Open Source MANO

OSM#11 Hackfest

Team Asterisk Unibo

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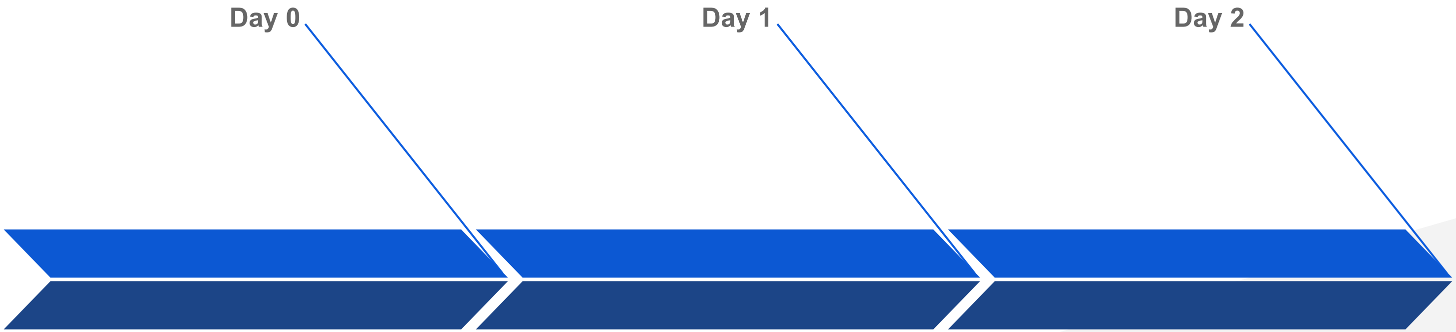
Asterisk

- Open Source software development project to implement universal tool for building communications applications
- Not only IP PBX systems but also VoIP gateways, call center systems, conference bridges, voicemail servers and all kinds of other applications that involve real-time communications.
- The address of a SIP device is generally referred to as its URI (Uniform Resource Identifier).
- To access the communications system it is needed a VoIP URI that sends calls to the server in the form “sip:number@sip_domain”..



Operations

- Description of VNF
- Description of NS
- Image Instantiation

Operations

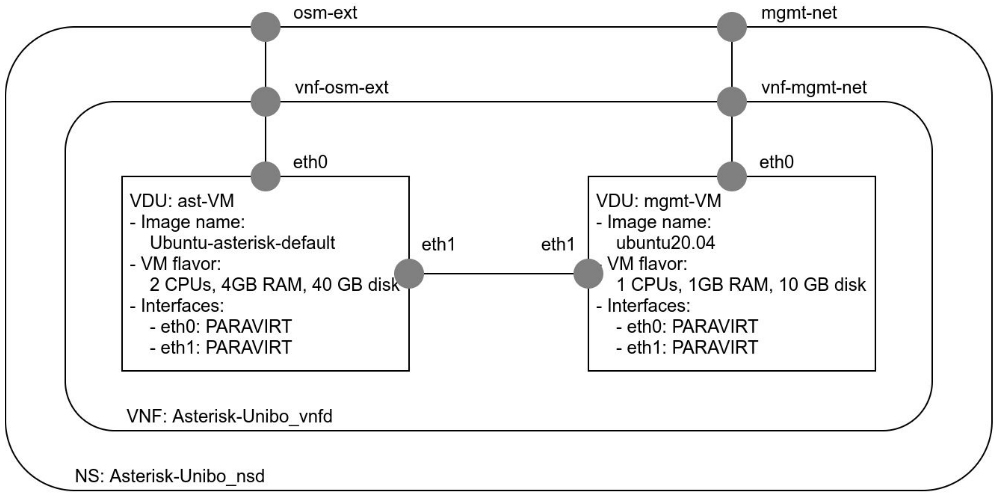
- Initial configuration
- Asterisk startup

Operations

- User Creation
- User Removal

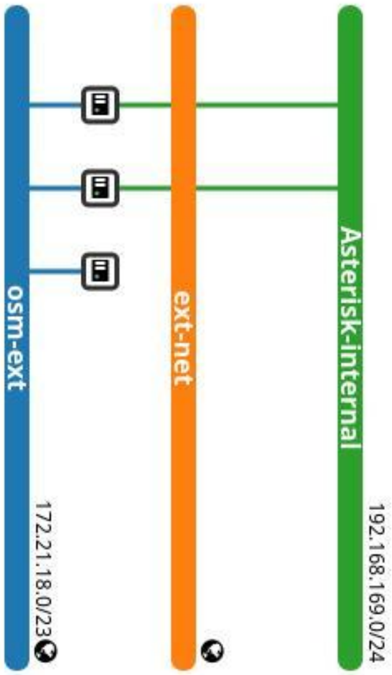
Day 0 - Modeling VNF and NS

- VNF Description
  - Topology description
    - VDUs and CPs
  - Execution environment list
    - Native Charms
- NS Description
  - Connection between VDUs and osm-ext



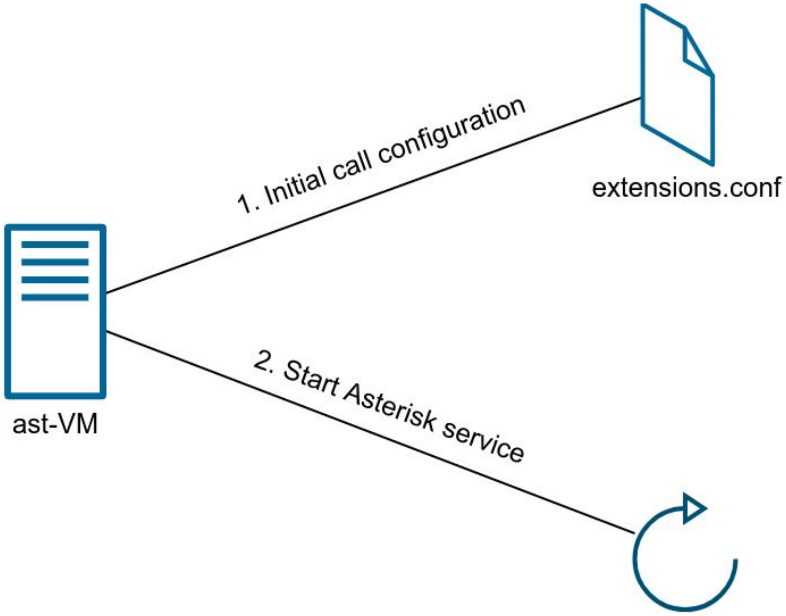
Day 0 - OpenStack Deployment

- Instantiation of the Image: Ubuntu 20.04
  - Installation and Configuration of Asterisk
  - Basic test on the instance
  - Snapshot of the instance
- VNF and NS instantiation



Day 1 operations – Asterisk startup

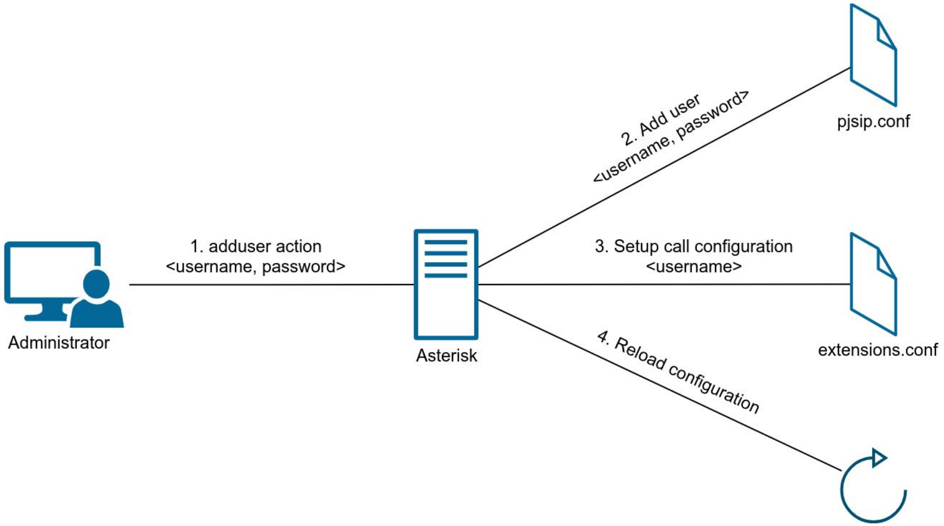
- Asterisk startup and initial configuration
- Juju-based execution environment with a Native Charm
- Adding the Day - 1 primitives
  - Action : startasterisk



Day 2 operations – Action: Add User

VNF Runtime Operations

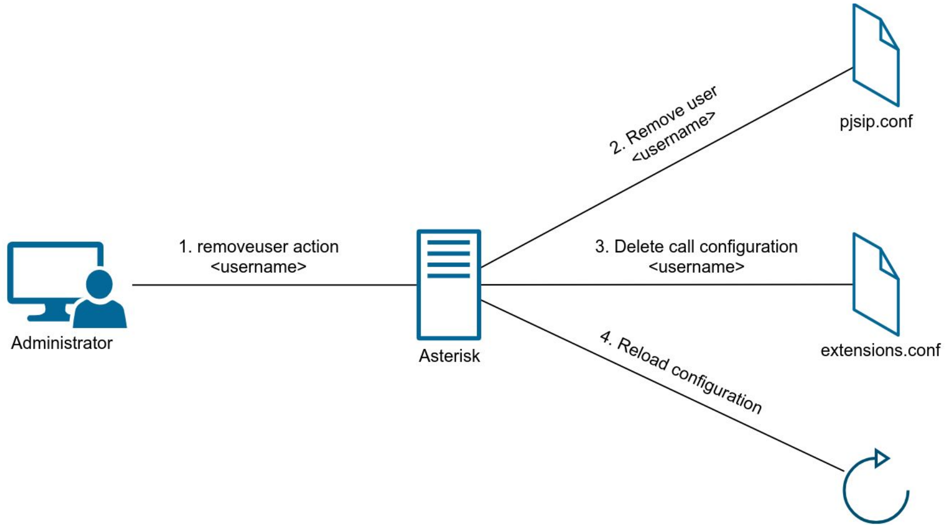
- Adding Day - 2 primitives
  - **Action - 2: adduser**
  - Action - 3: removeuser



Day 2 operations – Action: Remove User

VNF Runtime Operations

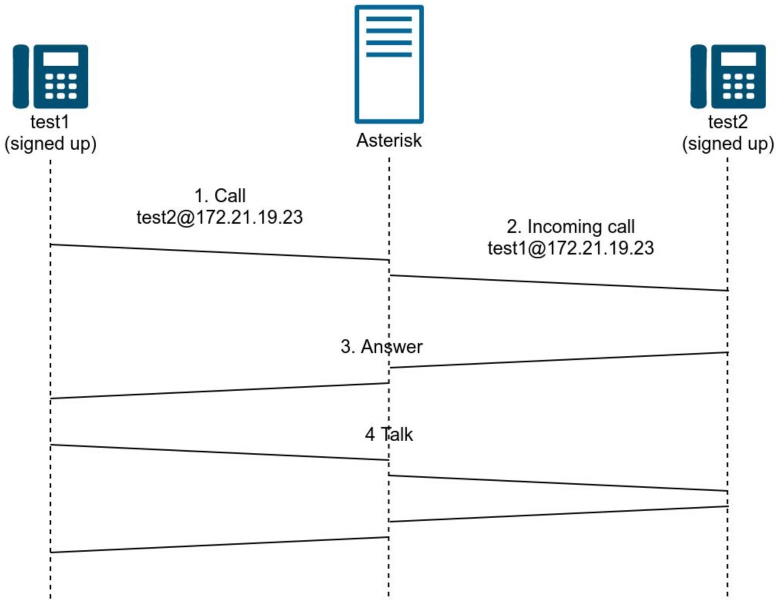
- Adding Day - 2 primitives
  - Action - 2: adduser
  - **Action - 3: removeuser**



Demo: Test Calls with Softphones

- test1 and test2 sign-up to the server.
    - test1@172.21.19.23
    - test2@172.21.19.23

- test1 starts the call request
- Server checks if the destination exists, then, the call is directed to the user



Further Development

- VoiceMail
- Load Balancing among multiple servers

Grazie per l’attenzione

Now follows a demo of the system