VoIP: Voice over Internet Protocol for Small Single-board Computers

Term Project Presentation

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College of Engineering

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- Obstacles
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Objective

- Determine optimal voice transmission paradigm in edge scenario
  - Architecture: IP-based PBX
  - Devices: Raspberry Pi, a cheap and energy efficient single board computer, can run IP-PBX software and control calls made over VoIP, but it is limited by its low memory and weak CPU.
  - Requirements:
    - Free and cheap communication within the local area, whiling maintaining egress access to remote entities.
    - Distributed implementation and centralized management.
- Implementation using IoT devices
  - Fulfills the IP-PBX functions with Raspberry Pi
  - Verifies the suitability of Raspberry Pis for IP-PBX software under a variety of stressors.
- Tests
  - Voice calls and transmission are made at different distances from the device and with the device working with different network difficulties.

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Introduction & Background: PBX & VoIP

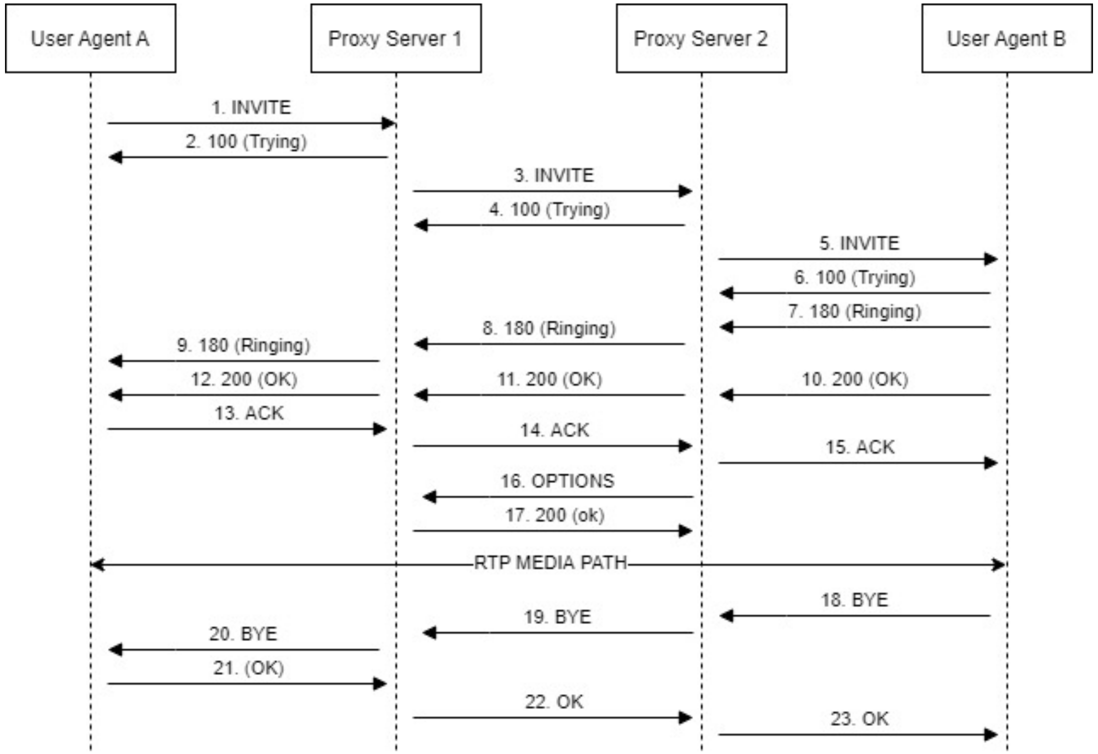
- Private Branch Exchange (PBX)
  - a technology which allows telephones to call each other, forwarding audio and controlling call behavior. While other technologies can provide similar services, PBX technology is designed for calls within a private institution.
- Voice over IP (VoIP)
  - allowing for telephone services to be provided over IP networks instead of standard telephone networks. Today, IP networks are so cheap and widespread that it is more cost efficient to use them for telephony than standard phone networks.
- Characteristics
  - Hosting communication within a limited area, while still maintaining access to the entities outside is a natural characteristics of IoT scenarios and the reason IP-based PBX paradigm is suitable for IoT.

Introduction & Background: IP-PBX

- IP-PBX was made to act as a PBX for VoIP services, as a traditional PBX does not interact with IP networks.
- IP-PBX is vital for IP telephony services, as without it, no two VoIP callers can connect to each other. IP-PBX can handle SIP messaging to manage call setup and RTP packages to relay audio data between callers and callees.
- SIP and RTP/RTCP act as the backbone of IP-PBX.
- SIP allows for calls to be started, connecting different callers together and allowing for complex behaviors such as putting callers on hold.
- RTP/RTCP allows for an effective means of forwarding audio data between callers and the control of this forwarding.

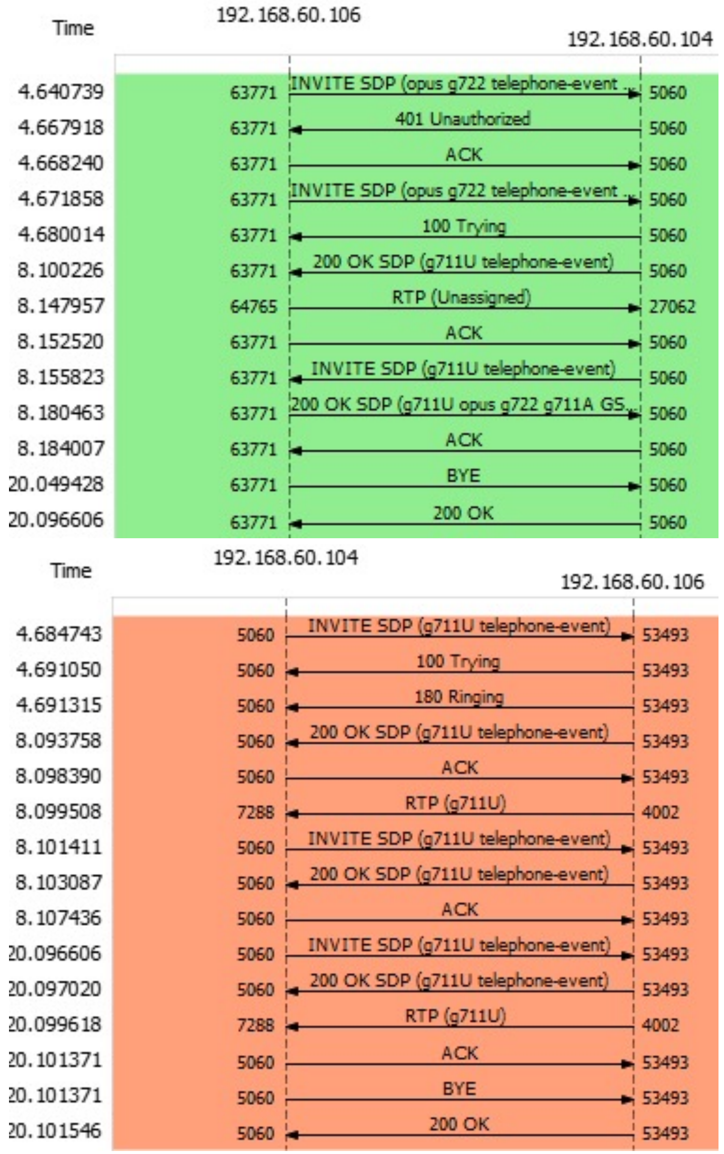
Introduction & Background: SIP

- Provides the fundamentals of the PBX communication
- In IP-based PBX implemented in the local network, the two proxy servers may merge into one.
- SIP components:
  - Model: session or pager
  - Format: request/status line, header, body
- From the SIP perspective, point out the core factors for a multimedia session: id, authentication, proxy, distance/hop count



Introduction & Background: RTP/RTCP

- RTP carries the media streams, while RTCP is used to monitor transmission statistics and quality of service
- RTP: sensitive to packet delays and less sensitive to packet loss
- Elements for RTP: sequence number, timestamp, source identifier, contributor identifier for multiple source
- RTCP allows the recipients so that they can send feedback to the source or sources.



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Obstacles

- PBX is complex
  - The IP-PBX software used was Asterisk PBX, a very robust and mature implementation, making it also very complicated
  - Much third-party software is involved: Asterisk PBX for the PBX, Zoiper to make calls, Wireshark for packet capture, SIP.US for SIP trunking, etc.
- Raspberry Pis could not initially access Eduroam without a large amount of additional setup

```
pi@raspberrypi:~ $ ls /etc/asterisk/
asterisk.conf      extconfig.conf     indications.conf
cdr.conf           extensions.conf    logger.conf
cli.conf           func_odbc.conf     manager.conf
codecs.conf        http.conf          modules.conf
confbridge.conf    iaxprov.conf       musiconhold.conf

pjsip.conf         res_parking.conf        sip.conf
queuerules.conf    res_snmp.conf           sip_notify.conf
queues.conf        res_stun_monitor.conf   udptl.conf
res_fax.conf       rtp.conf                users.conf
res_odbc.conf      say.conf                voicemail.conf
```

Obstacles

- Raspberry Pis could not initially access “**eduroam**” without a large amount of additional setup

```
#!/bin/bash
ifconfig wlan0 down
ifconfig eth0 down
killall wpa_supplicant
echo "

allow-hotplug wlan0
iface wlan0 inet manual
        wpa-conf /etc/wpa_supplicant/wpa_supplicant.conf

iface wlan0 inet dhcp

" >> /etc/network/interfaces
read -p 'Username(email-address): ' username
read -sp 'Password: ' password
echo "

network={
        ssid=\"eduroam\"
        key_mgmt=WPA-EAP
        eap=PEAP
        identity=\"$username\"
        password=\"$password\"
        phase2=\"auth=MSCHAPV2\"
}

" >> /etc/wpa_supplicant/wpa_supplicant.conf

```

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Analytical Method

- Primary metrics to be analyzed are loss and jitter.
  - The implementation of IP-PBX used, Asterisk PBX, provides a command to view **jitter** and **loss** for packets involved in telephony.
  - Wireshark was also used to capture SIP and RTP/RTCP packets involved in telephony.

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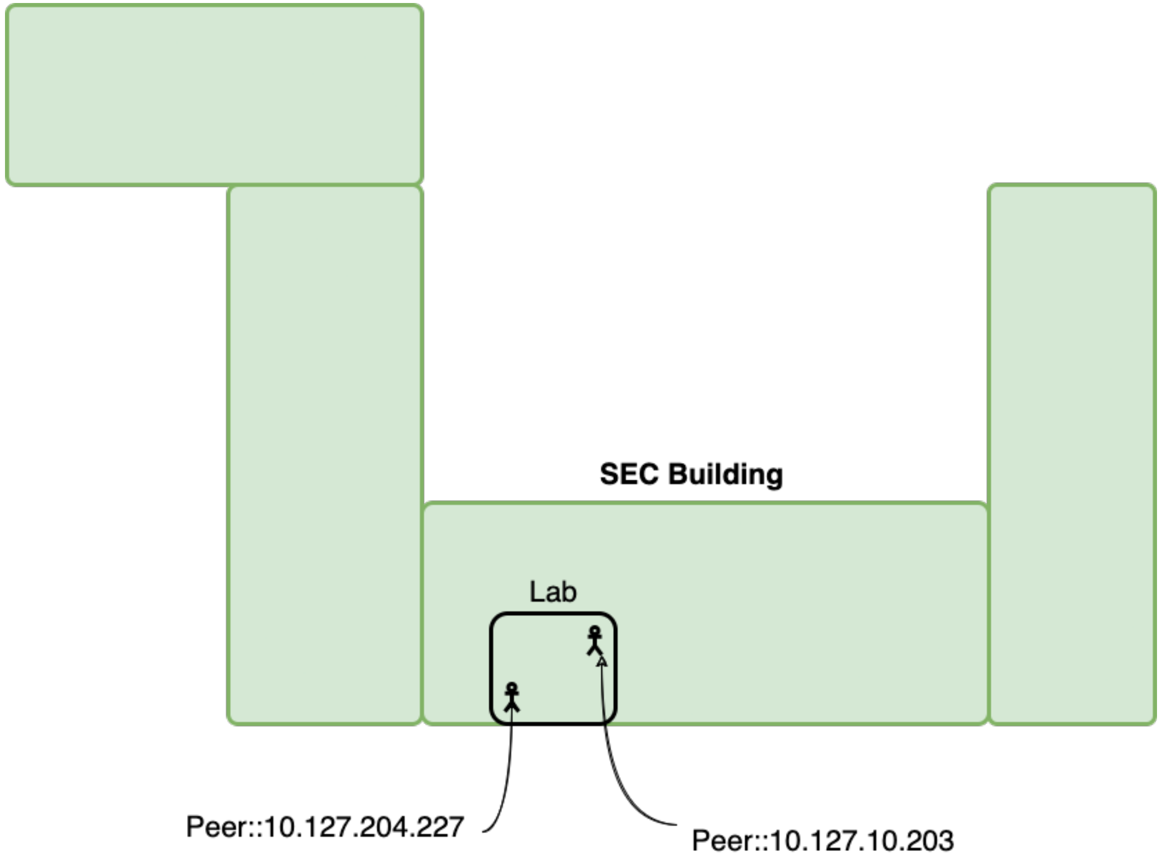
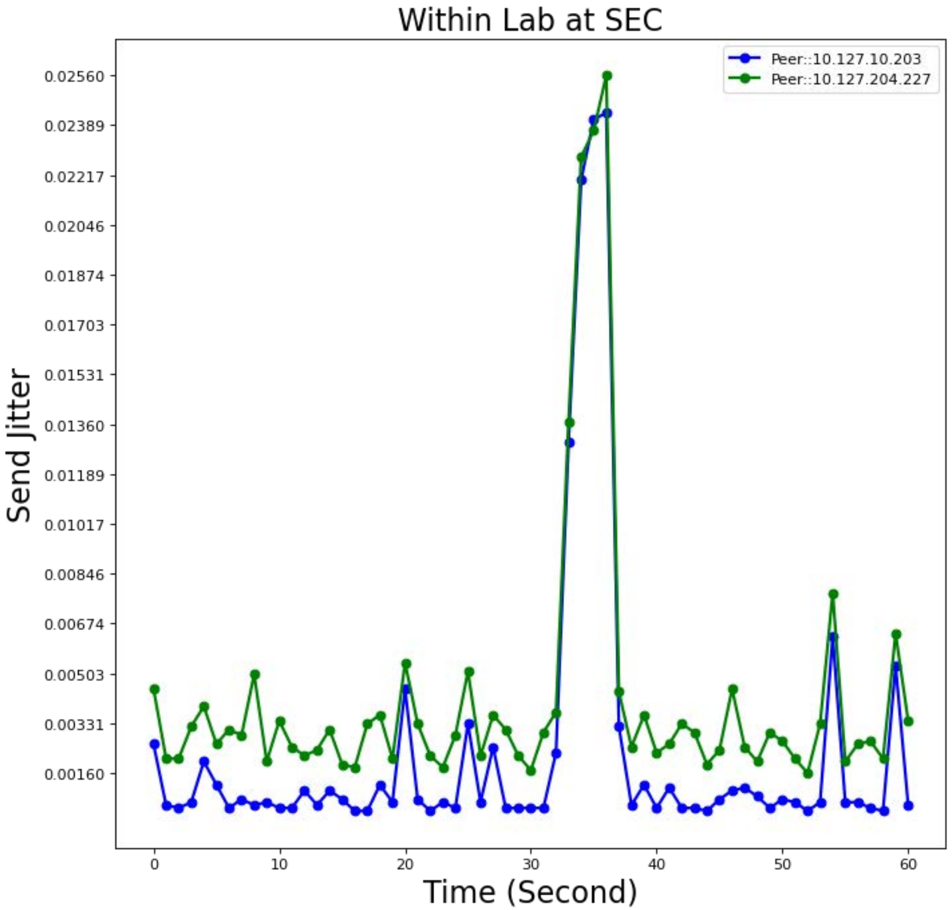
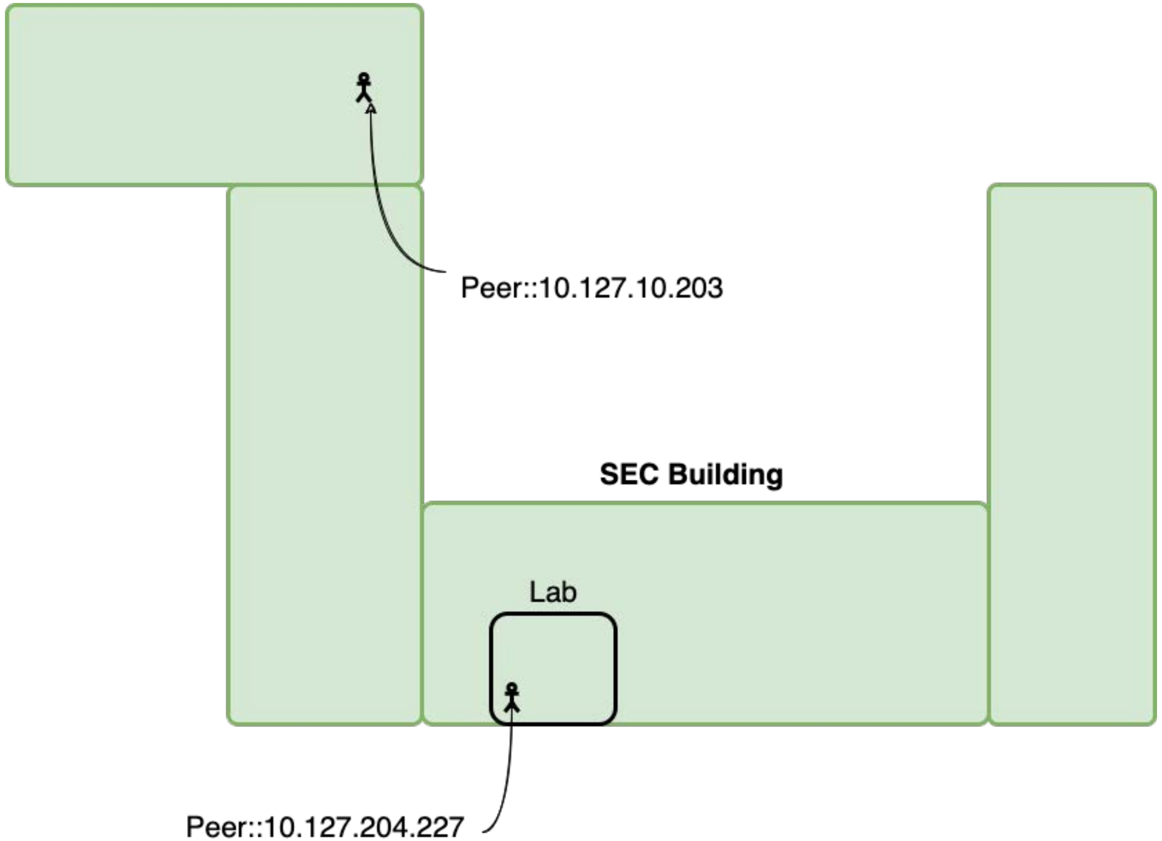
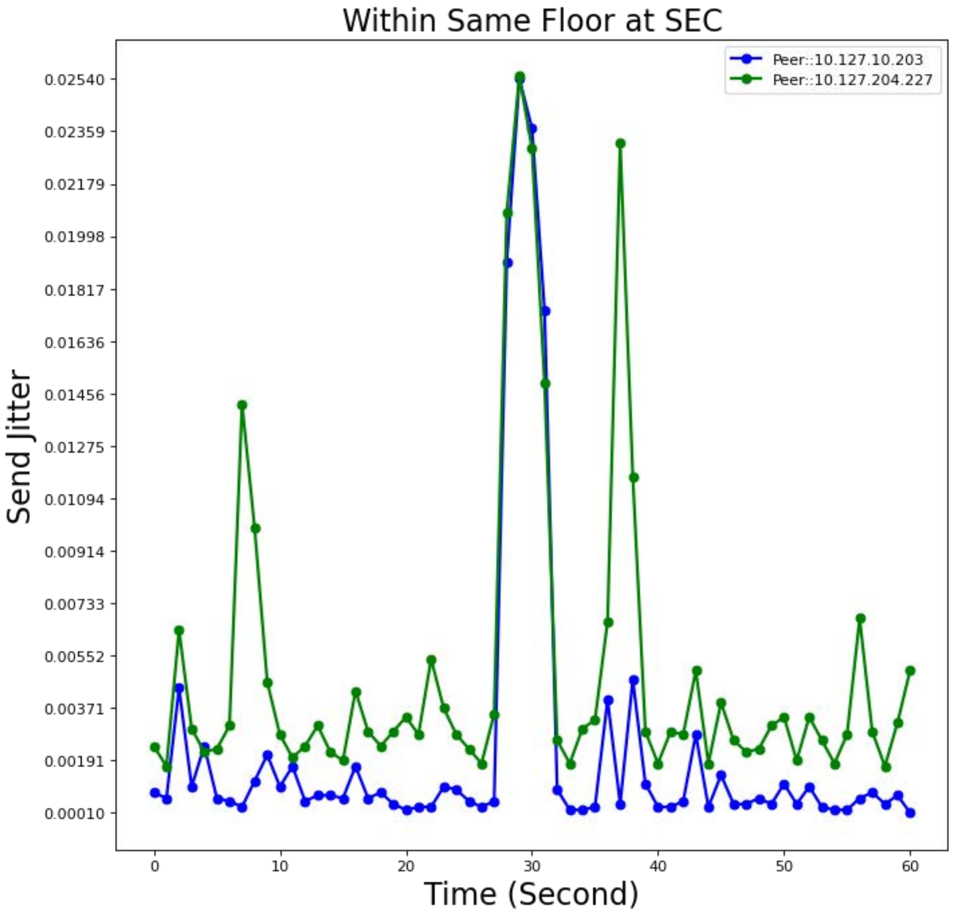
Experiments: Scenario 1



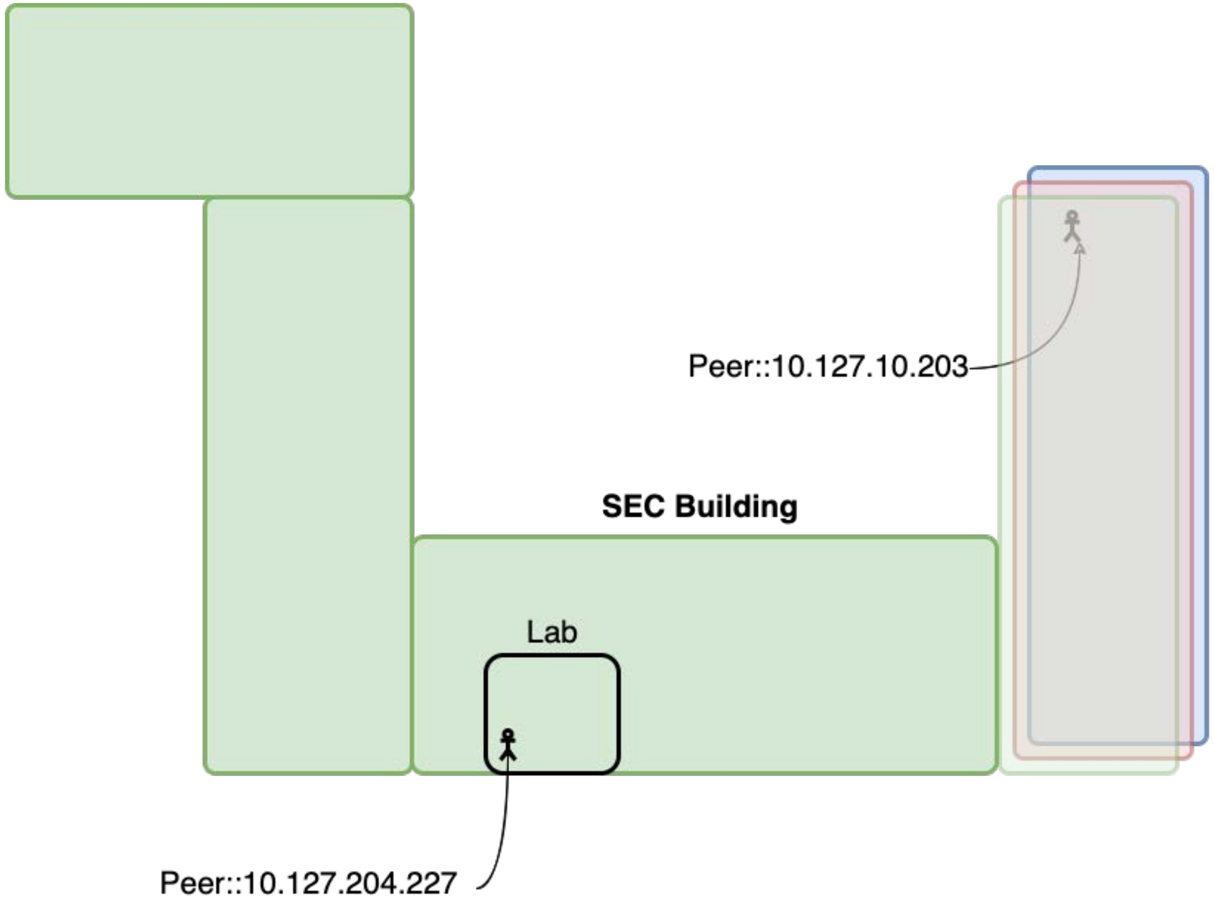


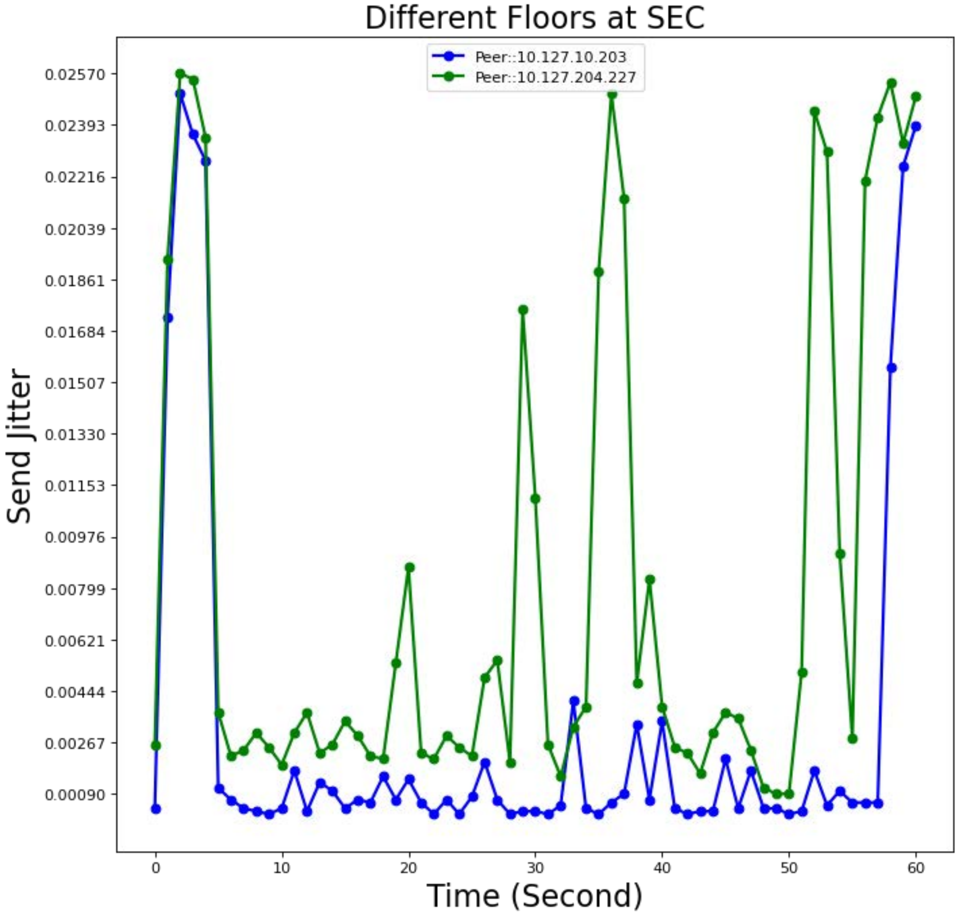
Experiments: Scenario 2



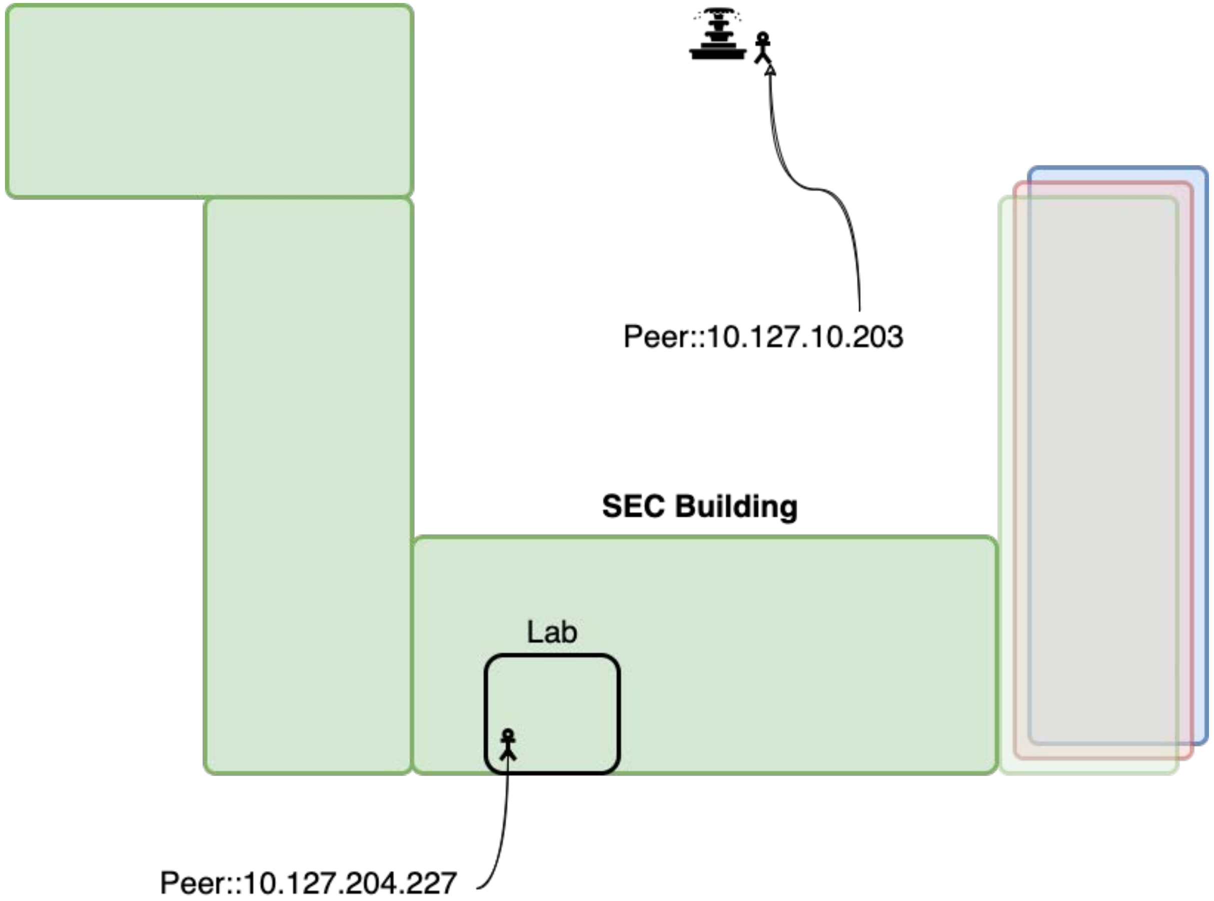


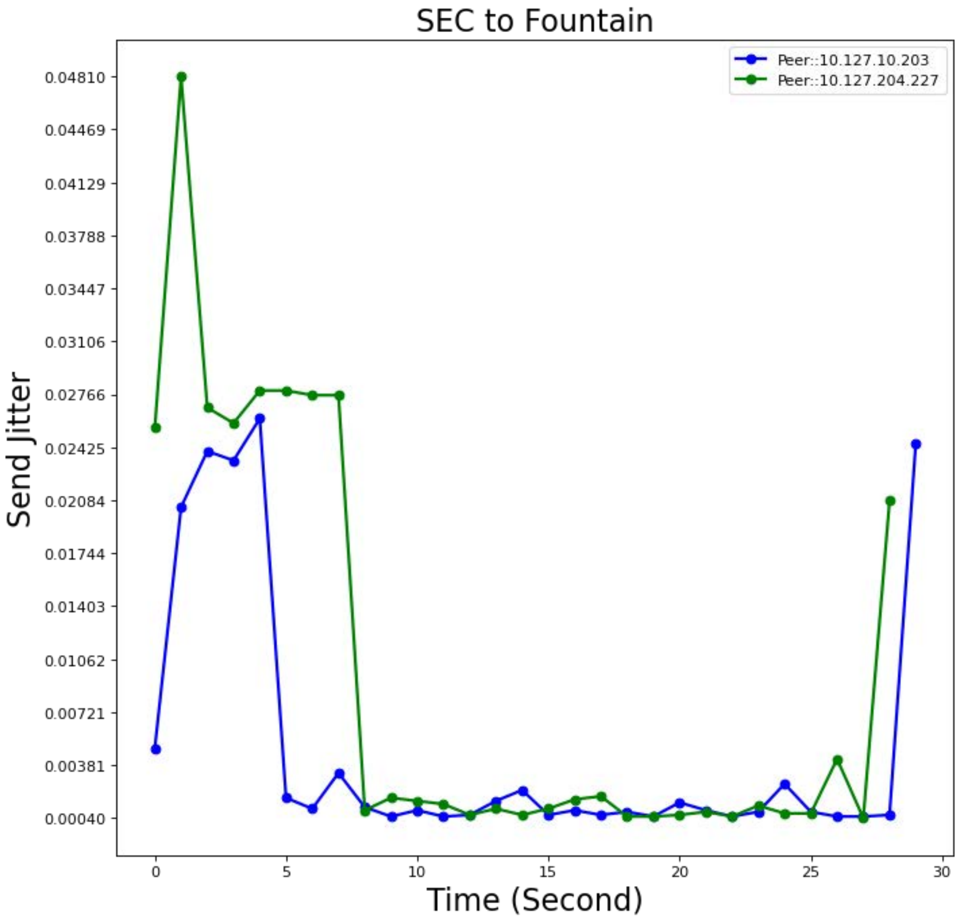
Experiments: Scenario 3



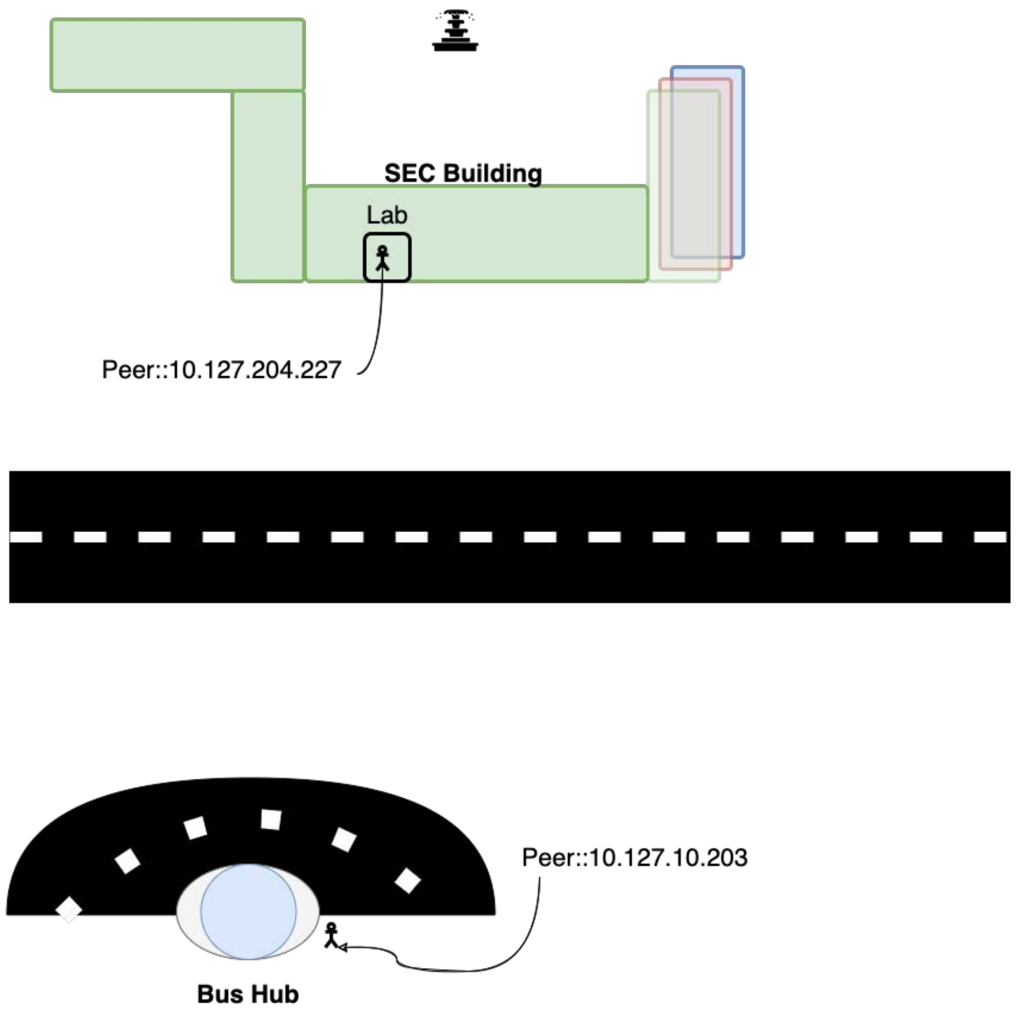


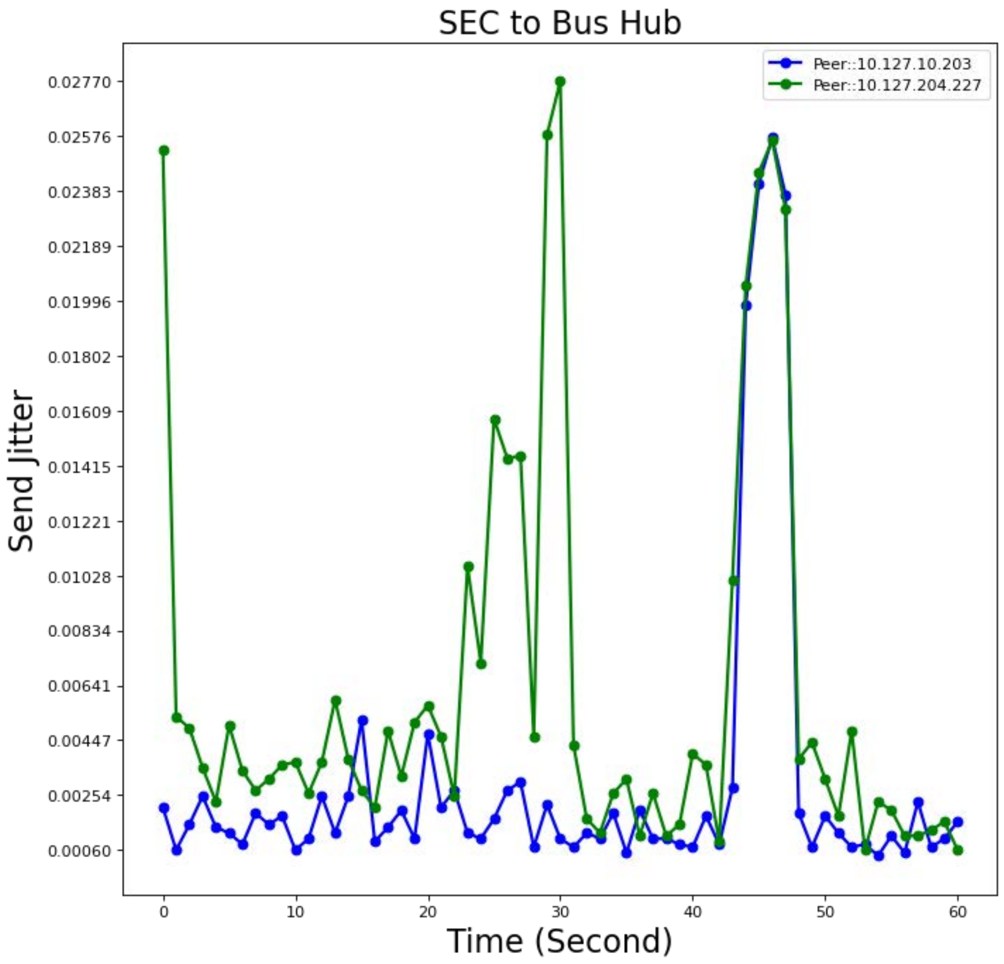
Experiments: Scenario 4





Experiments: Scenario 5





Expectation at Final Submission

- Currently, the experiment to make calls at different locations has been completed, and the results of which have been recorded.
- The experiment to test how well calls function while the Raspberry Pi is emulating different network conditions is to be completed soon.
- An analysis will be performed on data from both experiments, examining the loss and jitter under different conditions in the first experiment and the quality of telephony under the different conditions of the second experiment.

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Future Direction

- Add Interconnection between IP-PBX devices
  - Connect different Raspberry Pis running Asterisk PBX or some other implementation so that calls made to a phone registered on one can reach calls registered on the other
- Add Scalability
  - Make a group of Raspberry Pis, all of which run the same IP-PBX software, such that the workload of connecting phones and controlling calls is spread over them horizontally
- Test under More Conditions
  - How many callers can a Pi handle?
  - Effectiveness of Pi in different networks?
  - Making a call from an outside network?

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Conclusion

- VoIP is a cheap alternative to traditional telephone services currently.
- Raspberry Pis are cheap in terms of cost and energy usage, while still being powerful enough to easily handle IP-PBX software.
- When making VoIP calls, distance from the machine running the IP-PBX can matter.
  - Can cause more jitter and loss.
  - Disconnections can happen.
- Still need to test the Raspberry Pi with network emulation.

Special Thank You!

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Any Questions?

Experiments

- To test the Raspberry Pi, the machine ran an open source implementation of IP-PBX called Asterisk PBX. While Asterisk PBX was running, two phones were registered on the software using Zoiper, a VoIP implementation available on smartphones.
- There were two primary methods for testing the functionality of the Raspberry Pi running the IP-PBX software.
  - Make calls at a variety of locations of different distance from the Raspberry Pi
    - In the same room
    - On the same floor
    - In the same building
    - Outside same building
    - In a different building
  - Make calls while the Raspberry Pi is emulating stressful network conditions using the tc […] netem set of commands.
    - Add a delay to outgoing packets.
    - Add jitter to outgoing packets.
    - Add corruption to outgoing packets.
    - Add chance of loss to outgoing packets.