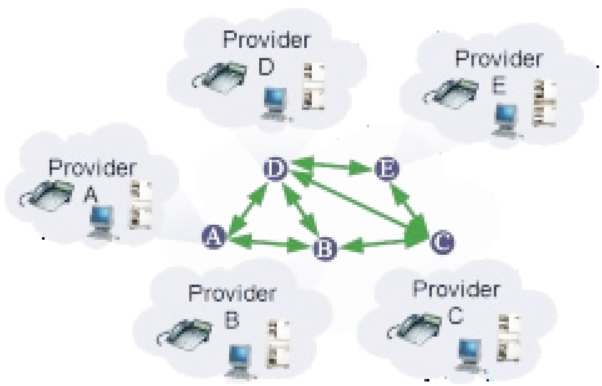
NGN Default Lab

VoIP Interconnection Analysis

Final Topology

Topology

In the end, you will run a VoIP service provider network and you will interconnect with another VoIP service provider. The aim of the project is to understand signaling traffic and media traffic for VoIP services, NAT routing solutions and VoIP interconnection.



Each team is one VoIP Service Provider. Each Team created their own VoIP Services with their own number space and domain.

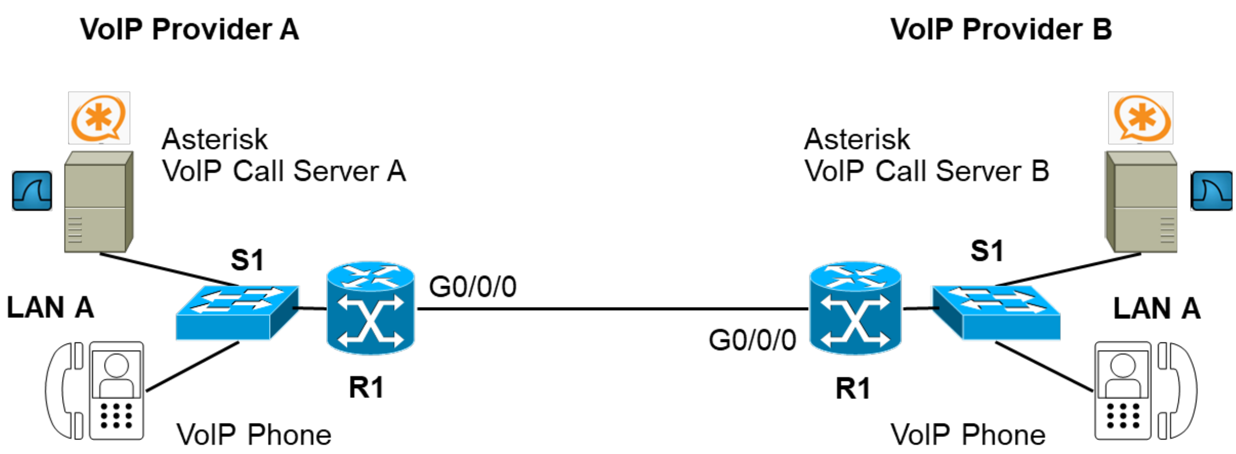
NGN Default Lab - Milestone 4

Task1: Create and analyze interconnected VoIP providers

In this Task, there will be interconnection of two or more VoIP Service Providers, and VoIP calls between VoIP Service Providers.

Topology

The following simplified topology is used:



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.

Step 1: Find a partnering team

a. Find a partner team within your class, to work on this jointly together.

b. Team MM (e.g. Team1) has LAN-A with network 10.1.0.0/24

c. Team NN (e.g. Team7) has LAN-A with network 10.7.0.0/24

Step 2: Interconnect your networks

a. Re-cable router R1 interface g0/0/0 in each team.

- Disconnect your g0/0/0 interface from the uplink switch
- Connect your router interface G0/0/0 to the router of your partner team.

b. Erase NAPT translation in router R1 in each team.

```
R1(config)# no ip nat inside source list 1 interface g0/0/0 overload
R1(config)# int g0/0/0
R1(config-if)# no ip nat outside
R1(config-if)# int g0/0/1
R1(config-if)# no ip nat inside
R1(config-if)# int s0/1/0                #<your serial interface>
R1(config-if)# no ip nat inside
```

c. New interconnecting LAN-X

- Design an additional LAN-X between the two routers in form of 10.<teamMM><teamNN>.0.0/24. E.g. for team 1 and team 7, LAN-X is network 10.17.0.0/24.
- The router interface R1 g0/0/0 of the team MM shall get the first available IP address and the router interface R1 g0/0/0 of the team NN shall get the second available IP address

d. Reconfigure router R1 interface g0/0/0 in each team.

```
R1(config)# int g0/0/0
R1(config-if)# description Connection to Team <your partner team>
R1(config-if)# no ip address 139.6.19.<your ip address> <your mask>
R1(config-if)# ip address 10.<LAN-X>.0.<1 or 2> 255.255.255.0
```

e. Test connectivity

With the default route of Task 1, you should have connectivity between LAN-A of team MM, LAN-X, and LAN-A of team NN.

- Test connectivity with a ping from Asterisk Server 1 (team MM) to Asterisk Server 2 (team NN). Does it work (y/n)? Yes
  (If not, resolve any false configuration or cabling.)

Part 2: Implement VoIP Interconnection by VoIP Trunk

Create VoIP interconnection, so that a VoIP client of team MM (e.g. local extension no. 1001) can dial and reach an extension of team NN (e.g. phone no. 7001).

Step 1: Use Asterisk Servers in Proxy Mode

a. Change Asterisk configuration to support Proxy Mode only. Configured

Step 2: Create a SIP Trunk between both Asterisk Servers

a. Read in the Asterisk instructions, e.g. in the Wiki, how to implement a SIP trunk. Implemented

b. Create a SIP trunk between the two Call Servers, which allows interconnected VoIP call routing

- The Call Server of team MM and the Call Server of team NN must be configured to route calls to external extension numbers.
- The two call servers must find each other.
- Test VoIP connectivity by placing a VoIP call from one team to the other team.
  Does it work (y/n)? Yes
  (If not, resolve any false configuration.)

Part 3: Evaluate SIP Interconnection

Step 1: Analyze SIP signaling with VoIP interconnection

a. Analyze the Wireshark captures at Asterisk Servers to evaluate the SIP Call Setup Process for any VoIP call from any caller (VoIP client A) to called party (VoIP client B)

   1. Record the SIP registration between Call Servers explained in final report
      - SIP messages for mutual registration
      - Record, which devices must register with remote VoIP Service Provider.
      - Record, which devices must authenticate with remote VoIP Service Provider.
   2. Record the SIP Call Setup for interconnected VoIP Services explained in final report
      - SIP messages of a VoIP call setup
      - Record, which devices must authenticate for a call setup
      - Measure call setup latency for interconnected VoIP Services

Step 2: Evaluate transcoding

a. Use one single, but different codec at VoIP client A and B

b. Allow all codecs in both Asterisk Servers

c. Evaluate RTP flows in case of transcoding

   3. Initiate phone calls from team MM, and from team NN
   4. Which device will be responsible to transcode samples?

d. Describe the transcoding process. What needs to be done to create a successful end-to-end call between phones with different codec capabilities?

Task2: Demonstrate the Results

Part 1: MS4 - Milestone 4

With Milestone MS4 you will demonstrate the analysis results.

MS4		**Approved / Corrections**
MS4 Demonstration Dates	**June 17, 20, 24 or 27 (latest)**	
Request an MS4 meeting by Email.		

Task 3: MS5 and MS 6 Tasks

Part 1: MS5 - Write your Technical Report

Create a Technical Report and summarize your tasks, your configurations, and your evaluation results.

1. The reader shall understand what you did and why you did it that way
2. Discuss your results

The report includes

1. Team members and their email addresses
2. Desciption of your approach and outcome
   - Create expressive graphs, MSCs and figures of your measurement results
3. References used by your team
4. Format: PDF, size DIN A4 and character size 11pt

MS5		**Approved / Corrections**
Upload your report in ILU	**July 3 (latest)**	

Part 2: MS6 - Cleanup your Workplace

When you finish your lab work at the end of your project, erase all startup-configuration on switch and router

1. Routers

   Router#erase startup-config and Switch-off the router
2. Switches

   Switch#erase startup-config
   Switch# delete vlan.dat and Switch-off the switches
3. Remove all cabling in your lab and return all cables to the cable holders.
4. Return VoIP phones and Console cables and adapters to your lab staff.
5. Cleanup PCs
   - Remove Asterisk configuration files
   - Remove Asterisk Server
   - Remove Wireshark capture files
   - Remove Softphone software
   - Shut down your PCs.

MS6		**Approved / Corrections**
Cleanup you lab workplace	**July 4 (latest)**	