SIMULATION OF VoIP IN NS2 SIMULATOR

Report submitted to the SASTRA Deemed to be

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CSE302:COMPUTERNETWORKS

Submitted by

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EXAMINER1 EXAMINER2

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Abstract

Voice over Internet Protocol (VoIP) is one of the most important technologies in the World of communication. VoIP is simply a way to make phone calls through the internet. VoIP transmits packet via packet-switched network in which voice packets may take the most efficient path. Basically, VoIP system can be configured in these connection modes respectively; PC to PC, Telephony to Telephony and PC to Telephony. In this project, we are going to simulate VoIP over SCTP and VoIP over UDP using NS2 simulator.

Keywords

Transport layer, VoIP, Soft phone, Hard phone, Telephony, SCTP, UDP, NS2.

Introduction to VoIP :-

Voice over Internet Protocol (VoIP) is one of the most important technologies in the World of communication. VoIP is simply a way to make phone calls through the internet. VoIP transmits packet via packet-switched network in which voice packets may take the most efficient path. On the other hand, the traditional public switched telephone network (PSTN) is a circuit-switched network which requires a dedicated line for telecommunications activity. Furthermore, Internet was initially used for transmit data traffic and it is performing this task really well. However, Internet is best-effort network and therefore it is not sufficient enough for the transmission of real-time traffic such as VoIP.

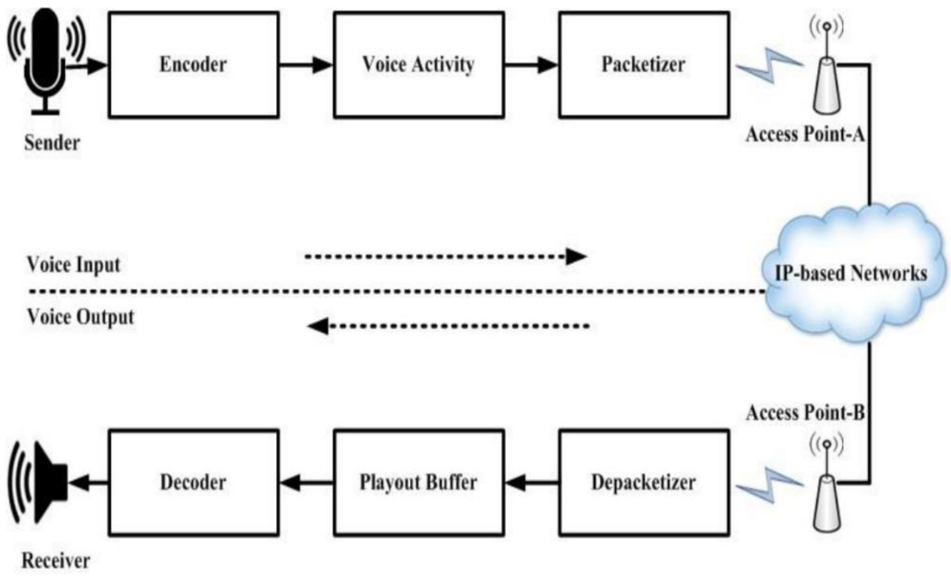
In addition, there are about 1 billion fixed telephone lines and 2 billion cell phones in the World that use PSTN systems. In the near future, they will move to networks that are based on open protocols known as VoIP . That can be seen from the increasing number of VoIP users. For instance there are more than eighty million subscribers of Skype; a very popular VoIP commercial application. VoIP has gained popularity due to the more advantages it can offer than PSTN systems especially that voice is transmitted in digital form which enables VoIP to provide more features. However, VoIP still suffer few drawbacks which user should consider when deploying VoIP system.

Basically, VoIP system can be configured in these connection modes respectively; PC to PC, Telephony to Telephony and PC to Telephony . VoIP consists of three essential components: CODEC (Coder/Decoder), packetizer and playout buffer.

At the sender side, an adequate sample of analogue voice signals are converted to digital signals, compressed and then encoded into a predetermined format using voice codec.

There are various voice codecs developed and standardized by International Telecommunication Union-Telecommunication (ITU-T) such as G.711, G.729, etc.

Next, packetization process is fragment encoded voice into equal size of packets. Furthermore, in each packet, some protocol headers from different layers are attached to the encoded voice. Protocols headers added to voice packets are of Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), and Internet Protocol (IP) as well as data link layer header. In addition, RTP and Real- Time Control Protocol (RTCP) were designed at the application layer to support real-time applications. Although TCP transport protocol is commonly used in the internet, UDP protocol is preferred in VoIP and other delay sensitive real-time applications. TCP protocol is suitable for less delay-sensitive data packets and not for delay-sensitive packets due to the acknowledgement (ACK) scheme that TCP applies. This scheme introduces delay as receiver has to notify the sender for each received packet by sending ACK.



Basic VoIP Components

For this project we are using NS2 to implement our VoIP network. We will be setting up several nodes on either side of two routers. We are going to be using an exponential traffic source to re-create a typical voice conversation over VoIP. Different protocols will be used to send voice information between terminals, starting with User Datagram Protocol (UDP), Transmission Control Protocol (TCP), and Realtime Transport Protocol (RTP). We are going to be measuring packet loss, throughput, end-to-end delay, and if the protocol permits, jitter. We will be plotting our results and comparing them to our theory based predictions.

Transport layer:

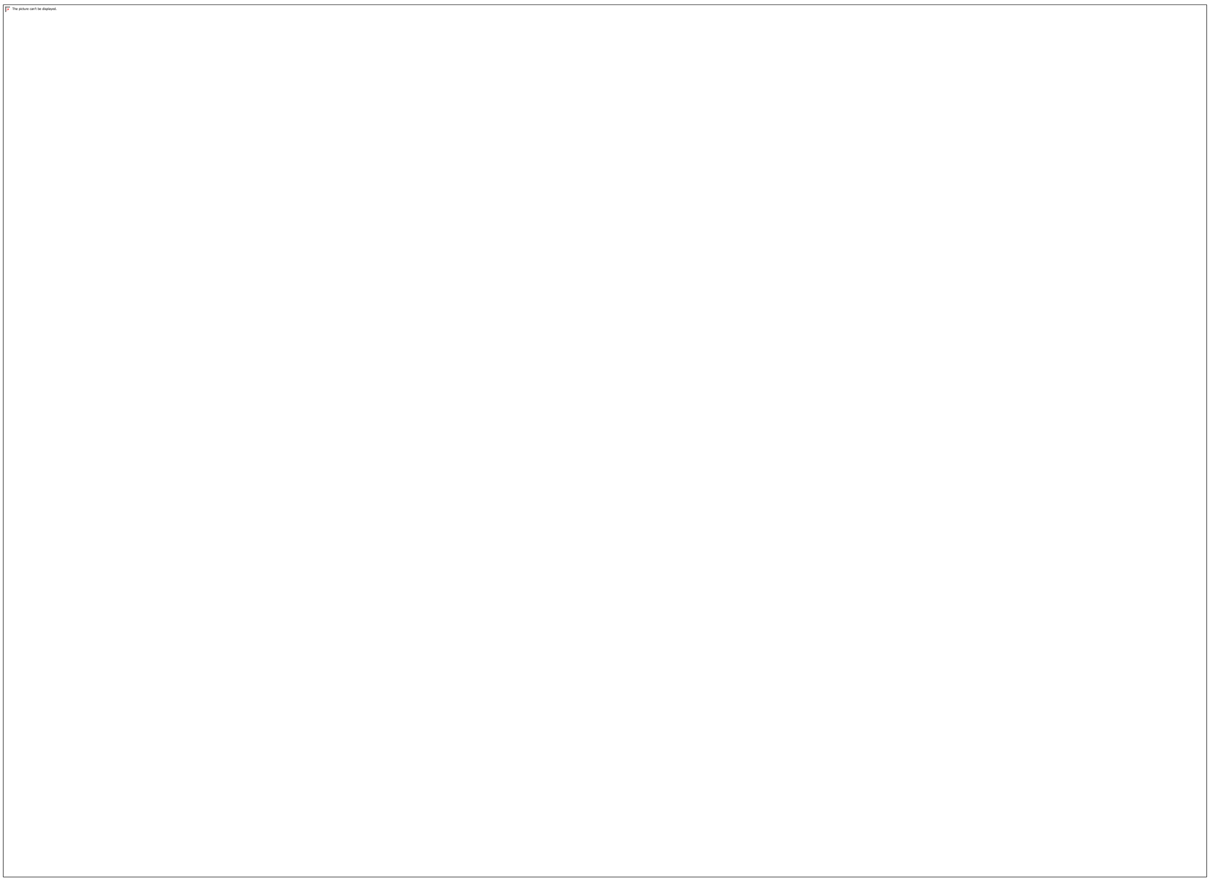
- The transport layer is a 4th layer from the top.
- The main role of the transport layer is to provide the communication services directly to the application processes running on different hosts.

Services provided by the transport layer:

The services provided by the transport layer are similar to those of the data link layer. The data link layer provides the services within a single network while the transport layer provides the services across an internetwork made up of many networks. The data link layer controls the physical layer while the transport layer controls all the lower layers.

Functions of Transport Layer:

1. **Service Point Addressing:** Transport Layer header includes service point address which is port address. This layer gets the message to the correct process on the computer unlike Network Layer, which gets each packet to the correct computer.
2. **Segmentation and Reassembling:** A message is divided into segments; each segment contains sequence number, which enables this layer in reassembling the message. Message is reassembled correctly upon arrival at the destination and replaces packets which were lost in transmission.
3. **Connection Control:** It includes 2 types:
    - Connectionless Transport Layer : Each segment is considered as an independent packet and delivered to the transport layer at the destination machine.
    - Connection Oriented Transport Layer : Before delivering packets, connection is made with transport layer at the destination machine.
4. **Flow Control:** In this layer, flow control is performed end to end.
5. **Error Control:** Error Control is performed end to end in this layer to ensure that the complete message arrives at the receiving transport layer without any error. Error Correction is done through retransmission.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Merits and Demerits of VoIP:-

VoIP: Advantages

Being VoIP providers ourselves at Nextiva, it would be fair to say we know a thing or two about VoIP. Here's a detailed guide into its pros and cons:

- Lower costs
- Increased accessibility
- Complete portability
- Higher scalability
- Advanced features for small and large teams
- Clearer voice quality
- Supports multitasking
- More flexibility with softphones

1) Lower Costs

The bottom line is vital for every business, large or small. So, you have to consider every cost-saving opportunity. One way companies can realize significant cost savings is by adopting a VoIP phone system.

Consider this:

On average, a landline phone system (POTS) costs businesses $50 per line each month. This rate comprises local (and sometimes domestic) calls only. VoIP plans, in contrast, are available for less than $20 per line.

Wait, what?

That's right. Going by these figures means, VoIP can lower your phone bill by more than half of what it is right now.

It's important to note that a shift to VoIP is not a guarantee that your phone expenses will plummet. Businesses differ, and so do their needs.

But what you can be sure about is that switching to VoIP will bring about considerable cost savings. Cost savings in VoIP come in two ways: direct and indirect.

Direct Cost Savings

When it comes to traditional phone service, a business incurs massive initial costs. Especially in the name of business phones and PBX hardware.

a) PBX Costs

A PBX (private branch exchange) is an on-premise physical piece of hardware. It connects many landline phones in an office and can cost a huge sum of money. We are talking tens of thousands of dollars — an amount you can amortize over several years.

You may argue that analog phones cost about the same as IP phones. The exact price will differ based on the desired features.

But, onsite PBX installations are a costly capital undertaking. It can go for anywhere between $500 and $2,000 per user. So even a small business with a handful of employees needs to invest in physical hardware.

VoIP networks do away with this need for extra hardware since a broadband connection powers the service.

To ease the transition in their phone system upgrades, such organizations could use a Session Initiated Protocol, known as a SIP Trunk. A SIP Trunk acts as a digital pathway for your voice services while maintaining existing phone hardware in your office. The key benefits of SIP tracking include lower costs, easier to manage, and you can activate service instantly.

Technology leaders use a SIP trunking provider for adding new voice capabilities to an existing phone system.

b) Copper Wiring Charges

Broadband connections also do away with the extra wiring because VoIP networks allow both voice and data on the same channel. In IT and telecom circles, the correct word for this is full-duplex. It's the ability to send and receive voice and data concurrently. Most VoIP desk phones need only one Ethernet to be connected to it.

Power over Ethernet (PoE) enables offices to be more modular with their office staff. Additionally, those offices won't need to make changes to the building's electrical wiring.

Going completely wireless? Professional VoIP service is also available as an app on your computer or smartphone. VoIP's flexibility is a big win for entrepreneurs and enterprises alike.

c) Calling Expenses

Direct costs also come in the form of the cost of calling. VoIP calls are cheaper compared to the Public Switched Telephone Network (PSTN) or the traditional circuit-switched telephone network by a stretch.

A large part of this has to do with the drastic fall in data carriage costs. Initially, data was priced out of the reach of most small businesses.

Even for large organizations, users had to contend with capping on enterprise internet bandwidth and broadband. Today, however, internet speeds have improved while data costs have correspondingly taken a nosedive.

Statistics reveal that small businesses using VoIP can reduce their company’s phone bill by up to 60%. They can also save up to 90% on international calls.

That’s a substantial number we are talking about in any given year.

Traditional phone service costs approximately twice as much as compared to VoIP. VoIP offers a significant benefit for businesses that operate globally.

d) Recurring Expenses

A VoIP service also enables businesses to cut other ongoing expenses such as taxes, repair and maintenance fees. VoIP providers usually roll these costs into subscription plans which, like in the case of Nextiva, can cost as little as $5 per user per month.

All these costs, combined, make VoIP service an appealing proposition for growing startups, small and medium-sized enterprises.

Indirect Cost Savings

Indirect savings are more difficult to quantify, but that doesn't make them any less critical for your business. Below are some of the most common areas where organizations save money long-term.

a) Savings with Remote Work

Switching to VoIP lets employees stay connected to the corporate phone system while working remotely. This is thanks to the long list of VoIP phone features like call waiting, auto-attendant, instant video calling, conference calling, and others not provided by traditional phones.

Studies show that this not only can increase employee productivity, but it can also cut down on utilities and office space.

A typical business can save $11,000 per person per year by merely letting them work from home 50% of the time, according to a recent analysis by Global Workplace Analytics.

b) Add-On Features at No Extra Cost

You probably might be quick to point out that even traditional PBX supports remote working through functionalities like call transfer, group ringing, call queuing, and so on.

In essence, these features are not inherent in a standard PBX system. Rather, they are add-on features that you have to pay for separately.

By comparison, VoIP phone include many of these features at no additional cost. No need to pay extra for whatever feature you think could be useful for your business.

c) Repurposed Manpower

If your business relies on a secretary to handle phone calls and take messages, the auto attendant feature lets you repurpose that role at no additional cost. (An in-house secretary hired on a full-time basis pockets about $45K a year, which is not cheap.)

Of course, a secretary does make sense for companies taking a large number of walk-ins or large corporations with sizeable budgets.

However, smaller businesses may find it difficult to justify this kind of salary. But with the auto attendant feature a click away, you’ve just waived this cost.

2) Increased Accessibility

Cost efficiency aside, accessibility is one of the biggest benefits of VoIP for business. One distinct advantage cloud-based VoIP service offers is the ability to make calls from anywhere.

If you have a decent data connection, you can make and receive calls for your business. And when you're unable to answer the call, you can direct calls to another person or get voicemails emailed to you. A noted benefit of VoIP is the ability to take your business phone with you with nothing more than a softphone app.

In an increasingly mobile workforce, remote accessibility allows your business to be flexible. Mobile employees can stay productive regardless of their location.

What’s more: VoIP adapts based on how your employees work. Employees don’t need to be physically present at the office. They can work on their smartphones and tablets from anywhere.

3) Complete Portability

A VoIP number, also known as a virtual number, is completely portable. This means you can use the same number wherever you go.

For people who travel a lot, this should be more than welcome news. Better yet, in the event your business changes address, you can retain the same VoIP number.

4) Higher Scalability

Scalability is another of the many VoIP advantages that make it an attractive proposition for growing businesses. While this is an often-talked-about aspect of VoIP, what does it mean exactly?

Given the option, every business owner would prefer a phone system that grows in step with their business.

A VoIP solution does away with having to purchase expensive hardware or dedicated line as you grow. Think of all the possible scenarios here like you're:

- Prepping for a spike in demand during the holidays
- Opening a new branch office

No matter what the scenario, toggle your preferences instantly without having to purchase additional lines or dedicated hardware.

6) Clearer Voice Quality

When VoIP service first rolled around, one of its most significant disadvantages was its weak call quality. Calls would drop for no reason, the voice quality itself was bogus, and latency was the order of the day.

Today, as long as you have a fast and stable Internet connection, voice quality should not be an issue. VoIP calls tend to be crisp and clear, with no latency issues, lag, or call dropouts. We've all been on a miserable voice and video conference before.

The key to VoIP call quality is a robust connection with good bandwidth. Without this, it can be a nightmare, especially if you often find your office making concurrent calls. There's always someone who calls in with a cell phone, and everyone else suffers through echos, delays, and background noise.

VoIP phones end those interruptions so you can focus on the meeting's agenda. Innovations such as noise-canceling microphones and advanced audio compression enable VoIP phones to achieve superior sound quality.

7) Supports Multitasking

Along with traditional phone calls, VoIP allows you to send documents, images, and videos all while simultaneously engaging in a conversation. So you can seamlessly hold more integrated meetings with clients or staff from other corners of the globe.

8) More Flexibility with Softphones

Despite the name, softphones are not hardware devices. Instead, they are programs installed on a computer or other smart devices like a tablet or smartphone.

The upside to having a softphone for your business communications is manifold:

- Frees up desk space
- Cuts additional equipment costs
- Allows for even greater portability
- Enables the constantly-connected workforce

More than that, softphones allow you to be flexible. They give you access to features that support your remote work style.

9) Increased Security

Most people don't care to spend more time than it's worth to think about the security of their phone system. Phone system security is a big deal, especially for businesses. Demand for personally identifiable information (PII) has never been higher.

A typical entry point is to trick staff through fraudulent phone calls, typically known as social engineering.

VoIP can mitigate such security threats by leveraging the advancements made in IP technology including encryption and improved identity management. Hosted VoIP providers work around the clock to protect their networks, so you don't have to.

Securing your VoIP system means you should work with a trustworthy VoIP provider that undergoes independent security audits, ensure staff practices healthy password habits, and configure automated alerts for questionable calling behavior.

Additionally, it is always advised to complete operating system updates consistently to ensure your business isn't at risk through newer vulnerabilities.

VoIP: Disadvantages

Everything that has an advantage has its disadvantages. VoIP telephony is not exempt from this rule.

Here are the downsides associated with the VoIP service you need to be aware of:

- Reliable Internet Connection Required
- Latency and Jitter
- No location tracking for emergency calls
- Not all VoIP providers offer secure transmissions
- Determining the actual location of people during emergencies can be difficult when using VoIP lines.
- No internet means no VoIP phone service

1) Reliable Internet Connection Required

For starters, your VoIP service is only as good as your internet connection. If your network bandwidth is low, the service is bound to suffer.

VoIP doesn't use as much bandwidth as you might expect. It's essential that VoIP devices receive low latency on your network. Each device should have at least 100 kbps upload speed available. A good connection has less than 70ms ping and jitter, which measures the latency and stability of your internet connection.

The bandwidth your business needs will depend on the number of concurrent calls you plan on making. The best way to determine this is to run a bandwidth test on your current network.

2) Latency and Jitter

Aside from speed, there are other connection issues any internet-based technology can face: latency and jitter.

When communicating online each message (whether it be email, video, or audio) is broken into bits of data called "data packets." These packets are then reassembled at their intended destination to create the original message.

Latency and jitter are when these data packets either hit delays in transmission or get improperly re-assembled. These issues might not even be with your network; major internet backbones modify data routes to deliver traffic reliably, fastest path to a destination. These changes happen automatically with no involvement on your part.

Why latency and jitter occur

- **Poor Internet connection** — VoIP requires more bandwidth than regular web surfing. So, if you find your Internet speed wanting, it might be a good time to have an honest conversation with your ISP.
- **Inadequate router** – For VoIP service to run smoothly, it needs a specialized VoIP router. This is a router configured for packet prioritization so that it affords higher priority to voice traffic over data.
- **Insufficient cables** – Ethernet cables come in a range of categories or power levels. For VoIP, it's best to use a Cat-5e Ethernet cable or higher. Lower cables may not be able to operate at high enough speeds.

How to fix latency and jitter issues

- **Enable jitter buffering** — This is easy to set up and comes pre-enabled with many of Nextiva's devices.
- **Opt for high-speed Internet** — Contact your internet provider about available bandwidth options.

- **Upgrade ethernet cables** — Use a CAT-5e or CAT-6 Ethernet cable on all VoIP devices.

3) Limited Location Tracking for Emergency Calls

Location tracking is the final con of VoIP. Because of VoIP's portability and accessibility, it's difficult for third parties to pinpoint where a call originates.

The calls come from an IP address with no GPS data or cell tower information to track. While 99% of callers don't need this information, this does create an issue for emergency services like 911. You'll need to communicate where you are in an emergency.

4) Not all VoIP providers offer secure transmissions.

The internet is a public network, so there is a possibility of data being intercepted during transmission. This includes voice data during VoIP calls. The danger of private voice calls being breached is very real and should not be ignored. Fortunately, when you take the time to discover what VoIP service providers are and how they help businesses, you will also find that the top companies implement security measures to protect your voice data during VoIP calls. This includes high-level encryption during transmission from end to end.

5) Determining the actual location of people during emergencies can be difficult when using VoIP lines.

One of the advantages of VoIP is that you can purchase a local number from any state, with any area code, even without a physical address. That gives you the chance to establish a virtual presence anywhere. The problem is that during emergencies, responders typically use your phone number to identify your location. With your phone number and actual physical location not necessarily matching, emergency responders will have trouble finding you when you need them to. Fortunately, this is where an e911 emergency address comes in. This allows businesses and users to implement changes to the physical address connected to a digital VoIP line. This same address will be used by 911 to trace the origin of a VoIP call within the US. The best VoIP providers give their users access to this feature. As you can see, the advantages of VoIP far outweigh the disadvantages. The disadvantages themselves are not really deal breakers, as there are workarounds to the problems ensuring they do not become big issues. So if you're asking whether businesses should shift to VoIP, the answer is a big YES.

6) No internet means no VoIP phone service

The main disadvantage of VoIP compared to traditional lines is that it is totally dependent on the strength of your broadband connection. No internet equals no VoIP phone service. It does not end there either. Poor internet connection can also affect call quality and lead to problems like jitter and latency. Thankfully, there are quite a few workarounds you can apply here.

• First, get a reliable internet provider. A lot of operational tasks right now use the internet anyway, so it is an investment you should not scrimp on in the first place. • Second, have a dedicated internet network solely for VoIP (if possible) so it does not compete for bandwidth with other online-based business tasks. • Last, you can also use QoS routers so VoIP transmissions are prioritized over other internet-related activities like downloads and streaming.

Using VoIP Services at Home

You need a VoIP phone to connect to a VoIP service provider. You can do this through any of the following three ways:

- Connect a dedicated VoIP to your wired ethernet of WiFi
- Use your conventional analog telephone but with an analog telephone adapter.
- Install a softphone application on your computer with a microphone and speaker, or headset.

VoIP Services for Businesses

Businesses tend to migrate from the traditional copper-wire telephone systems to VoIP systems for two reasons:

- Bandwidth efficiencies
- Reduced costs

Since VoIP allows both voice and data to run over the same network and because it works with your existing hardware, it’s an attractive alternative for businesses. Even if you were to extend your VoIP lines, it’s still more affordable as compared to Private Branch Exchange (PBX) lines.

Today, VoIP for businesses include the following and is called **Unified Communications**:

- Phone calls
- Faxes
- Voicemail
- Email
- Web conferences and more

Thus, VoIP can spur growth in both enterprises and SMBs, without much of a budget to work with.

What Are VoIP Codecs & How Do They Affect Call Sound Quality?

Thanks to Voice over Internet Protocol (VoIP), today's phone calls are crystal-clear and only need an internet connection. It's all possible because of VoIP codecs.

Read along as we discuss what a codec means, along with how you can select the right codec for your company's VoIP phone system.

Definition of a codec

Fundamentals of audio quality

How HD voice improves call quality

Three types of VoIP codecs

Organizing your codecs

What are VoIP codecs?

A VoIP codec is a technology that determines the audio quality, bandwidth, and compression of Voice over Internet Protocol (VoIP) phone calls. VoIP codecs use either proprietary or open-source algorithms.

The word codec is a portmanteau of two terms: Compression and Decompression.

Codecs are the reason why you can download a movie in minutes, not hours. Practical examples of codecs include image capture (JPEG), encryption software (AES), streaming media (H.264), and music and audio recording software (MP3). For instance, codecs determine the quality and bandwidth you need to watch videos on YouTube or Netflix.

In the case of a VoIP codec, it converts analog voice signals into digital packets or a compressed digital form for transmission and then back into an uncompressed audio signal. VoIP codecs determine the call quality and latency in a conversation since the call takes place through the internet.

You might encounter some VoIP problems since calls travel over the internet. If your VoIP provider has multiple data centers, reliability is a non-issue for a vast majority of phone calls.

The success of a codec depends on its compatibility with different devices.

VoIP codecs all share one purpose. That is, they compress data and move it quickly. Business VoIP applications ensure phone calls don't hog a ton of bandwidth and that calls are clear and crisp.

The only notable difference between the different VoIP codecs is the way they compress the audio. Compression is necessary for transmitting audio since it requires a lot of bandwidth, which is finite. Files and documents will take a lot of space in the absence of compression, proving to be troublesome for organizations.

Business owners should think about their VoIP codec protocols to ensure the compression is efficient and requires less bandwidth.

These precautions will help organizations maximize their capacity planning, reduce expenses, and ensure a safer long-term investment for the company.

cloud VoIP phone system can be particularly useful for remote teams. It helps employees stay connected by making on-the-go video conferencing, call recording, and long-distance calling possible.

Voice Codecs

A codec is an algorithm most of the time installed as a software on a server or embedded within a piece of hardware (ATA, IP Phone etc.), that is used to convert voice (in the case of VoIP) signals into digital data to be transmitted over the Internet or any network during a VoIP call.

The word codec comes from the composed words coder-decoder or compressor-decompressor.

Codecs normally achieve the following three tasks (very few do the last one): • Encoding - decoding

• Compression - decompression

• Encryption – Decryption

Common VoIP codecs

Codec	Bandwidth/kbps
G.711	64
G.722	48/56/64
G.723.1	5.3/6.3
G.726	16/24/32/40
G.729	8
GSM	13
iLBC	15
Speex	2.15 / 44
SILK	6 to 40

Table 1. Bandwidth/kbps of different codecs

The fundamentals of audio quality

A reliable VoIP system has become essential to millions of companies today. The foundation of excellent business communication is fidelity – it’s why audio quality over the phone is a big deal.

But before we discuss audio processing in more detail, you should be aware of the common terminology related to audio quality.

- **Sample rate** - Also known as sample frequency, it refers to the audio samples taken per second. Every individual sample will tell you the signal waveform's total amplitude value over a specific period. The higher the sample rate, the better the audio quality.
- **Bitrate** - The amount of data that is transferred into audio. Audio bitrates mean it captures more sound information per second. Generally, a higher bitrate indicates better sound quality.
- **Bandwidth** - While bandwidth is the speed that you receive or send data. Likewise, the transmission rate is the number of samples that are transmitted every second.

No matter how exceptional a sound you have, it will sound bad on the other end if you have a low bitrate. The same is true for sample rates.

In short, *bandwidth* is your bottleneck. VoIP codecs aim to conserve bandwidth while maintaining impressive sound quality.

How HD Voice improves call quality

You might have noticed how the sound differs when you’re on the phone instead of speaking to someone face-to-face. This difference is noticeable because the phone doesn’t pick up all of the frequencies that the human voice hits.

The range of human speech is between 80 to 14,000 Hz. The lower the frequency, the deeper the sound. A punchy beat in a pop song consists of lower frequencies. On the other end of the spectrum, vocalists are renowned for their 250 to 1500 Hz.

Phone conversations are different, though.

Phone audio typically involves two bands: narrowband and wideband. Narrowband covers audio frequencies ranging between 300 Hz and 3400 Hz. For audio between 50 Hz to 7000 Hz, those are considered wideband.

However, when you talk in wideband audio, also known as HD Voice, you'll be able to hear an expanded range of pitches, most closely resembling an in-person conversation.

This improvement is because wideband improves the sampled and transmitted audio spectrum, making the audio sound better.

Here's another example to consider. Automotive manufacturers engineer a vehicle's exhaust system for optimal acoustics. Luxury sports cars are stealthy because of the silencing effect of high frequencies canceling out lower frequencies.

VoIP codecs adopt a similar approach to minimize background noise for even better sounding phone conversations.

Three types of VoIP codecs

As there are plenty of codec choices, choosing a specific one can be tricky. Below, we've listed a few individual codecs to consider.

G.711

The International Telecommunication Union (ITU) introduced the G.711 codec back in 1972 for telephony use. This codec has two variants: μ-law and A-law. The United States and Japan use μ-law. Europe uses A-law.

This codec can squeeze 16-bit samples into 8 bits through logarithmic compression. As a result, the compression ratio becomes 1:2. The bitrate for both directions is 128 kbit/s (64 kbit/s for a single path), which is a lot.

While you get superior sound quality, the bandwidth requirement is relatively high. Plus, this codec doesn't support multiple phone calls as adequately as other codecs like the G.729.

You can use the G.711 codec for all kinds of VoIP applications as there aren't any licensing fees. There is also no digital compression, which is why it's considered the best VoIP codec to interface with the public switched telephone network (PSTN).

G.722 HD

G.722 is a high-definition codec, which means it's wideband. ITU approved this codec in 1988, and since the patent has expired now, it's free to use for everyone.

This codec helps improve speech quality without perceivable latency. HD voice has double the sample rate of G.711 at 16 bits. The transmission rate remains the same at 64 kbit/s.

G.729

If you're looking for a codec with low bandwidth requirements and acceptable audio quality, the G.729 is a good bet.

The codec encodes the audio in frames. Each frame is ten milliseconds long and contains 80 audio samples. The bitrate for one direction of this non-HD codec is 8kbit/s. Since the compression is higher, you're able to make more calls from your network at once.

That said, a few VoIP providers may not support the G.729 codec. Music and other non-verbal audio can sound choppy.

Organizing your codecs

Cloud VoIP providers determine which codecs are available for your hardware.

VoIP providers transmit the data packets, while IP phones need to compress and decompress the audio effectively. The caller and the called phones negotiate the proper codec whenever there is a call connection attempt. Both the caller and receiver phones have a prioritized list to agree on the right codec.

When it comes time to select the best codec for your phone system, opt for the one that works. Think about your team's real-world bandwidth capabilities and call volumes.

If you prefer better call quality, you should place the codec G.722 first and then G.711. But if lower bandwidth is your concern, set G.729 above G.711.

Since almost all VoIP phones and providers accept G.711, the codec G.722 is more limited. IT professionals prefer the G.722 codec for high-grade voice conversations without placing a significant burden on the local area network.

VoIP codecs deliver crystal-clear communication

VoIP systems enhance your productivity by assuring seamless communication between your team members and customers. Voice over IP codecs make it possible to speak clearly without the complexity of bulky telecom equipment.

Avoid the stress of trying to wrap your head around the technical details of VoIP codecs. When you select a cloud phone system leader like Nextiva, you leverage its expertise from the start.

HD voice calls

HD voice is an enhanced calling experience using a combination of wideband audio codecs and handsets optimized to reduce background noise. HD voice is only available on Voice over Internet Protocol (VoIP) phone systems.

With high-definition phone calls, you have more productive conversations due to the improved audio quality that VoIP provides.

Traditional phones don't support HD voice because calls travel over the Public Switched Telephone Network (PSTN) using narrowband codecs.

High-Definition Voice (HD voice) helps people hear each other better over the phone.

Popular benefits of HD voice include

Crystal-clear sound

Look forward to calling your next lead or customer by hearing them in a full range of their voice. With a headset, background noise is eliminated entirely.

Less repetition

Repeating yourself on business calls is expensive. Call quality influences customer satisfaction and a poor audio experience can cost you sales**.**

No additional setup

HD voice works perfectly without any extra setup. Calls work just as they usually would. Enjoy HD voice calls on many VoIP phones and apps.

Optimal bandwidth usage

HD voice uses a variety of codecs to upgrade the call to a higher quality automatically.

VoIP over different protocol

Voice over UDP

The UDP is one of important members of the Internet protocol suite. It uses a simple transmission without doing any error checks. Therefore, UDP is not suitable for applications which need error checking all the time. However, the advantage of UDP is less delay. Theoretically, packet loss must within certain range in order to deliver normal condition voice package.

Voice over SCTP

Similar to UDP, Stream Control Transmission Protocol (SCTP) is a transport layer protocol. It contains both feature from UDP and TCP. It uses minimal message-oriented Transport Layer protocol as UDP. Also, it is in-sequence transport of messages with congestion control like TCP. Therefore, it is reliable and fast.

Comparison between SCTP and UDP:

i. Message Orientation:

In SCTP, message boundaries are preserved. If an application sends a 100-byte message, the peer application will receive all 100 bytes in a single read: no more, no less. UDP provides a message-oriented service, but without SCTP's reliability.

ii. Un-Ordered Service:

In addition to ordered message service (and parallel ordered service discussed above), SCTP offers the reliable delivery of messages with no order constraints. UDP provides unordered service, but again without SCTP's reliability. Unordered reliable delivery will be useful for many applications, in particular disk over LAN services (iSCSI, RDMA, etc.) where the application already provides ordering.

iii. Stronger checksum:

SCTP uses a 32-bit end-to-end checksum proven to be mathematically stronger than the 16-bit ones-complement sum used by UDP. SCTP's better checksum provides stronger verification that a message passes end-to-end without bit errors going undetected.

About NS2:

NS2 is an open-source simulation tool that runs on Linux. It is a discreet event simulator targeted at networking research and provides substantial support for simulation of routing, multicast protocols and IP protocols, such as UDP, SCTP, TCP, RTP and SRM over wired and wireless (local and satellite) networks.

It has many advantages that make it a useful tool, such as support for multiple protocols and the capability of graphically detailing network traffic.

Additionally, NS2 supports several algorithms in routing and queuing. LAN routing and broadcasts are part of routing algorithms.

Queuing algorithms include fair queuing, deficit round-robin and FIFO.NS2 started as a variant of the REAL network simulator in 1989 REAL is a network simulator originally intended for studying the dynamic behaviour of flow and congestion control schemes in packet-switched data networks.

Currently NS2 development by VINT group is supported through Defence Advanced Research Projects Agency (DARPA) with SAMAN and through NSF with CONSER, both in collaboration with other researchers including ACIRI (see Resources).

NS2 is available on several platforms such as FreeBSD, Linux, SunOS and Solaris. NS2 also builds and runs under Windows. Simple scenarios should run on any reasonable machine; however, very large scenarios benefit from large amounts of memory.

Tcl –

Tcl is shortened form of Tool Command Language.

John Ousterhout of the University of California, Berkeley, designed it.

It is a combination of a scripting language and its own interpreter that gets embedded to the application, we develop with it.

Tcl was developed initially for Unix. It was Performance Analysis of Inelastic Traffic(VoIP) Department of Computer Science and Engineering, IUST Awantipora, J&K Page 11 then ported to Windows, DOS, OS/2, and Mac OSX.

Tcl is much similar to other unix shell languages like Bourne Shell (Sh), the C Shell (csh), the Korn Shell (sh), and Perl.

It aims at providing ability for programs to interact with other programs and also for acting as an embeddable interpreter. Even though, the original aim was to enable programs to interact, you can find full-fledged applications written in Tcl/Tk.

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Basic Working of VoIP :

VoIP stands for Voice over IP (Internet Protocol), a variety of methods for establishing two-way multi-media communications over the Internet or other IP-based packet switched networks. Although VoIP systems are capable of some unique functions (for example: video conferencing, instant messaging, and multicasting), this appendix concentrates on the ways in which VoIP can be used to replicate the voice conversation functionality of the public switched telephone network (PSTN). There are several competing approaches to implementing VoIP. Each makes use of a variety of protocols to handle signalling, data transfer, and other tasks. Data is moved between the two endpoints using a media protocol, the Real-time Transport Protocol (RTP). A codec (coder/decoder) is used to convert the sound of each caller's voice to digital data, then back to analog audio signals at the other end. Conversation ends and the call is torn down. Again, this involves the signalling protocols appropriate to the particular implementation of VoIP, along with any Gateway or Gatekeeper functions. The instructions governing the call-the call setup and call teardown-are handled separately from the transmission of the actual data content of the call, or the encoding and packetization of voice media. There are several protocols and methods for VoIP calls – the commonest standards are termed SIP and H.323 – but they all have some basic features in common. To the user phone calls are made and handled in the same way as they always have been except that VoIP phones often have more features available from menus and buttons than regular phones. When a call is dialed, the system takes the phone number, connects over the local network to whatever system is providing service. That system figures out if the call needs to go into the regular phone network and if so switches it to a gateway that connects the call over the regular phone network. If the call can be completed without going over the regular phone network (the number dialed is also a VoIP system) then

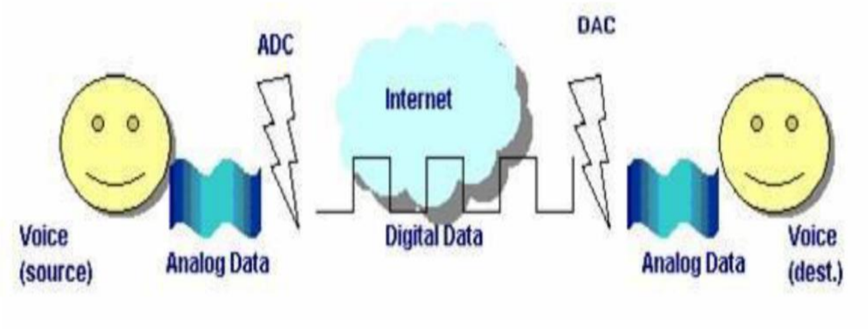


Figure 2.1. Basic Working of VoIP

the provider system will route the call directly, performing protocol translation (to a different kind of VoIP) if needed. When traveling on the network, VoIP calls are treated like any other network data – they are broken down into little pieces of digital information (packets) and sent by whatever route the network determines to be fastest. That means different pieces arrive and different times and out of order and then are reassembled back into the proper sequence at the destination. This is why the 100+ kbps transmission rate is needed – so that the signal can be sent and reassembled quickly enough so that human users on both ends don't notice any delay. It is also one of the weaknesses of VoIP – if the network goes down or has performance issues, so will your VoIP calls. Since the very early days of distance communication, signals were sent in analog form, in waves. Many years ago, the communication world discovered that sending a signal to a remote destination could have been done also in a digital fashion: before sending it we have to digitalize it with an ADC (analog to digital converter), transmit it, and at the end transform it again in analog format with DAC (digital to analog converter) to use it. VoIP works like that: digitalizing voice in data packets, sending them and reconverting them in voice at destination. Digital format can be better controlled: we can compress it, route it, convert it to a new and better format, and so on. We also saw that a digital signal is more noise-tolerant than its analog signal.

.

VoIP packet structure

A VoIP packet is composed of the IP header, followed by the UDP header, followed by RTP header, and finally followed by the payload.

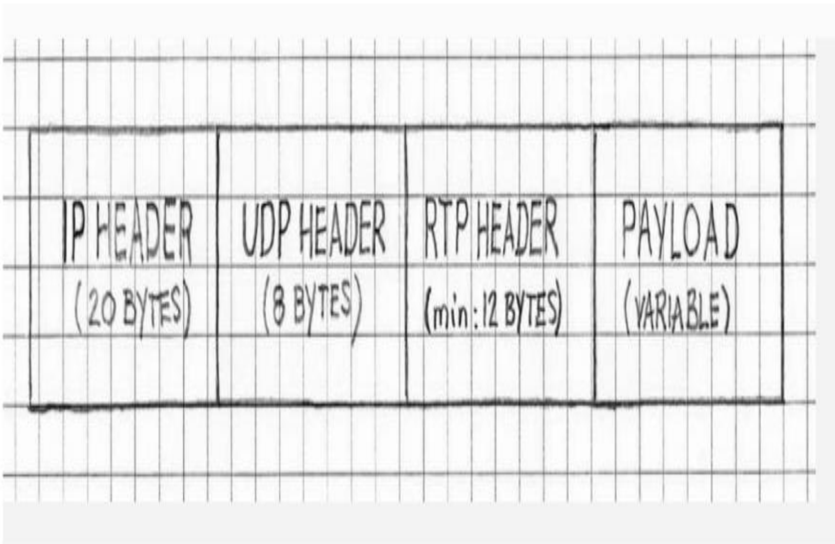
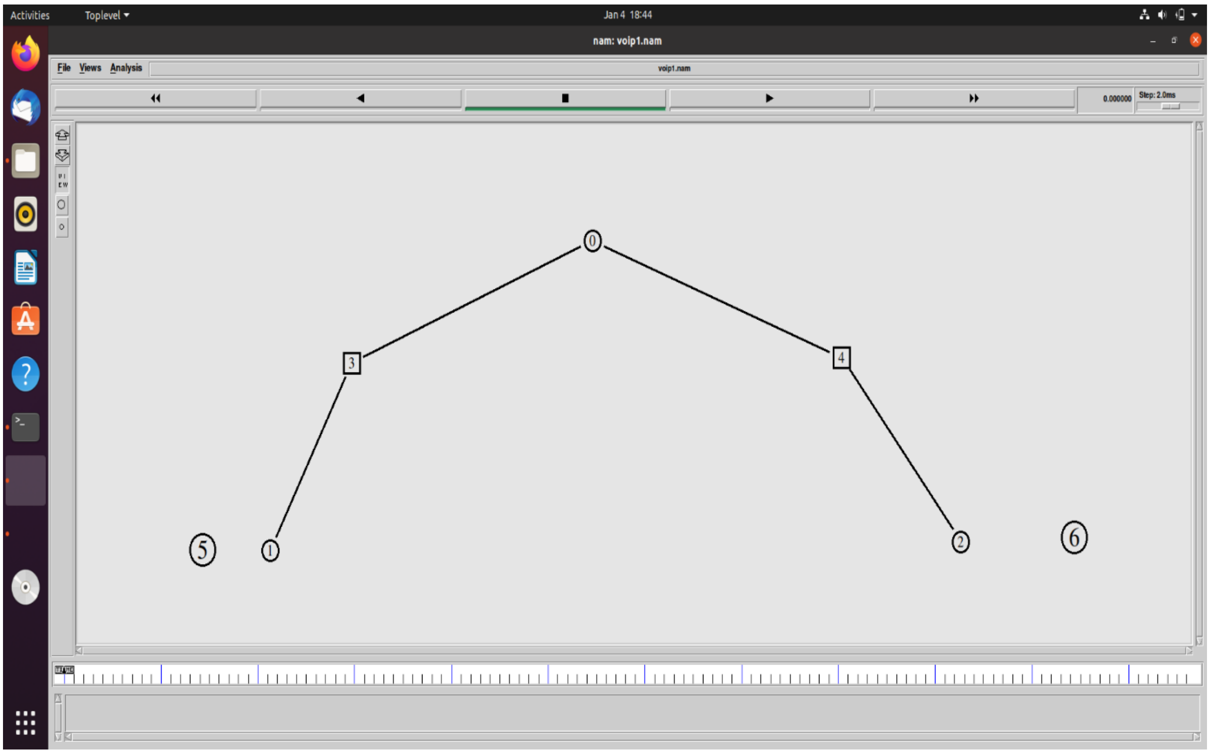


Figure 2.2. Structure of the VoIP packet (as in IPv4)

TOPOLOGY:


Activities
Toplevel
Jan 4 18:44
nam: voip1.nam
File
Views
Analysis
voip1.nam
0
1
2
3
4
5
6

SOURCE CODE:

PROGRAM 1:

In this program, the protocol is SCTP and the connections are between both HPs:

VoIP OVER SCTP (HARD PHONE TO HARD PHONE)

CODE:

```
set ns  [new Simulator]

global opt
set opt(chan)       Channel/WirelessChannel
set opt(prop)       Propagation/TwoRayGround
set opt(netif)      Phy/WirelessPhy
set opt(mac)        Mac/802_11
set opt(ifq)        Queue/DropTail/PriQueue
set opt(ll)         LL
set opt(ant)        Antenna/OmniAntenna
set opt(x)          500
set opt(y)          310
set opt(ifqlen)     50
set opt(wls_nodes)    2
set opt(adhocRouting)   DSDV
set opt(ap_nodes)        2

$ns color 1 orange
```

```
$ns color 2 red

# set up for hierarchical routing
$ns node-config -addressType hierarchical
AddrParams set domain_num_ 3
lappend cluster_num 1 1 1
AddrParams set cluster_num_ $cluster_num
lappend eilastlevel 1 3 3
AddrParams set nodes_num_ $eilastlevel

set tf  [open voip1.tr w]
$ns trace-all $tf

set nf [open voip1.nam w]
$ns namtrace-all $nf

set rtr [$ns node 0.0.0];

set w(0) [$ns node 1.0.2]
set w(1) [$ns node 2.0.2]

create-god [ expr $opt(wls_nodes) + $opt(ap_nodes) ]

set topo   [new Topography]
$topo load_flatgrid $opt(x) $opt(y)

set channel [new $opt(chan)]

$ns node-config -adhocRouting $opt(adhocRouting) \
                -llType $opt(ll) \
```

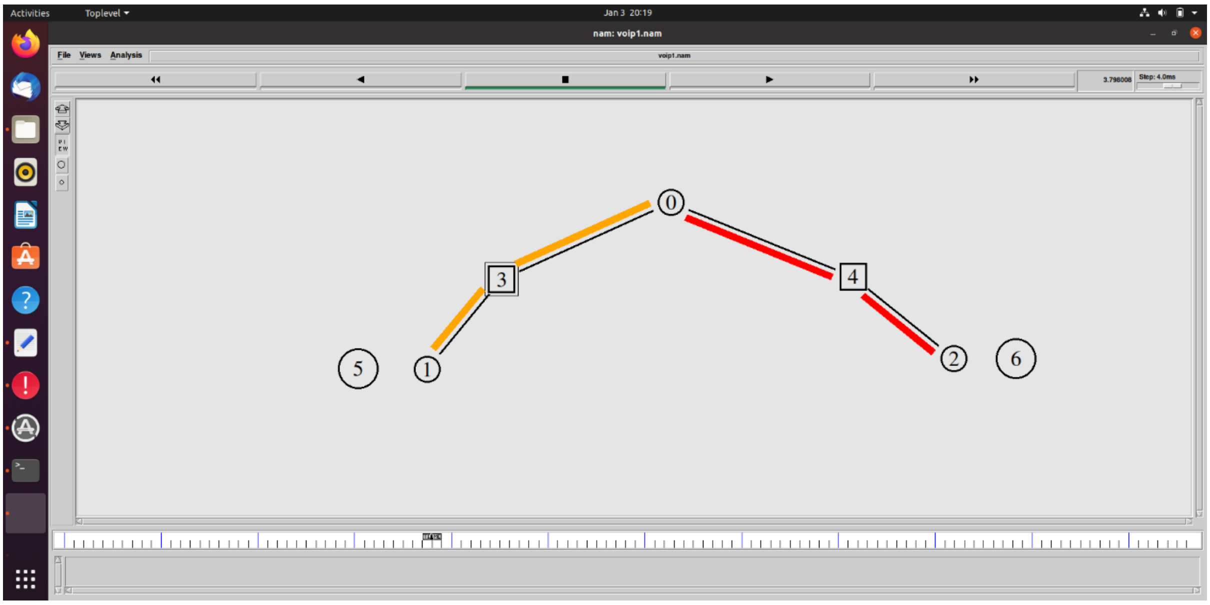
```
                -macType $opt(mac) \
                -ifqType $opt(ifq) \
                -ifqLen $opt(ifqlen) \
                -antType $opt(ant) \
                -propType $opt(prop) \
                -phyType $opt(netif) \
                -channel $channel \
                -topoInstance $topo \
                -wiredRouting ON \
                -agentTrace ON \
                -routerTrace ON \
                -macTrace OFF


set AP(0) [$ns node 1.0.0]
$AP(0) random-motion 0 ;
$AP(0) shape box

set AP(1) [$ns node 2.0.0]
$AP(1) random-motion 0 ;
$AP(1) shape box


#Locating the base station nodes in topography

$AP(0) set X_ 10.0
$AP(0) set Y_ 470.0
$AP(0) set Z_ 0.0
```

```
$AP(1) set X_ 400.0
$AP(1) set Y_ 470.0
$AP(1) set Z_ 0.0



#configure for mobilenodes
$ns node-config -wiredRouting OFF

set end_device(0) [$ns node 1.0.1]
$end_device(0) base-station [AddrParams addr2id [$AP(0) node-addr]]

$end_device(0) set X_ 5
$end_device(0) set Y_ 300
$end_device(0) set Z_ 0.0

$end_device(0) color black
$ns initial_node_pos $end_device(0) 15

set end_device(1) [$ns node 2.0.1]
$end_device(1) base-station [AddrParams addr2id [$AP(1) node-addr]]

$end_device(1) set X_ 490
$end_device(1) set Y_ 300
$end_device(1) set Z_ 0.0

$end_device(1) color black
$ns initial_node_pos $end_device(1) 15

$ns duplex-link $rtr $AP(0) 64kb 50ms DropTail
$ns duplex-link $rtr $AP(1) 64kb 50ms DropTail
```

```
$ns duplex-link-op $rtr $AP(0) orient left
$ns duplex-link-op $rtr $AP(1) orient right

$ns duplex-link $AP(0) $w(0) 64kb 50ms DropTail
$ns duplex-link $AP(1) $w(1) 64kb 50ms DropTail

$ns duplex-link-op $AP(0) $w(0) orient right
$ns duplex-link-op $AP(1) $w(1) orient left

set sctp0 [new Agent/SCTP]
$ns attach-agent $w(0) $sctp0
$sctp0 set fid_ 1

set cbr0 [new Application/Traffic/CBR]
$cbr0 set packetSize_ 128
$cbr0 set interval_ 0.05

$cbr0 set class_ 1
$cbr0 attach-agent $sctp0

set sink0 [new Agent/LossMonitor]
$ns attach-agent $w(1) $sink0

set sctp1 [new Agent/SCTP]
$ns attach-agent $w(1) $sctp1
$sctp1 set fid_ 2

set cbr1 [new Application/Traffic/CBR]
$cbr1 set packetSize_ 128
$cbr1 set interval_ 0.05
```

```
$cbr1 set class_ 2

$cbr1 attach-agent $sctp1
set sink1 [new Agent/LossMonitor]
$ns attach-agent $w(0) $sink1

$ns connect $sctp0 $sctp1

$ns at 0.1 "$cbr0 start"
$ns at 0.1 "$cbr1 start"

$ns at 7.0 "$cbr0 stop"
$ns at 7.0 "$cbr1 stop"

proc finish {} {

  global ns tf nf
  $ns flush-trace
  close $tf
  close $nf
  exit 0
}

$ns at 12.0 "finish"
$ns run
```

Output:



Activities
Toplevel
Jan 3 20:19
nam: voip1.nam
File
Views
Analysis
voip1.nam
0
1
2
3
4
5
6

PROGRAM 2:

In this program, the protocol is UDP and the connections are between both HPs.

VoIP OVER UDP (HARD PHONE TO HARD PHONE)

CODE:

```
set ns  [new Simulator]

global opt
set opt(chan)       Channel/WirelessChannel
set opt(prop)       Propagation/TwoRayGround
set opt(netif)      Phy/WirelessPhy
set opt(mac)        Mac/802_11
set opt(ifq)        Queue/DropTail/PriQueue
set opt(ll)         LL
set opt(ant)        Antenna/OmniAntenna
set opt(x)          500
set opt(y)          310
set opt(ifqlen)     50
set opt(wls_nodes)    2
set opt(adhocRouting)   DSDV
set opt(ap_nodes)       2

$ns color 1 orange
$ns color 2 red

# set up for hierarchical routing
$ns node-config -addressType hierarchical
```

```
AddrParams set domain_num_ 3
lappend cluster_num 1 1 1
AddrParams set cluster_num_ $cluster_num
lappend eilastlevel 1 3 3
AddrParams set nodes_num_ $eilastlevel

set tf  [open voip2.tr w]
$ns trace-all $tf

set nf [open voip2.nam w]
$ns namtrace-all $nf

set rtr [$ns node 0.0.0];

set w(0) [$ns node 1.0.2]
set w(1) [$ns node 2.0.2]

create-god [ expr $opt(wls_nodes) + $opt(ap_nodes) ]

set topo   [new Topography]
$topo load_flatgrid $opt(x) $opt(y)

set channel [new $opt(chan)]

$ns node-config -adhocRouting $opt(adhocRouting) \
        -llType $opt(ll) \
        -macType $opt(mac) \
        -ifqType $opt(ifq) \
        -ifqLen $opt(ifqlen) \
        -antType $opt(ant) \
```

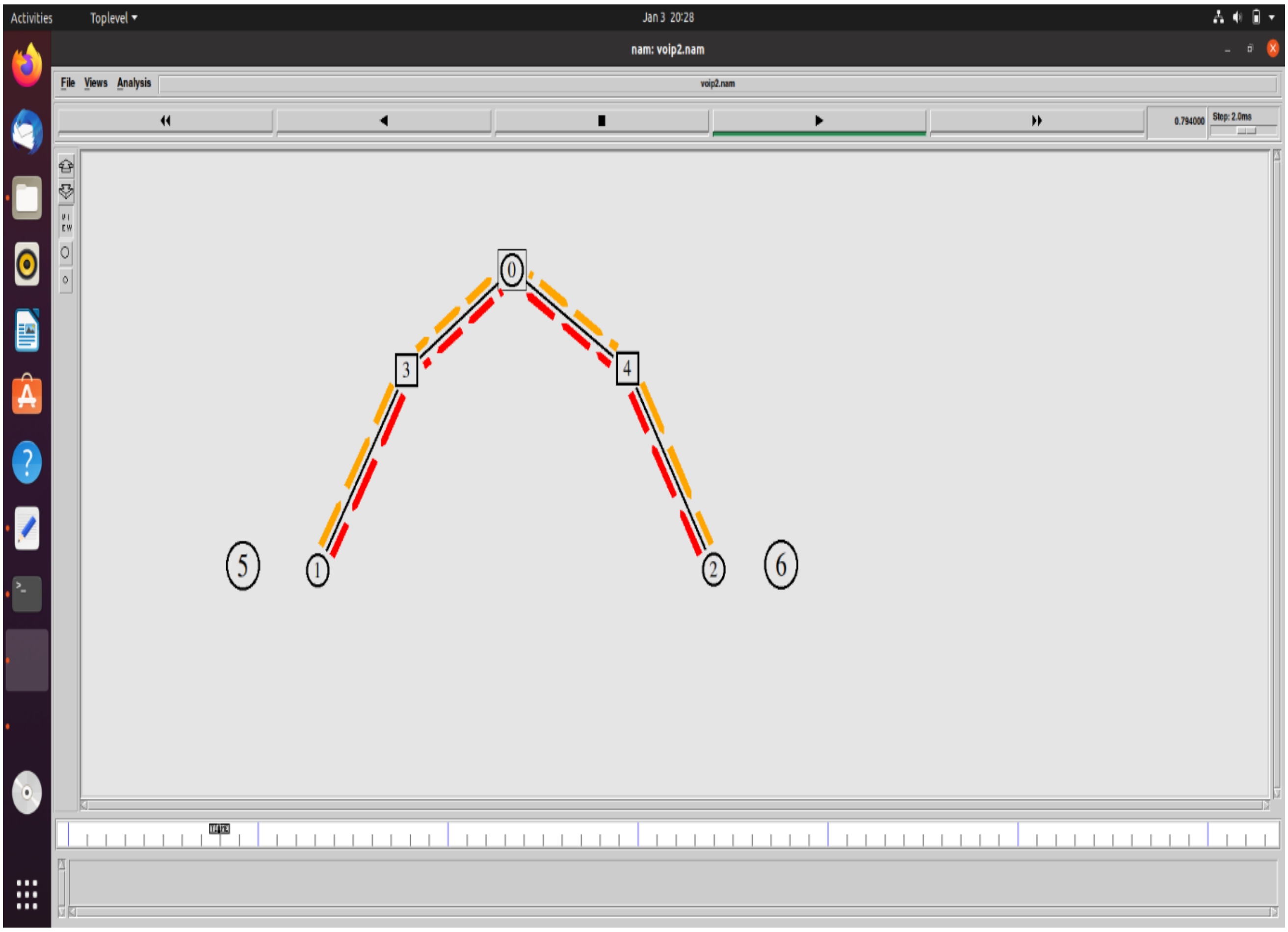
```
        -propType $opt(prop) \
        -phyType $opt(netif) \
        -channel $channel \
        -topoInstance $topo \
        -wiredRouting ON \
        -agentTrace ON \
        -routerTrace OFF \
        -macTrace OFF


set AP(0) [$ns node 1.0.0]
$AP(0) random-motion 0 ;
$AP(0) shape box

set AP(1) [$ns node 2.0.0]
$AP(1) random-motion 0 ;
$AP(1) shape box


#Locating the base station nodes in topography

$AP(0) set X_ 10.0
$AP(0) set Y_ 470.0
$AP(0) set Z_ 0.0


$AP(1) set X_ 400.0
$AP(1) set Y_ 470.0
$AP(1) set Z_ 0.0
```

```
#configure for mobilenodes
$ns node-config -wiredRouting OFF

set end_device(0) [$ns node 1.0.1]
$end_device(0) base-station [AddrParams addr2id [$AP(0) node-addr]]

$end_device(0) set X_ 5
$end_device(0) set Y_ 300
$end_device(0) set Z_ 0.0

$end_device(0) color black
$ns initial_node_pos $end_device(0) 15

set end_device(1) [$ns node 2.0.1]
$end_device(1) base-station [AddrParams addr2id [$AP(1) node-addr]]

$end_device(1) set X_ 490
$end_device(1) set Y_ 300
$end_device(1) set Z_ 0.0

$end_device(1) color black
$ns initial_node_pos $end_device(1) 15

$ns duplex-link $rtr $AP(0) 64kb 50ms DropTail
$ns duplex-link $rtr $AP(1) 64kb 50ms DropTail
$ns duplex-link-op $rtr $AP(0) orient left
$ns duplex-link-op $rtr $AP(1) orient right

$ns duplex-link $AP(0) $w(0) 64kb 50ms DropTail
```

```
$ns duplex-link $AP(1) $w(1) 64kb 50ms DropTail

$ns duplex-link-op $AP(0) $w(0) orient right
$ns duplex-link-op $AP(1) $w(1) orient left

set udp0 [new Agent/UDP]
$ns attach-agent $w(0) $udp0
$udp0 set fid_ 1

set cbr0 [new Application/Traffic/CBR]
$cbr0 set packetSize_ 128
$cbr0 set interval_ 0.020



$cbr0 set class_ 1
$cbr0 attach-agent $udp0



set sinknode1 [new Agent/LossMonitor]
$ns attach-agent $w(1) $sinknode1
$ns connect $udp0 $sinknode1

set udp1 [new Agent/UDP]
$ns attach-agent $w(1) $udp1
$udp1 set fid_ 2

set cbr1 [new Application/Traffic/CBR]
$cbr1 set packetSize_ 128
$cbr1 set interval_ 0.020
```

```
$cbr1 set class_ 2
$cbr1 attach-agent $udp1

set sinknode0 [new Agent/LossMonitor]
$ns attach-agent $w(0) $sinknode0
$ns connect $udp1 $sinknode0



$ns at 0.1 "$cbr0 start"
$ns at 0.1 "$cbr1 start"

$ns at 6.0 "$cbr0 stop"
$ns at 6.0 "$cbr1 stop"

proc finish {} {

  global ns tf nf
  $ns flush-trace
  close $tf
  close $nf
  exit 0
}

$ns at 22.0 "finish"
$ns run
```

Output:


Activities
Toplevel
Jan 3 20:28
nam: voip2.nam
File
Views
Analysis
voip2.nam
0
1
2
3
4
5
6

‘

PROGRAM 3:

In this program, the protocol is UDP and the connections are between HP and SP.

-

VoIP OVER UDP (SOFT PHONE TO HARD PHONE)

CODE:

```
set ns  [new Simulator]

global opt
set opt(chan)       Channel/WirelessChannel
set opt(prop)       Propagation/TwoRayGround
set opt(netif)      Phy/WirelessPhy
set opt(mac)        Mac/802_11
set opt(ifq)        Queue/DropTail/PriQueue
set opt(ll)         LL
set opt(ant)        Antenna/OmniAntenna
```

```
set opt(x)          500
set opt(y)          310
set opt(ifqlen)     50
set opt(wls_nodes)    2
set opt(adhocRouting)   DSDV
set opt(ap_nodes)        2

$ns color 1 orange
$ns color 2 red

# set up for hierarchical routing
$ns node-config -addressType hierarchical
AddrParams set domain_num_ 3
lappend cluster_num 1 1 1
AddrParams set cluster_num_ $cluster_num
lappend eilastlevel 1 3 3
AddrParams set nodes_num_ $eilastlevel

set tf  [open voip2.tr w]
$ns trace-all $tf

set nf [open voip2.nam w]
$ns namtrace-all $nf

set rtr [$ns node 0.0.0];

set w(0) [$ns node 1.0.2]
set w(1) [$ns node 2.0.2]

create-god [ expr $opt(wls_nodes) + $opt(ap_nodes) ]
```

```
set topo   [new Topography]
$topo load_flatgrid $opt(x) $opt(y)

set channel [new $opt(chan)]

$ns node-config -adhocRouting $opt(adhocRouting) \
                -llType $opt(ll) \
                -macType $opt(mac) \
                -ifqType $opt(ifq) \
                -ifqLen $opt(ifqlen) \
                -antType $opt(ant) \
                -propType $opt(prop) \
                -phyType $opt(netif) \
                -channel $channel \
                -topoInstance $topo \
                -wiredRouting ON \
                -agentTrace ON \
                -routerTrace OFF \
                -macTrace OFF



set AP(0) [$ns node 1.0.0]
$AP(0) random-motion 0 ;
$AP(0) shape box

set AP(1) [$ns node 2.0.0]
$AP(1) random-motion 0 ;
$AP(1) shape box
```

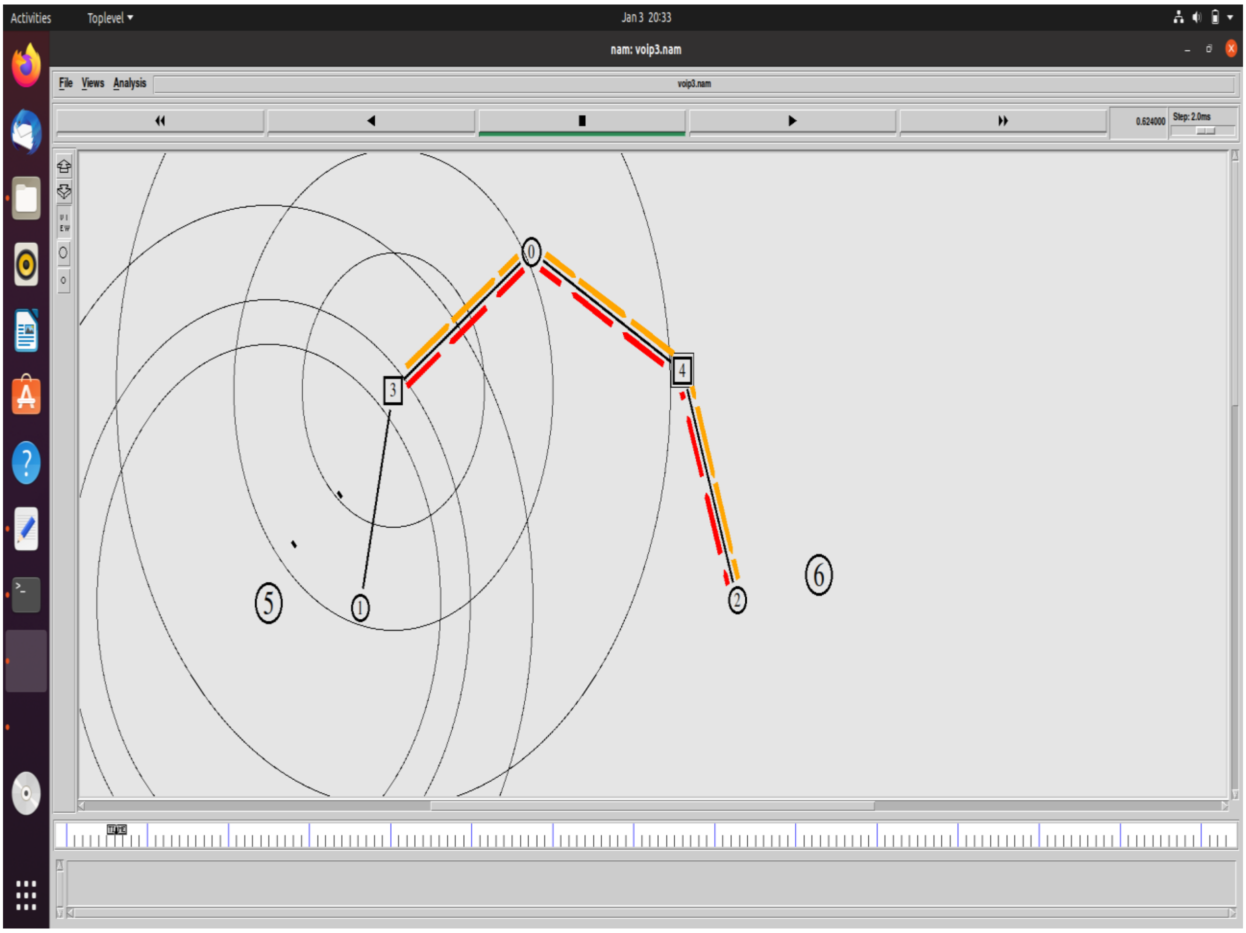
```
#Locating the base station nodes in topography

$AP(0) set X_ 10.0
$AP(0) set Y_ 470.0
$AP(0) set Z_ 0.0



$AP(1) set X_ 400.0
$AP(1) set Y_ 470.0
$AP(1) set Z_ 0.0



#configure for mobilenodes
$ns node-config -wiredRouting OFF

set end_device(0) [$ns node 1.0.1]
$end_device(0) base-station [AddrParams addr2id [$AP(0) node-addr]]

$end_device(0) set X_ 5
$end_device(0) set Y_ 300
$end_device(0) set Z_ 0.0

$end_device(0) color black
$ns initial_node_pos $end_device(0) 15

set end_device(1) [$ns node 2.0.1]
$end_device(1) base-station [AddrParams addr2id [$AP(1) node-addr]]

$end_device(1) set X_ 490
```

```
$end_device(1) set Y_ 300
$end_device(1) set Z_ 0.0

$end_device(1) color black
$ns initial_node_pos $end_device(1) 15

$ns duplex-link $rtr $AP(0) 64kb 50ms DropTail
$ns duplex-link $rtr $AP(1) 64kb 50ms DropTail
$ns duplex-link-op $rtr $AP(0) orient left
$ns duplex-link-op $rtr $AP(1) orient right

$ns duplex-link $AP(0) $w(0) 64kb 50ms DropTail
$ns duplex-link $AP(1) $w(1) 64kb 50ms DropTail

$ns duplex-link-op $AP(0) $w(0) orient right
$ns duplex-link-op $AP(1) $w(1) orient left

set udp0 [new Agent/UDP]
$ns attach-agent $w(0) $udp0
$udp0 set fid_ 1

set cbr0 [new Application/Traffic/CBR]
$cbr0 set packetSize_ 128
$cbr0 set interval_ 0.020


$cbr0 set class_ 1
$cbr0 attach-agent $udp0
```

```
set sinknode1 [new Agent/LossMonitor]
$ns attach-agent $w(1) $sinknode1
$ns connect $udp0 $sinknode1

set udp1 [new Agent/UDP]
$ns attach-agent $w(1) $udp1
$udp1 set fid_ 2

set cbr1 [new Application/Traffic/CBR]
$cbr1 set packetSize_ 128
$cbr1 set interval_ 0.020

$cbr1 set class_ 2
$cbr1 attach-agent $udp1

set sinknode0 [new Agent/LossMonitor]
$ns attach-agent $w(0) $sinknode0
$ns connect $udp1 $sinknode0



$ns at 0.1 "$cbr0 start"
$ns at 0.1 "$cbr1 start"

$ns at 6.0 "$cbr0 stop"
$ns at 6.0 "$cbr1 stop"

proc finish {} {

  global ns tf nf
  $ns flush-trace
```

```
  close $tf
  close $nf
  exit 0
}

$ns at 22.0 "finish"
$ns run
```

Output:



PROGRAM 4:

In this program, the protocol is SCTP and the connections are between HP and SP.

VoIP OVER SCTP (SOFT PHONE TO HARD PHONE)

CODE:

```
set ns  [new Simulator]

global opt
set opt(chan)       Channel/WirelessChannel
set opt(prop)       Propagation/TwoRayGround
set opt(netif)      Phy/WirelessPhy
set opt(mac)        Mac/802_11
set opt(ifq)        Queue/DropTail/PriQueue
set opt(ll)         LL
set opt(ant)        Antenna/OmniAntenna
set opt(x)          500
set opt(y)          310
set opt(ifqlen)     50
set opt(wls_nodes)    2
set opt(adhocRouting)   DSDV
set opt(ap_nodes)       2

$ns color 1 orange
$ns color 2 red

# set up for hierarchical routing
```

```
$ns node-config -addressType hierarchical
AddrParams set domain_num_ 3
lappend cluster_num 1 1 1
AddrParams set cluster_num_ $cluster_num
lappend eilastlevel 1 3 3
AddrParams set nodes_num_ $eilastlevel

set tf  [open voip4.tr w]
$ns trace-all $tf

set nf [open voip4.nam w]
$ns namtrace-all $nf
$ns namtrace-all-wireless $nf $opt(x) $opt(y)

set rtr [$ns node 0.0.0];

set w(0) [$ns node 1.0.2]
set w(1) [$ns node 2.0.2]

create-god [ expr $opt(wls_nodes) + $opt(ap_nodes) ]

set topo   [new Topography]
$topo load_flatgrid $opt(x) $opt(y)

set channel [new $opt(chan)]

$ns node-config -adhocRouting $opt(adhocRouting) \
		-llType $opt(ll) \
		-macType $opt(mac) \
		-ifqType $opt(ifq) \
```

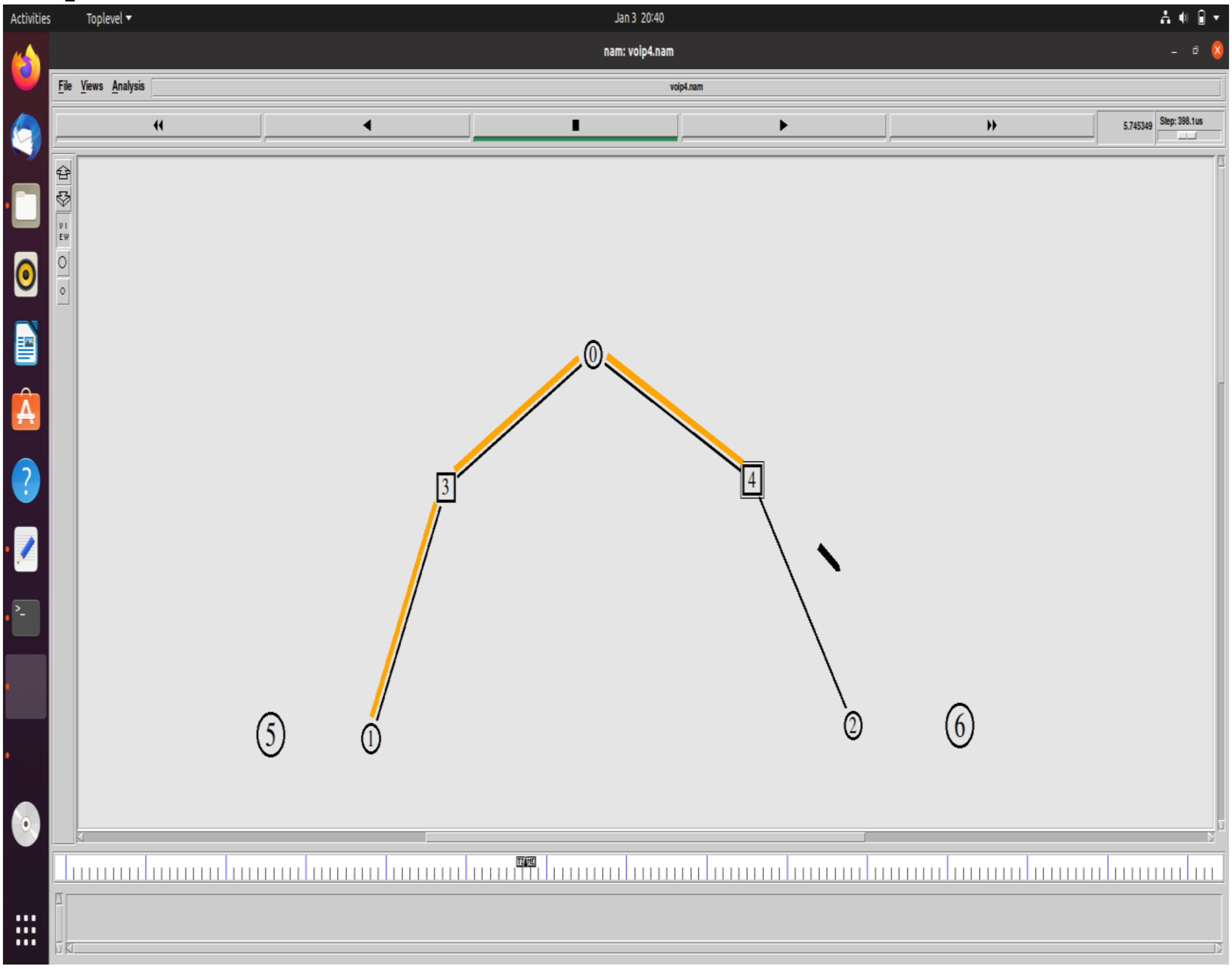
```
        -ifqLen $opt(ifqlen) \
        -antType $opt(ant) \
        -propType $opt(prop) \
        -phyType $opt(netif) \
        -channel $channel \
        -topoInstance $topo \
        -wiredRouting ON \
        -agentTrace ON \
        -routerTrace ON \
        -macTrace OFF


set AP(0) [$ns node 1.0.0]
$AP(0) random-motion 0 ;
$AP(0) shape box

set AP(1) [$ns node 2.0.0]
$AP(1) random-motion 0 ;
$AP(1) shape box


#Locating the base station nodes in topography

$AP(0) set X_ 10.0
$AP(0) set Y_ 470.0
$AP(0) set Z_ 0.0


$AP(1) set X_ 400.0
$AP(1) set Y_ 470.0
```

```
$AP(1) set Z_ 0.0


#configure for mobilenodes
$ns node-config -wiredRouting OFF

set end_device(0) [$ns node 1.0.1]
$end_device(0) base-station [AddrParams addr2id [$AP(0) node-addr]]

$end_device(0) set X_ 5
$end_device(0) set Y_ 300
$end_device(0) set Z_ 0.0

$end_device(0) color black
$ns initial_node_pos $end_device(0) 15

set end_device(1) [$ns node 2.0.1]
$end_device(1) base-station [AddrParams addr2id [$AP(1) node-addr]]

$end_device(1) set X_ 490
$end_device(1) set Y_ 300
$end_device(1) set Z_ 0.0

$end_device(1) color black
$ns initial_node_pos $end_device(1) 15

$ns duplex-link $rtr $AP(0) 64kb 50ms DropTail
$ns duplex-link $rtr $AP(1) 64kb 50ms DropTail
$ns duplex-link-op $rtr $AP(0) orient left
$ns duplex-link-op $rtr $AP(1) orient right
```

```
$ns duplex-link $AP(0) $w(0) 64kb 50ms DropTail
$ns duplex-link $AP(1) $w(1) 64kb 50ms DropTail

$ns duplex-link-op $AP(0) $w(0) orient right
$ns duplex-link-op $AP(1) $w(1) orient left

set sctp0 [new Agent/SCTP]
$ns attach-agent $w(0) $sctp0
$sctp0 set fid_ 1

set cbr0 [new Application/Traffic/CBR]
$cbr0 set packetSize_ 128
$cbr0 set interval_ 0.05

$cbr0 set class_ 1
$cbr0 attach-agent $sctp0

set sink0 [new Agent/LossMonitor]
$ns attach-agent $end_device(1) $sink0

set sctp1 [new Agent/SCTP]
$ns attach-agent $end_device(1) $sctp1
$sctp1 set fid_ 2

set cbr1 [new Application/Traffic/CBR]
$cbr1 set packetSize_ 128
$cbr1 set interval_ 0.05
$cbr1 set class_ 2
```

```
$cbr1 attach-agent $sctp1
set sink1 [new Agent/LossMonitor]
$ns attach-agent $w(0) $sink1

$ns connect $sctp0 $sctp1

$ns at 0.1 "$cbr0 start"
$ns at 0.1 "$cbr1 start"

$ns at 7.0 "$cbr0 stop"
$ns at 7.0 "$cbr1 stop"


proc finish {} {

  global ns tf nf
  $ns flush-trace
  close $tf
  close $nf
  exit 0
}

$ns at 22.0 "finish"
$ns run
```

Output:



Conclusion:

We have successfully simulated VoIP over SCTP and UDP using NS2 Simulator.

References:

1. https://www.sfu.ca/~ljilja/ENSC427/Spring13/Projects/team15/Final_Report.pdf

2. https://www.studytonight.com/computer-networks/osi-model-transport-layer

3. https://www.nextiva.com/blog/voip-codecs.html

Abbreviation

- HP-Hard Phone
- SP-Soft Phone
- VoIP – Voice over internet protocol
- SCTP- Stream Control Transmission Protocol
- UDP-User Datagram Protocol

THANK YOU!