Voice over Internet protocol (VoIP) Performance Enhancement over Wireless local area network (WLAN)

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***Abstract--*Nowadays the situation is all most WLAN application are data centric, and growing the popularity of internet as required in today are for voice conversation in all field over the network scenario. So in this paper we measure wireless local area network (WLAN) for voice performance and capacity of speech quality caused by Pocket delay and loss of voice quality. And also calculate the bandwidth for the VoIP.Due to the latest technology innovation popular packet switching networks, Voice over Internet Protocol (VoIP) has developed an industry favourite over Public Switching Telephone Networks (PSTN) with respects to voice communication. The cheap cost of making a call through computer to computer and computer to phone or phone to phone, VoIP globally and giving just one bill for data usage is a giantbenefit, Voice over IP (VoIP) technology has many advantages over the traditional Public Switched Telephone Networks. There are some issues in VoIP such as, packet loss, jitter and latency due to the connection through the computer internet. The main goal is to collect the required knowledge for VoIP planning and surveyproblem. An innovate methodology to calculate the system's actual throughput and a traffic model for mixed application users are proposed with a step by step description to derive an algorithm to determine the maximum number of subscribers that each specific VoIP for wireless local area network (WLAN) may support. The VoIP also using the IEEE standard 802.11 MAC sub layer.**

Keyword- capacity voice over internet protocol (VoIP), wireless local area network (WLAN), IEEE standard 802.11, Quality of service (QOS).

1. INTRODUCTION

Today voice over internet protocol are one to the fastest growing internet application it has two uses compare with voice over telephone network first voice techniques bandwidth showing packet switch network and the VoIP can improve bandwidth efficiency . Second one isit can creation of new service that combine with voice communication such asvideo,voice conference etc.

The first specification of Resident Area Wireless Network was accepted under the IEEE 802.11 standard with product authorization name of VoIP [1]. The IEEE 802.11e standard was developed to add an applications support to the basic standard. This standard serves fixed and wandering users in the frequency range of 2 – 11 GHz. In order to add mobility to wireless access, the VoIP IEEE standard 802.11e MAC layer specification was defined, utilizing frequencies below 6 GHz.

The VoIP is a technology for transmitting voice, video faxed, over the packet switch network. When we use the VoIP techniques,than voice information is converting into digital information packet and send on the internet [1]. And again the internetconverting back into analogy signals from source to destination.

When we call from one phone to another phone or one computer to anothercomputer than we transmitting voice single , video, etc. over internet than the voice must be efficiency, reliabilityand high quality and voice must not besuffer a delay light than 150ms [1][2].

OFDMA, PHY called the mobility VoIPsystem profile. The Mobile VoIP standard has been developed to be the best wireless broadband standard for the wireless local area network (WLAN)permitting a new era of high throughput and high delivered bandwidth together with exceptional spectral efficiency when compared to other 3G+ mobile VoIP wireless technologies.so we analysis the bandwidth and capacity for both direction downlink and uplink. We can use the multiple MAC layer with IEEE standard 802.11e, For MAC architecture, which we can use duplexing, frequency band operation etc.

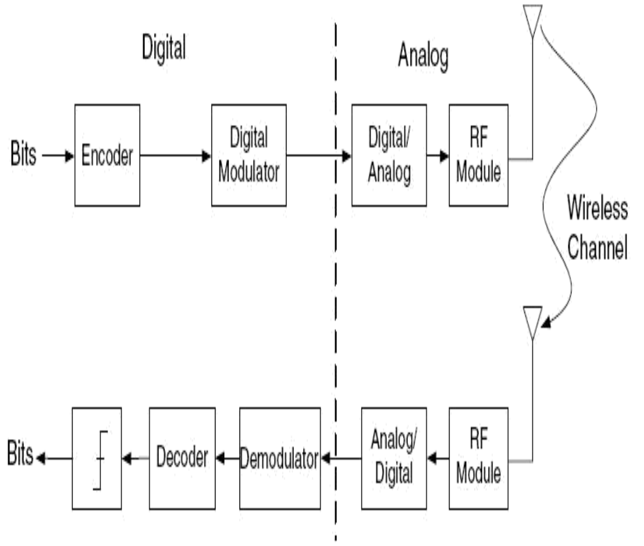

Figure 1:- Wireless channel of VoIP

The VoIP information in from of packet .and that information are converting into digital information packet over packet switching network which is called as digital modulation.Digital single are again converting into analog single which is go through over radio frequency (RF) and it's uphold by wireless channel .

1.1 PHYSICAL LAYER: - The 802.11 MAC works with a single First in First out (FIFO) transmission queue.The CSMA/CA constitutes a distributed MAC based on a local assessment of the channel status i.e. whether the channel is busy or idle. Due to high difference between transmitted and received power levels , traditional random channel access mechanism used in wired networks as CSMA/CD are not applicable in wireless networks.

1.1.1 OFDMA: - OFDM is the transmission scheme of select toallow high-speed data communications in broadband systems for VoIP over wireless local area network (WLAN). And physical layer base on OFDM(Orthogonal Frequency Division Multiplexing).its also called multicarrier modulation.

OFDM is based on the high-bit-rate datainto some similar lower bit-rate data and modulating each bit rate data on single subcarriers

In this enterprise wireless local area network (WLAN) uses are increasing in these field domestic commercial industrial and public areas like hotels airports coffee shop and university compose or conference setting also benefit from, WLAN since they provide flexible connection and network access at reduce costs voice over IP applications. Some other fields like construction, healthcareand banking etc. so it is more difficult to understand voice performance and capacity in WLAN.Mostly we focus in IEE802.11e.In this standard 802.11e, we also study MAC layer and queuing mechanisms that can improve voice performance.We calculatethe bandwidth and capacity for VoIP over wireless local area network (WLAN) [3].

2. VoIP process:-

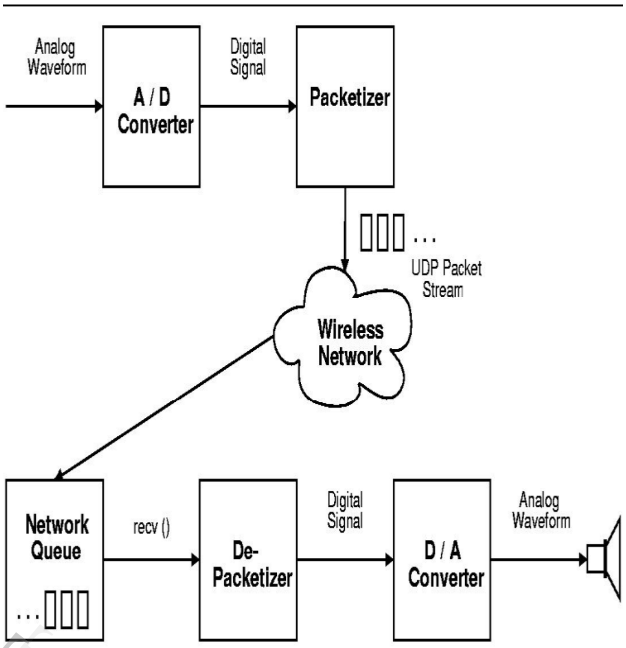

Figure 2:- VoIP process

The process of transmitting a conversation through VoIP includes some important steps that are not present in a PSTN. The first is converting an analog voice waveform to a digital signal that may be transmitted by a data network.Some codecs are using for encoding voice signals for VoIPover WLAN.Compressing CODECs are strongly suggested for saving bandwidth and increasing cell capacity. Using the basic non-compressing G.711 CODEC, which most VoIP gateways support, needs a bandwidth of 156 Kbps per direction at the basic rate (64 kbps raw data per call) [6]. A Codec is an algorithm that is used in VoIP process to converting voice signals into digital single to be transmitted over the Internet during a VoIP call. G.711 code is a common code which is use Pulse Code Modulation (PCM) of voice frequencies at the rate of 64 Kbps.

3. ADVANTAGE OF VOIP

- Save a lot of money
- More than two persons
  In the VoIP process we can communication with more than two people like conference with a whole team communicating in real time.
- Cheap user hardware and software
- Abundant, Interesting and Useful Features
- Free VoIP communication
- If you don't use VoIP for voice communication, then you are most certainly using the good old

phone line (PSTN– Packet-Switched Telephone Network).

4. DISADVANTAGE

- The most challenge is how to increase the systemcapacity for voice, when the noise traffic due to large overhead.
- Another challenge is how to increase quality of service (QOS) for voice user. And Voice traffic is sensitive to delay and day filter.

5. BACKGROUND/RELATED WORK

In the literature review the design of MAC protocol that is supporting voice traffic and also has drawn some interest.The related work on VoIP over wireless local area network (WLAN) can be classified according to which access mechanism are used for VoIP DCF or PCF, mode [7]. If we assumed the use of PCF mode but the PCF mode is not supported in most 802.11 products anda reason could bethat the market does not see a convincing need for PCF. In adding,DCF is a technology that has been well tested and establishedto be strong in the field. The main problem is, when there are two overlappingWLANs using the similar frequency channel, DCF willstay to work while PCF will not, since collisions is occur betweenstations of the two WLANs may occur during their theoreticallycontention-free periods. Because the DCF mode is base on collision Avoidance (CSMA/CA) protocol .The 802.11b standard provides two modes ofMAC operation [1] [4].

A). Distributed point coordination function (DCF) mode.

B). Optional point coordination function (PCF) mode. That is design for real time services like voice. In this paper we calculate the bandwidth and capacity in both direction downlink and uplink.

6. BANDWIDTH ALLOCATION

- In the downlink direction :-

The bandwidth allocation in downlink direction to allocate bandwidth by a base station (BS) without associationof mobile service based on connection identifier (CID). The connection identifier is 16 bit addressseparate between multiple UL channels (connections) linked with the similar DL channel.The Mobile Stations (BS) and Subscriber Stations (SS) dochecked connection identifier (CIDs) at the receiving packet data unit (PDUs) and recall only that packet data unit (PDUs) that are addressed to them [5] [8].ThePHY layer recourses assigned for PDU transmission in each connection is specified in DL-MAP messages.

- In uplink direction:-

The bandwidth allocation in uplink direction to allocate bandwidththe mobile station (MS)is requests resources by using a bandwidth request subheader on a media access control packet data unit (MAC PDU).The base station (BS) allocates committed (intended for a single SS) or shared (intended for a group of SSs) resources for the users intermittently which can be used to request bandwidth. In VoIP this process is called *Polling.*

7. RESULT ANALYSIS

The VoIP capacity and bandwidth is calculated in MATLAB. VoIP capacity in IEEE 802.11e WLAN are analysed on MATLAB and the DL capacity vs. UL capacity graphs are shown in Fig 3 and Fig 4.

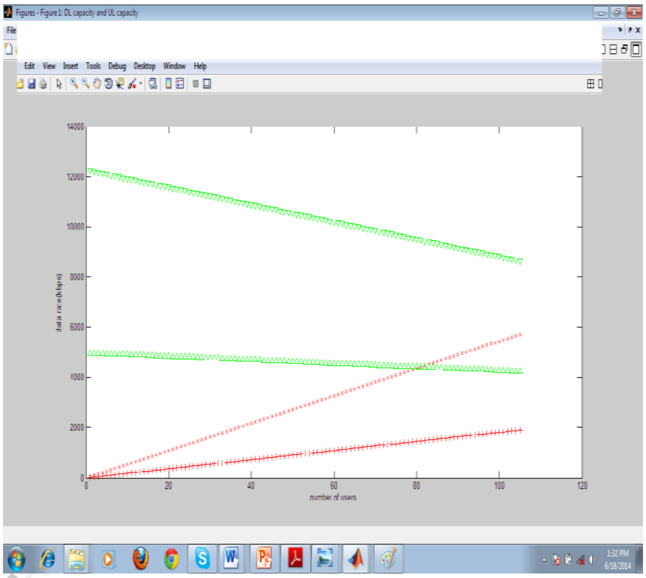

Figure3 capacity and bandwidth

X axis = number of user
Y axis = data rate in kbps
Green down arrow is DL capacity
Green up arrow is UL capacity
Red *is DL demand
Red + is UL demand

In the resulted figure it is highlighted in the data sheet for VoIP over WLAN the system is based on *5MHz* channel width and with DL/UL Traffic Ratio=2. In this figure wemeasured andmanipulate to the configurable system parameters in order to maximize the number of users. One of these parameters is DL: UL Ratio is 2:1 a DL: UL traffic ratio is = 2

Packet data unit is = 3
Packet data unit per data burst is = 2
Peak data rate in <DL> is = 16567.6 kbps
Peak data rate in <UL> is = 6643.33 kbps
All these are parameter which is run on MATLAB

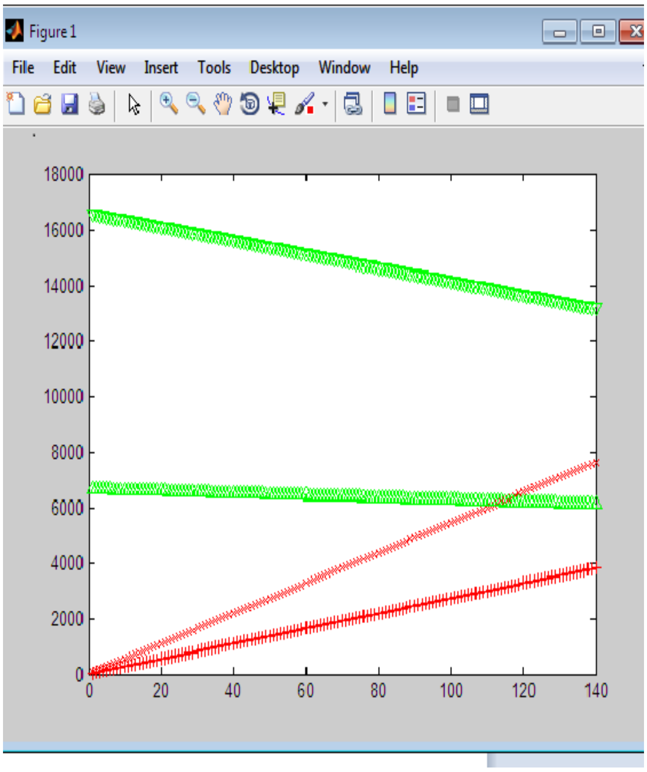

Figure 4:- DL capacity Vs. UL capacity

8.CONCLUSION

This paper investigates two critical technical problems in VOIP over WLAN. First is low VoIP capacity in a WLAN. And Second unacceptable VoIP performance and low quality of services over a WLAN. Due to the limitation of DCF and PCF, we defined the standard 802.11, which is single coordination function for quality of service QOS data transmission. When we use for more number of voice calls than it compare to DCF made and there are large polling over head and we try to overcome that problems in this paper. We are implementing my work on MAT lab and we done my programme with help c++ language MAT lab to implementing this work on graphics representation. In this paper I analysis the capacity and bandwidth for VoIP over WLAN. Future work is As we know that the voice is real time traffic and most of the time there is no need of acknowledgement. So when there is no Ack then AP sends the simple polling frame to the STAs (without using piggybacking) and then there will be some extra overheads due to these polling frames. In my future work new scheme will be developed to overcome this problem and developed a capacity algorithm which is increasing total through and resource efficacy of a system.

9.REFERENCE

[1] SuchiUpadhyay, S.K.Singh, Manoj Gupta, Ashok Kumar Nagawat "Improvement In Performance Of The VoIP Over WLAN"IEEE Standard Volume 12- No. 4 Dec 2010.

[2] Wei Wang, Student Member, IEEE, Soung Change Liew, Senior Member, IEEE and Victor O.K. Li, Fellow, IEEE "Solutions to performance Problem In VoIP Over a 802.11 Wireless LAN" IEEE Vol. 54, No. 1 Jan 2005.

[3] HongqianZhai, Student Member, IEEE, Xiang Chen, Member, IEEE, and Yuguang Fang, Senior Member, IEEE. "How Well Can the IEEE 802.11 Wireless LAN Support Quality of Service". Vol. 4, No. 6 Nov 2005.

[4] Hyewon Lee, SeonghoByoungjin Kim, Kwang Bok Lee, and Sunghyun Choi, department of Electrical and Computer Engineering and INMC, Seoul National University, Seoul, Korea, "Enhancing Voice Over WLAN Via Rate Adaption and Retry Scheduling ".

[5] FarooqAnjum, MoncefElaoud, David Famolari, AbhrajitGhosh, RavichanderVaidyanathan, AshushtoshDutta, PrathimaAgrawal. "Voice Performance In WLAN – An Experimental Study". IEEE 2003.

[6] Federico Coli,Marco Conti and EnricoGregori,"IEEE 802.11 protocol Design and Performance Evaluation an Adaptive Backoff Mechanism ".

[8] HongqionZhai, Xiong Chen and YuguangFang,"A call Admission and Rate Control Scheme of Multimedia Support Over IEEE 802.11 Wireless LANs".