SRI RAMACHANDRA
INSTITUTE OF HIGHER EDUCATION AND RESEARCH
(Category - I Deemed to be University) Porur, Chennai
SRI RAMACHANDRA FACULTY OF ENGINEERING AND TECHNOLOGY

DEPLOYMENT AND TESTING OF OPENSOURCE IP-PBX SOLUTION FOR REALTIME COMMUNICATION

INT 400 Internship

SECOND Review

Subrat Biswal

Project Guide : Mr. Subrat Biswal

Team Member:

Keerthana S (E0322005)

OVERVIEW

The project "Deployment and Testing of Open-source IP-PBX Solution for Real-time Communication" involves implementing a scalable IP-PBX system using FreePBX. It covers the full lifecycle from requirement analysis to system setup, configuration, testing, and deployment. Key tasks include setting up SIP trunks, configuring extensions, and implementing security measures. Extensive testing ensures secure and reliable real-time communication.

DOMAIN INTRODUCTION – COMPUTER NETWORKS

Telecom in Computer Networks integrates with computer networks through IP-based communication protocols like VoIP for seamless connectivity.

- Telecom Sector: Covers all communication technologies, transmitting voice, data, and video via wired, wireless, and satellite networks.
- Ensures real-time communication
- IP-PBX: Routes voice calls over IP networks, replacing traditional PBX systems for modern telecom.
- Industry Trend: Increasing shift to VoIP for scalable and efficient enterprise communication.

PROBLEM STATEMENT

The growing demand for scalable, cost-effective, and flexible communication solutions highlights the limitations of traditional PBX systems, which are expensive, difficult to manage, and lack adaptability. Organizations require a modern IP-based system that can handle high call volumes, ensure secure real-time communication, and integrate with existing infrastructure. This project aims to deploy and test an open-source IP-PBX solution, addressing these challenges by providing a customizable, secure, and efficient platform for real-time voice communication, while minimizing operational costs and improving system scalability.

OBJECTIVES

- **Implement Open-source IP-PBX:** Deploy a scalable open-source IP-PBX solution to replace traditional PBX systems.
- Ensure seamless and high-quality real-time voice communication.
- **Configure** and optimize SIP trunks, extensions, and dial plans.
- Implement robust **security measures** to protect the communication system.
- **Test** the system for functionality, load capacity, and failover performance.
- **Cost Optimization:** Leverage the open-source nature of the solution to reduce operational costs while enhancing overall communication infrastructure scalability and efficiency.

LITERATURE SURVEY

YEAR	AUTHOR/ DEVELOPER	Title of the Research Paper / Product	SUMMARY
2016	Dinesh Kumar, Suresh Babu, Prasanna Venkatesan	Implementation and Evaluation of IP-PBX System Using Asterisk	This study demonstrates the deployment of an Asterisk-based IP-PBX system, focusing on its cost-effectiveness and suitability for small to medium-sized enterprises.
2017	Ahmed Saeed, Mohamed Khattab, Osama Mokhtar	Security Issues in VoIP and IP Telephony	The paper explores security vulnerabilities in VoIP systems, highlighting essential measures for securing IP-PBX deployments.
2018	Rajesh Patel, Nilesh Kumar, Bhavik Joshi	Performance Analysis of Open-source IP-PBX System	This paper analyzes the performance of an open-source IP-PBX system under various network conditions, providing insights into its reliability and scalability.

LITERATURE SURVEY

YEAR	AUTHOR/ DEVELOPER	Title of the Research Paper / Product	SUMMARY
2019	Emily Wong, Robert Adams, Jonathan Lin	Comparison of Proprietary and Open-source IP-PBX Solutions	The authors compare open-source and proprietary IP-PBX solutions, emphasizing the benefits of open-source options in terms of flexibility and cost.
2020	Manish Gupta, Priya Sharma, Arjun Rao	Deployment Challenges and Best Practices for Open-source IP Telephony	This paper discusses common challenges in deploying open-source IP-PBX systems and outlines best practices to ensure successful implementation and operation.

TOOLS

- **Virtualization Tools: VMware -** For creating and managing virtual machines to host the CentOS environment and the IP-PBX software.
- **Operating System: CentOS -** A Linux-based operating system used to host the IP-PBX solution.
- **IP-PBX Software: FreePBX -** Popular open-source IP-PBX system used for managing and routing voice communications over IP networks.
- **PuTTY:** For secure remote access to the server via SSH.
- **WinSCP:** For secure file transfer between the local machine and the server.(here from linux to windows)
- **Wireshark -** For monitoring and analyzing network traffic to troubleshoot and ensure proper communication protocols are followed.
- **Softphones**: 3cx & Zoiper

TECHNOLOGIES

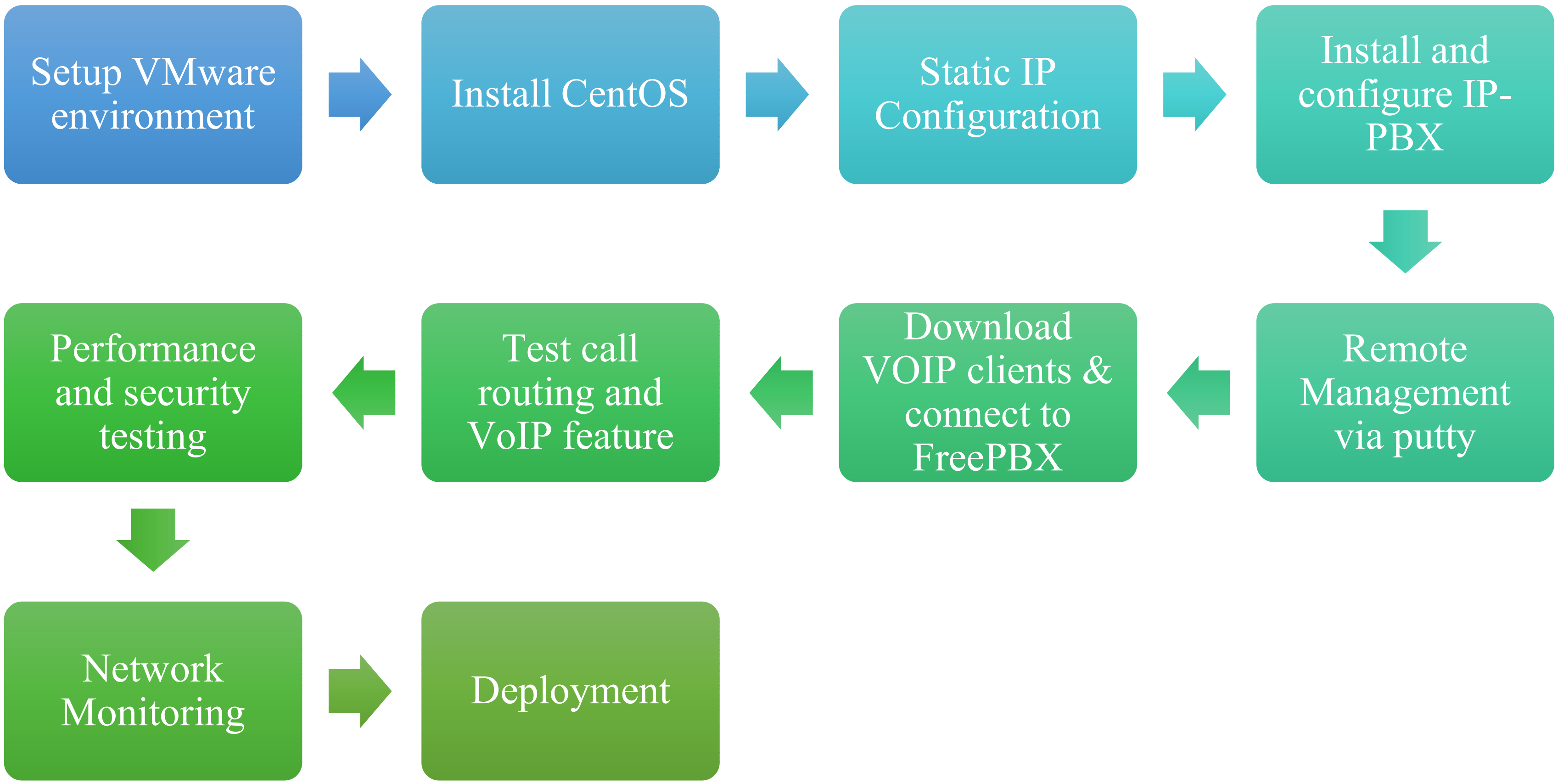
- **VoIP (Voice over IP):**
  - **SIP (Session Initiation Protocol):** A signalling protocol used to initiate, maintain, and terminate real-time communication sessions, such as voice and video calls.
  - **RTP (Real-time Transport Protocol):** Manages the transmission of media streams (voice and video) over IP networks during VoIP calls.
  - **VoIP Codecs (G.711, G.729, etc.):** Compression algorithms that convert voice into data packets for transmission over IP networks, optimizing bandwidth and call quality.
- **Networking:**
  - **Static IP Configuration:** Ensures consistent network connectivity by assigning a fixed IP address to the IP-PBX server.
  - **Firewall Configuration:** Ensuring the network is secure and only authorized traffic can reach the IP-PBX server.
- **SSH (Secure Shell):** A protocol used for securely accessing and managing remote servers (via PuTTY).

METHODOLOGY

1. **Environment Setup:**
   - **Install VMware:** Set up a virtual environment to host the CentOS VM.
   - **Install CentOS:** Deploy the CentOS operating system on the VMware virtual machine.
2. **Static IP Configuration:**
   - **Configure Static IP:** Assign a static IP address to the CentOS VM to ensure consistent network connectivity.
3. **Installation of IP-PBX Solution:**
   - **Download FreePBX:** Obtain the FreePBX installation package.
   - **Install FreePBX:** Follow the installation instructions to set up FreePBX on the CentOS VM.
4. **Tool Installation:**
   - **Install WinSCP:** Use WinSCP for file transfers between your local machine and the CentOS VM.
   - **Install Wireshark:** Set up Wireshark on your local machine for network traffic analysis.

5. **Download VoIP Clients:**
   - **Download 3CX:** Obtain the 3CX softphone client for managing calls.
   - **Download Zoiper:** Install the Zoiper softphone for additional VoIP functionalities.
6. **Integration and Configuration:**
   - **Configure FreePBX:** Set up extensions, SIP trunks, and user accounts within FreePBX.
   - **Connect Softphones:** Configure 3CX and Zoiper to connect to the FreePBX system.
7. **Network Monitoring:**
   - **Use Wireshark:** Monitor network traffic to troubleshoot any connectivity issues and ensure call quality.
8. **Testing and Validation:**
   - **Perform Functional Testing:** Test basic functionalities such as making and receiving calls using 3CX and Zoiper.
   - **Assess Performance:** Evaluate system performance and call quality under load.
9. **Deployment:**
   - **Go Live:** Transition the system to production and monitor its performance continuously for any issues.

WORKFLOW

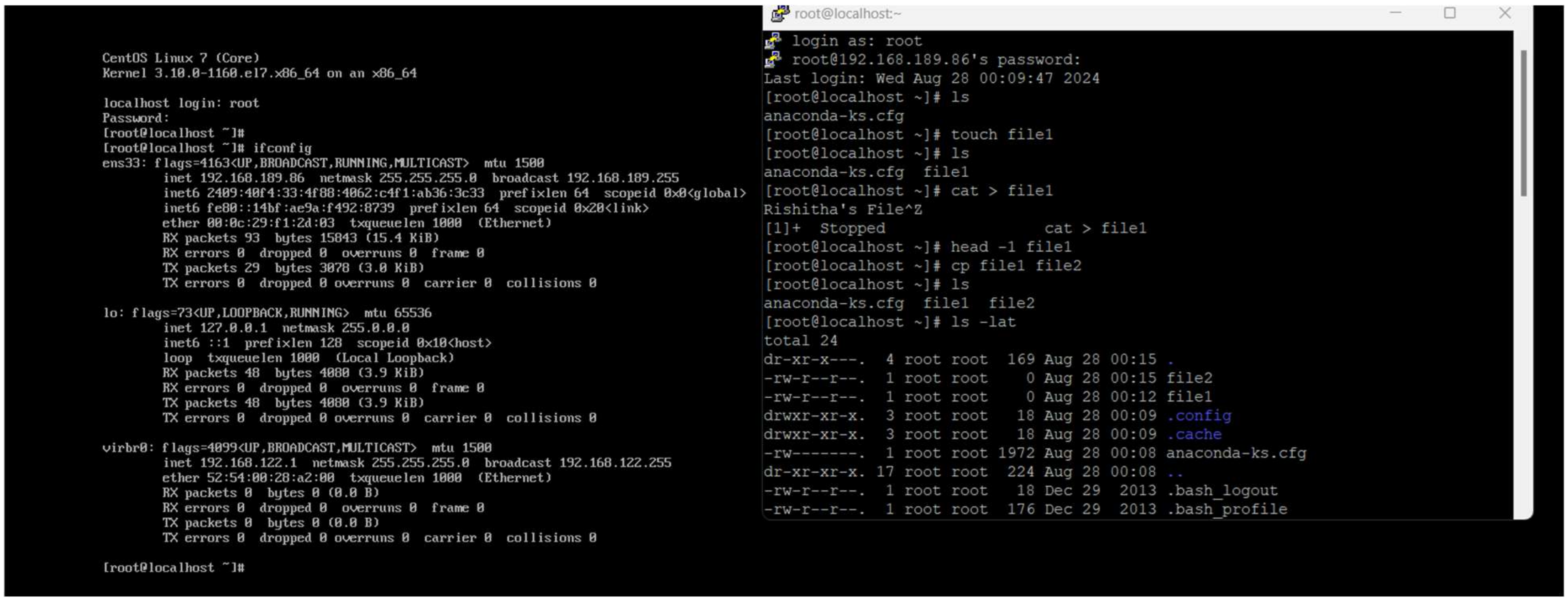

Setup VMware environment
Install CentOS
Static IP Configuration
Install and configure IP-PBX
Remote Management via putty
Download VOIP clients & connect to FreePBX
Test call routing and VoIP feature
Performance and security testing
Network Monitoring
Deployment

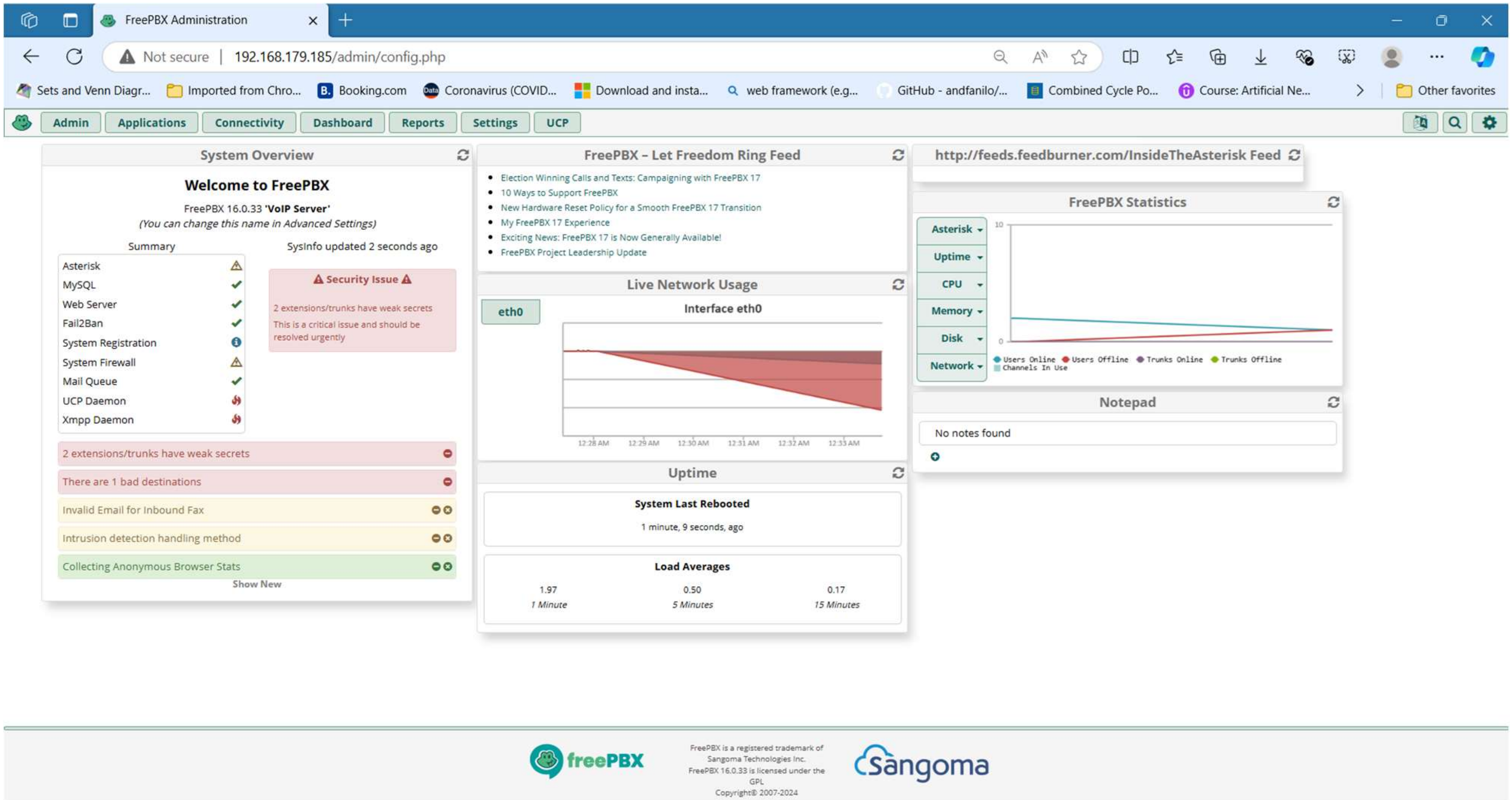
WORK DONE

Checking Voice Search In Home Page

Test ID	Test Case Name	Objective	Description	Pre - Requisite	Test Procedure/ Steps	Expected Result	Actual Result	Pass/Fail
TC-01	Checking Voice Search functionality	To verify that the voice search feature on the home page accurately recognizes and processes spoken commands to perform searches.	This test case checks whether the voice search feature on the home page correctly recognizes and interprets spoken input to return relevant search results. It ensures that the voice search is functional and responsive.	1.Access to the application home page. 2.User account credentials (if required for search). 3.Browser or application installed and configured for testing. 4.A microphone enabled and properly configured on the testing device.	1.Navigate to the application home page. 2.Ensure that the microphone is enabled and properly configured on the testing device. 3.Click on the voice search icon/button on the home page. 4.Speak a search query (e.g., "Search for the latest news"). 5.Observe the behavior and check if the spoken query is recognized and displayed in the search bar. 6.Ensure that the search results are displayed corresponding to the spoken query. 7.Repeat steps 3-6 with different search queries, including phrases, keywords, and questions. 8.Check if the voice search is responsive and returns relevant results for each query.	1.The voice search feature should accurately recognize and display the spoken query in the search bar. 2.Relevant search results corresponding to the spoken query should be displayed. 3.The voice search should be responsive, with minimal delay between the spoken command and the display of search results. 4.An appropriate error message should be displayed if the voice search fails to recognize the spoken command.	The voice search feature accurately recognized and displayed the spoken queries in the search bar. Relevant search results were displayed for each query, and the response time was quick. No errors were encountered, and the voice search performed as expected.	Pass

WORK DONE


CentOS Linux 7 (Core)
Kernel 3.10.0-1160.el7.x86_64 on an x86_64
localhost login: root
Password:
[root@localhost ~]#
[root@localhost ~]# ifconfig
ens33: flags=4163<UP,BROADCAST,RUNNING,MULTICAST> mtu 1500
inet 192.168.189.86 netmask 255.255.255.0 broadcast 192.168.189.255
inet6 2409:40f4:33:4f88:4062:c4f1:ab36:3c33 prefixlen 64 scopeid 0x0<global>
inet6 fe80::14bf:ae9a:f492:8739 prefixlen 64 scopeid 0x20<link>
ether 00:0c:29:f1:2d:03 txqueuelen 1000 (Ethernet)
RX packets 93 bytes 15843 (15.4 KiB)
RX errors 0 dropped 0 overruns 0 frame 0
TX packets 29 bytes 3078 (3.0 KiB)
TX errors 0 dropped 0 overruns 0 carrier 0 collisions 0
lo: flags=73<UP,LOOPBACK,RUNNING> mtu 65536
inet 127.0.0.1 netmask 255.0.0.0
inet6 ::1 prefixlen 128 scopeid 0x10<host>
loop txqueuelen 1000 (Local Loopback)
RX packets 48 bytes 4080 (3.9 KiB)
RX errors 0 dropped 0 overruns 0 frame 0
TX packets 48 bytes 4080 (3.9 KiB)
TX errors 0 dropped 0 overruns 0 carrier 0 collisions 0
virbr0: flags=4099<UP,BROADCAST,MULTICAST> mtu 1500
inet 192.168.122.1 netmask 255.255.255.0 broadcast 192.168.122.255
ether 52:54:00:28:a2:00 txqueuelen 1000 (Ethernet)
RX packets 0 bytes 0 (0.0 B)
RX errors 0 dropped 0 overruns 0 frame 0
TX packets 0 bytes 0 (0.0 B)
TX errors 0 dropped 0 overruns 0 carrier 0 collisions 0
[root@localhost ~]#
root@localhost:~
login as: root
root@192.168.189.86's password:
Last login: Wed Aug 28 00:09:47 2024
[root@localhost ~]# ls
anaconda-ks.cfg
[root@localhost ~]# touch file1
[root@localhost ~]# ls
anaconda-ks.cfg file1
[root@localhost ~]# cat > file1
Rishitha's File^Z
[1]+ Stopped   cat > file1
[root@localhost ~]# head -1 file1
[root@localhost ~]# cp file1 file2
[root@localhost ~]# ls
anaconda-ks.cfg file1 file2
[root@localhost ~]# ls -lat
total 24
dr-xr-x---. 4 root root 169 Aug 28 00:15 .
-rw-r--r--. 1 root root   0 Aug 28 00:15 file2
-rw-r--r--. 1 root root   0 Aug 28 00:12 file1
drwxr-xr-x. 3 root root   18 Aug 28 00:09 .config
drwxr-xr-x. 3 root root   18 Aug 28 00:09 .cache
-rw-------. 1 root root 1972 Aug 28 00:08 anaconda-ks.cfg
dr-xr-xr-x. 17 root root 224 Aug 28 00:08 ..
-rw-r--r--. 1 root root   18 Dec 29 2013 .bash_logout
-rw-r--r--. 1 root root 176 Dec 29 2013 .bash_profile



FreePBX Administration
Not secure | 192.168.179.185/admin/config.php
Sets and Venn Diagr...
Imported from Chro...
Booking.com
Coronavirus (COVID...
Download and insta...
web framework (e.g...
GitHub - andfanilo/...
Combined Cycle Po...
Course: Artificial Ne...
Other favorites
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
System Overview
Welcome to FreePBX
FreePBX 16.0.33 'VoIP Server'
(You can change this name in Advanced Settings)
Summary
SysInfo updated 2 seconds ago
Asterisk
MySQL
Web Server
Fail2Ban
System Registration
System Firewall
Mail Queue
UCP Daemon
Xmpp Daemon
Security Issue
2 extensions/trunks have weak secrets
This is a critical issue and should be resolved urgently
2 extensions/trunks have weak secrets
There are 1 bad destinations
Invalid Email for Inbound Fax
Intrusion detection handling method
Collecting Anonymous Browser Stats
Show New
FreePBX – Let Freedom Ring Feed
Election Winning Calls and Texts: Campaigning with FreePBX 17
10 Ways to Support FreePBX
New Hardware Reset Policy for a Smooth FreePBX 17 Transition
My FreePBX 17 Experience
Exciting News: FreePBX 17 is Now Generally Available!
FreePBX Project Leadership Update
Live Network Usage
eth0
Interface eth0
12:28 AM
12:29 AM
12:30 AM
12:31 AM
12:32 AM
12:33 AM
Uptime
System Last Rebooted
1 minute, 9 seconds, ago
Load Averages
1.97
1 Minute
0.50
5 Minutes
0.17
15 Minutes
http://feeds.feedburner.com/InsideTheAsterisk Feed
FreePBX Statistics
Asterisk
Uptime
CPU
Memory
Disk
Network
10
0
Users Online
Users Offline
Trunks Online
Trunks Offline
Channels In Use
Notepad
No notes found
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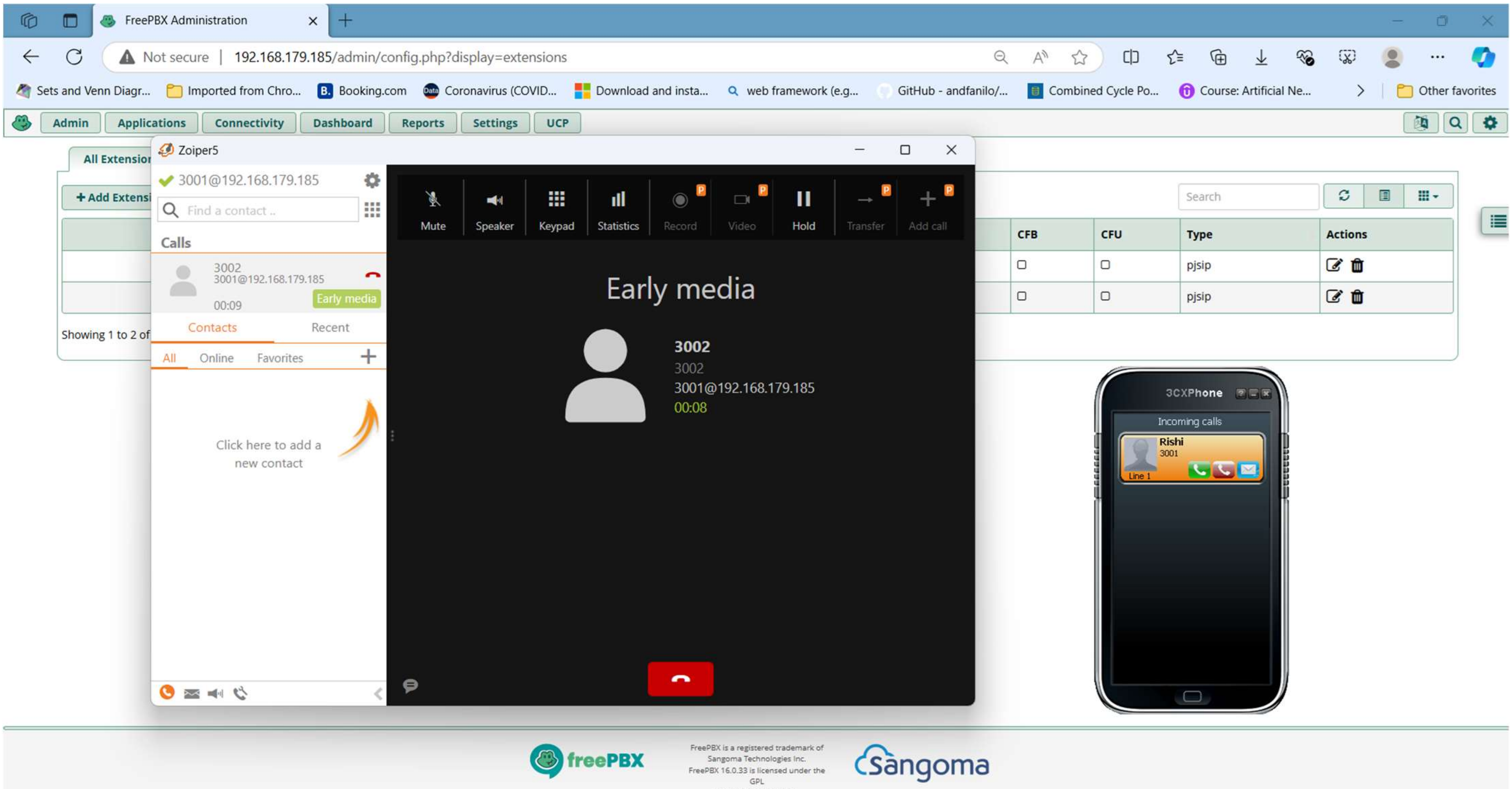
FreePBX Administration
Not secure | 192.168.179.185/admin/config.php?display=extensions
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
Zoiper5
3001@192.168.179.185
Find a contact ..
Calls
3002
3001@192.168.179.185
00:09
Early media
Contacts
Recent
All
Online
Favorites
Click here to add a new contact
Mute
Speaker
Keypad
Statistics
Record
Video
Hold
Transfer
Add call
Early media
3002
3002
3001@192.168.179.185
00:08
Search
CFB
CFU
Type
Actions
pjsip
pjsip
Showing 1 to 2 of
3CXPhone
Incoming calls
Rishi
3001
Line 1
freePBX
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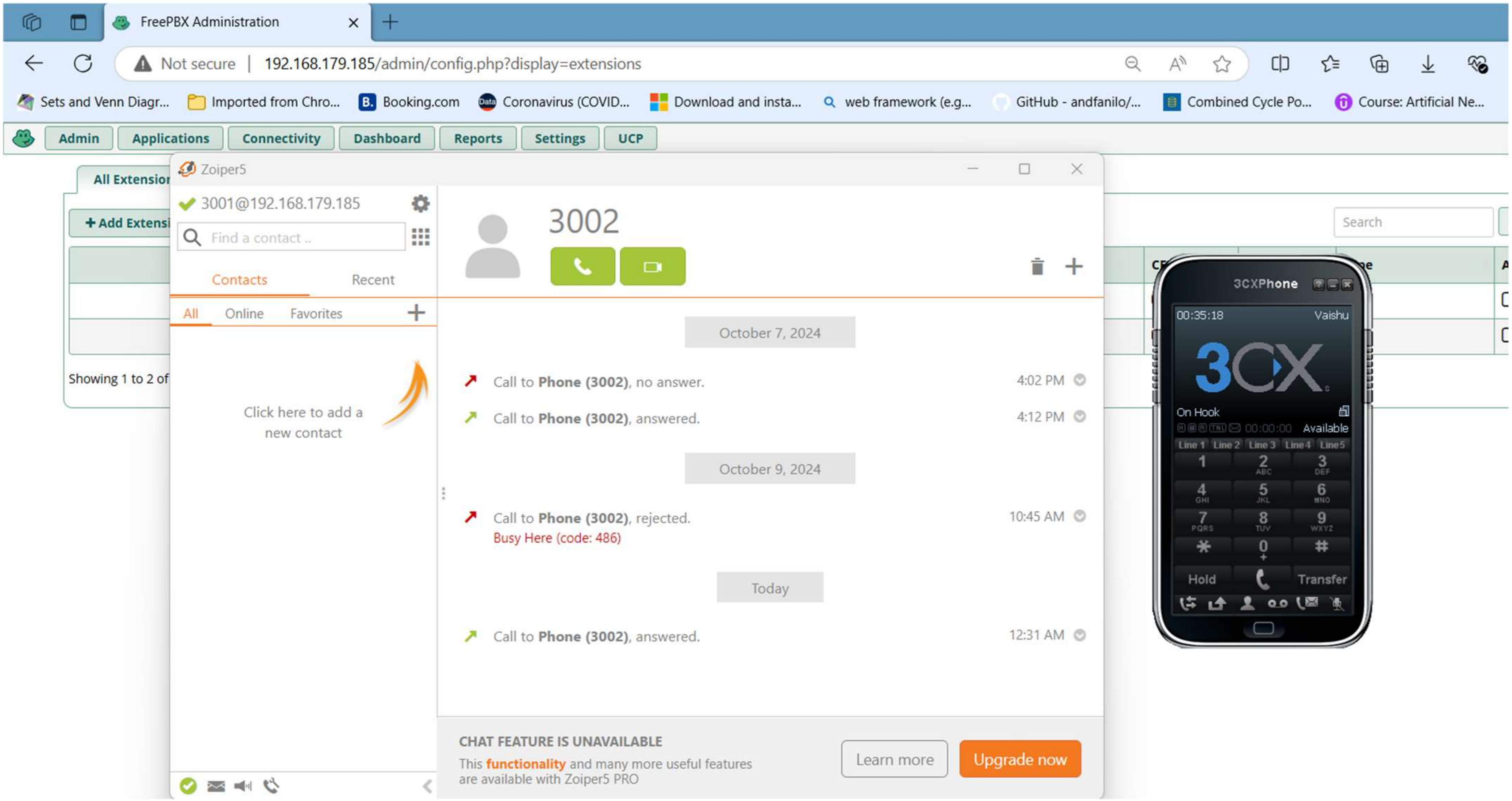
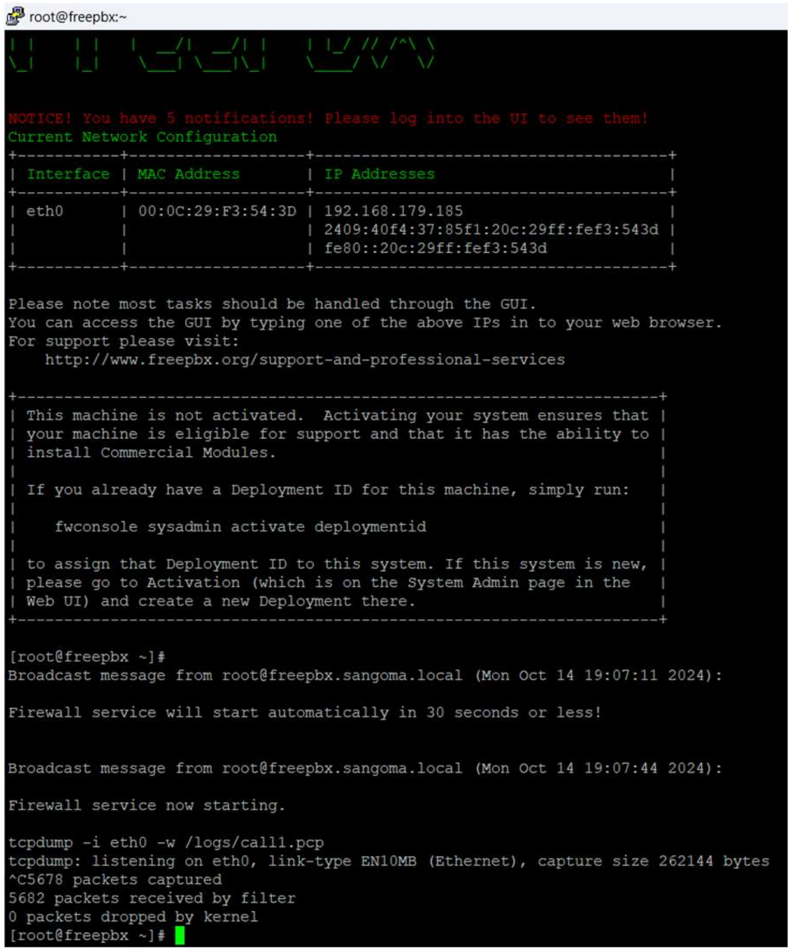
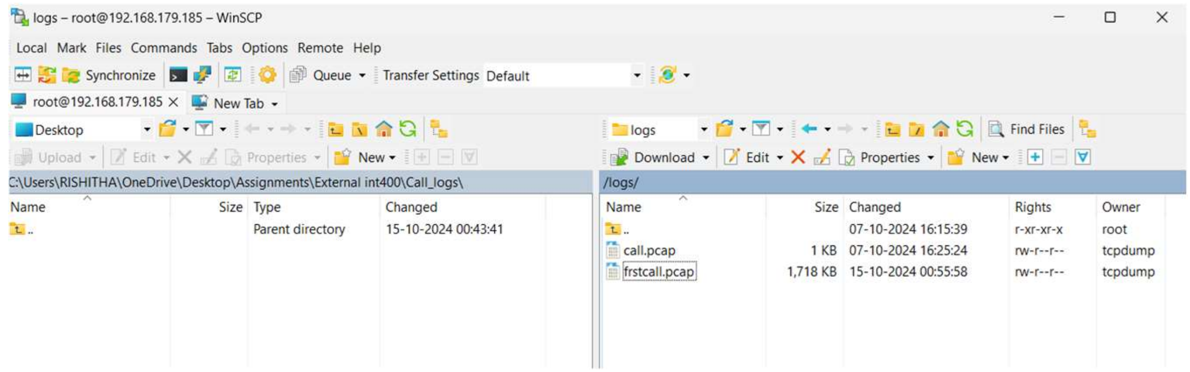
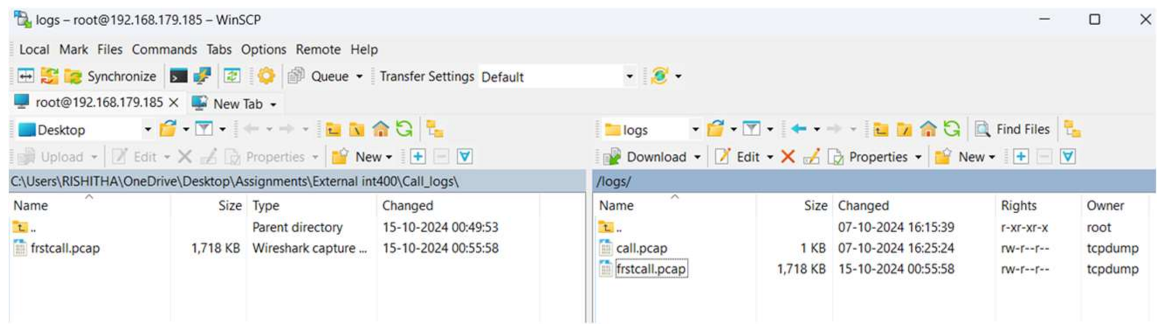
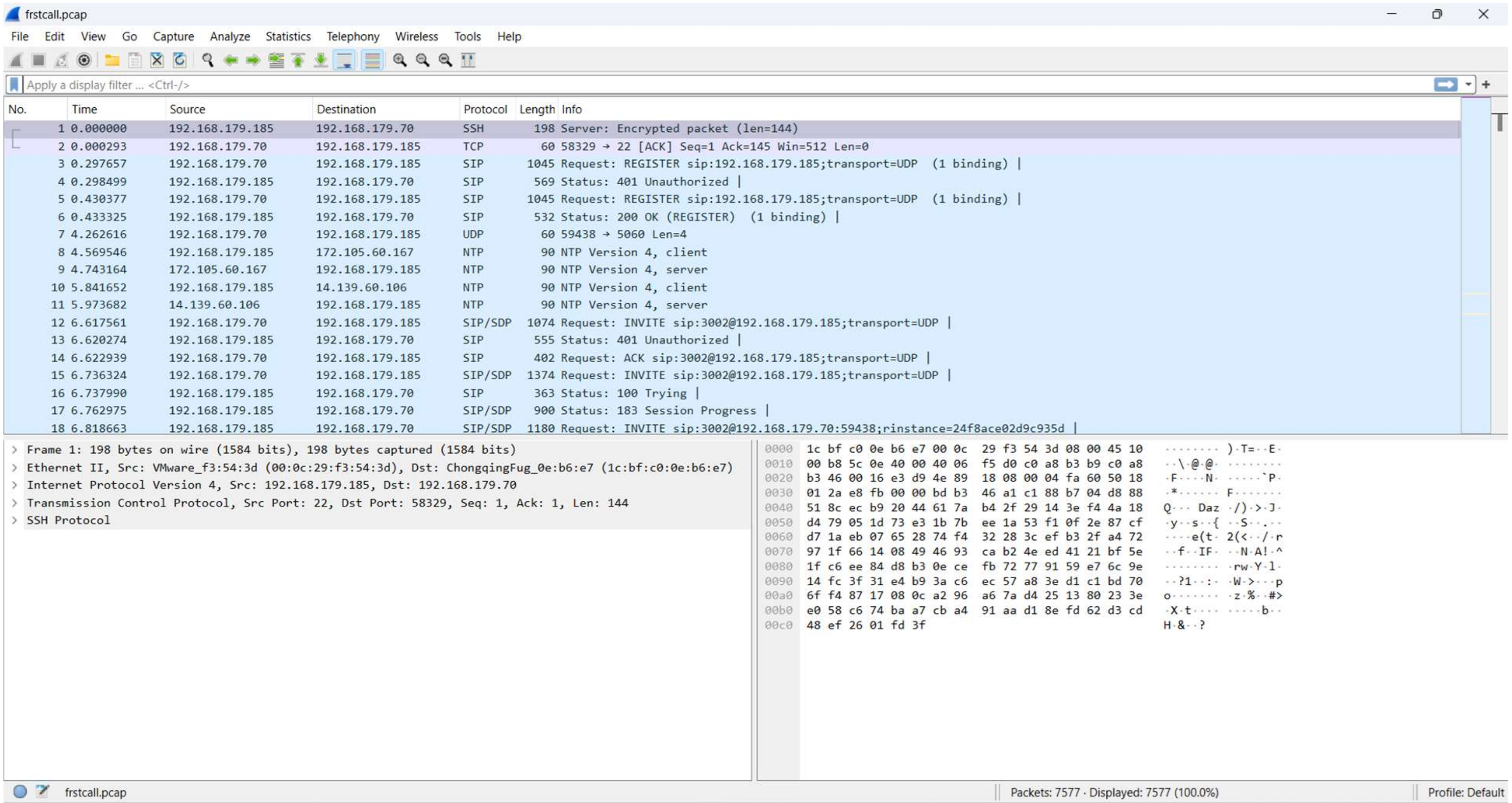
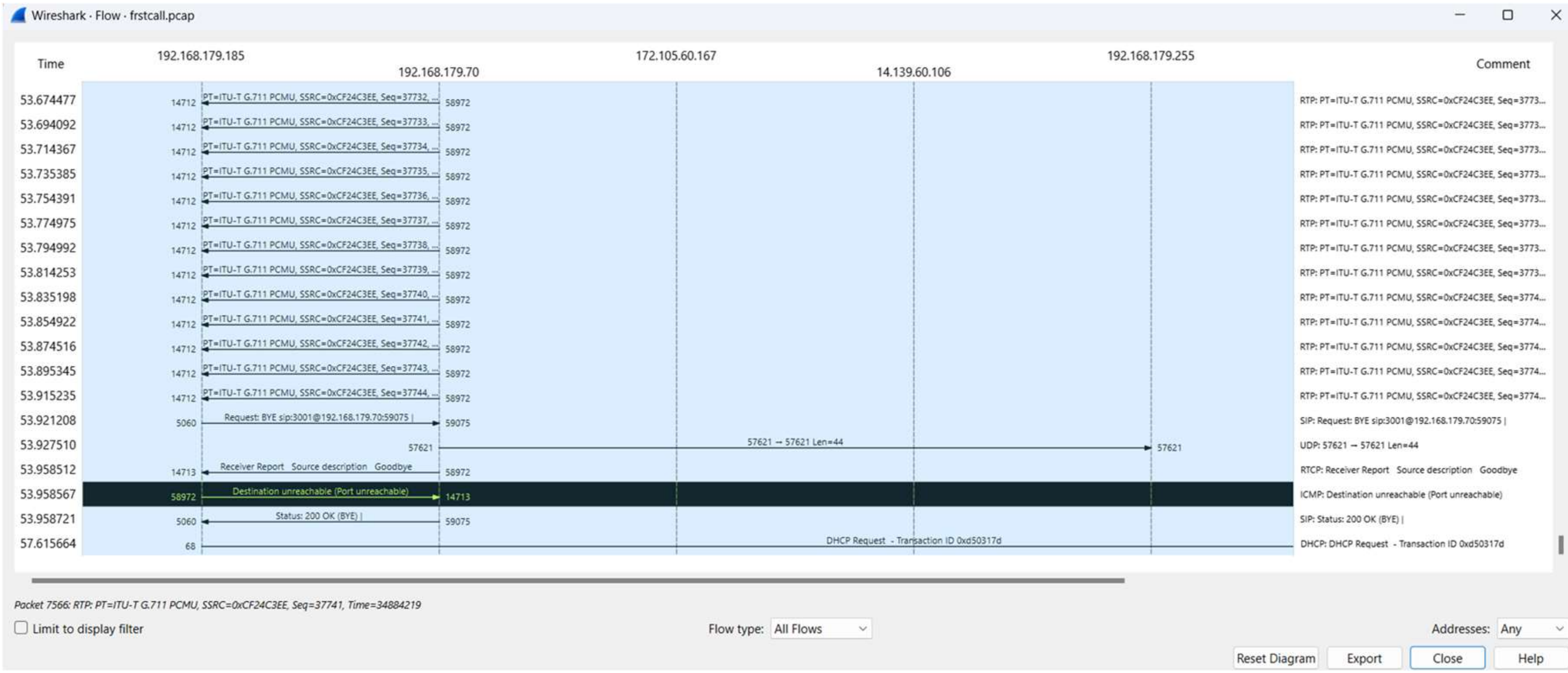

FreePBX Administration
Not secure | 192.168.179.185/admin/config.php?display=extensions
Sets and Venn Diagr...
Imported from Chro...
Booking.com
Coronavirus (COVID...
Download and insta...
web framework (e.g...
GitHub - andfanilo/...
Combined Cycle Po...
Course: Artificial Ne...
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
All Extensior
+ Add Extensi
Showing 1 to 2 of
Search
Zoiper5
3001@192.168.179.185
Find a contact ..
Contacts
Recent
All
Online
Favorites
Click here to add a new contact
3002
October 7, 2024
Call to Phone (3002), no answer. 4:02 PM
Call to Phone (3002), answered. 4:12 PM
October 9, 2024
Call to Phone (3002), rejected. 10:45 AM
Busy Here (code: 486)
Today
Call to Phone (3002), answered. 12:31 AM
CHAT FEATURE IS UNAVAILABLE
This functionality and many more useful features are available with Zoiper5 PRO
Learn more
Upgrade now
3CXPhone
00:35:18
Vaishu
3CX
On Hook
00:00:00
Available
Line 1
Line 2
Line 3
Line 4
Line 5
Hold
Transfer


root@freepbx:~
NOTICE! You have 5 notifications! Please log into the UI to see them!
Current Network Configuration
| Interface | MAC Address | IP Addresses |
| eth0 | 00:0C:29:F3:54:3D | 192.168.179.185
2409:40f4:37:85f1:20c:29ff:fef3:543d
fe80::20c:29ff:fef3:543d |
Please note most tasks should be handled through the GUI.
You can access the GUI by typing one of the above IPs in to your web browser.
For support please visit:
http://www.freepbx.org/support-and-professional-services
This machine is not activated. Activating your system ensures that your machine is eligible for support and that it has the ability to install Commercial Modules.
If you already have a Deployment ID for this machine, simply run:
fwconsole sysadmin activate deploymentid
to assign that Deployment ID to this system. If this system is new, please go to Activation (which is on the System Admin page in the Web UI) and create a new Deployment there.
[root@freepbx ~]#
Broadcast message from root@freepbx.sangoma.local (Mon Oct 14 19:07:11 2024):
Firewall service will start automatically in 30 seconds or less!
Broadcast message from root@freepbx.sangoma.local (Mon Oct 14 19:07:44 2024):
Firewall service now starting.
tcpdump -i eth0 -w /logs/call1.pcp
tcpdump: listening on eth0, link-type EN10MB (Ethernet), capture size 262144 bytes
^C5678 packets captured
5682 packets received by filter
0 packets dropped by kernel
[root@freepbx ~]#


logs – root@192.168.179.185 – WinSCP
Local Mark Files Commands Tabs Options Remote Help
Synchronize Queue Transfer Settings Default
root@192.168.179.185 New Tab
Desktop Upload Edit Properties New
C:\Users\RISHITHA\OneDrive\Desktop\Assignments\External int400\Call_logs\
Name Size Type Changed
.. Parent directory 15-10-2024 00:43:41
logs Find Files Download Edit Properties New
/logs/
Name Size Changed Rights Owner
.. 07-10-2024 16:15:39 r-xr-xr-x root
call.pcap 1 KB 07-10-2024 16:25:24 rw-r--r-- tcpdump
frstcall.pcap 1,718 KB 15-10-2024 00:55:58 rw-r--r-- tcpdump


logs – root@192.168.179.185 – WinSCP
Local Mark Files Commands Tabs Options Remote Help
Synchronize Queue Transfer Settings Default
root@192.168.179.185 New Tab
Desktop Upload Edit Properties New
C:\Users\RISHITHA\OneDrive\Desktop\Assignments\External int400\Call_logs\
Name Size Type Changed
.. Parent directory 15-10-2024 00:49:53
frstcall.pcap 1,718 KB Wireshark capture ... 15-10-2024 00:55:58
logs Find Files Download Edit Properties New
/logs/
Name Size Changed Rights Owner
.. 07-10-2024 16:15:39 r-xr-xr-x root
call.pcap 1 KB 07-10-2024 16:25:24 rw-r--r-- tcpdump
frstcall.pcap 1,718 KB 15-10-2024 00:55:58 rw-r--r-- tcpdump


frstcall.pcap
File Edit View Go Capture Analyze Statistics Telephony Wireless Tools Help
Apply a display filter ... <Ctrl-/>
No. Time Source Destination Protocol Length Info
1 0.000000 192.168.179.185 192.168.179.70 SSH 198 Server: Encrypted packet (len=144)
2 0.000293 192.168.179.70 192.168.179.185 TCP 60 58329 → 22 [ACK] Seq=1 Ack=145 Win=512 Len=0
3 0.297657 192.168.179.70 192.168.179.185 SIP 1045 Request: REGISTER sip:192.168.179.185;transport=UDP (1 binding) |
4 0.298499 192.168.179.185 192.168.179.70 SIP 569 Status: 401 Unauthorized |
5 0.430377 192.168.179.70 192.168.179.185 SIP 1045 Request: REGISTER sip:192.168.179.185;transport=UDP (1 binding) |
6 0.433325 192.168.179.185 192.168.179.70 SIP 532 Status: 200 OK (REGISTER) (1 binding) |
7 4.262616 192.168.179.70 192.168.179.185 UDP 60 59438 → 5060 Len=4
8 4.569546 192.168.179.185 172.105.60.167 NTP 90 NTP Version 4, client
9 4.743164 172.105.60.167 192.168.179.185 NTP 90 NTP Version 4, server
10 5.841652 192.168.179.185 14.139.60.106 NTP 90 NTP Version 4, client
11 5.973682 14.139.60.106 192.168.179.185 NTP 90 NTP Version 4, server
12 6.617561 192.168.179.70 192.168.179.185 SIP/SDP 1074 Request: INVITE sip:3002@192.168.179.185;transport=UDP |
13 6.620274 192.168.179.185 192.168.179.70 SIP 555 Status: 401 Unauthorized |
14 6.622939 192.168.179.70 192.168.179.185 SIP 402 Request: ACK sip:3002@192.168.179.185;transport=UDP |
15 6.736324 192.168.179.70 192.168.179.185 SIP/SDP 1374 Request: INVITE sip:3002@192.168.179.185;transport=UDP |
16 6.737990 192.168.179.185 192.168.179.70 SIP 363 Status: 100 Trying |
17 6.762975 192.168.179.185 192.168.179.70 SIP/SDP 900 Status: 183 Session Progress |
18 6.818663 192.168.179.185 192.168.179.70 SIP/SDP 1180 Request: INVITE sip:3002@192.168.179.70:59438;rinstance=24f8ace02d9c935d |
Frame 1: 198 bytes on wire (1584 bits), 198 bytes captured (1584 bits)
Ethernet II, Src: VMware_f3:54:3d (00:0c:29:f3:54:3d), Dst: ChongqingFug_0e:b6:e7 (1c:bf:c0:0e:b6:e7)
Internet Protocol Version 4, Src: 192.168.179.185, Dst: 192.168.179.70
Transmission Control Protocol, Src Port: 22, Dst Port: 58329, Seq: 1, Ack: 1, Len: 144
SSH Protocol
0000 1c bf c0 0e b6 e7 00 0c 29 f3 54 3d 08 00 45 10
0010 00 b8 5c 0e 40 00 40 06 f5 d0 c0 a8 b3 b9 c0 a8
0020 b3 46 00 16 e3 d9 4e 89 18 08 00 04 fa 60 50 18
0030 01 2a e8 fb 00 00 bd b3 46 a1 c1 88 b7 04 d8 88
0040 51 8c ec b9 20 44 61 7a b4 2f 29 14 3e f4 4a 18
0050 d4 79 05 1d 73 e3 1b 7b ee 1a 53 f1 0f 2e 87 cf
0060 d7 1a eb 07 65 28 74 f4 32 28 3c ef b3 2f a4 72
0070 97 1f 66 14 08 49 46 93 ca b2 4e ed 41 21 bf 5e
0080 1f c6 ee 84 d8 b3 0e ce fb 72 77 91 59 e7 6c 9e
0090 14 fc 3f 31 e4 b9 3a c6 ec 57 a8 3e d1 c1 bd 70
00a0 6f f4 87 17 08 0c a2 96 a6 7a d4 25 13 80 23 3e
00b0 e0 58 c6 74 ba a7 cb a4 91 aa d1 8e fd 62 d3 cd
00c0 48 ef 26 01 fd 3f
frstcall.pcap
Packets: 7577 · Displayed: 7577 (100.0%)
Profile: Default



Wireshark · SIP Flows · frstcall.pcap

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Duration	Packets	State	Comments
0.297657	53.661965	192.168.179.70	<sip:3001@192.168.179.185;transport=UDP>	<sip:3001@192.168.179.185;transport=UDP>	SIP	00:00:53	8	REJECTED	REGISTER 401
6.617561	53.958721	192.168.179.70	<sip:3001@192.168.179.185;transport=UDP>	<sip:3002@192.168.179.185>	SIP	00:00:47	12	COMPLETED	INVITE 401 200
6.818663	53.582300	192.168.179.185	"Rishi" <sip:3001@192.168.179.185>	<sip:3002@192.168.179.70;rinstance=24f8ace02d9c935d>	SIP	00:00:46	12	COMPLETED	INVITE 200
12.059417	12.171766	192.168.179.70	"3002"<sip:3002@192.168.179.185:5060>	"3002"<sip:3002@192.168.179.185:5060>	SIP	00:00:00	4	REJECTED	REGISTER 401
31.960806	32.078727	192.168.179.185	<sip:3002@192.168.179.185>	<sip:3002@192.168.179.70;rinstance=24f8ace02d9c935d>	SIP	00:00:00	2	CALL SETUP	OPTIONS 200
42.390620	42.397079	192.168.179.185	<sip:3001@192.168.179.185>	<sip:3001@192.168.179.70;rinstance=dae95e30e1a03e10>	SIP	00:00:00	2	CALL SETUP	OPTIONS 200

APPLICATIONS

- **Enterprise Communication:** Streamlines internal and external voice communication, enhancing call management and routing for businesses.
- **Customer Support Centers:** Provides a scalable platform for high-volume customer service calls, featuring IVR, call queuing, and voicemail.
- **Remote Work Solutions:** Facilitates secure VoIP communication for distributed teams, enabling effective collaboration from anywhere.
- **Educational Institutions:** Enhances communication between departments, staff, and students with cost-effective voice solutions.
- **Healthcare and Government**: Supports real-time communication for telemedicine and patient care, while enabling secure communication across public sector agencies for improved administration and safety.

TIMELINE

DATE	TASK / DESCRIPTION
21-08-2024	Introduction about the domain & the company
22-08-2024	Software Testing (Testing models, Testing lifecycle, Test cases)
23-08-2024	Task 1 presentation – Positive & Negative Scenarios
26-08-2024	CentOS and Linux Commands
27-08-2024	Installation Process
28-08-2024	Testing Linux Commands
29-08-2024	Task 2- Test cases Presentation
30-08-2024	Learnt about Communications (Telecom)
02-09-2024	Telecommunication Basics
03-09-2024	Basics Of networking

TIMELINE

DATE	TASK / DESCRIPTION
04-09-2024	OSI Layers & TCP/IP
05-09-2024	Routing & Doubt Clearing session
06-09-2024	Ip address & Mac Address
09-09-2024	Subnetting, CIDR
10-09-2024	Network Assignments
11-09-2024	First Review
12-09-2024	Static IP & Dynamic IP
13-09-2024	Setting up static IP
16-09-2024	Static IP Configuration
17-09-2024	Session initiation protocol
18-09-2024	Sip Servers(proxy,b2b,register,location,redirect)

TIMELINE

DATE	TASK / DESCRIPTION
19-09-2024	Sip stack & message headers
20-09-2024	Sip session, Transactions & Dialogs
23-09-2024	Sip messages & responses
24-09-2024	Session description Protocol
25-09-2024	SDP headers & Codecs
26-09-2024	Basic Call flows & call hold
27-09-2024	Forking, Call forwarding ,Call transferring
30-09-2024	Doubt Clearing session
1-10-2024	FreePBX installation
3-10-2024	Setting up FreePBX

TIMELINE

DATE	TASK / DESCRIPTION
4-10-2024	Installation of softphones(Zoiper & 3cx)
7-10-2024	Setting up the extensions & trunks
8-10-2024	Installation of WinSCP & Wireshark
14-10-2024	WinSCP & Wireshark usage demo
15-10-2024	Doubt Clearing session
16-10-2024	Second Review

CONCLUSION

In conclusion, the deployment and testing of an open-source IP-PBX solution significantly enhance communication efficiency within organizations. By providing a scalable, cost-effective, and customizable platform, this project addresses the limitations of traditional PBX systems. The integration of secure VoIP technology enables seamless communication, supporting diverse applications across various sectors. Comprehensive testing ensures high-quality voice transmission and robust security measures, ultimately improving overall communication management. This project positions organizations to adapt to evolving communication needs and foster better collaboration in a dynamic environment.

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