



VoIP + NAT



References

- “SIP, NAT and Firewalls”, Fredrik Thernelius
- Baruch Sterman and David Schwartz, “NAT Traversal in SIP”, Deltathree
- “STUN – Simple Traversal of UDP Through Network Address Translators”, RFC 3489, IETF
- “An Extension to the SIP for Symmetric Response Routing”, RFC 3581, IETF



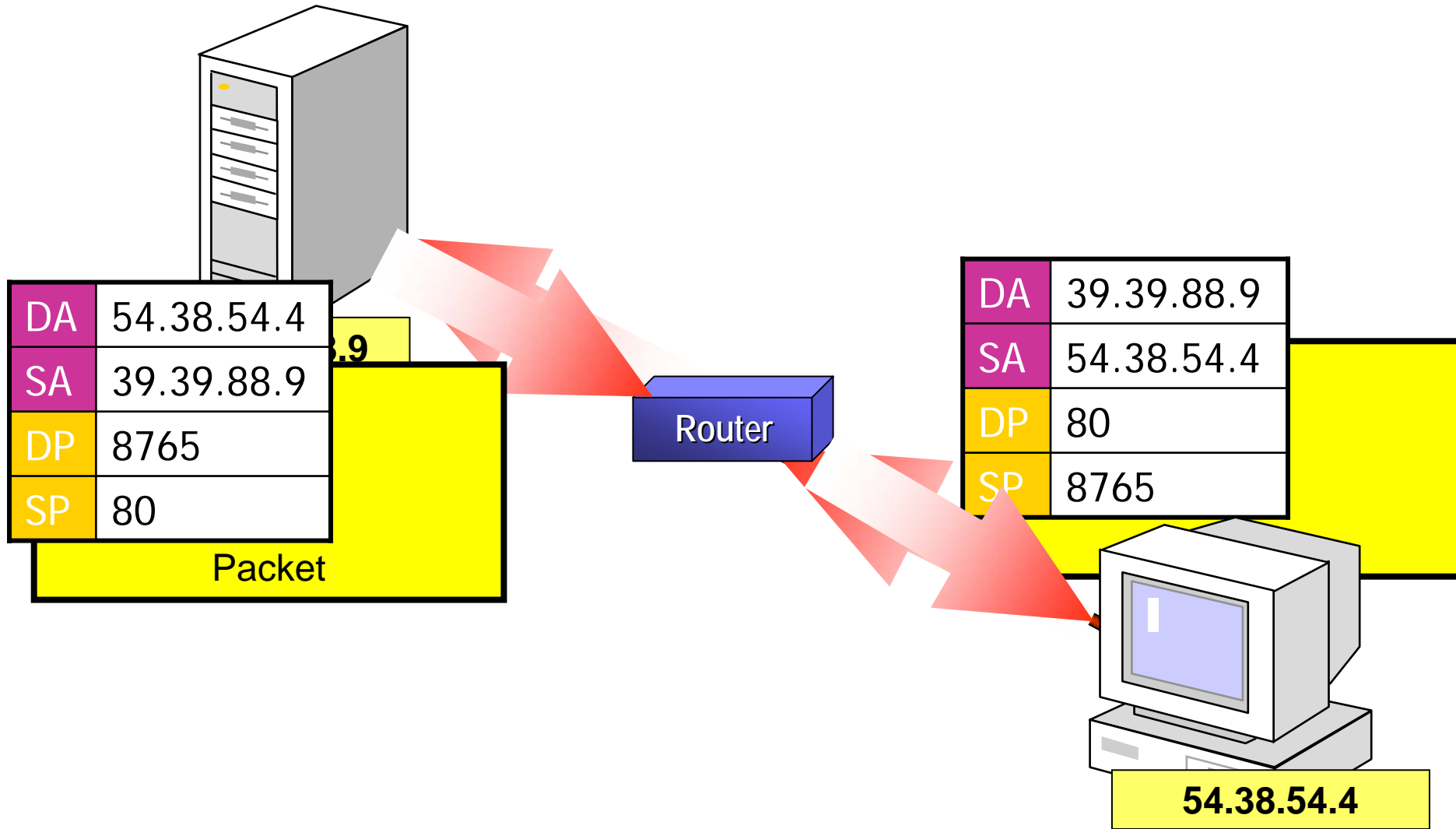
Outline

- Introduction
- The Problem of VoIP + NAT
- Possible Solutions for VoIP + NAT

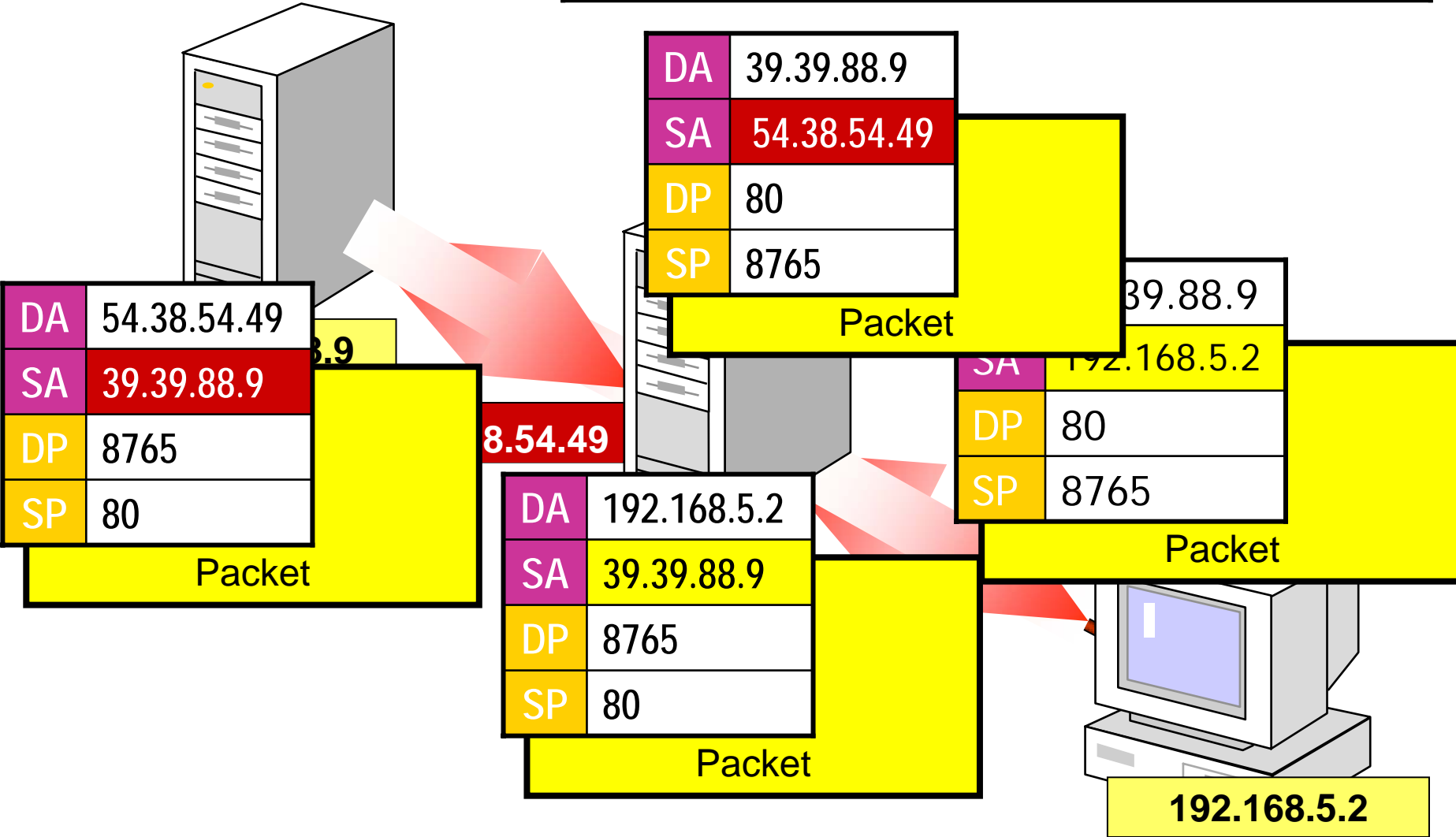


What is NAT?

- 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 to Application



DA	DP	SA	SP
39.39.88.9	80	192.168.5.2	8765





Flavors of NAT [1/3]

Static NAT

- Requiring same number of globally IP addresses as that of hosts in private environment
- Mapping between internal IP addresses and external addresses is set manually
 - This mapping intends to stay for a long period of time



Flavors of NAT [2/3]

Dynamic NAT

- Collect the public IP addresses into an IP address pool
- A host connecting to the outside network is allocated an external IP address from the address pool managed by NAT



Flavors of NAT [3/3]

NAPT (Network Address and Port Translation)

- A special case of Dynamic NAT
 - Use port numbers as the basics for the address translation
- The mechanism most commonly used

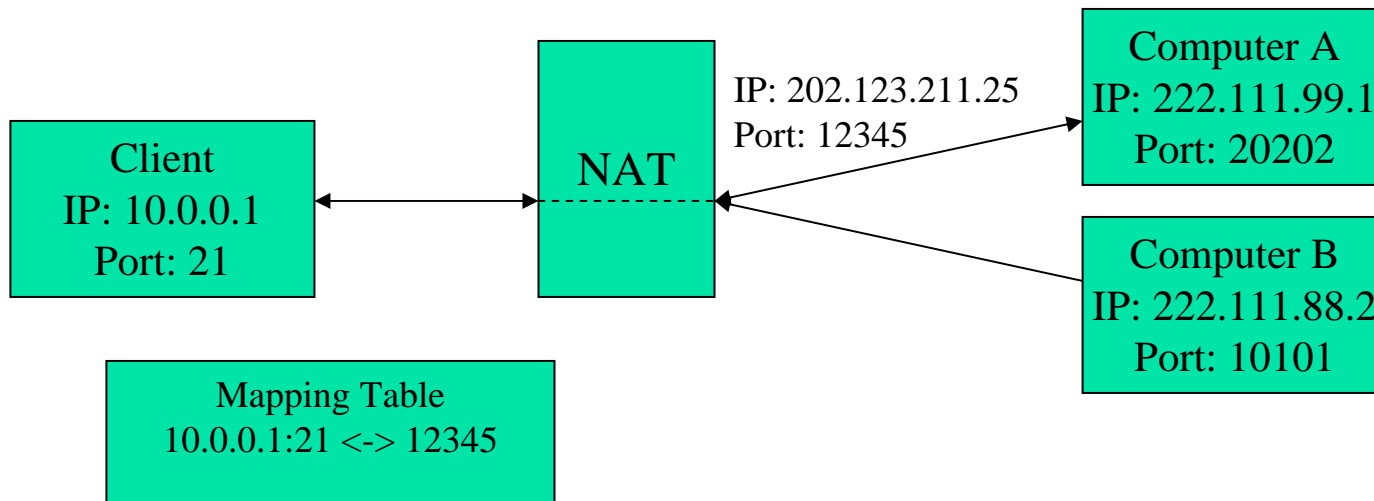


Types of NAT

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

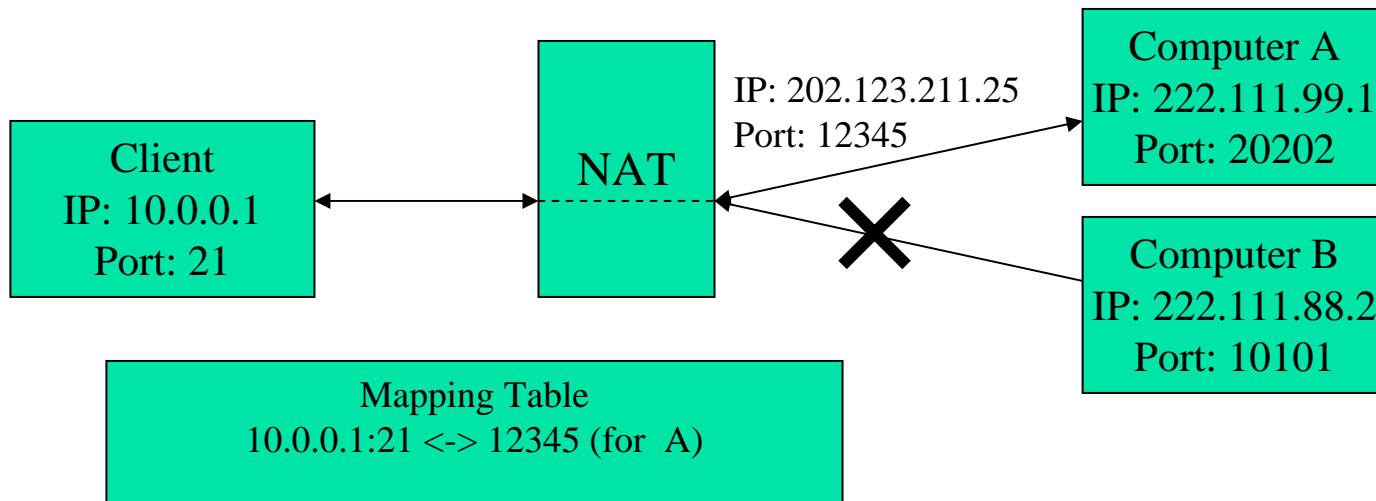
Full Cone NAT

- Client sends a packet to public address A.
- NAT allocates 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.



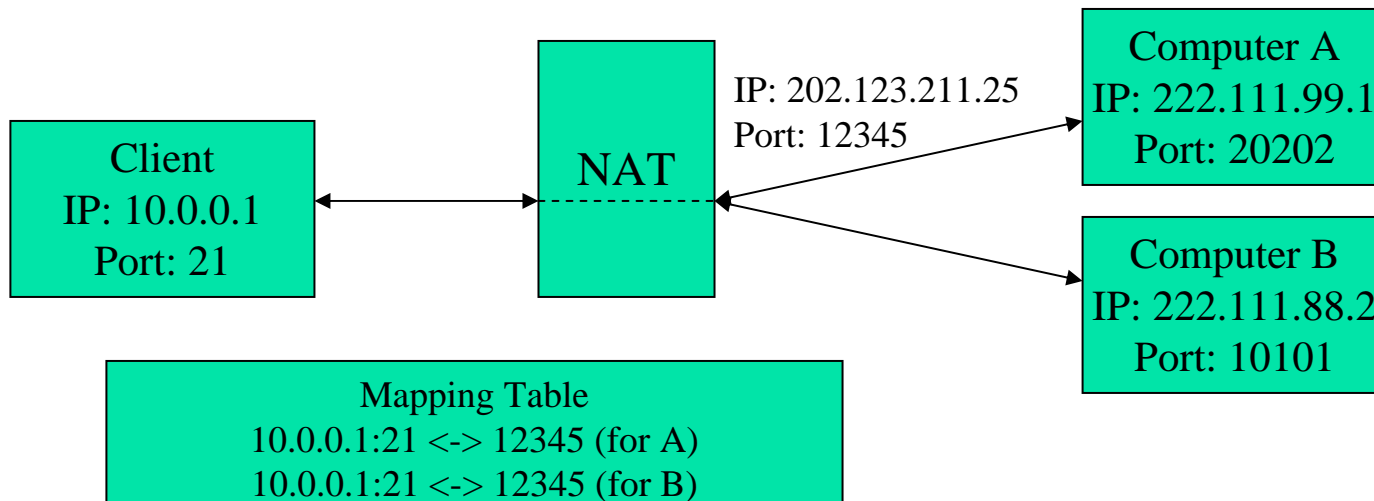
Restricted Cone NAT [1/2]

- Client sends 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.



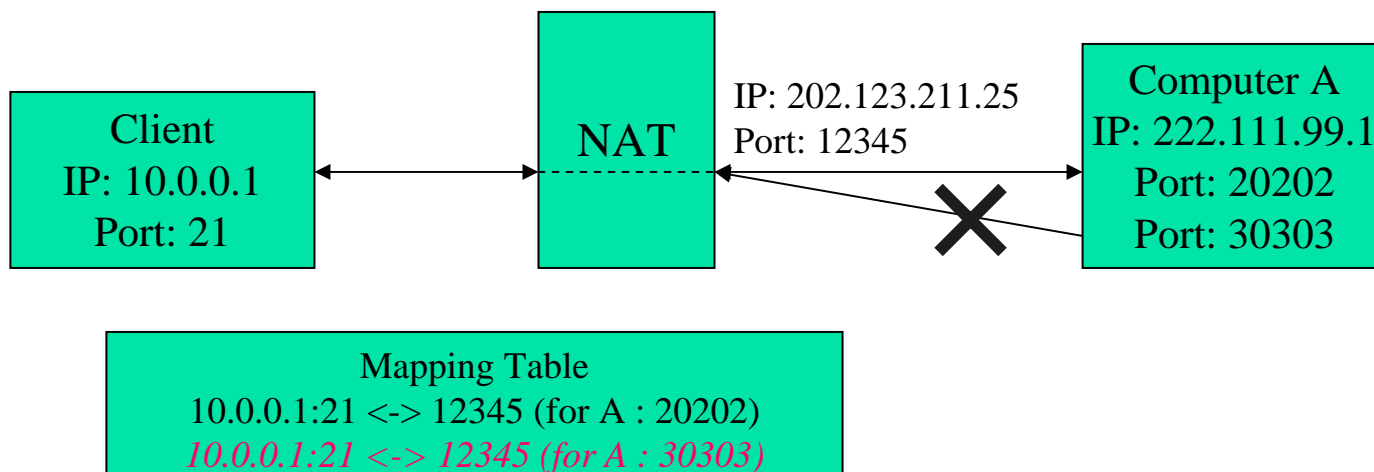
Restricted Cone NAT [2/2]

- Client sends 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.



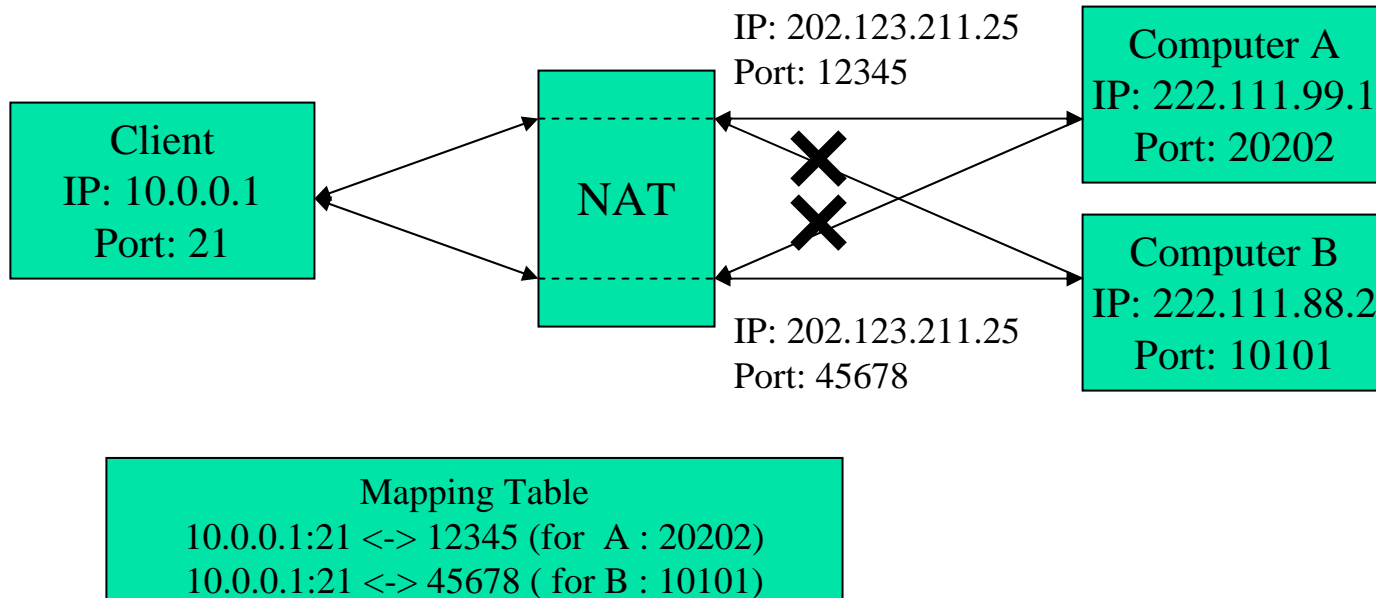
Port Restricted Cone NAT

- Client sends a packet to public address A at 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.



Symmetric NAT

- NAT allocates a public port each time the client sends 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





VoIP Protocol and NAT

- NAT converts 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



Solving NAT Traversal Problems

- Objectives

- Discover mapped public IP & port for private IP & port
- Use mapped public IP & port in application layer message
- Keep this mapping valid

- Issues

- NAT will automatically allocate a public port for a private address & port if needed.
- 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 (1 min~5 min)
- Keep a TCP connection to destination
- Send UDP packets to destination every specified interval

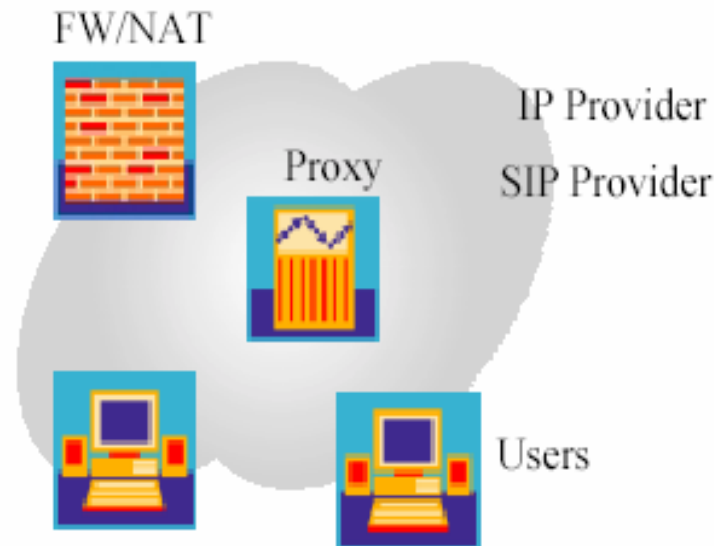


NAT Solutions

- IPv6 (Internet Protocol Version 6)
- UPnP (Universal Plug-and-Play)
 - UPnP Forum - <http://www.upnp.org/>
- Proprietary protocol by NAT/Firewall
 - SIP ALG (Application Level Gateway)
 - No standard now
- SIP extensions for NAT traversal
 - RFC 3581
 - 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-04
 - for symmetric NAT

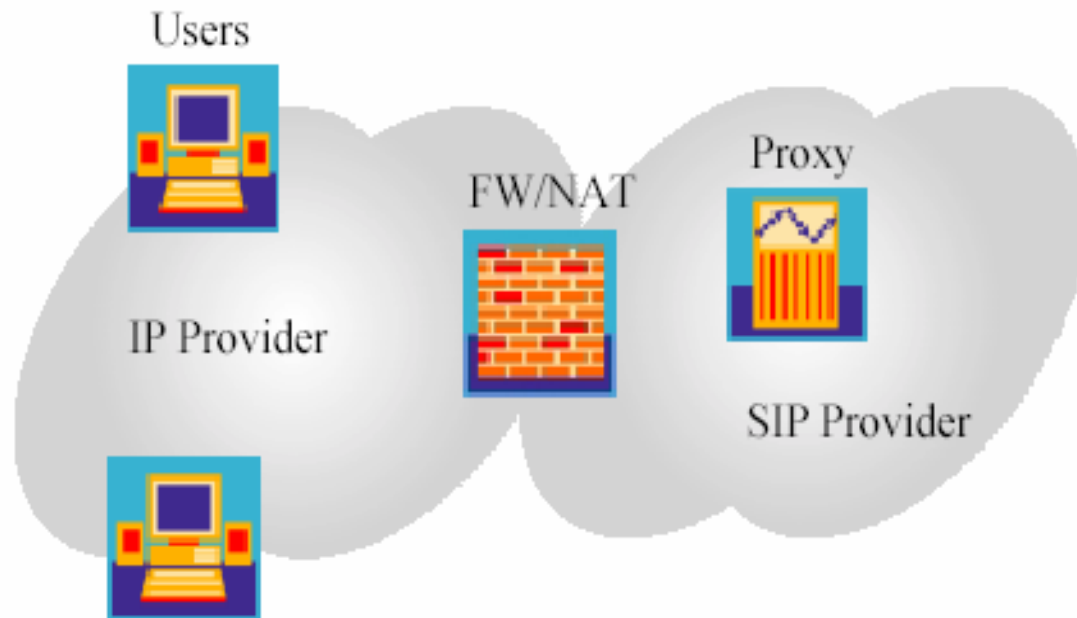
Two Distinct Cases – NAT Deployment [1/2]

Case I : SIP Provider is the IP Network Provider



Two Distinct Cases – NAT Deployment [2/2]

Case II : SIP Provider is **NOT** IP Network Provider



Solution for Case I – ALG [1/2]

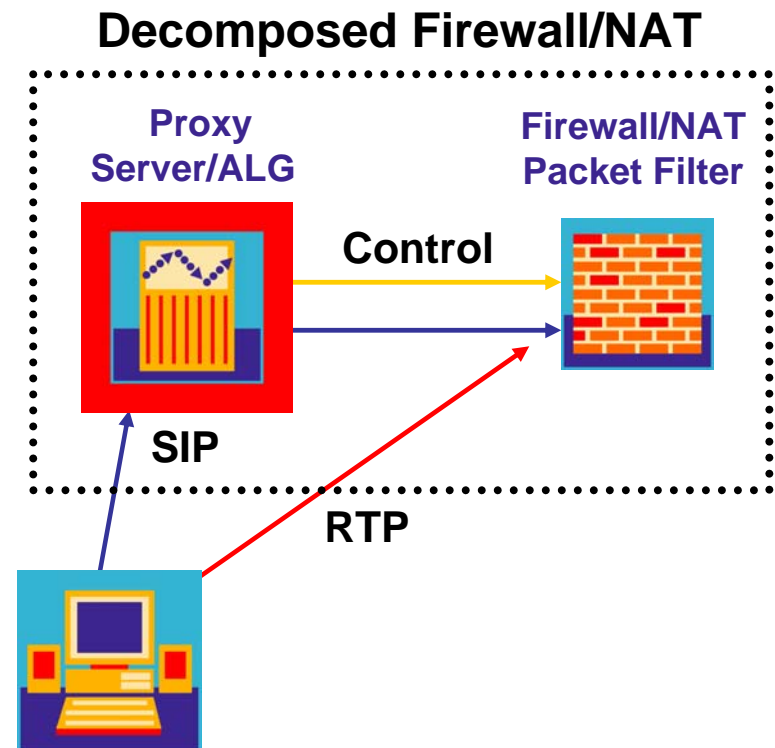
Separate Application Layer NAT from IP Layer NAT

- Like MEGACO Decomposition

- MG = Packet Filter
- MGC = Firewall Control Proxy

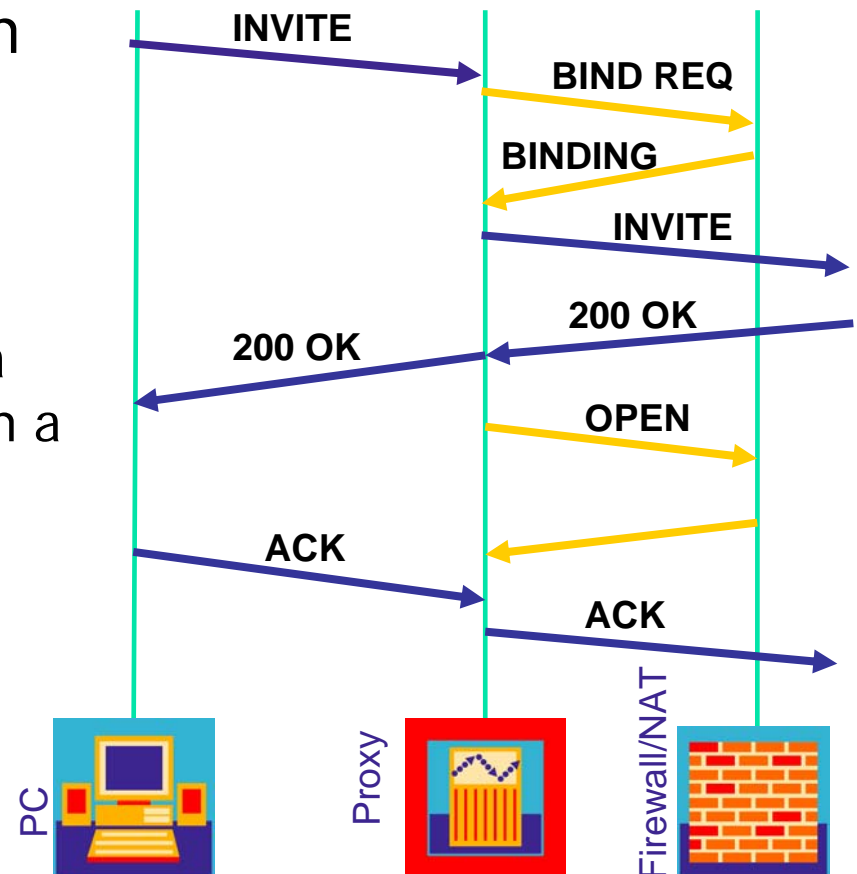
- Advantage

- Better scaling
- Load balancing
- Low cost
- Expertise problem solved



Solution for Case I – ALG [2/2]

- Control Protocol Between Application Layer NATs and IP Layer NATs
- Main Requirements
 - **Binding Request:** give a private address and obtain a public address
 - **Binding Release**
 - **Open Hole** (firewall)
 - **Close Hole** (firewall)





Proposed Solution for Case II

Much harder problem

- No way to control firewall or NAT
- Cascading NATs
- Variable firewall NAT behaviors

Proposed Solution

- Make SIP “NAT-Friendly”
 - Minor extensions
 - Address the issues for SIP only, not RTP
 - Accepted by IETF (RFC 3581)
- Develop a protocol for traversal of UDP through NAT
 - Work for RTP
 - Also support other applications



SIP Extension to NAT Friendly

Client Behavior

- Include an “**rport**” parameter in the Via header
 - This parameter **MUST** have no value
 - It serves as a flag
- The client **SHOULD** retransmit its INVITE every 20 seconds
 - Due to UDP NAT binding period and to keep the binding fresh



SIP Extension to NAT Friendly [2/2]

Server Behavior

- Examine the Via header field value of the request.
 - If it contains an “**rport**” parameter,
 - A “**received**” parameter
 - An “**rport**” parameter
- The response **MUST** be sent to the IP address listed in the “received” parameter, and the port in the “rport” parameter.



Example [1/2]

Client A: 10.1.1.1

Proxy B: 68.44.10.3

NAT C: 68.44.20.1

- A issues request
INVITE sip:user@domain SIP/2.0
Via: SIP/2.0/UDP 10.1.1.1:4540;**rport**;branch=z9hG4bKkjshdyff
- A→C (mapping port 9988)→B
INVITE sip:user@domain SIP/2.0
Via: SIP/2.0/UDP proxy.domain.com;branch=z9hG4bKkjsh77
Via: SIP/2.0/UDP 10.1.1.1:4540;
received=68.44.20.1;rport=9988;
branch=z9hG4bKkjshdyff



Example [2/2]

- 3) Server B receives the response
SIP/2.0 200 OK
Via: SIP/2.0/UDP proxy.domain.com;branch=z9hG4bKkjsh77
Via: SIP/2.0/UDP
10.1.1.1:4540;**received=68.44.20.1;rport=9988;**
branch=z9hG4bKkjshdyff
- 3) B (68.44.10.3:5060) → C (68.44.20.1:9988) → A
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.1.1.1:4540;**received=68.44.20.1;rport=9988;**
branch=z9hG4bKkjshdyff