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Improving QoS of IPTV and VoIP over IEEE 802.11n ☆

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ABSTRACT

Tremendous growth rates of Internet Protocol Television (IPTV) and Voice over Internet Protocol (VoIP) have demanded the shift of paradigm from wired to wireless applications. Increased packet loss with continuously varying wireless conditions make the transmission a challenging task in wireless environment. Our study investigates and proposes improvement in the transmission of combined IPTV and VoIP over the IEEE 802.11n WLAN. Our major contributions include the analytical and experimental investigations of (1) transport layer protocol UDP/TFRC for IPTV and VoIP, (2) optimal physical layer parameters for IPTV and VoIP, (3) proposition of wireless enhancement of TFMCC (W-TFMCC) to enhance the capacity and Quality of Service (QoS) of wireless IPTV and VoIP. Analytical and experimental evaluations show a 25% increase in capacity using TFRC with 167% more bandwidth share to TCP. Our study shows that use of W-TFMCC with optimal parameters can enhance IPTV and VoIP capacity by 44%.

1. Introduction

Internet Protocol Television (IPTV) is one of the fastest growing applications which has gained huge growth rates in the past few years. Number of IPTV users are expected to increase by 500% from 2011 to 2016 [1]. Large growth rate with increased user's interest motivate us to study transmission of IPTV with an aim to provide better Quality of Service (QoS). IPTV offers a number of advantages over its predecessor analog technologies. Major advantages of IPTV include user interaction, video on demand service, economic and better Quality of Service (QoS). Architecture of IPTV includes three entities: video head end, transport network and video receiver. Video head end is placed at the server side and it has the tasks of video encoding and transmission of video and audio to the user end. Transport network is the entity which plays the most crucial rule because it incorporates jitter, delay, scrambling and packet loss effects during the transmission of video. Transport network includes both wired and wireless medium. Inside the transport network, a number of queues having the parallel storing capabilities which shuffle the packets. Video receiver is the last entity which has the task of decoding information, eliminating delay and jitter factors and managing a reliable QoS at the user end.

Voice over Internet Protocol (VoIP) is another fastest growing internet application which has obtained huge growth rates in the past few years. There are 10 times more VoIP users than IPTV users [1]. Major factors for VoIP success are cheap calling rates, better QoS and better penetration among end users. VoIP uses bi-directional traffic and has more challenging

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requirements for packet loss and delay than IPTV. Transmission of VoIP requires limited packet loss and delay for all users which becomes challenging when optimum route changes for all users.

Currently, wired access links are preferred by service providers for transmission of IPTV and VoIP services owing to minimum packet loss and delay in the wired links. Transmission of IPTV and VoIP becomes challenging in the wireless environment because major bandwidth restriction occurs at the user end having wireless Access Point (AP) [2]. Packets drop from queues of the wireless AP which make it difficult to meet the QoS constraints of IPTV and VoIP. Moreover, range and data rate are also limited in wireless links owing to the continuously varying wireless conditions. Users demand, ease of access and freedom of mobility require an insight investigation for transmission of IPTV and VoIP over wireless networks.

IEEE 802.11n Wireless Local Area Network (WLAN) is the latest standard proposing data rates upto 600 Mbps (theoretically) and 300 Mbps (practically). IEEE 802.11n is equipped with a number of features which include its Multiple Input Multiple Output (MIMO) technology and frame aggregation mechanisms at Medium Access Control (MAC) layer and Physical (PHY) layer. Frame aggregation combines multiple frames at MAC layer and PHY layer level. Major advantage of frame aggregation includes the reduction of header over-head time and also the reduction in the collision time. Our previous study for IPTV and VoIP capacity over IEEE 802.11n shows that aggregation of 4 packets is the optimal aggregation size for capacity enhancement of IPTV and VoIP [3,4].

Enhancing QoS of IPTV and VoIP with increased capacity over IEEE 802.11n has been actively discussed in various studies due to the challenging IPTV constraints over WiFi. Packet loss is a major factor which results in capacity reduction for IPTV and VoIP. IPTV and VoIP are extremely susceptible to packet loss because both use User Datagram Protocol (UDP) at the transport layer.

UDP is a constant bit rate protocol. By use of UDP, packets accumulate at the AP which results in congestion at the AP. All packets crossing the limits of queue size are dropped at the AP. Our study shows that UDP provides less delay but it increases packet loss which becomes the bottleneck for other users. Our previous study [3,4] shows that Datagram Congestion Control Protocol (DCCP) is the better suited protocol for transmission of IPTV and VoIP. DCCP has two variants namely TCP-like and TCP Friendly Rate Control (TFRC). TCP-like offers high reliability and decreases its data rate much more rapidly than TFRC. This makes it suitable for all applications demanding less packet loss. On contrary, TFRC offers a nearly constant data rate by maintaining its data rate according to varying conditions of the network. Behaviour of TFRC makes it suitable for all applications which require less delay. Our investigations [3,4] reveal through simulations that TFRC gives better performance than UDP for transmission of IPTV and VoIP over IEEE 802.11n.

In this paper, we aim to develop an analytical model for transmission of IPTV and VoIP over IEEE 802.11n.[1] Transport layer protocols UDP and TFRC are modelled by their behaviour in the wireless environment. Extensive experiments are performed to validate the analytical results. Various physical layer parameters are modelled through SIFS, DIFS and default behaviour of wireless environment. Optimal values of queue size, contention window, SIFS and DIFS are proposed. Analytical values of physical layer parameters are compared with experimental results. We propose Wireless TCP Friendly Multicast Congestion Control Protocol (W-TFMCC). TFRC and TCP-like suffer low capacity because both use the unicast mechanisms. Capacity can be enhanced significantly by shifting unicast transmissions to multicast transmissions. In this paper we present the results of TCP Friendly Multicast Congestion Control Protocol (TFMCC). TFMCC is designed for wired networks which suffer low packet loss and all users are nearly in the same conditions. TFMCC keeps a track of the user facing worst packet loss conditions and adjusts its sending rate according to the worst case user. This is highly unsuitable for the wireless medium because all users are present in different environments. TFMCC forms channel groups based only upon users demands. Transmission for the worst case user would lead to lower data rate even if a single user is having high packet loss rates or Round Trip Times (RTT). We suggest a group based protocol which keeps a track of the various conditions experienced by different users. Our study shows that performance of W-TFMCC is greater than UDP/TFRC/TFMCC if at least two or more users are watching same channels. Performance of W-TFMCC is equal to TFRC/TFMCC when all users are watching different channels.

Our contributions in this work are (i) analytical and experimental evaluation of transport layer protocols UDP/TFRC for transmission of combined IPTV and VoIP over IEEE 802.11n, (ii) analytical and experimental investigation of optimum physical layer parameters of combined IPTV and VoIP over IEEE 802.11n, (iii) proposition of a new group-based multicast protocol W-TFMCC with performance analysis over UDP/TFRC/TFMCC through simulations and experiments.

The rest of the paper is organized as follows. Section 2 presents the related work for IPTV and VoIP over WLANs. Section 3 presents the experimental scenario and data rate estimation for IPTV and VoIP. Section 4 presents the IPTV and VoIP capacity analysis over UDP and TFRC along with fairness analysis with TCP traffic. In Section 5, we show the optimal values of IEEE 802.11n analytically and experimentally. Section 6 presents the performance improvement using multicast mechanism and our proposed W-TFMCC protocol. Section 7 presents the comparison of our results with previous state of the art approaches. Section 8 concludes the paper.

[1] Initial results of this research appeared in

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