

A Comprehensive Survey on Voice over Internet Protocol (VoIP)

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Abstract— Voice over Internet protocol (VoIP) is a new way of communicating. It is a technology that allows users to make telephone calls over an IP network. This paper will describe Voice over Internet Protocol (VoIP) to a level that allows business concerns of implementing VoIP, components of a VoIP system. The business concerns will be those that affect Quality of Service (QoS). VoIP components will include end-user equipment, network components, call processors, gateways and two of the more common architectures: Session Initiation Protocol (SIP), This paper gives a brief introduction of VoIP technology: the network structure, protocols, echo and delay, jitter, and packet loss in VoIP network. Finally, the survey concludes with a discussion on the feasibility of providing VoIP over challenging satellite links.

Keywords— VoIP, H.323, SIP, QoS, RTP, RTCP

I. INTRODUCTION

Today, One of the most dominant technology in the communication world is Voice Over Internet Protocol (VoIP). It is the easiest way to make a phone call through internet by sending packets through packet switched based network. VOIP has a benefit over the conventional form of communication: Public Switched Telephone Network (PSTN) since it sends packet through packet switched based network that voice data packets may take the best path while compared to PSTN which is a circuit-based switch technology that requires dedicated line for telecommunications services. Key benefits of using VOIP as over WIMAX networks to provide broadband services include, best quality of services, preferable cost and reliability. However, there are many threats attached to using VOIP network over WIMAX network when compared to its wire DSL network. Thus, VOIP over WIMAX witnesses various implantation issues such as network capacity and architecture, system design and quality of service.



Figure 1: Structure of Voice over Internet Protocol (VOIP)

II. VOIP SIGNALING PROTOCOLS

While making phone calls over internet, signalling protocol plays essential role since it empower the network components to communicate with each other, hence set up and tear down calls. For IP telephony, a call can be prescribe as the multimedia session between multiple participants, while on the other hand signalling conjoined with a call is referred to as a connection. Key roles of a signalling protocol can be divided into four functions:

- Session establishment: The callee decides, if to accept, reject or redirect the call.
- User location: The caller first has to find the location of the callee.
- Call participant management: It allows endpoints to join or leave an existing session.
- Session negotiation: The endpoints involved in the call should concur upon a set of properties for the session.

A. H.323

The procedures, elements and protocols specified by the H.323 standard that provides multimedia communication across packet-based networks. Multipoint multimedia or Point-to-point communication services is being provided by H.323 system when its four main elements – Multipoint control units (MCUs), gateways, Terminals, and gatekeeper, work together [12].

MCUs:

The function of MCUs is to manage multipoint conferences of three or more H.323 terminals. The MCU comprises of a mandatory multipoint controller (MC) and optional multipoint processor (MP). It assists the negotiation of capabilities with all terminals in order to insure a common level of communications.

Terminals:

Real-time bidirectional multimedia communications can be done by Terminals. A H.323 terminal could be a stand-alone device or personal PC that runs H.323 and multimedia applications. Owing to the fact of its basic service of audio communications, it plays a important role in IP-telephony.

Gateways:

Gateways create connectivity of H.323 networks to variant networks, such as ISDN, PSTN, H.320 systems, etc. The unlike connectivity of networks is attained by converting media format among various networks and translating protocols for call setup and release.

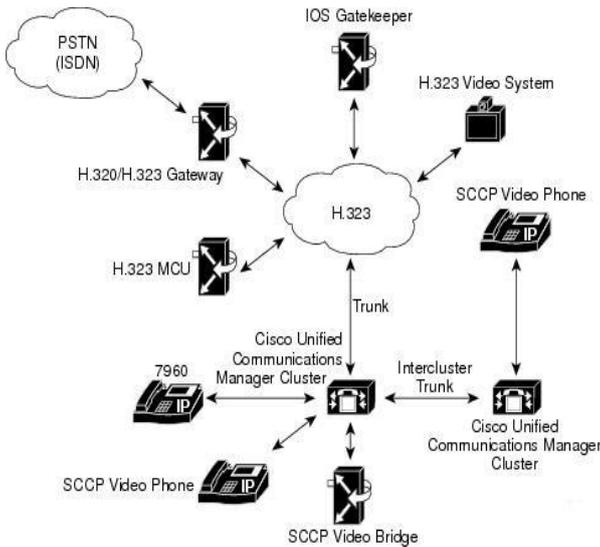


Figure 2: H.323 architecture

Gatekeepers:

Gatekeepers is for address resolution and admission control. It may allow calls either to be placed directly across endpoints or route the call signalling through itself. It also has the responsibility for the services of band control, billing and accounting. A collection of Terminals, Gateways, and MCUs forming a zone is managed by a single gatekeeper.

The H.323 Protocol Stack

H.323 suite comprising of a set of standards. G.711 (64 kbps channel) is the minimum requirement for audio applications. Various voice codec standards illustrated by H.323 are G.723 (5.3 and 6.3 kbps channels), G.728 (16 kbps channel), G.722 (48, 56, and 64 kbps channels), G.729 (8 kbps channel) [4]. The control protocol H.245 for multimedia communication is used during an initial deal among the machines to find the terminal capabilities, audio encoding algorithm and media channels. The Real-Time Control Protocol (RTCP) has a function to provide quality of the sessions and connections and also feedback information across the communication parties. The support and data packets can operate over UDP or TCP [4].

Audio I/O	Video I/O	Data App.	System Control User Interface
Audio Codec G.711 G.722 G.723 G.728 G.729	Video Codec H.261 H.263	Data Interface T.120	System Control CALL Control RAS Control H.245 Control H.225
RTP/RTCP			
UDP		UDP or TCP	
IP			
L_2 Varies			
L_1 Varies			

Figure 3: H.323 protocol Stack

B. Session Initiation Protocol (SIP)

SIP (Session initiation protocol) is a communication protocol used for signaling and controlling multimedia communication sessions such as online gaming, instant messaging and various services. It is similar to web protocol HTTP since messages comprises of headers and a message body. SIP generally uses port 5060 as its default protocol for either TCP or UDP. SIP can be interpreted as the authorize protocol for voice, telephony and video over IP (VoIP) services.

Network Elements

Server network elements are defined by SIP. Though multiple SIP endpoints can communicate without any involvement of SIP infrastructure but this commence is often not practical for a general service. The main network elements involved in the SIP communication can be illustrated as follows:

Proxy server

It is a mediator entity which reacts as server (UAS) as well as client (UAC) for raising requests on behalf of various clients. It also performs routing to transmit the job assigned to another entity next to the targeted user.

User Agent

The User Agent (UA) is used in generating or receiving SIP messages. It can also act as User Agent Client (UAC) for transmitting SIP messages and the receiver will act as a User

Agent Server (UAS). Elements of SIP network sometimes save this information, since it can be used in identifying SIP compatibility problems.

Redirect server

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

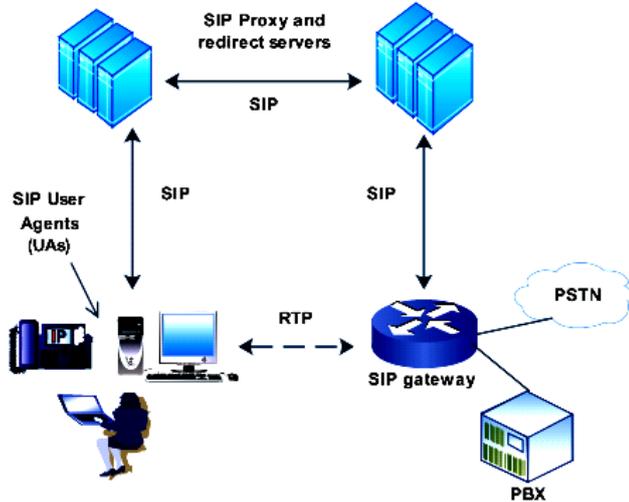


Figure 4: Session Initiation Protocol (SIP)

Gateway

It is used in connecting the SIP network various networks such as public switched telephone network (PSTN) that uses different protocols and technologies.

III. VOIP CODECS

Codec is a voice/video encoding algorithm through which process of compression is carried out that permit the call transmission over the IP network. Sound/video quality, required bandwidth, computational requirements, etc may vary.

Table 1: Audio Codecs

ITU No.	Rate[Kbit/s]	Algorithm	Frame Length
G.711	64	Pulse Code Modulation	0.125 ms
G.723.1	6.4/5.3	Multipulse Max. Likelihood quantization/algebraic-code-excited linear prediction	30 ms
G.726	40/32/24/16	Adaptive differential PCM	0.125 ms
G.728	16	Low-delay Code-excited linear prediction	2.5 ms
G.729	8	Conjugate-structure algebraic-code-excited linear prediction	10 ms

All programs, services, gateway etc. supports various Codecs and it also introduce a digitizing delay since every algorithm needs a certain amount buffering data before it is processed.

IV. REAL TIME PROTOCOLS

Real time protocols takes care of requirements of applications with real-time characteristics and delivers audio and video over IP network. It is generally used in communication and entertainment which involves streaming media like television services, telephony and web based push-to-talk features. It also manages the real time transmission of multimedia data multicast or unicast network services.

A. RTP – (Real Time Protocol)

RTP is drafted for all over real-time stream data transfer. It enables data transfer to multiple destinations through IP multicast and treated as primary standard for audio/video transport within IP networks. Generally RTP is used in alliance with a signalling protocol that assists in build up connections across the network. The headers of the RTP protocol are described below.

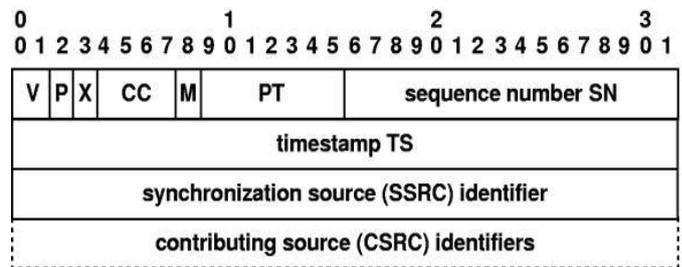


Figure 5: RTP Header

The RTP protocol is relevant for audio and video streaming. Two RTP sessions establishes for video streaming each with different SSRC identifiers out of which one is used for audio transmission whereas another for video transmission. Also, there is downside of RTP that it neither assure delivery of packets nor Quality of Service (QoS).

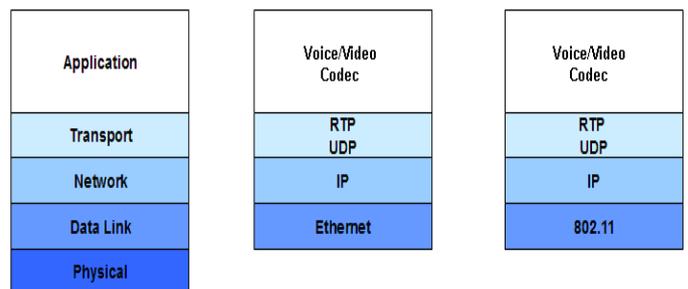


Figure 6: RTP Over different Network Conditions

B. RTCP – (Real Time Control Protocol)

RTCP stands for Real Time Control Transport Protocol and is defined as a protocol that works with real-time protol to observe delivery of data over large multicast network. RTCP can monitors the fraction lost, jitter, packet loss and one way delay. The basic functionality and structure of packet is defined in RFC 3550. One of the major drawbacks in the RTCP does not report the late arrival of packets. This has been

overcome in an improved version of RTCP-XR (Real Time Control Protocol Extended Reports).

V. VOIP ATTACKS

A. Denial of Services (DoS)

Denial of services (DoS) attack is an attack on an electronic network denying it of a service or connectivity. It can be done by consuming its bandwidth or overloading the network. DoS attack is attempt to make a network resource or devices unavailable to its intended end users. In VoIP DoS attack is carried out by the flooding. This causes calls to drop previously and interrupt the call. Once the target area is denied of the service, attacker can get remote control of administrative facilities of the organization. There are many types of DoS attack. A VoIP connection is established using two protocols, a media protocol and signaling protocol.

VoIP Signalling DoS attack

The attacker can attacks on signalling protocol to manipulate a Denial of Services attack. In the first type of attack the attacker sends the many "invites" request to B. At the same time A also sends the "invite" request to B, but B can't take request from A. In this type of DoS attack does not have same LAN requirement, only needs of large volume of request to flood the victim.

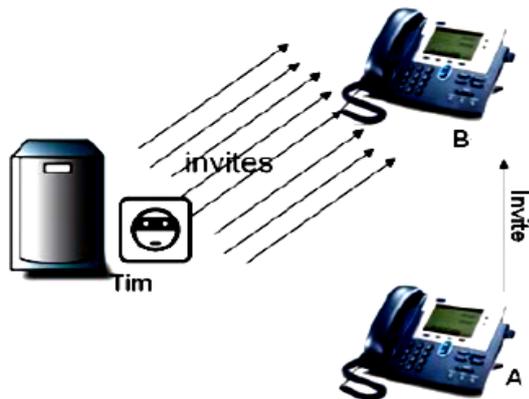


Figure 7(a): Start signal DoS

In the second case, the attacker use cancellation to delete all pending call set up signals by sending a GOODBYE, CANCEL or PORT UNREACHABLE message. The attacker wants to interrupt the landline calls by sending the malicious hang-up messages to the receiver as if they are equivalent from the caller. In the fig 7. (b) It shows an example where spoofed CANCEL message by the attacker to prevent call setup. In Fig 7. (c) Where GOODBYE message is spoofed by the attacker to prevent call setup or tear down the establish connections.

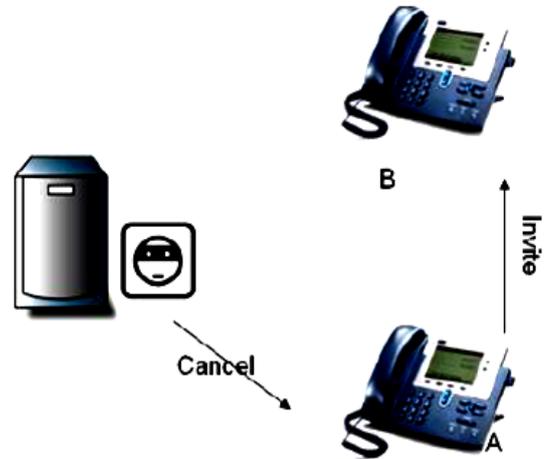


Figure 7(b) Teardown signal DoS

In this type of attack does require the attacker to be able to fill certain header of the appropriate message. The attacker can compile the web data.

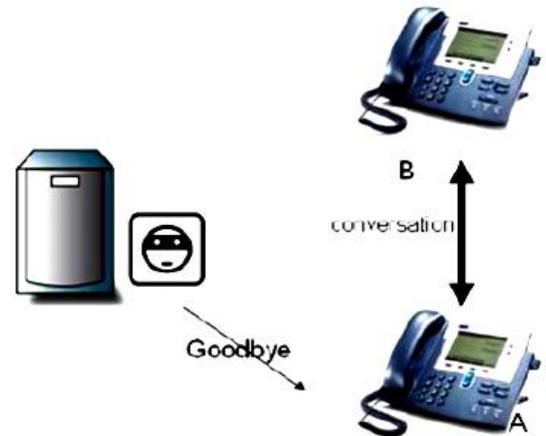


Figure 7(c) Teardown signal DoS

VoIP Media DoS attack

In this type of attack, the attackers can flood the IP Phone, gateway and other media VoIP components with huge numbers of Real Time Protocol packets. It is a popular and prominent way to revoke services to end users is to flood a network with traffic. When transmission is flooded; this can also disturb VoIP services. Because there is not sufficient transmission for the standard users of the services. Additionally the attacker might knock key components like gateway disconnected.

B. Man-in-the-middle attacks

Man-in-the-middle attack [2] in which the attacker prevents call signalling SIP message traffic and masquerades as the

calling party to the called party and once the attacker has gained the position he can hijack calls.

- a. User A Forwards an invite message to B and this message are identified by attacker.
- b. Attacker forwards a reply message to A spoofing from B with 301 moved permanently code. In the reply Attacker set the new location of A to his personal computer.
- c. A forwards a new invite message to Attacker.
- d. Attacker forwards back an acknowledgement to provide the connection between two.
- e. At the same time, Attacker forwards an invite message to B and he can fake the caller ID of A.
- f. Attacker responds with 200 ok and the connections between B and Attacker are established Attacker can also record the content of conversation. This is a man-in-the-middle.

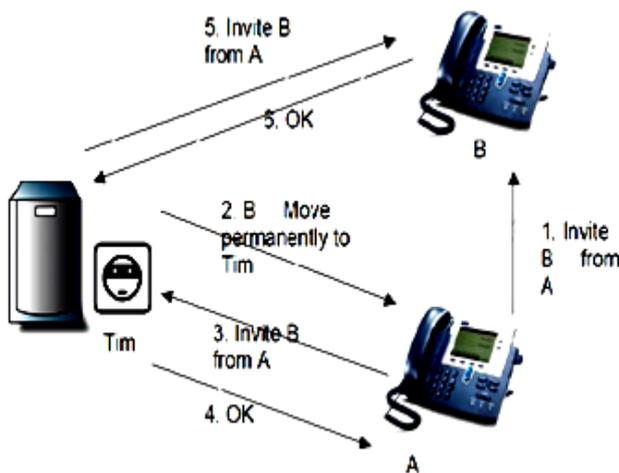


Figure 8: Man-in-the-middle attack

VI. LITERATURE REVIEW

Author	Title	Work
J Jaffar et al	Performance Analysis of VoIP Codecs over MPLS Networks [2009]	Here authors evaluates the quality of service of video transmission on Differentiated Services (Diff-Serv) with Multiprotocol Label Switching (MPLS) network is being simulated. The objective of this paper is to study the influence of the QOS mechanism via DiffServ-MPLS on network parameters such as packet loss, delay and throughput for different video resolutions.
Haniyeh Kazemitabar et al	A comprehensive review on VoIP over Wireless LAN networks [2012]	This paper has demonstrated a survey on VOIP over WLAN, its advantages and challenges and also VOIP capacity over WLAN and the number of calls for different voice codecs and intervals based on IEEE 802.11b standard.
M.A. Mohamed et al	Performance Analysis of VoIP Codecs	Here authors evaluates the performance of different VOIP codecs over the WIMAX network. Simulation results

	over WiMAX Networks [2012]	show that the G.723 is better than codecs G.711, G.726, G.728 and G.729 because it has lower delay and higher MOS, traffic received and throughput.
Mohd Nazri Ismail et al	Performance Evaluation of Audio Codecs using VoIP Traffic in Wireless LAN using RSVP [2013]	This paper deals with VOIP over VPN implementation between open source application and hardware device in campus environment. In this, the evaluation of performance of CPU Utilization, Jitter, Delay, Memory Usage and MOS(mean opinion score).
Elechi Onyekachi et al	Investigating the QoS of Voice over IP using WiMAX Access Networks in a Campus Network [2013]	This paper attempts to identify some of the network performance parameters that service providers will focus on to develop a VOIP over WIMAX communication tool that will serve as a voice communication broadband replacement technology to old circuit switch voice communication.
Mojtaba Seyedzadegan et al	SPEECH QUALITY EVALUATION BASED CODEC FOR VOIP OVER 802.11P [2013]	This paper provides the quick and technical overview of concept, standard, technology and architecture for IEEE 802.16 WiMAX.
D.Ramesh et al	Techniques to Improve Performance of VoIP over 802.11e WLAN [2013]	In this paper authors presented a Media Access Control Protocol that provides the quality of service for VOIP over wlan. In this ,the characteristics of our proposed protocol are No hardware modification of VOIP STA. Backward compability in order to minimize the cost of development no modification of access points.
Preetinder Singh et al	VOIP Over Wimax: A Comprehensive Review [2014]	This paper provides focusing on quality of service scheduling services and performance related metricses such as jitter ,packet end to end delay and MOS(mean opinion score).
Dr. Abdul-Bary R.Sulaiman et al	Performance Analysis of VoIP Codecs over MPLS Networks [2014]	This paper evaluated the performance measures such as delay variation, delay, page response time, throughput and packet drop for different types of traffic such as voice, video, data in their movement in a congested network for both MPLS-TE and Conventional IP Network.
Rahul Singh et al	A Review Paper: Voice over Internet Protocol [2014]	This paper deals with VOIP to a level that allows discussion of security purposes and concerns. In this, VOIP components will include network components, gateway, end user equipment, call processors and two of common architectures.
Lamia Bakri Abd Elhaleem Derar et al	Evaluation of Voice Codecs of VoIP Applications for UMTS [2014]	Here authors the network performance analysis to evaluate the effects of the application of different voice encoder schemes on quality of service of VOIP system which is deployed with the UMTS network.

CONCLUSIONS

VoIP nowadays enjoys the fruits of labours during the past few years and it can be considered a mature technology.

Moreover, behind the concept of a new broadband network that will be the amalgamation of existing and emerging fixed and mobile networks lies the need for network operators to provide new broadband services, as well as the desire of customers to be able to have access to their services from anywhere. This concept is termed Next Generation Network (NGN). ITU-T defined an NGN as a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies, and in which service-related functions are independent from underlying transport-related technologies. A standardized NGN architecture is the IP Multimedia Subsystems (IMS), which was defined by the ETSI and 3GPP. IMS will work with any fixed or wireless network based on packet-switching, including GPRS, UMTS, WiMAX, DSL, etc. Furthermore, IMS builds on SIP in order to ease the integration with the Internet. Therefore, VoIP's future appears bright since a terminal only needs to support IPv6 and SIP. However, traditional telephony systems, H.323 and other VoIP systems can also be integrated with the IMS network through gateways. Notwithstanding the advantages of VoIP, one of the problems that still remains are the provision of QoS guarantees to voice communication over IP networks.

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