Abstract

Voice over Internet Protocol is a way of communication in any network. By using this technology the user can make telephone calls over any IP network. There are some problems which are associated with this are delay in time, security issues, throughput, delay variation. This paper demonstrates the performance of Voice over Internet Protocol in 802.11 wireless networks. The performance of Voice over Internet Protocol can be increased by using enhanced 802.11e standard.

Keywords

IEEE 802.11, IEEE 802.11e, VoIP, WSN.

INTRODUCTION

Wireless Local Area Network

A wireless sensor network mainly consist of large number of low cost, low power, sensor nodes that can be used as an tool for gathering data in various situations. The nodes are densely distributed in a network. These sensors nodes have the ability to communicate either among each other or directly to an external base station (BS). A greater number of sensors allows for sensing over large geographical regions. The main components of the sensor nodes are controller, Transceiver, External memory, Power Source and one or more sensors. IEEE 802.11 Wireless local area network becomes a common network technology. It basically provide to architecture:- Infrastructure and Adhoc Mode. Both these modes are used to build the wireless network. In infrastructure mode there is a single or more Access points to establish the connection between the mobile stations. With the help to these Access points we can extent the network. Within the same network the wireless nodes can mode between the different access points. Adhoc network does not contain access points. The station can communicate directly. The Enhanced IEEE 802.11 mainly support two access mechanism that are distributed coordination function (DCN) and Point coordination function (PCF). PCF is optional whereas DCF is mandatory based on CSMA/CA.

Voice Over Internet Protocol (VoIP)

Voice over IP (VoIP) is a Technology used for the delivery of voice communication over Internet Protocol (IP) networks. It mainly associated with the IP telephony, Internet telephony, broadband telephony, and broadband phone service. It mainly transforms the analog signals into the digital signals over any computer network based on IP. Voice over Internet Protocol is the easiest way to make a phone call through internet by sending packets through packet switched based network. VoIP services mainly allow us to make and receive calls to and from landline numbers for a free of cost. In some VoIP a special type of adapter is used which require computer and a VoIP telephone. VoIP mainly refers to diffusion of voice traffic over the internet based network. The history of VoIP began with conversations by a few computer users over the Internet. Initially, VoIP required a headset to be plugged into the computer, and the participants could only speak with others who had a similar set up. They had to phone each other ahead or sent a text message, in order to alert the user at the other end of the incoming call and the exact time. VoIP technology mainly convert the analog voice into digital data packets that can stored, manipulated, copied, combined with other data and distributed to any device that connects to the IP network. This capability helps to achieve maximum flexibility in the transport or transmission of voice that has been transformed into data.

VoIP Packet Format

1. RTP HEADER:-RTP stands for Real time transport protocol. It is the one of the important parameter for the VoIP application. This protocol is mainly used for the transmission of audio and video. The size of the header is 12 bytes.

2. UDP HEADER:-Its function is to send the data actually to the destination. It is connectionless and does not support any sequence of information or the guarantee of delivery of information.

3. IP HEADER:-It stands for Internet Protocol. The function is this header is to send the data to the actual destination port. It is also the connectionless and does not guarantee to the delivery or sequence of order. It inserts 20 bytes for VoIP packets.
4. DATA: Its size varies between from 20 bytes to 160 bytes.
5. VoIP Codec: The function of VoIP is to convert the analog waveform to the digital form. Various real time applications are GSM 6.10, G.711, and G.729 etc.

QUALITY OF SERVICE

Quality of Service can be defined as the network ability to provide good services to the customer to satisfy it. When the user satisfaction is higher than the QoS is also higher. There are various factors that help in enhancing the quality of the voice service in any network are Packet End to End Delay, Jitter, Load, Throughput, and codec.

1. Delay: The main cause of end-to-end delays in VoIP packets the network congestion that helps to avoid the congestion. Next the encoding used by the sender and the decoding used by the receiver mainly depends on the codec used by the voice system. Thus there is a less delay time in voice service and more VoIP calls are allowed on the network. Delay can be categorized into three categories: delay at the source, delay at the receiver and delay at the network.

2. Jitter: Jitter is the variation in the latency on a packet flow between two systems, when some packets take longer to travel from one system to the other. It constraint the VoIP delivery to arrive at a constant rate and free of echo. To get overcome from this the jitter buffer can be used. The VoIP call is between 0msec and 50msec. Jitter has more negative effect on the voice quality.

3. Throughput: It basically defines the total received packets measured in bits per seconds. Both the packet loss and throughput are inversely proportional. The network which is strong having a lower degree of packet loss.

4. Packet Loss: The packets that are transmitted over the IP network may be lost in the network or corrupted or late. The packets that arrive late at the jitter buffer would be discarded or when there is an overflow in the jitter buffer or router buffer. The packet loss is equal to the total loss occur during the congestion of network.

5. Echo: In VoIP, Echo occurs when a caller at the sender side hears the reflection of his own voice after he talked on the phone or the microphone, whereas the caller does not notice the echo. Echo is the term of the reflections of the sent voice signals by the far end.

COMPARISON BETWEEN THE IEEE 802.11 and 802.11e

IEEE 802.11e is the enhanced version of the wireless network. It has better throughput as compared to the IEEE 802.11 and there is less amount of interference and deployment cost is also very low in this. It has a better response in a shorter range only. There is a less number of packet drop in jitter buffer for IEEE 802.11e network. There is more delay of end to end packets in VoIP of IEEE 802.11. The limitation of IEEE 802.11 is that DCF mechanism does not guarantee any type of quality of service.

CONCLUSION

The voice service is an essential tool in current area. The telephony systems are deployed to meet the customer requirement that are mobility, security, performance, and cost effectiveness. The method of packet switching is adopted so that to improve the efficiency of voice calls. For supporting Qos 802.11 is more powerful than 802.11 which are good for best effort service only. The IEEE 802.11e wireless network decreases the packet end to end delay. IEEE 802.11 has more robustness and stable communication system. Protocol TCP is not well adapted while UDP is more adapted.

REFERENCES


