Capacity Analysis of Combined IPTV and VoIP Over IEEE 802.11n

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***Abstract*—Internet Protocol Television (IPTV) and Voice over Internet Protocol (VoIP) have gained unprecedented growth rates in the past few years. Data rate and high coverage area of IEEE 802.11n motivate the concept of combined IPTV and VoIP over IEEE 802.11n. Transmission of combined IPTV and VoIP over a wireless network is a challenging task. In this paper, we deal with the capacity evaluation of combined IPTV and VoIP over IEEE 802.11n. We evaluate the use of Datagram Congestion Control Protocol (DCCP) at transport layer of IPTV and VoIP. Our study shows that DCCP can enhance capacity of IPTV by 25%. Our study confirms that performance of DCCP deteriorates severely in presence of any other UDP flow because of congestion-less mechanism of UDP. Our fairness analysis with TCP traffic shows that IPTV and VoIP using DCCP provides fair share in bandwidth to TCP with 19% increase in combined capacity. We study the effect of IEEE 802.11n parameters and obtain optimal values. We show the optimal values and trends of Access Point (AP) parameters including Queue size, Transmission Opportunity, Aggregation and Block ACK etc. Our study shows that nearly 9 more VoIP users are supported with a queue size of 70 packets and transmission opportunity of 9. Our study concludes that selection of DCCP and optimized parameters over IEEE 802.11n can enhance the capacity of IPTV and VoIP by atleast 25% and 19% respectively as compared to the use of UDP.**

***Keywords*- IPTV; DCCP; IEEE 802.11n.**

I. Introduction

Internet Protocol Television (IPTV) has gained tremendous growth rates in the past few years. Five times increase in IPTV traffic is expected from 2011 to 2016 [1]. Economic, video-on-demand and interactivity are key benefits of IPTV. IPTV transmits video, audio and data simultaneously to end users with satisfactory Quality of Service (QoS) [2]. Aside from IPTV, Voice over Internet Protocol (VoIP) is another fastly growing internet application. VoIP users are expected to increase to 928 millions by 2016 [1]. The Success of VoIP is attributed to cheaper rates, conference calls and less hardware cost. By 2016, half of the world would be using wireless Access Point (AP) at user end [1]. Among wireless standards, IEEE 802.11n is the latest standard providing maximum data rate with better coverage. Provision of wireless IPTV and VoIP at the user end with good QoS is a challenging task. Less packet loss, less delay, better QoS, more capacity and the fair allocation of resources to all traffic categories are key challenges for IPTV and VoIP.

Access link provides the major hindrance in providing satisfactory QoS because packets drop at the queues of AP and bandwidth at access links is limited. Freedom of mobility, less installation cost and ease of access demands the provision of wireless IPTV and VoIP. This requires an insight investigation on combined IPTV and VoIP users having good QoS with wireless access link. Evaluation of capacity is essential for deployment of real time applications including IPTV and VoIP over wireless networks. Our previous study [3] shows that use of DCCP with optimal frame aggregation size over IEEE 802.11n can enhance the capacity of IPTV.

IPTV and VoIP are delay sensitive applications. They use UDP at transport layer with no congestion control mechanism which causes large packet loss and delay in congested situations. In recent past, Datagram Congestion Control Protocol (DCCP) has emerged as a suitable transport layer protocol in place of UDP for streaming media applications. DCCP provides congestion control mechanism without any retransmissions. DCCP variant TCP Friendly Rate Control (TFRC) provides less reliability but gives a less variation in data rate which makes it suitable for time-sensitive applications. Previous studies [4]-[5] have studied the performance of DCCP and its fairness with TCP traffic but no evaluation has been made for combined IPTV and VoIP over DCCP to the best of our knowledge. Our study shows that use of TFRC for IPTV and VoIP can significantly increase network capacity by varying its data rate in congested situations. 80% of network users use TCP traffic so we also analyze IPTV and VoIP with TCP traffic [6]. In presence of UDP, TCP decreases its data rate which results in the starvation of TCP flows [7]. Our fairness analysis shows that TFRC for IPTV and VoIP gives fair share of bandwidth to TCP traffic than UDP.

IEEE 802.11n standard provides maximum data rates over IEEE 802.11 a/b/g. IEEE 802.11n has enhanced features including Frame Aggregation, Block Acknowledgement, Transmission Opportunity (TXOP) and Multiple-Input Multiple-Output (MIMO). Our study shows that optimal parameter selection of Queue size, TXOP and aggregation size can significantly enhance capacity of IPTV and VoIP over IEEE 802.11n. Our analysis deals with the optimal values and trends of SIFS, DIFS, Contention Window size and Physical header for IPTV and VoIP over IEEE 802.11n.

Our main contributions in this paper include, (i) To evaluate the capacity of combined IPTV and VoIP over IEEE 802.11n, (ii) To evaluate the effectiveness of TFRC for IPTV

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and VoIP over IEEE 802.11n, (iii) To determine the capacity of IPTV and VoIP using TFRC in presence of TCP traffic in the network, (iv) To estimate optimum values of IEEE 802.11n AP parameters e.g., Queue size, TXOP, Block Ack etc. for combined IPTV and VoIP traffic.

The remainder of this paper is organized as follows: In Section-II, we present the related work for wireless IPTV, VoIP and DCCP. Section-III explains the network and simulation parameters while Section-IV presents capacity of IPTV and VoIP using UDP. In Section-V, we evaluate the use of TFRC for IPTV or VoIP or both. In Section-VI, we present capacity and fairness analysis for IPTV and VoIP along with TCP traffic in the network. Section-VII evaluates optimal parameter values of IEEE 802.11n AP. In Section-VIII, we present the conclusion.

II. Related Work

The provision of wireless IPTV at end user has been a challenging task which made it a major focus of numerous studies. In [8], the authors suggest that IPTV performance metrics should be studied over Wireless Local Area Networks (WLANs) because IPTV is a killer application for next generation internet. In [9], the authors study IPTV capacity over IEEE 802.11b/g networks. They conclude that 2 and 6 IPTV streams can accommodate on IEEE 802.11b and IEEE 802.11g networks respectively. They show that a non-linear relationship exists between IPTV capacity and data rate of WLAN. In [10], the authors study the performance of wireless IPTV by varying number of hops and buffer size at AP. They compare the performance results with fluid model flow analysis and conclude that hop count and buffer size are major factors effecting capacity of IPTV users. In [11], the authors experimentally verify the performance of IPTV over IEEE 802.11n. They show that indoor environment can provide much better QoS than the outdoor environment.

VoIP has been studied over IEEE 802.11 WLANs in various studies. In [12], the authors evaluate upper bound capacity limit for VoIP users over IEEE 802.11b WLAN through simulations. They conclude that IEEE 802.11b can support 3 to 12 simultaneous VoIP calls based on codec and packetization interval and transmission rate in wireless medium. In [4], the authors evaluate VoIP capacity over IEEE 802.11b/g WLANs through simulations and experiments. They show that there is a 30% decrease in combined VoIP and tracking capacity at higher packetization intervals in comparison to VoIP only capacity. In [13], the authors evaluate experimental performance of VoIP over IEEE 802.11b testbed and compare it with theoretical and simulation results. They show that VoIP exhibits capacity of 15 calls with a packetization interval of 20ms.

Combined IPTV and VoIP over wireless LAN has been studied in [14]. The authors used reconfigured IEEE 802.11 b/g AP. They show through simulations and experiments that 3 IPTV streams can be provided along with VoIP connected with satisfactory QoS over 2 hops in the network. They conclude that IPTV and VoIP capacity is limited in IEEE 802.11b/g networks and it is expected to enhance in IEEE 802.11n network because of better data rate provision.

IEEE 802.11n MAC enhancements have been evaluated in [15]. The authors conclude that the enhanced MAC and physical layer features of IEEE 802.11n can significantly enhance capacity of VoIP. In [16], the authors study frame aggregation mechanism and show that upto 95% channel utilization can be increased for UDP traffic by using frame aggregation. Their study shows that combining multiple physical layer data units is more effective than combining multiple MAC layer data units because of physical bit error rate. In [17], the authors show that frame protection mechanism of legacy WLANs deteriorates the throughput performance of IEEE 802.11n WLAN. They show through simulations that IEEE 802.11n can provide better QoS than legacy WLANs.

In [4] and [5], the authors show that use of DCCP as transport layer protocol can enhance performance for delay sensitive applications including VoIP, video streaming and gaming. These studies show that change in data rate in congested situations can enhance network capacity by decreasing packet loss. They show that DCCP gives a fair share in bandwidth to TCP than UDP. Our previous study [3] shows that use of TFRC can enhance capacity of IPTV.

Comparison of work in existing literature [3]-[17] show that studies on combined wireless IPTV and VoIP are limited because of low data rates of IEEE 802.11b/g networks. High data rates of IEEE 802.11n motivate the concept of combined wireless IPTV and VoIP. To the best of our knowledge, very limited research exists on combined IPTV and VoIP using DCCP over IEEE 802.11n.

III. Network Scenario

We evaluate the capacity of wireless IPTV and VoIP through simulations by using open source simulation tool ns2 [18]. Currently, ns2 lacks implementation of IEEE 802.11n module. We develop an IEEE 802.11n module compatible with the DCCP module. At the physical layer, we enhance MIMO scheme with configurable TXOP and Queue size. At MAC layer, we develop aggregation support for multiple Medium access control Service Data Unit (MSDU). Our module is an enhancement of previous modules [15] and [19] which now contain features of TXOP, access categories with timers, enhanced distributed channel access mechanisms. We enhance aggregation mechanisms to provide Aggregate Medium access control Protocol Data Unit (A-MPDU) support. Our modification includes the packet format modifications for IEEE 802.11n and DCCP module.

Simulated architecture is shown in Fig. 1 with IPTV and VoIP server on one side of the internet and access nodes located in vicinity of AP. IEEE 802.11n AP has been implemented by use of default MAC and physical layer values from [20] with parameters as shown in Table-I. We use a physical layer data rate of 300 Mbps. Our previous study [3] shows aggregation of 4 packets as optimal size for IPTV traffic. We will use this aggregation size in all our simulations for all traffic types.

TABLE I
802.11N ACCESS POINT PARAMETERS

Parameter	Value	Parameter	Value
DIFS	34 μs	SIFS	16 μs
Slot time	9 μs	Physical Header	20 μs
Contention Window min	15	TXOP limit	5
Channel Bandwidth	40 MHz	Bit error rate	0.000008

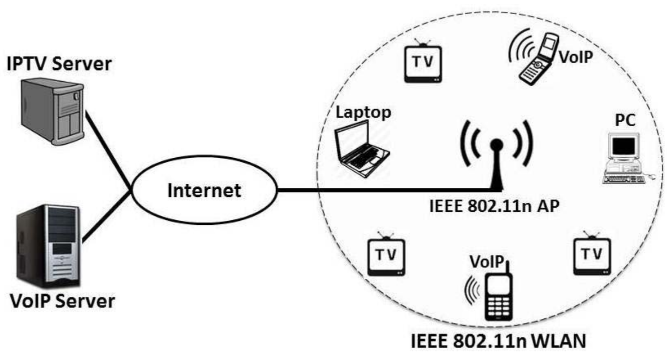


Fig. 1. IPTV and VoIP Network Scenario

To evaluate the capacity of IPTV and VoIP, we calculate the bandwidth requirements for IPTV and VoIP. We evaluate IPTV bandwidth requirement (D) by use of resolution (Horizontal Resolution R_H * Vertical Resolution R_V), frames per sec (F), bytes per pixel (B), Luminance and chrominance (C) and compression schemes. Equations (1) and (2) present the data rate required without and with compression schemes. N_{gop} and H_{comp} are number of pictures in compression group and efficiency of compression respectively. Table-II presents the data rates for IPTV using Standard Definition Television (SDTV) and High Definition Television (HDTV). We conduct analysis for both HDTV and SDTV.

$$D = R_H R_V C F B \tag{1}$$

$$D = R_H R_V C F B N_{gop} / H_{comp} \tag{2}$$

TABLE II
DATA RATE REQUIREMENT FOR VARIOUS COMPRESSION AND RESOLUTION SCHEMES

TV	F (fps)	C	Resolution $R_H * R_V$	Compression Scheme	Required Data Rate (Mbps)
SD	24	2	640 * 480	MPEG-2	3.93
SD	24	2	640 * 480	MPEG-4	2.36
HD	24	3	1920* 1080	MPEG-2	26
HD	24	3	1920* 1080	MPEG-4	15.92

We implement VoIP with the use of popular codec G.711 with a data rate of 64 kbps. VoIP packet size is 80 bytes (excluding headers) while packetization interval is 10ms.

IV. CAPACITY OF IPTV AND VOIP USING UDP

We simulate IPTV and VoIP in ns2 by turning on VoIP and IPTV users one-by-one. Both IPTV and VoIP use the default UDP protocol at transport layer. To evaluate the capacity of IPTV, we take a packet loss of 1% and delay of 200 ms as default threshold values [21]. Packet loss and delay threshold values for VoIP are 2% and 150 ms respectively [4][12]. Our objective in all simulations is to evaluate the effect and trends of VoIP users capacity by increasing IPTV users in the network. We aim to evaluate the capacity relationship between IPTV users and VoIP users.

Table-III shows the capacity evaluation results for HDTV and VoIP users. Our results show that 1 HDTV user and 36 VoIP users can be accommodated with IPTV packet loss of 0.89% and VoIP packet loss of 1.79%. As we increase the number of HDTV users, capacity of VoIP decreases by nearly 33%. Hence, 2 HDTV users support 24 VoIP users while 3 users can accommodate 11 VoIP users only. Maximum number of HDTV users appear to be 4 which allow 2 VoIP streams simultaneously. Analysis shows that addition of IPTV users decreases capacity for VoIP users. IPTV requires comparatively larger bandwidth than VoIP users which disrupts the support for large number of VoIP users. Net available bandwidth remains same in the system but distribution of bandwidth changes with the addition of IPTV users. Similarly, capacity of VoIP users decreases with the increase of SDTV users. Maximally, 17 SDTV users and 3 VoIP streams can be accommodated simultaneously. Our study shows that more SDTV streams can be accommodated with VoIP than HDTV because SDTV consumes less bandwidth than HDTV.

TABLE III
CAPACITY OF COMBINED IPTV AND VOIP

HDTV	VoIP	HDTV		VoIP	
Users	Users	Packet loss	Delay(ms)	Packet loss	Delay(ms)
1	36	0.89%	152	1.79%	132
2	24	0.92%	161	1.93%	125
3	11	0.87%	170	1.83%	131
4	2	0.96%	175	1.9%	141

V. CAPACITY OF IPTV AND VOIP USING TFRC

In this section, we evaluate the use of Datagram Congestion Control Protocol (DCCP) at the transport layer of combined IPTV and VoIP. DCCP has two famous variants namely TCP-like and TFRC [22]. Our previous study [3] shows that TFRC is the better suitable protocol for time sensitive applications like IPTV than TCP-like. TFRC provides less reliability with more timely response and uses no retransmissions. TFRC uses segment size, round trip time and packet loss rate to determine the sending rate continuously and adjusts itself according to varying congested situations [22].

The aim of our simulations is to study the suitability of TFRC for IPTV and VoIP. To test this, we take three scenarios for simulations. In the first scenario, TFRC based IPTV runs with UDP based VoIP. In the second scenario, UDP based IPTV runs with TFRC based VoIP. In the third scenario, TFRC based IPTV runs with TFRC based VoIP. Our study would reveal the optimum selection of transport layer protocol for IPTV and VoIP.

A. *IPTV using TFRC and VoIP using UDP*

We configure IPTV for using TFRC in place of UDP while VoIP uses UDP. VoIP users are increased with IPTV users until the delay and packet loss cross the threshold values. Table-IV shows the IPTV and VoIP capacity trends by varying number of users. Our results show that 1 HDTV user can accommodate upto 7 VoIP users. The 8^{th} VoIP user makes TFRC to decrease its sending rate and delay crosses threshold value for HDTV. Similarly, 2 and 3 HDTV users can accommodate 5 and 3 VoIP users respectively. 4^{th} HDTV user can run only if there is no VoIP running on UDP. Similar behavior is seen for SDTV streams where 4 SDTV channels can accommodate 6 VoIP users. Maximally, 17 SDTV users can accommodate no VoIP stream simultaneously. Analysis shows that addition of IPTV users deteriorates systems capacity by increasing queue congestion at wireless AP. As a result, packets drop from queue of AP and capacity of VoIP users is limited.

TABLE IV
CAPACITY OF COMBINED IPTV(TFRC) AND VOIP(UDP)

HDTV Users	VoIP Users	HDTV		VoIP	
		Packet loss	Delay(ms)	Packet loss	Delay(ms)
1	7	0.92%	171	1.9%	127
2	5	0.89%	188	1.85%	115
3	3	0.97%	179	1.89%	135
4	0	0.76%	165	2.52%	139

B. IPTV using UDP and VoIP using TFRC

In this subsection, we configure our simulations such that VoIP uses TFRC while IPTV uses UDP. Our results from Table-V show that net capacity deteriorates by using TFRC for VoIP. 1 and 2 HDTV users can accommodate 8 and 6 VoIP users respectively. Similarly, 3 and 4 HDTV users can arrange 4 and 1 VoIP calls respectively. Similar trends are observed for SDTV streams where 4 SDTV streams can accommodate 6 VoIP streams. Maximally, 17 SDTV users can arrange no VoIP connection simultaneously. Main reason for less capacity of VoIP is attributed to the use of TFRC. Our results also show that UDP based applications cannot run with fairness with TFRC based applications because of congestion-less mechanism of UDP.

TABLE V
CAPACITY OF COMBINED IPTV(UDP) AND VOIP(TFRC)

HDTV Users	VoIP Users	HDTV		VoIP	
		Packet loss	Delay(ms)	Packet loss	Delay(ms)
1	8	0.89%	163	1.86%	123
2	6	0.95%	174	1.82%	135
3	4	0.97%	169	1.91%	138
4	1	0.87%	182	1.96%	143

C. IPTV using TFRC and VoIP using TFRC

To test the performance of combined IPTV and VoIP over TFRC, we simulate IPTV and VoIP with TFRC as transport layer protocol. Same setup is simulated by varying number of users of IPTV and VoIP. Results from Table-VI show that 1 and 2 HDTV users can accommodate 35 and 22 VoIP connections respectively simultaneously. Maximum 5 HDTV users can accommodate 1 VoIP user simultaneously. Similar trends are seen for SDTV channels where 4 SDTV channels can accommodate 33 VoIP channels simultaneously. Maximally, 22 SDTV streams can allow 1 VoIP connection simultaneously. These results clearly show that TFRC increases capacity of IPTV and VoIP users if both applications use same protocol. Analysis shows that TFRC adjusts its data rate in congested situations. In the absence of UDP based applications, TFRC fits better for IPTV and VoIP by sharing fair bandwidth. Large IPTV packet size is also an advantage for IPTV which reduces header overhead and increases capacity of IPTV users.

TABLE VI
CAPACITY OF COMBINED IPTV(TFRC) AND VOIP(TFRC)

HDTV Users	VoIP Users	HDTV		VoIP	
		Packet loss	Delay(ms)	Packet loss	Delay(ms)
1	35	0.85%	176	1.84%	125
2	22	0.82%	181	1.79%	137
3	11	0.76	158	1.61%	144
4	2	0.8%	164	1.74%	122
5	1	0.74%	179	1.7%	139

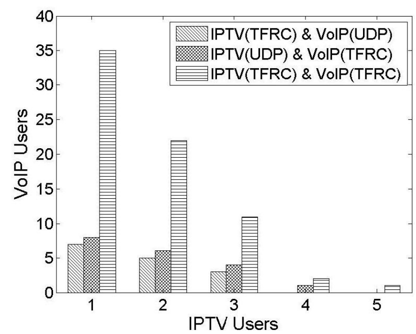


Fig. 2. IPTV and VoIP over multiple protocols

Our simulation results are summarized in Fig. 2. Comparison of various scenarios shows that TFRC is not suitable for IPTV only or VoIP only. Use of TFRC for combined IPTV and VoIP can increase net capacity. Based on these findings, we carry out further simulations using only UDP or TFRC for both IPTV and VoIP.

VI. CAPACITY AND FAIRNESS OF IPTV AND VOIP WITH TCP TRAFFIC

Studies suggest that non-real time traffic is four times greater than the real time traffic [6]. This motivates us to study TCP traffic along with IPTV and VoIP traffic in the network. Aim of our simulations is to evaluate the capacity and fairness of combined IPTV and VoIP traffic along with TCP traffic in the network.

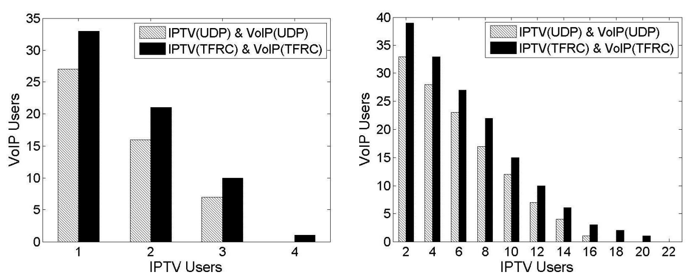
We have implemented File Transfer Protocol (FTP) at application layer which uses TCP at transport layer. We use packet size of 1000 bytes for simulations. Multiple connections (4) of TCP are established because multiple connections decrease their contention window sizes less abruptly than a single connection.

A. IPTV and VoIP using UDP

Our simulation results for IPTV and VoIP using UDP from Fig. 3(a) show that 1 HDTV user can accommodate 27 VoIP connections simultaneously. Similarly, 2 and 3 HDTV streams can accommodate 16 and 7 VoIP connections respectively. Throughput of TCP drops to nearly 0 in the presence of 4 HDTV streams. Fig. 3(b) shows similar trends for SDTV streams with a maximum capacity of 16 SDTV and 1 VoIP user. Our results show that VoIP capacity is significantly degraded in presence of TCP flows. Analysis shows that TCP goes to starvation in the presence of UDP flows which results in reduction of TCP congestion window size to nearly zero.

B. IPTV and VoIP using TFRC

We configure IPTV and VoIP using TFRC in an environment already running TCP traffic. Our results from Fig. 3(a) show that 1 and 2 HDTV users can accommodate 33 and 21 VoIP users respectively. Similarly, 3 and 4 HDTV users can arrange 10 and 1 VoIP user respectively. Fig. 3(b) shows similar trends for SDTV streams with a maximum capacity of 16 SDTV and 1 VoIP user. Our results show that capacity gained by IPTV and VoIP is much better by using TFRC than UDP. Analysis shows that congestion control mechanisms of TCP and TFRC adjust themselves according to the dynamic conditions of the network. This results in better bandwidth share between TCP and TFRC than TCP and UDP. Results show that VoIP capacity has increased with the use of TFRC than use of UDP.



(a) Using HDTV (b) Using SDTV

Fig. 3. Capacity of IPTV and VoIP with TCP traffic

TABLE VII
PERFORMANCE STATISTICS FOR COMBINED IPTV AND VOIP ALONG WITH TCP TRAFFIC

IPTV	Combined Traffic Flows	Average Throughput (Mbps)	Traffic Bandwidth Usuage(%)	VoIP capacity with TCP	VoIP capacity without TCP	Capacity decrease (%)
HDTV	UDP-TCP	UDP: 50.7 TCP: 6.8	UDP: 88.2 TCP: 11.8	27	36	25%
HDTV	TFRC-TCP	TFRC: 66.5 TCP: 14.2	TFRC: 82.4 TCP: 17.6	33	35	6%
SDTV	UDP-TCP	UDP: 51.3 TCP: 6.1	UDP: 89.4 TCP: 10.6	29	37	22%
SDTV	TFRC-TCP	TFRC: 64.7 TCP: 14.4	TFRC: 81.7 TCP: 18.3	39	41	5%

Table-VII presents the throughput results for IPTV and VoIP using UDP and TFRC with TCP traffic. Results show that TFRC for HDTV decreases VoIP capacity only by 6% while UDP for HDTV decreases VoIP capacity by 25% with TCP traffic. Throughput results show that TCP gets 14.2 Mbps in presence of HDTV (TFRC) while 6.8 Mbps in presence of UDP. These results show that TFRC is much more fairer to TCP traffic than UDP.

VII. Tuning IEEE 802.11n Access Point Parameters

Capacity and QoS can be significantly enhanced by selection of optimal parameters. Our aim is to estimate optimal values of Queue size, TXOP, Aggregation with Block Acknowledgement, SIFS, DIFS, Physical Header (PH) and Contention Window (CW) size. We use UDP at the transport layer of IPTV and VoIP for simulations. In last subsection, we would evaluate capacity with the use of DCCP along with optimal parameters of IEEE 802.11n and compare it with UDP for IPTV and VoIP. Since SDTV exhibits similar trends therefore from now-onwards IPTV refers to HDTV .

A. Effect of Queue Size

IEEE 802.11n AP uses queues to store data at AP before transmission to end user. Capacity is significantly degraded if packets drop at queues of AP. Our motivation is to find the optimal queue size for combined IPTV and VoIP over IEEE 802.11n. We vary the queue size from 5 to 500 packets and investigate the capacity of combined IPTV and VoIP.

Results of queue size trends from Fig. 4 show that maximum capacity is reached at a queue size of 70 packets. System capacity increases sharply initially by increasing queue size because larger queue size avoids packet loss at AP. There is a little increase in capacity from 50 to 70 packets because delay increases with large queue values. Above 70 packets, system's capacity remains constant to 4 IPTV users and 2 VoIP users because capacity to transfer packets to end user is the bottleneck. Our study shows that any increase in queue size above 70 packets is redundant. Optimal queue size for combined IPTV and VoIP over IEEE 802.11n is 70 packets.

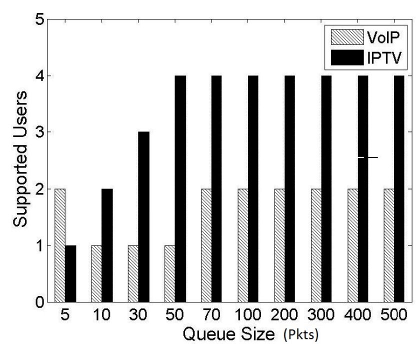


Fig. 4. IPTV and VoIP Capacity versus Queue size

B. Effect of Transmission Opportunity (TXOP)

TXOP for AP can provide contention free access to AP for a time equal to TXOP time limit. AP has load of all users packets so TXOP gives it more time for transmission. We vary TXOP in varying range of values to investigate IPTV and VoIP capacity.

Fig. 5 presents the results for IPTV and VoIP capacity versus TXOP of AP. Results show that capacity of IPTV becomes constant after TXOP of 5. Further increase in TXOP increases capacity by small margins which can translate to VoIP users only. TXOP of 9 is the maximum value which can accommodate 12 VoIP users with 4 IPTV streams simultaneously. Analysis shows that increase of TXOP allows the AP to send more packets with less collisions. After TXOP of 9, the bottleneck transfers from AP to user ends. TXOP of 10 gives less time to users for packet transmission which results in a constant capacity curve after TXOP of 9. VoIP trends are abrupt in Fig. 5 because any increase in throughput with TXOP accommodates the IPTV users first before arranging maximum VoIP users. Optimal TXOP value for combined IPTV and VoIP over IEEE 802.11n is 9.

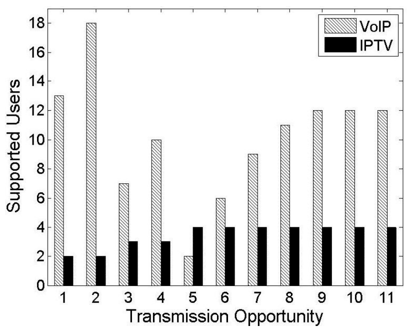


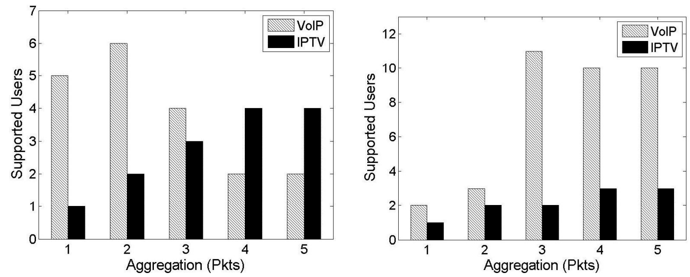
Fig. 5. IPTV and VoIP Capacity versus Transmission Opportunity

C. Effect of Aggregation and Block Acknowledgement (ACK)

IEEE 802.11n has been equipped with aggregation and Block ACK mechanism. Aggregation combines multiple frames at AP queues so that the overhead of transmission is reduced. Multiple MSDUs and MPDUs are combined at MAC layer and physical layer level. Acknowledgements of individual MPDUs can be sent using block ACK. This feature results in selective retransmission of MPDUs. To test the performance of aggregation and block ACK, we simulate two scenarios of IPTV and VoIP. The first scenario with low bit error rate while second scenario with high bit error rate.

1) Block ACK in Low Bit Error Rate: We evaluate IPTV and VoIP users capacity using block ACK in low bit error rate of 10^{-6}. Fig. 6(a) shows the results for IPTV and VoIP capacity without using block ACK. Results show that maximum capacity is reached at an aggregation of 4 packets. Maximally, 4 IPTV users can accommodate 2 VoIP users simultaneously. Results of IPTV and VoIP capacity using block ACK are shown in Fig. 6(b). Results show that Block ACK deteriorates capacity in low bit error rates. Maximally, 3 IPTV users can accommodate 10 VoIP users. Analysis shows that block ACK sends acknowledgements of individual MPDUs. In case of low bit error rate, corrupt packets are few in number so block ACK is not suitable. Time for block ACK can be utilized for data packets in low bit error rates.

2) Block ACK in High Bit Error Rate: We evaluate Block ACK performance by increasing bit error rate to 10^{-5}. Results of IPTV and VoIP without block ACK in high bit error rate are shown in Fig. 7(a). Results show that maximum capacity of IPTV and VoIP users is degraded in comparison to low bit error rate results. Maximally, 3 IPTV streams

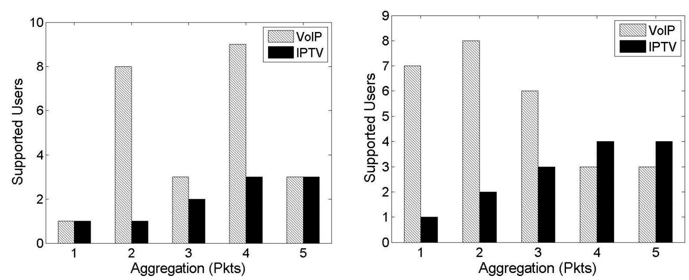


(a) Without Block ACK (b) Using Block ACK

Fig. 6. Capacity of IPTV and VoIP in Low BER

can accommodate 9 VoIP connections simultaneously. Same simulations are repeated by using block ACK in high bit error rate as shown in Fig. 7(b). Block ACK in high Bit Error rates provides 4 IPTV users with 3 VoIP users. Results show that block ACK increases capacity of IPTV and VoIP users. This trend occurs because only corrupted MPDUs have to be retransmitted instead of re-transmitting the whole packet containing multiple MPDUs. Trends of VoIP in Fig. 7 show variation because aggregation increases throughput. Increase in throughput is translated to a maximum number of IPTV streams. Only left over throughput is available for VoIP. Hence VoIP shows varying trend.

Block ACK results show that block ACK is suitable only in High bit error rate. Aggregation results show that aggregation of 4 packets is the optimal size for combined IPTV and VoIP over IEEE 802.11n.



(a) Without Block ACK (b) Using Block ACK

Fig. 7. Capacity of IPTV and VoIP in High BER

D. Effect of Short Interframe Spacing (SIFS)

SIFS duration is the idle time in the network immediately after transmission of RTS or data. We take a default SIFS value ($16 \mu s$) from [20] as a reference and modify default value in ratio to get the trends in capacity for SIFS. Our objective is to estimate the minimum change in SIFS which can increase network capacity.

Fig. 8 shows the IPTV and VoIP capacity trends for SIFS. Our results show that SIFS ratio of 1 gives a capacity of 4 IPTV users with 2 VoIP users. SIFS ratio of 0.9 gives 4 IPTV users with 3 VoIP users. This shows that 10% decrease in SIFS value increases 1 VoIP user. SIFS trends show that capacity of IPTV decreases to 3 users with 15 VoIP users at a ratio of 1.1. This occurs because increase in SIFS increases idleness in network which decreases the time for data transfer. Analysis shows that any decrease in SIFS value reduces the overhead by converting reduced time to data transfer time. Our study shows that optimal SIFS ratio is 0.9 which appears to be $14.4\mu s$.

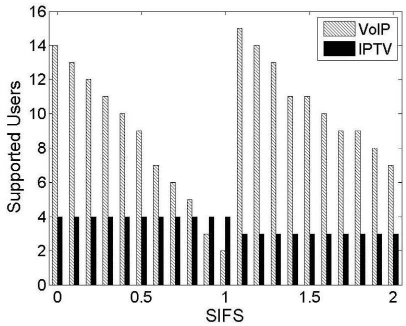


Fig. 8. IPTV and VoIP Capacity versus SIFS

E. Effect of Distributed Interframe Spacing (DIFS)

We simulate IPTV and VoIP by varying DIFS duration in the medium. DIFS duration is waited by all stations before going to back off mechanism for transmission of data. Default DIFS value is $34\mu s$ which is varied from 0 times to 2 times of original value [20].

Fig. 9 shows the IPTV and VoIP capacity trends for DIFS duration. Our results show that 4 IPTV with 2 VoIP users can be accommodated at a DIFS ratio of 1. DIFS ratio of 0.9 gives 4 IPTV users with 3 VoIP users. This shows that 10% decrease in DIFS duration increases nearly 1 VoIP user. Analysis shows that decrease in DIFS duration allows the communicating nodes to transmit immediately after transmission with less waiting time. These results are valid only if DIFS is equal to twice of SIFS duration plus the propagation time. Decrease in DIFS duration without changing SIFS results in a collision of packets between nodes which decreases capacity. Our study shows that decrease in DIFS duration increases IPTV and VoIP capacity. Optimal DIFS ratio is 0.9 DIFS which gives $30.6\mu s$ DIFS value.

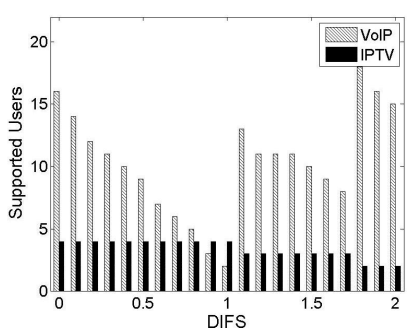


Fig. 9. IPTV and VoIP Capacity versus DIFS

F. Effect of Physical Header (PH)

We vary the time duration of PH transmission. Our aim is to estimate the effect of PH duration on the IPTV and VoIP capacity. Default PH value of $20\mu s$ is varied in ratio of original value [20].

Results of IPTV and VoIP capacity versus physical header duration are shown in Fig. 10. Results show that PH ratio of 1 can support 4 IPTV streams with 2 VoIP streams. With a PH ratio of 0.9, 4 IPTV streams can be supported with 3 VoIP streams. Trend of PH is similar to SIFS and DIFS. Major trend difference occurs at a ratio of 1.5. Capacity of IPTV streams reduces to 2 users with 16 VoIP streams. This shows that increase in PH adversely effects capacity than increase in SIFS or DIFS. Analysis shows that decrease in PH duration allows the nodes to send more data. Our results suggest that PH ratio of 0.9 is a reasonable ratio size with a value of $18\mu s$.

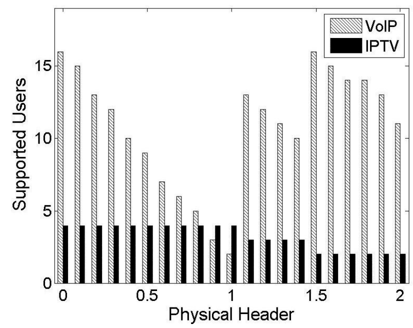


Fig. 10. IPTV and VoIP Capacity versus Physical Header

G. Effect of Contention Window (CW)

IEEE 802.11n AP uses back-off mechanism with a default minimum CW size of 15 [20]. We aim to estimate the optimal CW size because excessive CW size can result in redundant delays. CW less than optimal value results in repeated collisions between packets. We take CW of 15 as a reference and vary CW in ratio of original value. Fig. 11 shows our results of IPTV and VoIP capacity by varying CW size. Results show that optimal capacity is obtained at a ratio of 0.7. This shows that CW of 11 (0.7*15) is the optimal size. Decrease in CW size from 15 decreases the redundant waiting times of nodes till CW size of 11 is reached. CW size below 11 decreases performance because probability of selecting same window size by multiple nodes increases. This increases collisions and results in redundant delays. Results also show that no IPTV user is supported at a ratio of 0.2 or less than 0.2 CW. This suggests that lower CW size is the worst factor affecting IPTV and VoIP capacity. This also shows that

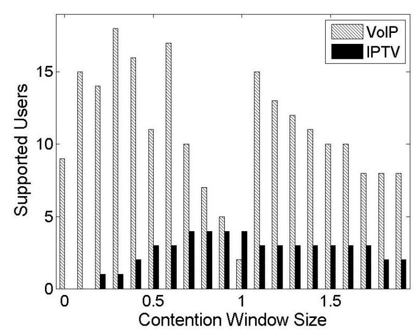


Fig. 11. IPTV and VoIP Capacity versus Contention Window

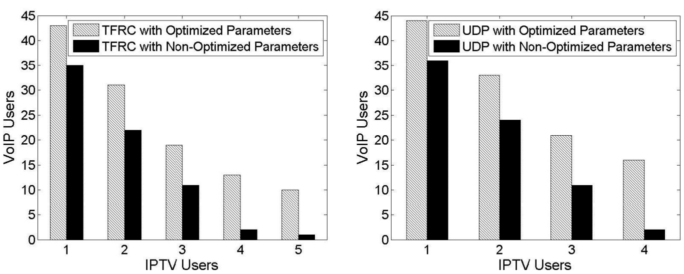
collisions in IEEE 802.11n AP effect performance of system by decrease in capacity of combined IPTV and VoIP users.

H. Discussion

Table-VIII presents the comparison of the optimal parameters obtained in this section and default parameters of IEEE 802.11n [20] for combined IPTV and VoIP. Since layers are independent, so optimal parameters do not change by the use of TFRC at transport layer of both IPTV and VoIP. Section-V shows that TFRC is the better suitable protocol for combined IPTV and VoIP. TFRC changes net capacity of IPTV and VoIP by use of congestion control mechanism. Results of both IPTV and VoIP over TFRC with optimized AP parameters from Fig. 12(a) show that maximally 5 IPTV and 10 VoIP flows can be accommodated. Our simulations show that TFRC and optimal parameters of IEEE 802.11n can increase capacity by atleast 23% for VoIP users as compared to TFRC with non-optimized parameters. Results from Fig. 12(b) show that there is atleast 22% increase in VoIP users with the use of optimized IEEE 802.11n parameters as compared to non-optimized parameters over UDP.

TABLE VIII
PARAMETERS FOR IEEE 802.11N

Parameters	Default Parameters [20]	Proposed Parameters
Queue Size	50 pkts	70 pkts
TXOP	5	9
SIFS	$16\mu s$	$14.4\mu s$
DIFS	$34\mu s$	$30.6\mu s$
Physical Header	$20\mu s$	$18\mu s$
Contention Window	15	11



(a) With TFRC (b) With UDP

Fig. 12. Capacity of combined IPTV and VoIP

VIII. CONCLUSION

In this paper, we evaluate the capacity of combined IPTV and VoIP users over IEEE 802.11n. Our evaluation show that use of TFRC at transport layer of combined IPTV and VoIP can increase capacity of IPTV users by 25%. Results show that use of TFRC for IPTV or VoIP in presence of UDP based applications can severely deteriorate the performance of IPTV and VoIP. Fairness analysis of combined IPTV and VoIP in presence of TCP flows shows that TFRC for IPTV and VoIP gives much more fair share in bandwidth to TCP traffic than UDP. Our study also evaluates optimal parameters for IEEE 802.11n including queue size and TXOP. Moreover, performance of block ACK is enhanced only in the presence of high bit error rate. We show that the selection of $14.4\mu s$, $30.6\mu s$ and $18\mu s$ for SIFS, DIFS and physical header respectively enhance the combined IPTV and VoIP capacity. Contention Window works best only at optimal point of 11. We conclude that use of TFRC with optimal parameters for IEEE 802.11n can increase capacity of IPTV and VoIP users by atleast 25% and 19% respectively in comparison to UDP with non-optimized parameters.

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