QUALITY OF SERVICE (QOS) FOR WIRELESS MULTICAST APPLICATIONS

by

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ABBREVIATIONS

3G:	Third Generation (Mobile Telephony Technology)
AP:	Access Point
AQM:	Active Queue Management
ATM:	Asynchronous Transfer Mode
BDL:	Bi-Directional Link
BER:	Bit Error Rate
DCF:	Distributed Coordination Function (802.11 protocol)
DSSS:	Direct Sequence Spread Spectrum
DuReTA:	Dual Receiver Transceiver Architecture
EDCF:	Enhanced Distributed Coordination Function (802.11e protocol)
EPP:	Equal Performance Profile
FHSS:	Frequency Hopping Spread Spectrum
FIFO:	First In First Out
HCF:	Hybrid Coordination Function (802.11e protocol)
IEEE:	Institute of Electrical and Electronic Engineers
IETF:	Internet Engineering Task Force
IGMP:	Internet Group Management Protocol
iMAP:	integrated Multimedia Appliance with Prioritization
ITU-T:	International Telecommunications Union Telecommunication Standardization Sector
MAC:	Media Access Control
M-LWDF:	Modified Largest Weighted Delay First
MOSPF:	Multicast Open Shortest Path First
MPEG:	Motion Pictures Expert Group
MPP:	Maximized Performance Profile

MQAA: Multicast QoS Adaptation Algorithm

OPP: Optimal Performance Profile

OSBU: Overall System Bandwidth Utilization

PCF: Point Coordination Function (802.11 protocol)

PCS: Personal Communication Service

PDV: Packet Delay Variation

PHB: Per Hop Behavior (DiffServ)

PIM: Protocol Independent Multicast

PTD: Packet Transfer Delay

QMACC: QoS Multimedia Adaptive Cell Controller

QoS: Quality of Service

RF: Radio Frequency

SAT: System Application Throughput

SIAT: System Inter-Packet Arrival Time

SOWL: System One Way Latency

SPER: System Packet Error Rate

UDL: Uni-Directional Link

WATM: Wireless Asynchronous Transfer Mode

WCDMA: Wideband Code Division Multiple Access

WLAN: Wireless Local Area Network

WMQF: Wireless Multicast QoS Framework

KUALITI PERKHIDMATAN UNTUK PERISIAN MULTICAST TANPA WAYAR

ABSTRAK

Penghantaran strim multimedia berkualiti tinggi melalui rangkaian tanpa wayar memerlukan kegunaan mekanisme Kualiti Perkhidmatan yang sesuai. Strim multimedia masa-nyata seperti "webcast" radio atau televisyen sesuai dilaksanakan melalui penghantaran multicast. Namun begitu, Kualiti Perkhidmatan multicast tanpa wayar masih merupakan suatu bidang penyelidikan yang aktif kerana mekanisme dan kaedah Kualiti Perkhidmatan telah dibangunkan terutamanya untuk trafik unicast yang dihantar melalui rangkaian berwayar. Tujuan penyelidikan ini adalah untuk mengoptimumkan truput sistem keseluruhan untuk menampung beberapa strim media multicast yang berkeupayaan Kualiti Perkhidmatan yang bersaing dalam rangkaian tanpa wayar jenis "best-effort."

*Tesis ini menerangkan beberapa mekanisme penyesuaian Kualiti Perkhidmatan yang baru serta kesan mekanisme tersebut terhadap prestasi sistem rangkaian berkongsi tanpa wayar jenis "best-effort." Rangka-kerja Kualiti Perkhidmatan Multicast Tanpa Wayar (*Wireless Multicast QoS Framework – *WMQF) direka untuk mencirikan dan mengukur Kualiti Perkhidmatan perisian multicast tanpa wayar. Kegunaan saluran multicast khusus dalam WMQF membolehkan pembekalan Kualiti Perkhidmatan Pembezaan Berkadar (Proportional Differentiation) untuk penyiaran multicast dalam rangkaian tanpa keupayaan Kualiti Perkhidmatan tersirat. Suatu Algoritma Penyesuaian Kualiti Perkhidmatan Multicast diperkenalkan dalam WMQF untuk mengatasi masalah penyesuaian Kualiti Perkhidmatan punca secara "closed-loop." Penyesuaian Kualiti Perkhidmatan punca dicapai melalui proses Jangkaan Kualiti Perkhidmatan yang dikawal dengan Profail Prestasi Multicast yang sesuai. Penjangka*

Kualiti Perkhidmatan menjanakan nilai normal yang digunakan oleh Pemilih Kualiti Perkhidmatan untuk memilih nilai Kualiti Perkhidmatan baru. Seterusnya, Pembentuk Kualiti Perkhidmatan membolehkan tumpuan kepada nilai Kualiti Perkhidmatan dipilih secara menjanakan nilai sasaran Kualiti Perkhidmatan baru berdasarkan nilai Kualiti Perkhidmatan yang sedia ada.

Model yang ditetapkan dalam rangka-kerja disahkan melalui simulasi komputer. Teknik Penyesuaian Kualiti Perkhidmatan "Predictive Adaptive" bersama Profail Prestasi Multicast Optimal dapat mencapai truput sistem keseluruhan yang paling baik di samping memenuhi keperluan Kualiti Perkhidmatan strim media multicast masing-masing.

QUALITY OF SERVICE (QOS) FOR WIRELESS MULTICAST APPLICATIONS

ABSTRACT

High quality multimedia streaming over wireless networks requires the use of suitable Quality of Service (QoS) mechanisms. Real-time multimedia streams such as radio or TV webcasts are well suited for multicast transmission. Nonetheless, QoS for wireless multicast remains an active research area since existing QoS mechanisms and methodologies were developed primarily for carrying unicast traffic through wired networks. The goal is to optimize overall system throughput for supporting several competing QoS-aware multicast media streams in best-effort wireless networks.

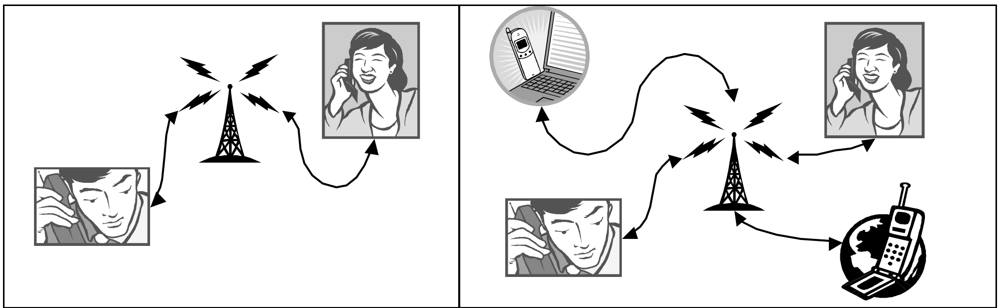
This thesis describes new QoS adaptation mechanisms and their impact on system performance for best-effort shared wireless networks. The Wireless Multicast QoS Framework (WMQF) was developed to characterize QoS support and measurement for wireless multicast applications. The use of dedicated multicast channels in WMQF enabled provisioning of *Proportional Differentiation* QoS for multicast transmission in networks that may not have inherent QoS capabilities. A new *Multicast QoS Adaptation Algorithm* (MQAA) was introduced in WMQF for addressing the problem of closed-loop source QoS adaptation. Source QoS Adaptation was achieved using QoS Estimation processes controlled by suitable *Multicast Performance Profiles*. The QoS Estimators generate normalized values that were used by a QoS Selector to select new QoS values. In addition, a QoS Shaper enabled smooth convergence towards the selected QoS values by generating new QoS target values based on existing QoS settings.

The models specified within the framework were verified using computer simulation. It was discovered that the *Predictive Adaptive QoS Adaptation* technique coupled with

the *Optimal Multicast Performance Profile* achieved the best overall system throughput while maintaining QoS requirements of the respective multicast media streams.

CHAPTER 1 INTRODUCTION

Providing access to information across wireless links is now the 'holy-grail' of computing. All types of services are proliferating, from unregulated IEEE 802.11 based wireless LAN running at 2 Mbps and 11 Mbps, to email and web-browsing via Second generation (2G) mobile cellular phones via Wireless Access Protocol (WAP), and future Third generation (3G) wideband mobile cellular access. The deployment of wideband mobile cellular devices is expected to herald new services such as Voice over IP (VoIP), wireless videophones and real-time delivery of multimedia news and entertainment over wireless links. Nonetheless, most cellular wireless links are designed for circuit switched (point-to-point) communications, whereas the services and applications that are envisioned require group-centric (multipoint-to-multipoint) communication infrastructure (Figure 1.1).



(a) One-to-One Communications **(b) Many-to-Many Communications**

Figure 1.1: Current vs. Future Wireless Communications Scenarios

The convergence of telephony, data and multimedia in mobile systems represents a fundamental shift in the way wireless communications is utilized. Current mobile communications devices such as cellular phones and Wireless Local Area Network (WLAN) enabled notebooks typically utilize one-to-one communications between two parties. Although one-to-one communications may be circuit-switched (e.g., cellular communications), or packet switched (e.g., WLAN), the basic requirement is to interconnect two endpoints. However, with the emphasis towards multimedia convergence and group collaborations, future communications links are required to provide one-to-many or many-to-many types of communications. Examples of one-to-many communications are multimedia streaming of news and entertainment, while many-to-many communications include videoconferencing and group endeavors such as networked games and team-based collaborative work.

The implementation of real-time interactive group communications in wireless systems is one of the goals of current research and development efforts. Although many of the building blocks exist to support such an environment, the combination of these blocks is not optimized. For example, multicasting, a mechanism to support group communications, is not easily supported in existing wireless communications protocols in a bandwidth efficient manner. More importantly, data communications protocols, often based on the Internet Protocol (IP), do not provide robust guarantees of quality or reliability (Quality of Service – QoS) that are desired for such applications such as real-time interactive multimedia streaming applications. In addition, the multimedia data must be delivered to participants that may reside in non-adjacent geographical areas. This requires the use of appropriate routing protocols to deliver the multimedia data to each participant effectively.

Addressing Bandwidth Efficiency and QoS requirements together in wireless environments that are unreliable and often prone to errors is a challenging problem. Nonetheless, the use of suitable wireless communications technologies, link protocols, multicasting, QoS-aware routing strategies, appropriate multimedia transport and encoding schemes, and adaptive application level QoS mechanisms are expected to provide the basis for solving this problem. The focus of this research is to develop a new methodology for optimizing system performance in the face of multiple competing multicast streams, so as to facilitate effective real-time group-based multimedia applications and services. The development of this optimization methodology will enable users to experience bandwidth-efficient "anytime, anywhere" access to multimedia communications with reasonable QoS performance, in contrast to the current "best effort" and bandwidth-inefficient methodologies. This is summarized in Figure 1.2.

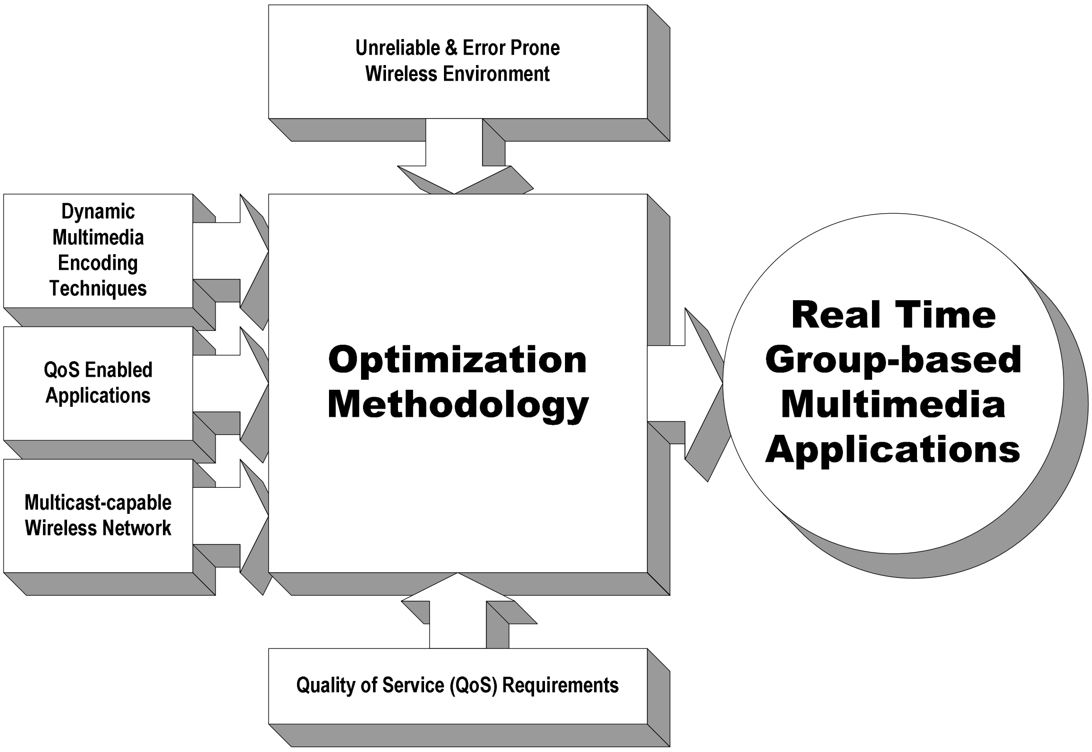


Figure 1.2: Formulation of Problem Domain

The following outline defines the scope of work for the research:

1. Quantify the requirements for multicasting over wireless links
2. Survey of current work in QoS, Multicasting, Mobility & Wireless Multicast
3. Specify a suitable framework for the specification, analysis and deployment of multicasting in wireless environments
4. Develop suitable algorithms to ensure that QoS provisioning for real-time media applications can be accommodated effectively over such links
5. Compare the performance of proposed algorithms using computer simulation techniques to determine their suitability

6. Determine the effectiveness of proposed algorithms and methodologies in achieving given goals

The organization of this thesis is as follows:

Chapter 2 covers background literature survey and discussion of relevant factors for wireless networks and QoS support in such environments, as well as issues relating to quantifying multicast QoS performance.

Chapter 3 defines the requirements and specifications for a comprehensive Wireless Multicast QoS Framework (WMQF) used to support the provisioning of QoS support for wireless multicast applications.

Chapter 4 describes the models and simulation scenarios for WMQF, while simulation results, analysis and discussion are given in **Chapter 5**.

Chapter 6 provides the overall conclusion, constraints and future research work.

CHAPTER 2 LITERATURE SURVEY

2.1 Terminology

When discussing wireless systems, the reader often encounters the problem of identical words with different meaning when used in different context. For example, a channel can refer to the Radio Frequency (RF) bandwidth occupied by a specific radio transmission signal (e.g., the radio channel has frequency 1430 MHz – 1435 MHz), as well as a logical entity derived from a portion of an assigned radio frequency bandwidth (e.g., in Time Division Multiplexing), or even a combination of several frequency bandwidths (e.g., in Orthogonal Frequency Division Multiplexing).

In this discussion, Shannon's concept of a Channel as a logical entity used to convey information from one or more senders to one or more receivers will be adopted. In addition, different Channels are logically distinct from each other. If a sender or receiver is not connected to a given channel, then it would not be able to directly affect the transfer of information within the channel. Conversely, a receiver would be able to observe all information conveyed via a given channel (even if it was intended for other receivers), while a sender is able to communicate information from itself to all receivers of that channel. Consequently, issues such as modulation techniques, coding, and other technology specific issues are not analyzed in detail except where absolutely necessary since they are outside the scope of this discussion.

Similarly, a Link usually refers to the logical or physical connection between two nodes in a given network, with or without specified attributes such as capacity, direction, and quality. In this discussion, a Link is defined to be a specific instance of a Channel, with defined Capacity, Direction, and Quality constraints. Capacity is defined in terms of bits per second (bps). Direction is defined in terms of Unidirectional (one-way) or

Bidirectional (two-way), and Quality is defined in terms of Quality of Service (QoS) parameters to be defined in a later part of this discussion.

The typical wireless network under consideration uses fixed Bases (such as Base Stations or Access Points) to provide Links with specific properties to mobile nodes for sending and receiving information. The Link properties may or may not meet the Link requirements of given mobile applications. This mismatch between the requirements and given properties results in the need for effective methodologies to minimize the discrepancies as well as to maximize the overall system utilization and hence is the focus of this research.

2.2 Characteristics of Wireless Networking Environment

The wireless environment is prone to fading (e.g., Rician, Raleigh, or multi-path fading), has high Bit Error Rates (BER, in the order of 10^{-3} to 10^{-5} compared to fiber optics that can achieve BER of 10^{-12}), and much lower bandwidths (typically from 10 kbps to 2 Mbps compared to 100 Mbps and higher). These issues hamper the deployment of multimedia applications in wireless environments. In addition to the channel characteristics, a wireless environment also creates additional issues, such as mobility and hand-over, in the case of cellular-type wireless systems.

2.2.1 Rayleigh and Rician Fading

The wireless environment experiences fluctuating signal conditions termed Rayleigh and Rician fading. Rayleigh fading is a long term fading phenomenon, whereas Rician fading is caused by multipath signals arising from movement and indirect propagation paths from the transmitter to the receiver (Sklar, 1997a, 1997b). These fading phenomena results in an environment that is non-deterministic and can have varying performance for network applications.

Formulae for Rayleigh and Rician fading (Hess, 1998) are given in 0 for reference. These formulae are commonly used for characterizing and simulating wireless network radio link behavior.

Strategies to overcome short term fading include Physical layer modifications to the radio system such as antenna diversity where the RF signals from multiple antennae are combined to provide a signal with much higher signal to noise ratio. In contrast, long term fading is often resolved by means of different modulation and coding (Forward Error Correction) techniques, both in the Physical as well as Data Link layers.

2.2.2 Bit Error Rates (BER)

The impact of signal fades on wireless transmission is manifested as errors in the bits received by the receiver. The Bit Error Rate (BER) in wireless environment is substantially higher compared to BER for wired networks due to fading effects (Hess, 1998). The variation in power levels causes the BER to vary as well. In general, wireless environments offer a BER of 10^{-3} to 10^{-6}, while data transmission over fiber optics offer a BER of 10^{-12} to 10^{-15}.

The ability to minimize the block errors due to fading phenomena such as Rician and Rayleigh fading would help greatly to reduce the retransmission overheads. One approach is to reduce the size of data frames transferred over the wireless links, thus ensuring that a high number of frames are received correctly. This is the approach taken by Wireless Asynchronous Transfer Mode (WATM). However, the drawback of short data frames is the overhead incurred by the frame header, which reduces the available bandwidth over the wireless link for actual data transmission. Another approach is to protect the data frame transmission in order to ensure that the recipient is able to recover the frame irrespective of errors. This implies that some form of Forward Error Correction (FEC) must be employed on the wireless link. The most powerful codes today include concatenated Reed-Solomon codes and Turbo codes that can address both random and block errors of a long duration (Sklar, 1997c).

Nonetheless, the use of effective FEC implies that frames can no longer be encoded for transmission independently and decoded independently of each other. Instead a transmission queue must be utilized to furnish the FEC coder with sufficiently sized data to enable it to function, while the receiver must have a ‘training phase’ in which it locks onto the FEC stream and be able to start decoding.

Thus, the need to improve the BER performance of wireless channels is in some ways conflicting with the desire to reduce the frame size to avoid excessive frame corruption. The use of FEC also introduces coding delay into the transmission stream. This additional delay may impact the suitability of FEC techniques for real-time applications.

2.3 Evolution of Wireless Network Connectivity

2.3.1 Early Wireless Systems

Early wireless systems were primarily radio frequency based point-to-point transmission networks. However, since radio transmission can be picked up by multiple receivers within its coverage area, RF-based broadcast networks for media dissemination as well became widespread. In addition, shared two-way radio systems with large area coverage using half-duplex mobile handsets were introduced for military, emergency, public safety, and industrial use. All such early radio systems suffer from limited capacity and little provision for channel access arbitration.

The development of trunked two-radio systems was the first step towards implementing channel access arbitration mechanisms. Base stations act as channel access arbitrators, dynamically assigning or denying access to the limited number of available channels, and extending the reach of a given user to others beyond its immediate radio range. However, the overall system capacity is still limited by the relatively high transmission power of the handsets and base stations.

2.3.2 Cell Based Infrastructure

The use of automated power control mechanisms and transmission bases with limited coverage made possible the development of cellular wireless communication networks. Cellular systems take advantage of frequency reuse to maximize the bandwidth efficiency of the wireless network. In addition, the use of advanced channel access arbitration protocols (also known as Media Access Control or MAC protocols) made possible the effective sharing of the given communication channels among a group of users. MAC protocols in such systems include Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), Code-Division Multiple Access (CDMA), Wideband CDMA (W-CDMA) and that have been implemented in first, second and currently third generation cellular telephony networks.

It is assumed that the network is subdivided into uniform cells, where a fixed Base control a set of shared channels within the cell, for this research. Shared channels necessitate the use of suitable Medium Access Control (MAC) mechanisms to arbitrate among the respective users. In addition, cell-based infrastructures require the specification of a suitable RF spectrum reuse plan. Typical hexagonal cell layouts use a three-color reuse structure (Steele & Hanzo, 1999), where three distinct sets of RF modulation parameters (e.g., frequency, spreading code) are used to provide non-interfering overlapping coverage across the entire area.

For example, 802.11-based wireless networks uses shared channels, while 3G systems provide dedicated channels as well as shared channels. In cell-based DSSS 802.11 networks, this results in only two (maximum of three where permitted by regulatory agencies) channels per cell due to frequency spectrum allocation and interference limits (IEEE, 1999a, 1999b, Andren, 1997). In comparison, FHSS 802.11

as well as 3G networks allow for multiple channels per cell (Andren, 1997, Steele & Hanzo, 1999), enabling the use of more flexible bandwidth allocation schemes.

2.3.3 Wireless Link Characteristics

The use of multicast for real-time multimedia applications helps to reduce the system bandwidth requirement for a given cell. The transmission of multicast traffic over wireless media can be an inherently spectrum efficient way to maximize the capacity of the wireless network, since radio signals can potentially be received by all nodes within its transmission range. However, cell-based systems require the transmission of data from the mobile node to the base station, which would then retransmit it to the intended receiver. For the systems under consideration, the deployment of real-time multimedia streaming only makes sense if the system is capable of providing a bandwidth of 2 Mbps. 802.11 specifies 2 Mbps operation (in addition to other data rates), while 3G systems can support several 2 Mbps dedicated channels per cell although current 3G implementations support only 384 kbps. Consequently, the practical upper limit per multicast stream is somewhat less than 2 Mbps.

While unicast Quality of Service (QoS) issues have largely been addressed by means of resource-reservation (Jorguseski et al., 2001), call admission control (CAC) (Jorguseski et al., 2001), and link level error-correction (IEEE, 1999a), most of the given approaches are not optimal for multicast traffic since they were designed for point-to-point flows. For example, the impact of adding an additional multicast receiver to a cell depends on whether other users are currently accessing the same multicast flow. In addition, the inherent QoS capabilities of the underlying wireless technology has a significant impact on how feasible are the given techniques for that particular system. In 3G systems, unicast traffic QoS requirements is addressed using physical and MAC Layer approaches (Fitzek, Morich & Wolisz, 2000):

Table 2.1: Physical & MAC Layer Solutions for Unicast Wireless QoS

Approach	Mechanism	Solves	Limitations
Error Correction	ARQ and FEC	PER, BER	Variable PTD and PDV
Power Control	Increase Tx power	PER, BER	Not suitable in 'bad' channels
Bandwidth Provisioning	DQRUMA, LIDA	PTD	Doesn't address PDV from high BER
Multi-Channel Tx (MC-CDMA, OFDMA)	SMPT (Simultaneous MAC Packet Tx)	PER, BER, PTD, PDV	SNR for other users, resource usage

A specific combination of the various approaches given is used to achieve specific targets for PTD, PDV and PER/BER arising from channel impairment in the wireless environment (Dixit, Guo & Antoniou, 2001).

For 802.11 networks, the options are more limited, since error correction, power control and multi-channel transmission are not provided for in the standard. The AP can operate in Point Coordination Function (PCF) Mode, to implement some form of bandwidth provisioning. However, this reduces the overall throughput on the shared channel since PCF is implemented via node polling (IEEE, 1999a). The recent IEEE 802.11e standard implements a Hybrid Coordination Function (HCF) for probabilistic QoS support in shared channels (Grilo, Macedo & Nunes, 2003, Gu & Zhang, 2003, Mangold et al., 2003, Pattarra-Atikom, Krishnamurthy & Banerjee, 2003). The HCF includes an Enhanced Distributed Coordination Function (EDCF) mode. While EDCF does not eliminate contention inherent in the 802.11 access protocol, it provides higher priority traffic a better chance of acquiring the channel compared to the existing Distributed Coordination Function (DCF) mode. This would be useful for enabling upstream (from Mobile to Access Point) QoS support.

In this research, no assumptions were made regarding the QoS capability of the underlying wireless network, hence only shared channels with limited or no QoS capability would be considered.

2.3.4 Dedicated Multicast Channels

Since we only consider shared channels, multicast streams must compete with other real-time and non real-time traffic carried on the shared channel. For wireless systems utilizing a single shared channel among the mobile nodes and the Base (such as found in the contention-based IEEE 802.11 networks), this mixture of unicast, multicast, uplink, downlink, real-time and non real-time traffic is not efficient in terms of QoS provisioning since competing traffic with different QoS requirements have to share access to the same link.

We do not consider QoS for unicast traffic in this research since effective solutions to unicast transmission are often closely coupled with specific RF modulation, coding and channel access techniques. An effective solution to the problem of arbitrating among different competing traffic types (unicast, multicast, real-time and non real-time data transfer) is to transmit all QoS sensitive multicast traffic from multiple sources separately over a dedicated unidirectional link that is received by all nodes within the cell. The existing shared channel would be used for bidirectional unicast traffic between mobile nodes and the Base, and for upstream multicast transmission from the mobiles.

This means that multicast traffic originating from a mobile node would encounter variable link conditions on the bidirectional link due to channel contention and traffic mix, while multicast traffic forwarded by the Base has much better control over the unidirectional link condition since traffic is one-way from the Base to all mobiles within the cell without any channel contention from the mobiles. Although there is link contention among different competing multicast streams, their QoS requirements are known and could therefore be managed using suitable bandwidth arbitration algorithms.

These dedicated unidirectional channels could be implemented as overlay macrocells where the coverage of the unidirectional multicast channel extends into the coverage area of several standard sized cells. However, to simplify the analysis, it is assumed that the unidirectional multicast link coverage is identical to the bidirectional link coverage. Consequently, two links are needed per cell.

The requirement for at least two links per cell is met in both 3G and 802.11-based systems. In order to have a common basis for comparison, an infrastructure consisting of a minimum configuration of two 2 Mbps channels per cell will be assumed.

The Base is equipped with two transmitters and one receiver to support this configuration. The increase in complexity for mobile nodes is in the addition of a second receiver, to create a one-transmitter two-receiver system. This proposed dual-receiver architecture is termed Dual Receiver Transceiver Architecture (DuReTA). While the DuReTA architecture is more complex than conventional transceivers, it is not significantly beyond the capabilities of current semiconductor processes. Recently, Engim Inc. has announced the development of Wideband Multi-Channel WLAN chipsets (Engim, 2004) which support simultaneous transmission on multiple 802.11 channels, which could be used to implement Base support for DuReTA equipped mobiles.

2.3.5 Mobility and Wired Network Integration

The deployment of cellular wireless systems necessitates the use of wireless transmission equipment, otherwise known as Bases. However, the issue arises where such Bases should be *confederated*, whereby they manage the mobility of different mobile nodes as they cross cell boundaries, forwarding traffic to adjoining cells as needed, thus operating as forwarding routers or Cellular Base Stations. Confederated Bases often appear as a single logical network to the mobile hosts, thus supporting seamless Layer 2 mobility. In contrast, *autonomous* Bases that only provide links to mobiles within its cell are much simpler in design and do not manage handoff and roaming issues. Such *autonomous* Bases (often known as Access Points) operate as bridges or edge routers where they primarily support media and protocol conversion from the wired to the wireless domain, and depend on upstream routers to perform the necessary mobility routing and enforce inter-network QoS guarantees. Roaming from one autonomous Base to another involves Layer 3 mobility.

The appeal of the autonomous Base approach is that they are easy to interface with existing equipment. Access Points (APs) attach to a port on the router, and perform the necessary physical layer conversion as well as any additional forward error correction, segmentation and reassembly requirements for the wireless link. However, such a setup places the burden on mobile nodes to handle handoffs and mobility issues, since standard routing protocols do not cater for mobility or handoff of mobile terminals. Additional signaling mechanisms such as Mobile IP (Perkins, 1997) need to be adopted to support mobile-initiated handoff. The advantage in adopting a mobile initiated handoff approach is the simpler infrastructure requirements. Conversely, the disadvantage is the significant impact on QoS requirements of the mobile applications, since delays are introduced during the setup and teardown of new connections.

2.4 Evolution of Quality of Service (QoS) Paradigms

2.4.1 Quality of Service Definition

Quality of Service (QoS) refers to the ability to provide a satisfactory experience for users of a given system. Since satisfaction is inherently a subjective criterion, it is very difficult to quantify and measure the performance of a multimedia multicast system using such criteria. Often application level and user level QoS parameters are translated into networking parameters to make the implementation and measurement feasible. The implicit assumption is that subjective QoS criteria can be converted into objective network-level QoS criteria; and this is the responsibility of the application or user. Consequently, we only address QoS as pertains to the specifications use at the network connectivity level.

Typically, QoS from the network perspective is defined using the following criteria:

- Delay
- Jitter (Delay Variation)
- Guaranteed Bandwidth
- Error Rates

2.4.2 Guaranteed QoS

The traditional definition of QoS is a rigid guarantee of the negotiated parameters on a per-flow basis. The network maintains QoS state information to regulate and meet the respective QoS guarantees; applications do not have to deal with QoS issues once the required QoS service level has been successfully negotiated with the network. The burden of QoS verification and enforcement falls on the network infrastructure. QoS Signaling protocols such as Resource Reservation Protocol (RSVP) are used by sending hosts to negotiate their QoS requirements with the network. Networks utilizing per-flow QoS signaling with hard network QoS guarantees are grouped under the Integrated Service (IntServ) architecture model. Examples include Asynchronous Transfer Mode (ATM) networks with inherent QoS support (Stallings, 1998), as well as bandwidth reservation based QoS techniques to support IntServ requirements over the best-effort Internet (Xiao & Ni, 1999).

2.4.3 Resource Reservation Protocol (RSVP)

RSVP (Braden, et al., 1997) is a background signaling protocol (Mankin, et al., 1997) that can be used by sending hosts to request specific QoS for data transmission. RSVP normally works in conjunction with the *Integrated Services* (*IntServ*) architecture that manages each traffic flow independently.

In order to setup a QoS enabled flow, the sender sends a RSVP PATH message to the receiver. Each PATH message contains details such as sender identity, data packet format, data flow traffic characteristics, as well as the profile of the destination host IP and port. The PATH state is installed on each of the routers along the route taken to the receiver. When the receiver receives the PATH message, it generates a RESV message that travels on the return path to the sender. The RESV message is examined by each Router's Admission and Policy Controls as it travels back to the sender. Admission control checks whether the router has sufficient resources to accept the RSVP flow and Policy Control determines whether receivers have the permission to receive the specified QoS. If the RSVP flow is accepted, then the requested flow will be classified by a Classifier into a specific class. The router's Packet Scheduler will then schedule the delivery of that specific data flow according to its class. If the reservation fails on any of the routers along its path, then an error message will be generated and all routers notified to terminate the RSVP flow request. The RSVP mechanism is illustrated in Figure 2.1.

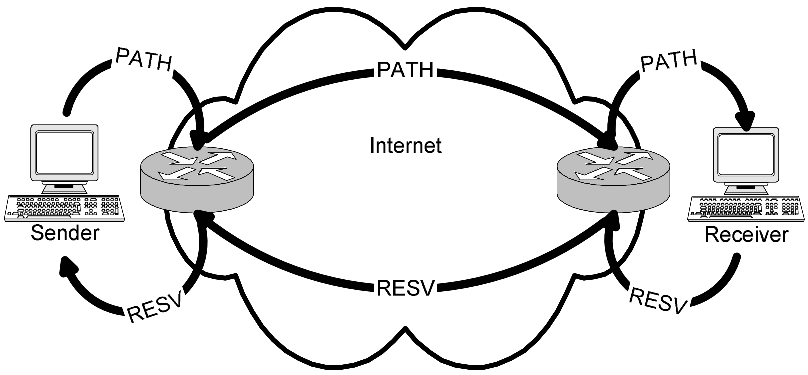


Figure 2.1: RSVP Mechanism

2.4.4 Aggregated QoS

The adoption of multimedia services over the Internet increased the burden significantly on network routers attempting to use *IntServ* as the QoS mechanism, since the number of networks and users can potentially be enormous. Consequently, the notion of 'elasticity' in QoS service classes has been introduced (Dixit, Guo & Antoniou, 2001). Traffic flows are categorized into different classes, such as *premium*, *gold* and *silver*. *Premium* classes have guaranteed QoS performance and are similar to wired QoS flows. Gold and Silver classes have relative QoS differentiation (Dovrolis & Ramanathan, 1999). When channel degradation occurs for *gold* and *silver* classes, QoS performance is reduced based on the *elasticity* associated with the respective class, such that the aggregate behavior of the various flows within the same service class maintains the overall QoS characteristics, although short term fluctuations in traffic patterns may cause instantaneous deviations from the aggregate QoS behavior (Dovrolis & Ramanathan, 1999). The aggregation of QoS flows maps well to the *Differentiated Services* (*DiffServ*) (Xiao & Ni, 1999) scheme under consideration for Internet-based QoS support.

DiffServ was designed to address the limitations of *IntServ* and enable scalable QoS in the Internet without the need for per-flow state and signaling at every hop (Nichols et al., 1998). A *DiffServ*-compliant network functions by using the TOS octet in IPv4 or Traffic Class octet in IPv6, termed the DS field, to perform service classification. Six bits of the DS field define various *DiffServ* Codepoints (DSCP) while another two bits remain unused. Routers use the DSCP to classify packets and provide appropriate packet forwarding termed Per-Hop Behavior (PHB). PHB of *DiffServ* can be implemented via several mechanisms such as Strict Priority Queuing, Class Based Queuing (CBQ), Weighted Fair Queuing (WFQ), Weighted Round Robin (WRR), etc. according to the needs of service provider (Blake et al., 1998). Sophisticated classification and conditioning functions are implemented only on boundary routers. Packet flows with the same DSCP markings are aggregated into a common queue and forwarded. This aggregation mechanism differentiates *DiffServ* from *IntServ* where per flow behavior is enforced. Consequently, aggregated traffic handling requires significantly less state and processing power in routers, especially those on large networks (Bernet, 2000). RSVP can still be used to negotiate QoS parameters between the end nodes and the *DiffServ* edge routers.

DiffServ faces some issues related to multicast traffic support. Since multicast group membership is dynamic, it is difficult to predict in advance the amount of network resources consumed by multicast traffic for a particular group. So, it may be difficult to provide quantitative service guarantees to multicast senders (Blake et al., 1998).

2.4.5 Proportional QoS

More recently, the provisioning of QoS in best-effort networks has been investigated by Wydrowski & Zukerman (2002). By definition, best-effort networks cannot provide guaranteed QoS, unlike traditional QoS approaches. QoS for best-effort networks is implemented using application-level closed-loop control using congestion feedback signals and suitable Active Queue Management (AQM) schemes. Further relaxation of the guaranteed and aggregated QoS assumption has resulted in the development of proportional service models such as Proportional Differentiation (Chen et al., 2003). In Proportional Differentiation schemes (Dovrolis & Ramanathan, 1999, 2000, Kumar, Kaur & Vin, 2001, Dovrolis, Stiliadis & Ramanathan, 2002), QoS parameters for each traffic class is not guaranteed; instead, the Proportional Differentiation scheme enforces class differentiation such that higher priority traffic always enjoys equal or better service than lower priority traffic. Best Effort and Proportional QoS service disciplines are well suited to the nature of wireless environments. This has been applied to the provisioning of Soft QoS for shared wireless channels using the Token Bank Fair Queuing (TBFQ) queue management discipline (Wong, Zhu & Leung, 2003).

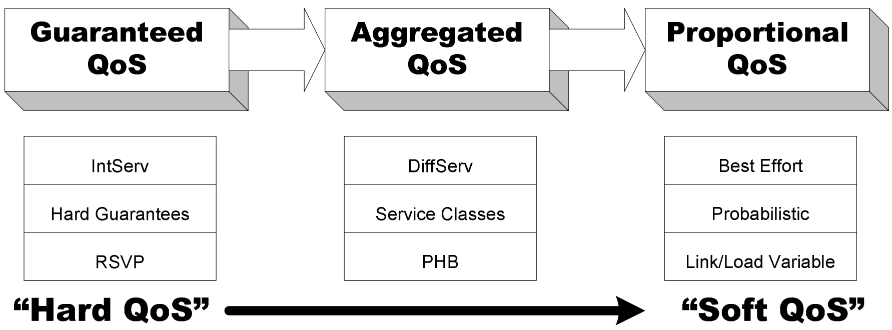


Figure 2.2: Evolution of QoS Concepts

2.4.6 QoS Monitoring and Enforcement

The different notions of QoS support provided by the network result in different requirements for QoS-enabled applications (Dovrolis & Ramanathan, 1999). Applications which assume an *IntServ* "Guaranteed QoS" network environment typically adopt open-loop QoS controls: The application determines the class of QoS support required, and this supported is either granted or denied at the start of a session. If the QoS request is unsuccessful, the application will either have to wait until the network is able to grant the request or else terminate the session. QoS parameters are fixed for the duration of the session. The network is solely responsible for ensuring that guaranteed QoS levels are maintained.

In contrast, applications targeted for Aggregated QoS support in DiffServ networks, as well as Proportional QoS or Best Effort QoS support typically adopt closed-loop QoS controls: feedback from the target endpoints (receivers) or the network are received periodically, and used to modify network layer QoS settings, as well as application QoS profiles in order to better match available network QoS support. Network layer adaptation occurs in the prioritization and buffer management policies for classifying different types of network traffic.

Possible approaches for application layer QoS adaptation is the use of Layered transmission (such as MPEG-4 (Motion Pictures Expert Group) (Turletti, Parisis & Bolot, 1997) with high priority substreams and lower priority enhancement streams), or change of encoding/decoding algorithms in the face of varying QoS profiles. A multi-layered QoS contract is proposed in (Naghshineh, 1999), where the transmission of multimedia data is subdivided into several sub-streams with different importance (Naghshineh & Willebeek-LeMair, 1997). Each substream negotiates for a different level of QoS, such that the most important elements in the fundamental transmission stream is given the highest QoS values while less important elements are assigned increasingly lower QoS values. If the wireless medium were unable to sustain the QoS of the entire transmission, it would still be able to meet the QoS requirements of one or more substreams such that the fundamental transmission streams would continue to be received correctly.

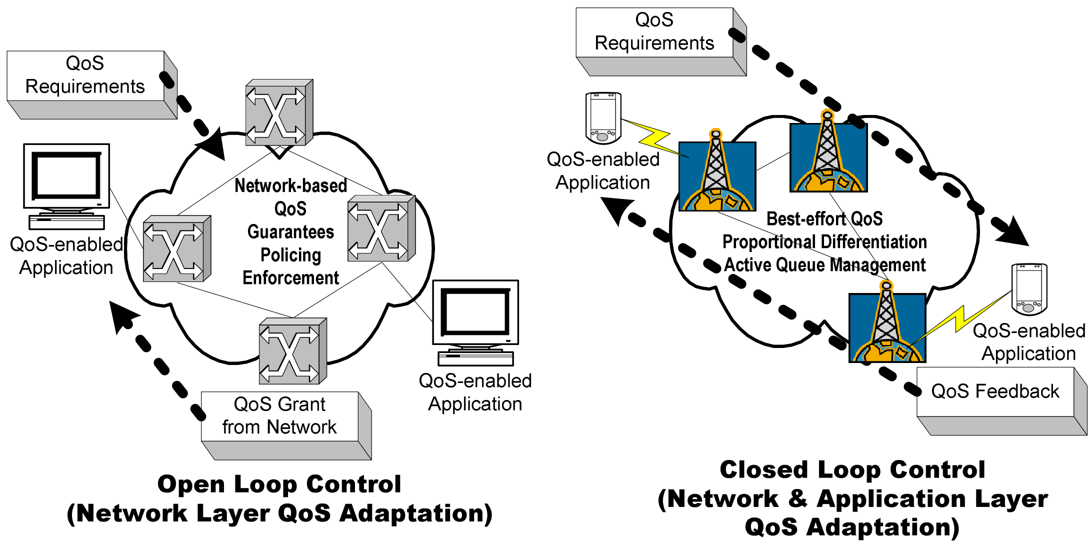


Figure 2.3: Open vs. Closed Loop QoS Control

2.4.7 QoS in a Wireless Environment

Hard QoS guarantees are very costly to implement in wireless systems since the link characteristics vary in availability and throughput due to mobility and environmental interference. In addition, wireless links are more susceptible to outages and link fluctuations. Consequently, the notion of 'guaranteed Quality of Service' has to be revised. Attempting to provide a blanket QoS guarantee would require over-provisioning of network resources, as well as incur heavy power requirements on the portable or mobile unit, while possibly not being able to achieve the stated QoS objectives in any case.

Consequently, the proportional QoS (Soft QoS) approach will be adopted for wireless networks. We will assume the use of layered (substream) transmission as a QoS adaptation strategy, since the use of alternative encoding/decoding algorithms is application specific. In addition, substream adaptation is performed at the application layer, resulting in an increase or decrease in bandwidth requirements for the transmitted multicast stream.

2.5 QoS Parameters and Metrics

In order to quantify QoS performance, we must specify quantitative parameters as well as suitable metrics for measuring the achieved QoS in the network. QoS Parameters are used by the multimedia source to request for network QoS support, while QoS metrics are used to measure the performance of the given multimedia flow.

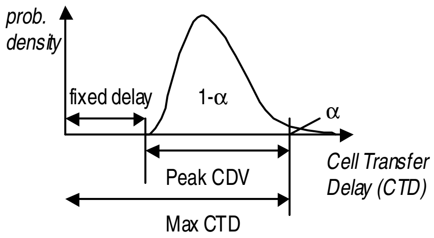
2.5.1 Standard QoS Metrics

Work on QoS evolved primarily from research into providing network performance guarantees in Asynchronous Transfer Mode (ATM) networks. The following parameters were used to specify QoS guarantees (Davis, 1999):

- Peak Cell Rate (PCR)
- Sustained Cell Rate (SCR)
- Maximum Burst Size (MBS)

The above bandwidth parameters are then measured via the following QoS metrics at the receiver (Davis, 1999):

- Max. Cell Transfer Delay (CTD)
- Peak Cell Delay Variation (CDV)
- Cell Loss Ratio (CLR)



where:

Max. CTD = $F_{(1-\alpha)}$(Delay)

Peak CDV = Max CTD – Fixed Delay

CLR = Lost Cells / Total Transmitted Cells

Figure 2.4: Definition of Cell Transfer Delay (CTD) (Davis, 1999)

Standardization of the various QoS and bandwidth parameters has resulted in International Telecommunications Union Telecommunication Standardization Sector (ITU-T) Recommendations Y.1221, Y.1540 and Y.1541 which define a QoS Framework with Traffic Parameters relating to network capacity requirements, QoS Classes and QoS metrics used to determine network performance targets (Seitz, 2003, Bain & Seitz, 2004).

- Peak Rate (Rp)
- Peak Token Bucket Size (Bp)
- Sustainable Rate (Rs)
- Sustainable Token Bucket Size (Bs)
- Maximum Packet Size (M)
- IP Packet Transfer Delay (IPTD)
- IP Packet Delay Variation (IPDV)
- IP Packet Loss Ratio (IPLR)
- IP Packet Error Ratio (IPER)
- Spurious IP Packet Rate (SIPR)

Six QoS classes (0 to 5) were defined, ranging from real time to unspecified QoS requirements.

Table 2.2: ITU-T 1540 & 1541 QoS Classes and Targets (Seitz, 2003)

QoS Class	Application	Max. IPTD	Peak IPDV	Max. IPLR	Max. IPER
0	Real-time, jitter-sensitive, high interaction (VoIP, voice teleconferencing)	100 ms	50 ms	1×10^{-3}	1×10^{-4}
1	Real-time, jitter-sensitive, interactive (VoIP, video teleconferencing)	400 ms	50 ms	1×10^{-3}	1×10^{-4}
2	Transaction data, highly interactive (e.g., signaling)	100 ms	U	1×10^{-3}	1×10^{-4}
3	Transaction data, interactive	400 ms	U	1×10^{-3}	1×10^{-4}
4	Low loss only (short transactions, bulk data, video streaming)	1 s	U	1×10^{-3}	1×10^{-4}
5	Traditional applications of default networks	U	U	U	U

A QoS-enabled flow would specify the bandwidth requirements and QoS class when requesting for a particular level of service from the network. Further work to harmonize the ITU-T Recommendations with Third Generation Partnership Project 2 (3GPP) cellular and ITU-T is underway (Bain & Seitz, 2004). Nonetheless, the standards assume that the network is capable of sustaining the QoS requirements that were negotiated at the beginning of the session.

2.5.2 Proportional QoS Parameters and Metrics

QoS parameters and metrics for proportional QoS wireless networks can be defined based on the standard (hard) QoS parameters given previously. However, the difference is that since no QoS guarantees are given, the following parameters are indicative only and would be used as prioritization parameters by the respective active queue management schemes:

- Min. Required Bandwidth (BW)
- Max. Packet Transfer Delay (PTD)
- Avg. Packet Error Rate (PER), alternatively Avg. Bit Error Rate (BER)

Nonetheless, the various standard QoS metrics would still be used as a means to evaluate the effectiveness of the network QoS mechanisms in meeting the requested QoS levels. Consequently, the following metrics analogous to those defined in ITU-T Y.1540 would be captured:

- Max. Packet Transfer Delay (PTD)
- Peak Packet Delay Variation (PDV)
- Avg. Packet Error Rate (PER)

It should be noted that the Packet Error Rate as defined for Soft QoS here does not distinguish between packets rejected due to errors as well as packets discarded (lost) in the network (i.e., PER $\cong$ IPLR + IPER).

2.5.3 Evolution of Multicast QoS Metrics

The issue of Multicast QoS metrics has not been addressed in most work into QoS metrics. One of the major difficulties is in quantifying what is meant as QoS for multicast, since QoS was initially defined with the notion of one-to-one semantics. As there are one-to-many and many-to-many semantics present in multicast environments, the definition of QoS must correspondingly be expanded to address these issues.

The definition of suitable metrics for measuring network performance has been spearheaded by the IETF IPPM (IP Performance Metrics) working group. However, the work to date has concentrated on unicast flows since the definition and collection of multicast metrics remains a complicated issue. Efforts to define multicast metrics are given in (Irey & Marlow, 1999, Stephan, 2002), and represent extensions of established unicast performance metrics. However, since multicast performance is of necessity an aggregate measurement, realistically the metrics can only be compiled offline, where analysis of system performance is done *a posteriori.* In order to support QoS within the network, real-time feedback of QoS estimation parameters is necessary for source adaptation and network resource reallocation to be successful. Consequently, traditional queue-based estimation parameters will be used for QoS adaptation. Nonetheless, the *a posteriori* QoS metrics are useful as analytical tools for evaluating the effectiveness of the queue-based estimation parameters in performing QoS adaptation. For the purpose of metric calculations, packets are assumed to be in error if they are late or lost (due to drop or channel error), therefore PER and BER are analogous (BER can be derived from PER).

2.5.4 Group Multicast Metrics

The concept of "Group QoS Metrics" (Table 2.3) is introduced to address the one-to-many semantics of multicasting. Since aggregation is the group statistics for a number of multicast nodes (1 transmitting node, N receiving nodes), we have to consider the mean (average), minimum, and maximum (peak) values:

Table 2.3: Group (One-to-Many) Multicast QoS Statistics

One-to-Many Multicast	Mean	Min	Max
Group Delay	$\sum_{i=1}^{N} Max\ PTD_i \Big/ N$	$\min(Max\ PTD_i)\big\vert_{i=1}^{N}$	$\max(Max\ PTD_i)\big\vert_{i=1}^{N}$
Group PDV	$\sum_{i=1}^{N} Peak\ PDV_i \Big/ N$	$\min(Peak\ PDV_i)\big\vert_{i=1}^{N}$	$\max(Peak\ PDV_i)\big\vert_{i=1}^{N}$
Group BER	$\sum_{i=1}^{N} BER_i \Big/ N$	$\min(BER_i)\big\vert_{i=1}^{N}$	$\max(BER_i)\big\vert_{i=1}^{N}$

These multicast group performance measures have been used to define various metrics that characterize the behavior of a given group (Irey & Marlow, 1999, Stephan, 2002). Notably, the following metrics correspond to the statistical measures specified in Table 2.3:

- Group Delay: Group One Way Latency (GOWL)
- Group PDV: Group Inter-Arrival Time (GIAT)
- Group BER: Group Packet Error Rate (GPER), PER is analogous to BER

The detailed definition of these and other metrics is given in Appendix B.

2.5.5 System Multicast Metrics

For the case of many-to-many semantics, "System QoS Metrics" has been defined to quantify the statistical behavior of the entire population of multicast nodes (M transmitting nodes, N receiving nodes each). The System Metrics in Table 2.4 are natural derivations from the Group Metrics defined in Table 2.3:

Table 2.4: System (Many-to-Many) Multicast QoS Statistics

Many-to-Many Multicast	Mean	Min	Max
System Delay	$\sum_{j=1}^{M} Group\ Delay_j \Big/ M$ $\Rightarrow \sum_{j=1}^{M}\sum_{i=1}^{N} Max\ PTD_{ji} \Big/ MN$	$\min(Max\ PTD_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$	$\max(Max\ PTD_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$
System PDV	$\sum_{j=1}^{M} Group\ PDV_j \Big/ M$ $\Rightarrow \sum_{j=1}^{M}\sum_{i=1}^{N} Peak\ PDV_{ji} \Big/ MN$	$\min(Peak\ PDV_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$	$\max(Peak\ PDV_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$
System BER	$\sum_{j=1}^{M} Group\ BER_j \Big/ M$ $\Rightarrow \sum_{j=1}^{M}\sum_{i=1}^{N} BER_{ji} \Big/ MN$	$\min(BER_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$	$\max(BER_{ji}) \Big\vert_{i=1}^{N} \Big\vert_{j=1}^{M}$

The System QoS metrics represents the actual system-wide conditions at the given time. By using the Node and Group metrics given in Appendix B, we can derive corresponding System-wide Multicast Metrics (Table 2.5) for quantifying the performance of multicast QoS algorithms. System-wide metrics aggregate the performance metrics of all active multicast groups in the system, and can be used to evaluate the performance of QoS adaptation algorithms under investigation. Consequently, optimal multicast QoS adaptation algorithms will result in optimum system-wide multicast metrics.

Table 2.5: System Multicast Metrics

System (Receiver) Metrics		
Metric	***Name***	***Definition***
$SOWL$	System One-Way Latency	System-wide Multicast One Way Latency
$SOWL_{avg}$	Average SOWL	$SOWL_{avg} = \sum_{j=1}^{g} \frac{GOWL_{avg_j}}{g}$, for g multicast groups
$SOWL_{\max}$	Max. SOWL	$SOWL_{\max} = \underset{\forall j}{MAX}\{GOWL_{\max_j}\}$, for g multicast groups
$SOWL_{\min}$	Min. SOWL	$SOWL_{\min} = \underset{\forall j}{MIN}\{GOWL_{\min_j}\}$, for g multicast groups
$SOWL_{sdev}$	Std. Dev. SOWL	$SOWL_{sdev} = \sqrt{\left(\sum_{j=1}^{g}(GOWL_{avg_j} - SOWL_{avg})^2\right) \Big/ (g-1)}$, for g multicast groups
$SIAT$	System Inter-Arrival Time	System-wide Multicast Inter-Arrival Time
$SIAT_{avg}$	Average SIAT	$SIAT_{avg} = \sum_{j=1}^{g} \frac{GIAT_{avg_j}}{g}$, for g multicast groups
$SIAT_{\max}$	Max. SIAT	$SIAT_{\max} = \underset{\forall j}{MAX}\{GIAT_{\max_j}\}$, for g multicast groups
$SIAT_{\min}$	Min. SIAT	$SIAT_{\min} = \underset{\forall j}{MIN}\{GIAT_{\min_j}\}$, for g multicast groups
$SIAT_{sdev}$	Std. Dev. SIAT	$SIAT_{sdev} = \sqrt{\left(\sum_{j=1}^{g}(GIAT_{avg_j} - SIAT_{avg})^2\right) \Big/ (g-1)}$, for g multicast groups
SAT	System Application Throughput	System-wide Application Throughput
SAT_{avg}	Average SAT	$SAT_{avg} = \sum_{j=1}^{g} \frac{GAT_{avg_j}}{g}$, for g multicast groups
$SAT_{\max}$	Max. SAT	$SAT_{\max} = \underset{\forall j}{MAX}\{GAT_{\max_j}\}$, for g multicast groups
$SAT_{\min}$	Min. SAT	$SAT_{\min} = \underset{\forall j}{MIN}\{GAT_{\min_j}\}$, for g multicast groups
SAT_{sdev}	Std. Dev. SAT	$SAT_{sdev} = \sqrt{\left(\sum_{j=1}^{g}(GAT_{avg_j} - SAT_{avg})^2\right) \Big/ (g-1)}$, for g multicast groups

Table 2.5, continued.

Metric	*Name*	*Definition*
$SPER$	System Packet Error Rate	System-wide Packet Error Rate
$SPER_{avg}$	Average SPER	$SPER_{avg} = \sum_{j=1}^{g} \frac{GPER_{avg_j}}{g}$, for g multicast groups
$SPER_{\max}$	Max. SPER	$SPER_{\max} = \underset{\forall j}{MAX}\{GPER_{\max_j}\}$, for g multicast groups
$SPER_{\min}$	Min. SPER	$SPER_{\min} = \underset{\forall j}{MIN}\{GPER_{\min_j}\}$, for g multicast groups
$SPER_{sdev}$	Std. Dev. SPER	$SPER_{sdev} = \sqrt{\left(\sum_{j=1}^{g}(GPER_{avg_j} - SPER_{avg})^2\right) \Big/ (g-1)}$, for g multicast groups

2.5.6 Mapping Proportional QoS Metrics to Specific Multicast Metrics

The three basic QoS parameters, PTD, PDV, and PER can be mapped to corresponding specific Multicast Metrics defined previously:

Table 2.6: Mapping Proportional QoS Metrics to Multicast Metrics

Proportional QoS Metrics	Multicast Metrics
Peak Packet Transfer Delay (PTD_{max})	$LOWL_{max}$, $GOWL_{max}$, $SOWL_{max}$
Peak Packet Delay Variation (PDV_{max})	$LIAT_{sdev}$, $GIAT_{sdev}$, $SIAT_{sdev}$
Average Packet Error Rate (PER)	$LPER_{avg}$, $GPER_{avg}$, $SPER_{avg}$

In general, we want to have low $SOWL_{max}$, low $SIAT_{sdev}$, high SAT, and low $SPER_{avg}$ for Constant Bit Rate (CBR) type traffic, while for Variable Bit Rate (VBR) type traffic, low $SOWL_{max}$, high SAT and low $SPER_{avg}$ values are optimal.

2.6 Multicast and QoS Support

2.6.1 Multicast Protocols and Group Membership Maintenance

Multicasting support in Internet Protocol (IP) has been a major driver for the development of real-time multimedia applications over IP-based networks. Several Multicast Routing algorithms have been developed, including Distance Vector Multicast Routing Protocol (DVMRP), Multicast Open Shortest Path First (MOSPF) Routing and Protocol Independent Multicasting (PIM), have been developed to address the issue of providing multicast support in the Internet (Stallings, 1998, Ramalho, 2000).

In order to provide nodes wishing to participate in a given multicast session, the Internet Group Management Protocol (IGMP) has been developed Nodes are able to join any active multicast group by responding for IGMP *Queries* initiated by the multicast router (Deering, 1989). IGMP *Reports* are used to maintain the multicast session, failing which the multicast router would inhibit multicast packet forwarding for subnets without any active multicast nodes. Periodic IGMP *Queries* would be issued into the subnet to probe for group members and refresh the group membership status. IGMPv2 and IGMPv3 protocol extensions provides additional support for node initiated *Leave* requests and source filtering to improve the performance of IGMP (Fenner, 1997, Cain et al., 2002).

2.6.2 QoS support in Wired Multicast Networks

Diot, Dabbous and Crowcroft (1997) provided a comprehensive survey of existing multicast protocols, as well as the application of QoS towards multicast transmission. It was noted that *"QoS constrained multicast route design is an NP-complete problem."* Multicast QoS controls could be source-based, sink-based, or a combination of both. The paper outlined three approaches towards multicast QoS implementation:

- Multicast source (Sender) defines QoS requirements, and sinks (Receivers) must accept the given requirements
- Multicast source (Sender) negotiates QoS requirements to be the minimum of each sink (Receiver) QoS
- Multicast source (Sender) sends at the maximum QoS negotiated with the group of sinks (Receivers), and each receiver performs receiver-based QoS control. Network support for QoS is required.

Nonetheless, the authors did not differentiate between QoS *Adaptation*, which refers to whether the QoS parameters remained fixed or variable throughout the lifetime of the multicast session, and QoS *Profiles*, which refers to the mechanism for determining the appropriate QoS parameters for negotiated or adaptive QoS.

In Wang & Hou (2000) and Striegel & Manimaran (2002), the issue of wired multicast QoS support was addressed as problems in deriving a QoS-satisfying multicast tree structure. The assumption was that the multicast tree experiences inherent network and resource bottlenecks and consequently tree construction and maintenance algorithms must account for required QoS constraints. In addition, the issue of source-driven QoS vs. sink-driven QoS was discussed.

The QoSMIC protocol (Yan, Faloutsos & Banerjea, 2002) was designed to support multiple-path selection to satisfy QoS requirements from groups with dynamic memberships. QMRP (Chen, Nahrstedt & Shavitt, 2000) attempted to improve on the overheads inherent in QoSMIC to create better QoS-compliant multicast trees. QUASIMODO (Bianchi et al., 2003) extended multicast protocols to utilize DiffServ for providing QoS support.

2.6.3 Network Bottlenecks for Wireless Multicast

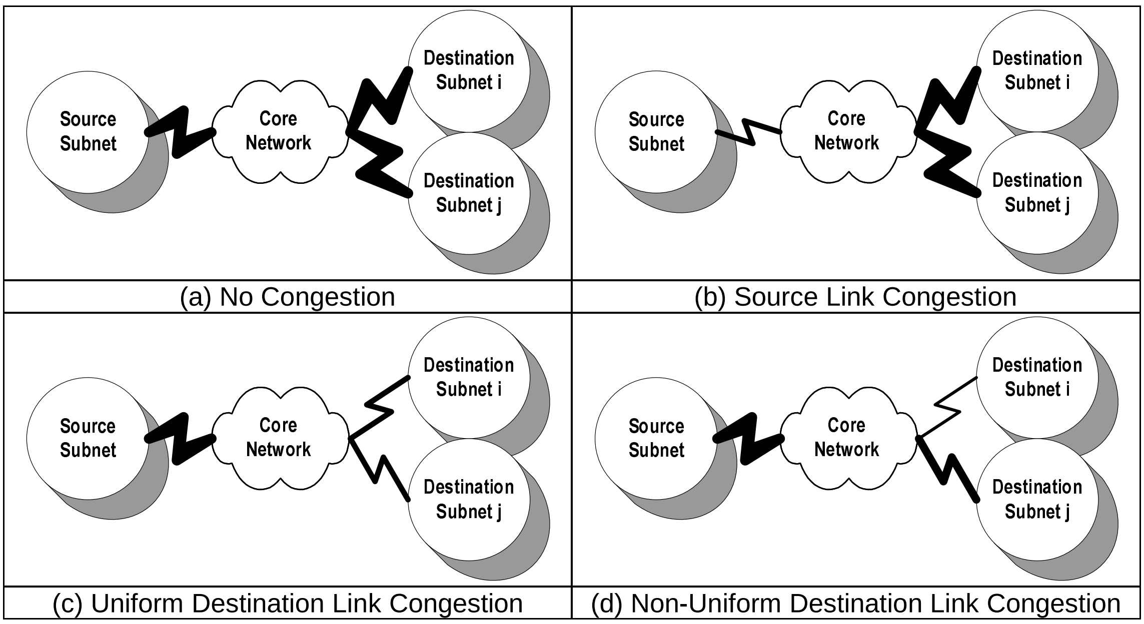
In Gossain, Cordeiro & Agrawal (2002), wireless multicast was discussed in the context of adapting mobile wireless nodes to existing wired multicast protocols such as MOSPF. Two approaches were proposed by IETF: *Remote Subscription* and *Bidirectional Tunneling* via Mobile IP extensions. *Remote Subscription* has inherently lower overheads since each mobile node subscribes to the multicast group directly as it roams into a new cell. In contrast, *Bidirectional Tunneling* requires the intervention of the Mobile IP Home Agent to tunnel multicast packets via unicast to the mobile node.

The *Remote Subscription* technique is able to better utilize wireless bandwidth since it uses native multicast, at the expense of higher group maintenance overheads. Native multicast can be carried efficiently over dedicated multicast channels shared among all multicast sessions (Chapter 2.3.4). Consequently, *Remote Subscription* is preferred over *Bidirectional Tunneling* for bandwidth constrained wireless links.

The multicast trees supporting cell-based wireless multicast networks therefore has bandwidth constrained wireless networks as leaf nodes in the multicast tree, while in comparison, the wired core network forming the trunk of the multicast tree has no bandwidth constraints. Consequently, the problem of QoS-constrained multicast tree creation and maintenance for the wired core network is well known and does not affect the wireless network environment under study significantly. The research is therefore focused on defining the QoS constraints in the cell-based wireless leaf networks.

2.6.4 QoS Constraints for Cell-based Wireless Multicast Networks

Given that all nodes participating in the multimedia multicast session are mobile wireless nodes, and the intermediate wired-infrastructure has sufficient bandwidth provisioning to forward all defined multicast streams without affecting their *in-transit* QoS, we would only need to consider the network conditions in the respective wireless cells (Figure 2.5).



(a) No Congestion | (b) Source Link Congestion | (c) Uniform Destination Link Congestion | (d) Non-Uniform Destination Link Congestion

Figure 2.5: Network Bottlenecks for Wireless Multicast Traffic

We can classify the various wireless QoS scenarios as follows:

- *No Congestion*: Both source and destination cells have sufficient resources to meet QoS requirements
- *Source Congestion*: Available Source Cell Bandwidth is less than all available destination cell bandwidths. QoS performance is Source constrained
- *Uniform Destination Congestion*: Available Source Cell Bandwidth is able to meet QoS requirements, while all destination cells are uniformly congested. QoS performance is Destination constrained
- *Non-Uniform Destination Congestion*: This is the general case. Available Source Cell Bandwidth is able to meet QoS requirements, while each destination cell has different congestion characteristics. QoS performance is subject to tradeoffs

Bandwidths are asymmetrical on the upstream and downstream, since upstream transmission is assumed to occur over a shared channel, while downstream traffic is carried over the dedicated multicast channel. Congestion measurements would therefore be different for source and destination bandwidths for a given cell.

2.7 Mobility and QoS support

A comprehensive survey of QoS issues for mobile environments is given in Chalmers & Sloman (1999). In order to provide seamless QoS support for mobile terminals, handoffs must be addressed in a way that avoids introducing delays and excessive packet loss above that provided for by the QoS guarantee. It is assumed that the new Base is able to maintain the required bandwidth requested by the QoS guarantee; otherwise the connection would be dropped on handoff. The multi-layered QoS guarantee would be helpful in such a scenario in avoiding a total loss of service. The main issue is then that of overcoming the delay in reestablishing a connection to the mobile terminal, as well as the loss of packets incurred while the handoff is taking place. Acharya et al. (1997) and Cheng et al. (1997) discuss several handoff strategies that could be implemented in a cellular hybrid PCS/ATM environment.

In order to avoid QoS violations during handoff, pre-provisioning of the data would be one method of avoiding loss and delays, assuming that actual handoff interval is minimal with respect to the QoS requirements (i.e., the actual switching time from one Base to another is less than one frame-time according to the delay and data rate requirements of the QoS contract). Pre-provisioning is a technique whereby the data stream is replicated to the AP once the handoff process is imminent and the new AP has been identified. Once the mobile terminal switches to the new AP, it would continue receiving data from the new stream, while the connection terminating at the previous AP is torn down. The issue here is one of re-synchronization, to determine which frame in the previous stream has been received and when the next frame is transmitted in the new connection. Marker cells for WATM have been proposed to address this issue (Acharya et al., 1997).

Full QoS support during handoffs can only be achieved with appropriate support from *confederated* Bases. However, with the assumption of having minimal coupling between networks (i.e., autonomous Bases), only some of the mitigation techniques could be employed. For example, data pre-provisioning would be possible by sending IGMP join messages to the new Base before leaving the existing Base. Several fast handoff techniques were used in MarconiNet (Dutta & Schulzrinne, 2004) for an overlay mobile content distribution network. Application layer signaling using Real-Time Transport Control Protocol (RTCP) is used to trigger Joins and Leaves from a given cell, while data pre-provisioning via Multicast Proxy agents would activate the necessary multicast streams prior to the mobile node entering the new cell.

2.8 Other Factors Affecting Wireless Multicast Support

2.8.1 Traffic Shaping

Traffic Shaping is often used at the Network and/or Data link layer to regulate the burstiness of certain traffic sources. By using Active Queue Management and Traffic Shaping algorithms in the Data Link (MAC) Layer, *Proportional Differentiation* QoS techniques is able to perform flow prioritization and regulation on outgoing multicast flows. The *Exponential rule* Modified Largest Weighted Delay First (M-LWDF) scheduling algorithm has been proposed to implement QoS support for multiple streams over a shared channel (Andrews, 2000, 2001, Shakkottai & Stolyar, 2001). This algorithm has been proven to be throughput optimal and is therefore inherently suitable for MAC-level QoS support over wireless channels. The M-LWDF algorithm can be classified as a *proportional differentiation* technique since it allocates available bandwidth among competing users based on packet delay requirements. Together with the use of the DuReTA RF architecture, we can provide QoS support for wireless multicast streams in an effective manner using such a *proportional differentiation* technique directly for downstream traffic. If necessary, M-LWDF can also be used to prioritize upstream traffic by modifying the polling order for PCF mode. In addition, M-LWDF can be used to prioritize different flows to allow high priority multicast streams better QoS than for lower priority multicast sessions.

To implement *proportional differentiation*, we set the non-compliance probability **Pr(·)** of a flow using M-LWDF as given in Shakkottai & Stolyar (2001):

$$\Pr(W_i > T_i) \le \delta_i \qquad \text{........ Equation 1}$$

where W_i is the instantaneous delay, T_i is the maximum delay, and δ_i is the non compliance probability for a given flow *i*. From these QoS requirements, we can schedule the appropriate flow for transmission at the UDL transmission queue using the *Exponential rule* M-LWDF algorithm (Equation 2).

Given a shared link of bandwidth μ carrying N flows, where a_i and γ_i are adjustable weights, $\mu_i(t)$ is the instantaneous available link bandwidth for flow *i*, and $W_i(t)$ is the in-queue wait time experienced by the Head of Line (HOL) packet for flow *i*, we can determine the scheduled flow *j* at time *t* using the *Exponential rule* M-LWDF algorithm via Shakkottai & Stolyar (2001):

$$j = \arg\max_i \gamma_i \mu_i(t) \exp\left(\frac{a_i W_i(t) - \overline{aW}}{1+\sqrt{\overline{aW}}}\right),$$
$$where\ \overline{aW} = \frac{1}{N}\sum_i a_i W_i(t),\ \gamma_i > 0,\ a_i > 0,\ and\ i = 1,2,...N$$

.... Equation 2

Good choices for a_i and γ_i are $a_i = \frac{-\log(\delta_i)}{T_i}$ and $\gamma_i = \frac{a_i}{\mu_i}$ (Shakkottai & Stolyar, 2001). If the flows were equal priority, $\mu_i(t) = \frac{\mu}{N}$.

For the rest of this discussion, the symbols used for defining the proportional QoS requirement of each flow will be represented as the following variables: $T_i \equiv D_{max}$, $\delta_i \equiv p_{max}$. A *Proportional Differentiation*-enabled flow would be specified using its Maximum Delay (D_{max}), Minimum bandwidth (BW_{min}), and Maximum non-compliance probability (p_{max}). Higher priority streams have lower p_{max}. Each multicast group is assigned a separate MAC-layer queue, and various MAC queues compete for access to the channel based on the bandwidth availability and prioritization using the *Exponential rule* M-LWDF algorithm. Nonetheless, MAC layer approaches such as M-LWDF are local solutions and only address QoS constraints within a cell. It is not sufficient to utilize MAC layer approaches to address end-to-end QoS issues arising from network congestion in other cells. Network Layer approaches are necessary for addressing system-wide QoS issues.

2.8.2 Wireless Multicast Group Maintenance

In analyzing wireless multicast streams, we consider the perspective of a single source node servicing multiple destination nodes (one-to-many). A many-to-many multicast session would be classified as a superposition of multiple one-to-many sources.

Wireless Multicast group maintenance can be performed using the IGMP protocol defined for wired multicast transmission. However, since the nodes under consideration are mobile, multicast groups are much more dynamic and hence the IGMP protocol overheads would have greater impact on wireless multicast groups.

Certain changes to the standard IGMP state protocol need to be made in order to reduce the joining latency (Xylomenos & Polyzos, 1997, Dutta & Schulzrinne, 2004). Explicit node initiated *Join* requests are implemented in order to reduce the time to reacquire the multicast stream after roaming.

2.8.3 Transport Layer Issues

Under the category of transport layer issues, multi-layered multimedia transmission approaches are used to adapt the multicast session to varying wireless network performance (Roca, 2000, Turletti, Parisis & Bolot, 1997, Wang & Hou, 2000). Buffering, layered quality adaptation, and TCP-friendly congestion control mechanisms (Clerget, 1999, Rejaie, Handley & Estrin, 2000) have also been proposed to address transport layer QoS requirements. Explicit Congestion Notification (ECN) has also been used to regulate data flows in the Internet (Ramakrishnan, 2001).

The Motion Pictures Expert Group (MPEG) defined MPEG-4 System Layer Model implements object-based multimedia substreams (FlexMux Streams) that can be prioritized and transmitted as multiplexed data (TransMux Streams) over standard transport mechanisms given in the Delivery Multimedia Integration Framework (DMIF) specifications (Koenen, 2002). The division of multimedia data into prioritized substreams simplifies the delivery of such real-time data over wireless links. In addition, MPEG-4 defines profiles for low-bit-rate transport in error-prone environments (Koenen, 2002).

Transport layer QoS approaches will not be considered in this research due to the fact that they're unicast flow oriented. Instead, extensions to IGMP to add QoS Feedback information generated by multicast routers provide an equivalent functionality, by performing flow adaptation at the application layer.

2.8.4 Application Layer Issues

2.8.4.1 Application Profiles

Several types of multicast applications are commonly defined. Media streaming is naturally suited for unidirectional multicast transmission. In addition, interactive multicast applications such as multi-way multimedia conferencing will be modeled as multiple simultaneous unidirectional multicast transmissions. Such applications both receive and transmit multimedia data to other nodes via multicast traffic.

2.9 Chapter Summary

Previous research in characteristics of the wireless environment, evolution of wireless communications, Quality of Service paradigms, QoS metrics, Multicasting, QoS for Wired and Wireless Multicasting, and Mobility Issues were surveyed. Notable results were found in the development of *Proportional Differentiation* QoS techniques, which was inherently suitable for wireless networks. Subsequently, the open issues for implementing multicast QoS in wireless networks were defined and explored.

This background survey led to the definition of a comprehensive wireless multicast QoS framework in Chapter 3 that aims to enable wireless multicast applications to achieve given QoS objectives despite the variability of the wireless environment.

CHAPTER 3 WIRELESS MULTICAST QOS FRAMEWORK

3.1 Framework Requirements

The proposed Wireless Multicast QoS Framework (WMQF) to address the needs of wireless multicast applications such as real-time multimedia delivery to a *mobile* multicast population is given below. The discussion is limited to mobile nodes; any other fixed nodes are assumed to have much higher available bandwidth since they are directly connected to the core network.

The QoS provisioning in this wireless network architecture assumes the use of a DuReTA (Dual Receiver Transceiver Architecture) physical layer for the mobile nodes. Mobile nodes are termed iMAP (*integrated Multimedia Appliance with Prioritization*) in the WMQF environment (Figure 3.1). The bandwidth and BER performance for the wireless connections are in the order of 2 Mbps and 10^{-3} respectively, suitable for real-time multimedia applications.

iMAPs connect to QMACCs (*QoS Multimedia Adaptive Cell Controller*) which provide access coverage, as well as facilitate roaming between different cells within the WMQF. QMACCs are autonomous Bases enhanced with *Proportional QoS* support for Multicast Routing. Mobile nodes are able to roam from one cell to another while still maintaining their membership in the given multicast population under consideration. The core network interconnecting the various QMACCs are assumed to be QoS-unconstrained, where sufficient resources exist to carry all offered traffic among the QMACCs and external network resources. Hence, the core network is able to provision required QoS profiles (service class) for any given multicast application executing among the mobile wireless population. Examples of core network technologies include ATM and DiffServ enabled Gigabit Ethernet.

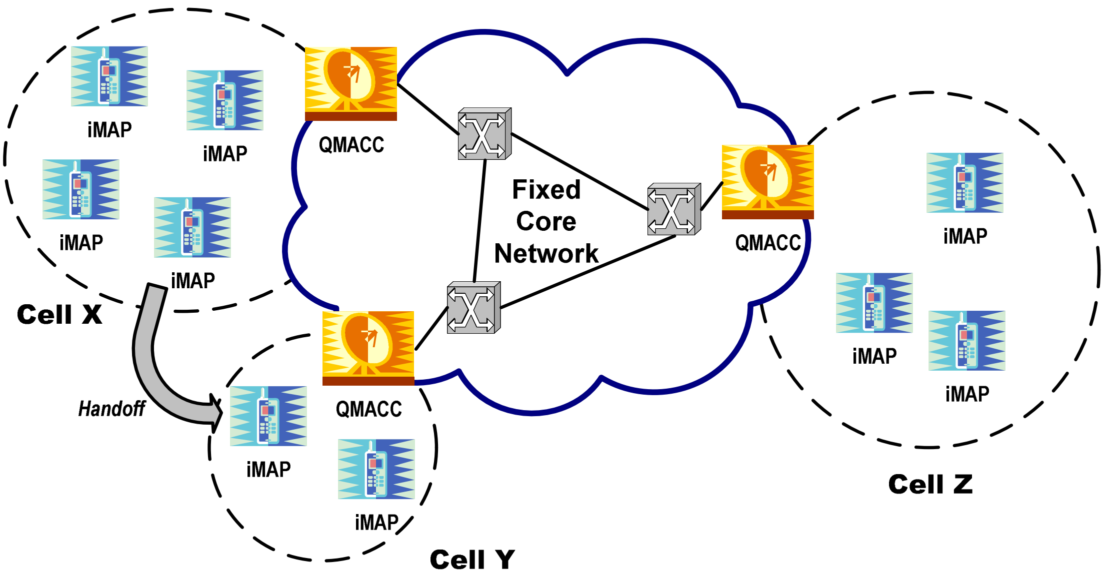


Figure 3.1: System Architecture for Wireless QoS Multicast Framework

Routing among different cells is occurs via the QMACC that is connected to the core network. Multicast routing protocols, such as MOSPF or PIM running in the QMACC ensure that group membership management is maintained regardless of the location of a particular iMAP via the Internet Group Management Protocol (IGMP).

While the WMQF can be applied towards existing and emerging wireless air-interfaces such as 802.11 and WCDMA (3G), the focus of this research is to model the Physical and Data Link layers using 802.11-based air interfaces with minimal or no QoS support. QoS capabilities will be implemented in the Application Layer instead.

Since the number of co-located channels required to support a roaming arrangement dictates the use of three distinct frequency plans to avoid adjacent channel interference, where each frequency plan utilizes two channels (one for bi-directional link, the other for the unidirectional link), therefore *Frequency Hopping Spread Spectrum* (FHSS) 802.11 is the more suitable air interface for supporting the frequency reuse plan for WMQF. Nonetheless, *Direct Sequence Spread Spectrum* (DSSS) 802.11b air interfaces could still be used with the limitation where the effective link bandwidth is reduced due to co-channel interference. However, since WMQF model link capacity as being 2 Mbps, this reduction in effective link bandwidth would not be detrimental to the frequency reuse scheme if WMQF were to be deployed using 802.11b technology, since 802.11b supports a nominal link bandwidth of 11 Mbps.

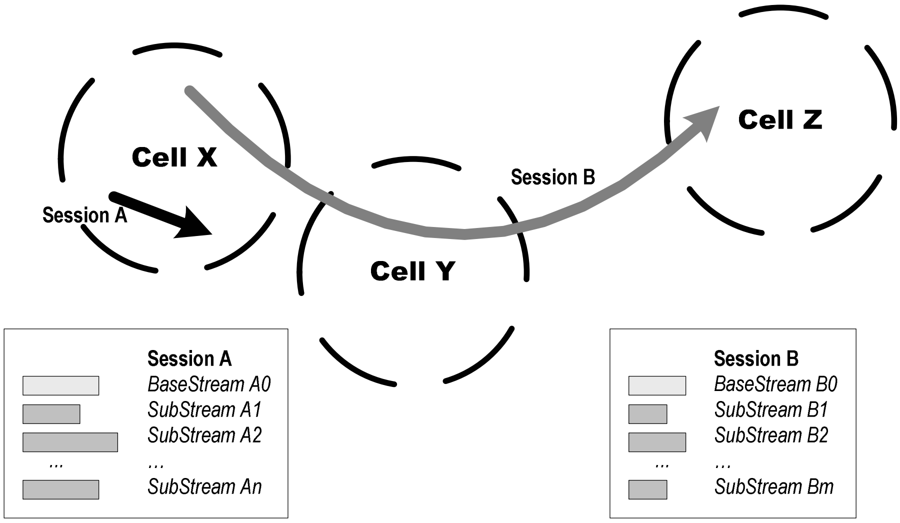


Figure 3.2: Intracell (Session A) vs. Intercell (Session B) Sessions

Several multicast sessions, identified via unique multicast addresses, carry multimedia traffic for different applications (Figure 3.2). iMAPs participate in one or more multicast session at any given time. Therefore, a given cell can support a number of multicast sessions, while a multicast session can encompass one or more cells. In addition, each multicast session is multilayered, comprising one base stream and additional substreams for carrying multimedia data with different priorities for the given session. The number of active substreams is adjusted based on the current network QoS.

There are some architectural similarities between WMQF and MarconiNet (Dutta & Schulzrinne, 2004), namely the use of application layer QoS mechanisms and handoff optimization via enhancements to the IGMP protocol. However, significant differences exist in WMQF, for example, dedicated multicast channels using DuReTA transceivers are used to enhance QoS delivery, and the use of QMACCs which perform QoS signaling, QoS monitoring and multicast routing to support mobile initiated multicast streams. Nonetheless, MarconiNet is focused on the detailed operation of the protocols required for supporting multicast applications, while WMQF focuses on optimization of system-wide bandwidth utilization via suitable QoS Adaptation schemes.

3.2 System Objective

3.2.1 Multicast QoS Adaptation Algorithms (MQAA)

The system objectives of the Wireless Multicast QoS Framework (WMQF) is to specify suitable Multicast QoS Adaptation Algorithms (MQAA) to address the following scenario, namely application layer QoS support for multiple multilayered multimedia streams competing for transmission bandwidth over dedicated wireless multicast links. The MQAAs are required to meet specified QoS Targets using appropriate Multicast QoS Profiles to process QoS Feedback messages from the network; subjected to the system constraints of high Bit Error Rates and Fading (Rician and Rayleigh). This is summarized in Figure 3.3.

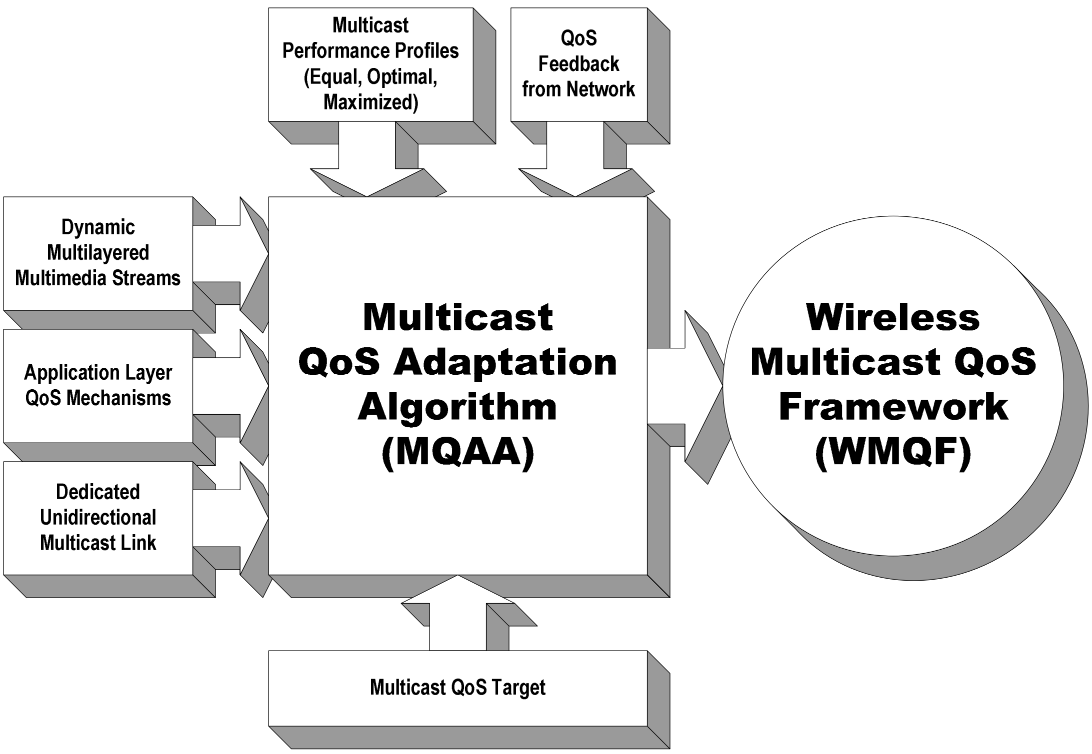


Figure 3.3: WMQF System Objective

3.2.2 Overall System Bandwidth Utilization (OSBU)

The MQAAs have a secondary objective, to attempt to maximize Overall System Bandwidth Utilization (OSBU). OSBU is defined as the achievable UDL throughput for all active multicast sessions within the system. Nonetheless, not all multicast sessions are carried over all UDLs due to randomized distribution of iMAPs within the network. Consequently, OSBU can only be estimated using the System Application Throughput (SAT) metric. However, it is possible to obtain a reasonably accurate estimate for OSBU for the case where all iMAPs subscribe to all active multicast sessions. The formula for OSBU is given in Equation 3.

$$OSBU = SAT \times g, \text{ where } g : no.\, of \;\; active\ multicast\ streams\ in\ system$$
$$and\ OSBU \leq BW_{UDL}, \text{ where } BW_{UDL} : UDL\ Link\ Bandwidth$$

.... Equation 3

3.3 System Architecture Overview

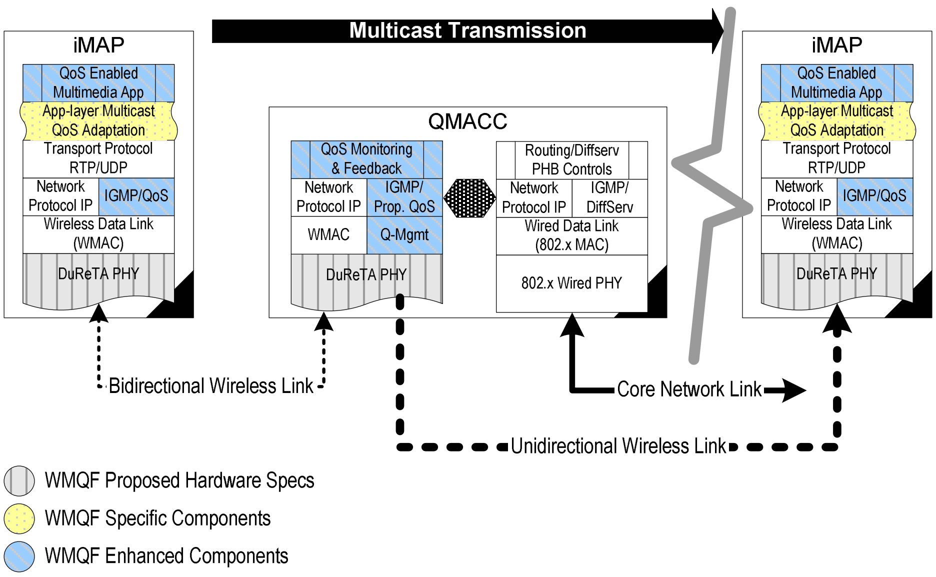


Figure 3.4: Details of WMQF Network Layers and Enhancements

The WMQF architecture (Figure 3.4) utilizes closed-loop QoS signaling among iMAPs as well as QMACCs using IGMP protocol extensions. However, within the core network, DiffServ provides the necessary QoS support, since DiffServ is much more scalable and easier to deploy (Bernet, 2000).

3.4 Framework Architecture

3.4.1 Physical Network Model

The WMQF environment is an Infrastructure-based wireless network, assumed to have complete signal coverage for mobile nodes roaming within the entire network. The network air interface is based on the IEEE 802.11 standard which provides minimal QoS support for shared communications channels. For simplicity, the Bases in the WMQF network are assumed to be in an idealized uniform grid formation with some overlapping coverage areas at the cell edges. In addition, the environment is assumed to be open terrain so fading would primarily come from Rayleigh components and not multipath (Rician) components. Two independent sets of non-interfering channels are implemented, one for Bidirectional Link (BDL) traffic, and the other for Unidirectional Link (UDL) traffic. Consequently, a standard three-color reuse pattern (Steele & Hanzo, 1999) will be implemented for each set of channels to ensure non-interference among adjacent cells.

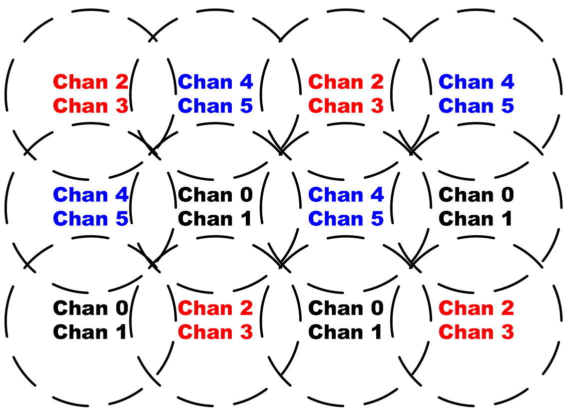


Figure 3.5: Idealized Uniform Grid Infrastructure-based Wireless Network

From Figure 3.5, we can assume that even numbered channels belong to the BDL set, while the odd numbered channels belong to the UDL set. A total of six distinct RF modulation parameters are needed to support the entire network.

Each cell controlled by a QMACC will be equipped with two transceivers. The first transceiver is used to control the UDL for sending downstream multicast traffic, while the second transceiver controls the BDL for upstream mobile access as well as downstream unicast traffic to the mobiles. The BDL and UDL interfaces on the QMACC are logically independent of each other. Consequently, we refer to the QMACC as being a BDL QMACC or a UDL QMACC depending on which link is used.

Mobiles have two transceivers based on the DuReTA design (Figure 3.6), where one of the transceiver is a Receiver-only device for accessing the UDL transmissions, while the other transceiver is a normal Transmit-Receive transceiver. Mobiles have prior knowledge of the frequency and channel assignments for BDLs and UDLs, it is not possible for mobiles to attempt to transmit on the UDL.

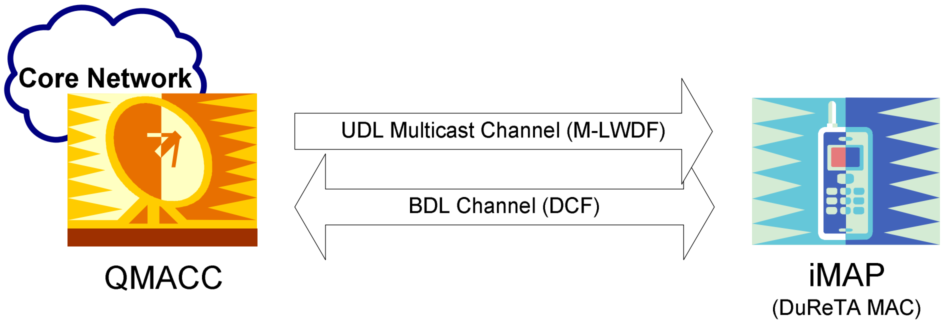


Figure 3.6: DuReTA implementing Unidirectional and Bidirectional support

3.4.2 Data Link Layer Model

3.4.2.1 QMACC MAC Layer Features

In WMQF, QMACCs are assumed to be autonomous Bases and do not provide roaming support. Hence, mobility and handoffs are the sole responsibility of the mobile iMAPs. Nonetheless, the QMACCs have a significant role to play in Multicast QoS support. The UDL QMACC used for multicast transmission and is equipped as a multicast router with *Proportional QoS* Active Queue Management capabilities based on the M-LWDF algorithm. However, the BDL QMACC is assumed to have little or no QoS support, since it implements the standard 802.11 PCF/DCF MAC protocol (Note:

only DCF mode is implemented for the simulation model, the achievable throughput would be higher using PCF mode).

3.4.2.2 iMAP MAC Layer Features

iMAPs are assumed to be moving with a given velocity and direction within the geographical area covered by the wireless network. Each interface in the iMAP operates semi-independently of each other, and can potentially associate with different QMACCs within its range. The iMAP MAC Layer implements three Mobile Roaming States to control its network access as it moves from one cell to another.

Upon activation, the mobile must first undergo a ***Channel Acquisition*** phase where it scans all available channels in the give channel set in order to obtain a list of suitable QMACCs. Subsequently, mobiles perform ***Base Association*** to the QMACC with the highest received power level among the QMACCs in range. If subsequent data transmission indicates that the received power level from the associated QMACC is less than the power received from another QMACC, the mobile will initiate ***Base Roaming*** to this target QMACC with the better signal quality. The mobile first disassociates from the current QMACC and then associates to the target QMACC.

For the BDL interface, Base Association and Roaming involves exchanging suitable Layer 2 station management protocol messages between the mobile and the BDL QMACC. For example, in 802.11, ASSOCIATE and DISASSOCIATE messages would be used for join and leave operations.

Layer 2 signaling is not used for the UDL. Instead, the mobile UDL transceiver will automatically associate and roam to the UDL QMACC with the best signal quality. Multicast membership management is performed using Layer 3 IGMP signaling sent via the BDL interface to the UDL QMACC to perform multicast group Join and Leave operations. During Base Roaming, IGMP signaling will be transmitted via the BDL interface to the target UDL QMACC prior to disassociation to activate all current multicast sessions. After the mobile UDL transceiver has associated to the target UDL QMACC, it will transmit IGMP signaling to the original UDL QMACC via the BDL interface to indicate that it no longer require the multicast streaming in the previous cell. This helps to reduce the multicast resynchronization time caused by the roaming event.

It is conceivable that both the UDL and BDL interfaces perform Base Roaming at the same time. In such a scenario, link reestablishment delay is unavoidable since the BDL needs to reestablish its connection before IGMP messages can be forwarded to the new UDL QMACC.

3.4.2.3 iMAP Roaming State Transitions

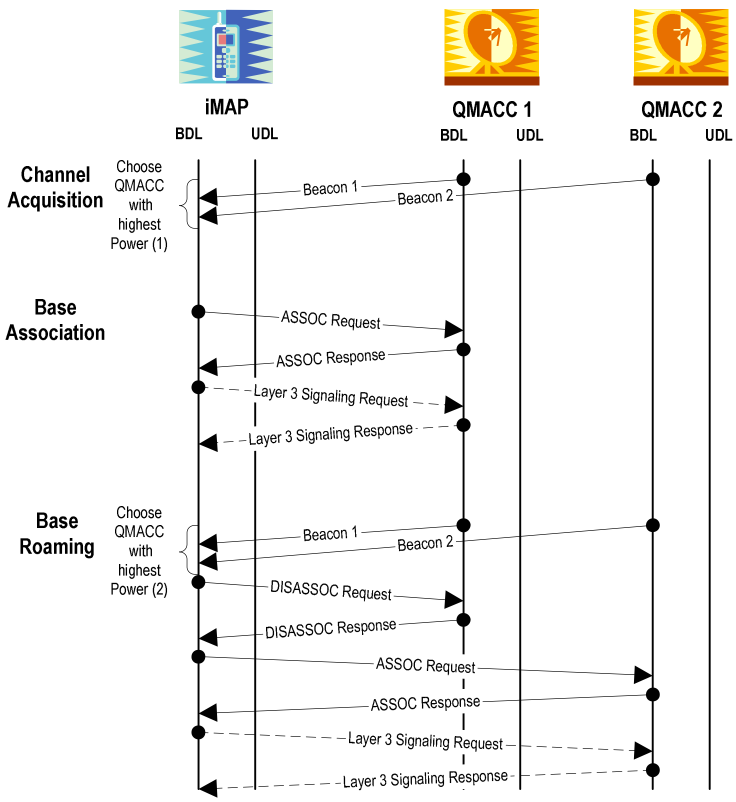


Figure 3.7: BDL Mobile Roaming States

An iMAP initiates *Channel Acquisition* upon entering the wireless network. In this mode, Beacons from different QMACCs within range are collected and their Receive power compared to determine which QMACC is preferred. From the example in Figure

3.7 which describes the behavior for the Bidirectional Link (BDL), QMACC 1 has higher power so the iMAP attempts to associate to it. The *Base Association* phase takes place using normal 802.11 signaling mechanisms. As the iMAP roams within the network, it will eventually experience decreasing Receive Power reception from its associated QMACC (QMACC 1). The iMAP then performs *Base Roaming* where it disassociates from its current QMACC and reassociates to the new QMACC with higher receive power (QMACC 2). Upon successful association to the new QMACC, upper layer signaling will take place to reestablish any active connections.

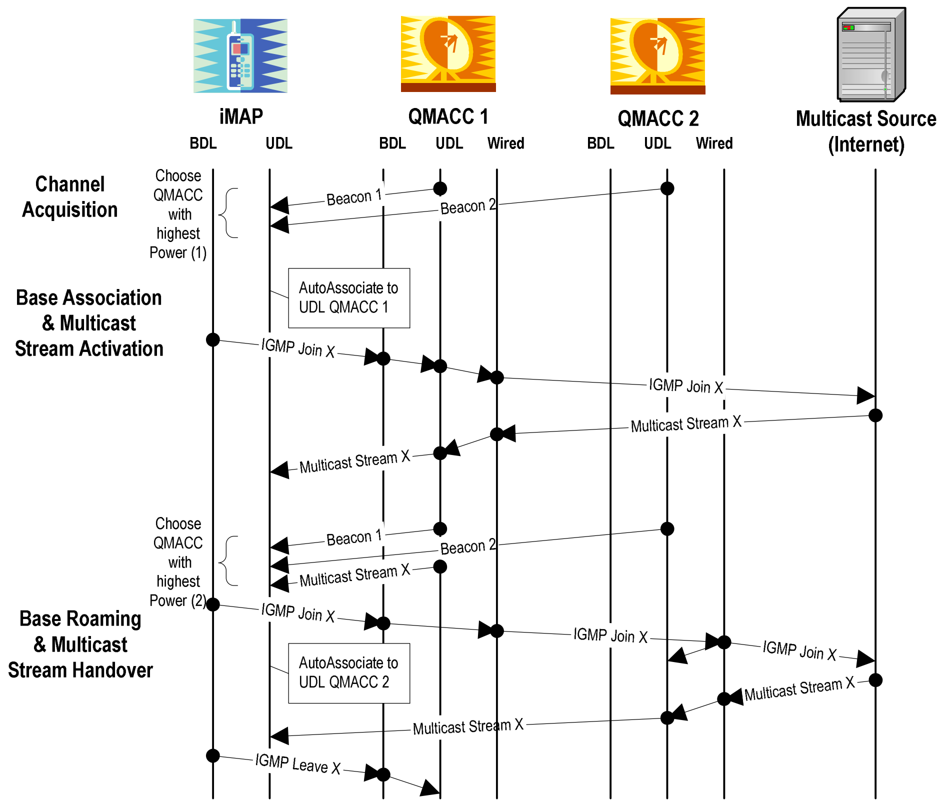


Figure 3.8: UDL Mobile Roaming States (assume BDL remains unchanged)

Similar transitions occur for the Unidirectional Link (UDL), except that iMAPs automatically associate to the UDL QMACC without performing any 802.11 signaling exchange (Figure 3.8). Instead, IGMP signaling will be used to inform the QMACC of *Joins* and *Leaves* that occur as the iMAP roams from one QMACC to another. The iMAP will attempt to Join the new QMACC (QMACC2) before switching QMACCS and sending the Leave message to its old QMACC (QMACC 1) to minimize disruption to ongoing multicast transmission streams.

3.4.3 Network Layer Model

3.4.3.1 Intracell Multicasting

Intracell Multicasting primarily involves Data Link level multicasting, since it is assumed that all nodes within a given cell are able to communicate effectively with the QMACC. Consequently, once a multicast stream has been established within a given cell, the QoS parameters are influenced only by the local conditions within the cell. This simplifies the process of determining which QoS target the source should use, since all nodes share the same UDL QMACC link for receiving the given multicast stream.

3.4.3.2 Intercell Multicasting

Intercell Multicasting involves the dissemination of multicast group information in order to construct a multicast tree spanning active multicast nodes in the network. Each QMACC acts as a multicast forwarding routing connected to the wired network (Internet). We assume that standard multicast tree configuration protocols, such as MOSPF and PIM would be utilized to configure the optimal multicast spanning tree for a given multicast session interconnecting various QMACCs, and that the wired network has sufficient QoS capacity to handle the multicast traffic originating from the wireless iMAPs. IGMP is used by the iMAPs to signal its membership in a given multicast session to its associated QMACC. Multicast traffic is forwarded into a given cell only if iMAPs within that cell has requested for the particular multicast stream. As multiple UDL QMACCs are involved, the various network congestion scenarios make the selection of suitable QoS target for the source vital to avoid transmitting excess data that could not be delivered to the respective receiving nodes.

3.4.3.3 IGMP and QoS signaling

While QoS feedback from actual receivers would provide the highest level of accuracy, it is not desirable for each receiver to transmit its QoS status as it would lead to inbound message implosion, as well as significant wireless channel signaling overhead. Consequently, the QoS estimate is generated by each end-QMACC in order to address the above concerns.

There is a two-level QoS control architecture. The first level is implemented by each UDL QMACC as an Active Queue Management scheme for the link. This ensures that competing multicast traffic over the UDL link experiences Proportional Differentiation based on the current QoS requirements. The second level is the application layer closed-loop end-to-end dynamic QoS control for adjusting the QoS requirements of various multicast flows in real-time. This is achieved using QoS Feedback messages sent by UDL QMACCs to the Multicast group. The metrics used for QoS control is of necessity less comprehensive and precise, but at the same time, more efficient. The IGMP and QoS signaling is illustrated in Figure 3.9.

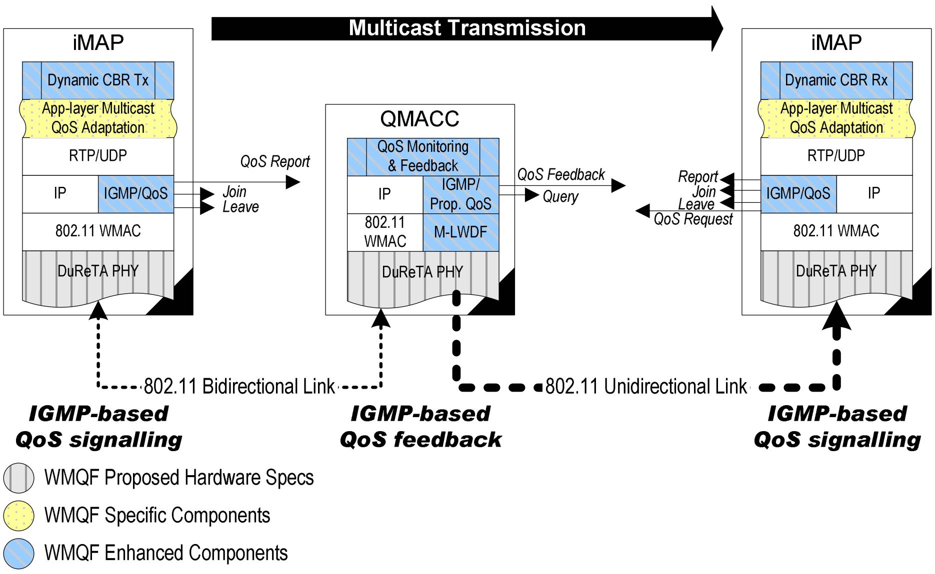


Figure 3.9: IGMP and QoS Signaling in WMQF

An end-to-end multicast QoS monitoring framework is implemented at the application layer, based on the multicast metrics defined in Chapters 2.5.4 and 2.5.5. This enables offline analysis and evaluation of the performance of various MQAA algorithms.

Each QMACC is responsible for periodically sending *QoS Feedback* packets to the multicast group so that the multicast source is able to perform application layer source QoS adaptation. The QoS Feedback message comprises of queue length of the given multicast group (Q_m), total queue length of all multicast queues (Q_a), total buffer space of allocated queues (B_a), and average delay of multicast packets in the group (D_m). These parameters can then be used by the QoS Estimators to derive the *Delay (D')* and *Available Bandwidth (BW')* estimates for a cell as normalized values (Table 3.1). Estimated *Delay (D')* is bounded by the base stream delay requirement D_{max}. The ratio of the multicast group queue length Q_m to the total queue length Q_a measures the contribution of a given flow towards the total traffic load, while the ratio of the Average Packet Delay D_m to maximum base stream delay D_{max} provides us a measure of the tolerable delay for a given multicast flow. The product of these two ratios gives us a normalized value for Estimated *Delay (D'). Available Bandwidth (BW')* is estimated using the ratio of the amount of free buffer space ($B_a - Q_a$) to total buffer space B_a.

Table 3.1: QMACC QoS Estimation Parameters Definition

QoS Est. Parameters	Definition	Purpose
Q_m	Multicast Flow Queue Length	Number of pending packets in flow for transmission
Q_a	Total MAC Queue Length	Total number of pending packets for transmission
B_a	Total MAC Buffer space	Total buffer space
D_m	$D_m = \frac{1}{Q_m}\sum_{j=1}^{Q_m}(T - T_j)$, where $T \geq T_j$	Average Packet Delay for pending packets in multicast flow, where T is the current time and T_j is the packet creation time
D'	$D' = \left\lceil \frac{Q_m}{Q_a} \times \frac{D_m}{D_{\max}} \right\rceil_0^1$, where $Q_m \leq Q_a \leq B_a$	Delay estimate (%), where D_{max} is the QoS Maximum Delay parameter; value of D' is truncated to 1: i.e., D' $\in$ [0,1]
BW'	$BW' = 1 - \frac{Q_a}{B_a}$, where $Q_a \leq B_a$	Available Bandwidth estimate (%), where BW' $\in$ [0,1]

3.4.4 Transport Layer Model

Multimedia multicast streams are carried via Real Time Protocol (RTP) packets based on Unreliable Datagram Protocol (UDP). Consequently, there is no inherent flow control available in the Transport Layer. In WMQF, Transport Layer QoS Adaptation is not implemented. Instead, it is approximated via the use of a Dynamic Constant Bit Rate (DynCBR) application model which changes its bandwidth and QoS requirements based on the number of active substreams.

3.4.5 Application Layer Model

3.4.5.1 Multicast QoS Signaling

Application layer issues focus primarily on end-to-end QoS adaptation. The adaptation process can be summarized into various Multicast QoS Profiles that are adopted by respective applications to satisfy the requirements of the users. While the ITU-T standards for QoS (Seitz, 2003) proposes various QoS Classes for six different types of unicast traffic, from real-time highly interactive traffic to default best effort Internet traffic, the focus of our discussion on multicast QoS will be on interactive real-time multimedia traffic for simplicity.

The *Dynamic Constant Bit Rate* (*DynCBR*) multicast source generates multicast packets to a group of receivers. It sends an IGMP Join to its UDL QMACC and specifies an initial QoS requirement (QoS target) to its associated UDL QMACC via a *QoS Report*. The UDL QMACC registers the multicast group, and performs multicast forwarding to interested receivers under its control and the control of other QMACCs. The multicast source receives its own multicast transmission in order to receive *QoS Feedback* messages sent periodically by participating QMACCs within the network.

The *Dynamic CBR* (*DynCBR*) multicast receiver act as sinks. It sends an IGMP Join to its associated QMACC, and performs a *QoS Request* to enable the QMACC to allocated required multicast bandwidth in the Unidirectional Link (UDL) for the associated group. Bandwidth allocation and packet scheduling in the UDL is performed using the M-LWDF algorithm. Periodically, the QMACC sends *QoS Feedback* to the multicast source based on the status of its UDL queues. These *QoS Feedback* messages are processed by the QoS Adaptation module in the multicast source, which causes the multicast application to adjust the target QoS parameters. After the QoS target adjustment, a new QoS Report is generated to enable all subscribed clients to update their QoS requirements with their associated QMACC (Figure 3.10).

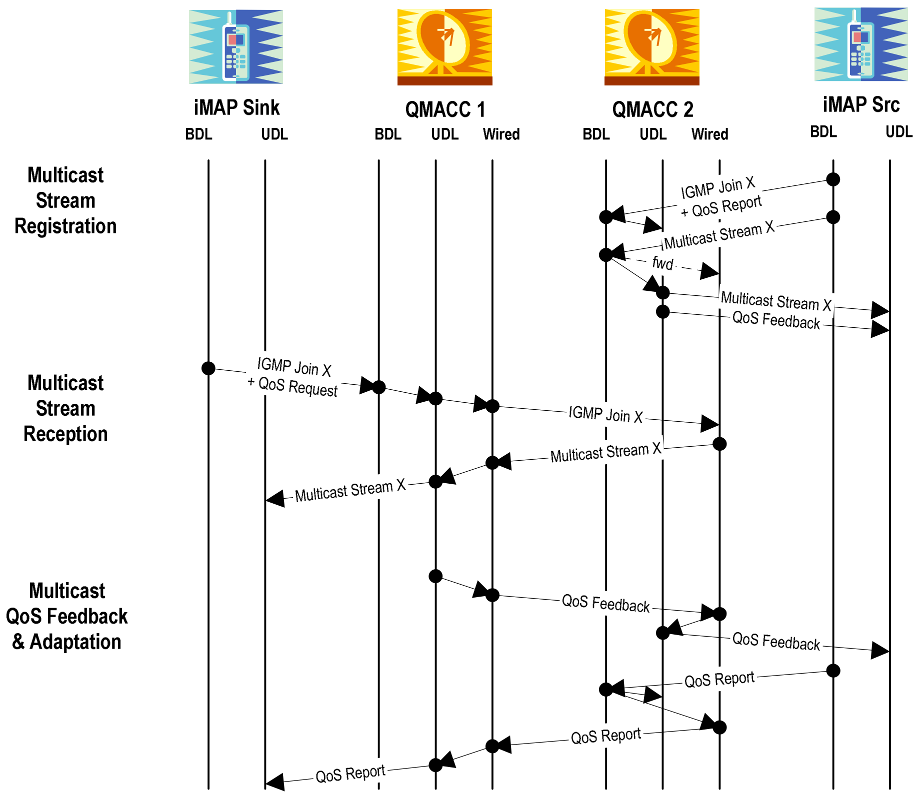


Figure 3.10: Application Layer QoS Signaling

3.4.5.2 Multicast Performance Profiles

While the QoS behavior of unicast flows has been precisely defined (Davis, 1999, Dixit, Guo & Antoniou, 2001), the QoS behavior of multicast flows is somewhat vague since it involves multiple nodes faced with differing network conditions (Diot, Dabbous & Crowcroft, 1997) described in Chapter 2.6.2. The notion of ***Multicast Performance Profiles*** is introduced to describe how a multicast source can respond to the varying network QoS conditions. By definition, *Multicast QoS Profiles* are proportional prioritization mechanisms for a given multicast stream.

Multicast Performance Profiles (Table 3.2) are used to control the multicast QoS Parameter Estimation. Given D' and BW' calculated from QoS Feedback messages (Table 3.1), we can select suitable QoS values (BW*, D*) for the next QoS Adaptation interval based on the current network condition.

- ***Maximized Performance Profile***: source adjusts QoS targets based on the Best QoS Feedback Statistics (Min. D', Max. BW')
- ***Optimal Performance Profile***: source adjusts QoS targets based on the Average QoS Feedback Statistics (Avg. D', Avg. BW')
- ***Equal Performance Profile***: source adjusts QoS targets based on the Worst QoS Feedback Statistics (Max. D', Min. BW')

Table 3.2: Multicast Performance Profiles Summary

Profile	Aggregate Statistics	Goal
Maximized Performance	Best case (Min Delay, Max Available BW)	Each node operates at its maximum QoS level Nodes meeting specified QoS Profile have highest performance. High QoS Profile Variance
Optimal Performance	Average case (Avg Delay, Avg Available BW)	Each node operates at an average QoS level A majority of nodes have reasonably good performance. Compromise between QoS Profile Variance and Performance
Equal Performance	Worst case (Max Delay, Min Available BW)	Each node operates at the minimum QoS level All Nodes have similar performance. Minimal or no QoS Profile Variance

Consequently, if different Multicast Performance Profiles are present in a given system, the priority of streams using the given profiles would be ordered according to: Maximized, Optimal, Equal.

3.4.5.3 *Application Layer QoS Requirements*

WMQF assumes that the Application layer is capable of generating layered multimedia data which is transmitted as different substreams within a given multicast group session. A multimedia stream has default delay (D_i) and bandwidth (BW_i) requirements for transmitting each of the layered substreams (S_i). A *layered* multimedia stream contains one base stream with multiple substreams for a given multicast group. QoS requirements for each multicast source are specified as:

$$\begin{aligned} QoS_{req} &= f(Delay, Bandwidth, Priority) \\ &= f(D_i, BW_i, P_i) \end{aligned} \qquad \textbf{..... Equation 4}$$

The Priority of a given multicast stream is fixed from the start of the multicast session and does not affect the QoS Adaptation process. The application specifies D_i and BW_i targets for the base stream and each of the layered substreams (S_i). Substreams have increasingly stringent QoS requirements.

Therefore, for a base stream and one substream, the bandwidth requirement is (BW_0 + BW_1) and the delay requirement is MIN (D_0, D_1), where $D_{i+1} < D_i$.

Nonetheless, actual multimedia layering and substream composition details will not be considered in this research. Instead, the effect of changing the number of substreams is to modify the current bandwidth and delay requirements of the application.

3.4.6 Multicast QoS Adaptation

Each QoS-aware Multicast Source Application is able to adapt its data rate according to the current QoS status of the network. The base substream is always present in a given multicast session. Additional substreams are added to the multicast flow when network QoS conditions allow.

The Application QoS requirements will be used by the QoS Selector to configure appropriate QoS parameters needed by the application at any given time. The process of Estimating, Selecting and Shaping is termed *QoS Adaptation* (Figure 3.11). The functions of the QoS Estimator, Selector, and Shaper are interdependent. The QoS Selector determines suitable QoS values (D*, BW*) based on current QoS Estimator output (D’, BW’). The selected QoS (D*, BW*) may differ greatly from the currently active QoS values (D_{curr}, BW_{curr}). The QoS Shaper serves to adapt current delay and bandwidth settings to the selected values using suitable convergence functions to generate new QoS targets (D_t, BW_t) for the next adaptation interval.

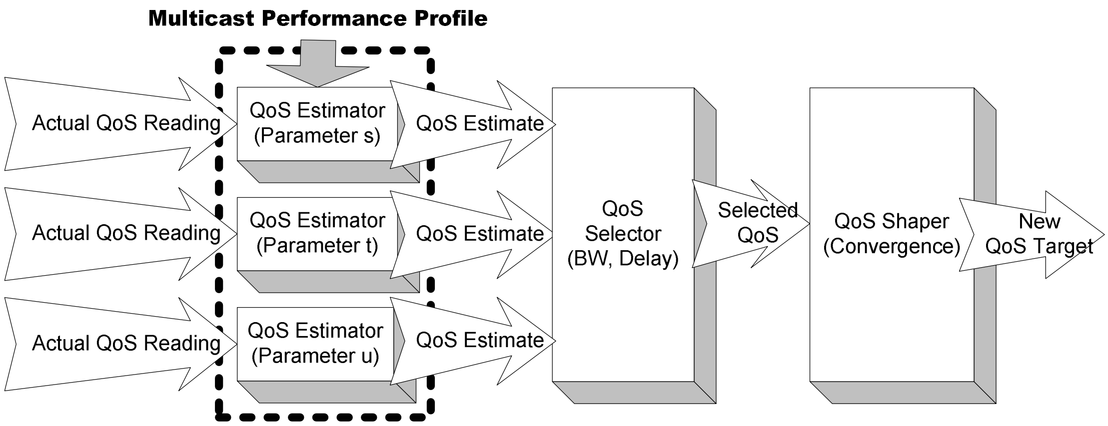


Figure 3.11: QoS Estimation, Selection, and Shaping Processes

The QoS Selector accepts normalized bandwidth and delay estimates to determine the number of substreams with relevant QoS parameters (BW_i, D_i) that can be carried over the network. Since actual QoS requirements for Bandwidth and Delay are not normalized, we must first normalize them.

Normalized Required Bandwidth for a substream ($BWreq_i \in [0,1]$) is given by Equation 5, while Normalized Required Delay for a substream ($Dreq_i \in [0,1]$) is given by Equation 6.

$$BWreq_i = \frac{BW_i}{\mu}, \text{ where } \mu : \text{Available Link Bandwidth} \quad \text{...... Equation 5}$$

$$Dreq_i = \beta \frac{D_i}{D_0}, \text{ where } D_0 : \text{Max Base stream Delay}, \beta : \text{Scaling factor} \quad \text{...... Equation 6}$$

The scaling factor β is required since there is no absolute upper bound for Delay, unlike Bandwidth which is limited by the link capacity. For $D_i < 1.0$ s, $\beta = D_0$ was found via simulation experiments to give the best performance.

Subsequently, given that a multicast flow has N substreams, we find the number of substreams *n* in order to determine (D*, BW*) suitable for the current QoS conditions estimated from (D', BW') using Equation 7 and Equation 8:

$$D^*: MIN(Dreq_0, Dreq_n) > D', \text{where } n \in 0,1,...,N$$
$$D^* = MIN(D_0, D_n)$$

..... Equation 7

$$BW^*: BWreq_0 + \sum_{i=1}^{n} BWreq_i < BW', \text{where } n \in 0,1,...,N$$
$$BW^* = BW_0 + \sum_{i=1}^{n} BW_i$$

..... Equation 8

The QoS Shaper attempts to converge the current QoS settings (D_{curr}, BW_{curr}) to the selected QoS settings (D*, BW*) using suitable convergence functions. This is accomplished by determining target QoS settings (D_t, BW_t) for the next adaptation interval which closes the gap between the old and new QoS values. The convergence function should exhibit smooth convergence behavior when fluctuations in (D*, BW*) are due to short term phenomena, while having the ability to adjust to long term changes by accelerating convergence when the differences are much greater.

The following QoS Adaptation techniques will be investigated:

- ***Static***: This is used for baseline QoS comparisons, where the QoS requirements are fixed at the start (i.e., n = N) and do not change in response to feedback from the network. Therefore the *QoS Estimator* and *QoS Shaper* are null functions.
- ***Simple Adaptive***: The target QoS requirements are adjusted using one of the given QoS profiles to the values given by the QoS Feedback in a single convergence step. Therefore, the *QoS Estimator* is a sample and hold function

based on the selected Multicast Performance Profile, while the *QoS Selector/Shaper* is a linear mapping (y=x) function (i.e., D_t = D*, BW_t = BW* for each adaptation interval).

- ***Predictive Adaptive***: QoS Feedback is not used directly in setting target QoS requirements. Instead, it is used by the *QoS Estimator* to predict future network conditions based on the selected Multicast Performance. The output of the *QoS Estimator* is used by the *QoS Selector and QoS Shaper* to achieve the selected QoS requirements using a convergence function.

3.4.6.1 Static QoS

In the static QoS technique, QoS Feedback is ignored (Figure 3.12). The QoS Selector is fixed to the maximum QoS Target values (i.e., base stream + all substreams). Hence no QoS Shaping occurs in this technique. This technique is used as a baseline for comparisons with the other techniques, to determine the worst-case network performance.

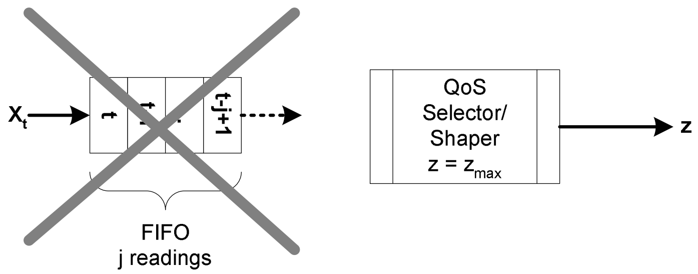


Figure 3.12: Static QoS Technique

3.4.6.2 Simple Adaptive QoS Adaptation Algorithm

QoS Feedback parameters (x_t) are stored in a FIFO queue of size j. The QoS Estimator uses the given Multicast Performance Profile to derive the Estimated QoS value (y). The estimated value (y) is normalized to [0,1] for use by the QoS Selector. The QoS Selector uses the estimate to determine the new selected quantized QoS settings (z*), which the Shaper also uses as the new target QoS settings (z). QoS changes are propagated as highest priority traffic through the network. Since the QoS Shaper implements the Selected QoS settings immediately, data packets currently in the various MAC queues may experience abrupt changes in available QoS especially if the new target QoS reduces the target bandwidth or increases the delay bounds. The size of the FIFO buffer has limited influence on the performance of the Simple Adaptive algorithm, as the Multicast Performance Profile weights each estimate equally (Figure 3.13).

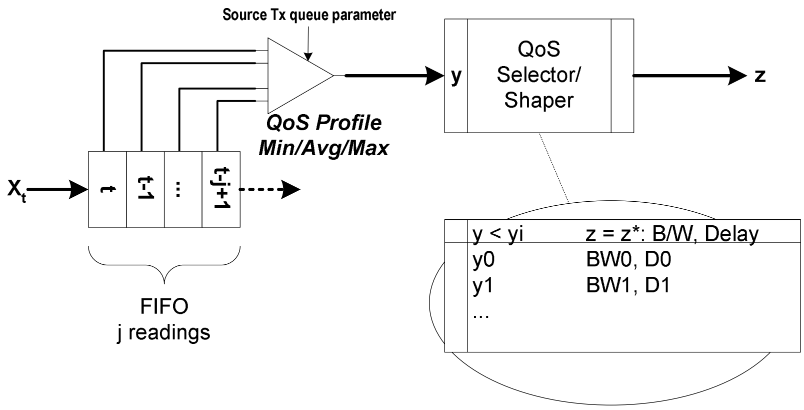


Figure 3.13: Simple Adaptive QoS Technique

3.4.6.3 Predictive Adaptive QoS Adaptation Algorithm

This is implemented using a neural network approach consisting of single element Perceptron/Adaline Estimators (Freeman & Skapura, 1991). Each QoS Parameter under consideration is modeled by its own Estimator. The Perceptron accepts a time-series stream of inputs comprising of the past *j* QoS Feedback readings (x), as well as the previous (*k* – 1) Estimator output values (y') in order to derive the new QoS estimate (y). All the weights (w_0 to w_{j+k-1}) are updated when new QoS Feedback (x) is received from the network. This is accomplished by calculating the error (δ) between the estimate (y) and the QoS Profile generated value (d). The readings (x) and estimates (y) are replaced in a FIFO manner to ensure that the system adaptation is done continuously (Continuous learning).

The size of the FIFO buffers has significant impact on the effectiveness of the Predictive Adaptive QoS algorithm since the algorithm should capture the QoS Feedback from the entire multicast population as represented by QMACC nodes in the multicast tree, in order to provide an accurate estimate of expected QoS parameters (Figure 3.16). For a multicast tree containing a small number of participating QMACCs (i.e., high iMAP to QMACC ratio, where clients are densely populated), the feedback messages could be processed with equal weighting. However, for very large multicast networks having many participating QMACCs (i.e., low iMAP to QMACC ratio, with sparsely distributed clients), feedback message aggregation would be necessary to keep the FIFO size from increasing uncontrollably. This research focuses on networks having a small number of participating QMACCs with densely populated clients for a given multicast stream, and hence does not consider feedback message aggregation.

From various simulation runs, it was found that as a rule of thumb, the number of feedback readings should at least equal to the number of QMACC nodes generating feedback messages in order for the estimator to capture the full behavior of the network. Keeping additional readings (j >> number of QMACC nodes) could be done at the expense of increased resource utilization of the Estimator, but this did not have much impact on the overall behavior of the Estimator compared with setting j equal to the number of QMACC nodes. The number of estimates (k) kept in the FIFO has an impact on how fast the Estimator senses changes in network conditions. Having too many old estimates would lead to slow reaction to change, while too few estimates would lead to a jerky response to change. From experiments, k = 2 was found to provide a reasonable rate of adaptation for the Estimator.

The Estimator output (y) value is normalized to [0,1]. It is then used by the QoS Selector to determine the appropriate QoS settings (z*) for the application. The selected QoS settings (z*) are then given to the QoS Shaper to determine the actual QoS target (z) to be propagated to the network. The QoS Shaper convergence function is a continuous function; whereas target QoS values (z) are quantized values similar to those generated by the QoS Selector. Consequently, the QoS Shaper performs the adaptation using the unquantized target values (z'), which then undergoes quantization to obtain z ($z = \lfloor z' \rfloor$ for maximized parameter; $z = \lceil z' \rceil$ for minimized parameter). Convergence is always determined based on the Unquantized QoS value.

The QoS Shaper uses a parameter *n* as a rate-increasing/decreasing factor (*Convergence Factor*) to modify the unquantized QoS value (z') to conform to the Selected QoS value (z*). Convergence for parameters that require *Maximization* (i.e., Available Bandwidth) uses the rate-increasing algorithm if $z' < z^*$ (parameter not exceeded), and the rate-decreasing algorithm if $z' > z^*$ (parameter exceeded). Conversely, parameters that require *Minimization* (i.e., Delay) would apply the rate-decreasing algorithm if $z' < z^*$ (parameter exceeded) and the rate-increasing algorithm if $z' > z^*$ (parameter not exceeded). Parameter *m* is the consecutive number of iterations where z' does not exceed z*. From Yamamoto et al. (1998) and Yamamoto et al. (2000) it is shown that the given rate increase and decrease mechanisms are effective in converging target parameters to the desired value. ε denotes convergence of *Unquantized QoS value (z')* to *Selected (z*)* value, and is expressed as a percentage of the Selected value (z*).

Figure 3.14 shows the algorithm for parameters needing *Maximization*:

If (| *Unquantized – Selected* | < ε): $Unquantized_i = Unquantized_{i-1}$; Incr.Iterations = 0
Else if (*Unquantized* > *Selected*): $Unquantized_i = Unquantized_{i-1} \div n$; Incr.Iterations = 0
Else if (*Unquantized* < *Selected*):
    If (*Incr.Iterations* == 0): $Increment = |(Selected - Unquantized) \times p|$
    Else if (*Incr.Iterations* < *m*): $Increment = Increment_{i-1} \div n$
    Else if (*Incr.Iterations* ≥ m): $Increment = Increment_{i-1} \times n$
    $Unquantized_i = Unquantized_{i-1} + Increment$; Incr.Iterations++
Target = $\lfloor Unquantized \rfloor$

Figure 3.14: QoS Shaper Algorithm (Maximization case)

Figure 3.15 shows the algorithm for parameters needing *Minimization*:

If (| *Unquantized – Selected* | < ε): $Unquantized_i = Unquantized_{i-1}$; Decr.Iterations = 0
Else if (*Unquantized* < *Selected*): $Unquantized_i = Unquantized_{i-1} \div n$; Decr.Iterations = 0
Else if (*Unquantized* > *Selected*):
    If (*Decr.Iterations* == 0): $Decrement = |(Selected - Unquantized) \times p|$
    Else if (*Decr.Iterations* < *m*): $Decrement = Decrement_{i-1} \div n$
    Else if (*Decr.Iterations* ≥ m): $Decrement = Decrement_{i-1} \times n$
    $Unquantized_i = Unquantized_{i-1} - Decrement$; Decr.Iterations++
Target = $\lceil Unquantized \rceil$

Figure 3.15: QoS Shaper Algorithm (Minimization case)

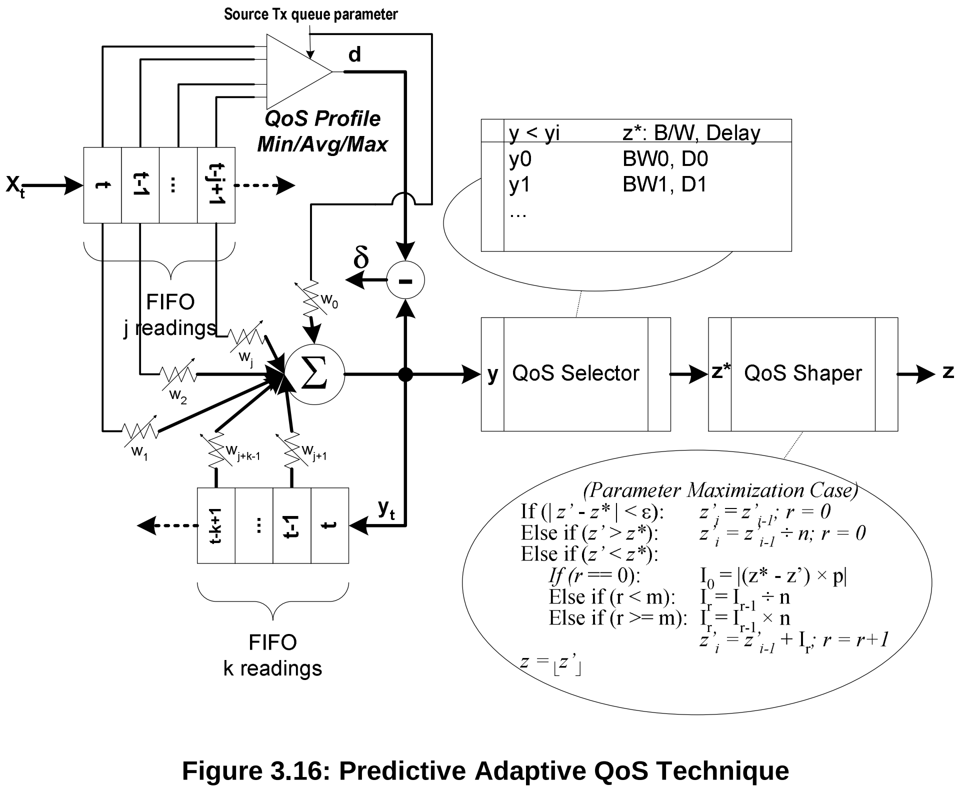


Figure 3.16: Predictive Adaptive QoS Technique

As there are 16 QMACCs defined for the various simulation scenarios, the FIFOs are configured with $j = 16$ and $k = 2$. The values for $n = 1.2$, and $m = 10$, were chosen experimentally (Yamamoto et al., 1998, Yamamoto et al., 2000) to achieve quick convergence. For WMQFSim, ε was set as 1%. The convergence function responds quickly to congestion in the network, while an increase in available bandwidth causes the QoS requirement to increase gradually at first. This is important since we want to avoid overloading the network with excess packets in case the increase in available bandwidth is only temporary.

3.5 Chapter Summary

The Wireless Multicast QoS Framework (WMQF) for mobile multicast applications was defined. WMQF used suitable Multicast QoS Adaptation Algorithms (MQAA) to perform application layer QoS Adaptation based on QoS Feedback indicating varying network conditions. In addition, the MQAA had a secondary objective of attempting to maximize Overall System Bandwidth Utilization (OSBU) for a given multicast link.

Details of the system architecture, various WMQF components, their interaction, and related algorithms were then specified and explained. These components would be used to design various validation experiments and specify several MQAA performance testing scenarios in Chapter 4.

CHAPTER 4 SIMULATION ENVIRONMENT AND WMQF SCENARIOS

The PARSEC simulation language and environment (Bagrodia et al., 1998) was developed at UCLA for the purpose of simulating complex systems and is available without cost for academic usage. The advantage of PARSEC over other freely available simulation environment is its ability to scale the simulated environment for large number of nodes, to distribute the simulation process to multiple processing hosts as well as the ability to integrate PARSEC code easily with code written in C.

The *Global Mobile Information Systems Simulation Library (GloMoSim)* (UCLA Parallel Computing Laboratory, 2001) for PARSEC was developed by the same research group in UCLA to support the specification, testing and verification of large scale ad hoc mobile networks. Consequently, reliable models for radio propagation, node mobility, standard TCP/IP network protocols and application layer models such as Constant Bit Rate (CBR), Telnet and FTP data sources and sinks were also made available.

Nonetheless, the provided models did not include 802.11 infrastructure mode, MAC layer Active Queue Management support, IGMP signaling, and application level QoS support. These features were developed as new models and incorporated with existing GloMoSim models as part of the WMQF simulation (WMQFsim) requirements.

4.1 Topographic Layout

A 2000 m by 2000 m open-plan grid topology with 16 stationary QMACCs and 30 mobile iMAPs which are moving randomly with speeds of up to of 2 m/s was used to facilitate the simulation experiments. iMAPs were distributed throughput the network topology using random placement techniques. Nonetheless, the initial positions of iMAPs and locations of QMACCs were identical for all simulation runs, in order to factor out different initial positions from affecting the simulation outcome (Figure 4.1).

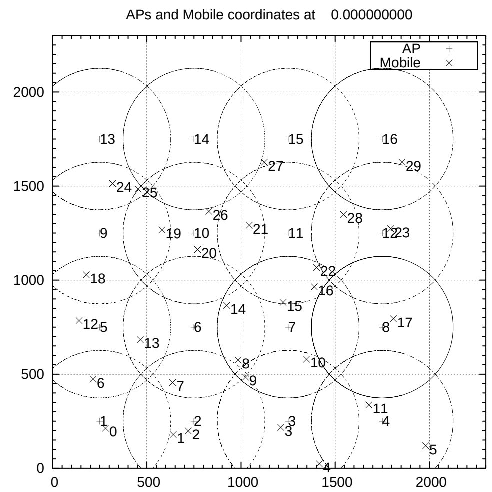


Figure 4.1: WMQF Initial Network Topography Layout

4.2 Physical (Radio) Layer Behavior

WMQFsim uses the default GloMoSim radio propagation models to simulate the environmental behavior of radio signal transmission, propagation, and reception. The default parameters settings as provided in GloMoSim were adopted to simulate the radio transceivers for both QMACCs and iMAPs. In order to simplify the physical layer modeling, equal performance transceivers were adopted for both QMACCs and IMAPs. In an actual hardware implementation, QMACC transceivers typically would have better sensitivity and transmission power compared to iMAP transceivers. The net effect of having better radio transceivers in QMACCs would result in better range as compared to the simulation results. However, this would also most likely result in larger cell sizes and more inter-cell interference.

Table 4.1: Physical (Radio) Layer Parameters

Simulation Parameters	Value
Propagation Model	Two-ray
Radio SNR	8.49583 dB
Tx Power	15 dBm
Antenna Gain	0 dB
Rx Sensitivity	-91.0 dBm
Rx Threshold	-81.0 dBm

Signals which arrive at a receiver above the Rx Sensitivity level would be registered as noise, while signals above the Rx Threshold could be decoded. The decoding of received packets was considered error-free if the Signal to Noise Ratio (SNR) was above the Radio SNR level. Otherwise, the packet would be considered to be in error and dropped.

4.3 Simulation Parameters

The following types of parameters were used to drive the various experiments:

- Node Mobility: Static / Mobile
- Link Congestion: Congestion / No Congestion
- Delay Requirements: Tight / Relaxed
- Performance Profiles: Equal / Optimal / Maximized / Mixed
- Packet Loss Requirements: 10% (constant)

4.3.1 Node Mobility

Node mobility controls whether the iMAP nodes are stationary, or moving. Since the initial node placement was performed identically for all simulation scenarios (Figure 4.1), nodes that were stationary would maintain the same network access configuration throughput the simulation run. In contrast, mobile nodes would travel randomly within the network topography, with a given speed and intermittent stops once they reach randomly generated waypoints. However, the random path generated for node mobility was identical for all simulation runs in order to standardize the environment for adaptation algorithm comparison by eliminating mobility as an independent variable.

Since mobility of individual nodes is difficult to plot, topographical snapshots every 600 s would be given to illustrate node movement throughout the simulation in Figure 4.2 and Figure 4.3:

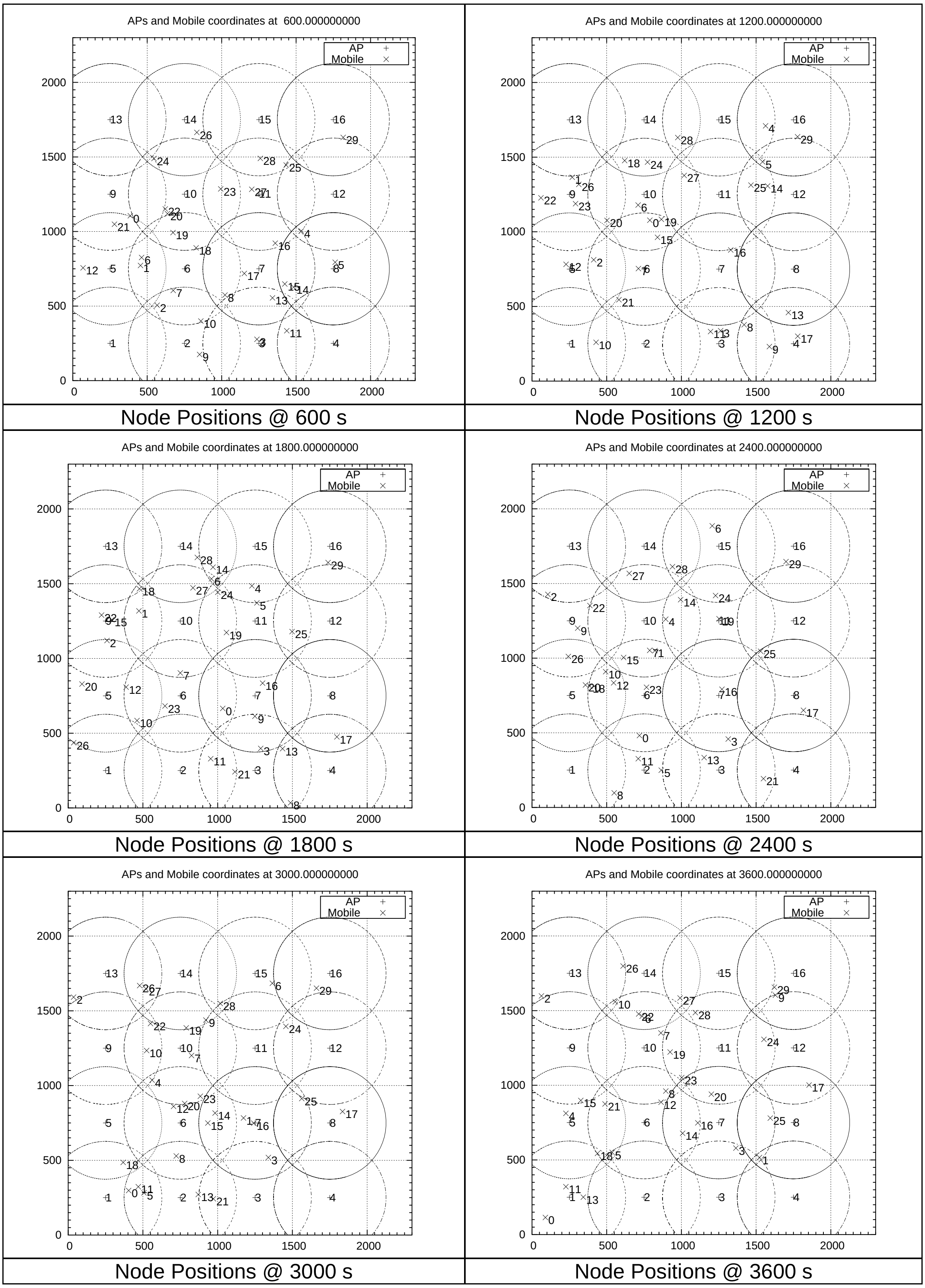
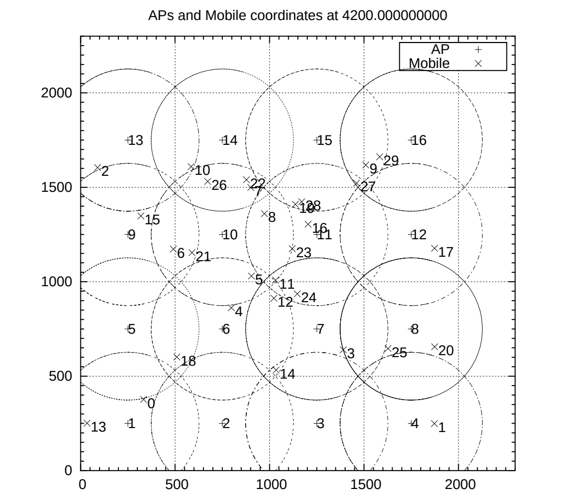
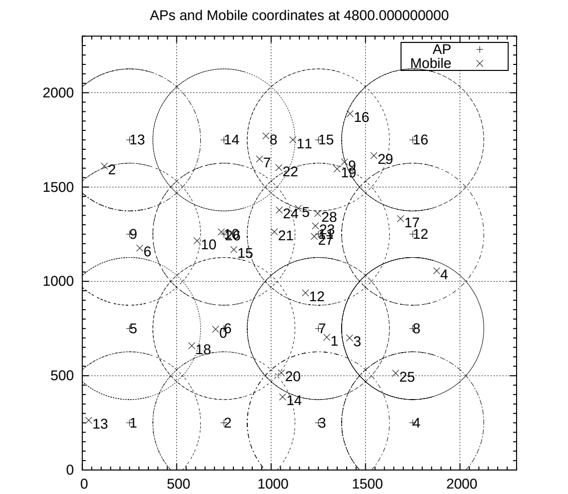


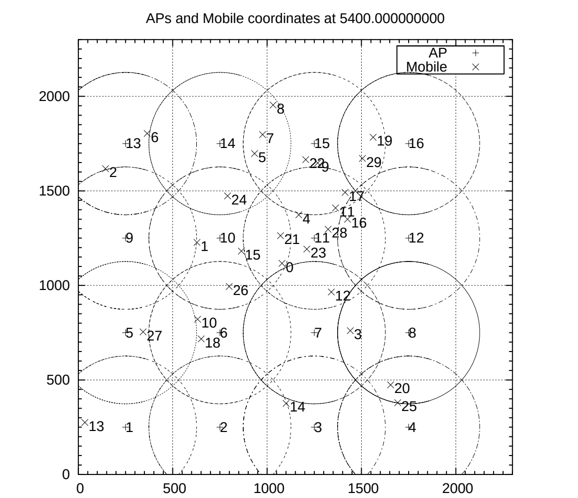
Figure 4.2: Node Position Snapshots 600s – 3600s



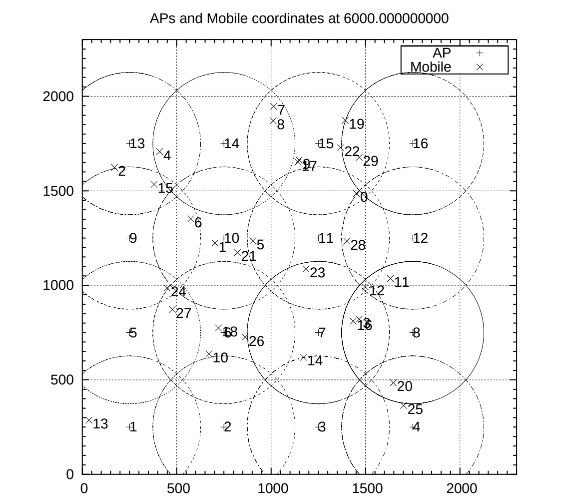
Node Positions @ 4200 s



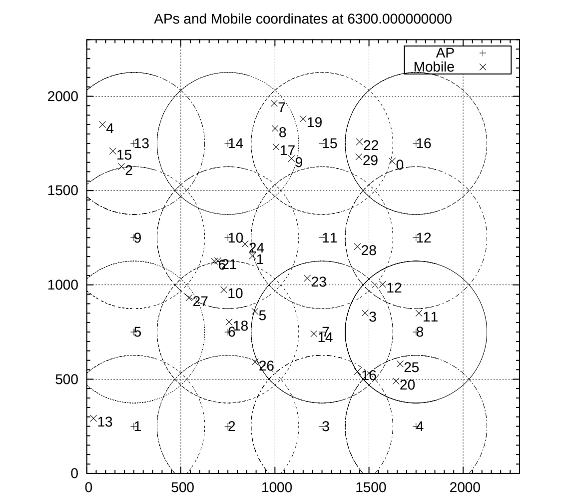
Node Positions @ 4800 s



Node Positions @ 5400 s



Node Positions @ 6000 s



Final Node Positions @ 6300 s

Figure 4.3: Node Position Snapshots 4200s – 6000s & Final Pos. at 6300s

4.3.2 Link Congestion

Link congestion occurs when the bandwidth requirements of all multicast flows carried over a given link exceed the available bandwidth. Since each flow consist of a base stream and several substreams, the No Congestion scenario was when the total required bandwidth of all base streams was less than the link capacity. Each multicast flow can potentially increase the number of substreams since excess capacity was available within the link. However, the flow should be able to transmit at its base stream bandwidth requirement without any problem. For most of the scenarios, three equal priority multicast flows were active. Given a base stream bandwidth of 300 kbps, the total bandwidth requirement for these three flows was 900 kbps, significantly less than the 2 Mbps link capacity.

In the case of the Congestion scenario, the base stream required bandwidth was set to 500 kbps. For three equal priority multicast flows, the total bandwidth requirement increased to 1.5 Mbps, which was close to or exceeded the effective link bandwidth given the 2 Mbps link capacity. Hence, most multicast flows would not be able to add substreams to their flows since the additional substreams would saturate the link.

4.3.3 Delay Requirements

Delay Requirements specify how close the application maximum delay threshold is to the inherent network delay. Since mobile iMAPs encounter inherent handover delays while roaming from one cell to another, the application maximum delay threshold should be greater than the inherent network delay. From Table 2.2, we observe that the ITU-T QoS Class 0 (real time, jitter-sensitive, high interaction) maximum delay requirements ($IPTD_{max}$ = 100 ms) exceeds the inherent network delay performance of WMQF. In contrast, ITU-T QoS Class 1 (real time, jitter-sensitive, interactive) has a less stringent delay requirement ($IPTD_{max}$ = 400 ms). For WMQF using *Proportional Differentiation*, the QoS targets for real time, jitter-sensitive, highly interactive multicast traffic were chosen to be comparable to ITU-T QoS Class 0 given the constraints of the network which has inherent network delay in the order of 200 ms. Consequently, Tight delay requirements meant that the application base stream maximum delay was set close to the inherent network delay (in the order of 200 ms), while Relaxed delay requirements meant that it was set somewhat higher than the inherent network delay (in the order of 250 ms). It should be noted that these maximum values are nominal goals in WMQF and not absolute values unlike the requirement for ITU-T QoS classes. However, the average delay values should not exceed the specified delay requirements to be considered as being complaint.

Similarly, Peak Delay Variation ($IPDV_{peak}$ = 50 ms) as measured by ITU-T QoS Classes for real time traffic is a nominal goal in WMQF. Nonetheless, the measured values of the equivalent WMQF parameter $SIAT_{sdev}$ (Table 2.6) appears to comply with this requirement generally (see Chapter 5).

4.3.4 Performance Profiles

The Multicast Performance Profiles under consideration were based on the following: Equal, Optimal and Maximized. These profiles were described in Chapter 3.4.5.2.

4.3.5 Packet Loss Requirements

Since media streams use the unreliable RTP protocol for transport, it is given that some amount of packet loss would occur. It is not possible to achieve ITU-T QoS requirements (Table 2.2) for Packet Error Rate ($IPER_{max} = 1 \times 10^{-4}$) and Packet Loss Rate ($IPLR_{max} = 1 \times 10^{-3}$) due to the wireless network environment as well as mobility triggered roaming. In WMQF, the packet loss rates would be much greater than that experienced by wired links. Consequently, each stream in WMQF was assumed to be able to tolerate up to 10% average packet loss (consisting of both packet error and packet loss) without any significant degradation in perceived quality. This Packet Loss Rate requirement is reflected in the value used for the M-LWDF Pr_{max} parameter which controls the probability of violating the QoS requirements of a given stream. The Packet Loss Rate requirement was identical for all simulations.

4.4 Simulation Categories

The simulation experiments were divided into four categories:

- Model Verification Experiments
- WMQF Performance Profiles Evaluation
- Optimal Performance Profile Mobility Trials
- Complex Scenarios

4.4.1 Model Verification Experiments

Model Verification experiments were used to test the static behavior of the system, whereby all portable nodes (iMAPs) are stationary and hence the delay, delay variance, throughput, and error rates are determined primarily by the multicast adaptation algorithms and not due to mobility, handover and dynamic network bottlenecks arising from variable node density as the simulation progressed. In addition, the Model Verification experiments would also determine if the Active Queue Management and Traffic Shaping algorithms used for *Proportional Differentiation* as well as the Application layer QoS Adaptation techniques were inherently stable.

4.4.2 WMQF Performance Profiles Evaluation

We evaluate the system performance of the three Multicast Performance Profiles (Equal, Optimal, and Maximized) in different scenarios to determine their behavior in maximizing overall system throughput while attempting to maximize individual multicast receiver performance. Since the Equal Performance Profile (EPP) used worst case network statistics to determine its QoS settings, it would by definition not be able to maximize individual receiver performance. However, it may help to achieve network stability while ensuring that the network did not saturate as quickly as for other profiles.

In contrast, the Maximized Performance Profile (MPP) is a *greedy* algorithm which attempts to maximize individual multicast receiver performance at the expense of other users and overall network stability. Consequently, it was not expected to be a viable profile for maximizing overall system throughput. Finally, the Optimal Performance Profile (OPP), which served to average network statistics in deriving its QoS settings, was expected to provide the needed balance between providing good individual receiver performance (which is available to most receivers) and achieving network stability while maximizing overall system throughput simultaneously (hence the name).

The scenarios investigated were:

- Equal scenario: Three multicast flows, each using EPP
- Maximized scenario: Three multicast flows, each using MPP
- Mixed scenario: Three multicast flows, each using a different performance profile (Equal, Optimal, and Maximized)

OPP was investigated in detail in the next series of experiments to determine its behavior since it was expected to provide the best performance of all the different Multicast Performance Profiles.

4.4.3 Optimal Performance Profile Mobility Trials

The behavior of the Optimal Performance Profile (OPP) for the case of mobile nodes experiencing different network conditions was explored in these experiments.

- No Congestion, Tight Delay scenario: 3 multicast flows, each using OPP
- No Congestion, Relaxed Delay scenario: 3 multicast flows, each using OPP
- Congestion, Tight Delay scenario: 3 multicast flows, each using OPP

4.4.4 Complex Scenarios

Complex scenarios were used to check the behavior of the QoS Adaptation Algorithms in high loading and high variance network conditions. The following scenarios were investigated:

- Static nodes, Many Multicast Streams (Multiple sources)
- Mobile nodes, Many Multicast Streams (Multiple sources)

4.5 Detailed Simulation Parameters

Some common simulation parameters identical in all simulation runs are listed in Table 4.2. Parameters indicated as *<Specific to Simulation>* are specified for a given experiment scenario in Appendix C.

Table 4.2: Default Simulation Parameters

Simulation Parameters	**Value**
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	<Specific to Simulation>
Base Stream Delay (max)	<Specific to Simulation>
Num. of additional Substreams	*<Specific to Simulation>*
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	*<Specific to Simulation>*
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s

4.6 Chapter Summary

Various validation test cases were defined to verify that the WMQF simulation models were stable. Subsequently, WMQF Performance Profiles were subjected to several test scenarios to determine their suitability. Finally, the most promising performance profile (OPP) was subjected to various mobility scenarios to verify its stability and convergence behavior under different network conditions. These scenarios would also be used to validate the performance of specified MQAA algorithms. Results and discussion of the validation and simulation tests are given in Chapter 5.

CHAPTER 5 RESULTS, ANALYSIS AND DISCUSSION

5.1 Simulation Results Presentation

In the following sections, only *Average* results for the System Multicast Metrics (SOWL, SIAT, SAT, SPER) will be presented since *Proportional Differentiation* QoS mechanisms were designed to meet average QoS targets and not absolute (Minimum, Maximum) targets. Nonetheless, details of the Minimum, Maximum, and Standard Deviation for SOWL, SIAT, SAT and SPER results were included in Appendix C for comparison purposes.

Experiments with No Congestion scenarios utilizes multicast source streams each with base stream requirement of 300 kbps, while Congestion scenarios uses multicast source streams each with base stream requirement of 500 kbps. Scenarios with Tight Delay requirement has SOWL target of 200 ms, while Relaxed Delay has SOWL target of 250 ms.

5.2 Model Verification Experiments

Model Verification Experiments were designed to validate the simulation models developed for WMQF as well as verify the stability of the Multicast QoS Adaptation Algorithms. Nodes in these experiments were distributed randomly throughout the simulation topology, and remained static for the duration of the experiments. Three scenarios were evaluated, and the detailed simulation parameters provided in Appendix C.2:

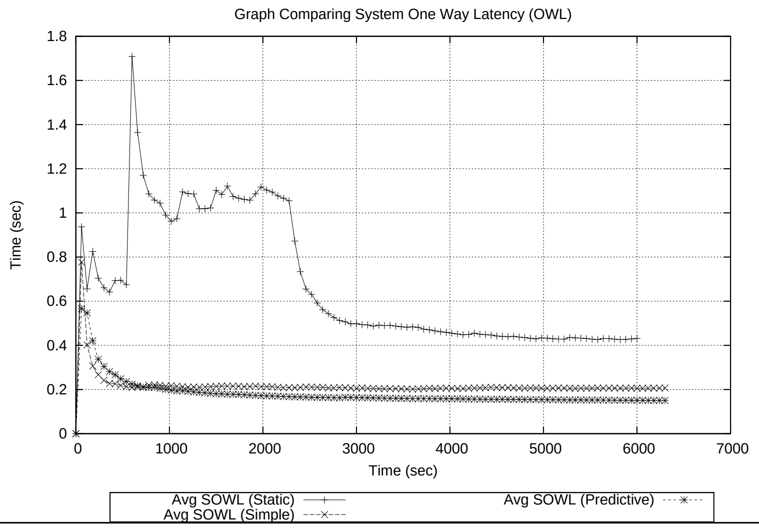
- Static Nodes, No Congestion, Tight Delay, OPP (Appendix C.2.1)
- Static Nodes, No Congestion, Relaxed Delay, OPP (Appendix C.2.2)
- Static Nodes, Congestion, Tight Delay, OPP (Appendix C.2.3)

5.2.1 Model Verification Experiment Multicast Stream Parameters

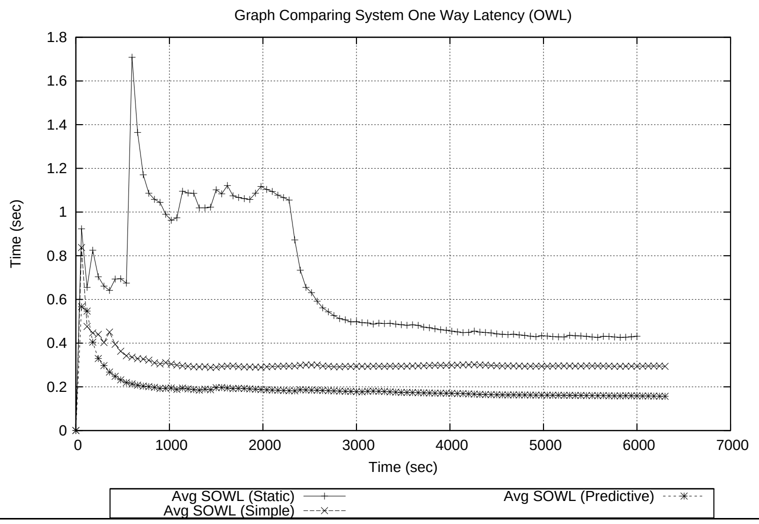
Three multicast streams would simultaneously be transmitted and compete for UDL bandwidth as they are transmitted to the entire group of 11 receivers. All cases involve multi-layered multicast streams, each with a maximum bandwidth of 1 Mbps. However, the no congestion cases involved multicast streams with base stream bandwidth requirements of 300 kbps, hence all 3 base streams should be carried over the UDL without any problems. A maximum of 7 additional substreams could be added depending on UDL bandwidth availability and network congestion.

The congestion case involved multicast streams with base stream bandwidth requirement of 500 kbps, and was therefore reaching the UDL bandwidth threshold where congestion started to take effect. A maximum of 5 additional substreams could be added depending on UDL bandwidth availability and network congestion.

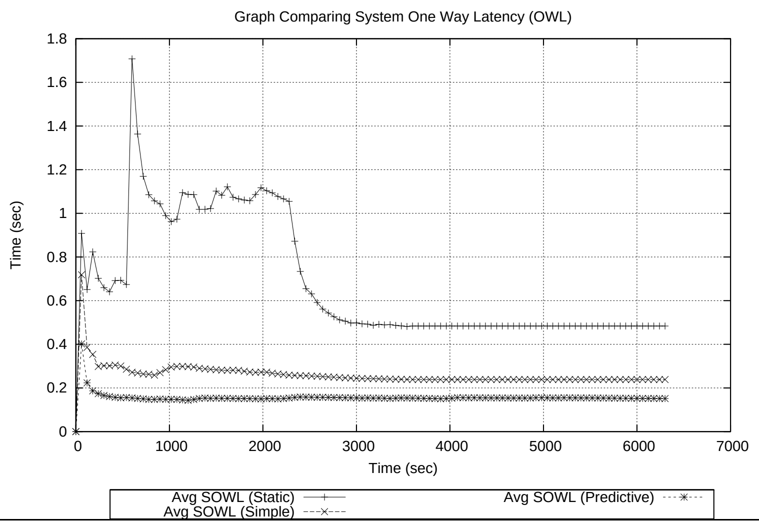
The Optimal Performance Profile (OPP) was used in these verification experiments as it was found to provide the best performance (for example, highest average SAT) after numerous trial and error attempts. This observation would be tested in much greater detail in Chapter 5.3. Scenarios involving other performance profiles would not be presented here as they do not change the steady state behavior of the WMQF models.



a. Avg. SOWL for Static Nodes, No Congestion, Tight Delay, OPP

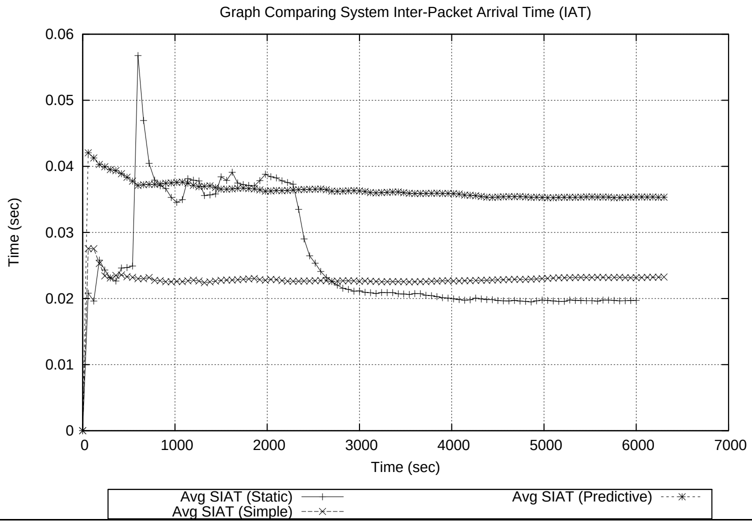


b. Avg. SOWL for Static Nodes, No Congestion, Relaxed Delay, OPP

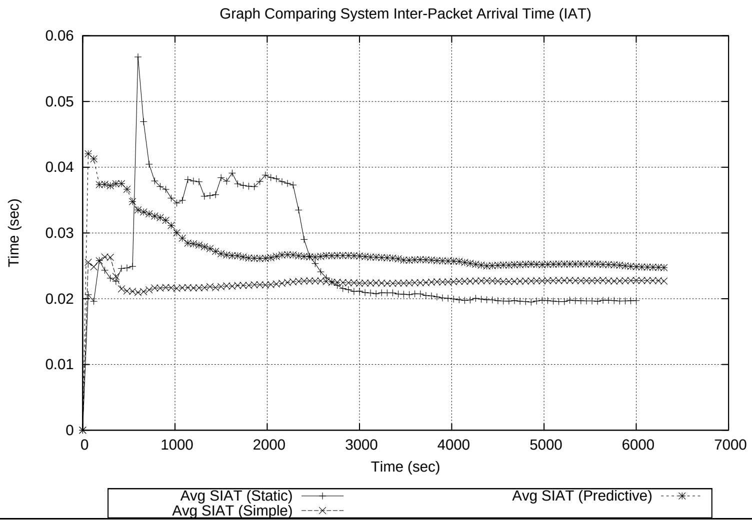


c. Avg. SOWL for Static Nodes, Congestion, Tight Delay, OPP

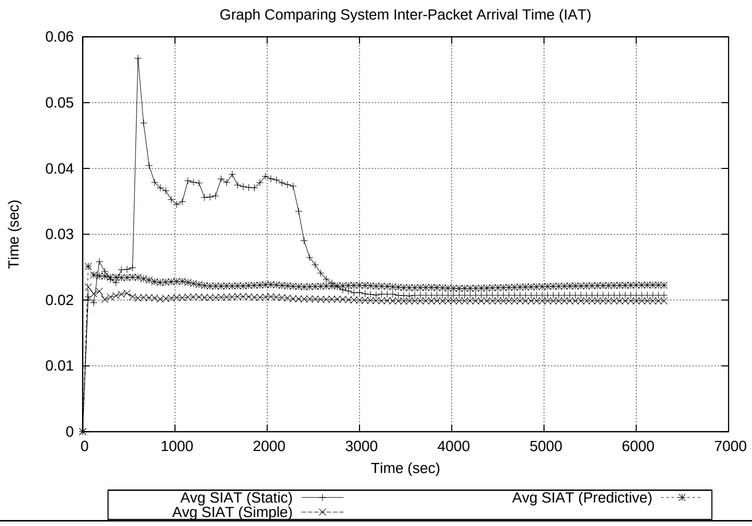
Figure 5.1: Avg. SOWL for Model Verification Experiments



a. Avg. SIAT for Static Nodes, No Congestion, Tight Delay, OPP

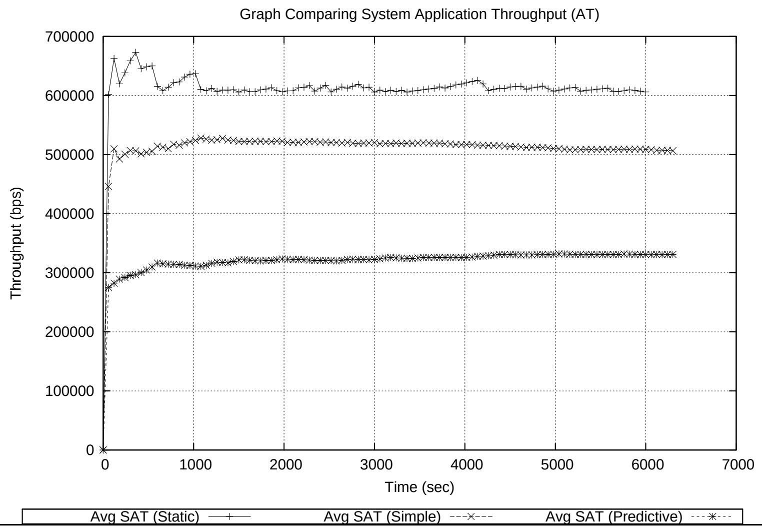


b. Avg. SIAT for Static Nodes, No Congestion, Relaxed Delay, OPP

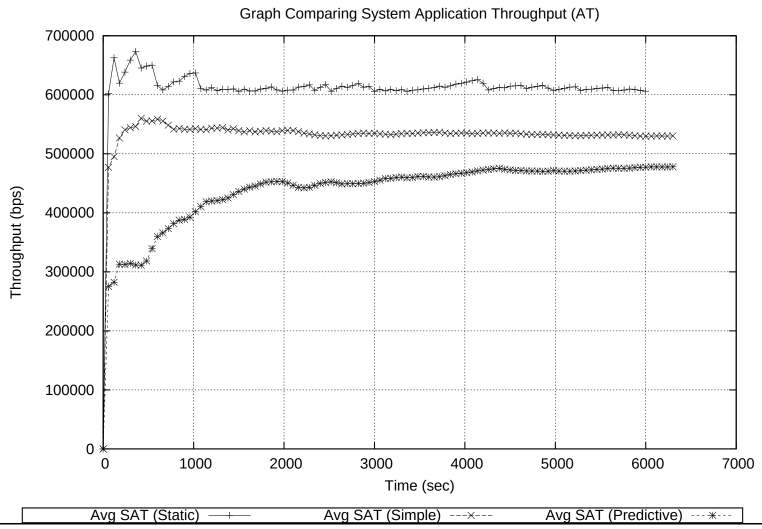


c. Avg. SIAT for Static Nodes, Congestion, Tight Delay, OPP

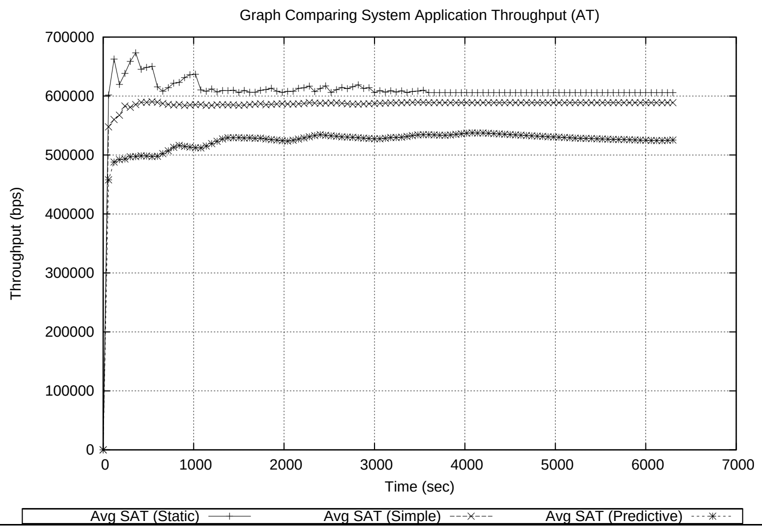
Figure 5.2: Avg. SIAT for Model Verification Experiments



a. Avg. SAT for Static Nodes, No Congestion, Tight Delay, OPP

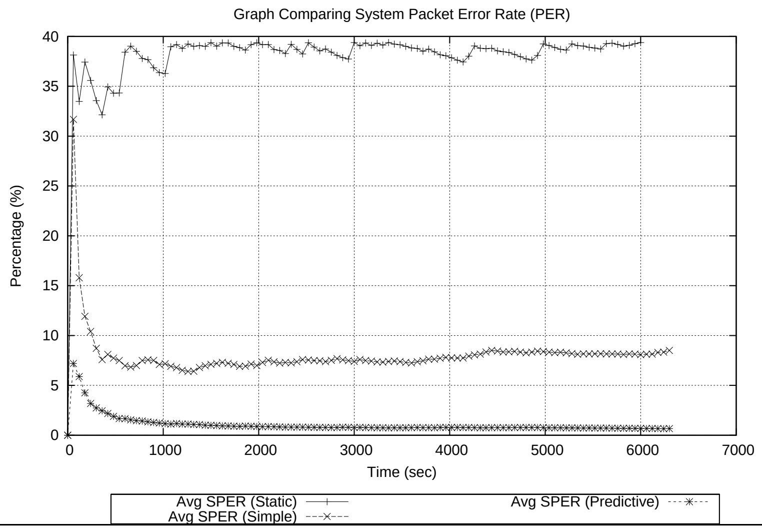


b. Avg. SAT for Static Nodes, No Congestion, Relaxed Delay, OPP

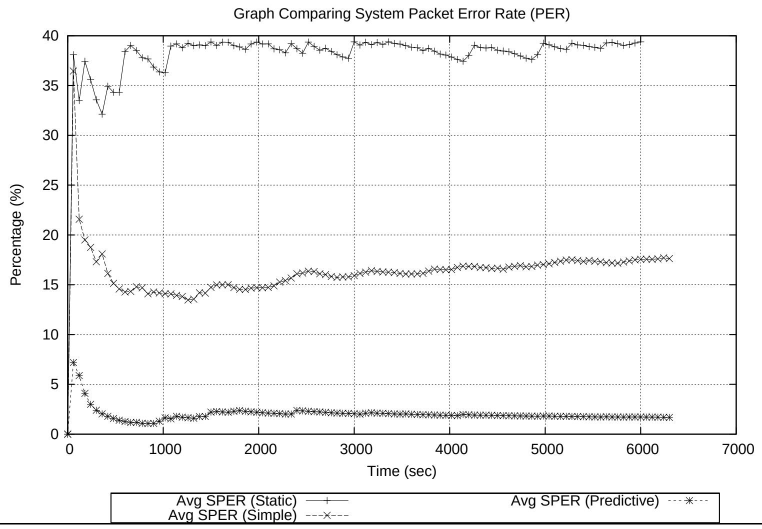


c. Avg. SAT for Static Nodes, Congestion, Tight Delay, OPP

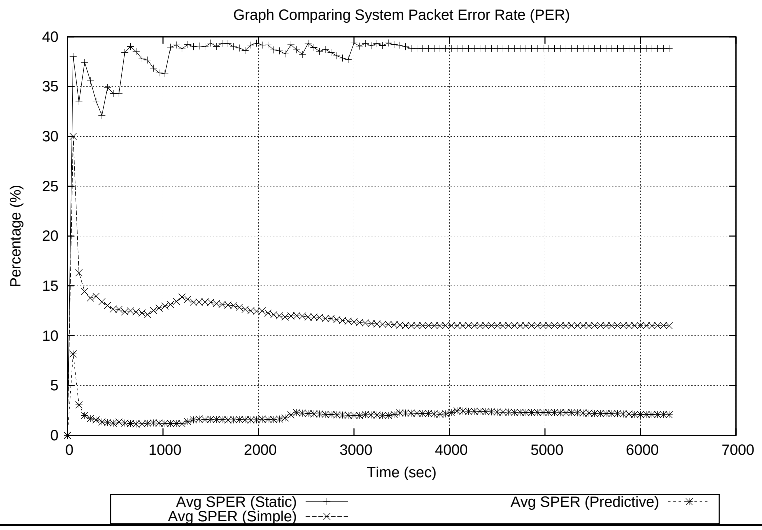
Figure 5.3: Avg. SAT for Model Verification Experiments



a. Avg. SPER for Static Nodes, No Congestion, Tight Delay, OPP



b. Avg. SPER for Static Nodes, No Congestion, Relaxed Delay, OPP



c. Avg. SPER for Static Nodes, Congestion, Tight Delay, OPP

Figure 5.4: Avg. SPER for Model Verification Experiments

5.2.2 Discussion of Model Verification Experiments Data

It is obvious that the Static QoS technique performs poorly for all measured metrics (Figure 5.1 – Figure 5.4), since the QoS settings were set to the maximum, requiring 1 Mbps for each multicast stream. Since there were three streams, the required bandwidth exceeds the link capacity.

However, it is interesting to note that the *Simple Adaptation* technique achieves much better Overall System Bandwidth Utilization (OSBU, Equation 3), of about 1.5 Mbps compared to the *Predictive Adaptive Adaptation* technique of about 960 kbps, for the "Static Nodes, No Congestion, Tight Delay, OPP" scenario (Figure 5.3.a). This occurred due to the rather stringent delay requirements for that scenario, causing the *Predictive Adaptive Adaptation* technique to adopt very conservative QoS settings. The advantage of doing so is in its ability to achieve better average SPER of 1% compared with the *Simple Adaptation* technique average SPER of about 9% (whereas target SPER is 10%) (Figure 5.4.a). Even though both techniques are within the SPER targets, the *Simple Adaptive Adaptation* technique provides better QoS to the multicast streams.

When the delay requirements was relaxed, as given in the "Static Nodes, No Congestion, Relaxed Delay, OPP" case, OSBU for *Predictive Adaptive Adaptation* rose to about 1.5 Mbps vs. 1.6 Mbps for *Simple Adaptation* (Figure 5.3.b). However, *Simple Adaptation* suffered in the area of average SOWL and average SPER, where average SOWL increased to 300 ms, exceeding the 250 ms QoS requirement (Figure 5.1.b), and average SPER rose to 17% exceeding the 10% QoS requirement (Figure 5.4.b). In comparison, *Predictive Adaptive Adaptation* achieved 200 ms average SOWL (Figure 5.1.b), and maintains less than 2% average SPER (Figure 5.4.b). Consequently *Predictive Adaptive Adaptation* technique is preferred over *Simple Adaptation* when QoS requirements are not so stringent.

For the case of “Static Nodes, Congestion, Tight Delay, OPP”, both *Simple Adaptation* and *Predictive Adaptive Adaptation* perform reasonably well, with Simple Adaptation exceeding the target SOWL and SPER by a small margin (Figure 5.1.c, Figure 5.4.c), but achieving OSBU of 1.8 Mbps (Figure 5.3.c), which is near the maximum achievable link utilization. In contrast, *Predictive Adaptive Adaptation* manages an OSBU of 1.5 Mbps (Figure 5.3.c), but manages to stay within the QoS requirement bounds (< 200 ms average SOWL, about 2% average SPER) (Figure 5.1.c, Figure 5.4.c).

The absolute value of SIAT is not so important since that depends on the Average SAT for the given stream (there is an inverse relationship between bandwidth usage / throughput and inter-packet arrival times). The SIAT performance (Figure 5.2) for *Simple Adaptation* was relatively stable for all scenarios. In contrast, Static QoS experiences significant fluctuation in SIAT at the earlier part of the experiment, while *Predictive Adaptive Adaptation* exhibited decreasing SIAT for the “Static Nodes, No Congestion, Relaxed Delay, OPP” scenario (Figure 5.2.b) due to ramping up of SAT (Figure 5.3.b) at the initial 1000 s of the experiment.

It can be summarized that *Static QoS* technique is completely unsuitable as a QoS adaptation scheme, while *Simple Adaptation* is more aggressive compared to *Predictive Adaptive Adaptation* in maximizing system utilization, at the risk of not achieving QoS requirements by a small amount. In contrast, *Predictive Adaptive Adaptation* is overly conservative and does not fully utilize available link capacity when network conditions are stable and when available bandwidth experiences minimal perturbation.

5.3 WMQF Performance Evaluation

Three multicast streams would simultaneously be transmitted and compete for UDL bandwidth as they are transmitted to the entire group of 11 mobile receivers. All cases involve multi-layered multicast streams, each with a maximum bandwidth of 1 Mbps. Each base stream has bandwidth requirements of 500 kbps. A maximum of 5 additional substreams could be added depending on UDL bandwidth availability and network congestion.

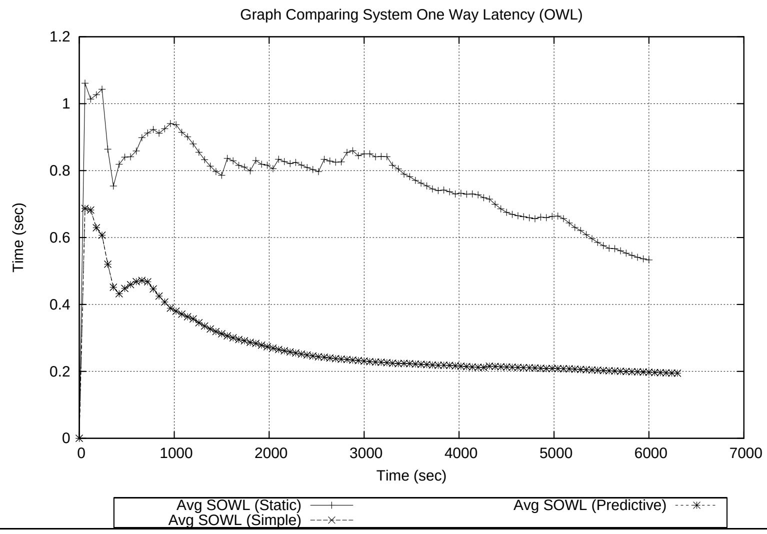
Essentially the WMQF Performance Evaluation experiments were meant for comparing the effectiveness of the various Multicast Performance Profiles in achieving QoS goals for the individual multicast stream in the face of mobility and link congestion, as well as their impact on system throughput and OSBU.

5.3.1 WMQF Performance Evaluation Scenario Parameters

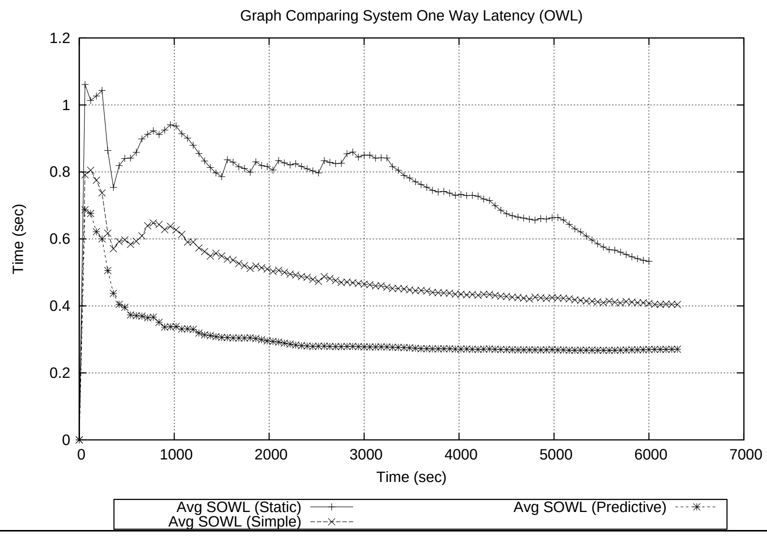
Three scenarios were tested, with detailed simulation parameters given in Appendix C.3:

- Mobile Nodes, Congestion, Tight Delay, EPP (Appendix C.3.1)
- Mobile Nodes, Congestion, Tight Delay, MPP (Appendix C.3.2)
- Mobile Nodes, Congestion, Tight Delay, Mixed PP (Appendix C.3.3)

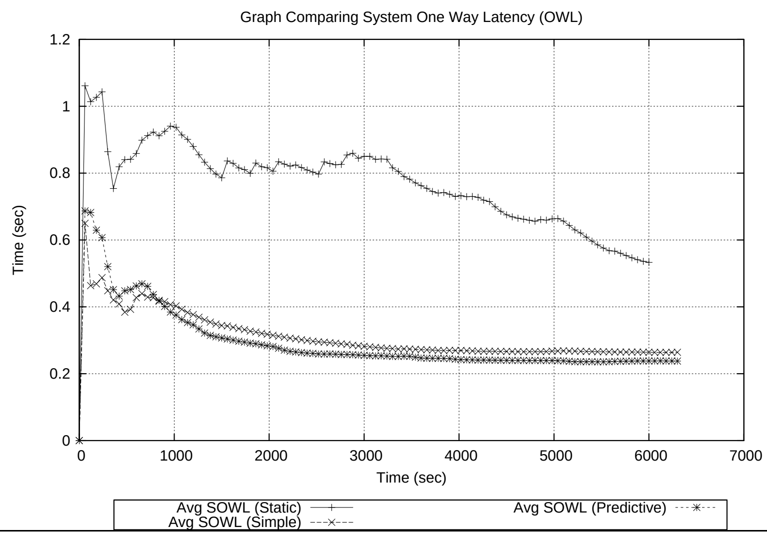
The Optimal Performance Profile (OPP) was not evaluated in this set of experiments as it would be explored in greater detail in Chapter 5.4.



a. Avg. SOWL for Mobile Nodes, Congestion, Tight Delay, EPP

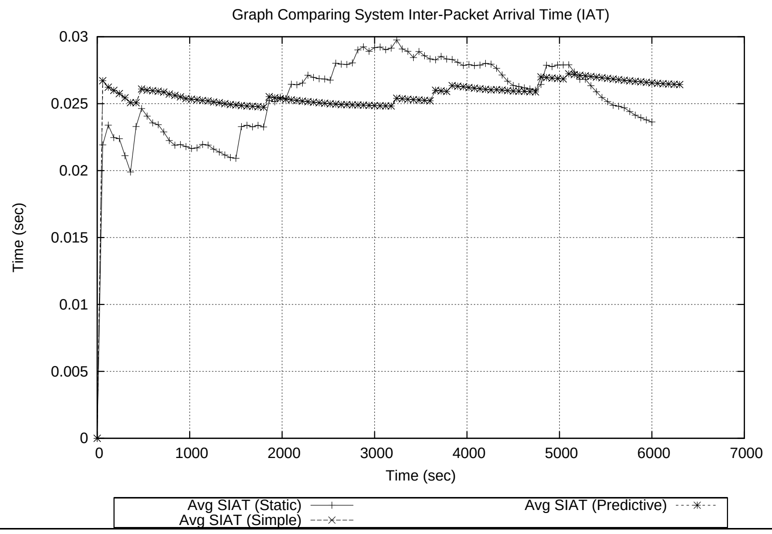


b. Avg. SOWL for Mobile Nodes, Congestion, Tight Delay, MPP

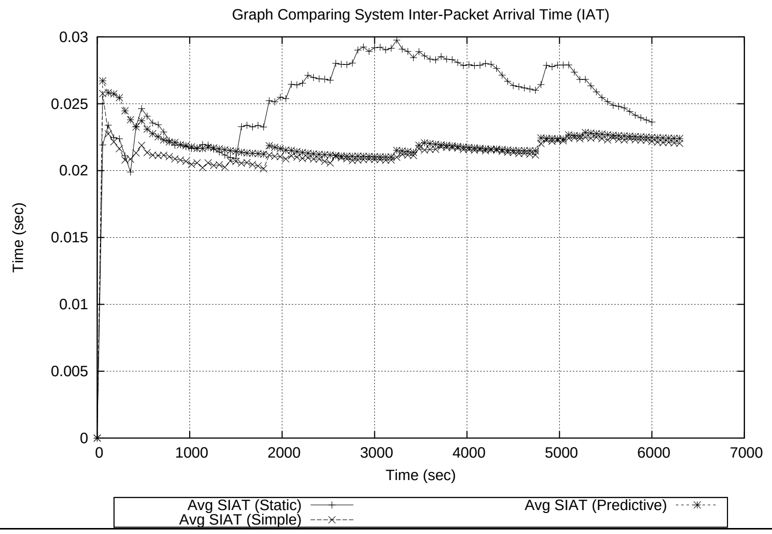


c. Avg. SOWL for Mobile Nodes, Congestion, Tight Delay, Mixed PP

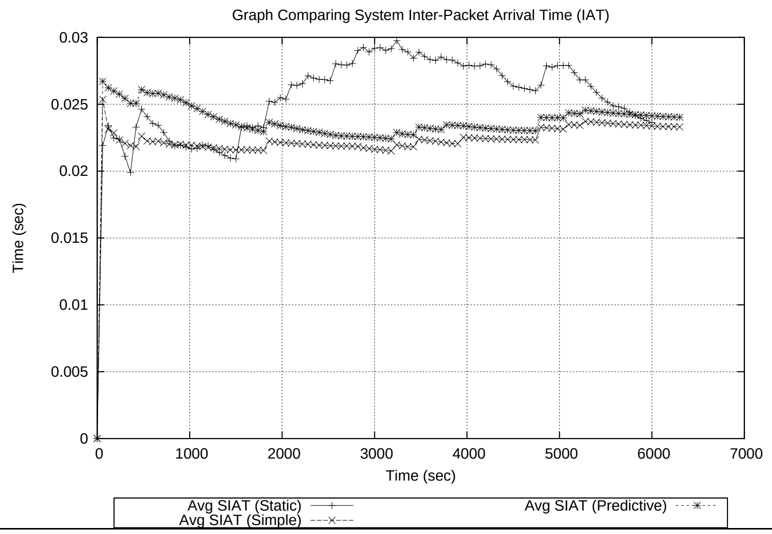
Figure 5.5: Avg. SOWL for WMQF Performance Evaluation



a. Avg. SIAT for Mobile Nodes, Congestion, Tight Delay, EPP

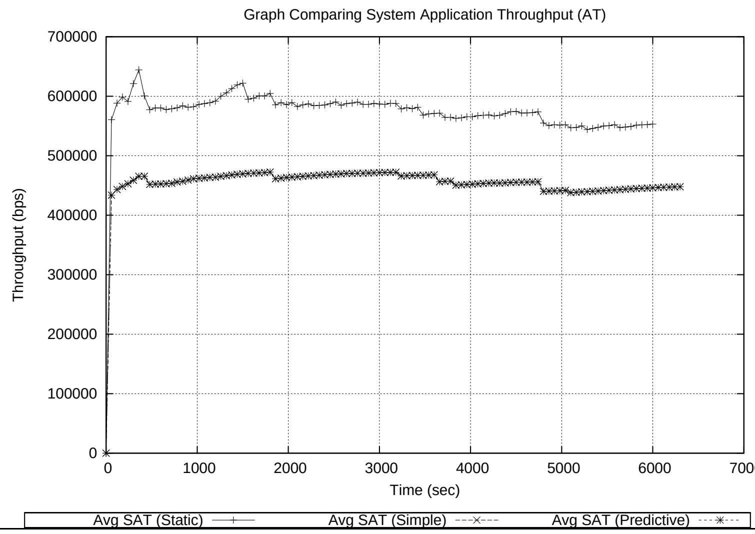


b. Avg. SIAT for Mobile Nodes, Congestion, Tight Delay, MPP

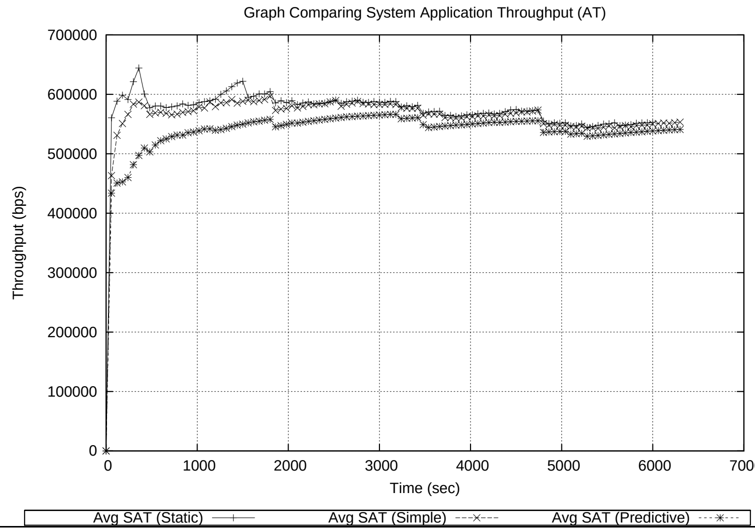


c. Avg. SIAT for Mobile Nodes, Congestion, Tight Delay, Mixed PP

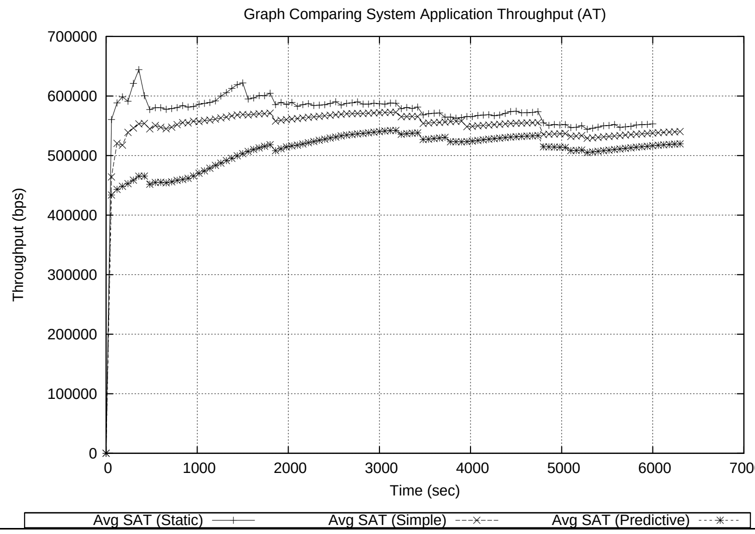
Figure 5.6: Avg. SIAT for WMQF Performance Evaluation



a. Avg. SAT for Mobile Nodes, Congestion, Tight Delay, EPP

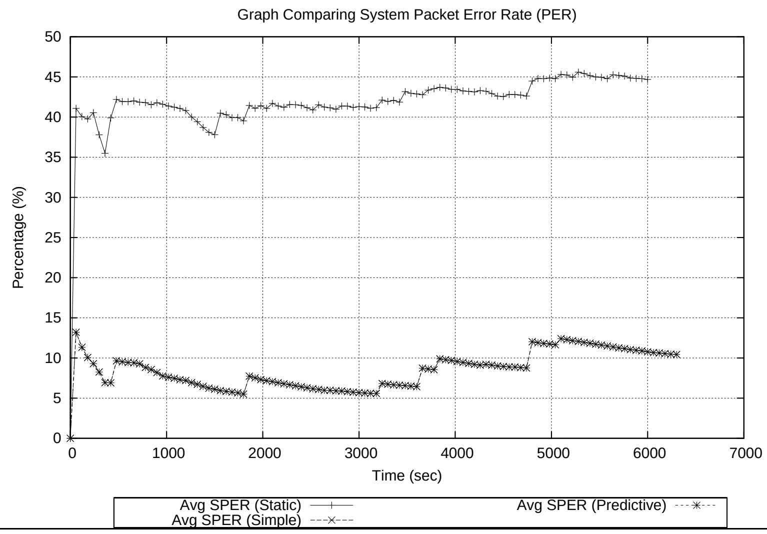


b. Avg. SAT for Mobile Nodes, Congestion, Tight Delay, MPP

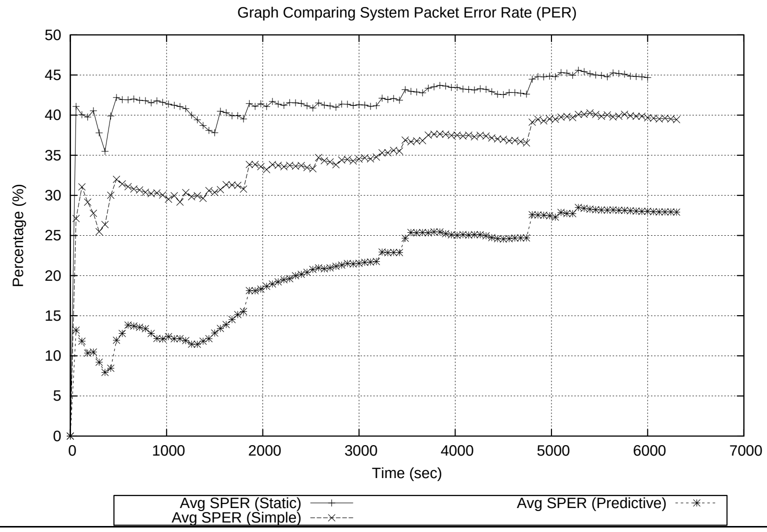


c. Avg. SAT for Mobile Nodes, Congestion, Tight Delay, Mixed PP

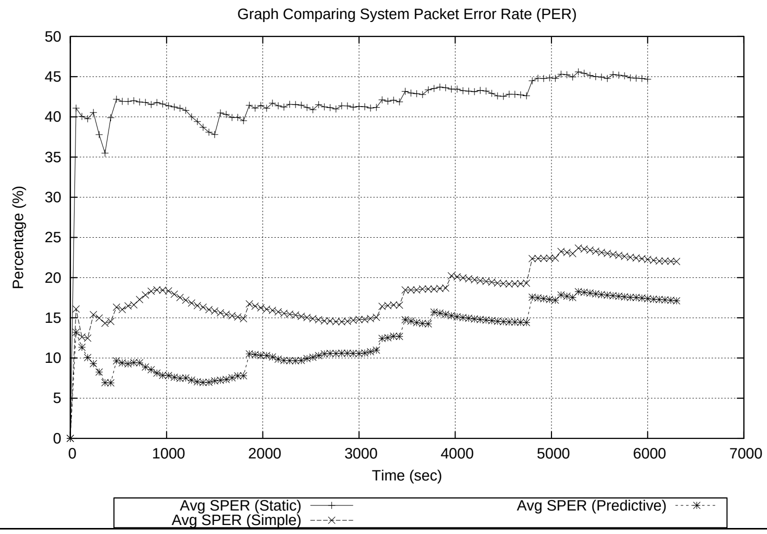
Figure 5.7: Avg. SAT for WMQF Performance Evaluation



a. Avg. SPER for Mobile Nodes, Congestion, Tight Delay, EPP



b. Avg. SPER for Mobile Nodes, Congestion, Tight Delay, MPP



c. Avg. SPER for Mobile Nodes, Congestion, Tight Delay, Mixed PP

Figure 5.8: Avg. SPER for WMQF Performance Evaluation

5.3.2 Discussion of WMQF Performance Evaluation Data

With the introduction of node mobility, the achievable performance of the system is impacted significantly. Firstly, SOWL (Figure 5.5) increases due to overheads in roaming, traffic spikes arising from densely populated cells, and other transient contention that occur when nodes move in and out of the respective cells. Consequently, none of the Multicast Performance Profiles and Adaptation techniques were able to meet the SOWL requirement for the entire duration of the simulation, and in some cases, it was not achieved at all. Consequently, we focus on the relative performance of the various adaptation techniques and behavior of the different performance profiles.

As expected, the Equal Performance Profile (EPP) resulted in the lowest OSBU of 1.35 Mbps (derived from Figure 5.7.a), but has the best average SOWL (Figure 5.5.a) and average SPER (Figure 5.8.a) performance. Both the *Simple Adaptation* and *Predictive Adaptive Adaptation* techniques were able to converge towards the QoS requirements towards the end of the simulation run. In addition, the behavior of both techniques was identical. Hence, for EPP, *Simple Adaptation* ≡ *Predictive Adaptive Adaptation*.

The results for Optimal Performance Profile (OPP) are discussed in *Chapter 5.4, "Optimal Performance Profile Mobility Trials*." It should be noted here that OPP provides better overall system stability compared with EPP and MPP, since it averages out extremes in reported QoS statistics.

Maximized Performance Profile (MPP) was observed to be unsuitable for achieving QoS requirements, even though OSBU was 1.65 Mbps (derived from Figure 5.7.b), since all of the adaptation techniques far exceeded the SOWL (Figure 5.5.b) and SPER (Figure 5.8.b) targets. While MPP might be viable for increasing QoS for a limited number of receivers, it causes significant negative impact on the entire system.

When three equal-priority multicast streams with differing performance profiles (Equal, Optimal, Maximized) were active in the same environment (Mixed PP), it was observed that the *Predictive Adaptive Adaptation* technique was able to adjust much better compared with the other two techniques. Even though *Predictive Adaptive Adaptation* technique was not able to meet the QoS requirements, it had a slight advantage compared to *Simple Adaptation* for average SOWL performance (Figure 5.5.c), and significantly better average SPER performance of 17% vs. 22% (Figure 5.8.c). This was due to the more conservative behavior of *Predictive Adaptive Adaptation* technique.

The SIAT performance (Figure 5.6) for *Simple Adaptation* and *Predictive Adaptive Adaptation* was relatively stable for all scenarios. However, there were fluctuations throughout the experiment duration as the various mobile receiver nodes experience varying network conditions and source multicast nodes encounter bottlenecks due to competing unicast traffic.

It can be summarized that Maximized Performance Profile is not suitable for maintaining network stability, while Equal Performance Profile achieves network stability at the expense of OSBU. In addition, *Predictive Adaptive Adaptation* is more robust compared to *Simple Adaptation* when dealing with multicast flows of differing priorities.

5.4 Optimal Performance Profile Mobility Trials

Three multicast streams would simultaneously be transmitted and compete for UDL bandwidth as they are transmitted to the entire group of 11 mobile receivers. All cases involve multi-layered multicast streams, each with a maximum bandwidth of 1 Mbps. The no congestion cases involved multicast streams with base stream bandwidth requirements of 300 kbps, hence all 3 base streams should be carried over the UDL without any problems. A maximum of 7 additional substreams could be added depending on UDL bandwidth availability and network congestion.

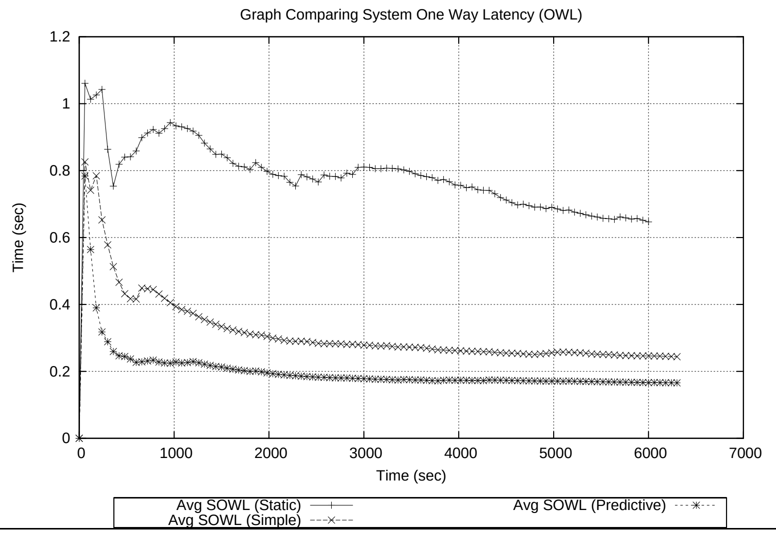
The congestion case involved multicast streams with base stream bandwidth requirement of 500 kbps, and was therefore reaching the UDL bandwidth threshold where congestion started to take effect. A maximum of 5 additional substreams could be added depending on UDL bandwidth availability and network congestion.

5.4.1 Parameters for Optimal Performance Profile Mobility Trials

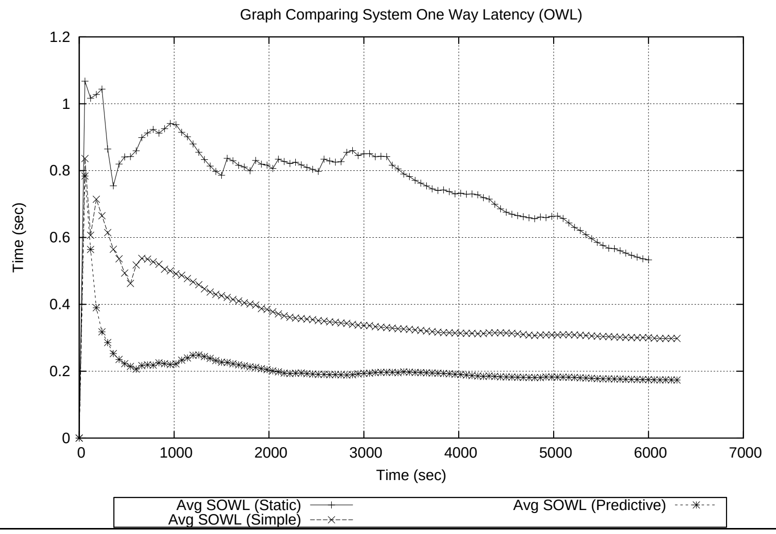
Three scenarios were evaluated, and the detailed simulation parameters provided in Appendix C.4:

- Mobile Nodes, No Congestion, Tight Delay, OPP (Appendix C.4.1)
- Mobile Nodes, No Congestion, Relaxed Delay, OPP (Appendix C.4.2)
- Mobile Nodes, Congestion, Tight Delay, OPP (Appendix C.4.3)

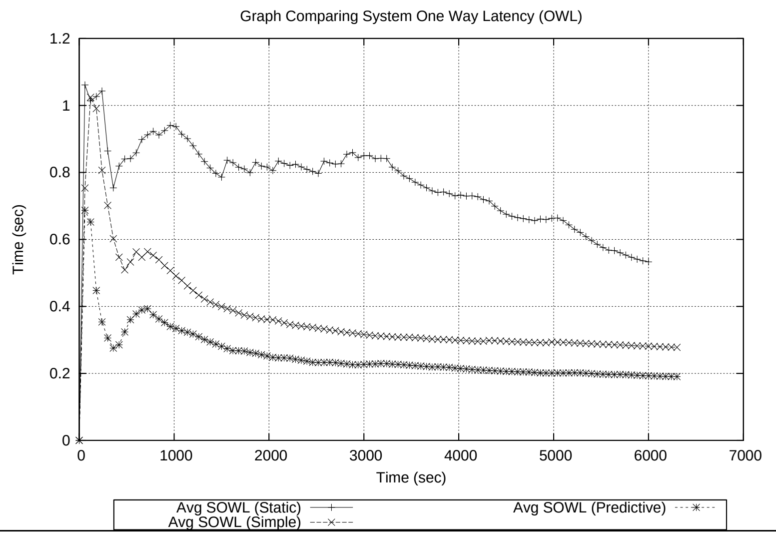
These scenarios test the performance of OPP in the face of node mobility and varying network conditions.



a. Avg. SOWL for Mobile Nodes, No Congestion, Tight Delay, OPP

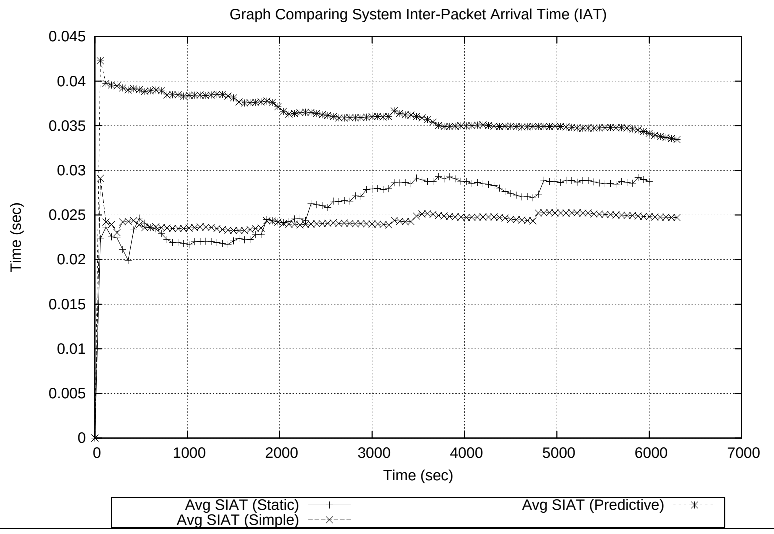


b. Avg. SOWL for Mobile Nodes, No Congestion, Relaxed Delay, OPP

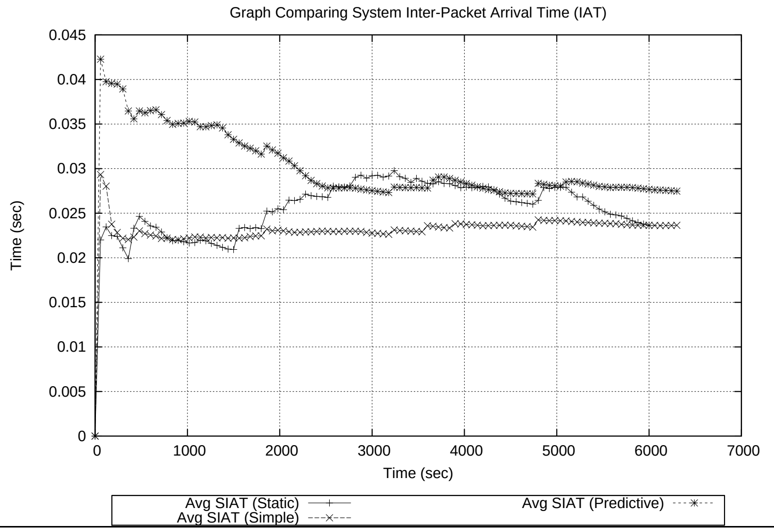


c. Avg. SOWL for Mobile Nodes, Congestion, Tight Delay, OPP

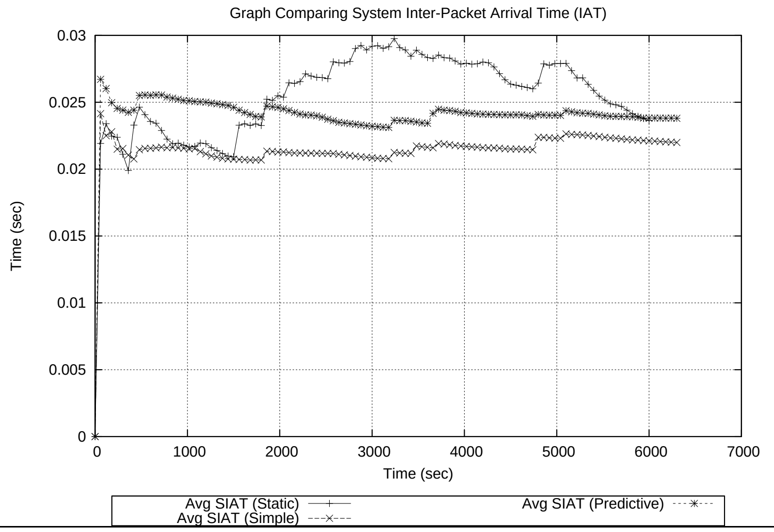
Figure 5.9: Avg. SOWL for Optimal Performance Profile Mobility Trials



a. Avg. SIAT for Mobile Nodes, No Congestion, Tight Delay, OPP

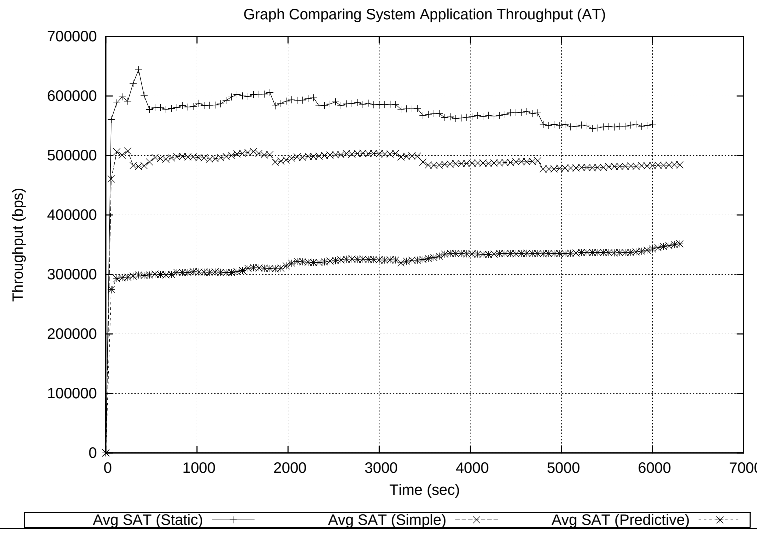


b. Avg. SIAT for Mobile Nodes, No Congestion, Relaxed Delay, OPP

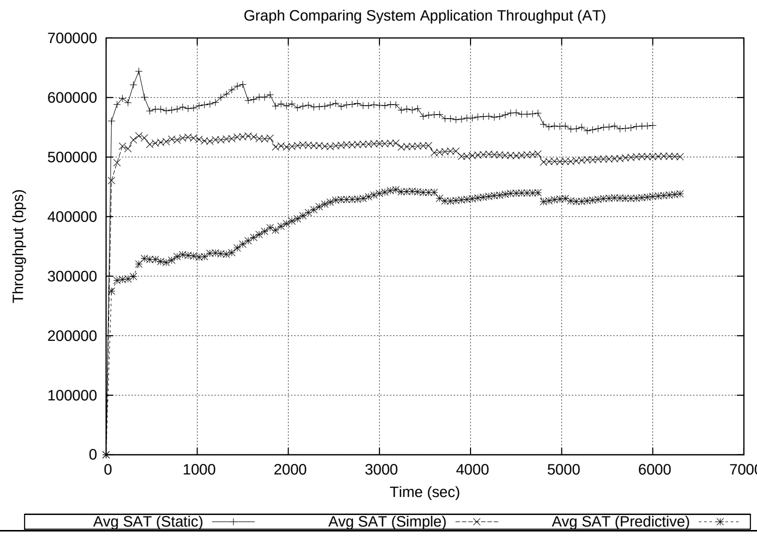


c. Avg. SIAT for Mobile Nodes, Congestion, Tight Delay, OPP

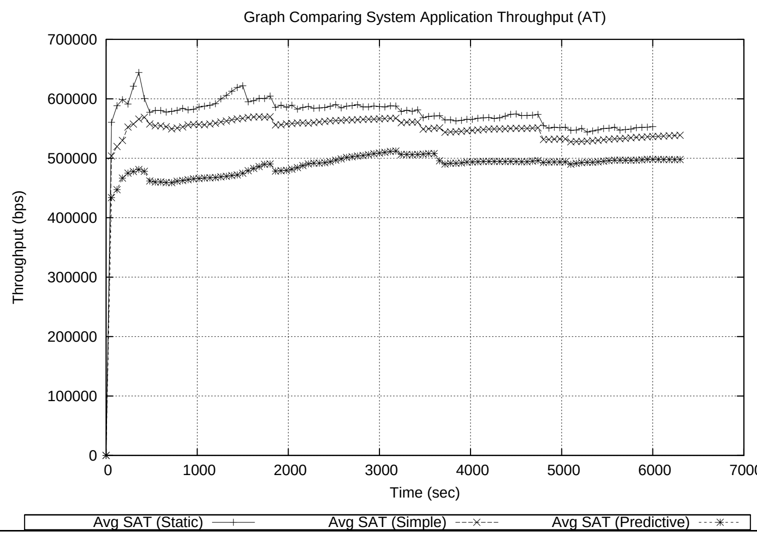
Figure 5.10: Avg. SIAT for Optimal Performance Profile Mobility Trials



a. Avg. SAT for Mobile Nodes, No Congestion, Tight Delay, OPP

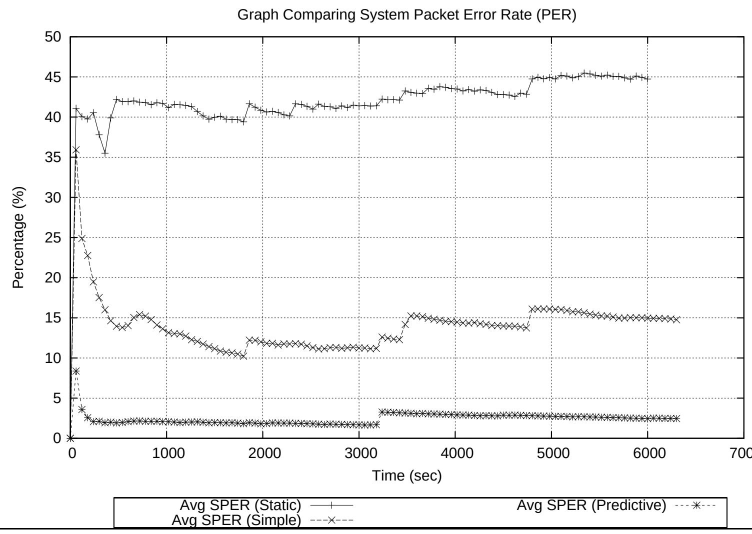


b. Avg. SAT for Mobile Nodes, No Congestion, Relaxed Delay, OPP

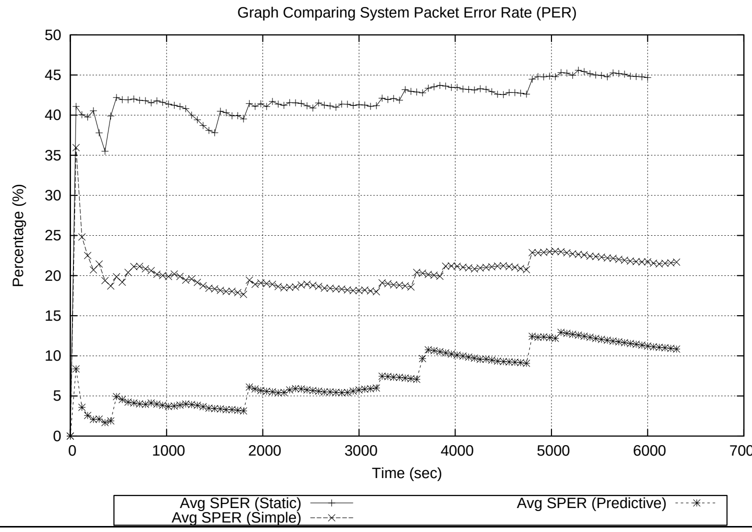


c. Avg. SAT for Mobile Nodes, Congestion, Tight Delay, OPP

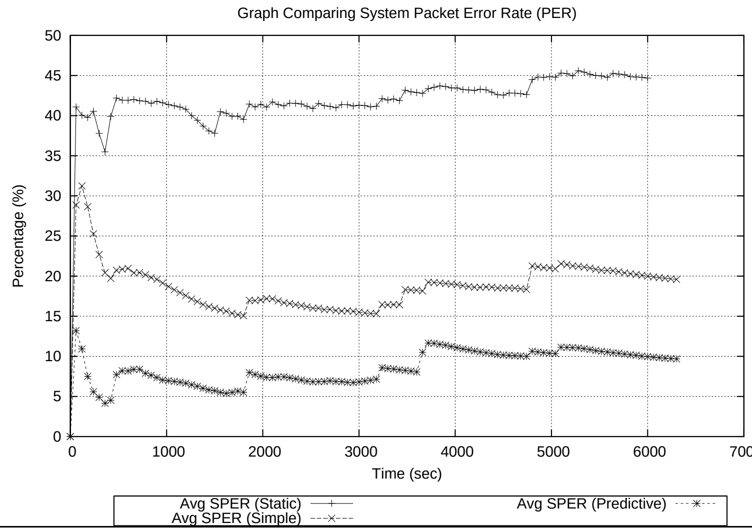
Figure 5.11: Avg. SAT for Optimal Performance Profile Mobility Trials



a. Avg. SPER for Mobile Nodes, No Congestion, Tight Delay, OPP



b. Avg. SPER for Mobile Nodes, No Congestion, Relaxed Delay, OPP



c. Avg. SPER for Mobile Nodes, Congestion, Tight Delay, OPP

Figure 5.12: Avg. SPER for Optimal Performance Profile Mobility Trials

5.4.2 Discussion of OPP Mobility Trials Results

These series of experiments were designed to compare the performance of various Adaptation Algorithms under different congestion and delay scenarios given the choice of the Optimal Performance Profile. As can be seen from the graphs, *Predictive Adaptive Adaptation* was able to meet the QoS settings without any problems for the case of "Mobile Nodes, No Congestion, Tight Delay, OPP", by maintaining average SOWL of 200 ms (Figure 5.9.a), and average SPER of 3% throughout (Figure 5.12.a), at the expense of OSBU which is at the minimum level of 900 kbps (derived from Figure 5.11.a). In contrast, *Simple Adaptation* achieved 1.5 Mbps OSBU (derived from Figure 5.11.a) but was not able to converge on average SOWL (250 ms, from Figure 5.9.a), nor average SPER (15%) (Figure 5.12.a).

This problem for *Simple Adaptation* persisted for the case of "Mobile Nodes, No Congestion, Relaxed Delay, OPP", although it was able to maintain 1.5 Mbps OSBU (derived from Figure 5.11.b), the average SOWL degraded to 300 ms (Figure 5.9.b), and average SPER rose to 20% (Figure 5.12.b). In contrast, *Predictive Adaptive Adaptation* maintained the average SOWL at 200 ms (Figure 5.9.b) while converging towards an average SPER of 10% at the end of the simulation (Figure 5.12.b). This enabled the OSBU to increase from the initial 900 kbps to a final OSBU of about 1.3 Mbps (derived from Figure 5.11.b). This ramping up of the OSBU was accomplished in 3000 s (50 minutes). While the duration seems long, the behavior of average SPER at 3000 s increased from 5% to 13% at 5000 s and finally settle down to 10% after 6000 s (Figure 5.12.b). This indicates that the OSBU of 1.3 Mbps is at the maximum sustainable utilization level barring any additional congestion caused by node mobility and roaming. It was observed in most of the mobility experiments that SPER experienced a jump at time instances around 3600 s and 4800 s. Consequently, the slow ramping up of offered traffic actually helped to prevent excessive fluctuations in average SPER and dampen the overshooting caused by roaming and mobility.

For the “Mobile Nodes, Congestion, Tight Delay, OPP” scenario, none of the algorithms were able to achieve targeted QoS goals, although *Predictive Adaptive Adaptation* managed to converge to the requirements towards the end of the simulation. This was achieved by constraining each multicast stream to its base stream traffic for most of the simulation, resulting in an OSBU of 1.5 Mbps (derived from Figure 5.11.c). In contrast, *Simple Adaptation* overestimated available network resources, allowing for more substreams. Hence, it achieved an OSBU of 1.65 Mbps (derived from Figure 5.11.c), at the expense of not meeting QoS requirements, having average SOWL of 300 ms (Figure 5.9.c) and average SPER of 15% (Figure 5.12.c). The overall system performance using *Static Adaptation* was poor, since average System One Way Latency (SOWL) exceeded 600 ms for most of the simulation (Figure 5.9.c), and maximum SOWL reached 400 s (Max SOWL, Figure C.26).

SIAT experienced fluctuation under all the QoS Adaptation techniques (Figure 5.10). *Static QoS* experiences high fluctuations, while *Simple Adaptation* has relatively smooth SIAT curves when offered data rates remained relatively unchanged. In comparison, the *Predictive Adaptive Adaptation* technique varied the offered data rate in response to QoS feedback, and hence experienced greater fluctuation in SIAT under the “Mobile Nodes, No Congestion, Relaxed Delay, OPP” scenario.

In summary, these experiments showed that *Predictive Adaptive Adaptation* was effective for converging QoS targets towards specified QoS requirements, at the expense of increased jitter (variability in SIAT). Less flexible QoS adaptation schemes such as *Simple Adaptation* encountered less jitter in general, but were not able meet other QoS requirements such as SPER due to its primitive QoS adaptation algorithm.

5.5 Complex Scenarios

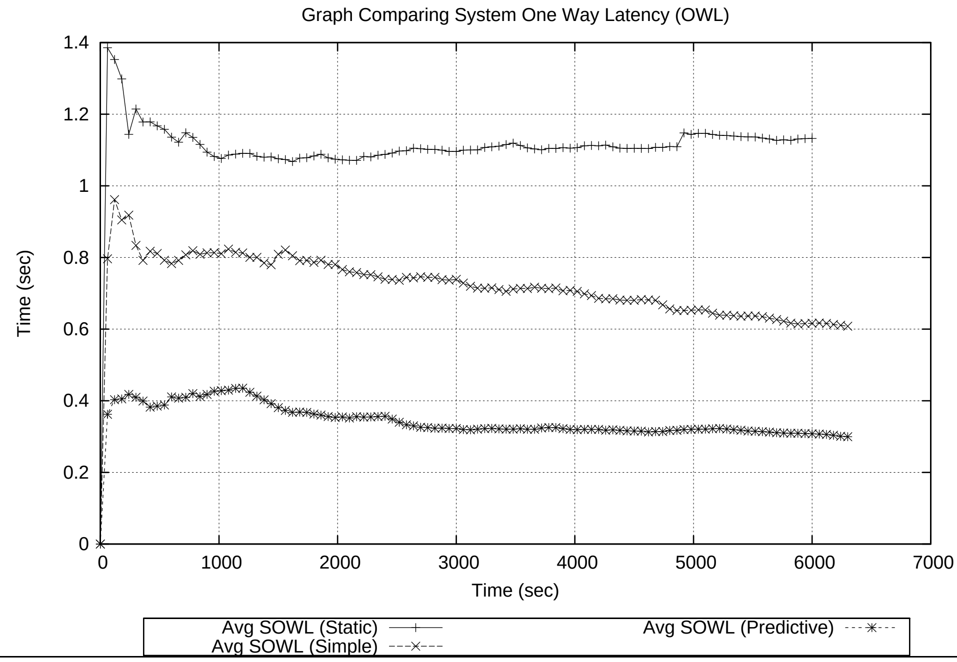
Ten multicast streams would simultaneously be transmitted and compete for UDL bandwidth as they are transmitted to selected receivers from a group of 20 mobile nodes. Each receiver will receive two out of the ten multicast streams. All cases involve multi-layered multicast streams, each with a maximum bandwidth of 1 Mbps. Each base stream has bandwidth requirements of 300 kbps. A maximum of 7 additional substreams could be added depending on UDL bandwidth availability and network congestion.

The complex scenarios were designed to stress test the Multicast Performance Profiles and Multicast QoS Adaptation Algorithms to determine their robustness and ability to handle extreme network conditions.

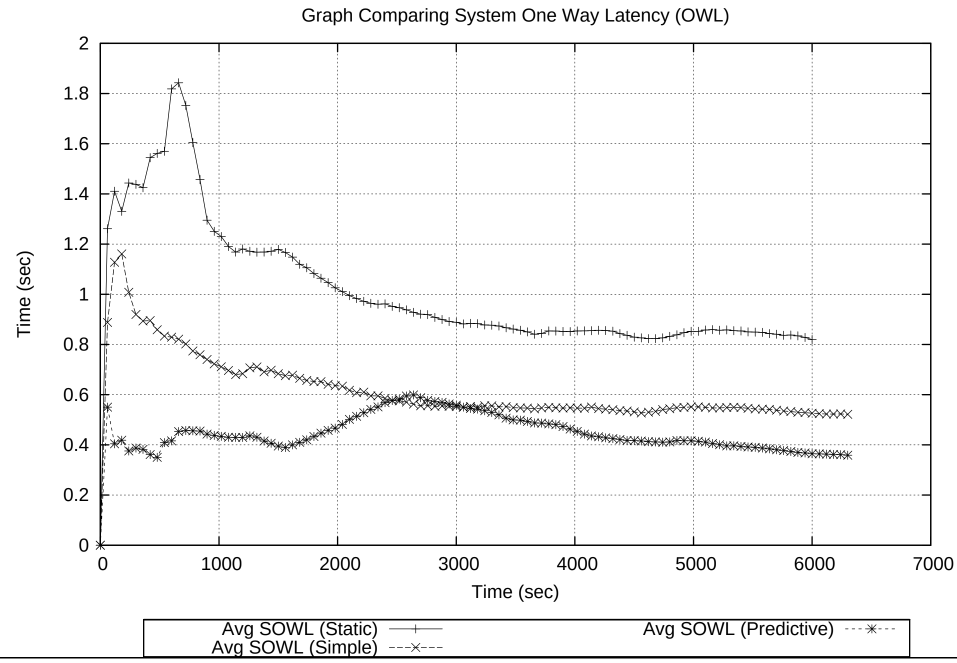
5.5.1 Complex Scenarios Simulation Parameters

Two scenarios were evaluated, and the detailed simulation parameters provided in Appendix C.5:

- Multiflows, Static Nodes, No Congestion, Relaxed Delay, OPP (Appendix C.5.1)
- Multiflows, Mobile Nodes, No Congestion, Relaxed Delay, OPP (Appendix C.5.2)

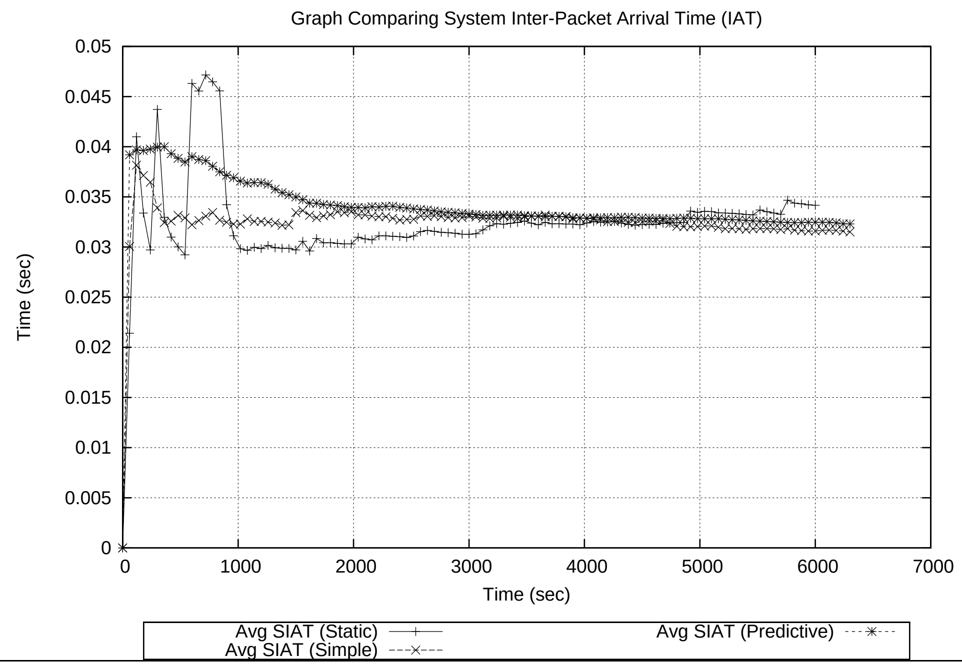


a. Avg. SOWL for Many Flows, Static Nodes, No Congestion, Relaxed Delay, OPP

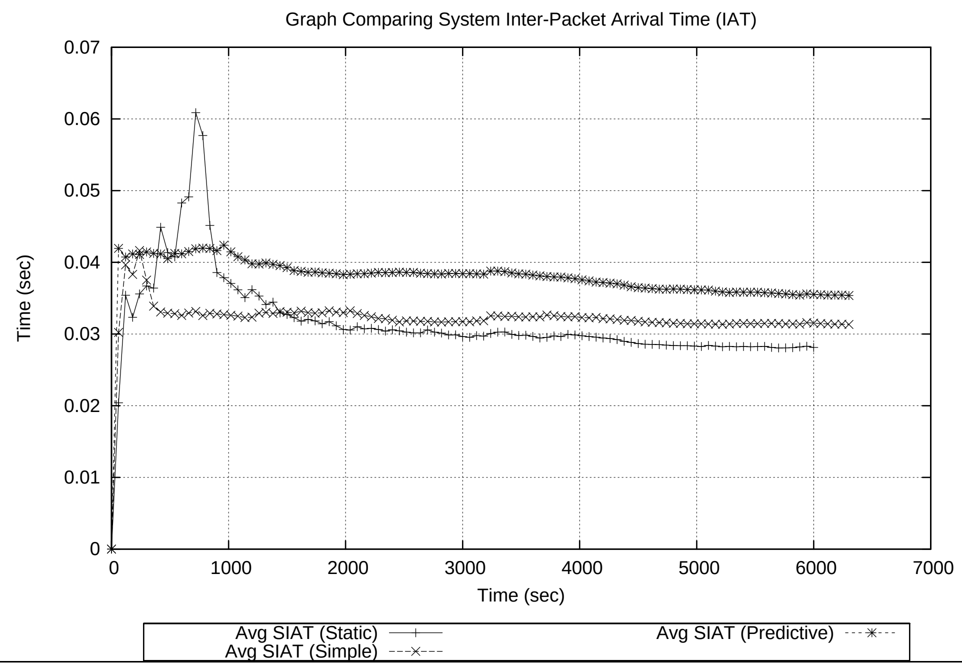


b. Avg. SOWL for Many Flows, Mobile Nodes, No Congestion, Relaxed Delay, OPP

Figure 5.13: Avg. SOWL for Complex Scenarios

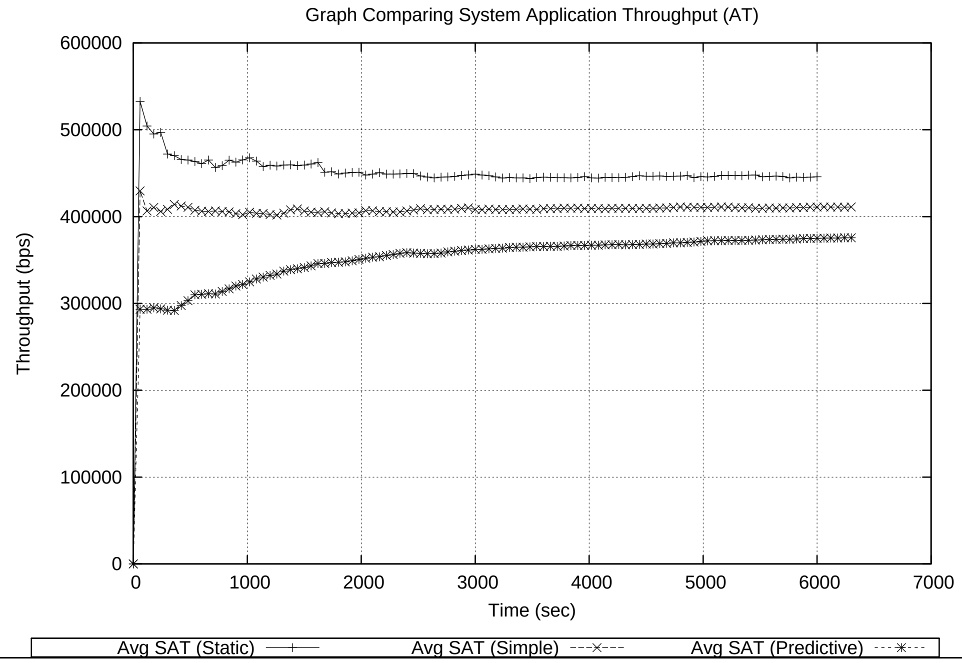


a. Avg. SIAT for Many Flows, Static Nodes, No Congestion, Relaxed Delay, OPP

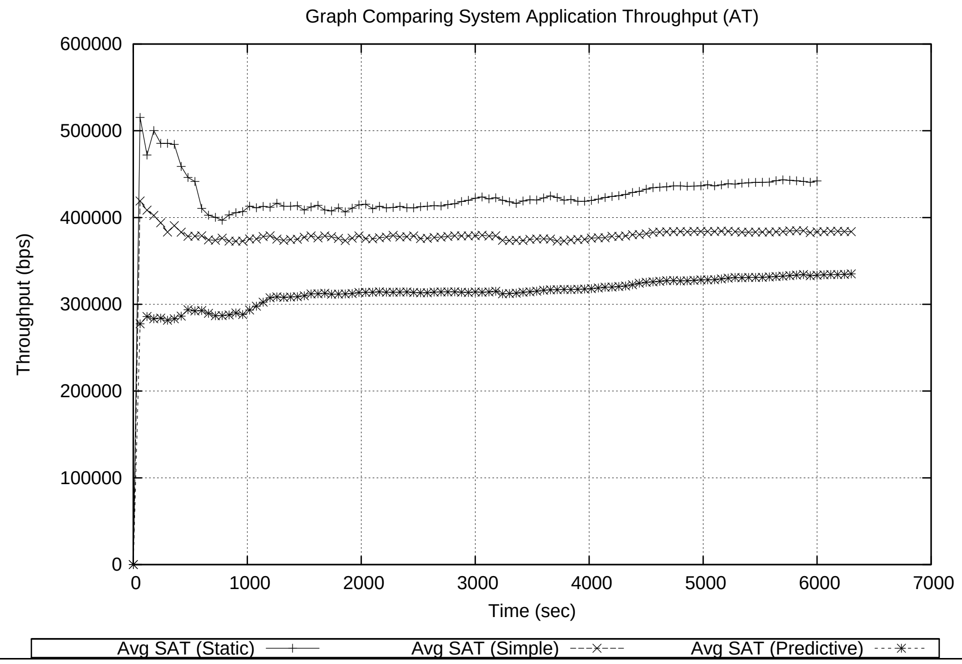


b. Avg. SIAT for Many Flows, Mobile Nodes, No Congestion, Relaxed Delay, OPP

Figure 5.14: Avg. SIAT for Complex Scenarios

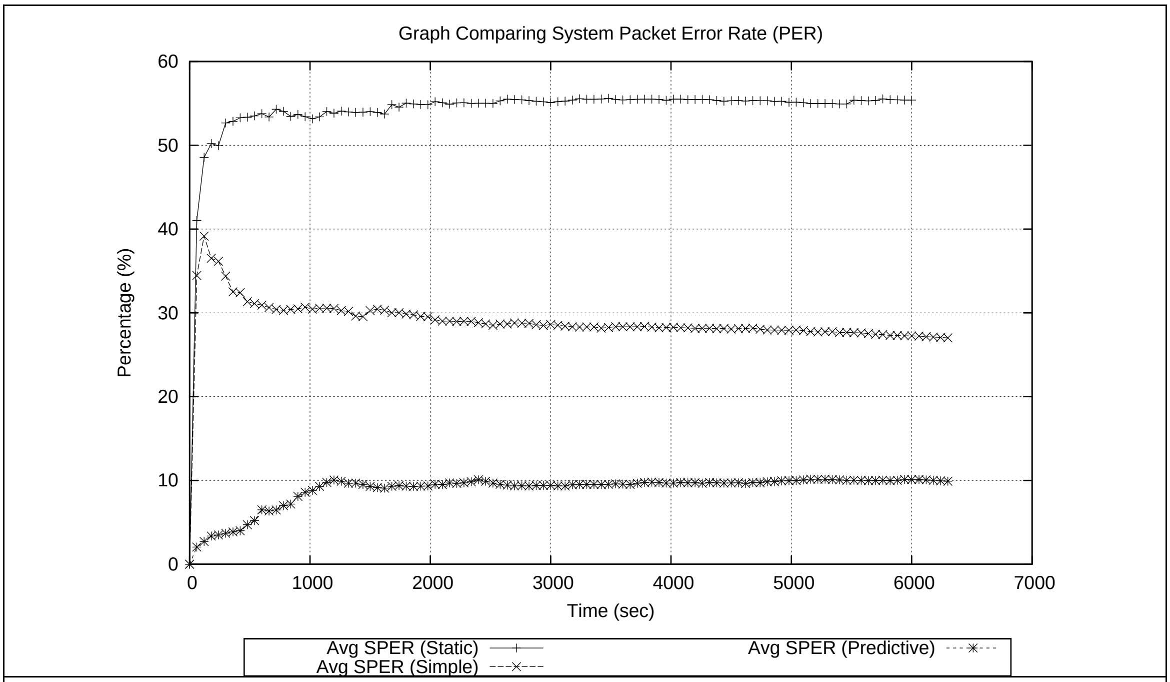


a. Avg. SAT for Many Flows, Static Nodes, No Congestion, Relaxed Delay, OPP

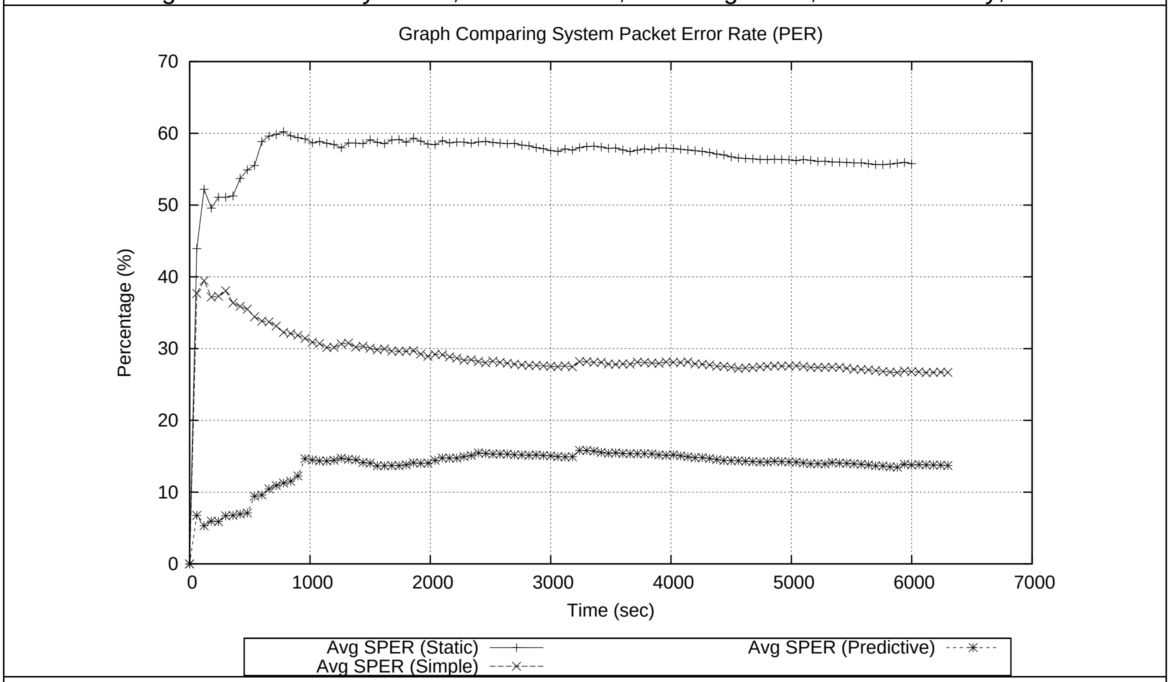


b. Avg. SAT for Many Flows, Mobile Nodes, No Congestion, Relaxed Delay, OPP

Figure 5.15: Avg. SAT for Complex Scenarios



a. Avg. SPER for Many Flows, Static Nodes, No Congestion, Relaxed Delay, OPP



b. Avg. SPER for Many Flows, Mobile Nodes, No Congestion, Relaxed Delay, OPP

Figure 5.16: Avg. SPER for Complex Scenarios

5.5.2 Discussion of Complex Scenarios Experiment Results

For the case of "Multiflows, Static Nodes, No Congestion, Relaxed Delay, OPP", none of the adaptation techniques were able to meet QoS settings, although *Predictive Adaptive Adaptation* came close with an average SOWL of 300 ms (Figure 5.13.a), and an average SPER of 10% (Figure 5.16.a). Since each multicast receiver only receive two of the ten active flows, it is not possible to derive a realistic OSBU for the UDLs. However, it could be seen that the slow ramping of average SAT from 300 kbps to approximately 380 kbps helped maintain system stability (Figure 5.15.a), whereas the *Simple Adaptation* technique had average SAT of 400 kbps throughput the simulation (Figure 5.15.a). This resulted in a high average SOWL of 600 ms (Figure 5.13.a), and average SPER of 30% (Figure 5.16.a).

When mobility is added to the picture, the overall results remained the same, although the absolute values achieved by *Predictive Adaptive Adaptation* deteriorated with 340 kbps average SAT (Figure 5.15.b), 400 ms average SOWL (Figure 5.13.b), and 15% average SPER (Figure 5.16.b). *Simple Adaptation* had some slight improvement in its absolute QoS performance with 390 kbps average SAT (Figure 5.15.b), 500 ms average SOWL (Figure 5.13.b), 27% average SPER (Figure 5.16.b), though it still did not match the performance of *Predictive Adaptive Adaptation*.

SIAT performance (Figure 5.14) was relatively stable for both *Simple Adaptation* and *Predictive Adaptive Adaptation* in both scenarios. This indicated that the Multicast QoS Adaptation Algorithms were stable in the face of congested network conditions.

5.6 Chapter Summary

The various experimental scenarios were presented and the data obtained from the different experiments discussed in detail.

Other than Static QoS, the MQAAs tested were shown to be stable under varying network loads and congestion conditions. However, some fluctuations in jitter (SIAT) resulted from the QoS adaptation process. It was shown that the *Predictive Adaptive Adaptation* technique was robust and flexible under varying network conditions and was able to deliver the best overall QoS performance compared with *Simple Adaptation.*

The summary of research findings, conclusion, and future work is given in Chapter 6.

CHAPTER 6 CONCLUSION AND FUTURE WORK

6.1 Summary of Research Findings

6.1.1 Research Contribution

QoS support for real-time interactive multimedia services over wireless networks is necessary for the development of the next “killer-application.” This support is made more difficult for the case of multicast traffic involving multiple receivers experiencing differing network conditions. Furthermore, the lack of suitable metrics for quantifying multicast performance was a limiting factor in the characterization of the problem domain. This research work sought to address the different constraints in a systematic manner:

- The *definition of new metrics* suitable for quantifying System-wide Multicast QoS Performance, via refinement of existing work done on multicast metrics
- The *definition of a comprehensive framework for wireless multicast QoS* via the specification of a Wireless Multicast QoS Framework (WMQF) and its corresponding components
- The *definition of a new Multicast QoS Adaptation Algorithm* (MQAA) for optimizing overall network utilization while achieving QoS targets of individual multicast sessions
- The *validation of the MQAA* for different *Multicast Performance Profiles* to determine its suitability and effectiveness via simulation techniques, with corresponding results and analyses

6.1.2 Key Features of WMQF

The use of dedicated multicast channels in WMQF enables provisioning of *Proportional Differentiation* QoS for multicast transmission in networks that may not have inherent QoS capabilities. By using a suitable Active Queue Management and Traffic Shaping scheme, namely the *Exponential rule Modified Largest Weighted Delay First* (M-LWDF) algorithm, link bandwidth is shared among competing multicast streams using *Proportional Differentiation* techniques.

Extensions to IGMP to add QoS related feedback messages and signaling enabled the use of close-loop QoS control to be implemented. QoS signaling support are added to QMACC and iMAPs in order to support QoS Adaptation requirements in WMQF. The signaling processes are detailed using signaling diagrams to explain the new features introduced in WMQF.

In addition, layered streams for application level source QoS adaptation are necessary for multicast applications to adapt to changes in network congestion for the source uplink as well as for the downlink to multiple receivers. A new *Multicast QoS Adaptation Algorithm* (MQAA) is introduced in WMQF for addressing the problem of closed-loop source QoS adaptation. Source QoS Adaptation is achieved using QoS Estimation processes controlled by suitable *Multicast Performance Profiles*. The QoS Estimators generate normalized values that are used by a QoS Selector to select new QoS values. In addition, a QoS Shaper enables smooth convergence towards the selected QoS values by generating new QoS target values based on existing QoS settings.

The QoS Adaptation technique used by the *Estimators, Selector* and *Shaper* to converge towards new QoS targets play an important role in ensuring that the overall system utilization and QoS performance is optimized. An incorrect choice of Performance Profiles and QoS Adaptation technique would lead to either low throughput and poor system utilization due to overly conservative and low QoS target values, or else excessive packet loss and long delays due to transmission of packets above the sustainable rate.

6.1.3 Effectiveness of the Multicast QoS Adaptation Algorithm

Through various simulation experiments, the ***Predictive Adaptive QoS Adaptation*** technique was shown to optimize overall system performance of multiple competing multicast streams when used in conjunction with the ***Optimal Performance Profile*** for QoS parameter estimation. This approach is expected to enhance the delivery of interactive multimedia streaming services in next generation wireless networks.

6.2 Constraints and Limitations

6.2.1 Air Interface Constraints

Since the Wireless Multicast QoS Framework was analyzed and simulated using the 802.11-based air interface standard, the results should be considered as a lower bound for achievable wireless QoS performance. This limitation arises because the 802.11 air interface does not inherently support QoS. Consequently, if air interface standards with QoS support such as W-CDMA for 3G cellular systems were adopted, it is expected that the ***Predictive Adaptive QoS Adaptation*** technique would be able to achieve much better compliance to specified QoS targets. At the same time, overall system performance is expected to improve as well.

6.2.2 Multicast Source Model

The multicast source model used in WMQF is based on a Dynamic Constant Bit Rate (DynCBR) model instead of a true multilayered multicast source model. Consequently, the bandwidth adaptation for DynCBR results in much coarser adaptation steps compared to that provided by an actual multilayered multicast source. Nonetheless, the DynCBR model is useful for a first approximation that is much simpler to implement. It is expected that the use of a true multilayered multicast source model would result in a smoother QoS Adaptation process.

6.2.3 Scalability Issues

Since the QoS Adaptation process relies on QoS Feedback messages received from QMACCs, it is subject to a phenomenon known as “feedback implosion” where the network cannot scale due to too many participating nodes. Nonetheless, the architecture in WMQF is inherently able to reduce the impact of “feedback implosion” since QoS feedback messages are generated only by QMACCs and not directly by receiving nodes. This reduced the scalability constraint to one where QMACCs can also play a role in aggregating QOS feedback messages received via the core network, before they are forwarded to the mobile source nodes. Two approaches can be taken for feedback message aggregation:

- In a hierarchical multicast tree, intermediate nodes on the multicast tree performs feedback aggregation for nodes beyond a specified depth. This approach requires in-tree multicast routers to be *QoS Feedback-aware*
- If only QMACCs are involved in QoS Feedback aggregation, *QoS Feedback* messages that are received via the core network for forwarding to a given cell are subjected to a token based rate-limiter. Multiple *QoS Feedback* messages received before a token is available are summarized into a single pending *Aggregated QoS Feedback* message, such that per cell *QoS Feedback* message forwarding is performed at a predefined rate for a given QoS Adaptation interval. This would not involve in-tree multicast routers

6.3 Future Work

The assumption of wireless networks having limited or no QoS support is slowly being addressed in the next generation of wireless network technologies, for example, in the IEEE 802.11e standard, to incorporate traffic prioritization mechanisms. Consequently, the inclusion of Hybrid Coordination Function (HCF) models in 802.11e into the WMQF simulation environment to determine its impact on overall system utilization and performance would be a logical extension of this research work.

As new wireless technologies and media streaming applications appear in the market, the comparison between predicted throughputs obtained through this research with achievable performance using actual deployed equipment would provide a good cross-validation of the WMQF models as well as identify areas of optimization that can be pursued within such systems.

Furthermore, the extension of this research towards ad hoc networks where the lack of high bandwidth infrastructure backbone networks and the presence of time-varying connectivity between nodes would increase the complexity of QoS adaptation requirements. It is expected that enhancements to the WMQF models and new techniques for QoS estimation and adaptation would be required for such ad hoc wireless network scenarios.

REFERENCES

Acharya, A. et al. (1997) Mobility Management in wireless ATM Networks. *IEEE Communications Magazine*. Vol. 35 No. 11, Nov. 1997. pp. 100-109.

Andren, C. (1997) A Comparison of Frequency Hopping and Direct Sequence Spread Spectrum Modulation for IEEE 802.11 Applications at 2.4 GHz [Internet]. Harris Corp. Available from: <http://www.sss-mag.com/pdf/ds-v-fh.pdf> [URL revised May 6, 2004]

Andrews, M. et al. (2000) CDMA Data QoS Scheduling on the Forward Link with Variable Channel Conditions. Bell Labs Technical Report. USA: Bell Labs.

Andrews, M. et al. (2001) Providing Quality of Service over a Shared Wireless Link. *IEEE Communications Magazine.* Vol. 39, No. 2, Feb. 2001. pp. 150-154.

Bagrodia, R. et al. (1998) Parsec: A Parallel Simulation Environment for Complex Systems. *Computer Magazine.* Vol. 31, No. 10, Oct. 1998. pp. 77-85.

Bain, G. & Seitz, N. (2004) *Mapping between ITU-T (Y.1541/Y.1221) and 3GPP (TS 23-107) QoS Classes and Traffic Descriptors.* ITU-T SG13 WP4 Q6/13 Meeting, Geneva, Switzerland. 3-13 Feb. 2004. Available from: <http://www.3gpp.org/ftp/tsg_sa/TSG_SA/TSGS_23/Docs/PDF/SP-040009.pdf> [URL revised May 6, 2004]

Bernet, Y. (2000) The Complementary Roles of RSVP and Differentiated Services in the Full-Service QoS Network. *IEEE Communications Magazine*. Vol. 38, No. 2, Feb. 2000. pp. 154-162.

Bianchi, G. et al. (2003) QUASIMODO: Quality of Service-Aware Multicasting over DiffServ and Overlay Networks. *IEEE Network.* Vol. 17, No. 1, Jan. 2003. pp. 38 – 45.

Blake, S. et al. (1998) *An Architecture for Differentiated Services [Internet].* RFC 2475. IETF. Available from: <http://www.ietf.org/rfc/rfc2475.txt> [URL revised May 6, 2004]

Braden, R. et al. ed. (1997) *Resource ReSerVation Protocol (RSVP) – Version 1 Functional Specification [Internet].* RFC 2205. IETF. Available from: <http://www.ietf.org/rfc/rfc2205.txt> [URL revised May 6, 2004]

Cain, B. et al. (2002) *Internet Group Management Protocol, Version 3 [Internet].* RFC 3376. IETF. Available from: <http://www.ietf.org/rfc/rfc3376.txt> [URL revised May 6, 2004]

Chalmers, D. & Sloman, M. (1999) A Survey of Quality of Service in Mobile Computing Environments. *IEEE Communications Surveys [Internet]* Second Quarter 1999, Vol. 2, No. 2. Available from: <http://www.comsoc.org/livepubs/surveys/index.html> [URL revised May 6, 2004]

Chen, S. Nahrstedt, K. & Shavitt, Y. (2000) A QoS-Aware Multicast Routing Protocol. *IEEE J. Selected Areas in Communications (JSAC),* Vol. 18, No. 12, Dec. 2000. pp 2580 – 2592.

Chen, Y. et al. (2003) Proportional Differentiation: A Scalable QoS Approach. *IEEE Communications Magazine.* Vol. 41, No. 6, June 2003. pp. 52-58.

Cheng, M. et al. (1997) PCS Mobility Support over Fixed ATM Networks. *IEEE Communications Magazine*. Vol. 35 No. 11, Nov. 1997. pp. 82-92.

Clerget, A. (1999) *TUF: A Tag-based UDP Multicast Flow Control Protocol.* Project Rodeo Research Report 3728. Rocquencourt, France: INRIA.

Davis, R. (1999) *ATM for Public Networks.* New York, NY, USA: McGraw-Hill.

Deering, S. (1989) *Host Extensions for IP Multicasting [Internet].* RFC 1112. IETF. Available from: <http://www.ietf.org/rfc/rfc1112.txt> [URL revised May 6, 2004]

Diot, C. Dabbous, W. & Crowcroft, J. (1997) Multipoint Communications: A Survey of Protocols, Functions and Mechanisms. *IEEE J. Selected Areas in Communications (JSAC),* Vol. 15, No. 3, April 1997. pp. 277 – 290.

Dixit, S. Guo, Y. & Antoniou, Z. (2001) Resource Management and Quality of Service in Third-Generation Wireless Networks. *IEEE Communications Magazine.* Vol. 39, No. 2, Feb. 2001. pp. 125-133.

Dovrolis, C. & Ramanathan, P. (1999) A Case for Relative Differentiated Services and the Proportional Differentiation Model. *IEEE Network.* Vol. 13, Sep./Oct. 1999. pp. 26 – 34.

Dovrolis, C. & Ramanathan, P. (2000) Proportional Differentiated Services, Part II: Loss Rate Differentiation and Packet Dropping. *IN: Proc. Int'l Workshop on Quality of Service (IWQoS)*, June 2000. pp. 52 – 61.

Dovrolis, C. Stiliadis, D. & Ramanathan, P. (2002) Proportional Differentiated Services: Delay Differentiation and Packet Scheduling. *IEEE/ACM Trans. Networking*, Vol. 10, No. 1, Feb 2002.

Dutta, A. & Schulzrinne, H. (2004) MarconiNet: Overlay Mobile Content Distribution Network. *IEEE Communications Magazine.* Vol. 42, No. 2, Feb. 2004. pp. 64-75.

Engim Inc. (2004) <http://www.engim.com> [URL revised May 6, 2004]

Fenner, W. (1997) *Internet Group Management Protocol, Version 2 [Internet].* RFC 2236. IETF. Available from: <http://www.ietf.org/rfc/rfc2236.txt> [URL revised May 6, 2004]

Fitzek, F. Morich, R. & Wolisz, A. (2000) Comparison of Multi-Code Link-Layer Transmission Strategies in 3G Wireless CDMA. *IEEE Communications Magazine.* Vol. 38, No. 10, Oct. 2000. pp. 58-64.

Freeman, J. & Skapura, D. (1991) *Neural Networks: Algorithms, Applications, and Programming Techniques*. Reading, MA, USA: Addison-Wesley.

Gossain, H. Cordeiro, C. & Agrawal, D. Multicast: Wired to Wireless. *IEEE Communications Magazine.* Vol. 40, No. 6, June 2002. pp. 116-123.

Grilo, A. Macedo, M. & Nunes, M. (2003) A Scheduling Algorithm for QoS Support in IEEE802.11e Networks. *IEEE Wireless Communications.* Vol. 10, No. 3, June 2003. pp. 36-43.

Gu, D. & Zhang, J. (2003) QoS Enhancement in IEEE802.11 Wireless Local Area Networks. *IEEE Communications Magazine.* Vol. 41, No. 6, June 2003. pp. 120-124.

Hess, G. (1998) *Handbook of Land-Mobile Radio System Coverage.* Boston, MA, USA: Artech House.

IEEE. (1999a) *Wireless LAN Medium Access Control and Physical Layer Specifications.* IEEE Std 802.11-1999. New York, NY, USA: Inst. Of Electrical & Electronics Engineering.

IEEE. (1999b) Wireless LAN Medium Access Control and Physical Layer Specifications, Higher-Speed Physical Layer Extension in the 2.4 GHz Band. IEEE Std 802.11b-1999. New York, NY, USA: Inst. Of Electrical & Electronics Engineering.

Irey, P. & Marlow, D. (1999) Metrics for the Evaluation of Multicast Communications. *IN: Proc. Workshop on Parallel and Distributed Real-Time Systems (WPDRTS)*, 10th Int'l Parallel and Distributed Processing Symposium (IPDPS) 1999. Apr. 12-16, 1999. San Juan, Puerto Rico.

Jorguseski, L. et al. (2001) Radio Resource Allocation in Third-Generation Mobile Communication Systems. *IEEE Communications Magazine.* Vol. 39, No. 2, Feb. 2001. pp. 117-123.

Koenen, R. ed. (2002) *Coding of Moving Pictures and Audio: Overview of the MPEG-4 Standard [Internet].* ISO/IEC JTC1/SC29/WG11 N4668 V.21. Available from: <http://www.chiariglione.org/mpeg/standards/mpeg-4/mpeg4-htm> [URL revised May 6, 2004]

Kumar, A. Kaur, J. & Vin, H. (2001) End-to-end Proportional Loss Differentiation. *Technical Report TR-01-33*, USA, Dept. of Computer Sciences, U. Texas at Austin.

Mangold, S. et al. (2003) Analysis of IEEE 802.11e for QoS Support in Wireless LANs. *IEEE Wireless Communications*. Vol. 10, No. 6, , Dec. 2003. pp. 40-50.

Mankin, A. et al. ed. (1997) *Resource ReSerVation Protocol (RSVP) – Version 1 Applicability Statement: Some Guidelines on Deployment [Internet]*. RFC 2208. IETF. Available from: <http://www.ietf.org/rfc/rfc2208.txt> [URL revised May 6, 2004]

Naghshineh, M. (1999) Elements of An End-to-End QoS Architecture in Wireless/Mobile Networks. *IN: Leung, K. & Vojcic, B. ed. Multiaccess, Mobility and Teletraffic for Wireless Communications: Vol. 3*. Boston, MA, USA: Kluwer Academic Publishers.

Naghshineh, M. & Willebeek-LeMair, M. (1997) End-to-End QoS Provisioning in Multimedia Wireless/Mobile Networks using an Adaptive Framework. *IEEE Communications Magazine*. Vol. 35 No. 11, Nov. 1997. pp. 72-81.

Nichols, K. et al. (1998) *Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers [Internet]*. RFC2474, IETF. Available from: <http://www.ietf.org/rfc/rfc2474.txt> [URL revised May 6, 2004]

Pattarra-Atikom, W. Krishnamurthy, P. & Banerjee, S. (2003) Distributed Mechanisms for Quality of Service in Wireless LANs. *IEEE Wireless Communications*. Vol. 10, No. 3, June 2003. pp. 26-34.

Perkins, C. (1997) Mobile IP. *IEEE Communications Magazine*. Vol. 35 No. 5, May 1997. pp. 84-99.

Ramakrishnan, K. (2001) *The Addition of Explicit Congestion Notification (ECN) to IP.* RFC 3168. The Internet Society. Available from: <http://www.ietf.org/rfc/rfc3168.txt> [URL revised May 6, 2004]

Ramalho, M. (2000) Intra- and Inter-Domain Multicast Routing Protocols: A Survey and Taxonomy. *IEEE Communications Surveys [Internet]* First Quarter 2000, Vol. 3, No. 1. Available from: <http://www.comsoc.org/livepubs/surveys/index.html> [URL revised May 6, 2004]

Rejaie, R. Handley, M. & Estrin, D. (2000) Layered Quality Adaptation for Internet Video Streaming. *IEEE J. Selected Areas in Communications (JSAC),* Vol. 18, No. 12, Dec. 2000. pp. 2530-2543.

Roca, V. (2000) *On-Demand Layer Addition (ODL): Making Multi-Layer Multicast Transmissions Cheaper.* Project PLANETE Technical Report 0239. Rocquencourt, France: INRIA.

Seitz, N. (2003) ITU-T QoS Standards for IP-Based Networks. *IEEE Communications Magazine.* Vol. 41, No. 6, June 2003. pp. 82-89.

Shakkottai, S. & Stolyar, A. (2001) Scheduling Algorithms for a Mixture of Real-Time and Non-Real-Time Data in HDR. *IN: Proc. 17th International Teletraffic Congress (ITC-17)* Sept. 2001. Salvador da Bahia, Brazil.

Sklar, B. (1997a) Rayleigh Fading Channels in Mobile Digital Communication Systems Part I: Characterization. *IEEE Communications Magazine.* Vol. 35 No. 9, Sept. 1997. pp. 136-146.

Sklar, B. (1997b) Rayleigh Fading Channels in Mobile Digital Communication Systems Part II: Mitigation. *IEEE Communications Magazine.* Vol. 35 No. 9, Sept. 1997. pp. 148-155.

Sklar, B. (1997c) A Primer on Turbo Code Concepts. *IEEE Communications Magazine.* Vol. 35 No. 12, Dec. 1997. pp. 94-102.

Stallings, W. (1998) *High-speed networks: TCP/IP and ATM design principles*. Upper Saddle River, NJ, USA: Prentice-Hall.

Steele, R. & Hanzo, L. ed. (1999) *Mobile Radio Communications. 2nd Ed.,* Chichester, West Sussex, England: J. Wiley & Sons.

Stephan, E. (2002) *Measuring the performance of Multicast Services [Internet].* 54th IETF- Meeting, MAGMA (Multicast and Anycast Group Membership) WG, Available from: <http://www.ietf.org/internet-drafts/draft-stephan-ippm-multicast-metrics-00.txt> [URL revised May 6, 2004]

Striegel, A. & Manimaran, G. A Survey of QoS Multicasting Issues. *IEEE Communications Magazine.* Vol. 40, No. 6, June 2002. pp. 82-87.

Turletti, T. Parisis, S. & Bolot, J-C. (1997) *Experiments with a Layered Transmission Scheme over the Internet.* Project Rodeo Research Report 3296. Rocquencourt, France: INRIA.

UCLA Parallel Computing Laboratory. (2001) *Global Mobile Info. Sys. Sim. Library (GloMoSim) 2.0 [Internet],* Available from: <http://pcl.cs.ucla.edu/projects/glomosim> [URL revised May 6, 2004]

Wang, B. & Hou, J. (2000) Multicast Routing and its QoS extension: Problems, algorithms, and protocols. *IEEE Network.* Vol. 14, No. 1, Jan/Feb 2000. pp. 22 – 36.

Wong, W. Zhu, H. & Leung, V. (2003) Soft QoS Provisioning Using the Token Bank Fair Queuing Scheduling Algorithm. *IEEE Wireless Communications.* Vol. 10, No. 3, June 2003. pp. 8-16.

Wydrowski, B. & Zukerman, M. (2002) QoS in Best-Effort Networks. *IEEE Communications Magazine.* Vol. 40, No. 12, Dec. 2002. pp. 44-49.

Xiao, X. & Ni, L. M. (1999) Internet QoS: the Big Picture. *IEEE Network.* Vol. 13, Mar. 1999.

Xylomenos, G. & Polyzos, G. (1997) IP Multicast for Mobile Hosts. *IEEE Communications Magazine.* Vol. 35, No. 1, Jan. 1997.

Yamamoto, K. et al. (2000) Performance Evaluation of ACK-based and NAK-based Flow Control Schemes for Reliable Multicast. *IN: Proc. TENCON 2000 Vol. I*. Sep. 24-27, 2000. Kuala Lumpur, Malaysia. IEEE. pp. 341-345.

Yamamoto, M. et al. (1998) NAK-based Flow Control Scheme for Reliable Multicast Communications. *IN: Proc. IEEE Globecom '98*. Nov. 1998.Vol. 5, pp 2611-2616.

Yan, S. Faloutsos, M. & Banerjea, A. (2002) QoS-Aware Multicast Routing for the Internet: The Design and Evaluation of QoSMIC. *IEEE/ACM Trans. Networking.* Vol. 10, No. 1, Feb. 2002. pp. 54 – 66.

APPENDIX A RAYLEIGH AND RICIAN FADING

A.1 Probability Density Functions

Typically Rayleigh and Rician fading phenomena are described using probability density functions (PDF) for average transmitted power (Hess, 1998).

Rayleigh PDF is defined as follows:

$$f_S(s) = \tfrac{1}{\alpha}\exp\left(-\tfrac{s}{\alpha}\right), \quad s \geq 0, \alpha = avg.\,power$$

..... **Equation 9**

Rician PDF is defined as follows:

$$\begin{aligned}
f_S(s) \quad &= \tfrac{1}{\alpha}\exp\left(-\tfrac{s}{\alpha}\right)\exp(-k^2)I_0\left(\tfrac{2k\sqrt{s}}{\sqrt{\alpha}}\right) \\
where \quad & s \geq 0, k^2 = \tfrac{(c^2/2)}{\alpha}, I_0():modified\ Bessel\ function\ order\ 0 \\
& \alpha = avg.diffused\ (Rayleigh)\ power, \tfrac{c^2}{2} = avg.direct\ (const.)\ power \\
Let\ (\alpha + \tfrac{c^2}{2}) &= 1\ (avg.\,total\ power) \\
we\ get\ \alpha &= 1/(1+k^2) \\
f_S(s) \quad &= (k^2+1)\exp\left[-s(k^2+1)\right]\exp(-k^2)I_0\left(2k\sqrt{k^2+1}\sqrt{s}\right), s \geq 0
\end{aligned}$$

...... **Equation 10**

When k is near zero, the signal behavior approximates Rayleigh fading. i.e., Rayleigh fading (Equation 9) is a special case of Rician fading (Equation 10)

A.2 Rayleigh Power Equation

In addition, the pdf for Rayleigh fading can be specified in dB, S_{dB} (Hess, 1998), where:

$$\begin{aligned}
f_S(s) &= \tfrac{1}{\alpha}\exp\left(-\tfrac{s}{\alpha}\right), \quad s \geq 0, \alpha = avg.\,power \\
f_{S_{dB}}(S_{dB}) &= f_s(s)/\left|\frac{ds_{dB}}{ds}\right| \\
&= \tfrac{1}{\kappa\alpha}\exp\left[\tfrac{s_{dB}}{\kappa} - \tfrac{1}{\alpha}\exp\left(\tfrac{S_{dB}}{\kappa}\right)\right]
\end{aligned}$$

...... **Equation 11**

Therefore, average (mean) power is given as:

$$
\begin{aligned}
\langle S_{dB} \rangle &= \langle 10\log(s) \rangle \\
&= \kappa \langle \ln(s) \rangle \\
&= \kappa \int_0^{\infty} \tfrac{\ln(s)}{\alpha} \exp\left(-\tfrac{s}{\alpha}\right) ds \\
&= \kappa \left[\ln(\alpha) - \gamma \right] \\
\textit{where} \quad & \gamma = 0.5722 \ (Euler's\ const), \\
& \alpha = Rayleigh\ dist.\ (avg.\ pwr),\ \kappa = 10/\ln(10)
\end{aligned}
$$

...... **Equation 12**

APPENDIX B MULTICAST METRICS DEFINITION

B.1 Multicast vs. Unicast Metrics

As explained in Chapter 2.5.3, work on defining multicast metrics is an ongoing research issue. In contrast, unicast metrics are well defined and widely adopted. (Davis, 1999). Consequently, the work of Irey & Marlow (1999) is an important step towards quantifying multicast QoS behavior.

Irey & Marlow defined multicast metrics as affecting single nodes (Local Metrics), and for a given multicast group (Group Metrics). Local Metrics were defined for the multicast source (Transmitter), as well as the various multicast sinks (Receiver). In contrast, Group Metrics were defined solely for multicast sinks since each multicast group was assumed to have a single source.

Nonetheless, Irey & Marlow did not define multicast metrics for Bit Error Rate (BER) or Packet Error Rate (PER). Consequently, in the process of defining suitable metrics to measure the performance of several competing multicast groups within a given wireless network (System Metrics) (Chapter 2.5.5), formulae for Local PER (LPER), Group PER (GPER), and System PER (SPER) were derived. These local, group and system PER metrics provide equivalent measures to well known unicast QoS PER metrics (Davis, 1999).

The various Local and Group Metrics are given in Chapters B.2 and B.3.

B.2 Local (Single Node) Metrics

B.2.1 Local Transmitter Multicast Metrics

Local Transmitter Multicast Metrics measure the multicast source performance for the transmitting node (Table B.1):

Table B.1: Local Transmitter Multicast Metrics (Irey & Marlow, 1999)

Transmit (Sender) Metrics		
Metric	***Name***	***Definition***
LMS	Local Messages Sent	Count of number of messages transmitted to a given address
$LIST_i$	Local Inter-Send Time	Time between successive message transmission for transmitter i
$LIST_{avg}$	Average LIST	$LIST_{avg} = \sum_{i=1}^{m} \frac{LIST_i}{m}$, given *m* messages
$LIST_{\max}$	Max. LIST	$LIST_{\max} = \underset{\forall i}{MAX}\{LIST\}$
$LIST_{\min}$	Min. LIST	$LIST_{\min} = \underset{\forall i}{MIN}\{LIST\}$
$LIST_{sdev}$	Std. Dev. LIST	$LIST_{sdev} = \sqrt{\left(\sum_{i=1}^{m}(LIST_i - LIST_{avg})^2\right) \Big/ (n-1)}$, given *m* messages, *n* receivers

B.2.2 Local Receiver Multicast Metrics

Local Receiver metrics measure the respective multicast sink performance (Table B.2):

Table B.2: Local Receiver Multicast Metrics (Irey & Marlow, 1999)

Receive (Receiver) Metrics		
Metric	***Name***	***Definition***
LMR_j	Local Messages Received	Count of number of messages received with sequence number (seqno) ≥ expected seqno at Receiver j
$LMRL_j$	Local Messages Received Late	Count of number of messages received with seqno < expected seqno at Receiver j
$LMRI_j$	Local Messages Received In-order	Count of number of messages received with seqno = expected seqno at Receiver j $LMRI_j = LMR_j - LGN_j$
$LPMR_j$	Local Percent Messages Received	$LPMR_j = LMR_j / LMS$
LGB_j	Local Gap Boundaries (Set)	Set of ordered pairs of (start, end) of gaps in the sequence space of received messages (LMR) at Receiver j.

Table B.2, continued.

Metric	***Name***	***Definition***
LGN_j	Local Gap Number	Count of the sequence space gaps observed for Receiver j
LGL_{j_k}	Local Gap Length	Size of sequence space gaps for Receiver j for k-th gap $LGL_{j_k} = LGB(end)_{j_k} - LGB(start)_{j_k}$
LGL_{avg_j}	Average LGL_j	$LGL_{avg_j} = \left(\sum_{k=1}^{LGN_j} LGL_{j_k} \right) \Big/ LGN_j$, for Receiver j given LGN_j sequence gaps
$LGL_{\max_j}$	Max. LGL_j	$LGL_{\max_j} = \underset{\forall k}{MAX}\{LGL_{j_k}\}$, for Receiver j given LGN_j sequence gaps
$LGL_{\min_j}$	Min. LGL_j	$LGL_{\min_j} = \underset{\forall k}{MIN}\{LGL_{j_k}\}$, for Receiver j given LGN_j sequence gaps
$LOWL_{j_i}$	Local One-Way Latency	Latency for receiver j from transmitter, $T_{r_j} - T_s$
$LOWL_{avg_j}$	Average $LOWL_j$	$LOWL_{avg_j} = \sum_{i=1}^{LMR_j} \frac{LOWL_{j_i}}{LMR_j}$, for receiver j given LMR_j messages
$LOWL_{\max_j}$	Max. $LOWL_j$	$LOWL_{\max_j} = \underset{\forall i}{MAX}\{LOWL_{j_i}\}$, for receiver j
$LOWL_{\min_j}$	Min. $LOWL_j$	$LOWL_{\min_j} = \underset{\forall i}{MIN}\{LOWL_{j_i}\}$, for receiver j
$LOWL_{sdev_j}$	Std. Dev. $LOWL_j$	$LOWL_{sdev_j} = \sqrt{\left(\sum_{i=1}^{LMR_j} (LOWL_{j_i} - LOWL_{avg_j})^2 \right) \Big/ (LMR_j - 1)}$, for receiver j given LMR_j messages
$LIAT_{j_i}$	Local Inter-Arrival Time	Time between successive message reception at receiver j, $LIAT_1$ undefined
$LIAT_{avg_j}$	Average LIAT	$LIAT_{avg_j} = \sum_{i=2}^{LMR_j} \frac{LIAT_{j_i}}{(LMR_j - 1)}$, for receiver j given LMR_j messages
$LIAT_{\max_j}$	Max. LIAT	$LIAT_{\max_j} = \underset{\forall i}{MAX}\{LIAT_{j_i}\}$, for receiver j
$LIAT_{\min_j}$	Min. LIAT	$LIAT_{\min_j} = \underset{\forall i}{MIN}\{LIAT_{j_i}\}$, for receiver j
$LIAT_{sdev_j}$	Std. Dev. LIAT	$LIAT_{sdev_j} = \sqrt{\left(\sum_{i=2}^{LMR_j} (LIAT_{j_i} - LIAT_{avg_j})^2 \right) \Big/ (LMR_j - 2)}$, for receiver j given LMR_j messages
LAT_j	Local Application Throughput	$LAT_j = \frac{Total\ Bytes\ Received}{T_{rlast} - T_{s1}}$, for receiver j
$LPER_j$	Local Packet Error Rate	$LPER_j = \left(\sum_{k=1}^{LGN_j} LGL_{j_k} \right) \Big/ \left(M_{SeqEnd} - M_{SeqStart} + 1 \right)_j$, for receiver j

B.3 Group (One Transmitter, Multiple Receiver) Metrics

B.3.1 Group Multicast Metrics

Group Multicast Metrics measure receiver multicast performance over a given multicast group (Table B.3):

Table B.3: Group (One-to-Many) Multicast Metrics (Irey & Marlow, 1999)

Group (Receiver) Metrics		
Metric	***Name***	***Definition***
GGN	Group Gap Number	Aggregated Gap Number
GGN_{avg}	Average GGN	$GGN_{avg} = \sum_{j=1}^{n} \frac{LGN_j}{n}$, for n receivers
$GGN_{\max}$	Max. GGN	$GGN_{\max} = \underset{\forall j}{MAX}\{LGN_j\}$, for n receivers
$GGN_{\min}$	Min. GGN	$GGN_{\min} = \underset{\forall j}{MIN}\{LGN_j\}$, for n receivers
GGN_{sdev}	Std. Dev. GGN	$GGN_{sdev} = \sqrt{\left(\sum_{j=1}^{n} (LGN_j - GGN_{avg})^2\right) \Big/ (n-1)}$, for n receivers
GGL	Group Gap Length	Aggregated Gap Length
GGL_{avg}	Average GGL	$GGL_{avg} = \sum_{j=1}^{n} \frac{LGL_{avg_j}}{n}$, for n receivers
$GGL_{\max}$	Max. GGL	$GGL_{\max} = \underset{\forall j}{MAX}\{LGL_{\max_j}\}$, for n receivers
$GGL_{\min}$	Min. GGL	$GGL_{\min} = \underset{\forall j}{MIN}\{LGL_{\min_j}\}$, for n receivers
GGL_{sdev}	Std. Dev. GGL	$GGL_{sdev} = \sqrt{\left(\sum_{j=1}^{n} (LGL_{avg_j} - GGL_{avg})^2\right) \Big/ (n-1)}$, for n receivers
$GOWL$	Group One-Way Latency	Aggregated One Way Latency
$GOWL_{avg}$	Average GOWL	$GOWL_{avg} = \sum_{j=1}^{n} \frac{LOWL_{avg_j}}{n}$, for n receivers
$GOWL_{\max}$	Max. GOWL	$GOWL_{\max} = \underset{\forall j}{MAX}\{LOWL_{\max_j}\}$, for n receivers
$GOWL_{\min}$	Min. GOWL	$GOWL_{\min} = \underset{\forall j}{MIN}\{LOWL_{\min_j}\}$, for n receivers
$GOWL_{sdev}$	Std. Dev. GOWL	$GOWL_{sdev} = \sqrt{\left(\sum_{j=1}^{n} (LOWL_{avg_j} - GOWL_{avg})^2\right) \Big/ (n-1)}$, for n receivers

Table B.3, continued.

Metric	***Name***	***Definition***
$GIAT$	Group Inter-Arrival Time	Aggregated Inter-Arrival Time
$GIAT_{avg}$	Average GIAT	$GIAT_{avg} = \sum_{j=1}^{n} \frac{LIAT_{avg_j}}{n}$, for n receivers
$GIAT_{\max}$	Max. GIAT	$GIAT_{\max} = \underset{\forall j}{MAX}\{LIAT_{\max_j}\}$, for n receivers
$GIAT_{\min}$	Min. GIAT	$GIAT_{\min} = \underset{\forall j}{MIN}\{LIAT_{\min_j}\}$, for n receivers
$GIAT_{sdev}$	Std. Dev. GIAT	$GIAT_{sdev} = \sqrt{\left(\sum_{j=1}^{n}(LIAT_{avg_j} - GIAT_{avg})^2\right) \Big/ (n-1)}$, for n receivers
GAT	Group Application Throughput	Aggregated Application Throughput
GAT_{avg}	Average GAT	$GAT_{avg} = \sum_{j=1}^{n} \frac{LAT_j}{n}$, for n receivers
$GAT_{\max}$	Max. GAT	$GAT_{\max} = \underset{\forall j}{MAX}\{LAT_j\}$, for n receivers
$GAT_{\min}$	Min. GAT	$GAT_{\min} = \underset{\forall j}{MIN}\{LAT_j\}$, for n receivers
GAT_{sdev}	Std. Dev. GAT	$GAT_{sdev} = \sqrt{\left(\sum_{j=1}^{n}(LAT_j - GAT_{avg})^2\right) \Big/ (n-1)}$, for n receivers
$GPER$	Group Packet Error Rate	Aggregated Packet Error Rate
$GPER_{avg}$	Average GPER	$GPER_{avg} = \sum_{j=1}^{n} \frac{LPER_j}{n}$
$GPER_{\max}$	Max. GPER	$GPER_{\max} = \underset{\forall j}{MAX}\{LPER_j\}$, for n receivers
$GPER_{\min}$	Min. GPER	$GPER_{\min} = \underset{\forall j}{MIN}\{LPER_j\}$, for n receivers
$GPER_{sdev}$	Std. Dev. GPER	$GPER_{sdev} = \sqrt{\left(\sum_{j=1}^{n}(LPER_j - GPER_{avg})^2\right) \Big/ (n-1)}$, for n receivers

B.3.2 Multicast Message Reception Metrics

Multicast Message Reception metrics provide correlation metrics to measure how many receivers were able to receive a given multicast packet directed towards the group (Table B.4):

Table B.4: Multicast Message Reception Metrics (Irey & Marlow, 1999)

Correlation (Receiver) Metrics		
Metric	***Name***	***Definition***
$\vec{V}_j$	Reception Vector	V_{ji} = 1 if message seqno i is received, 0 otherwise, for receiver j. $\vec{V}_j$ has m components (messages).
R	Reception Report	Matrix where each column is $\vec{V}_j$ for receiver j. R has n columns (receivers), m rows (messages). Row i is a reception report for message i.
MRC_i	Message Reception Correlation	$MRC_i = \left\| \left(\sum_{j=1}^{n} V_{ji} \right) - \left(n - \sum_{j=1}^{n} V_{ji} \right) \right\| \Big/ n$ $\Rightarrow MRC_i = \left\| 2\sum_{j=1}^{n} V_{ji} - n \right\| \Big/ n$ Measure of number of receivers which received message i vs. receivers which did not. 0 < MRC_i < 1. For MRC_i =1, either all receivers received the message or did not. For MRC_i = 0, half of the receivers received the message while the other half did not.
GRC	Group Reception Correlation	$GRC = \sum_{i=1}^{m} \frac{MRC_i}{m}$ $\Rightarrow GRC = \left(\sum_{i=1}^{m} \left\| 2\sum_{j=1}^{n} V_{ji} - n \right\| \right) \Big/ (n \times m)$ Measure of the degree of correlation in the messages received by a group. 0 < GRC < 1. A low GRC indicates that receivers are making measurements on different sets of samples. This affects the accuracy of gap-based metrics, e.g.,. LIAT.

APPENDIX C DETAILED SIMULATION RESULTS

C.1 Overview of Simulation Results

The following scenarios were simulated using WMQFsim:

- Model Verification Experiments
- WMQF Performance Profile Evaluation
- Optimal Performance Profile Mobility Trials
- Complex Scenarios

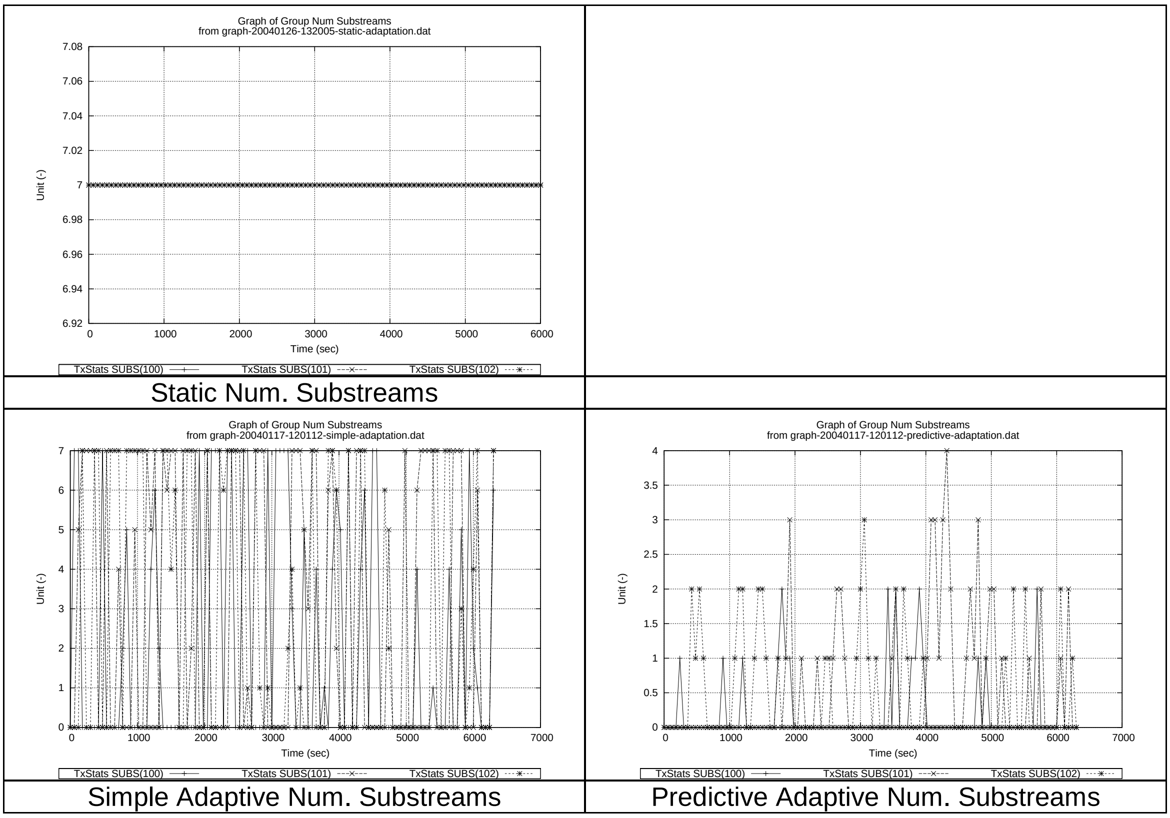
C.2 Model Verification Experiments

Model verification was performed on stationary nodes in order to verify that the Multicast Performance Profiles, Multicast QoS Adaptation Algorithms and underlying simulation models were functioning correctly. The verification experiments would also provide some indication regarding the suitability of the abovementioned algorithms for adapting Internet-based multicast traffic.

C.2.1 Static Nodes, No congestion, Tight Delay, OPP

Table C.1: Simulation Parameters for nomobility-300 kbps

Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams

Simple Adaptive Num. Substreams

Predictive Adaptive Num. Substreams

Figure C.1: Layered Substreams Count for nomobility-300 kbps

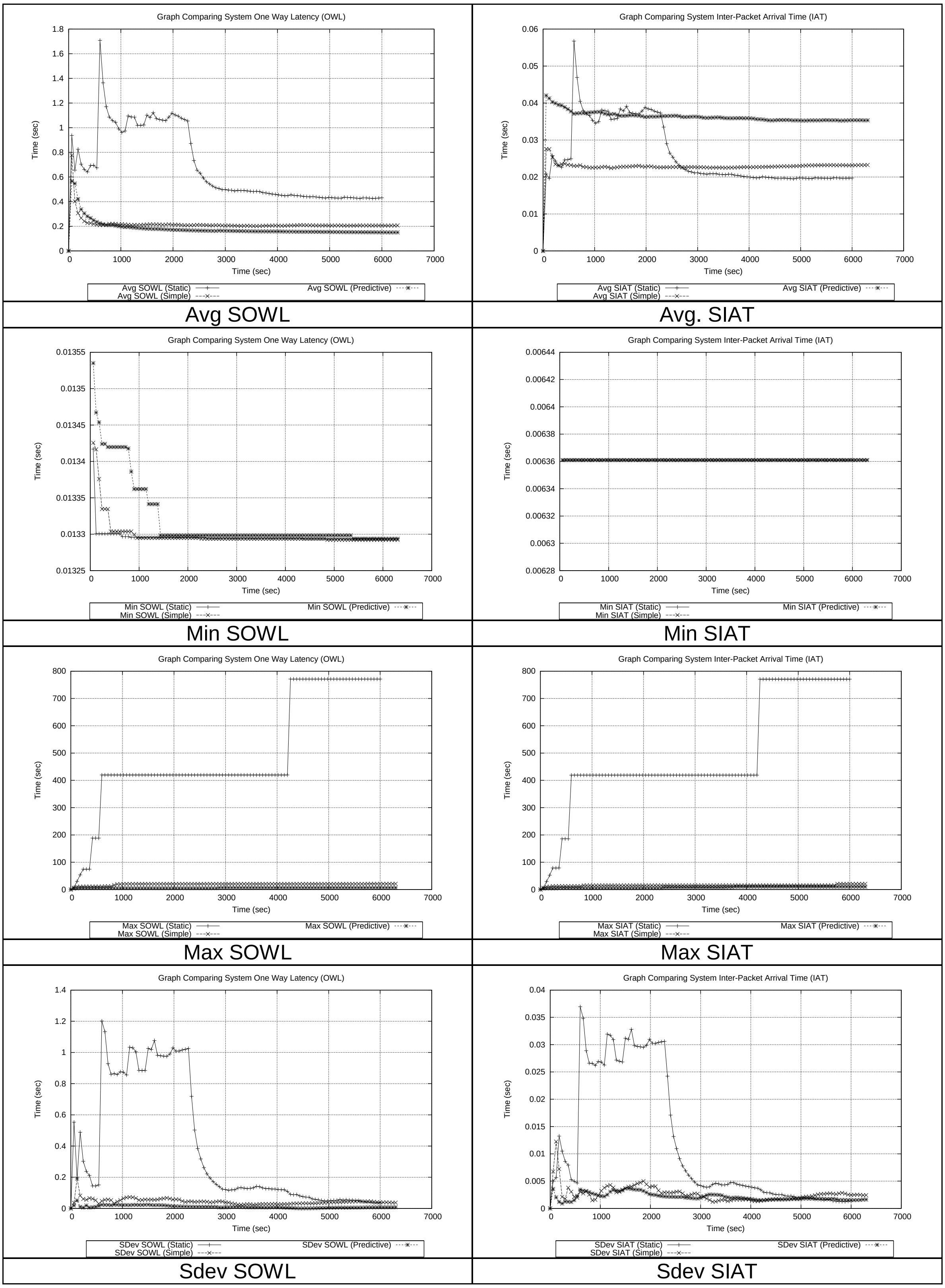


Figure C.2: SOWL and SIAT graphs for nomobility-300 kbps

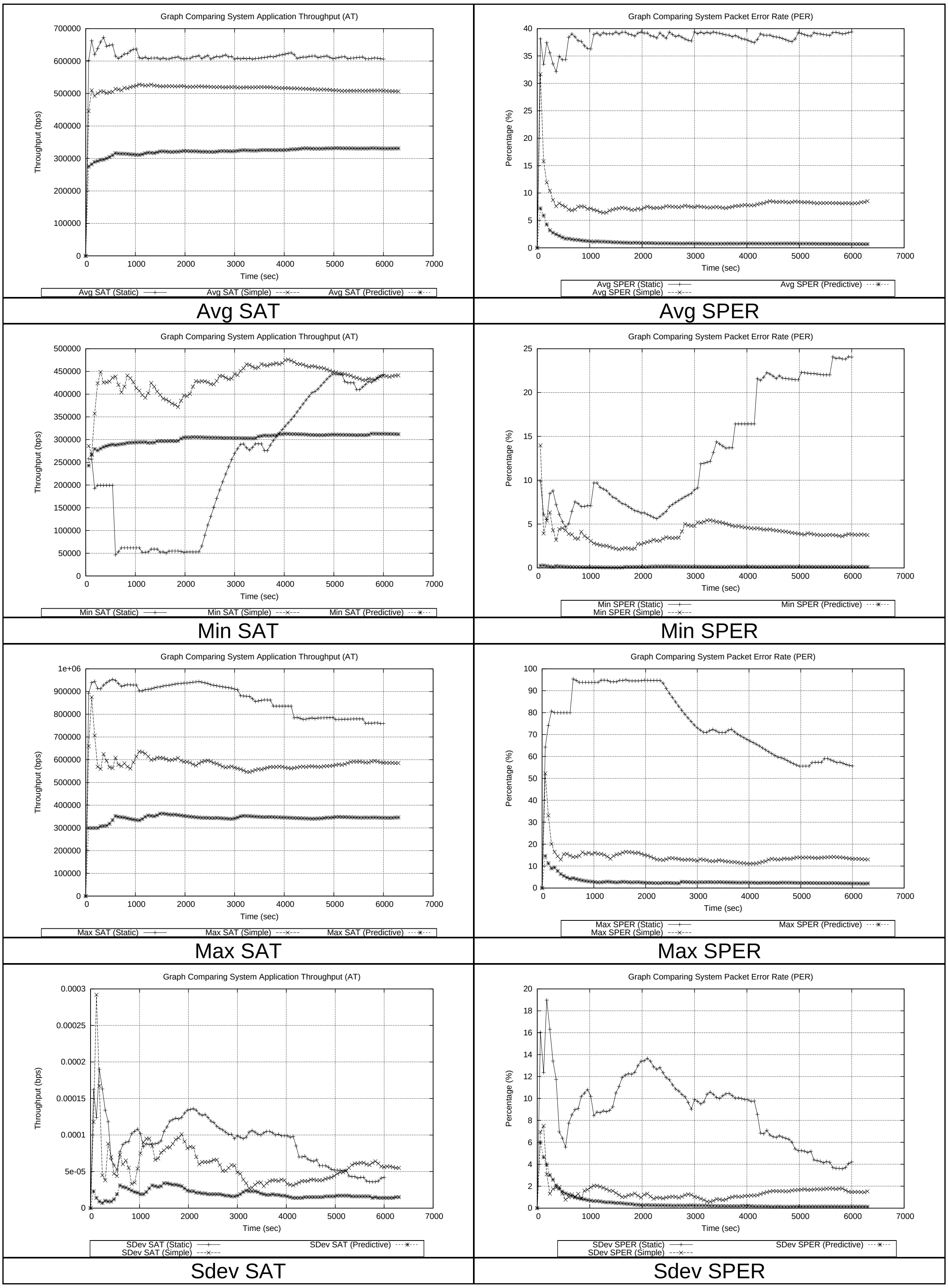
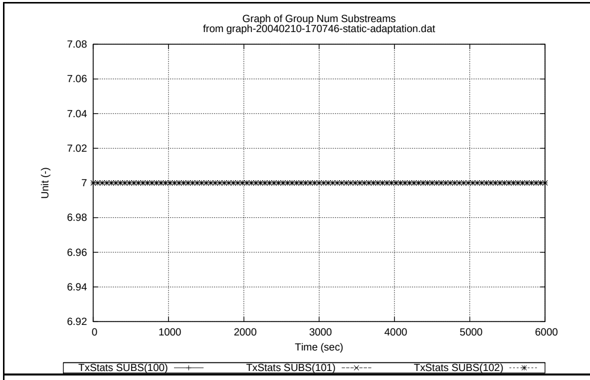


Figure C.3: SAT and SPER graphs for nomobility-300 kbps

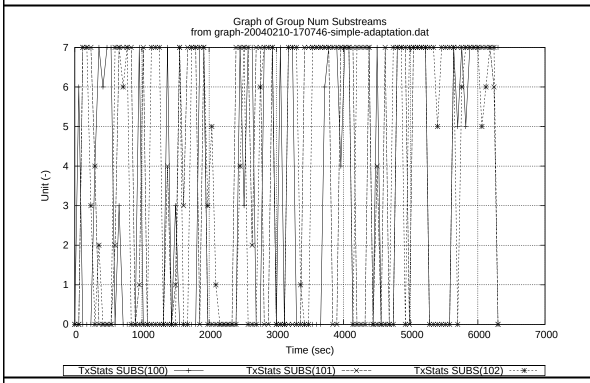
C.2.2 Static Nodes, No Congestion, Relaxed Delay, OPP

Table C.2: Simulation Parameters for nomobility-300 kbps- 250 ms

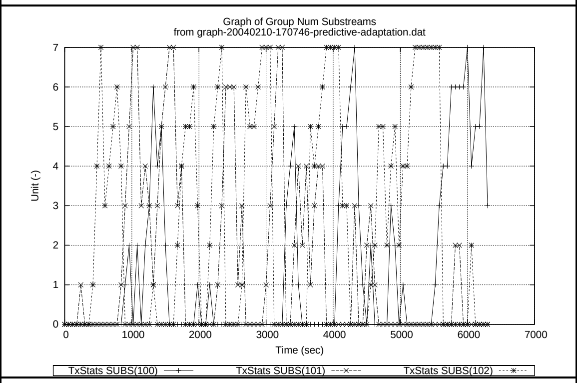
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	250 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	250×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams

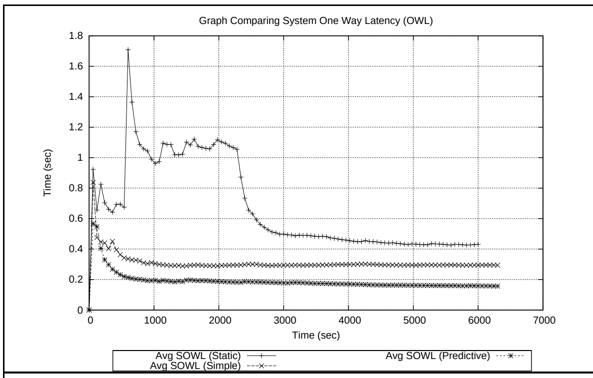


Simple Adaptive Num. Substreams

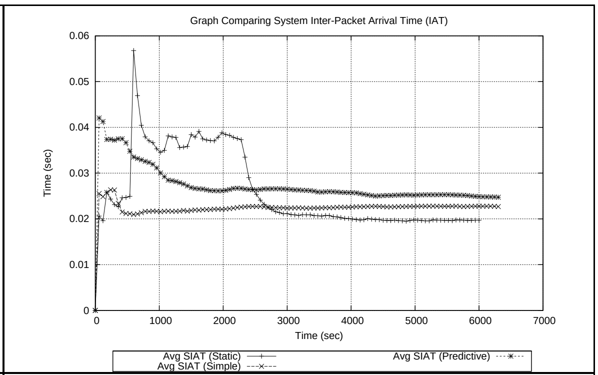


Predictive Adaptive Num. Substreams

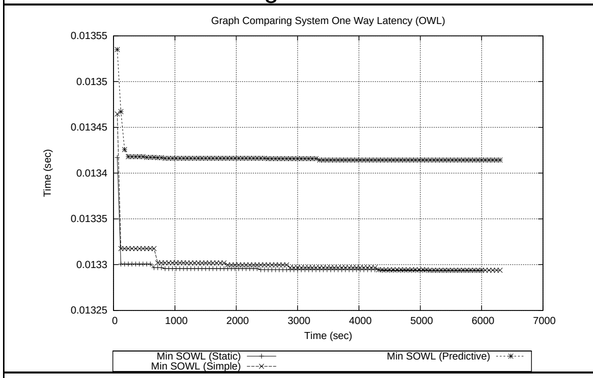
Figure C.4: Layered Substreams Count for nomobility-300 kbps- 250 ms



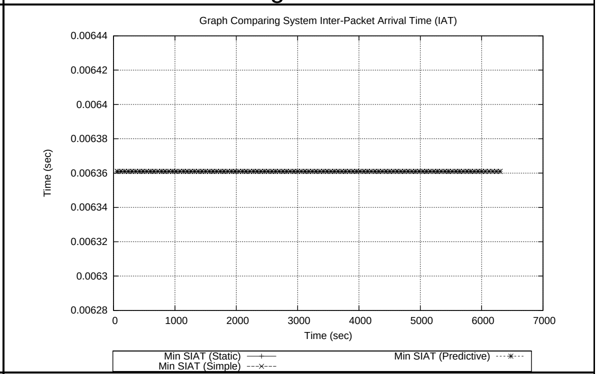
Avg SOWL



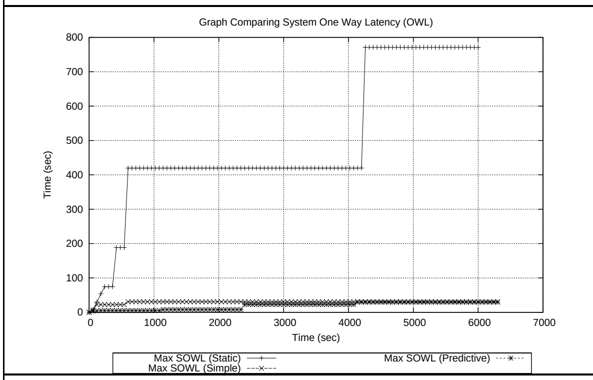
Avg. SIAT



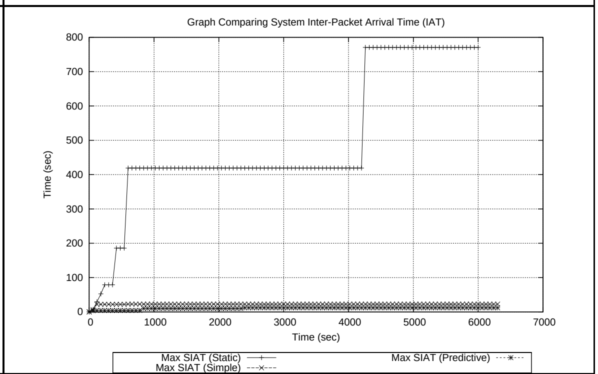
Min SOWL



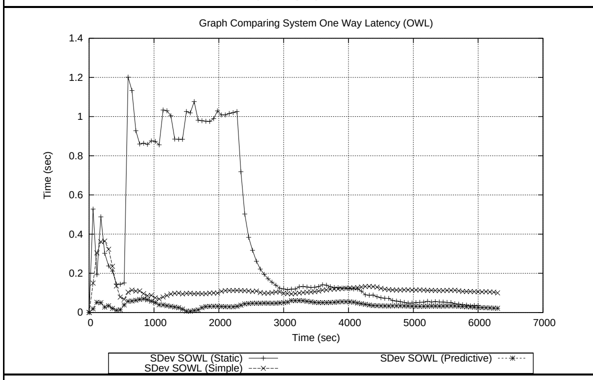
Min SIAT



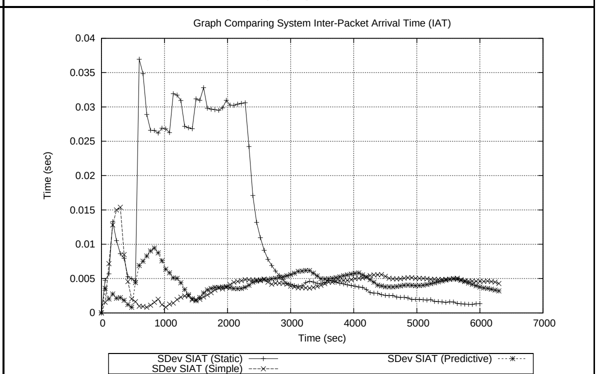
Max SOWL



Max SIAT

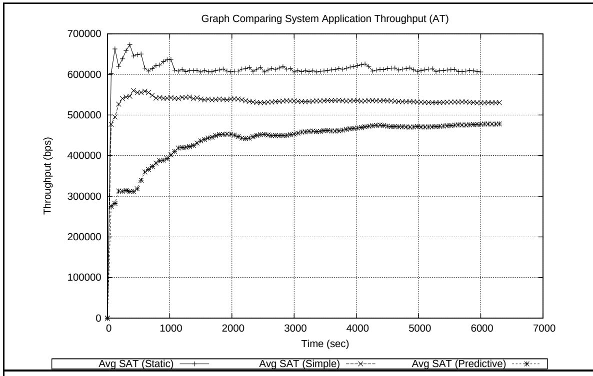


Sdev SOWL

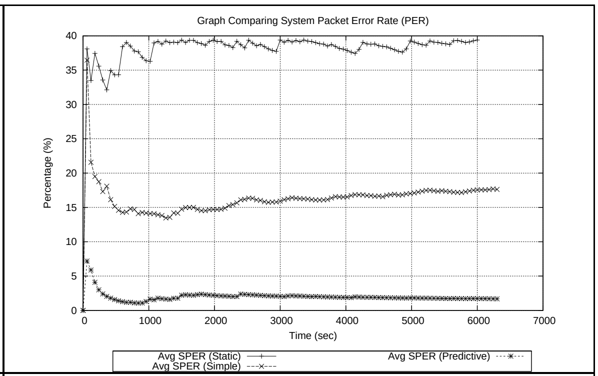


Sdev SIAT

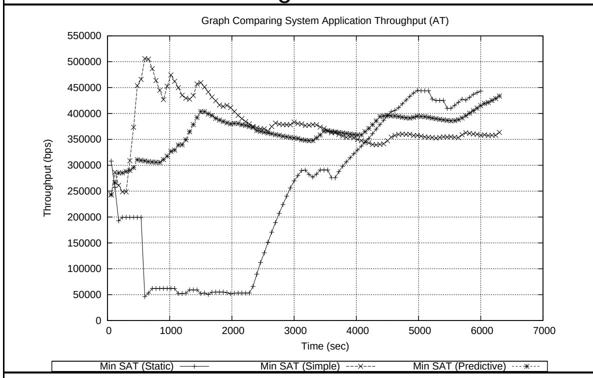
Figure C.5: SOWL and SIAT graphs for nomobility-300 kbps- 250 ms



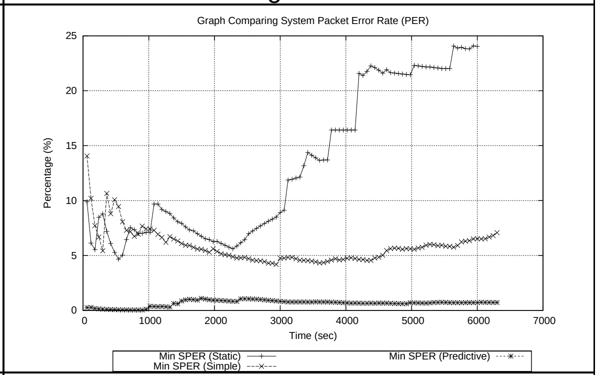
Avg SAT



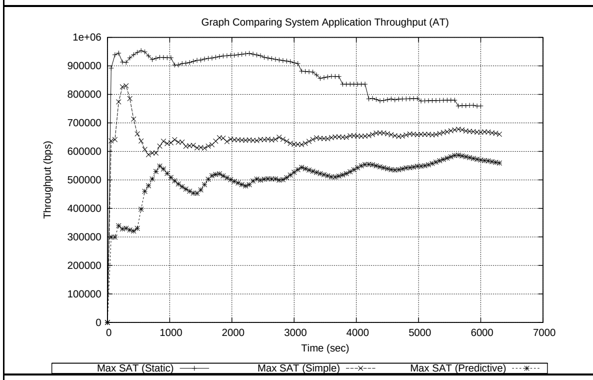
Avg SPER



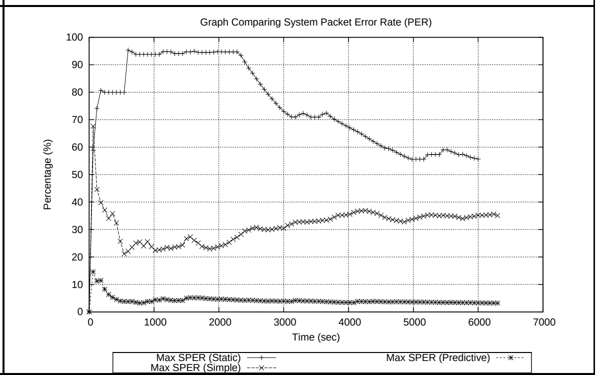
Min SAT



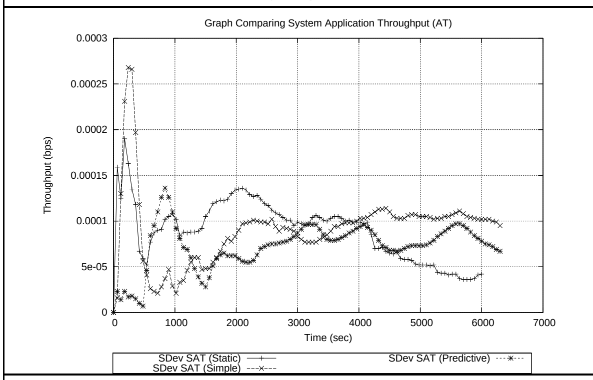
Min SPER



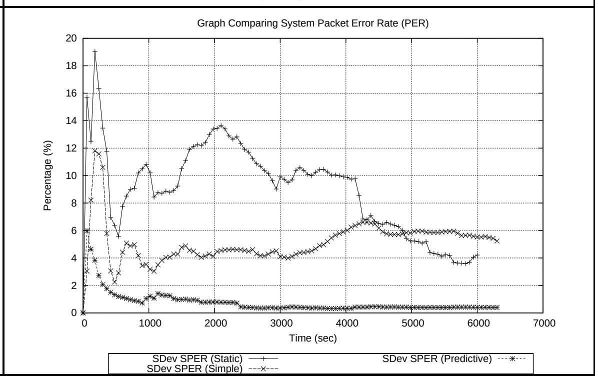
Max SAT



Max SPER



Sdev SAT



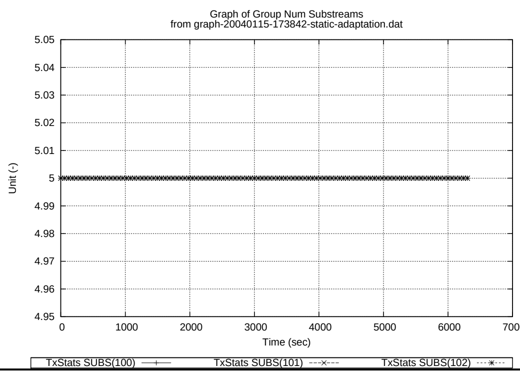
Sdev SPER

Figure C.6: SAT and SPER graphs for nomobility-300 kbps- 250 ms

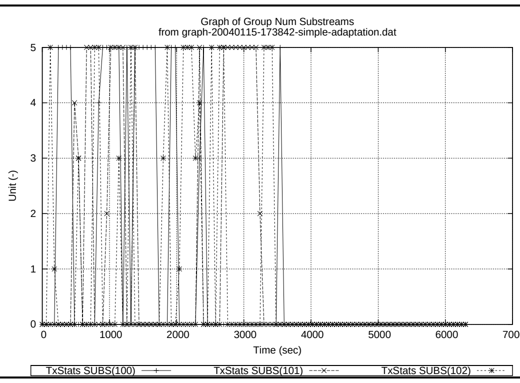
C.2.3 Static Nodes, Congestion, Tight Delay, OPP

Table C.3: Simulation Parameters for nomobility-90 min

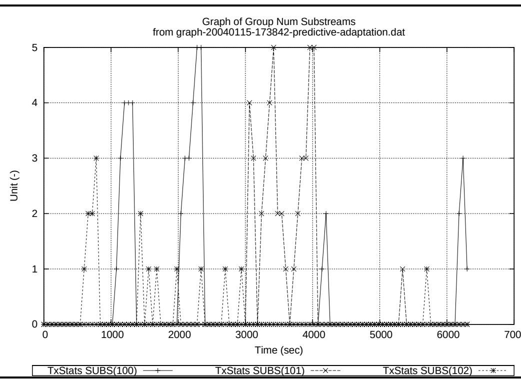
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	500 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	5
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.7: Layered Substreams Count for nomobility-90 min

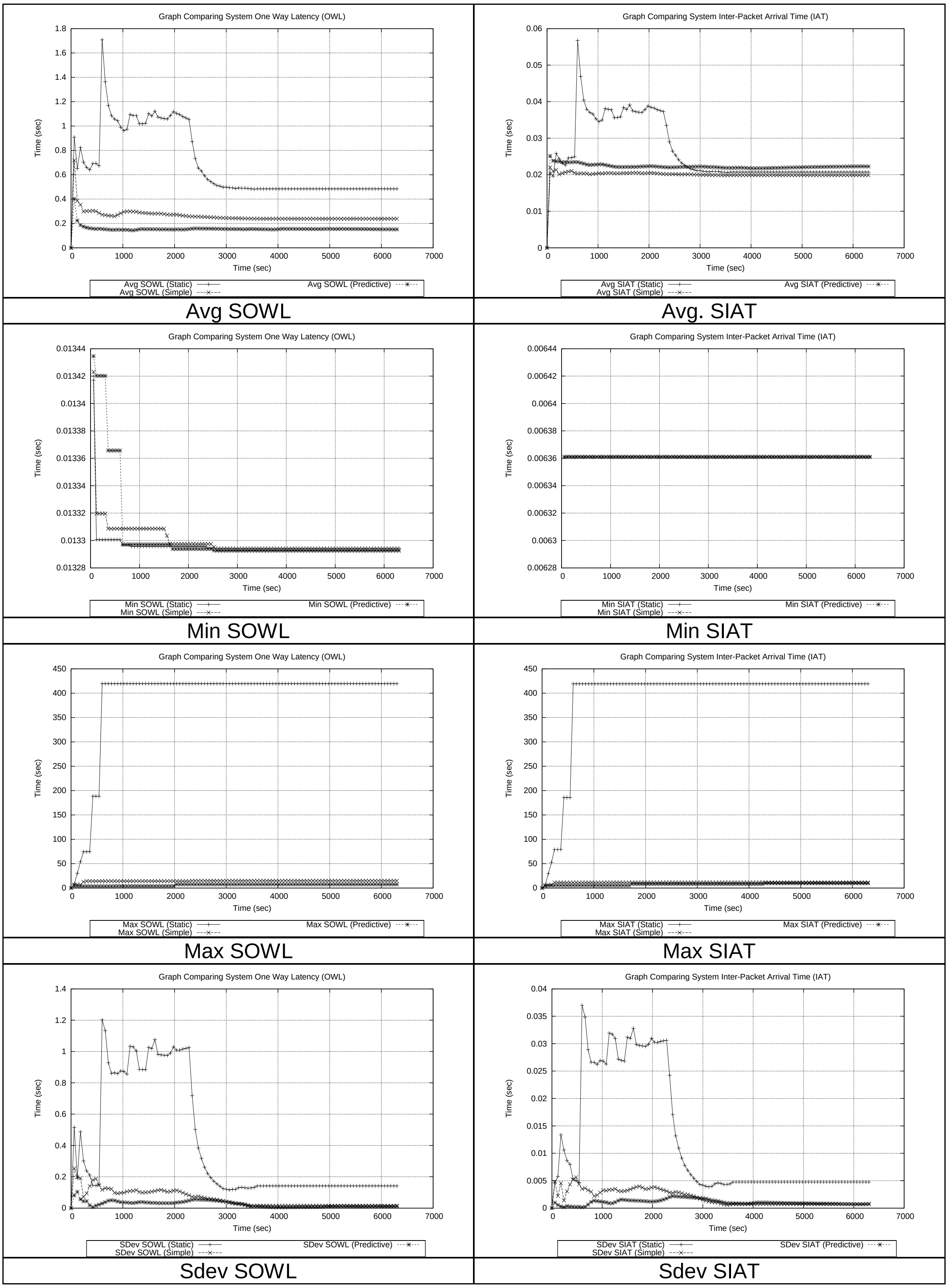


Figure C.8: SOWL and SIAT graphs for nomobility-90 min

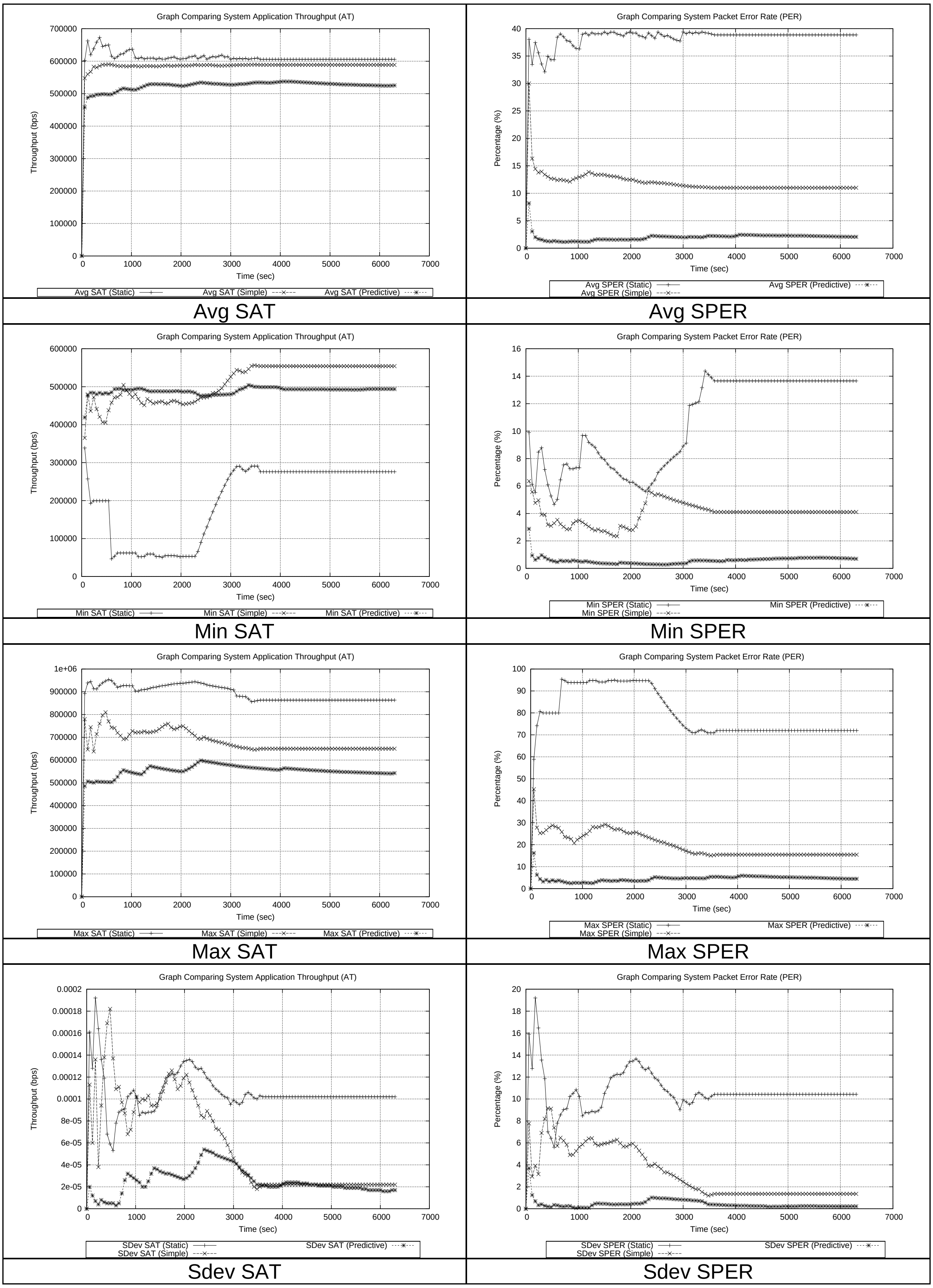


Figure C.9: SAT and SPER graphs for nomobility-90 min

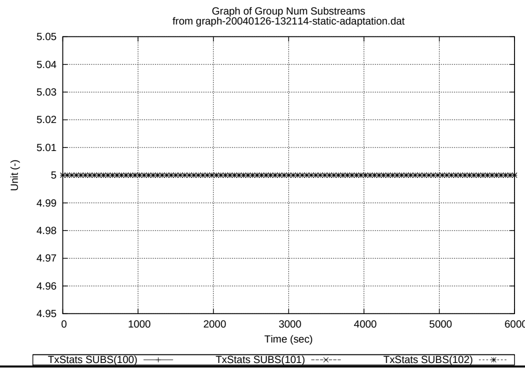
C.3 WMQF Performance Profiles Evaluation

The WMQF performance profiles evaluation compared the ability of different performance profiles to accommodate different priority requirements of competing multimedia streams. In addition, it was used to determine which performance profile was most suitable for use by competing equal-priority streams.

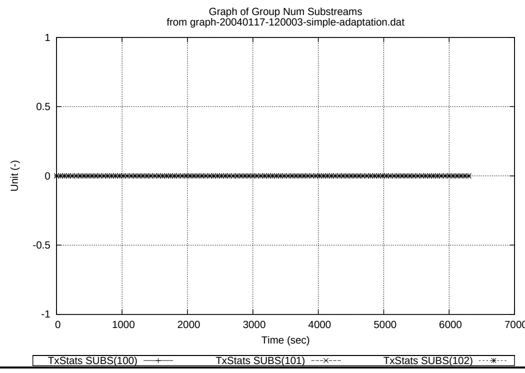
C.3.1 Mobile Nodes, Congestion, Tight Delay, EPP

Table C.4: Simulation Parameters for equal

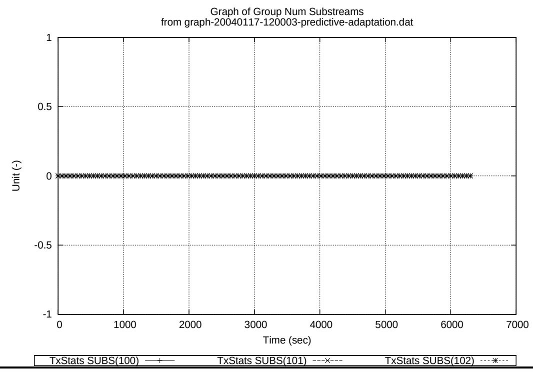
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Equal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	500 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	5
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.10: Layered Substreams Count for equal

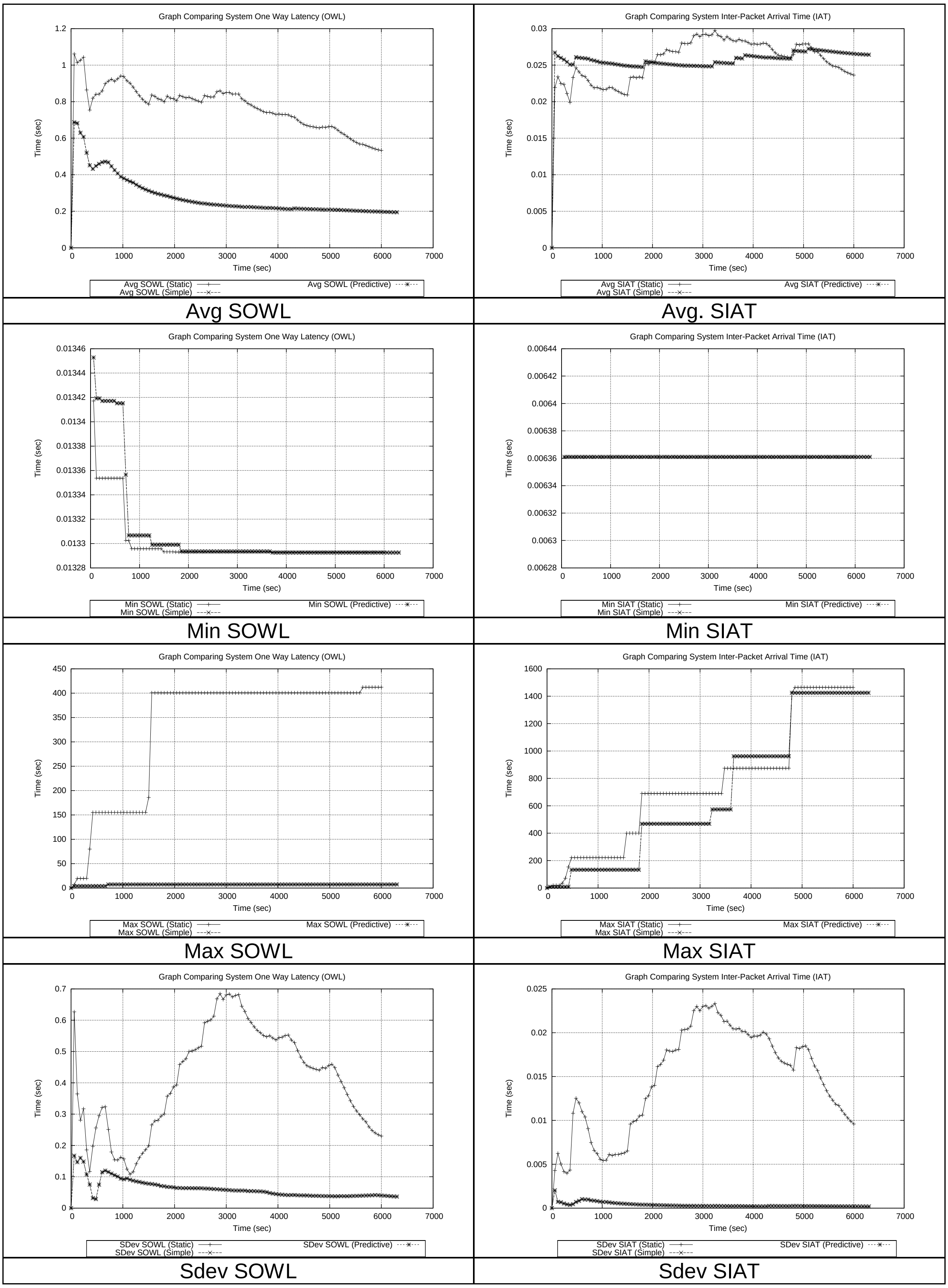


Figure C.11: SOWL and SIAT graphs for equal

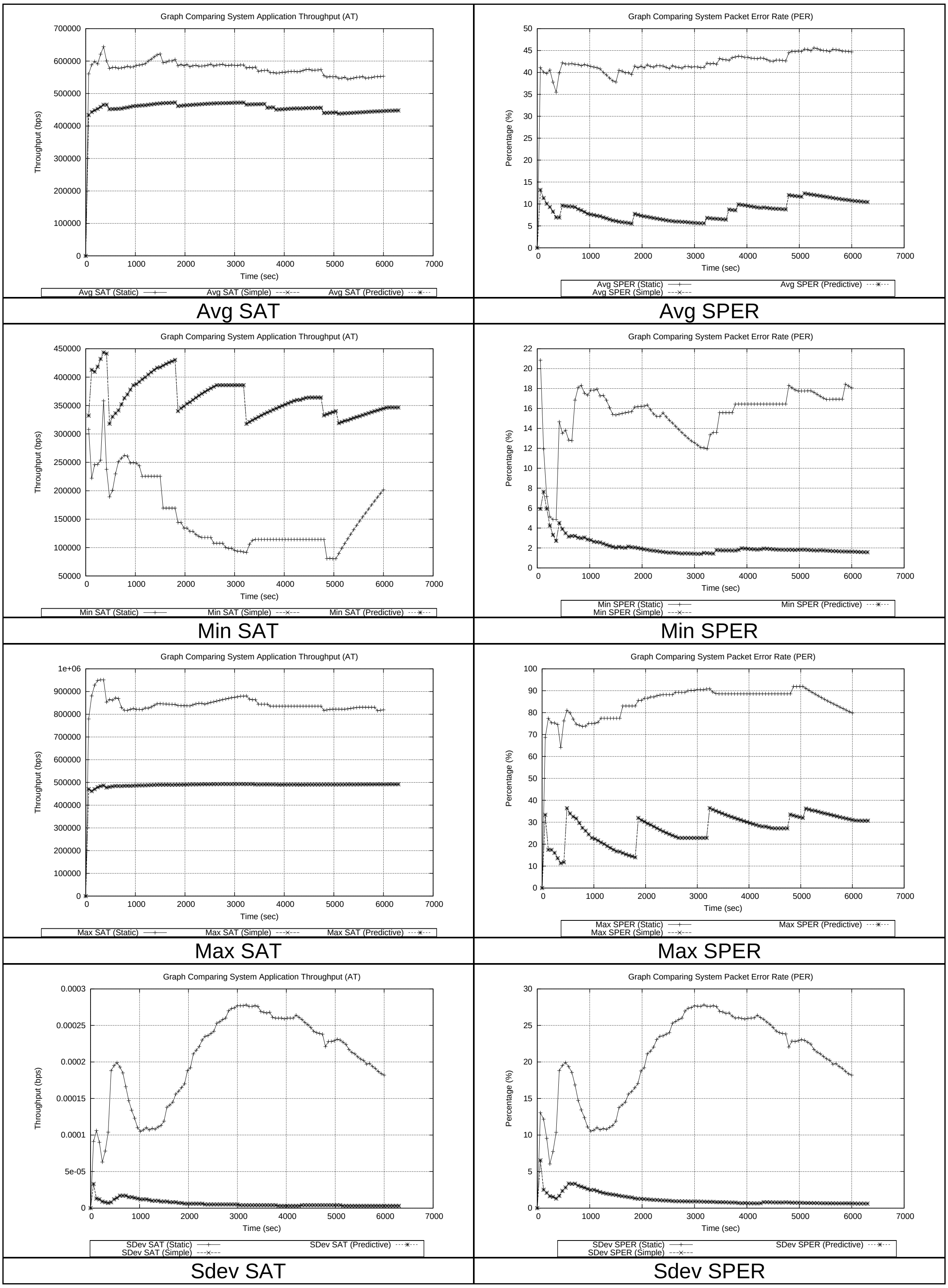
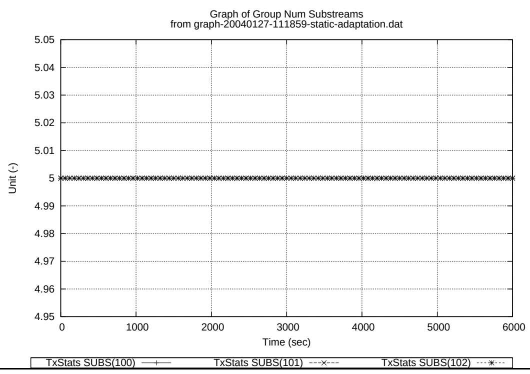


Figure C.12: SAT and SPER graphs for equal

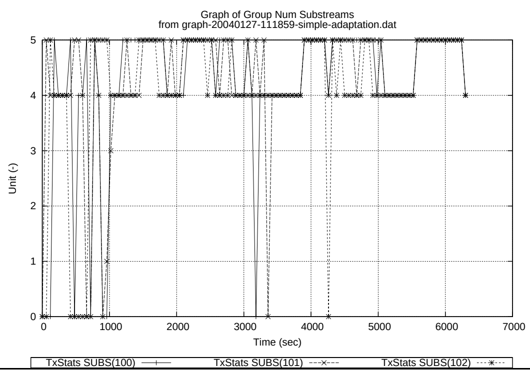
C.3.2 Mobile Nodes, Congestion, Tight Delay, MPP

Table C.5: Simulation Parameters for maximized

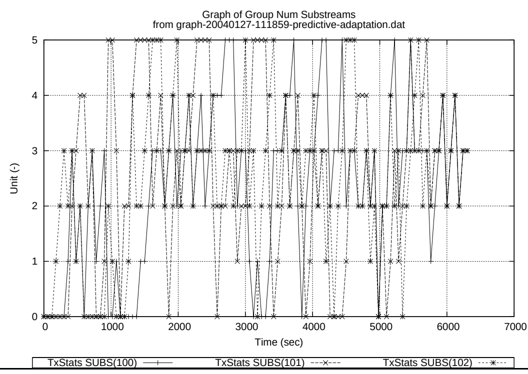
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Maximized
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	500 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	5
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams

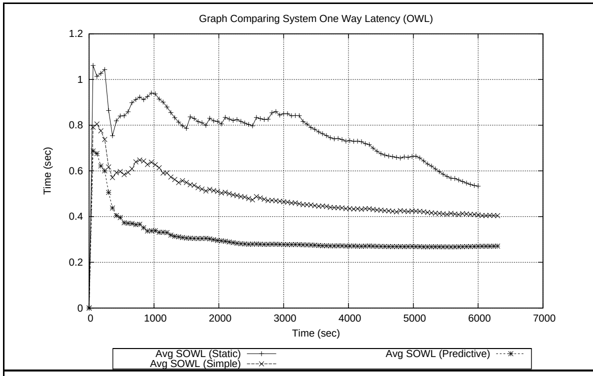


Simple Adaptive Num. Substreams

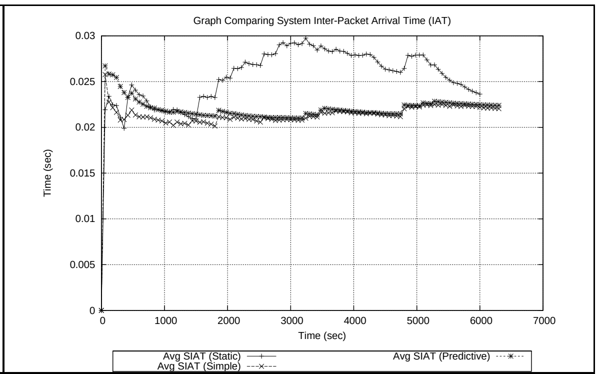


Predictive Adaptive Num. Substreams

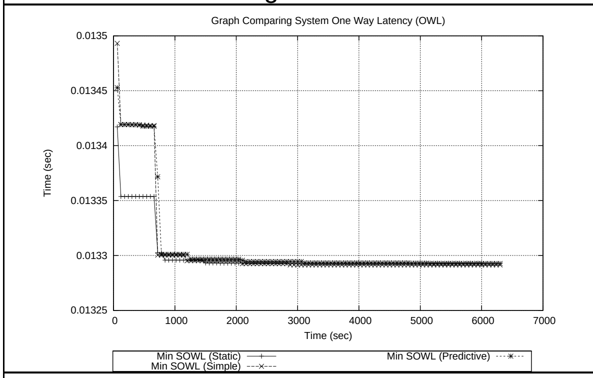
Figure C.13: Layered Substreams Count for maximized



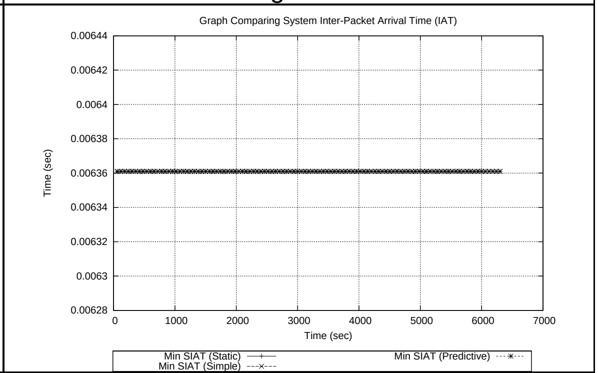
Avg SOWL



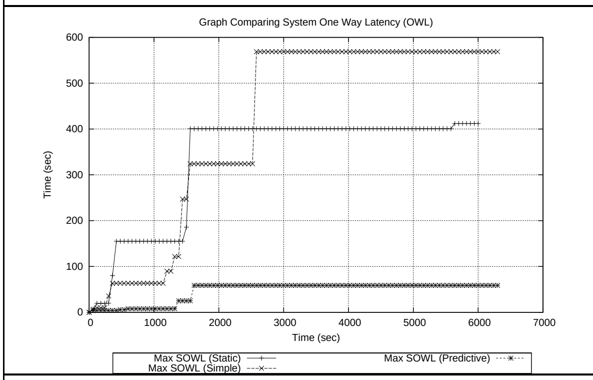
Avg. SIAT



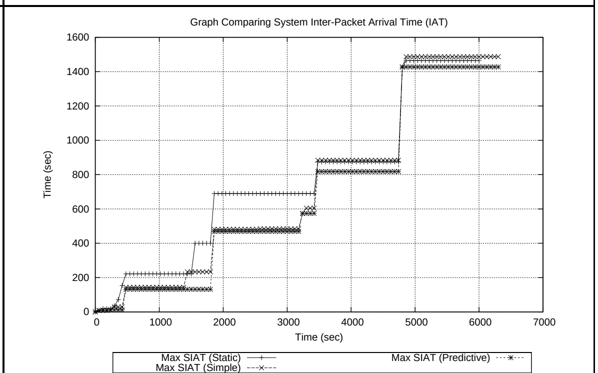
Min SOWL



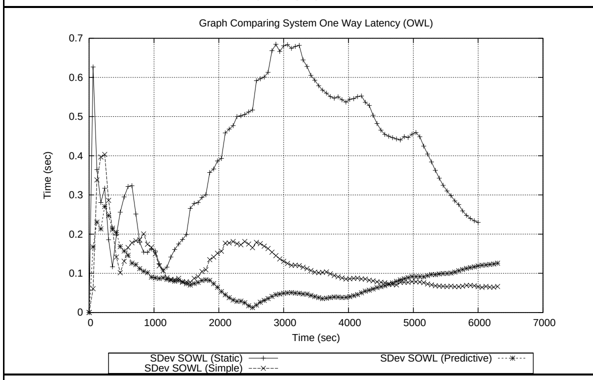
Min SIAT



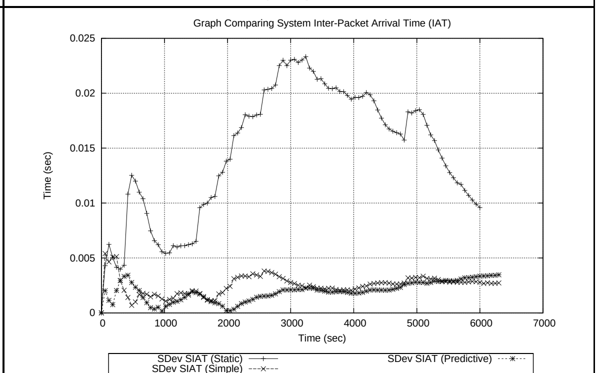
Max SOWL



Max SIAT



Sdev SOWL



Sdev SIAT

Figure C.14: SOWL and SIAT graphs for maximized

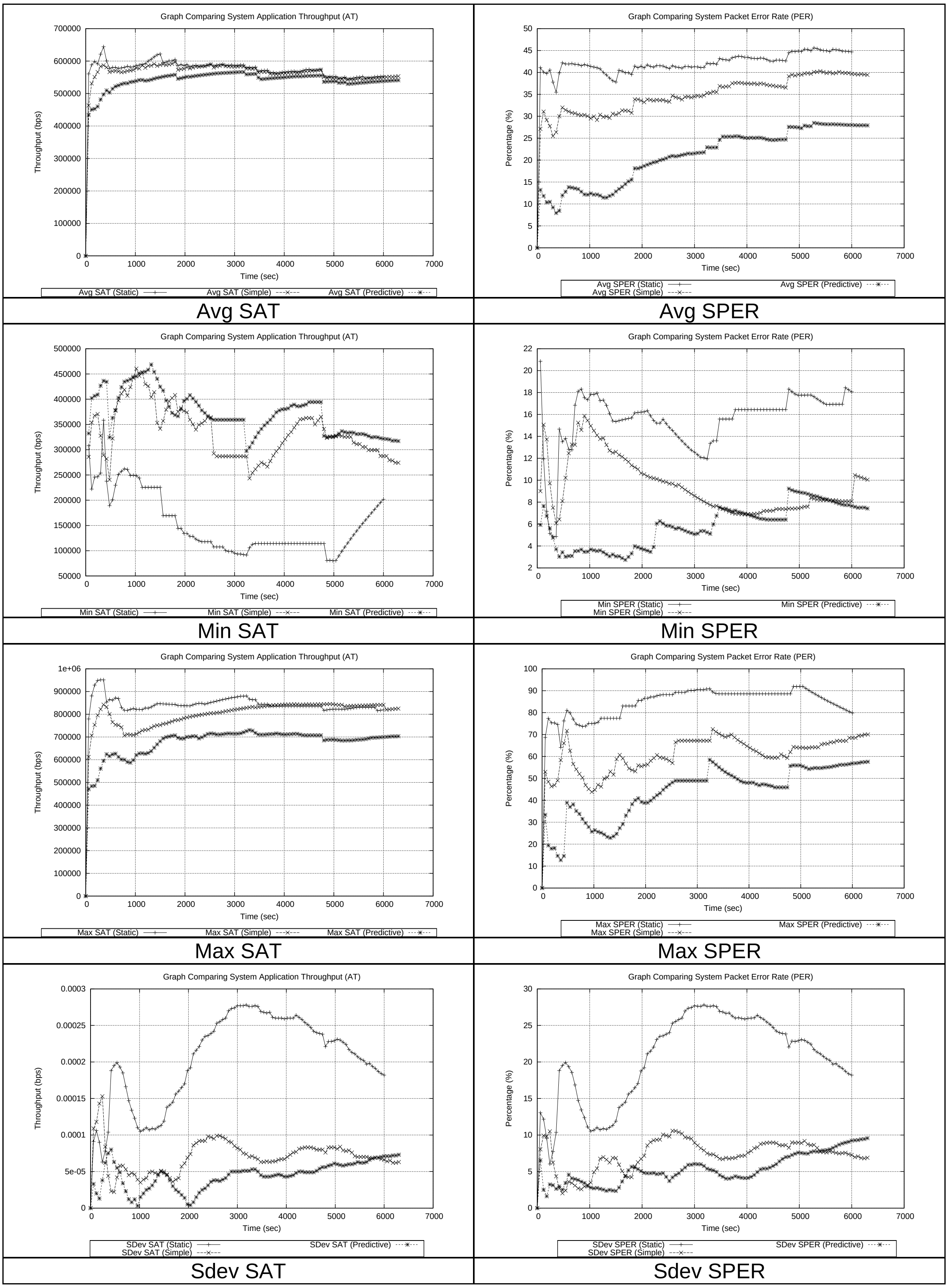
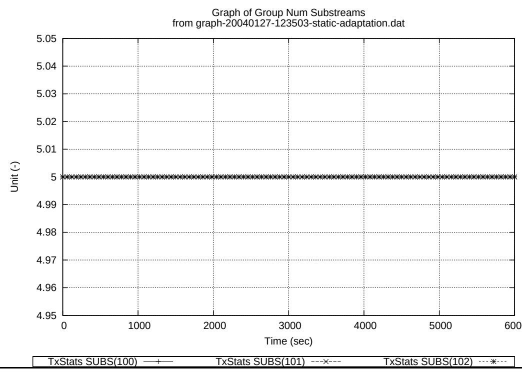


Figure C.15: SAT and SPER graphs for maximized

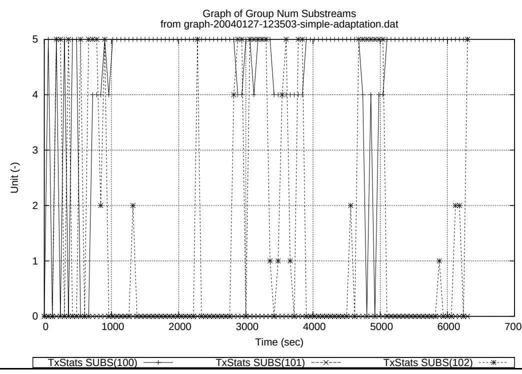
C.3.3 Mobile Nodes, Congestion, Tight Delay, Mixed PP

Table C.6: Simulation Parameters for mixed

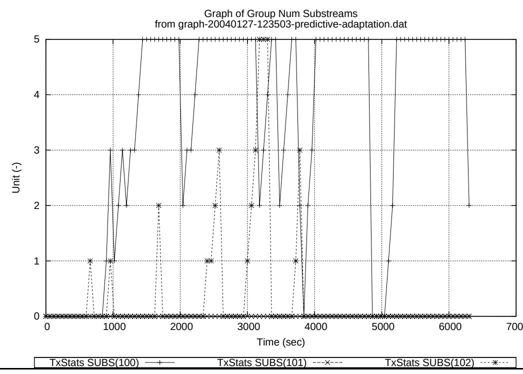
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Maximized (Node 20: Group 100), Equal (Node 21: Group 101), Optimal (Node 22: Group 102)
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (different-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	500 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	5
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.16: Layered Substreams Count for mixed

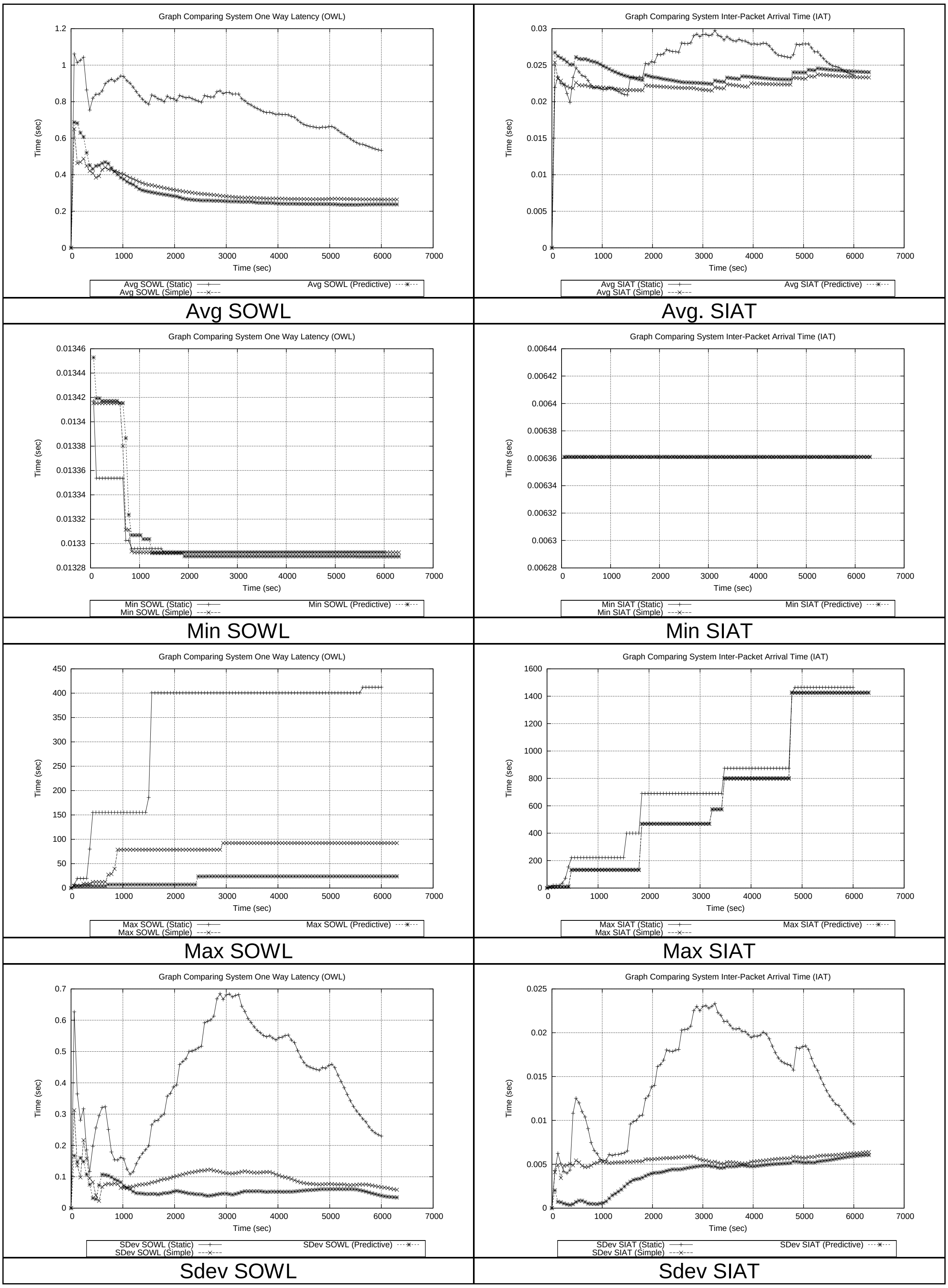
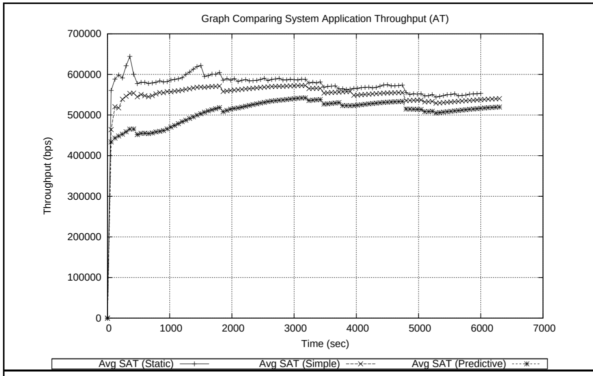
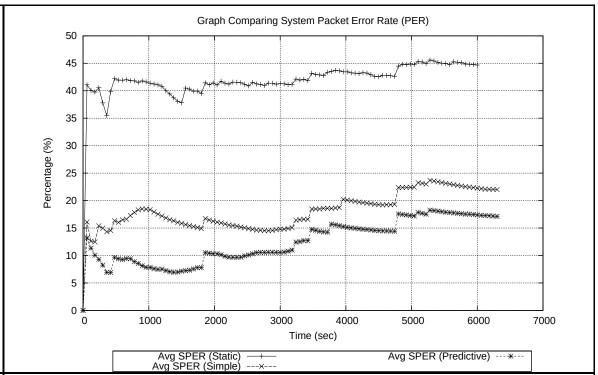


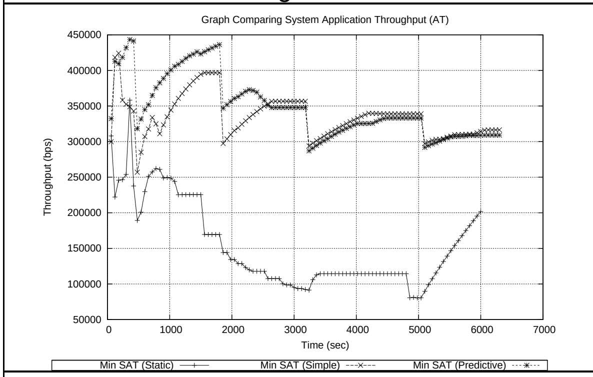
Figure C.17: SOWL and SIAT graphs for mixed



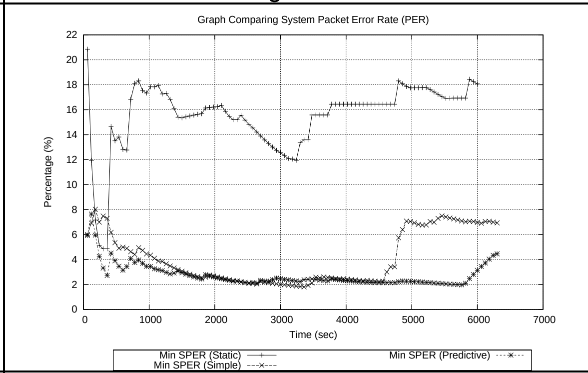
Avg SAT



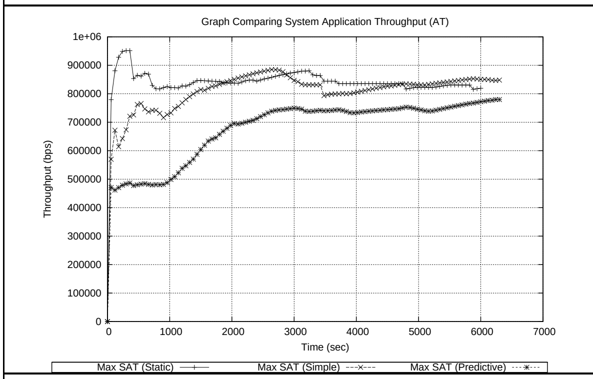
Avg SPER



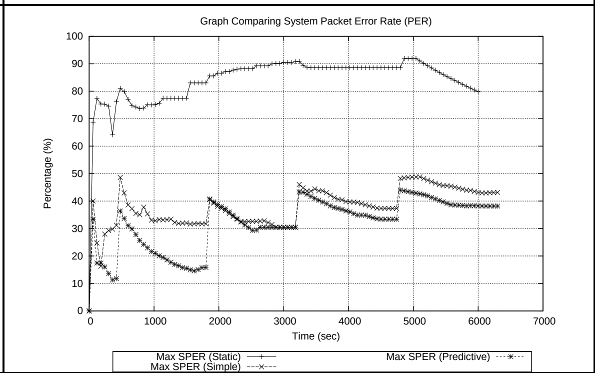
Min SAT



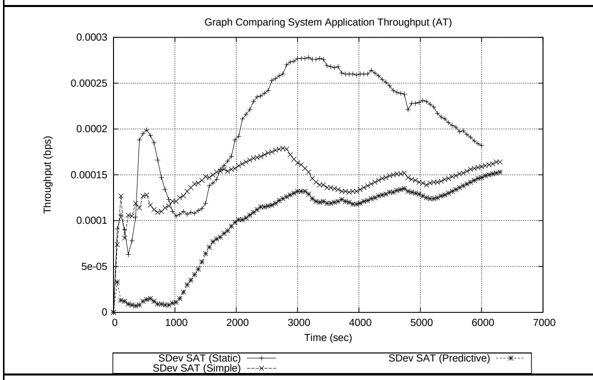
Min SPER



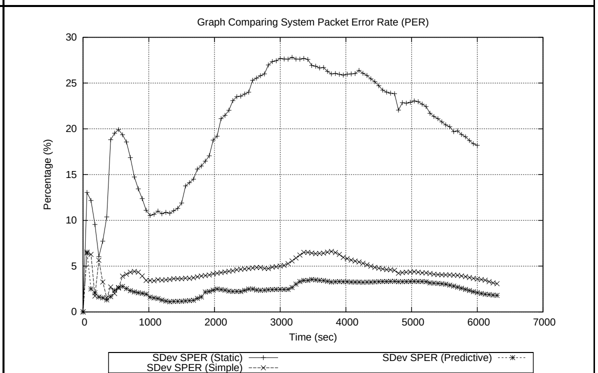
Max SAT



Max SPER



Sdev SAT



Sdev SPER

Figure C.18: SAT and SPER graphs for mixed

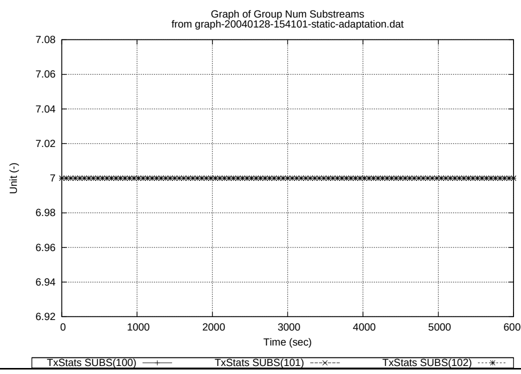
C.4 Optimal Performance Profile Mobility Trials

Mobility trials were designed to test the adaptability and robustness of the MQAA for dynamically changing network conditions present with mobile nodes.

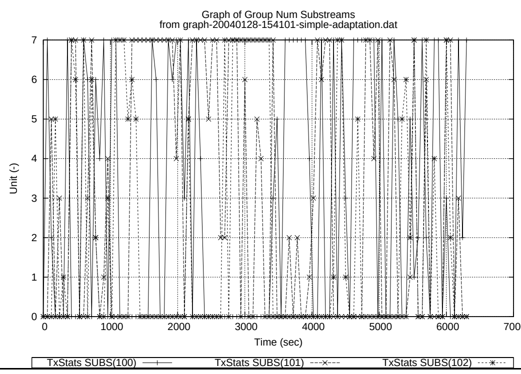
C.4.1 Mobile Nodes, No Congestion, Tight Delay, OPP

Table C.7: Simulation Parameters for optimal-300 kbps

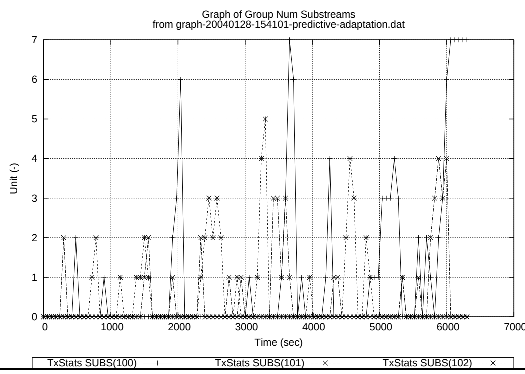
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.19: Layered Substreams Count for optimal-300 kbps

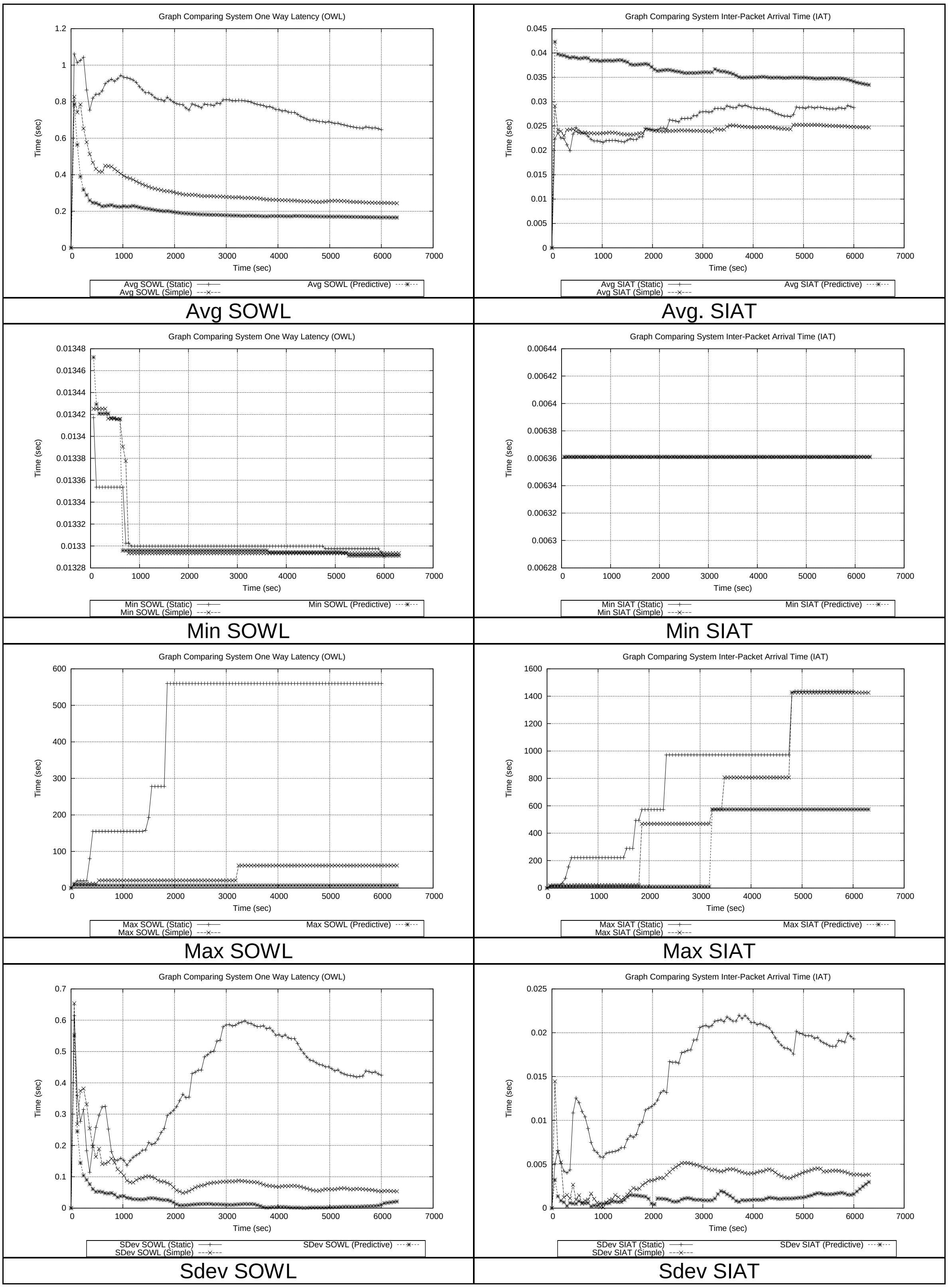


Figure C.20: SOWL and SIAT graphs for optimal-300 kbps

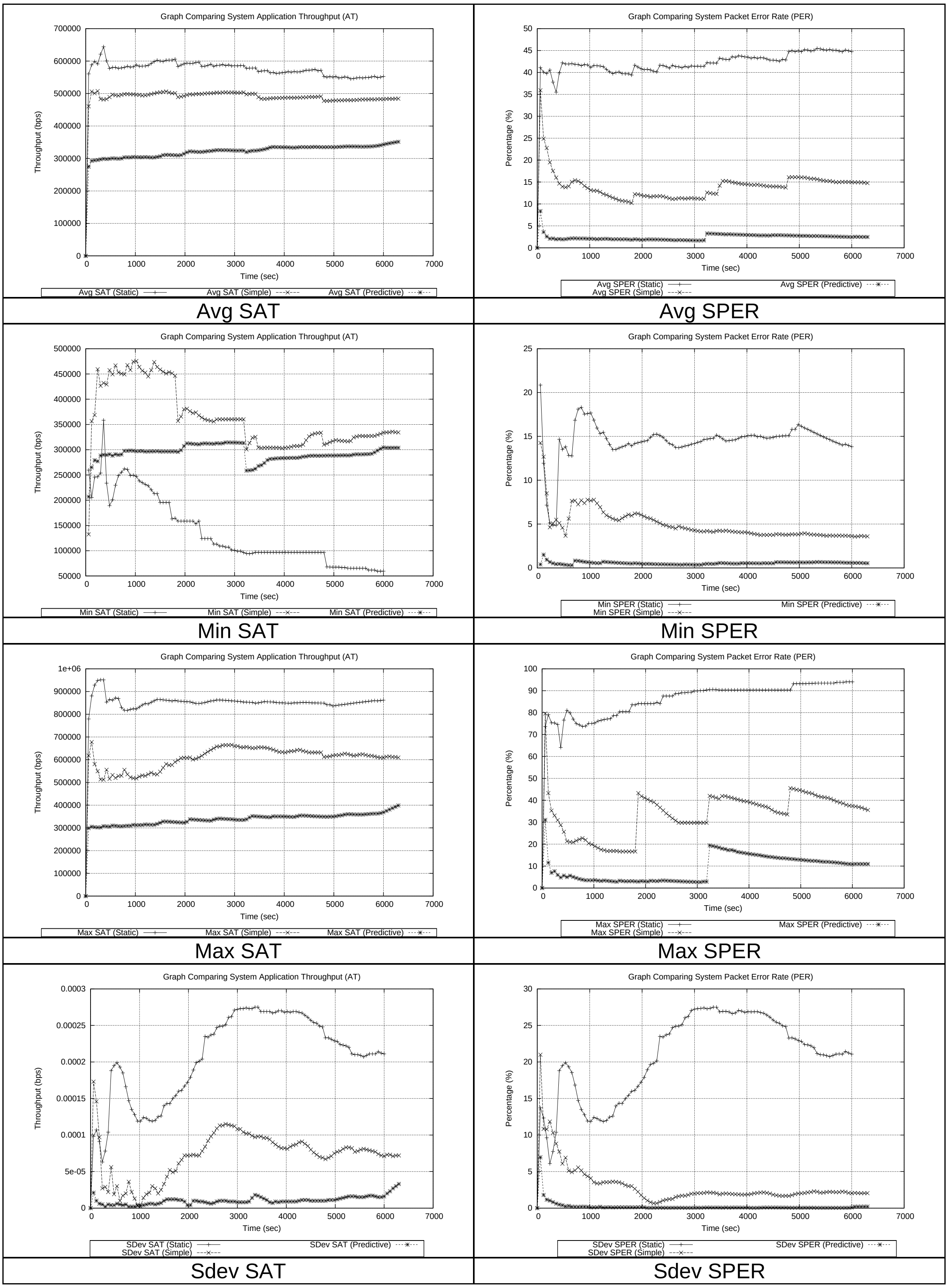
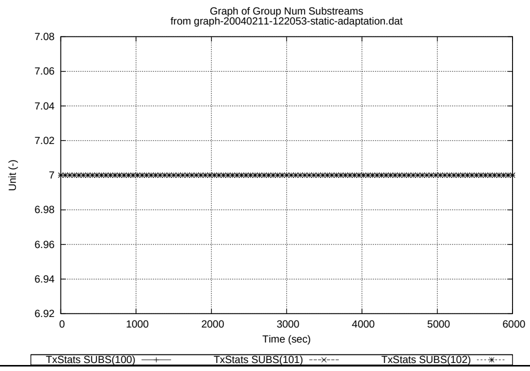


Figure C.21: SAT and SPER graphs for optimal-300 kbps

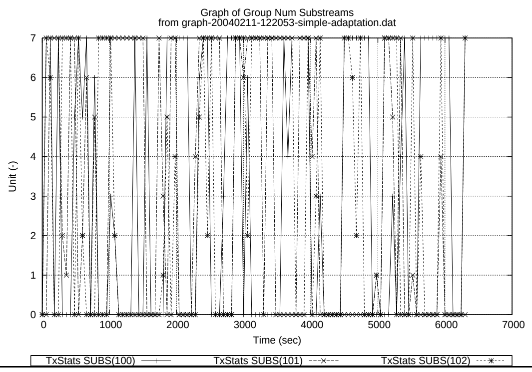
C.4.2 Mobile Nodes, No Congestion, Relaxed Delay, OPP

Table C.8: Simulation Parameters for optimal-300 kbps- 250 ms

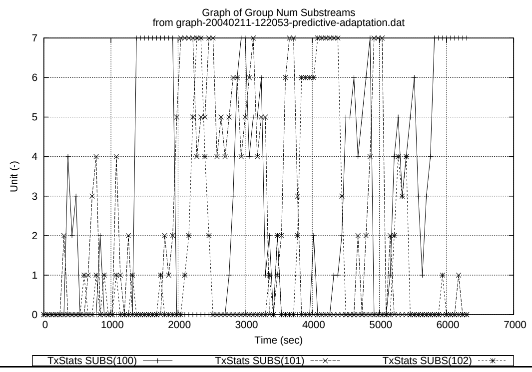
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	250 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	250×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.22: Layered Substreams Count for optimal-300 kbps- 250 ms

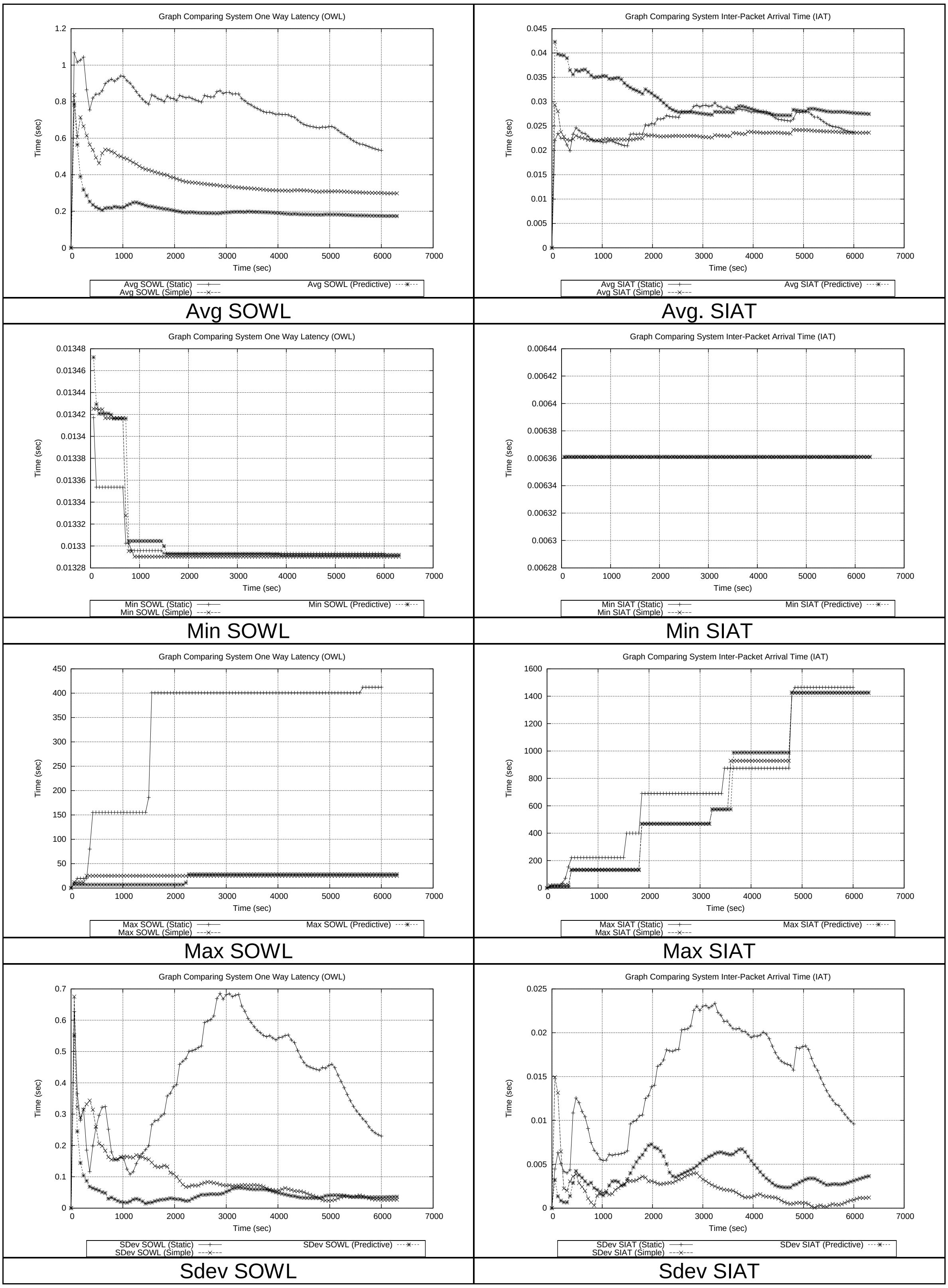


Figure C.23: SOWL and SIAT graphs for optimal-300 kbps- 250 ms

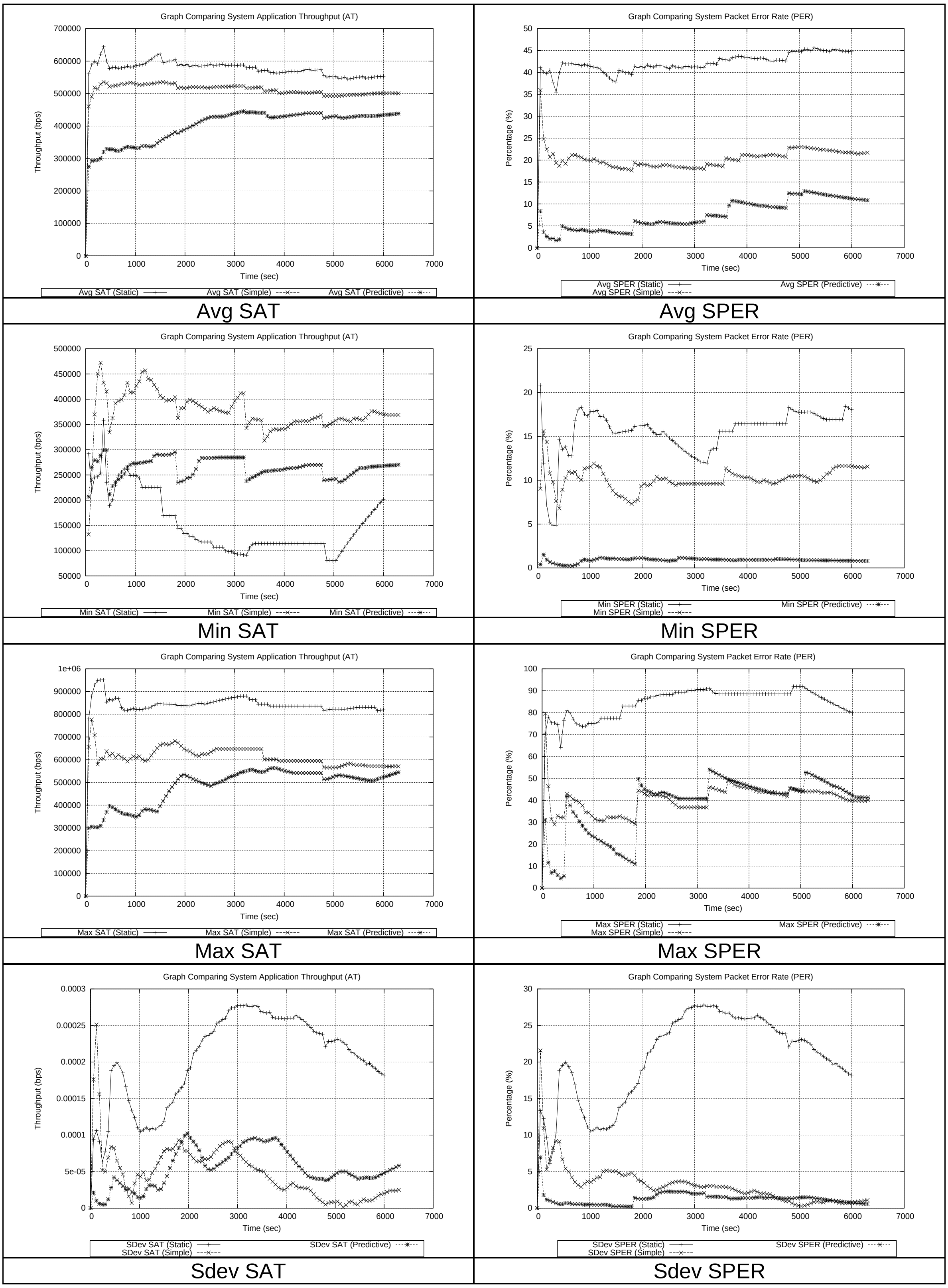
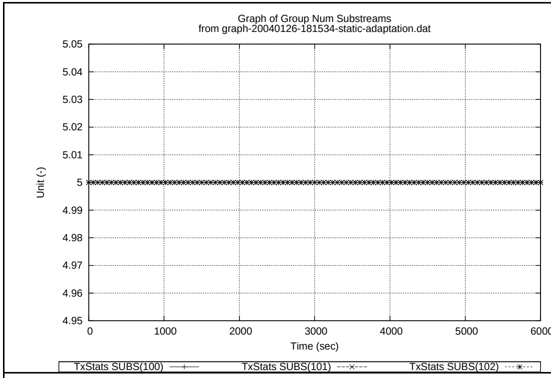


Figure C.24: SAT and SPER graphs for optimal-300 kbps- 250 ms

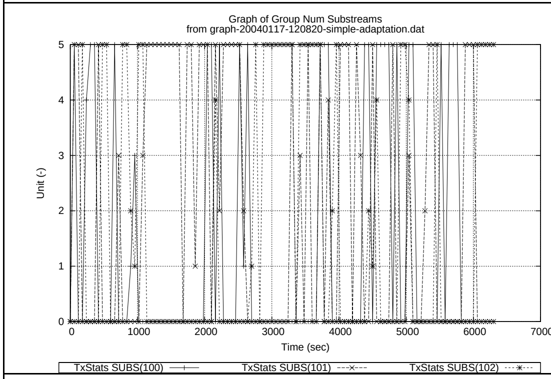
C.4.3 Mobile Nodes, Congestion, Tight Delay, OPP

Table C.9: Simulation Parameters for optimal

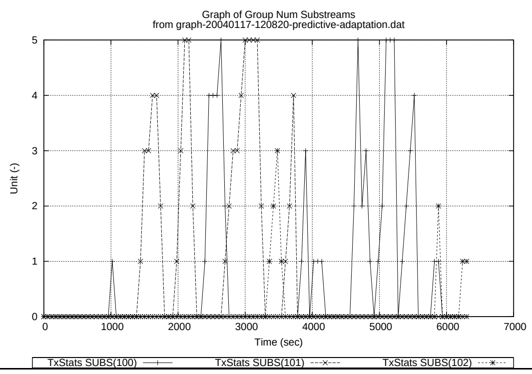
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20, 21, 22 (3 equal-priority streams)
Multicast Rx Nodes	0 to 10 (11 receivers, all 3 streams)
Base Stream Bandwidth (min)	500 kbps
Base Stream Delay (max)	200 ms
Num. of additional Substreams	5
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	200×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.25: Layered Substreams Count for optimal

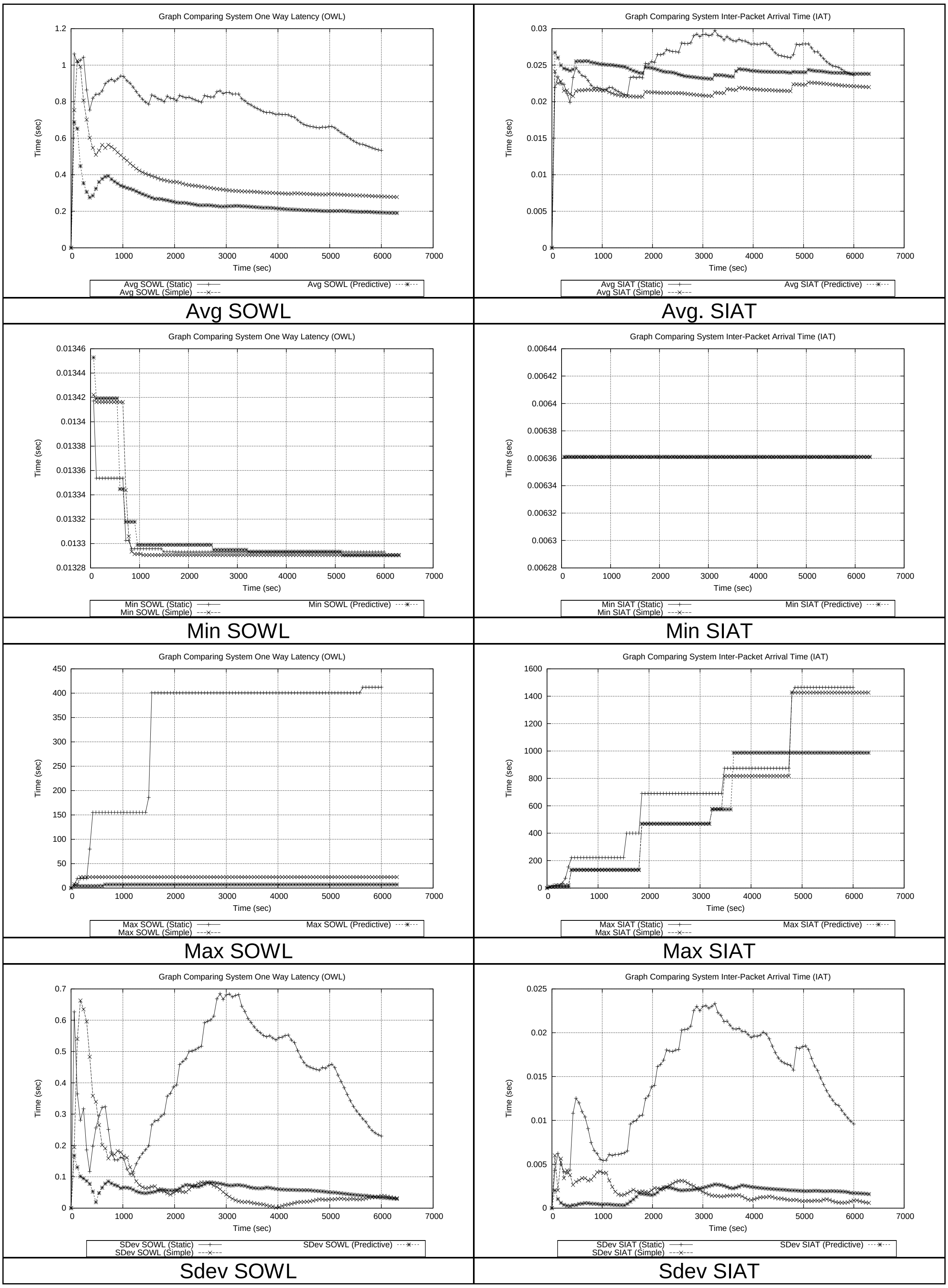


Figure C.26: SOWL and SIAT graphs for optimal

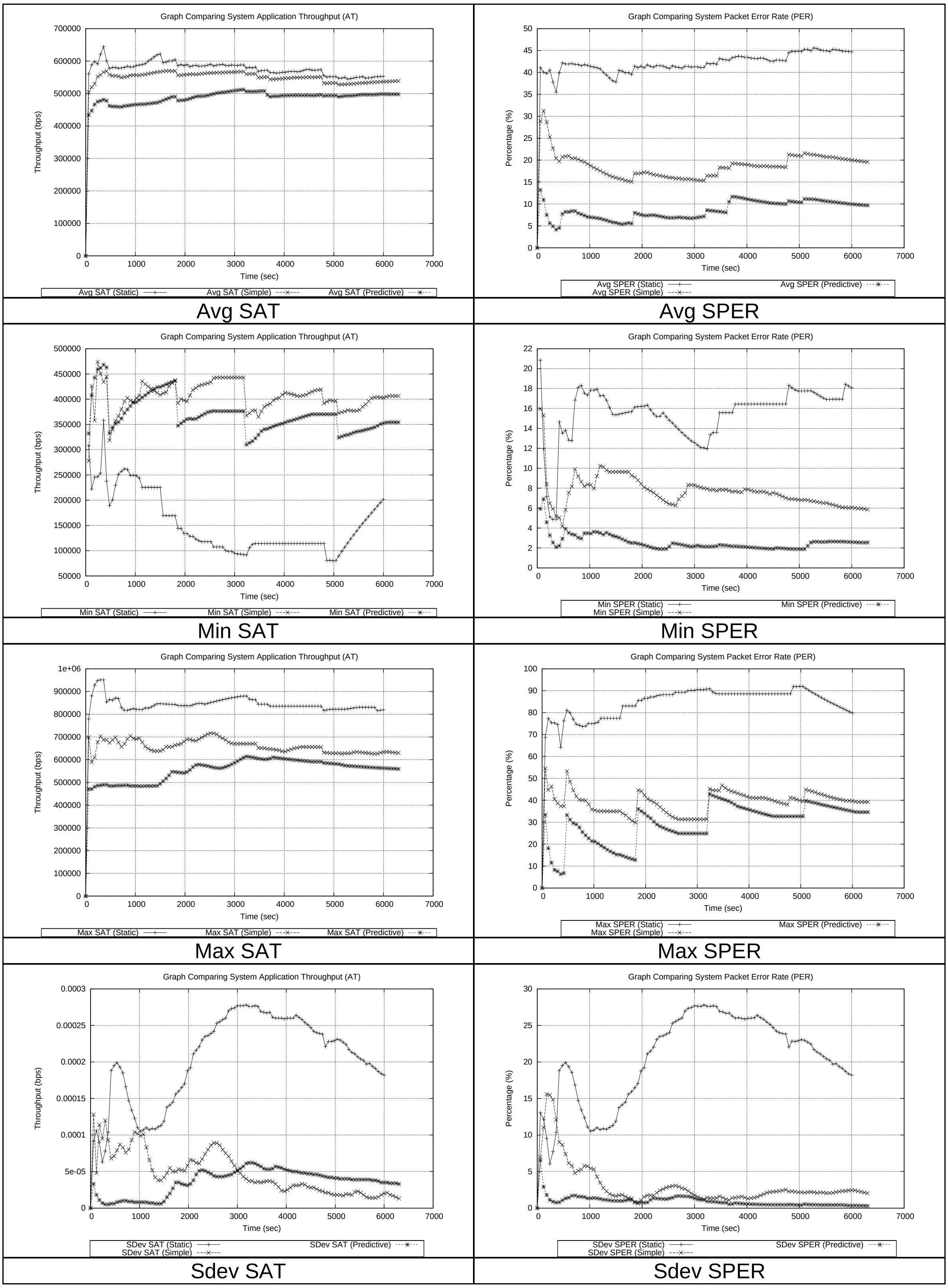


Figure C.27: SAT and SPER graphs for optimal

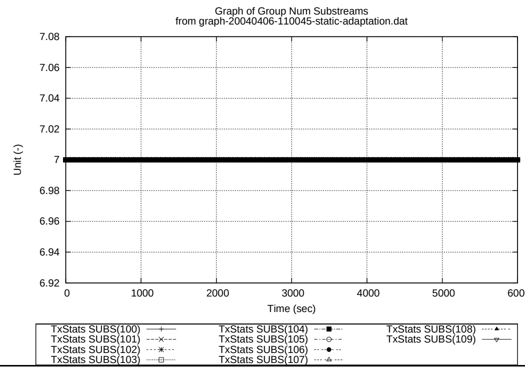
C.5 Complex Scenarios

Complex scenarios are designed to stress the adaptation algorithms and determine how robust they are when network conditions are dynamically fluctuating, as well as when network resources are unable to meet all QoS requirements from multiple streams.

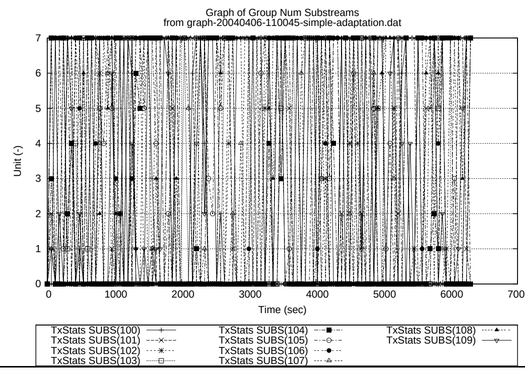
C.5.1 Multiflows, Static Nodes, No Congestion, Relaxed Delay, OPP

Table C.10: Simulation Parameters for manysrc-nomobility

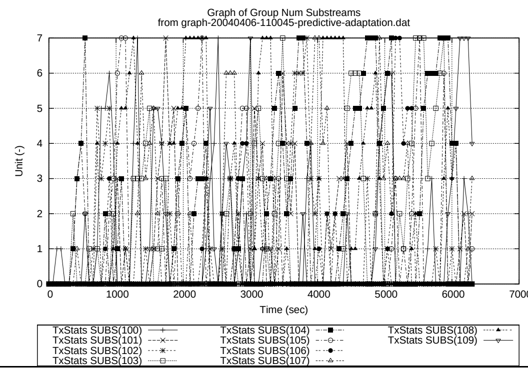
Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20 to 29 (10 equal-priority streams)
Multicast Rx Nodes	0 to 19 (20 receivers, 2 streams each)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	250 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	250×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams



Simple Adaptive Num. Substreams



Predictive Adaptive Num. Substreams

Figure C.28: Layered Substreams Count for manysrc-nomobility

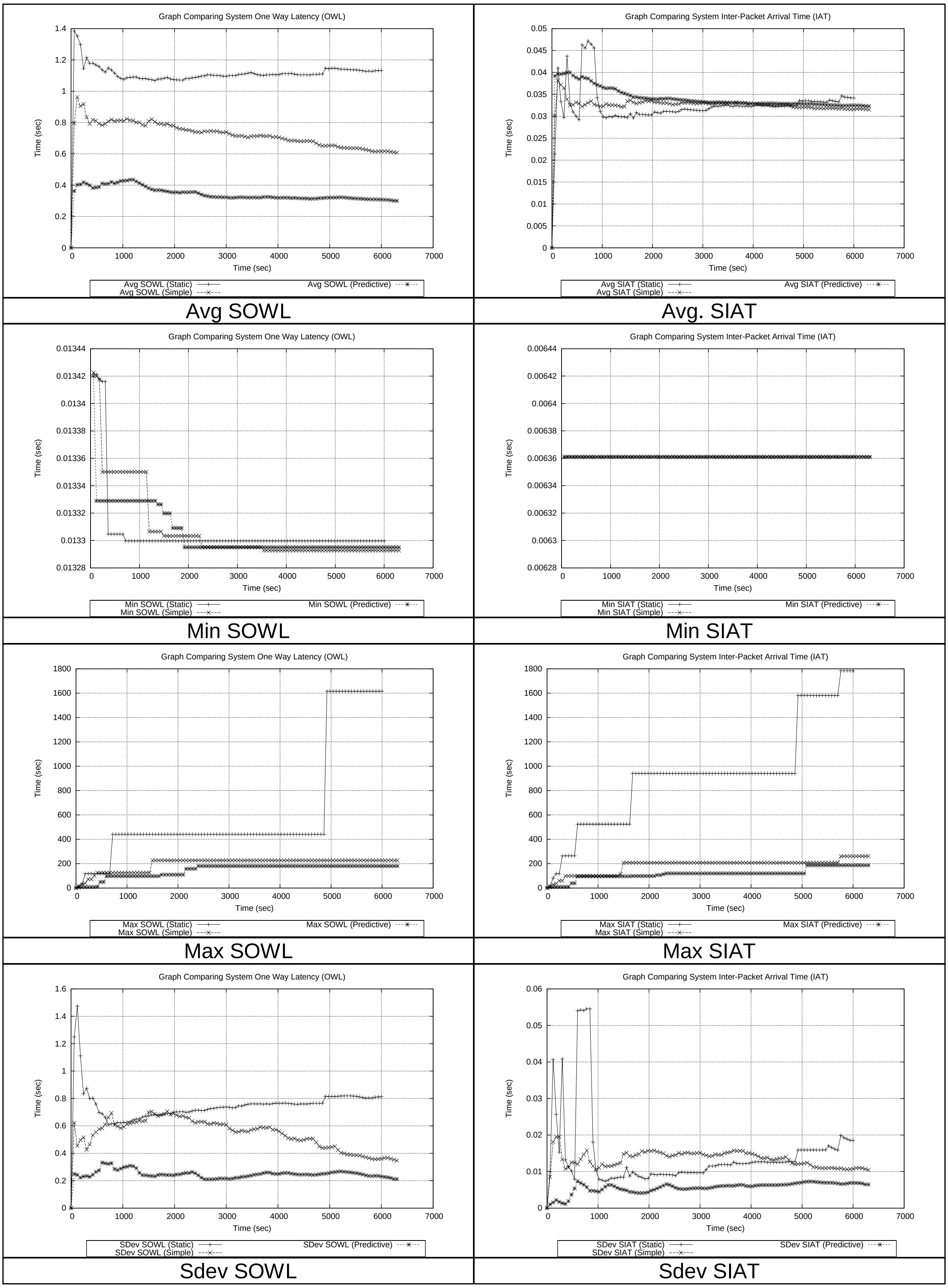


Figure C.29: SOWL and SIAT graphs for manysrc-nomobility

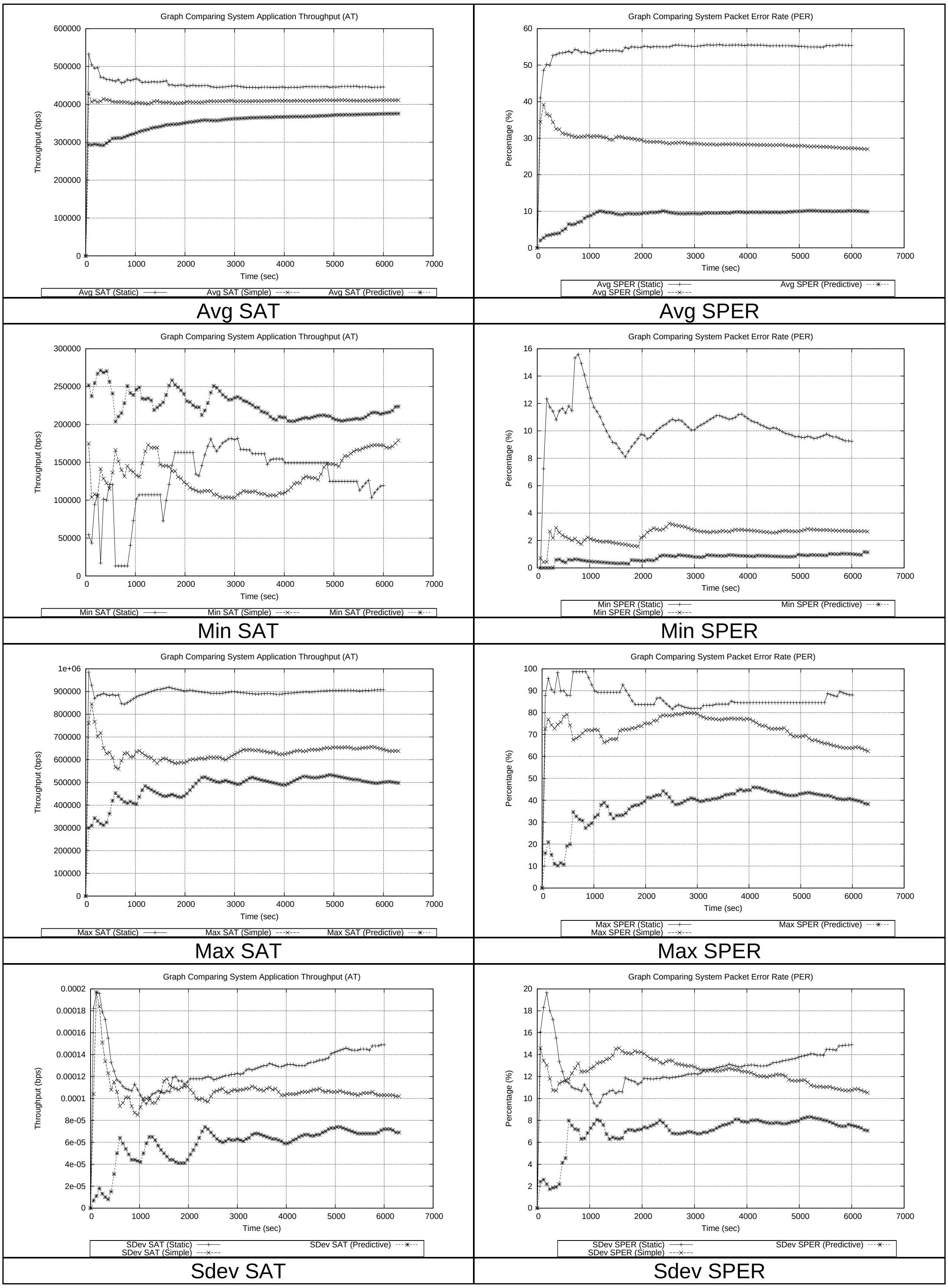
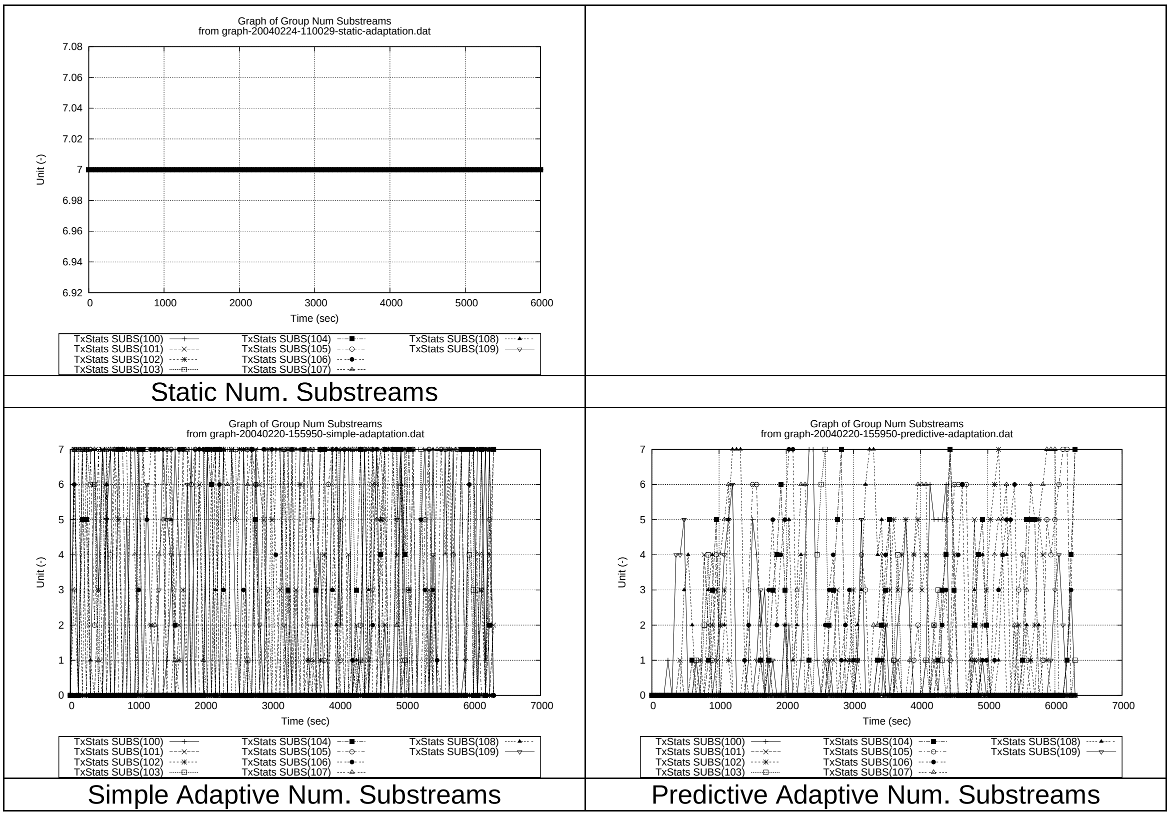


Figure C.30: SAT and SPER graphs for manysrc-nomobility

C.5.2 Multiflows, Mobile Nodes, No Congestion, Relaxed Delay, OPP

Table C.11: Simulation Parameters for manysrc

Simulation Parameters	Value
Simulation Duration	100 – 105 minutes (6000 – 6300 s)
Multicast Performance Profile	Optimal
QoS Adaptation Techniques	Static, Simple, Predictive
Multicast Tx Nodes	20 to 29 (10 equal-priority streams)
Multicast Rx Nodes	0 to 19 (20 receivers, 2 streams each)
Base Stream Bandwidth (min)	300 kbps
Base Stream Delay (max)	250 ms
Num. of additional Substreams	7
Substream Bandwidth Increment	100 kbps
Substream Delay Reduction	10 ms
UDL Avail. Link Bandwidth (μ)	2 Mbps (1.98 Mbps for Data)
Required Delay scaling factor (β)	250×10^{-3}
M-LWDF Pr_{max}	0.1
Convergence Func. FIFO size (j)	16
Convergence Func. FIFO size (k)	2
Convergence Factor (n)	1.2
Number of Increment steps (m)	10
Initial Increment Fraction (p)	0.1
Unicast Constant Bit Rate Flows (Flows 4-6 originate from Nodes 20, 21, & 22, competing with their respective multicast flows for uplink bandwidth)	Flow 1: 120 s – 1200 s Flow 2: 2700 s – 3900 s Flow 3: 4200 s – 5100 s Flows 4,5,6: 600 s – 1200 s



Static Num. Substreams

Simple Adaptive Num. Substreams

Predictive Adaptive Num. Substreams

Figure C.31: Layered Substreams Count for manysrc

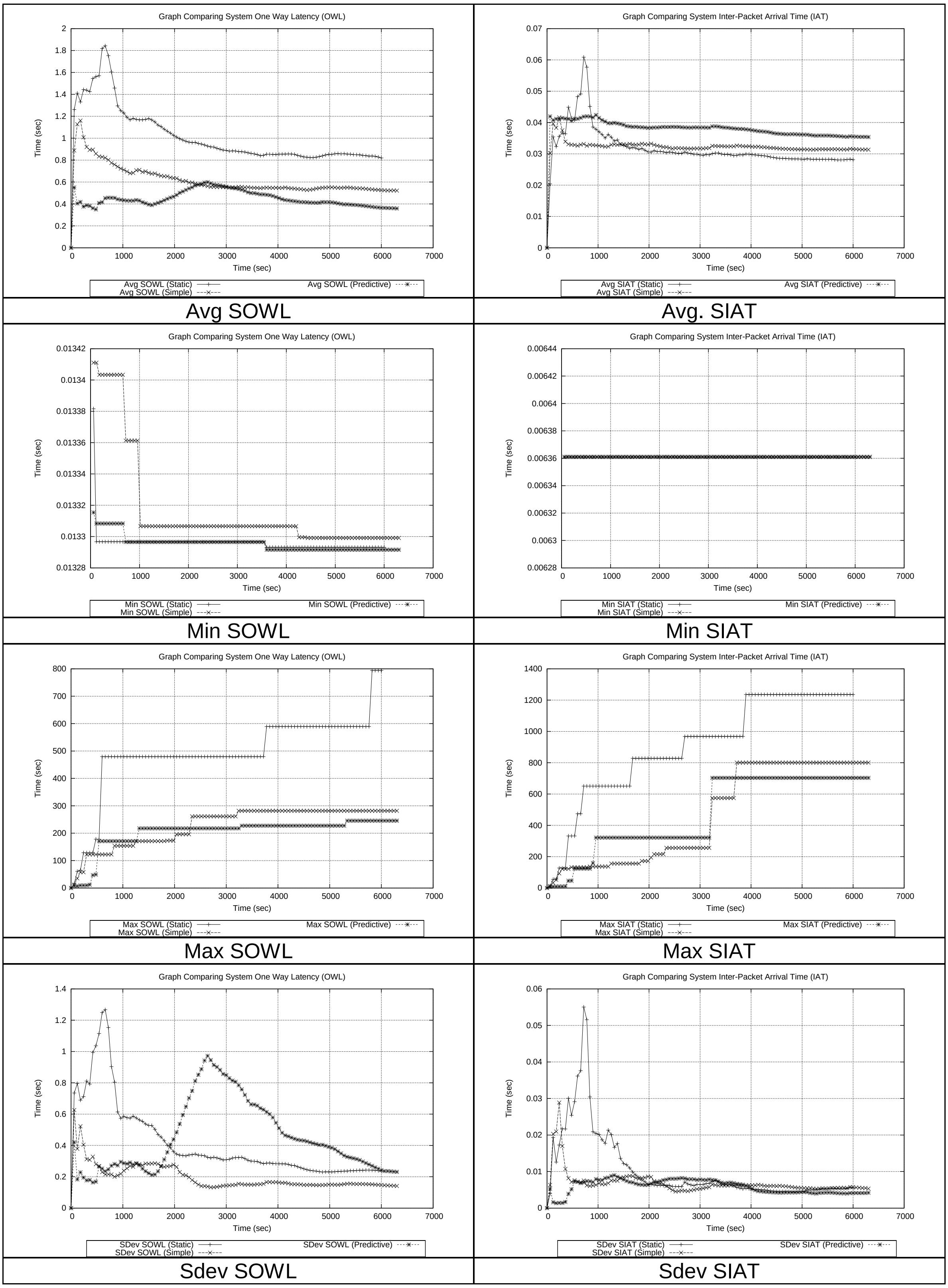


Figure C.32: SOWL and SIAT graphs for manysrc

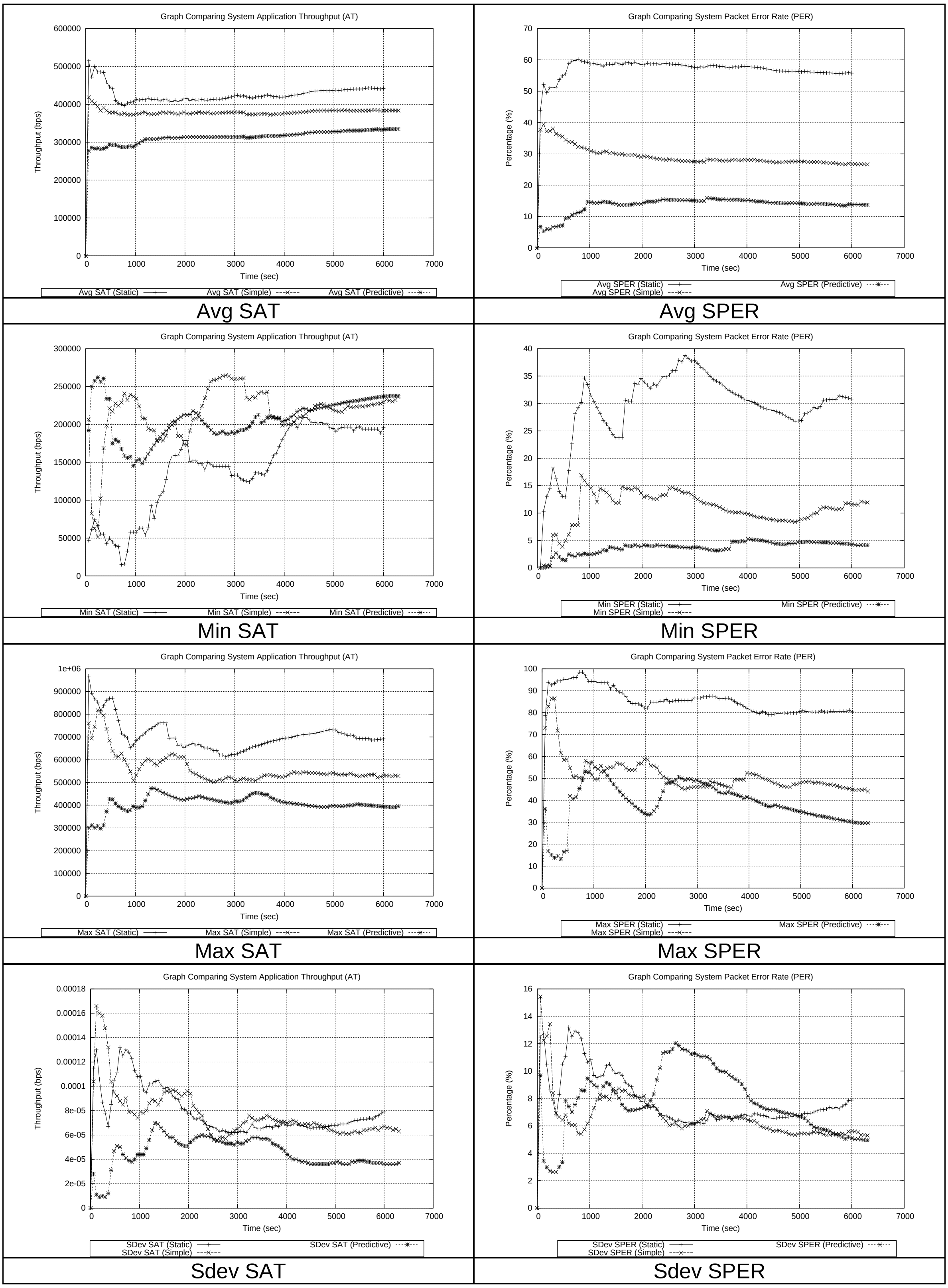


Figure C.33: SAT and SPER graphs for manysrc

VITA

Wan Tat Chee was born on December 30, 1967. He received his primary education at Kuen Cheng Primary School, Kuala Lumpur, Malaysia, and early secondary education at Bukit Bintang Boys Secondary School, Petaling Jaya, Selangor, Malaysia. He completed his secondary education at Raffles Institution, Singapore, and finished pre-university education at Raffles Junior College, Singapore.

In September 1987, he entered the University of Miami, Coral Gables, Florida, USA, and graduated *summa cum laude* with a Bachelor of Science in Electrical Engineering with Computer Option. He later obtained a Masters of Science in Electrical and Computer Engineering in August 1993 from the same university.

He joined Motorola Malaysia Sdn. Bhd. in 1994 as a software development engineer, and was promoted to senior software development engineer before leaving the company in 1998 to pursue his PhD in the area of wireless computing at the University of Science Malaysia (USM), Penang, Malaysia.

He currently resides in Penang, Malaysia with his wife and daughter.

LIST OF PUBLICATIONS

Tang, C. H. & Wan, T. C. (2000) Multicast Traffic Coordinator Framework for Real-Time Multimedia over Residential Networks. *IN: Proc. Int'l Symposium on Consumer Electronics (ISCE 2000),* Dec. 5-7, 2000, City Univ. of Hong Kong, Hong Kong. Available from: <http://www.ee.cityu.edu.hk/~ISCE2000/proceedings.html>.

Tang, C. H. & Wan, T. C. (2001) Performance of Multicast Traffic Coordinator Framework for Bandwidth Management of Real-Time Multimedia over Intranets. *IN: Proc. Joint Int'l Conf. MICC, LiSLO & ISCE 2001,* Oct. 22-24, 2001, Kuala Lumpur, Malaysia. IEEE Malaysia Section. pp. 223-229.

Wan, T. C. Leong, C. H & Thum, C. W. (1999) Multicast Firewall for Intranet Multimedia Applications. *IN: Borhanuddin, M. A. et al. ed. Computer and Communication Engineering: Proc. WEC '99,* Jul. 19-22, 1999, Subang, Kuala Lumpur, Malaysia. Universiti Putra Malaysia Press. pp 75-80.

Wan, T. C. Leong, C. H & Thum, C. W. (2001) Bandwidth Management for Intranets Using Multicast Firewalls. *PERTANIKA J. Science and Tech. Supp*. 9(2):159-172 (2001).

Wan, T. C. & Subramanian, R. K. (2004) Implementing QoS support for Wireless Multicast Nodes. IN: *Proc. Int'l Conf. on Communications 2004 (ICC 2004),* 20-24 June 2004, Paris, France, in press.

APPENDICES

VITA