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Analysis of Signaling Protocols and Packet Aggregation Techniques in VOIP Network

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ABSTRACT

Data networking and telecommunication are being transformed by today's developing and ongoing technologies one of which is VoIP (Voice over Internet Protocol). This technology has overtaken simple PSTNs by working through protocols over the internet and thus providing the same functionality as in the telephones in a feasible way through the public internet. This current technology makes us capable of attaining communication anywhere there is internet and thus not requiring any establishment of complex wired network, making it quite cost effective. The rapid growth and deployment of the technology has motivated us to discuss points about VoIP in this paper such as, the technical fundamentals of VoIP, how it is implemented on IP based wireless networks and the merits and demerits of using VoIP over wireless networks. This paper will help to understand the basics of VoIP and its status in the wireless networks.
Keywords: VoIP, PSTN, packet aggregation, wireless networks, MANET

I. INTRODUCTION

Earlier the main method of implementation of telecommunications was through public switched telephone network (PSTN) in which telephone lines are spread over the network and each call session is given a separate channel for passing. This way of providing telecommunication was very effective in terms of quality of service but not efficient in terms of cost as a complex network was required to be laid down for communication. After this, wireless mesh networks came into existence. In this type of network, the nodes are connected in a mesh topology through radio waves. Thus, these networks are more cost effective because they don't require any cable for connection of nodes. These networks are packet switched networks, where the data is sent in the form of packets with no predefined channel as in PSTN but according to the availability of channel and cost of transmission [18].

Due to the effectiveness of wireless mesh networks resulting from packet switching, in addition to sending data packets, networks nowadays, have also started transmitting voice packets over packet switched networks like internet. This technology is called Voice over Internet Protocol (VoIP). In this technology, the voice is sent as packets from one node to another, with the help of routers routing the packets through the shortest available path. This technique is effective in terms utilisation of bandwidth and cost but has got several shortcomings in terms of quality of service. This paper discusses various concepts of VoIP, how it deals with those shortcomings, and various applications of VoIP. The paper is divided into six sections About VoIP, Packet arrangement in VoIP, VoIP protocols, VoIP in wireless networks, VoIP Packet aggregation and finally Conclusion.

II. ABOUT VoIP

A reasonable quality internet can result in acquiring phone service through the internet connection that has been established. A varying percentage of world's population utilizes this methodology (VOIP) as an extension to their conventional phone service since it provides lesser rates than any standard phone firm [1].

The reasons that lead the people to opt for this technology can be:

a. Availability of this facility at lower Cost
b. Greater Functionality

Voice packets to be transmitted across a packet switched network are transmitted according to a rule or procedure which is used for this purpose; one such procedure fulfilling the requirements for the corresponding transportations is VoIP [19].

Using RTP/UDP/IP, VoIP fulfills the purpose. The abbreviated form when expanded, corresponds to Voice Over Internet Protocol, which during transmission of voice, when the voice stream is encoded, transmits this stream of voice as packets over the wireless networks or any Local Area Network [19].

The working of the VOIP can be explained as a two statements process, describing it as a way required to convert the analog signals into digital, so as to send them over the network. The functionality of this process can either be incorporated into the phone itself or in a distinct box like an ATA (Analogue Terminal Adapter).

The initial VoIP telephone calls contain the procedures and rule having similarity to the conventional digital phone network, and further require:

a. signalling
b. setting up of the channel
c. digitization of the analog signals
d. encoding

The transmission of the data packets is done through packetizing the digital information, thus forming IP packets and getting transferred over a packet-switched network instead of a circuit-switched network. The encoding of audio and video with audio codecs and video codecs is done by using some special media protocols that fetch media streams. Figure 1 shows the implementation of VoIP using IP phones [13].

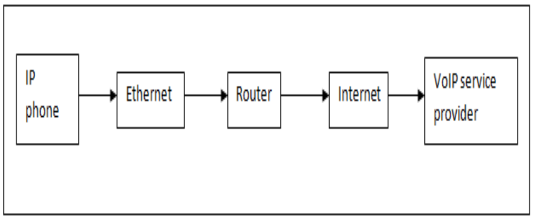

Figure 1. VoIP using an IP phone

An analog signal at the sender is translated to a stream of digital bits by a coder and the decoding of these digital bit streams into analog voice signal is done by a decoder at the destination [19].

The above conversion of the voice analog signal into a digital stream is accomplished by a 'Codec' whose use is often referred to as the 'encoding method'. This helps in performing its main task of compressing the voice traffic at the sender to save the network's bandwidth. The codec is also responsible for the conversion of analog to digital and digital to analog voice signal [19].

The continuation mechanism of VoIP is leading towards a developing technology generation by generation, thus imparting more relevant technical features and solutions. Further, in addition to these VoIP Phones, the facility of VoIP can be accessed on different PCs and also on other compatible gadgets [7].

III. PACKET TRANSMISSION IN VoIP

The VoIP packets are arranged in a manner which is formulated of the following parts (as shown in figure 2) [11]:

1. IP header (20 Bytes),
2. UDP header (8 Bytes),
3. RTP header (min: 12 Bytes),
4. Payload (Variable Bytes)

IP HEADER (20 BYTES)	UDP HEADER (8 BYTES)	RTP HEADER (min: 12 BYTES)	PAYLOAD (VARIABLE)

Figure 2. Structure of the VoIP packets as with reference to IPv4

There is a need for limited end-to-end delay. Therefore, if the packet is short then there will be a shorter delay i.e. if the packet creates a short delay then VoIP call is able to permit a greater network delay, resulting in fewer problems in case the packet is lost. Greater bandwidth is required when the packet is short, whereas the longer packets requiring less bandwidth contain more number of speech bytes but end up creating a longer delay, thus making it hard to recover when any packet is lost.

IV. VoIP SIGNALING PROTOCOLS

In order to transfer audio and video in packet switched networks in the form of packets, many protocols are used simultaneously. The protocol stacks and software tools which are used to achieve the transfer of voice packets over internet are of three types:

Signalling protocols: These are used for setting up as well as tearing down calls by sending signals to the participating terminals in the calling process. Two widely used signalling protocols for VoIP are H.323 and SIP (Session Initiation Protocol).

Media protocols: These are the protocols which are used to actually transfer the audio or video packets in real time between the terminals over the packet switched network without significant delay otherwise the quality of audio or video would be affected. For this purpose, RTP (Real time Transfer Protocol), RTCP (Real time Control Protocol) are used irrespective of the signalling protocol being used in that network.

Codec: Codec means 'Compression and Decompression' algorithms. This is required while transferring of voice packets on IP networks in order to minimise the bandwidth requirement as unlike PSTN (Public Switched Telephone Networks) the bandwidth is not dedicated for each call session. There exist several codecs for achieving this task like G.711, G.729, etc. [8][12].

A. H.323 PROTOCOL SUITE

A protocol suite means a collection of protocols. H.323 protocol suite also contains many protocols which are together used for signalling in VoIP. H.323 protocol stack has four components:

Terminals: Terminals are the devices which are used by the users to place call. They are the user interfaces e.g. mobile phones or telephones.

Gateways: These are the devices which help in translation of data from one format to another i.e. from circuit switched to packet switched data and thus form a gateway between two networks.

Gatekeeper: These are devices whose job is to monitor the terminals and gateways of their zone (i.e. a collection of terminals and gateways). These perform the management of terminals and bandwidth of their zone. These are optional components.

Multipoint control unit: These are used for controlling conference calls i.e. calls between more than two terminals. They have a multipoint controller which controls the signals and multipoint processor which processes the streams of voice data for conference.

H.323 protocol stack is used for both data and voice transfer thus it uses UDP (User Datagram Protocol) which is an unreliable protocol for transfer of voice packets in order to avoid delay due to acknowledgements and TCP (Transmission Control Protocol) which is a reliable protocol for transfer of data packets. It also provides protocols like H.225/Q.931 and H.225 Registration, Admission and Signalling (RAS) which serve the function of call signalling and establishment of call respectively. After establishing call, H.245 is used to send media streams using RTP/RTCP. It also has T.120 protocol for data conferencing [2].

B. SESSION INITIATION PROTOCOL (SIP)

SIP is another VoIP protocol which is widely used nowadays. It is a client-server protocol in which the client sends a request to connect to a user for call which goes to the server which processes the request and sends response to the client for initiating a call session. It is a reliable protocol due to the presence of messages like ACK and thus, it doesn't depend on TCP. SIP is composed of two important components:

User Agents: These are systems that work on behalf of user. They consist of client and server systems which send requests and responses respectively.

Network servers: These are of three types. Registration server receives information about the current location of the client. Proxy server forwards the request to the next hop server which has more information about the user which has been called. Redirect server redirects the address of the next hop server to the client and the client then further communicates directly with the next hop server [2].

V. VoIP IN WIRELESS NETWORKS

Voice Over Internet Protocol is a technology for transferring voice signals over broadband internet instead of the traditional public switched telephone networks. For it to work, the main requirement is that the network should have a wide bandwidth so that it can accommodate significant amount of simultaneous call sessions. Therefore, it was earlier being implemented in DSL, satellite internet, LANs, etc. But, now it is also being implemented in wireless LANs, as they also satisfy the requirement of wide bandwidth. As a result, the voice packets can be now sent over wireless channels using radio waves or other electromagnetic waves. In order to send voice usind packet switched networks, the following process is adopted;

The voice generated at the source is an analog signal which is digitized, compressed, and then organised into packets which are sent over the channel using real time transfer protocol (RTP) to the destination where the reverse process takes place and the packet is converted back to analog voice signal [3][16].

The main advantage of sending voice signals over wireless networks is that it provides mobility. Thus,

user can easily access the network and talk on IP phones even while travelling. But the limitation is that the range over which the signal can be found is limited. Thus, multiple access points are used at fixed distances to provide continuous signals [9].
Some problems in the implementation of VoIP either on wired networks or wireless networks are:

a. Quality of voice and other service parameters like delay, jitter etc. are not very good in comparison to PSTN because there is no dedicated channel for each call session. Thus, there are possibilities of congestion, queuing, and loss of packets [10].
b. Security of the voice data is relatively low as it is sent over the internet and thus can be accessed by any unauthorized person too. This can be avoided to some extent using encryption techniques.
c. The large bandwidth is also not used efficiently due to the retransmission process in case of loss of packets and due to ACK signals. These utilise extra bandwidth, thus the actual available bandwidth for call sessions is reduced.

Problems in implementation of VoIP over IEEE 802.11 (i.e. wireless LAN) are [9]:

1. **Handoff**: Handoff is the phenomenon of a user getting disconnected from one access point as signal strength decreases and getting connected to another access point while moving from one place to another. When the user using VoIP moves from the coverage area of one access point to that of another access point while calling, delay is observed for a small interval of time which results in diminishing voice quality. This happens due the Inter Access Point Protocol (IAPP) in which there are various procedures for scanning, authentication, association etc. which produce the delay. In order to avoid this problem, better algorithm designs are needed [4].
2. **Burst traffic**: This is the phenomenon of blockage of several voice packets for significant time and then releasing them simultaneously to the receiver sometimes with loss of some packets. This also results in delay in transfer of packets which is undesirable [4].

VI. VoIP PACKET AGGREGATION

In order to implement VoIP technology in wireless networks, some problems need to be solved to get efficient use of network and better quality of service. The most efficient technique which can used to optimise the performance of VoIP over wireless networks is packet aggregation [5].

Packet aggregation is the process of aggregating small packets into one large packet to be sent over the network. It is used in wireless LAN (IEEE 802.11) when the overhead associated with a relatively small sized packet is very high which leads to inefficiency of the network as the network may get congested by a very small number of packets. Thus, aggregating small packets into one packet and then sending the packet on the network reduces the overhead of preamble, header, trailer, cyclic redundancy checks (CRC) etc. and this also helps to increase the efficiency of the network by letting the transfer of more information at a time. 802.11 (wireless LAN) uses CSMA/CD protocol in MAC layer for transfer of packets from source to destination. In such a protocol, there exists possibility of packet loss due to collision. But using packet aggregation the possibility of collision is reduced as there is less number of packets in the network now.
Generally, aggregation is of two types:

a. Hop-to-hop aggregation: Here, the packets are aggregated and disaggregated at each hop, until it reaches its destination. Thus, it causes extra delay. This is used when the aggregated packet contains packets for different destination [18].
b. End-to-end aggregation: In this type of aggregation, the packets are aggregated once at source node and disaggregated at the destination node only. This causes less delay. This is useful where all packets in aggregation are going to same destination [18].

In wireless LAN, this process is used in two types:

a. To send the ACK packet: Instead of sending an ACK for each individual packet that reaches the source correctly, using packet aggregation, a block acknowledgement is given now to acknowledge for a block of packets in one go. This saves the time and space in network that was wasted in sending individual ACKs and the waiting period of the sender also. This is known as AM-PDU (Aggregated MAC Protocol Data Unit).
b. To send many data packets as one: Another way of packet aggregation is to send the payload of many packets together as one big packet having single header containing information of all packets. This is known as AM-SDU (Aggregated MAC Service Data Unit).

But there also exist some problems in adopting this scheme like:

a. More delay in packet transmission.
b. Loss is heavy if an aggregated packet is lost.

Research is currently underway to develop mechanisms capable of calculating the required aggregation size depending on the network characteristics to solve the above problems and allow the efficient use of packet aggregation mechanisms [6][14][15][17].

VII. CONCLUSION

In this paper, discussion about the technology called 'Voice over Internet Protocol' (VoIP) has been done. Various concepts behind this technology have been discussed and it lead to the conclusion that in this technology, analog voice signals are digitised and converted into packets and sent over internet instead of the traditional public switched telephone network (PSTN). Discussion about the VoIP packets, the signalling protocols and standards has also been done. Finally, the pros and cons of using VoIP technology over wireless networks like 802.11 are looked at along with ways in which the performance of VoIP may be optimised in wireless networks by packet aggregation. It can be concluded that there is a need for some more research in order to improve the existing algorithms and protocol design for better implementation of VoIP over wireless networks. If implemented with improved methods, it can make the telecommunication process a lot easier and cost effective and highly applicable in health, business and learning environments.

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