VoIP and Tracking Capacity over WiFi Networks

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Approval

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Abstract

Wireless VoIP is becoming an increasingly important application in recent years. This fact, coupled with the increasing interest in location based services, strongly suggest that tracking of wireless VoIP clients will become a widely deployed feature in emerging wireless applications. In this paper, we evaluate the performance of voice and tracking sessions analytically and via simulations for the scenario to especially support VoIP and tracking applications. We carry out experiments to further prove the validity and strengthen the proposed analytical and simulation models. We first determine an upper bound on a maximum number of wireless tracking clients that can be maintained under a single access point. We change the transmission frequency of tracking information and examine that it has a significant impact on the tracking capacity. We further extend this study to evaluate the capacity of location tracking of wireless VoIP clients, to investigate the effect of transmission frequency over the capacity of combined VoIP and tracking sessions. From various performance measurements; we develop an insight that at higher packetization intervals (e.g. 60ms) of VoIP traffic, the capacity of combined VoIP and tracking sessions decreases by 30% compared to the VoIP only capacity. The presented study further demonstrates that at lower transmission frequency (e.g. 500ms) of tracking information, the capacity of combined VoIP and tracking sessions coincides to the VoIP only capacity. Finally, we evaluate the capacity of the VoIP and tracking wireless clients using UDP (User Datagram Protocol) and DCCP (Datagram Congestion Control Protocol) in the presence of TCP (Transmission Control Protocol) traffic. Our studies propose that compared to UDP, DCCP not only improves the combined VoIP and tracking capacity but also enables TCP to get a reasonable bandwidth share.

Certificate of Originality

I hereby declare that this submission is my own work and to the best of my knowledge it contains no materials previously published or written by another person, nor material which to a substantial extent has been accepted for the award of any degree or diploma at National University of Sciences & Technology (NUST) School of Electrical Engineering & Computer Science (SEECS) or at any other educational institute, except where due acknowledgement has been made in the thesis. Any contribution made to the research by others, with whom I have worked at NUST SEECS or elsewhere, is explicitly acknowledged in the thesis.

I also declare that the intellectual content of this thesis is the product of my own work, except for the assistance from others in the project's design and conception or in style, presentation and linguistics which has been acknowledged.

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First and foremost, I am immensely thankful to Almighty Allah for letting me pursue and fulfill my dreams. Nothing could have been possible without His blessings.

I would like to thank my parents for their support throughout my educational career, specially in the last year of my Master degree. They have always supported and encouraged me to do my best in all matters of life. I dedicate this work to them. I would also like to thank my brothers and sisters for their love and prayers for the successful completion of this work.

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Imdad Ullah

Contents

List of Abbreviations

Abbreviations	Descriptions
RTP	Real Time Protocol
SIP	Session Initiation Protocol
URI	Universal Resource Identifier
VCE	Virtual Conferencing Environment
DIT-G	Distributed Internet Traffic Generator
ISDN	Integrated Services Digital Network
CBR	Constant Bit Rate
VoIP	Voice Over IP
TVA	Tracking/VoIP Architecture
ATS	Advanced Tracking Server
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
DCCP	Datagram Congestion Control Protocol
TFRC	TCP Friendly Rate Control
ECN	Explicit Congestion Notification
PMTU	Path Maximum Transmission Unit
PSAP	Public Service Answering Point
PSTN	Public Switched Telephone Netwrok
GPS	Global Positioning System
DCF	Distributed Coordination Function
DIFS	Distributed Inter Frame Space
DIFS	DCF Interframe Space
PCF	Point Coordination Function
CW	Contention Window
PLCP	Physical Layer Convergence Procedure
SIFS	Short Inter Frame Space
RSS	Received Signal Strength
MAC	Medium Access Control
CSMA/CA	Carrier Sense Multiple Access / Collision Avoidance
LoS	Line of Sight
WiFi	Wireless Fidelity
WLAN	Wireless Local Area Network
PCM	Pulse Code Modulation
QoS	Quality Of Service
ITU	International Telecommunication Unit
IETF	Internet Engineering Task Force

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Chapter 1

Introduction

1.1 Background and Motivation

In the recent past, there has been an enormous increase in the popularity of VoIP services in consequence of vast growth in wireless broadband access networks. VoIP technology allows making phone calls using public internet rather than the traditional Public Switched Telephone Network (PSTN). Since this technology offers major cost savings with additional flexibility and complex features over Plain Old Telephone System (POTS), most voice calls are now carried over at least partly via VoIP. The advancement in VoIP services has further accelerated the deployment of wireless VoIP and it begins to offer pragmatic improvements of standard VoIP along with the user mobility. The application layer protocol SIP [2] has by far been one of the most popular signaling protocols for VoIP due to the easiness of its configurability for a variety of real time applications. Besides the use of UDP as a transport protocol for multimedia applications; DCCP [3] with the congestion control mechanisms has been proposed to undertake the better transport of multimedia applications including VoIP calls.

The wireless VoIP traffic is based on IEEE 802.11 Distributed Coordination Function (DCF-based) wireless LAN, although the other optional mode of IEEE 802.11 i.e. Point Coordination Function (PCF) is specifically designed for real time traffic [18]. The PCF mode is not widely installed due to the limited Quality of Service (QoS) provisioning [19] and the implementation complexity. The wireless VoIP or the combined wireless VoIP along with tracking capacity of a DCF-based wireless LAN, defines the maximum number of connections that can be established with acceptable user-perceived quality. The quality of the VoIP calls is fine for certain number of VoIP connections, subject to the QoS constraints for VoIP calls [20][21][1], and is degraded after these constraints are violated. As the number of wireless clients cross the capacity threshold, all the wireless streams start suffering and the call quality become unacceptable for each call in wireless domain.

Location tracking in wireless networks is a leading area of research and is mainly used for continuously positioning the wireless clients. Location tracking leads to many interesting applications, mostly regards to the tracking of robots, vehicles and people e.g. [9]-[13]. A variety of location-based services provided in wireless communication networks depend on the position of a mobile node. These location-based services ranges from low-accuracy methods that are based on cell identification to high-accuracy methods that combines positioning based on information collected from wireless networks and the satellite positioning [10]. The location tracking of wireless VoIP clients is also an important feature for such services. One similar example of these services is the E911 emergency service where the location tracking of VoIP based terminals facilitates the emergency operations. For implementation of location-based services, the SIP Client/Server architectures are found to aid the requirements of call establishment and signaling. The VoIP SIP-based Client/Server architectures devised for location based services [14][15] mainly consider the availability of GPS all the time. However, due to non-line of sight effects in outdoor environments, GPS availability will come and go. SIP-based VoIP architecture [14] identifies, locates and routes emergency calls to appropriate Public Service Answering Points (PSAPs). The SIP based location tracking architecture [16] and location tracking architecture for wireless VoIP [17] works across different position technologies (e.g. GPS and wireless tracking filters). These architectures are deployed and provide best position tracking in real time environment for outdoor mobile users.

For better transport and due to growth of multimedia applications over the Internet, such as VoIP, Datagram Congestion Control Protocol (DCCP) [3] has been proposed by Internet Engineering Task Force (IETF) that is a good candidate intended for a replacement protocol for UDP in real-time multimedia applications. DCCP provides connection oriented and congestion-controlled transport layer transmission like TCP, with reliable connection establishment and teardown, but unreliable data transmission like UDP. DCCP provides two congestion control mechanisms, namely the TCP Friendly Rate Control (TFRC) [4] (Congestion Control ID 3 (CCID-3)) [5] and the TCP-Like also identified as Congestion Control ID 2 (CCID-2). TFRC is a rate-based congestion control mechanism [5] which provides a TCP-friendly rate by minimizing abrupt rate changes so that data is sent smoothly during the whole voice communication. On the other hand, the abrupt change in sending data rate (as with the TCP-Like) slows down the sending rate [4] thus the packets have to wait in sender's queue for some amount of time to be served to the destination. The congestion control mechanisms can be negotiated between the two communicating parties through the feature negotiation [3] at the connection setup or during communication with an already established connection. Another important feature that DCCP provides is the Late Data Choice [6] where DCCP can adapt to constantly changing network conditions. This feature is important for such applications that can adapt different kind of delays [7][8] which a particular data packet can experiment all the way throughout the network. Depending on the network delays, data packets become useless for the distant application.

1.2 Research Statement

Location tracking in wireless networks is a leading area of research and is mainly used for continuously positioning the wireless clients. The location tracking of wireless VoIP clients is also an important feature for such services. One similar example of these services is the E911 emergency service where the location tracking of VoIP based terminals facilitates the emergency operations. The location of a wireless client is traditionally found using GPS. However, due to non-line of sight effects in outdoor environments, GPS availability will come and go. Thus we are specifically interested in the worst scenario of unavailability of traditional GPS to quantify an upper bound on VoIP and tracking sessions analytically and via simulations for the tracking wireless clients for the scenario especially developed to support VoIP and tracking applications, one similar scenario is shown in Fig. (1.1). Similarly, performing experiments to further prove the validity and strengthen the proposed analytical and simulation models. Thus to determine an upper bound on a maximum number of wireless clients that can be maintained under a single access point. Likewise evaluating the capacity of the wireless tracking clients for wireless VoIP and then to investigate the effect of transmission frequency over the capacity of combined VoIP and tracking sessions. Finally, to evaluate the capacity of wireless tracking clients for wireless VoIP using UDP (User Datagram Protocol) and DCCP (Datagram Congestion Control Protocol) in the presence of TCP (Transmission Control Protocol) traffic.

1.3 Thesis Contributions

The main contributions drawn from this research work are

- To evaluate the capacity of wireless clients for location tracking using simulation and experimental model.

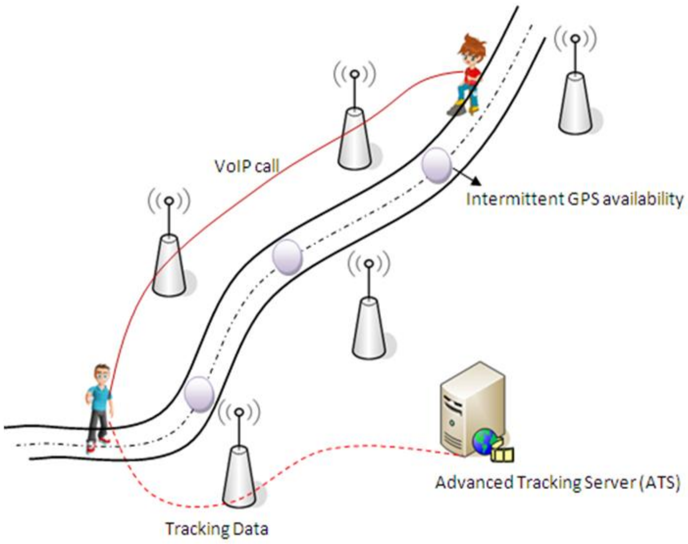

Figure 1.1: Network Problem Scenario

- To determine the effect of variation in the size of tracking data over the capacity of wireless tracking clients.
- To investigate the effect of transmission frequency of tracking data over the capacity of wireless tracking clients.
- To carry out a mathematical analysis for the capacity evaluation of the location tracking wireless clients.
- To determine the combined VoIP and tracking capacity of wireless VoIP clients using the proposed analytical model, simulations and experiments.
- To analyze the capacity of combined VoIP and tracking capacity of wireless clients in real environment using UDP and DCCP in the presence of simultaneous TCP flows.

1.4 Thesis Organization

Chapter 1 discusses an overview of the research work where we discussed the background and motivations of the research work. We discuss the problem statement and how it is being addressed in the current research work. We further discuss the contributions of the research work and how and through which process they are achieved.

Chapter 2 provides discussion on existing capacity evaluation of VoIP applications from simulations and analysis measurements, performance evaluation of IEEE 802.11 DCP-based and PCF-based model using analytical models, and performance evaluation of transport protocols like UDP, TCP and DCCP. This chapter also discusses various SIP-based location tracking and location tracking architectures designed for wireless VoIP.

In Chapter 3, we carried out a detailed analysis using simulations and experiments over tracking only, VoIP only and simultaneous VoIP and tracking sessions and a detailed comparison of their results. Further we evaluate the effect of the transmission frequency and effect of the size of the tracking information over capacity of tracking only and tracking and VoIP wireless clients.

Chapter 4 is dedicated to the implementation of the analytical model for wireless tracking clients and capacity of wireless tracking clients for wireless VoIP to determine the maximum number of simultaneous VoIP and tracking sessions. Results form the analytical model is also discussed for the queue utilization, the average waiting time, the frame's service rate and the backoff stages though which wireless nodes undergo.

Chapter 5 is dedicated to analyze the capacity of combined VoIP and tracking wireless clients in real environment using UDP and DCCP in the presence of simultaneous TCP flows. In this chapter, a portion of the UDP & TCP and DCCP & TCP flows is considered to mimic the real world network scenario.

Chapter 6 summarizes the contributions of this thesis.

1.5 Summary

In this chapter, we had an overview of the research work where we discussed the background and motivations of the research work. We discuss the problem statement and how it is being addressed in the current research work. We further discuss the contributions of the research work, which will be discussed in detail as how and through which process are they achieved. Further we discuss the thesis organization in various chapters.

Chapter 2

Literature Review

Recently, due to the excess of deployment of wireless LANs in offices and homes, wireless VoIP is achieving a considerable popularity. Wireless VoIP services can provide a caller with the ease of using portable phones with limited roaming/mobility inside the buildings. Providing the VoIP clients with accurate mobile services while having the choice of roaming requires wide area of wireless coverage. This chapter provides the review related work in enhancements to the VoIP system, the VoIP capacity domain and the SIP based location tracking and location tracking architectures for wireless VoIP. Moreover, it also discusses the emerging multimedia transport protocol like the Datagram Congestion Control Protocol (DCCP), its adaptive congestion control architecture and the mechanisms for providing the congestion control mechanisms.

2.1 Voice over Internet Protocol (VoIP)

Voice over Internet Protocol (VoIP) uses Internet Protocol for communication of voice data, just like any other data, over IP networks. This process includes digitization of voice, removal of unwanted noise signals and then the compression of voice signal using different codec schemes. After the compression, packets are defined to send over an IP network. Each packet includes a destination address and a sequence number and error checking scheme. At this stage, signaling protocols are also added to achieve requirements in addition to other call management requirements. When it reaches the destination, the sequence number helps to place the packet in correct order. The data is recovered from packet through the decompression algorithm. To make sure the correct spacing, synchronization and delay management is needed. The packets arriving out of the order are stored in the jitter buffer [43].

VoIP uses session control protocols to control establishment and teardown of calls as well as audio codec schemes, allowing transmission over an IP network as digital audio through an audio stream. It employs a combination of both TCP and UDP at the transport layer. TCP is the transport protocol which is responsible for basic call control functions and signaling including call setup, flow control, codec negotiation etc. Call control functions employ reliable communication facilities to ensure the completion and maintenance of calls correctly. Whereas actual voice data is very time-sensitive, so it relies on UDP due to low latency. As UDP is not concerned with sequencing of packets and any reliability mechanism like ACKs but it is responsible to reduce the delay associated with the VoIP transmission across a packet-switched network. As such, UDP employs that upper layer protocols will provide these features, namely RTP. RTP employs UDP in the transport layer, ensuring both sequencing information so that packets are delivered in the correct order and timing information is catered to reduce the network delay. UDP has less connection time as compared to TCP. As, TCP has overhead for acknowledgment mechanism for each and every packet sent. Moreover, throttling mechanism of TCP slows down the pace of transmission to adjust to the available bandwidth [3].

2.1.1 VoIP System

The working mechanism of a VoIP setup contains three components that are codec, digitizer, and a playout buffer. Before sending voice over IP network a voice signal is encoded using a particular codec scheme and produces a digital version of that analog signal, like in G711 codec scheme with 10ms of sampling interval it produces 80 bytes of digital data. After the digital data is being digitized is given to the packetizer where it generates CBR (Constant Bit Rate) traffic by appending 40 bytes overhead of RTP/UDP/IP headers so that this traffic can be recognized into the IP network. The IP/UDP/RTP header is a fixed overhead with each packet, however on point-to-point links RTP header compression can shrink this to 2 to 4 bytes [42]. The transmission medium, i.e. Ethernet and Physical layer, will add its own headers, checksums and calculations to the packet. These voice packets are then transmitted over the network, the reverse process is applied on the receiver side to decode the voice traffic. The quality of the VoIP calls depends on the playout buffer scheme implemented at the far end of the VoIP system which dramatically reduces the delay and packet loss [43] of the whole audio session. In a bi-directional call normally the wireless network is more responsive to network delays, packet loss and jitter, thus at the far end of the VoIP system a user endures through a poor quality of voice communication if these metrics cross their threshold. A playout buffer, for this reason, is to smoothen communication to get rid of the delay, jitter and packet loss while dropping packets where they arrive after the playout buffer timer expires. Some other components in VoIP system to progress the performance of the system are like silence detector, acoustic echo control, adaptive filters for cancellation of echo effects.

2.1.2 VoIP Setup

The working mechanism of a VoIP setup contains three components that are codec, digitizer, and a playout buffer. Before sending voice over the IP network a voice signal is encoded using a particular codec scheme and produces a digital version of that signal, like in G711 codec scheme with 10ms of sampling interval it produces 80 bytes of digital data. After the digital data is being digitized is given to the packetizer where it generates CBR (Constant Bit Rate) traffic by appending 40 bytes overhead of RTP/UDP/IP headers so that this traffic is recognized into the IP network. These voice packets are then transmitted over the network, the reverse process is applied on the receiver side to decode the voice traffic. The quality of the VoIP calls depends on the playout buffer scheme implemented at the far end of the VoIP system which dramatically reduces the delay and packet loss [43] of the whole audio session. In a bi-directional call normally the wireless network is more responsive to network delays, packet loss and jitter, thus at the far end of the VoIP system a user endures through a poor quality of voice communication if these metrics are in a greater amount. A playout buffer, for this reason, is to smoothen communication to get rid of the delay, jitter and packet loss while dropping packets where they arrive after the playout buffer timer expires. Some other components in VoIP system to progress the performance of the system are like silence detector, acoustic echo control adaptive filters for cancellation of echo effects, an adaptive algorithm LMS-type (Least Mean Square) has been devised in [44] that removes the fading effects of a frequency offset and renovates the functionality of echo cancellation, and error concealment techniques [45] for audio transmission over packet switched network for error free receiving of voice packets where it is applicable for networks with high packet loss rate.

2.1.3 VoIP Bandwidth Calculation

Different number of factors involved to find the amount of bandwidth required to carry voice traffic over IP network. Important factors are the use of codec which describes the actual amount of bandwidth that the voice data will occupy and also the rate at which the voice is digitized, the IP/UDP/RTP header and overhead due to the transmission medium. The IP/UDP/RTP header is a fixed overhead of 40 bytes each packet [1]. The transmission medium, i.e. Ethernet and Physical layer, will add its own headers, checksums and calculations to the packet.

A codec normally converts the analog signals coming from the voice conversation to a digital form using a specific codec. The analog signals are digitized on a regular interval and samples are taken 8,000 times a second and produces 80 bytes of data, this data is then sent over the transmission medium after adding essential overhead of each layer. A packetization interval of 20 ms is frequent; however different sample intervals are taken from 10 to 100ms, this selection is normally depends upon the codec like G.723.1 uses 30ms sapling period. The total packet size can be calculated as (Physical & Ethernet header) + (IP/UDP/RTP header) + (voice payload size), the number of packets per second are (codec rate / (voice payload size *8), the total rate required after adding layers overhead is (payload size along with overheads)*(8*number of packets per second)/1000. Following Table 2.1 contains some sample calculations for the bandwidth required to transmit voice data over IP network for voice codec G711. The G.711 is an international standard used for encoding telephone audio at a rate of 64kbps channel. It is a Pulse Code Modulation (PCM) scheme working at 8kHz sampling rate. It ensures the lowest lag between the conversations as there is no compression mechanism involved thus reducing processing power. The drawback is it uses more bandwidth as compared to other codec schemes. But this should not be a problem as broadband bandwidth is increasing. G.711 for VoIP will ensure the best voice quality; as there is no compression and it is the same codec used by the PSTN network and Integrated Services Digital Network (ISDN) lines.

Table 2.1: VoIP Bandwidth Calculation for Various Sampling Rates

Packetization Interval (ms)	Voice Payload Size (Bytes)	Packets per Second	Bandwidth per Conversation (Kbps)	Voice Payload with Overhead (Bytes)	Ethernet Bandwidth (Kbps)
10	80	100	96	158	126.4
20	160	50	80	238	95.2
30	240	33.33	74.66	318	84.79
40	320	25	72	398	79.6
50	400	20	70.4	478	76.48
60	480	16.67	69.35	558	74.41

2.2 IEEE 802.11 DCF and PCF Mode

The IEEE 802.11 defines two modes of MAC protocol to access wireless medium, the essential Distributed Coordination Function (DCF) mode and the optional Point Coordination Function (PCF) mode. Though the PCF mode is designed for real-time traffic [50] [49], it is not widely positioned due to its incompetent polling schemes, limited Quality of Service (QoS) provisioning [51], and implementation obscurity. On the other hand, sustaining VoIP traffic over DCF-based Wireless LANs opposes their own significant challenges, because the performance characteristics of wireless' Physical and MAC layers are much inferior to their wire line counterpart. The VoIP capacity of a DCF-based wireless networks, describes as the maximum number of voice connections that can be supported with good quality.

2.2.1 IEEE 802.11 DCF Mode

The IEEE 802.11 DCF is founded on the Carrier Sense Multiple Sense with Collision Avoidance (CSMA/CA) mechanism [48] where a station senses the medium before actually starting transmission. The wireless node suspends transmission until the medium is sensed inactive for duration of time equal to the DIFS (DCF InterFrame Space) if the medium is sensed busy. After the DIFS interval, it goes into the backoff period where it sets a random backoff counter and randomly selects a timer form the contention window (CW)that is uniformly distributed over [0,CW]. Each interval this timer is freezed if the channel is sensed busy else the backoff counter is decreased by one at each idle slot and when it reaches zero, the node start transmission. The size of the contention window becomes double after every unsuccessful transmission; similarly the sender reorganizes the transmission according to the previous backoff rule after it reaches the maximum backoff stage where the frame is dropped after retransmission limit expires. If the transmission becomes successful, the contention window is re-tuned to its low value. A SIFS (Short InterFrame Space) time wait is conducted before acknowledgment is sent by the receiver in response of a successful transmission. There are different MAC and physical layer parameters with different values for IEEE 802.11 b/g standards; e.g. the DIFS timer for 802.11 b is $50\mu s$ and for 802.11 g networks is $20\mu s$. Various parameters from MAC and Physical layers are give in Table 2.2.

Several works analyzed the performance of DCF-based networks in different mode of network operations using single-hop and multi-hop networks. In [23], Bianchi proposed a Markov chain-based analysis for the throughput of IEEE 802.11 DCF-based networks by considering all the network's nodes

Table 2.2: VoIP Bandwidth Calculation for Various Sampling Rates

Parameters	IEEE 802.11a	IEEE 802.11b	IEEE 802.11g
Data Rate	54Mbps	11Mbps	54Mbps
Basic Rate	6Mbps	2Mbps	6Mbps
Slot Time	$9\mu s$	$20\mu s$	Long = $20\mu s$, short = $9\mu s$
SIFS	$16\mu s$	$10\mu s$	$\mu s10$
DIFS	$34\mu s$	$50\mu s$	$\mu s20$
CWmin	15	31	15, 31
CWmax	1023	1023	1023
Retry Limit	7	7	7
PLCP & Preamble (Voice)	$24\mu s$	$192\mu s$	$20\mu s$
MAC Header + FCS	$5\mu s$	$24.7\mu s$	$6\mu s$
RTP/UDP/IP Header	$6\mu s$	$29.1\mu s$	$6\mu s$
Voice Payload	(Vpayload*8 / 54)	(Vpayload*8 / 11)	(Vpayload*8 / 54)
PLCP & Preamble (Ack)	$24\mu s$	$192\mu s$	$20\mu s$
ACK	$2.1\mu s$	$10.2\mu s$	$1.4\mu s$

in saturated mode. The single-hop network is assumed with an infinite number of possible retransmissions per frame. Another work has extended the work in [23] to include a finite number of retry limits as discussed in [48]. Similarly, another analytical model for the case of unsaturated sources with the assumption of very low traffic load is proposed in [47] for the channel throughput. Another work in [46] evaluated the performance of DCF-based networks using different approach by varying the traffic load. In this work, the CBR traffic with the on-off characteristics of the stations are modeled by using the state-dependent queues where the service time of different states Is estimated form the saturation throughput obtained in [23]. The work in [46], assumed that a station directly reaches the saturation throughput mode that is not a reasonable scenario if the active time of the nodes is compared with the stations in stable mode where the nodes are not transmitting data traffic. All together, the literature shows the work on DCF-based mode under the finite load condition where the MAC layer transmission queues are modeled using the M/G/1 and M/M/1 queuing system. However, this assumption does not stand well to fulfill the requirements for most of the quality of different real time applications, for instance the bidirectional VoIP traffic, due to the saturated mode condition. The Markov model mainly studies the network performance for the cases where the network's nodes are considered with saturation conditions such that all the stations present in the WLAN network always has data to transmit. Thus to support the VoIP traffic, keeping its unsaturated traffic transmission nature, all the stations must be in the unsaturated mode, that is, the stations should not always have packets to transmit the data. Therefore, the Markov model cannot be directly applied for the voice capacity analysis [21]. For this reason, the G/G/1 model is adopted to evaluate the network performance with the unsaturated stations, and it is assumed that all the stations have the same traffic load and the frame service rate. The existing WLANs are setup with the infrastructure mode, where the mobile nodes (MNs) access the internet through the access point (AP), which coordinates all the traffic to and from the WLAN. In the infrastructure based WLAN, the AP has much higher data traffic load and acts as a bottleneck. In the current work, we have evaluated the G/G/1 queuing system to evaluate the capacity of the tracking only architecture and then later extended to support maximum number of combined VoIP and tracking wireless clients where we utilized another architecture specifically designed for the tracking of wireless VoIP clients.

2.3 Deployment of SIP based Architectures

The Session Initiation Protocol (SIP) is standardized by the IETF and is continuously being implemented, mainly for Internet telephony, by a number of vendors. It has been extended to provide some enhanced functionality such as presence, event notification and instant messaging services [52][53]. The SIP protocol permits two or more parties to set up a session that consists of several media streams [54][55]. These media streams can be an audio, a video or any other Internet based communication mechanism, as an example the interactive games, internet telephone calls, audio and video streaming, and shared applications. The SIP architecture pursues the client-server model and is similar to the architecture defined for HTTP [56]. As an example of IP telephony, the caller and the callee act as a client and a server respectively. A single call session may involve several clients and several servers as requests may be routed [57] through different routes. SIP also enables different parties for multiparty interaction, such as Conference Servers (CS) [58] that is part of an audio service framework intended to offer Virtual Conferencing Environment (VCE), by providing support for various conferencing architectures [59]. The SIP based architectures are proposed for the location tracking of wireless clients and the location tracking of wireless VoIP that are mostly regards to the tracking of robots, vehicles and people e.g. [9]-[13]. A variety of location-based services provided in wireless communication networks depend on the position of a mobile node. The VoIP SIP-based Client/Server architectures devised for location based services [14][15] mainly consider the availability of GPS all the time. However, due to non-line of sight effects in outdoor environments, GPS availability will come and go. SIP-based VoIP architecture [14] identifies, locates and routes emergency calls to appropriate Public Service Answering Points (PSAPs). The SIP based location tracking architecture [16] and location tracking architecture for wireless VoIP [17] works across different position technologies (e.g. GPS and wireless tracking filters). These architectures are deployed and provide best position tracking in real time environment for outdoor mobile users and are discussed in details.

2.3.1 Tracking only SIP Based Architecture

The location tracking in wireless clients has directed towards many interesting applications and it is expected that in the coming years wireless devices are expected to be equipped with on-board Global Positioning System (GPS). However, due to the no line of sight the availability of the GPS will come and go as the wireless client traverses a region covered by an outdoor wireless network. There are some other methods, where this intermittent availability of GPS can be combined with the wireless tracking filters, to obtain richness of position information which is unavailable in modern wireless technologies. The architectural issues related to the deployment of an advanced wireless tracking system [11]. The key motivation for the issues related to the deployment of these tracking system comes from the fact that in [38][39] the authors have shown that the real time deployment of particle filters for position tracking in a WiFi network is very much possible. The architecture presented in [11] is based on Session Initiation Protocol (SIP) [2] and are shown in Fig. (2.1). The SIP based architectures have been extensively studied in the literature to support handoff in heterogeneous wireless access technologies [66]. In [61] the authors presented an architecture that uses SIP event mechanism to provide users with service portability between different communication end points. In [68] a SIP-based network mobility management architecture is presented. Similarly, in [60] the focus is on a location based push architecture that uses SIP. SIP has also been adopted in medical communication systems [69][67]. Of particular interest to the current work is the use of SIP in location-based services such as E-911 emergency location for VoIP [63]. For more SIP based architectures the reader is referred to [64][62][65]. The SIP-based architecture presented in [11] is specifically developed for the environments where it can works across different position technologies (GPS and Particle Filters). This architecture proposes employs the fusion of particle filter algorithm and GPS information to provide optimal position tracking in real time for outdoor mobile users. For further understanding of position tracking using particle filters, the reader is referred to the reports on simulated and experimental deployment of GPS-assisted particle filters in wireless networks [38][39].

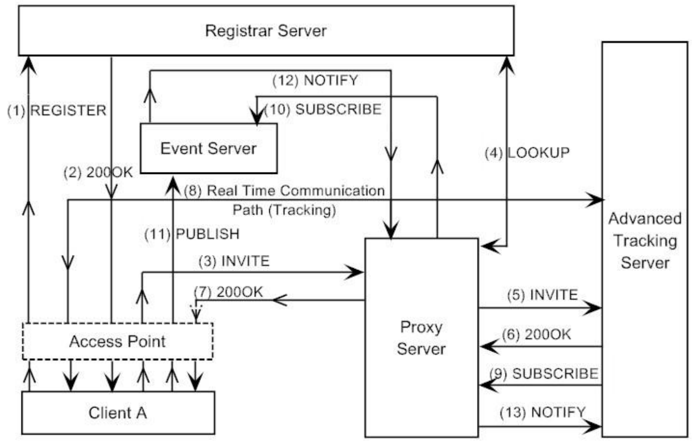

Figure 2.1: Tracking only Architecture

2.3.2 Tracking/VoIP (TVA) Architecture

We now discuss the SIP based location tracking architecture for real-time wireless VoIP, presented in [17], shown in Fig. (2.2)). The key motivation of this architecture is to provide optimal position tracking based on RSS and user profile data. Since the wireless VoIP is an emerging technology and in the coming years it is expected to gain further growth and popularity. This provided the motivation to further extend the architecture presented in Section 2.3.1 to cater for combined wireless VoIP and tracking. Because of the growing location based services, the location tracking in the absence of GPS is highly critical for wireless VoIP users. One of the prime aims of the work presented here is to design architectures which combine particle filter and intermittent GPS data to track the wireless VoIP users. This combination increases the position accuracy [17] and thus can enhance the wireless VoIP users experience by providing them with more services. Although SIP based architectures that combine location tracking and wireless VoIP [75][74] have been proposed, but intermittent nature of GPS has not been considered. The architecture proposed by the authors in [17] is specifically designed for real outdoor environments where GPS is not available all the time. The work presented in [17], the authors focus on analyzing the call setup time which is defined as the total time elapsed from a call request and the establishment of the call as per the recommendation of the ITU [71] an average call setup delay should not exceed 3s and 8s for local and international VoIP calls, respectively. Several studies have been carried out to analyze the call setup time. In [73] an analytical model is proposed to measure call setup time for a SIP based VoIP system. In [72] the call setup delay over public internet is evaluated for SIP. Call setup latency is also evaluated in [70] using 3G network emulator. However, the call setup time for architectures that combine VoIP and tracking has not been studied. The investigation of call setup time is critical for real time deployment of such architectures.

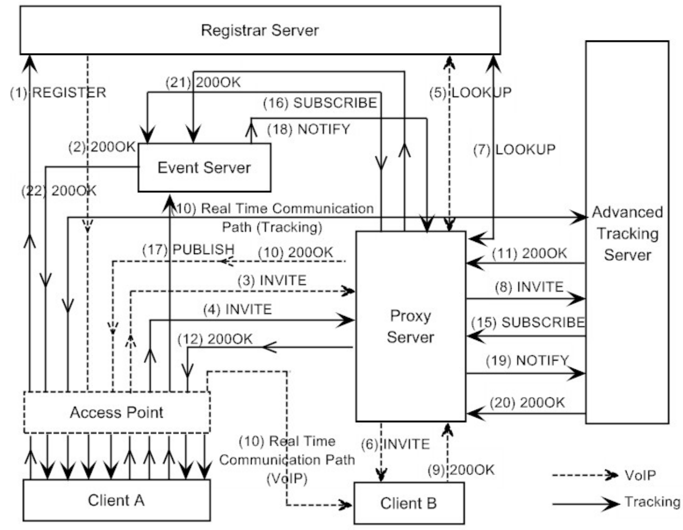

Figure 2.2: Tracking/VoIP Architecture (TVA)

The intention for the proposal of TVA architecture is to permit the network to track the wireless VoIP clients owing to the unavailability of GPS. In such a case, the location tracking is achieved using the tracking information obtained from the wireless VoIP clients (in the same manner as presented in Tracking Only architecture). However, with a difference, when GPS is available then the wireless VoIP client can send the GPS location information to the Advanced Tracking Server via a separate real time communication path (step 10 in Fig. (2.2)). Another design goal of the TVA architecture, in the absence of GPS, is to provide location tracking information to wireless VoIP clients themselves. For this purpose the position obtained at Advanced Tracking Server using wireless tracking filters is also sent back to the wireless

VoIP clients. For more details on the objectives of TVA architecture readers are referred to [17].

2.4 VoIP Capacity Evaluation

The literature shows an extensive study for the capacity analysis of wireless VoIP [20][1] and the analysis for the channel throughput over IEEE 802.11 DCF-based networks [23]. These studies reveal the measurement of capacity of wireless VoIP from simple simulations to complex capacity analysis. In [20] and [1], the 802.11b networks are evaluated through simulations for the VoIP capacity in the presence of multivariate constrains such as those of delay, maintenance of voice quality and channel conditions. In [21] the authors carried out a capacity analysis of IEEE 802.11 DCF and have taken the queue utilization ratio as a metric to calculate the upper bound of wireless VoIP clients. In [22], the authors provide analytical model for the voice capacity of a single Access Point (AP) for two different scenarios under the best effort traffic and prioritized traffic. In [24] the authors used the stochastic Petri Nets to model the behavior of DCF-based wireless LAN and derived the performance measures such as channel throughput. However, we found that the combine VoIP and tracking capacity analysis has not been carried out in the existing literature.

Accommodating the VoIP clients with mobility, while considering different kinds of mobility ranging from slow to medium and fast location changing clients, we carry out the capacity evaluation of the architectures proposed in the literature for location tracking architecture [16] and location tracking architecture for wireless VoIP [17]. These architectures optimally calculates location tracking of wireless clients across different position technologies (e.g. GPS and wireless tracking filters) in real time environment for outdoor mobile clients. These architectures are especially intended for the environment where GPS is intermittently available, that is a characteristic commonly found in outdoor wireless networks. In our present work, we develop a mathematical model and then carry out simulations and experiments to prove the validity of the proposed mathematical model and to determine the capacity of the proposed location tracking architecture. We further determine the effect of transmission frequency on tracking capacity in scenarios where users have diverse speeds of mobility. This investigation to cater for different kind of mobility directs us to demonstrate and analyze the tradeoff that has to be provided for while determining the effect on capacity in relation to the transmission frequency of tracking information.

The wireless VoIP traffic is based on IEEE 802.11 Distributed Coordination Function (DCF-based) wireless LAN, although the other optional mode of IEEE 802.11 i.e. Point Coordination Function (PCF) is specifically designed for real time traffic [18]. The PCF mode is not widely installed due to the limited Quality of Service (QoS) provisioning [19] and the implementation complexity. The wireless VoIP or the combined wireless VoIP along with tracking capacity of a DCF-based wireless LAN, defines the maximum number of connections that can be established with acceptable user-perceived quality. The quality of the VoIP calls is fine for certain number of VoIP connections, subject to the QoS constraints for VoIP calls [20][21][1], and is degraded after these constraints are violated. As the number of wireless clients cross the capacity threshold, all the wireless streams start suffering and the call quality become unacceptable for each call in wireless domain.

Besides the wireless VoIP capacity analysis, the IEEE 802.11 DCF has been extensively studied in the literature for the channel throughput and delay. These studies are mainly based on two analytical models i.e. the bi-dimensional Markov model [23] and G/G/1 model [25] where both of these models consider a homogeneous scenario in an ad hoc network. In [23], the author considered different number of wireless nodes (with saturated stations) and developed a model for channel throughput and loss rate using Markov process. However, for the VoIP traffic, wireless nodes do not have traffic all the time [21]. In all of the above work, the objective has been either to study the performance of DCF itself or has taken the assumptions made on the traffic load generated specifically for VoIP traffic only. For the capacity analysis of combined VoIP and tracking wireless clients, we utilize the analytical model presented in [21]. We adopted the G/G/1 model instead of Markov model because all the stations do not have voice traffic or the tracking traffic to transmit all the time. We note that this capacity analysis has not been addressed in the existing literature.

2.5 Datagram Congestion Control Protocol (DCCP)

Most real-time multimedia applications as well as internet telephony use either Transmission Control Protocol (TCP) or User Datagram Protocol (UDP). Both of these protocols have few drawbacks. TCP transmits data reliably, which is important for interactive data that losses importance for the receiver the longer it is delayed. TCP relies on retransmissions of already stale data which can hence introduce more delay in transmission of more current audio, thus reducing voice quality. On the other hand, UDP is a connectionless transport protocol which has been widely used for multimedia applications in the internet. Although UDP reduces delays and provides delivery of multimedia data on time. With wide use of multimedia applications, UDP fails to offer fairness over different transport protocols' data. Because UDP does not supports any congestion control mechanism at all, although congestion control mechanism can be implemented by higher layers, but this will increase burden to application programmers [3].

On the other hand, DCCP is particularly proposed for streaming applications that can take advantage from its control mechanism which reduces the tradeoffs between delay and reliability. It employs built-in congestion control at the transport layer, together with ECN support, for unreliable datagram flows, eliminating the long delays associated with TCP. So, programmers can concentrate only on the applications and not on the implementation of the congestion control mechanism at lower layers. This will be helpful in the development of applications in terms of time consumed.

2.5.1 Need of DCCP for Multimedia Applications

DCCP was first introduced by Kohler et al. in July, 2001, at the IETF transport group. It bridges the gap between TCP and UDP protocols for the requirements of multimedia application. It employs congestion controlled with unreliable data delivery at the transport layer. DCCP supports a framework which enables addition of a new congestion control mechanism that can be specified during the connection initiation procedure, or can be implemented in already established connection. DCCP also supports a feature to get connection statistics, which describes about packet loss, a congestion control mechanism with Explicit Congestion Notification (ECN) support and Path Maximum Transmission Unit (PMTU) discovery. Like TCP, DCCP supports the connection-oriented and congestion-controlled features, and like UDP, it enables an unreliable data transmission. The main reasons to give a connection-oriented protocol is to assist the achievement of congestion control mechanism.

For better transport and due to growth of multimedia applications over the Internet, such as VoIP, Datagram Congestion Control Protocol (DCCP) [3] has been proposed by Internet Engineering Task Force (IETF) that is a good candidate intended for a replacement protocol for UDP in real-time multimedia applications. DCCP provides connection oriented and congestion-controlled transport layer transmission like TCP, with reliable connection establishment and teardown, but unreliable data transmission like UDP. DCCP provides two congestion control mechanisms, namely the TCP Friendly Rate Control (TFRC) [4] (Congestion Control ID 3 (CCID-3)) [5] and the TCP-

Like also identified as Congestion Control ID 2 (CCID-2). TFRC is a rate-based congestion control mechanism [5] which provides a TCP-friendly rate by minimizing abrupt rate changes so that data is sent smoothly during the whole voice communication. On the other hand, the abrupt change in sending data rate (as with the TCP-Like) slows down the sending rate [4] thus the packets have to wait in sender's queue for some amount of time to be served to the destination. The congestion control mechanisms can be negotiated between the two communicating parties through the feature negotiation [3] at the connection setup or during communication with an already established connection. Another important feature that DCCP provides is the Late Data Choice [6] where DCCP can adapt to constantly changing network conditions. This feature is important for such applications that can adapt different kind of delays [7][8] which a particular data packet can experiment all the way throughout the network. Depending on the network delays, data packets become useless for the distant application.

One of the main features of DCCP is the modular congestion control framework [3]. It was designed to allow broadening the congestion control mechanism with addition or elimination of some congestion control mechanism based on the application requirements. All of these features can be performed before or during the connection setup through the feature negotiation mechanism. Each congestion control algorithm has an identifier called CCID. The main motivation behind designing new protocol is to bridge gap between TCP and UDP. As TCP guarantees reliable delivery with the tradeoff of unwanted delay. When a packet is lost in TCP, it decreases its transmission rate and increases it after the successful delivery of the packet. But this introduces delay in already queued packets as they cannot be transmitted until the first lost packet is recovered or retransmitted. As a result unwanted lag between the multimedia applications is observed. These characteristics of TCP make it suitable to be used in application which requires reliable data transfer without the requisition of in time delivery like data transfers, such as web browsers, instant messengers, email, file sharing, and so forth.

On the other hand, UDP is a very simple protocol which provides unreliable but in time delivery of packets, running on top of the best-effort IP protocol. It employs a connectionless service without caring about data packets delivery and network congestion control. In addition, it does not take care of packet sequencing on the receiver end. Due to the absence of any type of congestion control, it may lead to a network congestion collapse. This means UDP can deliver as much data as it can without caring reliable delivery of packets in applications like VoIP applications, video conferencing, and Internet radio. So the new option is DCCP, which includes best features of both protocols to support better quality for multimedia data streaming, as well as to share network bandwidth with TCP [80].

2.5.2 DCCP Congestion Control Identifiers

DCCP provides two congestion control mechanisms, namely the TCP Friendly Rate Control (TFRC) [4] (Congestion Control ID 3 (CCID-3)) [5] and the TCP-Like also identified as Congestion Control ID 2 (CCID-2). The main idea of this feature is to give a mechanism to control the flow of packets according to its data type. A CCID may be used at any time of a DCCP connection, and it is possible to have two different CCIDs in sender and receiver side. It provides flexibility which is important due to difference in the characteristics of the transmitted multimedia flows. Like in VoIP, there is a burst of small packets e.g. when sender says somethingbetween periods of silence, the sender stops talking and waits the peer at the receiver side to talk. In addition to these initially standardized CCIDs, DCCP IETF identifies the CCID-4, which is a new congestion control algorithm for DCCP to be used by applications that transmits small bursts of data in a short period, such as VoIP applications.

TCP-Like employs window flow control and resembles TCP congestion control mechanism. After receiving a packet, it sends an ACK back to the sender. The sender adjusts the window size and expiration time after receiving the ACK. Like TCP, the window size used in the algorithm is given as the congestion window size, which is equivalent to the upper limit of in-transit packets acceptable in the network at any time. The sender adjusts its congestion window size through congestion evaluation according to the sequence of the ACK packet received. The TFRC employs a receiver-based congestion control algorithm where the sender maintains its rate by analyzing the factors, like receive rate, loss intervals and the time taken by packets to reside in the queue before being acknowledged. This is suitable for applications that smoothly support data rate changes. As this change is not abrupt so it responds more slowly than CCID-2. The transmission rate can be varied by changing the number of packets sent. In the CCID-3 implementation, the sending rate is set by evaluating the loss event rate based on a throughput equation named TFRC Equation [81].

2.6 Summary

In this chapter, we discussed a detailed related work carried out in the previous work. We have divided the whole work in different parts: the literature work related to the VoIP system and improvement to the VoIP system and the conduction of the IEEE 802.11 DCF and PCF mode. Further we discuss the SIP-based location tracking architecture and the location tracking architecture for wireless VoIP, we discuss the call establishment process for both architectures. We carried out a detailed literature review of the work carried out over the performance of the Datagram Congestion Control Protocol (DCCP), its need for multimedia applications and the different identifiers it defined to its basic structure. Further we carry out a detailed analysis of the research work over the capacity evaluation of the VoIP only applications. In the capacity evaluation, we discussed the work carried out with analytical models and simulation models.

Chapter 3

VoIP and Tracking Capacity Evaluation

3.1 Proposed Methodology

For calculating the upper bound on capacity of tracking only and combined VoIP and tracking wireless clients to implement the real-time VoIP and tracking system, we developed a simulation model using NS-2 where significant changes were made to NS-2 modules. The contributions to the existing work includes the capacity of tracking users only, the upper bound on the combined VoIP and tracking capacity, the effect of transmission frequency of tracking data on tracking capacity, and the effect of packet size of tracking data on tracking capacity. In the current study, we analyzed that with higher packetization intervals (e.g. 50ms and 60ms) there is a significant decrease in combined VoIP and tracking capacity in association to only VoIP capacity, the variation of the size of tracking data has not been a major factor on the capacity of tracking users, and the transmission frequency of tracking data observed to be a major aspect on the network capacity, as a consequence, lowering the transmission frequency from 100 ms to 500 ms in IEEE 802.11b network; it observed a 4 times gain in the tracking capacity. Furthermore, we verified all results, which we already obtained with simulation, using a Distributed Internet Traffic Generator (DITG) [36]. The outcomes obtained with this traffic generator were found to be in close contact with the simulation results. All these consequences are discussed in the following sections.

3.2 Quality of Service (QoS) Constraints

To maintain the Quality of Service (QoS) of bi-directional VoIP traffic and to accurately send and receive the tracking data to efficiently track a wireless client, literature shows that most of the real-time traffic can accept some packet loss and one way end-to-end delay though still delivering satisfactory quality. Voice quality is affected by delay, delay jitter and unreliable packet delivery all of these are characteristics of the basic IP-network service [76]. These QoS constraints for VoIP traffic is discussed in details:

3.2.1 End-to-End Delay

Real-time applications are constrained by end-to-end delay. Maximum amount of one way end-to-end delay for VoIP is 150ms [20][21][1]; since in our case the playout deadline is fixed, although adaptive playout [43] mechanisms are promising to balance the length of buffer and to make end-to-end delay lesser along with the possibility of less packet loss, so any packet arrived after their playout becomes time limit is dropped by the receiver. Since the delay is more dominant in case of wireless rather than wired [21], so the end-to-end delay accounts for the packetization delay, the propagation delay over wired network, propagation delay and the channel access delay over wireless medium, queuing and packets processing delay at the access point [23]. The buffer size plays an important role for the packets to be delivered before playout deadlines, if the buffer size is larger a delay occurs, with the smaller size of buffer packets are loosed. Delays has also been observed due to the back off a node goes into, while accessing the channel at the same time, this collision increase the size of the contention window and a node selects DIFS timer with greater spanned contention window and gets less chances to transmit its data more frequently.

3.2.2 Packet Loss

Packet loss plays an important role to maintain or disturb the quality of voice codec [43]. To maintain the Quality of Service (QoS) of VoIP sessions, literature shows that voice conversations can accept up to 2% packet loss [20][21][1] though still delivering satisfactory voice quality. To sustain the packet loss threshold to this amount, the VoIP calls can maintain the Quality of Service of voice traffic; because the real-time bi-directional VoIP traffic is very sensitive to packet loss that significantly impact the quality of VoIP calls. To quantify the quality of voice calls the packet loss threshold is calculated by the percentage of sent packets and the packets that are dropped during communication. Packet loss are observed at the wireless side and more specifically at the intermediate node (AP), if at AP the packet arrival rate is greater rather than the service rate; packet loss encounters [21]. Due to simultaneous VoIP connections medium is the wireless frequently accessed by the clients; thus if the channel is accessed by two or more clients then packets are not decoded correctly and at the receiver are dropped.

3.3 Simulation Model

To evaluate the capacity of VoIP and tracking wireless clients under a single access point under the various QoS constraints, we carry out a wide range of simulations using the Network Simulator (NS-2.34) [33]. The NS-2 simulator is a discrete event network simulator which is extensively used by the networking research community. NS-2 is widely used in simulation for the performance evaluation of different routing and multi-cast protocols and is extensively used in ad-hoc/wireless networking research. NS-2 supports a large collection of popular network protocols, offering simulation results for wired and wireless networks alike. Due to Its open source model NS-2 is popular in academia for its extensibility and enhanced usage. To implement the proposed system, significant additions and modifications are done in various modules of standard NS-2 distribution.

Default NS-2 only has 802.11b implementation and it lacks support for 802.11g. Following the IEEE standard [34], we first of all extended it for 802.11g. This extension requires the modification of MAC and physical layer implementation to reflect the IEEE 802.11 standards that includes modification in data sending rate module, setting of physical layer transmission frequency to 2.4 GHz and changing of inter frame spacing timers to $10\mu s$ and $20\mu s$.

Secondly, to generate packets according to ITU standards [35], we implement a packetizer module and G.711 codec to ensure codec rate of 64Kbps and a packet size according to the packetization interval e.g. 160bytes for a 20ms packetization interval. The packetizer module generates packets after set intervals and specified parameters. For example, with 20ms packetization interval, using the formula below, 50 packets per seconds are generated.

$$pps = R_{codec} / V_p * 8 \tag{3.1}$$

where pps is the packets per second, R_{codec} is the codec rate and V_p is the voice payload size. The packetizer then further encapsulates these encoded packets in RTP/UDP/IP headers. To simulate two VoIP calls we use bi-directional Constant Bit Rate (CBR) sources and the traffic is generated from both sides i.e. wired and wireless. To send these packets with all the overheads that are mentioned above, the packetizer calculates the rate at which packets can be sent over the media. This rate R_t is calculated as follows.

$$R_t = V_o * (8 * pps) /1000 \tag{3.2}$$

where V_0 is the packet size with all overheads.

Thirdly, a VoIP caller module is implemented. The main function of this module is to linearly increase VoIP users to check how many users are supported by the system. Finally, to analyze the behavior of these calls in real network scenario a loss monitor module is implemented. The packet drops are monitored at the queue of the access point and the wireless nodes. Real-time bi-directional traffic is very sensitive to delay and packet loss that significantly impact the quality of VoIP calls. To quantify the quality of voice calls the packet loss threshold (see Sections IV, V, VI for details) is calculated by the percentage of sent packets and the packets that are dropped during communication.

3.4 Experimental Setup

In this section, we describe the experimental setup which we used to measure the capacity of the tracking only and combined VoIP and tracking wireless clients. To carry out the experiments, we used the Distributed Internet Traffic Generator (DITG) [36]. The DIT-G platform is capable to produce traffic using its different utilities at packet level precisely replicating correct processes for both inter departure time and packet size. These utilities are characterized as; the ITGSend is used for sending traffic and ITGRecv is the receiver component of the DITG Platform. ITGSend and ITGRecv stores information like received time of the packet, delay, port numbers, number of packets dropped and destination and sender addresses in the log files. ITGDec is used to process log files to calculate delay or packet loss.

We used Cisco Aironet 802.11b/g wireless AP, this AP is tuned to IEEE 802.11b and IEEE 802.11g WiFi standards for the experiments. The experimental setup consists of two PCs; one consists of wired link (connected through wire ink with AP) and the other node connected through wireless link with the AP. Bi-directional VoIP traffic is generated from each wired and wireless link with the G.711 codec. The voice codec is generated according to the specifications discussed in Chapter 1. These experiments are generated with UDP, DCCP and TCP according to the scenario discussed in following corresponding sections. We used the Script files to send multiple flows using same ITGSend, where each line in the script file determines the number of flows. The payload for each flow is considered as the basic VoIP decoded file for VoIP traffic to be transferred over the IP network. Further we ignored headers as they are added at each layer after it follows the OSI model at both nodes.

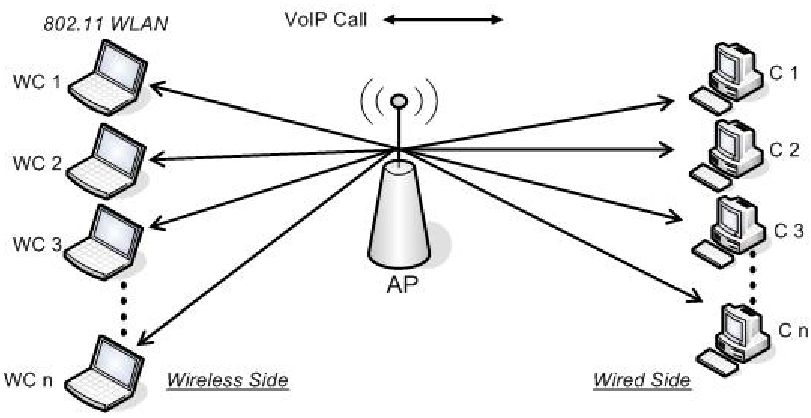

Figure 3.1: Network Scenario

3.5 VoIP Capacity of IEEE 802.11b/g Networks

In this subsection, we use the simulation model discussed in Section (3.3) to calculate the upper bound on the number of VoIP clients over WiFi networks. The proposed network scenario with equal number of wireless clients (WCs) and wired clients and an AP, as shown in Fig. (3.1). To maintain the Quality of Service (QoS) of VoIP sessions, literature shows that voice conversations can accept up to 2% packet loss and 150ms one way delay though still delivering satisfactory voice quality [20][21][1]. While evaluating the capacities of location tracking and location tracking for wireless VoIP clients, we adopt these QoS constraints for the VoIP traffic. The upper bound on maximum number of VoIP connections that can be maintained with IEEE 802.11b/g standards are summarized in Table 3.1. This table shows that the capacity of wireless VoIP clients increases as the packetization interval is increased, such an observation also presented in [20][21][1]. Further to verify the simulation results we also carried the experiments. The results we obtain through simulations and experiments are in complete accord with those presented in [20][21][1][37].

Table 3.1: VoIP Capacity over WiFi Networks using Simulations & Experiments

Packetization Interval (ms)	IEEE 802.11b		IEEE 802.11g	
	Simulation Results	Experimental Results	Simulation Results	Experimental Results
10	7	6	21	19
20	12	10	44	40
30	19	16	66	62
40	24	21	85	81
50	29	27	106	101
60	34	32	125	121

3.6 Tracking Capacity of Wireless Clients

For investigation of the capacity of wireless tracking clients using simulations and experiments and further with the analysis (as discussed in Chapter 4); we take a worst scenario where fix GPS is not available most of the time. We carry out the capacity analysis of the tracking wireless clients using the tracking architecture shown in Fig. (2.1). In this scenario, the Advanced Tracking Server receives tracking information from all wireless clients, and at the same time it also sends location tracking information to wireless clients through the same real-time path in the form of a user profile. We take the tracking information from wireless clients to Advanced Tracking Server in uplink direction as 300bytes while the size of down-link tracking information from Advanced Tracking Server to wireless clients is taken as 200bytes. We take the uplink tracking data that contains information about the wireless client's SIP Uniform Resource Identifier (URI), past 15 GPS positions, Received Signal Strength (RSS) values of each AP along with their MAC addresses that are in the range of WCs. Since a WC is connected to a particular AP but we consider a case where each WCs obtain RSS values and MAC from up to 10 nearby APs. In the same manner, for the creation of user profile, the down-link information consists of SIP URI of WCs and the location calculated by wireless tracking filter in Advanced Tracking Server. To make the proposed the capacity evaluation more flexible and for future research enhancements; we keep an extra 100bytes in both uplink and down-link tracking information. The mathematical analysis, simulations and experiments are performed at a transmission frequency of 100ms that is taken as the default beacon interval in WiFi networks. This corresponds to instantaneous transmission of all RSS values obtained from each APs heard. We take the packet loss threshold as 2% and 150ms of one way end-to-end delay for tracking data, since the real time tracking can suffer some packet loss and delay like other similar (real time) applications [1][21]. These constraints can further be relaxed to increase the wireless tracking clients' capacity.

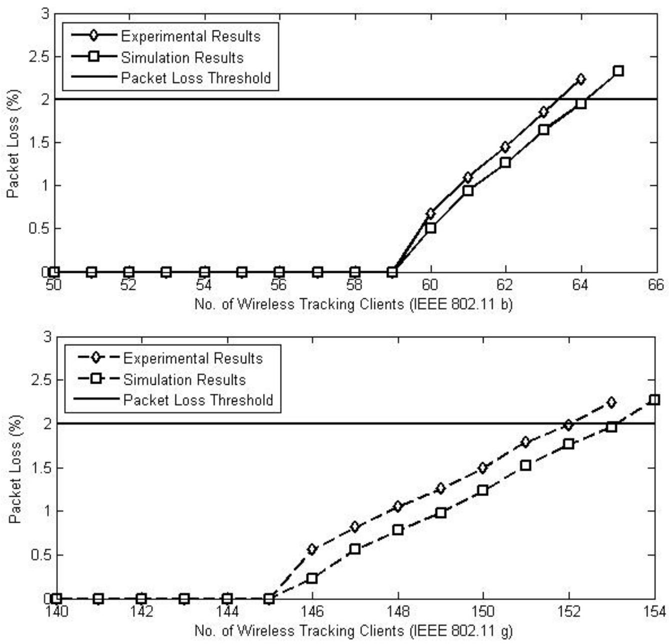

Figure 3.2: Effect of Packet Loss on Tracking Capacity in IEEE 802.11b/g

Fig. (3.2) and Fig. (3.3) show the packet loss and one way end-to-end delay acquired during simulations and experimentations for both IEEE 802.11b and IEEE 802.11g networks. During simulations and experiments it is observed for both WiFi standards that the packet loss as well as delay (specifically in downlink direction) intensifies after a certain minimum number of WCs take part in communication (for example 59 and 145 wireless clients for both IEEE 802.11b and IEEE 802.11g standards, respectively). The rationale behind this fact is that the queue of AP overflows (as here the AP acts as a major source of bottleneck). The analytical results obtained from the terracing capacity is limited to 65 and 154 IEEE 802.11b and IEEE 802.11g respectively before the queue of the AP is fully utilized. In the same manner, the simulation results shows the tracking capacity to be restricted to 64 and 152 for IEEE 802.11b and IEEE 802.11g respectively before the QoS constraints are violated. Besides simulation, the capacity of the wireless tracking clients is also calculated through the analytical model and is sub-

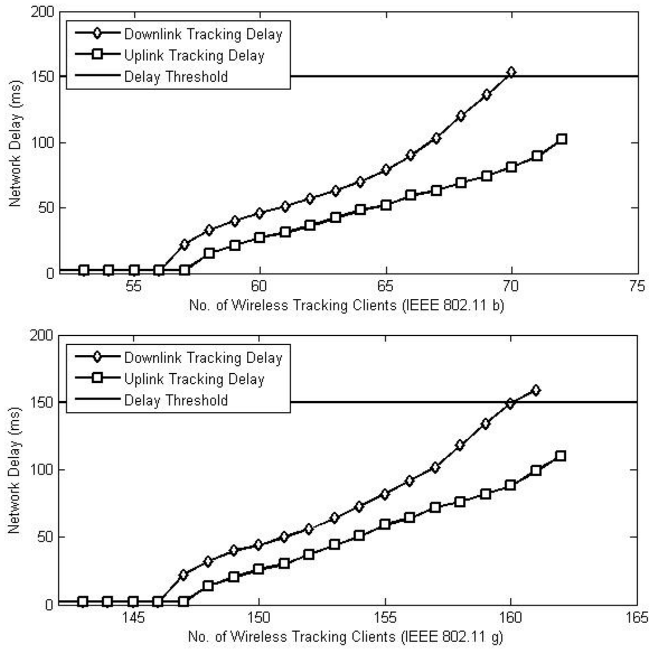

Figure 3.3: Effect of End-to-End Delay on Tracking Capacity in IEEE 802.11b/g

ject to the instability of the queue of AP. Along with the simulation results some similar results are also obtained when experiments; carried out using the DITG. The simulation and experimental results along with the analytical results are also shown in Table 3.2.

Table 3.2: Tracking Capacity over WiFi Networks

Results	IEEE 802.11b	IEEE 802.11g
Simulations	64	152
Experiments	63	151

3.6.1 Effect of Packet Size on Tracking Capacity

In above discussion, we observed the capacity of location tracking wireless clients by taking a logical size of tracking data packets to gain the basic insight into the behavior of QoS for wireless tracking clients in relation to capacity. We now evaluate to observe the variation of size of tracking data packet and examine its effect in estimating the capacity of WiFi networks. For this reason, we first increase the tracking data packet size in uplink and downlink direction from 300bytes to 400bytes (the scenarios where the client is located in area of densely clustered APs) and then we decrease from 300bytes to 200bytes (for areas where there exist less number of APs). The results from these experiments are tabulated in Table 3.3. It is observed that the variation in packet size is not seem to have a considerable impact on the number of WCs that can be obtained for location tracking in IEEE 802.11b/g networks.

From above analysis, it is possible that the transmission frequency of tracking information in both directions may have an important factor in determining network capacity. We further investigate this fact and observe its impact over the capacity of wireless tracking clients.

Table 3.3: Tracking Capacity with variation in Uplink and Downlink Packet Size

Variation in Uplink Packet Size			
WiFi Standards	200bytes	300bytes	400bytes
IEEE 802.11b	67	64	62
IEEE 802.11g	153	152	151
Variation in Downlink Packet Size			
WiFi Standards	200bytes	300bytes	400bytes
IEEE 802.11b	66	65	63
IEEE 802.11g	154	153	152

3.6.2 Effect of Transmission Frequency on Tracking Capacity

This section illustrates the effect of the transmission frequency and observes how it can have an impact over the evaluation of tracking capacity. The transmission frequency at which the tracking information is being sent by the WCs is enormously associated with their mobility. They can be classified according to their location changing while moving either frequently or slowly, whereby the RSS values could be used to determine the intervals at which the transmission frequency of tracking information is set. Accordingly, the instant RSS value (taken at 100ms that is also the default beacon interval of WiFi standards) is involved in modeling the transmission frequency for fast WCs. In the same manner, an average of several RSS's[1] (taken after 500ms) differentiate well for WCs with minor mobility requirements. In this study, we incorporate the change in transmission frequency from 100ms to 500ms. Our current study observes an increase in capacity (for example, 272 wireless clients for IEEE 802.11b) for WCs of slower speeds. This capacity is approximately four times more than the capacity observed for the case of 100ms (64 tracking clients). This analysis advocates a trade-off to be carried out to achieve the upper bounds on the desired capacity. Thus a slower transmission rate allows for a significantly higher tracking capacity at the cost of limited mobility whereas a faster transmission frequency supports greater user mobility at the cost of lesser tracking capacity.

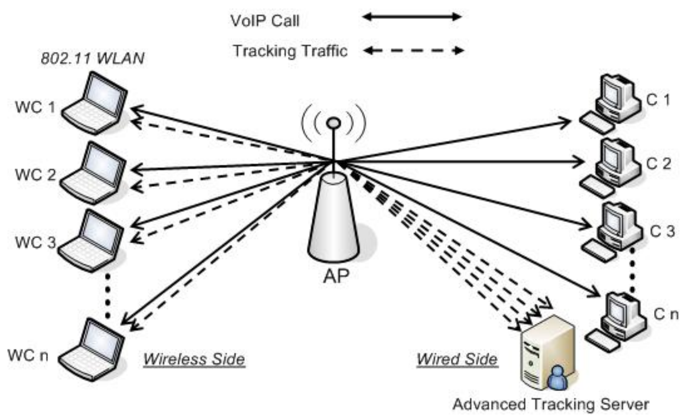

Figure 3.4: Network Scenario (combined VoIP and Tracking)

3.7 Combined VoIP and Tracking Capacity Evaluation

In this section we discuss the combined VoIP and tracking capacity using simulations and experiments. A high level view of the proposed network scenario is shown in Fig. (3.4). The size of the tracking data is the same as 300

[1]A feature advantage of using the average values of RSS is the intrinsic contest against small scale fading. Another benefit of using this strategy is the improved tracking accuracy [38][39].

Table 3.4: Combined VoIP and Tracking Capacity at the Transmission Frequency of 100 ms

	IEEE 802.11b			IEEE 802.11g		
	Simulation Results		Experimental Results	Simulation Results		Experimental Results
Packetization Interval (ms)	VoIP Capacity	VoIP & Tracking Capacity	VoIP & Tracking Capacity	VoIP Capacity	VoIP & Tracking Capacity	VoIP & Tracking Capacity
10	7	6	5	21	19	17
20	12	11	9	44	35	32
30	19	14	12	66	50	47
40	24	18	15	85	61	58
50	29	21	19	106	71	68
60	34	24	21	125	80	76

bytes for uplink and 200 bytes for down-link (as described above) and we use the instantaneous transmission frequency of 100 ms. Table 3.4 illustrates the simulation and experimental results. In order to facilitate a comparative analysis, VoIP only capacity is also shown in Table 3.4. It is recorded that the packet loss increases as we increase the number of wireless clients as shown in Fig. (3.5) for IEEE 802.11b and IEEE 802.11g. It is observed that the capacity for cases of VoIP only and combined VoIP and tracking does not tend to differ significantly for lower packetization interval (10 ms, 20 ms). However, the number of users that can be supported gets profoundly limited as the packetization interval increases for both WiFi standards *e.g.* In 802.11b, at 60 ms the VoIP only capacity is 34 and that of combined VoIP and tracking is 24. The rationale for the above observation is that at higher packetization interval greater number of connections are established resulting in higher generation of traffic volume. As a consequence, the contention in the wireless medium becomes severe. Not only do the wireless clients contend for medium access among themselves but also a self contention is triggered at each wireless client. To make the situation worse, as there is an extensive traffic load on the AP it is forced for continual medium access attempts amidst grave contention that is posed by all the wireless clients. This causes the queue of the AP to overflow which results in heavy packet loss on the wireless side. It is also noted that loss of VoIP packets is more in comparison to tracking packets. This is because more VoIP packets are sent (at lower packetization interval) as compared to the tracking data (instantaneous transmission frequency of 100 ms). This fact is demonstrated for IEEE 802.11b in Fig. (3.6). In Fig. (3.6) both lines show irregular behavior,

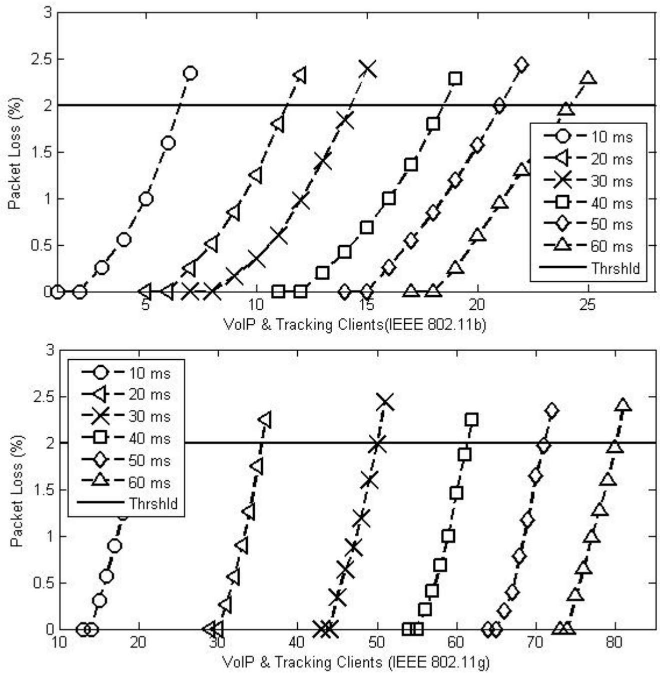

Figure 3.5: Combined VoIP and Tracking Capacity over IEEE 802.11b/g

because these statistics are obtained under the packet loss threshold for each packetization intervals.

3.7.1 Effect of Transmission Frequency over Capacity of Combined VoIP and Tracking Wireless Clients

We now evaluate the effect of transmission frequency on the combined VoIP and tracking capacity since it is extremely associated with the mobility of the wireless client. The traffic parameters are the same as discussed in previous section, however, the transmission frequency is changed to 500ms. As with this lower frequency; lesser number of tracking packets are sent so it lessens the contention on wireless medium and accordingly increases the capacity. We noticed that the combined VoIP and tracking capacities are comparable to that of capacities obtained at 100ms (Table 3.4 for IEEE 802.11b and IEEE 802.11g), an increase as high as 25% is observed in newly obtained capacity. Furthermore, it is also noted, that at 500ms the capacities obtained for VoIP only and combined VoIP and tracking traffic shows insignificant differences for IEEE 802.11b. Conversely, this difference in capacities is somewhat more and is related to more number of VoIP and tracking connections established

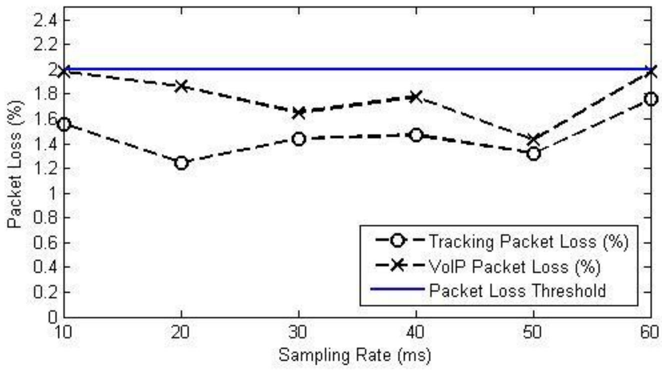

Figure 3.6: Comparative Packet Loss of Tracking and VoIP Traffic in IEEE 802.11b

which in turn also increases the congestion at the AP.

This observation strongly recommends that using this strategy, the capacity of the desired system can be maximized even if the tracking applications are run over VoIP. The intrinsic trade off to be made, however, to support more or less number of users and depends whether the speed of user is slow or fast, respectively. To further increase the capacity of combined VoIP and tracking wireless clients, the transmission frequency can be further increased by another 500ms. However, this will result in an increased total delay of 1sec and is not suitable for real time tracking.

3.8 Summary

In this chapter, we discussed the proposed methodology and the obtained results using simulations and experiments. We discussed the Quality of Service (QoS) constraints for the multimedia applications, where we discuss the constraints of end-to-end delay, the packet loss along with the work carried out in the literature. We discussed the simulation and experimental models which we used for the capacity evaluation of the tracking only and VoIP and tracking wireless VoIP clients. Using the simulation and experimental model; we carry out the VoIP only capacity analysis under the single AP and compared the obtained results with some of the literature work. We further discuss the effect of the transmission frequency of tracking information over the capacity of the tracking only and combined VoIP and tracking wireless clients. We also discuss the effect of the size of the tracking packets over the capacity of tracking only and VoIP and tracking wireless clients.

Chapter 4

VoIP and Tracking Capacity: Analytical Modeling

4.1 Introduction

In this chapter, we develop an analytical model for the capacity evaluation of tracking clients and combine VoIP and tracking clients, considering the IEEE 802.11 DCF based networks. Several works have analyzed the performance of DCF based networks by taking into account different applications and for single-hop and multi-hop networks. As an example, Bianchi [23] proposed a Markov chain-based analysis for the throughput of IEEE 802.11 DCF-based networks by considering all the network's nodes in saturated mode. Another work has extended the work in [23] to include a finite number of retry limits as discussed in [48]. The analytical model presented in these works considers DCF-based mode modeled using the M/G/1 and M/M/1 queuing system. However, this assumption does not stand well to fulfill the requirements for most of the quality of different real time applications, for instance the bidirectional VoIP traffic, due to the saturated mode condition. The Markov model mainly studies the network performance for the cases where the network's nodes are considered with saturation conditions such that all the stations present in the WLAN network always has data to transmit. Therefore, the Markov model cannot be directly applied for the voice capacity analysis [21]. For this reason, the G/G/1 model is adopted to evaluate the network performance with the unsaturated stations, and it is assumed that all the stations have the same traffic load and the frame service rate. We present an analytical model for the capacity evaluation of the location tracking wireless clients and VoIP only wireless clients, we then further extend this model to investigate the capacity of combined VoIP and tracking wireless clients.

4.2 Queuing System

Queuing theory is the mathematical study of queues or the waiting lines that facilitate mathematical analysis of several processes that are related to one another. This process includes arriving of a process at the queue, waiting states in the queue, and being served at the front of the queue. The mathematical calculation is carried out by several performance measures that include the expected number waiting or receiving service, the average waiting time in the queue or the system, and the probability of coming across the system in certain states. For instance the states may be empty, full, having an available server or having to wait a certain time to be served. A queue model is characterized by:

The Arrival Process: The process are normally assumed as their inter-arrival times are independent and they have a common distribution. In practical circumstances the processes arrive according to a Poisson distribution (i.e. exponential inter-arrival times). Process may arrive individually one by one, or in batches.

The Waiting Room: The waiting states of the system depends upon the number of the process in the system. For example, in a data communication network, a finite number of packets can be buffered in a switch or a router. The design of the network depends of the issues related to the determination of good buffer sizes during the queue stability. The process can be executed at the queue of the system to be served to the destination to take a particular packet to the destination for the next hop or the system waits to get a chance to get the service time.

The Service Times: Since the inter-arrival of the processes in the system are independent of one another so these process are usually assumed that the service times are independent and equally distributed. For example, the service times can be determined mathematically or is exponentially distributed. It can also be determined that the service times are dependent of length of the queue. For example, the processing rates in a production system can be increased once the number of jobs that are waiting in a queue becomes too large.

4.3 Capacity Analysis for Real-Time VoIP Traffic

For the capacity analysis of the proposed location tracking wireless clients; we utilized the analytical model presented in [21] and further extended for capacity evaluation of the combined VoIP and tracking wireless clients. The proposed network scenario, consider a DCF-based WLAN, consists of N nodes with $N-1$ wireless clients (WCs) and an AP, as shown in Fig. (3.1). All the WCs access the internet though AP where AP has much higher traffic loads than the WCs. The AP takes the coordination of all the traffic to and from the internet and WCs and acts as a bottleneck. Keeping in view the VoIP traffic nature, the nodes are not in saturated mode and do not have packets all the time to transmit.

The collision probability, denoted as P_i, refers to the probability where two nodes transmit at the same in a randomly chosen slot. For instance, the collision probability of the WC denotes the probability when the tagged WN transmit in slot T and either of the AP or the remaining $N-2$ wireless clients also transmit in the same slot. The collision probabilities of AP and WC characterized as

$$p_1 = 1 - (1 - p_2[T])^{(N-1)}$$
$$p_2 = 1 - (1 - p_2[T])^{(N-2)}(1 - p_1[T])$$

The collision probability contributes T_c (successful transmission time) and the success probability (i.e. $1 - P_i$) contributes T_s (time spent due to collision). T_c and T_s consists of the time for data transmission with their corresponding $ACKs$ or $ACK_timeout$ and various timers like SIFS, DIFS etc. The average collision time of a frame transmitted by a WC and AP is calculated for its geometric distribution. During the transmission, each frame suffers the collision time and remains in the backoff stage until they are successfully transmitted. The backoff procedure consists of the backoff window timer taken from the current contention window i.e. CW during the transmission of a frame until it is successfully transmitted. The average backoff timer for the AP point and WCs is modeled as geometrically distributed variable. Before the successful transmission of a data frame, it remains in the queue of a node and faces backoff stages and contributes in the saucerful and colistin timers until successfully served.

We evaluate the analytical model to calculate the upper bound on the number of VoIP clients over WiFi networks. We analyze the queue utilization ratio of the AP and WCs that acts as a metric for the capacity evaluation of wireless VoIP clients, as shown in Table 4.1.

The simulation and experimental resutls are obtained under the Quality of Service (QoS) constraints for VoIP sessions, i.e. 2% packet loss and 150ms one way delay though still delivering satisfactory voice quality [20][21][1], as discussed in 3.5. Results from analytical model, simulations and experiments are presented in Table 4.1. This table shows that the capacity of wireless VoIP clients increases as the packetization interval is increased, such an observation also presented in [20][21][1]. Further to verify the simulation results we also carried the experiments. The results we obtain through analysis, simulations and experiments are in complete accord with those presented in [20][21][1][37].

Table 4.1: VoIP Capacity over WiFi Networks using Analysis, Simulations & Experiments

	IEEE 802.11b			IEEE 802.11g		
Packetization Interval (ms)	Analytical Results	Simulation Results	Experimental Results	Analytical Results	Simulation Results	Experimental Results
10	7	7	6	22	21	19
20	13	12	10	45	44	40
30	20	19	16	68	66	62
40	25	24	21	87	85	81
50	30	29	27	108	106	101
60	36	34	32	127	125	121

4.4 Capacity Analysis of Tracking Only Wireless Clients

We now evaluate the capacity of the tracking only wireless clients discussed, here we analyze the queue utilization ratio of the AP and WCs that acts as a metric for the capacity evaluation of wireless tracking clients, as shown in Table 4.2. The analytical results obtained from the tracking capacity is limited to 65 and 154 IEEE 802.11b and IEEE 802.11g respectively before the queue of the AP is fully utilized. The queue utilization ratio of the AP and WCs continuously increases while remains stable until a specific number of wireless clients are tracked. Fig. (4.1) shows the capacity of wireless tracking client form the Tracking Only architecture. In the same manner, the simulation results shows the tracking capacity to be restricted to 64 and 152 for IEEE 802.11b and IEEE 802.11g respectively before the QoS constraints are violated. Along with the analysis and simulations some similar results are also obtained when experiments are carried out using the DITG. The simulation and experimental results along with the analytical results are also shown in Table 4.2.

Table 4.2: Tracking Capacity over WiFi Networks

Results	IEEE 802.11b	IEEE 802.11g
Analysis	65	154
Simulations	64	152
Experiments	63	151

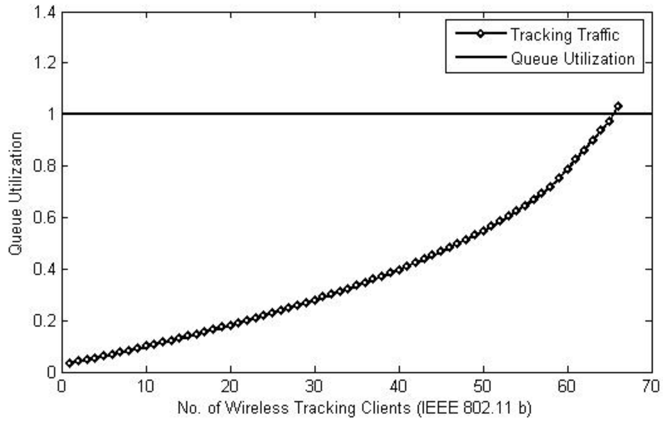

Figure 4.1: Effect of Queue Utilization Ratio on Tracking Capacity in IEEE 802.11b/g

4.5 Analytical Model for combined VoIP and Tracking Capacity

In this section, we present an analytical model for the capacity evaluation of combined VoIP and tracking traffic under a single Access Point (AP) of a Distributed Coordinated Function (DCF-based) WLAN in an infrastructure-based or ad hoc network. In the proposed network scenario (as shown in Fig. (3.4)), since AP is the transit node and the entire internet traffic goes through it, so it acts as a bottleneck. The WCs are not in saturated mode and do not have the packets all the time. For the capacity evaluation, we have N nodes with $(N-1)$ mobile nodes (WCs) and wired nodes, one AP and SIP Servers. All the wired nodes and AP are connected through a wired backbone. The WCs and wired nodes establish VoIP connections through the AP while the tracking server (a SIP server) sends and receives tracking information to and from all the WCs. The traffic through the AP is categorized as the

VoIP traffic and the tracking traffic, where each of them has their own traffic arrival rate. The VoIP and tracking traffic arrival rates are denoted as δ_d and δ_t respectively. Since in infrastructure-based WLAN, all the traffic is transmitted through the AP so the traffic arrival rates over the AP are $(N-1)\delta_d$ for VoIP traffic and $(N-1)\delta_t$ for tracking traffic. The frame service rate of the AP and the WC is taken as η_1 and η_2. We are not differentiating the service rate of AP and WCs for VoIP or tracking traffic separately, because both types of traffic are serviced according to the resources available for serving them. From the combined VoIP and tracking traffic arrival and service rate, we analyze the queue utilization ratio of the AP and WCs. This queue utilization acts as a metric for the capacity of combined VoIP and tracking traffic under a single AP. The queue utilization for both AP and WC is denoted as ε_1 and ε_2 respectively. The queue utilization ratios for both VoIP and tracking traffic are represented taken with the subscript d and t respectively. These are denoted as $\varepsilon_{1d} = \frac{(N-1)\delta_d}{\eta_1}$, $\varepsilon_{1t} = \frac{(N-1)\delta_t}{\eta_1}$, $\varepsilon_{2d} = \frac{\delta_d}{\eta_2}$, and $\varepsilon_{2t} = \frac{\delta_t}{\eta_2}$. Various network symbols, representing the model, are shown in Table 4.3.

During wireless when the AP transmits in a given slot, either the VoIP packet or the tracking packet, and at the same time slot at least one of the WC also transmit, then they result in collision. Let we denote $p_{id}[T]$ and $p_{it}[T]$, the probability that a WC or the AP transmit either the VoIP packet or the tracking traffic in a randomly chosen slot. The collision probability of the AP due to VoIP traffic (p_{1d}) or the tracking traffic (p_{1t}) is determined as

$$p_{1d} = 1 - (1 - p_{2d}[T])^{(N-1)}(1 - p_{2t}[T])^{(N-1)} \tag{4.1}$$

$$p_{1t} = 1 - (1 - p_{2t}[T])^{(N-1)}(1 - p_{2d}[T])^{(N-1)} \tag{4.2}$$

The collision probability of the WC at a particular time slot is calculated as, when either the AP or any of the remaining WCs *(N-2)* transmit either the VoIP or tracking packet at the same time slot.

Table 4.3: Network Model Symbols

Symbols	Description
ε	Queue utilization ratio
η	Traffic service rate
δ	Traffic arrival rate
$\mathbb{Q}$	Queue empty
$\mathbb{Q}'$	Queue not empty
p	Conditional collision probability

$$p_{2d} = 1 - (1 - p_{2d}[T])^{(N-2)}(1 - p_{2t}[T])^{(N-2)} \\ (1 - p_{1d}[T])(1 - p_{1t}[T]) \quad (4.3)$$

$$p_{2t} = 1 - (1 - p_{2t}[T])^{(N-2)}(1 - p_{2d}[T])^{(N-2)} \\ (1 - p_{1t}[T])(1 - p_{1d}[T]) \quad (4.4)$$

In the proposed scenario the WCs are not in saturated mode and do not have packets all the time to be served, same with the AP as well, so the transmission probability of the VoIP or tracking traffic over the AP or WCs depends on the condition of the queues being empty or non-empty.

$$p_{id}[T] = p_{id}[T/\mathbb{Q}]p_{id}\,[\mathbb{Q}] + p_{id}[T/\mathbb{Q}']p_{id}[\mathbb{Q}'] \quad (4.5)$$

$$p_{it}[T] = p_{it}[T/\mathbb{Q}]p_{it}\,[\mathbb{Q}] + p_{it}[T/\mathbb{Q}']p_{it}[\mathbb{Q}'] \quad (4.6)$$

Here $p_{id}\,[\mathbb{Q}]$ and $p_{it}\,[\mathbb{Q}]$ represents the probabilities of the AP (i=1) or the WC (i=2) when the queue is empty due to VoIP and tracking traffic respectively. Similarly, $p_{id}[T/\mathbb{Q}]$ and $p_{it}[T/\mathbb{Q}]$ represents the transmission probabilities of the AP or the WC when queue is empty due to VoIP and tracking traffic. In the same way, the $p_{id}\,[\mathbb{Q}']$ and $p_{it}\,[\mathbb{Q}']$ represents the probabilities when the queue is not empty due to VoIP and tracking traffic along with their transmission probabilities represented as $p_{id}[T/\mathbb{Q}']$ and $p_{it}[T/\mathbb{Q}']$ for both AP and WCs.

If queue is not empty then the queue is being utilized either due to tracking traffic $p_{it}\,[\mathbb{Q}'] = \varepsilon_{it}$ or due to VoIP traffic $p_{id}\,[\mathbb{Q}'] = \varepsilon_{id}$, while queue is not utilized with the probability of $1 - \varepsilon_{id}$ and $1 - \varepsilon_{it}$ due to VoIP and tracking traffic. On the other hand, when the queue of either AP or the WCs is empty then their transmission probabilities are zero i.e. $p_{id}[T/\mathbb{Q}] = p_{it}[T/\mathbb{Q}] = 0$. The above Eq. (5) and Eq. (6) then becomes $p_{id}[T] = p_{id}[T/\mathbb{Q}']p_{id}[\mathbb{Q}']$, $p_{it}[T] = p_{it}[T/\mathbb{Q}']p_{it}[\mathbb{Q}']$. The transmission probability of the AP or the WCs when queue is not empty due to either VoIP traffic $p_{id}[T/\mathbb{Q}']$ or tracking traffic $p_{it}[T/\mathbb{Q}']$ is analyzed in [40], given below.

$$p_{id}[T/\mathbb{Q}'] = \frac{2\,(1 - 2p_{id})\,(1 - p_{id}^{m+1})}{W\left(1 - (2p_{id})^{m'+1}\right)(1 - p_{id}) + (1 - 2p_{id})\,\epsilon_{id}} \quad (4.7)$$

$$p_{it}[T/\mathbb{Q}'] = \frac{2\,(1 - 2p_{it})\,(1 - p_{it}^{m+1})}{W\left(1 - (2p_{it})^{m'+1}\right)(1 - p_{it}) + (1 - 2p_{it})\,\epsilon_{it}} \quad (4.8)$$

Here, $\epsilon_{id} = w2^{m'} p_{id}^{m'+1}\left(1 - p_{id}^{m-m'}\right) + 1 - p_{id}^{m+1}$ and $\epsilon_{it} = w2^{m'} p_{it}^{m'+1}\left(1 - p_{it}^{m-m'}\right) + 1 - p_{it}^{m+1}$, m is the retransmission limit, m' is the maximum backoff stages and W is the minimum backoff window size, according to IEEE 802.11 standard their values are 7, 5 and 32 respectively. Let $\rho_{id} = p_{id}[T/\mathbb{Q}']$ and $\rho_{it} = p_{it}[T/\mathbb{Q}']$, put them in Eq. (5) and Eq. (6), we have

$$p_{id}[T] = \varepsilon_{id}\rho_{id} \tag{4.9}$$

$$p_{it}[T] = \varepsilon_{it}\rho_{it} \tag{4.10}$$

Expending these equations for VoIP and tracking traffic over both AP and WCs we have

$$p_{1d}[T] = \varepsilon_{1d}\rho_{1d},\ and\ p_{1t}[T] = \varepsilon_{1t}\rho_{1t}\ (forAP) \tag{4.11}$$

$$p_{2d}[T] = \varepsilon_{2d}\rho_{2d},\ and\ p_{2t}[T] = \varepsilon_{2d}\rho_{2d}\ (for\ WCs) \tag{4.12}$$

Putting Eq. (11) in Eq. (1), (2) and Eq. (12) in Eq. (3) (4), we have Eq. (13), (14) and Eq. (15) (16) respectively.

$$p_{1d} = 1 - (1 - \varepsilon_{2d}\rho_{2d})^{(N-1)}(1 - \varepsilon_{2t}\rho_{2t})^{(N-1)} \tag{4.13}$$

$$p_{1t} = 1 - (1 - \varepsilon_{2t}\rho_{2t})^{(N-1)}(1 - \varepsilon_{2d}\rho_{2d})^{(N-1)} \tag{4.14}$$

Similarly for the WCs

$$\begin{aligned} p_{2d} = 1 - (1 - \varepsilon_{2d}\rho_{2d})^{(N-2)}(1 - \varepsilon_{2t}\rho_{2t})^{(N-1)} \\ (1 - \varepsilon_{1d}\rho_{1d})(1 - \varepsilon_{1t}\rho_{1t}) \end{aligned} \tag{4.15}$$

$$\begin{aligned} p_{2t} = 1 - (1 - \varepsilon_{2d}\rho_{2d})^{(N-2)}(1 - \varepsilon_{2d}\rho_{2d})^{(N-1)} \\ (1 - \varepsilon_{1t}\rho_{1t})(1 - \varepsilon_{1d}\rho_{1d}) \end{aligned} \tag{4.16}$$

The combined VoIP and tracking traffic transmissions probabilities of both AP and WCs is represented as single probabilities

$$p_1 = p_{1d} + p_{1t} \tag{4.17}$$

$$p_2 = p_{2d} + p_{2t} \tag{4.18}$$

These are the conditional collision probabilities of the AP and WC due to combined VoIP and tracking traffic. Once we determine these probabilities then we can evaluate the queue utilization of both AP and WCs, but we need to calculate the average service rate of combined VoIP and tracking traffic of both AP and WCs. In the same manner the queue utilization ratio of the combined VoIP and tracking traffic can also be represented as a single ratio for both AP and WCs.

$$\varepsilon_1 = \varepsilon_{1d} + \varepsilon_{1t} \tag{4.19}$$

$$\varepsilon_2 = \varepsilon_{2d} + \varepsilon_{2t} \tag{4.20}$$

During the frame service time a node can suffer collisions, caused by either simultaneous transmission by various nodes or the channel is sensed busy, and can go in backoff stages. Alternatively, a node successfully transmits if any of the remaining nodes do not transmit. The time duration due to the successful transmission of either VoIP or tracking traffic is denoted as T_s while the collision time is calculated as T_c. In DCF-based WLAN, the successful transmission time of a frame consists of the transmission of data frames (VoIP or tracking frames), the transmission of ACK frame (due to either VoIP or tracking frames), a SIFS and then a DIFS timer. Similarly, the busy time due to collision consists of the transmission of data frames, the ACK timeout and a DIFS timer. These can be determined as

$$T_s = T_{d/t} + SIFS + T_{d_{ack}/t_{ack}} + DIFS \tag{4.21}$$

$$T_c = T_{d/t} + ACK_{d_{timeout}/t_{timeout}} + DIFS \tag{4.22}$$

The collision T_c and successful T_s transmission times are contributed by the conditional collision probability p_i and successful transmission probability $1 - p_i$ respectively. The average collision time of either AP or WC is approximated in [21] by considering the collision time of a frame transmitted by AP or WC as a geometrically distributed random variable.

$$T_{c_1} = \frac{p_1(1 - (m+1)\, p_1^m + m p_1^{m+1})\, T_c}{(1 - p_1)} \tag{4.23}$$

$$T_{c_2} = \frac{p_2(1 - (m+1)\, p_2^m + m p_2^{m+1})\, T_c}{(1 - p_2)} \tag{4.24}$$

In the basic access mode of IEEE 802.11; the channel is sensed busy due to successful transmission of either AP or any of the WCs, or due to collision that occurs between WCs or between the AP and WCs. In the same way, a node undergoes backoff stages when channel is sensed busy due to either collision or successful transmission times. We need to evaluate the average backoff time when the AP and WC experiences until it successfully transmits a frame. In DCF mode, a backoff window timer is selected from [0, CW_i] where CW_i is the current contention window. The average backoff timer for the AP is considered as geometrically distributed and is approximated in [21] as

$$w_i = (1 - p_i)\frac{W}{2} + ... + p_i^{m'}(1 - p_i)\frac{\sum_{i=0}^{m'} 2^i W}{2} +$$
$$... + p_i^m \frac{\sum_{i=0}^{m'} 2^i W + (m - m')2^{m'} W}{2} \tag{4.25}$$

Here the first term determines the successful transmission of a frame by randomly selecting a timer from backoff window. Similarly the fifth term calculates the average backoff timer by calculating five times the conditional collision probability along with the randomly selected backoff timer from respective backoff window and the success probability after the frame is successfully transmitted. From the above analysis we have the collision probabilities, the average successful and collision times, and the average backoff times of both the AP and WCs. This analysis contributes the service time of a frame that can be transmitted through the AP and WCs. The service time of the AP is approximated in [21] as

$$\frac{1}{\eta_1} = \left((N-1)\frac{\delta 1}{\eta_1} + 1\right) T_s + \frac{1}{2}\left[(N-1)\frac{\delta 1}{\eta_1} T_{c_2} + T_{c_1}\right] + w_1 \tag{4.26}$$

Here $\delta 1$ is composed of the arrival rate of combined VoIP and tracking traffic. On average all the WCs except the AP successfully transmits $(\mathrm{N}-1)\frac{\delta 1}{\eta_1}$ frames and it takes $(\mathrm{N}-1)\frac{\delta 1}{\eta_1}\mathrm{T}$ time slots because each successful transmission takes T_s time slots. The average collision time suffered by each node is $(N-1)\frac{\delta 1}{\eta_1} T_{c_2} + T_{c_1}$. However collision occurs because of the simultaneous transmission of two nodes, which means that both nodes have to wait for the duration of T_c. So the time duration for the channel to be busy due to collision of two nodes is $\frac{1}{2}(N-1)\frac{\delta 1}{\eta_1} T_{c_2} + T_{c_1}$. The service time of a frame transmitted through WC is also calculated as

$$\frac{1}{\eta_2} = \left((N-2)\frac{\delta 2}{\eta_2} + 1 + \frac{(N-1)\,\delta 2}{\eta_2}\right) T_s + w_2 +$$
$$\frac{1}{2}\left(\left[(N-2)\frac{\delta 2}{\eta_2} + 1\right] T_{c_2} + \frac{(N-1)\,\delta 2}{\eta_2} T_{c_1}\right) \tag{4.27}$$

Eq. (17), (18), (23), (24), (25), and (26) can be solved for the conditional collision probabilities and the queue utilization ratio of the AP and the WCs.

4.5.1 Results from Analytical Modeling

We now present the capacity of combined VoIP and tracking wireless clients from the analytical model presented in previous section. For the evaluation of the combined VoIP and tracking capacity we utilize Maple 12 to calculate the analytical results. The physical and MAC layer parameters from IEEE 802.11b [41] and IEEE 802.11g [34] networks are taken from their respective standards e.g. the channel rate for data and various slot times like SIFS and DIFS. In the same manner all the layers overheads are considered according to layers specifications. As an example, the physical layer overheads includes Physical Layer Convergence Protocol (PLCP) consists of 48 μs and 144 μs preamble. Similarly overheads form MAC layer and RTP/UDP/IP headers are also included. Fig. (4.2) shows the conditional collision probabilities of AP using the VoIP only and combined VoIP and tracking sessions. Both traffics show increase in collision probabilities as the number of VoIP only or the combined VoIP and tracking wireless clients take part in communication. However, it is observed that this increase in conditional collision probabilities is more likely to occur rapidly with combined VoIP and tracking sessions than with the VoIP only sessions. This is due to the fact that an increase number of both VoIP and tracking sessions are established which results in higher generation of both types of traffic volume. For VoIP only and combined VoIP and tracking sessions, the AP has traffic load of all the WCs i.e. $(N-1)$, however, this traffic load increases more severely in case of combined VoIP and tracking sessions. As a consequence, the AP contends the medium access for not only VoIP sessions but also for the tracking sessions among all the wireless clients. This in turn increases the collision probabilities of not only the VoIP sessions (p_{id}) but also the collision probabilities of the tracking sessions (p_{it}) over the AP. These individual collision probabilities of both sessions have the combined effect on the collision probabilities to be increased for combined VoIP and tracking sessions. It is also examined that this increase in conditional collision probabilities of VoIP sessions is more in comparison to the tracking sessions. This is because of the arrival rate of VoIP sessions is more as compared to the tracking sessions (instantaneous transmission frequency of 100ms). This observation is illustrated in Fig. (4.3).

The conditional collision probability of the AP increases as the wireless clients takes part in capacity evaluation of VoIP only or combined VoIP and tracking sessions. In the same manner the AP is more likely to go into backoff stages due to high traffic load which in turn decreases the service rate of AP. Fig. (4.4) shows the service rate of AP with VoIP only and combined VoIP and tracking sessions. It can be observed that the service

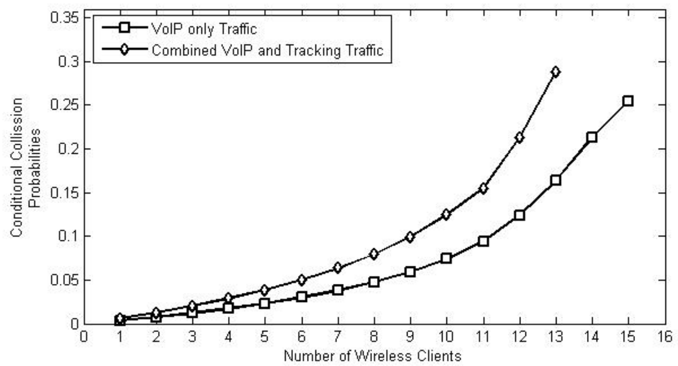

Figure 4.2: Comparison of conditional collision probabilities of VoIP only and combined VoIP and tracking sessions

rate in case of combined VoIP and tracking sessions is less as compared to the VoIP only traffic as soon as the number of wireless clients increases. In the same way, the individual service rate of both the traffics also decreases with the increase in VoIP only or combined VoIP and tracking wireless clients. However, this decrease is observed more rapidly in case of combined VoIP and tracking sessions. This rapid decrease in service rate also limits the capacity of combined VoIP and tracking sessions to less number of wireless clients as compared to the VoIP only sessions.

The combined effect of the traffic arrival and frame service rate defines the capacity of VoIP only or combined VoIP and tracking wireless clients with effect of queue utilization ratio of AP. The queue utilization ratio increases as the number of wireless clients increases and describes whether the node is stable or unstable. If the queue utilization ratio of a node is less than 1 (i.e. $\varepsilon_1 < 1$), then the node is stable. However, a node is considered unstable if its queue utilization ratio exceeds 1 (i.e. $\varepsilon_1 \geq 1$). Fig. (4.5) shows the comparison of queue utilization ration of AP for VoIP only and combined VoIP and tracking traffic. It can be observed from figure that the queue utilization ratio for combined VoIP and tracking traffic quickly approaches 1 as compared to the VoIP only traffic as the number of wireless clients increases. Besides the increase in number of wireless clients, the AP observes more traffic load due to the combined VoIP and tracking traffic which in turn also restricts its capacity to less number of wireless clients.

The combined VoIP and tracking capacity of the wireless clients is achieved under the acceptable constraints defined for voice quality. Real time multimedia applications are sensitive to delay, jitter and packet loss. The capacity

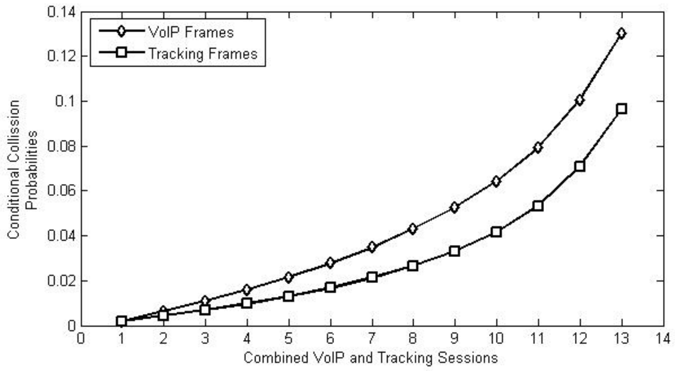

Figure 4.3: Comparison of conditional collision probabilities of VoIP frames and tracking frames in combined VoIP and tracking traffic

of the wireless clients is achieved when the traffic arrival rate is less than the service rate. When the traffic arrival rate is larger than the service rate then the queue of the nodes build up more frequently, thus due to queuing delay and overflow the real time applications severely losses their quality. To illustrate this we use 30ms of packetization interval. As the number of wireless clients increases, the traffic arrival rate of the AP increases linearly while the frame service rate shows a non linear behavior, as shown in Fig. (4.6). Due to the characteristics of the voice traffic, the traffic arrival rate of the WCs is constant while its service rate degrades more rapidly than the AP because of higher collision probability. The service rate of the AP is more than the WC due to which it enters the saturation mode early before the WCs because of its high traffic volume. It can be observed from Fig. (4.6), as soon as the thirteenth connection joins the network the queue of the AP is not stable any longer. Thus at 30ms of packetization interval a maximum of fourteen connections can be established over an AP.

We observed that the combined effect of increasing the number of wireless clients and the traffic load over a node utilizes the queue more rapidly. This fact is profoundly observed with the AP rather than the WC because here the AP is bottleneck. This utilization of the queue defines the capacity of wireless clients that can be accommodated under a single AP. Fig. (4.7) shows the capacity of combined VoIP and tracking sessions with various packetization intervals. The maximum number of wireless clients is achieved as far as the queue utilization of AP is less than 1 with each packetization interval. It can be observed from Fig. (4.7), with G.711 voice codec and 10ms of packetization interval a maximum of six wireless clients can be sup-

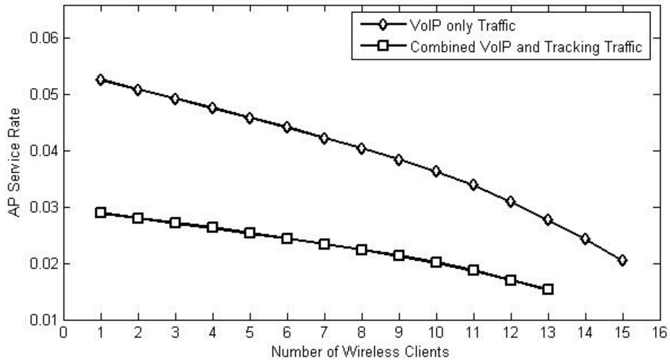

Figure 4.4: Comparison of AP service rate of VoIP only and combined VoIP and tracking sessions

ported. When sixth connection takes part in, the queue of the AP is not stable and it crosses 1 i.e. $\varepsilon_1 > 1$. Similarly, with 20ms of packetization interval eleven wireless clients can be supported and thirteen with 30ms in IEEE 802.11b DCF-based WLAN. Additionally, more wireless clients can be accommodated with higher packetization interval, as with higher packetization interval less number of VoIP and tracking traffic is triggered as compared to lower packetization intervals.

4.5.2 Capacity Evaluation of Combined VoIP and Tracking Wireless Clients

In this subsection, we discuss the capacity of combined VoIP and tracking wireless clients from the above analysis for IEEE 802.11 DCF-based WLAN, as shown in Table 4.4. For the comparison purpose of combined VoIP and tracking capacity; the VoIP only results are also presented in Table 4.4. We are specifically interested in IEEE 802.11b and IEEE 802.11g standards. The analytical results are obtained as far as the queue of the AP is stable i.e. $\varepsilon_1 < 1$. It is observed that the conditional collision probability increases as the number of wireless clients for both VoIP only as well as for combined VoIP and tracking increases as shown in Fig. (4.2). In the same manner, it can also be observed from Table 4.4, that the capacities of VoIP only and combined VoIP and tracking do not show a significant difference for lower packetization intervals e.g. 10ms and 20ms. However, an observation found in both WiFi standards, the number of supported wireless clients is limited as the packetization interval increases. As an example in IEEE 802.11b

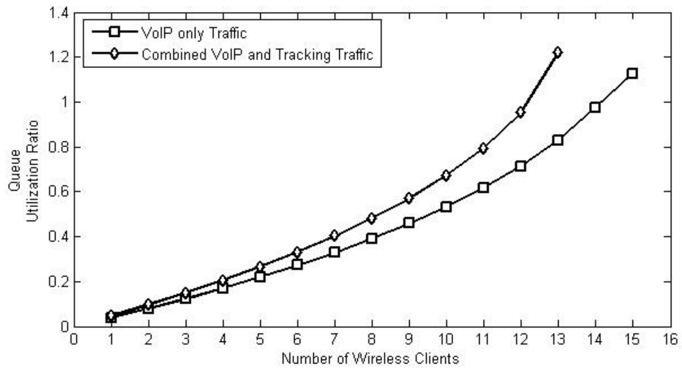

Figure 4.5: Comparison of queue utilization ratio of AP for VoIP only and combined VoIP and tracking sessions

and IEEE 802.11g, at 60ms the VoIP only capacity is 36 and 127 while the combined VoIP and tracking capacities are 24 and 82 wireless clients respectively. The reason behind this observation is that lower packetization intervals, less number of wireless connections are established while at higher packetization intervals higher number of wireless connections are established and results in generation of huge amount of traffic volume. As a result, for the case of combined VoIP and tracking sessions, it found a high contention in wireless medium as a result the queue of the AP become frequently utilized as shown in Fig. (4.5).

Table 4.4: Combined VoIP and Tracking Capacity over WiFi Networks

	IEEE 802.11b		IEEE 802.11g	
Packetization Interval (ms)	VoIP Only Capacity	VoIP & Tracking Capacity	VoIP Only Capacity	VoIP & Tracking Capacity
10	7	6	22	19
20	13	11	45	36
30	20	14	68	51
40	25	18	87	61
50	30	22	108	73
60	36	25	127	82

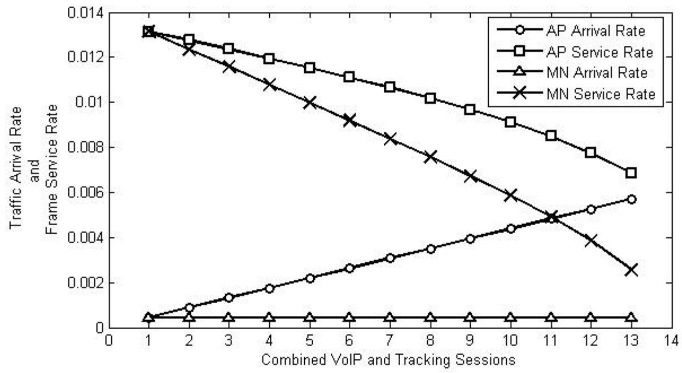

Figure 4.6: Traffic arrival and frame service rate of combined VoIP and tracking traffic

Table 4.5: Combined VoIP and Tracking Capacity at the Transmission Frequency of 100ms over IEEE 802.11b

Packetization Interval (ms)	Analysis VoIP & Tracking Capacity	Simulations VoIP & Tracking Capacity	Experiments VoIP & Tracking Capacity
10	6	6	5
20	11	11	9
30	14	14	12
40	18	18	15
50	22	21	19
60	25	24	21

4.6 Comparative Analysis form Analytical Model, Simulations and Experiments

We now carry out a comparison of the capacity for the combined VoIP and tracking under a single AP from the proposed analytical models, simulations and experiments. Table 4.5 and Table 4.6 illustrates the simulations and experimental results obtained from IEEE 802.11b and IEEE 802.11g networks respectively, analytical results are also expressed for comparison and validity purposes. During the simulations and experiments that as soon as the number of wireless clients becomes increase and take part in communication, then it is observed that the packet loss also increases, as shown in Fig. (3.5) for both standards. As observed during the mathematical analysis that the capacity for VoIP only and combined VoIP and tracking does not

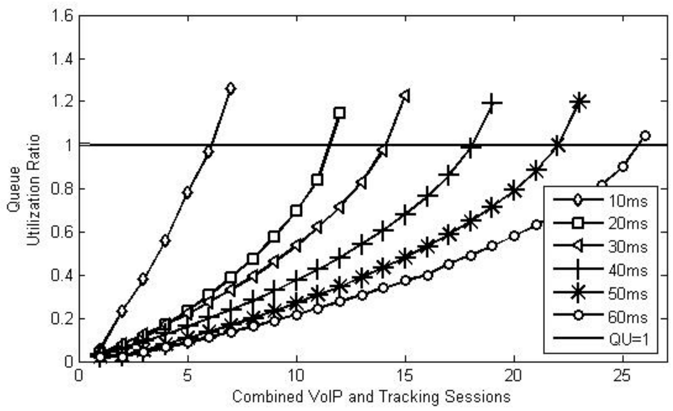

Figure 4.7: Queue utilization ratio of combined VoIP and tracking sessions

Table 4.6: Combined VoIP and Tracking Capacity at the Transmission Frequency of 100ms over IEEE 802.11g

Packetization Interval (ms)	Analysis VoIP & Tracking Capacity	Simulations VoIP & Tracking Capacity	Experiments VoIP & Tracking Capacity
10	19	19	17
20	36	35	32
30	51	50	47
40	61	61	59
50	73	71	69
60	82	80	77

tend to show significant separation in wireless capacities for lower packetization intervals (e.g. 10ms and 20ms). The same observation is also found with simulations and experiments. Thus the capacity of combined VoIP and tracking for higher packetization intervals (like 60ms) are profoundly limited to less number of wireless clients. This observation is found for both IEEE 802.11b and IEEE 802.11g standards. As an example, in IEEE 802.11b, the VoIP only capacity is 34 while the combined VoIP and tracking capacity is 24. The reason behind this verity is the generation of high traffic volume by the establishment of more number of combined VoIP and tracking sessions. As a result the wireless clients not only trigger a self contention but also they contend for the wireless medium access among themselves. The contentions in the wireless medium, in this way, become more severe. Since the AP is the bottleneck and there is a higher work load posed by all wireless clients, consequently the queue of AP overflows which results in heavy packet loss on wireless side. This degrades the voice quality and limits the combined VoIP and tracking capacity by violating QoS constraints for VoIP sessions.

4.7 Summary

In this chapter, we carried out a detailed analysis of the IEEE 802.11 DCF-based networks, by exploiting the queuing system. We had an introduction of the queuing system and its different states though the processes can undergoes. First we carry out the capacity analysis for the VoIP only and the tracking only wireless clients under a single AP and then we extended the same analytical model for the capacity evaluation of combined VoIP and tracking applications. Building upon the same model we discussed the results obtained for tracking only applications, VoIP only applications and the combined VoIP and tracking applications. We also carried out the comparison of the results obtained from the three models i.e. analytical model, simulation model and the experimental model. The results from the analytical model has also been discussed where we evaluated the behavior of the queue of the AP and wireless clients, the collision probabilities of the AP and the wireless clients for the collision of the packets during transmission, the arrival and service rate and time, and the backoff stages through which the wireless clients and the AP undergo.

Chapter 5

VoIP and Tracking Capacity Analysis in Real Networks

5.1 Introduction and Motivation

Datagram Congestion Control Protocol (DCCP) has been proposed by Internet Engineering Task Force (IETF) that is a good candidate intended for a replacement protocol for UDP in real-time multimedia applications over the Internet, such as VoIP. It provides connection oriented and congestion-controlled transport layer transmission like TCP, with reliable connection establishment and teardown, but unreliable data transmission like UDP. For the performance evaluation of the proposed tracking only and combined VoIP and tracking wireless clients, we utilized the whole network for UDP traffic in the absence of cross traffic. However, according to [26], this assumption does not stand valid and the authors further observed that major part of the internet is being utilized by data-oriented traffic (i.e. TCP traffic). In effect, they examined that 20% of the traffic is non-data oriented and 80% is data oriented. In this section we carry out this observation and advocate the use of DCCP for real time traffic. To carry on the performance study over DCCP, we presented the problem of starvation being observed by TCP flows caused by the real time UDP flows. Subsequently, we demonstrated the performance gains that are achieved by using DCCP. Since our focus for the capacity evaluation of combined VoIP and tracking wireless clients is to maintain minimum end-to-end delay for VoIP traffic and to allocate fair resources to TCP traffic. For this reason, we utilized TFRC as the congestion control mechanism of DCCP. This is because TFRC is a receiver based congestion control mechanism and minimizes the abrupt rate changes [5], so that it keeps data rate smoothly for the entire communication. In the UDP, DCCP and TCP performance evaluation (see next section), we showed that if UDP is used in the presence of TCP (a real network scenario) then UDP does not consume the capacity of network due to high packet loss, the QoS constrains of VoIP sessions are violated, and TCP also goes in starvation and does not get a reasonable throughput. Conversely, if DCCP is used as a transport protocol, the voice quality of VoIP sessions is maintained and TCP also get a sound throughput by successfully transmitting through the entire communication sessions.

5.2 Need for Capacity Evaluation using DCCP

In this section, we evaluated the need for the capacity evaluation of location tracking only and combined VoIP and tracking wireless clients using DCCP as a multimedia transport protocol. We carried out the performance study of UDP and DCCP by using similar number of bi-directional concurrent VoIP calls on both UDP and DCCP in the existence of same number of TCP traffic. We examined the network capacity usage by each flow, the preservation of voice quality of simultaneous VoIP sessions and the obtained TCP throughput. The different parameters and the specifications of simultaneous VoIP traffic using UDP and DCCP are similar to those discussed in Section V, whereas the TCP traffic is simulated with File Transfer Protocol (FTP traffic). The average TCP throughput obtained from the simulations, as shown in Fig. (5.1), deviates between 13Mbps to 33Mbps for the first 50s. However, during when bi-directional VoIP calls with UDP are introduced, this fluctuation in throughput is considerably decreased to about few Kbps and is observed to be the same until the end of whole communication (as shown in Fig. (5.1) with an arrow). In the mean while, a 34.87% of packet loss is observed by bi-directional UDP traffic which is extremely high packet loss than the loss threshold classified for VoIP traffic (i.e. 2%). Due to high packet loss, it severely degrades the quality of voice and also wastes network's bandwidth. Conversely, a fair allocation of resources is observed by using bidirectional VoIP calls simulated with DCCP in the presence of TCP traffic (as shown in Fig. (5.2)). In the same way, TCP traffic also gets a good share until the end of communication, though the obtained TCP throughput is a bit lower as compared to the first 50s. Likewise, the observed QoS constraints experienced by bi-directional DCCP traffic are within the acceptable threshold for VoIP calls e.g. the packet loss and delay is 1.34% and 142ms respectively.

Above performance analysis motivates for the need for the capacity evaluation using DCCP . Now we first investigate the capacity of combined VoIP

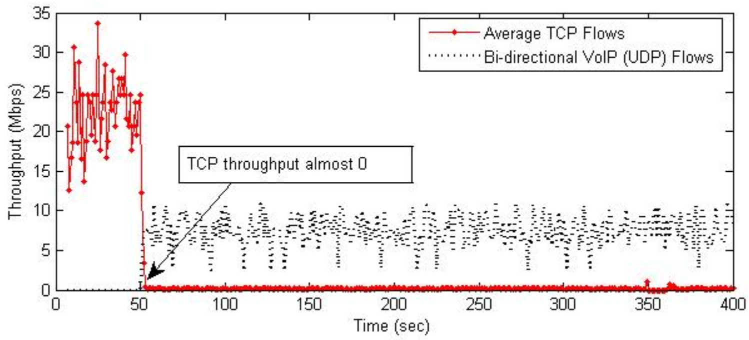

Figure 5.1: Simultaneous UDP and TCP Traffic

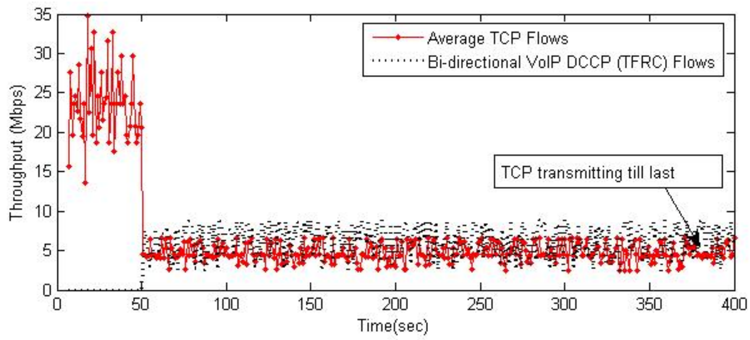

Figure 5.2: Simultaneous DCCP and TCP Traffic

and tracking wireless clients using DCCP only and then a detailed capacity analysis using UDP and DCCP in the presence of TCP traffic.

5.3 Capacity Evaluation using DCCP

For the capacity analysis of combined VoIP and tracking wireless clients, we take the traffic specifications from simulations and experiments the same as discussed in Section III and IV. This evaluation contains capacities from simulations and experiments using both DCCP's congestion control mechanisms, under the constraints defined for QoS for voice sessions in [20][21][1], as shown in Table 5.1 and Table 5.2 for IEEE 802.11b and IEEE 802.11g respectively. For a comprehensive analysis of the capacities obtained with DCCP, combined VoIP and tracking capacity using UDP is also presented in Tables 5.1 and 5.2 . It is observed that the packet loss and end-to-end delay increases as the number of wireless clients are increased. However, due to the congestion control mechanisms provided by DCCP, the end-to-end delay seems to be more prominent rather than the packet loss for the case when DCCP is used as a transport protocol. The delay encountered for DCCP (TFRC) in IEEE 802.11b for each packetization intervals is shown in Fig. (5.3). It is further examined that the delay for TCP Like for each packetization intervals is more than the delay obtained with TFRC, which restricts the combined VoIP and tracking capacity with TCP Like to less number of wireless clients and can be observed from Table 5.1 and Table 5.2. The reason behind this fact is that TFRC is a receiver based congestion control mechanism and provides such rate that minimizes abrupt rate changes [5] so that data is sent smoothly during the entire communication which in turn also minimizes end-to-end delay. On the other hand, the abrupt change in sending data rate (as with the TCP Like [4]) slows down the sending rate thus the packets have to wait in sender's queue at transport layer for some amount of time to be served to the destination. Another observation from Table 5.1 and Table 5.2 is that for both IEEE 802.11b and IEEE 802.11g, the capacity decreases with DCCP for each packetization interval when they are compared with the capacities obtained with UDP in section V. As an example, with 10ms of packetization interval, six combined VoIP and tracking wireless clients are supported while it reduces to 3 and 5 wireless clients with TCP Like and TFRC respectively. The reason behind these lower capacities with both congestion control mechanisms of DCCP is that the network is being utilized by VoIP data packets, tracking packets and their corresponding acknowledgements (due to congestion control mechanism). Since both WiFi standards process upper layer packets without considering data or acknowledgement packets, so both standards apply the basic channel access mechanism and both types of packets are kept in AP's buffer. The actual data packets are thus delayed due to the existence of non-data packets, whereas using UDP as a transport layer protocol the AP need not to process such non-data packets.

5.4 VoIP and Tracking Capacity Evaluation using UDP and DCCP in the Presence of TCP Traffic

We now evaluate the capacity of proposed TVA architecture using UDP and DCCP for real time VoIP traffic along with the tracking traffic in the presence of TCP traffic. The capacity of the TVA architecture is determined from 20% of UDP and DCCP traffic in the existence of 80% of TCP traffic

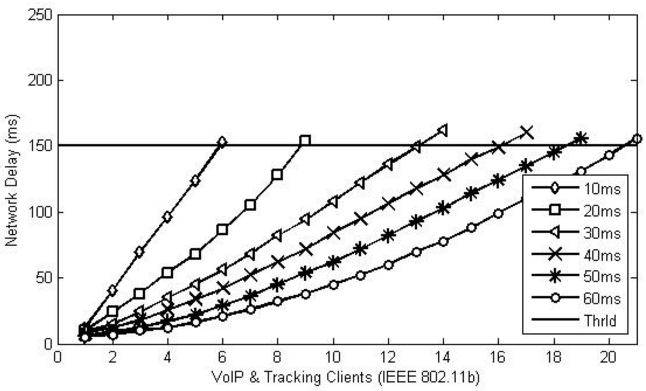

Figure 5.3: Simultaneous DCCP and TCP Traffic

Table 5.1: Combined VoIP and Tracking Capacity over IEEE 802.11b using DCCP at the Transmission Frequency of 100 ms

Packetization Intervals (ms)	Simulations			Experiments	
	UDP	TCP Like	TFRC	TCP Like	TFRC
10	6	3	5	3	3
20	11	7	10	5	8
30	14	10	13	9	11
40	18	12	16	11	14
50	21	15	18	13	16
60	24	17	20	15	18

is shown in Table 5.3 using simulations and Table 5.4 using experimental setup; such fractions of data traffic usage is also given in [26]. The last three columns from the same table demonstrate the obtained capacities (within the thresholds for QoS constraints of VoIP traffic as discussed in Section V) in the presence and absence of TCP traffic together with their corresponding percentage decrease in capacity. It can be seen that the percentage reduction in capacities with UDP traffic is in greater amount as compared to the capacities obtained with DCCP traffic. As an example, the packetization interval of 10ms, UDP reduces the capacity of supported wireless clients to 50% in the presence of TCP traffic compared to the obtained capacity, conditioned to the situation when there is no TCP traffic. Contrary to the reduction in capacities with UDP, this decrease in supported wireless clients is far less with DCCP and is only 20%. Likewise, DCCP supports more wireless clients

Table 5.2: Combined VoIP and Tracking Capacity over IEEE 802.11g using DCCP at the Transmission Frequency of 100 ms

Packetization Intervals (ms)	Simulations			Experiments	
	UDP	TCP Like	TFRC	TCP Like	TFRC
10	19	15	16	14	14
20	35	22	29	20	27
30	50	31	33	29	31
40	61	35	42	32	39
50	71	42	55	40	53
60	80	47	67	45	64

Table 5.3: Performance Statistics from bi-directional VoIP and Tracking UDP/DCCP traffic along with TCP traffic: Simulation Results

Packetization Interval (ms)	Combined Traffic Flows	Average Throughput (Mbps)	Traffic Bandwidth Usage (%)	Data Loss (%)	With TCP Capacity	Without TCP Capacity	Decrease in Capacity (%)
10	UDP-TCP	UDP:6.35 TCP:1.14	UDP:57.72% TCP:10.36%	UDP:1.86% TCP:0.28%	3	6	50.00%
	DCCP-TCP	DCCP:5.36 TCP:2.85	DCCP:48.72% TCP:26.00%	DCCP:1.04% TCP:0.46%	4	5	20.00%
20	UDP-TCP	UDP:5.85 TCP:1.42	UDP:1.69% TCP:12.90%	UDP:1.86% TCP:0.17%	4	11	63.64%
	DCCP-TCP	DCCP:4.16 TCP:2.77	DCCP:37.81% TCP:25.18%	DCCP:0.85% TCP:0.27%	6	10	40.00%
30	UDP-TCP	UDP:5.22 TCP:1.61	UDP:47.45% TCP:14.64%	UDP:1.74% TCP:0.67%	7	14	50.00%
	DCCP-TCP	DCCP:3.93 TCP:2.91	DCCP:35.73 TCP:26.45%	DCCP:0.96% TCP:0.71%	8	13	38.46%
40	UDP-TCP	UDP:4.78 TCP:1.85	UDP:43.45% TCP:16.82%	UDP:1.84% TCP:0.99%	11	18	38.89%
	DCCP-TCP	DCCP:3.63 TCP:3.09	DCCP:33.00% TCP:28.09%	DCCP:1.03% TCP:0.79%	12	16	25.00%

than the clients maintained by UDP in the presence of TCP traffic. Table 5.3 also illustrates the packet loss observed by all respective flows (5th column), the obtained throughput (3rd column), and their corresponding bandwidth usage (4th column). This can be examined that at the packetization interval of 10ms, the average throughput obtained by TCP is only 1.14Mbps when UDP is active. This TCP throughput keeps up a correspondence of 10.36% of the entire network bandwidth. On the other hand, with the same packetization interval, TCP obtains a reasonable bandwidth share of 26% when run it with DCCP. The TCP obtained throughput share is around 2.5 times more when compared to the TCP throughput with UDP. Similar results are also shown by the experimental setup as shown in Table 5.4. Another important observation is that at lower packetization intervals, there are more

Table 5.4: Performance Statistics from bi-directional VoIP and Tracking UDP/DCCP traffic along with TCP traffic: Experimental Results

Packetization Interval (ms)	Combined Traffic Flows	Average Throughput (Mbps)	Traffic Bandwidth Usage (%)	Data Loss (%)	With TCP Capacity	Without TCP Capacity	Decrease in Capacity (%)
10	UDP-TCP	UDP:5.92 TCP:1.05	UDP:53.82% TCP:9.55%	UDP:1.98% TCP:0.48%	2	5	60.00%
	DCCP-TCP	DCCP:5.04 TCP:2.55	DCCP:45.82% TCP:23.18%	DCCP:0.86% TCP:0.73%	2	3	33.33%
20	UDP-TCP	UDP:5.61 TCP:1.13	UDP:51.00 % TCP:10.27%	UDP:1.82% TCP:0.49%	3	9	66.67%
	DCCP-TCP	DCCP:3.92 TCP:2.45	DCCP:35.64% TCP:22.27%	DCCP:1.34% TCP:0.63%	5	8	37.50%
30	UDP-TCP	UDP:5.00 TCP:1.34	UDP:45.45 % TCP:12.18%	UDP:1.88% TCP:1.02%	6	12	50.00%
	DCCP-TCP	DCCP:3.60 TCP:2.51	DCCP:32.73% TCP:22.82%	DCCP:1.27% TCP:1.03%	7	11	36.36%
40	UDP-TCP	UDP:4.44 TCP:1.46	UDP:40.36% TCP:13.27%	UDP:1.93% TCP:1.45%	9	15	40.00%
	DCCP-TCP	DCCP:3.57 TCP:2.66	DCCP:32.45% TCP:24.18%	DCCP:1.48% TCP:0.96%	10	14	28.57%

numbers of packets sent as compared to the higher packetization intervals, thus UDP consumes major part of the network bandwidth and TCP gets a smaller amount of the total bandwidth. On the other hand, in case of higher packetization intervals, less numbers of UDP packets are sent that results in a better bandwidth share for TCP traffic. Thus at higher packetization intervals (like 40ms, 50ms, 60ms), it is observed that TCP is able to get more throughput as compared to the lower packetization intervals. This fact is further demonstrated strongly when TCP is operational with DCCP. Based on this analysis, we bring to close that DCCP improves the capacity of the proposed TVA architecture and it also fairly coexist with TCP due to its improved congestion control mechanism.

5.5 Summary

In this chapter, we discuss the motivation for the capacity evaluation using DCCP, where we discuss the greedy approach adopted by the UPD while it is being simulated with the TCP traffic. We further discuss the need for the evaluation of capacities using DCCP and their results with both congestion control mechanisms of DCCP. We further discuss the capacity evaluation using UDP and DCCP in the presence of the TCP traffic to simulate the real-network scenario by taking a proportion of the two types of traffic.

Chapter 6

Conclusion

6.1 Summary of Contributions

Following are the main contributions drawn from this research work:

- We evaluate the capacity of location tracking wireless clients using simulations and experiments over IEEE 802.11b and IEEE 802.11g networks.

- We determined the effect of transmission frequency of the tracking data on tracking capacity which refers to different situation regarding the mobility of the wireless clients, which are categorized as fast moving wireless clients and wireless clients with less mobility.

- We investigated the effect of the variation of the tracking data over the capacity of the wireless tracking clients. This variation in size of the tracking data with different sizes meant for the wireless clients situated in a dense area of access points or the area where there are less number of wireless clients, thus showing a good mean of flexibility for the location tracking applications.

- We developed a mathematical probabilistic model for the capacity evaluation of VoIP and tracking wireless clients to quantify the upper bound on the number of wireless clients under a single access point. We utilized the queuing theory to analyze the behavior of different process used in IEEE 802.11 DCF-based networks for their waiting times, serving time and backoff mechanism.

- The capacity of the combined VoIP and tracking wireless clients is also evaluated using UDP and DCCP in the absence and presence of the

TCP traffic, this referred to the real network scenario where a portion of the TCP and UDP traffic exists.

6.2 Conclusions from Research Work

In the recent past, it has not been determined the capacity of the combined VoIP and tracking applications in the literature. In this paper, we determine the capacity in WiFi networks through analysis, simulations and experiments for the location tracking applications. We then quantify the consequences on tracking capacity by varying the transmission frequency of tracking information. We inferred that at higher packetization intervals (50ms, 60ms), the capacity of combined VoIP and tracking sessions reduces by a greater amount. We deduced that the transmission frequency of tracking information relates to the speed of wireless clients that results in greater capacity for slow users. The capacity of the combined VoIP and tracking sessions is further calculated in the presence of TCP traffic. We examined that significant performance are achieved by utilizing DCCP at low packetization intervals. In addition, it is noticed that DCCP performs better than UDP in terms of improving the capacity of combined VoIP and tracking wireless clients and providing faire share to TCP traffic.

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