VSB TECHNICAL UNIVERSITY OF OSTRAVA | FACULTY OF ELECTRICAL ENGINEERING AND COMPUTER SCIENCE | DEPARTMENT OF TELECOMMUNICATIONS

VoIP Exercise
Basic configuration of Asterisk PBX

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Course: Telecommunication Networks

1) Introduction

Voice over IP (VoIP) technology is used to carry voice signal over the IP network. This allows us to use IP phones connected via Ethernet or Wi-Fi or software clients instead of dedicated telephone lines. Thus, voice is transmitted in the form of packets over a data network that is not specifically designed to transmit voice traffic.

The main goal of the exercise is to get familiarized with a software implementation of a telephone private branch exchange (PBX) called Asterisk. PBX acts as the central switching system for phone calls. The content of this practical exercise is to install, configure and start the Asterisk PBX, with following verification of functionality. Another goal of the exercise is to analyze and understand the call flow that contains the Session Initiation Protocol (SIP) signaling messages.

2) Realization

Students have a total of 10 virtual machines that are running on Debian without a GUI. By mutual agreement, each will choose one to work on the exercise. These machines are available through SSH protocol and a tool like Putty.

Virtual machines

IP addresses of the machines:

158.196.244.181 - 158.196.244.190

Example of assignment for 3 groups of students (optional):

Group 1: 158.196.244.181

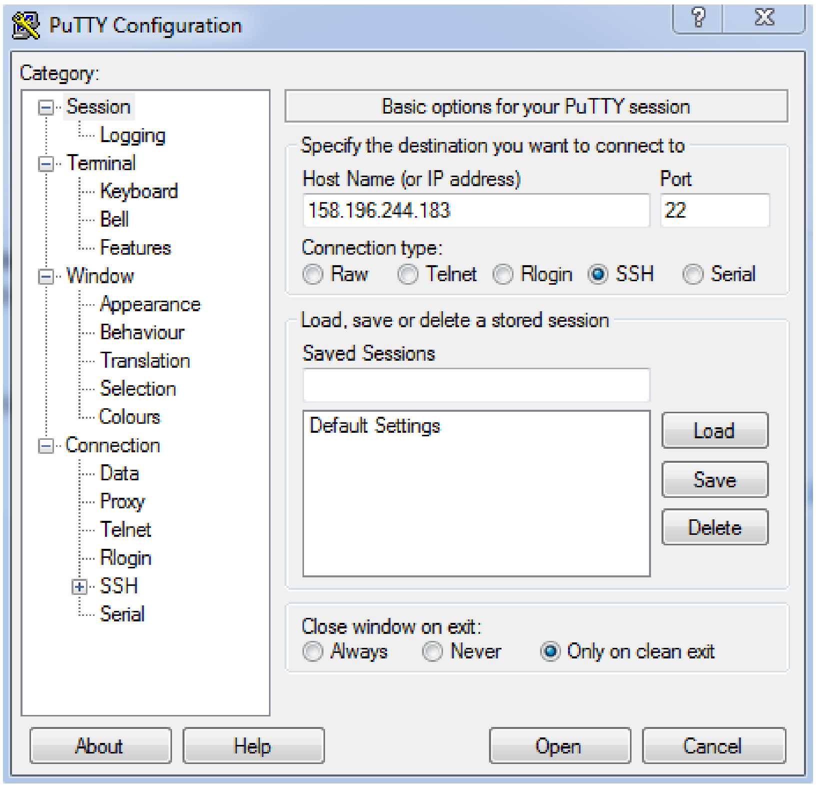
Group 2: 158.196.244.182

Group 3: 158.196.244.183

Enter your connection settings:

- **Host Name:** IP address of Virtual Machine (eg. 158.196.244.183)
- **Port:** 22 (leave as default)
- **Connection** Type: SSH (leave as default)

Click Open to start the SSH session.



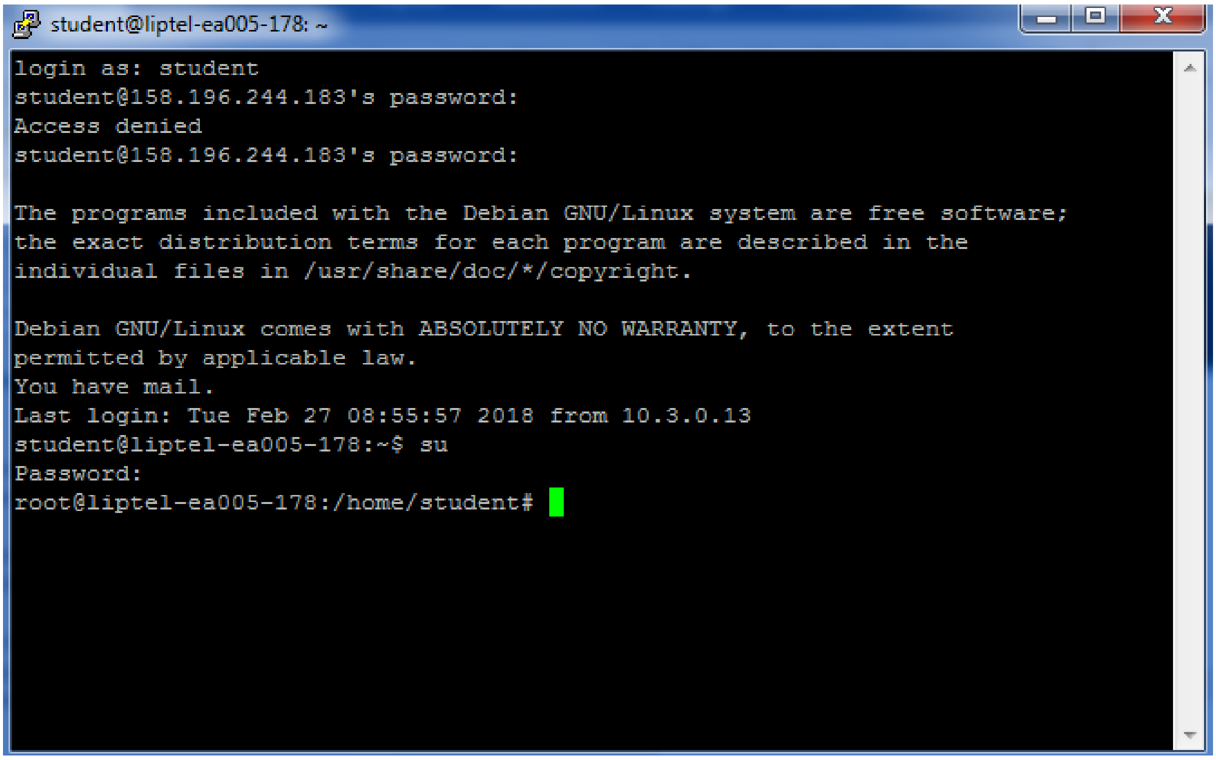
Once the SSH Connection is open, you should see a terminal prompt asking for your username:

Login: student

Password: k440

Change the ownership of a login session to root with following command (**password:** k440):

```
su
```



Asterisk installation:

Update Debian packages with following command:

```
apt update
```

First, we need to install missing system libraries and other dependencies. Unless you've installed all of the System Requirements for your version of Asterisk, the configure script is likely to fail.

```
apt install build-essential openssl libxml2-dev libncurses5-dev uuid-dev sqlite3 li
bsqlite3-dev pkg-config libjansson-dev
```

Change directories to the /usr/src source directory:

```
cd /usr/src/
```

Now download Asterisk using following command:

```
wget http://downloads.asterisk.org/pub/telephony/asterisk/asterisk-13-current.tar.g
z
```

Extract downloaded tar file:

```
tar xvf asterisk-13-current.tar.gz
```

Change directory to the extracted directory:

```
cd asterisk-13*
```

The next step in building and installing Asterisk is configuring it using following command:

```
./configure
```

The last step is to build and install Asterisk:

```
make && make install && make samples && make basic-pbx
```

Asterisk configuration

Assuming your asterisk server is up and running, we will only need to edit two files: sip.conf and extensions.conf.

sip.conf configuration

In the sip.conf file we can configure everything related to the SIP protocol, add new sip users or define sip providers. All phone numbers (users) must be registered in sip.conf.

Delete the content of sip.conf file:

```
rm /etc/asterisk/sip.conf
```

Open sip.conf file. It should be empty:

```
nano /etc/asterisk/sip.conf
```

The following is an example of the sip.conf configuration file content to create a single account with 5010. Always keep in mind that the number of accounts will correspond to the number of students in the group + one account for the hardware phone. Thus, for four students in a group, you create five accounts with the appropriate numbers according to the information from the instructor:

```
[5010]

type=friend

context=outgoing

host=dynamic

secret=5010

username=5010
```

[5010] - This means we are registering user '5010' extension
type=friend - This means the user can place or receive calls. For INBOUND calls only, use ‘peer’ as type. For outbound calls only use ‘user’ as type.
username=5010 - This declares that our user will be named '5010'
secret=5010 -This creates the password for the user to login/authenticate on Asterisk
host=dynamic -This sets dynamic IP for the host. You may also define this as a static IP
context=outgoing -This defines the dial context for the user which in this case is tutorial. In Asterisk, outgoing numbers are divided in groups called contexts in order to separate/define different needs

for different user types. For example, a context for local calls, another for within the city, and another for international calls and so on.

extensions.conf configuration

The configuration file "**extensions.conf**" contains the "dial plan" of Asterisk, the master plan of control or execution flow for all its operations. It controls how incoming and outgoing calls are handled and routed. This is where you configure the behavior of all connections through your PBX.

Delete the content of sip.conf file:

```
rm /etc/asterisk/extensions.conf
```

Open extensions.conf file. It should be empty:

```
nano /etc/asterisk/extensions.conf
```

Copy and paste following configuration into this file. The following configuration line tells Asterisk to call to all dialed extensions that start with number 5, and are 4 digits long.

```
[outgoing]

exten => _5XXX,1,Dial(SIP/${EXTEN})
```

To load the correct channel driver, you need to open modules.conf:

```
nano /etc/asterisk/modules.conf
```

Inside the modules.conf configuration file, find the line that contains loading of the chan_pjsip.so (load = chan_pjsip.so) and change it as follows:

```
load = chan_sip.so
```

After performing all this configuration, the Asterisk PBX is ready to start. You can start Asterisk in the foreground, with an attached root console, using the *-c* option. Asterisk provides several mechanisms to control the verbosity of its logging. One way in which this can be controlled is through the command line parameter *-v*. For each *-v* specified, Asterisk will increase the level of *VERBOSE* messages by 1. The following will create a console and set the *VERBOSE* message level to 3:

```
asterisk -cvvv
```

To reload the Asterisk, you can use this command within the Asterisk CLI:

```
core restart now
```

To show all defined accounts inside the sip.conf file, you can use this command within the Asterisk CLI:

```
sip show peers
```

To stop the Asterisk, you can use this command within the Asterisk CLI:

```
core stop now
```

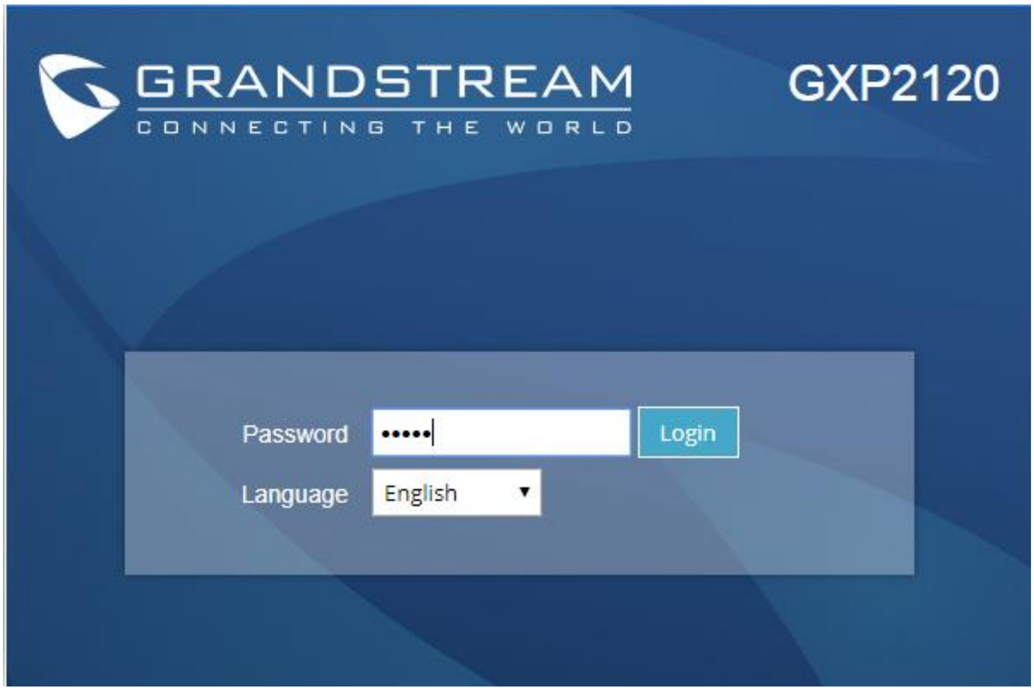
Grandstream hardware VoIP phone configuration

Connect the phone to a LAN port. The phone should obtain a valid IP Address (DHCP enabled by default). You can find the IP Address on the display of this phone (eg. 158.196.244.200).

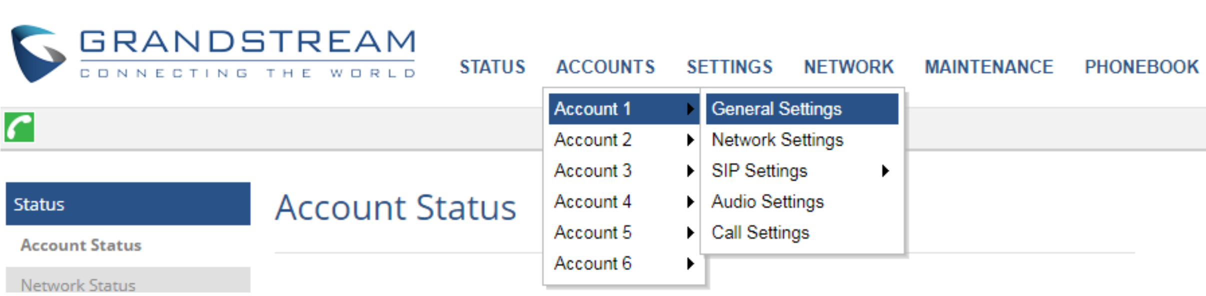
Enter the IP address in a browser and log into the web console of the Grandstream VoIP phone. The username and passwords are:

Username: admin

Password: admin

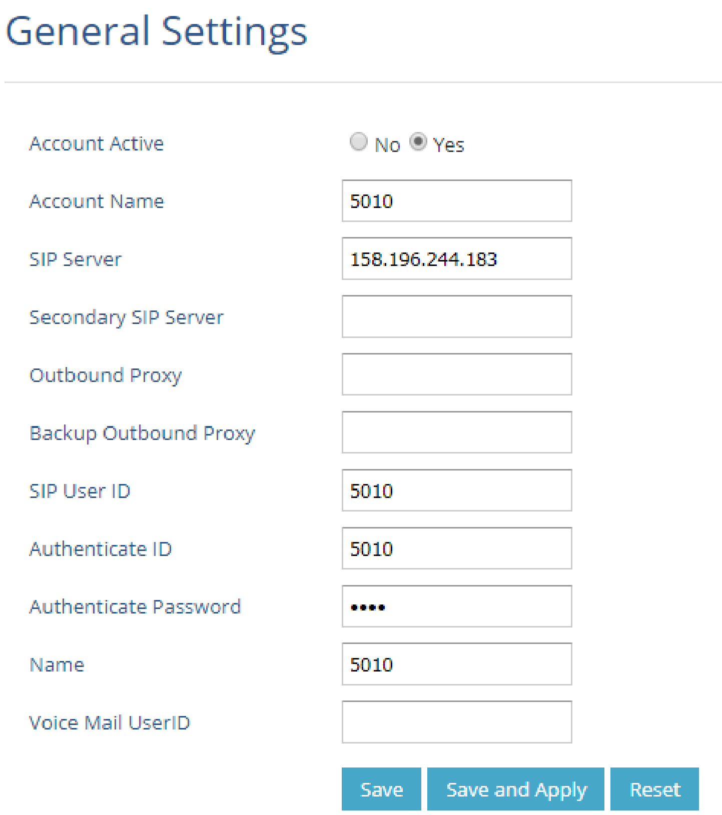


Navigate to the Grandstream Accounts webpage. Select first account to modify and open the **General Settings**.



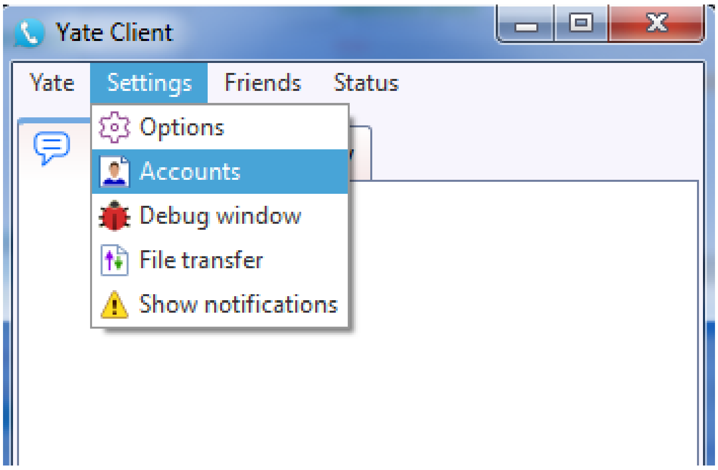
On the screen below enter the following details:

- **Account Name:** Your extension from sip.conf file.
- **SIP Server:** IP address of Asterisk server.
- **SIP User ID:** This is the username. Same as Account Name.
- **Authenticate ID:** This is the username of the credential. Same as SIP User ID.
- **Authenticate Password:** This is the password corresponding to the username.
- Save and Apply

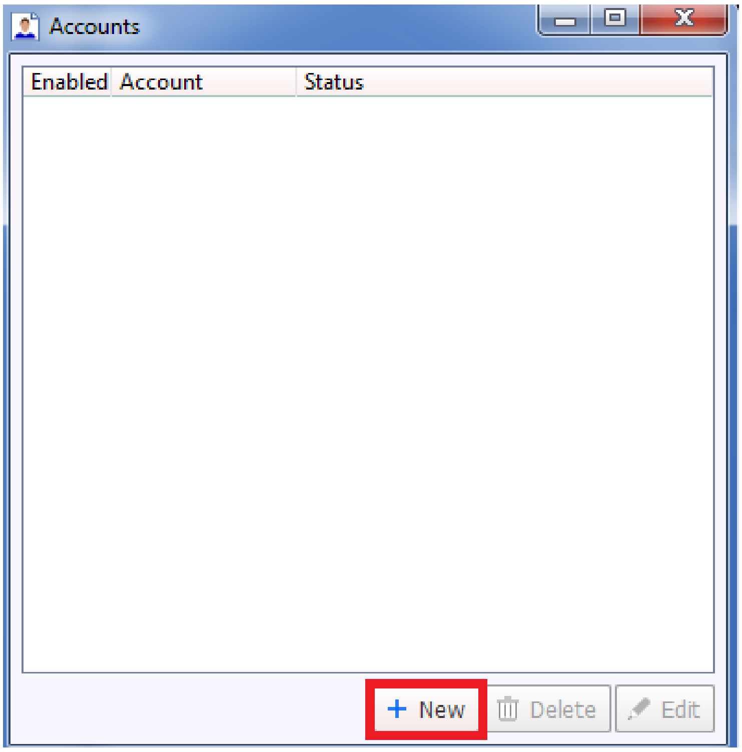


Yate Client configuration

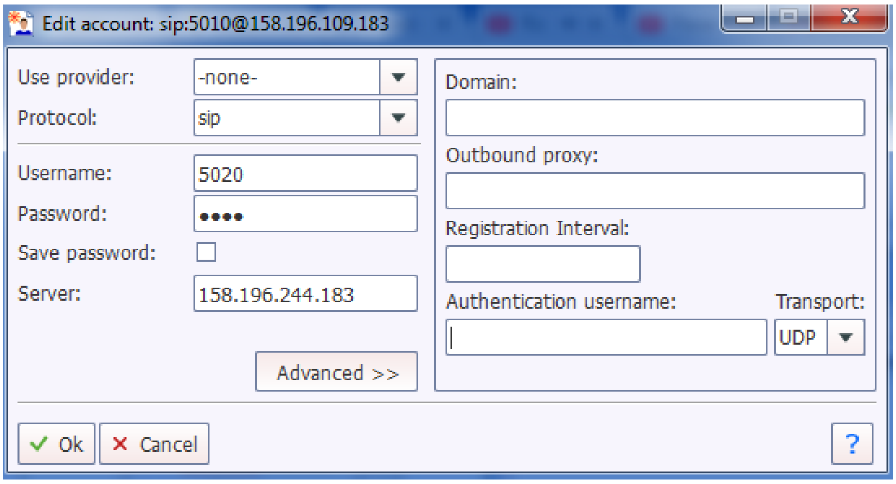
Open Yate Client softphone. In YateClient main window, go to Yate->Settings->Accounts:



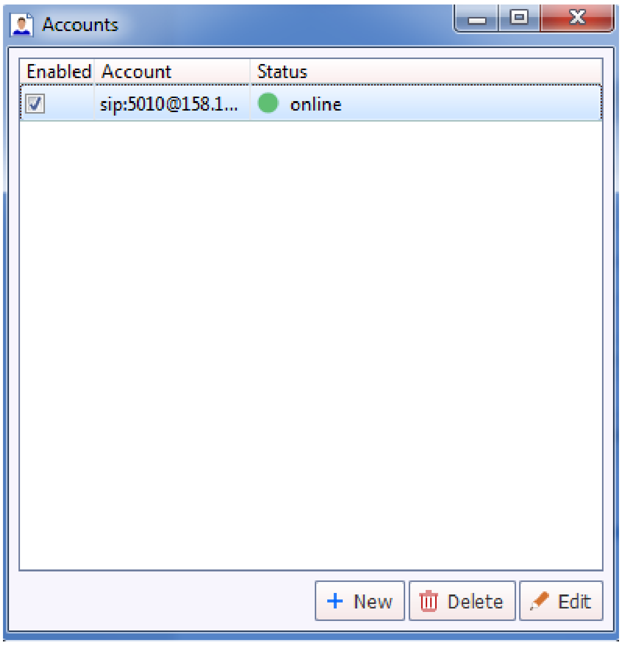
Open configuration form by clicking on the **NEW** button.



Select the SIP protocol, insert the SIP username, the SIP password and the Server IP. Apply configuration with OK button.

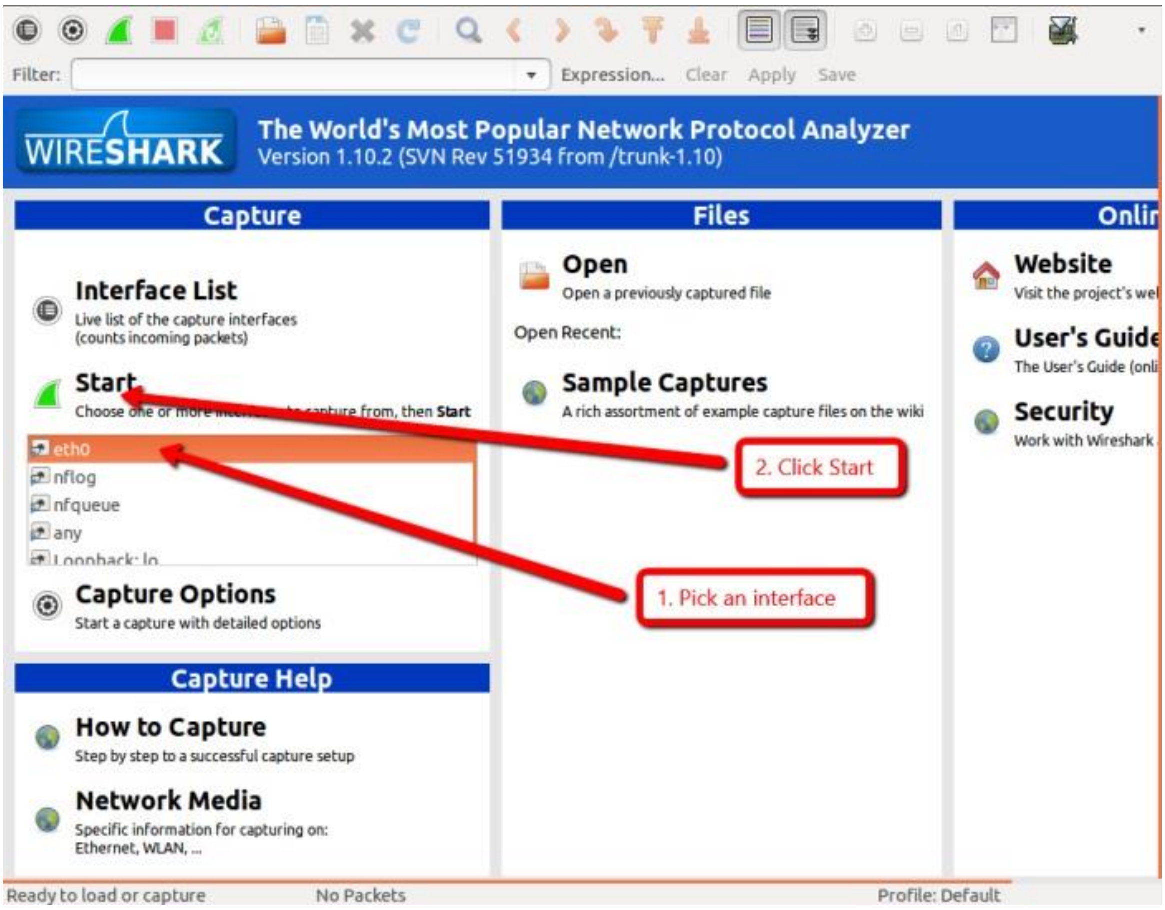


Be sure your SIP client is registered on the network, in Yate Client for example, you will see an “online” status.



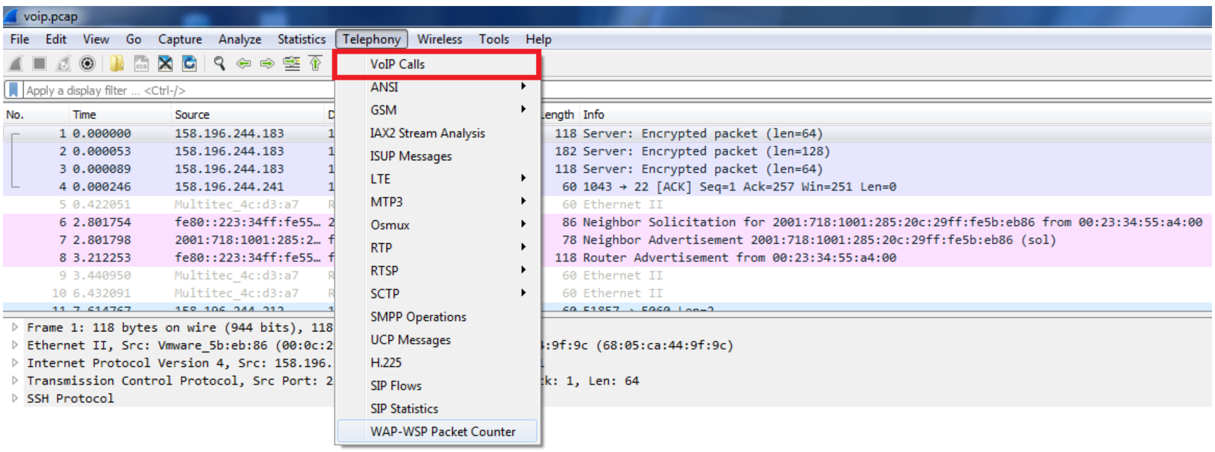
Analysis

Open the **wireshark** software. After launching wireshark application choose the name of a network interface you want to listen on and click on the **Start** button (see the picture below) .

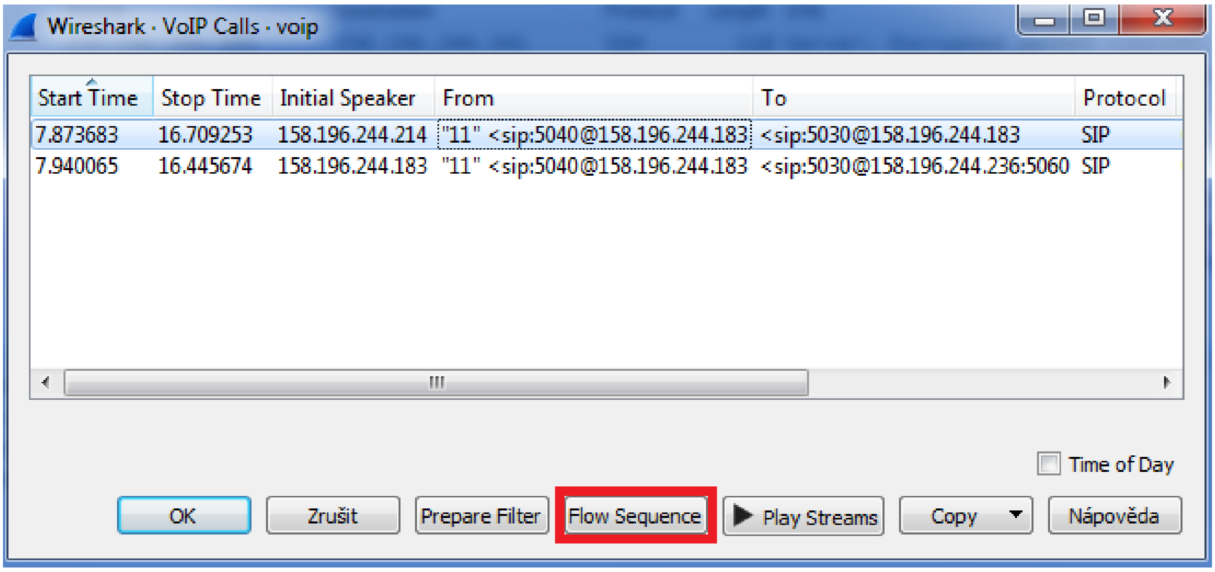


Now make a call between hardware phone and softphone Yate.

In main window, go to Telephony->VoIP Calls:



Select the phone call that was made between Yate Client and hardware phone. Then click on the Flow Sequence button.



The Flow graph displaying SIP messages should appear.