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Review: Voip Architecture and Its Applications

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***Abstract*—The advent of internet has become an eminent part of our life as well as the services that it offers has also been increased since a decade. It provides a number of services like video streaming, e- commerce, mailing etc. Out of which Voice over Internet Protocol is one of the most popular and widely used technology of today. A form of communication which is carried out using voice with the help of Internet Protocol (IP) is termed as Voice over Internet Protocol (VoIP).VoIP is prominently flexible technology providing lost cost services to the users, giving it an upper hand over Public Switched Telephone Network (PSTN). This technology offers more robustness and reliability. Two major issues has been encountered in the VoIP architecture as security and quality of service (QoS) which have been emphasized upon and possible attacks have been discussed. This paper gives an insight of the VoIP architecture which has proceeded the legacy of PSTN further. Applications related to this topic are also discussed.**

***Index Terms*—VoIP, SIP, PSTN, LTE, WLAN**

I. Introduction

The circuit switched PSTN have been replaced with the more potential packet switched application layer protocol termed as VoIP which is capable enough of delivering voice as well as multimedia services. VoIP serves the advantageous purpose of generating least packets for call such as in Session Initiation Property (SIP) which utilizes this property of VoIP and uses User Datagram Packet- UDP and Transmission Control Packet- TCP as its transport layer protocol [1]. The two main components of VoIP [2] are:

- End system
- Signaling server

Since VoIP offers a number of services like voice mails, video and voice calls, instant messaging, interactive voice response etc. by utilizing the internet as a communication medium, its popularity has touched the high of the sky. But, the security threats have been encountered due to the medium in VoIP. Since SIP offers weak communication due to the transmission of information in the form of plain text, security is a major concern for which the security based architecture has also been discussed. The standard protocol for signaling i.e. SIP is vulnerable to man- in- the- middle attack, eavesdropping as well as message modification.

As far as the SIP is concerned, there is a need of authentication on gateway server interface [3]. Considering the deployment in real networks, QoS is also a major concern for the transfer of voice over IP. As far as the applications are concerned, the non- real time applications are definitely not vulnerable to problems of jitter or packet loss and delay [4] and offers a great performance and these are the only factors along with bandwidth that can degrade the quality if occurs.

VoIP uses its hand greatly in a number of applications such as in Long Term Evolution (LTE) which has a peak data of 50 Mbps and 100 Mbps for uplink and downlink respectively can be made more efficient by deploying VoIP. When merged with Wireless LAN offers a great benefit by cost saving [5]. VoIP also shares an application in the field of forensics where it has offered a dual streaming approach for improvement of the speech quality. This also offers reduction in the lost packets. Therefore in this paper, the SIP based VoIP architecture in its general form is discussed in Sec III and applications in Sec IV.

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II. RELATED WORK

Several VoIP architectures have been proposed by various authors in order to provide security and QoS to accommodate a security of the user data and to improve the quality which results in reduction in packet loss and various attacks to which VoIP is vulnerable.

As the work carried out in [5] [6] [7] [8] [9] [10] presents various amendments in the VoIP architecture by comparing it with a general VoIP architecture. The author [11] presents a novel method of secure architecture by putting forward the architecture in which the honeypot idea is given to counter various attacks to provide quality and reliability. The QoS architecture has been proposed in [12] which the QoS architectures for VoIP are compared to finalize the best out of all. Basic three VoIP vulnerabilities: integrity, confidentiality and integrity. The proposed scheme of VoIP over LTE and WLAN has been proposed in [12] which presents the application of VoIP in the field of networks as in 3GPP. The idea given in [2] revolves around the VoIP application in LTE which utilizes block preserve scheduling algorithm. Section- III discuss the basic architecture behind the VoIP i.e. the SIP protocol based architecture. The end to end architecture has also been defined which presents a pure VoIP implementation.

III. GENERAL ARCHITECTURE

Since, VoIP requires the signaling protocols to provide services. Hence the services signaling protocols are SIP and H.323 providing high scalability with the ease of integration with other IP based services. The basic architecture of VoIP is based on SIP which includes User agent (UA) and various servers such as proxy server, redirect server and registrar server as shown in Fig. 1.

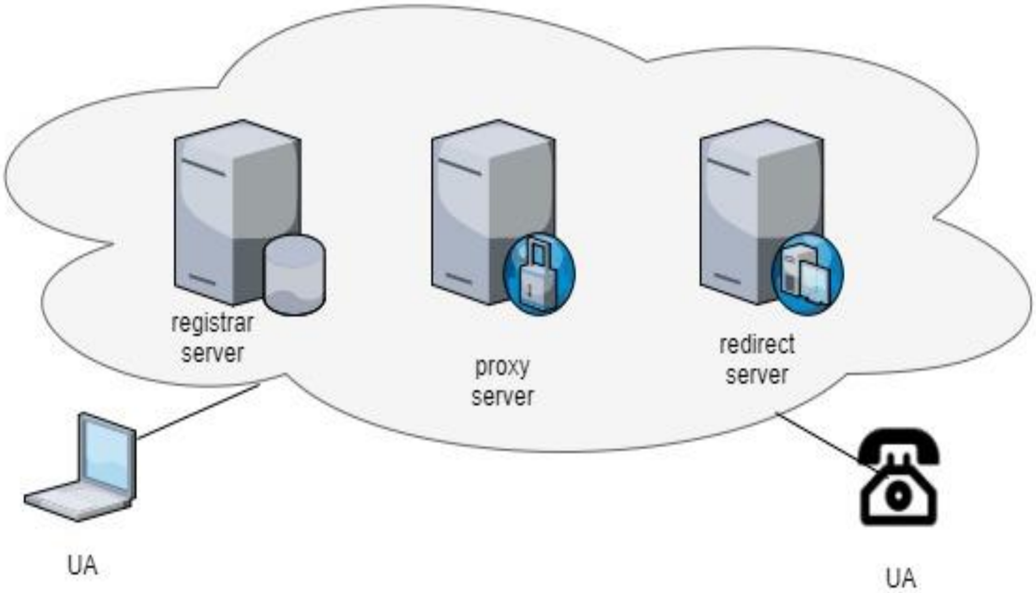

Fig. 1 General SIP Architecture

TABLE I.DESCRIPTION OF SIP COMPONENTS

COMPONENTS	DESCRIPTION
User Agent	• Can be client or server. • Initiates register requests, invite, and bye or cancel requests.
Registrar Server	• Maintains database of locations and UA’s priorities.

Proxy Server	• Receives request from caller. • Forwards to correspondent callee directly or through another server.
Redirect Server	• Provides the information about the next hop server for the caller.

SIP offers reliability and efficiency but on the other hand it is susceptible to various threats in terms of security concerns [5]:

- Lack of cryptographic assurance and message origin authentication leads to *registration hijacking* in which the attacker registers himself and perform modifications in message. Lack of authentication and confidentiality leads to *message tempering* where a malicious server cause changing message characters. As a solution, Elliptic Curve Diffie- Hellman algorithm is used.
- Lack of secure SIP authentication results in *denial of service attack* where the attackers makes a certain requested service unavailable by redirecting many register requests. Hence, the illegal message are discarded using the min- max- fair- share algorithm as a solution.
- In *Man in the middle attack* the attacker spoof the identity of the system to monitor or hijack VoIP traffic. Public key cryptography can be used for its prevention.
- *Parser attack* use distorted messages so as to recover the data and to analyze the behavior of VoIP systems. [5].

IV. VoIP ARCHITECTURE

The actual implementation of VoIP uses IP enable end user equipment as depicted in Fig. 2 [8] like computer or IP phones. These IP phones have the capability to digitize and encode the speech and are connected to LAN over which voice calls can be made. VoIP gateways are used to make PSTN connections.

A. Signalling

VoIP uses signalling which creates and manages the calls in voice and multimedia communication between two or more parties. In accordance with the RFC- 3261, SIP based VoIP contains layer models as: transaction layer, transport layer and syntax and encoding. VoIP follows SIP as a signalling protocol which also facilitates the call set up as described:

When call base on SIP is initiated, the caller firstly registers itself with a suitable VoIP server which behave as a gateway to forward the caller request to the corresponding party. A VoIP session is established on the request of User- A to User- B. with the help of a proxy server the caller the caller sends an *INVITE* for the corresponding User- B containing useful information like caller ID, request line from source to destination contact by virtue of which the intermediate proxy server authenticates the received Invite to User.

Callee responds to the to the Invite from proxy server with an *OK* when the User- B wants to proceed the communication. After receiving the response from the callee, the caller responds with *ACK* to tell that he is also willing to exchange information and hence a VoIP call is established within the two parties. The call is then terminated with *BYE* message sent by any of the user and the call is finally terminating by acknowledging the Bye message from other party.

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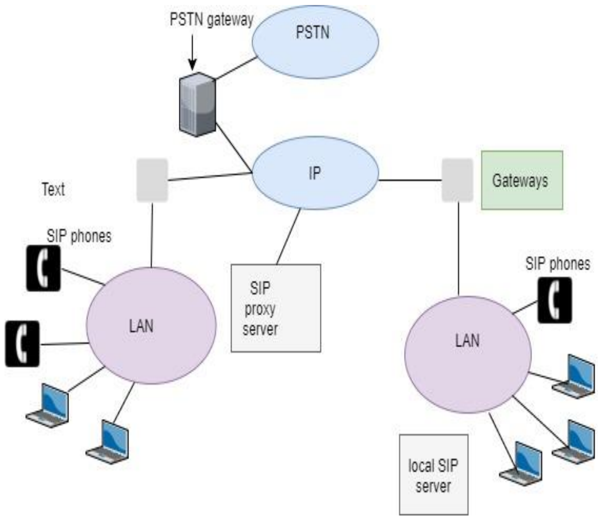

Fig. 2 End- to- End VoIP architecture

V. VOIP APPLICATIONS

With the evolution of technology and advancement in the field of internet, VoIP has constructed its pillars deep in the field of communication.

A. LTE and WLAN

VoIP serves a wide range of applications out of which the usage of mobile VoIP architecture over LTE and WLAN (VALWN) is proposed as shown in Fig. 3 [7]. It is been suggested to VALWN architecture.

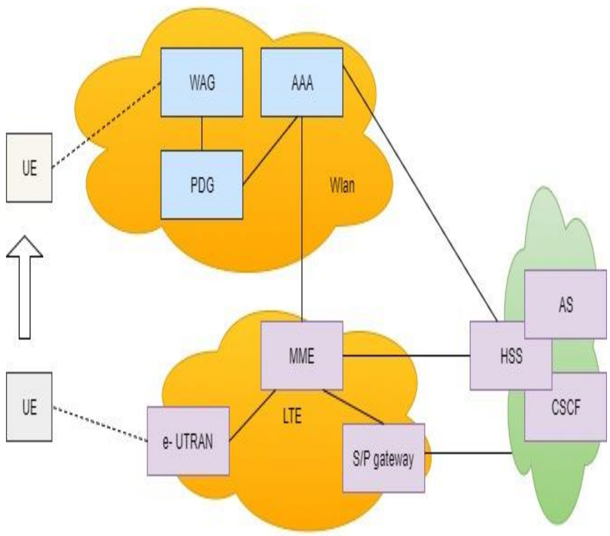

Fig. 3 VoIP architecture (3GPP LTE and WLAN network)

It provides an internetworking model to support VoIP services over the LTE and WLAN where IP Multimedia Subsystem (IMS) acts as a signalling mediator, including the application server to tackle handovers and is the future central network. Application server is responsible for terminating the IMS session such that voice service can be provided. This scheme has pointed out the a vital feature of 3GPP including the voice call continuity

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B. Forensic Applications

One of the advance and novel technique presented as an enhancement of VoIP is the High Definition VoIP (HD- VoIP) used for environmental wiretap applications. This application [3] follows the approach of dual VoIP over 3G/HSPA access network for the improvement of QoS. Dual VoIP technique is used to reduce the packet loss. Considering the 3G- HSPAA networks, they undergo packet loss and delay, reduced radio link quality which are responsible for the degradation of quality of VoIP calls as a solution to which the dual streaming scheme has been proposed [3] which handles the delay and loss of packets by duplicating the single data flow and then sending it through two unalike network access which are guaranteed by two dissimilar telephone operators.

The data flow is divide into two IP flows at the application layer. These IP flows will be regarded with different source addresses but their destination address will be same. The receiver will combine the data flow using sequence number to retrieve correct information by discarding the duplicate packets. Conclusively, it provides dual streaming provides a feasible solution for the improvement of audio signal quality.

C. Traffic handling

A novel QoS aware packet scheduling algorithm has been proposed [2] to handle the real time traffics using RB preserver in VoIP. The proposed algorithm comprises of two layers i.e. the upper layer which utilizes the LTE frame features of multiple sub- frames and the lower layer which opts for the Proportional Fair Algorithm for scheduling flows which are non- real time in nature.

To overcome the challenge of more NRT packets, the upper is reassigns the free resource blocks (RB) of real time flow.

Depending upon the flow type, the scheduler allocated the RBs depending on the type of flow. Proportional fair algorithm is used for NRT which assigns RBs to downlink connections related with User equipment’s to achieve fairness and the Exponential/ Earliest Deadline algorithm is used for RT flow which is a channel aware algorithm to satisfy QoS.

D. Cloud Computing

Cloud computing is a popular technology in the field of networks and wireless communication. This has the ability to change the future of information and communication technology with cloud native service model of ICT. It serves as an application of VoIP as it uses mobile VoIP as a computing paradigm. This approach [4] has the potential to improve security of VoIP. Also, to assure the security of voice communication over IP, QoS is a dire need. To guarantee the QoS, minimum response strategy is satisfied along with the network tuning of delay, jitter and packet loss.

E. Social networking applications

Quality is a crucial parameter in voice communication. It is the collective effect of performance of services hence determine the degree of customer or user satisfaction. As now a days, VoIP has become the most popular source of communication because it offers cheap cost for domestic and international services. This technology has become a necessary and popular part of our daily life because it provides an easy access to the consumer. It supports various VoIP applications such as Viber, Line, and Skype whose major focus is on the voice quality. The VoIP applications such as social networking apps facilitates the user to use it for the voice, text or video communication. Operating systems like MAC, Android, Linux, Windows etc. supports VoIP applications.

F. Digital Signal Processing

VoIP also shares its application in the field of DSP. The proposed scheme [7, 9] shares the digital signal processing on the ground of VoIP. Voice quality plays a major role in any kind of voice communication.

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Voice quality enhancement is a major area of concern in the field of telecommunication so [9] talks about the background noise reduction acoustic echo cancellation for VoIP. Hence speech intelligibility will be boosted and enhanced to reduce the listening efforts. Removal of unwanted echoes and signal noise leads to better voice quality.

Video quality assessment [7] for VoIP evaluates blurring of the video and blocking artefacts caused by packet loss. The video quality is measured in terms of the two parameters termed as blocking artefact score and the weighted sum of blurring score.

VI. Conclusion

The present paper presents the knowledge about the VoIP architecture and its applications in various fields such as DSP, telecommunication, social networking, forensics and data traffic handling, 3GPP networks etc. The VoIP architecture is driven from that of SIP which is the protocol that works behind the voice over IP. The end to end architecture is a general form of VoIP architecture which offers end to end communication. The two major challenges encountered are the security and QoS which have been counter-measured using encryption techniques and algorithms respectively. The future work on these technologies has already been done in [12].

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