ENHANCED BANDWIDTH UTILIZATION IN WLAN FOR MULTIMEDIA DATA

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ABSTRACT

Deployment of wireless local area networks (WLANs) is growing consistently and demanding the support of multimedia applications with acceptable quality of service (QoS). This is attracting the interest of researchers globally. Under the optimum QoS, a number of VoIP calls can be supported by a WLAN. Distributed Coordination Function (DCF) and Point Coordination Function (PCF), two MAC protocols specified in the IEEE 802.11 standard have upper bound on VoIP connections. Under DCF mode 12 calls and in 20 calls in PCF mode[1,2,3,4]. In this paper we are proposing an access media mechanism in which audio data is transmitted in PCF mode and best-effort traffic in DCF mode. In the proposed access media mechanism, polling list is dynamically updated so that only those stations are polled which have voice packets ready to transmit. We have proposed a multi-queued MAC architecture for the access point. We considered voice traffic in CBR mode. The simulation results show that the maximum number of VoIP calls supported by 802.11b is 26 and 14 when inter arrival time for voice packets is 20 ms and 14 ms respectively.

KEYWORDS: *Medium Access, Mechanism, Multimedia Data, QoS, Bandwidth.*

I. INTRODUCTION

In future generations of WLANs the IEEE 802.11 WLANs will influence the style of daily life of people. The 802.11 technology provides flexible and cheap wireless access capability. Deployment of an 802.11 WLAN is very easy also in hospitals, stock markets, campuses, airports, offices and many other work places. Multimedia applications are increasing very fast in number as well as in size. Demand of voice and broadband video services through WLAN connections is growing day by day. Real time multimedia applications require strict QoS support such as guaranteed bandwidth and bounded delay/jitter etc[,6,7,8,17]. In the organization of this paper section-2 describes the functioning of DCF and PCF modes, section -3 discusses the past work done in this direction, section-4 focuses on the limited QoS support in PCF, section-5 describes the proposed approach, in section-6 simulation environment and results are discussed.

II. BACKGROUND

In this section, we are discussing an overview of the IEEE 802.11 standard that provides two different channel access mechanisms, namely the Distributed Coordination Function (DCF) and Point Coordination Function (PCF). Our scheme introduces the enhancements in the PCF access scheme.

2.1 Distributed Coordination Function (DCF)

Most of the wireless LANs in the Industrial Scientific and Medical (ISM) band uses CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance) as the channel access mechanism. The basic principles of CSMA are to listen before talk and the contention. This is asynchronous message passing mechanism (connectionless), delivering a best effort of service, and no bandwidth and latency are guaranteed. CSMA is fundamentally different from the channel access mechanisms used by

cellular phone systems (i.e. TDMA).
CSMA/CA is derived from the channel access mechanism CSMA/CD (*Collision Detection*) employed by Ethernet. However, collisions waste valuable transmission capacity, so rather than the collision detection (CD) used in Ethernet, CSMA/CA uses collision avoidance. Collision Avoidance (CA), on a wire, the transceiver has the ability to listen while transmitting and so to detect collisions (with a wire all transmissions have approximately the same strength). But, even if a radio node could listen on the channel while transmitting, the strength of its own transmissions would mask all other signals on the air. Thus, the protocol cannot directly detect collisions like with Ethernet and only tries to avoid them.
The 802.11 standard defines the Distributed Coordination Function (DCF) as its fundamental access method and is based on CSMA/CA. DCF allows each multiple independent stations to interact without central control. Figure 1 illustrates the basic access method used in the DCF protocol. If a station finds a channel idle for at least a DIFS period, it sends the first frame from its transmission queue. If channel is busy, the station waits till the end of current transmission and then starts the contention. It selects a random slot time, so called back-off time from a Contention Window (CW) and waits for DIFS and its back-off time.

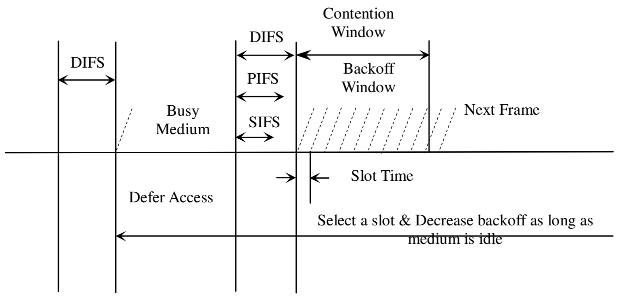

Figure 1: DCF (CSMA/CA) basic access method

The backoff time is calculated as

$$T_{backoff} = \text{Rand}\ (0, \text{CW}) * T_{slot},$$

where T_{slot} is a time slot specific to physical layer and
Rand () is a uniform distribution random function[2].

The back-off time is computed to initialize the back-off timer and this timer is only decreased when the medium is idle. When the medium is sensed to be busy, this timer is frozen.

When its back off timer expires, and if the channel is still idle, the node sends the frame. Thus, the node having chosen the shortest backoff time wins and transmits its frame. The other nodes just wait for the next contention (after waiting for the end of this packet transmission). Because the contention period is derived from a random number chosen with a uniform distribution, and done for every frame, each station is given an equal chance to access the channel.

2.2 Point Coordination Function (PCF)[10,11,15]

Periods of contention free service arbitrated by the Point Coordinator (PC) alternate with the standard DCF-based access (or contention period). The duration of the contention free period can be configured. 802.11 describes the contention-free period as providing near asynchronous service because the contention-free period will not always start at the expected time.
The Contention-free service uses a centralized access control method. Access to the medium is restricted by the Point Coordinator, a specialized function implemented in access points. Associated stations can transmit data only when they are allowed to do so by the point coordinator. Contention-free access under the PCF resembles token-based networking protocols, with the point coordinator's polling taking the place of a token. Despite, access is under the control of a central entity, all transmissions must be acknowledged. The figure 2 illustrates the PCF access method.
When the PCF is used, time on the medium is divided into contention-free period (CFP) and the contention period (CP). Access to the medium during the CFP is controlled by the PCF, while access

to the medium in CP is controlled by the DCF[12,13,14]. In order to be fair with contending traffic, the contention period must be long enough for the transfer of at least one maximum size frame and its associated acknowledgement. Alternating periods of contention-free service and contention based service repeat at regular intervals, called the contention-free repetition interval (known also as super frame).

At the beginning of the CFP, the PC (which resides in AP) transmits a management frame, called beacon. One of the beacon role components is the maximum duration, *CFPMaxDuration,* of the CFP. The PC generates beacons at regular beacon frame intervals, thus every station knows when the next beacon frame will arrive. This time is called target beacon transmission time (TBTT). All stations receiving the beacon set the NAV to the maximum duration to lo ck out DCF based access to the wireless medium. The access point maintains a polling list of associated stations and polls any station in this list. Since time in the CFP is precious, acknowledgements, polling, data transfer may be

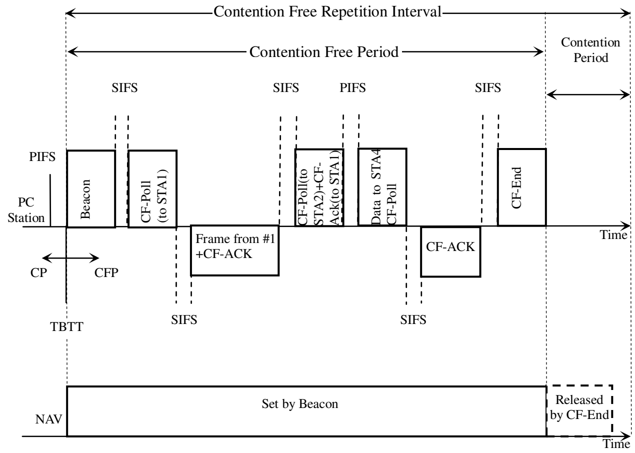

Figure 2: PCF Access Scheme

combined to improve efficiency (as shown in figure 2. All CFP transmissions are separated by short inter frame spaces (SIFS), where PCF waits some time if there is no response from a polled station, to prevent interference from DCF traffic (both are shorter than DCF inter frame space, DIFS).

III. RELATED WORK

In the past many researchers have given analytical, simulation based or experimental results on transmitting Voice over IEEE 802.11 WLAN. MAC protocols defined in IEEE 802.11 standards [1,2,19] are DCF and PCF. Performance of transmitting Voice over WLAN in DCF mode as well as PCF mode has been evaluated. In PCF mode various polling algorithms have been used to transmit Voice over WLAN in the form of the interactive human speech.

In [3] Ali Zahedi and Kevin Pahlavan obtained the capacity of IEEE 802.11 WLAN with voice and data services analytically in DCF mode. Precisely the question which they answered is the number of network telephone calls that can be carried over WLAN with a predefined amount of data traffic or what is the maximum data traffic per user for a given number of voice users. Priority to voice is given by assigning UDP protocol for voice and TCP protocol for data. They found that in 1Mbps bandwidth, a maximum 18 voice users are supported with upper bound for delay of 100ms and decreasing upper bound for delay to 50ms which reduces maximum voice users to 14. Data traffic is assumed to be less than 10Kbps.

Sachin Garg and Martin Kappes found an upper bound on the number of simultaneous VoIP calls that can be placed in a single cell of an 802.11b network in [4]. They performed an experiment in which multiple Wireless PCs running Windows 2000, were associated with the same 802.11b AP, which was connected to a 100Mbps Ethernet. The setup was used to make full duplex VoIP calls between a wireless PC and a wired PC using IP phones. For each call the ITU G.711 codec was used where frames are sent every 10ms. Each call results in two RTP streams, from wired to wireless and vice-versa. Number of VoIP connections with acceptable voice quality is tested by successively establishing new calls in addition to the ongoing calls. The quality of connections was monitored measuring of loss, jitter and round trip time by a commercially available tool. For the first five calls, the quality of all the calls was acceptable. Loss (0%), round trip time (around 5ms) and jitter (around 7ms) were all in acceptable ranges for a good quality of VoIP. When the sixth call was placed, except for an increase in the round-trip time for some of the connections the quality of all six simultaneous connections was still acceptable. As soon as the seventh call was placed, all seven wired to wireless streams started suffering approximately 16% loss and the call quality became unacceptable for all calls in this direction. All wireless to wired streams still exhibited acceptable quality. In addition to this experiment they obtained an upper bound for simultaneous calls analytically. They found that when a G.711 Codec with 20ms audio payload is used, an 802.11b cell could support only 3 to 12 simultaneous VoIP calls. The actual number depends on the effective transmission rate of the wireless station, which for 802.11b can be 1Mbps, 2Mbps, 5.5Mbps and 11Mbps.

In [6,16] practical investigation of the IEEE 802.11b MAC layer's ability to support simultaneous voice and data applications is done. DCF mechanism is modified and new mechanism is called Back-off Control with Prioritized Queuing (BC-PQ). BC-PQ addresses the two shortcomings of the DCF mode with respect to voice. First, it distinguishes voice packets from data packets and provides a higher priority to the voice traffic. Allocating separate prioritized queues for voice and non-traffic does this. Secondly, in addition to priority queuing, the enhanced AP transmits voice packets using zero back-off instead of random back-off as required by the 802.11b standard. The key parameter used to quantify voice performance is packet loss.

Jing-Yuan Yeh and Chienhua Chen in [7] proposed three polling schemes (RR, (FIFO, Priority and Priority Effort-Limited Fair) combined with the Point Coordination Function (PCF) to improve the utilization of the wireless channel and support certain Quality of Service of multimedia traffic. The polling schemes proposed in [8] are-Round-Robin Scheme (RR), First-In-First-Out Scheme (FIFO), Priority Scheme, Priority Effort-Limited Fair Scheme. All the above-mentioned polling schemes are simulated with the network simulator OPNET in [7]. It is found through simulations that all these schemes perform better than the DCF mode. To achieve the maximum throughput in a BSS, the FIFO scheme is found to be the best. Priority Scheme provides a simple way to support QoS of traffic; however this scheme can exhaust all the bandwidth of best-effort traffic. The Priority-ELF scheme achieves high utilization of wireless channel in case of the bursty traffic.

In [9,10,15] the capability of the Point Coordination Function (PCF) to support Voice over IP (VoIP) applications is evaluated. The capability of PCF mode in support of variable bit rate (VBR) VoIP traffic is investigated, where the silence suppression technique is deployed in voice codec so that no voice packets are generated in silence periods. Simulation shows that under the PCF using VBR mode for the VoIP traffic may effectively reduce the end-to-end delay of VoIP. Simulation is carried out in the OPNET network simulator. The upper bound for number of VoIP connections in CBR mode is found to be 15 and in VBR mode it is 22. Brady's model and May and Zebo's models are used for VBR voice traffic.

E. Ziouva and T. Antonakopoulos in [12] proposed a new dynamically adaptable polling scheme for efficient support of voice communications over IEEE 802.11 networks. They proved analytically that when silence detection is used their scheme improves the capability of IEEE 802.11 wireless LANs for handling voice traffic efficiently. Their polling scheme is called Cyclic Shift and Station Removal Polling Process (CSSR).

In [11] a distributed fair queuing scheme called Distributed Deficit Round Robin (DDRR) is proposed which can manage bandwidth allocation for delay sensitive traffic. The question which is answered in this paper is how many voice and video connections can be accommodated in an 802.11 WLAN satisfying the imposed QoS requirements under different scheduling schemes.

Xiyan Ma, Cheng Du and Zhisheng Niu in [14] proposed two adaptive polling list arrangement schemes called Dynamic Descending Array (DDA) and Hybrid DDA and RR (HDR) in order to decrease the average delay of the voice packets, by means of reducing the possibility of null polls. One scheme is called Dynamic Descending Array (DDA) and the other scheme is a combination of DDA and traditional Round-Robin scheme and is called Hybrid DDA and RR (HDR).

IV. PCF WITH LIMITED QOS SUPPORT

Although the contention-free service is designed in 802.11 networks to provide QoS for real-time traffic, this service also have some limitations[8,9]. In the following we describe main limitations related PCF-

- **Unpredictable beacon delay**-The problem is related to the uncontrolled length of CP. Indeed, the minimum length of the CP is the time required to transmit and acknowledge one maximum frame size. It is possible for the contention service to overrun the end of the CP, due to transmission of a contending traffic. When the contention based service runs past the TBTT, the CFP is foreshortened, and hence the beacon is delayed.
- **Unknown transmission time of polled stations-** A station which is polled by the PC is allowed to send a single frame that may be fragmented and of arbitrary length, up to maximum of 2304 bytes (2312 bytes with encryption). Furthermore, different modulation and coding schemes are specified in 802.11a, thus the duration of the MSDU delivery that happens after the polling is not under the control of PC. This may destroy any attempt to provide any QoS to other stations that are polled during the rest of the CFP.
- **No Knowledge of the offered traffic at the stations-** With CFP, the access point (AP) has any knowledge of the offered traffic at the polled stations. Thus, when polling the different stations with a round-robin scheduling algorithm, the PC may waste a lot of time until polling a special station having critical time traffic (e.g., CBR traffic). This may affect the QoS parameters for these traffic categories. Hence, with PCF there is no efficient scheduling algorithm, which has the knowledge of the different traffic categories at associated stations and uses this knowledge in order to meet the requirements (e.g., latency, bandwidth) of these different traffic categories.

V. PROPOSED MECHANISM TO ACCESS MEDIA

5.1 Overview

In this section we will discuss our proposed access media mechanism to transmit voice over WLAN. In our scheme also we use PCF mode to transmit voice and DCF mode to transmit Best-Effort traffic. A lot of valuable bandwidth is wasted when a station which has nothing to transmit is polled. Keeping all the stations in polling list and polling them in Round-Robin manner hampers the channel utilization severely. We propose a dynamic polling list arrangement to poll only those stations which have data to transmit. Also in our scheme, the downlink voice traffic from AP is combined with CF-Poll which can significantly reduce the overhead and thus may improve the channel utilization. We give the MAC architecture of a WLAN station and the MAC architecture of the access point respectively. How the polling list is handled dynamically is discussed in the section 5.4.

5.2 MAC Architecture of WLAN Station

MAC architecture of a WLAN station in our scheme is shown in the figure 3. Each station maintains two queues at its MAC layer, one for voice traffic and the other for Best-Effort traffic. When a frame arrives at the MAC layer of a WLAN station from its upper layers a classifier checks whether it is a voice frame or Best-Effort data frame, and accordingly it is put in the voice queue or Best-Effort queue. When a station is polled in the CFP it transmits the voice frame from the head of the voice queue. The frames from the Best-Effort queue are transmitted in DCF mode in the CP.

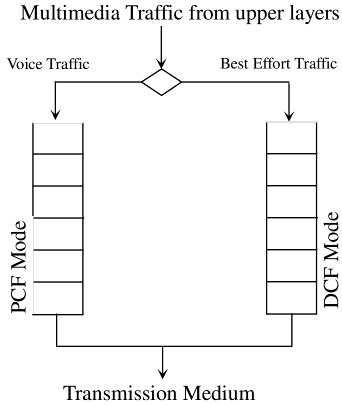

Figure 3: MAC Architecture of Station

5.3 MAC Architecture of the Access Point

MAC architecture of the access point, in our scheme is shown in the figure 4. The AP maintains one queue for Best-Effort traffic and (Poll Limit + 1) queues for voice traffic at its MAC layer. Poll Limit is the maximum number of stations which can be kept in the polling list in our Dynamic Polling arrangement scheme discussed in 4.4. When a frame arrives at the MAC layer of the AP, a classifier checks whether it is a voice frame or data frame and accordingly it takes its path as shown in the figure 4. If it is a data frame, it simply enters the Best-Effort queue and is transmitted in CP using DCF method. Otherwise, if it is a voice frame, further it is checked whether the destination station is in polling list or not. In case destination station is not in the polling list, it is checked whether number of stations in polling list exceed Poll_Limit or not. If they exceed then the frame enters the Non_Poll queue, otherwise destination station is added in the polling list and the frame takes the path as shown in the figure 4. The frame enters the Non_Poll queue as shown in figure 4. On the other hand, if the destination station is in the polling list, then the frame enters queue for its destination station among Poll_Limit queues.

5.4 Dynamic Polling list arrangement

The main issues in maintaining a polling list are how to add a station in the polling list and how to remove a station from the polling list.

5.4.1 Adding a station in the polling list:

The stations which want to transmit voice frames in CFP send Resource Reservation (RR) request in CP in controlled contention interval (CCI). A CCI is started when the PC sends a specific control frame. Only those stations which want to transmit voice in the CFP contend in CCI using CSMA/CA. In case of collision between RR requests no retransmission is done. Corresponding stations can send their RRs in the next CCI. If a downlink frame arrives at the MAC layer of the AP and the destination station is not in the polling list, in this scenario if number of stations in the polling list is less than Poll Limit then the destination station is added to the polling list.

5.4.2 Removing a station from the polling list:

A station is removed from the polling list when its connection time is over.

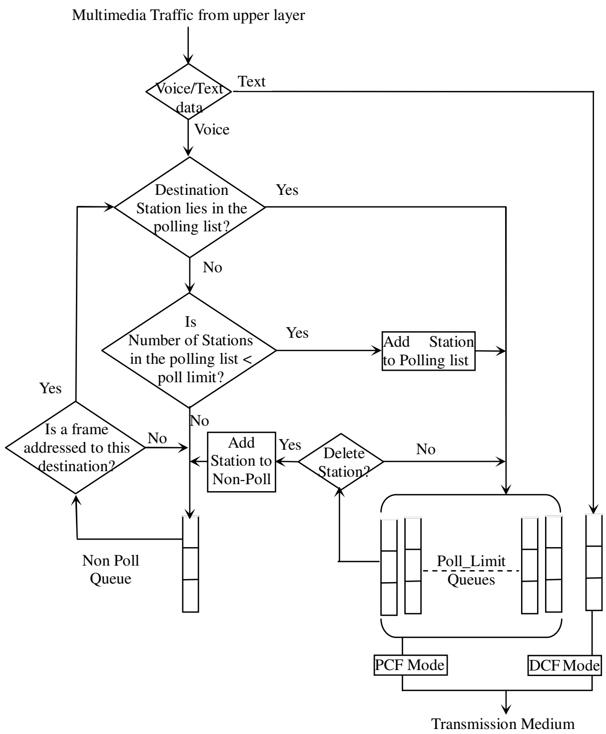

Figure 4: MAC Architecture of a Polling List

VI. SIMULATION STUDIES ON THE PROPOSED ACCESS MEDIA MECHANISM

6.1 Overview

We have simulated the proposed access media mechanism in C programming language. All the parameters considered for simulation are also given in the table 1. Finally simulation results are discussed in the section 5.3.Parameters used for simulation are shown in the table 1. The simulation is run for 85000 cycles of contention-free repetition interval. The results of initial warm-up period of 5000 cycles are ignored.

6.2. Maximum Number of VoIP Connections in 802.11b

Calculation shows that given the 11 Mbps rate and 128kpbs needed for duplex VoIP calls. But due to network layer protocol, inter frame space, beacon frame, poll frame, CF-end frame and channel utilization is around 30% of the total payload. Figure 5 shows the plot between number of VoIP connections and average packet delay when the inter arrival time is 20ms. We see in that case the maximum number of voice calls is 26; the uplink average packet delay goes up abruptly. We see when there is no best-effort traffic the maximum number calls supported by 802.11b is 26.

Table 1: Simulation Parameter

PHY Layer Specifications	DSSS
Transmission Rate	11 Mbps
Beacon Interval	60ms
Beacon Size	106 bytes
QoS Acknowledgement	14 bytes
CF-Poll Frame Size	34 bytes
CF-End Frame Size	20 bytes
PLCP Preamble	18 bytes
PLCP Header	6 bytes
SIFS Time	10µs
PIFS Time	30µs
DIFS Time	50µs
A Slot Time	20µs
Nccop	5
RTP Header	8 bytes
UDP Header	20 bytes
IP Header	8 bytes
MAC Header	34 bytes
CWmin	7
CWmax	255
Retry limit	5

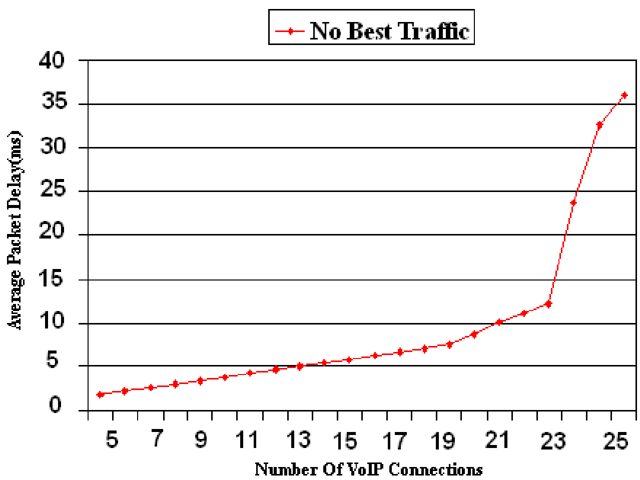

Figure 5. Number of VoIP vs. Average Packet Delay
(Inter Arrival Time is 20 ms)

6.2 Simulation Results

In another case, Figure 6 shows the plot between number of VoIP connections and average packet delay when the inter arrival time is 10ms. We see in that case the maximum number of voice calls is 14; the uplink average packet delay goes up abruptly. We see when there is no best-effort traffic the maximum number calls supported by 802.11b is 14.

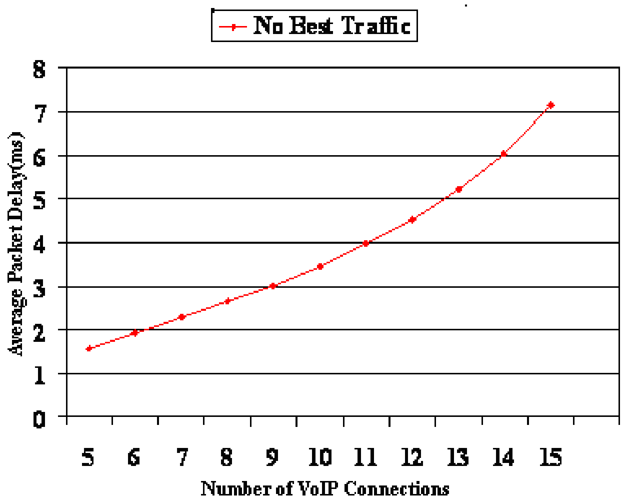

Figure 6. Number of VoIP vs. Average Packet Delay
(Inter Arrival Time is 10 ms)

As the best-effort traffic increases up to 10% the numbers of voice calls in this case are 24 and 13. This is shown in figure 7 and figure 8.

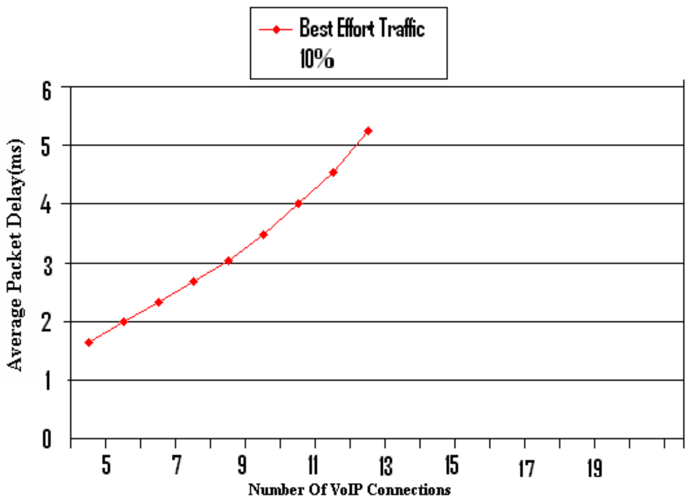

Figure 7. Number of VoIP Connections vs. Average Packet Delay
(Inter arrival Time is 20 ms)

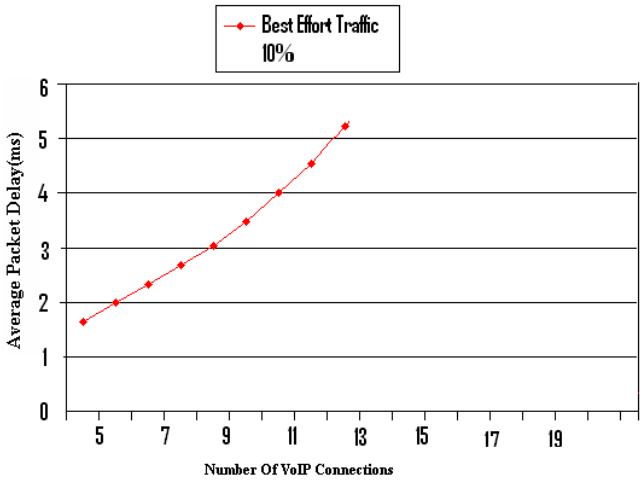

Figure 8. Number of VoIP Connections vs. Average Packet Delay
(Inter arrival Time is 10 ms)

VII. CONCLUSION

In this work we studied the VoIP over the 802.11 networks, for the perspective of number of the connections that an access point can support. We found that the maximum number of full duplex VoIP connections supported by 802.11b is 26. Our access media mechanism improves on the maximum number of VoIP connections supported. This enhancement is due to efficient utilization of the available bandwidth hence will support to multimedia real time applications as well.

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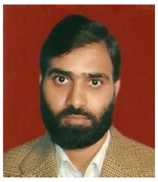

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