



Voice Over Internet Protocol Over Wireless Local Area Network: A Review

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Abstract

The use of Voice over Wireless Local Area Network is seeing a meteoric rise in popularity as a result of its simplicity, non-intrusiveness, and cheap cost of implementation, as well as its low cost of maintenance, universal coverage, and fundamental roaming capabilities. Nevertheless, deploying Voice over Internet Protocol (VoIP) over Wireless Local Area Network (WLAN) is a challenging task for many network managers, architects, planners, designers, and engineers. Because of this, there is a need for a guideline to design, model, and simulate the network before it is deployed. In this work, a variety of models, including mathematical, theoretical, statistical, and graphical models, that are used to measure the quality and features of VoIP are discussed.

Introduction

Voice over Internet Protocol, sometimes known as VoIP, is the process of transmitting voice calls via data networks (Nisan et al., 2011). Packet switching is the technology that underpins data networks like the internet, local area networks (LANs), and wide area networks (WANs), which all make use of internet Protocol (IP) (Zhang et al., 2011). This does not mean that Voice over Internet Protocol (VoIP) traffic is sent over the internet; rather, it indicates that the same Internet Protocol is used (Olejniczak, 2009). Speech over Internet Protocol, which is also known as IP Telephony, is a technology that integrates many modes of communication, such as voice, video, and data, into a single network. Because of this, it is now possible to control both data and voice traffic on the same network. These many kinds of IP packetized communication are sent across an IP-based network that is privately controlled (Costello & Lassman, 2004) As a result of the fact that Voice over Internet Protocol (VoIP) technology evolved with various favorable characteristics over the older standard Public Switched Telephone Network (PSTN), VoIP is rapidly replacing PSTN.

In contrast to PSTN, Voice over Internet Protocol (VoIP) is not limited to only making and receiving telephone conversations; rather, it incorporates all other types of communication, such as videoconferencing, voice mail, email, call waiting, call identification, faxes, and so on. Because of its many benefits, Voice over Internet Protocol (VoIP) is progressively replacing the more conventional PSTN. The use of Voice over Internet Protocol (VoIP) has helped break down barriers in international communication by offering an alternative to traditional phone conversations and facilitating easy access to low-cost calls using a computer and the internet. In contrast to the PSTN, which requires a significant investment of cash, the deployment of VoIP networks is quite affordable. The vast majority of calls made using VoIP services may

be made at no cost, regardless of the distance. It is not dependent on distance or location; it does not require any additional cabling; and a virtual number enables you to make calls from anywhere in the world as long as there is an available broadband connection. VoIP is inexpensive and simple to use, with the ease of upgrading; it is not dependent on distance or location; it does not require any additional cabling. There are many different communication functionalities that may be implemented with VoIP. The service is also adaptable in the sense that it enables users to connect traditional landline phones to the VoIP network. However, owing to degradations in the network, the quality of a communication conducted through VoIP is worse than that of a conversation conducted over a fixed-line. According to Rodriguez et al. (2018), the use of MIMO technology, which uses multiple transmit and/or receive antennas, helps to increase the quality of speech signals. A wireless system's information capacity gain may be boosted by deploying several transmit and/or receive antennas in a communication system. This can be done either independently or in combination (Oluwafemi, 2015).

A Wireless Local Area Network, often known as WLAN, is a network that links two or more devices across a greater distance utilizing wireless communication networks such as radio or infrared signal. This kind of network also provides a connection to the larger Internet by way of an access point. WLANs have exploded in popularity because they enable users to roam freely inside a certain location while remaining connected to a network even while they do so. It enables high-speed data connectivity in confined spaces like workplaces, residential and commercial buildings, and other such locations. The IEEE 802.11 standards serve as the foundation for wireless LANs, which were formerly known as LAWNs (Local Area Wireless Network). The use of voice over internet protocol (VoIP) is seeing widespread adoption of WLAN technology. The implementation of Voice over WLAN is simple, low-cost, and unobtrusive; it provides universal coverage; has a low maintenance cost; and has the fundamental roaming capabilities (Ganguly & Bhatnagar, 2008). The fact that the network is not well prepared to fulfill the quality of service (QoS) standards of VoIP is one of the drawbacks of using a wireless local area network (WLAN) (Ifijeh et al., 2015)

The implementation of voice over data networks has proven to be very difficult for companies, making it extremely difficult for these networks to properly be merged and unified into a single network (AlAlawi & Al-Aqrabi, 2015). This article provides an in-depth analysis of the technique involved in the step-by-step implementation of VoIP over a WLAN, which may be used in any organization moving forward. In this article, you will find in-depth descriptions of the components, properties, and needs of VoIP.

Related Work

Since the inception of VoIP, a number of studies have been conducted on its implementation. Even though Voice over Internet Protocol (VoIP) over wireless local area networks (WLAN) is one of the newest applications in high-speed packet-switched networks, many important research projects are still being conducted in order to guarantee a high Quality of Service.

This article presents an IP Telephony framework in order to highlight the common restrictions that apply to using an IP Telephony delivery platform for interactive content applications. The framework integrated Voice over Internet Protocol (VoIP) technology with support for content delivery by using a standard unified communication device in conjunction with networking devices such as routers and switches. Using the OPNET network simulator, Salah & Alkhoraidly (2006) developed a detailed simulation technique for installing an end-to-end VoIP component from sender to receiver. This approach was applied in both of their publications.

Ign'acio de Matto et al. (2010) and Komolafe & Gardner (2003) both detail the creation of a practical simulation tool that can be used to create and analyze communication networks. In the articles, the implementation techniques and applications of the developed simulator

enabling VoIP were the primary topics of discussion. Instead of modeling aggregated traffic, the new model in (Ign'acio de Matto et al., 2010) focuses on simulating the behavior of individual users. This represents a significant step forward. Research in this area looked towards the modeling and aggregation of VoIP streams (Komolafe & Gardner, 2003).

There was discussion of network design standards, needs, and problems that arose during the development of a VoIP network in the. The papers analyzed the usefulness of Internet Protocol (IP) telephony over the internet as well as searching for other means that may incur additional communication overhead cost committed to reducing any risk associated with IT investment through the appropriate design, development, configuration, and implementation of the system.

Several mathematical models for measuring the qualities of VoIP calls, such as the Mean Opinion Score (MOS), the E-model, and the Perpetual Evaluation of Speech Quality (PESQ) score, were analyzed in (Beuran, 2006; Tsompanidis et al., 2010; Chhabra and Singh, 2011). These studies were cited in (Beuran, 2006; Tsompanidis et al., 2010; Chhabra and Singh, 2011). We contrasted the results of our experimental speech quality measurements taken in wired and wireless environments with the results of our mathematical speech predictor. According to the findings, WLAN QoS parameters have a high degree of fluctuation in real-world contexts, which has a major impact on the performance of applications.

To assess the performance of VoIP services using various Codecs, a theoretical technique was proposed in (Cao & Gregory, 2008; Segar, 2003; Anouari & Haqiq, 2012) to model an aggregated flow of VoIP connections on a packet basis. This flow was formed owing to tele-traffic factors. Both the one hybrid UMTS in (Cao & Gregory, 2008) and the Network Simulation 2 in (Cao & Gregory, 2008) were used in the simulations that were run (Segar, 2003; Anouari & Haqiq, 2012).

On the basis of statistical studies of VoIP traffic in the multiplexed process, the development and validation of models for the multiplexed process were provided (Xi et al., 2010; Dang et al., 2004; Gustafson & Lindahl, 2008; Hassan et al., 2005). The models are versatile enough to be used for the simulation of any IP network design, whether wired or wireless. Importing the data that was obtained from the post processing into the MATLAB program allowed the analysis to be carried out that was described in (Gustafson & Lindahl, 2008).

The findings of a Quality of Service (QoS) research for voice over internet protocol (VoIP) service carried out over 3G WCDMA networks were given in (Cuny & Lakaniemi, 2003). The research gives thorough features and precise modeling of VoIP Quality of Service (QoS) metrics. The jitter and packet loss behavior of VoIP traffic was analyzed using network measurement and simulation findings.

Hassan et al. (2005) also established a general hierarchical model for multimedia traffic sources, which was then utilized to build VoIP, video, and web traffic sources. This model was used to generate the traffic sources. The models may also be used to assess the Quality of Service (QoS) of the various apps operating under a variety of network conditions. In order to build traffic models, a traffic source design tool known as Traffic Source Modeler (TSM) was used. These models were then utilized by a simulation program known as Distributed Hybrid Simulator (DHS).

A hardware-fitted simulation model was built in order to establish the maximum number of voice over internet protocol (VoIP) conversations that a wireless local area network (WLAN) of type IEEE802.11 can handle (Feign, 2000). MATLAB is used to do an analysis on two capture files, each of which contains the inter-arrival timings and the unique fragment identifier for each packet. In order to expand the VoIP measurement findings beyond what could be accomplished with real hardware, the OPNET model was used in the process of developing and simulating the wireless network.

An method to modeling and simulating a P2P VoIP system based on graph transformations was suggested by the authors of the Khan, (2010) study. Graph transformations are a visual rule-based transformation that has recently been backed by facilities for stochastic modeling and simulation. The architecture of a P2P network was conceptualized as a graph, with the vertices of the graph standing in for network nodes and the edges of the graph standing in for connections between the nodes. A recently developed program called GraSS (Graph-based Stochastic Simulation) was used in order to carry out simulations of the model.

The majority of the previous research work that was done consisted on modeling or simulating VoIP networks, traffic, and performance using NS-2 or UMTS. In terms of the determination of the Quality of Service, the deployment of a VoIP network in convergence with an already existing Public Switched Telephone Network has not yet been given sufficient investigation.

Ethernet Based VoIP Network

The ability to communicate effectively is essential to the success of any company, organization, or society. The introduction of voice over internet protocol (VoIP), which combines speech and data transmissions on a single network, has the potential to improve the quality of communication. The growing number of mobile phone lines throughout the globe is evidence that the next generation of technology is transitioning toward IP network-based technologies such as VoIP (Nisar et al., 2011). It provides the advantages of large cost reductions, a rise in income, and improved service to customers (Olejniczak, 2009).

Although the majority of VoIP deployments are based on Ethernet and operate on the IP protocol, the use of VoIP over Wireless Local Area Networks (WLAN) is becoming more common. It is possible that Voice over Internet Protocol (VoIP) over Wireless Local Area Networks (WLAN), which sends IP-based telephone conversations via WLANs, will become a prominent application in conjunction with the Third Generation (3G) (Nisar et al., 2011).

Figure 1 depicts a VoIP network that is built on Ethernet. The network of the company is referred to in the next sentence as the Corporate Network. In the vast majority of companies, the data network and the voice network are kept completely separate.

An Internet Protocol Private Branch Exchange (IP PBX) that is capable of executing the VoIP protocol has to be installed into the organization's network server before it will be possible to properly implement VoIP. Every workstation, laptop, desktop, and personal digital assistant (PDA) that is currently connected to the network will have a Session Initiation Protocol (SIP) phone, a VoIP protocol phone, and other codecs loaded. The router makes it possible for portable mobile devices to communicate wirelessly with one another. In addition, VoIP gateways will be linked to the network in order to allow conversion from the circuit switched network PSTN to the packet switch network VoIP. This will make it possible for various technologies to communicate with one another. On top of that, telephone adapters are connected to the IP-PBX and then configured so that communication may take place between analogue phones and SIP phones.

VoIP Over WLAN

The capability of wireless access networks to both enable the mobility of VoIP users and provide extensive network coverage is one of the most significant benefits offered by such networks. Despite these benefits, there is a possibility that the changeover may not be easy due to the intrinsic qualities of WLANs (Beuran, 2006). Different wireless access networks provide varying degrees of functionality for voice over internet protocol (VoIP). However, IEEE 802.11-based Wireless LAN (WLAN) and IEEE 802.16-based WiMAX are two key wireless access technologies that are well suited to enable VoIP service for customers. WLAN and WiMAX are both based on IEEE 802.11.

Short-range coverage is often provided by IEEE 802.11-based Wireless LANs, such as those found in businesses and households. These networks typically have a bandwidth of up to 54 Mb/s in the 5GHz band and use an OFDM Modulation scheme. WiMAX is a standard for fixed broadband wireless metropolitan access networks (MANs) that allows wide-range connectivity to mobile users. Its bandwidth can reach up to 75 Mb/s. The frequency range it covers is 10-66 GHz (Beuran, 2006). Because of their usage as packet-switched IP networks, both of these technologies are recognized as viable options for enabling voice over internet protocol (VoIP).

VoIP Over WLAN Equipment

Wireless access points are the most important elements in establishing and maintaining wireless connection (AP). Access points are devices that service numerous stations directly and operate as a bridge between the wired network and the IEEE 802.11-based wireless network. They consist of a radio transceiver, communication and encryption software, and a wireless Network Interface Card that is based on the IEEE 802.11 standard (NIC). A solitary access point in an open area that is devoid of obstructions is capable of covering a circle with a diameter of about 100 meters. Because the quality of the signal and the speed of the connection both suffer proportionally with the distance between the computer and the access point, big workplaces sometimes make use of many access points with ranges that overlap. In order to make VoIP calls via a WLAN connection, you will also need VoIP phones. Recent advances in technology have resulted in the production of mobile phones that are both GSM and WLAN capable. These phones can now make and receive VoIP calls. Call servers are also required in order to successfully handle several calls taking place at the same time and to successfully create connections. Call servers provide duties that are analogous to those of the PSTN switch.

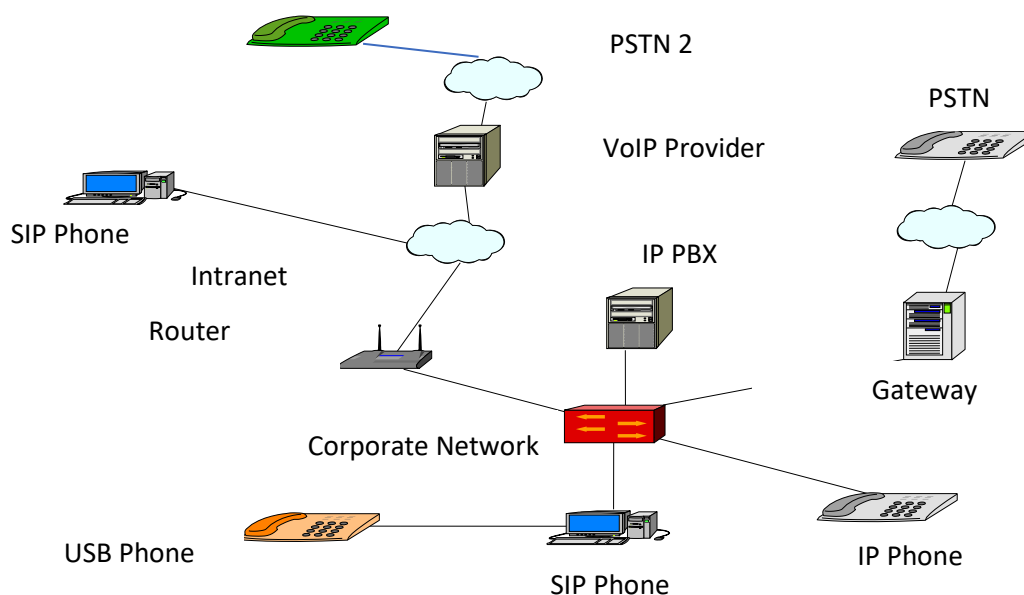


Figure 1. An Ethernet -Based VoIP Network

VoIP Over WLAN Protocols

One of the components that makes it possible for end users to connect with one another is the VoIP protocol. Communication and the manner in which network traffic is packed for transmission across a network are governed by a set of rules known as protocols. They specify the procedures that users in a communication session are required to follow, based on the service that the end user demands, and they do this for each individual communication session. The protocols are designed to establish voice calls that proceed all the way to their destinations (Costello & Lassman, 2004; Cao & Gregory, 2008) Signaling Protocols and Media Transport

Protocols are the two categories of protocols that are relevant to voice over internet protocol (VoIP).

Signaling protocols provide the establishment, management, and termination of sessions, making it possible for call information to be sent beyond the confines of individual networks. These protocols are used to carry out auxiliary functions that are associated with the process of establishing and sustaining calls (Costello & Lassman, 2004). SIP and H.323 are the two signaling protocols that are used the most often.

The second kind of protocol that VoIP employs for the transfer of voice traffic is known as the Media Transport Protocol. The real task of transporting the digitized speech data in the form of packets is the responsibility of the protocol. It guarantees that voice packets will travel successfully via the network. The Real-time Control Protocol (RTCP) and the Real-time Transport Protocol (RTP) are now the most widely used transport protocols (RTCP).

VoIP Over WLAN Codecs

A voice codec is one of the most important parts of a voice over internet protocol (VoIP) system. It transforms the input speech signal into digital form, send the signal to the receiver and rebuild the original voice signal. The waveform is sampled at predetermined intervals by the codec, and each sample is assigned a value that the codec creates. Either 8000 times per second at a sampling rate of 8kHz or 16 000 times per second are samples collected (16kHz sampling rate). After that, these values are quantized in order to convert them into discrete-finite values that may be expressed using bits. This mapping process is called quantization.

The following algebraic relationships are self-evident:

$$f_s = 8fbps \quad (1)$$

$$T_{SI} = \frac{N_S}{f_s} ms \quad (2)$$

$$f_{SI} = \frac{1}{T_{SI}} = \frac{f_s}{N_S} Slps, \quad (3)$$

where,

- f sampling rate (Bps),
- f_s (codec) bit rate (bps),
- N_S (codec) sample size (Bytes),
- T_{SI} (codec) sample interval (ms),
- f_{SI} sample intervals per second (Slps).

G.711 Codec

The G.711 codec is the most widely used one that is specified by the ITU-T Recommendation. PCM, or pulse code modulation, is applied to the voice frequencies, and the data transfer rate is 64 kbps. The μ -law and the A-law algorithms are used to do the analysis on it. The μ -law procedure is used, in which the input variable x is quantified with 14 bits of uniformity, and then it is converted using a memoryless function $f(x)$ that lowers the amount of inaccuracy caused by voice distortion. An example of this may be found in the following paragraph. (Cincunegun, 2011);

$$f(x) = A \frac{\ln(1+\mu \frac{|x|}{A})}{\ln(1+\mu)} \operatorname{sgn}(x), |x| \leq A \quad (4)$$

$$f^{-1}(y) = \frac{A}{\mu} \left[\exp\left(\frac{\ln(1+\mu) \cdot |y|}{A}\right) - 1 \right] \operatorname{sgn}(y), |y| \leq A, \quad (5)$$

where A is the input magnitude's peak and μ is a compression control degree. To decode such output, the inverse function is Real implementation for G.711 adopts a linear approximation through tables with $\mu = 255$ where only 8 Most Significant bits (MSB) are taken into consideration, resulting in a bit rate of 64 kbps at 8Khz.

Wideband and Multirate Codec

In order to facilitate the adoption of voice over internet protocol (VoIP) over broadband access networks, which have significant levels of accessible bandwidth, wideband and multirate codecs were created. The sound quality produced by wideband codecs is superior since they have a greater sampling rate (16 kHz). The most widely used category of wideband codecs is also offered with multirate adaptation, which enables them to transmit data at a low bit rate in addition to a high bit rate. These codecs guarantee that they may be applied to any underlying network state, and they also avoid the need for transcoding when the voice is routed from a network with a high bandwidth to one with a low bandwidth. Adaptive Multi-rate Wideband (AMR-WB) and Speed are the names of the two most current wideband multi-rate codecs to be developed.

VOIP Quality of Service

The quality of service, often known as QoS, is an extremely important component of VoIP systems. There are substantial performance concerns that have a negative impact on the voice quality of the network, despite the fact that the cost of establishing VoIP is far lower. Latency, jitter, packet loss, codec, bandwidth, throughput, voice data length, and de-jitter buffer size are some of the metrics that may be used to evaluate the quality deterioration that might occur across a particular connection.

The term "latency" refers to the cumulative amount of time that a packet must wait as it moves from its origin to its final destination. According to the International Telecommunication Union (ITU), one-way, end-to-end telephone applications in echo-free settings should have a latency of less than 150 milliseconds to assure customer satisfaction (Wu, 2008). Voice coding method, queuing algorithm of communication equipment, and variable delay caused by external variables such as network conditions, weather, etc. are the three components that are responsible for delay.

The term "jitter" refers to the variation in delay that is seen across many packets that are part of the same end-to-end connection. It has an impact on the quality if it is more than the maximum value that is permitted, which ranges from 0 to 50 milliseconds (Wu, 2008; Beuran, 2006). A jitter buffer is used to keep the packets for a time and then release them at a steady rate to the application so that it can play them. This helps to reduce the amount of jitter that occurs. Following is a calculation that may be used to determine the difference in arrival times between consecutive voice packets:

$$\begin{aligned} J(K) &= (R_K - S_K) - (R_{K-1} - S_{K-1}) & (6) \\ &= (R_K - R_{K-1}) - (S_K - S_{K-1}) & = IAT(K) - IDT(K) \end{aligned}$$

$$IAT(K) = J(K) + IDT(K), \quad (7)$$

where

S_K = departure time in RTP timestamp units

R_K = arrival time in RTP timestamp units

K and $K - 1$ two successive packets

$IDT(K)$ = inter-departure time

$IAT(K)$ = inter-arrival time

Loss of data packets along the data line leads to the phenomenon known as "packet loss," which drastically lowers the quality of the speech transmission. When there are poor channel conditions, packet loss may happen in a wireless network. This happens when attempts to send the packet to the receiver via retransmission or error recovery mechanisms are unsuccessful. Early research indicates that the acceptable range for packet loss rates is between one and three percent, and that the voice quality will no longer be acceptable when the packet loss rate is more than three percent (Wu, 2008).

A time dependence may be seen in the sporadic character of the packet loss. If packet n is lost, then there is a greater chance that packet $n + 1$ is also lost. If both packets are lost, then the transmission was unsuccessful. Because of this, there is a connection between successive packet losses, which ultimately leads to a behavior characterized as bursty packet loss. This temporal dependence may be adequately approximated by utilizing the 2-state and 4-state Markov chain that are described further down;

$$P = \begin{bmatrix} S_1 & S_2 & \dots & S_m \\ S_1 & S_2 & \dots & S_m \\ \vdots & \vdots & \ddots & \vdots \\ S_1 & S_2 & \dots & S_m \end{bmatrix}, \quad (8)$$

where

$S = S_1, S_2 \dots S_m$ is the m state of an m -state Markov chain

P_{ij} = the probability of the chain to pass from state S_i to S_j

P = Transition matrix, the probability of transitions between states.

VOIP over WLAN performance Issues

This section briefly highlights some root causes of performance issues in delivering VoIP over WLAN.

The Channel Access Delay (CAD) is the amount of time that elapses between the moment that a data packet reaches the end of the transmission queue and passes through the channel and the amount of time that elapses before the data packet is successfully received at the station that it was intended for. It is a consequence of the amount of time required to get access to the channel and release the awaiting VoIP packet into the atmosphere. The Distributed Coordination Function, or DCF, that is used by IEEE 802.11 was not intended to give latency assurances when it was created.

Because of the qualities that are unique to the DCF protocol, the amount of time it takes for a packet to be sent might vary quite a little (Ganguly & Bhatnagar, 2008). There are two possible explanations for this varying latency. The media access controller (MAC) waits until the channel is idle in the first scenario. The amount of time spent waiting is determined by the size of the packet that is presently being broadcast. In the second case, if there is any collision, the MAC will wait for an arbitrary period of time before resending the VoIP packet. This happens in the event that any collision occurs. As a direct result of this issue, a single VoIP packet may have an unknown amount of delay when traveling from an access point (AP) to a client station or vice versa.

The research presented in Sakurai & Vu's (2007) paper used a more straightforward analysis that was based on a one-dimensional Markov chain as opposed to a two-dimensional Markov chain analysis. An equation for the access latency was generated from the packets of a particular station, and its essential components are detailed in the following paragraphs:

$$D = A + T \quad (9)$$

where,

D = the random variable representing the access delay

T = the random variable representing the channel

occupancy of the transmitted packet

A = the random variable representing the sum of durations

of collisions and back-offs involving the tagged

station, and the durations of successful transmissions

and collisions by non-tagged stations.

The number of backoff periods depends on the number of retransmissions, hence the value of A strongly depends on the number of retransmissions. The number of retransmissions before success obeys a truncated geometric distribution, so that the probability of i retransmissions is ηp^i , where $\eta = (1 - p)(1 - p^k)^{-1}$

Therefore:

$$A = A^{(i)} \text{ with probability } \eta p^i, 0 \leq i \leq k - 1 \quad (10)$$

The generic component $A^{(i)}$ consist of i collisions, $i+1$ backoff intervals and their associated interruptions. Therefore;

$$A^{(i)} = \sum_{j=0}^i B_i^{(j)} + \sum_{j=1}^i C_{ij} \quad (11)$$

where,

C_{ij} = the channel occupancies of collisions involving the

tagged user

$B_i^{(j)}$ = represents the backoff intervals and their interruptions.

When there are more nodes in close proximity to one another, a phenomenon known as co-channel or internal interference may take place. Interference is caused by ongoing traffic that is supplied on the same channel that the VoIP is connected to its AP, whether it be given by neighboring APs or by the same AP. Internal interference from other continuing VoIP conversations or data traffic may have a major influence on the quality of any given ongoing VoIP connection, particularly if the interference comes from both. The active flow of packets from neighboring access points in the same channel might cause a collision of received packets or a slowdown in the sending rate. Since of the collision of packets at the receiver, the latency will be increased because the packets will need to be resent. Because of the decrease in the sending rate, the packet will have to wait in a queue at the sender and will experience extra lag time before it can be sent into the air.

The quality of the wireless medium is reflected in the signal-to-interference and noise ratio, which is generated by interference from external sources such as multipath fading effects, microwaves, neighboring or parallel overlap channels, and so on (SINR). It is possible to lessen the impact by doing careful planning and design work on the network before it is deployed. Calculations are done off-line to determine the optimal access point placements, transmitted wattage, and channel allocations in order to achieve the highest possible level of performance. Additionally, this ensures that adequate capacity and complete coverage are provided.

During the handoff between access points (APs), the wireless communication is lost, which causes an ongoing VoIP conversation to be interrupted. This disruption in connectivity results in the loss of connectivity. If just a little amount of time passes between handoffs, the packets

may be buffered and resent, which will result in an increase in jitter. When the time of the handoff is too long, a surge of VoIP packets will be dropped without any possibility of recovering them.

Another factor that might influence the level of pleasure of users is security. The majority of wireless local area networks (WLANs) encrypt the data that is sent over their networks using Wired Equivalent Privacy (WEP). WEP is not particularly secure since it is readily broken, which presents a significant challenge for commercial users (Beuran, 2006). There is also a solution known as Wi-Fi Protected Access (WPA), which is an alternative to WEP. WPA is intended to prevent any unauthorized user from accessing the network. Constructing Virtual Private Networks, sometimes known as VPNs, or encrypting data using SSL are two further options.

However, for voice over Internet Protocol (VoIP) over wireless local area networks (WLAN), the IEEE802.11 Medium Access Control (MAC) protocol outlines the technique for communication via wireless stations (Ganguly & Bhatnagar, 2008; Beuran, 2006). The Several Access Controller (MAC) protocol assures that multiple stations may use the same wireless channel to send data without interfering with one another. When it comes to accessing the shared wireless channel for the purpose of data transmission, there are two approaches available.

The first option is known as the Distributed Coordination Function (DCF), and it is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol. The protocol is a distributed protocol, which means that it does not need a centralized body in order to coordinate channel access. Because it provides coordination but does not support any kind of priority access to the wireless medium, DCF only offers best-effort service. It does not have a mechanism to offer better service for real-time multimedia traffic in comparison to data traffic. This is because DCF only offers coordination (Beuran, 2006).

The Point Coordination Function (PCF) for the second mode helps to accommodate time-sensitive traffic flows. Because PCF is a centralized protocol, its implementation necessitates the presence of a centralized coordinator entity that is referred to as a point coordinator. Point coordinators are located inside of access points and are responsible for determining which stations are allowed to broadcast within a specific window of time.

Track for Future Work

The adoption of voice over internet protocol (VoIP) over wireless local area networks (WLAN) is growing; nevertheless, the quality of service continues to be a problem; as a result, there is still room for improvement in terms of the quality of service provided by VoIP over wireless networks. It is possible to do more research on methods of lowering channel access latency on the network, as well as internal and external interferences, jitter, and packet loss, etc., all of which may contribute to an improvement in speech quality sent over the network.

This work proposes to model and simulate the deployment of VoIP over WLAN using a suitable network simulation tool such as OPNET. While lots of research works are currently being conducted on the modeling and simulation of VoIP over Ethernet, this work proposes to model and simulate the deployment of VoIP over WLAN.

Conclusion

This work reviewed VoIP over WLAN; it briefly analysed equipment required for the deployment, protocols that will enable communication of voice through the wireless data network, codecs for compression and quality of service issues. The work also presents mathematical models to ascertain the quality of service of the network. Security challenges were also enumerated briefly and solutions were proffered.

References

- AlAlawi, K., & Al-Aqrabi, H. (2015, February). Quality of service evaluation of VoIP over wireless networks. In *2015 IEEE 8th GCC Conference & Exhibition* (pp. 1-6). IEEE.
- Anouari, T., & Haqiq, A. (2013). Performance analysis of VoIP traffic in WiMAX using various service classes. *arXiv preprint arXiv:1308.0223*.
- Beuran, R. (2006). VoIP over Wireless LAN survey. *Internet Research Center Japan Advanced Institute of Science and Technology (JAIST)*, 1-40.
- Cao, J., & Gregory, M. (2008, December). Performance evaluation of VoIP services using different CODECs over a UMTS network. In *2008 Australasian Telecommunication Networks and Applications Conference* (pp. 67-71). IEEE.
- Cincunegui, I. L. (2011). *Quality of Service for VoIP in Wireless Communications* (Doctoral dissertation, Newcastle University).
- Costello, R., & Lassman, J. (2004). IP Telephony for Enterprise Networks: Technology Overview. *Technology Overview*, 21.
- Cuny, R., & Lakaniemi, A. (2003, April). VoIP in 3G networks: An end-to-end quality of service analysis. In *The 57th IEEE Semiannual Vehicular Technology Conference, 2003. VTC 2003-Spring*. (Vol. 2, pp. 930-934). IEEE.
- Dang, T. D., Sonkoly, B., & Molnár, S. (2004, June). Fractal analysis and modeling of VoIP traffic. In *11th International Telecommunications Network Strategy and Planning Symposium. NETWORKS 2004*, (pp. 123-130). IEEE.
- Ganguly, S., & Bhatnagar, S. (2008). *VoIP: wireless, P2P and new enterprise voice over IP*. John Wiley & Sons.
- Gustafson F. & Lindahl, M. (2008). *Evaluation of Statistical Distribution For VoIP Traffic Modeling*. B.Sc. project, Dept. of Economics and IT, Sweden.
- Hassan, H., Garcia, J. M., & Brun, O. (2005). Generic modeling of multimedia traffic sources. *Proc. HET-NETs*, 5.
- Ifijeh, H. A., Idachaba, F. E., & Oluwafemi, I. B. (2015, July). Performance evaluation of the quality of VoIP over WLAN codecs. In *Proceedings of the world congress on engineering* (Vol. 1).
- Khan, A., Heckel, R., Torrini, P., & Ráth, I. (2010, June). Model-based stochastic simulation of P2P VoIP using graph transformation system. In *International Conference on Analytical and Stochastic Modeling Techniques and Applications* (pp. 204-217). Springer, Berlin, Heidelberg.
- Komolafe, O., & Gardner, R. (2003, April). Aggregation of VoIP streams in a 3G mobile network: A teletraffic perspective. In *2003 5th European Personal Mobile Communications Conference (Conf. Publ. No. 492)* (pp. 545-549). IET.
- Nisar, K., Md Said, A., & Hasbullah, H. (2011). A Voice Priority Queue (VPQ) Fair Scheduler for the VoIP over WLANs. *International Journal on Computer Science and Engineering (IJCSE)*, 3(2), 506-518.
- Olejniczak, S. P. (2009). *VoIP deployment for dummies*. John Wiley & Sons.
- Oluwafemi, I. B. (2015). Hybrid Concatenated Coding Scheme for MIMO Systems. *International Journal of Electrical & Computer Engineering (2088-8708)*, 5(3).
- Sakurai, T., & Vu, H. L. (2007). MAC access delay of IEEE 802.11 DCF. *IEEE Transactions on Wireless Communications*, 6(5), 1702-1710.

- Salah, K., & Alkhoraidly, A. (2006). An OPNET-based simulation approach for deploying VoIP. *International Journal of Network Management*, 16(3), 159-183.
- Wu, D. (2008). *Performance Studies of VoIP over Ethernet LANs* (Doctoral dissertation, Auckland University of Technology).
- Zhang, J., Chan, S., & Fang, X. (2004). Enterprise User Adoption of VoIP. *School of Computer Science, Telecommunications, and Information Systems. Chicago, DePaul University*, 10.