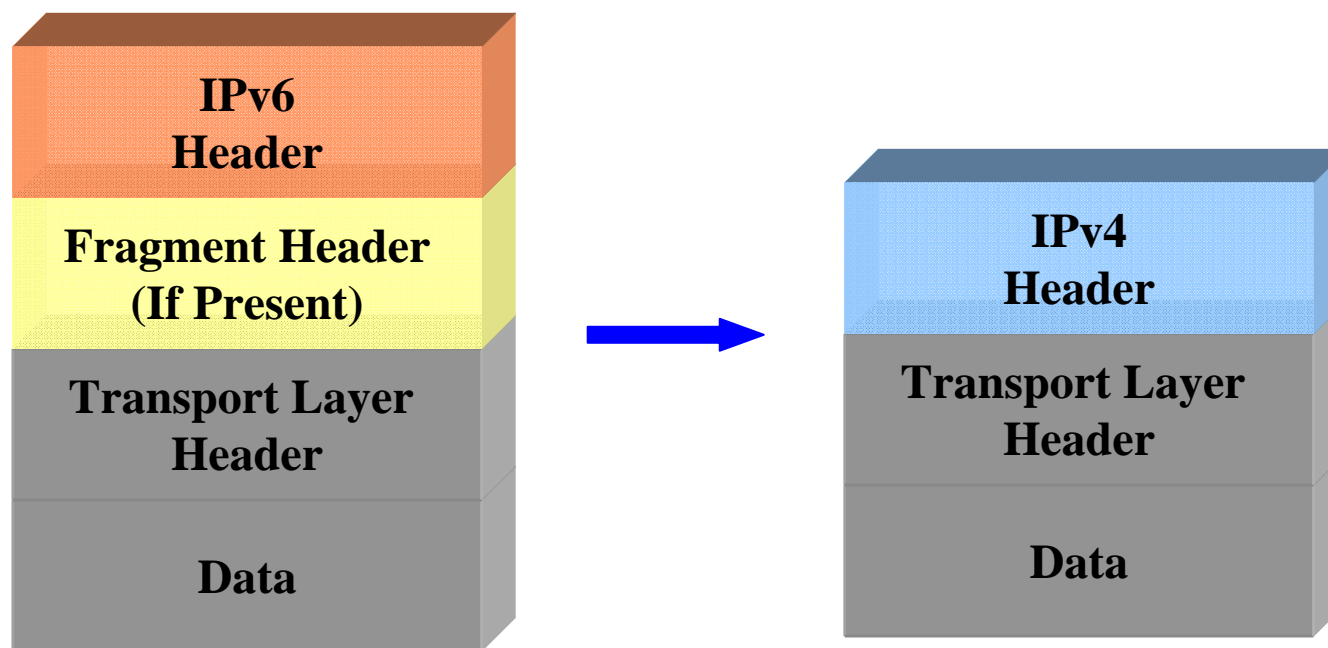


ICMPv4(Type)	ICMPv6(Type)
Echo Request(8) / Reply(0)	Type = 128/129
Information Request(15) / Reply(16)	Silently drop (Undefine)
Timestamp Request(13) / Reply(14)	Silently drop (Undefine)
Address Mask Request(17)/ Reply(18)	Silently drop (Undefine)
ICMP Router Adver.(9)	Silently drop (Single Hop)
ICMP Router Solicit.(10)	Silently drop (Single Hop)
Unknown ICMP4 types	Silently drop

Translating ICMPv4 Headers into ICMPv6 Headers - ICMP Error Message

ICMPv4(Type)	Code	ICMPv6
3 (Destination Unreachable)	0/1	Type = 1, Code = 0 (no route to destination)
	2	Type = 4, Code = 1, Pointer to 6NexHead
	3	Type = 1, Code = 4 (port unreachable)
	4	Type = 2, Code = 0 (Packet too Big)
	5	Type = 1, Code = 0
	6/7/8	Type = 1, Code = 0
	9/10	Type = 1, Code = 1 (Administratively Prohibited)
	11/12	Type = 1, Code = 0
5 (Redirect)		Silently drop (Single Hop)
4 (Source Quench)		Silently drop (Obsoleted in ICMPv6)
11 (Time Exceed)	0/1	Type = 3, Code=0/1
12 (Parameter Problem)	0	Type = 4, Code=0, Pointer to the correspon. Field
Unknow error message		Silently drop

Translating from IPv6 to IPv4



Translating IPv6 Headers into IPv4 Headers

- If there is no IPv6 Fragment header.

IPv4 Header Fields	Value
Version	4
Internet H. L.	5 (no IPv4 option)
ToS and Precedence	0
Total Length	Payload Length + IPv4 IHL
Identification	0
Flags	MF = 0, DF = 1
Fragment Offset	0
Time to Live	IPv6 Hop Limit – 1
Protocol	IPv6 Next Header
Header Checksum	Recompute
Source Address	IPv4 in NAT-PT Mapping table
Destination Address	Destination address of IPv6 Low-order 32 bits

Translating IPv6 Headers into IPv4 Headers

- If there is a IPv6 Fragment header.

IPv4 Header Fields	Value
Total Length	Payload Length - IPv4 IHL + 8(Fragment header)
Identification	Identification in Fragment header
Flags	MF = M in the Fragment header, DF = 1
Fragment Offset	Fragment Offset in Fragment header
Protocol	Next Header in Fragment header

Translating ICMPv6 Headers into ICMPv4 Headers

- ICMPv6 Query Message

ICMPv6(Type)	ICMPv4(Type)
Echo Request(128) / Reply(129)	Type = 8/0
MLD Multicast Listener Query/Report/Done (130/131/132)	Silently drop (Single Hop & Undefined)
Neighbor Discover messages (133 – 137)	Silently drop (Single Hop)
Unknown ICMPv6 types	Silently drop

Translating ICMPv6 Headers into ICMPv4 Headers - ICMPv6 Error Message

ICMPv6(Type)	Code	ICMPv4
1 (Destination Unreachable)	0	Type = 3, Code = 1 (host unreachable)
	1	Type = 3, Code = 10 (Admin. Prohibited)
	2	Type = 3, Code = 1 (host unreachable)
	3	Type = 3, Code = 1 (host unreachable)
	4	Type = 3, Code = 3 (port unreachable)
2 (Packet Too Big)	0	Type = 3, Code = 4 (fragmentation needed)
3 (Time Exceeded)	0/1	Type = 11, Code = 0/1
11 (Parameter Problem)	1	Type = 3, Code = 2 (protocol unreachable)
	other	Type = 12, Code = 0
Unknown error message		Silently drop

Checksum Modification

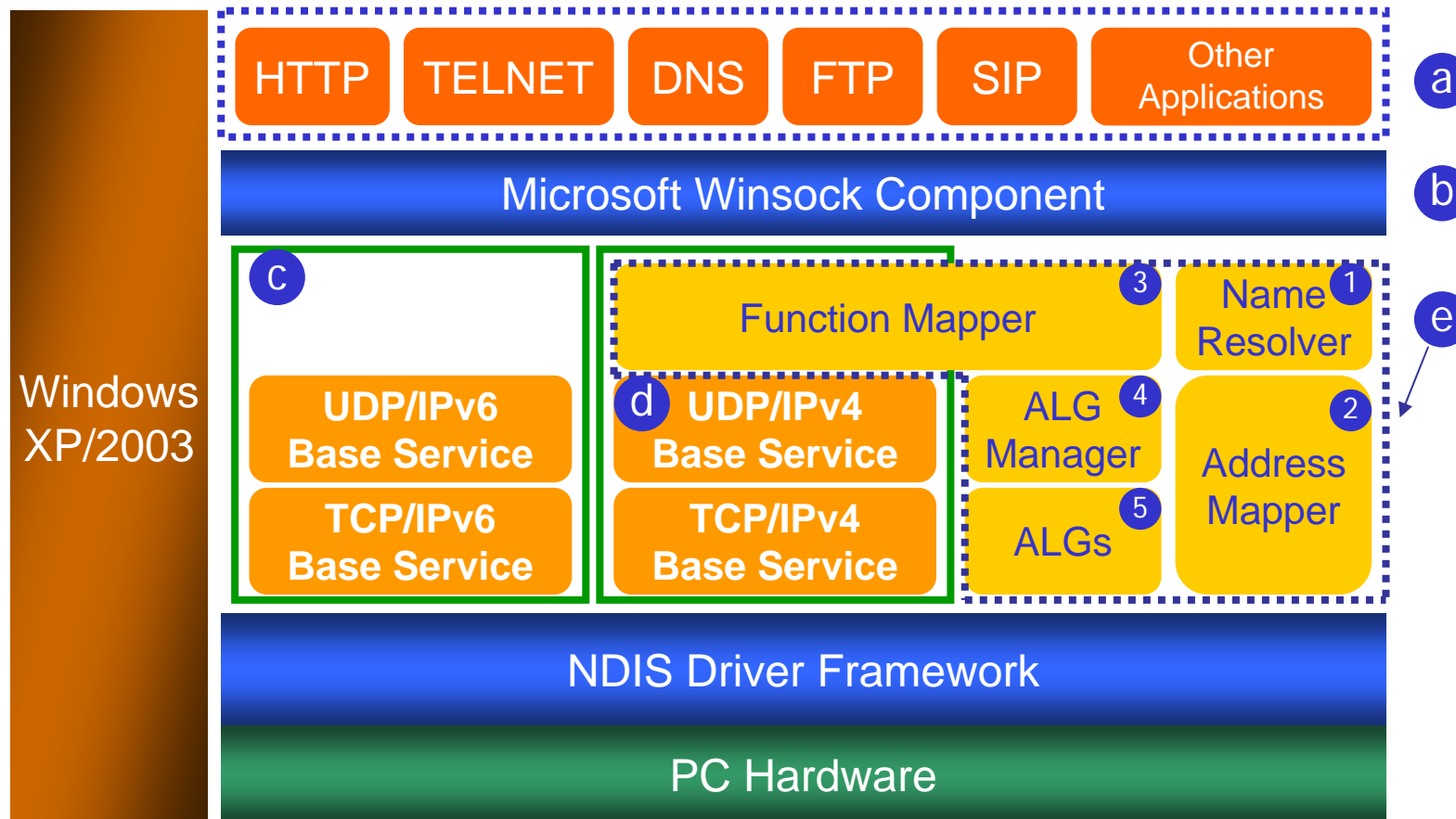
- Internet checksum use 16-bits 1's complement checksum
- 2 kinds of policy
 - Re-Compute Algorithm
 - Adjustment Algorithm

Case	Policy
IPv6 Header ↔ IPv4 Header	Re-Compute Algorithm
ICMPv6 Header ↔ ICMPv4 Header	Adjustment Algorithm
TCP	Adjustment Algorithm
UDP	Adjustment Algorithm

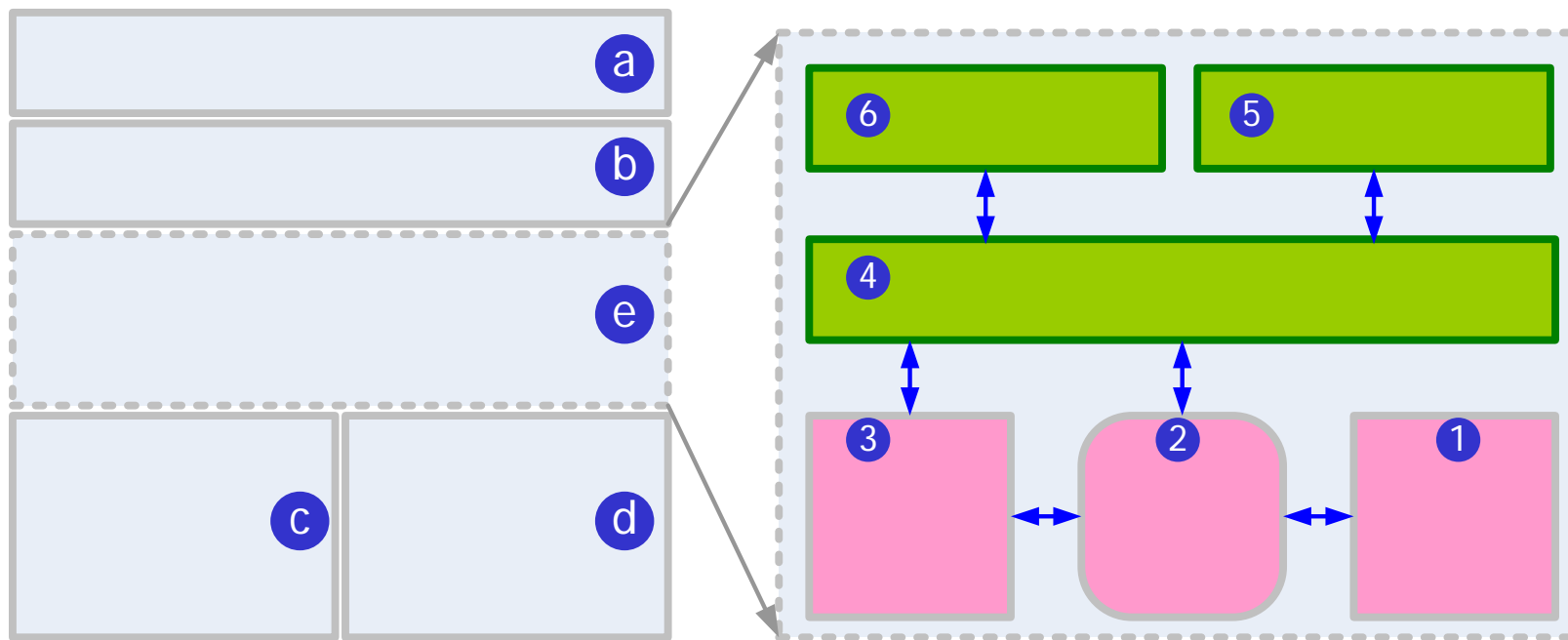


Socket-Layer Translator

System Architecture of SLT and ALGs



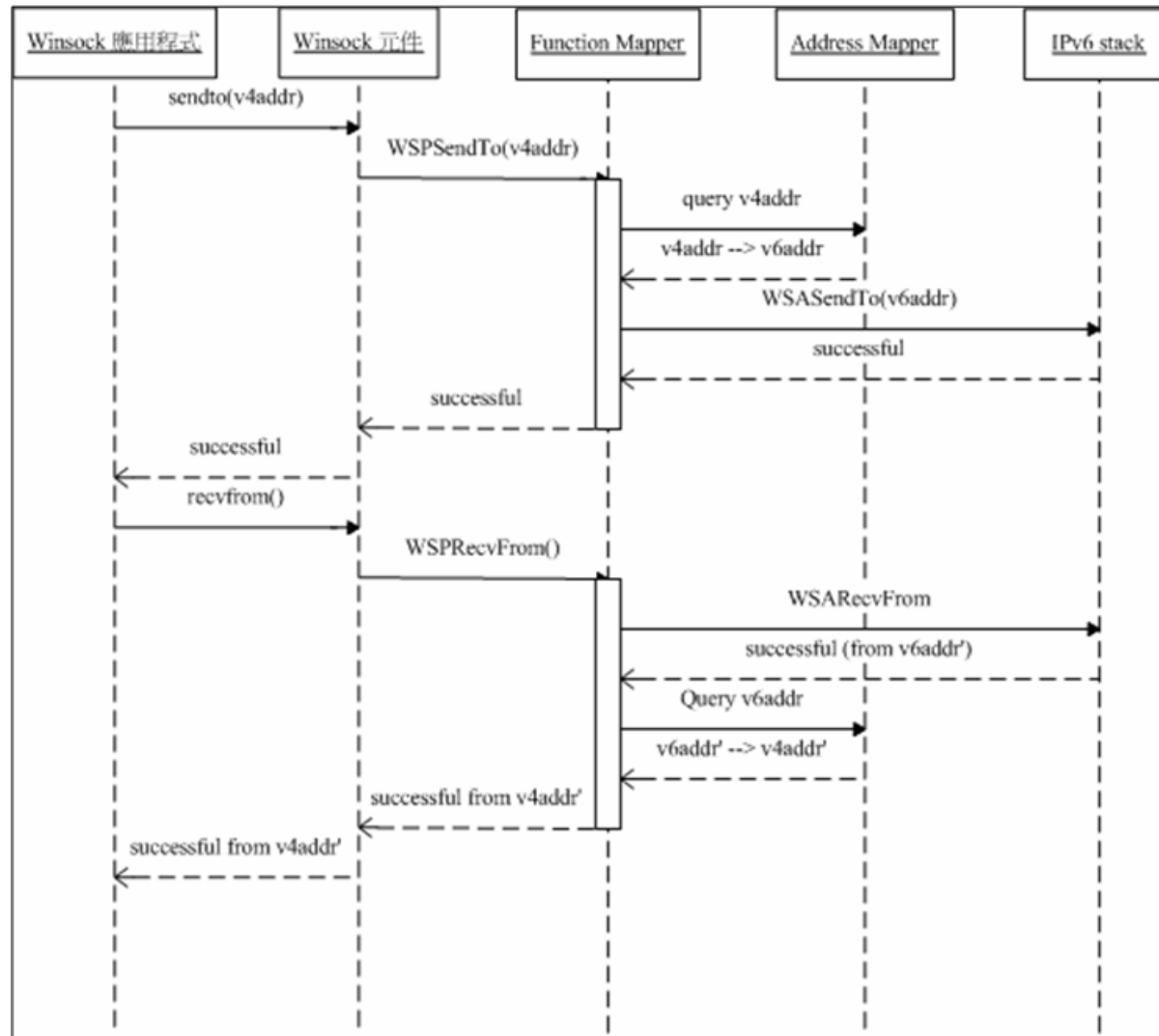
System Architecture of SLT and ALGs



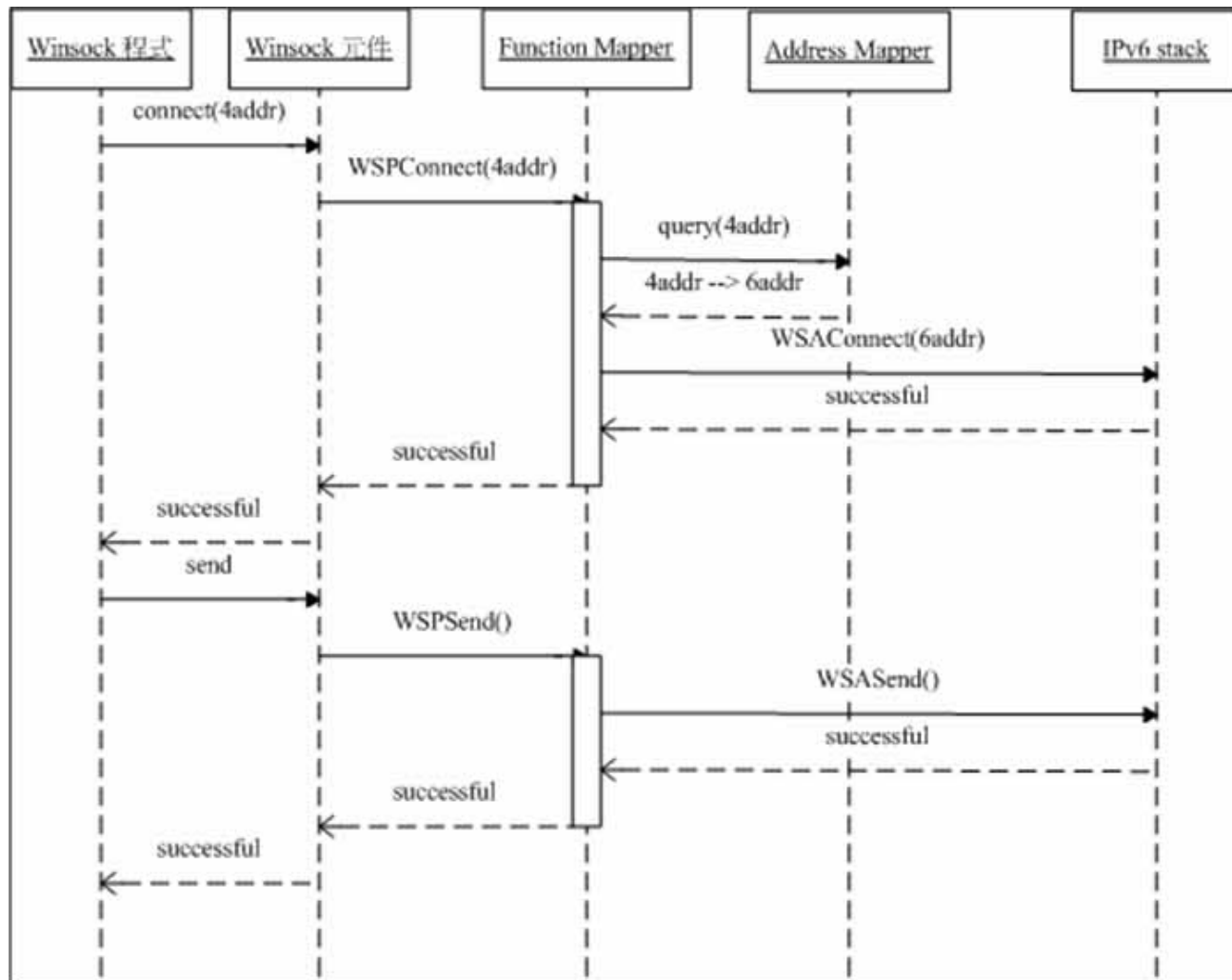
Network Applications

Winsock Component

Example of UDP Call Flow



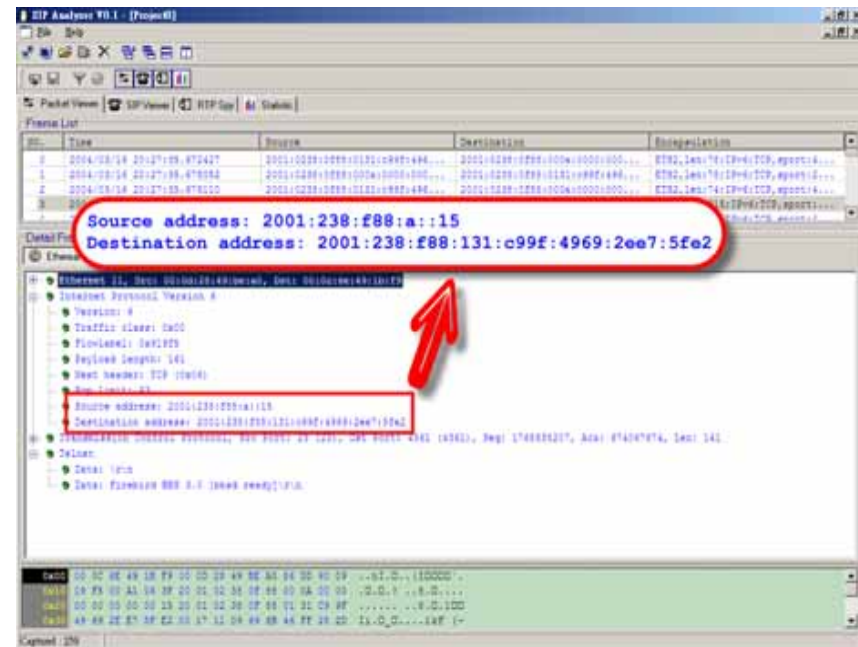
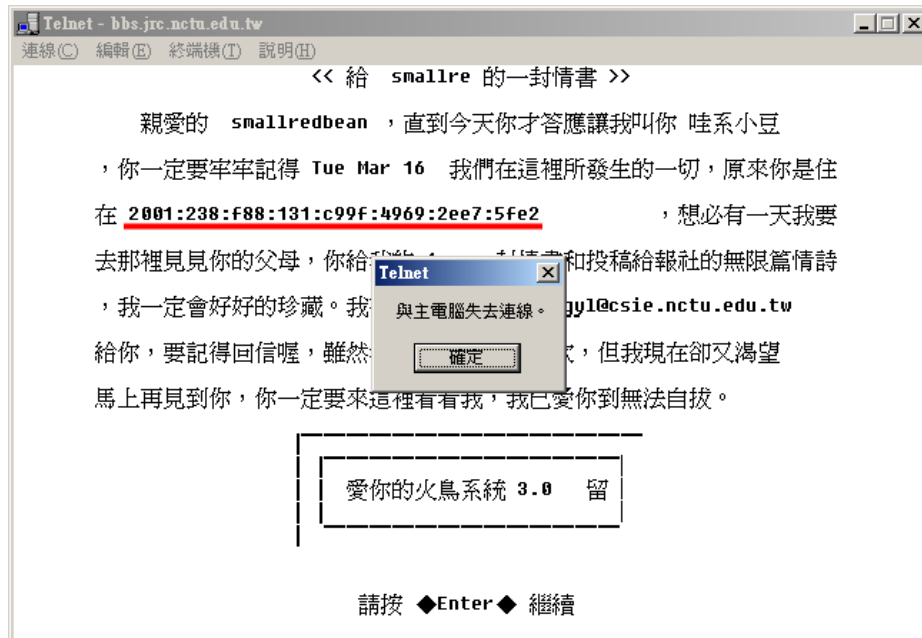
Example of TCP Call Flow



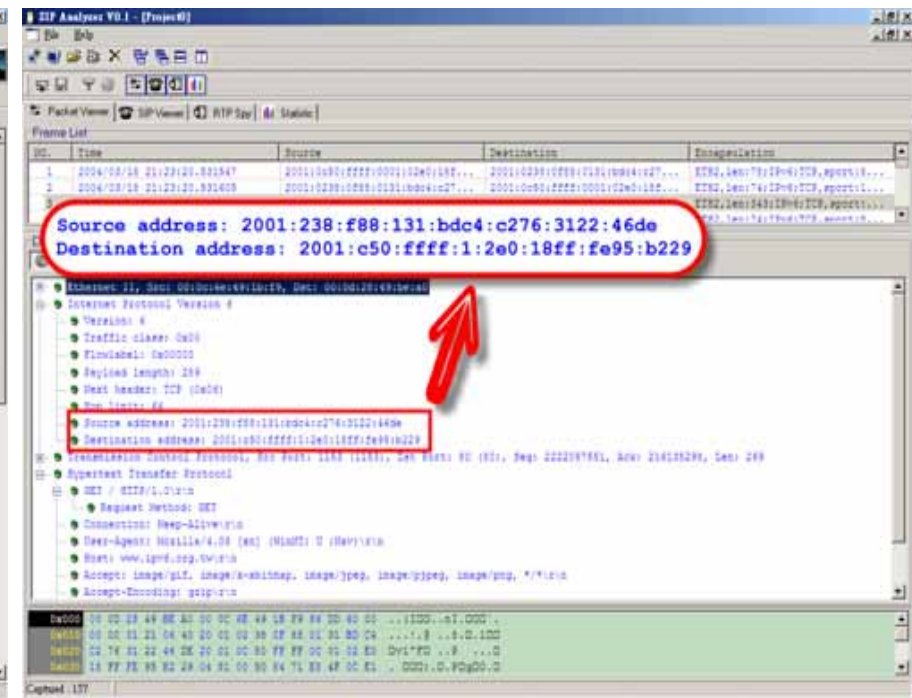
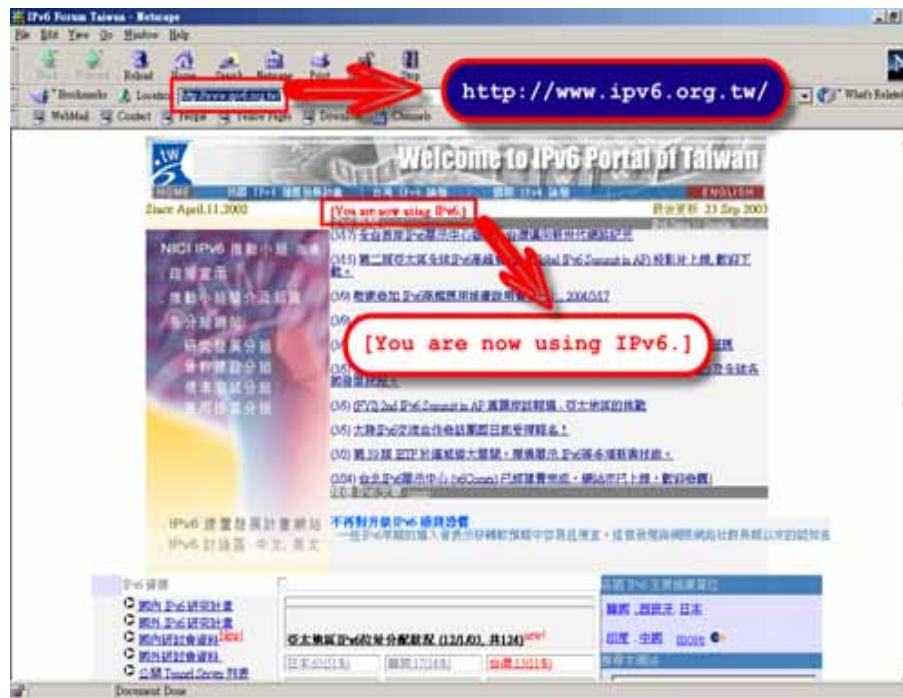
Development Status

- *Socket-layer Translator* can translate most Winsock applications using NAT-friendly protocol
- *FTP-ALG* and *SIP-ALG* can translate FTP and SIP applications respectively.
- Tested Applications
 - NAT-friendly: Netscape (HTTP), Telnet
 - DNS
 - FTP: CuteFTP, FileZilla, SmartFTP
 - SIP: CCL SkinUA 0.8

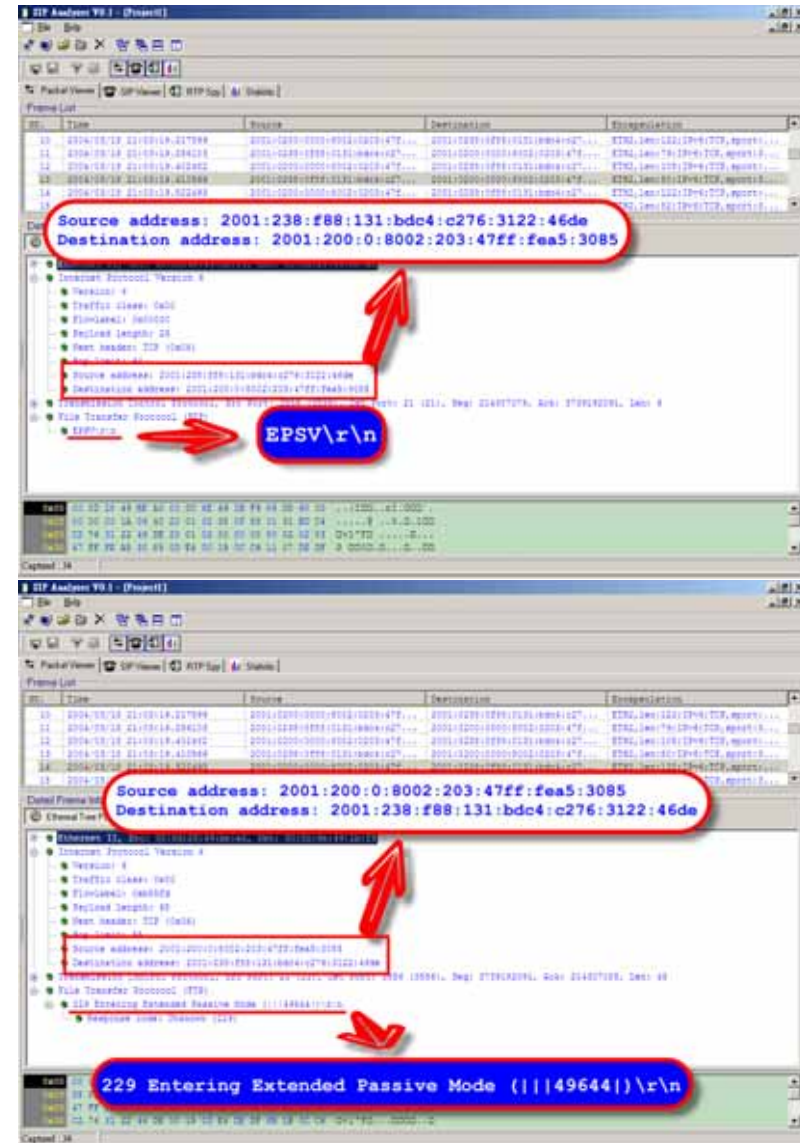
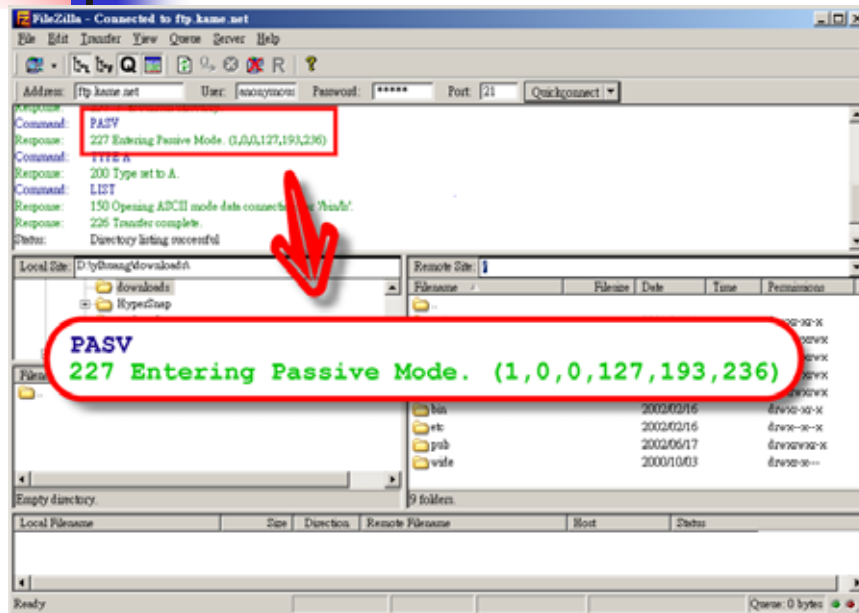
Snapshot of TELNET and Analyzer



Snapshot of NETSCAPE and Analyzer



Snapshot of FileZilla and Analyzer



Compare IPv4 and IPv6 FTP Commands

Reference \ Item	FTP Command	Successful Response
RFC-959	PORT a1,a2,a3,a4,p1,p2	200 OK
RFC-2428	EPRT <net-prt> <addr> <port>	200 OK
RFC-959	PASV	227 Entering...(a1,a2,a3,a4,p1,p2)
RFC-2428	EPSV	229 Entering...(<port>)

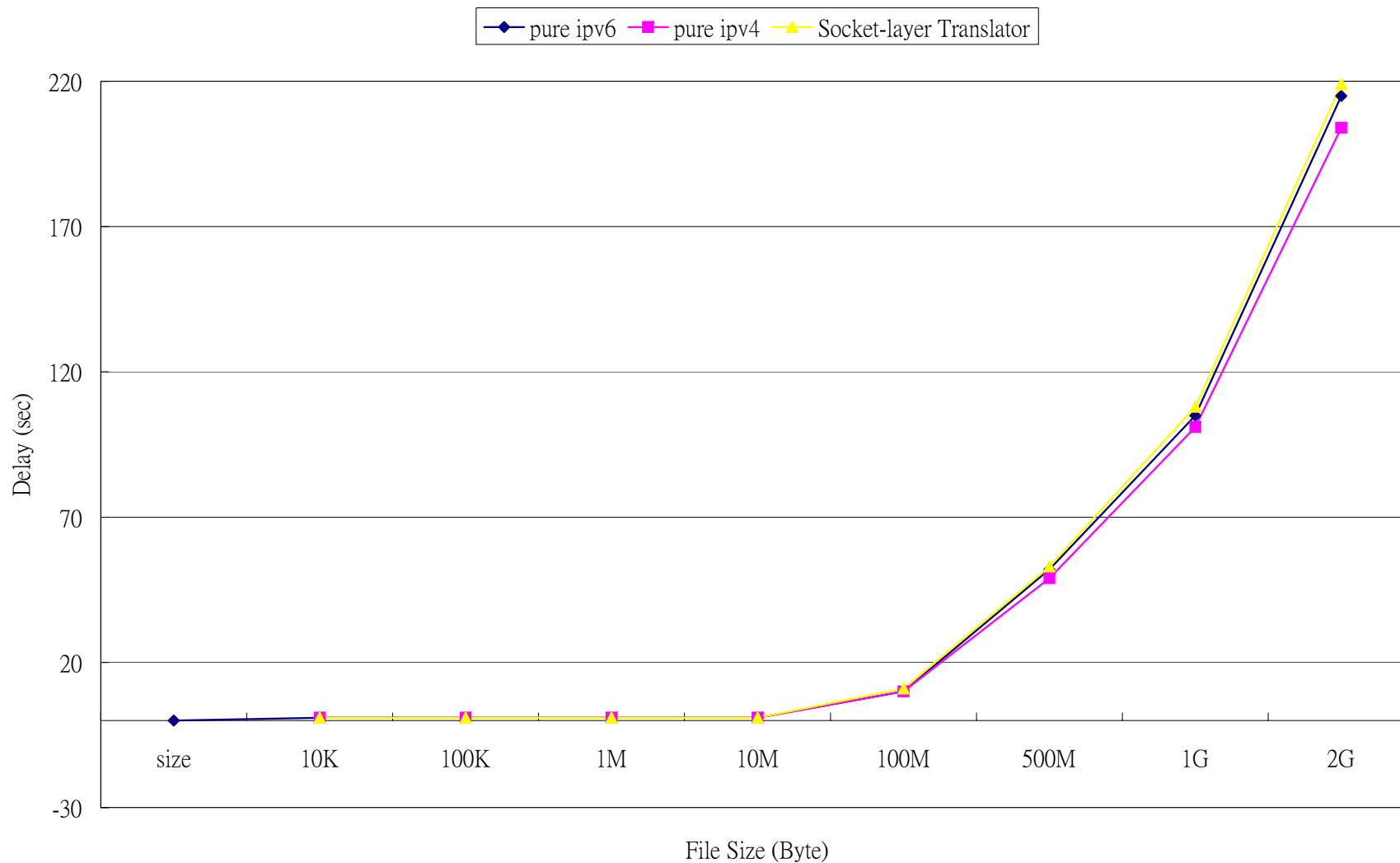
Performance Evaluation of SLT and FTP-ALG

- Client
 - OS: Windows XP SP1
 - FTP Client
 - SmartFTP IPv4 mode
 - SmartFTP IPv6 mode
 - SmartFTP IPv4 mode with SLT and FTP-ALG
 - Actions: Download files with size from 10KB to 2G
- Server
 - OS: linux Fedora Core 1
 - FTP Server: vsftpd

Transmission Delay (sec)

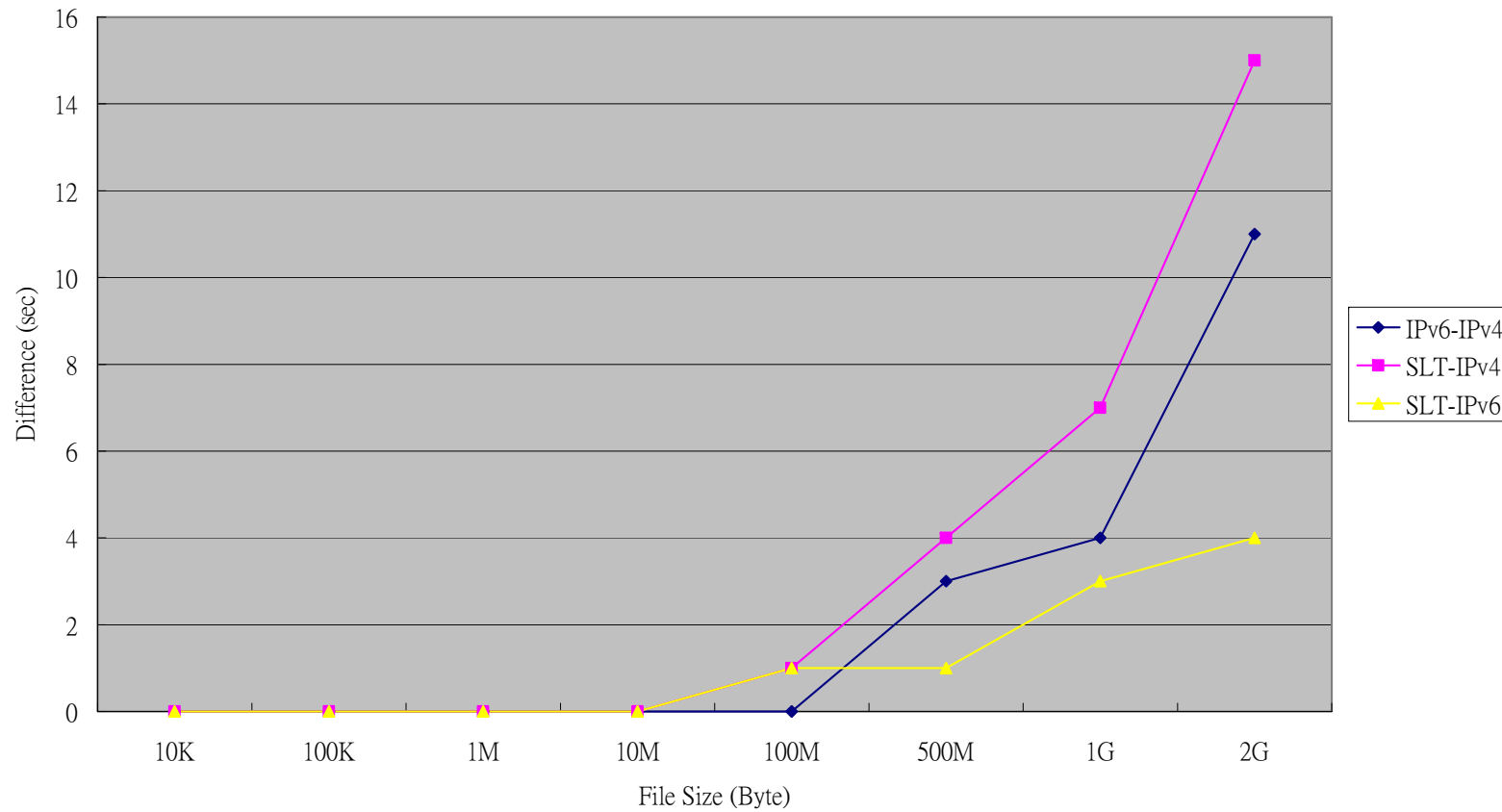
Mode \ Size	Size							
	10KB	100KB	1MB	10MB	100MB	500MB	1GB	2GB
IPv4	1	1	1	1	10	49	101	204
IPv6	1	1	1	1	10	52	105	215
SLT	1	1	1	1	11	53	108	219

Transmission Delay



Performance Test (FTP) cont.

Difference among Three Modes



Comparison of Transition Mechanisms

Mechanisms Items	Manual Modification	BIA	BIS	Socket-layer Translator
Source Codes	Required	Not required	Not required	Not required
Applicability	Modified codes	All socket-based programs	All IP-based Programs	All socket-based programs
P2P Security Support	Yes	Yes	No	Yes
Translation of NAT friendly protocols	Yes	Yes	Yes	Yes
Support of SIP and FTP	Yes	No	No	Yes



SIPv6 Translator

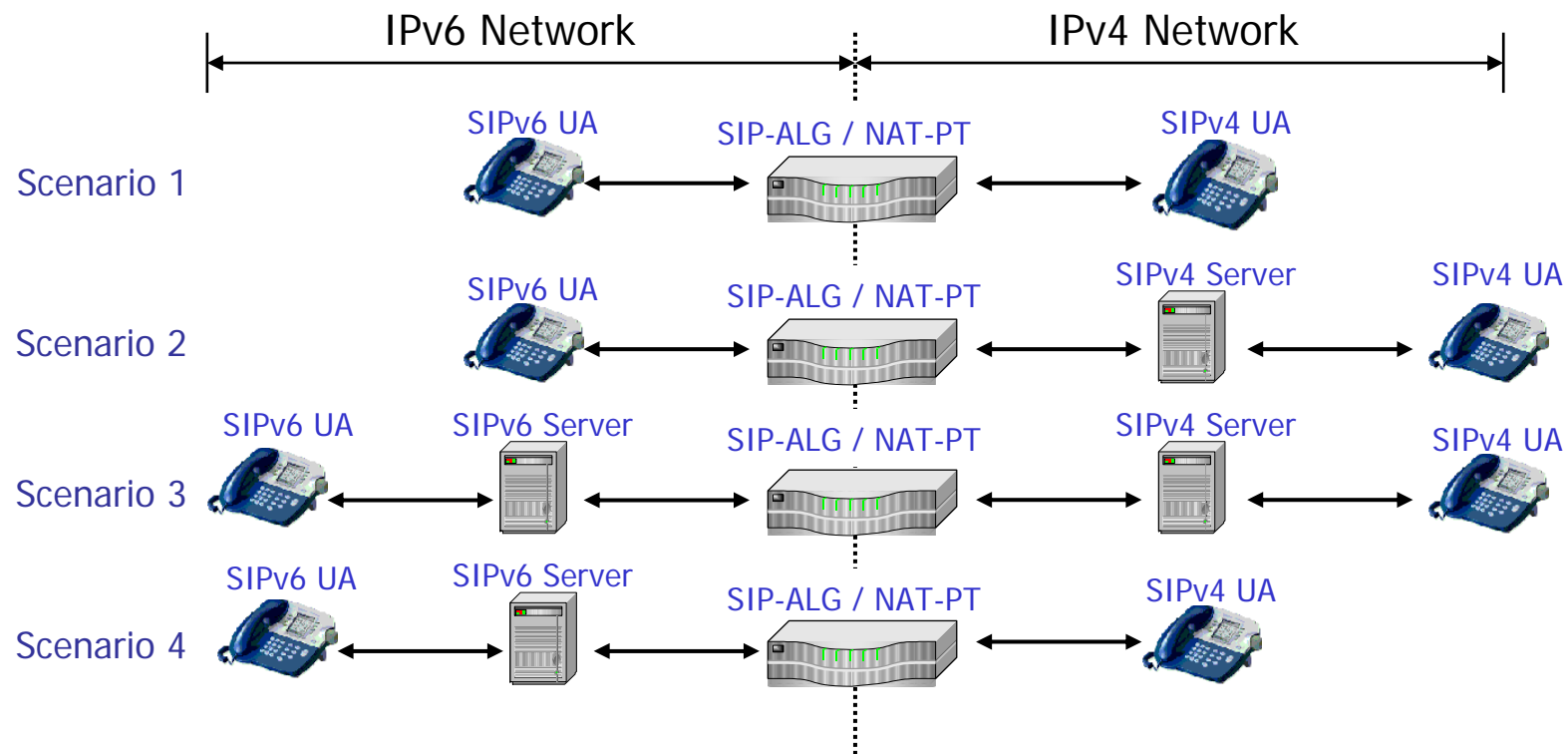
Abstract

- Since Voice over IP (VoIP) phones can efficiently utilize the network and save a lot of money for users, there are increasingly people employ personal computers, notebooks, and PDA with VoIP Phone to communicate with each other.
- VoIP is a peer-to-peer service, and connecting through Internet requires large number of global Internet identifiers.
- Though network address translation (NAT) can help to reduce the address requirements, NAT cannot offer incoming calls directly.
- IPv6 provides large addressing space and is the best choice for VoIP phones.
- However, VoIPv6 phones should be able to communicate with existing VoIPv4 systems.
- Consequently, we will propose an VoIPv6 translator to connect VoIPv6 phones with popular Internet.

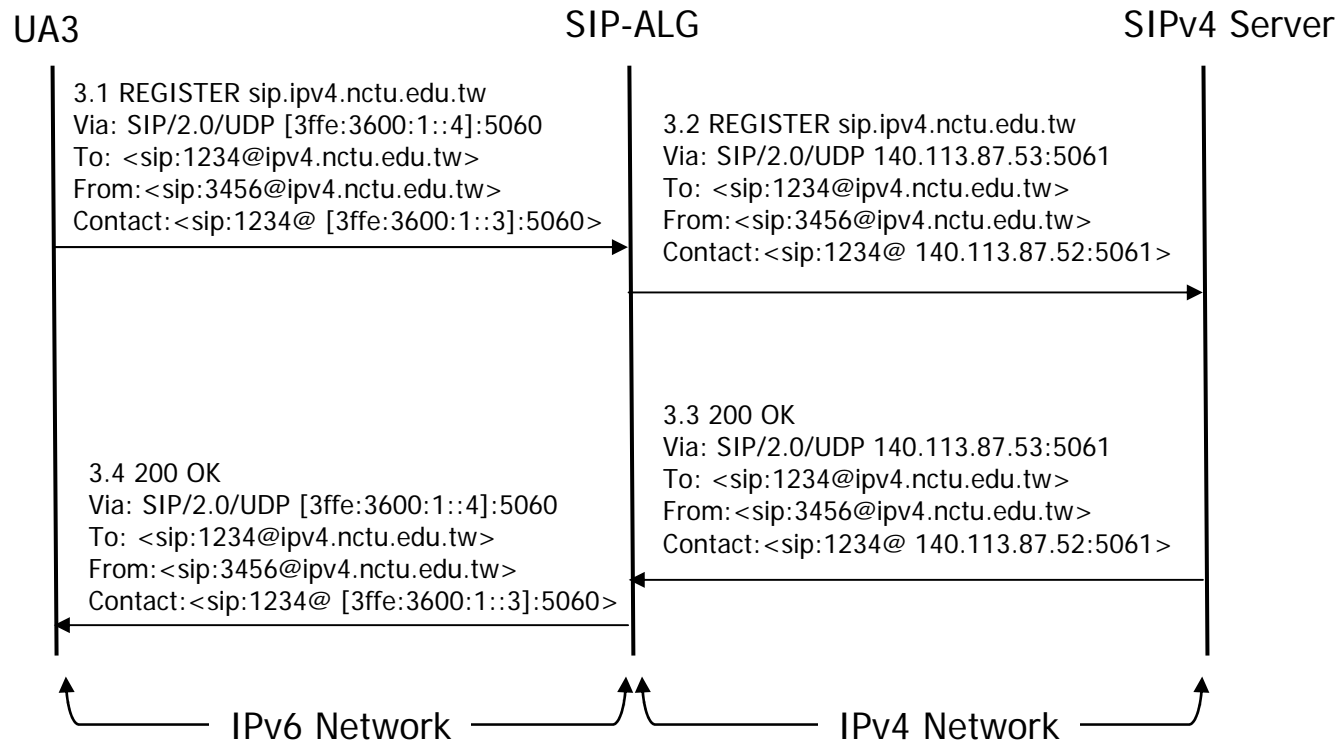
Proposed VoIPv6 Solutions

- Since IP addresses are not enough for peer-to-peer service such as VoIP, the IPv6 is the best answer for next generation Internet.
- Though VoIPv6 phones hold global identifications, VoIPv6 phones cannot directly connect VoIPv4 phones through Internet.
- Therefore this paper proposed an NAT-PT translator with SIP-ALG to assist VoIPv6 phones in communicating with VoIPv4 phones.
- The target of the proposed translator should be capable of handling incoming and outgoing calls within IPv6 domain and simultaneously providing transparent translation for users.
- The SIP-ALG should cooperate with existent modules such as DNS-ALG and FTP-ALG.

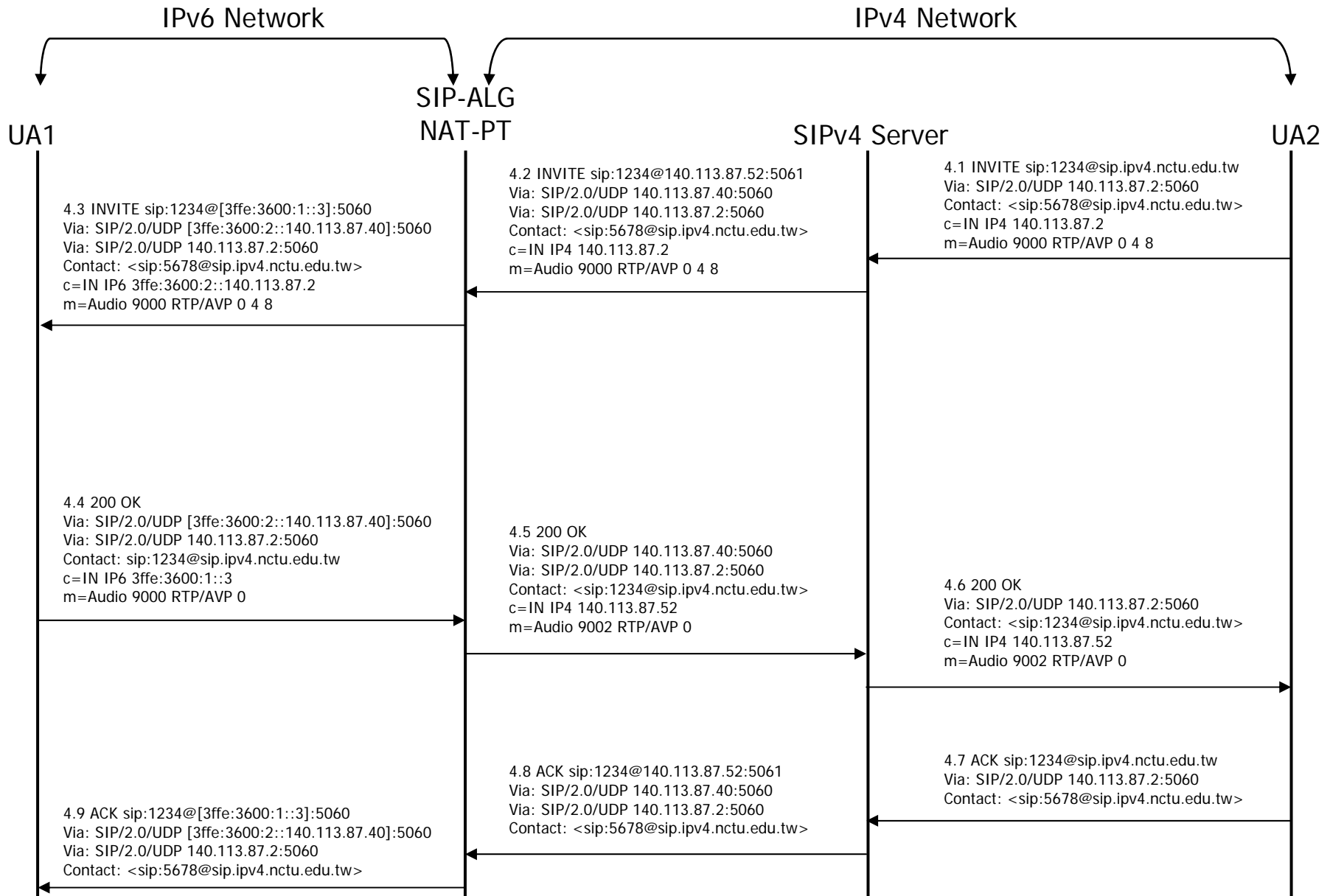
SIP-based VoIPv6 Evolution Scenarios



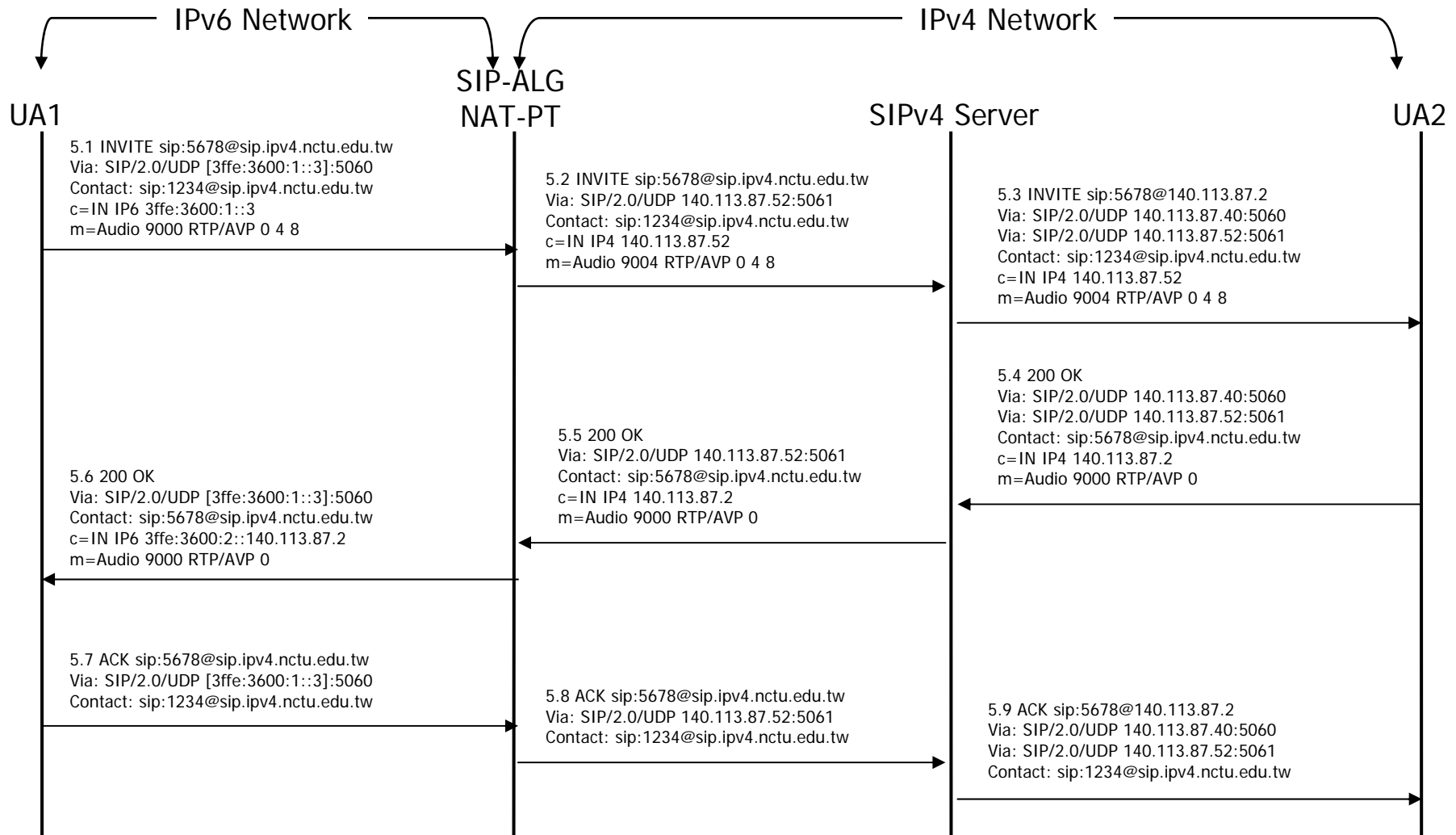
SIP UA registration while Initiating



(IPv4->IPv6)



(IPv6->IPv4)



SIP Message (initiated from SIPv6 UA)

SIP Signaling

```

INVITE sipv4@[3ffe:3600:2::10.113.131.4]:5060 SIP/2.0
Expires: 180
Content-Type: application/sdp
Via: SIP/2.0/UDP [3ffe:3600:1:0:20c:6eff:fe1e:55da]:5060;branch=1FV1xhfvxGJOK@rWckdAKOA
To: <sipv4@[3ffe:3600:2::10.113.131.4]>
From: sipv6@[3ffe:3600:1:0:20c:6eff:fe1e:55da]
Call-ID:
CSeq: 100 INVITE
Contact: sipv6@[3ffe:3600:1:0:20c:6eff:fe1e:55da]:5060
Content-Length: 249
User-Agent: CCL's SIPv6 UA
Accept: application/sdp
    
```

SIP的標頭中，
含有Callee的IPv6位址，
3ffe:3600:2::10.113.131.4

SDP Signaling

```

v=0
o=CCL'sSIPv6UA 17045 11864 IN IP6 3ffe:3600:1:0:20c:6eff:fe1e:55da
s=SIP Call
c=IN IP6 3ffe:3600:1:0:20c:6eff:fe1e:55da
t=0 0
m=audio 29118 RTP/AVP 0 101
a=rtpmap:0 pcmu/8000
a=rtpmap:101 telephone-event/8000
    
```

SIP與SDP標頭中，
含有Caller的IPv6位址，
3ffe:3600:1:0:20c:6eff:fe1e:55da

SIP Message (translated by SIP-ALG)

SIP Signaling

```

INVITE sipv410.113.131.4:5060 SIP/2.0
Expires: 180
Content-Type: application/sdp
Via: SIP/2.0/UDP 10.113.131.101:5060;branch=1FV1xhfvxGJOK@rWckdAKOA
To: <sipv4@10.113.131.4>
From: sipv6@10.113.131.101
Call-ID:
CSeq: 100 INVITE
Contact: sipv6@10.113.131.101:5060
Content-Length: 196
User-Agent: CCL's SIPv6 UA
Accept: application/sdp
    
```

SIP的標頭中，
Callee經過SIP-ALG轉換的IPv4位址，
10.113.131.4

SDP Signaling

```

v=0
o=CCL'sSIPv6UA 17045 11864 IN IP4 10.113.131.101
s=SIP Call
c=IN IP4 10.113.131.101
t=0 0
m=audio 2918 RTP/AVP 0 101
a=rtpmap:0 pcmu/8000
a=rtpmap:101 telephone-event/8000
    
```

SIP與SDP標頭中，
Caller經過SIP-ALG轉換的IPv4位址，
10.113.131.101



SIPv6 UA Initiated Registration

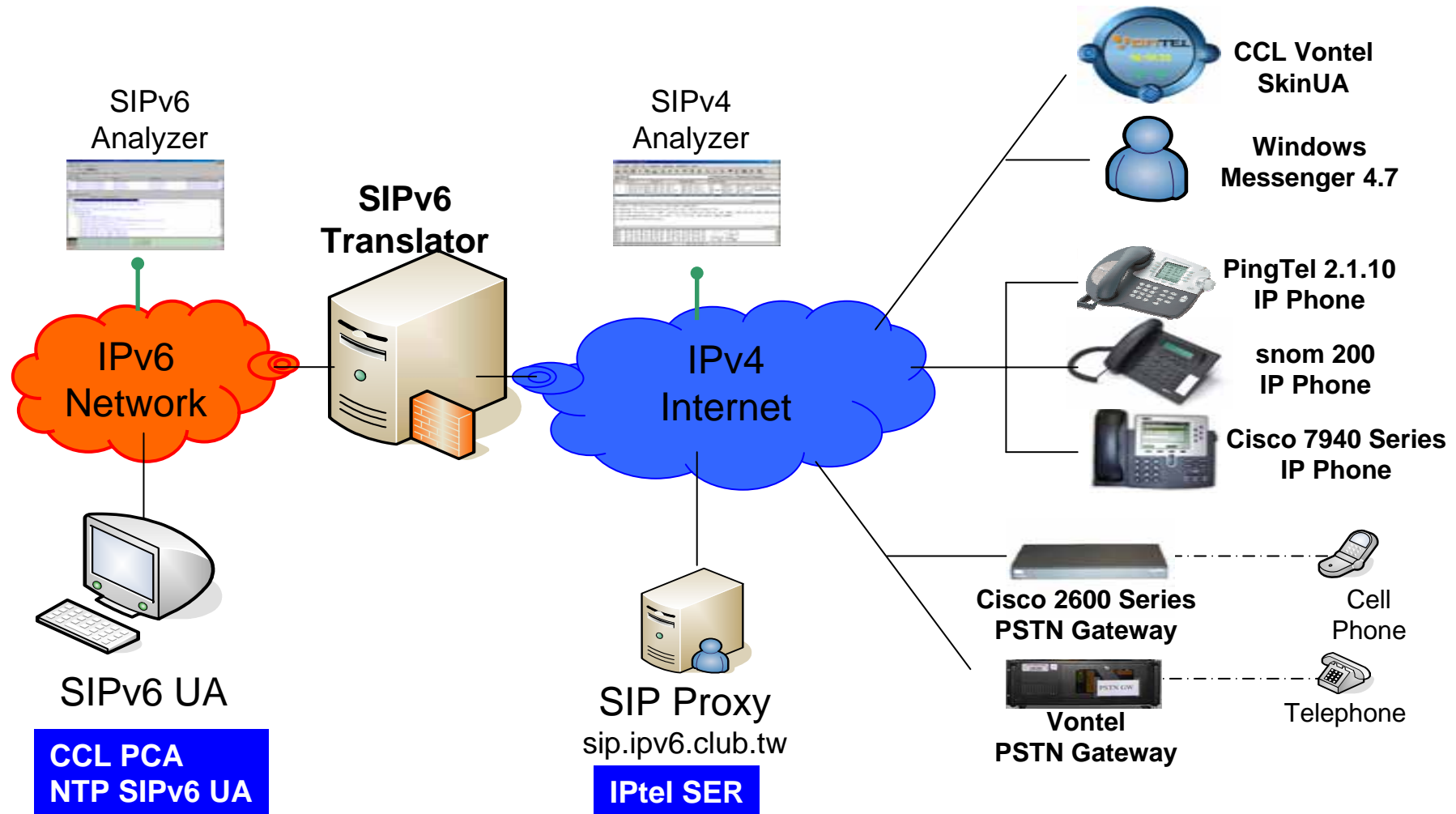
Header	REGISTER	200 OK (REGISTER)
<i>Request-URI</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	-
<i>Via</i>	IP Address (IPv6->IPv4)	IP Address (IPv4->IPv6)
<i>Contact</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	Domain Name (No Action) Or IP Address (IPv4->IPv6)
<i>To</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	Domain Name (No Action) Or IP Address (IPv4->IPv6)
<i>Content-Length</i>	Value is 0	Value is 0
<i>c</i>	--	-
<i>m</i>	--	-



SIPv6 UA Session Setup

Header	INVITE	200 OK (INVITE)	ACK
<i>Request-URI</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	--	Domain Name (No Action) Or IP Address (IPv6->IPv4)
<i>Via</i>	IP Address (IPv6->IPv4)	IP Address (IPv4->IPv6)	IP Address (IPv4->IPv6)
<i>Contact</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	Domain Name (No Action) Or IP Address (IPv4->IPv6)	Domain Name (No Action) Or IP Address (IPv6->IPv4)
<i>Content-Length</i>	Re-calculate	Re-calculate	Value is 0
<i>c</i>	Domain Name (No Action) Or IP Address (IPv6->IPv4)	Domain Name (No Action) Or IP Address (IPv4->IPv6)	--
<i>m</i>	No Action	Change Port Number	--

Network Architecture of SIPv6 Translator



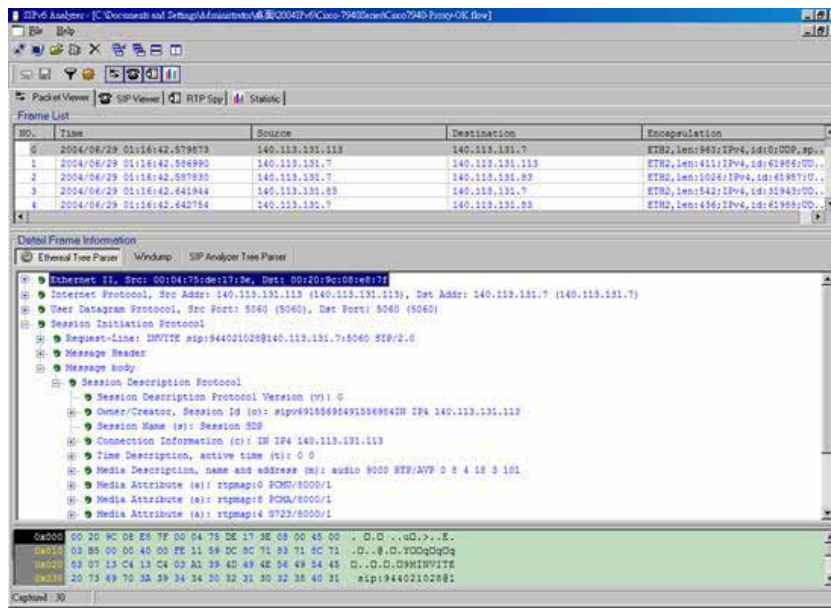
SIP User Agents and SIPv6 Translator



PSTN Gateways

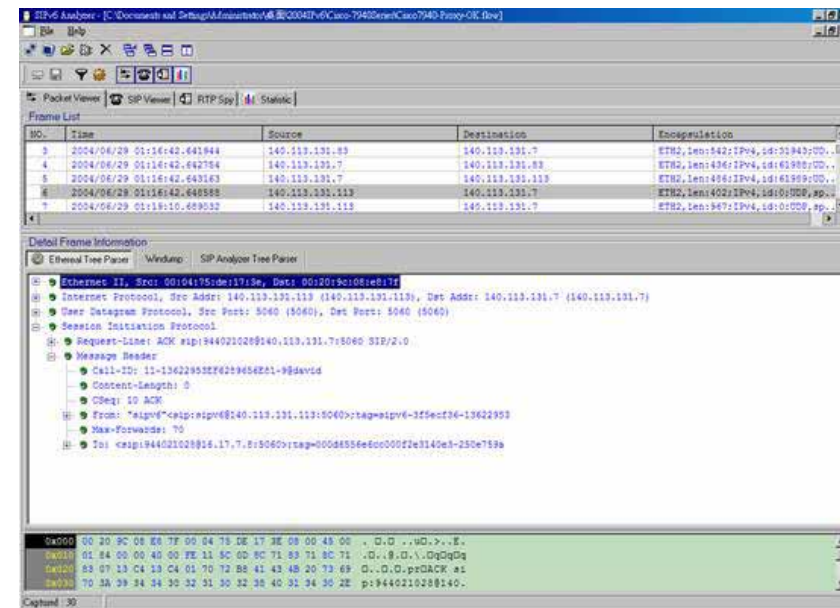


SIPv4 UA (Pingtel) to SIPv6 UA



Wireshark capture of a SIP INVITE message. The packet list shows the message arriving at 140.113.131.113 from 140.113.131.7. The details pane shows the following structure:

- Ethernet II, Src: 001041754de1171e, Dst: 002019c108e81114
- Internet Protocol, Src Addr: 140.113.131.113 (140.113.131.113), Dst Addr: 140.113.131.7 (140.113.131.7)
- User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
- Session Initiation Protocol
 - Request-Line: INVITE sip:944021028@140.113.131.7:5060 SIP/2.0
 - Message Header
 - Message body
 - Session Description Protocol
 - Session Description Protocol Version (v): 0
 - Offer/Creator, Session ID (s): sipv4915589491589843H IP4 140.113.131.113
 - Session Name (n): Session 528
 - Connection Information (c): IP 194 140.113.131.113
 - Time Description, Active time (t): 0 0
 - Media Description, name and address (m): audio 9000 RTP/AVP 0 8 4 18 3 101
 - Media Attribute (a): stpmapi0 PCMA/8000/1
 - Media Attribute (a): stpmapi8 PCMA/8000/1
 - Media Attribute (a): stpmapi4 G723/8000/1

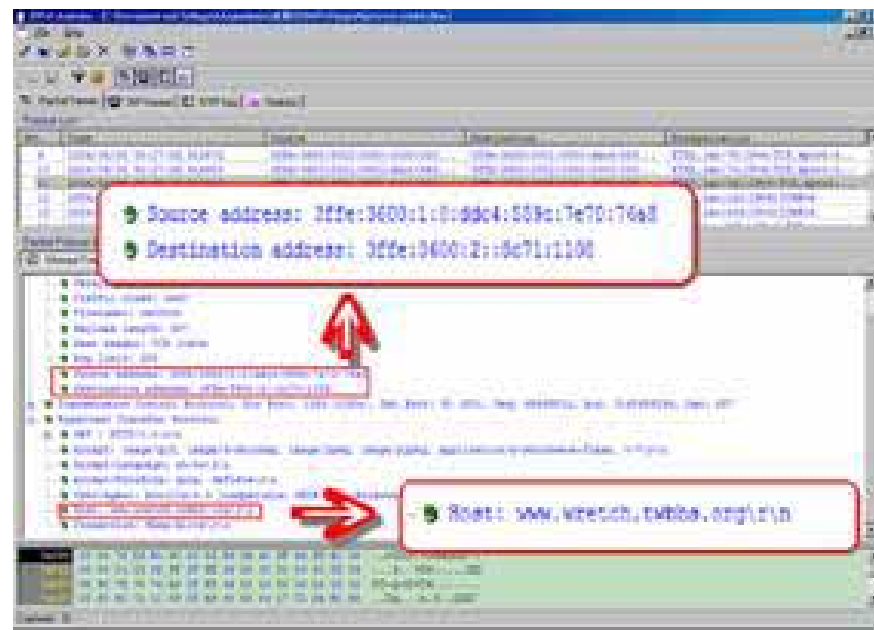


Wireshark capture of a SIP ACK message. The packet list shows the message arriving at 140.113.131.7 from 140.113.131.113. The details pane shows the following structure:

- Ethernet II, Src: 001041754de1171e, Dst: 002019c108e81114
- Internet Protocol, Src Addr: 140.113.131.113 (140.113.131.113), Dst Addr: 140.113.131.7 (140.113.131.7)
- User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
- Session Initiation Protocol
 - Request-Line: ACK sip:944021028@140.113.131.7:5060 SIP/2.0
 - Message Header
 - Call-ID: 11-13622955276229856281-98david
 - Content-Length: 0
 - CSeq: 10 ACK
 - From: "sipv4" <sipv4@140.113.131.113:5060>;tag=sipv4-3f5ecf36-13622955
 - Max-Forwards: 70
 - To: <sip:944021028@140.113.131.7:5060>;tag=00d8856e6e0002f6b140eb-250e789e

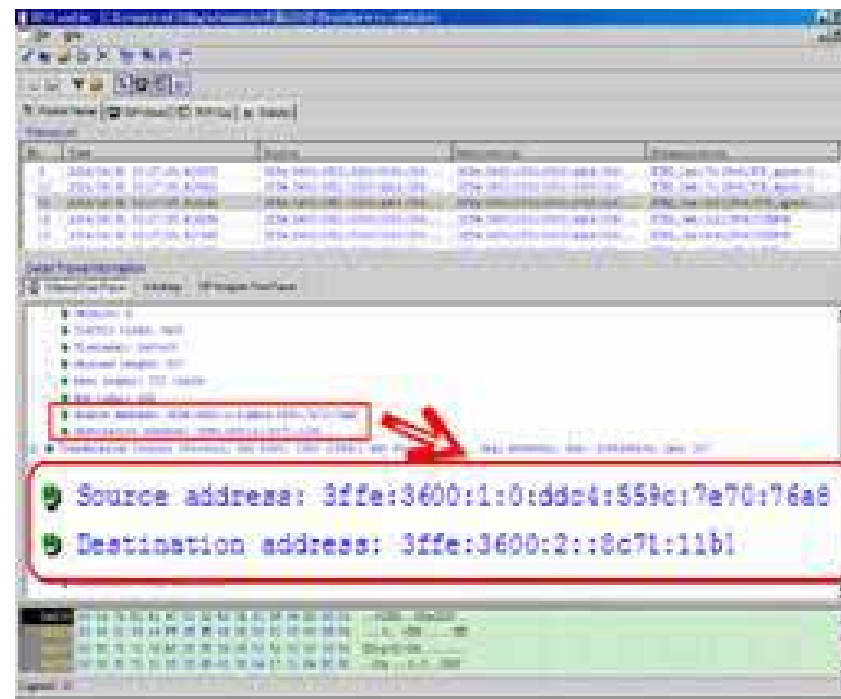


HTTP Test

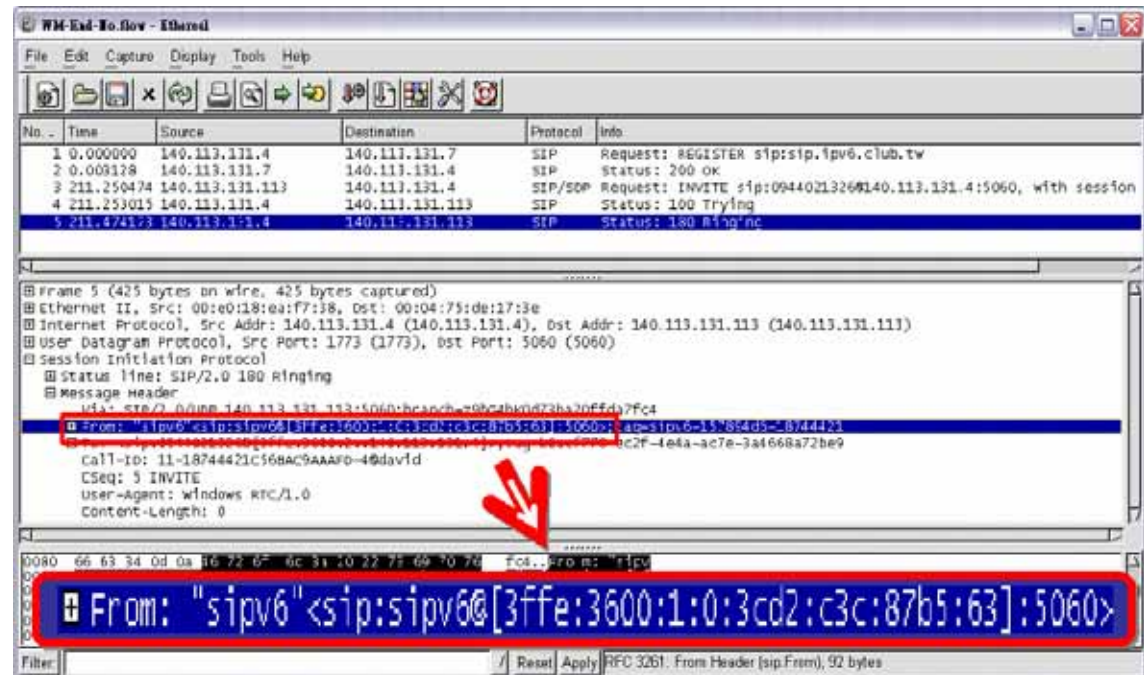
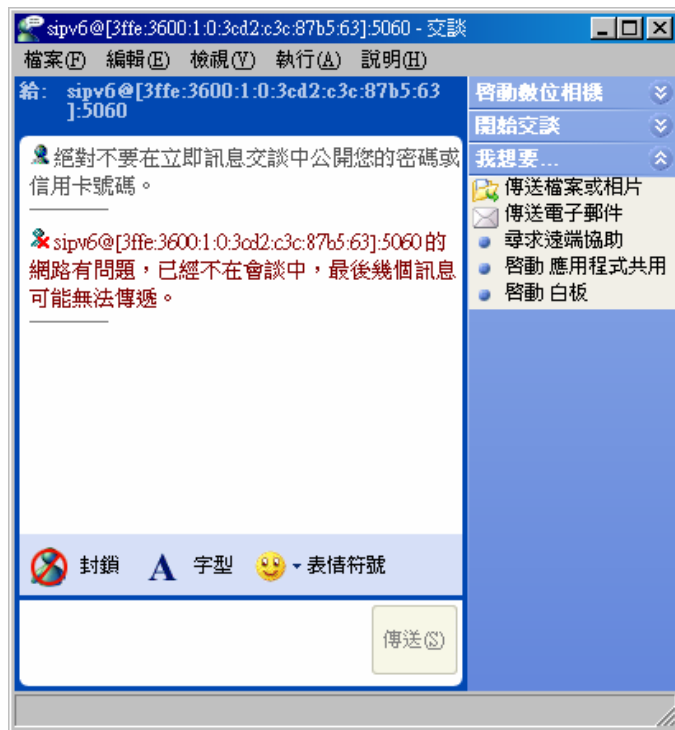




Telnet Test

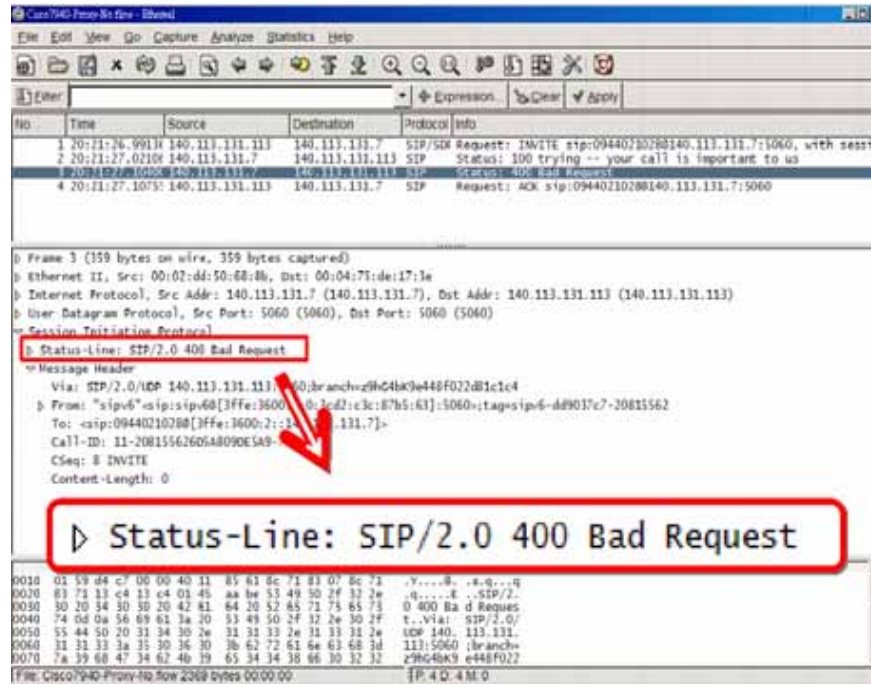


Windows Messenger 4.7



- Solution : modified **“From”** header field in SIP.

Cisco IP Phone 7940 Series



```

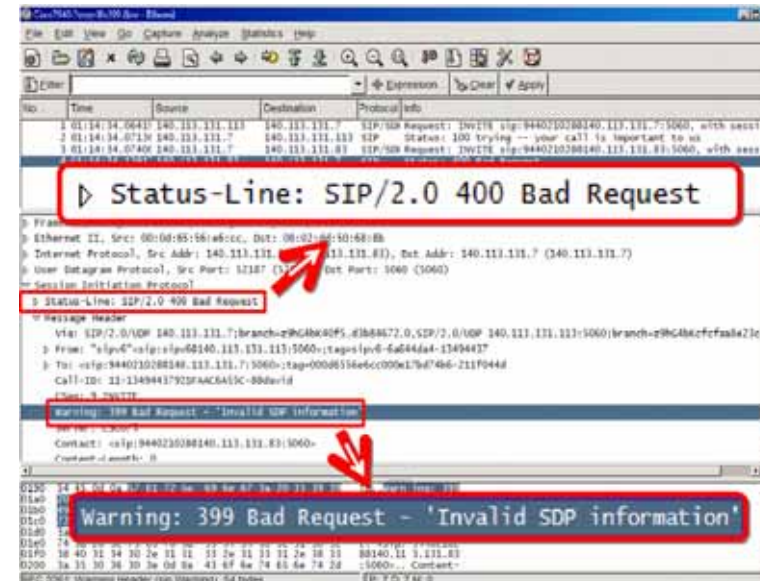
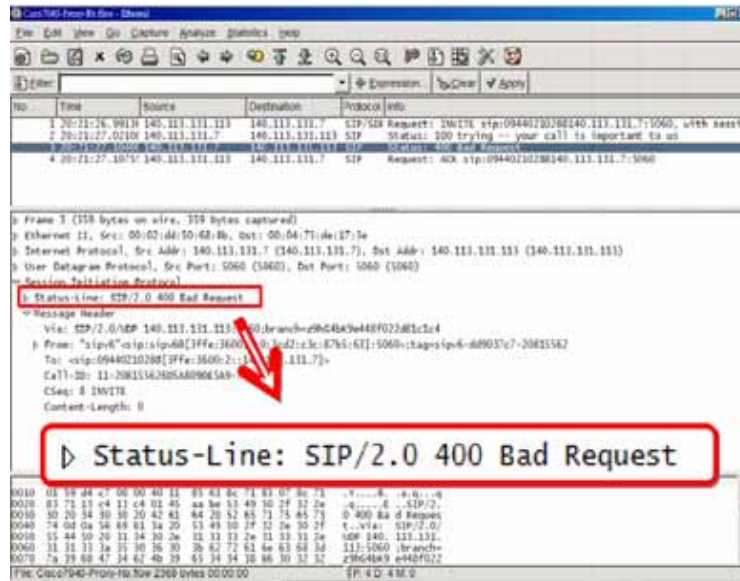
Cisco7940-Proxy-84-80w - libpcap
File Edit View Go Capture Analyze Statistics Help
Filter:
No. Time Source Destination Protocol Info
1 20:11:26.99114 140.113.131.113 140.113.131.7 SIP/SIP Request: INVITE sip:09440210280140.113.131.7:5060, with sessio
2 20:11:27.02104 140.113.131.7 140.113.131.113 SIP Status: 100 trying -- your call is important to us
3 20:11:27.10753 140.113.131.113 140.113.131.7 SIP Status: 400 Bad Request
4 20:11:27.10753 140.113.131.113 140.113.131.7 SIP Request: ACK sip:09440210280140.113.131.7:5060

0 Frame 3 (359 bytes on wire (359 bytes captured)
  Ethernet II, Src: 00:02:dd:50:68:86, Dst: 00:04:71:de:17:1e
  Internet Protocol, Src Addr: 140.113.131.7 (140.113.131.7), Dst Addr: 140.113.131.113 (140.113.131.113)
  User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
  Session Termination Protocol
  Status-Line: SIP/2.0 400 Bad Request
  Message Header
  Via: SIP/2.0/UDP 140.113.131.113;branch=z9hG4kQ9e448F022d81c1c4
  From: "sipv6"<sip:sipv6@3ffe:3600:0:3cd2:c3c:87b5:c13:5060>;tag=sipv6-d89017c7-20811562
  To: <sip:09440210280140@3ffe:3600:2::3:140.113.131.7>
  Call-ID: 11-20811562605A809DE5A9
  CSeq: 8 INVITE
  Content-Length: 0
  Status-Line: SIP/2.0 400 Bad Request
  0010 01 59 d4 c7 00 00 40 11 85 83 8c 71 83 07 8c 71 .Y....@..s.g...q
  0020 81 71 13 c4 01 45 aa be 53 49 50 2f 32 2e .q.....E..SIP/2.
  0030 30 20 34 30 30 20 42 81 64 20 52 85 71 75 65 73 0 400 Ba d Reques
  0040 74 68 0a 58 69 61 3a 20 33 48 50 2f 32 2e 30 2f t.Via: SIP/2.0/
  0050 55 44 50 20 31 34 30 2e 31 31 33 2e 31 33 31 2e UDP 140.113.131.
  0060 31 31 33 3a 31 30 36 30 3b 62 72 61 6a 63 68 3d 113:5060 (branch=
  0070 7a 29 68 47 34 62 4b 19 65 34 34 38 66 30 32 32 z9hG4kQ9 e448F022
  File: Cisco7940-Proxy-84-80w.pcap [Ethernet II] [IP: 4 D: 4 M: 0]
  
```

* 未修改SIP訊息中的From與To欄位

- Solution : modified “From” and “To” header fields in SIP.

Cisco PSTN Gateway



* 未修改SIP訊息中的From與To欄位

* 未修改SDP訊息中的o欄位

- Solution : modified “**From**”&”**To**” header fields in SIP and “**o**” header field in SDP.

Problem: when SIPv6 UA dials my cellular phone number, Cisco PSTN Gateway returns “200 OK” to SIPv6 UA. But SIPv6 UA doesn’t send “ACK” to PSTN Gateway.

Test Results

- Modified SIP and SDP header fields for SIP routing
 - Request-URI、Contact、Via
 - c、m
- Success: SIP message flow is correct and RTP sessions are set up.
 - CCL SkinUA, Pingtel 2.1.10, snow 200, CCL PSTN Gateway
- Fail: different error message for each case
 - Windows Messenger 4.7 (show Error Message in Windows Messenger but no response)
 - Cisco IP Phone 7940 Series (return “400 Bad Request”)
 - Cisco PSTN Gateway (return “400 Bad Request” and “399 Invalid SDP Information”)

Conclusions

- Currently, NTP SIPv6 UA can communicate with all SIPv4 UAs and CCL PSTN gateway that are deployed in NTP VoIP platform.
- However, NTP SIPv6 UA cannot communicate with CISCO PSTN gateway, and CCL PCA (IPv6 SIP UA) cannot communicate with CISCO PSTN gateway and Pingtel hardware-based SIP phone.
- In next week, we will have more tests and find out the problems. Then we will fixed the problems in one week.
- Finally, the *SIPv6 Translator* can bridge the traffic between SIPv6 network and SIPv4 network (i.e., NTP VoIP platform), and we will start to write down our contributions as a paper.

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