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VoIP over Internet Protocol

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Abstract— **VoIP is a technology that digitizes voice packets and enables telephony communication by using internet as a backbone. The broad development of Internet had witnessed the concept of VoIP and the technology is evolving to replace the Public Switched Telephone Network. VoIP is composed of Signalling and Media. The use of Signalling is for controlling communication and it does the call set up, locate users, and tear down sessions. Media is used for transporting the voice packets. A Comparative study can be done for analysing the performance of VoIP over wireless networks. In earlier days VoIP merges over devices like IP Telephony, Palms, PDAs, etc over IP network for communication. Though there are major factors affecting VoIP quality such as delay, jitter and loss it plays a vital role in day-day- life.**

Keywords- **PDA, PSTN, RTP, Telephony, VoIP.**

I. INTRODUCTION

VoIP is a technology that digitizes voice packets and enables telephony communication by using internet as a backbone. The paper focuses on providing voice over ad hoc wireless networks. When transmitting voice data, continuous delivery with limited packet loss rate is of primary importance. The voice and data networks are existing separately, along their separate paths. One of the main reasons is the technology used for voice and data communication. To overcome this problem, voice over wireless networks can be applied.

Many protocols were invented to overcome the network traffics. Protocols like Session Initiation Protocol, Real Time Transport Protocol, H.323 standards, etc., were also developed.

I-A Experimental Setup

VoIP implementation follows H.323 standard. It consists of gateways, terminals, gatekeepers, etc. It is also known as IP Telephony. The protocols used are Session Initiation Protocol discovered by IETF and H.323 by ITU. The data and voice gets encoded which is packetized in the sender side and in the receiver side it gets decoded and depacketized.

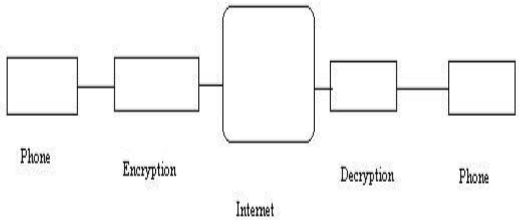

Fig1. Network Communication

Once when a voice is transmitted it is encrypted by the sender with the help of keys and decrypted by the receiver with the same key. Hence a secured communication can be provided. The users can use the techniques like Forward Error Correction techniques, Multiple Description, Layered Coding along with their algorithms.

II. INCREASED USAGE OF VOIP

- Lower cost
- Optimised functionality

II-A Lower Cost

People pay lot for monthly fee for local telephone calls and by using VoIP the price can be reduced. ISD and STD calls get charged to the maximum. But by using VoIP the rate can be minimized. It can be used from computers through Microphones and Speakers.

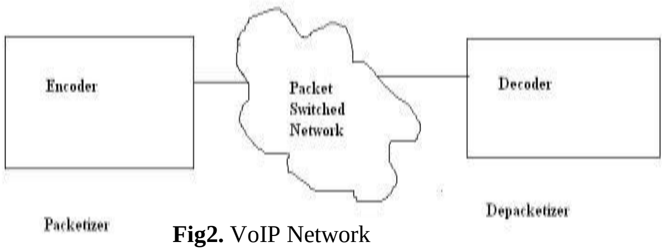

Fig2. VoIP Network

II-B Optimized Functionality

Calls can be connected anywhere on the Internet. Outsources like BPO, KPO can make their job simple by having VoIP. Network Conference can be at low cost. Both the video and audio chats are at low cost. VoIP works either in single or on multiple environments.

III. ROUTING IN VoIP

Router determines whether the packets from source node are affordable to reach the destination node. It can be analysed in two ways.

III-A Performance

Many tools have been invented for network administration but they fail to provide security for both voice and data.
In case of VoIP it provides the following features.

- Poor Quality of speech can be detected.
- RTP associated with voice protocol monitors the call.
- Good QoS can be provided.
- Provides the solution for both voice and data application.
- It finds how call is being hacked / traced by various domains on the network.

III-B Stability in VoIP

The vital role play in VoIP is the bandwidth and the Quality of Service required for each call.

- To ensure stability the nodes or the links must be connected well without any failures.
- The links must be maintained throughout the entire call.
- A Standard Protocol must be used for transmitting the packet over network.
- Hence high speed connection with low packet loss, poor jitter and retransmission must be made.

IV. SERVICES

There are three types of services provided in VoIP.

PC to PC: Service from VoIP client to another client.

PC to Phone: Service from Client to Voice.

Phone to Phone: Service from Phone to another Phone through IP network.

V. DRAWBACKS

1. Network Connection is must for communication.

2. VoIP can be subjected to vulnerable attacks easily.

3. Network Traffic may occur.

4. If there is no power then VoIP calls cannot be connected.

VI. CONCLUSIONS

The foundation block of VoIP is set on two technology - the telephone (PSTN network) and Internet (IP technology). Tremendous works and research on these two fields has made the existence of VoIP possible. This paper has been a brief study on the protocols used to support VoIP technology, the threats that may occur in a VoIP communication and the security measures taken to avoid these threats. Security for a VoIP system should implement concrete security on the internal network. It should be protected from the threats of hostile networks and any threats to the internal network. The load of the VoIP system should be accommodated by the network and the servers involved. Packet loss, delay jitter and throughput all contribute to degraded voice quality. Additionally, because network congestion can occur at any time in any portion of the network.

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