SISTEMAS

E REDES MULTISERVIÇO – 2019/20

Practical Evaluation #3 – SIP, SDP, RTP and RTCP

The purpose of this work is to familiarize the students with the operation of SIP, SDP, RTP and RTCP protocols and their functions in a VoIP architecture. For the voice server we will use *3CX Phone System* in Cloud (http://www.3cx.com). It is a VoIP server, based on Asterisk, with a browser management console. For telephone terminals (extensions), we will use 3CX softphones (https://www.3cx.com/voip/softphone/) on PC's and smartphones.

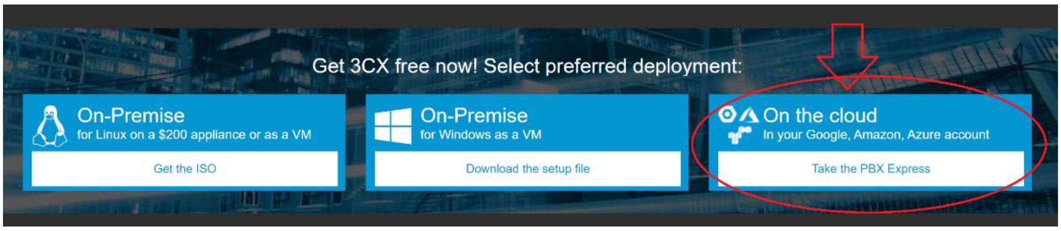
The capture of information exchanged is done by the software "Wireshark" (https://www.wireshark.org/download.html).

PREVIOUS NOTES:

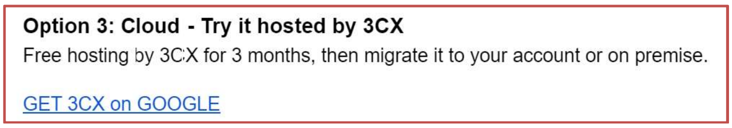
For the execution of this work you should not use the Wifi networks "*eduroam*", "Lab8_5G" or "Lab8" since some necessary ports of communication are blocked. You can, for example, use the "mstio" network.

INSTALLATION OF VOIP SERVER AND CLIENTS WITH SOFTPHONE

1. This work requires that, in each group, a VoIP server and two softphones that will function as clients; it is not necessary that they are on the same network; one of the clients must be on a PC where you must also install wireshark software (this software should always run in Administrator mode).
2. For the clients: the 3CX softphone software for PC can be obtained in https://www.3cx.com/voip/softphone/. For Android just search for "3CX" in the Play Store or App Store.
3. For the installation and activation of the voice server (only one per group):
   - I. Start the process of creating the voice server hosted in the 3CX Cloud. At the bottom of the page www.3cx.com select the option "On The*Cloud*":
   - II. Perform the registration process – use a true email – and wait for the reception of an email with a software activation key;



   - III. In this mail you will find the software activation key and instructions for various options available. Choose 3CX Cloud server hosting (via Google):


Option 3: Cloud - Try it hosted by 3CX
Free hosting by 3CX for 3 months, then migrate it to your account or on premise.

GET 3CX on GOOGLE

IV. During the installation process, you will need to enter some data:

1. Country, time zone and language
2. A name for the server that will serve to create the access URL
3. How many digits will have the internal numbers (2, 3, or more digits)
4. A Google Datacenter for hosting the server

V. At the end, you will receive an email with the general information of your system! Sign in to your Browser with the credentials created earlier.

CONFIGURATIONS AND TESTS

1. Open the VoIP server management page (indicated in the mail from the previous points) and use the credentials configured in the installation process;

2. A screen with menus on the left side will appear:

3. Creation and activation of clients (extensions):
   a) Access the "Ramais" menu and "Add" to create a new extension.
   b) Set the internal extension number (example: 21), a name and a existing mail address that will use the softphone. The other fields are not required.
   c) It is important to remove the prohibition on use outside the LAN. In the "Options" tab remove the selection of "Not authorize the use of the extension outside the LAN..." as the next image. In the end do OK.

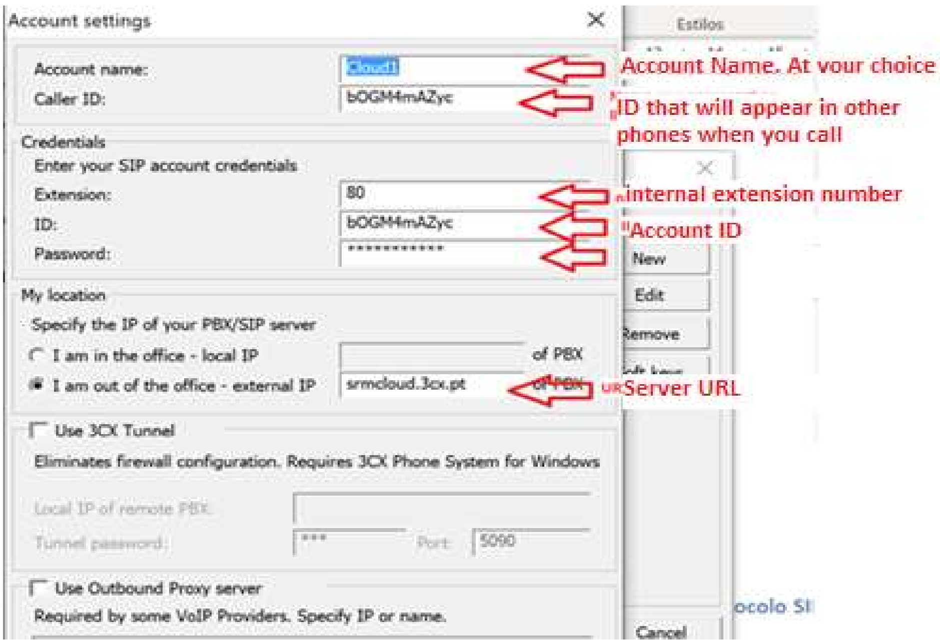
   d) Repeat the process to create a second user (with another extension number and email address);

e) Returning to the overview of this “Ramais” menu, all extensions already created appear. Select the two created users and click "Send Welcome Email".

f) Both users will receive an email with instructions and attachments that allow them to automatically configure the 3CX softphones. <u>Do not do it now!</u>

g) Start a packet capture on Wireshark from the PC where you installed the softphone. The goal is to capture packets exchanged between the softphone and the VoIP server at the time of SIP registration.

h) Activate the user through the automatic configuration file received in point f). If this operation fails, you can manually configure the softphone:



i) Verify that the customer completes the registration successfully.

4. Question for report:

a) Finish the capture on wireshark. Identify and explain SIP protocol information exchanges in the user registry on the VoIP server.

5. Reboot wireshark with a new packet capture on the user's server and PC and make a call between the two extensions. After you hang up the call, terminate packet captures in Wireshark and find those that pertain to the call (SIP, SDP, RTP, and RTCP).

(NOTE): Wireshark makes it easy to view the packets involved in a VoIP communication. Just go to the menu “Telephony” -> VoIP Calls -> Flow Sequence

6. Questions for report:

a. In SIP packets: interpret and explain the call establishment sequence and information exchange between users and the server;

b. In the same wireshark capture, locate a SIP / SDP packet. Identify and explain the main fields regarding the action of this protocol;

c. RTP packets: identify the main fields introduced by this protocol; What is the function of these packets during VoIP communication?

d. RTCP packets: interpret the information they carry; What is the role of this protocol during VoIP communication?

e. Indicate which session protocol (UDP or TCP) is used during the call between the server and the client.

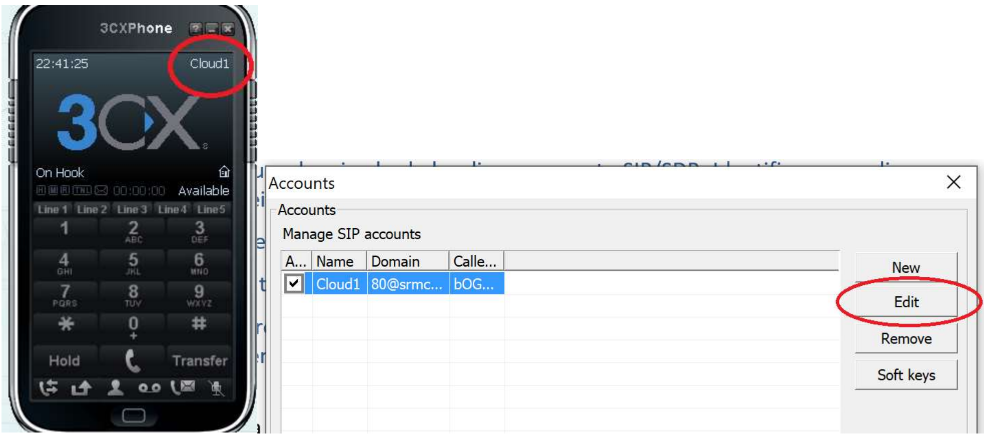
i. What are the advantages of using this protocol in a VoIP session?

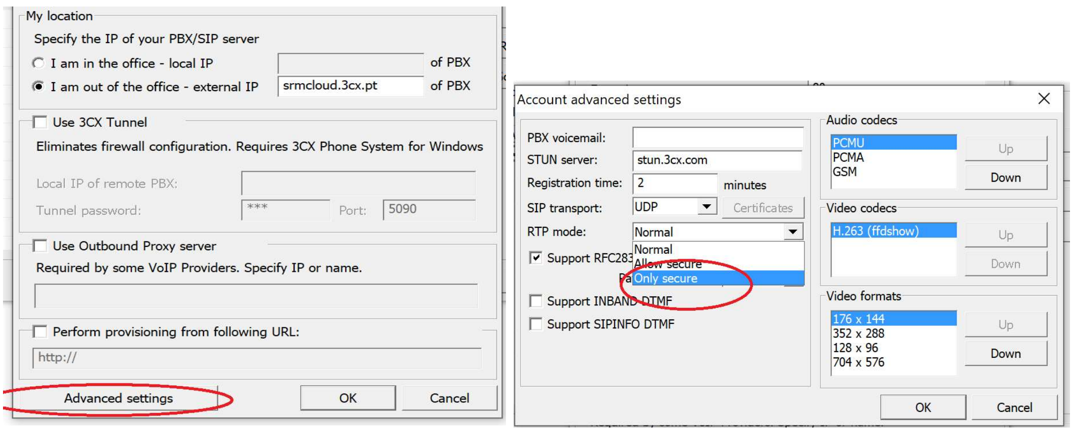
ii. What are the disadvantages? How are they outdated?

7. Make a new call between the two users, but in this case, the receiver should reject the call.

a) Reporting question: Interpret SIP signaling on call decline.

8. On the client softphone installed on PC change transport mode to safe mode as following images:





a) Capture packets from a new call originated by this user.

b) Reporting Question: What changes do you find in SIP / SDP and RTP packets? Explain the purpose of these changes.

9. Reporting Question - What are the advantages and disadvantages of using a VoIP server in a cloud environment?