Assignment 3

Asterisk and VoIP

Administrative Info	
Mode of delivery	SLATE
Due	See SLATE

Purpose

- **Lab Part:**
    - Deploy VoIP "phones"
    - Build a basic VoIP exchange service off an Asterisk PBX node
    - Interconnect multiple nodes via PBX "trunking"
    - Design and implement a call plan to allow local and trunked calls between PBX sites
    - Analyze SIP traffic at the session layer as well as headers and payloads
    - Analyze at least two media traffic types
    - Explore SIP security with SIPP

Deliverables

- **Week 9**: Lab Part due
    - Lab part review – hands on
    - Lab part write up – succinct!:
        - An executive summary
        - A copy of your dial plan
        - Answers to parts 2-7 of the lab procedure (labeled and properly sequenced)

Lab Part Procedure

1. Build an Asterisk PBX on the Linux of your choice.
    a. Install 1 softphone (i.e. ZoiPer, 3CX, X-Lite, Microsip) and one HW IP phone (Cisco/Siemens)
    b. You may use an analog phone using an ATA device
    c. Configure the PBX to support the softphones/IP phones via SIP – put all phones on a private LAN to simulate a NATted system
    d. Manually (via command line!) create a dial plan (DP) to allow for calls between the soft and IP phones□including conferencing

   (i) Do not use a GUI to generate a dialplan!
   (ii) You must be able to explain the dialplan config script

e. Connect to as many other PBX's as possible via trunks. Extend your dial plan to allow inter-PBX calls between your system and other groups
f. Include a diagram of working PBX topology in your write up
g. Provide a commented copy of your dial plan with comments in your write up

2. Capture and analyze a SIP signaling session for a device to device call.
   a. First examine and discuss signaling from the perspective of the session level – i.e. SIP packet types from the first to last – from caller to server to collie. Provide a diagram.
   b. Next, on a packet by packet basis, analyze the key headers in the session dialog – do a bit of research to enrich your understanding – make sure you include the registration & authentication part.
   c. Now extend your packet analysis to the SDP payload. Do a breakdown and explanation of SDP contents and session.
   d. Once the VoIP steam starts to flow, capture and analyze an RTP header. Explain what it is doing. List 5 key characteristics of RTP.

3. Capture a conference call.
   a. Obtain the SIP signaling. Explain the basics of how the conference is signaled.
   b. Now capture the media stream for a conference call. Note the individual flows. Explain how conferencing is being achieved (i.e. how are all the streams merged?)

4. Capture and analyze trunking traffic.
   a. With capture on bring up a trunk. Illustrate the packets and with some level of detail explain what you see.
   b. Now capture host to host signaling on an inter-PBX call over the trunk. Compare this to the signaling between hosts on a non trunked call.
   c. What exactly is the function of the "trunk". What does it do? Why is it needed?
   d. Note the paths of the signaling and media flows in a "trunked" call. Are both servers kept in the media flow? How is the trunk involved?
   e. In the sip.conf file, explore what happens when you modify the "canreinvite" (or "directmedia") and "nat" options. Explain how this works. Prove that it works!

5. For G.711 and one other codec of your choice use Wireshark to measure the following. Tabulate your results.
   a. the packet rate (pps),
   b. packet size (bytes)
   c. IPG (interpacket gap in milliseconds)
   d. jitter (average in milliseconds)
   e. BW (bps)

6. Sign out a Linktropy device. Plot MOS score for two different codecs across the following independent variables: latency, packet loss, bit error, jitter. Draw some conclusions base on what you see.

7. Load SIPP onto one of your Linux servers. Review how SIPP works and get a good sense of the scripting language.

   a. Carefully design a staged stress test for your Asterisk server. Remember that signaling taxes the system most. Try not to vary more than one variable at a time. Structure your approach.
   b. Implement your stress test using SIPP. Record critical data and present it professionally in the report.
   c. Analyze results and draw conclusions on the success of your test.

8. Briefly explain each of the following
   a. Asterisk Modules
   b. Asterisk Extensions Language (AEL)
   c. Asterisk Gateway Interface (AGI)

Resources

There is much online support for Asterisk, SIPP and VoIP in general. The best all round source for concepts and practical advice on installation continues to be “Asterisk, The Definitive Guide” by Madsen.