Mobile Wireless System Interworking with 3G and Packet Aggregation for Wireless LAN

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submitted to the Faculty of the

Virginia Polytechnic Institute and State University

in partial fulfillment of the requirements for the degree of

Doctor of Philosophy

in

Computer Engineering

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April 22, 2004

Blacksburg, Virginia

Keywords: WLAN, Aggregation, 3G, Protocol

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(ABSTRACT)

This research considered the efficient transmission of data within a wireless local area network (WLAN) system. A simulation model was developed to study the performance of our protocol, AGG-MAC (aggregated medium access control). AGG-MAC is a simple and elegant medium access control (MAC) protocol designed to improve performance by transmitting a maximal quantity of data with minimal overhead. Our enhancement to IEEE 802.11, AGG-MAC yields dramatic improvements in both local and global throughput. It furthermore reduces jitter in support of real time communications requirements such as voice over IP (VoIP). In support of heterogeneous roaming between Third Generation (3G) Wideband CDMA (WCDMA), specifically Universal Mobile Telecommunications System (UMTS) and WLAN systems, we constructed a simulation environment which allowed the evaluation of AGG-MAC in such a system. We further demonstrated the suitability of AGG-MAC throughout a range of infrastructure and ad hoc based WLAN scenarios. The AGG-MAC protocol enhancement provides significant performance improvements across a range of wireless applications, while interoperating with standard IEEE 802.11 stations. Performance is commensurate to original WLAN MAC performance for applications that do not benefit from packet level aggregation.

The key contributions of this research were two-fold. First was the development of an OPNET^{TM} simulation environment suitable for evaluation of future protocols supporting tightly coupled, heterogeneous WLAN and 3G systems. Secondly was the implementation and testing of the AGG-MAC protocol which aggregates suboptimal size packets together into a single frame, thereby amortizing the overhead.

Dedication

This work is dedicated to my wife, Valerie and my two children, Brittany and David. The sacrifices you all have made during the past three years in support of this work are greatly appreciated. You made it possible for me to complete this arduous task and there is no way I would ever be able to have done it without you. Your understanding, love, and support brought me through the challenges and the long nights and provided the support necessary to finish this project and maintain what was left of my sanity. I love you all, always and forever. Thanks for being there for me, now let me be there for you.

Acknowledgments

I would like to acknowledge the many people without whom I could not have completed this dissertation. First, I thank my advisor, Dr. Nathaniel J. Davis, IV. His guidance throughout the research and editorial help throughout the writing were invaluable. I would like to thank my committee members Scott Midkiff, Tom Hou, Brian Woerner, and Srinidhi Varadarajan for their support of this research. I would especially like to thank Tracy Mann for his superb effort in the development of the initial models, and to Grant Jacoby, Tracy Mann, Mike Brownfield and the scores of others for providing a sounding board for the many varied ideas and approaches and for the occasional loaned simulation platform. I would like to thank the United States Army and the United States Military Academy for their funding support and for allowing me this opportunity. Finally, I would like to thank my wife, Valerie, and children, Brittany and David for their love and support throughout three long years of researching and writing.

Contents

List of Figures

List of Tables

Chapter 1

Introduction

The continued growth of wireless communications networks and their increased ubiquity have transformed our society. The exponential growth of the Internet and the proliferation of cellular mobile systems and WLAN systems throughout both home and business applications generated both competition and cooperation among the different systems. Academic researchers and service providers have both sought ways to integrate the WLAN systems with mobile cellular systems. The desire was to gain the increased capacity and indoor service quality provided by WLAN working together with the mobility provided by cellular systems. This research effort focused first on interworking high capacity WLAN "hot spots" with the emerging 3G, specifically Universal Mobile Telecommunications System (UMTS), cellular networks and subsequently on a WLAN MAC protocol enhancement to improve performance for small packets.

This chapter defines the problem investigated by this research. The remainder of this chapter is organized as follows. Section 1.1 states the research problem investigated. A brief background is provided in Section 1.2 with motivation for this work provided in Section 1.3. Section 1.4 presents the research questions addressed by this work and a Section 1.5 describes the intended purpose of this research. Finally, Section 1.6 summarizes the significant results.

1.1 Problem Statement

This research was two-fold in its focus and efforts. The initial phase of work focused on the design, implementation and testing of a tightly coupled network level system designed to provide WLAN "hot spots" interworked with 3G cellular mobile communications system, specifically UMTS networks. The resultant network level simulation tools provided a foundation for future research into the issues and trade-offs associated with protocol design and parameterization in this type of system. The second phase of this work was the design of a AGG-MAC, a WLAN MAC layer, protocol designed to significantly improve throughput and reduce latency for data traffic consisting of packets of less than maximal size. The identification of the bottleneck associated with the transmission of small frame sizes and its severity was identified as the result of UMTS-WLAN interworked system simulations. The utility of AGG-MAC is not restricted to interworked UMTS-WLAN systems or even to infrastructure based systems, but AGG-MAC benefits quite a wide range of WLAN configurations.

The primary contributions of this research were two-fold. The first contribution was the design and implementation in OPNET^{TM} of a network level simulation tool to support future research in tightly coupled interworked system consisting of WLAN "hot spots" and UMTS cellular systems. The second significant contribution was the design, implementation and testing of AGG-MAC, a WLAN MAC layer protocol enhancement designed to improve performance for a broad range of wireless LAN configurations and traffic patterns.

1.2 Background

Cellular telephone systems were originally analog systems designed to provide voice communications to the outdoor, traveling user. Cellular systems have evolved significantly based on technological advancements. The coverage and the quality of the service have both increased dramatically, but as it was not within the original scope of the design of cellular networks, indoor coverage and data capacity are still significantly limited. WLAN systems were designed for indoor, data traffic and have demonstrated their ability support the needs of limited mobility indoor clients. For these reasons, many supported the eventual convergence of the two communications networks to provide better service and improved coverage for their users [6]. The standards developing bodies attempted to define standards for the interoperation of the two systems [7] and several researchers sought to determine the best methods to interwork the two systems.

Capacity of a WLAN system is dependent on many factors. One factor, which became increasingly apparent during our early research, was the overhead associated with small packet sizes. Under traffic conditions commonly found in current cellular data traffic, we observed that the throughput on an $11 Mbps$ WLAN with 35 users was significantly less than expected. The per frame overhead in the IEEE 802.11 WLAN standard [3] significantly limits capacity for all users on the WLAN network for collections of small data packets. Much work has been done to improve WLAN performance by selecting an appropriate size for a data frame based upon limiting the largest allowable frame size, but little work has been done to improve performance for smaller data objects. Much of the conducted WLAN research avoided this issue by selecting a fixed (relatively large) packet size for all data on the WLAN network. We sought to address the issue of overhead for small frame sizes and proposed a solution that improved performance and was applicable for a wide range of network situations.

1.3 Motivation

The first phase of this work was motivated by the need to have a network level simulation tool capable of testing, evaluating and refining protocols and standards for the interworking of tightly integrated UMTS and WLAN systems. The potential benefits of a tightly integrated system are not yet fully understood. While some believed that using current cellular authentication, authorization and accounting (AAA), roaming, security and other subsystems may yield significant benefit, it was not yet clear and warranted further study. The trade-offs and best design decisions associated with an interworked system were not yet known.

The 3rd Generation Partnership Project (3GPP) was one of the primary sources of standardization efforts for 3G cellular systems. Much of their work focused on the convergence of voice and data communications. WLAN is capable of providing significantly higher capacity for indoor environments albeit of limited range. For this reason, the 3GPP Services and Systems Aspects (SA1) working group published a feasibility study and further sought to develop standards for the interworking of 3GPP systems with WLAN systems [7]. Their study identified six scenarios ranging in amount of interoperability between the two systems. These scenarios are presented in Section 3.1.1. We sought to provide a robust network level simulation environment capable of supporting future research on UMTS-WLAN interworked systems.

The second phase of this research was motivated by the desire to improve the realizable capacity over a WLAN channel for commonly occurring traffic patterns. While it is possible that some equipment vendors are researching aggregation, it is equally likely that commercial interests will prevent the timely publication of their results. This research was conducted after simulation of a heterogeneous wireless system consisting of hosts accessing via either UMTS cellular system or WLAN system indicated that performance over an $11Mbps$ WLAN was less than expected. Upon observing that the fixed overhead associated with the WLAN protocols was the primary contributor to the lower performance, we sought to reduce the per packet overhead. We further realized that, for our target system, all traffic from a client station was routed through the Access Point (AP). Because the design was for a public access "hot spot," the probability of traffic among stations on the WLAN was near zero. In such systems, all of the traffic was routed directly from the client station (STA) to the AP, and that in such a case, the routing problem is trivial. Based on these criterion, we designed the AGG-MAC protocol and found that its benefits went far beyond our original, limited scenario.

1.4 Research Questions

This research addressed the following questions regarding wireless network performance. In each case, the performance metrics were either throughput, end-to-end delay or they were compared to previously designed metrics for an existing network level simulation. We compared our enhancements to original, baseline system performance where possible.

1. What are the benefits associated with a tightly-coupled UMTS-WLAN interworked system?
2. How does a tightly-coupled, interworked UMTS-WLAN system perform?
3. What is the design for an aggregated WLAN MAC (AGG-MAC) protocol?
4. How does AGG-MAC performance compare with original WLAN MAC protocol for a single deterministic source with point-to-point, WLAN communications pair?
5. How does AGG-MAC performance compare with original WLAN MAC protocol across a wide range of traffic loads and higher layer protocols?
6. Can AGG-MAC improve WLAN performance when carrying traffic loads designed for UMTS systems?
7. What are some potentially useful system configurations for incorporating AGG-MAC?

1.5 Purpose

The purpose of this research was two-fold. We first defined, implemented and tested a UMTS-WLAN network level simulation tool which could be used for testing future potential standards and protocols. The second phase of our work was to develop an extension to the existing WLAN protocol set that limits protocol overhead and improves performance while providing support for quality of service (QoS) dependent applications. The original impetus of this research was to design a mechanism that ensured that performance was not reduced when wireless systems operated with traffic patterns optimized for UMTS channels via the much higher capacity, albeit shared, channel of a WLAN.

1.6 Summary

This research effort focused on tools to evaluate and test wireless communications networks designed to support the convergence of two complementary access layers. The specific problem being addressed was two-fold. First, we created a simulation tools set for UMTS-WLAN interworked network systems, and finally, we designed, implemented and tested AGG-MAC, a WLAN protocol enhancement designed to improve performance for commonly occurring traffic patterns in which the frame size is suboptimal. The research questions presented in Section 1.4 are answered by the design of simulations which can be analytically confirmed and through comparison of simulation results with analytical results. For cases of more complex, stochastic simulations, a comparison with existing work from other researchers or from performance of the un-enhanced system simulation was performed.

This chapter defined the research problems addressed in the remainder of the document. The rest of the document is organized as follows. Chapter 2 presents the theoretical background. Chapter 3 presents a review of relevant literature for both UMTS-WLAN interworking and packet level aggregation in the context of WLAN. Chapter 4 describes the methodology while Chapter 5 presents results specific to the UMTS-WLAN interworking simulation tools. Chapter 6 describes the design and methodology used to develop and test the WLAN packet aggregation protocol enhancement and provides an overview of the simulation design with results specific to AGG-MAC. Results for AGG-MAC testing are provided in Chapter 7. Finally, Chapter 8 discusses the results and their significance in the form of a conclusion.

Chapter 2

Background

This chapter provides the theoretical background for this work in the area of wireless networks, aggregation, and traffic patterns and sources.

2.1 Wireless Networks

This section provides the theoretical background in both the area of Wireless Local Area Networks (WLANs) and in data networks in Third Generation (3G) wireless networks. A basic theoretical background in 3G wireless networks and the WLAN family of technologies is required to address the topic of this research effort. Section 2.1.1 provides an introduction to 3G mobile communications and concludes with an overview of the evolution in the mobile communications. Section 2.2 provides an introduction to the Universal Mobile Telecommunications System (UMTS), which includes a detailed description of the UMTS 3GPP standards, specifically in the UMTS packet switch (PS) domain. The UMTS-WLAN interworking research was conducted in conjunction with masters student Tracy Mann. Most of the UMTS-WLAN simulation work was presented in [8]. Section 2.3 presents an introduction to the WLAN family of technologies that includes a detailed discussion of the IEEE 802.11

standards, the WLAN system architecture, and protocol architecture, focusing on the WLAN medium access control (MAC) layer. Current related work in the areas of UMTS-WLAN interworking and in WLAN MAC protocol enhancements are presented later in Chapter 3.

2.1.1 Cellular Communications Systems

3G wireless refers to the developing technology standards for the next generation of mobile communications systems. One of the main goals of the standardization efforts of 3G is to create a universal infrastructure that is able to support existing and future services. This requires that the infrastructure be designed so that it can evolve as technology changes, without compromising the existing services on the existing networks. Separation of access technology, transport technology, service technology and user applications from each other make this demanding requirement possible [9].

Mobile Communications Technology Evolution

Mobile communications technology evolution is typically classified in terms of generations of the technology. First Generation (1G) is the term used for the analog mobile systems that were deployed in the mid 1980s. Second Generation (2G) cellular systems refer to the systems deployed in the early 1990s that utilized digital communications techniques like TDMA and CDMA. The 1G and 2G cellular systems were initially designed to provide circuit-switched voice service, and were later modified to provide low bit rate data service. Current Second Generation (2G) cellular systems are primarily voice systems, with data rates up to $19.2kbps$ [10]. These include Global System for Mobile Communications (GSM), IS-136 (TDMA) and IS-95 (CDMA). Efforts aimed at increasing the data rates provided by 2G systems, typically through the use of higher order modulation, are termed 2.5G systems. Emerging 3G cellular standards will replace the 2G radio interface to provide higher data rates and provide inherent support for packet data, while leaving the 2G core network basically unchanged. GSM's General Packet Radio Service (GPRS) is the leading packet data network proposal for Third Generation (3G) wireless systems. The goal of 3G systems is to offer data rates up to $2Mbps$ to support wireless packet data. The focus of Fourth Generation (4G) systems is on future modifications to the core network, possibly moving to an IP-based infrastructure. 4G system standards have not yet been defined at this time.

First Generation (1G)

During the early 1980s, analog cellular telephone systems were deployed. At that time, each country developed its own system, limiting usage to within country boundaries [6]. In North America, the Advanced Mobile Phone System (AMPS) was the first commercial cellular system to be deployed. It was a frequency modulated (FM) analog mobile radio system using frequency division multiple access (FDMA) with $30kHz$ channels occupying the $824MHz - 894MHz$ frequency band [10]. Analog cellular systems were limited in both system capacity, and in support for data traffic. In 1993, Cellular Digital Packet Data (CDPD) added support for data traffic in the AMPS frequency band. CDPD was a packet switched network that used a $30kHz$ AMPS channel to provide mobile access to packet data networks at a rate of $19.2kbps$ [10]. Mobile data users were able to transmit on idle AMPS voice channels, but were required to hop to a different channel when a voice user began to transmit. CDPD was termed a packet overlay network since it utilized the existing base station equipment of cellular network, and required no additional frequency spectrum. The core network of CDPD consisted of packet switches, separate from the AMPS Mobile Switching Center (MSC), which provided connectivity to public data networks. CDPD was also utilized as a packet overlay network for 2G digital voice systems.

Second Generation (2G)

The digital cellular systems deployed in the early 1990s are called 2G wireless systems. 2G systems use digital modulation schemes, as well as digital multiple access schemes such as time division multiple access (TDMA) and code division multiple access (CDMA). The three primary 2G systems are TDMA (IS-136), CDMA (IS-95), and GSM.

The United States Digital Cellular (USDC/IS-54) system was deployed in North America in 1991. It had a digital TDMA data channel, but utilized the AMPS analog control channel. In 1993, IS-136 added a digital control channel for a completely digital system. These TDMA 2G systems operated in the AMPS frequency band of $824MHz - 894MHz$ [10]. CDMA (IS-95) systems using Direct Sequence Spread Spectrum (DSSS) were deployed in the United States in 1993. They also operated in the AMPS frequency band. Additional spectrum was added in the 1850-1990 MHz frequency band to support CDMA carriers. This spectrum is commonly called Personal Communications Services (PCS) [11].

GSM was introduced in Europe in 1990. The goal of GSM was to replace the various European 1G analog systems with a single digital system. GSM operates in the $890MHz - 960MHz$ frequency band, and a PCS version termed Digital Cellular System (DCS) 1800 operated in the $1.8GHz - 2.0GHz$ band [10]. GSM used slow frequency hopping, and a $200kHz$ carrier bandwidth. Its frame structure supported eight users per carrier at a maximum data rate of $24.7kbps$ [10]. GSM is the most widely used 2G standard, accounting for about 66 percent of the world market [6]. Standards were developed to provide both data service, and increase the data rate in GSM networks. General Packet Radio Service (GPRS) was a packet overlay network designed to provide data services in a GSM network. GPRS utilized the same frame structure as GSM, and supported a maximum data rate of $21.4kbps$ [12]. The GPRS infrastructure added two new nodes to the GSM network, the Gateway GPRS Support Node (GGSN) and the Serving GPRS Support Node (SGSN). The GGSN served as a gateway to external data networks and tunneled data traffic to the appropriate SGSN. The SGSN was responsible for delivering data packets to the mobile user. The 3G GPRS architecture is discussed later in Section 2.2.2 . Enhanced Data rates for GSM Evolution (EDGE) was an ongoing effort to increase the data rate of GSM. EDGE will provide data rates up to $384kbps$ over the basic GSM $200kHz$ channel by using higher order modulation schemes. Both GPRS and EDGE brought the capabilities of GSM closer to the stated goals of 3G, and they were often referred to as 2.5G standards. In order to be commercially feasible, 3G systems must exceed the capabilities of these enhanced 2G systems.

Third Generation (3G)

3G was a term coined by the global cellular community to indicate the next generation of mobile service capabilities, including higher capacity and enhanced network functionalities. The goal of 3G wireless systems was to provide wireless data service with data rates of $144kbps$ to $384kbps$ in wide coverage areas, and $2Mbps$ in local coverage areas [13][14]. Possible applications included wireless web-based access, E-mail, as well as video teleconferencing and multimedia services consisting of mixed voice and data streams. 3G wireless standards were being designed with the intent to support a variety of services and applications.

IMT-2000 (International Mobile Telecommunications-2000) is the International Telecommunication Union (ITU) globally coordinated definition of 3G wireless systems covering key issues such as frequency spectrum use and technical standards. The ITU IMT-2000 standards are being developed with input from regional standards organizations, including the European Telecommunications Standards Institute (ETSI) in Europe, the Telecommunications Industry Association (TIA) in the United States, the Association of Radio Industry and Business (ARIB) in Japan, and the Telecommunications Technology Association (TTA) in Korea. These regional organizations together have proposed IMT-2000 standards that take into account regional concerns such as evolution from existing 2G systems, and current allocation of frequency spectrum [14].

Two international bodies were established to resolve the differences between these regional proposals. The Third Generation Partnership Proposal (3GPP) worked to standardize the proposals from ETSI, ARIB, and TTA into a single standard termed WCDMA (Wideband Code Division Multiple Access). The European proposal for IMT-2000 was UMTS (Universal Mobile Telecommunications System). UMTS was an evolutionary system based upon the currently fielded GSM. UMTS packet data services (PS) are based on the 2.5G GSM General Packet Radio Service (GPRS) framework. The 3GPP2 was likewise standardizing the proposals from TIA and TTA into a single standard called CDMA2000. The North American CDMA2000 proposal was strongly influenced both by compatibility issues with the current IS-95 CDMA systems, and existing frequency allocations.

2.2 UMTS

Universal Mobile Telecommunications System (UMTS) is a Third Generation (3G) wireless protocol that is part of the ITU's IMT-2000 vision of a global family of 3G mobile telecommunications systems. Its specifications were created by the 3GPP and the ETSI. It is an evolutionary system based upon GSM with its packet domain based on the 2.5G General Packet Radio Services (GPRS) framework. UMTS is expected to deliver low-cost, high-capacity mobile communications with data rates up to $2Mbps$. This research effort focused on the 3GPP Release 1999 standards.

The following sections provide an overview of UMTS and its signaling protocols. The intent of these sections is to provide the necessary background information on the UMTS signaling protocols to understand the simulation design. The GPRS Mobility Management and Session Management protocols that were used to realize the primary design goal of interworking WLAN with UMTS in accordance with the 3GPP scenario 2 are covered in detail in Sections 2.2.4 and 2.2.6.

The main system components of the UMTS network architecture are the user equipment (UE), the UMTS Terrestrial Radio Access Network (UTRAN) and the core network (CN) as shown in Figure 2.1.

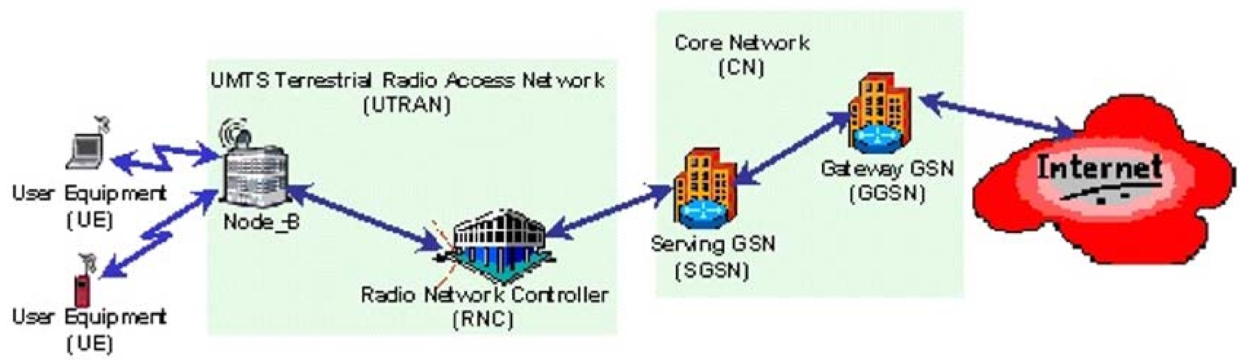

Figure 2.1: UMTS System Overview

The user equipment (UE) consists of the mobile terminal (MT), the terminal equipment (TE), and the Subscriber Identity Module (SIM). The UTRAN is comprised of the Node-B and the Radio Network Controller (RNC). The RNC is in charge of the overall control of logical resources provided by the Node-Bs. The RNC manages the air interface resources between its Node-Bs and their associated UEs. The Node-B provides logical resources, corresponding to the resources of one or more cells, to the RNC. It is responsible for the radio transmission and reception in the cells it controls. A Node-B can control several cells, managing the network air interface for its associated UEs. It is responsible for relaying packets between the UEs and its controlling RNC. The Node-B is also responsible for assisting the RNC with radio resource management through the Node-B Application Protocol (NBAP) signaling messages. The Serving GPRS support node (SGSN) keeps track of the location of individual UEs and performs security functions and access control. The Gateway GPRS support node encapsulates packets received from external packet networks (IP) and routes them to the SGSN.

2.2.1 UMTS Air Interface

3GPP TS 25.302 defines the access scheme for UMTS as Wideband Code Division Multiple Access (WCDMA) with a chip rate of $3.84Mcps$ and a $5MHz$ carrier for both the uplink and downlink. The standard supports both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) with FDD mode considered to be the main technology for UMTS. In the FDD mode, the uplink and downlink transmissions use different frequency bands. A radio frame has a length of $10ms$ and is divided into 15 slots. The spreading factors vary from 4 to 256 for the uplink and up to 512 for the downlink. Under limited coverage and mobility, data rates up to $2Mbps$ are achievable using these spreading factors. The TDD mode defined is Time Division-Synchronous Code Division Multiple Access (TD-SCDMA). It operates on a low-chip rate carrier, with a $1.6MHz$ carrier instead of a $5MHz$. It can also offer the end user data rates up to $2Mbps$ under optimal conditions [6].

2.2.2 GPRS – UMTS Packet Domain Architecture

3GPP TS 23.060 defines the packet domain architecture for both GSM and UMTS using Figure 2.2 [1]. The shaded elements highlight the key elements of the UMTS packet domain. The standard provides a specification for each of the key elements of the architecture, as well as a detailed specification for the interfaces that connect these key elements to ensure interoperability between multiple vendor implementations.

UMTS packet data services are based on the 2.5G GSM General Packet Radio Service (GPRS). GPRS uses packet switched technology to transfer high and low-speed data and signaling in an efficient manner. The packet overlay architecture maintains a strict separation between the radio subsystem and the network subsystem, allowing the reuse of other radio access technologies. GPRS optimizes the use of network and radio resources and defines the protocols for interworking with external IP networks (represented generically as a packet data network (PDN) on the reference diagram) through the G_i interface.

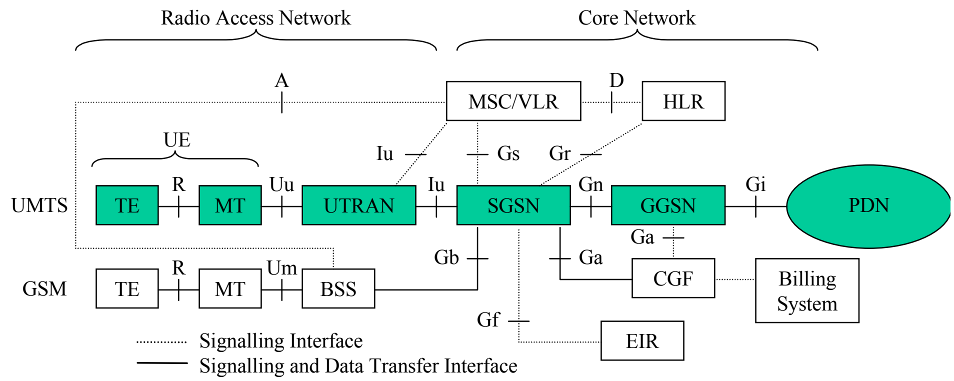

Figure 2.2: UMTS Packet Domain Architecture [1]

Both GSM and UMTS use a common packet domain Core Network (CN) to provide packet switched (PS) services. The packet domain is designed to support several quality of service levels to allow efficient data transfer of application traffic ranging from non-real-time, intermittent and bursty data to real-time voice and video. The Serving GPRS Support Node (SGSN) is the node that is serving the UE. It supports GPRS for UMTS via the I_u interface. The SGSN performs location management, security, and access control functions for the UEs. The GGSN provides interworking with external packet switched networks, and is connected with SGSNs via the ATM (Asynchronous Transfer Mode)-based interface, G_n. It contains routing information for PS-attached users. This routing information is used to tunnel data to the UEs current point of attachment (i.e., the SGSN).

The common circuit switched (CS) CN elements include the Mobile Switching Center/Visitor Location Register (MSC/VLR), the Home Location Register (HLR), the Charging Gateway Functionality (CGF), and the Equipment Identity Register (EIR). The MSC/VLR is used to provide efficient coordination of PS and CS services (i.e., combined GPRS and non-GPRS location updates). The HLR contains GSM and UMTS subscriber information. The CGF collects charging records from the SGSNs and GGSNs. The EIR stores information about user equipment identity.

In order to access the PS services, a UE must make its presence known to the network by performing a GPRS attach. This makes the UE available via the SGSN for notification of incoming PS data. In order to send and receive PS data, the UE must activate the Packet Data Protocol (PDP) context that it wants to use. This operation makes the UE known to its GGSN and to the external data networks through this gateway. User data is transferred transparently between the UE and the external data networks with a method known as encapsulation and tunneling. Data packets are equipped with PS-specific protocol information and transferred between the UE and the GGSN. This transparent transfer method enables easy introduction of additional interworking protocols in the future.

2.2.3 UMTS Protocol Stack (Control and User Plane)

3GPP TS 23.060 defines the layered protocol structure for both the user and control planes in Figure 2.3 and Figure 2.4, respectively [2]. The protocols of both the user and control planes can be grouped into one of three more general UMTS internetwork protocol layers: transport network layer, radio network layer, and the system network layer. The protocols within each of these network layers extend across multiple interfaces and work together in the execution of many common system-wide functions [9].

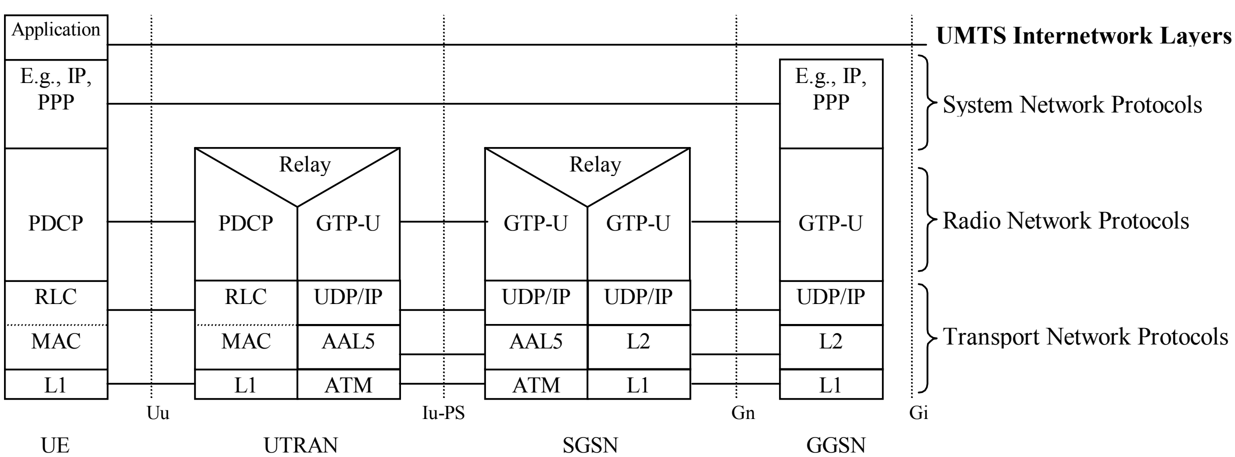

Figure 2.3: UMTS User Plane [2]

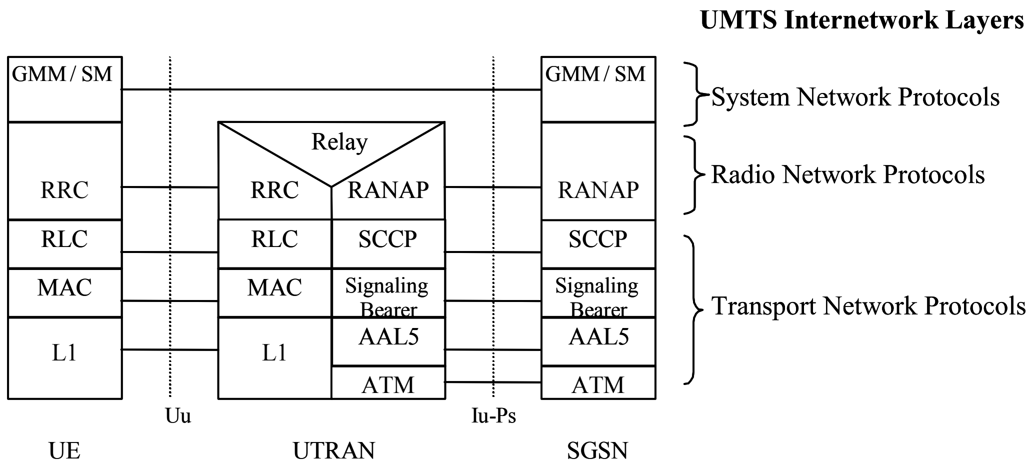

Figure 2.4: UMTS Control Plane [2]

Transport Network Protocols

The UMTS transport network protocols are responsible for providing a general-purpose transport service to both the user and control planes. It is actually a network within the UMTS network. The logical end-to-end transport network is composed of: the Medium Access Control (MAC), the Reliable Link Control (RLC), the physical layer (PHY), as well as the Asynchronous Transfer Mode (ATM) and ATM Adaptation Layer 5 (AAL5) [9].

The PHY protocol, specified in 3G TS 25.302, controls the use of the WCDMA physical channels on the U_u interface. It is responsible for mapping the logical transport channels to the physical channels.

The MAC protocol, specified in 3G TS 25.321, controls the access signaling procedures (request and grant) for the radio channel. It provides its service as a set of logical channels, which are characterized by the type of data that they transport. The MAC has the overall responsibility of controlling the communications over WCDMA transport channels provided by the PHY layer. In order for multiple users to share the capacity of the transport channels, the MAC uses transport blocks as units of transmission. It multiplexes and de-multiplexes protocol data units (PDUs) from the RLC into these transport blocks that are delivered to and from the physical layer. In the user plane, the MAC handles the real-time protocol issues associated with priority handling between data flows of a single UE [9].

The RLC protocol, specified in 3G TS 25.322, provides logical link control over the radio interface in both the user and control planes. There may be several simultaneous RLC links per UE with each link being identified by it own radio bearer (RB) ID. The standard supports transfer of user data in three modes: transparent, unacknowledged, and acknowledged modes. The transparent mode supports segmentation and reassembly (SAR) of higher layer PDUs into/from smaller RLC payload units (PUs) and forwarding to/from the MAC. The unacknowledged mode supports SAR, concatenation, padding, forwarding to/from the MAC, as well as ciphering and sequence number checking. The acknowledged mode supports all of the functionality of the unacknowledged mode plus error correction, in sequence delivery of higher-layer PDUs, duplicate detection, and flow control [1][9].

The ATM protocol, specified in [15], divides the information from the higher layer into fixed-size cells (53 octets), multiplexes, and forwards the cells to the next node. The AAL5 protocol, specified in [16], provides support for variable bit rate connection-oriented or connectionless data services to the ATM protocol.

Radio Network Protocols

The radio network protocols form the layer on top of the general-purpose transport network protocols discussed in the previous section. These protocols are needed to control the establishment, maintenance and release of radio access bearers (RABs) and to transfer user data between the UE and the CN using the RABs. The RAB represents a logical channel between the SGSN and the UE. The protocols of the radio network extend from the UE across the UTRAN to the SGSN and create this logical channel association. The radio network protocols in the control plane are the Radio Access Network Application

Part (RANAP) and the Radio Resources Control (RRC) protocols. See Figure 2.5 for the relationship between the RRC, RANAP, signaling connection and RAB.

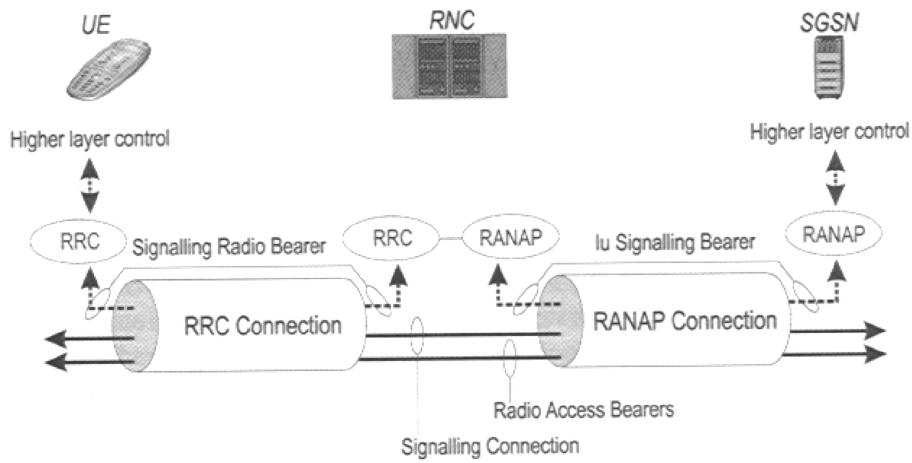

Figure 2.5: Relationship Between the RRC, RANAP, Signaling Connection and RAB

The RANAP protocol, specified in 3G TS 25.413, controls the resources in the I_u interface, between the SGSN and the RNC. It has the main responsibility for overall RAB management, providing the SGSN with the control mechanisms for the establishment, modification and release of RABs between the UE.

The RRC protocol, specified in 3G TS 25.331, is the key radio resource control protocol in the UTRAN. It supports the Radio Resource Management (RRM) functionality by coordinating the execution of resource control requests. RRM is covered in greater detail in Section 2.2.5 . It also has the responsibility to transport RANAP messages as their payload between the RNC and UEs. The major function of the RRC is to control the radio bearers, transport channels and physical channels. The RRC provides a logical connection between the UE and the RNC.

As stated earlier, the RAB represents the logical channel connection between a UE and the SGSN. It is the RNCs responsibility to provide the RAB connection, creating an illusion for the SGSN of a fixed radio bearer (connection) to the UE. The RNC analyzes the RAB requests, evaluates radio resources needed to support the requests, reconfigures the radio channels, called radio bearers (RB) and maps the requested RAB to the RBs. As shown in Figure 2.5, the RAB is carried within the RRC connection between the RNC and UE over the radio interface and within the RANAP connection between the RNC and the SGSN. In this association, the RNC acts as a protocol converter between the UTRAN and the SGSN [17].

In the user plane, the radio network protocols are: the Packet Data Convergence Protocol (PDCP) and the GPRS Tunneling Protocol-User (GTP-U). The PDCP layer handles transmission and reception of PDUs using the services provided by the RLC. Its main functionality is to compress and decompress the headers of the higher layer protocol data units in order to save valuable radio link resources. The most common user to user packet data protocol is TCP/IP. The IP header compression specified for the 3GPP is RFC2507 defined by the Internet Engineering Task Force (IETF) [9].

The GTP-Us, specified by 3G TS 29.060, main functions are data packet transfer between the UTRAN and SGSN, as well as between the SGSN and GGSN. On both of these interfaces the GTP-U operates on top of UDP (User Datagram Protocol). It uses encapsulation and tunneling to provide a connectionless data transfer service to the higher layers. Since its main purpose is to transfer user data, it has been optimized for that specific task, leaving the signaling messages required to set up the GTP endpoints to the control plane.

System Network Protocols

The radio network protocols make it possible for the UE to communicate across the UTRAN subnetwork by maintaining the communications path between the UE and the SGSN, forming a radio access network. The system network protocols operate on top of the radio network protocols and allow the UMTS SGSN to provide communications services to the UE. This group of control plane protocols is carried transparently through the radio access network, allowing the radio access network to be interchanged seamlessly without affecting the communications services that the SGSN provides to the UEs. The main system network protocols in the UMTS control plane are the GPRS Mobility Management (GMM) and the Session Management (SM). The GMM protocol, which is covered in more detail in Section 2.2.5 , operates between the UE and the SGSN. The GMM protocol provides the basic signaling mechanisms for controlling mobility management and authentication functions for the UEs in the UMTS PS domain. The SM protocol, which is covered in more detail in Section 2.2.4 , is responsible for establishing and releasing packet data sessions, called packet data protocol (PDP) contexts with the UMTS network.

2.2.4 GPRS Mobility Management and Session Management

GPRS Mobility Management

As stated in Section 2.2.2 , in order for a UE to utilize the resources provided by the UMTS GPRS PS domain, it must first register into the GPRS network. The GMM protocol provides the signaling mechanisms for controlling this mobility management registration process, termed a GMM context. Figure 2.6 shows the Packet Mobility Management (PMM) States of the GMM context [1].

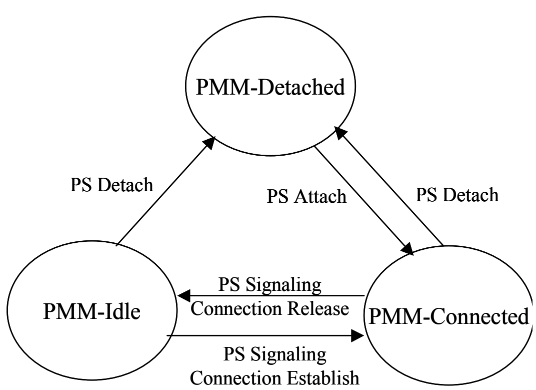

Figure 2.6: Packet Mobility Management (PMM) States [1]

The procedures for the establishment and release of the GMM context are: GPRS Attach and the GPRS Detach. The signaling mechanisms for realizing a GPRS Attach are covered in more detail in Section 2.2.6 . Prior to registering into the GPRS network, a UE is in the Packet Mobility Management Detached (PMM-Detached) state as shown in Figure 2.6. In that state, a GMM context does not exist and subsequently, the SGSN cannot reach the UE because it does not have any valid location or routing information about the UE. After successful completion of a GPRS Attach procedure, the UE will be in the PMM-Connected state, is authorized into the network, and has established a PS signaling channel with the SGSN. As shown in Figure 2.4, the signaling connection consists of an RRC connection between the UE and the UTRAN and RANAP connection between the UTRAN and the SGSN. If the PS signaling connection is released, the UE will move to the PMM-Idle state [2].

Session Management

In the PS domain the packet connections are called sessions—they are established and managed by the SM protocol. Its main function is to establish and release packet data sessions by providing support for Packet Data Protocol (PDP) context handling between the SGSN and the UE. The SM has two logical states: inactive and active, as depicted in Figure 2.7.

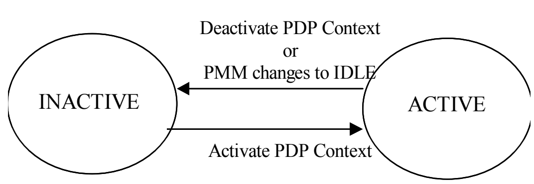

Figure 2.7: Session Management (SM) States

When the UE is in the Session Management Active (SM-Active) state, a PDP context exists and contains the necessary information for routing user data packet from the GGSN (i.e., the gateway) to the UE and vice versa. When a UE is in the SM-Inactive state, the PDP context does not exist and therefore, there is not any valid routing information for it. The basic SM procedures are: PDP context activation, modification, and deactivation. The signaling mechanisms for realizing a PDP context activation are covered in more detail in Section 2.2.6 .

The goal of the SM protocol is to create the illusion of an “always on” type of connection between the UE and the SGSN—this must be accomplished in an effective way in order to conserve network resources whenever possible. Section 2.2.5 discusses how the SM and RRC protocols interact to accomplish this.

Referring back to Figure 2.6, once in the PMM-Connected state, the UE is authenticated into the network and can request to transfer data using the UMTS PS domain. In order to exchange packet data with external data networks, the UE must be in the SM-Active state, meaning that the UE has a PDP address in the Packet Data Network (PDN). The UE accomplishes this by requesting a PDP context activation. The PDP context characterizes a data transfer session and includes information like: the PDP type (i.e., IPv4 or IPv6), the PDP address, and the requested Quality of Service (QoS). With an active PDP context, the UE is known to external packet data networks (PDN) and can send and receive PDP protocol data units (PDUs) with the external PDN through its gateway (i.e., the GGSN).

When a UE transitions to the PMM-Idle state, it remains attached to the GPRS network and has a GMM context, but data transmission and reception are not possible, because there is not a PS signaling connection between the UE and the SGSN; therefore, there is no PDP context (i.e., the UE transitions to the SM-Inactive state). In order to reestablish the PS signaling connection between the UE and the SGSN, the UE must perform a Service Request procedure [1][2]. The signaling mechanisms for realizing a Service Request procedure are covered in more detail in Section 2.2.2 . Upon successful reestablishment of the PS signaling connection, the UE moves back to the PMM-Connected state and has the resources necessary to initiate a PDP context activation.

2.2.5 Radio Resource Management

UMTS uses the RRC procedures to allocate radio resources to the UE in a very flexible manner depending on the state of the UE (both PMM context and PDP context) and the amount data that it needs to send. Each physical channel in UMTS is called a transport channel. The following are uplink transport channels:

- **Physical Random Access Channel (PRACH)**: A Slotted Aloha contention-based uplink channel used for transmission of small amounts of data and control from UE.
- **Physical Common Packet Channel (PCPCH)**: A contention-based uplink channel for data.
- **Dedicated Physical Dedicated Channel (DPDCH)**: Physical channel dedicated to UE and is used to transmit large amounts of data.

For small amounts of data, the PRACH is normally used. For small to medium amounts of data, the PCPCH is preferred and for large amounts of data, the DPDCH can be used [1].

On receipt of data packets from the higher layers (i.e. IP), the UE begins the Radio Access Bearer (RAB) assignment procedure in order to establish a Radio Bearer (RB) between the UE and the UTRAN. The signaling mechanisms for realizing a RAB assignment are discussed in greater detail in Section 2.2.6 .

RRM Interaction with SM and GMM

If these PDUs belong to a QoS for which a PDP context does not exist (i.e., SM-Inactive), the UE will first initiate a PDP context activation. If the PDP context already exists, the UE will initiate the Radio Access Bearer (RAB) assignment process by sending a service request to its SGSN. The Service Request procedure, discussed in greater detail in Section 2.2.6 , is used to set up a PS connection to the SGSN if the UE is in the PMM-Idle state or is used to request resource reservation, if the UE is in the PMM-Connected state. To initiate the RAB assignment process, the SGSN sends a RAB Assignment Request message to the UTRAN to establish a RAB. If there is sufficient uplink and downlink capacity to support the request, the UTRAN will establish a RB by sending a RB Setup message to the UE. Once the RB is established the UE can begin sending and receiving PDUs on the uplink and downlink. Section 2.2.6 discusses the RB Setup procedure in greater detail.

2.2.6 UMTS Protocol Interaction Diagrams

This section details how the UMTS control protocols work together to realize common management functions. The common management functions described are: the RRC Connection Setup, the GPRS Attach procedure, the PDP Activation procedure, the RAB Assignment procedure, and the RB Setup procedure.

RRC Connection Setup

All of the following management functions begin with a Radio Resource Control (RRC) connection setup procedure if one does not already exist between the UE and RNC. Figure 2.8 illustrates the principle of how the radio connection between the UE and the RNC is established over the U_u (radio) interface and the access network internal interface, I_{ub} [17]. The RRC connection always starts with the UE sending a RRC Connection Setup Request (*1*) over the common control channel (CCCH). The RRC entity in the RNC receives this message and changes state from Idle to Connected and establishes the I_{ub} bearer between the Node-B and RNC according to the requirements of the request.

The RRC connection request contains all the information necessary for the RNC to coordinate the setup of the radio connection between the UE and the RNC. The message contains the UE identity, its location and routing information, and the requested QoS. Depending on the requested QoS, the RNC makes the decision whether to allocate dedicated or common resources to the request. The RNC will negotiate a I_{ub} bearer with the Node-B.

When the I_{ub} communication is ready, the RNC sends the RRC Connection Setup (*2*) message to the UE with the appropriate information to establish the radio channel with the Node-B. The message informs the UE of the transport format, power control, and channel codes so that it can configure it RLC with the appropriate settings. The UE confirms that the RRC connection is established by sending the RRC Connection Setup Complete message (*3*) [17].

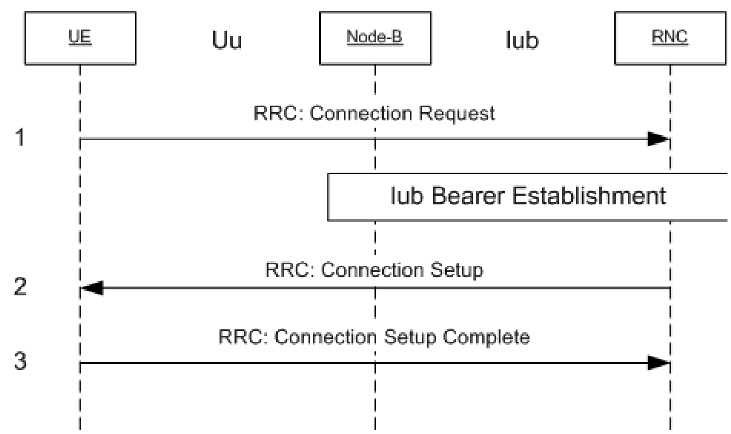

Figure 2.8: RRC Connection Setup

Once the RRC connection has been established, the UE can proceed with its subsequent management function. The RNC provides the protocol translation for the RRC and the RANAP protocols giving the UE a virtual PS signaling connection shown in Figure 2.5. Figure 2.9 illustrates how this virtual PS signaling connection is actually realized. The UE sends the message RRC: Initial Direct Transfer to the RNC. It carries as its payload the first system network message corresponding to the management function task from the UE to the network. Upon receiving this message the RNC prepends a header with additional control information and forwards it to the SGSN as a RANAP: UE Initial Message. This message contains as it payload the original RRC direct transfer message with the first system network message generated by the UE [17]. The following figures will use the notation RRC DT (payload) and RANAP DT (payload) to illustrate a network management message being transferred as payload in a RRC and RANAP direct transfer, respectively.

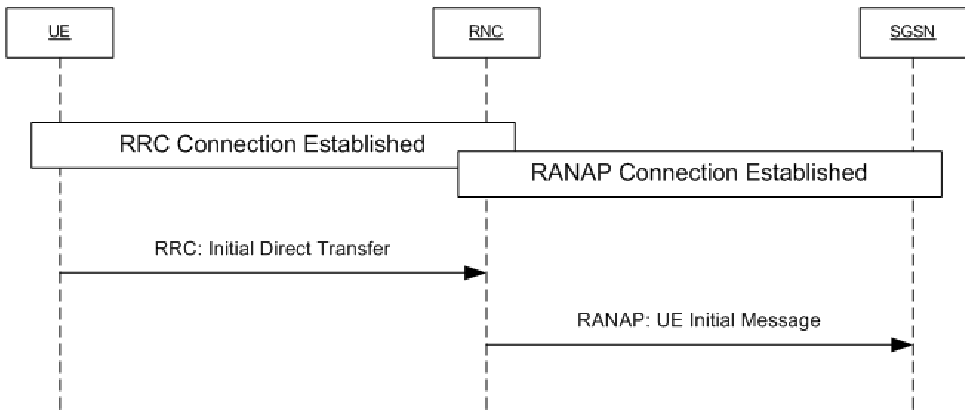

Figure 2.9: Transaction Reasoning

GMM Attach Procedure

The GPRS Mobility Management (GMM) Attach procedure is performed to register the UE for the GPRS network services by notifying the SGSN of its location and by establishing a PS signaling connection between the UE and the SGSN. After successfully executing the GMM Attach procedure the UE is in the PMM-Connected state, Figure 2.6, and GMM context is established in both the UE and the SGSN. The UE may then activate PDP contexts as described below. The GMM protocol makes use of the signaling connection provided by the radio network layer protocols shown in Figure 2.5. This signaling connection, referred as the radio resource (RR) connection, consists of the RRC protocol connection over the signaling radio bearer and the RANAP connection over the I_u signaling bearer.

The three-way handshake signaling mechanism for the GMM Attach procedure is illustrated in Figure 2.10. The UE initiates the GMM Attach procedure by sending a GMM: Attach Request (*1*) message to the SGSN. This message includes the UE's IMSI (International Mobile Subscriber Identity), the type of attach requested and a follow-on request indication that specifies whether the I_u connection should be released or kept after the GMM attach procedure. Upon receipt of the GMM: Attach Request message, the SGSN will perform the authentication procedures with the HLR and ensure that the UE is authorized GPRS network services. The SGSN will then send GMM: Attach Accept (*2*) message back to the UE indicating that the UE is accepted into the GPRS network. The three-way handshake is completed with the UE acknowledging the attach accept message with a GMM: Attach Complete (*3*).

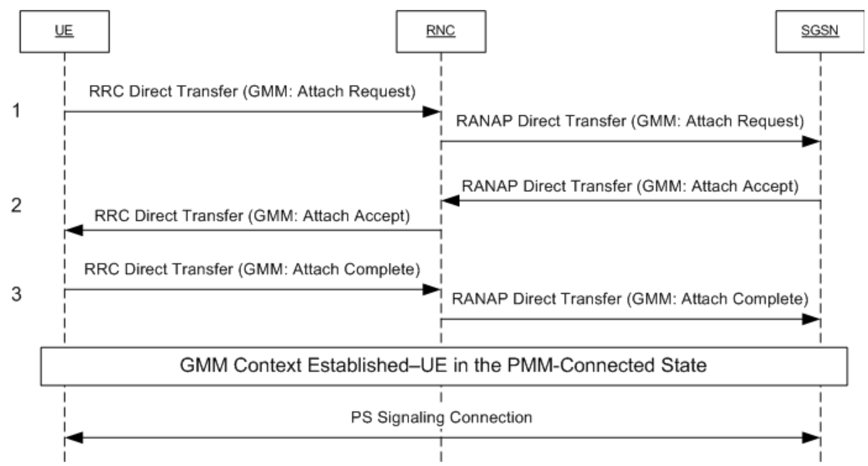

Figure 2.10: GMM Attach Procedure

PDP Activation Procedure

The PDP (packet data protocol) activation procedure is required when a RAB is requested for a class of service whose PDP context is inactive (i.e., the UE is the SM-Idle state for that QoS, see Figure 2.6). Upon successful completion of the PDP activation procedure, the UE will be in the SM-Active state, a RAB will exist between the SGSN and the UE with the associated PDP context parameters, and the UE will be able to exchange PS data with the CN. Figure 2.11 illustrates the PDP activation procedure initiated by the UE.

When the UE receives PDUs from the higher layers for a QoS that does not have an activated PDP context, the UE initiates PDP context activation by sending a SM: Activate PDP Context Request (*1*) message to the SGSN. Upon receipt of this message, the SGSN will initiate the RAB assignment procedure (*2*) described in the next section. On receipt of the RANAP: RAB Assignment Response (*2.4*) message, the SGSN will request that the GGSN create a PDP context with the negotiated parameters. The GGSN will add the UE's PDP address—IP address pair to its routing table, and advertise the UE's IP address to the external IP network. After the RAB has been established the SGSN confirms the packet session establishment by sending a SM: Activate PDP Context Accept (*3*) message to the UE.

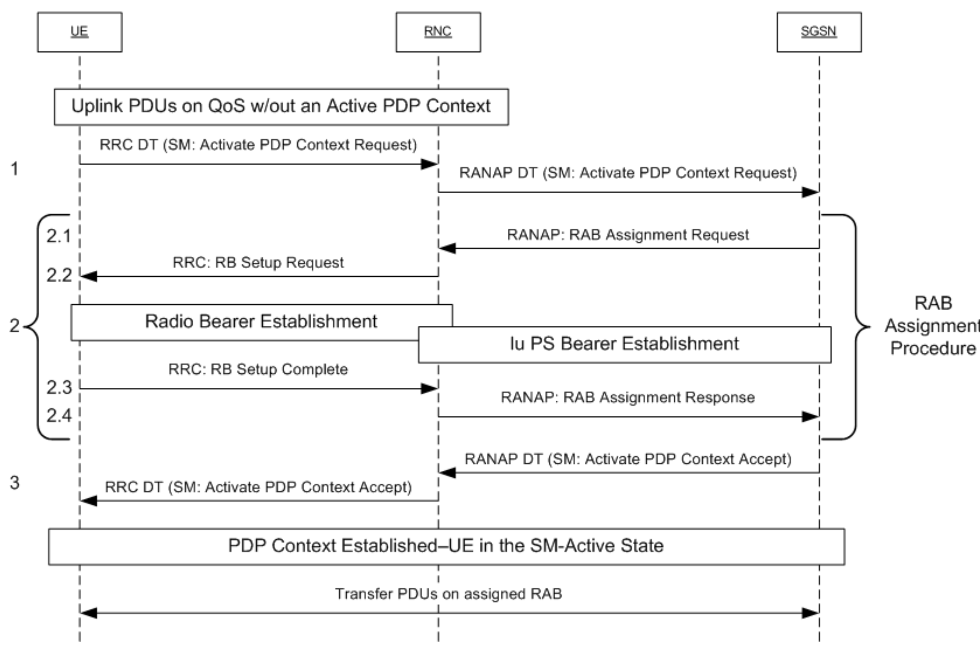

Figure 2.11: PDP Activation and RAB Assignment Procedures

RAB Assignment Procedure

The RAB Assignment procedure is illustrated in step *2* of Figure 2.11. The RAB assignment procedure can be initiated by either a PDP activation, when a PDP context has not yet been established for the requested QoS or by a service request, as in Figure 2.12, when a PDP context is already active, but no RAB exists.

Upon receiving either a PDP activation or a service request, the SGSN initiates RAB assignment activities. First, the SGSN ensures that the UE under its current subscription has the rights to perform the operation, and then begins RAB allocation by assigning a unique RAB ID and requests a RAB for the given QoS by sending a RANAP: RAB Assignment Request (*2.1*) message to the RNC. The RNC performs an admission control procedure checking the availability of uplink and downlink capacity. If sufficient capacity is available, the RNC establishes the appropriate radio bearer (RB) by sending the UE a RRC: RB Setup Request (*2.2*) message. The UE establishes the appropriate RB as specified by the RNC and sends a RRC: RB Setup Complete (*2.3*) message to the RNC. The RNC completes the transaction by notifying the SGSN of the established RAB-RB mapping with a RANAP: RAB Assignment Response (*2.4*) message.

Service Request Procedure

When a UE requires a RAB for a PDP context that is already active, it initiates the RAB assignment procedure with a SM: Service Request (*1*) message to the SGSN as shown in Figure 2.12. First, the SGSN notifies the UE that the request is granted by sending a SM: Service Accept (*2*) message. It then initiates a RAB assignment in step *3* as specified in detail in the previous section.

2.2.7 UMTS Summary

The previous sections provided an overview of UMTS and its signaling protocols. Section 2.2.1 discussed the standards of the WCDMA air interface. GPRS, the backbone of the UMTS packet domain architecture was discussed in Section 2.2.2 . Section 2.2.3 gave an overview of the UMTS protocol stack, with Section 2.2.4 covering the details of the GMM and Session Management protocols.

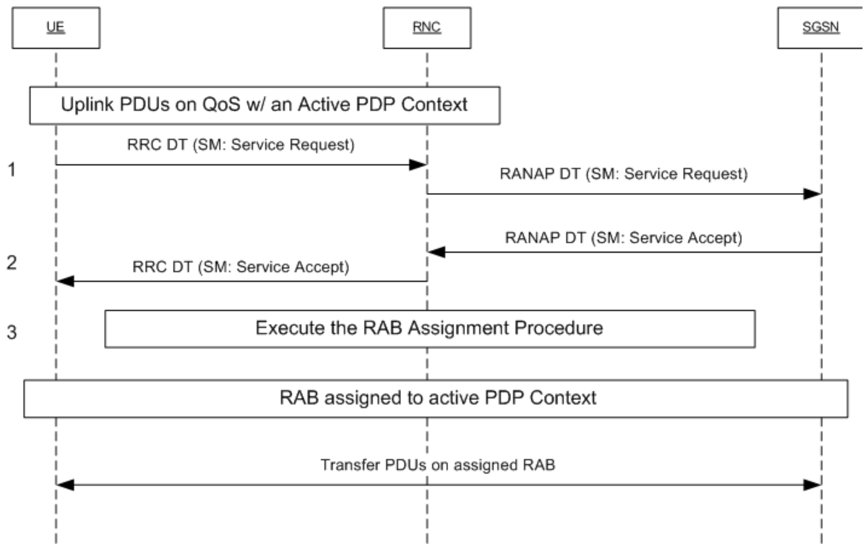

Figure 2.12: Service Request

The simulation design covered in Chapter 4 integrated these signaling protocols into the WLAN access point in order to create an alternate radio access network. The UMTS protocol interaction diagrams and detailed discussions of Section 2.2.6 should provide the background information necessary to understand the design. Chapter 4 contains references to these sections for the reader's convenience.

2.3 IEEE 802.11 Wireless Local Area Networks

Wireless Local Area Networks (WLANs) continue to proliferate in corporate and residential environments. The IEEE 802.11 family of WLANs have dominated in the United States and are now becoming dominate throughout Europe. This work focused on the IEEE 802.11 family of wireless LANs. The original IEEE 802.11 standard [3] and each of its supplemental standards Table 2.1 provides a basic overview of the current versions of the 802.11 technologies. In Europe, the European Telecommunications Standards Institute (ETSI) originally developed a competing standard, HiperLan2. HiperLan2 also operated in the 5GHz range, just as IEEE 802.11a. The IEEE 802.11 family of protocols define the

Medium Access (MAC) layer and physical (PHY) layer protocols required to deliver MSDUs (MAC Service Data Units) between peer LLCs (Logical Link Controls). An overview of the IEEE 802.11 system architecture is presented in Section 2.3.1. Section 2.3.2 defines the frame structure of an IEEE 802.11 frame and Section 2.3.3 presents the protocol architecture. The latter sections present more specific information about the operation of IEEE 802.11. Section 2.3.4 and 2.3.5 present information about the operation of the physical (PHY) layer and the MAC layer of the protocol. The WLAN modes of operation are presented in Section 2.3.7 and 2.3.8. One key issue for the second phase of our research was the per frame overhead associated with the protocol. In IEEE 802.11 networks, it is not possible to both transmit and receive on the same channel using a single radio transceiver. The 802.11 standard uses CSMA/CA (carrier sense multiple access with collision avoidance); whereas, standard Ethernet uses CSMA/CD (carrier sense multiple access with collision detection). Another way to say this is that an 802.11 wireless LAN only takes measures to avoid collisions, not to detect them. CSMA/CA requires somewhat extensive periods of inactivity called interframe spaces (IFS) to occur between transmission. IFS will be discussed further in Section 2.3.6. A method to amortize these periods of inactivity to improve performance is one of the key aims of AGG-MAC. This work is presented in Chapters 6 and 7.

Table 2.1: Comparison of IEEE 802.11 WLAN Standards

	802.11	**802.11b**	**802.11a**	**802.11g**
Raw Data Rate	1,2	1,2,5.5,11	$6, 9, 12, 18, 24, 36, 48, 54$	$1, 2, 5.5, 6, 9, 11, 12, 22, 24, 33, 36, 54$
Frequency	2.4GHz	2.4Ghz	5GHz	2.4GHz
Available Spectrum	83.5MHz	83.5MHz	300MHz	83.5MHz
Modulation Encoding	$FHSS/FSK$ $DSSS/PSK$ IR/PPM	DSSS/CCK	OFDM	DSSS/OFDM
# Channels/nonoverlapping	11 / 3	11 / 3	12 / 8	11 / 3
maxMSDU	2304	2304		

2.3.1 System Architecture

The IEEE 802.11 architecture is comprised of several components and services that interact to provide station mobility transparent to the higher layers of the network stack.

The Wireless LAN station (STA) is the most basic component of the wireless network. A station is any device that contains the functionality of the 802.11 protocol, consisting of MAC, PHY, and a connection to the wireless media. Typically the 802.11 functions are implemented in the hardware and software of a network interface card (NIC). A station could be a laptop PC, handheld device, or an Access Point (AP). Stations may be mobile, portable, or stationary and all stations support the 802.11 station services of authentication, de-authentication, privacy, and data delivery.

802.11 defines the Basic Service Set (BSS) as the basic building block of an 802.11 wireless LAN. The BSS consists of a group of any number of stations. The BSS is not a very interesting topic until we take the topology of the WLAN into consideration.

Topology

The IEEE 802.11 topology consists of components, interacting to provide a wireless LAN that enables station mobility transparent to higher protocol layers, such as the LLC. The 802.11 standard supports the following two topologies: ad hoc mode and infrastructure mode.

The most basic wireless LAN topology is a set of stations which have recognized each other and are connected via the wireless media in a peer-to-peer fashion. This form of network topology is referred to as an Independent Basic Service Set (IBSS) (see Figure 2.13) or an Ad hoc network. Ad hoc mode (also called peer-to-peer mode or an Independent Basic Service Set, or IBSS) is simply a set of 802.11 wireless stations that communicate directly with one another without using an access point or any connection to a wired network. This mode is useful for quickly and easily setting up a wireless network anywhere that a wireless infrastructure does not exist or is not required for services, or where access to the wired network is barred. A STA is free to move about the BSS, but can no longer communicate with other stations if it leaves the IBSS. Every mobile station may not be able to communicate with every other station due to the range limitations. There are no relay functions in an IBSS therefore all stations need to be within range of each other and communicate directly.

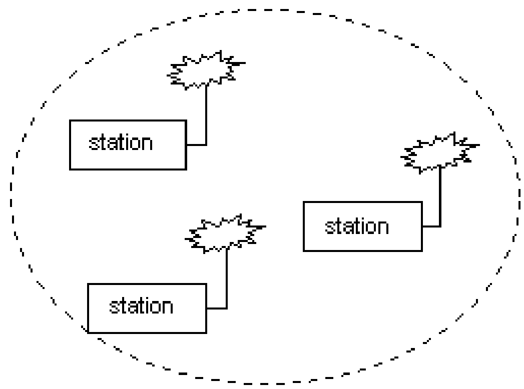

Figure 2.13: WLAN Infrastructure Basic Sevice Set (IBSS)

In infrastructure mode (see Figure 2.14), the wireless network consists of at least one access point (AP) potentially connected to the wired network infrastructure and a set of wireless end stations. This configuration is called a Basic Service Set (BSS). An Extended Service Set (ESS) is a set of two or more BSSs forming a single subnetwork. The 802.11 standard defines the distribution system (DS) as an element that interconnects BSSs within the ESS via APs. The distribution system supports the 802.11 mobility types by providing logical services necessary to handle address-to-destination mapping and seamless integration of multiple BSSs. An access point is an addressable station, providing an interface to the distribution system for stations located within various BSSs. The independent BSS and ESS networks are transparent to the LLC Layer.

The 802.11 standard does not constrain the composition of the distribution system; therefore, it may be 802-compliant or some non-standard network. If data frames need transmission to and from a non-IEEE 802.11 LAN, then these frames, as defined by the

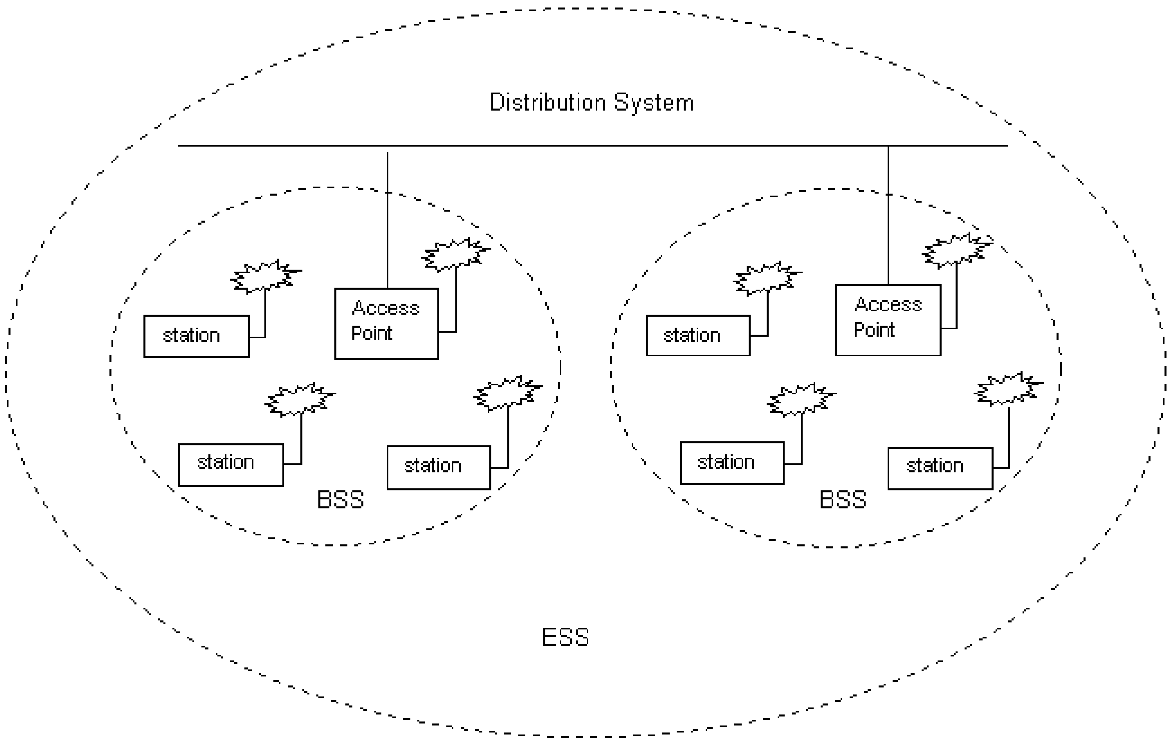

Figure 2.14: WLAN Infrastructure Mode

802.11 standard, enter and exit through a logical point called a portal. The portal provides logical integration between existing wired LANs and 802.11 LANs. When the distribution system is constructed with 802-type components, such as 802.3 (Ethernet) or 802.5 (Token Ring), then the portal and the access point become one and the same.

Logical Architecture

A topology provides a means of explaining necessary physical components of a network, but the logical architecture defines the network's operation. The logical architecture of the 802.11 standard that applies to each station consists of a single MAC and one of multiple PHYs: frequency hopping spread spectrum, direct sequence spread spectrum, and infrared light.

The goal of the MAC Layer is to provide access control functions (such as addressing, access coordination, frame check sequence generation and checking, and LLC PDU delimiting) for shared-medium PHYs in support of the LLC Layer. The MAC Layer performs the addressing and recognition of frames in support of the LLC.

2.3.2 WLAN Frame Structure

[3] specifies the basic frame structure shown in Figure 2.15. Each frame consists of the following basic components:

a) A MAC header, which comprises frame control, duration, address, and sequence control information;

b) A variable-length frame body, which contains information specific to the frame type;

c) A frame check sequence (FCS), which contains an IEEE 32-bit cyclic redundancy code (CRC).

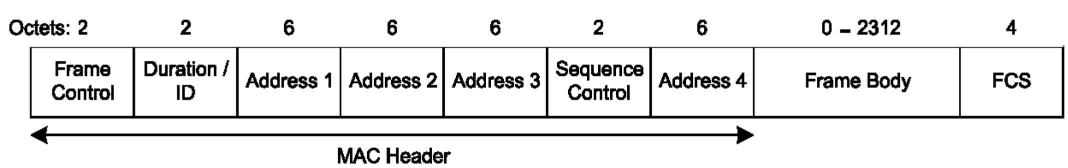

Figure 2.15: IEEE 802.11 Frame Structure [3]

Frame Control

Within the MAC Header, the first two octets define the Frame Control (FC) field. The Frame Control field consists of the following subfields: Protocol Version, Type, Subtype, To DS, From DS, More Fragments, Retry, Power Management, More Data, Wired Equivalent Privacy (WEP), and Order. The format of the Frame Control field is illustrated in Figure 2.16.

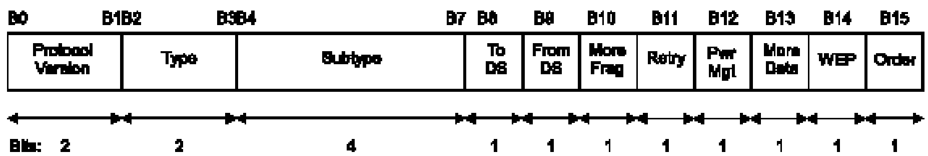

Figure 2.16: IEEE 802.11 Frame Control Field [3]

Protocol Version subfield The Protocol Version field is 2 bits in length and is invariant in size and placement across all revisions of this standard. For this standard, the value of the protocol version is 0. All other values are reserved. The revision level will be incremented only when a fundamental incompatibility exists between a new revision and the prior edition of the standard. A device that receives a frame with a higher revision level than it supports will discard the frame without indication to the sending station or to LLC.

Type and Subtype subfields The Type field is 2 bits in length, and the Subtype field is 4 bits in length. The Type and Subtype fields together identify the function of the frame. There are three frame types: control, data, and management. Each of the frame types have several defined subtypes. Table 2.2 defines the valid combinations of type and subtype.

Table 2.2: IEEE 802.11 Valid Type and Subtype Combinations

Type value b3 b2	**Type description**	**Subtype b7 b6 b5 b4**	**Subtype description**
00	Management	xxxx	(Management Frames)
01	Control	xxxx	(Control Frames)
10	Data	0000	Data
10	Data	0001	Data + CF-Ack
10	Data	0010	Data + CF-Poll
10	Data	0011	Data + CF-Ack + CF-Poll
10	Data	0100	Null function (No Data)
10	Data	0101	CF-Ack (No Data)
10	Data	0110	CF-Poll (No Data)
10	Data	0111	CF-Ack + CF-Poll (No Data)
10	Data	1000-1111	Reserved (see [18])
11	Reserved	0000-1111	Reserved

To DS subfield The To DS field is 1 bit in length and is set to 1 in data type frames destined for the DS. This includes all data type frames sent by STAs associated with an AP. The To DS field is set to 0 in all other frames.

From DS subfield The From DS field is 1 bit in length and is set to 1 in data type frames exiting the DS. It is set to 0 in all other frames..

More Fragments subfield The More Fragments field is 1 bit in length and is set to 1 in all data or management type frames that have another fragment of the current MSDU or current MMPDU to follow. It is set to 0 in all other frames.

Retry subfield The Retry field is 1 bit in length and is set to 1 in any data or management type frame that is a retransmission of an earlier frame. It is set to 0 in all other frames. A receiving station uses this indication to aid in the process of eliminating duplicate frames.

Power Management subfield The Power Management field is 1 bit in length and is used to indicate the power management mode of a STA. The value of this field remains constant in each frame from a particular STA within a frame exchange sequence defined in [3]. The value indicates the mode in which the station will be after the successful completion of the frame exchange sequence. A value of 1 indicates that the STA will be in power-save mode. A value of 0 indicates that the STA will be in active mode. This field is always set to 0 in frames transmitted by an AP.

More Data subfield The More Data field is 1 bit in length and is used to indicate to a STA in power-save mode that more MSDUs, or MMPDUs are buffered for that STA at the AP. The More Data field is valid in directed data or management type frames transmitted by an AP to an STA in power-save mode. A value of 1 indicates that at least one additional buffered MSDU, or MMPDU, is present for the same STA. The More Data field may be set to 1 in directed data type frames transmitted by a contention-free (CF)-Pollable STA to the point coordinator (PC) in response to a CF-Poll to indicate that the STA has at least one additional buffered MSDU available for transmission in response to a subsequent CF-Poll. The More Data field is set to 0 in all other directed frames. The More Data field is set to 1 in broadcast/multicast frames transmitted by the AP, when additional broadcast/multicast MSDUs, or MMPDUs, remain to be transmitted by the AP during this beacon interval. The More Data field is set to 0 in broadcast/multicast frames transmitted by the AP when no more broadcast/multicast MSDUs, or MMPDUs, remain to be transmitted by the AP during this beacon interval and in all broadcast/multicast frames transmitted by non-AP stations.

WEP subfield The WEP field is 1 bit in length. It is set to 1 if the Frame Body field contains information that has been processed by the WEP algorithm. The WEP field is only set to 1 within frames of type Data and frames of type Management, subtype Authentication. The WEP field is set to 0 in all other frames. When the WEP bit is set to 1, the Frame Body field is expanded as defined in [3]. The future of WEP appears to be limited at this point. The security provided by WEP is limited and new solutions are currently in progress. IEEE 802.1x is currently replacing most WEP solutions and is the security solution of the foreseeable future.

Order subfield The Order field is 1 bit in length and is set to 1 in any data type frame that contains an MSDU, or fragment thereof, which is being transferred using the StrictlyOrdered service class. This field is set to 0 in all other frames.

Duration/ID field

Immediately following the Frame Control field in the IEEE 802.11 header is the Duration/ID field. The Duration/ID field is 16 bits in length. The contents of this field are as follows:

1. In control type frames of subtype Power Save (PS)-Poll, the Duration/ID field carries the association identity (AID) of the station that transmitted the frame in the 14 least significant bits (lsb), with the 2 most significant bits (msb) both set to 1. The value of the AID is in the range 1–2007.

2. In all other frames, the Duration/ID field contains a duration value as defined for each frame type in section 7.2 of [3]. For frames transmitted during the contention-free period (CFP), the duration field is set to 32 768.

Whenever the contents of the Duration/ID field are less than 32 768, the duration value is used to update the network allocation vector (NAV).

2.3.3 WLAN Protocol Architecture

As indicated by the standard number, IEEE 802.11 fits seamlessly into the other 802.x standards for wired LANs. Applications should not notice any difference apart from a lower bandwidth and possible a higher access time from the WLAN. Therefore, the higher layers (application, TCP/IP, etc.) in a wireless node look the same as the wired node [17].

The IEEE 802.11 standard only covers the medium access (MAC) and physical (PHY) layers like the other 802.x LAN standards. Figure 2.17 is the basic reference protocol reference model from [3].

The physical layer is subdivided into a physical layer convergence protocol (PLCP) and the physical medium dependent (PMD) sublayers. The basic tasks of the MAC layer are

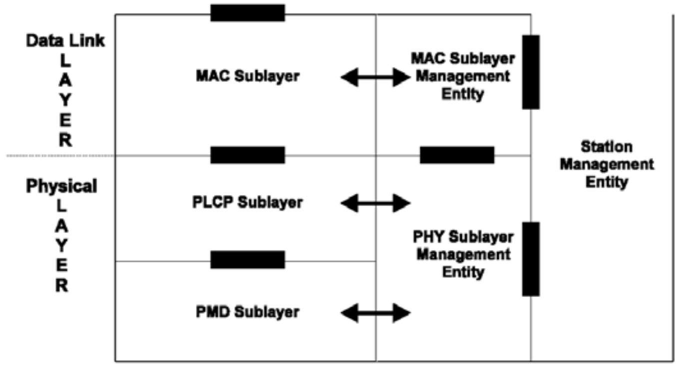

Figure 2.17: IEEE 802.11 Protocol Reference Model [3]

medium access, fragmentation of user data, and encryption. The PLCP sublayer provides a carrier sense signal, called clear channel assessment (CCA), and provides a common PHY interface for the MAC that is independent of the transmission technology. The PMD sublayer handles modulation and encoding/decoding of signals. The PHY is covered in more detail in Section 2.3.4 and the MAC in Section 2.3.5.

2.3.4 WLAN Physical Layer (PHY)

The WLAN physical (PHY) layer is out of the scope of this research effort because it will not be explicitly modeled—the basic physical layer information will be included for completeness. [3] supports three different PHYs: one based on infrared and two layers on the basis of radio transmission, specifically the 2.4 GHz Industrial, Scientific, and Medical (ISM) band. All PHY variants include the provision of the clear channel assessment (CCA) signal introduced in Section 2.3.3 . This signal is needed by the MAC in order to determine if the medium is idle. The three PHYs specified in [3] are Frequency Hopping Spread Spectrum (FHSP), Direct Sequence Spread Spectrum (DSSS), and Infrared. Our research focused on the DSSS version of the PHY because it was the foundation of all current versions of the IEEE 802.11 series of WLAN protocols. For the UMTS-WLAN interworking, we note that all WLAN "hot spot" employments are based on DSSS protocols. In our AGG-MAC protocol work, we note that our MAC layer enhancements will function properly on any of the PHY layers as defined in all of the IEEE 802.11 standards versions.

2.3.5 WLAN Medium Access Control (MAC)

The primary responsibility of the WLAN MAC is to control medium access, but it can also provide optional support for roaming, authentication, and power conservation. The basic services provided by the MAC layer are the mandatory asynchronous data service and an optional time-bounded service. The standard specifies that 802.11 only offer the asynchronous service in the ad hoc network mode, while both services work together in an infrastructure based network with an access point coordinating medium access and defines the following three basic access mechanisms: the mandatory basic method based on a version of carrier sense multiple access with collision avoidance (CSMA/CA), an optional method avoiding the hidden terminal problem, and a contention-free polling method for time-bounded service. The first two methods are termed the distribution coordination function (DCF), and the third is the point coordination function (PCF). Figure 2.18 [3] shows the interaction of the waiting time between frames, termed interframe spaces (IFS), used for controlling the waiting time before accessing the medium.

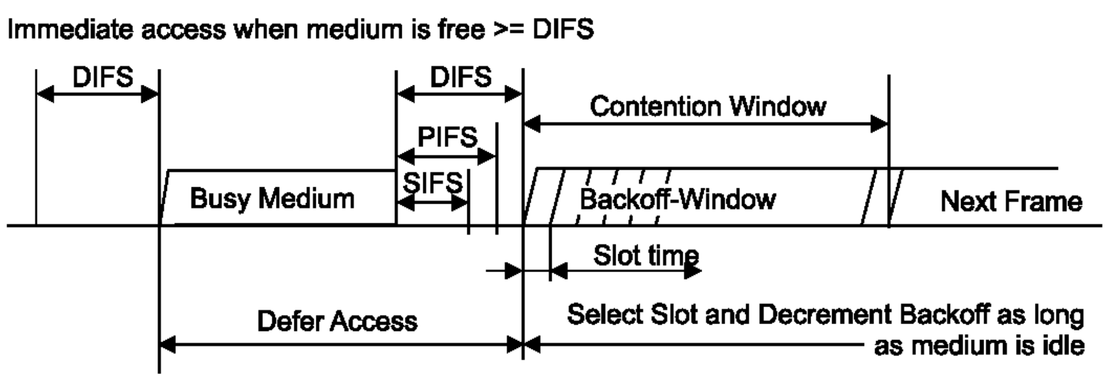

Figure 2.18: Medium Access and Interframe Spacing [3]

2.3.6 Interframe Spacing

The time interval between frames is called the IFS. A station will determine that the medium is idle through the carrier sense mechanism, specifically the CCA in the PLCP sublayer. Four different IFSs (Figure 2.18) are specified in [3] to provide priority levels of access to the wireless media; the main three that are relevant to this research are listed below in order, from shortest to longest. A station must wait an IFS after sensing an idle medium before it can attempt to access the medium.

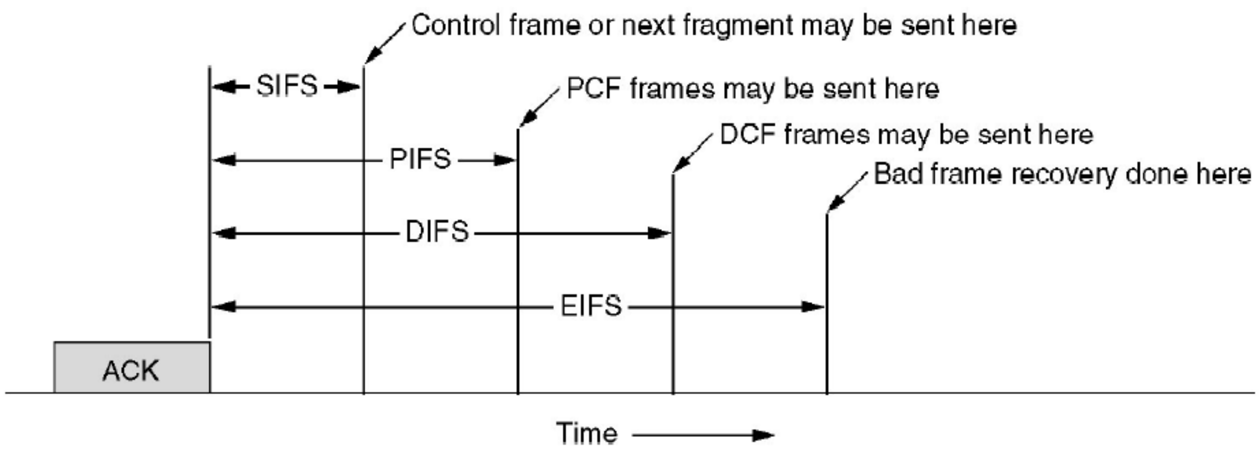

Figure 2.19: Interframe Spacing in IEEE 802.11 [3]

Short Interframe Spacing (SIFS)

SIFS is the shortest waiting time for medium access and thus the highest priority for medium access. It is defined for short control messages, such as acknowledgements for data packets or polling responses. SIFS is covered in greater detail in Sections 2.3.7 and 2.3.8.

PCF Interframe Spacing (PIFS)

PIFS is the waiting time between DIFS and SIFS (and thus medium priority). It is defined for the time-bounded service used in PCF. An access point only has to wait PIFS after sensing an idle medium in order to access. Both PCF and PIFS are covered in greater detail in Section 2.3.8.

DCF Interframe Spacing (DIFS)

DIFS is the longest waiting time (and thus the lowest priority). This IFS is defined for the asynchronous data service within a contention period used in the DCF. Both DCF and DIFS are covered in greater detail in Section 2.3.7.

2.3.7 Distributed Coordination Function (DCF)

[3] defines the basic medium access protocol as the distributed coordination function (DCF). It allows stations in the same BSS to share the medium using CSMA/CA and a random back-off time following a busy medium. It also specifies that a receiving station will respond with an immediate positive acknowledgement (ACK frame) following successful receipt of a frame, while the sender schedules immediate retransmission if the ACK is not received.

CSMA/CA

CSMA/CA with binary exponential backoff is the basic access mechanism specified by the DCF. Using the physical carrier sense mechanism provided by the CCA, a station will listen to the medium before beginning a transmission. If the medium is already carrying a transmission, the station will not begin its own transmission. This is the CSMA portion of the access mechanism. If two or more stations sense an idle medium and begin their transmission at the same time then there will be a collision, which may result in one or more frames being corrupted.

The IEEE 802.11 MAC uses collision avoidance rather than collision detection in order to transmit and receive simultaneously. For this reason, the IEEE 802.11 MAC implements a virtual sensing mechanism termed the network allocation vector (NAV). The NAV is a value that indicates to a station the amount of time that remains before the medium will become available. The NAV is kept current through duration values that are transmitted in all frames. By combining this virtual sensing mechanism (using the NAV) with the physical sensing mechanism (using the CCA), the MAC implements the collision avoidance portion of the CSMA/CA access mechanism.

In order to avoid collision, a station listens to the medium before beginning its own transmission. If it detects an existing transmission then it will enter into a deferral period determined by binary exponential backoff algorithm.

DCF and DIFS

The DCF uses the physical and virtual carrier sense mechanisms to determine if the medium is idle. If both mechanisms indicate that medium in not in use for an interval of DIFS then the station will begin to transmit the frame. However, if the medium is busy then the random backoff algorithm is applied to prevent access collisions. The transmission is considered to be unsuccessful if an ACK is not received, resulting in the retransmission of the frame.

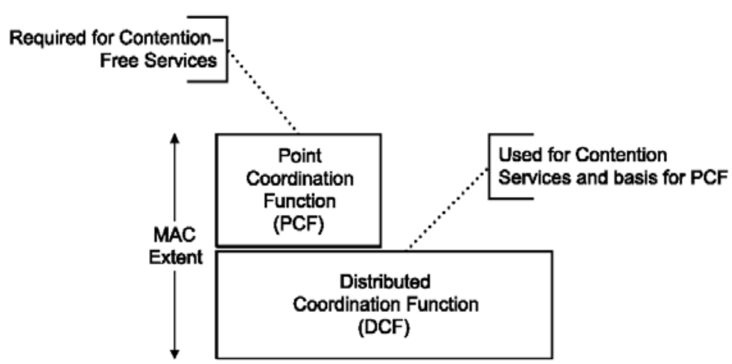

Figure 2.20: MAC Reference Model [3]

2.3.8 Point Coordination Function (PCF)

The DCF cannot guarantee a maximum access delay or minimum transmission bandwith. To provide a time-bounded service, [3] specifies the PCF on top of the basic DCF access mechanism (see Figure 2.20). Using PCF requires an AP that controls medium access by polling individual stations. The polling mechanism in the AP is termed the point coordinator (PC). The PC splits the access time into superframe periods shown in Figure 2.21 [3]. A superframe comprises a contention-free period (CFP) and a contention period. The contention period is used for stations that are not accessing the AP and are using the DCF access mechanism [17].

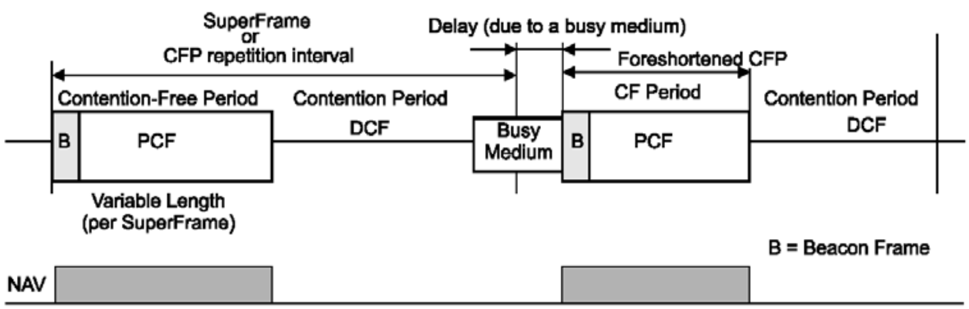

Figure 2.21: Contention-Free Period using SuperFrames [3]

PCF operates by requiring stations to request that the PC register them on a polling list. Once a station is registered with the PC, it becomes contention-free pollable (CF-Pollable). During the CFP, the PC polls and delivers traffic to the CF-Pollable stations at regular intervals. Figure 2.22 [3] is an example of PCF data transfer with four stations.

In this example, there are four CF-Pollable stations. The AP has data to send to all four stations, but only stations 1, 2, and 4 have data to send to the AP. The PC transmits a Beacon to mark the beginning of the CFP. The PC will transmit data to, respond with an ACK, and/or poll a CF-pollable station using the SIFS as is the cases when it sends data to stations 1, 2, and 3. Stations 1 and 2 wait a SIFS before responding to the AP with an ACK and subsequent data transmission. The data exchange attempt with station 3 highlights

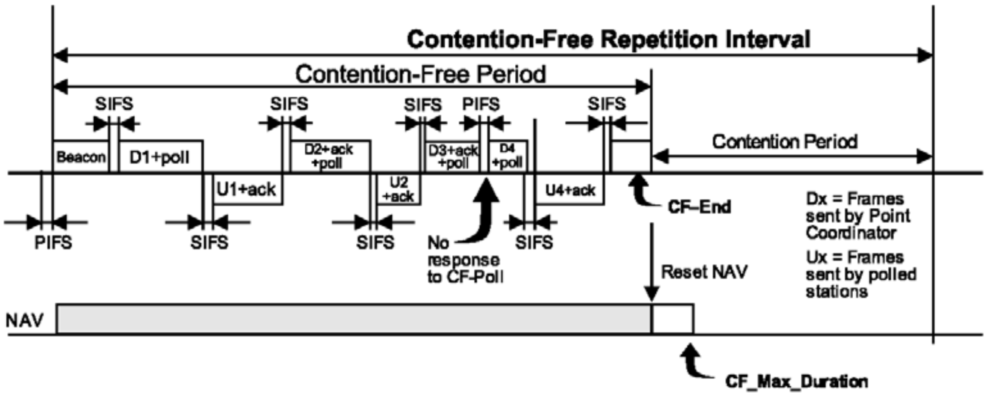

Figure 2.22: PCF Data Transfer with 4 Stations [3]

what happens when a CF-Pollable station does not respond to its poll. The PC expects station 3 to respond with an ACK and transmit any data it might have after waiting a SIFS. When a SIFS period elapses without the receipt of the expected transmission, the PC may send its next pending transmission as soon as one PIFS after the end of its last transmission. This permits the PC to retain control of the medium in the presence of an overlapping BSS. The PC ends the CFP with a CF-End Beacon.

WLAN "hot spots" provide access to the Internet for clients in public locations. This is accomplished by use of an AP providing PCF WLAN service.

2.3.9 Fragmentation Threshold

The MAC may fragment and reassemble directed, but not multicast or broadcast, MSDUs or MMPDUs. The length of a fragment shall never be larger than *aFragmentationThreshold* unless the application of WEP creates a fragment exceeding the threshold size. Once a fragmented MSDU is transmitted for the first time, its frame body content and length will remain fixed, regardless of any required retransmission or intermediate relays. The size of each fragment, with the possible exception of the final fragment, in a fragmented MSDU or MMPDU will be set to the value *aFragmentationThreshold*. The size of the final fragment will vary based on the amount of data remaining to send, but will not be larger than a*FragmentationThreshold*. The More Fragments subfield of the Frame Control field will be set to 0 for the last (or only) fragment of the MSDU or MMPDU. Both source and destination will establish a timer to limit the lifetime of each fragmented MSDU. If the source timer expires before successful transmission of all fragments, the STA will discard remaining segments and make no further attempts to transmit the MSDU. The destination will maintain timers for at least three fragmented MSDUs (more is permitted). Destination is permitted to discard fragments for fragments for which the STA is not maintaining a receive timer and for all fragments received for an expired timer.

2.4 Mobile IP

Providing seamless, transparent data services to the applications running on mobile devices is one of the key issues with integrating WLAN into 3G. Mobile IP is a proposed standard protocol that builds on the Internet Protocol by making mobility transparent to applications and higher level protocols like TCP. Perkins', "Mobile IP," [19] is the seminal paper for this body of research.

This article describes some of the technical obstacles that must be overcome before mobile networking can become widespread. The author says that the most fundamental technical issue is the way that IP routes packets to their destination according to IP address. The main problem being that IP addresses are associated with a fixed network location. When a packet's destination is a mobile node then, as the mobile moves, each new point of attachment is associated with a new network number and, hence, a new IP address, making transparent mobility impossible.

The author presents, Mobile IP (RFC 2002 [4]), a standard proposed by a IETF working group, that was designed to solve this problem by allowing the mobile node to use two IP addresses: a fixed home address and a care-of address that changes at each new point of attachment. Perkins explains Mobile IP in moderate detail. In Mobile IP, the home address is static and is used to identify TCP connections. The care of address changes at each new point of attachment and is the mobile node's network location significant address; it indicates the network number and thus identifies the mobile node's point of attachment with respect to the network location. The home address makes it appear that the mobile node is always attached to its home network. Mobile IP requires the existence of two additional network nodes, known as the home agent and foreign agent. Whenever the mobile node is not attached to its home network, the home agent forwards all of its packets to its current point of attachment, termed a foreign network. A foreign agent acts on behalf of the mobile nodes at the foreign network. Whenever the mobile node moves, its new foreign agent registers its new care-of address with its home agent. To get a packet to a mobile node from its home network, the home agent delivers the packet from the home network to the care-of address, using IP tunneling shown in Figure 2.23 [4].

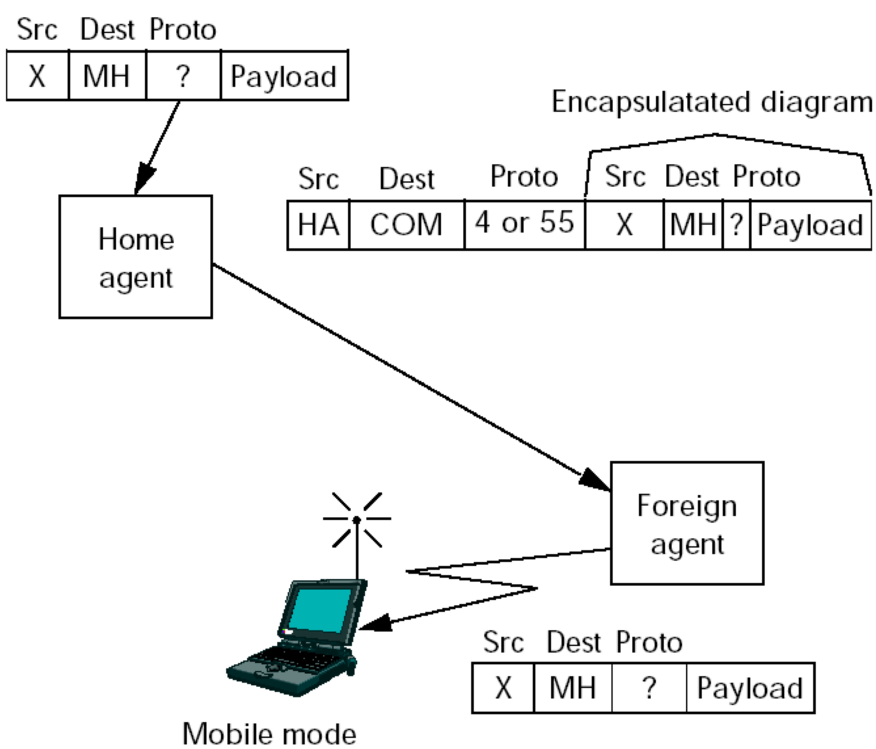

Figure 2.23: Mobile IP Tunneling [4].

The home agent constructs a new IP header that contains the mobile node's care-of address as the destination IP address. This new header then encapsulates the original packet, causing the mobile node's home address to have no effect on the encapsulated packet's routing until it arrives at the care-of address. Its new header is stripped off revealing the original packet, that is forward to the mobile node.

In the IEEE Communications Magazine, 50th Anniversary Commemorative Issue [20], Perkins gives an author's introduction, "Mobile IP—Updated," to his original article [19] that was being recognized as one of the Ten Landmark Articles for the previous decade. Perkins highlights the successes and the existing shortcomings of Mobile IP for both Mobile IPv4 and IPv6 [21]. The author highlights the existing shortcomings as: routing efficiencies caused by triangle routing; security issues caused by Mobile IP traffic being blocked by firewalls; egress router filtering issues that caused IP packets without a source address of an internal network to be blocked by border routers; and a slow growth in the WLAN market that he does not provide a reason for. The existing work on Mobile IPv6 is focused on solutions to all of these problems. The problem with IPv6 is that it is having difficulties spreading with a highly deployed IPv4 infrastructure in place.

The design approach presented in Chapter 4 does not require Mobile IP in order to make mobility transparent to applications and higher layer protocols. It uses the PDP context address functionality for the UMTS GPRS in order to provide this transparent data service. In the future, as UMTS transitions to an all IP framework, Mobile IP will certainly be key protocol to allow this transition to happen. The simulation framework presented in this research was designed with an open architecture so that future investigators will be able to easily modify the models in order to incorporate Mobile IP and allow seamless data transfer over an all IP backbone.

2.5 Aggregation

Aggregation is not a new concept. It has been used primarily in the core network to reduce overhead, primarily computational overhead, in routers. Aggregation can be employed in a number of cases to amortize the cost associated with network, in our case protocol, overhead. The IEEE 802.11e working group considered the inclusion of an aggregation or container frame to improve performance [22]. This approach differs from our work in that every packet is first encapsulated in its own individual WLAN frame and then the collection of frames is subsequently encapsulated into a container frame. This approach reduces the amount of interframe spacing incurred, but not the MAC layer frame overhead. The 802.11e approach was deemed to be overly complex due to issues arising from how to respond to both container and interior frame protocol responses and has subsequently been removed from consideration in the standard [18][23]. AGG-MAC avoids the issues encountered by the 802.11e working group by aggregating multiple packets into a single WLAN frame. In our case, there is only one frame and the handling requirements of that frame merely follows existing 802.11 specifications [3].

2.6 Summary

This chapter presents the theoretical background and foundation necessary to fully understand this research work. We started with the presentation of a historical perspective of the evolution of wireless cellular networks in Section 2.1. Section 2.2 provided details on the UMTS system. An overview of WLAN, specifically IEEE 802.11, systems was presented in Section 2.3. As it represents the most likely network methodology to employ loosely-coupled interworked cellular and WLAN systems, an overview of Mobile IP was presented in Section 2.4. And finally, Section 2.5 contained a basic introduction to the concept of aggregation.

Chapter 3

Related Work

3.1 Interworked Wireless Systems

The convergence of different wireless network systems and the increased desire for flexible access mechanisms and increased capabilities of handsets and associated terminal devices continued to generate significant research interest. While mobile IP mechanisms seemed to hold the most favor, Mobile IP was not without its issues. Much research continued in providing differing levels of interoperability in mobile systems. Although cellular systems best support mobility, those systems were limited in capacity. For this reason, work was continued to attempt to determine the best methods of interworking high-capacity broadband systems with high-mobility cellular systems.

The initial framework defined by the 3GPP systems architecture (SA) working group for interworking WLAN and UMTS systems is discussed in Section 3.1.1. This framework defines six levels of interoperability between the two systems. In order to achieve the highest levels of interworking, a tightly coupled system would be necessary. Two research activities supported by the European Information Society Technologies (IST) program focus on various aspects of cellular and WLAN interworking. These initiatives will be discussed further in

Section 3.1.2 and Section 3.1.3. An integral assumption of these projects was that cellular systems consist of a pure IP core network. This appeared to be the trend, but was not yet realizable at the time of this work or for the foreseeable future.

3.1.1 3GPP Feasibility Study

3GPP standards continue to evolve. One of the 3GPP stated goals is to support data rates of up to 2Mbps in the indoor environment [24]. This requirement is stated for the 3GPP system and is not based upon the integration of WLAN. The 3GPP realized that mobile cellular systems remain limited in their ability to provide consistent quality coverage inside buildings. They understand that high quality, indoor coverage would allow them to directly compete with infrastructure based systems, and that WLAN systems are currently much better suited to provide such service.

Table 3.1: 3GPP Scenarios for Interworking WLAN

Scenarios: Service and operational Capabilities:	Scenario 1: Common Billing and Customer Care	Scenario 2: 3GPP system based Access Control and Charging	Scenario 3: Access to 3GPP system PS based services	Scenario 4: Service continuity	Scenario 5: Seamless services	Scenario 6: Access to 3GPP system CS based services
Common billing	X	X	X	X	X	X
Common customer care	X	X	X	X	X	X
3GPP system based Access Control		X	X	X	X	X
3GPP based Access Charging		X	X	X	X	X
Access to 3GPP system PS based services from WLAN			X	X	X	X
Service Continuity				X	X	X
Seamless Service Continuity					X	X
Access to 3GPP system CS based services with seamless mobility						X

The 3GPP SA1, TR 22.934 [7], presented their initial work defining standards for the integration of 3G and WLAN systems. The most significant result of this initial work was the definition of potential levels of Interworking requirements for UMTS and WLAN systems. Table 3.1 presents 6 scenarios exactly as stated in [7]. The scenarios range in Interworking from two completely disparate systems connected only through offline billing and support services and subsequently progress to a completely seamless integration of the two systems. There is very limited information presented beyond the basic taxonomy for any of the scenarios with the exception of scenario #2. Scenario #2 defines Interworking to consist of sharing the 3GPP provided mechanisms for access control (authentication and authorization) and for the charging or billing functionality.

3.1.2 Wireless IP Network as a Generic Platform for Location Aware Service Support (WINE GLASS)

The stated purpose of the WINE GLASS project was "to exploit enhanced and/or new IP-based techniques to support mobility and soft-guaranteed QoS in a wireless Internet architecture based on UMTS and incorporating WLANs, and to explore their potential in enabling location- and QoS-aware application services for wireless mobile users." [25] As IST was a European Union affiliated activity, their WLAN focus was on HiperLan2 which was initially more prevalent in Europe than other WLAN solutions. Although IEEE 802.11 systems were becoming increasingly common in Europe and are expected to emerge. The research developed a testbed, based on system emulation, designed to support protocol design and validation in support of location-aware applications and soft QoS application services.

3.1.3 Broadband Radio Access for IP-based Networks (BRAIN)

The BRAIN project aimed at providing a broadband extension of up to $20Mbps$ for hot spot applications into a cellular system. Their cellular target was GSM/GPRS/EDGE and UMTS. Like WINE GLASS, their research focused on an all IP realization of the cellular network. As this research was also conducted in Europe, HiperLan2 was again the broadband system employed with this work. BRAIN research sought to define and influence standard definitions for IP cellular systems and HiperLan2 hot spots support for "piconets."

Handover management, path updates and support to idle mobile hosts were the focus of their work.

3.1.4 Broadband Radio Access Network (BRAN)

The ETSI Broadband Radio Access Network (BRAN) standardization body [26] investigated, for HiperLan2, two approaches for interconnection of WLAN and UMTS networks. The first was a tight coupling scheme, to offer a seamless handover and the same level of security in WLAN and UMTS networks. This approach would require a simplified I_u interface for interconnection of WLAN network to UMTS core network. The second approach was a loose coupling scheme, which would rely on IP protocols to organize mobility and roaming between access networks. Interworking between WLAN and the core network is performed between the authentication, authorization, and accounting server (AAA) and the home location register (HLR). A major conclusion was Mobile IP and the home agent/foreign agent concept will extend mobility to any network while preserving seamless operation.

3.1.5 Vriendt et. al.

Vriendt et. al. were some of the early advocates for convergence of cellular and WLAN technologies. In [6], the authors present the evolution of mobile communications. They focus on the evolution of various radio technologies, and the evolution toward pure IP-based networks, and the interworking of varied wireless access technologies. The authors make the case that WLAN should not be perceived as a competitor or replacement for mobile cellular systems, but rather a complementing component of the network. They state that close interworking will provide the necessary mechanisms to ensure proper handling of traffic based upon the most appropriate available access network.

The authors delineate the goals of 3G cellular systems to provide more capacity, new frequencies and higher bit rates within a single global wireless architecture and how the efforts of 3GPP seek to accomplish those goals. The article provides good insight into the evolution of GSM, to include GPRS, EDGE and how it will evolve to UMTS and finally to an all IP-based network infrastructure.

Vriendt et. al. present the interworked network as three layers: the cellular layer, the hot spot layer and the personal network layer. They identify the need for a new medium access layer to interconnect the three layers and ensure transparent delivery across this architecture.

The authors, like many others, define a two level taxonomy for interworking UMTS and WLAN. These approaches are labeled a tight coupling scheme and a loose coupling scheme. The tight coupling scheme offers seamless handover and common level of security in either the WLAN or the UMTS network. This approach requires interconnection of the WLAN to the core network. The loosely coupled scheme would provide integration at the IP layer and would rely on IP layer protocols to facilitate mobility and roaming within such a network. The current trend is toward the loosely coupled scheme and the IP protocols currently supporting those efforts are mobile IP which will be discussed further in section 2.4.

The UMTS-WLAN simulator presented in this work was based upon the tightly coupled approach, and the WLAN AGG-MAC protocol could be incorporated into either type of system. The design of the interworking simulation tool was done to fully support the more complex tight coupling while remaining adaptable to support loose coupling work as well with limited modification.

3.1.6 Tsao and Lin

Tsao and Lin evaluated possible UMTS-WLAN interworking strategies in [27]. Like [6], they follow the tightly coupled and loosely coupled taxonomy, but their work employs three UMTS-WLAN interworking strategies. The authors term their strategies mobile IP approach, gateway approach, and emulator approach. The loosely coupled strategy is to employ mobile IP, and they use mobile IPv4 without route optimization for consideration of this system. The authors point out the need for mobile IP client software in every device, and the need for home agents and foreign agents in each type of network for this strategy. The authors point out the additional cost for handoff operations for this approach.

The other strategies more tightly couple the two networks. The gateway approach employs a gateway between the two networks to handle routing for users who roam from one network to another. The final and most tightly coupled approach is the emulator, in which the WLAN is merely an access stratum into the UMTS network. In this case, the authors connect a WLAN access point to an RNC emulator to access the network. The emulator approach most closely resembles our approach in that our UMTS_aware AP includes the functionality of a UMTS RNC node. The authors' simulation results and conclusions favor the emulation approach for having the least overhead during internetwork handover.

3.2 WLAN Performance

The onslaught, growth and subsequent successes of WLAN stimulated a significant amount of research in the technology. Much of this research focused on mitigating the negative affects of the wireless transmission medium. Because available bandwidth and maximum transmission power levels are constrained, many researchers have worked to limit the size of a frame in order to reduce the probability of error during the transmission of that frame. It has been shown that there is benefit, especially under noisy conditions, to limiting the size of a frame on a WLAN channel. But as in most engineering problems, trade-offs exist. Just as a frame that is too large wastes resources when retransmitted, small frames waste resources due to protocol overhead.

Wireless LAN in our work was based upon IEEE 802.11 protocol standards. This is actually a family of standards and comes in many different flavors. The original IEEE 802.11 [3] supported $1Mbps$ and $2Mbps$ data rates with frequency hopping spread spectrum. IEEE 802.11b [28] was backward compatible with the original and added support for $5.5Mbps$ and $11Mbps$ using DSSS. 802.11b is the focus of our current work. The newest variants, 802.11a [29] and 802.11g, increase the maximum data rates to $54Mbps$. The MAC layer for all versions of the 802.11 standard is based upon the well-known Ethernet paradigm. Unlike wired Ethernet, a WLAN station is unable to listen during transmission to determine whether or not a frame transmission was successful. For this reason, a collision avoidance mechanism is employed by the protocol, CSMA/CA. The protocol utilizes interframe spacing (IFS) to provide a required period of inactivity between frames in order to avoid collisions. This IFS must be considered a part of the protocol overhead as it occupies the channel and reduces capacity available for data transmission. This increases the per frame communications overhead incurred by a transmitted frame.

Several researchers have conducted work to improve WLAN MAC layer performance. Research in WLAN MAC layer performance improvements can be classified into basically three areas of interest: higher-layer protocols, backoff or contention algorithms, and frame sizes. Most accepted work in the area of higher layer protocols has focused primarily on improving TCP performance for a wireless system in which generally bursty transmission errors rather than congestion cause frame loss. [30] compared end-to-end solutions, link-layer solutions, and split protocol solutions and recommended enhancements to the link layer to improve TCP performance. Research in [31] attempts to address bidirectional flow of packets between a source and destination. This work modified the contention based DCF mechanism to support embedding the reservation of the next transmission slot in the MAC layer ACK. Their argument was that this would support TCP ACKs, but it would seem that insufficient time would occur following reception of a frame to immediately respond with an ACK. This would hold more promise in a bidirectional traffic flow type of scenario.

Several researchers have looked at mechanisms to improve performance under contention. One of the first was [32]; the authors proposed a dynamic Adaptive Contention Window based upon estimated number of stations to dynamically control the size of the backoff window.

It is well understood that frame size affects network performance. As discussed in 2.3.9, the original IEEE 802.11 specification defines a fragmentation threshold which establishes an upper limit on the size of a frame transmitted over the WLAN. In [3], the fragmentation threshold is a static parameter. Because channel conditions such as noise and interference are chiefly responsible for packet loss in WLAN systems, a number of researchers are working to establish mechanisms to dynamically adjust frame size based upon current channel conditions [33][34]. Most of the available research seeks to reduce the maximum allowable frame size in order to improve the probability of successful transmission. Any of these works are a good compliment to our research, but the focus of this research is to increase frame size in order to limit wasted potential caused by protocol overhead. This work recognizes that an appropriate value of fragmentation threshold should be the upper bound on frame size, but that the MAC layer should increase the frame size up to such a limit where possible.

3.2.1 Adaptive Fragmentation

As previously identified, fragmentation of WLAN frames that are too large for a reasonable probability of successful transmission has been widely recognized as an important performance consideration. Fragmentation mechanisms, albeit static, are included in the original 802.11 IEEE standard [3]. The standard defines a "Fragmentation Threshold" parameter, ranging in size from $256 Bytes$ up to the maximum allowable size of the MAC Service Data Unit (MSDU), currently $2,304 Bytes$. The fragmentation threshold as defined in the IEEE standard is a fixed parameter to be set at system configuration. Additional research is ongoing to support the dynamic sizing of WLAN fragments and respond to changing channel characteristics.

3.2.2 Lettieri and Srivastava

One work that is especially relevant to our initial idea for AGG-MAC was [33]. This work provided the initial impetus for us to consider aggregation of smaller packets to generate the optimal or near optimal frame size to balance the negative effects of protocol overhead and error rates. In [33], goodput is defined to be "the throughput the user will see." Although the authors argued for the need to limit the size of large frames, they provided the foundation for our contention that frames sizes too small are also problematic. The focus in [33] was to reduce the size of a frame, causing the operating point to the right of the elbow of the curve in Figure 3.1 [33], to be reduced down to a smaller, more optimal frame size through fragmentation. Our work sought to move the operating point from the left of the elbow of the curves in Figure 3.1 to a larger, more optimal frame size through packet aggregation.

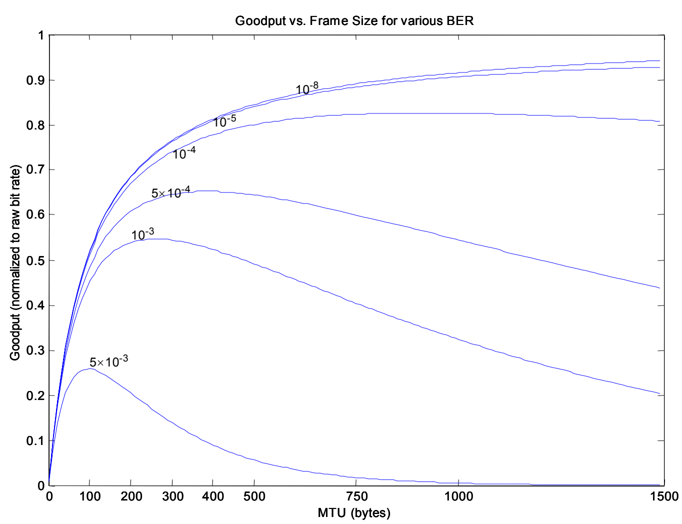

Figure 3.1: WLAN Goodput vs. MSDU for various BER

3.2.3 Reduction of Fixed Overhead

Tourrilhes

Jean Tourrilhes was a researcher at Hewlett-Packard and maintained the LINUX driver set for wireless LANs. [35] presented a scheme to concatenate a sequence of frames into a frame burst. The basic concept was to permit only a SIFs between frames of a frame burst in order to retain control of the medium and avoid releasing to contention. This research only considered the DCF operational mode. Throughput and latency were the primary metrics. The factors considered were load and frame size. The results indicate that WLAN MAC protocols are not very efficient for small packet sizes and the authors propose their Packet Frame Grouping as a possible solution to the problem.

IEEE 802.11e

The IEEE 802.11e draft standard was designed to augment the existing WLAN standards to integrate support for QoS. This draft standard was included in Related Work because it is currently a work in progress subject to changes rather than a standard point of reference. The 802.11e working group released a draft copy of the standard [18]. The current draft contains a significant number of changes from the previous, indicating that the standard has not yet solidified and may yet be far from ratification.

The relevant portion of IEEE 802.11e is actually one that was omitted from further consideration in the standard. The working group considered frame level aggregation, also called containerization, during their design process. [22] are minutes from a teleconference dedicated to aggregation issues in which their concept is delineated. The intent was to encapsulate multiple MAC frames into a single container frame. According to the minutes [22] and subsequent notes available on the IEEE 802.11 web site, aggregation was omitted due to the increased complexities of properly handling multiple layers of WLAN MAC headers.

Some of the issues cited were: whether CRC was done on the internal frames or on the aggregate frame, whether an ACK was required for each internal frame, and how to define the length field and sequence numbers of the container frame.

3.3 Summary

This chapter presented an overview of related work. In Section 3.1, we presented work related to the interworking of UMTS and WLAN systems to support "hot spots" in a cellular domain. We presented worked related to WLAN performance in Section 3.2. The next two chapters covers our work in UMTS and WLAN technologies in a tightly-coupled interworked environment. Chapters 6 and 7 presents our AGG-MAC work to improve WLAN performance and Chapter 8 presents identifies the remaining experimentation currently planned in the completion of this research effort.

Chapter 4

UMTS-WLAN Design and Methodology

This chapter presents the methodology used throughout this research and the design of the UMTS-WLAN interworking simulation model. As discussed in Section 1.5, the purpose of the first portion of this research effort was to create a network system-level simulation to allow investigators to study the issues and tradeoffs for interworking the infrastructure-based wireless LAN (WLAN) technologies into UMTS. The key contribution of this research was the UMTS-WLAN simulation environment, specifically the enhanced user equipment node model (UW) and the 3G-aware WLAN AP (UWLAN_AP). The techniques for interworking these two technologies are described in Sections 4.9.4 and 4.9.2.

The interworked UMTS-WLAN research was conducted jointly with masters student Tracy Mann. During the course of my studies, I guided him in the specification, design, model implementation and initial testing of the simulation model set. Our initial implementation was presented in [36] and was well received. Following the updated implementation and subsequent validation and verification testing, Tracy and I outlined the testing design and he went on to further study the performance of the UMTS-WLAN interworked system based on our models while my research efforts moved forward in the WLAN MAC layer enhancements presented later in Chapter 6. The design and model implementation results for the UMTS-WLAN interworking simulation model presented herein were also presented in his Masters thesis [8]. The reader is urged to read [8] for additional performance related results of our UMTS-WLAN interworked model set.

Selecting an appropriate, proven research methodology is a critical step in any research endeavor. Resource constraints were prohibitive for implementing and testing the interworking of UMTS and WLAN in prototype 3G wireless equipment. Therefore, a simulation model was designed as the primary goal of this research. The simulation model was developed using the commercial discrete simulation package OPNET Modeler^{TM}. Initial implementation was done in version 9.0.A and updated to OPNET^{TM} version 9.1.A. The sections in this chapter describe the simulation model developed for this research effort. Jain presents a ten-step method of systematic performance evaluation, which is well suited for evaluating the performance of a communications system through simulation [37]. This method was used to create the UMTS-WLAN simulation environment.

1. State goals and define the system
2. List services and outcomes
3. Select metrics
4. List parameters
5. Select factors to study
6. Select evaluation technique
7. Select workload
8. Design experiments

9. Analyze and interpret data

10. Present results

4.1 System Definition and Assumptions

The purpose of this portion of the research was to develop a network system level simulation (shown in Figure 4.1) to allow investigators to study the effects of interworking WLAN and UMTS. The research problem was defined in Section 1.1.. OPNET Modeler™ was chosen as the simulation environment due to its flexibility and extensive model library sets. The author created user-defined models as well as modified and augmented the existing models in order to develop the UMTS-WLAN simulation environment. This research effort leveraged the existing OPNET™ model libraries, specifically the UMTS and the 802.11b WLAN model sets. Several assumptions were necessary to limit the scope of the problem. The intent of these limiting assumptions was to keep the simulation complexity manageable, while still meeting the research goals. This section describes assumptions made in modeling both the UMTS and WLAN data networks, as well as the interworking of these two technologies.

4.1.1 UMTS Model Assumptions

The UMTS-WLAN simulation environment designed in the research leverages the OPNET™ UMTS model set. The following sections describe the assumptions of this model set.

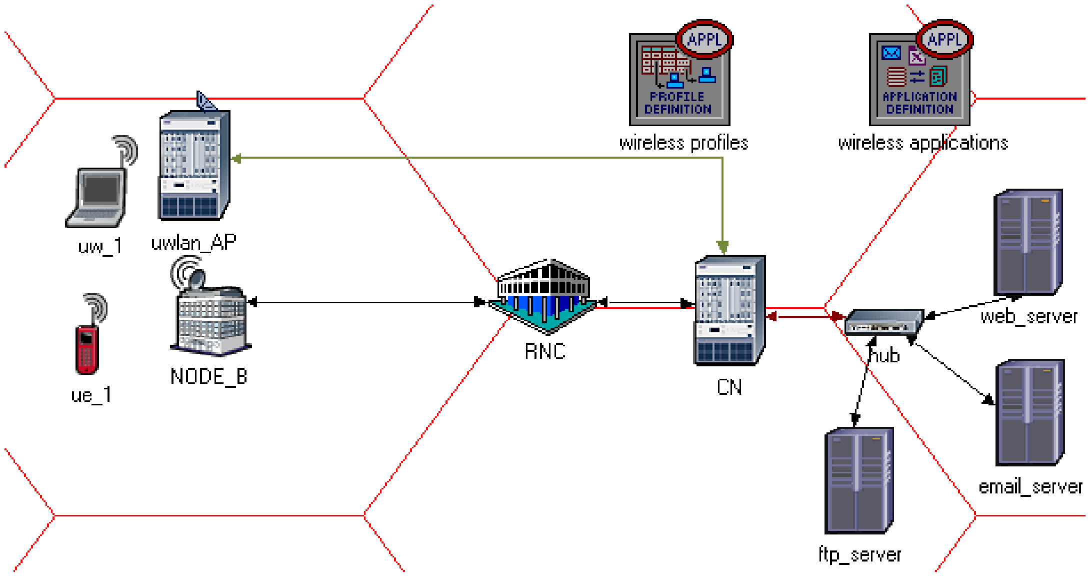

Figure 4.1: UMTS-WLAN Network System Level Simulation

Packet Switch Domain

The OPNET™ UMTS simulation environment models the packet domain of the 3GPP Release 1999 UMTS standard. The voice, circuit switched (CS) domain is not explicitly modeled. The focus of this research is the interworking of WLAN to provide data services; therefore, the CS domain is not required.

The main system components of the UMTS packet domain architecture, described in detail in Section 2.2, are modeled as the following OPNET™ nodes: the user equipment (UE), the Node-B, the Radio Network Controller (RNC) and the core network (CN).

The UE node models the full TCP/IP stack with both data/multimedia application models and hi-fidelity TCP/IP layer models, the GMM/SM layer, the RLC/MAC layer, and the physical/air-interface layer. The GMM/SM layer handles both mobility management and session management. The RLC/MAC layer models the three transport modes, the four transport channels, and the segmentation and reassembly of higher layer data discussed in Section 2.2.3.

The Node-B models the radio interface (physical/air-interface) with the UE, the radio link management functionality, and the ATM interface layer that provides connectivity to the RNC. The RNC models the RLC/MAC layer protocol interaction with the UE, handles both UE admission control and radio bearer (RB) assignment, and provides the ATM interfaces to the Node-B and SGSN.

The CN models the GMM/SM layer protocol interaction with the UE, handles GPRS attach, PDP context activation, and Service requests, and provides both an IP interface to the Internet and ATM interfaces to multiple RNCs. It models the CN functionality in the SGSN. The GGSN as well as the tunneling between the SGSN and GGSN are not explicitly modeled. The IP gateway functionality is modeled with an IP router stack connected to the SGSN.

GPRS Mobility Management / Session Management

With the exception of the GPRS attach procedure, the UE synchronization that occurs when a user powers-on is not modeled. The PS (packet-switched) signaling connection is not modeled explicitly. The model assumes that a RRC Connection Setup procedure occurs when a user powers-on and thus a PS signaling connection is established. This connection is maintained for the entire simulation. The GPRS detach procedure is not modeled—a UE will remain attached for the entire simulation [38].

Upon receipt of PDUs, the UE or network activates a PDP context if one is not already activated. The PDP context activation includes the requested QoS profile associated with its class of traffic. Once activated, a PDP context remains active for the remainder of the simulation. Multiple PDP contexts per QoS, and the PDP context deactivation and reactivation procedures are not modeled. Only one PDP context for each QoS is modeled. The PDP context is not deactivated and is reused for subsequent data traffic associated with the QoS. The Service Request procedure is used to establish a RAB for a PDP that is active.

Radio Resource Management

Both the RAB Setup and Release procedures are modeled. As stated above, the model assumes that a PS signaling connection is already established for the PDP context activation procedure. RABs are set up by the network and are later released after being idle for a period of time. The model does not support QoS negotiation; the SGSN model either grants the UEs requested QoS in its entirety or rejects the request.

UMTS Air Interface

The WCDMA air interface for the FDD mode only is modeled. OPNET's Wireless$^{\mathrm{TM}}$ module includes 13 pipeline stages to model radio interfaces. The UMTS model set developers modified the following pipeline stages to model specific WCDMA air interface behavior: received power, background noise, interference noise, bit error rate. The packet dropping probability is based on curves obtained from another set of simulations of the WCDMA air interface. This simulation was accurate to the waveform level [38].

4.1.2 WLAN Model Assumptions

The UMTS-WLAN simulation environment designed in the research leverages the OPNET$^{\mathrm{TM}}$ WLAN model set. The OPNET$^{\mathrm{TM}}$ WLAN simulation environment models the WLAN MAC and PHY specifications of the IEEE 802.11b Release 1999 standard. The following sections describe the assumptions of this model set.

The main system components of the WLAN infrastructure-based network architecture, described in detail in Section 2.3.1, are the station and the WLAN AP (access point) modeled as OPNET^{TM} nodes.

The Station models the full TCP/IP stack with application models and hi-fidelity TCP/IP layer models, the WLAN MAC layer, and the physical/air-interface layer. The

WLAN AP models an IP routing stack, the WLAN MAC layer, the WLAN physical/air-interface layer and an Ethernet MAC and physical layer. The OPNET^{TM} 802.11 model provides a complete representation of the MAC layer protocol, but much of the PHY is abstracted.

Medium Access Control

The WLAN MAC supports the three access mechanisms described in Section 2.3.5: DCF, RTS/CTS, and PCF. The interframe spacings (IFS) and binary exponential backoff are modified based on the selection of the physical layer. The frame exchange sequence is modeled in accordance with the standard requiring a data and acknowledgement exchange to ensure the reliability of data transfer. The model supports data rates of 1, 2, 5.5, and 11 Mbps.

The WLAN MAC model provides tunable attributes based on the physical layer impacts on the MAC protocols.

WLAN Air Interface

The WLAN air interface is not explicitly modeled. It is modeled with the thirteen pipeline stages of the OPNET's Wireless^{TM} module. The WLAN model set developers modified the following pipeline stages to model the specific WLAN air interface behavior: radio receiver, channel match, propagation delay, and error correction stages.

4.1.3 UMTS-WLAN Interworking Assumptions

Coupling Scheme

Using the coupling taxonomy defined in [6], the design approach presented in this research used the tight coupling scheme. The UMTS-WLAN system was tightly coupled at the RNC using the WLAN technology as an alternate radio access technology for "hot spots." The physical layer radio interface and the transport network layer protocols of the RNC were replaced with WLAN MAC and PHY, while the system network and radio network layer protocols remained with little modification. This provided a simplified interface for interworking the WLAN network with the UMTS core network.

The design decision to use the tight coupling scheme was made for two reasons. First, the author considers the loosely coupled scheme to be a "solved problem" by Lucent Technologies, Inc., as stated [39]. Second, the loosely coupled scheme would integrate the two systems at the IP layer and would rely on the IP protocols, specifically Mobile IP, to handle mobility and roaming between the access networks. This internetwork mobility was beyond the scope of this research.

The tightly coupled design leveraged the mobility management features of UMTS and therefore, required a Mobile IP implementation to handle mobility and roaming. The design using the tight coupling scheme created a controlled, baseline simulation framework to study the interworking of WLAN and UMTS. This simulation framework was designed to allow future investigators to easily modify the models in order to investigate the loosely coupled scheme.

Technology Generation Independence

The simulation model was designed using the 3GPP UMTS Release 1999 as the representative cellular phone technology and the IEEE 802.11b Release 1999 as the representative

WLAN technology. The model was designed so that as future technologies are released, the effect of these technologies can be studied with minimal modifications to the current design.

Mobility and Handoff

The simulation does not model UW mobility or call handoff between the UWLAN_AP and the UMTS system. Modeling call handoff for IP data users between the two technologies would require extensive modifications to TCP in order to "throttle back" the transmission rate in order to facilitate handoff from WLAN to UMTS. These modifications would enhance the model, but were not within the scope of this research. The intent of the open architecture design of this simulation model was to allow future investigators to modify the current model set in order to study interworking affects such as this.

4.2 System Services and Outcomes

The UMTS-WLAN simulation environment will allow an investigator to study the protocol effects and evaluate the overall system performance for an interworked UMTS-WLAN system. The specific statistics and effects that can be studied with this simulation environment are the control plane signaling protocol interaction, the UMTS dedicated channels (DCH) utilization, and the application response time for a variety of application traffic.

As with the 3GPP UMTS standards, the UMTS-WLAN model separates the user and control planes so that protocol enhancements and interactions can be studied for both data and signaling messages independently. The model was designed so that it can be extended to allow investigators to study the protocol effects of virtually any component or system service.

4.3 Performance Metrics

The UMTS-WLAN system was evaluated based upon protocol interaction, dedicated channel utilization (DCH), data session set-up delay, and application response time (FTP download response time and Web page response time). These performance metrics were defined as follows:

4.3.1 Protocol Interaction

Protocol interaction was defined as the control plane signaling messages required to complete a system level control transaction (i.e., GPRS Attach). The protocol interaction was expressed as a sequence of signaling messages between system nodes required to complete a transaction. The protocol interaction metric was used in the model verification process to ensure that the simulation accurately modeled the 3GPP UMTS and IEEE 802.11 WLAN protocol standards.

4.3.2 DCH Utilization

DCH utilization was defined as the number of active UMTS dedicated channels. DCH utilization was measured in number of channels, and a smaller value was considered better.

The DCH utilization was used to demonstrate how the proposed interworked UMTS-WLAN system allows the conservation of the limited UMTS cellular capacity.

4.3.3 Data Session Setup Delay

Data session setup delay was defined as the time between a client requesting an application service to the time when the first response packet was received. The data session setup delay was measured in seconds, and a smaller value was considered better. The data session setup delay included the time to establish a TCP connection, a PDP context, a Radio Access Bearer and process a Service Request as applicable. The data session setup delay was used to demonstrate how the UMTS-WLAN system differed from the existing WLAN system, specifically it demonstrated the additional connection setup delay as a result of the UMTS signaling procedures.

4.3.4 Application Response Time

FTP and Web application response times were chosen because they belong to two different UMTS QoS profiles. Both response times were measured in seconds, and a smaller value was considered better.

FTP download response time was defined as the time that elapses between a client sending a "get" request and receiving the entire file. It was measured from the time a client application sent a request to the server to the time it received the file. The FTP download response time included the time required to transmit the entire file—it was dependent on the file size. This time included the signaling delay for connection setup. As described in Section 4.8, FTP download response time was used to demonstrate how the UMTS-WLAN system differs from the existing UMTS and WLAN systems. It was also used to show how the UMTS-WLAN system performs over a range of normal operating conditions.

Web page response time was defined as the time required to retrieve an entire HTML page with all of its inline objects. Similar to the FTP response time, the web page response time was measured from the time a client browser application sent the request to the web server and ends when the client received the entire HTML page.

Both application delay performance measures were used to demonstrate the benefit of shifting data users to the WLAN access network in terms of reduced application delays. These performance measures were also used to demonstrate the capabilities of the inter-worked UMTS-WLAN system over a range of normal operating conditions.

4.4 Simulation Parameters

Inputs to the simulation model that were not varied during different simulation runs are termed simulation parameters. The values selected for these parameters affected how accurately the simulation modeled the actual system. The simulation parameters are given in Table 4.1, and are discussed below.

Table 4.1: UMTS-WLAN Simulation Parameters

Simulation Parameters	**Values**
WLAN PCF Functionality	Enabled
WLAN Access Point Functionality	Enabled
WLAN Physical Layer Characteristics	Direct Sequence
WLAN Data Rate	$11 Mbps$
Beacon Interval	0.02
CFP Interval (sec)	0.018
CFP Beacon Multiple	1
UMTS Cell State	DCH

4.4.1 WLAN PCF Functionality

The simulation was run with the WLAN PCF (Point Coordination Function) functionality enabled. This value specified whether the Point Coordination Function was enabled for the WLAN MAC. When enabled for the WLAN AP, its MAC acted as the point coordinator for the BSS. The WLAN AP polled client nodes with the WLAN PCF functionality enabled during the CFP, contention free period.

4.4.2 WLAN Access Point Functionality

The simulation was run with the WLAN Access Point functionality enabled for the UWLAN_AP and disabled for all UWs. This attribute was used to assign the WLAN MAC within the UWLAN_AP as the access point of its BSS. BSSs that deploy PCF required an access point.

4.4.3 WLAN Physical Layer Characteristics

The simulation was run with the WLAN physical layer characteristics set to Direct Sequence Spread Spectrum. The value of this attribute determined the physical layer technology in use. As stated in Section 4.1.2, the WLAN PHY was not modeled explicitly; the WLAN MAC configured the values of the following parameters as indicated in the IEEE 802.11 WLAN standard:

a) SIFS time,

b) SLOT time,

c) Minimum and Maximum contention window sizes,

d) and any other parameter value derived from the values of these parameters (like DIFS).

4.4.4 WLAN Data Rate

The simulation was run with the WLAN data rate set to 11 Mbps. This value specified the data rate that was used by the MAC for the transmission of the data frames via the physical layer. The WLAN model set supported data rates specified in IEEE 802.11b standards.

4.4.5 CFP and Beacon Intervals

The beacon interval specified the $TBTT$ (Target Beacon Transmission Time) in seconds. The AP attempted to transmit beacon frames (and subsequently mark the beginning of a CFP for a CFP beacon multiple of 1) with a period specified by the value of this attribute. As stated in Section 2.3.8, the actual start of a CFP can be slightly off, since the AP may need to wait for the shared medium to become idle before it can transmit the beacon frame.

The CFP interval specified the maximum duration of the CFP in seconds. The value of the CFP interval was dependent on the beacon interval and the CFP beacon multiple. Based on these values, the CFP interval was calculated so that the duration of the CFP was maximized and the CP, contention period, minimized. The reason that the duration of the CFP was maximized was that in an infrastructure BSS (i.e. WLAN "Hot Spots"), most of the client traffic was directed to the Internet via the AP. Therefore, one of the goals of the system was to maximize throughput to/from the AP, and the way to accomplish this was to minimize contention for the WLAN channel by controlling access as much as possible (i.e. minimize the CP window).

The IEEE 802.11 Standard [3] stated that the maximum CFP interval must be limited to allow the coexistence between the CP and the CFP (Equation 4.1).

$$Max\_CFP\_Interval = TBTT - Min\_CP\_Dur \tag{4.1}$$

The $TBTT$ was defined in Equation 4.2.

$$TBTT = Beacon\_Interval * CFP\_Beacon\_Multiple \tag{4.2}$$

The minimum CP duration required sufficient time to send at least one data frame and receive a positive ACK. To send a data frame during the CP, a station must wait a period of a DCF interframe spacing (DIFS) to sense if the medium is idle. If idle, station A will transmit the data frame to station B. After receiving the entire data frame, station B will wait a period of a short interframe spacing (SIFS) and then transmit a positive ACK. Equation 4.3 defines this relationship for the minimum CP duration.

$$Max\_CP\_Dur = DIFS + Data + SIFS + ACK \tag{4.3}$$

Using the parameter values specified in Table 4.1, the optimal CFP Interval was calculated to be 0.018 seconds.

4.4.6 UMTS Cell State

The simulation was run with the UEs and UWs in the UMTS DCH cell state. This value indicated that all UMTS uplink and downlink traffic was sent on dedicated channels (DCH).

4.5 Simulation Factors

Inputs to the simulation that are varied during different simulation runs are termed simulation factors. The simulation was run with different combinations of these factors as described in Section 4.8. The simulation factors are summarized in Table 4.2 and discussed below.

Table 4.2: UMTS-WLAN Simulation Factors

Simulation Factor	**Values**
Application Profile/Traffic Model	Wireless Client Light and FTP Fixed File Size
FTP File Size	$1 - 100 MBytes$
Client Access Mode	UMTS, WLAN
Number of Clients	1, 2, 10, 16, 20, 25, 28 and 30

4.5.1 Application Profile

The simulation was run using the application profiles “Wireless Client Light,” “Wireless Client Heavy” and “FTP Fixed File Size” as specified in Section 4.7, Traffic Models.

This value specified the names of the application profile that were enabled for a UW.

Each application profile was defined in the profile configuration object at the OPNET^{TM} network level. An application profile described user behavior in terms of what applications were being used and the amount of traffic each application generated. More than one profile can be configured for a particular client.

4.5.2 FTP File Size

The simulation was run varying the FTP file size from 1 to $100 MBytes$. The combination of the Application Profile, FTP file size, number of clients, and their corresponding client access mode determined the traffic loads for both the WLAN channel and the UMTS DCH channels. These combinations were varied in order to exercise the model to demonstrate the system's performance over a range of normal operating conditions.

4.5.3 Client Access Mode

The simulation was run with the UW in the UMTS mode and the WLAN mode. This value was used to determine if the UW used the WLAN or UMTS interface. In the UMTS mode, the UW functioned as a UMTS UE. In the WLAN mode, the UW used the WLAN interface to gain access to the UMTS PS services via the UWLAN_AP.

4.5.4 Number of Clients

The simulation was run with 1, 2, 10, 16, 20, 25, 28 and 30 clients. These values represented the number of clients sharing access to the WLAN channel or sharing UMTS DCH channels depending on the Client Access Mode setting. These values were selected because they represent the number of clients that a typical UMTS cell or WLAN AP is expected to support. The combination of the application profile, packet size, number of clients, and their corresponding client access mode setting determined the traffic loads for both the WLAN channel and the UMTS DCH channels.

4.6 Evaluation Technique

The selection of a particular evaluation technique can significantly impact the outcome of a performance evaluation. Three possible techniques of performance evaluation are analytic, simulation, and measurement [37]. These methods differ in terms of accuracy, cost, and required time. Based upon these factors, simulation was the most appropriate technique for this research effort. Measurement was easily ruled out as a feasible technique based upon both cost and required time. Commercial 3G equipment was not yet available, and the development of prototype 3G hardware was not possible within the financial and time constraints of this project. Analytic solutions typically offer less accuracy than simulation, but are also less costly and often more time consuming. In this case, the cost of simulation was negligible because the hardware and software required was already on-hand. Therefore, simulation was used to conduct this performance analysis. Analytical methods were used in the model verification process.

4.7 Traffic Models

The simulation model used the OPNET^{TM} built-in application distribution models. The built-in OPNET^{TM} application profiles were used to closely simulate traffic generated by a wireless data user. The application profiles used were the FTP, E-mail, and HTTP profiles. These profiles, discussed below, were combined and parameterized in order to define the Wireless Client application profile. The Wireless Client application profile provided a realistic model of application data traffic, but could not be analyzed with traditional queuing theory because the stochastic nature of the traffic does not offer closed form solutions.

To facilitate the verification process, the FTP application was parameterized such that a client requested a single file transfer. This configuration might provide a less realistic model of actual FTP application data traffic, but allowed the results to be analyzed using analytical methods and was useful in verifying correct operation of the model.

4.7.1 Wireless Client Traffic Model

The Wireless Client traffic model was defined by using the OPNET™ Standard Network Application models. The Standard Network Application models are a set of models that capture specific characteristics of the application that they represent. They are defined within the application configuration object. The built-in standard network applications are implemented in a two-tier (client-server) architecture, wherein: (1) the client issues a request; and (2) the server receives the request and returns a response. The request-response exchange, termed a conversation, represent a sequence of activity between the client and the server within the context of a given application. The conversations include a pattern of data exchanges that are defined in a statistical manner to repeat over time [40].

The Wireless Client profile described a wireless data user's activity over a period of time. The profile consisted of the standard network applications: FTP, E-mail, and HTTP. The applications were parameterized in order to create a Wireless Client profile that was representative of the data traffic generated by a typical wireless data user. These profiles were used to generate traffic across the network.

4.7.2 File Transfer Protocol (FTP) Application

The standard OPNET™ FTP application allows file transfers between a client and server. It permits the user to issue two basic commands for transferring a file: "get" and "put." TCP is the default transport protocol for this application. The FTP application was parameterized using the attributes in Table 4.3 [40].

Table 4.3: UMTS-WLAN FTP Application Attributes

Attribute	**Description**
Command Mix (get/total)	Ratio of "get" (downloads) to the total number of commands
Inter-Request Time	Time between subsequent file requests
File Size (Bytes)	Average size of a file being transferred
Type of Service	QoS parameter for assigning priority to this application's traffic

4.7.3 E-mail Application

TCP is the default transport protocol used in the standard OPNET™ E-mail application. SMTP (Simple Mail Transport Protocol) and POP (Post Office Protocol), which usually use TCP as the underlying transport protocol are not explicitly modeled. E-mail is modeled as a single TCP connection between the client and the server, and the data is transferred based on the configured send and receive attributes. The message transfer is modeled between the client and the server, not from a client to another client. The E-mail application was parameterized using the attributes in Table 4.4 [40].

Table 4.4: UMTS-WLAN E-mail Application Attributes

Attribute	**Description**
Send Interarrival Time (sec)	Time between E-mails sent from the client to the server
Send Group Size	Number of E-mail messages grouped before transmission
Rcv Interarrival Time (sec)	Time between E-mails rcv'd at the client from the server
Receive Group Size	Number of E-mail messages grouped before reception
E-mail Size (bytes)	Average size of an E-mail message
Type of Service	QoS parameter for assigning priority to this application's traffic

4.7.4 HTTP (Web) Application

The HTTP application models web browsing. The user downloads a page from a server. The page contains text and graphics (termed inline objects). TCP is the default transport protocol. Each HTTP page request may result in opening multiple TCP connections for transferring the contents of the inline objects embedded in the page. The Web application was parameterized using the attributes in Table 4.5 [40].

Table 4.5: UMTS-WLAN HTTP (Web) Application Attributes

Attribute	**Description**
HTTP Specification	HTTP Version
Max Connections	Max number of simultaneous connections
Max idle period	Max idle time before a connection is torn down
Pipeline Buffer Size(requests)	Number of HTTP request that can be buffered together into a single application message
Page Interarrival Time (sec)	Time between subsequent pages that a user browses
Page Properties- Object Size (bytes/object)	Average size of an object
Num Objects (Object/page)	Number of objects per page
Server Selection- Initial Repeat Probability	Probability that the user will request the next page from the same server
Pages per Server	Number of pages accessed consecutively on the same server
Type of Service	QoS parameter for assigning priority to this application's traffic

4.8 Simulation Scenarios

The simulation was run under three different scenarios termed Existing WLAN vs UMTS-WLAN Scenario, Existing UMTS vs UMTS-WLAN Scenario, and Mixed Client Access WLAN—UMTS Scenario. The primary difference between these scenarios was the access network used. All three scenarios varied the simulation factors discussed in Section 4.5 and the traffic models discussed in Section 4.7 in order to vary the network traffic loads. The performance metrics defined in Section 4.3 were used to measure performance in each scenario.

4.8.1 Existing WLAN vs UMTS-WLAN Scenario

The Existing WLAN vs UMTS-WLAN scenario simulated a single wireless client data user running the FTP application. The purpose of this scenario was to demonstrate how the existing WLAN differs from the proposed interworked UMTS-WLAN system. Both systems were evaluated in terms of data session set-up time and application response time. The FTP file size was varied in order to systematically vary the network traffic load.

The existing WLAN system simulated a single WLAN client accessing the network using a WLAN access point (AP). The UMTS-WLAN system simulated a single UW in the WLAN Client Access Mode accessing the network via the UWLAN_AP.

The data session set-up time was chosen as a performance measure because it demonstrated the additional set-up delay that a UW encountered as a result of the UMTS signaling procedures. The FTP download response time was chosen as a performance measure in this scenario because it demonstrated how UMTS-WLAN system differs from the existing WLAN systems in terms of application response time. Specifically, it demonstrated the accumulative application delay for large file transfers as a result of the PDP header overhead.

4.8.2 Existing UMTS vs UMTS-WLAN Scenario

The Existing UMTS vs UMTS-WLAN scenario simulated a number of wireless client data users running a mix of client applications. The purpose of this scenario was to demonstrate how the existing UMTS system differs from the proposed interworked UMTS-WLAN system. Both systems were evaluated in terms of their DCH utilization and application response time.

The existing UMTS system simulated a number (1, 2, 4, 6,. . . , 18, 20) of UEs accessing the network using UMTS dedication channels (DCH). The UMTS-WLAN system simulated a number of UWs accessing the UMTS network via the UWLAN_AP.

The DCH utilization was measured for both systems with 20 clients running the Wireless Client application profile. The DCH utilization was chosen as a performance measure because it demonstrated how the proposed interworked UMTS-WLAN system differs from the existing UMTS system in terms cellular capacity utilization.

The FTP download response time was measured for both systems for a single user running the FTP application. The FTP file size was varied in order to systematically vary the network traffic load. The FTP download response time was chosen as a performance measure in this scenario because it demonstrated how the UMTS-WLAN system differs from the existing UMTS system in terms of application response time. Specifically, it demonstrated the significant differences in application response times for even relatively small file sizes.

4.8.3 Mixed Client Access WLAN—UMTS Scenario

The Mixed Client Access UMTS-WLAN scenario simulated twenty UWs running a UMTS parameterized Wireless Client application profile. The UWs accessed the UMTS data services via both the UWLAN-AP and the UMTS access network. The purpose of this scenario was to exercise the interworked UMTS-WLAN system over a range of normal operating conditions in order to demonstrate system capabilities. The client access modes were varied from 20 WLAN and 0 UMTS to 0 WLAN and 20 UMTS varying the client access mode by 2 each simulation run. The DCH utilization and application response time for both FTP and HTTP traffic were collected.

Varying the client access mode had the result of systematically varying the network traffic load for both the UWLAN_AP and the UMTS access networks. The results of this scenario were analyzed in terms of DCH utilization, FTP and HTTP application response time.

The DCH utilization was chosen as a performance measure because it demonstrated how the proposed interworked UMTS-WLAN system allows wireless data users to be shifted to the WLAN access network so that cellular capacity is conserved and available to support cellular voice subscribers.

Both the HTTP and FTP application response times were chosen as a performance measures because they both belong to different QoS profiles. This allowed us to analyze the effect of varying the network traffic load in terms of application response time.

4.9 UMTS-WLAN Interworked System Design

The simulation was designed in OPNET Modeler^{TM} using a top-down design approach. Modeler^{TM} uses a hierarchical structure of network scenarios, nodes and processes. The top level of the simulation graphically depicted the network containing the Enhanced WLAN Access Point (termed UWLAN_AP), the Enhanced User Equipment (termed UW), the Core Node (CN), the Radio Network Controller (RNC), the Node Base Station (Node-B), the UMTS User Equipment (UE), and the Application server. The UWLAN_AP and UW were designed at the node level using both built-in OPNET^{TM} processes and user-defined processes. User defined processes, UWLAN_AP Controller and UW_MAC_IF_Control, were defined at the process level using state diagrams. The simulation actions in each state of the user-defined processes were defined with a combination of C code and built-in OPNET functions. This section presents an overview of the simulation design at the network, node, and process levels.

The UMTS-WLAN system was designed in accordance with the second scenario as defined by the 3GPP Feasibility Study for Interworking WLAN into 3G Wireless Systems [7]. This design leveraged the resources of the OPNET^{TM} UMTS specialized model set as the representative 3G cellular system [1] and 802.11b model set as the representative WLAN system.

The design created an enhanced UE (termed UW- discussed in detail in Section 4.9.4).

The UW was augmented with the capability to selectively gain network access through either a UMTS Node-B or through a 3G-aware WLAN network The WLAN network was made 3G aware by tightly coupling the WLAN access point (AP) at the UMTS Radio Network Controller (RNC) to create an enhanced WLAN access point (termed UWLAN_AP - discussed in detail in Section 4.9.2). The UWLAN_AP was augmented with the RNC functionality and the capability to process UMTS control messages in order to build an access control table to support UMTS authentication and access control. Together, the UW and UWLAN_AP created a simulation framework for interworking the WLAN technology into UMTS as an alternate radio access network for supporting "hot spots." This would conceptually be an airport, or office building or other indoor area with high concentration of wireless data users.

4.9.1 Network Level

The simulation network level, depicted graphically in Figure 4.1, consisted of a number of UWs with the capability to access the UMTS packet network via either the UWLAN_AP or the UMTS Node-B depending on their client access mode setting. In the WLAN client access mode, the UW gains network access through the UWLAN_AP to the CN. In the UMTS client access mode, the UW gains access through the Node-B to the RNC and CN.

At simulation start up, the network initializes. Once the network is initialized, the UWs "power on" and perform a GMM_Attach procedure in order to authenticate with the UMTS network. The GPRS Mobility Management (GMM) Attach procedure, discussed in detail in Section 2.10, is performed to register the UW for the GPRS network services by notifying the SGSN of its location and to establish a PS signaling connection with the SGSN. The GMM protocol provides a logical connection between the UW and the CN, specifically the SGSN, for this procedure to occur. The OPNET™ UMTS model set explicitly models the GMM attach procedure between the UE and the SGSN. The WLAN AP had to be modified extensively in order to process the GMM control messages. The modifications are described in detail in Section 4.9.2.

4.9.2 Enhanced WLAN Access Point Node (UWLAN_AP)

The Enhanced WLAN AP node model, Figure 4.2, consisted of the WLAN medium access control (MAC) and physical (PHY) layers, a full ATM stack, a UMTS specific ATM interface module, and a user-defined process module, UWLAN_AP_Control.

The WLAN MAC and PHY are part of the standard model set and provide a detailed protocol level implementation of the 802.11 standard as discussed in Sections 2.3.5 and 4.1.2. The ATM stack implements the ATM and AAL5 protocols, specified in [15] and [16], respectively. The UMTS specialized model set provides a UMTS specific ATM interface that maps UMTS QoS traffic to ATM QoS. It also manages the virtual circuit between UMTS components.

The user-defined UWLAN_AP_Control process module is responsible for interworking the WLAN infrastructure BSS at the RNC, specifically by controlling RAB (Radio Access Bearer) and RB (Radio Bearer) installation and tear down. It also provides the UW with a mechanism for authentication and provides access control into the UMTS network. A detailed description of the UWLAN_AP_Control process model is given below.

4.9.3 UWLAN_AP_Control Process Module

The UWLAN_AP_Control process module contains the control logic to interwork WLAN into the UMTS system as an alternate radio access network. It maintains the additional state required, in the form of an Access Control Table (Figure 4.4), for the UWLAN_AP to support access control (authentication and authorization) in the UMTS system. The user-defined process model, UWLAN_AP_Control employs a state diagram to

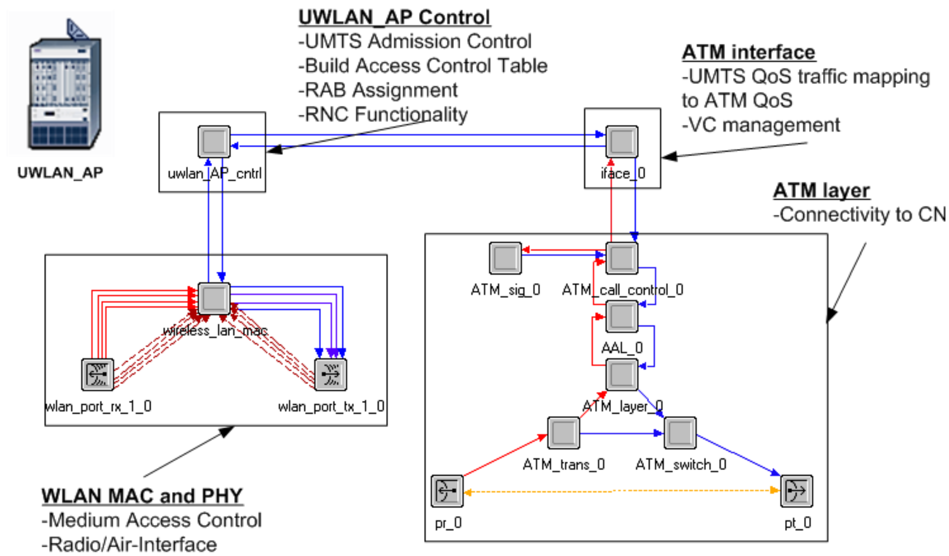

Figure 4.2: Enhanced WLAN AP (UWLAN_AP) Node Model

define the control logic. The state diagram, Figure 4.3, contains 3 initialization states and 4 control states. The function of each state is a follows:

init: This state initializes the state variable associated with the model and registers it as a process in the model-wide process registry so that statistics can be collected. The two subsequent initialization states, **init2** and **wait** schedule self-interrupts to allow the lower layer WLAN MAC and ATM PVCs to initialize and register with the model-wide process registry. Once initialized, the state machine waits in the idle state.

idle: This state processes stream interrupts associated with packet arrivals from the WLAN MAC and ATM interface. It is responsible for determining the UMTS message type (mt) and the UW's IMSI associated with the packet. It routes the packet to the From_CN and From_UW state based on id of arrival stream.

From_UW: This state performs an Access Control Table lookup based on the source IMSI to determine if the UW is authorized to transmit UMTS data traffic. It processes data packets and the signaling messages, Activate_PDP_Request and SM_Service_Request messages by forwarding them to the CN over the ATM interface. It processes the Modified_GMM_Attach_Request and RAB_Assignment_Request message by transitioning to the ADM_CNTL state.

From_CN: This state performs an Access Control Table lookup based on the destination IMSI to determine if the UW is authorized to receive UMTS data traffic. It processes data packets and the signaling messages, Activate_PDP_Context_Accept and SM_Service_Request_Accept messages by forwarding them to the UW over the WLAN interface. It processes the GMM_Attach_Complete and RAB_Assignment_Request messages by transitioning to the ADM_CNTL state.

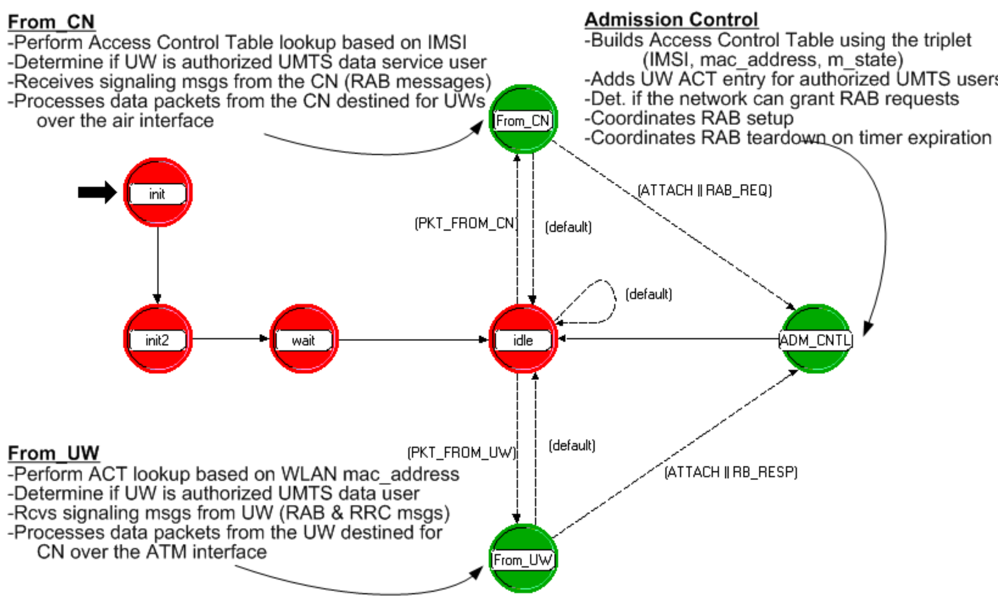

Figure 4.3: UWLAN_AP Control Process Model

ADM_CNTL: This state contains the bulk of the UWLAN_AP_Control process functionality. It handles all signaling messages that require an action be taken by the UWLAN_AP. The ADM_CNTL state builds the Access Control Table that the UWLAN_AP uses to keep track of the UMTS authorized UWs. It also coordinates RAB setup procedures. The functions of the ADM_CNTL state can best be explained by walking sequence diagrams associated with these UMTS signaling procedures.

As a UW performs a GMM Attach, the UWLAN_AP uses the Modified GMM Attach

Request message to build the Access Control Table. The UWLAN_AP adds a UW entry, Figure 4.4, using the triplet (*IMSI, MAC_Address, M_State*). The Access Control Table provides the UWLAN_AP with the ability to look at a packet. Based on its IMSI either support access or drop the packet thereby denying access. The Modified_GMM_Attach procedure is discussed in detail in Section 4.9.3. The UWLAN_AP Control process also implements the PDP_Context_Activation, and RAB_Assignment procedures both of which are discussed later in Section 4.9.3.

$$(IMSI, MAC\_Address, M\_State)$$

Figure 4.4: UWLAN_AP Access Control Table Entry

Modified GMM Attach Procedure

The GMM protocol, which logically operates between the UE and the SGSN, provides the basic signaling mechanisms for controlling mobility management and authentication into the UMTS PS domain. The GMM Attach procedure is the signaling required to implement this protocol. In order for the UWLAN_AP to build the Access Control Table and thus control access to the UMTS network, the GMM Attach procedure was modified. The sequence diagram for the Modified GMM Attach procedure is shown in Figure 4.5.

The GMM Attach process is a 3-way handshake process between the GMM modules in the UW and the SGSN, where the UW requests access, the SGSN accepts, and the UW acknowledges. The steps of the Modified GMM Attach procedure and the actions taken by the key processes are below:

1. The GMM in the UW generates a GMM_Attach_Request.

1a. The UW_MAC_IF_Control process in the UW destroys the original request and generates a Modified_GMM_Attach_Request and forwards it to the UWLAN_AP. The Modified_GMM_Attach_Request message, Figure 4.6,uses 32 bits of the 144 unused

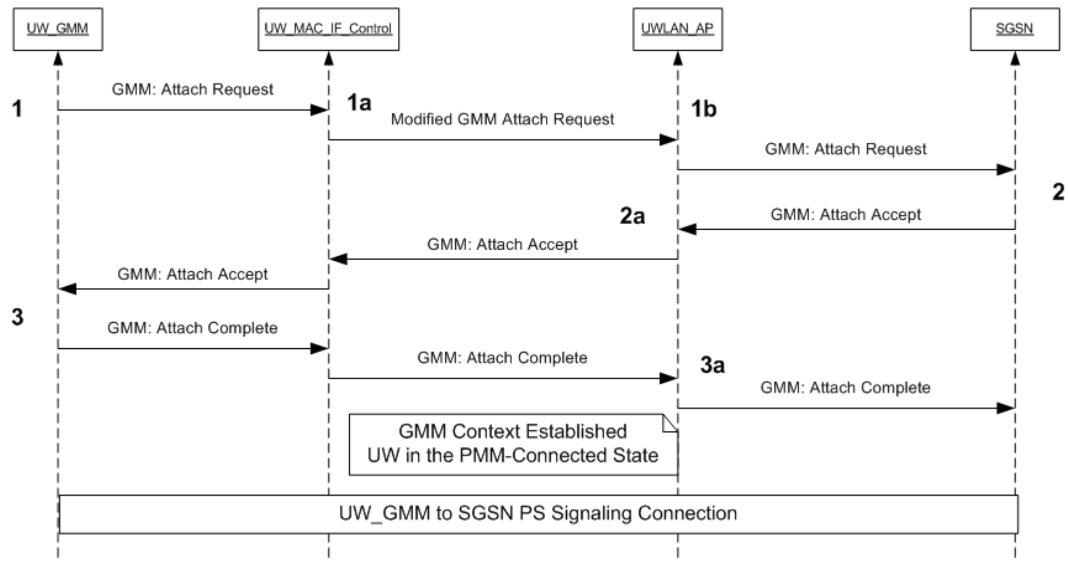

Figure 4.5: Modified GMM Attach Procedure

bits in the original UMTS GMM_Attach_Request to include the WLAN MAC address.

1b. The UWLAN_AP_Control process in the UWLAN_AP will add an entry to the Access Control Table using the triplet *(imsi, mac_address, m_state)* with a value of M_State = M_PENDING. In an attempt to limit modifications to the SGSN, the UWLAN_AP will destroy the Modified_GMM_Attach_Request and generate a standard GMM_Attach_Request and forward it to the SGSN.

2. The second step in the 3-way handshake process is for the SGSN to verify that the UW is an authorized user and send a GMM_Attach_Accept message to the UW.

2a. The UWLAN_AP will verify the UW's IMSI and M-State = M_PENDING within the Access Control table and forward the "accept" message to the UW.

2b. The UW_MAC_IF_Control forwards the GMM_Attach_Accept up the protocol stack unmodified.

3. To complete the 3-way handshake of the GMM Attach process, the GMM in the UW changes to the PMM_Connected state, generates a GMM_Attach_Complete message and forwards it to the SGSN.

3a. The UWLAN_AP will once again verify the UW's IMSI and M_State = M_PENDING; change the M_State = M_CONNECTED and forward the message to the SGSN. At this point the UW is authorized into the UMTS network and is allowed to use the UMTS resources via the WLAN access network.

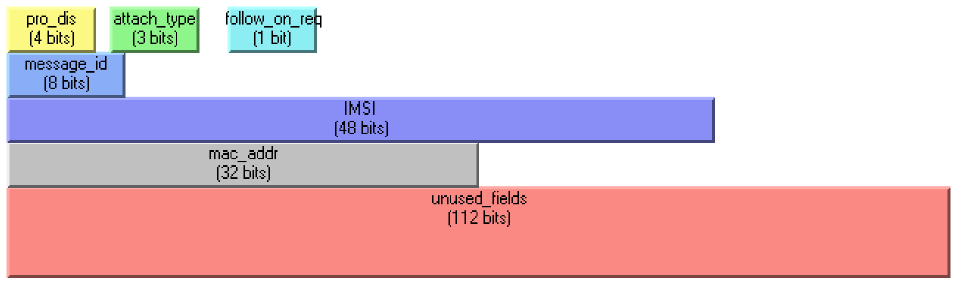

Figure 4.6: Modified GMM Attach Request Message Format

Modified RAB Assignment Procedures

The PDP Context Activation and Modified RAB Assignment Procedures are shown below in Figure 4.7. Upon receiving PDUs on a QoS for which the UW does not have an active PDP Context, the GMM in the UW will request an active PDP Context using the PDP Context Activation procedure. This procedure is unmodified from the standard, except that the UWLAN_AP_Control process (see 1a in Figure 4.7) inspects the IMSI and performs an Access Control Table lookup to ensure that the UW is authorized services. This procedure is discussed in detail in Section 2.2.6. The modification of the UWLAN_AP inspecting the IMSI and performing an Access Control Table lookup applies to all packets (both data and signaling); therefore, for brevity, this modification will not be repeated for every message.

The modifications to the RAB Assignment Request procedure occurs at 2a in Figure 4.7. The standard RAB Assignment procedure requires the RNC to reserve resources for a Radio Bearer (RB) between the UE and the Node-B as discussed in Section 2.2.6. The WLAN protocol does not support resource reservation; therefore, this portion of the procedure is not required. A future enhancement to this system could be to modify the WLAN protocol to support channel reservation.

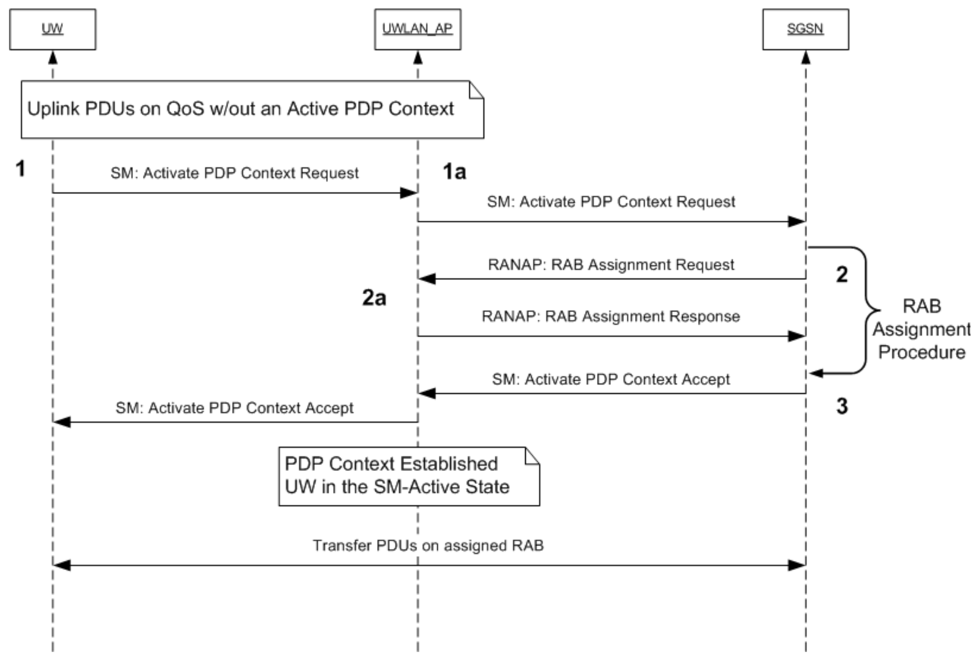

Figure 4.7: PDP Context Activation and Modified RAB Assignment Procedures

4.9.4 Enhanced User Equipment Node (UW)

The UW model assumes the availability of a software radio system or some comparable means of switching between two radio stacks based on the setting of the user device. Conceptually, the UW model augments the UMTS model by adding a WLAN protocol stack to the UE workstation model while adding the necessary control logic to selectively configure the desired client access mode. The control logic was implemented in the user-defined process module, UW_MAC_IF_Control.

The Enhanced User Equipment (UW) node model, Figure 4.8, consists of an application layer, a full TCP/IP protocol stack, the UMTS GMM with Layer-1 Mobility Management, RLC/MAC and PHY layers, the WLAN MAC and PHY layers, and the user-defined UW_MAC_IF_Control process module.

The application process module works in conjunction with the wireless applications and profiles utilities, shown in the network model (Figure 4.1), to define the application traffic that will be applied to the simulation. The standard applications utility module, discussed in detail in Section 4.7, defines a set of client-server based application transactions, including E-mail, Web browsing (HTTP), and File Transfer Protocol (FTP) traffic. The profiles utility allows the user to define a mix of applications from the application utility to form a profile that is representative of a client's application use.

The standard OPNET^{TM} TCP/UDP/IP protocol stack provides a high-fidelity implementation of the transport and network layers.

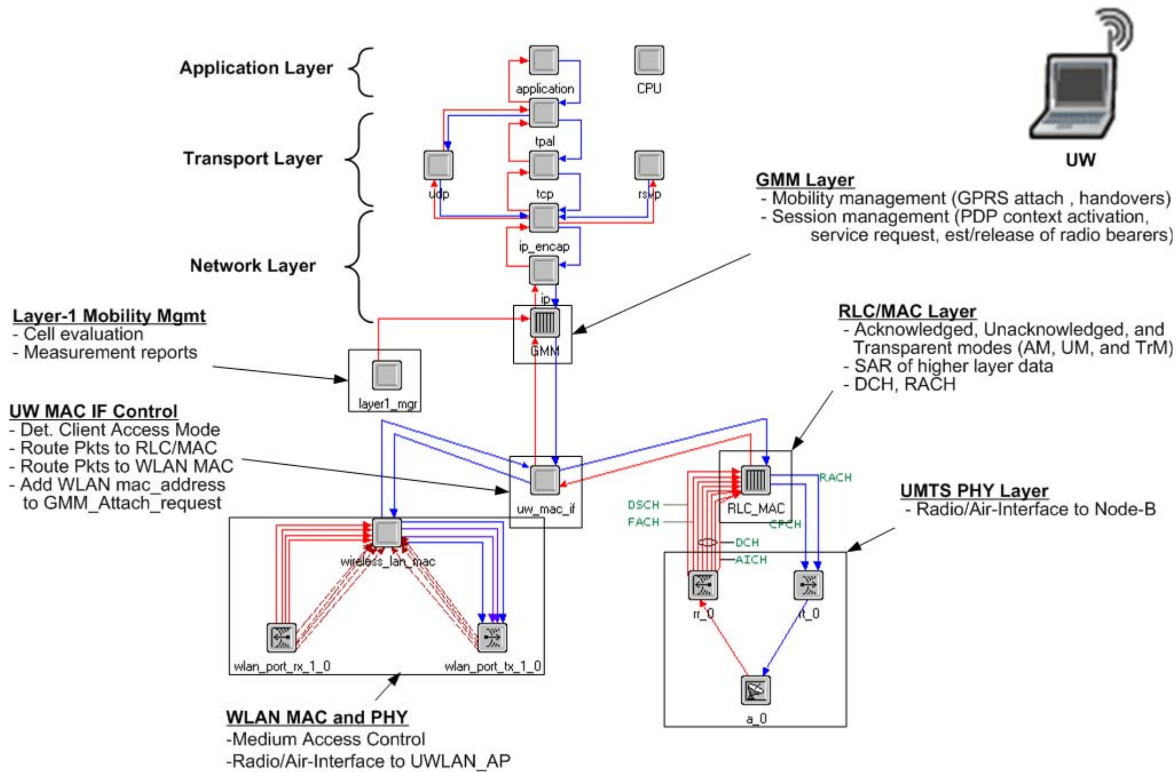

Figure 4.8: Enhanced User Equipment (UW) Node Model

The GMM module provides a detailed implementation of the GPRS Mobility Management/Session Management (GMM/SM) layers of the UMTS control plane, discussed in detail in Section 2.2.4. The GMM, which logically operates between the UW and the SGSN, provides the basic signaling mechanisms for controlling mobility management and UW authentication into the UMTS PS domain. The SM protocol is responsible for establishing and releasing packet data sessions, called packet data protocol (PDP) contexts within the UMTS network.

The RLC/MAC module provides a detailed implementation of the UMTS Reliable Link Control and Medium Access Control layer, discussed in detail in Section 2.2.5. The RLC portion of the module supports the transfer of user data in three modes: transparent, unacknowledged, and acknowledged modes as specified by the protocol in 3G TS 25.322. The module provides logical link control over the radio interface and can provide several simultaneous RLC links per UW with each link being identified by it own radio bearer (RB) ID. The MAC portion of the module implements the radio channel access signaling as specified in 3G TS 25.321. It provides this service as a set of logical channels. It (de)multiplexes protocol data units (PDUs) from the RLC into transport blocks that are delivered to/from the physical layer. The PHY portion of the module maps the logical transport channels to the physical channels to be processed through the Wireless™ module's 13-stage pipeline air interface model.

The WLAN MAC and PHY were implemented as stated above in the UWLAN_AP discussion, Section 4.9.2.

The user-defined UW_MAC_IF_Control process module is responsible for selectively switching the radio air interface between the UMTS radio stack and the WLAN radio stack. It does this by inspecting the model attribute Client Access Mode. When the Client Access Mode is set to UMTS, the UW_MAC_IF_Control routes packets to the UMTS RLC/MAC layer and the UW functions as a standard UE.

When the Client Access Mode is set to WLAN, the UW_MAC_IF_Control routes packets to the WLAN MAC layer and the UW uses WLAN as the radio access network via the UWLAN_AP. The modification to the GMM_Attach_Request, discussed in Section 4.9.3, is the only modification to the UMTS signaling.

4.9.5 Core Network (CN)

The following three sections give a brief node model overview of the existing OPNET™ UMTS specialized model set that was leveraged for this research.

The UMTS Core Network (CN) node model, Figure 4.9, consists of the SGSN process model, eight ATM layer interfaces to provide connectivity to the RNCs and UWLAN_AP, and a full IP router stack to interface with the Internet at large.

The CN model abstracts the operation of the Serving and Gateway GPRS Support Nodes (SGSN/GGSN), as well as the authentication center (AuC) and visitor location register (VLR) into one node model. The internal interaction between these entities is not simulated. The IP gateway functionality in the GGSN is modeled as IP routing stack and the SGSN process module interfaces the IP process module of this IP routing stack. The ATM stacks provide connectivity to the RNCs and the UWLAN_AP.

4.9.6 Serving GPRS Support Node (SGSN) Process Module

The SGSN process module models the two system network protocols in the UMTS control plane: the GPRS Mobility Management (GMM) and the Session Management (SM) protocols. The GMM protocol signaling is modeled between the UE and the SGSN with the various GMM_Attach_* messages to provide the basic signaling mechanisms for mobility management and authentication functions within UMTS PS domain. The SM protocol signaling is modeled between the UE/UW with the various PDP_Activate_* and Service_Request_* messages in order to establish/release packet data sessions, called packet data protocol (PDP) contexts within the UMTS network. The SM protocol signaling is modeled between the RNC/UWLAN_AP with the various RAB_Assignment_* messages in order establish/release Radio Access Bearer (RAB) assignments between the UE and its SGSN. The RAB assignment process triggers the establishment of a Radio Bearer (RB), the

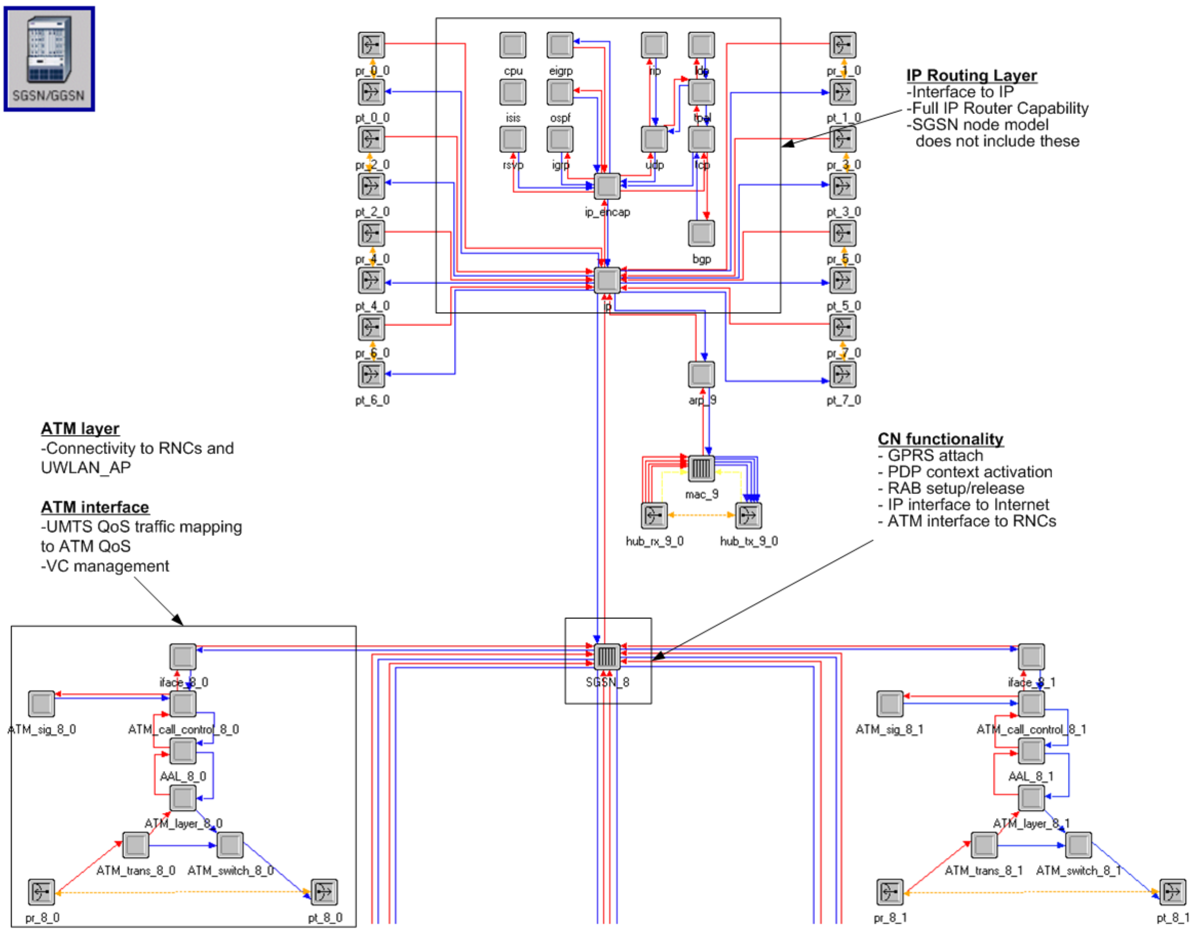

Figure 4.9: UMTS Core Network (CN) Node Model

physical channel between the UE and its serving Node-B. The RNC functionality was designed into UWLAN_AP in order to leverage these messages to use the WLAN infrastructure BSS as an alternate radio access network as discussed in Section 4.9.3 .

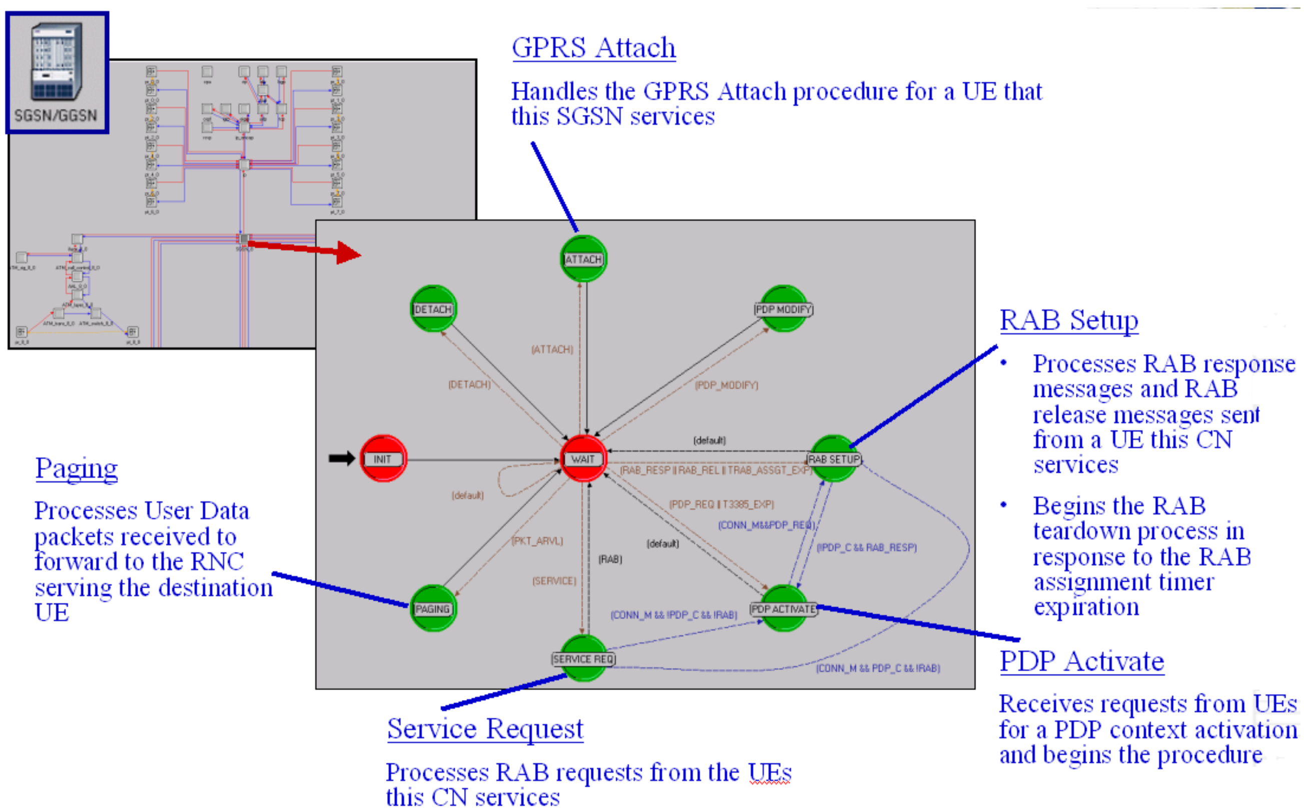

Figure 4.10: Serving GPRS Support Node (SGSN) Process Module

4.10 Summary

This chapter presented the design of the simulation model and the methodology used throughout this phase of the research. The chapter began with a discussion of Jain's ten-step method of systematic performance evaluation [37]. Jain's ten-step method was determined to be well suited for evaluating the performance of communications systems through simulation; therefore, it was chosen as the design methodology for this research. The following sections described the design of the simulation model using this methodology.

Section 4.1 presented the definition of the system to be modeled and detailed the assumptions that were necessary to limit the scope of the problem. It described the assumptions made in modeling both the UMTS and WLAN data networks, as well as the interworking of these two technologies. The selection of these assumptions was a critical step in order to keep the simulation complexity manageable, while still meeting the research goals.

The system services and outcomes were discussed in Section 4.2. This section explained the possible usages of the UMTS-WLAN simulation environment. These included the investigation of the protocol effects and overall system performance for an interworked UMTS-WLAN system. The specific statistics and effects that can be studied with this simulation environment were listed as the control plane signaling protocol interaction, the UMTS dedicated channels (DCH) utilization, and application response time for a variety of application traffic.

Section 4.3 presented the performance measures used to evaluate the UMTS-WLAN system. These performance measures included protocol interaction, dedicated channel utilization (DCH), data session set-up delay, and application response time.

Inputs to the simulation model that were not varied during different simulation runs were termed simulation parameters. Section 4.4 presented the simulation parameters used to model the UMTS-WLAN system. The selection of these values was important because they affected how accurately the simulation modeled an actual system. The simulation parameters were given in Table 4.1.

Inputs to the simulation that were varied during different simulation runs were termed simulation factors. Section 4.5 presented the simulation factors used to exercise the simulation environment across a spectrum of normal operating conditions. The simulation factors were summarized in Table 4.2.

Section 4.6 presented the decision to use simulation was the primary tool used throughout this research. It also highlighted that OPNET Modeler™ 9.1.A was used to implement and test the UMTS-WLAN simulation environment.

The selection of the appropriate traffic models to test the simulation environment was a critical step to ensure that the system modeled an actual UMTS-WLAN system. Section 4.7 presented the traffic models used in this research endeavor. The traffic models were designed using the OPNET™ built-in application models. These profiles were parameterized in order to closely simulate traffic generated by a wireless data user. The application profiles used were the FTP, E-mail, and HTTP profiles.

Section 4.8 presented the three scenarios that where used to test the simulation model. The three scenarios were termed Existing WLAN vs UMTS-WLAN Scenario, Existing UMTS vs UMTS-WLAN Scenario, and Mixed Client Access WLAN—UMTS Scenario. The primary difference between these scenarios was the access network used. All three scenarios varied the simulation factors and the traffic models in order to vary the network traffic load.

Section 4.9 described the design of the simulation in detail. The simulation was designed in OPNET Modeler™ 9.1.A using a top-down design approach. Modeler™ uses a hierarchical structure of network scenarios, nodes and processes. This section presented an overview of the simulation design at the network, node, and process levels. A detailed description of how the UWLAN_AP and UW were designed at the node level using both built-in OPNET™ processes and user-defined processes was also given.

The next chapter provides results from the UMTS-WLAN network simulation models.

Chapter 5

UMTS-WLAN Results

This chapter presents the simulation results and analysis and is the product of work jointly conducted with Tracy Mann. The chapter begins with a discussion of the simulation results' statistical accuracy in Section 5.1. Subsequent sections represent work jointly conducted with Tracy Mann. These results are also presented in his Masters Thesis [8]. We jointly designed and implemented the UMTS-WLAN simulation tools and conducted the initial validation and verification testing presented in Section 5.2. Additional performance testing results are presented in [8], as I was involved in the design of the performance tests, but not in the actual execution of those tests. The reader is urged to reference [8] for additional performance testing results of the UMTS-WLAN interworking simulation set. While Tracy continued testing the UMTS-WLAN models, my research activities branched to AGG-MAC which is presented later in Chapters 6 and 7.

The research questions were intended to provide a context in which to exercise the UMTS-WLAN models over a range of normal or anticipated operating conditions and to demonstrate the functionality of the UMTS-WLAN simulation environment. The research questions for this phase of the research were originally stated in Section 1.4. The questions related to the UMTS-WLAN interworked phase of our research are restated here for ease of reference:

1. What are the benefits associated with a tightly-coupled UMTS-WLAN interworked system?

2. How does a tightly-coupled, interworked UMTS-WLAN system perform?

5.1 Statistical Accuracy

The use of stochastic processes introduces a measure of uncertainty in simulation results. Running a simulation with different random number generator seeds will produce different results. In this research, the simulation model was run with five different seed values for each set of input parameters. The statistical accuracy of the results was measured as a confidence interval. The 95-percent confidence interval was calculated with Equation 5.1, where $\overline{x}$ is the mean, s is the standard deviation, n is the number of samples, and $t_{[1-\alpha/2;n-1]}$ is the $(1-\alpha/2)$ -quantile of the Student t distribution with $n-1$ degrees of freedom [37].

$$100(1-\alpha)\%CI = \left(\overline{x} - t_{[1-\alpha/2;n-1]}\frac{s}{\sqrt{n}}, \overline{x} + t_{[1-\alpha/2;n-1]}\frac{s}{\sqrt{n}}\right) \quad (5.1)$$

The simulation results for the FTP application traffic had very tight confidence intervals. Table 5.1 gives the FTP download response time results for a single UW in the WLAN Client Access Mode with FTP application traffic. As an example, consider the right-most column for an FTP file size of 60 MBytes. The average FTP download response time for all five seeds was 448.6591 seconds with a 95-percent confidence interval of 448.5574 to 448.7607 seconds. There was a less than 1-percent difference between the minimum and maximum FTP response times for all of file sizes.

The simulation results for the Wireless Client application traffic also had very tight confidence intervals. Table 5.2 gives the HTTP page response time results for the UMTS-

Table 5.1: FTP Download Response Time (sec) for a UW in WLAN Client Access Mode

FTP Filesize	1B	100KB	800KB	1.5MB	7MB	12MB	15MB	30MB	40MB	60MB
Seed 1	0.194574	1.177744	6.337744	11.71774	52.45774	89.95774	112.5177	223.9377	298.0577	448.7377
Seed 2	0.188612	1.190725	6.350725	11.73073	52.47073	89.97073	112.5307	223.9307	298.0707	448.7507
Seed 3	0.183707	1.203707	6.363707	11.74371	52.48371	89.78371	112.4086	223.7437	297.8837	448.5637
Seed 4	0.190725	1.068612	6.428612	11.60861	52.54861	89.84861	112.3437	223.8546	297.9546	448.6086
Seed 5	0.197744	1.094574	6.454574	11.63457	52.57457	89.87457	112.4346	223.8086	297.9486	448.6346
Mean:	0.191072	1.147072	6.387072	11.68707	52.50707	89.88707	112.4471	223.8551	297.9831	448.6591
95% CI:	0.184352	1.071135	6.323218	11.61113	52.44322	89.79029	112.3503	223.7529	297.8847	448.5574
	0.197793	1.22301	6.450927	11.76301	52.57093	89.98386	112.5439	223.9573	298.0815	448.7607

WLAN scenario with the Wireless Client application traffic. As an example, consider the right-most column for 2 UWs in the UMTS access mode and 18 UWs in the WLAN access mode. The average HTTP page response time for all five seeds was 7.8262 seconds with a 95-percent confidence interval of 7.1727 to 8.4796 seconds.

Table 5.2: HTTP Response Time (sec) for UWs in UMTS Client Access Mode

Number of UWs in UMTS Mode	20	18	16	14	12	10	8	6	4	2
Seed 1	12.432839	9.167257	8.251739	8.116110	8.385436	8.096770	7.359587	7.627617	8.387927	7.525191
Seed 2	13.416985	8.911012	8.658675	8.086360	7.534312	8.096770	7.651648	7.722649	7.751405	8.666523
Seed 3	13.667978	8.865238	9.023893	8.796989	8.571726	7.673093	7.694435	8.272729	7.446080	7.268593
Seed 4	12.710351	9.249344	8.772352	10.255774	8.036825	8.029945	7.695623	7.315788	7.261536	7.848843
Seed 5	13.161796	8.891231	8.517017	8.068062	8.243471	7.538070	7.461991	7.967559	7.586461	7.821761
Mean:	13.077990	9.016817	8.644735	8.664659	8.154354	7.886930	7.572657	7.781268	7.686682	7.826182
95% CI:	12.450649	8.795917	8.287612	7.496688	7.660177	7.560854	7.382510	7.333671	7.151045	7.172729
	13.705330	9.237717	9.001858	9.832630	8.648531	8.213006	7.762803	8.228865	8.222319	8.479636

The results for all simulation runs exhibited similar 95-percent confidence intervals. Both the FTP application traffic and the Wireless Client application traffic produced very tight confidence intervals. Therefore, five simulation runs with different seed values was determined to be sufficient for 95-percent confidence intervals. All data points presented throughout the rest of this chapter represent the mean value of five simulation runs with different seed values. Complete tables with the results of individual simulation runs and the 95-percent confidence intervals are provided in the Appendix.

5.2 Model Verification and Validation

This section describes the methods used to ensure the simulation model was both correctly implemented and representative of the real system. These two steps were termed model verification and model validation [37]. Section 5.2.1 discusses the use of a single FTP file transfer to input signals to verify correct operation of the simulation. Section 5.2.2 presents the theoretical analysis that was used to validate the simulation model. The analysis compared theoretical and simulation results for a single client with the FTP traffic inputs.

5.2.1 Model Verification

Model verification is the process of determining if a simulation model functions correctly. This includes such tasks as debugging the computer code, testing for logic errors, and testing the functionality of different modules. As discussed in Section 4.9, the simulation was designed in OPNET Modeler 9.1.A using a top-down, modular approach. This approach simplified the task of model verification since each module was tested independently.

Each node and process in the simulation was tested to verify that it functioned correctly. This was accomplished by running short simulations in the OPNET™ ODB (OPNET Simulation Debugger) mode. The ODB provides an environment where the user has interactive control of the simulation in order to investigate its behavior by setting breakpoints and traces to print out detailed information about events or objects. The user-defined authentication and access control protocol was verified in detail by creating traces for each of the messages. The results of these traces can be seen in the appendix. Each of the protocol transactions was verified by inspecting the proper message flow to ensure that it corresponded with message flow defined in 2.2.6. The user-defined authentication and access control protocol passed the verification process. The signaling protocol mechanisms of the model functioned in accordance with the 3GPP standards as designed.

Short simulations were also run to collect statistics at various points in the model to ensure that the model was functioning properly. The standard OPNET^{TM} processes were tested to ensure they functioned as described in the software documentation. This required that both the UMTS and WLAN model sets to be tested.

The UMTS model set was developed jointly by Telcordia and OPNET^{TM} Technologies. [1], by Demers, is the seminal paper highlighting the capabilities of the baseline UMTS model set. The results from the short verification simulation tests verified the correct operation of the UMTS models against the results of this paper. The UMTS models passed the verification and validation process.

The WLAN model set was verified in the same manner as the UMTS models. Short simulations were run to verify the correct operation of the various WLAN nodes. The results of these simulations were compared against the results presented in the OPNET^{TM} Technologies model documentation [41]. The WLAN models passed the verification and validation process.

5.2.2 Model Validation

Model validation is the process of determining if a simulation model is representative of the real system. A simulation can be validated using expert intuition, real system measurements, or theoretical results [37]. Comparing simulation outputs and measurements from a real system is the most reliable way of validating a simulation model. Real system measurements were not available in this research since resources were not available for prototype 3G wireless equipment. Comparing simulation and theoretical results was the primary method used to validate the simulation model. Theoretical analysis of the system was conducted using FTP application traffic.

To the validate the PCF data transfer within the WLAN access network, a test was conducted with one wireless client data user running the FTP application. The system

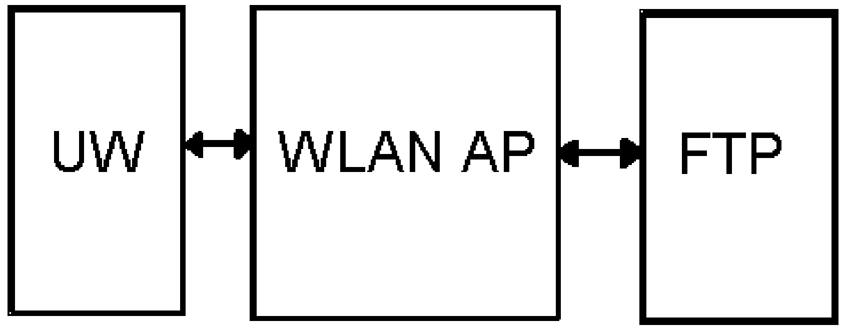

Figure 5.1: Simplified Analytical Model

consisted of a WLAN node operating in an infrastructure BSS. The FTP file size was systematically increased in order to vary the network traffic load. The FTP download response time was measured for each increment of the FTP file size and compared with the analytical results in Figure 5.3.

The block diagram in Figure 5.1 was used to construct the analytical model and determine all of the delays that an FTP packet will encounter in the system. The following simplifying assumptions were made in order to make the analytical model tractable.

Backside Server Network

The delays on the backside server network would be implementation dependent. The focus of this analysis was on the access network and not the backside server network; therefore, the backside network delay was considered to be constant. From 3GPP TR 25.853, the one-way delay between a 200 km link between the SGSN and the GGSN would be $800\mu s$; therefore, a total round-trip delay of $2ms$ was chosen between the access network and the server.

TCP Setup, Slow-start, and Window Size

The delays from TCP connection setup and the slow-start algorithm were assumed to be constant. By assuming an infinite TCP window size, the delays from the TCP throttling were minimized. Ignoring the non-deterministic effects of TCP affected the accuracy of the analytical model, but this assumption was necessary to make the model tractable.

Node Processing Delay

The node processing delays were considered to be small as compared to the link transmission delays and the PCF signaling delays. Therefore, the node processing delays were assumed to be zero.

Number of Packets

The number of packets was calculated by dividing the file size by the maximum segment size (MSS). The MSS for the access network was assumed to be 1500 bytes.

$$Num\_Pkts = \left\lceil \frac{File\_size}{MSS} \right\rceil \tag{5.2}$$

WLAN CFP Timing and Overhead

The most significant source of delay was from the WLAN CFP Timing and packet overhead as result of the point coordination function (PCF) protocol. The CFP timing delays were calculated using Figure 5.2.

Once all of the CFP timing delays and overhead were accounted for, the wireless link transmission delay was calculated for a single packet using the packet size divided by the client goodput. The client goodput referred to the bandwidth the user actually received after

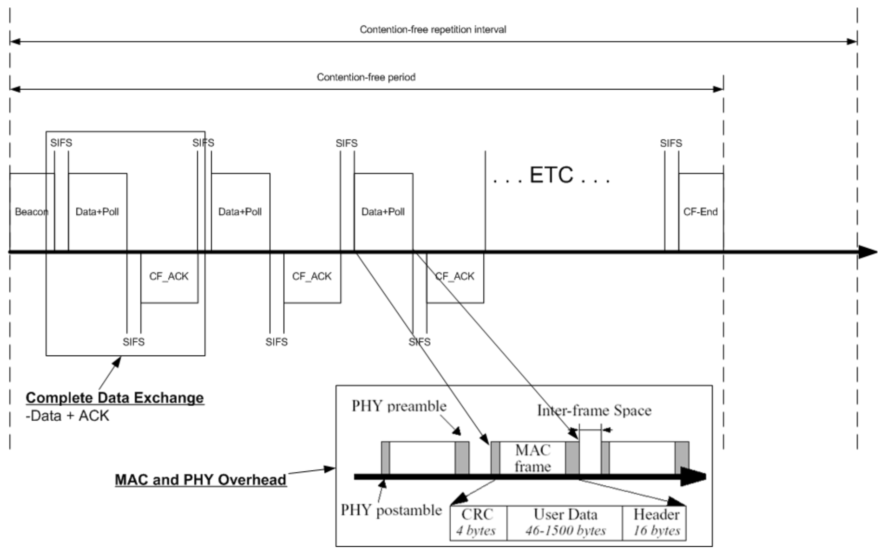

Figure 5.2: CFP Timing and MAC/PHY Overhead

all overheads were accounted for, including the MAC and PHY overheads of Figure 5.2. The client goodput was calculated by multiplying the raw channel throughput (11 Mbps) times the ratio of packet payload over the total packet overhead (including timing overhead). The ratio was normalized for an 11 Mbps channel in terms of seconds.

To simplify the CFP timing overhead calculations, the term data exchange slot was defined as a complete data exchange including $Data + 2 * SIFS + ACK$ (see Figure 5.2). Using the PCF parameters from Section 4.4, the CFP Interval included ten data exchange slots. The Data_Exchange_Slot_Ratio was defined as the ratio of the number of data exchange slots actually used to transmit divided by the total number of data exchange slots within a CFP Interval Repetition. The term $Slot\_Data\_OH\_Ratio$ was defined as the ratio of the data to overhead within a data exchange slot. Equation 5.3 defines this ratio.

$$Slot\_Data\_OH\_Ratio = \frac{MSS}{Data + (2 * SIFS) + ACK} \tag{5.3}$$

where,

$$Data = \left[\frac{WLAN\_MaxSDU + WLAN\_MSDU\_Header}{Data\_Rate} + plcp\_overhead\right] \tag{5.4}$$

Packet Delay

Using the terms defined above, the per packet delay was defined in Equation 5.5.

$$Pkt\_delay = \frac{Pkt\_size}{Data\_rate} * Slot\_Data\_OH\_Ratio * Data\_Exchnge\_Slot\_Ratio \tag{5.5}$$

FTP Response Time

The FTP response time was defined in Equation 5.6.

$$FTP\_Rspns\_Time = Conn\_Setup + Num\_Pkts * Pkt\_delay \tag{5.6}$$

The theoretical results were calculated in MATLAB using the relationships defined above and the parameters defined in Section 4.4. Figure 5.3 shows a comparison between the theoretical and simulation results for various file sizes. The simulation results are very close to the theoretical results. The difference between the theoretical and the simulation results can be accounted for by the inability of the simple analytical models to capture the full effects of TCP. The agreement between the analytical and simulation results validates the correct operation of the PCF data transfer for the WLAN access network. All aspects of the model passed the verification and validation process.

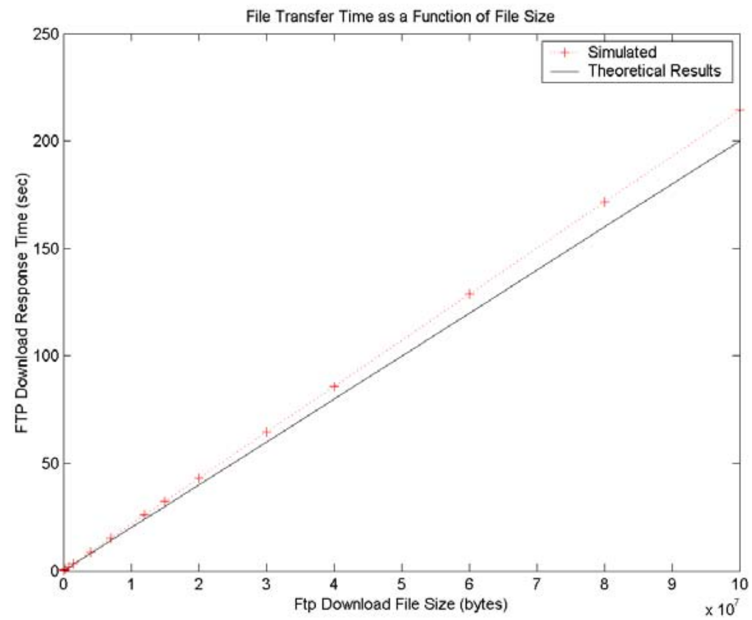

Figure 5.3: Simulated vs Theoretical Results

5.3 Benefits of Interworking UMTS-WLAN

The intent of this scenario was to demonstrate the benefits of the interworked UMTS-WLAN system as compared to the existing UMTS system. Specifically, this scenario demonstrated the conservation of limited dedicated cellular channels by shifting data users to the WLAN access network. This had the effect of "freeing" limited resources for revenue producing voice calls. It also demonstrated the significant improvement in application response time of the interworked UMTS-WLAN system as a result of the larger WLAN channel.

Since the primary motivation for integrating WLAN into the UMTS system was to free limited resources while continuing to provide service, the number of reserved dedicated channels (DCH) in the Node-B was examined. Because of the importance of voice channels, dedicated channels were considered the critical, finite resource in the UMTS network.

The interworking UMTS-WLAN experiment was conducted using twenty wireless client data users running a mix of client applications; specifically the mix of Web, E-mail and FTP applications (Wireless_Client application profile) discussed in Section 4.7. The baseline system consisted of twenty UW nodes operating in a UMTS cell. Sixteen of these clients were operating within 300m of the UWLAN_AP, and the other five were not within the range of the WLAN access network. The simulation was first run with the Client Access Mode set to UMTS for all 20 clients. The Client Access Mode was then toggled to WLAN for the sixteen clients that were in range of the WLAN access network and the results were compared.

Based on the quality of service requirements for the data session, the UMTS allocates one or more DCH channels to support a user session. It can be observed from Figure 5.4 that reducing the number of contenders for service from twenty to four reduced the average number of DCH channels in use by approximately 65%. It is possible to reallocate these unused DCH channels to voice services or to additional packet data services. Thus, it can be asserted that a service provider could use these unused resources to generate additional revenue.

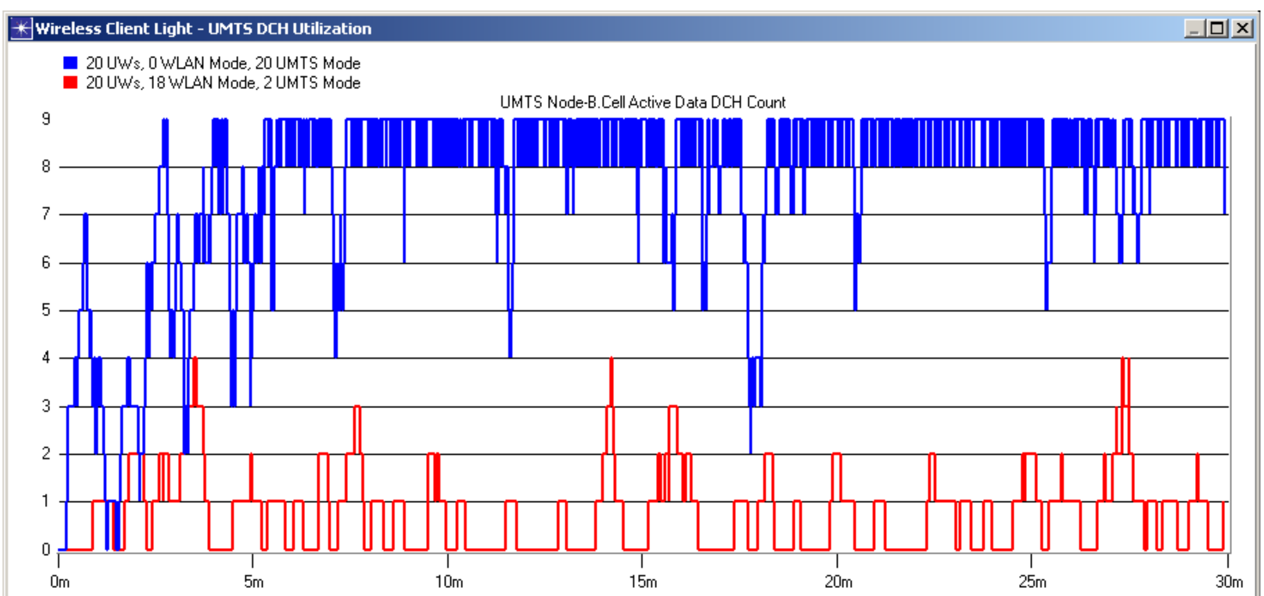

Figure 5.4: Reserved DCH Channels (DCH Count vs Time (min))

One might question why the reduction in DCH utilization was only 65%, and the results of Figure 5.5 offer that insight. In the case where the sixteen clients were shifted to the WLAN access network, there were 0 queued requests for services, while the heavily loaded system using strictly UMTS dedicated channels appears to be saturated and requests for service are being queued in the CN. This equates to delayed or possibly even denied service to the user and potentially lost revenues to the service provider.

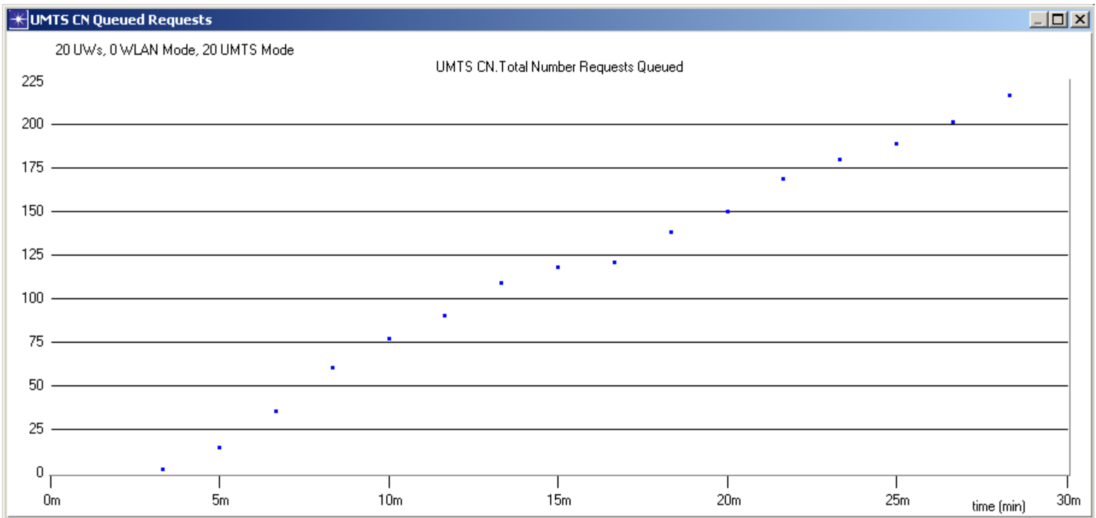

Figure 5.5: Queued Requests for DCH Channels (Delayed or Denied Service)

5.4 Summary

Section 5.2 described the methods used to ensure the simulation model was both correctly implemented and representative of a real system. These two steps were termed model verification and model validation. A series of short simulations were run in the OPNET^{TM} Debugger (ODB) environment to verify correct operation of the models. The results from both the WLAN and UMTS models sets were compared against the documentation results in order to validate the OPNET^{TM} defined models. Comparing simulation and theoretical results was the primary method used to validate the user-defined models. Both the OPNET^{TM} and user-defined models passed the verification and validation process.

The benefits associated with a UMTS-WLAN interworked system were demonstrated in Section 5.3. The performance of the UMTS-WLAN interworked system is more fully covered in [8], but significant results are included herein in Sections 5.2 and 5.3. The second phase of our work was initiated when we discovered that, for some cases, the achievable throughput was significantly less than expected. This led us to consider packet aggregation for which the design and methodology is presented in Chapter 6. The results for the AGG-MAC protocol are presented in Chapter 7.

Chapter 6

AGG-MAC Design and Methodology

During the second phase of our research, we sought to identify a protocol enhancement in the WLAN MAC which would provide performance improvements in the WLAN component of a UMTS-WLAN interworked system. In this context, the mode of operation was PCF and traffic was routed through the AP to an external network. At the time of this research, it was unclear what characteristics a client device would exhibit. A client station might resemble a wireless phone with rather limited capabilities or a fully featured mobile computer. Because the future was unclear, we focused our efforts on the limited capability system with the realization that our enhancements would benefit either type of client. We further assumed that the access point (AP) in a "hot spot" scenario would generally be significantly more capable than the client station. With these considerations, we attempted to identify which aspect of WLAN performance would yield the most performance improvement.

We focused our efforts on the reduction of per frame fixed overhead because it offered the most potential for improvement in our defined environment. Contention based, or backoff algorithm, enhancements are more relevant for DCF mode operation. During the CFP, the backoff algorithm is not invoked, and although backoff does function during the CP, the CP is limited in such a system. We did not focus on higher layer protocols because we wanted our enhancement to support a range of traffic conditions from CBR, voice traffic, to VBR, data traffic such as E-mail, HTTP, and FTP traffic. We saw frame size referenced as a factor commonly identified as affecting WLAN performance [42][33][5]. As our focus was a limited capability client station, we considered the traffic in such a system and observed that at least as many, if not more, small packet sizes than referenced in traffic research [43][44][45][5] occurred under these conditions. For these reasons, we sought to reduce the per frame overhead for small packet sizes.

For this phase of our research, we again followed the ten-step method of systematic performance evaluation using simulation [37].The ten-step method was covered in the design discussion of our first research phase in Section 4. In the upcoming sections, we present our WLAN MAC layer protocol enhancement. Section 6.1 provides an overview of the protocol at a higher level of abstraction. Additional specifics on the design choices and considerations made in the design of AGG-MAC are presented in Section 6.2. Performance metrics are presented in Section 6.3, simulation parameters are in Section 6.4, and simulation factors are presented in Section 6.5. The simulation design is presented in Section 6.7. We subsequently present the results of validation, verification and other simulated experiments in Chapter 7.

6.1 AGG-MAC Protocol Design

The packet aggregation protocol was implemented in the MAC layer (AGG-MAC). An illustrative example of the functionality of AGG-MAC wireless LAN MAC layer protocol design is presented in Figure 6.1. Packets received from the higher layer are aggregated into a single frame up to the maximum allowable size. The fragmentation threshold was defined as the maximum allowable aggregate size in order to avoid creating frames that were too large for existing channel conditions. Our protocol works with either a statically assigned fragmentation threshold or any of the several dynamic assignment mechanisms currently being researched as discussed in Section 3.2.2 [33][34]. As illustrated, the protocol provides performance improvements for a collection of packets having a total aggregate size less than or equal to the fragmentation threshold. The protocol does not offer improvement, nor does it degrade performance, for a series of packets that would exceed the threshold (such as packets 3 and 4 shown in Figure 6.1). Notable from this illustration is the dependence on not only distribution of packet sizes, but also the specific ordering of packet sizes and on the arrival time of packets from the higher layer.

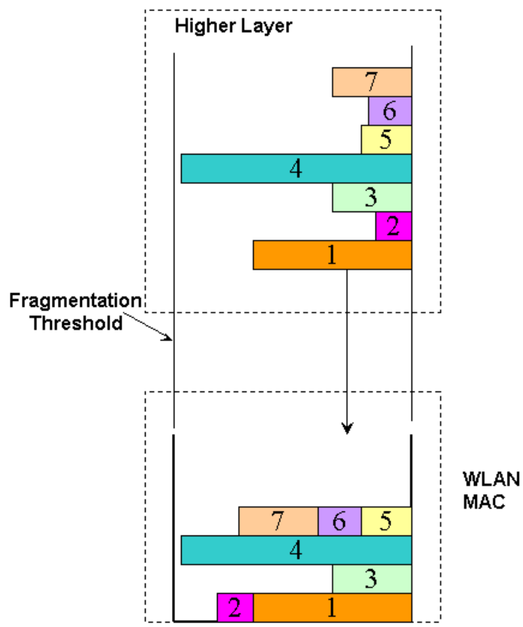

Figure 6.1: AGG-MAC Protocol Overview

The algorithm implementing AGG-MAC is quite straightforward. The basic design choices in the algorithm are below:

- When to perform aggregation operation
- What is the upper limit for aggregate frame size
- Which packets selected as candidates for aggregation

6.1.1 When to perform aggregation operation

The timing for when to perform the aggregation will have an impact on performance of the algorithm. In the processing of packets from the higher layer for transmission, traditional designs of MAC layer protocols respond to an interrupt from above, immediately encapsulate the packet into its appropriate frame and then place it in a queue for transmission by the physical layer. In the very first local area network (LAN) systems, this approach was a necessity because protocol-processing delays were the limiting factor rather than capacity of the medium. AGG-MAC benefits most from having as many packets as possible awaiting transmission and available to aggregate. Therefore we sought to construct the frame just prior to transmission while insuring that a frame is ready for transmission when the physical layer is ready to transmit. The WLAN protocol implements a more complex protocol than most traditional MAC layers in order to provide IEEE 802.2 based LAN services via a wireless medium. This added complexity generally allows for a range of instants in which a packet or set of packets could be prepared and passed to the physical layer.

6.1.2 What is the upper limit for aggregate frame size

With the realization in actual system implementations that bigger is not always better, we leveraged the existing fragmentation threshold parameter to limit the upper bound of an aggregated frame. This allowed better performance of AGG-MAC under varied channel conditions. The purpose of the fragmentation threshold was to establish the maximum allowable size of a frame in order to have a high probability of successful transmission. We recognize that this is the appropriate upper limit for an aggregated frame as well. As stated in [46] and restated in many subsequent works, fragmentation can significantly degrade performance. It would be counterproductive to create an aggregate frame that was allowed to trigger fragmentation. We are just beginning to explore the possibility of using various values of the fragmentation threshold to apportion capacity to different classes of clients.

In OPNET^{TM}, the fragmentation mechanism is implemented using the Segmentation and Reassembly (SAR) package in the function *wlan_prepare_frame_to_send()*. Because of the flexible design of the SARs package, we were able to implement the aggregation and subsequent recovery of packets in the AGG-MAC enhancement using the existing *wlan_mac* kernel procedures (KPs) and data structures with only minor modification to parameters used to create the SARs lists. The SARs package greatly simplifies the receive process in the simulation. An actual receiving MAC layer would be required to parse the length information from the higher layer (IP) header in order to identify the end of a packet and the start of the next within an aggregated frame. In the simulation, the KP *op_sar_rsmbuf_pk_remove()* allowed us to remove a single packet at a time from an aggregate frame and pass that packet up to the higher layer.

6.1.3 Which packets selected as candidates for aggregation

Concerning the selection of packets for consideration in the AGG-MAC protocol, this implementation considered packets in strict order received from the higher layer. Packets from the higher layer are enqueued in order. Our iterative operation first selected the packet at the head of the queue for preparation to transmit and considered the next packet to aggregate with the current set of packets. If the destination address of the packet under consideration was the same as the destination address of the current working frame and if the total size does not exceed the maximal size limitation of the AGG-MAC frame, then the packet is aggregated with the current set. This selection process iterates until either condition is false. The aggregated collection of packets is then encapsulated into the WLAN frame for transmission.

We considered more complex scheduling algorithms, but selected simple FIFO to ensure strict ordering is maintained. We could search from through the queue until we either created a maximal frame or encountered the end of the queue. This would potentially create larger aggregate frames, but could negatively impact the fairness of our system.

We have also considered relaxing the requirement for aggregated packets to be addressed to the same destination. For a system in which all traffic is directed through the AP such as our "hot spot", we could aggregate packets with different destinations. This would be a reasonable and potentially desirable improvement because every frame is transmitted to the AP as its next hop. We have considered aggregation from the AP to a set of AGG-MAC aware stations, but have deferred this consideration because it would be unclear how to acknowledge such a frame and ensure that every destination received the information.

Our current system is the most conservative implementation of AGG-MAC. We aggregated strictly in order of arrival to the MAC layer and then only for packets with the same destination address.

6.2 AGG-MAC Definition and Assumptions

6.2.1 AGG-MAC Association

We assume that during association with the AP, an AGG-MAC-capable station will report its ability to support the AGG-MAC enhancement by setting bit 5 of the capability information field (Figure 6.2) of the association request frame. This frame is transmitted to the AP in order to initiate association. The AP will respond with an association response frame. The AP will use the same bit 5 in the capability information field to declare its ability to support AGG-MAC.

6.2.2 AGG-MAC Frame Format

An AGG-MAC frame can be based on any of the previously defined data frame types and conforms to IEEE 802.11 [3] requirements with the following exceptions.

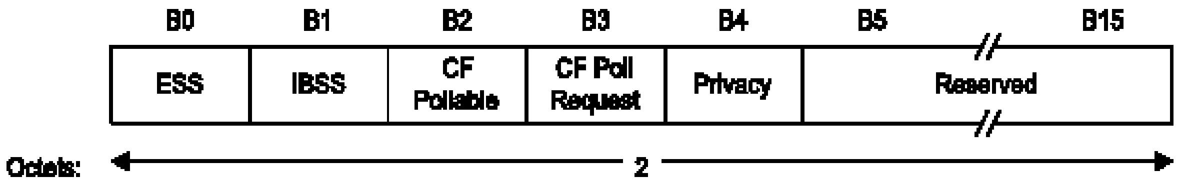

Figure 6.2: WLAN Capabilities Info Field [3]

In the MAC Header (described in 2.3.2), the following items are specific to the AGG-MAC protocol:

Three elements of the frame control field, as described in Section 2.3.2, are related specifically to AGG-MAC:

- **Type/Subtype** fields will be used to indicate that this frame is an AGG-MAC frame. The type field will be set to the previously reserved value (11), and the subtype will be used to indicate any of the accepted data frames (0000-0111), or indicating the proposed IEEE 802.11e frame (1000-1111) [18]. AGG-MAC is compatible with either variant.

- **More Fragments** field should always be zero for a correct implementation of the AGG-AMC protocol. The protocol implementation should ensure that an AGG-MAC frame is not subjected to fragmentation.

- **Duration/ID** is set to the size of the total aggregate frame body, rather than to any sort of individual packet sizes during CP, and is set to the specified value of 32,768 for all CFP transmissions. Special cases are handled in accordance with [3] where the aggregate MSDU is treated as a single entity.

- **Fragment Number** field should always be set to zero, for reasons indicated above in More Fragments subfield.

- **Frame Check Sequence (FCS)** will be computed over the entire aggregate header and payload.

6.3 Performance Metrics

The AGG-MAC protocol was evaluated based upon throughput, delay, dropped data, and voice over IP performance metrics. Voice over IP metrics consisted of voice packet end-to-end delay and voice packet delay variation. These performance metrics were defined as follows:

6.3.1 Throughput

Relative to the WLAN MAC layer, **throughput** was defined as the total number of bits sent to the higher layer from the MAC layer. The data packets received at the physical layer were sent to the higher layer if they were destined for this station. We measured this value in terms of bits per second. Throughput represents an average rate of traffic flow where higher values were better.

6.3.2 End-to-End Delay

End-to-End Delay was the period of time between arrival of a packet at a protocol layer and the completion of processing of the packet at the peer protocol on the receiving node. The receiving node reported the value of End-to-End delay. This delay included all processing, queueing, and transmission times from source to destination. End-to-End delay was reported for voice protocol packets and for other traffic flows.

6.3.3 Data Dropped

The metric, **Data Dropped**, measured the rate of data lost in the WLAN MAC and in unsuccessful transmission between peer MACs. This value was measured in bits per second and lower was better.

The packets dropped in bits by the WLAN MAC due to:

a) the overflow of higher layer buffer, or

b) failure of all retransmissions until retry limit.

In an error-free channel, buffer overflow was the cause of dropped data. In such a case, the reported value can be considered representative of congestion.

6.3.4 Voice Packet Delay Variation (Jitter)

Delay Variation measured variance among end-to-end delays for voice packets received by a node. End-to-end delay for a voice packet was measured from the time it was created to the time it was received. This value was measured in seconds and smaller is better. Delay variation impacts the amount of additional delay required and the size of the buffers necessary to support real-time, voice applications in order to limit the effects of the network on the user.

6.3.5 Application Response Time

HTTP and FTP application response times were chosen because they represented two different classes of traffic. Both response times were measured in seconds and a smaller value was considered better.

HTTP Page Response Time was defined as the time required to retrieve an entire HTML page with all associated inline objects. The web page response time was measured from the time a client browser application sent the initial request to the web server and terminated when the client received the last element of the entire HTML page. Embedded within this measurement were the setup and termination signaling required for the TCP connections. HTTP1.1 protocols were employed in order to allow connection reuse to limit this overhead. It was believed that version 1.1 of the HTTP protocol was more likely representative of current and future browsers.

FTP Upload Response Time was defined as the time that elapsed between a client sending a "put" request and when acknowledgement of the final segment signaling successful upload of the entire file to the FTP server. This metric was dependent upon the file size. Like the HTTP response time, this metric includes the TCP protocol setup and termination of the TCP connection.

6.3.6 Amount of Aggregation

Packets per frame represents the number of individual data packets combined into a single data frame transmitted, or the amount of aggregation. If IP Aggregation is inactive, this value should always be 1. With IP Aggregation, the value should be greater than or equal to 1. The higher the value, the more aggregation. More is better. This value was measured in packets per frame (ppf).

6.4 Simulation Parameters

Inputs to the simulation model that were not varied during different simulation runs are termed simulation parameters. The values selected for these parameters affected how accurately the simulation modeled the actual system. The simulation parameters are given in Table 6.1.

Table 6.1: AGG-MAC Simulation Parameters

Simulation Parameters	**Values**
WLAN PCF Functionality	Enabled
WLAN Access Point Functionality	Enabled
WLAN Physical Layer Characteristics	Direct Sequence
WLAN Data Rate	$11Mbps$
Beacon Interval	0.02
CFP Interval (sec)	0.018
CFP Beacon Multiple	1
UMTS Cell State	DCH

6.5 Simulation Factors

Inputs to the simulation that are varied during different simulation runs are termed simulation factors. The simulation was run with different combinations of these factors as described in Section 6.7. The simulation factors are summarized in Table 6.2 and discussed below.

Table 6.2: AGG-MAC Simulation Factors

Simulation Factor	**Values**
Packet Size	32B - 2,304B
Number of Transmitting Stations	1, 10, 50
BER	0, 10^{-8}, 10^{-5}, 10^{-4}, and 10^{-3}
Fragmentation Threshold	$150B$, $300B$, $576B$, $1,500B$, $2,304B$
Application Profile/Traffic Model	FTP, E-mail, HTTP, and Voice_pcm

6.5.1 Packet Size

Packet Size affects not only traffic loads as related to capacity of the channel, but also directly affects the potential for aggregation. If all packets are larger than one half the size of the fragmentation threshold, the AGG-MAC protocol will perform exactly like the original WLAN MAC protocol. The amount of aggregation depends on the arrival rate, arrival sequence and packet size offered to the MAC layer protocol.

The packet size was varied from $32B$ to $2,304B$. These values were selected to represent small control packets from higher layer protocols up to the maximal allowable MSDU. Research has shown that a significant percentage of internet traffic is less than $100B$ in size [44][5]. The IP protocol recommends an upper limit of $576B$. All IP networks are required to support at least this limit [47]. For application based traffic, packet sizes are set within the definition of the protocol in the form of a probabilistic distribution, and the sizes are impacted by the TCP MSS value and the IP maximum MTU value. In those simulations, the full IP protocol stack was employed with its default values. This implies that the IP maxMTU was set to $576B$, and the TCP MSS was set based upon the value of the IP maxMTU.

6.5.2 Number of Transmitting Stations

The basic simulation was run with 1, 10, and 50 transmitting stations. These values represented the number of stations sharing access to the WLAN channel. These values were selected because they represented a reasonable range of clients that a typical WLAN might be expected to support [48]. The combination of the application profile, packet size, and number of stations determined the traffic loads for the WLAN channel.

6.5.3 BER

In a digital transmission, bit error rate (BER) is the percentage of bits with errors divided by the total number of bits that have been transmitted, received or processed over a given time period. The rate is typically expressed as 10 to the negative power. For example, four erroneous bits out of 100,000 bits transmitted would be expressed as 4 x 10-5, or the expression 3 x 10-6 would indicate that three bits were in error out of 1,000,000 transmitted. BER is the digital equivalent to signal-to-noise ratio in an analog system. We varied the bit error rate (BER) of the wireless channel to assess the performance of AGG-MAC under more realistic conditions. The channel model we used employed additive white Gaussian noise

(AWGN) based models for BER. We evaluated the performance using a perfect channel, and BERs of 10^{-8}, 10^{-5}, 10^{-4}, and 10^{-3} to represent increasingly worsening channel conditions.

This factor will demonstrate the overhead incurred from attempting to transmit larger frames which encounter an increased probability of error in transmission of an aggregate frame. This factor is varied in conjunction with fragmentation threshold and will be compared with results from [33]. The characteristics of the noise additionally had a measurable effect on WLAN performance. For this reason, we also considered the effects of a single periodic bit error that would produce the desired BER and of a periodic burst error. The effect of periodic burst errors were evaluated for various noise ON/OFF periodocities. We considered time periods where a burst of noise yielding the desired BER was repeated every $1sec$, every $0.5sec$, and for the case where the noise was active occurred for exactly one_bit. Not surprisingly, the same BER with more distributed errors affected more frames, and therefore negatively impacted performance more than longer burst errors.

6.5.4 Fragmentation threshold

As stated in Section 6.1, the fragmentation threshold establishes the upper limit on the size of a frame transmitted over the WLAN channel. AGG-MAC will not create frames larger than the size established by the fragmentation threshold in order to avoid triggering the fragmentation algorithm, thereby inducing significant additional overhead.

In the evaluation of AGG-MAC performance under varied channel conditions, we varied the value of fragmentation threshold based upon data presented in [33]. We compared the performance of AGG-MAC with both the original WLAN MAC performance in our simulation, and with the computed values presented in [33].

6.5.5 Traffic models

We ran validation and verification testing with basic station models transmitting generic frames. Subsequent simulations were conducted with workstation models capable of simulating the entire protocol stack. Our traffic was based on the OPNET defined traffic application profiles "wireless_client," "Voice_pcm," "Web," and "FTP." Section 4.7 presented an overview of most of the applicable traffic models, which will again be used in simulations evaluating the AGG-MAC protocol enhancement. Section 6.7.2 describes the simulation environment for the Infrastructure Mode simulation designed to represent shared access for multiple stations with differing traffic requirements.

Voice_pcm Traffic Models

The *Voice_pcm* application models PCM quality VoIP traffic. A voice application enables two clients to establish a virtual channel over which they can communicate using digitally encoded voice signals. UDP is the default transport protocol used for this application. The voice data arrives in spurts that are followed by a silence period. Encoding schemes can be specified for the voice-to-packet translation. The Voice_pcm application was parameterized using the attributes in Table 6.3 [40].

Table 6.3: AGG-MAC VoIP Application Attributes

Attribute	**Description**
Silence Length (sec)	Silence length for the incoming and outgoing calls along with the associated distributions.
Talk Spurt Length (sec)	Length of a talk spurt for the incoming and outgoing calls along with the associated distributions.
Symbolic Destination Name	Symbolic destination name of the client.
Encoder Scheme	Encoding scheme in effect at the client.
Voice Frames per Packet	Number of voice frames that can be sent in a single packet.
Type of Service	Quality-of-service parameter for assigning priority to this application's traffic.
RSVP Parameters	RSVP parameters for making bandwidth reservations.

6.6 Evaluation Techniques

As stated in Section 4.6, the selection of a particular evaluation technique can significantly impact the outcome of a performance evaluation. Three possible techniques of performance evaluation are analytic, simulation, and measurement [37]. These methods differ in terms of accuracy, cost, and required time. We again choose simulation to conduct this performance analysis due to limitations in available resources. Analytical methods were used in the model verification process.

6.7 Simulation Design

The rationale and design overview of the simulation-based experiments employed to evaluate the AGG-MAC protocol enhancement are contained in this section. The results and analysis from each of those simulations are located in Chapter 7. We again used standard OPNET^{TM} models and modified them to support our AGG-MAC protocol. Our initial work was to conduct verification and validation testing to ensure the proper and correct functioning of the AGG-MAC protocol and the associated model. Those results are presented in Section 7.1 and in Section 7.2. In addition to basic verification and validation testing, we conducted a range of simulation experiments to better analyze and characterize the performance of the AGG-MAC protocol enhancement as compared to the original WLAN MAC protocol. We conducted simulations to better understand the effects of noise on the channel. The design of our testing system operation over a channel subject to errors due to noise is presented in Section 6.7.1 and results and analysis of various BER situations are presented in Section 7.3. We present the simulation design for a reasonable office network scenario in Section 6.7.2 with the results of those simulations presented in Section 7.4.

Section 6.7.3 presents an overview of our rationale and approach for using live trace data to evaluate and compare the performance of AGG-MAC on actual network traffic. We present two very distinct cases in order to better understand the performance of the AGG-MAC protocol enhancement. Section 6.7.4 provides an overview of the problem space and a discussion of the approach to analyzing the performance of AGG-MAC on traffic generated by the U.S. Army's Land Warrior proof of concept exercise. The results from the Land Warrior traffic trace, representing a small data set collected from an ad hoc network configuration are presented in Section 7.5. Our second live data set involves over 1GB of collected trace data from an infrastructure-based network at the IEEE SIGCOMM Conference 2001; the overview of the situation and discussion of our simulation design are found in Section 6.7.5 with results from these simulations included in Section 7.6.

Our final experimental scenario involves the incorporation of the AGG-MAC WLAN protocol enhancement incorporated into a UMTS-WLAN interworked system. The rationale and design of the UMTS-WLAN interworked system were discussed at length in Chapter 4 with results from the baseline system without the AGG-MAC protocol enhancements included in 5. In Section 6.7.6 we identify the rationale and methodology we employed to consider the effect of incorporating the AGG-MAC WLAN protocol enhancement into the UMTS-WLAN interworked system. Our results from these simulations are located in Section 7.7.

6.7.1 Effect of Errors on the Channel

Noise in a wireless radio channel is a common occurrence. From the perspective of the MAC-layer protocol, loss of data during transmission is the result of noise on the channel. Wired networks generally have a very low noise level and therefore, the probability of error on a wireline channel is often negligible. Most higher layer protocols were designed to operate under conditions associated with wired networks. For this reason, frame loss is generally assumed to be the result of increased congestion along the path rather than due to bursty losses associated with noise on the channel. The issues associated with using these assumptions on a wireless network have been well studied [30][31][33][35]. The BER-related simulation experiments were designed to better understand how the employment of the AGG-MAC protocol would function across a range of noise conditions.

One of the potential negative results from packet aggregation is that a larger frame incurs a greater probability of encountering an error during transmission which results in the loss of a larger number of bytes of data. Lettieri, et al. [33] presented analytical and experimental results (summarized in Section 3.2.2) pertaining to the performance of WLAN systems under various BER conditions. We sought to confirm their results with an original WLAN and to evaluate AGG-MAC under the same conditions. We additionally sought to identify an appropriate value for the fragmentation threshold for a given channel BER. We used workstation models to evaluate the achievable throughput across the range of factors as identified in Table 6.4. We selected the values of BER to best correlate to the results presented in [33].

Table 6.4: AGG-MAC Simulation Factors for BER

Simulation Factor	**Values**
Packet Size	$32B$ - $2,304B$
Number of Transmitting Stations	$1, 10, 50$
BER	$0, 10^{-8}, 10^{-5}, 10^{-4}$, and 10^{-3}
Fragmentation Threshold	$150B, 300B, 576B, 1,500B, 2,304B$

Modeling Noise on WLAN Channel

Our WLAN MAC-layer model, AGG-MAC, was further modified to include the highlighted parameters listed in Figure 6.3. The parameter "*Is Noise Active*" was a BOOLEAN value toggled and globally monitored by each station. "*Noise_ Burst_ OFF*" and "*Noise_ Burst_ ON*" were each double precision numeric values which established the length of time, or periodocity, of noise bursts. Code was added to the model to perform the required logic to process the noise on the channel. The model updated the status of "*Is Noise Active*" based upon the values of the ON/OFF parameters. During reception

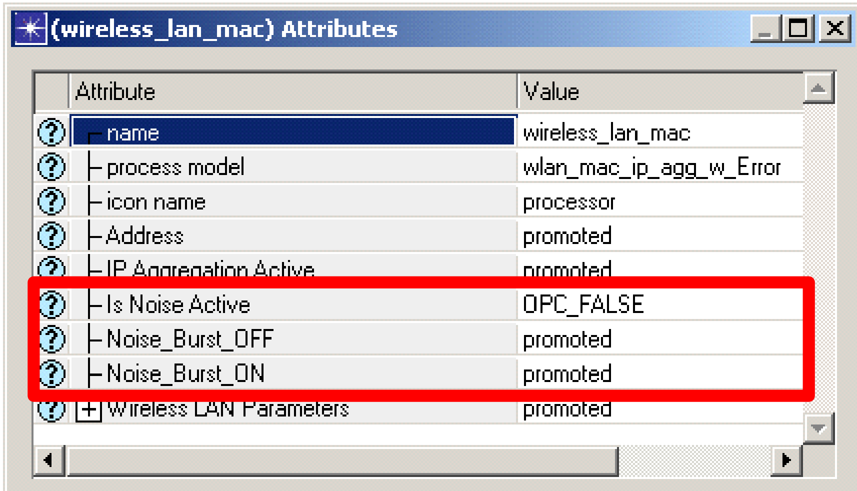

Figure 6.3: MAC-layer model parameters for BER

of a packet, the receiver compared the time period of the packet's transmission to the time period of the most recent burst of noise. If there was any overlap in the packet transmission and the noise burst, the packet was marked to be dropped due to errors. This processing occurred at the receiver to ensure that packet collisions were still properly considered in all cases. Statistics were updated and the packet was subsequently dropped when collision or noise dictated it.

BER Simulation Configuration

We based the network simulation design used to test the effects of bit error rate on the same fundamental design as was used for validation testing, see Section 7.2. In order to test the effect of errors on the channel, we added periodic bursts of errors into the simulation using the mechanisms discussed in the previous subsection. We additionally repeated the simulations with 1, 10 and 50 transmitting stations operating in DCF mode. The network diagram for a single transmitting station was the same as presented in Figure 7.2, and the diagram for 50 stations is presented in Figure 6.4. In each case, the receiver is labeled *wkstn_2* and each of the other nodes in the diagram is a transmitter. We positioned the transmitters in close physical proximity to one another in order to provide the fairest

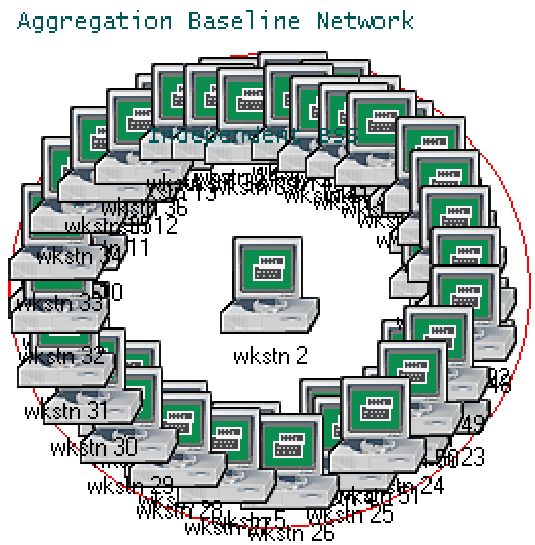

Figure 6.4: Baseline 50 Station AGG-MAC Network

opportunity for successful transmission among competing stations. In our environment, each of the transmitting stations generated enough data to saturate the shared, contention-based channel. This provided us with an opportunity to observe, in effect, 1, 10 and 50 continuous sources, each attempting to saturate the channel. When periodic errors were introduced, we examined the effect of the errors for a range of both number of clients and a range of BERs. We additionally sought to obtain confirmation of the analytically derived appropriate value for fragmentation threshold for a given channel BER as presented in [33] while testing our BER scenarios. The results of our BER simulation are presented in Section 7.3. The effects of various burst characteristics of the noise are presented in Section 7.3.3.

6.7.2 AGG-MAC Infrastructure Mode

In order to determine the performance of AGG-MAC under more realistic and varied conditions, we compared its performance with OPNET™ provided scenarios. We have focused our efforts on the scenarios provided in the project Wireless_LAN [41] and in the OPNETWORK 2002 Tutorial 1332 titled “Maximizing Performance for Wireless LANs””

[49]. As we anticipated, for traffic streams with small packet sizes, we were able to improve throughput and generally decrease delay.

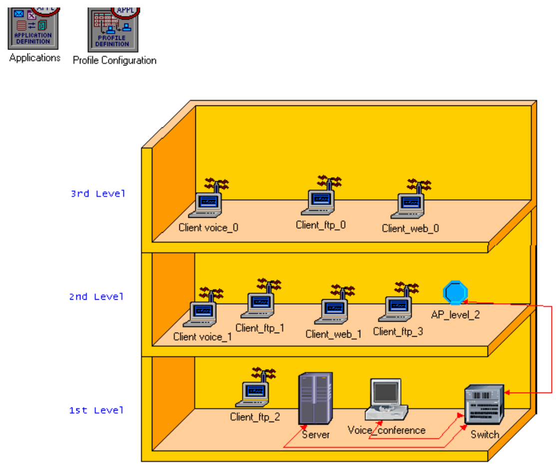

Figure 6.5: Network Configuration

Figure 6.5 presents the network configuration. In this office environment, a single AP provides network connectivity to eight client machines via the WLAN. Each client was designed to use a single application. Two were PCM quality voice, two were HTTP traffic and four were transferring FTP traffic. In the scenario *PCF_ disable*, all clients were operating in DCF mode in which all traffic was pure contention based. In the *PCF_ enable* mode, preference was given to the voice clients by configuring a $20mSec$ contention-free period (CFP) approximately every $30mSec$ in which voice stations were the only clients permitted access to the medium. We used the provided reference model in the tutorial and strictly replaced the *wlan_ mac* process model with our *wlan_ mac_ ip_ agg* process model to test our enhancements. AGG-MAC requires a value for fragmentation threshold in order to actually aggregate the traffic; we set this to the maximal value of $2,304B$. This represented performance on a perfect channel with the best possible results, just as the case was in the original tutorial.

Our analysis included result from four different scenarios and are presented in Section 7.4. Two scenarios were the original unchanged MAC protocol called *PCF_ disable* and

PCF_ enable, and the other two scenarios were our AGG_MAC protocol called *PCF_ disable_ agg* and *PCF_ enable_ agg*.

6.7.3 Trace Traffic

We evaluated the performance of AGG-MAC for live data. We acquired WLAN traffic traces from two sources. The first source was a US Army platoon-level exercise testing prototype systems which employed the Land Warrior communications system, see Section 6.7.4. The second was trace data collected during the ACM SIGCOMM'01 conference held at UCSD in August 2001, see Section 6.7.5. Each of the traces provided an opportunity to evaluate AGG-MAC performance on actual user traffic. They represented two very different types of communication. The Land Warrior data was a very small data set of an ad hoc network in DCF mode. The communications were almost exclusively multicast. The SIGCOMM'01 trace was an extremely large data set consisting of PCF mode clients that accessed the Internet via four Access Points. While the Land Warrior data was collected over a period of approximately thirty minutes, the time necessary to conduct the military operation, the SIGCOMM'01 traffic was collected over three full days of the conference and represented over 300,000 flows from 195 distinct users.

We used a manual analysis of the Land Warrior traffic data in order to better understand the effects of AGG-MAC on such a data set and we ran OPNET™ simulations using the actual traffic flows with the SIGCOMM'01 traffic captures. The Land Warrior data offered little statistical significance, but did provide a single sample of an interesting class of real traffic in a real situation. The SIGCOMM'01 trace data offered a much better statistical significance in terms of both users and time. The results collected from these two scenarios significantly improved our understanding of the benefits and limitations offered by AGG-MAC over a wide range of conditions.

6.7.4 Land Warrior Trace Traffic

Land Warrior Background

The US Army's Land Warrior (LW) system was designed to interconnect the individual infantry soldier on the ground to the Tactical Internet. The purpose of this system was to enhance the ground forces' ability to deploy, fight and win on the battlefields of the 21st century. The Land Warrior design considered the soldier to be a system of systems, tightly integrated to form a complete weapons system. The soldier generated and consumed data from many different sources. This data was transferred among the soldiers in the platoon via a WLAN. Separate radio links served as gateways to send information from the platoon to headquarters, to other units and to the rest of the Tactical Internet. Bandwidth limitation was a major consideration throughout the design of the system. The current prototype systems were quite judicious in limiting the size and quantity of the data transmitted over the system. We believed that as the system continued to mature, the capacity requirements would continue to increase significantly, but that many artifacts of the small message sizes would persist. Since WLAN was a shared bandwidth medium, an increase in traffic for an individual soldier reduced the total available capacity for all elements of the WLAN.

Figure 6.6 illustrates the hierarchical unit organization. Eight individual soldiers and a squad leader comprised a squad. Three squads were grouped along with additional leadership and a headquarters section to form a platoon. The platoon was led by the platoon leader and was generally comprised of approximately 30-40 soldiers. The next level higher, the company, was comprised of three platoons and a headquarters element.

Traditionally, communications among soldiers at the platoon-level had been conducted via either non-technical means or through the use of voice radio systems. The LW system was designed to enhance the individual soldier through an infusion of information technology [50]. The communications subsystem provided integrated voice and data com-

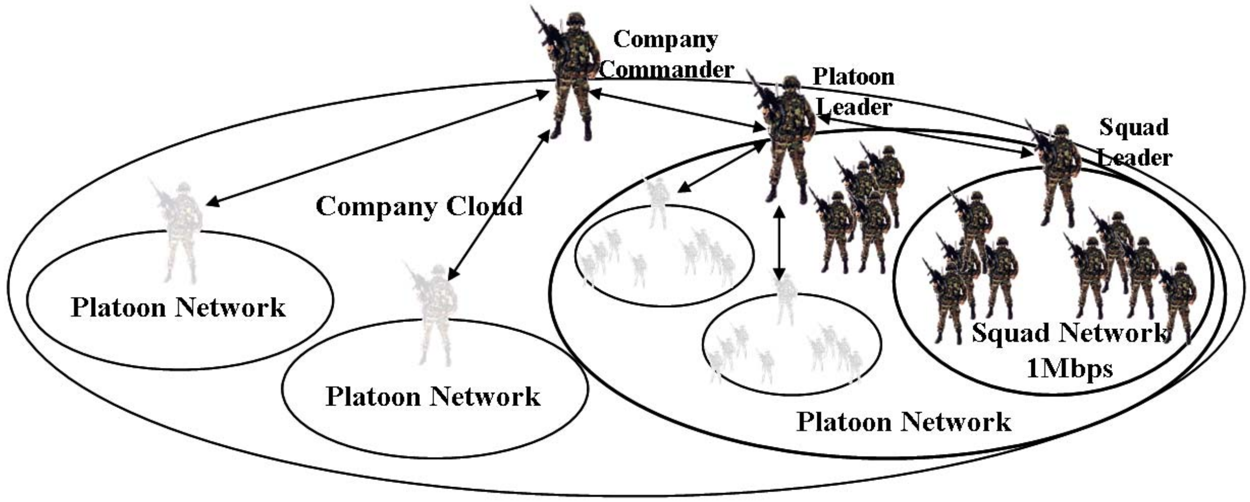

Figure 6.6: Land Warrior Network Heirarchy

munications capabilities to dismounted Infantry soldiers within a platoon. At the individual level, each soldier was a personal area network (PAN) that interconnected the various sensors, communications devices and automated systems through a combination of wired and wireless channels [51]. This information allowed the Land Warrior system to collect and process information available to each individual soldier. A significant portion of the data generated by those sensors, communications devices and systems was useful to other soldiers within the platoon, or potentially even by soldiers beyond that platoon. In order to support the squad and platoon communications, each individual soldier was also a communications node within an 802.11 WLAN interconnecting members of the platoon. Because of the dynamic nature of soldiers' movement in the platoon, an ad hoc network was employed in which each WLAN also acted as a router to forward messages within the platoon. To support communications beyond the platoon, LW augmented the leader's configuration to act as a gateway to higher headquarters or other units via Single Channel Ground and Airborne Radio System-Advanced System Improvement Program (SINCGARS-ASIP) radio [51].

Exercise Data Origin

We were provided with captured traces from an actual Land Warrior exercise conducted in 1999 at Fort Polk, Louisiana by a platoon of approximately 40 soldiers from the 82nd Airborne Division. The exercise was a proof of concept and used prototype equipment. The results were considered a major success and the program continued to move forward although it was still in the process of significant modifications and enhancements. An open area of concern was and continues to be scalability of the system to ultimately support multiple and larger units on the battlefield.

Though useful, the exercise data, presented several issues for our purposes. The quantity of data itself was limited. Our sample data covered only a period of approximately fifteen minutes of actual network operation. This provided only a single sample of a single type of operation – and communications patterns could differ greatly for other types of operations.

The only data in the trace was internal platoon communications. In a fully developed Land Warrior system, additional data from external sources would enter and propagate through the network. Presumably, additional reported data would also propagate outward beyond the platoon. Only IP-level header information was available in the trace. No WLAN specific traffic such as collisions, or protocol management traffic was collected. One significant limitation for our purposes was having only transmission time available. For this reason, we assumed a $30ms$ window to account for variations in queuing in the MAC or PHY layers. The data also did not include retransmissions for subsequent hops across the network, so it is unclear how many retransmissions of multicast traffic occurred. Each of these factors should be considered when evaluating the overall performance of the WLAN network.

There were four distinct types of data packets in the current Land Warrior system: Active Soldier data, E-mail data, Overlay data, and Voice over IP (VoIP) data. There were currently several future data sources planned, including biometric and other sensor data, as well as data interconnecting additional peripherals. Active Soldier data was GPRS location data transmitted every minute to update soldier's location in the common operating picture. E-mail data format was used to transmit text messages including predefined messages such as SALUTE report, (SALUTE report includes {Size, Activity, Location, Unit or Uniform, Time, and Equipment} information of potential intelligence value and was transmitted to higher headquarters upon observation), call for fire, request for medical support, and several others. Overlay data consisted of graphics images transmitted to other soldier's such as map graphics or battlefield images. VoIP packets were used to transmit voice communications among the members of the platoon. Each of these packet types had differing constraints of service requirements for the servicing network. Research into QoS attempted to address these requirements. Our approach recognized these benefits and was intended to fully support and potentially enhance QoS-aware implementations.

The current LW architecture employed a multicast-based addressing scheme. This created a number of effects on the resultant network traffic. As most of the data generated was intended for multiple recipients, repetitive unicast transmissions to each destination were avoided. This design decision also prohibited the use of reliable MAC layer protocol mechanisms, so UDP packets were used in the transmission of all the data across the network. Application layer protocols were used to handle reliability and retransmission. The mechanics of these application protocols were not available to the authors, and was therefore not considered. We considered each packet for transmission as an individual entity and assumed it was processed accordingly. Results of our analysis of the Land Warrior trace data are presented in Section 7.5.

6.7.5 ACM SIGCOMM'01 Trace Traffic

Researchers at UCSD captured user traffic during the conduct of the ACM sponsored SIGCOMM'01. The duration of both the conference and the resulting trace was three days.

The researchers analyzed the trace and presented their results in [5]. Table 6.5 [5] presents high-level observations derived from the traffic trace.

Table 6.5: Overall Statistics for SIGCOMM'01 Traffic Trace

Attribute	**Values**
Number of wireless users	195
Maximum users at an AP	32
Total hours of trace	52
Total bytes transmitted	$3.5GB$
Total flows	$298,995$
Peak throughput at an AP	$3.2Mbps$

The configuration of the network is presented in Figure 6.7 [5]. The four APs were installed in the conference auditorium and provided overlapping coverage within the auditorium. The client hardware was provided by the users and the trace data included traffic from eight different vendors of WLAN [5]. The paper [5] analyzed the traffic by protocol type and application type. They defined three types of sessions: light, medium, and heavy and evaluated traffic based on their session definitions. As indicated in Table 6.5, the peak throughput at any AP was $3.2Mbps$.

We used this rich traffic trace to better understand the potential for AGG-MAC to provide improved throughput and delay on real traffic patterns. We sought to confirm potential for performance improvement using packet aggregation in a WLAN network using real, rather than statistical, traffic patterns. We analyzed more than $1GB$ of trace data and used dumps from MIB (Management Information Base) data collected using a SNMP (Simple Network Management Protocol) dump from the access points to locate clients within the WLAN network.

The MIB provided us with the MAC address of all associated stations. Using this data, we knew how many stations were associated with each of the access points under consideration at the sampled instance in time. In order to make our simulation more tractable, we only simulated one of the four APs in the network. This simplification was reasonable because the APs neither interfered nor interacted with one another. We identified

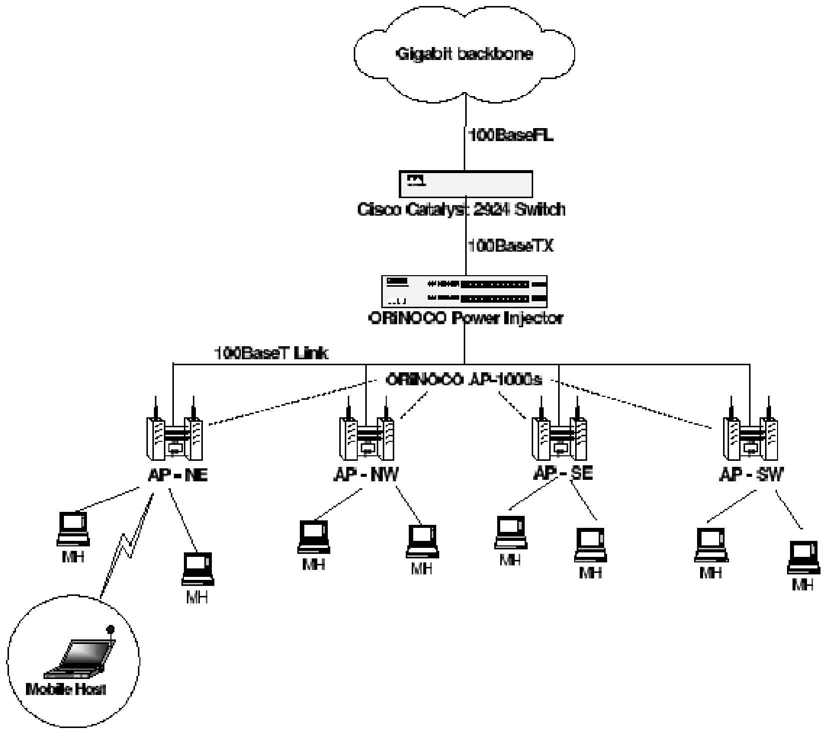

Figure 6.7: SIGCOMM'01 Network Configuration [5]

all of the clients associated with AP-NW at 11:00, the peak-hour, on Thursday, the busiest day of the three-day conference. We randomly selected the AP-NW, one of the two busiest APs, and conducted a limited search of the other APs in order to ensure that AP-NW was representative of the overall population. Our analysis of the trace data also revealed that due to the use of DHCP-based addressing, the IP addresses of the clients was allowed to change throughout the conference. We realized therefore that we were required to use the client adapter's MAC address in order to uniquely identify an individual client machine.

Using the MAC address of each client associated with the AP-NW as a filter, we used a C++ program to create an individual packet trace for each of the 25 clients associated with the AP-NW. The C++ program incorporated the WinPcap Developer's pack library routines to read each packet from the original trace and write it out to another trace file if the desired client was either the source or the destination address. This produced 25 trace files, one for each desired client, that contained all captured packets where the desired client was either a source or destination in the ethernet packet - in effect, a trace file for each client. This allowed us to use the OPNET^{TM} Application Characterization Environment (ACE) to analyze traffic flows for each specific client from which we created an application profile to generate traffic on the network representative of the trace traffic generated by each client.

6.7.6 UMTS-WLAN w/AGG-MAC

We evaluated the performance of AGG-MAC in the interworked UMTS-WLAN system we discussed in Chapters 4 and 5. As previously stated, the focus of Tracy Mann's thesis work [8] was to develop simulation experiments to compare the performance of an interworked system against each of the baseline technologies, UMTS and WLAN. In order to evaluate the performance of an interworked system, Section 4.9, which has been augmented with AGG-MAC, a WLAN enhancement, we compared the performance of the original UMTS-WLAN system with the performance of the same system using AGG-MAC at the WLAN MAC protocol layer. We based our comparison on Mann's simulation scenario *UW-Heavy* [8] because it evaluated the performance of the WLAN portion of the network for a varied number of clients passing data traffic based upon a mix of *http*, *email*, and *ftp* sessions. The original purpose of these experiments was to compare the performance of the UMTS-WLAN interworked system against a baseline original WLAN system. The original UTMS-WLAN interworked system was considered to be the baseline for our purposes in order make a comparison with a similar system enhanced with AGG-MAC. The network diagram for these experiments is shown in Figure 6.8. The results of these simulation experiments are presented in Section 7.7.

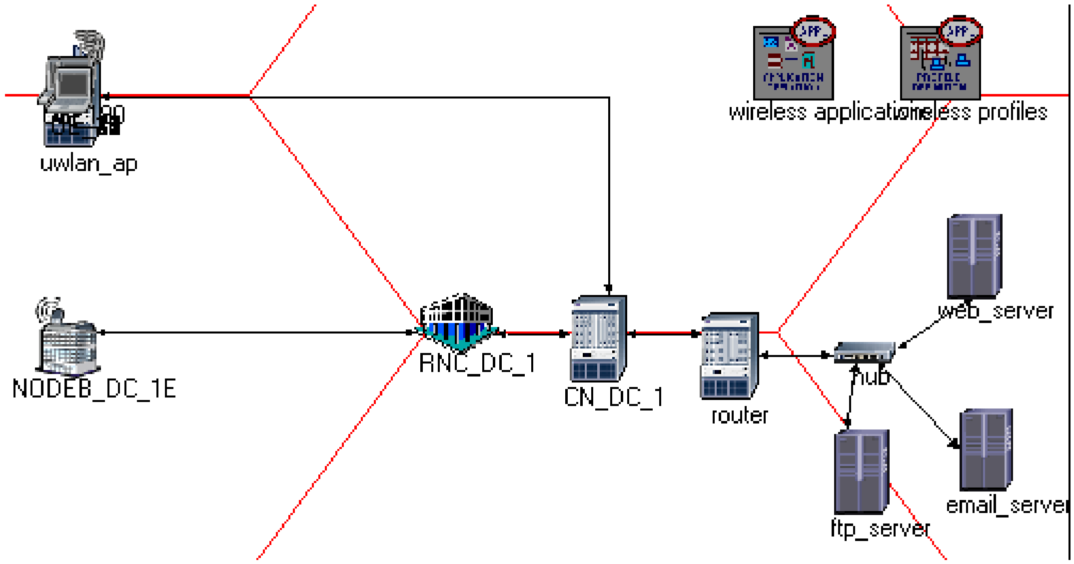

Figure 6.8: UMTS-WLAN with AGG-MAC Network Diagram

6.8 Summary

This chapter presented the design of the simulation model and the methodology used in the development and evaluation of the AGG-MAC protocol. Section 6.1 provided a high-level overview and discussion of the design space. More detailed discussion of the protocol was presented in Section 6.2. We presented values for performance metrics used in our simulation environment in Section 6.3, simulation parameters and simulation factors were presented in Section 6.4 and Section 6.5 respectively, and the design of each of the simulation experiments were presented in Section 6.7.

We tested AGG-MAC in both the PCF and the DCF modes, and determined that it could be useful in a very broad range of situations. We simulated performance using both statistically generated and actual captured live trace traffic data. Our analysis considered the performance of the protocol across a wide range of conditions. We present the results of validation, verification and our other simulated experiments next in Chapter 7.

Chapter 7

AGG-MAC Results

7.1 Model Verification

Model verification is the process of determining if a simulation model functions correctly. This includes such tasks as debugging the computer code, testing for logic errors, and testing the functionality of different modules. As discussed in Section 6.1 and Section 6.2, the simulation was designed in OPNET^{TM} Modeler 9.1.A using a top-down, modular approach. This approach simplified the task of model verification since each module can be tested independently.

Each node and process in the simulation was tested to verify that it functioned correctly. This was accomplished by running short simulations and collecting statistics at various points in the model. Standard OPNET^{TM} processes were tested to ensure they functioned as described in the software documentation.

In order to verify correct operation of the AGG-MAC model, we created a basic scenario in which the network consisted of only two simple, custom stations (Figure 7.1). The only traffic on this network was generic packets transmitted from *station_1* to *station_2*. We based our source on the process model *comm_station*, and made modifications to compute

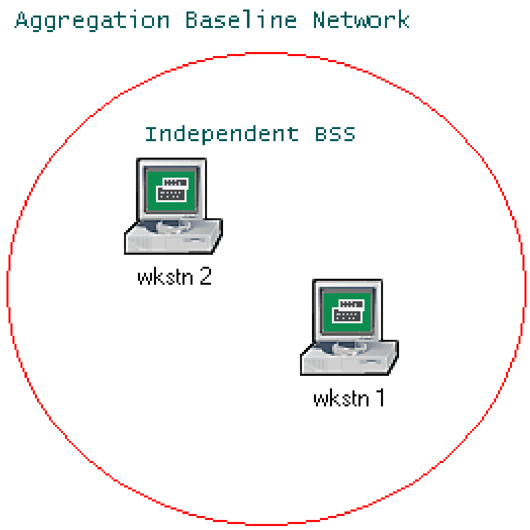

Figure 7.1: Aggregation Baseline Network

and set the *arrival_rate* based on the promoted parameter, *packet_size*. We designed the source process to produce packets of the requested size at a rate of 2 $Mbps$, based on either a constant or exponential distribution.

Our initial verification was to ensure that packets coming into our *wlan_mac_ip_agg* process model were correctly leaving our process model and being sent to the correct destination. Also we ensured that packets were not in any way resequenced by *wlan_mac_ip_agg* to ensure that current in order delivery assumptions were not modified. We recognize, see Section 6.1.3, that additional performance enhancements could be obtained through resequencing, but that mechanism remains an open topic for future research. We collected each packet transmitted, verifying both correct and in-order transmission and receipt of packets from transmitting node to receiving node over the wireless channel were correctly performed. In addition to basic verification, our subsequent testing included operation with a fully simulated protocol stack (workstation) in which we confirmed correct operation during interaction with higher layer protocols.

7.2 Model Validation

Model validation is the process of determining if a simulation model is representative of the real system. A simulation can be validated using expert intuition, real system measurements, or theoretical results [37]. Comparing simulation outputs and measurements from a real system is the most reliable way of validating a simulation model. Real system measurements were not available in this research, since this was not yet a currently implemented protocol. Comparing simulation and theoretical results was the primary method used to validate the simulation model. Theoretical analysis of the system was conducted using both deterministic and exponential traffic sources.

To validate the expected performance improvements, we computed the expected throughput and delay based on overhead associated with a packet in the original and in our aggregated wireless LAN protocol. Our results were consistent with the computed values, as Figure 7.2 indicates by plotting the throughput for the original WLAN and for AGG-MAC. As expected, and predicted analytically, the original throughput was severely limited for small packet sizes while AGG-MAC provided near optimal throughput at high load.

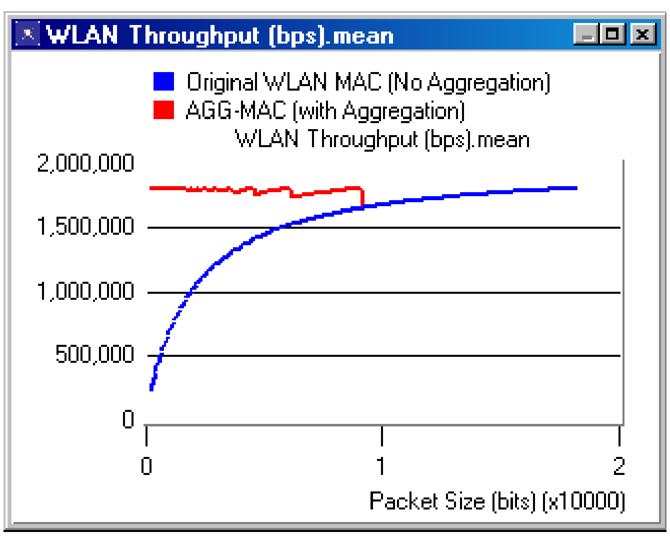

Figure 7.2: Avg. Throughput (Constant 2Mbps Source)

Figure 7.3 presented the average bits per second dropped due to buffer capacity and Figure 7.4 similarly presented the average delay for both cases. Clearly AGG-MAC provided significant performance benefit for small data streams with small packet sizes and performance consistent with the original WLAN for packet sizes too large to aggregate. In such a case, throughput was near optimal, and delay was lower and more constant with less data dropped.

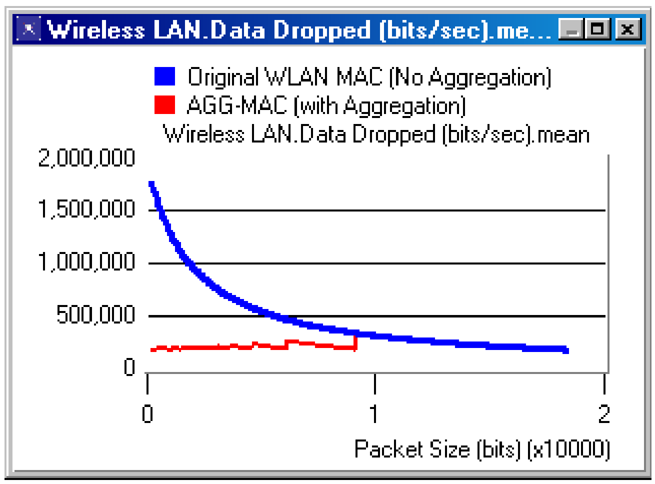

Figure 7.3: Avg. Dropped Data (Constant 2Mbps Source)

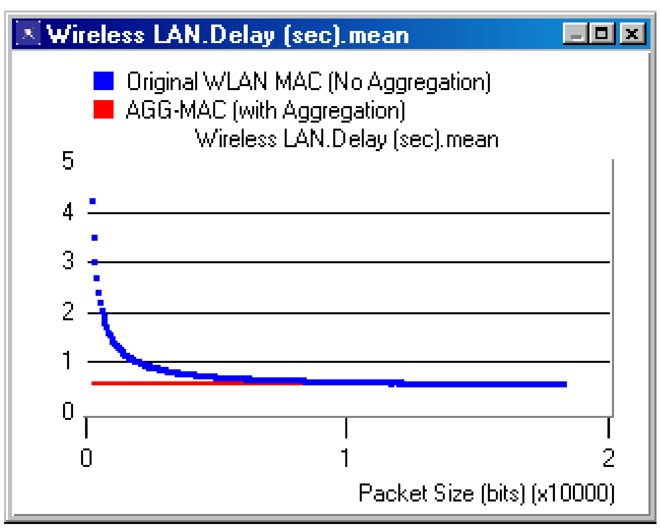

Figure 7.4: Avg. WLAN Delay (Constant 2Mbps Source)

The saw tooth effect in the red AGG-MAC traces (Figures 7.2 and 7.3) was due to the increasing step sizes where the number of packets that was aggregated changed. Best performance occurred where the total size of the aggregate frame was equal to the maximum fragmentation threshold, $2,304$ *Bytes*. The drop in the saw tooth occurred where an additional incremented byte in packet size caused fewer packets to be aggregated. Larger packet sizes created larger saw tooth because the step size between number of packets aggregated was larger.

7.3 Results for Effect of Errors on the Channel

In this section, we present the results of analysis of effect of noise on the WLAN channel. As presented in Section 6.7.1, we ran an extensive set of simulations designed to better characterize the performance of AGG-MAC protocol and to determine its suitability under anticipated operating conditions. Section 7.3.1 demonstrated the performance of a single transmitting station over varied error conditions. In Section 7.3.2, we expanded on the insights gleaned from Section 7.3.1 to consider the effects of multiple stations competing for the shared channel. We incorporated results for scenarios with 10 stations and for 50 stations attempting to transmit data over the shared WLAN channel. Because the characteristics of a source of noise that produced errors at a given value of BER and their resultant effect on performance can vary significantly, we addressed this situation in Section 7.3.3.

7.3.1 Single Transmitting Station

This section considered our most basic error-prone scenario. As identified in Section 6.7.1, we began with the same system used in our Validation testing, see Section 7.2, and injected periodic bursts of noise on the channel. The results presented in this section represent a single bit error where the frequency of occurrence of the error, or the length of time the channel was noise free, was set to produce the desired BER. The results produced

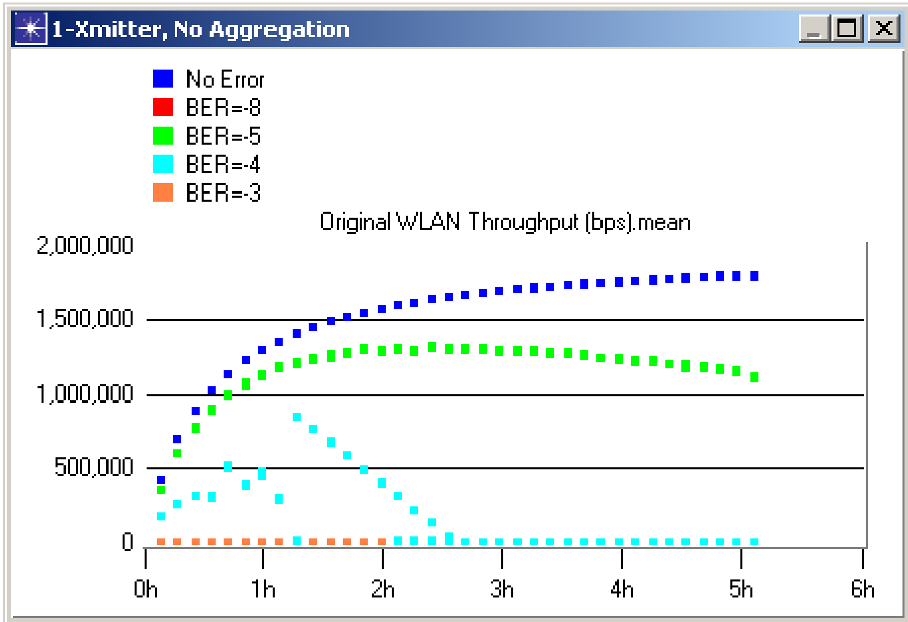

Figure 7.5: Original WLAN Throughput vs. Packet Size under varied BER

in this set of simulations were expected to be consistent with those published in [33] and discussed in Section 3.2.2. The results for the original WLAN MAC system for various Frame Sizes for varied values of BER are depicted in Figure 7.5. The achieved results are consistent with those presented [33] and discussed in Section 3.2.2 where the reader can refer back to Figure 3.1. These results illustrate the analytical results published in [33]. While the focus of [33] was on establishing the *fragmentation_threshold* as an upper limit to the size of large frames based on a low probability of successful transmission; we approached the problem from the other extreme. AGG-MAC used the *fragmentation_threshold* as, still an upper limit, but also as a target value to aggregate together smaller packets so the total system functions as close to the optimum operating point, *fragmentation_threshold*, as possible.

Figure 7.6 illustrates the performance of the same system augmented with the AGG-MAC protocol. These results are presented for a maximal value, $2,304$ *Bytes*, of *fragmentation_threshold*. Under conditions of low noise, BERs above approximately 10^{-5}, AGG-MAC with a maximal *fragmentation_threshold* outperformed or at least equaled the performance

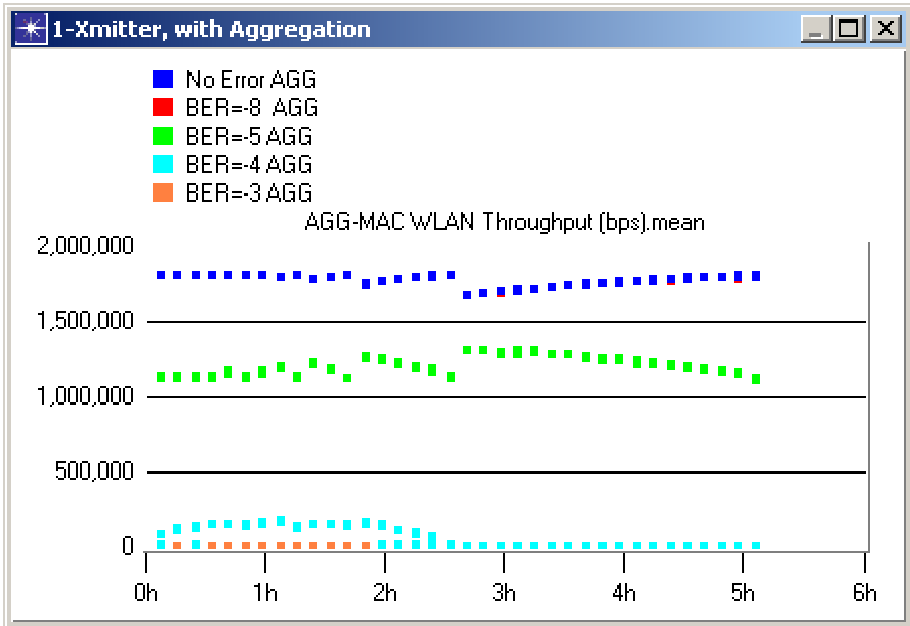

Figure 7.6: AGG-MAC WLAN Throughput vs. Packet Size under varied BER

of the original WLAN MAC. At any BER, the probability of encountering an error was higher for a larger frame than for a smaller frame. Each additional bit transmitted incurred an additional probability of encountering an error. In the comparison of AGG-MAC performance with the original WLAN MAC protocol, under varied error conditions, the real issue was identifying the operating point that allowed successful transmission of the most user data. At lower BERs, the probability of error was lower and therefore the better approach was to send as much data as possible. At very high values of BER, the frequency of occurrence of errors can be sufficiently high that the probability of transmitting a maximal, or even a near maximal, frame approaches zero. For the case of an error prone channel, the interesting question was 'can we identify the optimal frame size for a given BER?'

As seen in both Figure 7.5 and Figure 7.6, the performance of either protocol operating with a BER of 10^{-4} or worse was severely limited. In Figure 7.7, we observed that the original WLAN MAC outperformed AGG-MAC when the *fragmentation threshold* was configured for the maximal, $2,304$ *Bytes*, frame size. This result was not unexpected. This figure also identified the optimal *fragmentation_ threshold* for this system operating at BER $= 10^{-4}$,

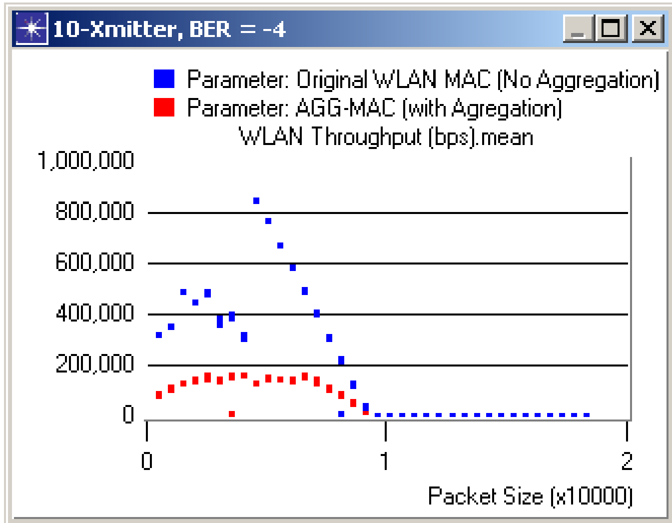

Figure 7.7: WLAN Throughput for BER $= 10^{-4}$

we observed the maximal throughput occurs where the frame size equaled 576 *Bytes*. This value is the optimal setting for the *fragmentation_threshold* for this set of conditions. This result was consistent with those published in [33]. Our observed maximal throughput was approximately $845,110$ *bps* which was slightly lower than the analytically derived value presented in [33], but was consistent with the experimental results produced in the same work. In Section 7.3.3 we will explore this finding for varied noise characteristics with the same value of BER. Before we do that, the next section considered the effects multiple competing stations had on performance.

7.3.2 Multiple Transmitting Stations

It was and is reasonable to assume that WLAN network resources will continue to be shared among a number of users or stations. This portion of our work considered how AGG-MAC performed under noisy conditions compounded with contention for the shared WLAN channel. As Section 6.7.1 described, we employed a DCF-based WLAN system operating under a range of BER noise conditions. These simulations incorporated multiple

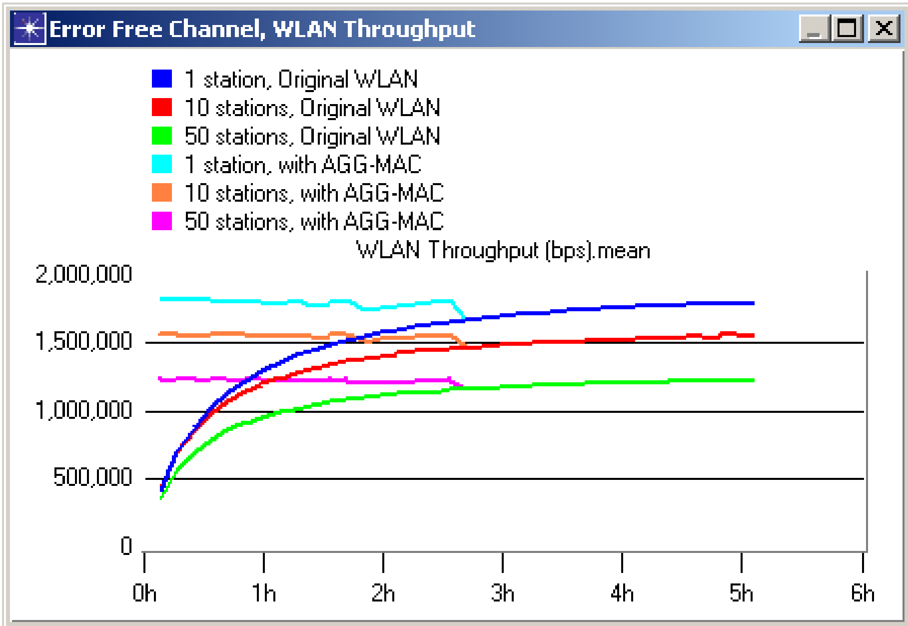

Figure 7.8: WLAN Throughput, Multiple Stations, Error Free

stations concurrently generating data and attempting to transmit over the shared channel. Each station generated sufficient traffic to saturate the system, so each station consistently had frames to transmit. We considered 10 stations to a be a nominal, reasonable operating condition and used 50 stations for a purely saturated condition.

As illustrated in Figure 7.8, under the examined set of operating parameters, the number of transmitting stations had a measurable impact on the maximum achievable throughput, but the characteristics of the throughput curves were consistent for the cases of 1, 10, and 50 transmitting stations. Figure 7.8 presents only the error free case for clarity and simplicity, however comparison of Figure 7.5 and Figure 7.9 illustrates the point for the original WLAN MAC protocol. Similarly for the case of the performance of WLAN systems augmented with the AGG-MAC protocol, compare Figure 7.6 with Figure 7.10 to observe that the characteristic curves are similar.

Under the least favorable conditions, we observed that the performance of the systems was more dependent upon the errors in the system than on the amount of contention in the

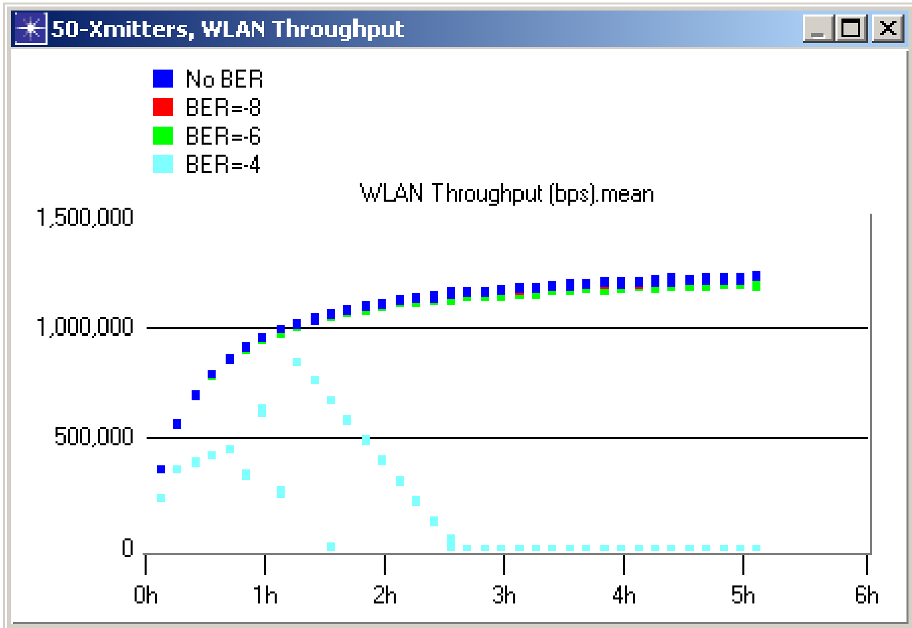

Figure 7.9: Original WLAN Throughput vs. Packet Size under varied BER (50 stations)

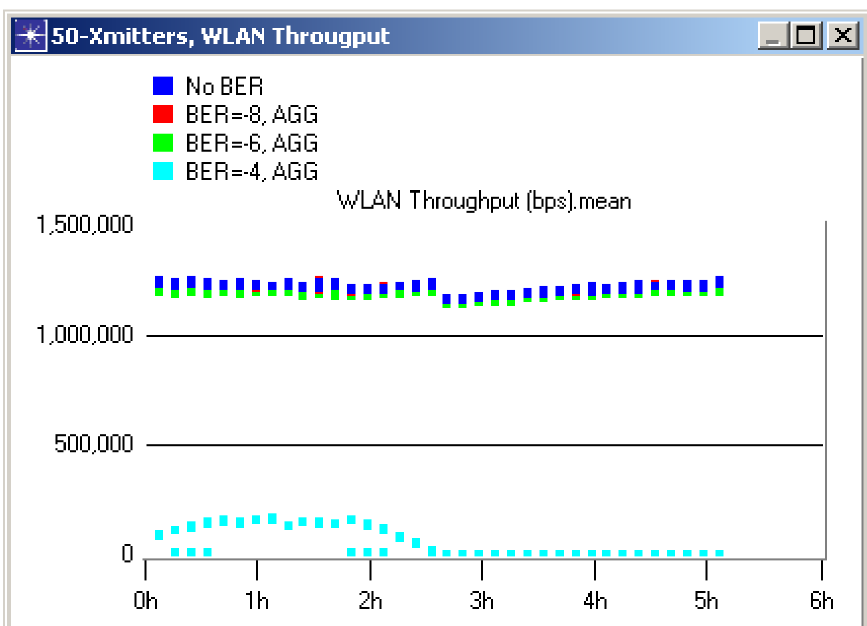

Figure 7.10: AGG-MAC WLAN Throughput vs. Packet Size under varied BER (50 stations)

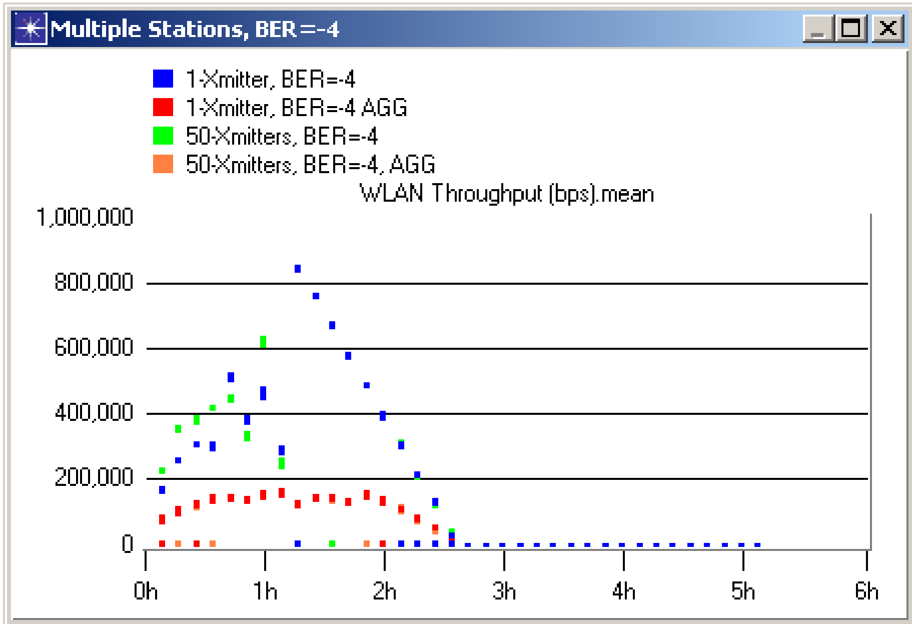

Figure 7.11: Multiple Station Total WLAN Throughput for BER=10^{-4}

system. Figure 7.11 illustrated the performance of the system, regardless of number of users, was directly connected to the BER encountered on the channel. In the next section, we examine the effect longer or shorter bursts of noise on the channel.

7.3.3 Burstiness of the Noise

As bit error rate is, in fact, a measure of the rate and a single dimensional value, it necessarily abstracts away much information about the characteristics of the source of the errors. In order to produce a BER = 10^{-4}, a single bit error would occur every $10,000\ bits$, while a repeated burst error of $1,000\ bits$ in length would be repeated only once every $10,000,000\ bits$. In this section, we examined the effects of differing lengths of burst errors, or the characteristics of the noise on the channel. We identified this as one aspect of [33] that could have been more explicitly described in better understanding the parameters they used. As seen in Figure 7.12, the characteristics of the length of the noise burst had a significant impact on the performance of the channel. Distributed single bit

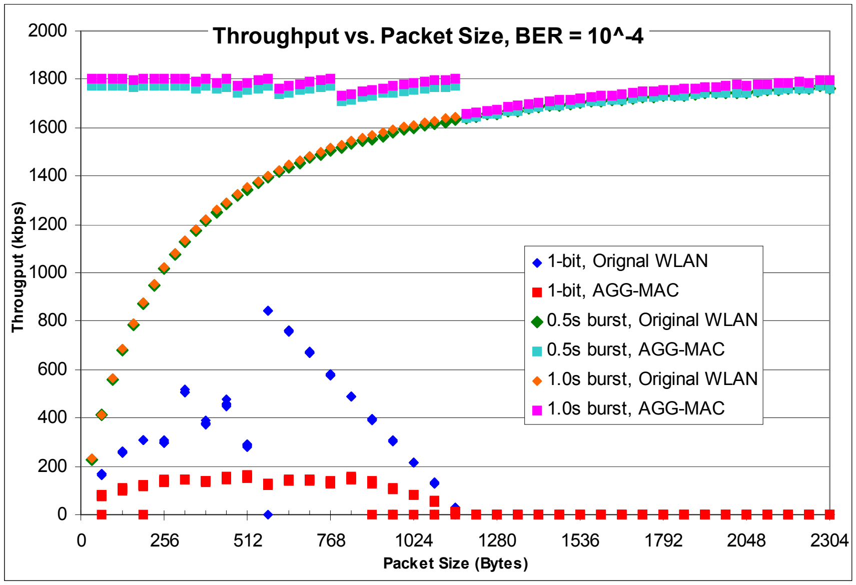

Figure 7.12: WLAN Throughput vs. Packet Size under varied length burst errors (BER = 10^{-4})

errors potentially generated errors in several frames and had a significantly greater negative effect on the potential throughput of the WLAN than a burst of bit errors that collided with a single packet or group of adjacent packets. This work did not consider the higher-order effects of transport level protocols, such as TCP, that might respond to differently to a single packet loss than to loss of a sequence of packets.

7.4 AGG-MAC Infrastructure Mode

Below we present some of the results from the OPNETWORK 2002 Tutorial 1332 titled "Maximizing Performance for Wireless LANs" [49]. The scenario of this tutorial was selected because it represented a very diverse traffic load with consideration for both DCF and PCF modes of operation. Refer to [49] for additional specifics on the simulation parameters.

Because the system operated well below capacity and was designed to provide successful communication in its original configuration, the throughput was not significantly

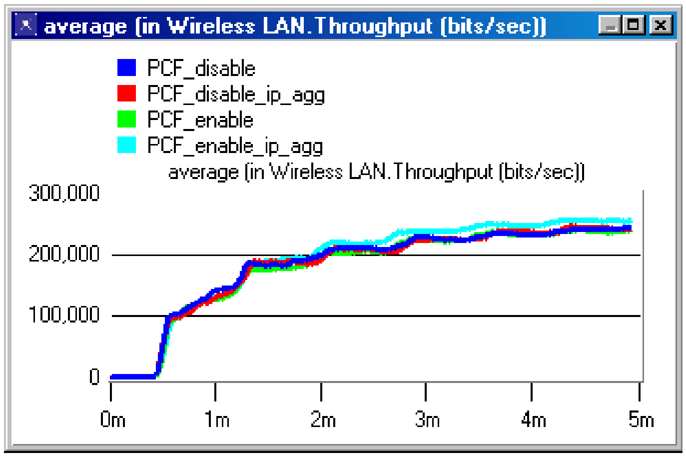

Figure 7.13: Average Throughput vs. Time

affected. Figure 7.13 highlights the average throughput for the four scenarios over time. You can observe a slight improvement in the PCF with aggregation but even this improvement was limited. This represented a system generally in steady-state in as much as the usable capacity was more than sufficient for the offered load.

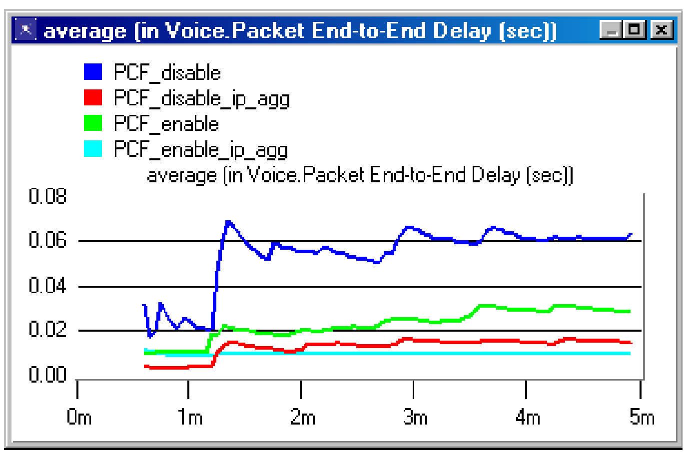

Figure 7.14: Voice End-to-End Delay vs. Time

The *end-to-end delay* and the *delay variation* permitted in the original DCF (*PCF_ disable*) scenario was the original problem addressed in this exercise. As seen in Figure 7.14, the *end-to-end delay* for all cases other than *PCF_ disable* was acceptable for voice traffic. It was notable that the end-to-end delay in both of the AGG-MAC traces of Figure 7.14 was better than either of the original traces. This indicated that even in the contention-based DCF mode of operation, AGG-MAC could support voice traffic in this scenario and while either MAC layer could support voice traffic in PCF mode, AGG-MAC performance was better under these conditions.

Figure 7.15 indicated that *delay variation* or jitter was significantly reduced when using AGG-MAC. Because AGG-MAC attempted to send as much data as was available under the constraints of the fragmentation threshold, the small packet sizes of the voice traffic clearly benefited from our approach. From these graphs, it appeared that aggregation in either mode of operation was more beneficial than simple PCF mode operation for voice communications under these conditions.

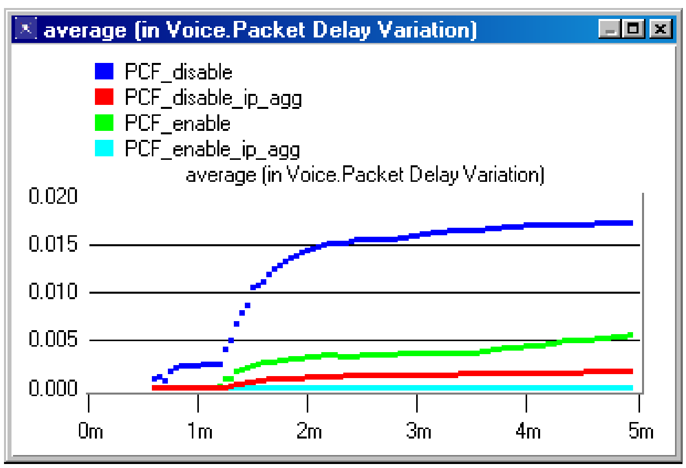

Figure 7.15: Voice Delay Variation vs. Time

We additionally observed from Figure 7.16 that HTTP traffic performance was comparable and acceptable in all cases with generally a slightly better performance under AGG-MAC than with the original protocols.

In Figure 7.17, we observed that FTP performance was also comparable with or without aggregation. FTP performance was slightly worse for the AGG-MAC protocol during peak traffic periods and for the QoS generally associated with FTP protocols. We considered this a desirable trait because it had the effect of rewarding smaller streams and penalizing

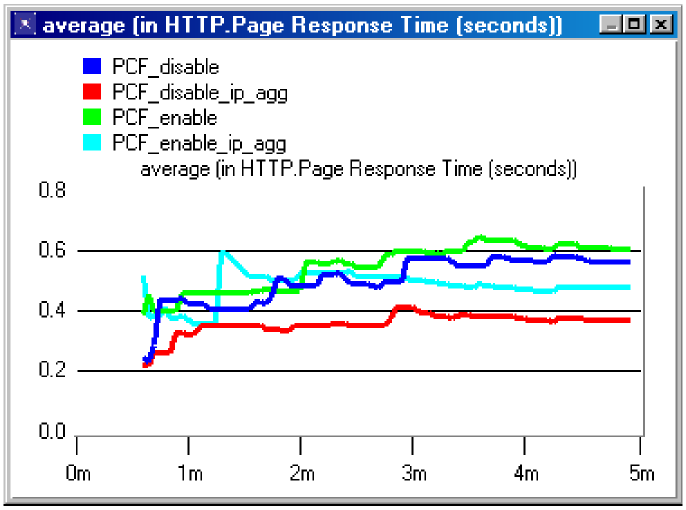

Figure 7.16: Avg. HTTP Page Response

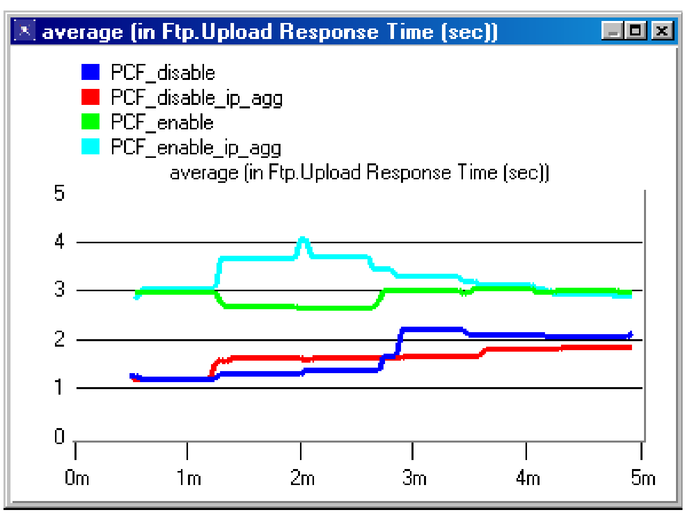

Figure 7.17: Avg. FTP Upload Reponse

those streams that consumed a greater share of the resources through improved response times. Figure 7.18 illustrated this penalty and reward effect by plotting the amount of aggregation achieved by traffic model type. This graph plotted the average number of higher layer packets contained within a frame. FTP traffic typically created larger packets, up to the MTU as set in the IP layer. Accordingly in this simulation, the MTU was justifiably set to 2,304 *Bytes* in the FTP client (WLAN based), and 1,500 *Bytes* in the FTP server (Ethernet based). This left little room for aggregation to benefit FTP. In our scenario, FTP performance was actually reduced because the other clients tended to use more network resources when they transmitted their larger frames, thereby reducing the amount of capacity available to FTP traffic.

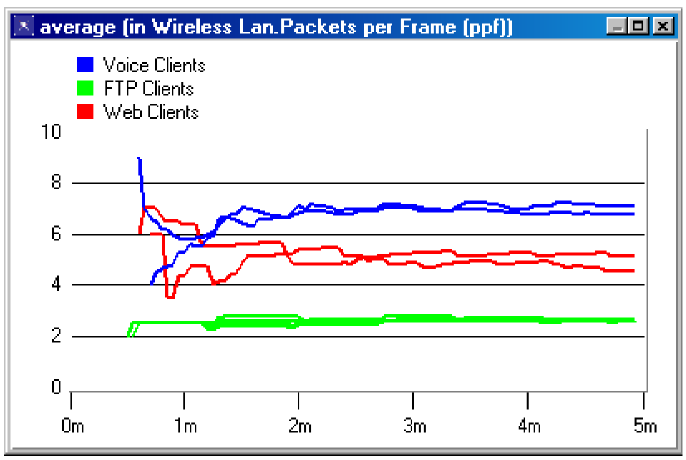

Figure 7.18: Avg. Aggregation (Packets per Frame)

7.5 Land Warrior Trace Results

In the consideration of our Land Warrior trace data, we manually processed the small data set and the results were promising for those nodes that generated appreciable amounts of the traffic. In our data set, five of the nodes generated 94.8% of the total offered traffic. Table 7.1 presents performance improvements for those five nodes. Despite the limited data set, the results did indicate the potential existed for packet aggregation to improve performance.

The six nodes with the fewest transmissions each had only one transmission. It was not clear whether these were frames received in error but not identified or whether six nodes transmitted only one frame during the entire exercise. Our inclination was that these frames were received in error.

Table 7.1: Top 5 Land Warrior Senders AGG-MAC Improvment

Originating Station	**Number Frames Transmitted**			**Percent Reduction**
	Original	**Enhanced**	**Reduction**	
Platoon Leader	2322	1511	811	34.93%
Unidentified (192.169.0.60)	563	384	179	31.79%
Unidentified (192.169.0.83)	70	50	20	28.57%
Unidentified (192.169.0.37)	158	116	42	26.58%
Unidentified (192.169.0.42)	126	97	29	23.02%

One additional enhancement that would be worth consideration for the Land Warrior system was to set a minimum and maximum timer for the transmission of Active Soldier data on the order of $30s$ - $55s$. If, upon receipt of a packet for transmission after the timer reached the minimum value but before reaching the maximum value, the desired GPS position location information could then be opportunistically generated and aggregated into the other data traffic. Then the system would only generate an additional frame to support position location in the event the maximum timer were allowed to expire without the occurrence of a packet for transmission. Similar approaches could be used for other periodic, non-real-time or near real-time data transmissions.

The platoon leader generated the most data during the exercise, and therefore presented the most potential for improved performance. Because the trace data was captured during actual transmission, we could not exactly determine when the packet was available to the MAC layer and whether or not internal queuing had occurred. For this reason, we assumed that packets transmitted within a $30msec$ window could potentially be aggregated.

In order to consider the effect of a noisy channel and to conform to stated design considerations [50], we constrained the maximum size of a data element to be 256 *Bytes*. Selection of this value corresponded to the most restrictive possible value for *fragmenta-*

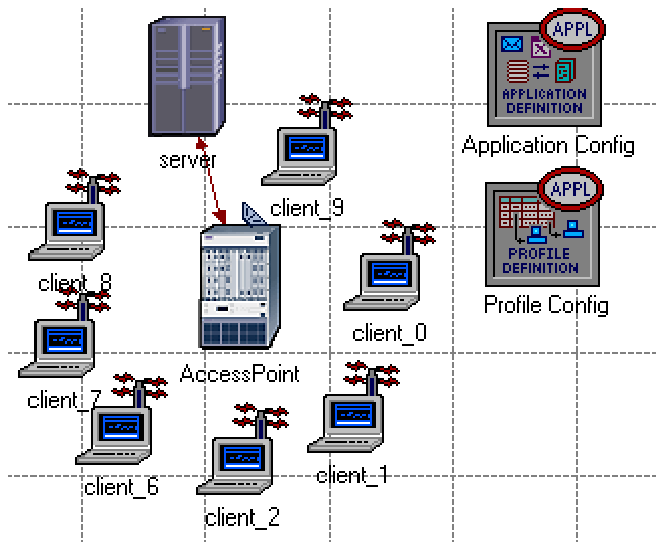

Figure 7.19: SIGCOMM'01 Network Simulation Layout

tion_threshold in 802.11 [28]. Even under these limited constraints, the platoon leader achieved a 34.9% reduction in the number of frames transmitted. This resulted in additional realizable capacity of 338 *kbits* of data traffic for a fully utilized channel. The total improvement achieved 7.5% increase in throughput overall, even with the inclusion of the six nodes that transmitted only a single frame. Based upon our operational experience, we believe that as the Land Warrior system develops, the traffic intensity will only increase and could derive even greater performance improvement from the AGG-MAC packet aggregation technique.

7.6 ACM SIGCOMM'01 Trace Results

Section 6.7.5 provided an overview of our methodology for using the rich trace data collected at the IEEE SIGCOMM 2001 conference. This real-world traffic was the foundation for the simulation scenario depicted in Figure 7.19. In this section, we present the results of our simulation using trace data collected from the IEEE SIGCOMM'01 conference.

As indicated in Section 6.7.5 , our simulation was constrained to only one, AP_NW, of the four access points. In order to limit the volume of data considered, we further

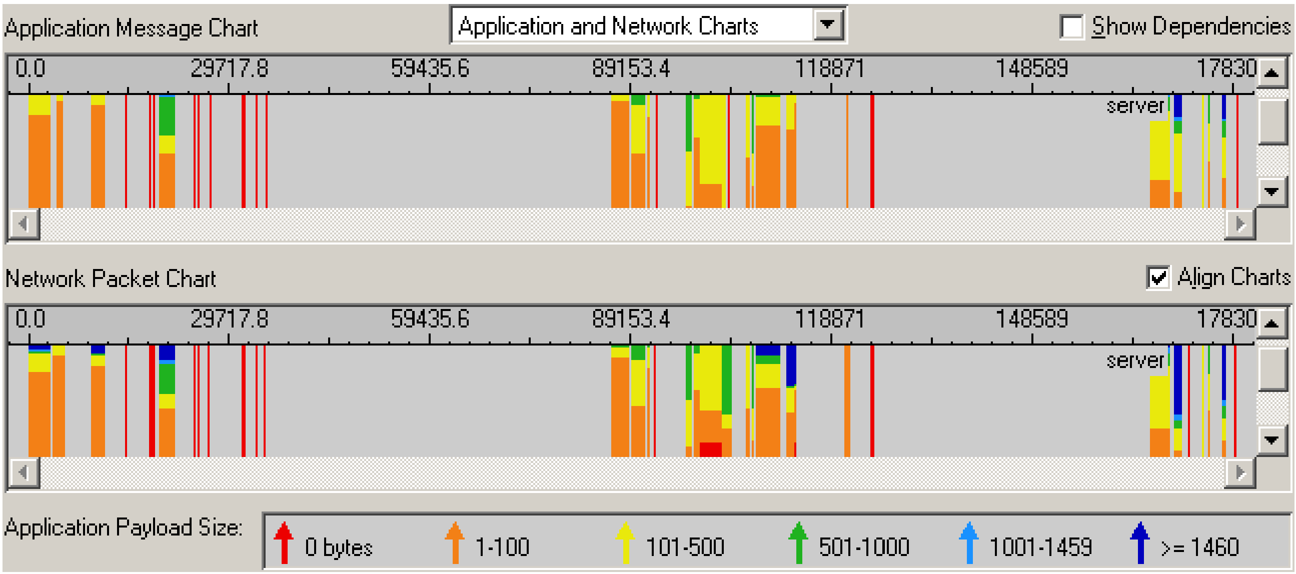

Figure 7.20: SIGCOMM'01 Client Data Exchange (52 hour Trace)

constrained our focus to the busiest hour of the conference. While users were free to move among the different access points throughout the conference, [5] noted that mobility was not a factor during scheduled sessions. It was noted that clients only moved to a different AP following breaks in the program. We filtered the original trace file using a C++ program we created using the WinPCAP libraries, and created trace files for each of the 25 clients that were associated with the AP-NW as identified in the MIB files at the start of the busiest hour. Because our focus was on the performance of the WLAN portion of the network, we abstracted the 473 remote machines contacted by the conference attendees into a single conceptual destination.

An example filtered trace file for the client with MAC address = 00:e0:63:50:6c:d4 is provided in Figure 7.20. This trace represented the entire 52 hours of the conference. This client was identified as one of the heavy traffic clients throughout the conduct of conference. From this trace, we can clearly identify the times the conference was in session and when it was not. It can also be observed that the preponderance of traffic was less than 500 *Bytes* in size. These two observations were consistent for each of the 25 clients we analyzed.

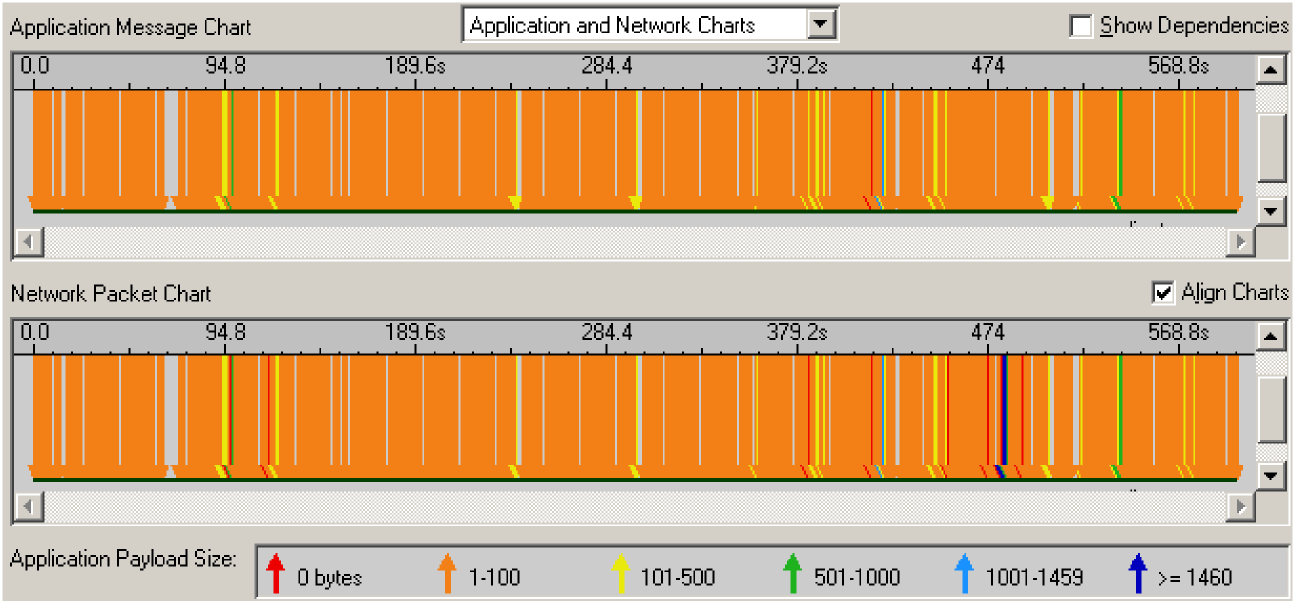

Figure 7.21: SIGCOMM'01 Client Data Exchange (11:00 - 11:10 Trace)

Figure 7.21 presents a view of the first ten minutes of communications during the busiest hour for this heavy traffic client. This was a ten minute subset of the 52 hours depicted in Figure 7.20. This ten minute period represented the time simulated in our OPNET^{TM} model. Analysis of the communications traces for each of the 25 clients associated considered in this simulation identified that although 25 clients were associated, only seven of those communicated any traffic during the period of observation. Four of those clients were characterized as medium traffic clients and three were identified as heavy traffic clients. We simulated this network with nodes using the original WLAN MAC and then repeated it using our AGG-MAC protocol enhanced nodes. The results of those simulations are presented in Table 7.2.

The throughput of our simulated system was consistent with the expected peak throughput for the original system from which the trace data originated. Average throughput during the simulation was 3.04 $Mbps$ with an instantaneous peak throughput of 4.1$Mbps$. As indicated in Figure 6.5, [5] observed a peak data rate of 3.2 $Mbps$ and our system was designed to represent the busy hour of the conference. When AGG-MAC was active, the enhancement achieved an aggregation level of 3.23 packets per frame. Because the system capacity was 11 $Mbps$, the achieved throughput was actually more representative of the offered load or even the actual load since no data was lost. Using AGG-MAC with the representative traffic loads, we were able to reduce the throughput on the system to an average of 2.27 $Mbps$, a 25.3% reduction in the number of bytes on the channel which yielded improvements in the performance of the application traffic.

The overall average download response time for Email was improved by 32.1% as indicated in Table 7.2. HTTP Page download response time was improved by 27.4%, and FTP download response time was improved by 21.7%.

Table 7.2: SIGCOMM'01 AGG-MAC Performance Improvement Results)

	WLAN Load	Email Download Response	HTTP Page Response	FTP Download Response
Original MAC	$3.04 Mbps$	$0.1068s$	$0.1607s$	$0.3407s$
AGG-MAC	$2.27 Mbps$	$0.0725s$	$0.1166s$	$0.2667s$

This simulation indicated a measurable potential performance improvement using AGG-MAC techniques to mitigate the fixed overhead associated with interframe spacing in a wireless LAN network. Coupled with the results from the Section 7.5, AGG-MAC appeared to be beneficial for a wide range of network topologies and traffic situations. While further study is warranted to fully understand all the issues associated with this approach, the potential is clear.

7.7 UMTS-WLAN with AGG-MAC Results

This section presents the results of our simulation of a UMTS-WLAN interworked system which was enhanced with the AGG-MAC, packet aggregation, protocol compared with an unenhanced, interworked UMTS-WLAN system using only the original WLAN MAC

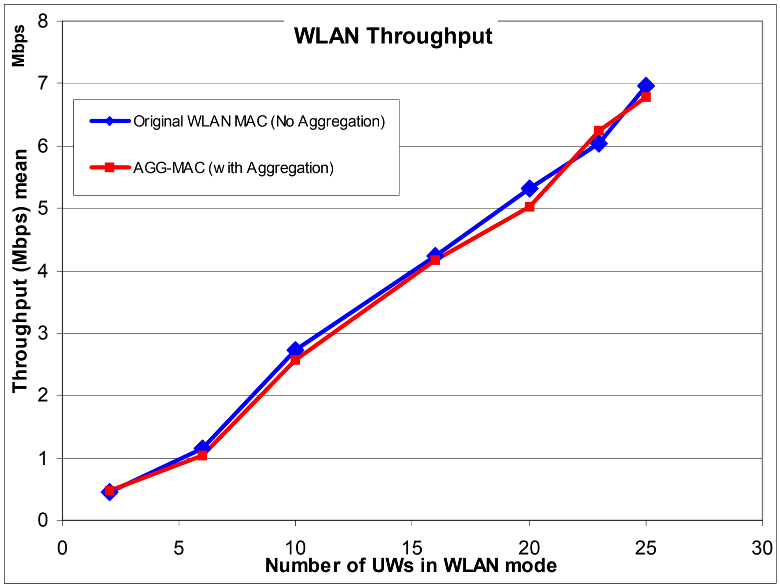

Figure 7.22: UMTS-WLAN with AGG-MAC: WLAN Average Total Throughput (Mbps)

protocol. The original purpose of Tracy Mann's thesis experiments [8] was to compare the performance of the UMTS-WLAN interworked system with an original baseline WLAN system. We compared his results with those produced by a UMTS-WLAN interworked system enhanced with the AGG-MAC packet aggregation protocol. While the simulations in [8] had three portions, our focus is currently on the performance of the WLAN component. In order to evaluate the performance of an AGG-MAC enhanced UMTS-WLAN system, our focus was strictly on UWs operating in WLAN mode.

In order to assess the overall performance of our two systems, we examined total system throughput by varying the number of UWs that were attempting to communicate on the channel. The results of these simulations are presented in Figure 7.22. At first glance, it appeared that AGG-MAC performed worse than the original WLAN based on total system throughput. However this was not the case, due to the aggregation of packets, fewer bits were transmitted over the channel. In this case, the channel supported all of the offered communications requirements while transmitting fewer bits. In the cases consisting of a small number of clients, fewer than ten, the network was idle for measurable periods of time while waiting for a client to offer data for transmission. For those situations where ten or more clients were sharing the network, the utilization of the network was higher which caused some level of queuing in the clients. Due to this queuing, clients were able to aggregate packets together and send larger frames over the network when AGG-MAC was enabled, therefore the number of bits transferred per frame was higher. The plot in Figure 7.22 illustrates the benefits of AGG-MAC from two different perspectives: fewer bits transmitted to accomplish same amount transfer of information and an increased achievable capacity due to more bits transmitted per unit of overhead.

Performance of email download with and without AGG-MAC enabled are presented in Figure 7.23. We expected these results to be more significant for an email download because we anticipated the email traffic characterization to represent a number of small data transfers that could be conducted concurrently. Upon investigation, we discovered that the UW-Heavy traffic load employed in [8] was based upon the original UMTS traffic load which had been optimized for 32kbps channels. The email traffic patterns are an equal mix of upload and download traffic, further reducing the amount of traffic available to aggregate in any given direction. In order to provide a sufficient load on the WLAN channel, [8] increased the size of each data object in order to increase traffic on the channel. These larger transfers generated larger packets and therefore provided fewer opportunities for packet aggregation to provide benefit. Increasing the frequency of occurrence of smaller email data transactions or providing a greater variance in the size of the objects would have been a reasonable alternative approach. Had we made these modifications to the traffic stream, the resultant benefits from AGG-MAC would have been considerably more striking. We did not modify the traffic characteristics from the original set of assumptions in order to avoid an potential to skew the results in favor of AGG-MAC. Even under the stated conditions, AGG-MAC still improved performance as indicated in Figure 7.23. We observe a 14% or better reduction in average Response Time for email transactions with 16 or more clients.

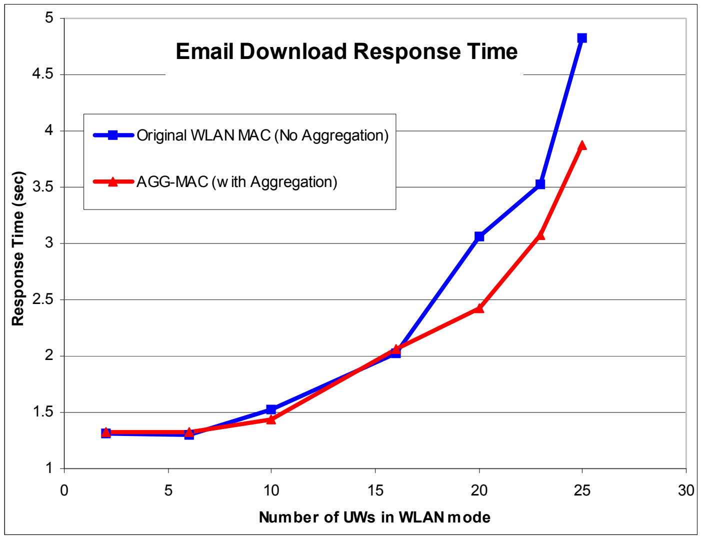

Figure 7.23: UMTS-WLAN with AGG-MAC: Email Download Response Time

HTTP traffic in characterized as an interactive data stream; smaller delays are preferred, therefore the response time should be as low as possible. Figure 7.24 depicts the average response time for a client to download all of the objects on a web page. The response time for HTTP traffic was reduced by as much as 22% with packet aggregation. Consideration of these results from the perspective of a service provider, it could be observed that the incorporation of AGG-MAC could support between four to six additional users with the expectation of receiving similar responsiveness from the system as the original MAC protocol. This additional provisioning could yield increased revenue or could be used to create differentiation among the service provider's competition.

Figure 7.25 presents the results of the average download response time for FTP file downloads across a differing number of clients. The FTP Download using AGG-MAC in our test cases requires an average of 10.9% less time overall to download across the range of cases considered in these simulations. One reason the FTP downloads benefitted from packet aggregation was that the server was located on an ethernet segment, therefore the *fragmentation_threshold* from that station was set to the appropriate value for ethernet which is approximately $1,500$ *Bytes*. These packets could be aggregated at the AP with

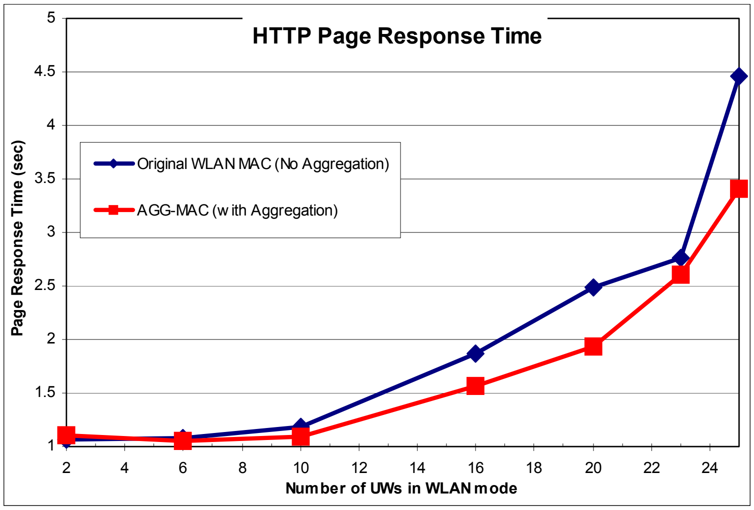

Figure 7.24: UMTS-WLAN with AGG-MAC: HTTP Page Response Time

other smaller packets.

The situation where performance was not generally improved was in the FTP files upload. In this case, the frames were often maximal in size and therefore were unable to benefit from aggregation. Because competing stations were able to aggregate, those other stations generally sent larger quantities of data when it was their turn to transmit which increased the station's waiting time. For this reason, Figure 7.26 illustrates an example of a traffic stream that did not benefit from packet aggregation. These results were consistent with those observed in Section 7.4 which considered a PCF-based WLAN office network with multiple types of competing data streams.

7.8 Summary

AGG-MAC provided a more efficient use of WLAN capacity. Section 7.1 verified the correct operation of the AGG-MAC protocol and its associated simulation model while Section 7.2 validated the model's operation within the context of both the network and the protocol stack. We demonstrated that when sufficient traffic was offered, AGG-MAC

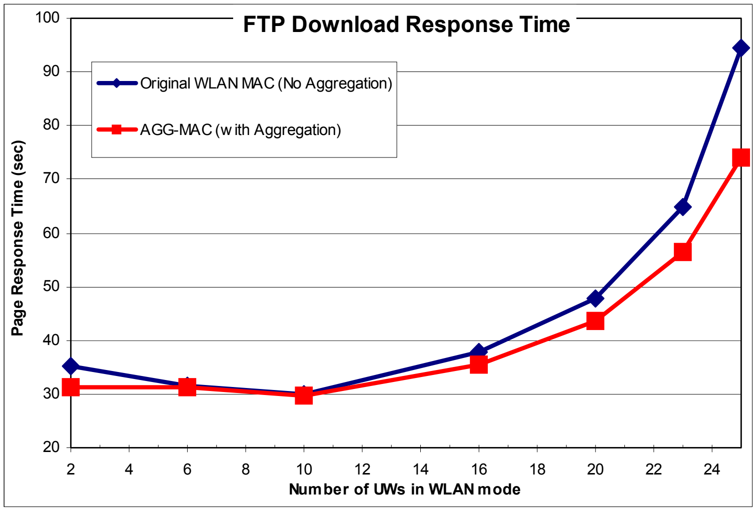

Figure 7.25: UMTS-WLAN with AGG-MAC: FTP Download Response Time

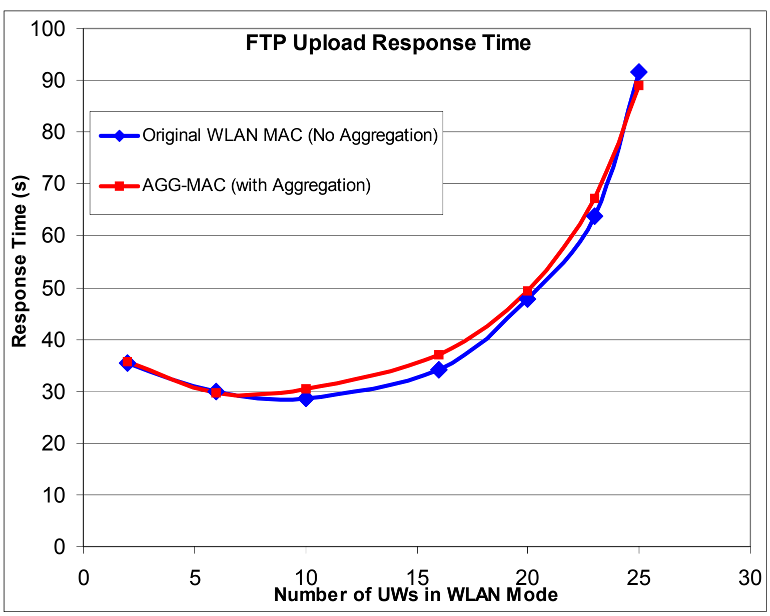

Figure 7.26: UMTS-WLAN with AGG-MAC: FTP Download Response Time

systems could achieve 90% of the advertised capacity throughout the range of frame sizes compared to only 11% achievable for small frame sizes, $32B$, for the original WLAN MAC. Furthermore, *end-to-end delay* was as much as $0.777s$ on the original system and varied significantly for different frame sizes; but on the AGG-MAC system, for a range of frame sizes from $32B$ to $2,304B$, the *end-to-end delay* remained within the range of $0.101s$ - $0.144s$. This result indicated not only lower delays, but also limited delay variation or jitter which was particularly important to interactive or real-time systems.

The effects of noise on the system was examined in Section 7.3. On the original WLAN MAC, our simulation results were consistent with those published in [33]. We observed that for both the original and the AGG-MAC enhanced systems, BERs of 10^{-5} or better benefitted from maximal frame sizes. For BERs of 10^{-4} or lower, frames larger than the optimal size reduced achievable performance as did frames smaller than optimal size, but the performance of the system improved using AGG-MAC with an appropriately chosen fragmentation threshold The shape of the curve seemed to be influenced more by the amount of errors and by the burstiness of those errors than by contention from competing stations.

We additionally demonstrated the benefits of AGG-MAC in both infrastructure and ad hoc networks. Our infrastructure mode simulations illustrated a situation where AGG-MAC improved performance more dramatically than a proposed prioritization scheme providing dedicated PCF communications opportunities for voice traffic. AGG-MAC improved the original, problematic, pure-DCF system by 60% for *end-to-end delay* and by 80% for *delay variation* or jitter. Furthermore, AGG-MAC even improved on the original proposed solution of using dedicated PCF opportunities by 30% and 27% respectively. We confirmed similar results for a number of DCF implementations, to include multi-cast based systems in the US Army's Land Warrior WLAN system. The Land Warrior evaluations demonstrated the utility of AGG-MAC for ad hoc network systems and for a very lightly loaded system, provided a potential improvement in achievable capacity of 7.9%.

The SIGCOMM'01 simulations were based upon live collected data and yielded a 32.1% improvement for *email download response time*, a 27.4% reduction for *HTTP Page response time* and a 21.7% reduction for *FTP download response time*. From this trace data, we observed that the size of almost all of the frames on this network were less than $576B$. This provided significant support to the potential for derived benefit from the AGG-MAC approach.

Our final results in this section sought to assess the performance of AGG-MAC when employed in an interworked UMTS-WLAN network. These results produced as much as a 14% reduction in the *Email response time*, up to a 22% reduction in the *HTTP Page response time*, and a 10.9% improvement in the *FTP download response time*.

Because WLAN was a shared channel, the impact of such improvements generally provided better performance not only to the user who implemented the change, but to all users on the system. We observed that AGG-MAC protocol generally rewarded data streams that did not overuse the shared channel and yielded slightly lower performance for those data streams that consumed significant resources on a high demand channel. While additional developmental testing to include prototype evaluation systems should be considered, our results were quite favorable.

Packet level aggregation, especially in the context of PCF-based WLAN, yielded considerable performance improvement provided appropriate value for *fragmentation_threshold* was set for the existing channel conditions. If the *fragmentation_threshold* was too high, then AGG-MAC may increase the probability of frame error be generating larger frames, and actually degrade system performance. While it was possible to create a monitor capable of dynamically controlling the *fragmentation_threshold*, it was beyond the scope of our current research effort.

Chapter 8

Conclusions

The wireless communications industries experienced significant growth over the past decade. Cellular communication devices and wireless LAN systems have become so pervasive that they are considered commonplace in most households in North America. While a number of scheduled UMTS system fieldings had been delayed due to economic factors, systems were available at the time of this publication and more were programmed for implementation. Despite the economic factors, the proliferation of WLAN systems experienced near phenomenal growth in both the US and in Europe. WLAN "hotspots" were widely available. While WLAN capacities are marketed as 11Mbps and 54Mbps in current systems, this level of performance is generally not reached in practice. Means to improve the performance of WLAN systems under various conditions and constraints was an ongoing research issue. At the time of this publication, systems supporting 3G cellular and WLAN access mechanisms were based upon Mobile IP. Mobile IP-based systems would be considered loosely-coupled systems and are not capable of supporting all of the levels of interoperability defined for an interworked UMTS-WLAN system as defined by the 3GPP, see Section 3.1.1. Research is ongoing to fully define the standard implementation of a tightly-coupled UMTS-WLAN interworked system. Our research effort in this integration sought to produce a viable UMTS-WLAN interworking model and simulation framework for evaluating future systems. We additionally sought to improve the performance of the WLAN portion of this interworked system.

8.1 Significant Results

The primary contributions of this research were two-fold. The first contribution was the design and implementation in OPNET^{TM} of a network level simulation tool to support future research in tightly-coupled interworked system consisting of WLAN "hot spots" and UMTS cellular systems. The second significant contribution was the design, implementation and testing of AGG-MAC, a WLAN MAC layer protocol enhancement designed to improve performance for a broad range of wireless LAN configurations and traffic patterns. The UMTS-WLAN interworked system was described in Section 4 and the design of the WLAN enhancement, AGG-MAC, was presented in Section 6.

Simulation was the primary tool used to implement and test each of these mechanisms. The simulation was validated using analytic mechanisms for a baseline implementation using deterministic and exponential traffic sources, as described for AGG-MAC in Section 7.2. For each of the two primary portions of this research, simulations were conducted to assess the performance and evaluate the suitability of the proposed system under consideration in order to address the research questions posed in Section 1.4. A summary of significant results for each of those questions is presented below.

8.2 UMTS-WLAN System Significant Results

The initial phase of work focused on the design, implementation and testing of a tightly-coupled network-level system designed to provide WLAN "hot spots" interworked with 3G cellular mobile communications system, specifically UMTS networks. The resultant network level simulation tools provided a foundation for future research into the issues and trade-offs associated with protocol design and parameterization in this type of system.

The 3rd Generation Partnership Project (3GPP) was one of the primary sources of standardization efforts for 3G cellular systems. Much of their work focused on the convergence of voice and data communications. WLAN is capable of providing significantly higher capacity for indoor environments albeit is of limited range. For this reason, the 3GPP Services and Systems Aspects (SA1) published a feasibility study and further sought to develop standards for the interworking of 3GPP systems with WLAN systems [7]. Their study identified six scenarios ranging in amount of interoperability between the two systems. These scenarios are presented in Section 3.1.1. We sought to provide a robust network level simulation environment capable of supporting future research on UMTS-WLAN interworked systems.

8.2.1 Benefits to a tightly-coupled UMTS-WLAN interworked system

The benefits associated with a UMTS-WLAN interworked system were demonstrated in Section 5.3. The intent of Section 5.3 was to demonstrate the benefits of the interworked UMTS-WLAN system as compared to the existing UMTS system. Specifically, this scenario demonstrated the conservation of limited dedicated cellular channels by shifting data users to the WLAN access network. It also demonstrated the significant improvement in application response time of the interworked UMTS-WLAN system as a result of the larger WLAN channel.

Since the primary motivation for integrating WLAN into the UMTS system was to free limited resources while continuing to provide service, the number of reserved dedicated channels (DCH) in the Node-B was examined. Because of the importance of voice channels, dedicated channels were considered to be the finite, limiting resource in the UMTS network.

The interworking UMTS-WLAN experiment was conducted using twenty wireless client data users running a mix of client applications; specifically the mix of Web, E-mail and FTP applications (*Wireless_ Client* application profile) discussed in Section 4.7. The baseline system consisted of twenty UW nodes operating in a UMTS cell. Sixteen of these clients were operating within 300m of the UWLAN_AP, and the other five were not within the range of the WLAN access network. The simulation was first run with the Client Access Mode set to UMTS for all 20 clients. The Client Access Mode was then toggled to WLAN for the sixteen clients that were in range of the WLAN access network and the results were compared.

Based on the quality of service requirements for the data session, the UMTS allocates one or more DCH channels to support a user session. It can be observed from Figure 5.4 that reducing the number of contenders for service from twenty to four reduced the average number of DCH channels in use by approximately 65%. It is possible to reallocate these unused DCH channels to voice services or to additional packet data services. Thus, it can be asserted that a service provider could use these unused resources to generate additional revenue.

Beyond these basic performance metrics, the simulation platform provides a foundation for development of protocols to leverage a shared packet network, an alternative path to migrate to an all IP network, and the potential for service differentiation throughout the core network.

8.2.2 UMTS-WLAN System Performance

The first issue to consider in the performance of our proposed system was its ability to perform as intended. Because we proposed a new and novel system approach based upon the integration of a WLAN AP and a 3G RNC as the coupling point, we first had to ensure proper operation. The design considerations, rationale and decisions were presented in Section 4.

Verification and validation of the protocols and models was discussed in Section 5.2. The performance of the UMTS-WLAN interworked system was more fully covered in [8] which was derived from this research effort. The jointly conducted significant results were also presented in Sections 5.2 and 5.3 herein. The design of the performance testing was conducted jointly and involves comparison of the interworked UMTS-WLAN system and both baseline UMTS and baseline WLAN networks. The potential performance improvements over pure UMTS continued to fuel efforts to identify best practices and appropriate configurations to enhance consumer interest in such systems. While systems were being commercially fielded, the absence of a key application for which this feature becomes critical was limiting the adoption and growth of integrated UMTS-WLAN systems.

The UMTS-WLAN interworked simulation environment provided a rich framework in which to conduct future research. The two systems to be interworked each provided mutually supportive benefits and should be combined to improve performance for a wide range of situations. The information gained from understanding the issues with a tightly-coupled system are beneficial in any interworked system to include loosely-coupled using technologies such as mobile IP. A service provider could gain competitive advantage through the additional capabilities offered in a tightly-coupled environment. The UMTS-WLAN framework provides a foundation for conducting additional research in order to better understand the benefits and trade-offs associated with a range of interworked system configurations.

8.3 AGG-MAC Protocol Enhancement Significant Results

The second phase of this work was the design of AGG-MAC, a WLAN MAC layer protocol designed to significantly improve throughput and reduce latency for data traffic consisting of packets of less than maximal size. The identification of the bottleneck associated with the transmission of small frame sizes and its severity was identified as the result of UMTS-WLAN interworked system simulations. The utility of AGG-MAC is not restricted to interworked UMTS-WLAN systems or even to infrastructure based systems. We have demonstrated that AGG-MAC can benefit quite a wide range of WLAN configurations.

8.3.1 Design of the WLAN Packet Aggregation MAC protocol (AGG-MAC)

The design of the protocol was discussed in Section 6.1. AGG-MAC does not adhere to the traditional MAC-layer protocol paradigm of handling exactly one packet per frame. We made a conscious design decision to relax this requirement while ensuring transparency and compatibility with the layers above and below the MAC in the protocol stack. We also wanted to ensure that the transmitted frame and transmission parameters did not interfere with and could communicate with the existing, legacy WLAN systems. The basic design choices for the implementation consist of the three questions below. Refer to Section 6.1 for discussion of the options, rationale, and design choices made in our implementation. Any or all of these three design options could be pursued for further research.

- When to perform aggregation operation
- What is the upper limit for aggregate frame size
- Which packets selected as candidates for aggregation

8.3.2 Baseline Performance

In order to evaluate baseline performance for our AGG-MAC enhanced system, we considered the most basic possible WLAN network. We employed a single transmitter and a single receiver operating on an error-free channel. In the baseline system, we evaluated the performance using a deterministic and an exponential source. In each of these cases, the results were compared to analytically derived results, see Section 7.1. Validation of the model included demonstration that the AGG-MAC protocol could potentially achieve 90% of the stated network capacity whereas for small frames, 32B, the original system would be limited to not more than 11% of the system capacity. We further illustrated that the *end-to-end delay* and the *delay variation* or *jitter* would both be significantly improved with the AGG-MAC enhancement.

Because noise on the channel is a significant difference between wired and wireless network systems, we developed simulations to better understand the effects of noise on the WLAN system. We considered the original WLAN MAC protocol and our AGG-MAC protocol enhancement. Our simulations for the original system served to further validate the simulation implementation. Our results were compared with analytical and laboratory measured results as published in [33]. Our results were consistent. From these simulations, we observed that low BER conditions strongly favored larger frame sizes. While higher BERs, 10^{-4} or worse, favored limiting the frame size. Our results indicated that the characteristics of the noise on the channel was the determining factor in determining the appropriate value for the maximum frame size, *fragmentation_threshold*, and a poorly chosen frame size can severely reduce, potentially to nothing, the achievable throughput on the channel. Multiple competing stations reduced the achievable throughput on a channel, but their effect was one of scaling the results rather than changing the characteristics of the performance curve, Section 7.3.2.

8.3.3 AGG-MAC Performance Over a Range of Scenarios

We simulated the AGG-MAC protocol over a range of diverse network configurations. The results are presented throughout Chapter 7. AGG-MAC demonstrated the potential to improve overall system performance as well as individual client performance. Performance metrics considered, described in Section 6.3, were *throughput*, *end-to-end delay*, *data dropped*, *voice packet delay variation* (*jitter*), *application response time*, and the *amount of aggregation*.

Our simulations demonstrated example configurations in which the AGG-MAC protocol enhancement can benefit real-time, interactive, voice data. The results of our simulation reduced the *end-to-end delay* from $0.06s$ to less than $0.02s$, a 60% improvement. *Delay variation* or *jitter* was reduced from $0.017s$ to $0.002s$, nearly an order of magnitude improvement. These simulations also identified a property of this approach to provide greater improvement to smaller packets while providing less benefit to larger packet sizes; a desirable trait in many QoS systems. In our SIGCOMM'01 trace data, we observed that over 97% of the packets transmitted on the network were smaller than $576B$ in size and therefore could benefit significantly from the AGG-MAC approach. This observation was the result of over $1GB$ of data collected over three days from 195 distinct users employing their own equipment and communicating with several thousand distinct distant host machines. The SIGCOMM'01 and the Land Warrior results were derived from live trace data rather than purely statistically generated traffic. The Land Warrior system, while of limited statistical significance demonstrated the utility of AGG-MAC in a distributed, ad hoc environment where the traffic load was severely limited. The SIGCOMM'01 data was statistically much more robust and it provided a real example of an infrastructure network with a wide range of clients, similar to a public "hotspot." The SIGCOMM'01 data rates were higher, generally loading the AP at approximately 20% - 25%. Due to the reduction in the number of bytes transmitted because of both protocol retransmissions and from reduction in the total number of bytes of frame overhead, the load on the SIGCOMM'01 AP was reduced from 3.2 $Mbps$ to 2.27 $Mbps$ while application response times for the range of applications was improved. This is an example of accomplishing more while consuming less resources, specifically 25.3% less resources. The bandwidth saved could be used to support additional users.

8.3.4 AGG-MAC Performance Incorporated in a UMTS-WLAN System

We evaluated AGG-MAC within the context of a UMTS-WLAN interworked system and the results were promising. This research effort provided additional information available for consideration of future 3G and WLAN systems. At the time of this publication, the biggest drawback in the employment of AGG-MAC seemed to be a reluctance to permit the MAC layer protocol to operate on more than one packet at a time. The general rationale seemed to be reluctance to change commonly accepted paradigms rather than technological issues.

Our evaluation of the performance of AGG-MAC within the context of a tightly coupled, heterogeneous, combined 3G cellular (UMTS) and WLAN system [7][36] were presented in Section 7.7. These simulations yielded improvements to the applications with as much as a 14% reduction in the *Email response time*, up to a 22% reduction in the *HTTP Page response time*, and a 10.9% improvement in the *FTP download response time*. Although these traffic patterns were designed for and optimized for a UMTS client using a dedicated 32*kbps* circuit, we illustrated the potential for AGG-MAC to provide increased performance on such a client communicating with a WLAN "hotspot". These improvements could allow the "hotspot" to support a greater number of users and therefore increase the earning potential of the service provider while additionally providing better service to the user.

8.4 Recommendations for Future Research

This research effort has successfully developed a feasible initial architecture and resultant simulation framework to further evaluate a tightly-coupled integrated 3G cellular and WLAN networking system. We have also developed an enhancement to the WLAN MAC layer protocol, AGG-MAC, capable of providing measurable performance enhancements for a wide range of situations. In each case, there are several aspects of this work which offer the potential for meaningful future research.

8.4.1 Loose Coupled Design

The simulation environment could be expanded by creating UW and UWLAN_AP node models to implement a loosely-coupled design. The future trend for 3G and 4G wireless systems is for the network between the Node-B and CN to evolve into an all-IP based architecture. This would allow service providers to provide data services at various locations in the network without all of the data traffic flowing through the GGSN. A loose-coupled implementation would provide the necessary simulation environment to study the signaling protocol effects of an all-IP network. The loosely coupled design would integrate the two systems at the IP layer. Both the control and data traffic would be routed over the IP network. The ability of IP protocols, such as Mobile IP, to handle mobility and roaming between the access networks would also need to be studied. This would allow more direct and meaningful comparison between the two approaches to further define strengths and weaknesses.

8.4.2 QoS or Alternative Queuing over the WLAN Access Network

The simulation environment could be expanded to study QoS mechanisms over the WLAN access network. While the current simulation environment and 3GPP standards provide inherent support for quality of service over the UMTS access network, the simulated WLAN access network and 802.X standards do not provide these QoS mechanisms. WLAN does not distinguish between data streams of differing QoS. The current Internet service model supports only best-effort traffic. Therefore, as UMTS networks are deployed there may initially be little incentive to support different QoS levels for data traffic. However, as both IP QoS and the 3G wireless networks develop there will be increasing requirements to support data traffic with different QoS requirements. For WLAN to be a viable alternate access network it will also have to provide inherent support for QoS. The OPNET^{TM} application model set provides the necessary hooks to support QoS. Different QoS scheduling mechanisms could be studied within the framework of the UMTS-WLAN simulation environment. The model could be used to explore management, overhead, and scheduling and control issues to support QoS. It could also be used to study how the PCF

As identified in Section 6.1.3, our research effort implemented a single FIFO queue for packets received from the higher level for processing. Multiple queues could easily be implemented to facilitate either a QoS approach, or many other alternatives. The control traffic could be prioritized, as could potentially routing or security protocol packets. In addition to considering strictly prioritization as the reason to use a more advanced queuing mechanism, one could employ either logically or physically multiple queues to each address, or could sort queues based on packet size in order to increase the probability of finding a packet suitable to aggregrate. Each of these approaches would need to be considered in the context of their interaction with higher level protocols and with their impact on both the local and other nodes in the network. (point coordination function) in the WLAN access point can be leveraged to support multiple clients each with different data streams and mixes of QoS requirements.

8.4.3 Adaptive Fragmentation Threshold

The selection of appropriate value for the fragmentation threshold has been identified throughout this work. It was originally discussed here in Section 3.2.2 from the original document [33]. Although we have demonstrated the utility of finding the right fragmentation threshold, the value is still defined as a static value in the IEEE 802.11 series of standards and proposed standards. A fragmentation threshold that dynamically adjusts to changing channel conditions would have significant impact on noisy channels, see Section 7.3. AGG-MAC was designed to support a dynamic fragmentation threshold and would benefit from such an advancement.

8.5 Summary

This chapter presented conclusions based upon research results and recommended future areas of research. The goal of this research was to, first, design and implement a framework suitable for evaluation of mechanisms necessary to realize a tightly-coupled, UMTS-WLAN interworked network system; second, to implement a WLAN MAC-layer protocol enhancement capable of mitigating the costly per frame overhead per unit of user data associated with sequences of small packets. The results demonstrated that the research goals were achieved. The UMTS-WLAN system configuration is viable and the existing simulation framework provides a platform for future research issues. The AGG-MAC WLAN protocol enhancement provides a mechanism to aggregate suboptimal size packets into a larger frame in order to amortize the per frame overheads associated with a WLAN channel. We demonstrated the suitability of such a system for a variety of both theoretical and actual situations. The results of this research effort could be expanded by further protocol development in UMTS-WLAN interworked systems.

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Appendix A

Simulation Data Tables

Table A.1: FTP Download Response Time vs. FTP Filesize (1UE vs. 1UW)

1 UMTS UE — FTP Download Response Time (sec) vs FTP Filesize

FTP Filesize	1B	50KB	100KB	200KB	500KB	700KB	900KB	1MB	1.5MB	1.8MB	2MB	3MB
Seed 1	1.252	21.943	37.783	66.343	182.963	238.849	390.563	370.063	533.349	653.749	724.449	1053.533
Seed 2	1.252	37.369	38.849	61.249	179.069	250.249	344.869	360.849	538.449	646.435	731.235	1071.933
Seed 3	1.253	47.949	38.649	73.249	179.849	252.533	323.049	364.249	561.335	630.633	716.333	1063.932
Seed 4	1.253	27.249	37.749	70.855	176.855	259.273	326.749	363.055	556.533	665.433	711.533	1082.632
Seed 5	1.253	27.835	39.735	77.273	178.653	249.252	324.735	368.733	554.533	651.749	743.053	1061.932
Mean:	1.253	32.469	38.553	69.794	179.478	250.031	341.993	365.390	548.840	649.600	725.321	1066.792
95% CI:	1.252	19.700	37.527	62.093	176.701	240.880	306.571	360.566	533.690	633.868	709.851	1053.123
	1.253	45.238	39.579	77.495	182.255	259.183	377.415	370.214	563.989	665.331	740.790	1080.462

1 UW (WLAN Mode)

FTP Filesize	1B	100KB	250KB	800KB	1.5MB	4MB	7MB	12MB	15MB	20MB	30MB	40MB
Seed 1	0.195	1.178	2.358	6.338	11.718	30.278	52.458	89.958	112.518	149.798	223.938	298.058
Seed 2	0.189	1.191	2.171	6.351	11.731	30.291	52.471	89.971	112.531	149.831	223.931	298.071
Seed 3	0.184	1.204	2.184	6.364	11.744	30.369	52.484	89.784	112.409	149.624	223.744	297.884
Seed 4	0.191	1.069	2.249	6.429	11.609	30.304	52.549	89.849	112.344	149.715	223.855	297.955
Seed 5	0.198	1.095	2.275	6.455	11.635	30.395	52.575	89.875	112.435	149.689	223.809	297.949
Mean:	0.191	1.147	2.247	6.387	11.687	30.327	52.507	89.887	112.447	149.731	223.855	297.983
95% CI:	0.184	1.071	2.153	6.323	11.611	30.263	52.443	89.790	112.350	149.627	223.753	297.885
	0.198	1.223	2.341	6.451	11.763	30.391	52.571	89.984	112.544	149.835	223.957	298.081

Table A.2: SIGCOMM 2001 AGG-MAC Performance Results

	WLAN Throughput (bps)		Packets per Frame		Email Download Response (sec)		HTTP Page Response (sec)		FTP Download Response (sec)	
	Original	AGG-MAC	Original	AGG-MAC	Original	AGG-MAC	Original	AGG-MAC	Original	AGG-MAC
Seed 1	2890.753	4190.187	1	3.5000	0.1073	0.0917	0.2402	0.1024	0.4261	0.3848
Seed 2	1740.220	1503.707	1	3.5000	0.1170	0.0522	0.0890	0.1385	0.1929	0.2874
Seed 3	4273.840	3025.453	1	3.7143	0.1240	0.0744	0.1752	0.1079		0.3066
Seed 4	3451.787	1598.787	1	3.3333	0.1088	0.0725	0.1881	0.1374		0.1735
Seed 5	2871.507	1050.220	1	2.1111	0.0767	0.0716	0.1110	0.0967	0.4032	0.1814
Mean	**3045.621**	**2273.671**	**1.0000**	**3.2317**	**0.1068**	**0.0725**	**0.1607**	**0.1166**	**0.3407**	**0.2667**
95% C.I.	1895.696	656.981	1.0000	2.4361	0.0843	0.0551	0.0850	0.0919	0.1812	0.1558
	4190.714	3230.428	1.0000	4.0062	0.1292	0.0839	0.2138	0.1393	0.4748	0.3433

Table A.3: WLAN Throughput (bps) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	624698.3	1337410	2372332	3931098	5166313	6121838	6803098
Seed 2	369216.1	1175165	2629577	4085539	5077751	5922392	7311414
Seed 3	400841.7	875810.7	3025562	4818576	5268652	6423834	6966020
Seed 4	384531.7	1169035	2847130	4151827	5724907	5713984	6741078
Seed 5	1099556	882155.5	2656270	3926646	5174739	6826022	7025860
Mean	**444822**	**1139355**	**2718650**	**4246760**	**5309406**	**6045512**	**6955403**
95% CI	58876.7	888157.1	2413489	3789099	4991103	5503879	6677830
	830767.2	1390553	3023812	4704421	5627709	6587145	7232975

AGG-MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	453047	999017.7	2905132	4295042	4677677	6687835	6789057
Seed 2	370907	970794	2631792	4098125	5083766	6463385	6839448
Seed 3	457072.6	1068595	2049095	4439709	5236814	5919574	6596546
Seed 4	593332.7	1102662	2690511	3806671	5108206	5938991	6919918
Seed 5	289193.4	882293.8	3118303	4389574	5683152	5683152	6518607
Mean	**468589.8**	**1035267**	**2569133**	**4159887**	**5026616**	**6252446**	**6786242**
95% CI	328102	928032.5	2071282	3838933	4578700	5732217	6576663
	609077.7	1142502	3066984	4480841	5474532	6772675	6995821

Table A.4: Email Download Response Time (sec) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	1.338187	1.276127	1.273018	1.660419	2.979936	3.06995	4.43885
Seed 2	1.247118	1.296025	1.501833	2.007365	3.286699	3.806128	6.0522
Seed 3	1.355304	1.347258	1.623578	2.254882	3.149737	2.658883	3.963652
Seed 4	1.331513	1.298409	1.400458	1.993467	3.394444	3.373943	4.498285
Seed 5	1.284629	1.260143	1.842715	2.181432	2.491643	4.737488	5.194292
Mean	**1.31135**	**1.295593**	**1.52832**	**2.019513**	**3.060492**	**3.529278**	**4.829456**
95% CI	1.256	1.255	1.258	1.734	2.621	2.542	3.821
	1.367	1.336	1.799	2.305	3.500	4.516	5.838

AGG-MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	1.318933	1.31706	1.421392	1.96496	2.21556	3.214532	4.453271
Seed 2	1.313237	1.386206	1.507734	1.87695	2.21556	3.165747	3.621121
Seed 3	1.338043	1.319151	1.280306	1.734024	2.343017	3.329029	3.700737
Seed 4	1.333889	1.267478	1.323334	2.362701	2.444014	2.640791	3.861176
Seed 5	1.318673	1.301968	1.624324	2.398806	2.895512	3.054623	3.728114
Mean	**1.324555**	**1.318373**	**1.431418**	**2.067488**	**2.422733**	**3.080944**	**3.872884**
95% CI	1.311	1.265	1.258	1.698	2.074	2.752	3.456
	1.338	1.372	1.604	2.437	2.772	3.410	4.290

Table A.5: HTTP Page Response Time (sec) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	1.126953	1.088874	1.11312	1.617066	2.893603	2.564647	3.809593
Seed 2	1.012171	1.062954	1.319026	1.995069	2.386861	2.520937	3.486958
Seed 3	1.051496	1.113623	1.138121	2.170647	2.576208	3.03189	4.752442
Seed 4	1.068473	1.072801	1.158159	1.713091	2.099019	2.958858	5.768314
Seed 5	1.155275	1.059954	1.182349	1.624993	2.032074	3.871353	4.44427
Mean	**1.064773**	**1.084563**	**1.182107**	**1.873968**	**2.488923**	**2.769083**	**4.454327**
95% CI	0.993	1.057	1.082	1.567	2.050	2.095	3.350
	1.137	1.112	1.282	2.181	2.928	3.444	5.559

AGG-MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	1.08571	1.055486	1.173096	1.587889	2.063021	2.69573	3.92433
Seed 2	1.082376	1.077153	1.041581	1.545024	1.608402	2.82449	3.192896
Seed 3	1.098433	1.075701	1.101508	1.584995	2.274205	2.346	3.438289
Seed 4	1.148125	1.024209	1.06979	1.524214	1.796345	2.547238	3.072442
Seed 5	1.113574	1.101822	1.482568	1.662411	2.300332	2.467062	3.105449
Mean	**1.103661**	**1.058137**	**1.096494**	**1.560531**	**1.935494**	**2.603364**	**3.406989**
95% CI	1.070	1.022	0.874	1.495	1.561	2.370	2.968
	1.137	1.094	1.319	1.626	2.310	2.837	3.846

Table A.6: FTP Download Response Time (sec) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	37.98518	32.17791	29.60495	39.84076	51.94955	54.58895	72.89831
Seed 2	34.44464	32.9196	30.40111	46.13049	49.94332	67.44773	101.9493
Seed 3	35.19232	30.28213	30.66728	34.12649	45.90415	76.51637	93.40438
Seed 4	33.20475	30.70411	29.43784	34.07965	45.2318	61.05606	109.2755
Seed 5	35.2	31.5	29.8975	34.938	45.74584	64.9	94.3
Mean	**35.20538**	**31.51675**	**30.00174**	**37.82308**	**47.75493**	**64.90182**	**94.36551**
95% CI	33.028	30.187	29.354	31.340	44.020	54.844	77.462
	37.383	32.847	30.649	44.306	51.490	74.959	111.269

AGG-MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	32.32675	32.72118	30.13524	31.09142	41.17497	53.94636	83.38348
Seed 2	30.52336	30.92515	29.21571	38.54042	43.73469	51.43869	72.30913
Seed 3	36.17891	31.30436	27.85806	35.88355	41.00852	55.88482	70.33524
Seed 4	26.05137	30.13165	29.02273	36.59116	43.91356	64.17476	69.89879
Seed 5	31.2	31.2	31.7505	34.66156	47.56636	56.3	73.9
Mean	**31.25608**	**31.25647**	**29.59645**	**35.35362**	**43.47962**	**56.34892**	**73.96533**
95% CI	26.737	30.091	27.795	31.919	40.173	50.416	67.133
	35.775	32.422	31.398	38.788	46.786	62.282	80.798

Table A.7: FTP Upload Response Time (sec) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	31.9483	29.02267	27.48263	32.67273	45.15917	55.25635	106.8355
Seed 2	35.47446	28.51647	28.65674	32.24532	47.65609	78.00263	90.37302
Seed 3	38.0658	30.38613	28.53809	38.73574	50.31432	69.6465	94.49867
Seed 4	35.79855	31.36712	28.9126	32.44519	50.33898	51.70481	74.76741
Seed 5	30.7024	30.0102	29.65165	34.49985	44.91267	68.64636	61.79836
Mean	**34.39791**	**29.86052**	**28.64834**	**34.11976**	**47.67625**	**64.65133**	**91.61866**
95% CI	30.662	28.462	27.677	30.728	44.390	51.122	69.789
	38.134	31.259	29.619	37.512	50.962	78.181	113.448

AGG-MAC

Number of UWs in WLAN Mode	**2**	**6**	**10**	**16**	**20**	**23**	**25**
Seed 1	34.0256	31.00059	28.74363	38.35409	51.97187	71.72923	96.03442
Seed 2	38.5473	30.88335	29.70175	35.75054	46.32221	67.58896	74.97303
Seed 3	33.37654	28.63718	31.80237	37.58979	46.52963	65.54268	86.02254
Seed 4	36.90343	27.842	29.90335	34.65876	44.91328	63.72706	98.49381
Seed 5	33.81959	30.63383	32.10356	38.22709	56.41674	70.08661	116.8579
Mean	**35.33449**	**29.79939**	**30.45093**	**36.91606**	**49.23075**	**67.73491**	**88.88095**
95% CI	32.515	27.990	28.659	34.886	43.227	63.694	69.535
	38.154	31.609	32.243	38.946	55.234	71.776	108.227

Table A.8: Amount of Aggregation (packets per frame) vs. Number of Clients

Original MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	1	1	1	1	1	1	1
Seed 2	1	1	1	1	1	1	1
Seed 3	1	1	1	1	1	1	1
Seed 4			1	1	1	1	1
Seed 5	1	1				1	1
Mean	**1**	**1**	**1**	**1**	**1**	**1**	**1**
95% CI	1	1	1	1	1	1	1
	1	1	1	1	1	1	1

AGG-MAC

Number of UWs in WLAN Mode	2	6	10	16	20	23	25
Seed 1	1.175439	1.58209	2.686567	3.507463	4.41791	5.522388	5.925373
Seed 2	1.209677	1.671642	2.074627	3.791045	4.283582	5.164179	5.477612
Seed 3	1.265625	1.626866	2.597015	3.328358	4.298507	5.208955	5.671642
Seed 4			2.850746	3.791045	4.61194	5.149254	6.104478
Seed 5	1.153846	1.545455				5.208955	6.074627
Mean	**1.216914**	**1.626866**	**2.552239**	**3.604478**	**4.402985**	**5.261194**	**5.794776**
95% CI	1.156397	1.558941	2.135981	3.322045	4.214629	5.069794	5.459708
	1.27743	1.69479	2.968496	3.886911	4.591341	5.452594	6.129845

Vita

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