CHAPTER 1

INTRODUCTION

1.1 GENERAL

The fourth generation (4G) wireless networks are targeted to support various data and multimedia applications over packet-switched networks (Dapeng 2005). Quality of Service (QoS) guarantee for various applications is the main objective in the design of next generation wireless networks. The remarkable trend of network evolution towards convergence of wireless and wired networks is developed in addition to convergence of voice and data into a common packet switched network infrastructure. Although IEEE 802.11 Wireless Local Area Network (WLAN) is the most widely used standard today, it cannot provide QoS support for the increasing number of multimedia applications (Qiang et al 2004). The QoS requirements set by forthcoming applications such as video chat and mobile TV are not met by the current communication systems. The main task of providing QoS is to ensure that users’ requirements are satisfied throughout the entire service period. The most common QoS requirements include hand-off latency, minimum / maximum throughput, delay bound or delay jitter and packet loss rate.

Compared with a wired infrastructure, WLAN has unique advantages, such as broadband bandwidth capability and low deployment cost. The WLAN market is experiencing explosive growth in hotspots and in places like hotels, hospitals, airport etc. With WLANs being deployed in unlimited ways as Access Points (AP), wireless users can access real-time and Internet services virtually anytime, anywhere, while enjoying the flexibility of mobility and guaranteed connectivity. Aditya et al (2005) presented an
analysis of a data set with 28475 APs; the data was collected by the Intel Place lab project in six US cities in June 2004. The analysis shows that more than half of the APs have three or more neighbours within transmission range. The maximum number of neighbours for one AP was 85 in Boston city. Due to the still rising popularity of WLANs, it is expected that these densities will continue to increase. The lack of inbuilt mechanism for the support of real-time services makes it very difficult to provide QoS for throughput-sensitive and delay-sensitive multimedia applications. Therefore, modification in existing 802.11 standards are necessary. Although IEEE 802.11e is being proposed as a standard for the enhancement of service differentiation, QoS guarantee in 802.11 is still a very challenging problem and needs further study (Li et al 2003).

The maintenance of QoS in a wireless environment is difficult. Most researchers have already acknowledged adaptive techniques as the appropriate methods to deal with the fluctuation of QoS in a wireless network. However, it is not easy to implement these services with optimum QoS provision as mobile wireless networks pose significant challenges due to the mobility of a user, limited radio spectrum and radio channel impairment. A good mechanism is needed to provide a guaranteed QoS to the system as what wireline networks are capable of providing. Call Admission Control (CAC) is a control mechanism with a specific goal to maintain a fine balance between maximizing network utilization and delivery of QoS to an ongoing connection. It is the first control step in the provisioning of network resources to connections. Based on a decision algorithm, Dynamic Call Admission Control (DCAC) framework determines whether to admit a new connection or not, based on the current usage level of network resources.

Thus, mobility management along with QoS provisioning in wireless and mobile networks has been a topic of significant research in recent years.
1.2 CHALLENGES IN PROVIDING QoS

Quality of service (QoS) support for multimedia services in the IEEE 802.11 wireless LAN is an important issue. The issues and effects are summarized in Figure 1.1. Some of the QoS issues attributed to the successful implementation of wireless/mobile environment are outlined in the literature (Qiang et al. 2004 and Hua et al. 2004):

- Seamless Mobility
- Quality of Service
- Call Admission Control
- Resource Reservation and Load Sharing

![Figure 1.1 Challenges in providing QoS in IEEE 802.11 WLAN](image-url)
The following sections describe each of the challenges in detail and emphasize the need to address the challenges by discussing their impact on providing QoS.

1.2.1 Seamless Mobility

Voice over WLANs allows voice users to roam without restriction. Implementing this capability involves Layer-2/Layer-3 roaming, in a very fast manner and with strong security all the way. As the users roam throughout the enterprise or a “hot zone,” their calls will be handed off to multiple APs. If these APs belong to the same IP subnet, this hand-off is performed at Layer-2; if the move is across subnets, then the hand-off is at Layer-3. Roaming between APs inevitably involves some delay or latency. In a voice call, hand-off latency should be less than 50 milliseconds. Otherwise the connection will be dropped or the call quality will severely degrade. The Layer-2 roaming latency is mainly due to three processes: scanning, re-association and re-authentication. Using current standards (active scanning, 802.11F Inter-Access Point Protocol, and 802.1x authentication) hand-off latency can vary from a few hundred milliseconds to several seconds, which is unacceptable for voice calls.

Hand-off management is responsible for monitoring the active session of the mobile devices, when it moves across the coverage areas of various APs. The hand-off results in a change of the network layer connectivity of the mobile host. A hand-off control protocol should ensure that hand-offs are fast and smooth without significant delays and without packet losses. Hand-off delay is defined as the time between the delivery of the last packet to the mobile device from the old AP and the delivery of the first packet from the new AP.
Regardless of the hand-off mechanism used within an IP access-network, it should be stressed that the hand-off performance is also highly dependent on the underlying radio technology and the information the latter provides to the IP layer. For example, a radio layer that provides indications to the IP layer of an impending hand-off enables the preparation and possible completion of the IP hand-off before the mobile device loses its layer-2 connectivity. Alternatively, hand-off decision can solely be based on layer-3 indication and completely independent of the layer-2 technology, resulting in greater hand-off delay and a greater possibility of service disruption.

1.2.2 Quality of Service (QoS)

Maintaining call quality and ensuring end-to-end QoS is a complex task. It is required to map and extend QoS features across wireless environment as per the QoS standard and power-saving mechanisms. To provide QoS guarantees in the wireless network, a network architecture should contain the six components namely a) Traffic specification (Tspec), b) QoS routing, c) Call Admission Control, d) Wireless channel characterization, e) resource reservation, and f) packet scheduling.

Wired VoIP deployments have made it clear that call quality depends primarily on the voice codecs used (such as G.711, G.729, or iLBC) and on end-to-end latency, jitter (variations in delay) and packet loss. Voices over WLANs are subject to the same end-to-end parameters. The main differences are that a WLAN relies on a shared airlink, using different access mechanisms than wired Ethernet (CSMA/CA 802.11 vs. CSMA/CD 802.3), exhibiting relatively limited bandwidth, higher packet-error rates and packet overhead.
WLAN QoS mechanisms are critical when voice and data share the same Radio Frequency (RF) band. Data traffic typically consists of bursty transmissions and various-sized packets. Some of these can be very large (e.g., file downloads) and take more time to transmit over the airlink. This delay is smaller, for continuously transmitted voice packets. The various traffic content on the shared medium using collision avoidance increases both latency and jitter. QoS mechanisms rectify this situation by prioritizing voice over other traffic (data and video) by providing voice with faster and more time access to the medium.

1.2.3 Call Admission Control (CAC)

Overburdened APs degrade the overall voice quality and drop calls. These problems can be prevented using call admission control, load sharing and other techniques. When planning a Voice over WLAN, network capacity is a crucial design parameter. Capacity depends on many factors, the specific RF environment, infrastructure and handheld devices being used. Except for a few single-channel or array architectures, WLAN capacity is mainly determined by the number of available non-overlapping channels and the density of deployed APs. Aggregating several neighbouring APs with non-overlapping channels increases the capacity within a given area. If the channels do overlap, RF interference, packet collisions and retransmissions can result in degrading the overall performance. Collisions and retransmissions can result in degrading the overall performance. 802.11b protocol, which is a backward-compatible protocol of 802.11g standards is currently supported by most Wi-Fi phones. They both operate in the 2.4-GHz frequency band. They have only three non-overlapping channels.

Since VoIP codecs use relatively little bandwidth (the least efficient, G.711 consumes about 80 Kbps), an 802.11b AP could theoretically support
up to 23 simultaneous calls. In practice, wireless network support 7 to 10 calls. If the number of calls exceeds the practical limit, the resultant MAC collisions and retransmissions would increase delay and degrade voice communications. WLAN infrastructure vendors are implementing CAC mechanisms to limit the total number of calls. Subsequent requests are either rejected or redirected.

1.2.4 Resource Reservation and Load Sharing

Future generation communication systems are required to support multimedia services of various QoS requirements. One of the important issues in the wireless multimedia networks is the user mobility that leads to variation in wireless resource demands. This change in resource request can result in a major fluctuation in the availability of network resources, especially the resource in wireless networks. This is because wireless channels have limited available bandwidth and can easily be overloaded if many users attempt to access an AP at the same time. The overloaded situation results in forced termination of the connection due to lack of bandwidth. Therefore, mobility has a direct impact on the connection level QoS in terms of New Call Blocking Probability (NCBP) and Hand-off Call Dropping Probability (HCDP).

To guarantee an uninterrupted connection for a call during its entire lifetime, dynamic resource reservation scheme is proposed to allocate part of the network resources for high priority calls (hand-off and real-time calls). This implies that flows can be accepted as long as lower bounds can be satisfied. However, in order to increase the utilization of the network and provide better service to applications, excess resources need to be distributed effectively among the flow. These mechanisms are basically proprietary, although some are ready to accept some traffic specification (TSpec)
signaling from Wireless Multimedia (WMM) capable clients for admission control. Ultimately, WMM-capable APs will only approve TSpec requests from clients when they have enough bandwidth and resources to meet the client’s QoS requirements. Emerging standards like 802.11k and 802.11v assist handheld devices by enabling neighbouring APs to take overflow connections. Thus AP shares the load with better network utilization.

1.3 MOTIVATION

To satisfy the diverse demands of wireless communication, the 4G wireless networks provide users/applications with certain QoS. The main task of providing QoS ensures that user’s requirements are satisfied throughout the entire service period. The continuous support of minimum QoS for multimedia applications poses a major challenge in a heterogeneous wireless environment.

QoS mechanisms for 802.11 are classified as:
- Service Differentiation
- Admission Control and Resource Reservation
- Link Adaptation

1.3.1 Service Differentiation

Basically, service differentiation is achieved by two main methods: priority and fair scheduling (Pattara-Atikom et al 2003). The priority-based service differentiation method binds channel access to different traffic classes by prioritized contention parameters. The fair scheduling-based service differentiation method partitions the channel bandwidth fairly by regulating wait times of traffic classes in proportion to the given weights. Potentially, fair-scheduling-based schemes are advantageous to allocate bandwidth fairly
among traffic classes and prevent starvation of a specific class. However, they often require substantial modification of existing 802.11 standards. Compared to the fair-scheduling-based schemes, priority-based mechanisms require less modifications of the existing Distributed Co-ordination Function (DCF) access method for providing better QoS support for real-time applications.

1.3.2 Admission Control and Resource Reservation

Service differentiation is helpful in providing QoS for multimedia data traffic under low to medium traffic load conditions. However, due to the limited functionality of IEEE 802.11 Media Access Control (MAC), service differentiation does not perform well under high traffic load conditions (Lindgre et al 2001). In this case, admission control and bandwidth reservation become necessary to guarantee QoS of existing traffic. Otherwise, the extremely large saturation delay may fail to support multimedia applications. However, an IEEE 802.11 wireless network, a Mobile Client (MC) has no knowledge of the exact condition of the network. This cannot make an accurate decision to admit or deny it.

In general, admission control schemes require less modification to the 802.11 standards than bandwidth reservation schemes. Basically, admission control schemes can broadly be classified into measurement-based and calculation-based methods. In measurement-based schemes, admission control schemes decisions are made based on the measurements of existing network status, such as throughput and delay. On the other hand, calculation-based schemes construct certain performance metrics or criteria for evaluating the status of the network. Li et al (2004) introduced a novel framework for admission control with priority reservation and allocation, which focuses on optimizing the usage of priority resources. It is necessary to have a framework
for effective coordination of wireless users based on the reserved transmission rate, which uses appropriate link layer priority and allocation of sufficient resources.

1.3.3 Link Adaptation

In WLAN 802.11 network the transmission rates differ from the channel conditions. An appropriate link adaptation mechanism is desirable to maximize the throughput under dynamically changing channel conditions. Metrics used in existing link adaptation algorithms include the channel received signal strength and average payload length.

Pavon and Choi (2003) proposed an algorithm based on Received Signal Strength (RSS) as a metric for switching between appropriate transmission rates. A rate adaptation algorithm at every station maintains its own 12 RSS thresholds and corresponding transmission rates. Based on the measured RSS value, the station dynamically switches to an appropriate transmission rate.

Qiao et al (2002) proposed to use a combination of Signal to Noise Ratio (SNR), average payload length and frames retry count as the metric for the link adaptation algorithm. The proposed algorithm pre-establishes a table of best transmission rate for decision making.

In this research, active channel scanning mechanism is proposed. The probe response signal consists of its current channel number. The MC performs an active scanning and switchover to the neighbouring AP’s current channel number rather than of performing hard scanning of all the channels in the AP.
1.4 PROBLEM DESCRIPTION

The study examines QoS parameters for a mobile node when it is roaming in a campus wide wireless environment:

- **Hand-off Latency Minimization:** The key idea is to minimize the channel scanning time when the mobile devices are moving between different networks. The focus is to identify (1) the nearest AP’s channel number and (2) the threshold limit of the signal strength in the overlapping region and to analyze the capacity of the candidate AP.

- **Call admission control and resource sharing:** The Call admission control and bandwidth reservation become necessary to provide guaranteed QoS for the existing traffic. Otherwise, the extremely large saturation delay may lead to failure in supporting high priority multimedia applications. Acceptable QoS is provided to individual users by allocating system resources based on the network conditions and Tspec. The admission control decisions are made dynamically at both source and destination stations in a fully distributed way.

- **Mobility Management:** The task of mobility management in the wireless internet is basically to enable network applications to continually operate, at the required QoS, in the wireless mobile nodes throughout an IP-layer hand-off (Layer-3). The process of routing through the home agent is inefficient and is known as triangular routing. Therefore, routing optimization enhancement has to be introduced to solve this problem. Tunneling of packets from home agent to
foreign agent need to be encapsulated. During hand-off process, there is a delay resulting in increase of hand-off latency. The home agent is considered as single point of contact for communication between the mobile host and the corresponding node. When a mobile host is moving to a foreign network, and if the home agent is unavailable, then the packets designated to the mobile hosts are lost.

1.5 LITERATURE SURVEY

Due to the advancement in technology and societal changes, people are more dependent upon mobile devices. Besides the mobility of devices, they also need high-speed and reliable networks to support various applications. There has been immense growth in IEEE 802.11 based WLANs in the last few years. The latency is often considered as a hindrance in providing seamless roaming. The QoS mechanism considered under this research works are classified as (i) Hand-off latency (ii) Call admission control, and (iii) Resource sharing.

1.5.1 Hand-off Latency Minimization

The IEEE 802.11 WLAN offer wireless connectivity to user at high data rates. This technology is now widely spread at home and work environment. APs provide wireless connectivity by bridging packets from the wireless domain to an internal network. The wireless nature of 802.11 devices allows the user to freely move within AP’s coverage area, which is commonly known as a hotspot, while in communication with the network. Due to user mobility, a user may reach the boundary of its current hotspot and lose the signal from its current AP. In that case, the mobile user should change to a new AP in order to maintain the wireless connectivity. The operation to
change an association from one AP to another is known as hand-off (O’Hara and Petrick 2005). In order to determine whether an AP is operating on a particular channel, the mobile device periodically scans the channels by sending probe request message. The duration of the scanning stage strongly depends on the number of channels a mobile device has to probe (Montavont and Noel 2006). In wireless IP node is involved in two types of hand-offs: link-layer hand-off that is between two base stations and IP-layer hand-off that is between two access routers. In most cases, an IP-layer hand-off is accompanied by a link-layer hand-off.

Ramani and Savage (2005) described Sync-scan, a low cost technique for continuously tracking the nearby base stations by synchronizing short listening periods at the MC. It demonstrates a better hand-off decision but with a regular overhead of the scanning phase. It is observed that scanning time is a major overhead towards the total hand-off delay.

Brik et al (2005) introduced a hand-off scheme utilizing multiple radios called MultiScan. Similar to Sync-Scan, MultiScan obtains information on neighbouring APs by scanning opportunistically. However, MultiScan requires an additional radio interface for the channel scanning. In MultiScan, the primary interface is associated with the current AP and used for data transmission. At the same time, the secondary interface is performing the channel scanning. If a hand-off to a new AP is required, the second interface is associated with the new AP while the primary interface is still employed for data transmission. After the completion of a new association by the secondary interface, interface switches from the secondary interface to the primary interface which is triggered. As a result, the formerly secondary interface becomes primary for data transmission and the formerly primary interface is used for channel scanning. Consequently, MultiScan achieves a make-before-break hand-off by using multiple radio interfaces.
Singh et al (2005) had done GPS-assisted low latency hand-off scheme for 802.11 based wireless networks. In their work, the coordinates of the neighbouring APs are stored for assisting the mobile device during hand-off process. Their approach is extravagant on power issues as MCs are GPS-equipped and consume more power.

Mishra et al (2004) proposed a technique called neighbour graph. This technique dynamically captures pre-positioning station state. Every station is very well suited for low mobility conditions. Hence, neighbour graph technique is not recommended for high mobility and for ping-pong type of users.

Mishra et al (2003) had done an empirical analysis of 802.11 MAC layer hand-off process and observed that the hand-off latency is significant enough to affect the QoS in real-time applications. They concluded that there are variations in hand-off latency because of AP and mobile devices that are produced from different manufacturers.

Mobile IP version 4 (MIPv4) had been developed with IETF standard for mobility management (Perkins et al 2000, Perkins and Johnson 2002). MIPv4 provides a solution to mobility management for IPv4 networks. A router in the Mobile Node's home subnet is called the Home Agent (HA). This routes the packets addressed to the Mobile Node's home network address through a tunnel to the Mobile Node's local IP address in the foreign subnet, called the Care of Address (CoA), when the Mobile Node is away from home. The Mobile Node's care of address is maintained by the last hop router in the foreign subnet, called the Foreign Agent (FA). The FA decapsulates packets tunneled by the HA to the CoA. This delivers them to the link level address of the Mobile Node on the foreign subnet.
Due to tunneling, the packets heading for the mobile devices are forwarded from the home network to the foreign network. The high latency, may adversely affect application QoS during hand-off. As defined by the Mobile IPv4 specification, movement detection and CoA change depend completely on information available from the network layer. The algorithm does not have any provision for maintaining IP service while the mobile node no longer has a connection to its old subnet but before the routing has changed to the new subnet. As a consequence, packets sent to the old subnet are dropped until the Home Agent is updated with the new routing information. Packet loss of this nature is a particular problem for real-time services, such as voice over IP, video, or interactive games, as the gaps in packet delivery covering eighty milliseconds or more are noticeable by user. Non-real time services utilizing Transmission Control Protocol (TCP) can also be disrupted if the number of dropped packets is significant.

A technique to reduce packet loss is to buffer traffic while the link switch and routing changes are underway. Proposals have been made to buffer at the AP (Balakrishnan et al 1995 and Caceres and Padmanabhan 1998) and at the Foreign Agent (Perkins et al 1996). The buffering algorithms do not directly address the source of handover latency in a standard Mobile IP handover, rather are attempting to mitigate it. If TCP traffic results have been good, buffering introduces artifacts into streaming media and real-time traffic. Excessively large buffers of streaming media or real-time traffic can cause "ringing" on slow links after the handover. The buffering agent is required to buffer again while the original buffer is downloaded and then buffered again, etc. until the effect of the handover is dissipated. However, a limited amount of buffering may be useful in conjunction with link synchronous approaches for packet drops that cannot be eliminated.
The concept of spatial locality is crucial for the environment in hospitals, offices and schools. Such environments always utilize the same APs over and over hence not requiring their continuous discovery. In such environments the mobile user deals with the same subnets and more importantly the number of layer-3 hand-off required is very much lower than the number of layer-2 hand-off. Identifying the layer-3 hand-off process is the detection of subnet change and the new Care of Address (CoA) acquisition time via Dynamic Host Configuration Protocol (DHCP). According to the spatial locality principle and its other considerations, enhanced version of the cache mechanism is considered (Shin et al 2004). This allows having a layer-2 assisted layer-3 hand-off to reduce the latency time in MIP hand-off in layer-3.

The proposed Enhanced Active Channel Scanning (EACS) protocol supports the link layer synchronization approach for MIPv4 hand-off. In the link synchronous approach, information on the progress of switching the link is used to drive handover at the IP level. The link synchronization reduces the packet losses substantially for real-time communications.

1.5.2 Call Admission Control and Resource Sharing

The QoS management on hotspots environment becomes vital. Many emerging applications such as mobile information access, real-time multimedia communications, networked games, immersion world and cooperative works require a minimum level of QoS (Qiang et al 2004, Lin and Gerla 1989, Lindgren et al 2001 and Mangold et al 2002). Some of the recent works on CAC algorithms have been proposed in wireless/mobile networks using the adaptive framework (Bharghavan et al 1998). They proposed an overall framework including CAC seeking to achieve optimal revenue over the whole network. However, the message overhead ensuring optimal
bandwidth adaptation is inherently high. It also assumes continuous values of bandwidth in the adaptive framework.

Yee et al (2007) proposed a model for conservative approach for call admission. In this model the bandwidth was conserved even though there are less number of calls. In this proposed model the optimal utilization of bandwidth is not possible. The statistical reference model samples the number of calls and their respective profile.

Prihandoko et al (2003) and Malla et al (2003) proposed partition-based approach for bandwidth splitting in a cell. Bandwidth is segmented into three parts namely, available, reserved and allocated bandwidth. In their approach, if the available bandwidth does not satisfy the new calls, then they are blocked irrespective of their priority. The QoS bandwidth reallocation algorithm is invoked to degrade the bandwidth of the ongoing connections by pre-determined bandwidth degradation unit, which follows the connection priority. They reported that smaller degradation unit yields better performance in terms of call blocking and call dropping. This work is less scalable because of fixed reservation mechanism, failing to adjust the reserved bandwidth due to increase in the traffic rate.

Zhao and Fan (2003) presented a class-based analytical model for multiple QoS supports in IEEE 802.11. In their approach, QoS guarantee is formulated as an optimization problem with constraints on bandwidth requirements of each class. However, this approach consumes more time in computation, especially when the number of active stations is large.

Kwon et al (2003) proposed a distributed call admission control algorithm that guarantees the upper bound of the cell overloaded probability.
Only a single class of the adaptive multimedia networking was investigated. Multiple classes of various adaptive multimedia applications are not supported in their work.

Hong and Rappaport (1986), Guerin (1988), Chang et al (1994) and Ramjee et al (1997, 1998) proposed the well-known guard channel (cutoff priority) scheme and its variations to give high priority to hand-off connections over new connections by reserving a number of channels called guard channels for hand-off call connections. All these schemes are static in the sense that the number of guard channels is determined by a prior knowledge of the traffic patterns. This makes it unable to cope with network dynamics.

The ongoing call dropping probability and new call blocking probability are considered as two major challenging factors in wireless networks (Wang 2004). Most of the work concentrates towards minimizing the probability of call dropping and call blocking (Kwon et al 1998, Li et al 2004, Ahmed et al 2004, Zhai et al 2004) without considering the priority of the incoming calls. Yee et al (2007) have proposed a model for conservative approach for call admission, in which the bandwidth is conserved even though there is less number of calls.

Aljadhar and Znati (2000) proposed a bandwidth adaptation scheme using user-perceived QoS satisfaction. In this scheme, ongoing calls with high priority are allocated with the maximum bandwidth but calls with low priority are allocated with only the minimum level of bandwidth. Campbell and Hutchison (2002) formally defined the concepts of flow and flow management. They proposed a QoS architecture that is a layered architecture of services and mechanisms for QoS management and control of continuous media flow in multi-service networks. They argued that meeting QoS
guarantees in distributed multimedia systems is fundamentally an application-to-application (end-to-end) issue. The architectures that they reviewed focus on QoS in the context of individual layers of a network protocol. QoS management currently resides primarily in the policies and mechanisms to route packets of data (Chaskar 2001).

Chen et al (2004) proposed an important bandwidth reservation scheme, allocating bandwidth to the cell at which the MC is going to enter, primarily based on the decision of whether or not the call be supported by that cell. During this process, the call may have to gain a channel in the new cell to continue its service; otherwise the call must be dropped. From the user’s perspective, however, dropping an ongoing call is far more undesirable than blocking a new call.

Singh (1996) proposed a graceful degradation mechanism to increase bandwidth utilization by adaptively adjusting bandwidth allocation according to the user-specified loss profiles. Most multimedia applications service can be degraded in case of congestion as long as it is still within the pre-specified tolerable range. Chang and Chen (2004) attempted to address the scalability issue by employing a fuzzy inference mechanism with timing based reservation strategy. Better performance was achieved with such fuzzy approach at the cost of processing overheads at the base station.

1.6 RESEARCH OBJECTIVE

This research attempt to examine the above challenges and requirements based on various QoS parameters. The parameters considered are latency minimization, throughput maximization and effective resource utilization. It also provides solutions to these problems based on various techniques:
To support mobile users, it is critical to provide a seamless hand-off capability. The total hand-off latency includes the data link layer (layer-2) and the IP layer (layer-3) and should meet the strict time constraints of interactive multimedia applications. 802.11 supports MCs within an 802.11 Extended Service Set (ESS) to roam among multiple APs. This roaming capability is achieved through MCs’ beacon scanning in a channel sweep. When an MC enters a new basic service area, it first scans across all channels, remaining on each channel for a specified period of time to detect the signal radiated from the AP, and then acquires the channels from the AP.

However, an AP with large coverage area does not necessarily guarantee high-speed connectivity. Due to the unavoidable channel contention, throughput may degrade when an AP is overcrowded. Therefore, it is more desirable to have overlapped small coverage area with fast hand-off mechanisms.

The proposed pre-hand-off initiation algorithm is used to achieve proactive hand-off process. The layer-2 hand-off latency is minimized with the proposed enhanced active channel scanning protocol. The proposed route optimization technique eliminates the triangular routing problem in layer-3 (Mobile IPv4). The network efficiency and utilization have been improved.
The proposed RMIP is reliable for real-time communications and providing high throughput with minimized latency time.

1.6.2 Dynamic Call Admission Control

Basically, service differentiation is achieved by priority and fair scheduling. The proposed framework along with the DCAC algorithm provides efficient utilization of the network resources. It controls new call blocking and hand-off call dropping based upon the service differentiation. In the dynamic call admission control portion of the proposed framework, a new mechanism is introduced to achieve the above objectives by allocating/reallocating the bandwidth of connections based on the priority and Tspec. A high priority has been assigned to hand-off calls and real-time calls. The proposed dynamic bandwidth partition method accepts 25% more calls compared to existing conventional bandwidth partition methods. The method provides better QoS for high priority calls, and also prevents low priority calls from starvation.

1.6.3 Dynamic Resource Sharing

The proposed DRM framework estimates the capacity of all the APs in the campus wide network for load sharing process. A capacity-based load sharing among wireless base stations improves the overall efficiency of bandwidth utilization. The proposed distributed load sharing mechanism is incorporated in the AP. The MC receives capacity information of the neighbouring APs and invokes a pre-emptive hand-off to an AP with larger capacity. The performance benefits of the capacity based load sharing mechanisms are presented in terms of the traffic drop ratio. In the existing RSS-based load sharing, the traffic drop ratios are 30% more than the proposed capacity-based load sharing method.
1.7 ORGANIZATION OF THE THESIS

The thesis is organized as follows: Chapter 2 discusses the problem related to conventional layer-2 channel scanning process and the existing solutions. The enhanced active channel scanning protocol is implemented in the firmware for handling layer-2 hand-off process. The performance of EACS protocol is investigated and the results are analyzed.

Chapter 3 introduces a unique model for call admission control. The existing static resource allocation method is compared with the proposed dynamic resource allocation method. The priority-based resources sharing mechanism reduces the hand-off call and real-time call dropping. The performance of the DCAC model is investigated for high priority calls and the results are analyzed.

Chapter 4 describes DRM framework model based on capacity of the APs. The dynamic resource-sharing algorithm provides required resources for the MCs when they are in the overlaid region. The model is compared with the existing RSS-based resource sharing mechanism and capacity-based resource-sharing mechanism. Analyses of the results from analytical and experimental models are discussed.

Chapter 5 presents a route optimization technique for mobile IP hand-off. The various IP hand-off mechanisms are investigated and compared with the proposed route optimization mechanism. The simulation results depict the effect caused by triangular routing in mobile IP during hand-off process and the proposed route optimization technique.

Chapter 6 provides the conclusion of the thesis and the future prospects of research in wireless/mobile environment.