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Performance Analysis of Multi Traffic over Multi Rate EDCA of WLAN Network

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Abstract

Enhanced distributed channel access (EDCA) is used for transmitting good quality of service of multimedia traffic in wireless local area network (WLAN). Applications like Constant bit rate (CBR), file transfer protocol (FTP) and voice over internet protocol (VoIP) are used in wireless LAN network. Audio data is assign as high priority (AC_VO) and other such as CBR & FTP type of data are assigned as low priority (AC_BE) in IEEE 802.11b network. G.711 provides a very good perceived audio quality in comparison to other audio CODECs. The G711 codec is used in VoIP call with 20 msec of audio data. In this paper, analysis of the effects on CBR& FTP traffic with increased call rate of VOIP and payload sizes of CBR & FTP traffic in WLAN environment for real time applications are made. Jitter buffer used in RTP protocol is improving the quality of service (QoS) of multi traffic in comparison to RTP protocol without jitter buffer.

Keywords: *EDCA, G.711 Codec, VoIP, CBR, FTP, priority, RTP protocol, jitter buffer*

1. Introduction

Wireless LAN [1] technology provides a very good business model as it uses free unlicensed frequencies and provides a wireless last hop to IP networking which is also free. The IEEE 802.11 standard provides two MAC methods: Distributed Coordination Function (DCF) and Point Coordination Function (PCF). The original standard IEEE 802.11 failed to provide the required quality of service (QoS) performance as it serves all transmitted frames with the same level of priority. To provide a better QoS a new standard called IEEE 802.11e was deployed by enhancing the original standard [2].

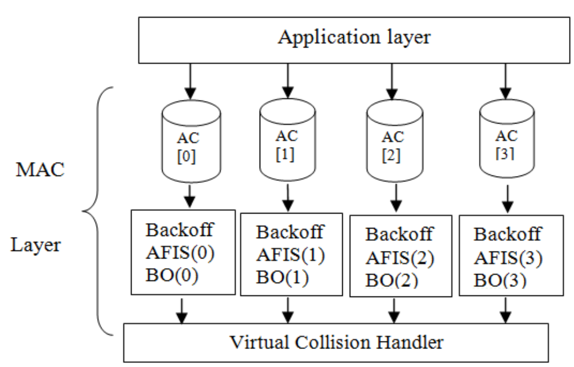

Figure 1. IEEE 802.11 Architecture

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Table 1. 802.11e EDCA Parameters

Parameter	AC [0] =BK	AC [1]= BE	AC [2]= VI	AC[3] =VO
Precedence	0	1,2	3,4,5	6,7
AIFS	7	3	2	2
TXOPlimit	0	0	6.016 ms	3.264 ms
Cwmin	31	31	15	7
Cwmax	1023	1023	31	15

In multi-rate operation, Access point (AP) selects the appropriate transmission rate from a set of possible rates based on wireless channel conditions. Algorithms for multi-rate operation are broadly classified into two categories [3]: Signal-to-Noise-Ratio (SNR) measurement based and Statistical Count based. SNR measurement based schemes include Automatic Rate Fallback (ARF) algorithm which measures SNR at the receiver and convey it to the transmitter. ARF algorithm is implementing with channel parameters as shown in Table 2.

Table 2. 802.11b Physical Layer Parameters for Multi-Rate

data rate	Receiver sensitivity
1 Mbps	-94.9 dbm
2 Mbps	-91.0 dbm
5.5 Mbps	-87.0 dbm
11 Mbps	-83.0 dbm

The paper is organized as follows. Section 2 represents related work. Section 3 defines proposed scenario. Section 4 defines performance evaluation parameters. Section 5 shows results and discussion. Section 6 includes conclusion and future work.

2. Related Work

The performance of the IEEE 802.11b wireless local area networks has been analyzed in previous research work. The performance of IEEE 802.11e EDCA is very effective service as compared with IEEE 802.11 DCF From the results, it was concluded that IEEE 802.11e introduces a very effective service differentiation mechanism to provide QoS support. Evaluated at heavy load of data has been high priority ACs *i.e.*, AC_VO and AC_VI suffer from greater number of collisions due to small CWmin and CWmax values [4]. Different traffic types like CBR, VBR and FTP are used to completely analyze the performance differences between two standards (DCF and EDCA). The enhanced standard (EDCA) is found better in protecting and providing less delay in medium access to high priority flows [5]. In case of EDCF, as there was a separate queue for each priority traffic, a minor change in throughput, latency and drop ratio was observed for voice as compared to DCF [6].The EDCA QoS mechanism produced significant improvements in the transmission of voice, which was consequential in the case of highly loaded network. Measurement results had shown that the use of QoS mechanisms reduced values of the mean and maximum delay, jitter and lost packets as compared to the case without QoS mechanism [7].When G711 VoIP connections is reduction in the bandwidth available for data traffic by approximately 900 Kbps [8].The MOS (mean opinion score) value of different VOIP CODECs (G.711) found was 4.48322 without background traffic (data) over 802.11e and 802.11b [9] Dynamic jitter buffer schemes give the highest user satisfaction in VoIP codec (G.711) and WLAN environments where compared without dynamic jitter buffer schemes [10].

The current work is distinguished from a majority of the previous literature. First, the performance of the CBR and FTP generic traffic by changing the parameters CBR and FTP generic traffic respectively. Finally, we measurement the performance of CBR & FTP generic traffics by increasing connection of VoIP with and without jitter buffer is done. Therefore the research work represents performance of different types of traffics by using or not jitter buffer in VoIP traffic.

3. Proposed Scenario

Simulations have been performed in QualNet 5.2 to deploy multi traffic application in a single QoS-enabled Basic Service Set (QBSS). Multi traffic such as constant bit rate (CBR), file transfer protocol (FTP) and voice over internet (VoIP) traffic studies are done in this paper. Eight stationary nodes (stations) are considered with single Access point, a single channel and single basic service set (BSS), which is a reasonable assumption of capacity. All even stationary nodes are assumed as sender and all odd nodes are assumed as destination whereas one node is assumed as AP. We have developed a system model for evaluating the performance of multi traffic over multi rate EDCA of WLAN.

A generic term used to describe the techniques used to carry voice traffic over IP. G.711 codec used in VoIP. In order to establish H.323 connections, two protocols RTP and RTCP must be set up between the two devices. RTP/RTCP protocols are defined in RFC 3550 [11].

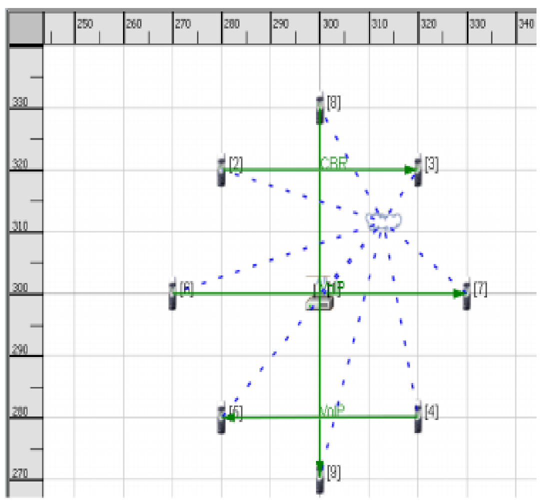

Figure 2. Network Topology for Stationary Nodes with Multi Traffics

Single CBR or FTP traffic transmits packets from node 2 (server) to node 3 (client) with different parameters of CBR or FTP traffic respectively. Analysis of the parameter of QoS of CBR & FTP is made. Evaluation of the performance of a single traffic with increasing call rate over multi rate EDCA of WLAN is done. Total three calls are established in it. First call is established between Node 6 (client) to node 7 (server). Second call is established between Node 4 (client) to node 5 (server).Third call is established between node 8 (client) to node 9 (server). A single CBR traffic transmits 1000 packets with 1472 bytes in 5 msec. A single FTP traffic transmits 1000 packets with 1472 bytes. Time for simulation is 6 minutes.

Table 3. Parameters of Multi Traffic

Type of traffic	Parameter value		
	Payload size	Number of packet	Time interval
CBR	1472 bytes	1000 packets	5 ms
FTP/GEN	1472 bytes	1000 packets	-
VoIP	G.711 CODEC (160 bytes per packet at 20 ms)		

Table 4. Simulation Parameters

Parameters	Value
Terrain Size	600 *600 meters
Physical protocol	IEEE 802.11b
MAC protocol	IEEE 802.11e,EDCA enable
Short Packet Transmit Limit	7
Long Packet Transmit Limit	4
MAC Propagation Delay	1 µs
Beacon Interval	200 TU
Network Protocol, IPV4	Fragment Unit 2048 bytes, Hold time 15sec
Number of IP Output Queues	3 with strict priority
Routing Protocol	Bellman Ford
IP Input Queue Size	150 Kbytes
Transport Protocol	RSVP, UDP and TCP-lite enabled
Maximum Segment Size	512 Byte
Send/Receive Buffer Size	16384 bytes buffer
QoS application	VoIP, CBR & FTP generic
Average time	20 sec
Connection Delay	8 sec
Call Timeout	60 sec
Packetization Interval	20 msec
Total Loss Probability	5.07
Multimedia signaling	H323: Direct call model, No Gatekeeper
RTP	Enabled
RTCP Session Bandwidth	64 Bps

4. Performances Evaluation

The following QoS performance metrics of multi traffic are considered for performance evaluation [11].

4.1. Throughput: It is the number of bits passed through a network in one second. The throughput of the network is finally defined as the average of the throughput of all nodes involved in data transmission. Therefore, the throughput can be stated as:

$$\text{Throughput } = \frac{(\text{total bytes received} * 8)}{(\text{session ended at} - \text{session initiated at})}$$

4.2. Average End-to-End Delay: Average End-to-End delay indicates the length of time taken for a packet to travel from source to the destination. The average delay is calculated by taking the average of delays for every data packet transmitted.

$$\text{Avg. delay}_{\text{end}-\text{end}} = \frac{(\text{Total trans. delay of all received pkts})}{(\text{Number of pkts received})}$$

$$\text{Trans. delay} = (\text{Time}_{\text{pkts received at server}} - \text{Time}_{\text{pkts transmitted at client}})$$

4.3. Average Jitter: Jitter is the difference of packet transit delays between two consecutive packets. Thus early, late, or out of sequence arrival of packets will cause jitter.

$$\text{Avg. Jitter}_{\text{end}-\text{end}} = \frac{(\text{Total pkt jitter of all received pkts})}{(\text{Number of pkts received} - 1)}$$

4.4. Packet Loss Rate: The difference of total packets sent and total packets received give the total packet loss. Packet loss is related to delivery ratio as follows:

$$\text{PDR}\ (\%) = \frac{(\text{Total pkt received})}{(\text{Total pkts sent})} \times 100$$

$$\text{PLR}\ (\%) = 100 - \text{packets Delivery Ratio(PDR)}$$

5. Results and Discussion

This section include following results of multi traffic with different parameters of multi traffic over EDCA.

5.1. Increase Payload Size of Traffic

The simulation results indicate that the maximal achievable payload data rate of CBR and FTP traffic in this scenario is approximately 2.4 Mbps and 0.741 Mbps respectively which is achieved when the payload size of traffic is chosen to be 1472 bytes. With an increase the payload size in each packet of traffic, the number of bits in each packet also increases. As a result, increasing the packet size of traffic will increase the throughput of traffic whereas average delay and average jitter of traffic decreases.

As show in the graph payload size of each packet of CBR & FTP traffic is varies 50 bytes to 1500 bytes. Figures 4, 5 and 6 shows the result of different parameters of QoS of multi traffic (CBR & FTP) when payload size of each packet is increased in a single sender node over multi rate EDCA and 802.11b.

5.1.1. Effect on Throughput of Multi Traffic: The throughput of CBR traffic increases and varies from 80.8 kbps to 2.40 Mbps and also FTP traffic varies from 0.83 Mbps to 0.741 Mbps.

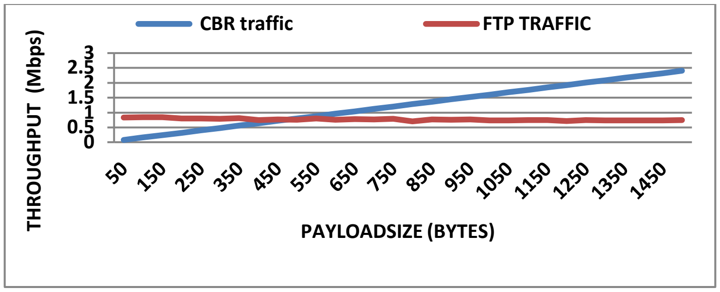

Figure 4. Throughput Vs Payload Size of Traffic

5.1.2. Effect on Average Delay of Multi Traffic: Average delay of CBR traffic varies 2.61 msec to 11.66 msec and also FTP traffic varies from 119.23 msec to 98.73 msec.

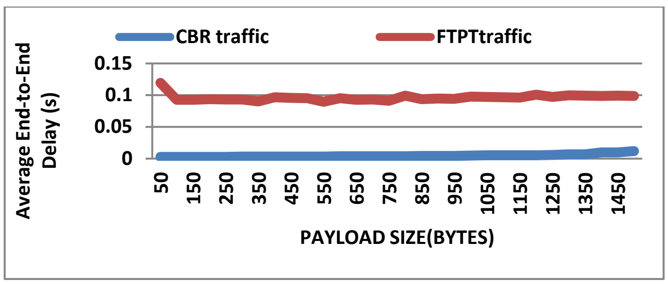

Figure 5. Average End to End Delay Vs Payload Size of Traffic

5.1.3. Effect on Average Jitter of Multi Traffic: Average jitter of CBR traffic varies from 0.313 msec to 2.22 msec and also FTP traffic varies from 0.827 msec to 8.20 msec.

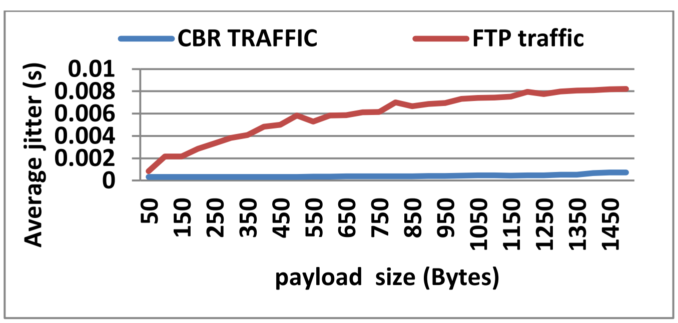

Figure 6. Average Jitter Vs Payload Size of Traffic

5.2. Increased Calls Rate

Firstly three VoIP calls are established without using background data such as CBR & FTP traffic in WLAN network. G711 codec used in VoIP, each frame is transmitted from source to destination within 20ms. IP/UDP/RTP protocols are used to send the codec source to destination. The audio data packets are sent every 20ms and payload of each packet is 160 bytes. Analysis show that maximum throughput of G 711 codec is 39.293 Kbps, Mean opinion score (MOS) of G711 is 3.29 and Packet loss is 2% when three calls are established. For the three calls, the quality of all connections is good. Thus, the experiment indicates number of VoIP connections is 3 in a single cell of an IEEE 802.11b network.

Multi traffic such as voice and data traffic are considered in the wireless network. One station transmits the data traffic such as CBR & FTP, whereas other stations transmit voice data from sender to destination. All connection terminals are established in the wireless network. Table 3 shows observation of different performance of CBR and FTP traffic with increasing VoIP call rate. Figures 7, 8, 9 and10 shows the result of different parameter of QoS of multi traffic when increase call rate using with and without jitter buffer.

5.2.1. Effect on Throughput of Multi Traffic: Throughput of CBR & FTP traffic decreases and average delay and average jitter increased with increase call rate. Throughput of CBR and FTP traffic is 2.36 Mbps and 0.679 Mbps respectively without call established. Throughput of CBR traffic is 2.36 Mbps, 2.06 Mbps, 1.977 Mbps and 1.624 Mbps without using jitter buffer in RTP protocol and 2.36 Mbps, 2.06 Mbps, 2.04 Mbps and 1.719 Mbps with using jitter buffer in RTP protocol when increase call rate (0, 1, 2, and 3) in this scenario.

Throughput of FTP traffic is 0.679 Mbps, 0.635 Mbps, 0.563 Mbps and 0.442 Mbps without considering jitter buffer and 0.679 Mbps, 0.635 Mbps, 0.566 Mbps and 0.456 Mbps with jitter buffer when increase call rate (0, 1, 2, and 3). All observation shows in Figure 7. It is analyzed that with an increased call rate, the number of packets in VoIP traffic also increases. It affects the throughput of traffics such as CBR & FTP.

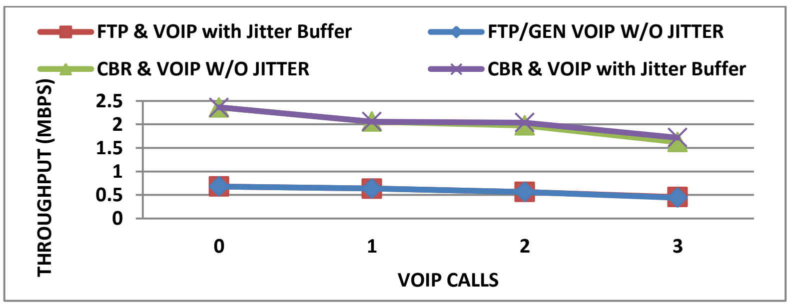

Figure 7. Throughput Vs VoIP Calls

5.2.2. Effect on Average Delay of Multi Traffic: Average delay of CBR and FTP traffic varies with increased call rate. Average delay of CBR traffic is 0.038598 s without call establishment. Average delay of CBR traffic is 0.038598 s, 0.399503 s, 0.532191 s and 0.562358 s without consider jitter buffer in RTP protocol and 0.038598s, 0.399503s, 0.4974s and 0.537055s with consider jitter buffer in RTP protocol when increased call rate.

Average delay of FTP traffic is 0.008889s, 0.009586s, 0.010801s and 0.013773s with increase call rate without jitter buffer and 0.008889s, 0.009586s, 0.010743s and 0.013348s with increase call rate with jitter buffer. Average delay of CBR traffic is more increase as compare FTP traffic. Average delay is increasing with decrease throughput.

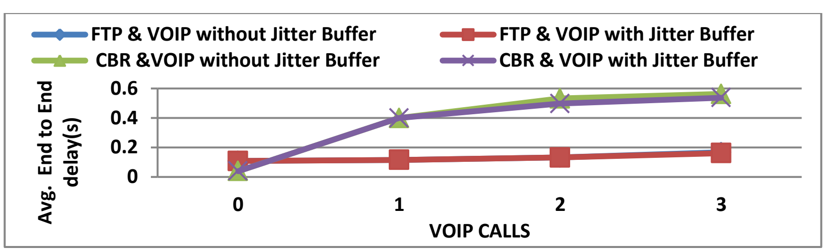

Figure 8. Average End to End Delay Vs VoIP Calls

5.2.3 Effect on Average Jitter of Multi Traffic: Average jitter of CBR and FTP traffic also changes with increased call rate. Average jitter of CBR and FTP traffic is 0.001091s and 0.107346s respectively without any call establishment. Average jitter of CBR traffic is 0.001091s, 0.00276s, 0.003064s and 0.004124s with increase call rate (0, 1, 2, and 3) without jitter buffer and 0.001091s, 0.00276s, 0.002684s and 0.003887s with increase call rate with jitter.

Average jitter of FTP traffic is 0.107346s, 0.114899s, 0.131762s and 0.166481s with increase call rate without jitter buffer and 0.107346s, 0.114899s, 0.131416s and 0.160754s with increased call rate with jitter buffer. Average jitter of CBR and FTP traffics are decreasing with increasing throughput of CBR and FTP traffic.

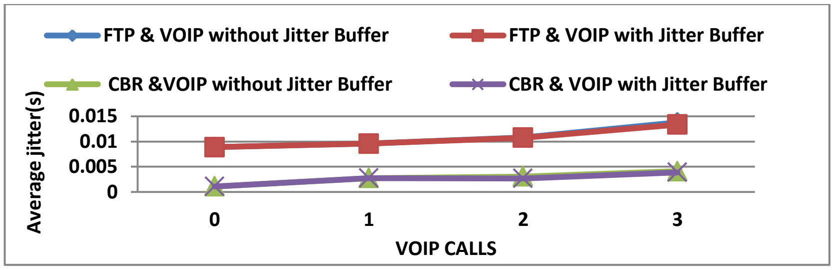

Figure 9. Average Jitter Vs VoIP Calls

5.2.4. Effect on Packet Loss Rate of Multi Traffic: Variation of packet loss rate of CBR is also observed with increase call rate. No packet loss rate in CBR and FTP traffic without any calls is observed. Packet loss rate of CBR traffic is 0%, 5.2%, 6.9% and 19.7% without assuming jitter buffer and 0%, 5.2 %, 5.3% and 18.3% with jitter buffer and increase call rate. No Packet loss rate in FTP traffic with increase call rate.

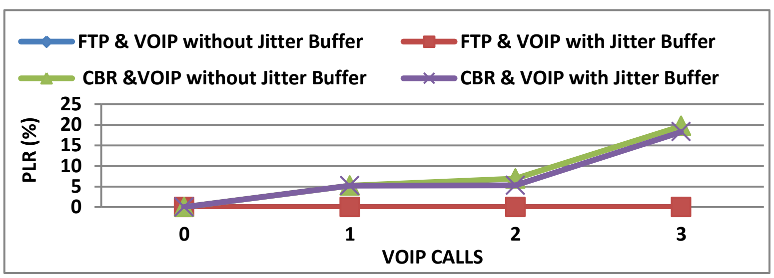

Figure 10. Packet Loss Rates Vs VoIP Calls

Therefore only one channel is shared between voice and data traffic. Large number of packets of voice is send as compared with data traffic sender to destination. Data traffic sends only 1000 packets. Each voice sender sends more than 4000 packets (4000 *160 bits = 640000 bits) of G711 audio data when call is established between senders to destination for 5 minute. With heavy load of data due to increased calls, throughput of data traffic decreases whereas average delay and average jitter of data traffic increases.

5. **Conclusion and Future Work**

This paper presents the performance evaluation of multi traffic over a key emerging technology (IEEE 802.11e) by using the network modeling and simulation tool QualNet 5.2. Simulations have been performed to investigate the effect of quality of service of multi traffic with changing parameters of multi traffics (CBR / FTP generic). It is observed that throughput of single CBR traffic is greater than a FTP generic. The throughput, average delay and average jitter of single CBR & FTP traffic are 2.36 Mbps, 0.038595 s and 0.001091s respectively and 0.67989 Mbps, 0.008889 s and 0.107346s respectively. Increased call rate is affects the QoS of multi traffic (CBR & FTP traffic). Throughput of traffic increase with decrease average delay and average jitter when payload size of CBR & FTP traffic increased. Throughput of traffic decreases with increasing average delay, average jitter and packet loss rate when call rate increased. Dynamic jitter buffer used in VoIP codec has subtle improve quality of service of multi traffic and WLAN environments considered.

For future work, the performance can be considered and it can also be performance for HCCA with real time application. This technique can be further applied on mobiles nodes.

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