NAT Traversal for VoIP

Dr. Quincy Wu

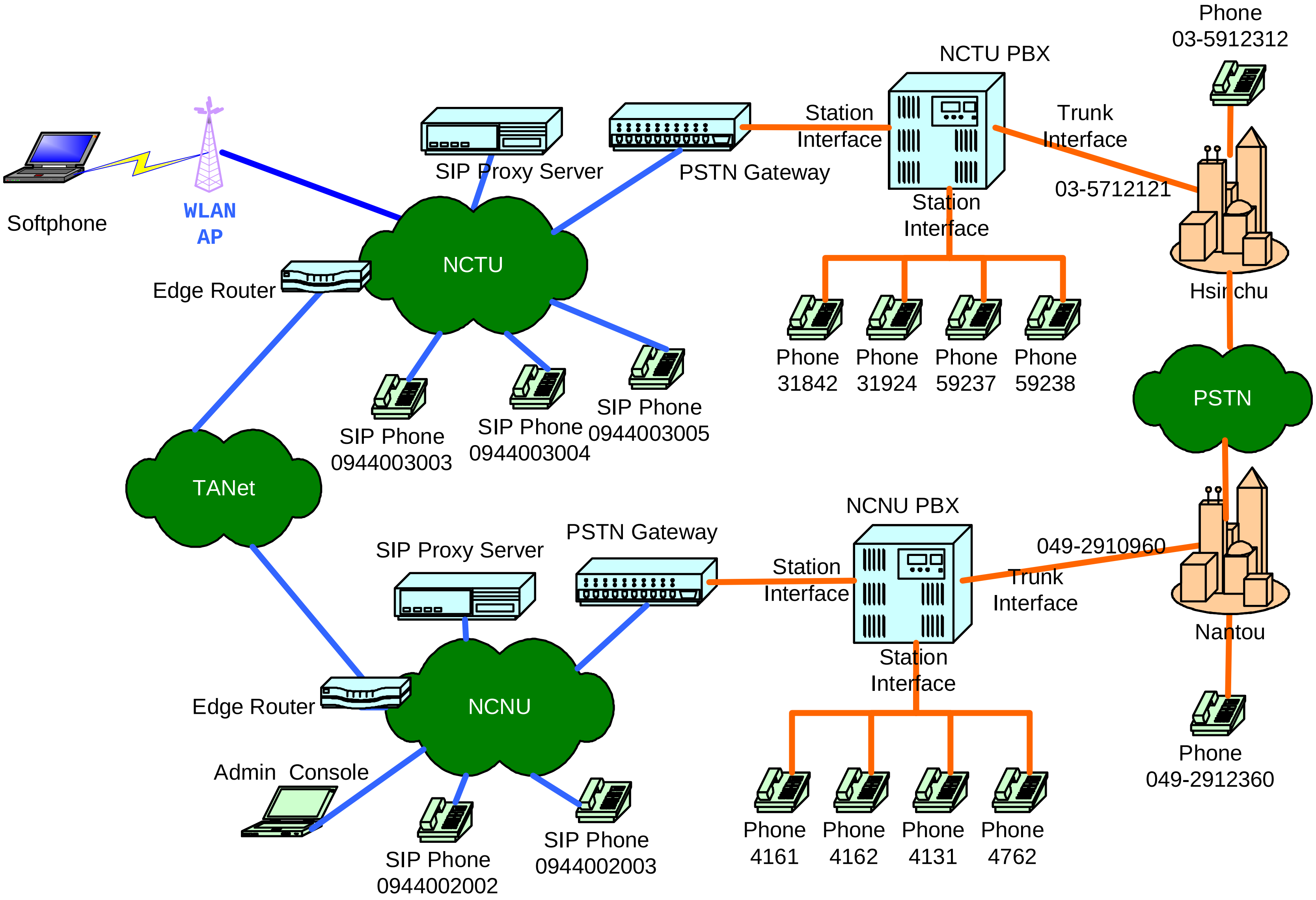
National Chi Nan University

Email: solomon@ipv6.club.tw

NAT Traversal

- Where is NAT
- What is NAT
- Types of NAT
- NAT Problems
- NAT Solutions
- Program Download

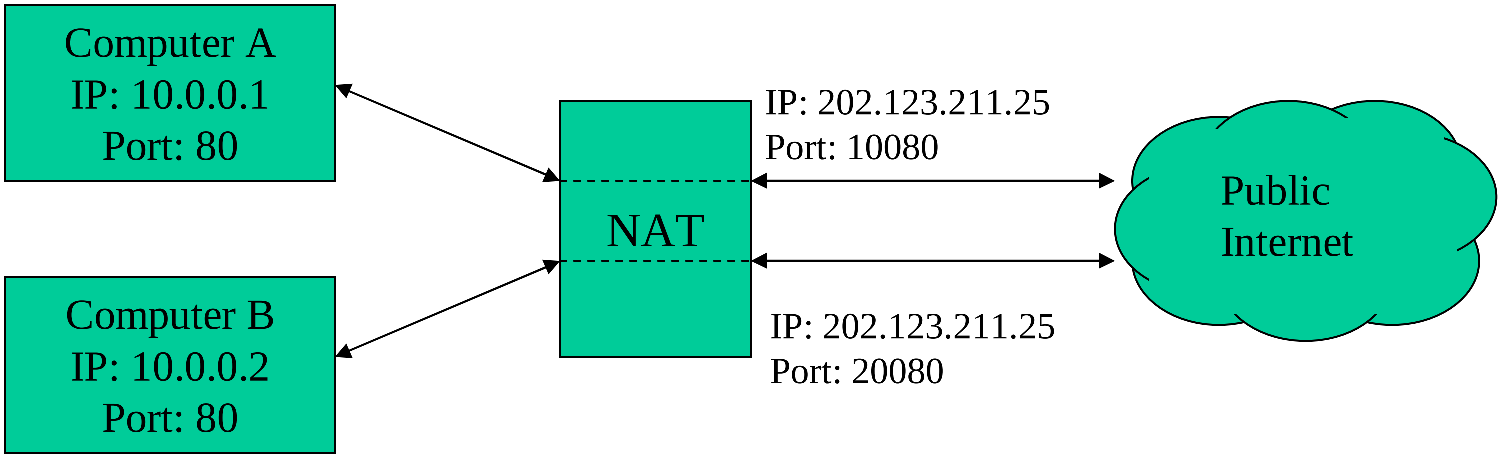
NTP VoIP Platform



What is NAT

- NAT - Network Address Translation
  - RFC 3022 - Traditional IP Network Address Translator (Traditional NAT)
  - RFC 1918 - Address Allocation for Private Internets (BCP 5)
  - RFC 2993 - Architectural Implications of NAT
  - RFC 3027 - Protocol Complications with the IP Network Address Translator
  - RFC 3235 - Network Address Translator (NAT)-Friendly Application Design Guidelines
- Convert Network Address (and Port) between private and public realm
- Works on IP layer
- Transparent for Application

NAT Schematic



Computer A
IP: 10.0.0.1
Port: 80

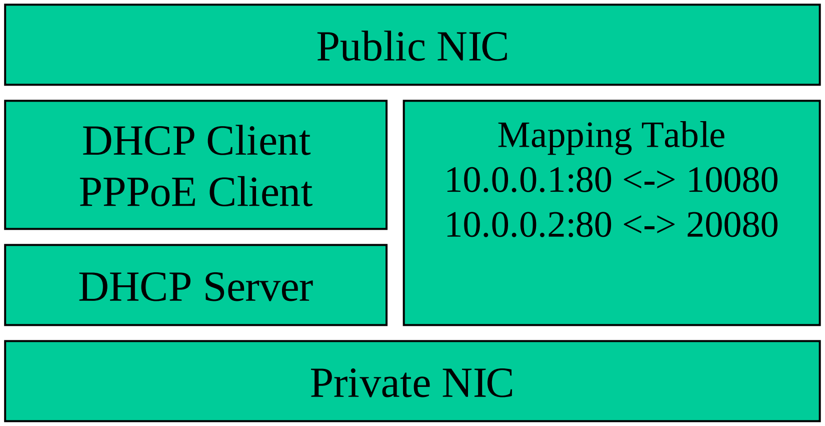
Computer B
IP: 10.0.0.2
Port: 80

NAT

IP: 202.123.211.25
Port: 10080

IP: 202.123.211.25
Port: 20080

Public
Internet



Public NIC

DHCP Client
PPPoE Client

DHCP Server

Mapping Table
10.0.0.1:80 <-> 10080
10.0.0.2:80 <-> 20080

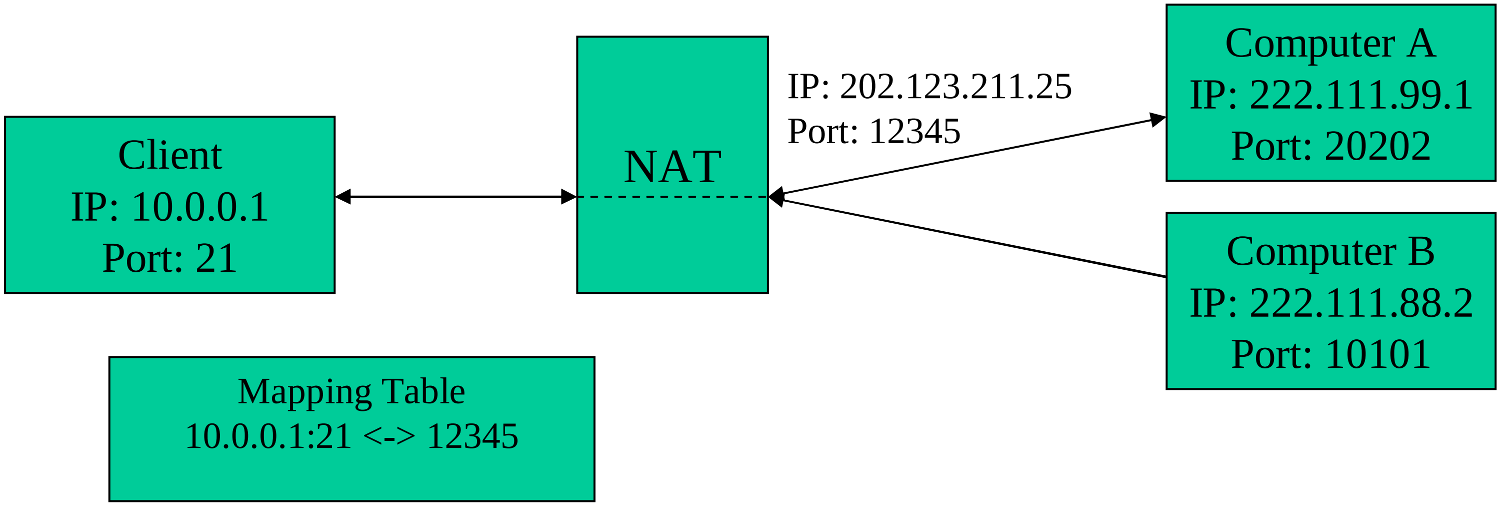
Private NIC

Types of NAT

- Full Cone
- Restricted Cone
- Port Restricted Cone
- Symmetric

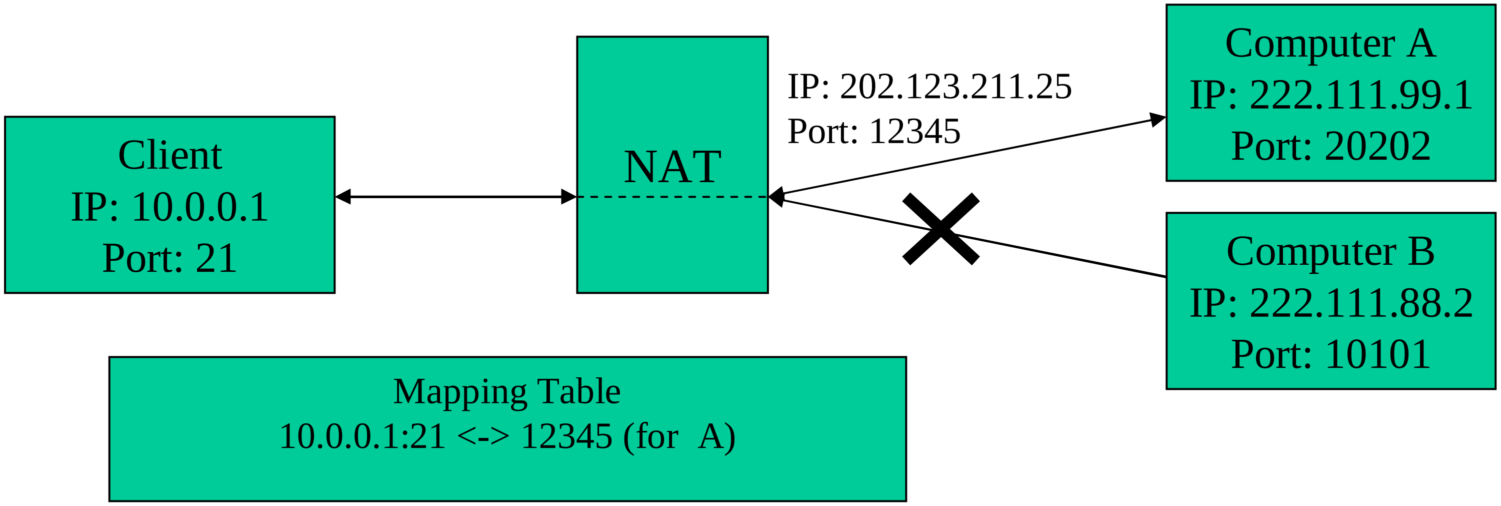
Full Cone NAT

- Client send a packet to public address A.
- NAT allocate a public port (12345) for private port (21) on the client.
- Any incoming packet (from A or B) to public port (12345) will dispatch to private port (21) on the client.



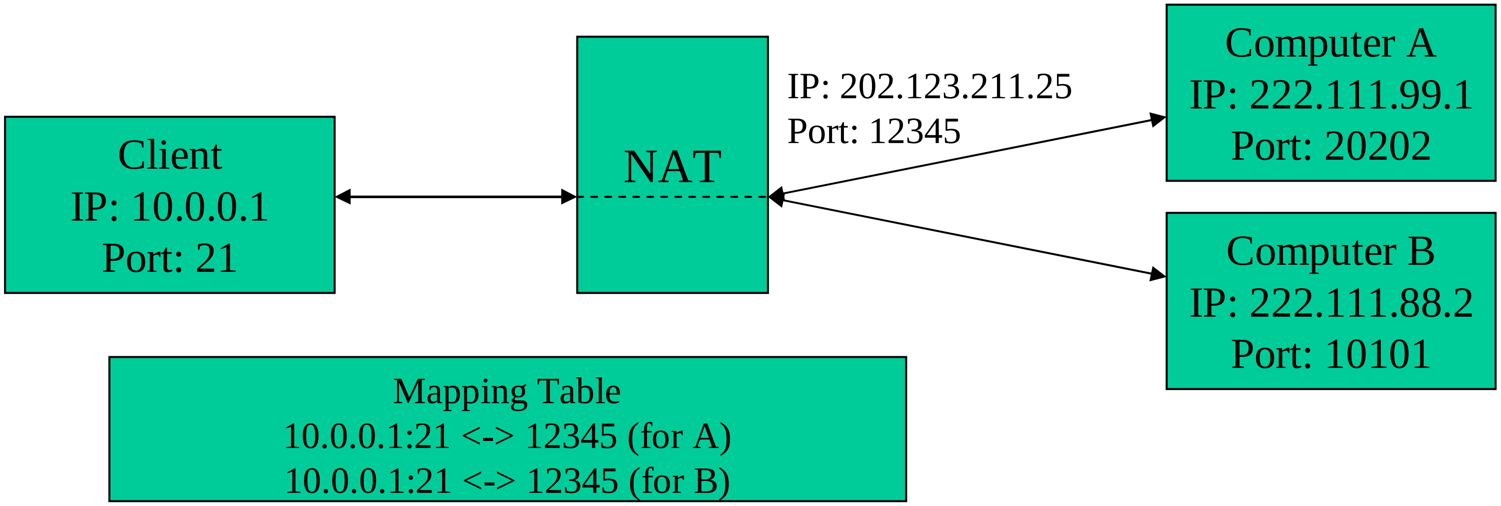
Restricted Cone NAT (1/2)

- Client send a packet to public address A.
- NAT allocate a public port (12345) for private port (21) on the client.
- Only incoming packet from A to public port (12345) will dispatch to private port (21) on the client.



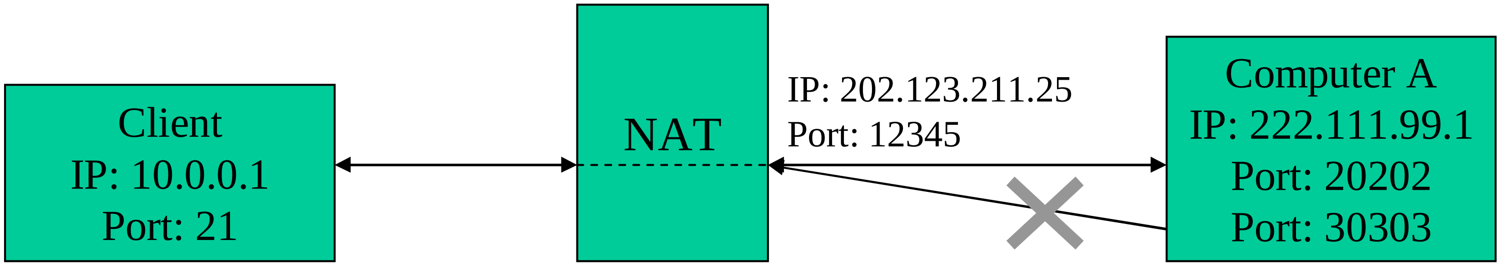
Restricted Cone NAT (2/2)

- Client send another packet to public address B.
- NAT will reuse allocated public port (12345) for private port (21) on the client.
- Incoming packet from B to public port (12345) will now dispatch to private port (21) on the client.



Port Restricted Cone NAT

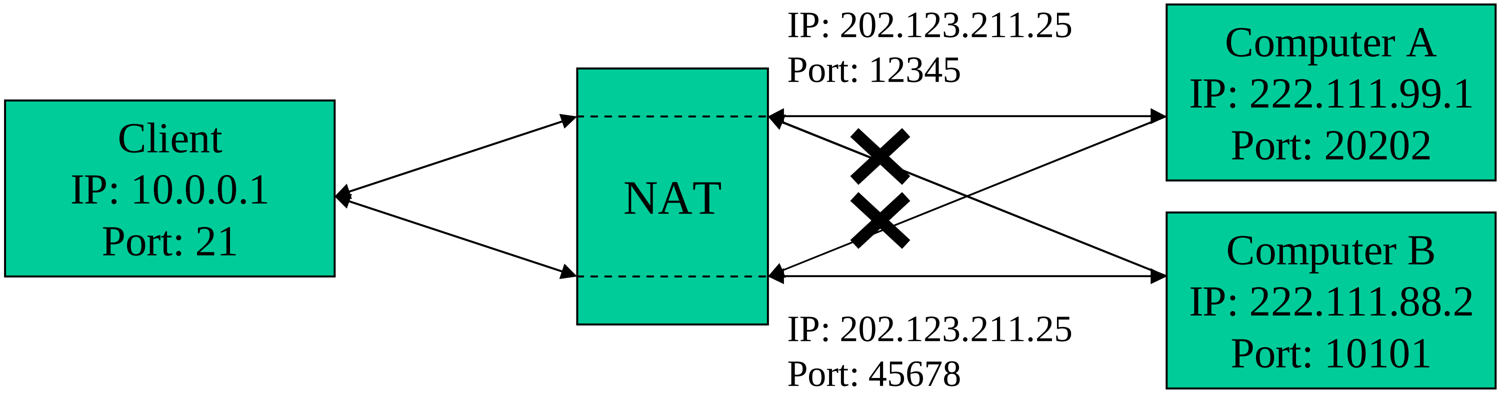
- Client send a packet to public address A port 20202.
- NAT will allocate a public port (12345) for private port (21) on the client.
- Only incoming packet from address A and port 20202 to public port (12345) will dispatch to private port (21) on the client.



Mapping Table
10.0.0.1:21 <-> 12345 (for A : 20202)
10.0.0.1:21 <-> 12345 (for A : 30303)

Symmetric NAT

- NAT allocate a public port each time the client send a packet to different public address and port
- Only incoming packet from the original mapped public address and port will dispatch to private port on client



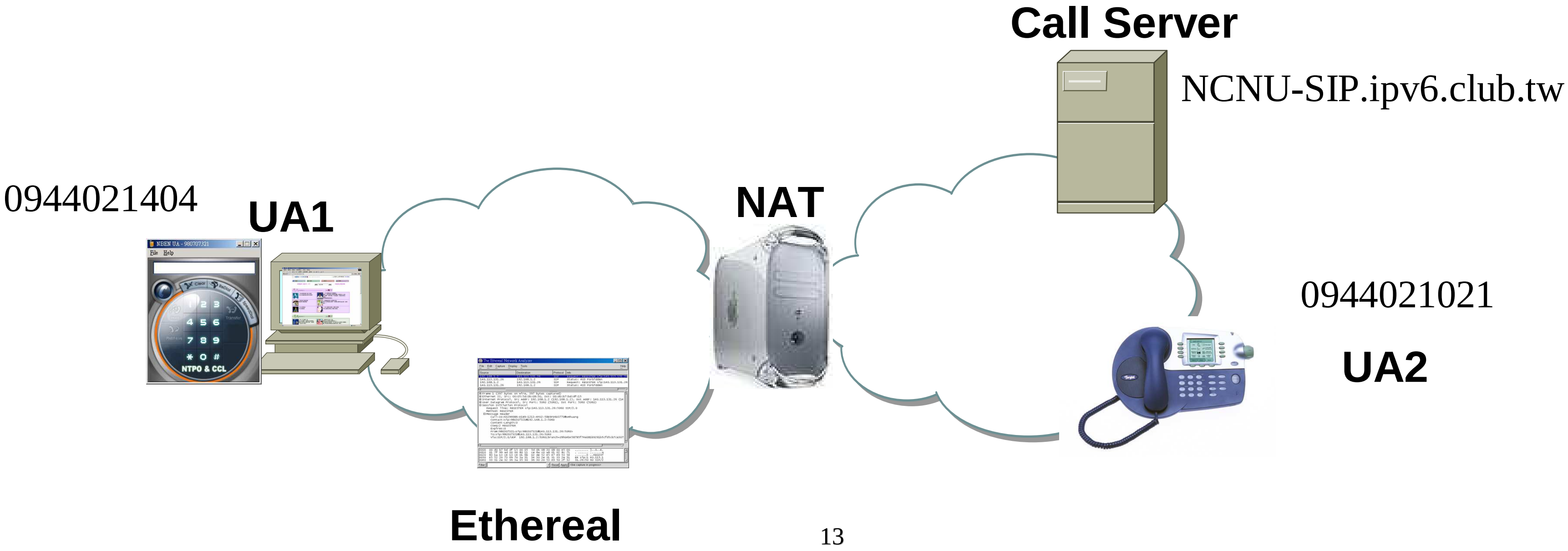
Mapping Table
10.0.0.1:21 <-> 12345 (for A : 20202)
10.0.0.1:21 <-> 45678 (for B : 10101)

VoIP Protocol and NAT

- NAT convert IP addresses on IP layer
- Problem 1:
  - SIP, H.323, Megaco and MGCP are application layer protocol but contain IP address/port info in messages, which is not translated by NAT
- Problem 2:
  - Private client must send a outgoing packet first (to create a mapping on NAT) to receive incoming packet

Lab Environment

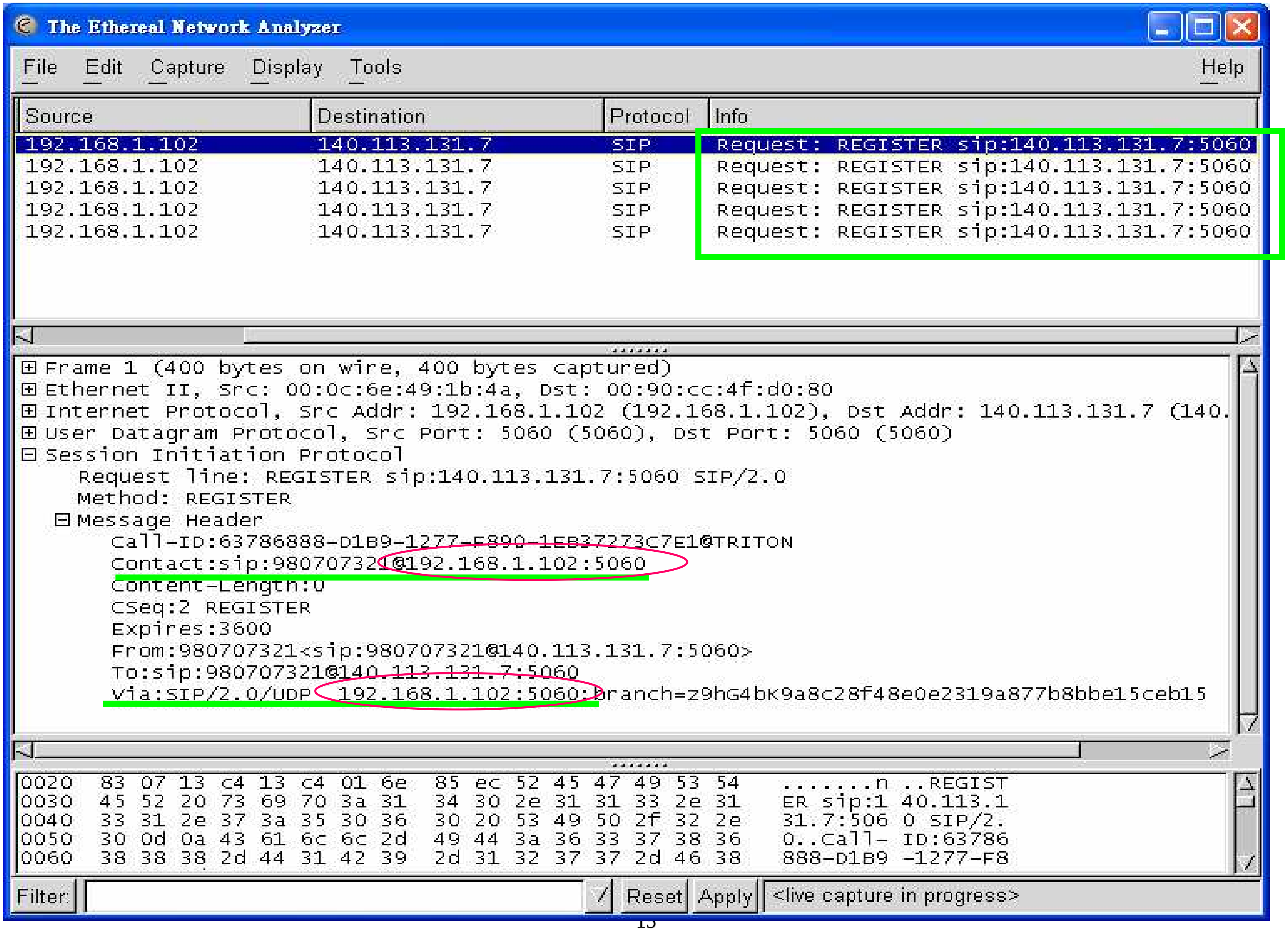
- UA1: UA behind NAT.
- UA2: SIP device outside NAT.
- Call Server: SIP-express router 0.8.12.
- NAT: Linux Fedora Core 2.
- Packet Capturer: Ethereal-0.9.15.



The Problem (1/2)

- Due to private address, the Via header and Contact address in SIP messages sent by UA1 are incorrect.
  - With incorrect Via header, responses of messages sent by UA1 cannot be routed back.
  - With incorrect Contact address in REGISTER messages, call server cannot inform UA1 the incoming calls.
    - ☞UA1 can only act as a calling party.

Incorrect REGISTER Message


The Ethereal Network Analyzer
File Edit Capture Display Tools Help

```
Source          Destination     Protocol  Info
192.168.1.102   140.113.131.7   SIP       Request: REGISTER sip:140.113.131.7:5060
192.168.1.102   140.113.131.7   SIP       Request: REGISTER sip:140.113.131.7:5060
192.168.1.102   140.113.131.7   SIP       Request: REGISTER sip:140.113.131.7:5060
192.168.1.102   140.113.131.7   SIP       Request: REGISTER sip:140.113.131.7:5060
192.168.1.102   140.113.131.7   SIP       Request: REGISTER sip:140.113.131.7:5060
```

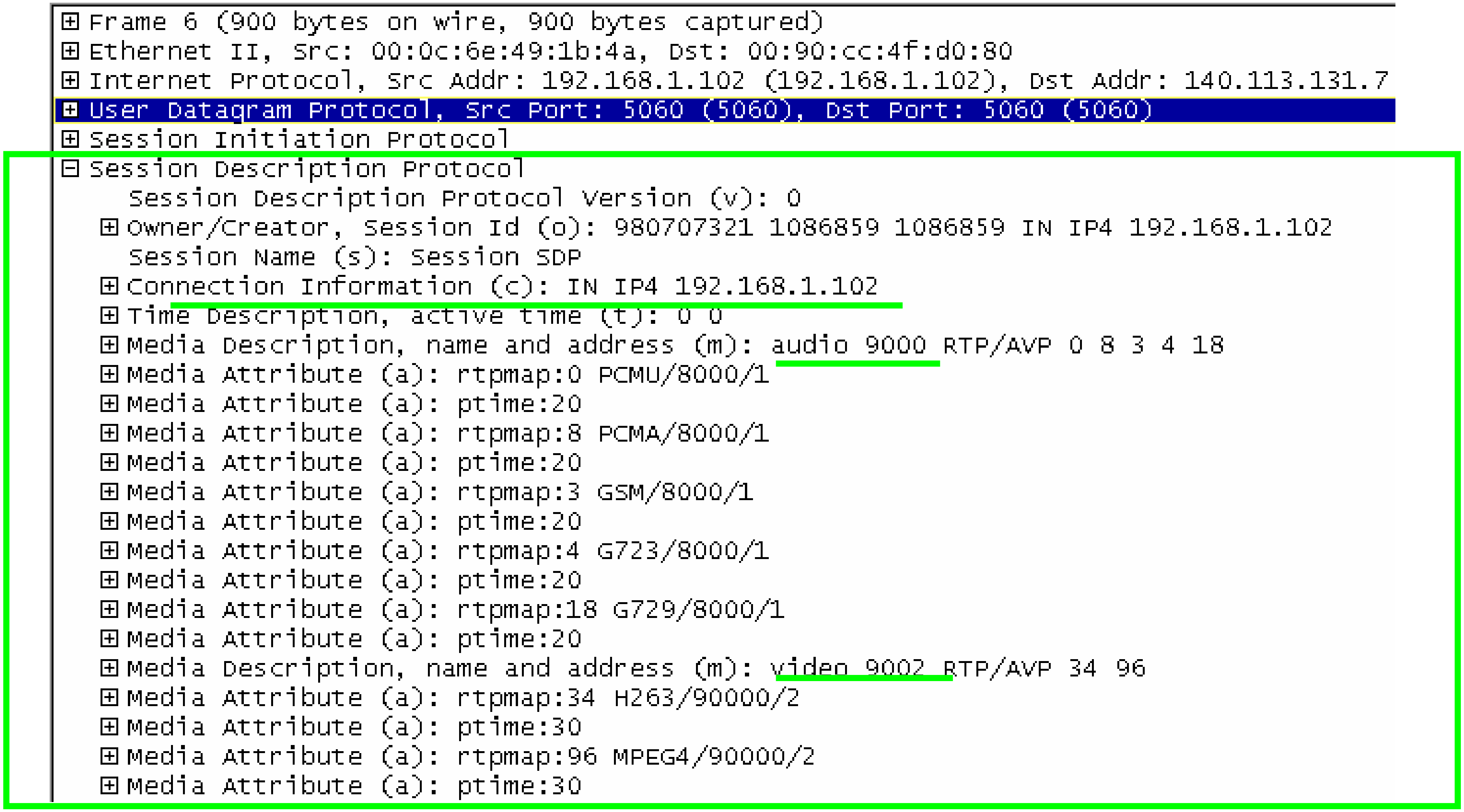
```
⊞ Frame 1 (400 bytes on wire, 400 bytes captured)
⊞ Ethernet II, Src: 00:0c:6e:49:1b:4a, Dst: 00:90:cc:4f:d0:80
⊞ Internet Protocol, Src Addr: 192.168.1.102 (192.168.1.102), Dst Addr: 140.113.131.7 (140.
⊞ User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
⊟ Session Initiation Protocol
    Request line: REGISTER sip:140.113.131.7:5060 SIP/2.0
    Method: REGISTER
  ⊟ Message Header
      Call-ID:63786888-D1B9-1277-F890-1EB37273C7E1@TRITON
      Contact:sip:980707321@192.168.1.102:5060
      Content-Length:0
      CSeq:2 REGISTER
      Expires:3600
      From:980707321<sip:980707321@140.113.131.7:5060>
      To:sip:980707321@140.113.131.7:5060
      Via:SIP/2.0/UDP  192.168.1.102:5060;branch=z9hG4bK9a8c28f48e0e2319a877b8bbe15ceb15
```

```
0020   83 07 13 c4 13 c4 01 6e  85 ec 52 45 47 49 53 54   .......n ..REGIST
0030   45 52 20 73 69 70 3a 31  34 30 2e 31 31 33 2e 31   ER sip:1 40.113.1
0040   33 31 2e 37 3a 35 30 36  30 20 53 49 50 2f 32 2e   31.7:506 0 SIP/2.
0050   30 0d 0a 43 61 6c 6c 2d  49 44 3a 36 33 37 38 36   0..Call- ID:63786
0060   38 38 38 2d 44 31 42 39  2d 31 32 37 37 2d 46 38   888-D1B9 -1277-F8
```

The Problem (2/2)

- When UA1 initiate a call, the connection information for media establishment in SDP are also incorrect.
  - UA2 gets a private peer address, the RTP packets from UA2 cannot be routed to UA1.
  - Media can only be sent from UA1 to UA2.

Incorrect Fields in SDP of INVITE Message



```
⊞ Frame 6 (900 bytes on wire, 900 bytes captured)
⊞ Ethernet II, Src: 00:0c:6e:49:1b:4a, Dst: 00:90:cc:4f:d0:80
⊞ Internet Protocol, Src Addr: 192.168.1.102 (192.168.1.102), Dst Addr: 140.113.131.7
⊞ User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
⊞ Session Initiation Protocol
⊟ Session Description Protocol
    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): 980707321 1086859 1086859 IN IP4 192.168.1.102
    Session Name (s): Session SDP
  ⊞ Connection Information (c): IN IP4 192.168.1.102
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 9000 RTP/AVP 0 8 3 4 18
  ⊞ Media Attribute (a): rtpmap:0 PCMU/8000/1
  ⊞ Media Attribute (a): ptime:20
  ⊞ Media Attribute (a): rtpmap:8 PCMA/8000/1
  ⊞ Media Attribute (a): ptime:20
  ⊞ Media Attribute (a): rtpmap:3 GSM/8000/1
  ⊞ Media Attribute (a): ptime:20
  ⊞ Media Attribute (a): rtpmap:4 G723/8000/1
  ⊞ Media Attribute (a): ptime:20
  ⊞ Media Attribute (a): rtpmap:18 G729/8000/1
  ⊞ Media Attribute (a): ptime:20
  ⊞ Media Description, name and address (m): video 9002 RTP/AVP 34 96
  ⊞ Media Attribute (a): rtpmap:34 H263/90000/2
  ⊞ Media Attribute (a): ptime:30
  ⊞ Media Attribute (a): rtpmap:96 MPEG4/90000/2
  ⊞ Media Attribute (a): ptime:30
```

Solving NAT Traversal Problems

- Target:
  - Discover mapped public IP & port for private IP & port
  - Use mapped public IP & port in application layer message
  - Keep this mapping valid
- Timing Issue
  - NAT will automatically allocate a public port for a private address & port if need.
  - NAT will release the mapping if the public port is “idle”
    - ☞No TCP connection on the port
    - ☞No UDP traffic on the port for a period (45 sec ~ 5 min)
  - Keep a TCP connection to target
  - Send UDP packet to target every specified interval

NAT Solutions

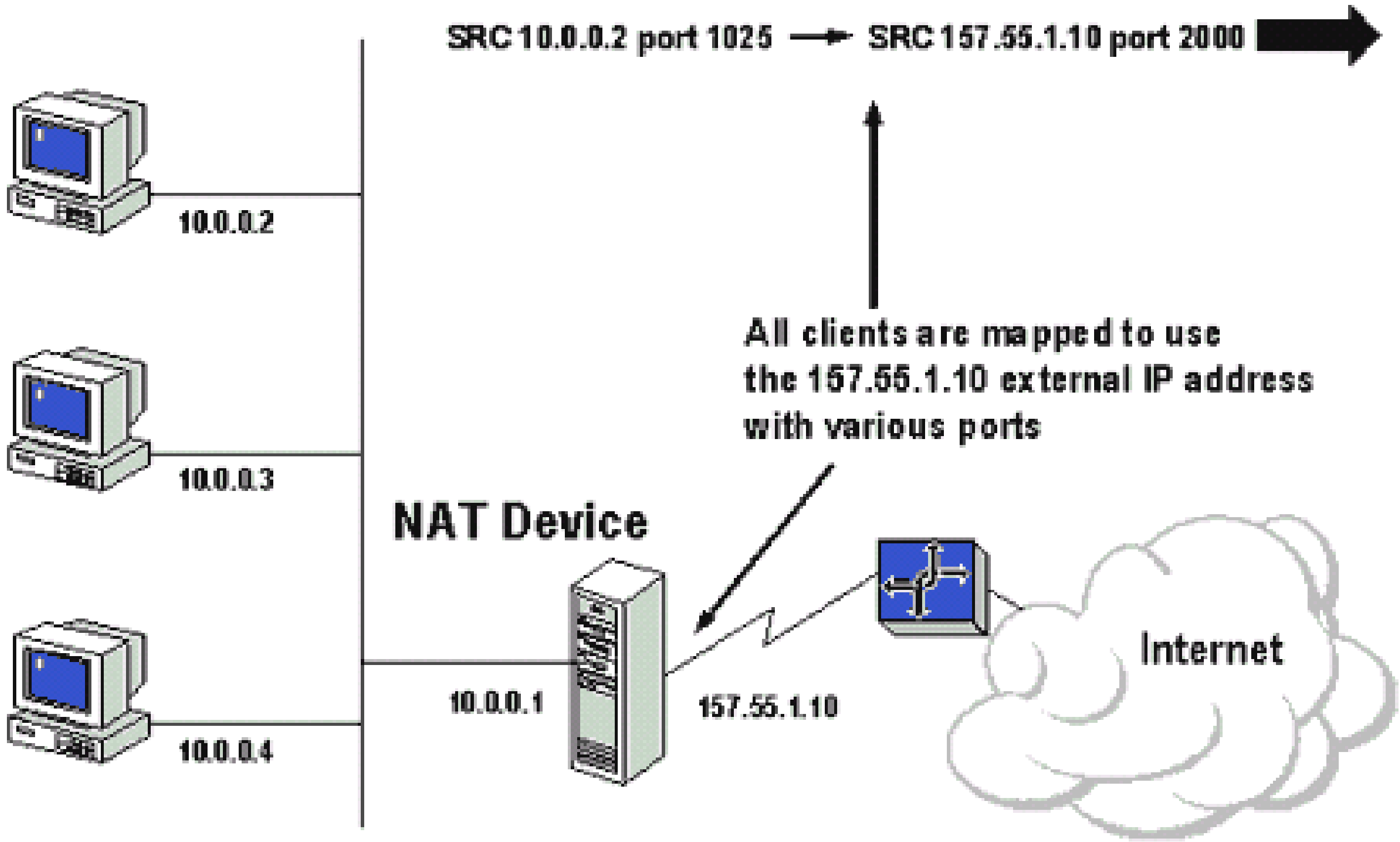
- IPv6 (Internet Protocol Version 6)
- UPnP (Universal Plug-and-Play)
  - UPnP Forum - http://www.upnp.org/
- VPN (Virtual Private Network)
- Proprietary protocol by NAT/Firewall
  - SIP ALG (Application Level Gateway)
  - No standard now. Not applicable for existing NATs.
- SIP extensions for NAT traversal
  - RFC 3581 - rport
  - Works for SIP only, can not help RTP to pass through NAT
- STUN (Simple Traversal of UDP Through Network Address Translators)
  - RFC 3489
  - Works except symmetric NAT
- TURN (Traversal Using Relay NAT)
  - draft-rosenberg-midcom-turn-08
  - for symmetric NAT

UPnP – Universal Plug-and-Play

NAT Traversal with UPnP

- 目的
  - 使 NAT 網路內的機器確切知道對外所用的 Public IP 位址資訊
- 解法
  - 讓 NAT Device 可與 NAT 內的應用程式溝通, 交換位址資訊
  - 定義 NAT Device 爲一 UPnP Device (IGD)
    - ☞IGD -- Internet Gateway Device

NAT 網路架構


SRC 10.0.0.2 port 1025
SRC 157.55.1.10 port 2000
10.0.0.2
10.0.0.3
10.0.0.4
All clients are mapped to use the 157.55.1.10 external IP address with various ports
NAT Device
10.0.0.1
157.55.1.10
Internet

UPnP IGD

- 提供以下UPnP 功能
  - 取得 public IP 位址
  - 取得現有 port mapping
  - 新增/移除 port mapping
  - 指定 mapping 的存續時間

利用 UPnP 取得位址資訊

- NAT 內主機可利用 UPnP *Control* Message 通知 IGD 增加一 Port Mapping
- 範例:
  - 本機位址: 192.168.0.14
  - 正在本機 port 10001 上聽 UDP 封包
  - 希望能在 IGD 新增一 port mapping

IGD Control Message

```
POST /upnphost/udhisapi.dll?control=uuid:c3038e95-ea88-4d5c-98ff-3ad68f7aaa32+urn:upnp-org:serviceId:WANIPConn1 HTTP/1.1
Host: 192.168.0.1:2869
Content-Length: 734
Content-Type: text/xml; charset="utf-8"
SOAPAction: "urn:schemas-upnp-org:service:WANIPConnection:1#AddPortMapping"

<SOAP-ENV:Envelope
xmlns:SOAP-ENV="http://schemas.xmlsoap.org/soap/envelope/"
    SOAP-ENV:encodingStyle="http://schemas.xmlsoap.org/soap/encoding/">
  <SOAP-ENV:Body>
    <u:AddPortMapping xmlns:u="urn:schemas-upnp-org:service:WANIPConnection:1">
      <NewRemoteHost></NewRemoteHost>
      <NewExternalPort>17769</NewExternalPort>
      <NewProtocol>UDP</NewProtocol>
      <NewInternalPort>10001</NewInternalPort>
      <NewInternalClient>192.168.0.146</NewInternalClient>
      <NewEnabled>1</NewEnabled>
      <NewPortMappingDescription>s2EAYp (192.168.0.146:10001) 17769 UDP</NewPortMappingDescription>
      <NewLeaseDuration>0</NewLeaseDuration>
    </u:AddPortMapping>
  </SOAP-ENV:Body>
</SOAP-ENV:Envelope>
```

Current Defects of UPnP

- 目前尚未解決的問題
  - Aging 問題
    - ☞程式需自行清除 port mapping
  - 安全性問題
    - ☞UPnP 尚未提供認證機制
  - Multi-level NAT
    - ☞NAT 內的裝置只能存取前一層的 IP 位址

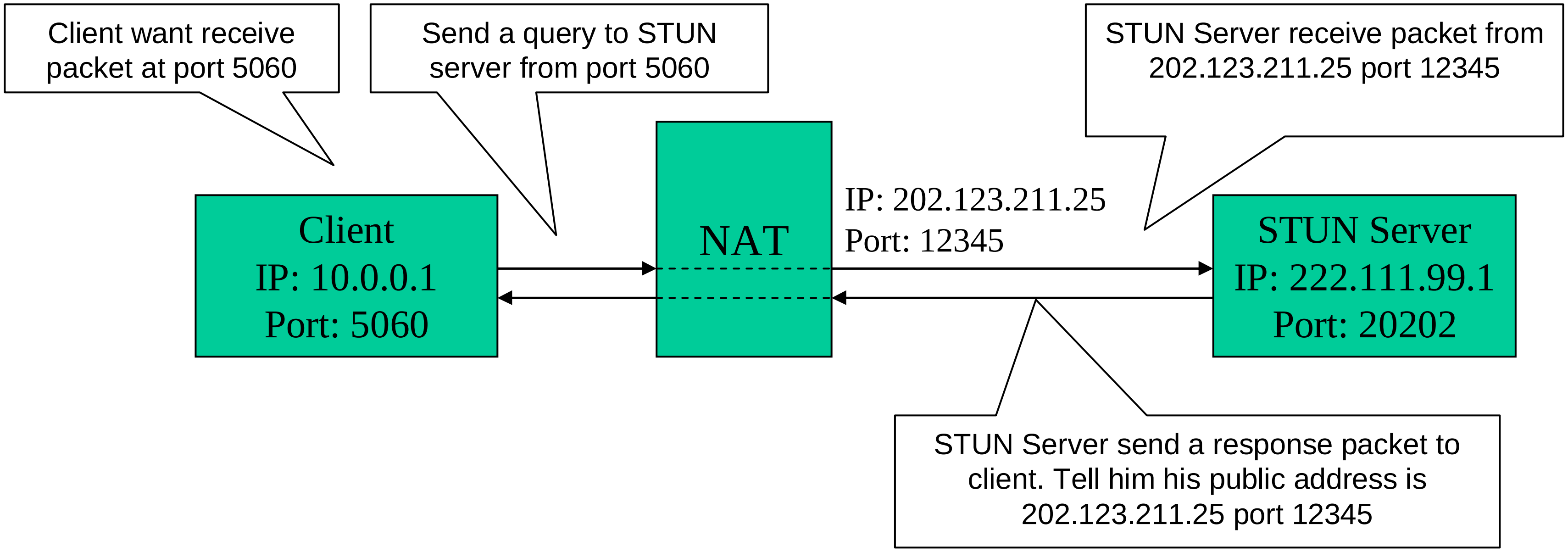
Simple Traversal of UDP Through Network Address Translators (STUN)

STUN (RFC 3489)

- A mechanism for a socket behind NAT(s) to get its mapped (IP,port) on Internet.
- Check whether UA is behind NAT.
  - If not true, the STUN mechanism is not applied.
- When new socket is created, use this socket to request its mapped (IP,port) from STUN server.
  - The response IP is stored in a string buffer.
  - The response port is saved in a table, using source port as key.
- When UA wants to stuff local IP or port in a message, it will first look up mapped IP or port in the table.

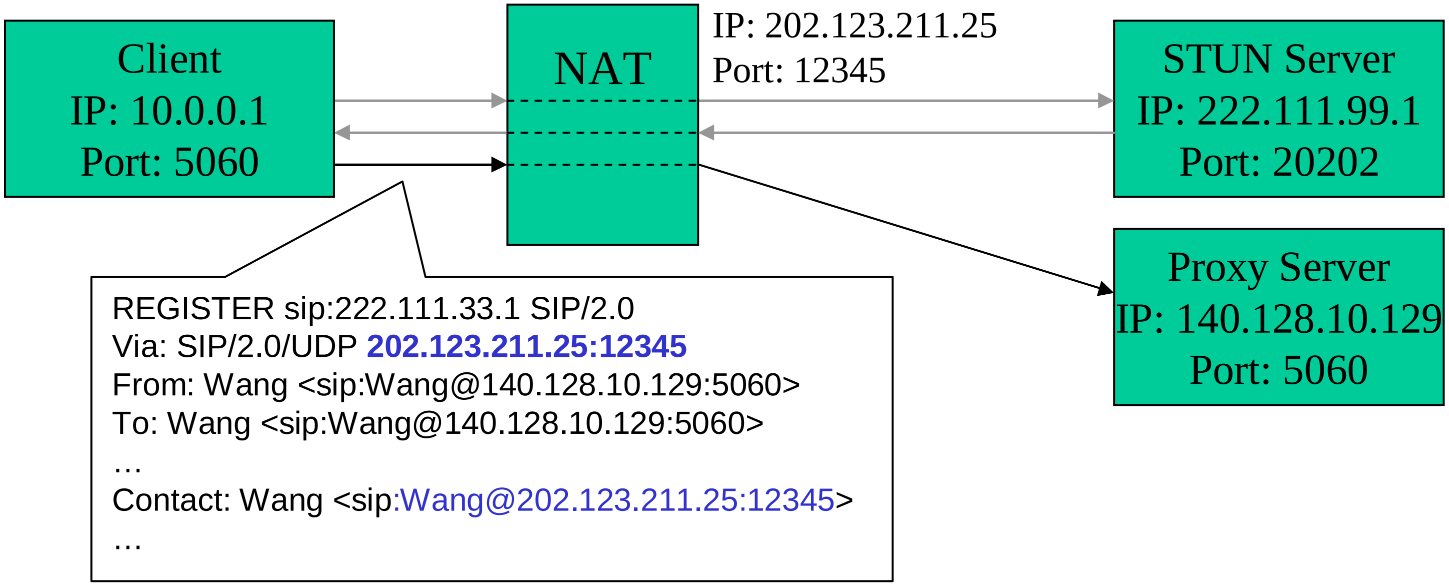
STUN Server

- Allow clients to discover if it is behind a NAT, what type of NAT it is, and the public address & port NAT will use.
- Very Simple Protocol, Easy to implement, Little load


Client want receive packet at port 5060
Send a query to STUN server from port 5060
STUN Server receive packet from 202.123.211.25 port 12345
Client
IP: 10.0.0.1
Port: 5060
NAT
IP: 202.123.211.25
Port: 12345
STUN Server
IP: 222.111.99.1
Port: 20202
STUN Server send a response packet to client. Tell him his public address is 202.123.211.25 port 12345

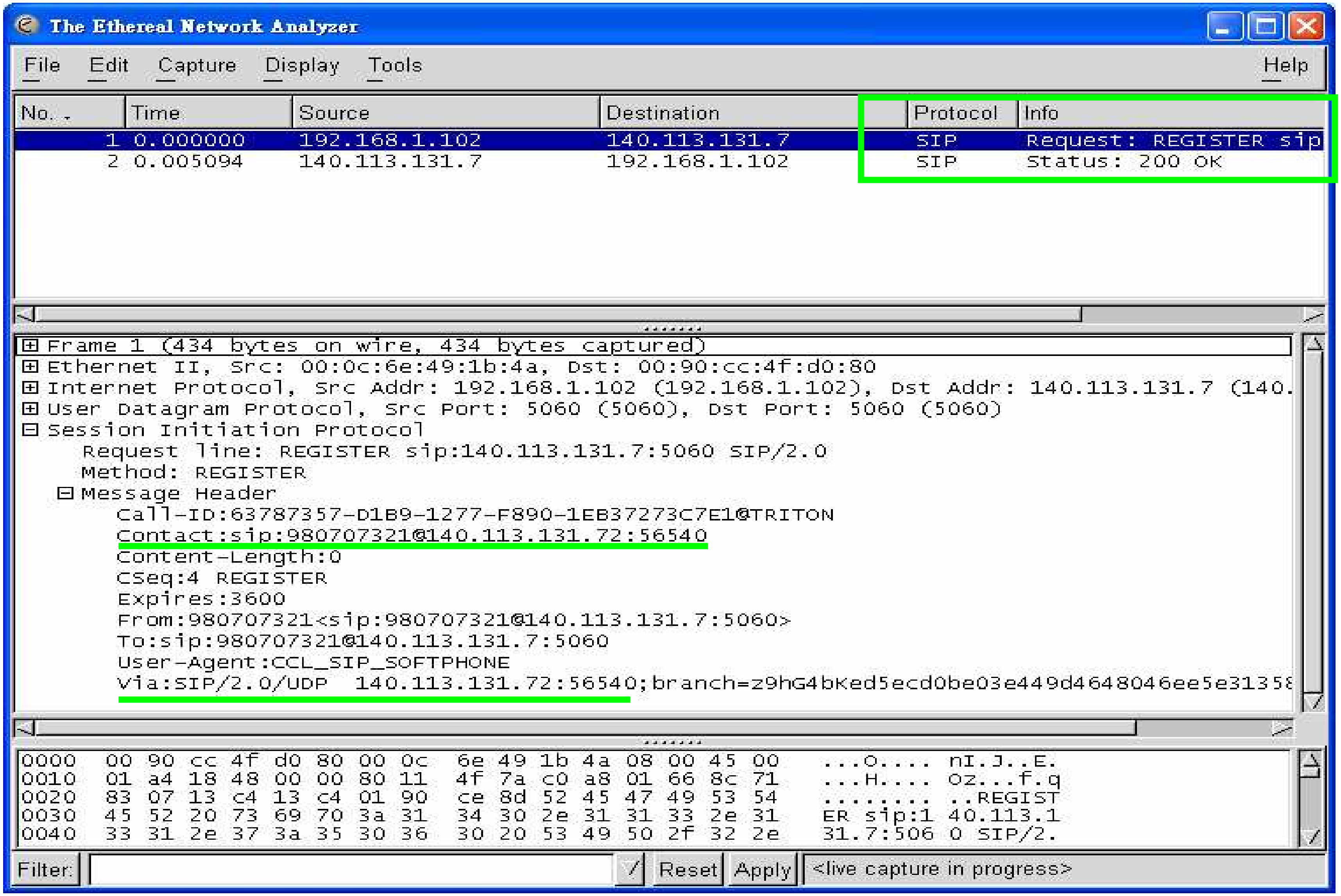
Use STUN for SIP Registration

- Use port 5060 to send a packet to STUN Server
- Receive public address & port mapped to client:5060 from STUN Server
- Fill the SIP register message with client’s public address & port, send to proxy server



```
REGISTER sip:222.111.33.1 SIP/2.0
Via: SIP/2.0/UDP 202.123.211.25:12345
From: Wang <sip:Wang@140.128.10.129:5060>
To: Wang <sip:Wang@140.128.10.129:5060>
…
Contact: Wang <sip:Wang@202.123.211.25:12345>
…
```

Corrected SIP Message


The Ethereal Network Analyzer
File Edit Capture Display Tools Help

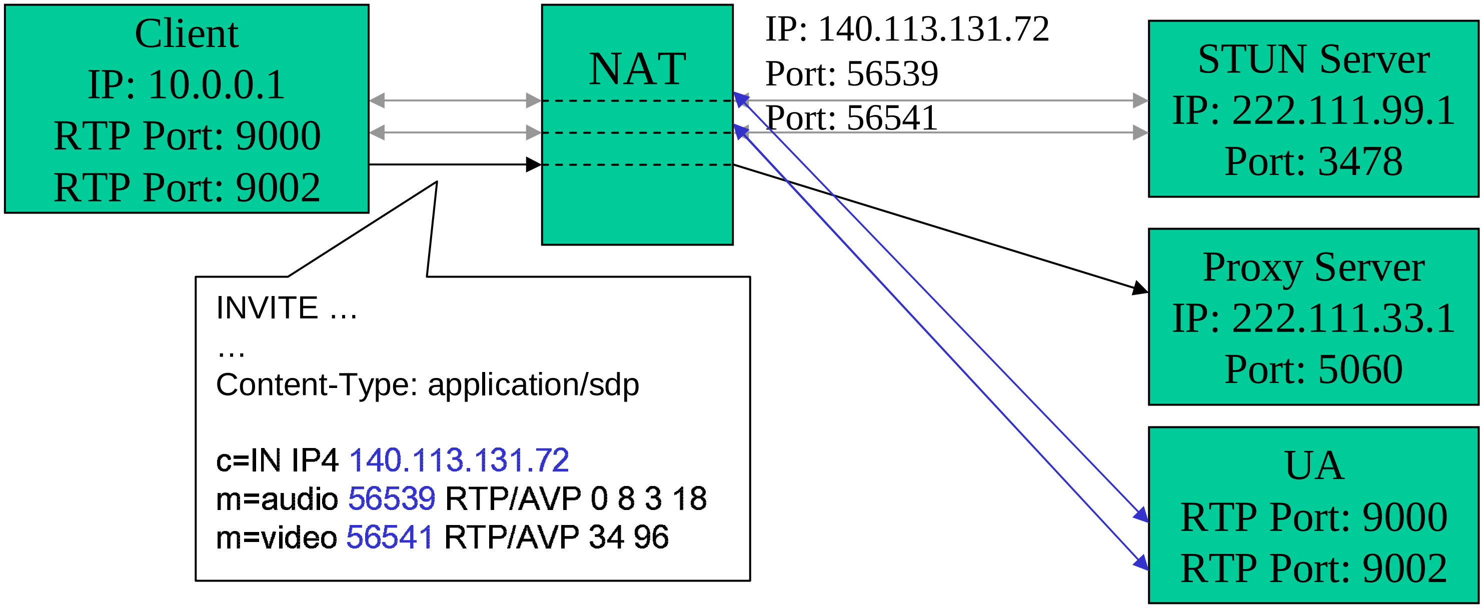
```
No. .    Time        Source            Destination       Protocol  Info
    1 0.000000    192.168.1.102     140.113.131.7     SIP       Request: REGISTER sip
    2 0.005094    140.113.131.7     192.168.1.102     SIP       Status: 200 OK
```

```
⊞ Frame 1 (434 bytes on wire, 434 bytes captured)
⊞ Ethernet II, Src: 00:0c:6e:49:1b:4a, Dst: 00:90:cc:4f:d0:80
⊞ Internet Protocol, Src Addr: 192.168.1.102 (192.168.1.102), Dst Addr: 140.113.131.7 (140.
⊞ User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
⊟ Session Initiation Protocol
    Request line: REGISTER sip:140.113.131.7:5060 SIP/2.0
    Method: REGISTER
  ⊟ Message Header
      Call-ID:63787357-D1B9-1277-F890-1EB37273C7E1@TRITON
      Contact:sip:980707321@140.113.131.72:56540
      Content-Length:0
      CSeq:4 REGISTER
      Expires:3600
      From:980707321<sip:980707321@140.113.131.7:5060>
      To:sip:980707321@140.113.131.7:5060
      User-Agent:CCL_SIP_SOFTPHONE
      Via:SIP/2.0/UDP  140.113.131.72:56540;branch=z9hG4bKed5ecd0be03e449d4648046ee5e3135
```

```
0000  00 90 cc 4f d0 80 00 0c  6e 49 1b 4a 08 00 45 00   ...O.... nI.J..E.
0010  01 a4 18 48 00 00 80 11  4f 7a c0 a8 01 66 8c 71   ...H.... Oz...f.q
0020  83 07 13 c4 13 c4 01 90  ce 8d 52 45 47 49 53 54   ........ ..REGIST
0030  45 52 20 73 69 70 3a 31  34 30 2e 31 31 33 2e 31   ER sip:1 40.113.1
0040  33 31 2e 37 3a 35 30 36  30 20 53 49 50 2f 32 2e   31.7:506 0 SIP/2.
```

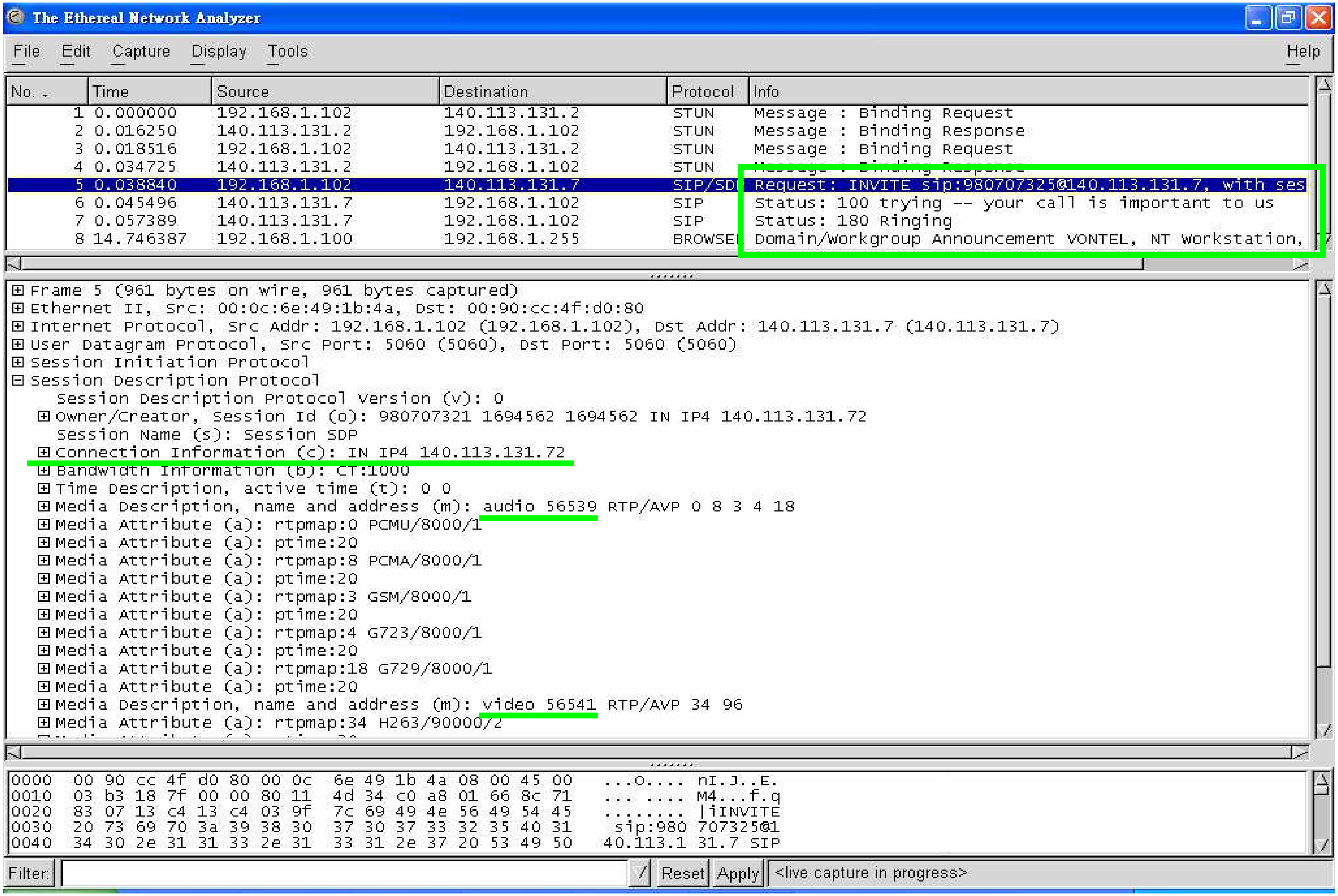
Use STUN for RTP

- Send two STUN queries from RTP port (9000 & 9002) to STUN Server
- Use replied public address & port in SDP



```
INVITE …
…
Content-Type: application/sdp

c=IN IP4 140.113.131.72
m=audio 56539 RTP/AVP 0 8 3 18
m=video 56541 RTP/AVP 34 96
```

Corrected SDP


The Ethereal Network Analyzer
File Edit Capture Display Tools Help
Filter:
Reset
Apply

```
No. .    Time        Source           Destination      Protocol  Info
      1 0.000000    192.168.1.102    140.113.131.2    STUN      Message : Binding Request
      2 0.016250    140.113.131.2    192.168.1.102    STUN      Message : Binding Response
      3 0.018516    192.168.1.102    140.113.131.2    STUN      Message : Binding Request
      4 0.034725    140.113.131.2    192.168.1.102    STUN      Message : Binding Response
      5 0.038840    192.168.1.102    140.113.131.7    SIP/SD    Request: INVITE sip:980707325@140.113.131.7, with ses
      6 0.045496    140.113.131.7    192.168.1.102    SIP       Status: 100 trying -- your call is important to us
      7 0.057389    140.113.131.7    192.168.1.102    SIP       Status: 180 Ringing
      8 14.746387   192.168.1.100    192.168.1.255    BROWSE    Domain/Workgroup Announcement VONTEL, NT Workstation,
```

```
⊞ Frame 5 (961 bytes on wire, 961 bytes captured)
⊞ Ethernet II, Src: 00:0c:6e:49:1b:4a, Dst: 00:90:cc:4f:d0:80
⊞ Internet Protocol, Src Addr: 192.168.1.102 (192.168.1.102), Dst Addr: 140.113.131.7 (140.113.131.7)
⊞ User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
⊞ Session Initiation Protocol
⊟ Session Description Protocol
     Session Description Protocol Version (v): 0
   ⊞ Owner/Creator, Session Id (o): 980707321 1694562 1694562 IN IP4 140.113.131.72
     Session Name (s): Session SDP
   ⊞ Connection Information (c): IN IP4 140.113.131.72
   ⊞ Bandwidth Information (b): CT:1000
   ⊞ Time Description, active time (t): 0 0
   ⊞ Media Description, name and address (m): audio 56539 RTP/AVP 0 8 3 4 18
   ⊞ Media Attribute (a): rtpmap:0 PCMU/8000/1
   ⊞ Media Attribute (a): ptime:20
   ⊞ Media Attribute (a): rtpmap:8 PCMA/8000/1
   ⊞ Media Attribute (a): ptime:20
   ⊞ Media Attribute (a): rtpmap:3 GSM/8000/1
   ⊞ Media Attribute (a): ptime:20
   ⊞ Media Attribute (a): rtpmap:4 G723/8000/1
   ⊞ Media Attribute (a): ptime:20
   ⊞ Media Attribute (a): rtpmap:18 G729/8000/1
   ⊞ Media Attribute (a): ptime:20
   ⊞ Media Description, name and address (m): video 56541 RTP/AVP 34 96
   ⊞ Media Attribute (a): rtpmap:34 H263/90000/2
```

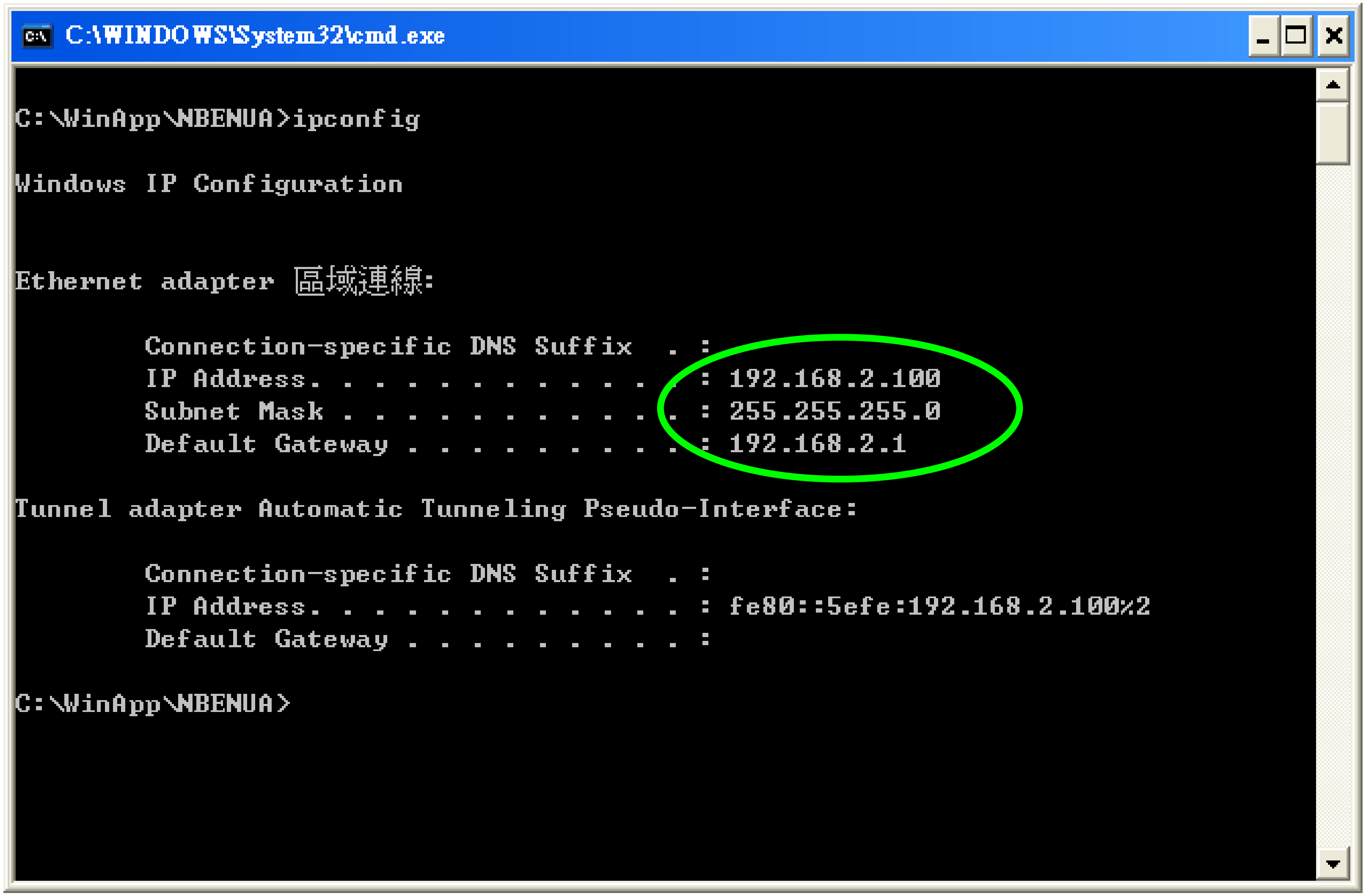
```
0000   00 90 cc 4f d0 80 00 0c  6e 49 1b 4a 08 00 45 00   ...O.... nI.J..E.
0010   03 b3 18 7f 00 00 80 11  4d 34 c0 a8 01 66 8c 71   ........ M4...f.q
0020   83 07 13 c4 13 c4 03 9f  7c 69 49 4e 56 49 54 45   ........ |iINVITE
0030   20 73 69 70 3a 39 38 30  37 30 37 33 32 35 40 31    sip:980 707325@1
0040   34 30 2e 31 31 33 2e 31  33 31 2e 37 20 53 49 50   40.113.1 31.7 SIP
```

<live capture in progress>

Download

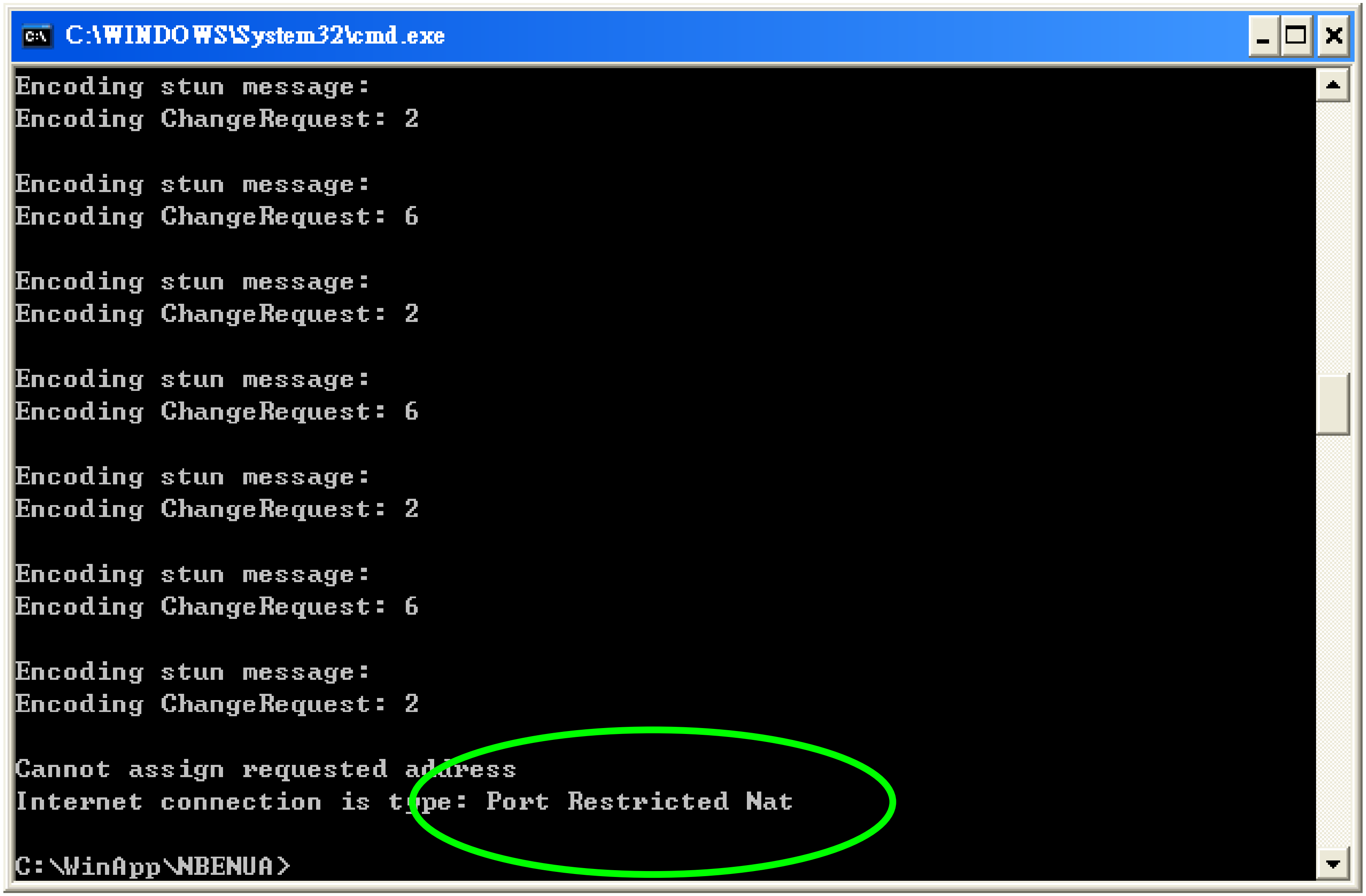
- STUN Client
  - A diagnosis tool which utilizes STUN mechanism to find out the type of NAT.
  - Download at http://voip.ipv6.club.tw/Download/
  - Usage:
    - ☞stun-client STUN.ipv6.club.tw
    - ☞stun-client –t STUN.ipv6.club.tw
    - ☞stun-client –p 5060 STUN.ipv6.club.tw
  - Note: Be sure to close any running SIP UA before you run the STUN client. (why?)
- Many commercial SIP UAs support STUN
  - X-Lite (softphone)
  - Snom (hardphone)

Running STUN Client on a PC



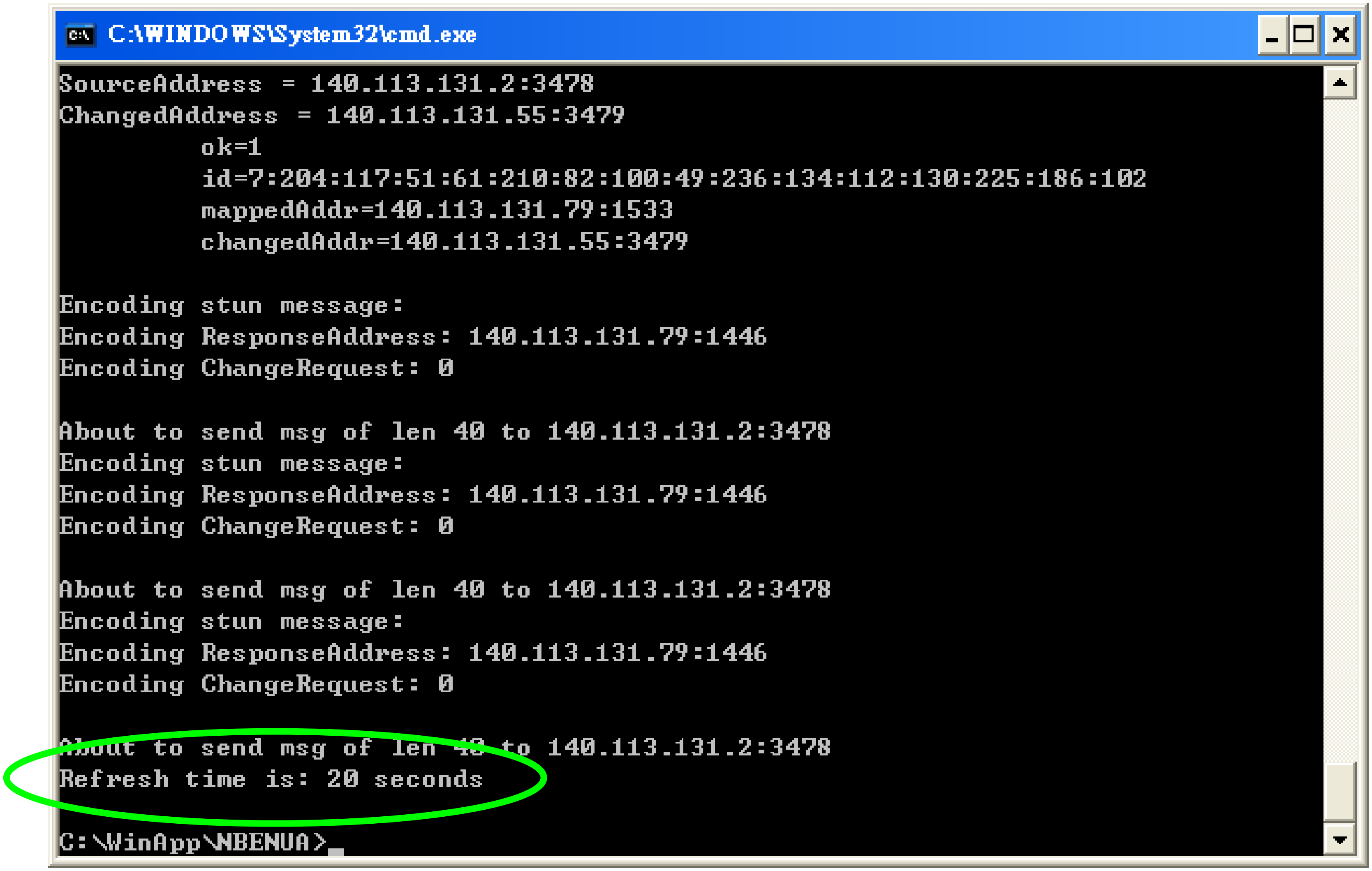
```
C:\WinApp\NBENUA>ipconfig

Windows IP Configuration


Ethernet adapter 區域連線:

        Connection-specific DNS Suffix  . :
        IP Address. . . . . . . . . . . . : 192.168.2.100
        Subnet Mask . . . . . . . . . . . : 255.255.255.0
        Default Gateway . . . . . . . . . : 192.168.2.1

Tunnel adapter Automatic Tunneling Pseudo-Interface:

        Connection-specific DNS Suffix  . :
        IP Address. . . . . . . . . . . . : fe80::5efe:192.168.2.100%2
        Default Gateway . . . . . . . . . :

C:\WinApp\NBENUA>
```

stun-client STUN.ipv6.club.tw



```
Encoding stun message:
Encoding ChangeRequest: 2

Encoding stun message:
Encoding ChangeRequest: 6

Encoding stun message:
Encoding ChangeRequest: 2

Encoding stun message:
Encoding ChangeRequest: 6

Encoding stun message:
Encoding ChangeRequest: 2

Encoding stun message:
Encoding ChangeRequest: 6

Encoding stun message:
Encoding ChangeRequest: 2

Cannot assign requested address
Internet connection is type: Port Restricted Nat

C:\WinApp\NBENUA>
```

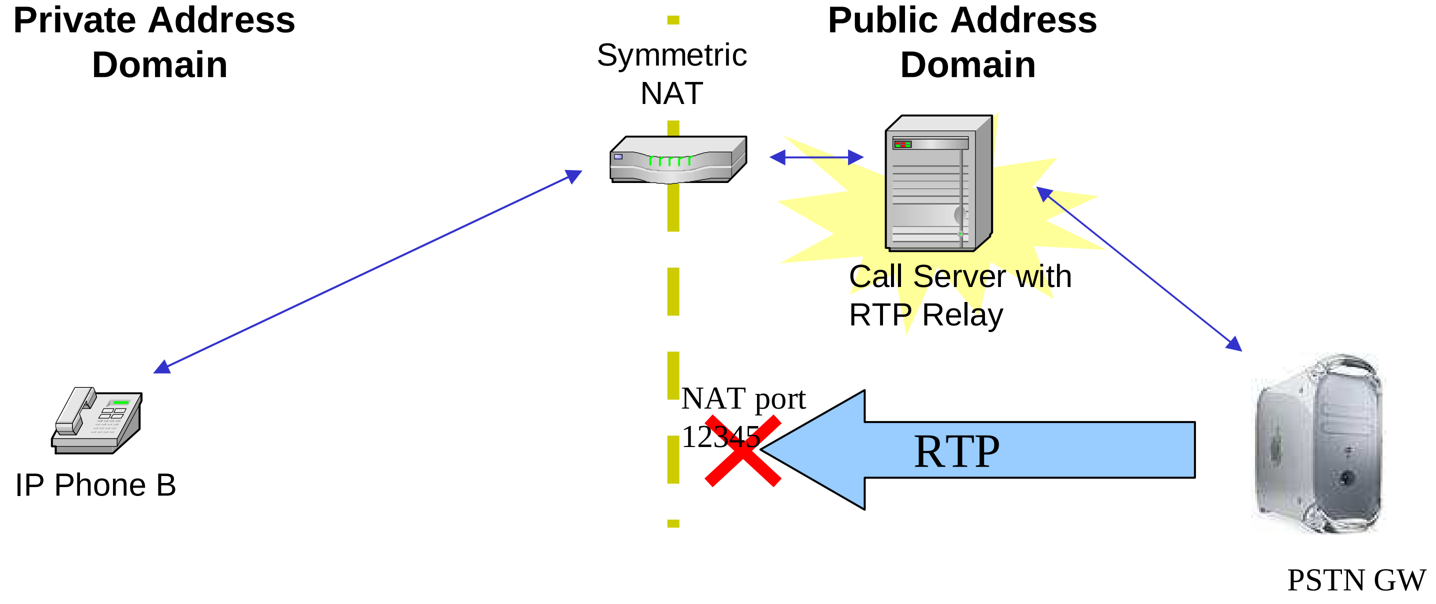
stun-client –t STUN.ipv6.club.tw



```
SourceAddress = 140.113.131.2:3478
ChangedAddress = 140.113.131.55:3479
         ok=1
         id=7:204:117:51:61:210:82:100:49:236:134:112:130:225:186:102
         mappedAddr=140.113.131.79:1533
         changedAddr=140.113.131.55:3479

Encoding stun message:
Encoding ResponseAddress: 140.113.131.79:1446
Encoding ChangeRequest: 0

About to send msg of len 40 to 140.113.131.2:3478
Encoding stun message:
Encoding ResponseAddress: 140.113.131.79:1446
Encoding ChangeRequest: 0

About to send msg of len 40 to 140.113.131.2:3478
Encoding stun message:
Encoding ResponseAddress: 140.113.131.79:1446
Encoding ChangeRequest: 0

About to send msg of len 40 to 140.113.131.2:3478
Refresh time is: 20 seconds

C:\WinApp\NBENUA>_
```

Clients Behind Symmetric NAT

- Provide a Call Server with RTP relay for non-upgradeable IP phone or Softphone
  - The loading for this server would be terribly heavy



Private Address Domain

Symmetric NAT

Public Address Domain

Call Server with RTP Relay

IP Phone B

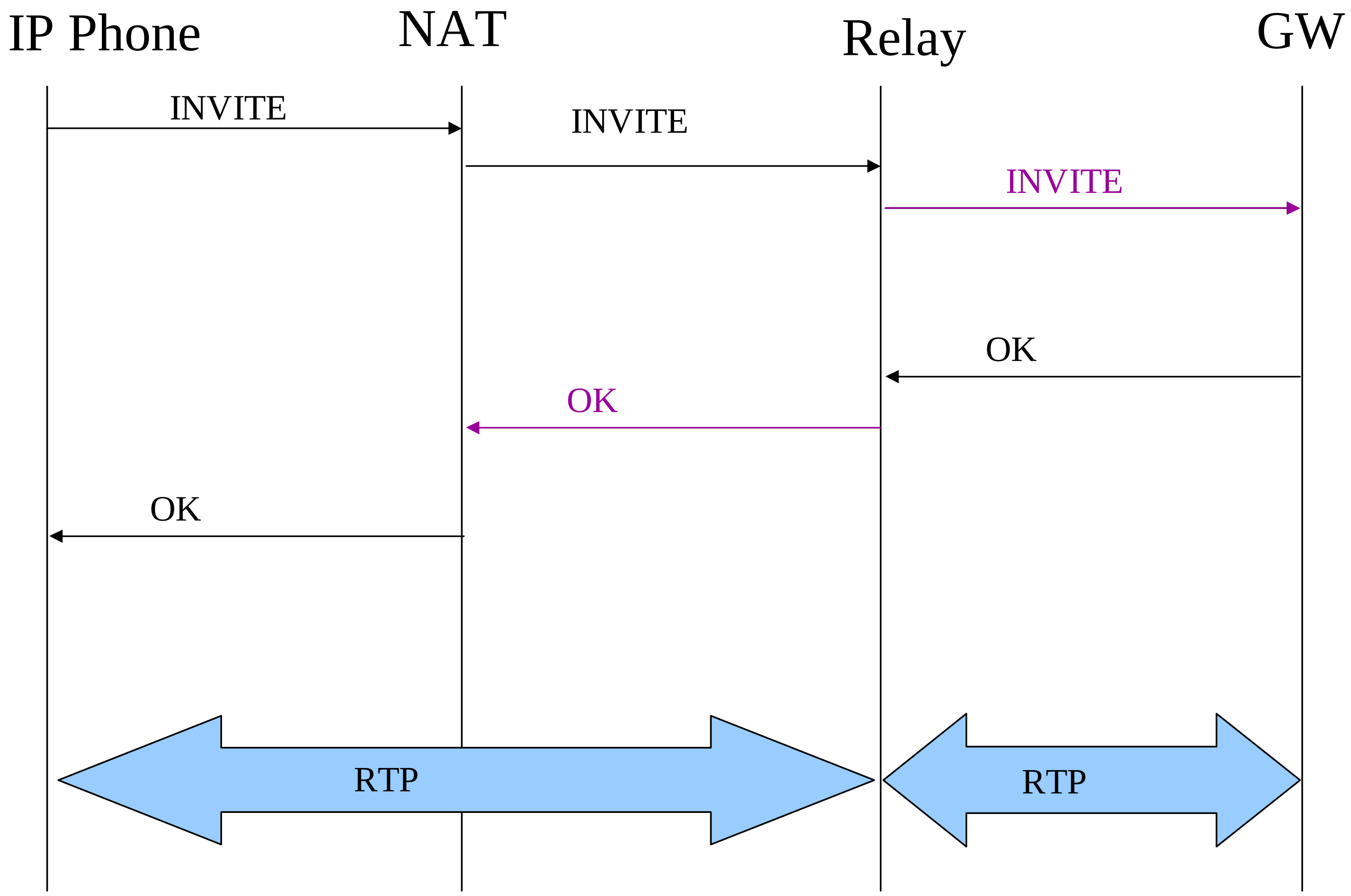
NAT port 12345

RTP

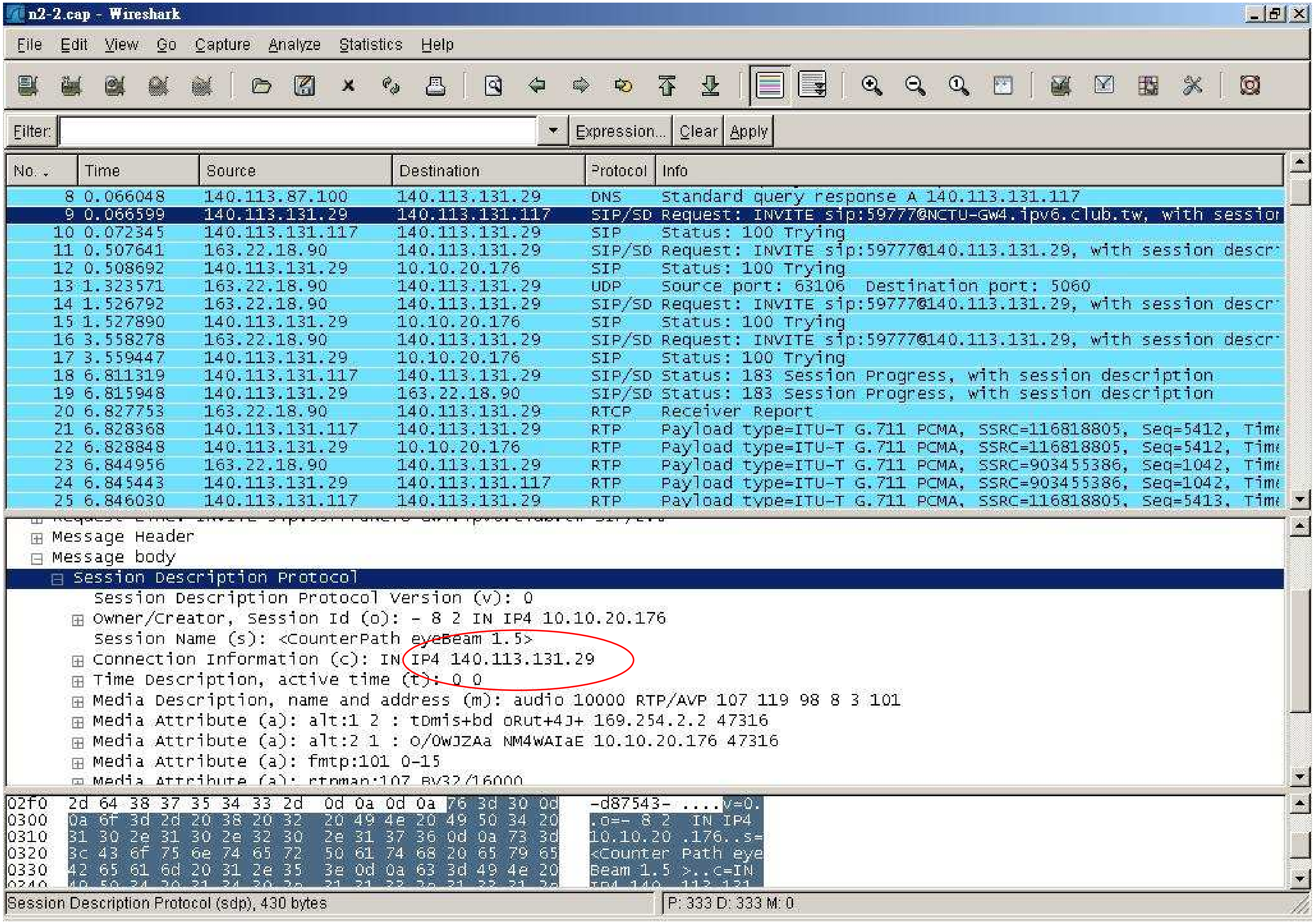
PSTN GW

Mapping Table
192.168.10.1:5060 <-> 10120 (for Call Server : 5060)
192.168.10.1:9000 <-> 12345 (for Call Server : 9000)

Signaling Flow



Relay -> PSTN GW


n2-2.cap - Wireshark
File Edit View Go Capture Analyze Statistics Help
Filter:
Expression... Clear Apply

```
No. .  Time      Source            Destination       Protocol Info
     8 0.066048  140.113.87.100    140.113.131.29    DNS      Standard query response A 140.113.131.117
     9 0.066599  140.113.131.29    140.113.131.117   SIP/SD   Request: INVITE sip:59777@NCTU-GW4.ipv6.club.tw, with sessior
    10 0.072345  140.113.131.117   140.113.131.29    SIP      Status: 100 Trying
    11 0.507641  163.22.18.90      140.113.131.29    SIP/SD   Request: INVITE sip:59777@140.113.131.29, with session descr
    12 0.508692  140.113.131.29    10.10.20.176      SIP      Status: 100 Trying
    13 1.323571  163.22.18.90      140.113.131.29    UDP      Source port: 63106  Destination port: 5060
    14 1.526792  163.22.18.90      140.113.131.29    SIP/SD   Request: INVITE sip:59777@140.113.131.29, with session descr
    15 1.527890  140.113.131.29    10.10.20.176      SIP      Status: 100 Trying
    16 3.558278  163.22.18.90      140.113.131.29    SIP/SD   Request: INVITE sip:59777@140.113.131.29, with session descr
    17 3.559447  140.113.131.29    10.10.20.176      SIP      Status: 100 Trying
    18 6.811319  140.113.131.117   140.113.131.29    SIP/SD   Status: 183 Session Progress, with session description
    19 6.815948  140.113.131.29    163.22.18.90      SIP/SD   Status: 183 Session Progress, with session description
    20 6.827753  163.22.18.90      140.113.131.29    RTCP     Receiver Report
    21 6.828368  140.113.131.117   140.113.131.29    RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5412, Time
    22 6.828848  140.113.131.29    10.10.20.176      RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5412, Time
    23 6.844956  163.22.18.90      140.113.131.29    RTP      Payload type=ITU-T G.711 PCMA, SSRC=903455386, Seq=1042, Time
    24 6.845443  140.113.131.29    140.113.131.117   RTP      Payload type=ITU-T G.711 PCMA, SSRC=903455386, Seq=1042, Time
    25 6.846030  140.113.131.117   140.113.131.29    RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5413, Time
```

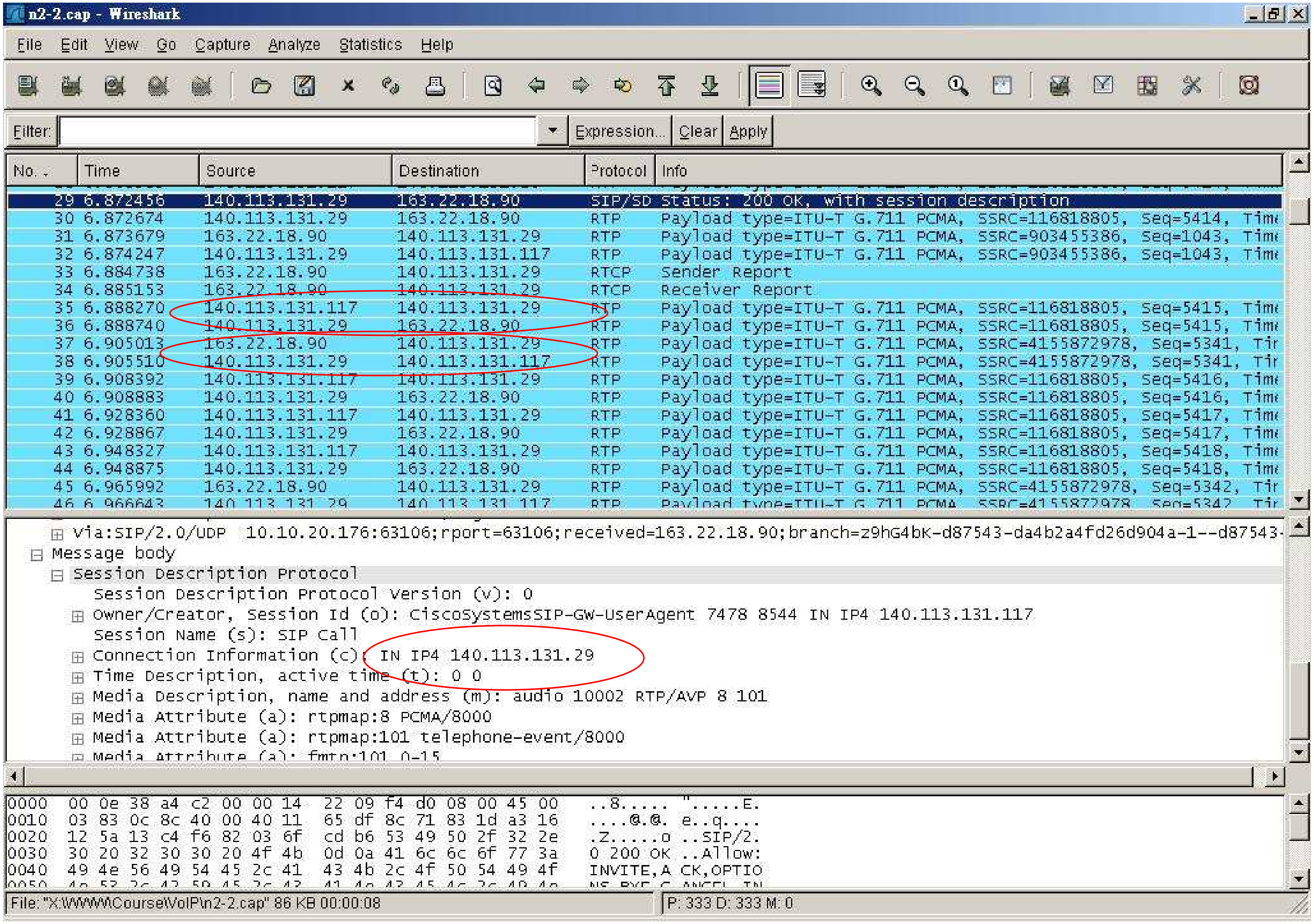
```
⊞ Message Header
⊟ Message body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 8 2 IN IP4 10.10.20.176
      Session Name (s): <CounterPath eyeBeam 1.5>
    ⊞ Connection Information (c): IN IP4 140.113.131.29
    ⊞ Time Description, active time (t): 0 0
    ⊞ Media Description, name and address (m): audio 10000 RTP/AVP 107 119 98 8 3 101
    ⊞ Media Attribute (a): alt:1 2 : tDmis+bd oRut+4J+ 169.254.2.2 47316
    ⊞ Media Attribute (a): alt:2 1 : o/0wJZAa NM4wAIaE 10.10.20.176 47316
    ⊞ Media Attribute (a): fmtp:101 0-15
    ⊞ Media Attribute (a): rtpmap:107 BV32/16000
```

```
02f0  2d 64 38 37 35 34 33 2d  0d 0a 0d 0a 76 3d 30 0d   -d87543- ....v=0.
0300  0a 6f 3d 2d 20 38 20 32  20 49 4e 20 49 50 34 20   .o=- 8 2  IN IP4
0310  31 30 2e 31 30 2e 32 30  2e 31 37 36 0d 0a 73 3d   10.10.20 .176..s=
0320  3c 43 6f 75 6e 74 65 72  50 61 74 68 20 65 79 65   <Counter Path eye
0330  42 65 61 6d 20 31 2e 35  3e 0d 0a 63 3d 49 4e 20   Beam 1.5 >..c=IN
```

Session Description Protocol (sdp), 430 bytes

P: 333 D: 333 M: 0

IP Phone B <- Relay


n2-2.cap - Wireshark
File Edit View Go Capture Analyze Statistics Help
Filter:
Expression... Clear Apply

```
No. .  Time        Source           Destination      Protocol Info
   29 6.872456    140.113.131.29   163.22.18.90     SIP/SD   Status: 200 OK, with session description
   30 6.872674    140.113.131.29   163.22.18.90     RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5414, Time
   31 6.873679    163.22.18.90     140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=903455386, Seq=1043, Time
   32 6.874247    140.113.131.29   140.113.131.117  RTP      Payload type=ITU-T G.711 PCMA, SSRC=903455386, Seq=1043, Time
   33 6.884738    163.22.18.90     140.113.131.29   RTCP     Sender Report
   34 6.885153    163.22.18.90     140.113.131.29   RTCP     Receiver Report
   35 6.888270    140.113.131.117  140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5415, Time
   36 6.888740    140.113.131.29   163.22.18.90     RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5415, Time
   37 6.905013    163.22.18.90     140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=4155872978, Seq=5341, Tir
   38 6.905510    140.113.131.29   140.113.131.117  RTP      Payload type=ITU-T G.711 PCMA, SSRC=4155872978, Seq=5341, Tir
   39 6.908392    140.113.131.117  140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5416, Time
   40 6.908883    140.113.131.29   163.22.18.90     RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5416, Time
   41 6.928360    140.113.131.117  140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5417, Time
   42 6.928867    140.113.131.29   163.22.18.90     RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5417, Time
   43 6.948327    140.113.131.117  140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5418, Time
   44 6.948875    140.113.131.29   163.22.18.90     RTP      Payload type=ITU-T G.711 PCMA, SSRC=116818805, Seq=5418, Time
   45 6.965992    163.22.18.90     140.113.131.29   RTP      Payload type=ITU-T G.711 PCMA, SSRC=4155872978, Seq=5342, Tir
   46 6.966643    140.113.131.29   140.113.131.117  RTP      Payload type=ITU-T G.711 PCMA, SSRC=4155872978, Seq=5342, Tir
```

```
    ⊞ Via:SIP/2.0/UDP  10.10.20.176:63106;rport=63106;received=163.22.18.90;branch=z9hG4bK-d87543-da4b2a4fd26d904a-1--d87543-
  ⊟ Message body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): CiscoSystemsSIP-GW-UserAgent 7478 8544 IN IP4 140.113.131.117
        Session Name (s): SIP Call
      ⊞ Connection Information (c): IN IP4 140.113.131.29
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 10002 RTP/AVP 8 101
      ⊞ Media Attribute (a): rtpmap:8 PCMA/8000
      ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
      ⊞ Media Attribute (a): fmtp:101 0-15
```

```
0000  00 0e 38 a4 c2 00 00 14  22 09 f4 d0 08 00 45 00   ..8.....  ".....E.
0010  03 83 0c 8c 40 00 40 11  65 df 8c 71 83 1d a3 16   ....@.@.  e..q....
0020  12 5a 13 c4 f6 82 03 6f  cd b6 53 49 50 2f 32 2e   .Z.....o  ..SIP/2.
0030  30 20 32 30 30 20 4f 4b  0d 0a 41 6c 6c 6f 77 3a   0 200 OK  ..Allow:
0040  49 4e 56 49 54 45 2c 41  43 4b 2c 4f 50 54 49 4f   INVITE,A  CK,OPTIO
0050  4e 53 2c 42 59 45 2c 43  41 4e 43 45 4c 2c 49 4e   NS,BYE,C  ANCEL,IN
```

File: "X:\WWW\Course\VoIP\n2-2.cap" 86 KB 00:00:08 | P: 333 D: 333 M: 0

Summary

- STUN is a good solution for non-symmetric NAT
  - Suitable for small-scale solution
    - Client-side
    - Enterprise-server
  - Compatible with most NATs
  - STUN server is easy to implement and low-cost
- Call Server w/ RTP Relay may be needed, if the users cannot make sure whether they are behind a symmetric NAT
  - Capacity is limited
  - Centralized server is expensive
    - That’s why Skype distributed the loading to individual users
- UPnP is a promising solution, but its nature is competing with IPv6.
  - Peer-to-Peer vs. Gateway/Device model