NGN Default Lab

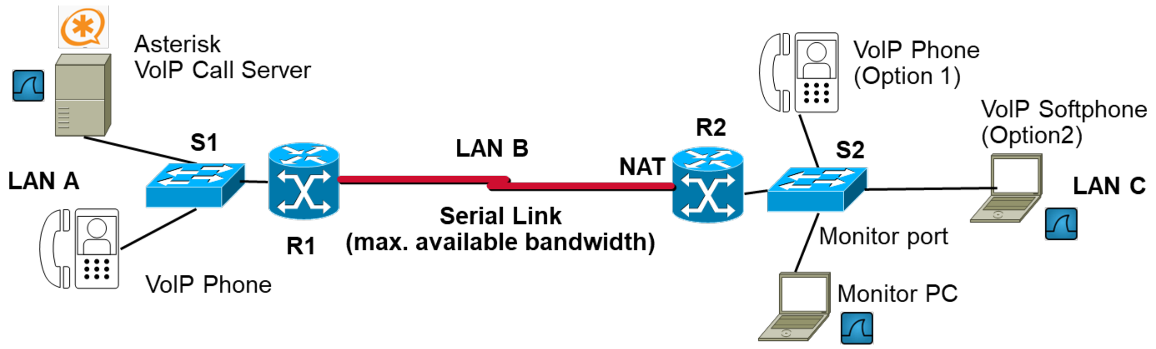
Examine VoIP calling with NAT

NGN Default Lab - Milestone 3

Task1: Change LAN C to become a Remote Site

Topology

The following simplified topology is used for Task 1:



Note: Some routers have Gigabit Ethernet interfaces **G0/0** and **G0/1 (see above)**, and others have Fast Ethernet **F0/0** and **F0/1** interfaces. When you use the **show ip interface brief (sh ip int br)** command, you see which type of interfaces are installed on your router.

Part 1: LAN C becomes Remote Site LAN with NAT

Step 1: Delete IP static route to LAN-C in Router R1

a. At router R1, erase the static IP route to LAN-C, in this example with **interface s0/1/0**

```
R1(config)# no ip route 10.<team no.>.2.0 255.255.255.0 s0/1/0
```

Step 2: Create NAT in router R2 for LAN-C

a. The static default router 0.0.0.0/0 shall still be valid for routing to other networks via router R1.

b. Limit NAT translation to LAN-C IP addresses with an Standard ACL

```
R2(config)# access-list 1 permit 10.<team-no.>.2.0 0.0.0.255
```

c. Create a NAPT translation including port translation to the interface, which is connected to router R1 (e.g. s0/1/0)

```
R2(config)# ip nat inside source list 1 interface s0/1/0 overload
```

d. Set router R2 GigabitEthernet interface to LAN C as NAT inside and R2 serial interface to R1 as NAT outside. Sample:

```
R2(config)# int s0/1/0
R2(config-if)# ip nat outside
R2(config-if)# int g0/0/0
R2(config-if)# ip nat inside
```

e. Test connectivity.
   - From router R2 ping router R1 serial interface. It should work. (y/n) Yes
   - From router R2 ping Asterisk server in LAN A. It should work. (y/n) Yes
   - From Router R1 ping VoIP phone in LAN-C. It should fail. (y/n) No

Task2: Inspect Peer-to-Peer VoIP traffic

Part 1: Enforce VoIP RTP traffic to be routed peer-to-peer through NAT

Step 1: Check Asterisk default NAT operation.

By default, an Asterisk server does no special NAT handling other than RFC3581. Check the RFC 3581 to understand this type of operation.

RFC 3581 Title	An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
Description of Asterisk behavior in case NAT	By default, Asterisk does minimal NAT handling based on RFC 3581. This means it uses the rport parameter in SIP headers to ensure responses are sent back to the correct IP and port, accommodating for NAT traversal.

Step 2: Reconfigure your Asterisk server for Peer-to-Peer operations

a. Enable peer-to-peer routing in **pjsip.conf** configuration:

```
directmedia=yes
direct_media_method= invite
disable_direct_media_on_nat= no
```

b. Restart the VoIP service
   1. Reload your configurations or restart your Asterisk server to apply the changes. sudo systemctl restart asterisk
   2. (Re-)start your Asterisk terminal in verbose mode. This will display state changes and will provide information about your VoIP service.

```
asterisk -rvvvvv
```

Task2: Evaluate Peer-to-Peer VoIP Traffic with NAT

Part 1: VoIP Calls from LAN-A to LAN-C

Step 1: Erase NAT translations

In router R2, no NAT translations shall be active.

a. Stop any communication from LAN-C to other networks

b. Erase NAT translations in router R2.

```
R2# clear ip nat translations *
```

c. Check all active NAT translations by displaying the translation table. They should be empty.

```
R2# show ip nat translation
```

Step 2: Call from VoIP phone in LAN-A to VoIP phone in LAN-C

a. Immediately after erasing NAT translations try to call your VoIP phone in LAN-C.
   - Start Wireshark captures to check signaling flows.
   - Erase NAT translations in router R2.
   - Initiate a VoIP call from LAN A phone to LAN-C phone and capture corresponding signaling and media traffic. It should be **not successful**.

b. For each protocol involved in VoIP call explain, why this call is not working

VoIP call LAN-A to LAN-C		
Protocol	**Success (yes / no)**	**Description**
SIP signaling traffic	no	SIP protocol is working between caller and server but not between called party and server. The Asterisk server forwarded the INVITE to the VoIP phone in LAN-C. SIP TRYING/RINGING/OK: There are no response messages (TRYING,RINGING, OK) sent back to the server and subsequently to the VoIP phone in LAN-A, since there are no NAT to traverse the network correctly with translations. The Asterisk server and the VoIP clients depend on these translations to correctly route SIP messages between the internal and external networks. SIP ACK: Without the OK message, the ACK message cannot be sent, and the call setup process is not completed.
RTP media traffic	no	RTP Stream: RTP media traffic relies on the SIP signaling to establish the media path. If the SIP signaling fails due to the lack of NAT translations, the RTP media stream cannot be established. Media Flow: Even if the initial SIP signaling succeeded by some means, the media flow would fail because the NAT translations are required to properly route the RTP packets between the networks. Even if SIP signaling somehow succeeds, the media stream (audio) cannot be established.This results in a call where the participants cannot hear each other, effectively making the call unsuccessful Here, caller sending the audio codecs in RTP traffic to server, but server is unable to reach the called party.

Part 2: VoIP Calls from LAN-C to LAN-A

Step 1: Create some SIP traffic from LAN-C

Check NAT translations after your LAN-C VoIP phone had any communication with the Asterisk server. This may be a SIP registration or a SIP VoIP call.

a. Check NAT translations in router R2, which are relevant for SIP VoIP calls from the router's NAT table.

b. Record the necessary translations in the following table:

SIP traffic from LAN-C to Asterisk server (LAN-A)			
Protocol	**Inside local address:port**	**Inside global address:port**	**Outside local address:port** **Outside global address:port**
UDP	10.3.2.3:5060	10.3.1.2:5062	10.3.0.2:5060 10.3.0.2:5060
TCP	10.3.2.3:54694	10.3.1.2:5064	23.201.240.195:80 23.201.240.195:80
TCP	10.3.2.3:34664	10.3.1.2:5068	2.18.64.13:80 2.18.64.13:80
TCP	10.3.2.3:46089	10.3.1.2:5063	108.177.15.188:5228 108.177.15.188:5228

Step 2: Evaluate a VoIP traffic from LAN-C to LAN-A

a. Create a call from VoIP phone in LAN-C to VoIP phone in LAN-A.

- Start Wireshark captures to check signaling flows.
- Initiate a VoIP call from LAN C phone to LAN-A phone and capture corresponding signaling and media traffic. It should be **successful**.
- You should be able to start a video call as well, if codecs match.

b. For each protocol involved in VoIP call explain, why this call is working

VoIP call LAN-C to LAN-A		
Protocol	**Success (yes / no)**	**Description**
SIP signaling traffic	yes	SIP is responsible for call setup, modification, and teardown. The static route in R1 is deleted which blocks routing from LAN A to LAN C, NAT config (setting NAT and NAPT interfaces) and ACLs (creating ACL to limit NAT translations) allow SIP messages to traverse correctly between LAN-C and LAN-A.
RTP media traffic	yes	RTP carries the actual media (voice/video) of the call. NAT translates RTP ports correctly, enabling media stream between the two phones.

c. After a successful call from LAN-C to LAN-A a successful call in reverse direction should be possible. Test this as well. Is it working? (y/n) yes

Evaluate Asterisk NAT debug information.

- Check debug messages at Asterisk terminal. What do you learn from these messages? explained in final report

Part 3: Evaluate SIP Call Setup Process in SIP Peer-to-Peer Mode

Step 1: Capture VoIP traffic

Initialize a peer-to-peer VoIP call and check the SIP signaling for call setup.

- Analyze Wireshark captures at Monitor PC and Asterisk Server to evaluate the SIP Call Setup Process for any VoIP call from caller (VoIP client A) to called party (VoIP client B)
- Record the following
  - SIP messages which are used for call setup through Asterisk Server.
  - Compare and discuss this peer-to-peer behavior with Asterisk Server in Proxy Mode.

Task3: Demonstrate the Results

Part 1: MS3 - Milestone 3

With Milestone MS3 you will demonstrate peer-to-peer traffic and your solution.

MS3		**Approved / Corrections**
MS3 Demonstration Dates	**June 3, 6, 10 or 13 (latest)**	
Request an MS3 meeting by Email.		