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Voice over Internet Protocol (VoIP) - A Review

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Abstract:- Voice over Internet protocol (VoIP), is a modern method of communication. The field of "IP Telephony" is growing more and more popular today. Voice over Internet Protocol (VoIP) is the term used to describe the transmission of voice messages over packet-switched IP networks. One of the most recent forms of communication is VoIP. VoIP presents opportunities as well as security challenges, as is the case with the majority of modern technology. In contrast to the traditional circuit-based telephony, it features a distinctive architecture. VoIP is hence vulnerable to a variety of security threats.

I. INTRODUCTION

Voice over IP is referred to as VOIP. A group of technologies and procedures that make it easier to transport voice messages over IP networks are collectively referred to as "voice over Internet Protocol." VoIP networks are also referred to by other terms, such as Internet telephony, IP telephony, and broadband phone service. Any type of computer can be used to run VoIP systems. Additionally, VoIP services canbe adapted to work with traditional phone handsets. Voice Over Internet Protocol is currently a technology that dominates the communication sector (VoIP). The simplestapproach to make a phone call over the internet is to send packets across a network that uses a packet switching architecture. VOIP is superior than traditional communication in some ways. Though it incurs cost for network connectivity, many applications like Skype, Yahoo Messenger, Google Chat, etc. allow “free" calls to their users. Long distance phone calls such as Jumblo, are also cheaper than traditional phone calls [3]. VOIP also reduces the infrastructure cost as an only a single network isrequired to carry the data as well as voice packets. Packets are delivered when media content is to be delivered. The user can receive and make calls from anywhere.

II. VOIP SIGNALING PROTOCOLS

Signalling protocol is crucial when conducting phone calls online since it enablesnetwork elements to communicate with one another and establish and terminate calls.A call in IP telephony can be defined as a multimedia session involving several participants, whilst signalling associated with a call is referred to as a connection. A signalling protocol's primary duties can be broken down into four categories: [1]

- **Session Establishment**: The call recipient chooses whether to accept, reject, or divert the call.
- **User Location**: The caller must first determine the callee's location.
- **Call Participant Management**: Endpoints can join or exit an existing session usingthis feature.
- **Session Negotiation**: A set of session attributes should be agreed upon by all call-involved endpoints.

There are two distinct types of protocols that allow Internet Telephony:1. H.323

A. *Session Initiation Protocol (SIP) [1]*

➢ *H.323*

Basically, H.323 may be a standard set by International Telecommunication Union (ITU)which allows telephones on the general public telephone network to speak to computers connected to internet. The architecture of H.323 is shown:

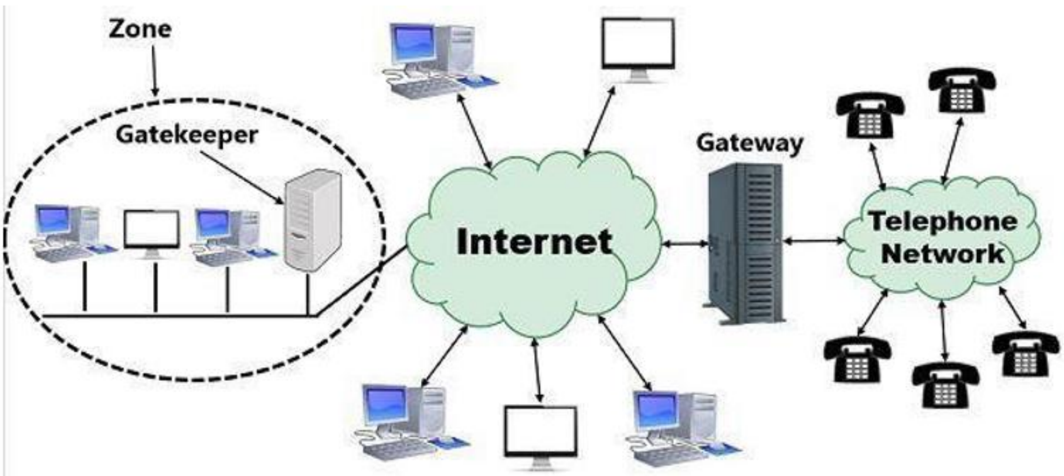

Fig 1: H.323 Architecture

Four main elements of H.323 are Multipoint control units (MCUs), Gateways,Terminals, and Gatekeeper. [2]

- MCU:

MCUs are used to control multipoint conferences between three or more H.323 terminals. A required multipoint controller (MC) and an optional multipoint processor make up the MCU (MP). It aids in the capability negotiations with each terminal to ensure a uniform degree of communications.

- Terminals:

Terminals are capable of conducting real-time bidirectional multimedia communications. A standalone device or a personal computer running H.323 and multimedia programmes are both examples of H.323 terminals. Because it providesthe fundamental service of audio communications, it is crucial to IP telephony.

- Gateways:

In order to connect H.323 networks to other networks, such as ISDN, PSTN, H.320 systems, etc., gateways are used. Networks' disparate connectivity is achieved by converting media formats between them and interpreting call setup and release protocols.

- Gatekeepers:

Admissions control and address resolution are handled by gatekeepers. Calls may be made directly between endpoints or via its own internal call signalling system. In addition, it is in charge of bookkeeping, billing, and band control services. A single gatekeeper controls a group of Terminals, Gateways, and MCUs that makeup a zone. [1]

B. SESSION INITIATION PROTOCOL (SIP)

An Internet Telephony protocol called SIP (Session Initiation Protocol) is used for signalling and managing multimedia communication sessions, including online gaming, instant messaging, and other services. Since messages consist of a message body and headers, it is analogous to the web protocol HTTP. SIP often utilises TCP or UDP as its default protocol on port 5060. SIP can be thought of as the authorization protocol for VoIP (voice, telephone, and video) services. [3]

- *SIP Server*

In order to effectively direct requests issued from one user agent to another, the SIP server converts usernames to IP addresses. By registering with the SIP server and giving it their username and current IP address, a user agent can determine where they are now located on the network. Additionally, this confirms their onlinestatus so that other user agents can check if they are available and invite them to asession. A request is sent to the SIP server to invite another user into a session because the user agent is unlikely to be aware of the IP address of another user agent [4]. The SIP server then determines whether the user is online and, if so, usesa comparison of the user's username and IP address to pinpoint their location. It willalso route requests to other servers if the user uses a differentSIP server because they are not apart of that domain. The SIPserver will behave in any of the following different roles whencarrying out these varied duties ofdelivering client requests:

- *Registrar server*
- *Proxy server*
- *Redirect server [3]*

- *Registrar Server*

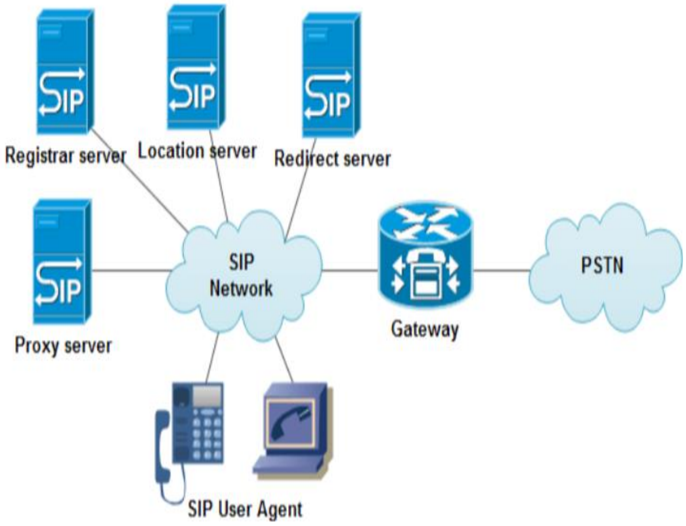

Fig 2: SIP architecture

The location of a user agent that has logged onto the network is recorded by registrar servers. It acquires the user's IP address and connects it to their system username. This generates a directory of everyone who is connected to the network right now, along with their whereabouts. The Registrar server's data is resorted to when someone wants to start a connection with one of these users, allowing the IP addresses of individuals taking part in the session to be determined. [3]

- *Proxy Server*

Computers called proxy servers are employed to transmit requests on behalf of other machines. When a client sends a request to a SIP server, that server may passthe request on to another SIP server on the network. The SIP server can perform proxy server duties as well as network access control, security, authentication, andauthorization. [3]

- *Redirect Server*

It allows proxy servers to connect SIP session invitations to external domains. [1]

III. VOIP CODECS

The compression procedure that enables call transmission over an IP network is carried out by the codec, a voice/video encoding method. There may be variations in the sound/video quality, needed bandwidth, computing needs, etc. Since every algorithm requires a certain amount of buffering data before it is processed, all programmes, services, gateways, etc. support different Codecs. This also introducesa digitising delay. [1]

Codec Information				Bandwidth Calculations			
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	55.2 Kbps
G722_64k (64 Kbps)	80 Bytes	10 ms	4.13	160 Bytes	20 ms	50	87.2 Kbps

Table 1: Some popular Audio Codecs

IV. REAL TIME PROTOCOL

Real-time protocols transport audio and video over IP networks and take care of theneeds of applications with real-time characteristics. It is typically utilised in push-to- talk functions on the web, telephone, and other streaming media for communication and enjoyment. It also controls the multicast or unicast network services for the real-time delivery of multimedia data. [1]

V. COST

The VoIP system is popular because, in comparison to other communication methods, it is less expensive. The switched telephone networks are not utilised by the VoIP system. As a result, sending voice communications over great distances is inexpensive. The standard telecommunications line configuration is not used by the VoIP system to transmit voice traffic. The internet or private data network lines are used instead to carry the traffic. [2]

VI. VOIP ATTACKS

An electronic network may be the target of a denial of service (DoS) assault, which prevents it from receiving services or connectivity. It may be accomplished by usingall of its bandwidth or by overtaxing the network. A DoS attack is an attempt to render a network resource or device unavailable to the users for whom it isdesigned. VoIP DoS attacks are conducted through flooding. This results in earlier calls dropping and cutting the call short. Once the service is denied to the target location, the attacker can take remote control of the company's administrative facilities. A DoS attack can take many different forms. A media protocol and a signalling protocol are used to establish a VoIP connection. [1]

VII. LATENCY

Latency in Voice over Internet Protocol refers to the duration that it takes a voice data to travel from the sender to the receiver. High latency translates toa delay or echo caused by slow network links. Latency is measured in milliseconds (ms), which are thousandths of a second. A latency of 20 ms is normal for VoIP calls;a latency of 150 ms is barely noticeable and therefore acceptable. Any higher than that, however, and quality starts to diminish. At 300 ms or higher, latency becomesunacceptable. [4]

➢ *Effects of Latency on Voice Calls*

- The negative effects of latency on call quality include:
- Slow and interrupted phone conversations.
- Overlapping noises, with one speaker interrupting the other.
- Echo.
- Disturbed synchronization between voice and other data types, especiallyduring video conferencing. [4]

VIII. JITTER

When specific information packets are lost or sent out of order, it causes jitter, which results in a disorganised dialogue. Jitter, to put it simply, is when information is delivered out of sequence and is not received by the intended receiver in the same order. This phenomenon is often measured in milliseconds, and call quality willdrastically decline if the jitter exceeds 40 or 50ms. [4]

➢ *Reasons for Jitter:*

- Network Congestion
- Wireless Networks
- Bad Hardware

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IX. CONCLUSION

VoIP is currently enjoying the benefits of years of hard work, and it may be regarded as a mature technology. In addition, the idea of a new broadband network that combines fixed and mobile networks that are already in existence and those that are just starting to emerge is motivated by the need for network operators to offer new broadband services as well as the desire of customers to have access to their services from anywhere. VoIP is device-flexible. Your smartphone, tablet, computer, or even a dedicated VoIP telephone can keep you connected at all times. The majority of internet-based phone services offer unlimited long-distance calling. Minutes tracking is unnecessary, and the sole expense is a set monthly fee. The versatility of the VoIP phone number has aided in its uptake in a variety of contexts, from home-based businesses to transportation companies. A virtual phone line allows remote workers to speak with one other from any device, full-time travellers to stay in contact while travelling, and business owners to easily scale up or down their operations. Cloud- based numbers don't need any additional gear because they run on your existing internet. The cost reductions for business owners are significant, calling is unlimited, and setup charges are negligible.

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