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A Survey on Enhancing the QoS through voice Quality for Voice over Wireless LANs (VoWLAN)

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Abstract— Voice over Internet Protocol (VoIP) is one of the important technologies that allow voice transmission over the IP network. Various voice codec are available for VoIP as this is a rapidly changing technology. It can be an effective renewal for the traditional telephone systems (PSTN) because of extreme utilization of its sources as well as to provide very low cost. Apart from, Wireless Local Area Network (WLAN) has become apparent as a durable networking technology. Hence, the combination of these two popular technologies is growing so fast all over the world. Voice over WLAN will be a tool to provide low-cost and reliable voice services on wireless media. However just like other wireless applications, VoWLAN has also faced few challenges that need to be considered. Quality of Service (QoS) is one of the primary requirements in different kind of wireless applications. In this survey some of the important QoS requirement (latency, delay, jitter etc) have been analyzed, and it also has the introspection of the E- model and MOS (Mean Opinion Score) value for voice quality while using of different ITU-T codec. Therefore it makes Voice over WLAN a challenging research topic. In this study we will address all VoWLAN issues.

Keywords- *Wireless LAN,Codec, Quality of Service, Voice over WLAN.*

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I. INTRODUCTION

A. Voice over IP (VoIP)

The major classification of VoWLAN is IEEE 802.11 WLAN (wireless local area network) and VoIP (Voice over IP). Combining both of these technologies into a new variety of technology Voice over WLAN (VoWLAN) [1] has been an infrastructure for the provision of low-cost wireless voice services. With the advent of wireless networks in the communication technology field, there are many applications which are benefiting from out of exploiting robust features of wireless networks. Among various applications of wireless network, Voice over IP (VoIP) is one of such applications. The main reason behind this fact is its ability of accepting the challenges of real time delivery of voice packets across networks while utilizing the present internet protocol (IP). Since the past two decades, IP telephony service has reached to new high, one can anticipate it as a viable alternative to the conventional voice service being exercised over public switched telephone networks (PSTN) due to its cost effectiveness factor.

B. VoIP components

VoIP consists of three components - CODEC (Coder/Decoder), packetizer and playout buffer [2]. At the transmitter side, an appropriate sample of analog voice signals are converted into digital signals, compressed and then encoded into a prearranged format using voice codec. There are voice codecs according to the standards of International Telecommunication Union-Telecommunication (ITU-T) such as G.711 (data rate 64 kbps), G.729 (data rate 8 kbps), etc. Next, process performed is packetization. In which voice is encoded into equal size of packets. Moreover, in each packet, some protocol headers from unfamiliar additional layers are fastened to the encoded voice [3]. These protocol headers are Data link layer header, IP (Internet Protocol), RTP (Real-time Transport Protocol), and UDP (User Datagram Protocol). RTP and RTCP (Real-Time Control Protocol) also support various multimedia applications. Since TCP transport protocol is generally used in the internet, UDP protocol is mainly preferred in VoIP and other delay sensitive real-time applications. TCP protocol is suitable for very less delay sensitive data packets due to the acknowledgement (ACK) scheme. This acknowledgement scheme introduces some amount of delay as receiver has to notify the sender that each packet is correctly received by sending ACK. The data packets (voice) are transmitted over the IP based network to the destination (receiver) side, and the decoding and depacketization are accomplished at the receiver. When the packet is transmitted, time variations of packets delivery may occur which are also known as jitter [3] [4]. So there is a playout buffer is used by the receiver to smoothen the speech by getting rid of delay jitter. Packets arriving after than the allotted playout era testament are discarded [5]. The principle components of a VoIP over WLAN system covers the transmission of voice is shown in figure.

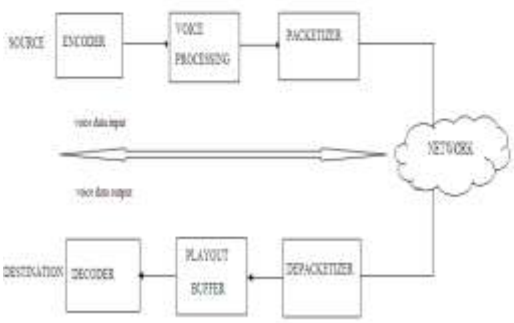

Figure 1.VoIP over WLANs components

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C. IEEE 802.11Wireless LAN

WLANs (Wireless Local Area Networks) are a novel approach in networking which is applied across the continents [6]. The main benefits of WLANs are it is easily manageable, simplicity, flexibility and cost effectiveness. Since, a decade ago, the IEEE 802.11WLAN founds gaining its vogue until it suits a versatile networking approach resulting toward its bunch layer deployment around the world. But majority of existing Wireless LAN applications are serving in the data transfer domain, web browsing and electronic mail; thus there is an ever increasing demand of multimedia services over WLANs [6].

IEEE 802.11(WLAN) standard supports two different modes i.e. infrastructure mode and ad hoc mode (infrastructure less mode) [7]. In the infrastructure mode, mobile nodes communicate with each other via an AP, while in the ad hoc mode, nodes communicate in a peer-to-peer fashion. VoWLAN applications accessed only at the infrastructure-based WLANs (communication via AP). At present, 802.11b is the most used standard all over world, whereas 802.11g has high data rate and compatible with previous versions such as 802.11b.

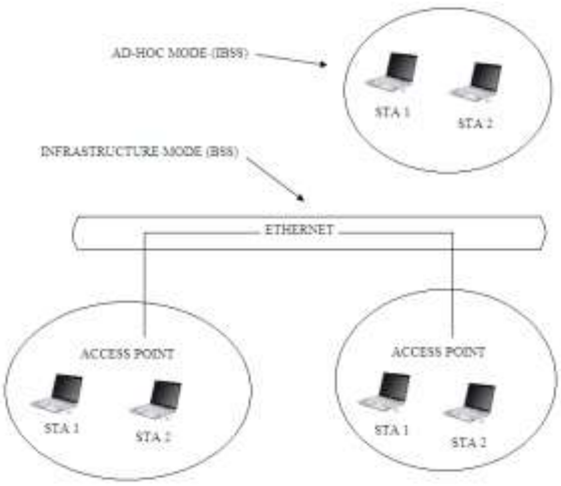

Figure 2. IEEE 802.11 WLAN

D. IEEE 802.11 MAC Layer

The 802.11 MAC protocol designed with two modes of communication, the DCF (Distributed Coordination Function) and the PCF (Point Co-ordination Function) (PCF) [2].

DCF (Distributed Coordination Function) uses the old policy of CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance), which usually checks the shared medium for the transmitted packet. Transmission is possible only when no other station is contending for channel. Other station has to wait until the channel is free from contention process. As when the channel is idle only then packet transmission is initialized after appropriate delay (back-off time). It prevents different stations from tying the channel immediately after completion of the previous transmission. It uses principle of acknowledgement of packet reception (ACK) without errors in the received frame. Data Packet reception in DCF needs an acknowledgment as shown in Figure (3). The timeframe between start of the ACK frame and completion of packet transmission is one Short Interframe Space (SIFS) [9]. To avoid congestion the data frames having less priority than the ACK frames. Transmissions other than ACKs should wait minimum one DCF Interframe Space (DIFS) [10] before transmission of data. If a transmitter senses a busy channel, it determines a random back-off period by setting an internal timer to an integer number of time slots [11]. On ending of a DIFS, the timer begins to decrement and when the timer reaches zero, the station may begin transmission [12]. If the timer is decreasing then it setup at a minimum value for upcoming transmission if the channel is linked to other station. [13].

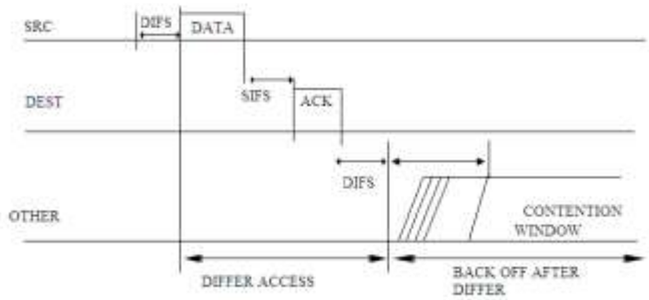

Figure 3. CSMA/CA Back Off Algorithm

In point coordination function total duration of packet transmission is divided in to a contention free interval and contention based interval. A station performing the traffic management is called the Point Coordinator (PC). The PC uses signal to transmit duration for a contention free period to all its associated station. Hence it has to wait for their Network Allocation Vector (NAV) for a contention free period. Further more all stations have to wait for Point Coordination Interframe Space (PIFS) interval to avoid any collisions in data. The PC may attach both acknowledgements of previous transmissions and polling messages for new traffic to a data frame. This allows the transmission to avoid waiting the interframe interval specified for individual frame transmissions

II. VoWLAN QoS Issues

The VoWLAN technology suffers from a few shortcomings which highly require be addressing and resolving. An important requirement of VoIP is obtaining desired Quality of Service (QoS). QoS can be defined as the ability of the network to support good services in order to satisfy its customers. In other words, QoS is a measure of user satisfaction and network performance i.e. services related to selected network protocol yield different elementary technologies, additionally to IP routed networks. Traditionally networks did not require strict measures for QoS because the data wasn't multimedia and the end-user could not notice or be materially affected by latencies. But, as the use of WLANs spread far beyond simple data transfer to intense multimedia applications, the need to focus on Quality of Service (QoS) issues becomes extremely important. As the technology is enhancing very rapidly over the past decade the consumers

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markets are now beginning to demand data concerted, time concerted movement of things like audio and video around a WLAN. We in fact, can introduce QoS within a given network at various stages and levels. The QoS can be establishing at the network adaptor, an event of an application level, a router, or at a server. However, for wireless networks, QoS can be implemented only at the network adaptor level because the routers and applications (codec) are not usually well aware of the connection medium.

Moreover, bandwidth and queuing are priorities to ensure a correct level of data integrity, when wireless network are used for applications such as VoIP and Media Streaming. There are numerous factors which can affect the quality of voice in VoWLAN; these include IP network service problems which are – delay (latency), delay jitter and packet loss. All of the above are important issues for the implementation of Voice over WLANs. The physical and MAC layers of wireless local area network has lower level of performance characteristics as when compared to wired local area network. That is the reason VoWLAN application evokes numerous implementation issues relating the network QoS stipulation, capacity, admission control and system architecture as well. So far, in this section, the basic introductory information about VoWLAN along with some of its QoS issues is described. In the next section the QoS requirements and challenges for VoWLAN are highlighted. First QoS parameters for VoIP in next subsection described. In Section III voice quality measurements mechanisms are presented. Finally in Section IV we discuss the conclusion.

A. QoS Requirements

All Intractability of obtaining guarantee in QoS is one of the nontrivial issues with the deployment of VoIP. Categorization of the parameters related to the quality of a single voice conversation with VoIP is done. These parameters are as follow Latency (end to end packet delay), Delay jitter (delay variation), and Packet loss.

a. Latency

Latency is the time delay induced in voice transmission by the Internet Protocol (IP) telephone system. One-way latency is the amount of time measured from the moment the speaker utters a sound until the listener hears it.

VoIP applications are very delay sensitive but packet loss can be negotiated by them up to some measure not like the data applications. End-to-end or mouth to ear delay is one of the primary constituents affecting QoS and should be inferior to 150ms for good network connection as defined by ITU G.114 [4]. Furthermore, delay is affected by several parameters or algorithms which can be categorized into: delay at the source, delay at the receiver, and network delay. Some of the delay parameters are recognized while some others are still not compatible.

1) Delay at the source - At the sender end before transmitting the voice data over the network, the delay caused by the whole process performed is known as delay at the source, moreover it is due to codec; packetization and process (playout buffer). The third component of source delay is when the computer passes the packets into the network for transmission to other side.

2) Delay at the receiver - The reverse process that carried out at the sender is performed at the receiver adding more delay: process delay and decoding delay including decompressing delay.

3) Network delay - Network delay in WLAN environment is the total delay of both WLAN and backbone networks. Queuing, transmission and propagation are other components of network delays. The propagation delay is the delay in the physical media of the network, while transmission delay includes router's delay and MAC retransmission delay.

b. Jitter

In much cases IP networks cannot guarantee the delivery time of data packets (or their Order), the data will arrive at very inconsistent rates. The variation in inter-packet arrival rate is jitter, which is introduced by variable transmission delays over the network and it has more negative effects on voice quality.

c. Packet Loss

Voice data Packets transmitted over IP network may be lost if it is transmitted late after a particular interval then it is discarded; when the late arrival take place at the jitter buffer of the receiver then packet loss occurs due to network or data traffic. In case of packet loss, the sender is informed to retransmit the lost packets and this is causing more delay and thus affecting transmission QoS. Packet losses more than 10% are generally unacceptable, unless the encoding arrangement allows extraordinary distinguishing quality [4].

III. VOICE QUALITY MEASUREMENTS IN VOWLAN

Quality of voice plays an important role in the QoS. As the VoIP industry got growth in recent past. So it is highly requires that the quality of voice should be as per standard to allows higher degree of user satisfaction. There are two tests for the measurement of voice quality: first is Mean Opinion Score (MOS) and the other has emerged with the rise in popularity of VoIP and is known as perceptual speech quality measurement (PSQM).

a. Mean Opinion Score

Voice quality as a function of QoS can be measured scientifically. A kind of rating factor known as the Mean Opinion Score is adopted by the communication industry to measure the quality of its connections [14][22]. These measurement techniques are defined in ITU-T P.800 Recommendation [15][21] as Mean Opinion Score (MOS) based on user perception that is ranged from 1 (poor) to 5 (excellent) for subjective determination of voice quality. The different factors considered such as loss, circuit noise ratio, side tone, echo, distortion, delay, and other transmission problems while calculating the quality of voice. A MOS of 4 is considered good quality or satisfied user that is, equal to the PSTN. VoIP applications characteristics (voice quality) are better than the PSTN and; it is also a challenge to deliver similar QoS over IP network to guaranty a good voice quality. This section explains how measures can be taken to more voice-specific solutions into a wireless network to ensure voice quality equal to that of the PSTN.

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User View	R Factor	MOS
Maximum obtainable for G.711[18]	93	4.4
Users very satisfied	90-100	4.3-5.0
Users Satisfied	80-90	4.0-4.3
Some users satisfied	70-80	3.6-4.0
Many users dissatisfied	60-70	3.1-3.6
Nearly all users dissatisfied	50-60	2.6-3.1
Not recommended	0 – 50	1.0-2.6

Table 1: voice quality levels with respect to user satisfaction (Referred from ITU-T) [15] [16]

MOS is a scale which is used to judge the quality of voice. The MOS is primarily derived from the R-factor. Thus the MOS can be estimated from the R-factor as follows:

$MOS = 1 + R * .035 + R * (R – 60) * (100 – R) * 7 * 10 ^ (-6)$

For $R = 0$, $MOS = 1$

For $R = 100$, $MOS = 4.5$

Where value of R is $0<R<100$ is the estimated conversational quality of voice it's assumed that R value will lie between this according to ITU-T standards. R factor can be premeditated by the E model. E-model is an instrument wearied in the planning and designing of networks for conveying voice communication applications [16]. The model estimates the relative impairments to voice quality. It compounds the impairments caused by several communication parameters against R element (total communication merit rating).The factor R is given by

$R = Ro – Is – Id – Ie + A$: (1)

Where: R (R Score) represents the resulting voice quality (from 0 to 100), Ro refers to signal to noise ratio, Is characterizes the simultaneous impairment factor such as too load speech level, Id represents mouth-to-ear delay, Ie is equipment impairment factor (e.g. codec specific characteristics), and A is advantage of access (advantage factor; for wire bound communication system factor A is equal zero).

The default values (bit rate, inter packet interval, payload, MOS) for some of the standard codec are shown in table 2

Table 2 Various VoIP Codec Parameters

Codec	Bit rate	Inter packet interval	Payload	MOS	Rmax
G.711	64 kb/s	20 ms	1280 bits	4.14	93.2
G.729a	8 kb/s	10 ms	80 bits	3.65	93.2
G.723.1	6.3 kb/s	30 ms	189 bits	3.9	93.2
G.722	64 kb/s	20 ms	1280 bits	4.17	129
AMR-WB	23.85 kb/s	20 ms	477 bits	4.14	129

Where, Rmax is the maximum R-score.

b. PSQM

It is another measurement test for voice quality in VoWLAN networks known as Perceptual Speech Quality Measurement (PSQM). It is based on ITU-T Recommendation P.861 [15], which specifies a model to map actual audio signals to their representations. Voice quality changes with different frequency component which ultimately are dependent on different coding mechanisms used for transmission. In PSQM measurements of processed (compressed, encoded) signals derived from a speech sample are collected and an analysis is performed comparing the original and the processed version of the speech sample. In MOS where the comparison of two signals is performed whereas in PSQM the score result will be an absolute number.

Some other important standard issued by ITU for improving the quality of voice: P.862 recommendation - **PESQ** (Perceptual Evaluation of Speech Quality) [17] and **PAMS** (Perceptual Analysis Measurement System) [18]. The PESQ tool was designed to calculate the MOS-LQO (Listening Quality Objective) values. It impacts on consistent samples with subjective calculation. For achieving this PESQ derives scores from the difference between reference signal and an output signal in the signal path. The greater the differences then lower the MOS value. It provides a reliable, inexpensive and quick performing codec performance. Mapping to an estimate of subjective mean opinion score on the ACR (Absolute Category Rating) scale is performed by using a combination of two parameters – symmetric and asymmetric disturbance. The mapping used in PESQ is given by

MOS = 4.5 – 0.1 - symmetric disturbance - 0:0309 - asymmetric disturbance

Now the PAMS (Perceptual Analysis Measurement System) is an improved version of the Hollier model (''the error surface.'') [19].The main improvement focuses on end-to-end measurements utilizing time, level alignment and equalization respectively. The steps initiated by the division of signals into aligning equalizing and utterances. It figures out delay changes due to transmission and processing. Equalization is used to set the listening level to a certain acceptable level before performing the required transforms. It follows with required transforms of both reference and error signals. The perceptual layer is performed by quantifying the errors and using the measured distortions as data for analysis. The results are depicted to two quality scores – an accepted MOS based on the ACR listening-quality opinion scale and other is based on ACR listening-effort opinion scale [20].

IV. CONCLUSION

In this survey paper demonstration on Voice over WLAN has been given, besides its confronts. As a primary concerned issue, QoS was explained moreover its contributing constituents were stated in features. Albeit, VoIP can permit packet loss to some extent, still it is very fragile to delay factor. Jitter also acts as a predominant function on voice quality therefore jitter buffer is introduced to adept the playout of packets. Different techniques for voice QoS assessment were mentioned besides speech quality measurements tools and standards are stated as well. While the voice quality depend on complex network parameters and it is highly desired to consider these parameters so E model and MOS standards are discussed, along there default values. Yet, WLAN is a bandwidth restricted network which advantages to a narrow count of VoIP calls. It can likewise cause problems therefore, security issues in VoIP stances a question. It is concluded that VoWLAN is a hopeful nevertheless much taxing technology that requisites extra efforts to realize capability success in the hereafter. As this research area is quite broad a share of hopeful views can be reviewed encore for its speech quality improvements.

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