QoS Enabled IP Based Wireless Networking: Design, Modelling and Performance Analysis

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Errata/Addenda

1) **IP micromobility management techniques**
The network architecture presented in chapter 3 of this thesis is based on the candidate's published work [34]. As [34] was published while most work on IP micromobility protocols management was still in their infancy, these are not referenced in the thesis and hence this addendum. Differentiated by operational scope, there are two basic types of mobility in Internet Protocol (IP) based wide-area wireless networks (W2ANs): macromobility and micromobility. Macromobility protocols manage mobility of mobile wireless users across domains (i.e. inter-domain mobility) or networks (i.e. inter-network mobility). Micromobility, on the other hand, deals with user mobility across subnetworks belonging to a single domain or network, and hence also referred to as intra-network or intra-domain mobility. The large-scale mobility (i.e. macromobility) is supported by the Mobile IP (MIP) protocol. The MIP protocol cannot support micromobility due the following reasons: (i) transferring large MIP control overhead over a wireless link each time a mobile user moves from one subnet to another is inefficient; (ii) long handoff delay inherent in MIP causes packet losses which can destabilise TCP operation; (iii) re-establishment of QoS reservations between FA and HA after every move of a QoS-enabled mobile host. Techniques proposed in the literature to support micromobility in IP based W2ANs fall into two categories: (a) exploitation of link layer signalling of the radio-networking standard used or (b) the application of autonomous micromobility protocol. The networking architecture (referred to as MOWINTA) proposed in Chapter 3 of this thesis adopts the former approach by exploiting wireless link layer signalling to reduce mobile IP handoff delay and saving TCP instability caused by packet losses during long handoff. Several micromobility architectures falling into the latter category are proposed, such as HAWAII [a], Cellular IP (CIP) [b] and Hierarchical Mobile IP (HMIP) [c].

2) (a) Page 76, Sect. 3.2.2, 1st paragraph: all "minimum" should be replaced by "average" and the average movement detection delay should be $t_{ECS} = t/2$ (b) Page 76, Sect. 3.2.2, 2nd paragraph: Should read " ... in contrast to the 1 sec Mobile IP inter-agent advertisement time. Therefore ... can be reduced by a factor of 10 (i.e. 1sec/100 ms)."

3) Contradiction between Sections 3.2.1 and 3.2.2: It should be modified as "The designed network architecture protects modification of TCP by moving the protocol modifications via intercommunication down one layer, i.e. intercommunication between layer 3 and layer 2 as this seems easier and cheaper. The reason being that TCP is in wider use than any given wireless link layer, as both wired and wireless networks can use TCP."

4) The architecture, MOWINTA, is actually designed to run any application-oriented QoS mechanism, such as IntServ or DiffServ. Table 3.2 in page 93 shows how one can be mapped onto the other. Hence, page 84 should have stated, "although the presented architecture can use either IntServ or DiffServ, most part of the presentation is based on IntServ with Table 3.2 aiding mapping."

5) Page 86: It is assumed that the features of the wireless standard (here IEEE 802.11) are exploited to support the necessary QoS of internal sessions. The IEEE 802.11 has inbuilt mechanisms such as priority schemes to support QoS. How efficient this approach works is still a research topic.

6) Figure 3.5: the transport protocol can also be UDP instead of TCP. Hence, should read TCP/UDP.

7) Page 96, 7th line from top: Replace Figure 10 by Figure 3.6.

8) Section 3.5.2: The database aspect and paging format of location management are outside the scope of the thesis. However, implications for location databases are potential areas for further studies.

9) Page 107, Sect. 4.2.2: The functions $f_1(/cdots)$ and $f_2(/cdots)$ are objective functions which are defined in Equations (4.4) and (4.5). Figure 4.2 illustrates the cumulative service (bottom figure) received by two mobiles undergoing difference instantaneous channel qualities (top figure). This figure assumes that mobile 1 is backlogged with traffic of higher QoS class GeS_1 than mobile 2 that is backlogged with traffic belonging to GeS_2. By normal priority scheduling mobile 2's traffic would never be scheduled until all packets belonging to mobile 1 are scheduled. However, the proposed channel-aware scheduler schedules packets of mobile 2 if its channel quality is much better than that of mobile 1 at the scheduling instant, although mobile 1 is still backlogged. Hence, the scheduler inherently protects both active mobiles from complete service starvation.

10) Equation 5.12 is derived assuming that ACK/NACK never gets corrupted. This seems a reasonable assumption given the low information rate of ACK/NACK feedback channel and the high error immunity. Such assumption is reflected in the literature.

11) Page 112: The discussion in Page 112 assumes the existence of a feedback wireless channel. Feedback of estimated wireless channel state is commonly used in many radio communications standards for purposes such as power control, load control and handoff decisions. The mobile terminal usually communicates such feedback information using the uplink pilot channel. However, a pilot signal need not be transmitted on a dedicated channel. Hence, the designed scheduling scheme does not increase the system complexity by requiring link quality feedback, as it can use the feedback channel of the respective wireless standard. Hence, the complexity associated with the feedback mechanism is not expected to have considerable additional implementation and cost implications to the entire communication system.

Additional References

[a] R. Ramjee, K. Varadhan, L. Salgarelli, S.R. Thuel, S-Y Wang and T. La Porta, "HAWAII: A domain-based approach for supporting mobility in wide-area wireless networks," *IEEE/ACM Trans. Netw,* vol 10, no. 3, June 2002, pp. 396–410.

[b] A Valko, "Cellular IP: a new approach to Internet host mobility," *ACM SIGCOMM Comp. Commun. Rev.,* vol. 29, no. 1, Jan 1999, pp. 50-65.

[c] E. Gustafsson, A. Jonsson and C. Perkins, "Mobile IP Regional Registration Registration," Internet draft, July 2000.

[d] A. T. Campbell, J Gomez, S Kim, and C-Y Wan, "Comparison of IP micromobility protocols," *IEEE Wir. Commun,* Feb 2002, pp. 72–82.

Supervisor: **Date:** June 13, 2003

Professor Reginald P. Coutts

In memory of my late dad, Mr Kofi Gyasi-Agyei

Declaration of Originality

This thesis contains no material which has been accepted for the award of any other degree or diploma in any university and that, to the best of the candidate's knowledge and belief, the thesis contains no material previously published or written by another person, except where due reference is made in the text of the thesis.

The author consents to the thesis being made available for loan and photocopying.

Signed: **Date:** February 21, 2003 **Place:** Adelaide, SA

Acknowledgments

I am greatly indebted to the Almighty God and His Son Jesus Christ and the Holy Spirit—The Creator of Heaven & Earth and everything in it, The Omniscience & The Omnipresence, The Alpha & The Omega—for His unfathomable mercy, grace and love to me throughout my life, especially during the period of my PhD candidature at Adelaide University. I will forever praise Him so long as He gives me breath.

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A/Prof. S. -L. Kim of RAMO Lab, Information and Communications University (ICU), South Korea, hosted me in his laboratory for 6 weeks in mid-2002. Some academics I met right from the beginning of my career merit acknowledgement. Notable amongst them are Prof. Timo Laakso of Helsinki University of Technology (HUT), Finland, through whom I learnt the "hows" of basic research; Prof. Seppo Halme, also of HUT, provided the opportunity to develop my research career. Dr Hochhaus, Prof. Ackermann and Prof. Schuneman, all of Hamburg-Harburg University of Technology (TUHH), Germany, employed me as Engineering Asssistant (HIWI) throughout my undergraduate degree at TUHH. This enabled me finance my education without a formal scholarship, as well as giving me insights into practical engineering right from undergraduate level.

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Thanks, Merci, Danke Schöne, Tack, Kiitos, Aseda: Amoakoh

Abstract

Quality of service differentiation has never achieved much attention and relevance until the advent of the convergence of mobile wireless network and the fixed Internet, that is, Internet Protocol (IP) based mobile wireless networks, or wireless Internet. These networks are poised to support multimedia applications' traffic with diverse QoS sensitivities. To date, most traffic transferred over the Internet still undergo best-effort forwarding, which does not guarantee whether or not traffic sent by a source gets to the intended destination, let alone loss and timing bounds. The major contribution of this thesis is three-fold.

First, the thesis proposes a QoS-enabled wireless Internet access architecture, which leverages the micromobility in wireless standards to reduce mobile IP weaknesses, such as long handoff delay, to achieve effective interworking between mobile wireless networks and the global, fixed Internet. Although the idea here is applicable to any wireless standard, the design examples in this thesis are based on the IEEE 802.11b wireless local area network (WLAN) standard.

Second, it proposes a framework for a class of wireless channel state dependent packet scheduling schemes, which consider the QoS requirements of the applications' traffic; the wireless channel state (reflected in instantaneous data rate or noise level); and optimises the usage of the expensive wireless resource. The operation of the QoS-enabled, channel state-dependent packet scheduler is analysed using optimisation theory, eigenanalysis and stochastic modelling.

Third, the thesis analyses the effects of wireless channel properties on differentiated QoS (DQoS) schemes, using two-dimensional, channel-state-dependent queuing theory, matrix analytic methods to stochastic modelling and eigenanalysis. The analytical model of DQoS schemes, especially models accounting for user scenarios such as speed of motion and wireless channel properties, such as fading, spatio-temporarily varying quality and low rate, is not properly covered in the open literature, and hence was a motivation for this part of the thesis. The wireless channel is discretized into discrete-time Markovian states based on the received signal-to-noise plus interference ratio (SNIR), which also reflects on the instantaneous link quality. The link quality, in turn, influences the QoS experienced by the transported applications sitting on top of the ISO/OSI protocol hierarchy. The parameters of the Markovian states are evaluated using realistic physical channel noise models and transceiver characteris-

tics, such as modem. [Different modems (modulator/demodulator) yields different transceiver properties such as sensitivity. The analysis in the thesis adopts QPSK and BPSK modulation.] Source traffic models are used in the analysis.

Lastly, the thesis provides an extensive introduction to, and provides a detailed background material for the new area of mobile wireless Internet systems, upon which considerable future research can be based.

Publications

The following are some of the publications of the candidate which are related to the theme of this thesis.

1. A. Gyasi-Agyei, "Mobile IP-DECT Internetworking architecture supporting IMT-2000 applications," *IEEE Network*, vol. 15, no. 6, Nov/Dec 2001, pp. 10-22.

2. A. Gyasi-Agyei and R. Coutts, "Analytical model of a Differentiated Service scheme over Wireless IP Links," In *Proceed. IEEE Int. Conf. on Networks (ICON'02)*, Singapore, 27 - 30 August 2002, pp. 223-228.

3. A. Gyasi-Agyei, "Service differentiation in wireless Internet using multi-class RED with drop threshold proportional scheduling," In *Proceed. IEEE Int. Conf. on Networks (ICON'02)*, Singapore, 27 - 30 August 2002, pp. 175-180.

4. A. Gyasi-Agyei, "Performance Analysis of a Differentiated Services over Wireless Links," In *Proceed. 5th IEEE Int. Conf. on High-Speed Networks and Multimedia Commun. (HSNMC)*, Jeju Island, Korea, July 2002, pp. 86-90.

5. Amoakoh Gyasi-Agyei, "A Differentiated Services Scheme for Wireless IP Networks," In *Proceed. IEEE Int Conf. on Telecom (ICT'02)*, vol. 2, Beijing, China, 23-26 June 2002, pp. 361-366.

6. Amoakoh Gyasi-Agyei, "Performance Analysis of a Differentiated Services Scheme over Fading Channels," In *Proceed. IEEE Int Conf. on Telecom (ICT'02)*, vol. 1, Beijing, China, 23-26 June 2002, pp. 1155-1160.

7. A. Gyasi-Agyei, "EGPRS/EDGE random access performance using M-PSK in AWGN and Nakagami-m fading channel," In *Proceed. Int. Conf. on Inform., Commun. & Signal Proc. (ICICS)*, Singapore, Oct. 2001, 5 pp.

8. A. Gyasi-Agyei, "QoS guarantees in IP based wireless/mobile networks," Research Proposal, EEE Dept, Adelaide University, Australia, 5th October 2001, 9 pp.

9. A. Gyasi-Agyei and S. J. Halme (eds), *Network and telecommunications signaling architectures for contemporary and future broadband intelligent networks*, ISBN 951-22-4982-0, ISSN 1456-3835.

10. A. Gyasi-Agyei and S. J. Halme, "A novel planning of 3G all-wireless access network," in *Proc. IEEE Int. Conf. on Inform., Commun. & Sig. Proc. (ICICS)*, Singapore, Dec. 1999, 5 pps.

11. A. Gyasi-Agyei, S. J. Halme and Sarker J, "GPRS–Features and Packet Random Access Channel Performance Analysis," in *Proc. IEEE Int. Conf. on Networks (ICON'00)*, Singapore, Sept. 2000, pp. 13–17.

12. A. Gyasi-Agyei and S. J. Halme, "Mobile IP based DECT multimedia architectures for IMT-2000," in *Proc. IEEE Veh. Techn. Conf. (VTC'00)*, vol. 2, Boston, MA, Sept. 2000, pp. 963–970.

Papers in Review

1. A. Gyasi-Agyei, "Performance analysis of differentiated services QoS model over multi-state wireless Links," submitted to *Wiley Wireless Communications & Mobile Computing J.*, Dec. 2001, 20 pps.

2. A. Gyasi-Agyei, "Fluid Analysis of a Channel State Dependent Packet Scheduler over Fading Wireless Channels," submitted to *Kluwer Wireless Networks J.*, March 2002.

3. A. Gyasi-Agyei, "MOWINTA—A QoS enabled wireless Internet access architecture based on Mobile IP/IEEE 802.11 interworking," submitted to *special issue* on QoS in next-generation multimedia communications systems *IEEE Wireless Commmun. Mag.*, Dec. 19, 2002.

4. A. Gyasi-Agyei, "Some wireless effects on QoS provisioning in IP based networks," submitted to *IEEE/ACM Trans. on Netw.*, Jan. 21, 2003.

Notable Seminars and Workshops

The following is a list of some of the lectures, seminars and workshops I presented during my PhD candidature besides presentation at international conferences.

1. Gyasi-Agyei, A., "Performance analysis of the packet random access channel of GPRS," Seminar, Institute for Communications Research (ICR), National University of Singapore (NUS), 2 Feb. 2001.

2. A. Gyasi-Agyei, "QoS guarantees in IP based wireless/mobile networks," Research Proposal Seminar, EEE Dept, Adelaide University, 5 Oct. 2001.

3. A. Gyasi-Agyei, "Performance analysis of DiffServ QoS model over multi-state wireless links," 6th Annual Melbourne-Adelaide Teletraffic Workshop, 12-14 Dec. 2001, Grampians, Vic, Australia.

4. A. Gyasi-Agyei, "Differentiated QoS schemes for wireless internet," 1st Workshop for CRC Smart Internet Technology scholars, Sydney, 8-10 May 2002.

5. A. Gyasi-Agyei, "BL^4DF – Wireless Channel State Dependent Packet Scheduling for QoS Provisioning in Multiservice Wireless Networks," Seminar, Computer Science & Eng (CSE) Dept, UNSW, Sydney, 18 Sept. 2002.

6. A. Gyasi-Agyei, "The Innovative Utilisation of Communications/Electronic Engineering in the Development and Growth of Central Queensland Industry and Education," Lecture, Central Queensland University (CQU), Rockhampton, 11 Dec. 2002.

Other publications of the candidate can be found at the bibliographic section.

Nothing tends so much to the advancement of knowledge as the application of a new instrument.The native intellectual powers of men in different times are not so much the causes of the different success of their labours, as the peculiar nature of the means and artificial resources in their possession – Sir Humphrey Davy.

Contents

When I was still a rather precocious young man, I already realized most vividly the futility of the hopes and aspirations that most men pursue throughout their lives – Albert Einstein.

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List of Symbols

B_c	Coherence bandwidth
B_s	Signal bandwidth
β_n	Bandwidth efficiency in link state n
C	Cluster size
c	Number of traffic classes/generic streams differentiated
$D_i,\ d_i$	Delay parameter of GeS_i
d_i^*	Optimum value of d_i
f_c (Hz)	Operating radio frequency (carrier)
$f_\gamma(\gamma)$	Probability density function
γ	Signal-to-noise ratio
$\mathbf{I}$	An identity matrix of a given order
$\mathbf{I}(n)$	An identity matrix of order n
$K,\ N_s$	Number of traffic sources
κ	Information part of L2 codeword
λ	Average packet arrival rate
$L_i,\ l_i$	Loss parameter of GeS_i
L_p	Layer 3 (IP) packet size
$L2hdr$	Layer 2 frame header field for error control
$m,\ m \geq 1/2$	Nakagami$-m$ fading figure
μ (dB)	Mean of $10\log_{10}\gamma$
$N,\ N_c$	Number of discrete wireless channel states
ν	L2 codeword size
$N_0/2$ (W/Hz)	Two-sided power spectral density of AWGN
$\mathbf{P}$	Transition probability matrix
$\mathbf{p}$	A vector of probabilities
$P_{b,e}$	Bit error probability

P_e	Error probability
p_i	Service cost (premium) for traffic belonging to GeS_i
p_{ij}	Elements of $\mathbf{P}$
$\hat{r}_{arq},\ \hat{r}$	Maximum number of ARQ retransmissions
$R_b,\ R_p$	Interface transmission rate
$r_m(t)$	Link rate of mobile m at time t
ρ	$E[\gamma]$, i.e. mean of γ
σ (dB)	Standard deviation of $10\log_{10}\gamma$
s_n	Wireless channel state (condition)
$\mathbf{T}$	Infinitesimal generator (state transition rate matrix)
t_{ij}	Elements of $\mathbf{T}$
τ	# number of bit errors per L2 frame correctable by an FEC code
$v,\ v_m$	Mobile user velocity of motion
v_0	Speed of light in vacuum ($\approx 3\cdot 10^8$ m/s)

It is often stated that of all the theories proposed in this century, the silliest is quantum theory. In fact, some say that the only thing that quantum theory has going for it is that it is unquestionably correct – Michio Kaku.

List of Acronyms

AFD	Average Fade Duration
AF	Assured Forwarding PHB
AP	Access Port
API	Application Programming Interface
ARQ	Automatic Repeat reQuest
ATM	Asynchronous Transfer Mode
AWGN	Additive White Gaussian Noise
BEF	Best Effort Forwarding
BER	Bit Error Rate
BL^2DF	Best Link Lowest Delay First scheduler
BL^4DF	Best Link Lowest Loss Lowest Delay First scheduler
BL^2PF	Best Link Largest Premium First scheduler
BM	Buffer Manager
BPSK	Binary Phase-Shift Keying
BS	Base Station
BSS	Basic Service Set
CBB	Class Based Buffering
CDMA	Code Division Multiple Access
CIDR	Classless Inter-Domain Routing
CL	Controlled Load service of IntServ
CN/CH	Corresponding Node/Host
CO	Connection-Oriented (service of 802.11's LLC)
CS	Circuit Switching
CSDPS	Channel State Dependent Packet Scheduling
DB (dBase)	Database (Register)
DECT	Digital Enhanced Cordless Telecommunications
DiffServ	Differentiated Services architecture
DQoS	Differentiated QoS
DS	Same as DiffServ
DT-FSMM	Discrete-Time Finite State Markov Model
EF	Expedited Forwarding PHB

FCFS	First Come First Served scheduling
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FWR	Fixed Wireless Router (= IBS)
GEC	Two state Gilbert-Elliot Channel model
GeS	Generic Stream
GPRS	General Packet Radio Service
GPS	Generalized Processor Sharing scheduling
GoS	Grade of Service
GS	Guaranteed Service of IntServ
GW	Gateway
HM	Handoff Management
HLC	Home Location Register
IBS	IP enabled Base Station
IEEE	The Institute of Electrical & Electronic Engineers
IMS	IP enabled Mobile Station
IntServ	Integrated Services architecture
IP	Internet Protocol
IP-RAN	IP based RAN
IS	Same as IntServ
ISO	International Standards Organization
Lk	ISO/OSI protocol layer k
LCR	Level Crossing Rate
LLC	Logical Link Control
LM	Location Management
MA	Mobility Agent
MAC	Medium Access Control
MH/MN	Mobile Host/Node
MIP	Mobile IP
MM	Mobility Management
MMPP	Markov-Modulated Poisson Process
MODEM	Modulator/Demodulator
MOWINT	Mobile Wireless Internet
MOWINTA	Mobile Wireless Internet Access Architecture(s)
MPLS	Multi-Protocol Label Switching

MRSVP	Mobile RSVP
MS	Mobile Station
MWR	Mobile Wireless Router
NAT	Network Address Translation
NLN	Nakagami Lognormal channel impairment
NP^3A	Non-Pre-emptive Priority with Partial Assurance scheduling
OSI	Open Systems Interconnection
PBSP	Proportional Buffer Sharing with Pushout
pdf	Probability Density Function
PHB	Per-Hop Behavior
POA	Point of Attachment
PS	Packet Scheduler / Packet Switching
QDS	QoS enabled Distribution System
QoS	Quality of Service
QPSK	Quadrature Phase-Shift Keying
RAN	Radio Access Network
RED	Random Early Detection/Discard
RLC	Radio Link Control
RSIP	Real Specific Internet Protocol
RSSI	Received Signal Strength Indicator
RSVP	Resource ReSerVation Protocol
SNIR	Signal-to-Noise plus Interference Ratio
SNR	Signal-to-Noise Ratio
STA	Station (i.e. handset)
TC	Traffic Class/Classifier
TCP	Transmission Control Protocol
TD	Tail/Threshold Dropping
TDMA	Time Division Multiple Access
3G	Third Generation Wireless Networks
UC	Unacknowledged Connectionless service of 802.11's LLC
VLR	Visitor Location Register
WLAN	Wireless Local Area Network
WR	Wireless Router

Chapter 1

Introduction

1.1 Motivation

The unstoppable convergence of *fixed* Internet and *mobile wireless* networks [34] to mobile wireless Internet (abbreviated in this thesis as MOWINT) requires engineering professionals with knowledge in both converging industries. This is because, just as the two industries were traditionally separated, so have been their issues and their professionals. A lot of research has been done in both fields of the fixed Internet and the wireless networks. However, research integrating the issues in both fields is scanty, especially in the analysis of wireless channel dependent architectures to support differentiated quality of service (DQoS). MOWINT systems need to support heterogeneous multimedia traffic, and hence need mechanisms to provide differentiated quality of service (DQoS). Also, there are only few analytic models [18], as well as performance analysis of DQoS schemes, accounting for wireless inherent features, such as temporarily and spatially (i.e., spatio-temporarily) varying state (condition) due to fading, a dynamically changing user/network interface due to mobility, and limited bandwidth.

This thesis examines wireless effects on differentiated QoS, designs a QoS enabled wireless Internet architecture, and analyses the performance of some QoS-aware wireless protocols. The originality of this thesis partly lies in the integration of wireless physical layer issues (e.g. radio fading, error control and modulation) and higher layer and networking issues (e.g. packet scheduling), endeavours which are conventionally separated. This thesis deals with only *terrestrial* wireless networks.

1.2 Mobile Wireless Networks

The beginnings of Telecommunications engineering can be traced back to the invention of the *wireline* telephone by Alexander Graham Bell in 1870. In 1873, Maxwell postulated wireless communications by radio transmission, which was demonstrated practically by Heinrich Hertz in 1888, when he produced and detected radio waves over short distances of the order of few meters across his laboratory. Motivated by Hertz's work, Guglielmo Marconi (04/1874–07/1937), an Irish Italian, then pioneered a practical radio transmission and detection over long distances in the order of several kilometres in 1895 [46]. Radio then extended the traditional telephone network to moving vehicles, pedestrians, and to the home. Since then wireless communications has not been confined to ship-to-shore and ship-to-ship applications. Rather, it has evolved through applications in land and space vehicles to the more sophisticated personal cellular and cordless applications we use today. Kindled by the desire for freedom from *tethered* communications by mankind, Marconi's work paved the way for further development in wireless technologies, which has enjoyed exponential growth in the past decade. The desire for *anytime, anywhere* (i.e. ubiquitous) information exchange at affordable tariffs is not new, as our forerunners used such primitive means like smoke signals, talking drums, birds, petragraphs, etc. for communications in the early days.

The first mobile radio system was lunched in 1928 by the US Police Department in Detroit. However, mobile cellular telecommunications, as it is known today, began only in the late 1970s. The Japanese NTT (Nippon Telephone and Telegraph) became the first operational analogue cellular system in 1979. The Swedish Ericsson Radio Systems AB then introduced the Nordic Mobile Telephone (NMT) in 1981. In 1983, the North American AMPS was field tested in Chicago and Baltimore by AT&T [97]. The standardization of GSM, the first digital cellular system, began around 1984, but its deployment occurred not until around 1992. Then followed IS-54 (US TDMA), Japanese JDC, US E-TDMA, US CDMA (IS-95) voice and IS-99 data standards, in that order.

The wireless access industry has caught much attention in the industrialized world mainly because of the desire for *mobility*—an enhanced service. This has resulted in a paradigm shift in mobile communications— mobile terminals from being social status symbols into essential communications tools for everyone. However, in the economically yet-to-blossom communities, mobility is not as important as having a quick and cheap access to a basic telephony (*lifeline*) service since the infrastructure is hitherto not there, or not well developed. Wireless communications ranges from a simple fixed wireless access (FWA) without mobility, as in conventional wireless local loop (WLL) applications, to sophisticated, multicellular mobile networks with handoff and mobility. In FWA systems, the entire local loop is absolutely replaced with radio paths, microwave paths, infrared paths or combinations thereof, instead of using wires. Here, the local loop is assumed to be the section of the telecom infrastructure between the central office (CO) or switch and the customer premise equipment (CPE). It consists of the feeder, the distribution and the drop sections. This offers several inherent advantages as discussed in this thesis.

If wireless networks are to compete with the legacy wired access technologies (e.g., toll quality POTS, ISDN, cable 'phone') as well as new technologies, such as, digital subscriber lines (DSL) and fiber to the home (FTTH)), for a sustainable market share, then they should also deliver competent services. An advantage of wireless, in contrast to wireline, communications is the fact that wireless can "go" everywhere and can offer person-to-person, rather than desk-to-desk, communications. This requires wireless system to have comparable grade of service (GoS) and quality of service (QoS) as in wired systems. High GoS is characterized with low network access blocking of the network, marginal call dropping rate due to handoff failure and low outage or high area coverage probability. High QoS, on the other hand, is inherent in low end-to-end service latency, sufficient network capacity and acceptably small errors (bit/frame error rate, or BER/FEP).

The challenge in planning today's wireless networks is compounded by the requirements of multi-standard and heterogeneous 3^{rd} generation (3G) networks. The 3G systems enhance the air interface of 2G (or 2.5G) systems so rendering them capable of supporting higher-rates to support novel services and applications. For example, EDGE (Enhanced Data for GSM Evolution) uses 8-PSK (Phase-Shift Keying) modulation instead of GSM's Gaussian minimum shift keying (GMSK) to offer a factor three-bandwidth efficiency enhancement. The major new capabilities of 3G networks are *larger bandwidth* and flexible (asymmetric) bearers to *support multimedia* (variable rate) services; bearer service classification; provision of service-supportive standard-bearer capabilities instead of standard services.

Table 1.1 compares several features and the developments in mobile wireless networks from the last three to the next decade. The generations of terrestrial wireless networks, as shown in Table 1.1, have mostly been evolutionary, rather than revolutionary, in order to leverage existing infrastructure and return on capital. These service-enriching features are to be provided alongside *service portability, terminal* and *personal mobility* to everyone, everywhere, and at every time at consumer tariffs. Everywhere interconnectivity promotes global roaming, as mobile users can be connected seamlessly over large geographical regions.

Service-proof[1] wireless systems are required for long-term return on investment to enhance the delivery of affordable telecommunications services, which, in turn, promotes network penetration resulting in greater profitability of the operator. Service-proof networks permit speedy and yet efficient introduction of new services into the network at minimal costs. Mobile wireless communications services have enjoyed dramatic uptake (Fig. 1.1) but with some reduction with the recent techno-economic downturn from year 2000.

[1] Unlike e.g. the PSTN, which was designed for a single voice service, service-proof networks are not designed for a specific service. This extends the life of the system as it continually accommodates new services.

Table 1.1: Generations of mobile wireless networks and features.

Generation	1^{st}	2^{nd}	2.5^{th}	3^{rd}	4^{th}
Air interface	analog	digital	digital	digital	digital
	CS	CS	CS & PS	CS & PS	PS (all-IP)
	FDMA	TDMA	TDMA	W-CDMA	OFDMA?
		DS-CDMA	DS-CDMA		CDMA?
RAN backbone	CS	CS	CS	CS & PS	PS
Core network	CS	CS	CS & PS	CS & PS	PS (all-IP)
Major services	voice	voice	voice	voice	packet voice
			text-only Internet	text & images Internet	rich Internet
System example	AMPS	GSM	GPRS	WCDMA	high-speed wireless Internet
	TACS	IS-95 (cdma)	IS-136 (D-AMPS)	EDGE	
	NMT	PDC	i-mode		wireless LANs
	NTT	HSCSD			
Data rates	9.6 kb/s	9.6-64 kb/s	28.8-164 kb/s	144-2000 kb/s	~11-25 Mb/s
Scope of std.	Regional	Continental	Continental	Global	Global
Roll-out dates	1980’s	1990’s	2000’s	2000^{+}	2010’s

1.2.1 Mobile Wireless Network Architectures

Mobile wireless networks can be classified into infrastructure and infrastructureless networks. Examples of infrastructure networks are the traditional mobile networks which have fixed network elements such as base stations, radio network controllers and routers. Such networks require sophisticated network planning prior to roll-out, and they have stable topology. Fig. 1.2 shows an architecture of an infrastructure network.

Infrastructureless networks, on the other hand, require no prior planning, no fixed network nodes, and have changing topologies. Such networks are commonly referred to as ad hoc networks, and they form a key technology for pervasive or mobile computing. They are most useful during emergencies and disasters. Fig. 1.3 shows an architecture of an infrastructureless network. Mobile

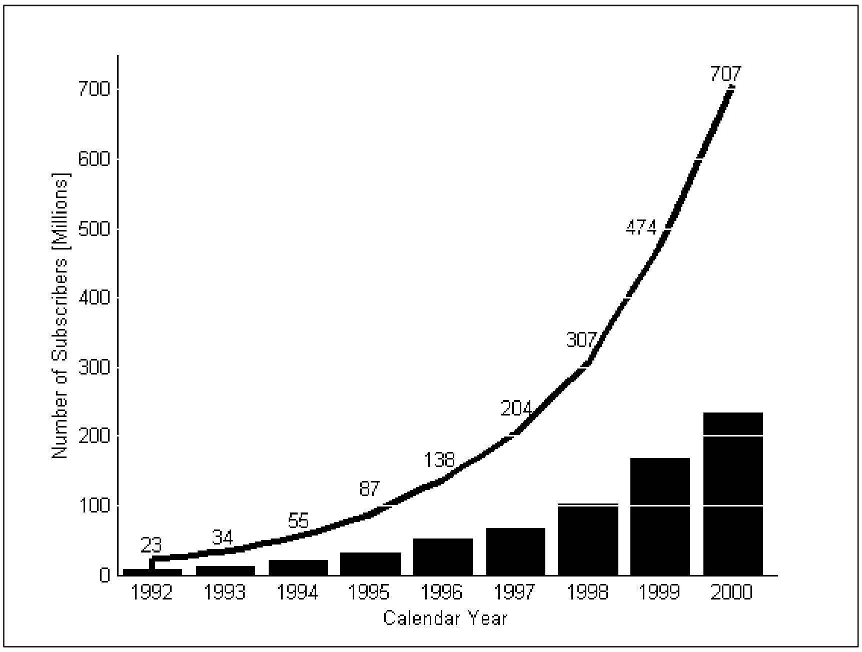


Figure 1.1: Mobile cellular technology subscription worldwide, shaded bars show in-year net gains [Source: [34] ©IEEE].

handsets used in infrastructureless networking need packet forwarding (routing) capabilities, and hence the coining of the term *Mobile Wireless Routers* (MWR). This thesis, however, focuses on infrastructure networks.

1.2.2 Basics of Mobile Cellular Networks

The cellular architecture, as illustrated in Fig. 1.4, was adopted in mobile networks to allow the usage of a limited radio frequency (RF) spectrum over a wide geographical area. The available RF spectrum is first partitioned into discrete frequency bands or carriers (except CDMA systems), and these bands are, in turn, grouped into, say, C groups. Radio nodes, or base stations (BSs), separated spatially as wide as possible are positioned in the coverage area, each of which is assigned one of the C channel groups. A bunch of C tessellating cells with different RF carrier groups is called a *cluster*, and the same set of

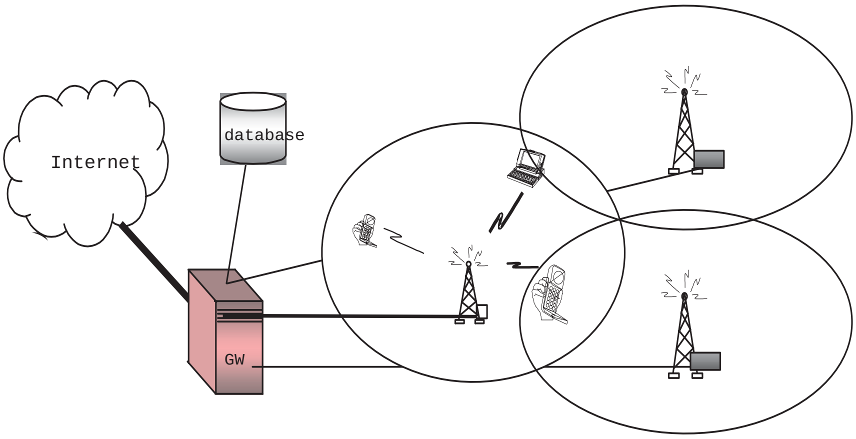


Figure 1.2: An architecture of an IP based infrastructure mobile network.

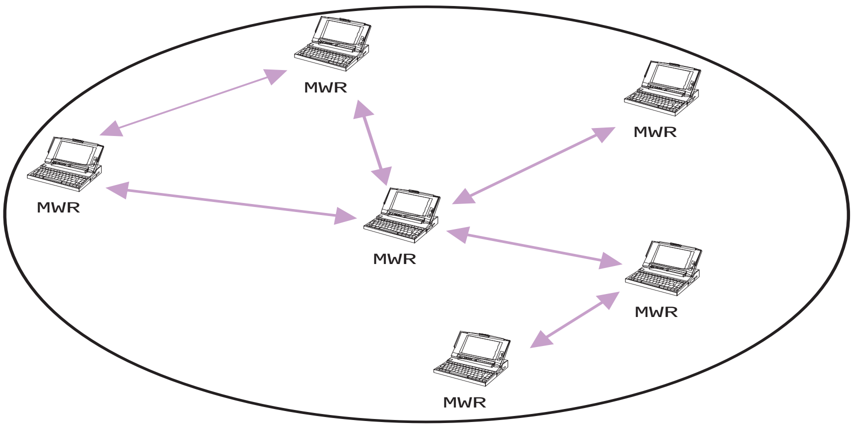


Figure 1.3: An architecture of an IP based infrastructureless mobile network.

frequencies are used in each cluster. However, cells using the same set of RF frequencies need be separated sufficiently to reduce co-channel interference. The *cluster size* obey the relation

$$C = m^2 + mn + n^2, \quad m, n \in Z^+, m \geq n \tag{1.1}$$

Commonly used cluster sizes are $C = 3, 4$ and 7. For example, AMPS and IS-54 (commonly called TDMA) use $C = 7$. It is noted that CDMA systems have *frequency reuse* of $C = 1$. Figure 1.4 is based on a 7-cell frequency reuse, where $f_k, k = 1, 2, ..., 7$, in general, represents a group of RF carriers. The

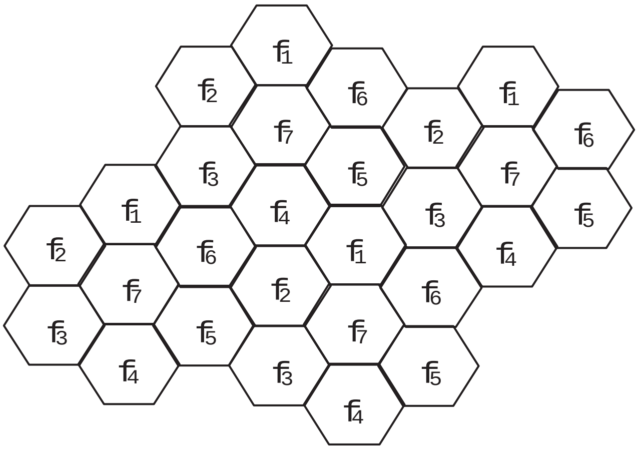


Figure 1.4: An idealized cellular topology with a 7-cell frequency reuse.

hexagonal cell representation is just an idealization, and it is adopted for two reasons: it closely approximates a circle, and offers a wide range of tessellating cluster sizes to cover a cellular region. A more realistic infrastructure network architecture with oval cells is shown in Fig. 1.2.

Multiple access schemes are used by the Media Access Control (MAC) protocols (see Section 2.3) to allow as many mobiles as possible to access the radio base stations. The number of BSs used to cover a geographical area is a matter of trade-offs between network quality (performance), expected network capacity and costs. For economic reasons, the coverage area may be covered with as few radio nodes as possible. However, physical radio propagation effects limit the minimum number of fixed radio nodes needed for a specified geographic region. Radio cells can be partitioned into smaller cells to increase network capacity with the same RF spectral bandwidth. This approach, however, is penalised by frequent intercell handoffs, increased network interference, and increased costs of radio equipment and transmission/interconnection links. Cellular systems can be said to be *coverage* (*range*) limited or *capacity* (or *interference* or *bandwidth*) *limited*. In contrast with coverage-limited systems, in capacity-limited systems, the number of transmitters is large compared to the available bandwidth [114].

Antennas. Wireless devices (e. g., MWRs, FWRs) require antennae to transmit/receive radio signals. Antennas are of two basic types: directive and omnidirectional (or simply, omni) antennae. In contrast to omni, directive antennas concentrate power (energy) in specific directions where the interesting radio signals are propagating. As such, directive antennas have lower power requirements, have smaller delay spread, and reduced interference from/to other antennas or cells. An antenna, which adaptively adjusts its radiation patterns to track the location of mobiles, have caught much attention in the recent years. Such antennas are referred to as *smart antennas,* as they are equipped with more signal processing intelligence than the conventional antennas.

1.2.3 Drivers of Wireless Networks

Besides *mobility* (i.e. 'everywhere' connectivity), there are a number of stimuli promoting the penetration of wireless networks. Below the most significant ones are briefly touched upon [46].

Just-in-time rollout. Unlike fixed network services, which are delivered when structures are built whether there will be a need for them or not, wireless access services are delivered to the customer when and where they are needed. This just in time (early enough) service rollout reduces the upfront investment as outlined below. This guarantees minimal investment in resources that generate no revenue in the short term. In many countries state laws mandate the installation of wires by the local Telcos in all buildings during construction.

Low up-front investment in network infrastructure. Avoidance of digging reduces the initial investment needed to provide the wireless access services. In some cases no special masts are even needed for the radio nodes. This is largely due to the ever-increasing reduction in size of the radio nodes which permits installation on existing structures (houses, utility pylons, etc.) by paying little or no rent. This reduces financial risk and the initial capital requirements before network rollout. The network can even be installed in phases and revenue from earlier phases could finance subsequent ones. Furthermore, as reported by Prasad in [90], linking homes to the core network by wire amounts to about 46% of the total network infrastructure cost. Thus migrating to, or choosing a wireless access network, could achieve a remarkable cost savings.

Fast network deployment and service delivery costs. Fast network (service) rollout is also enhanced by the avoidance of digging. Rapid service delivery is needed to win more customers in the ever-competitive telecommunications market, or prevent long waiting times due to long waiting queues in untapped markets, as is the case in some emerging communities. Penetration of wireless access networks is predicted to reach 50% of all installed access lines by 2005. A high pricing has largely hindered wireless access networks' penetration. To date most operators use revenue from long-distance telephony traffic to subsidize local calls, especially, in rural areas with low teledensity. This makes, in most cases, short-distance wired telecom services cheaper than wireless services—an indispensable network penetration driver.

However, the evolution of cheaper Internet telephony, for instance, will force operators to rebalance their tariffs to reflect the delivery cost of wired telecommunications services to the home. This, perhaps, will make local calls more expensive promoting churn of customers to a wireless access technology, thereby enhancing wireless access penetration. This churn impacts customers in all demographic areas: urban, sub-urban and rural/isolated areas. Furthermore, as regulatory and standard bodies work towards the standardization of wireless access networks to promote compatibility and multi-sourcing, large economies of scale through worldwide network penetration would be achieved. This would lead to lower infrastructure costs, further enhancing the cost saving advantage of wireless access networks.

Cost-effective system reconfiguration. Adapting a wireless system to the service environment is easier than a wireline system. Adaptability is crucial for fast changing service environment, which requires continual network reconfiguration and terminal relocations. Moreover, it is not possible or permissible to drill through some structures (e.g., historic buildings, buildings containing asbestos) in order to lay cables needed for a wired system. Simple reconfiguration prevents loss of productivity during reconfiguration and costs that would otherwise result from re-wiring wired terminals. Furthermore, it is shown that about 70% of all wired network problems are associated with faulty wiring. System reconfiguration would be more cost-effective if portable radio nodes or base stations are employed, which would require dynamic system reconfiguration.

Key drivers of 3^{rd} generation of mobile wireless networks include:

1. increased data rates
2. increased capacity due to newly introduced RF blocks to serve bandwidth-hungry applications
3. a new range of mobile services
4. integrated voice, data and video (i.e. mobile multimedia) capability
5. potential for a global mobile standard.

A global standard potentially allows travellers, either for business or leisure, to use the same handset both at home and in most parts of the world. However, ITU's aim of achieving a single global wireless standard has proved unattainable, but has successfully achieved compatible multi-standard systems which "may" permit the most desirable *global roaming.* All these factors contribute to the dramatic uptake of mobile cellular services (Fig. 1.1) over the last decade.

1.2.4 Challenges in Mobile Wireless Networks

There are many challenges in the design and rollout of mobile wireless networks, including both technical and non-technical issues. The skyrocketing RF spectrum acquisition costs, crowding of the spectrum, and the growing concern of possible RF hazards to humans are some of the non-technical challenges. Of course, these non-technical issues often translate to technical issues. For instance, RF acquisition costs and spectrum scarcity dictate design of spectrally efficient wireless protocols, while RF-related health concerns dictate more investment in research to prove or disprove the concerns. The good news is the creation of jobs.

Technical issues include *multiaccess interference* arising from uncoordinated multi-user transmissions, *power limitation* and *fading* (both slow and fast fading), and background noise. Multiple access is required for simultaneous access of a base station by multiple mobiles. However, the multi-user environment raises the issue of co-channel interference (CCI). In TDMA multiacces systems, there are both intercell CCI from overlapping timeslots and intracell CCI due to channel dispersions. CCI interference is mitigated in TDMA systems by introducing *guard times,* and by conservative frequency reuse patterns [108]. These strategies, however, degrades spectral efficiency.

In synchronous CDMA systems, orthogonal signals lose their orthogonality after traversing the radio channel due to dispersion. CCI in asynchronous CDMA systems are unpreventable even on distortionless wireless channels [108]. Interference in CDMA systems are counteracted using tight power control schemes to achieve equal received power to prevent the so-called near/far problem, where interference from a powerful transmitter overwhelms weaker signals. On low signal-to-noise ratio (SNR) channels, spectrally inefficient error-control schemes (see Sect. 2.2) are also employed to combat channel errors. Second generation cellular networks characterise multiaccess interference (MAI) as structureless white background noise with flat power spectrum. However, future networks will exploit the structure in MAI in multi-user detectors (MUD) to boost spectral efficiency, receiver sensitivity and system capacity [108].

Power limitation of mobile handsets due to transmitted power and power dissipated in electronic circuitry is a major concern in the handset industry. Circuit-dissipated power limits complexity of wireless schemes as more complexity translates to more power dissipation in electronic devices and circuitry. This issue is being tackled by the increasing usage of CMOS technology.

Fading comprises both short-scale fast fading and large-scale slow fading (See also Sect. 2.1.2.). Fading renders the wireless channel spatio-temporarily varying. Fading on wireless channels are usually mitigated using *diversity, which has three variants:* time, frequency and space diversity. Time diversity can be achieved by using interleavers and error-control coding (see Section 2.2), in which the interleaver debursts the bursty channel for better operation of the error code. In the frequency domain, spread spectrum (SS) techniques are used to spread the information-bearing signals over much wider RF spectrum. Combining SS and RAKE receiver can mitigate frequency-selective multipath fading. In the space domain, however, fading is combated by separating the transmissions antennas in a way that allows multiple copies of the same signal to propagate over different independent fading paths. Such signals are then detected and combined to decode the original message.

1.3 The Internet and Internet Protocol

1.3.1 The Internet

The Internet was invented by the US military-industrial unit called the ARPA (Advanced Research Projects Administration) in the late 1960's to allow electronic sharing of research findings among researchers engaged in military-funded projects [60]. It was then called the *ARPAnet.* Today's Internet is a network of networks interconnecting LANs, MANs and WANs. Isolated computers, used mainly in homes, normally gain access to the Internet via an Internet Service Provider (ISP), or via a dial-up MODEM through a corporate Intranet or Extranet. The Internet had as few as only 213 hosts in 1979 [34], and it became available commercially in 1988 when it was freed of "techno-political" restrictions.

Before 1988 its usage was restricted to only a few research and governmental institutions. The invention of the Web (also known as the WWW) then followed in 1991. In 1992, the US NSF (National Science Foundation) sponsored installation of access points to the T3 links of the Internet [112]. The first graphic-enabled web browser, Mosaic, was introduced in 1993 by Netscape Communications (then called Mosaic Communications). Later on Netscape Navigator, also designed by Netscape Communications, replaced Mosaic. Microsoft later introduced the Internet Explorer, which is competing fiercely with Netscape Navigator for market share. The graphic browsers made the Web attractive to use by many. This, coupled with fallen prices of computers, boosted the Internet and Web uptake. The number of hosts attached to the Internet then grew exponentially from 213 in 1979 to 60 million in 2000, and by January 2001 had reached its forecasted *practical* limit of 100 million hosts [34]. The evolution of the number of internet hosts over the last decade is depicted in Fig. 1.5. The cornerstone of the Internet is the Internet Protocol suite, especially IP, which is discussed next.

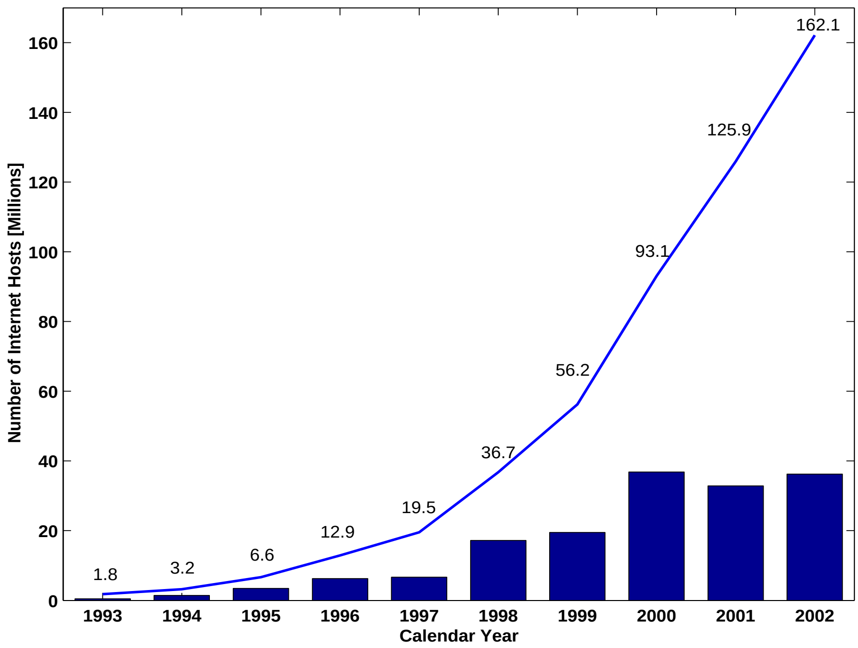


Figure 1.5: Number of Internet hosts worldwide, shaded bars indicate in-year net gains. [Source: ISC (http://www.isc.org/), as at 30 Sept 2002]

1.3.2 Internet Protocol (IP)

The IP [87] was designed in 1981 for end-to-end packet routing (i.e. from packet source to destination) [16]. It is a software protocol by which data units called *packets*[2] are forwarded by routers from one point (source) to another (destination) through the Internet. The forwarding of packets is *connectionless,* i.e., packets belonging to the same source/destination pair are forwarded independently, and can traverse different routes. The connectionless trait of IP makes the Internet *robust,* as one link cannot be a point of failure. The network does not guarantee the time taken for a forwarded packet to reach its intended destination, and whether or not it really gets there at all. This art of unguaranteed packet forwarding service is widely referred to as *best effort.* The best-effort IP forwarding, although robust and efficient (due to statistical multiplexing), raises the issue of unavailability of service guarantees (QoS), especially now that the

[2]Before the invention of IPv6, an IP packet was referred to as a *datagram.*

Internet handles traffic of various applications, some of which are strictly QoS-sensitive. Initiatives are underway to introduce better QoS-enabled forwarding mechanisms into the Internet (see Section 2.5).

The current IP is IP version 4 (IPv4) with a 32-bit address space, which is theoretically equivalent to circa 4.3 billion IP addresses. However, due to inefficient (and perhaps unfair) allocation, only about 10 million addresses are available today. This practical address limit was supposedly reached in Jan. 2001 (Fig. 1.5). The IP address exhaustion (*scalability*) issue, coupled with the lack of security and autoconfiguration, inefficient use of addresses, and the lack of service quality in IPv4, spawned the introduction of the next-generation IP (IPng) or IP version 6 (IPv6).

IPv6 has a 128-bit address space and a variable packet size in the range (576, 65575) bytes, including a 40-byte header. An IPv6 node must be able to receive a packet of size 1500 bytes, but it must not send fragments which when assembled exceeds 1500 bytes, unless it knows that the receiving node is capable of processing packets of such lengths. The minimum IPv6 header has the size of 40 bytes as optional extension headers may be in use. The IPv6 header includes a 4-bit priority field and a 24-bit flow label field. The priority bits can be used by a source to indicate the relative delivery priorities of packets it injects into the network, while the source can use the flow label fields to label packets for differentiated QoS. Also, mechanisms such as packet encapsulation increase the packet size. Unlike IPv4, IPv6 restricts packet fragmentation to only packet sources and routers are not allowed to do so.

Most of the IPv4 drawbacks are solved and the address exhaustion issue is also being tackled aggressively recently. Mechanisms used to mitigate the address exhaustion of IPv4 and extend its lifetime include [112]:

- reclaiming of allocated but unassigned IP addresses;
- usage of network address translation (NAT);
- deployment of classless interdomain routing (CIDR) [92], and
- introduction of the real specific IP (RSIP).

For example, Stanford University relinquished its Class-A network IP addresses in 2000. NAT assigns IP addresses dynamically to hosts of a network for internal usage. NAT's weaknesses include: (a) violation of IP's fundamental design (i.e. end-to-end routing), requiring an application layer gateway by some applications, (b) failure of some security protocols to operate with modified TCP socket, and (c) possible problems with address translation servers between the packet source and destination hosts. These NAT-related issues promoted the design of RSIP by 3Com in 1998 [112]. CIDR [92] unfortunately increases the size of core routers' routing table, and complicates packet routing. The IPv4 enhancement mechanisms, however, can only delay the transition to IPng.

1.4 Mobile Wireless Internet

Considering the unprecedented success of both mobile wireless networks (Fig. 1.1) and the fixed Internet (Fig. 1.5), as well as the strong desire to shift the service focus from conventional voice telephony to Internet access and electronic mailing, convergence of these two industries into mobile wireless Internet (*MOWINT*) is not unexpected. Fig. 1.6 compares the mobile handset market of 1992 (23 mio. units) with the growth of Internet hosts (450 000), and other consumer electronics markets—black and white (B&W) TV (66 000), color TV (120 000) and VCR (240 000)—after six years of commercial introduction. It can be observed from the figure that Internet and mobile markets really have uncomparable uptake. The rollout of MOWINT systems will be a driving force of the transition from IPv4 to IPng, as besides conventional routers and hosts, radio nodes (mobiles and base stations) will require unique IP addresses. However, delayed migration to IPng may hinder the uptake of MOWINT services.

The fundamental issues in mobile wireless Internet include the issues of both converging technologies. Example of the key issues are:

1. Provision of QoS in the randomly and spatio-temporarily varying wireless channel quality, i.e. QoS in a hostile environment.

2. Micromobility with Mobile IP due to long MIP handoff. Chapter 3 proposes a possible solution.

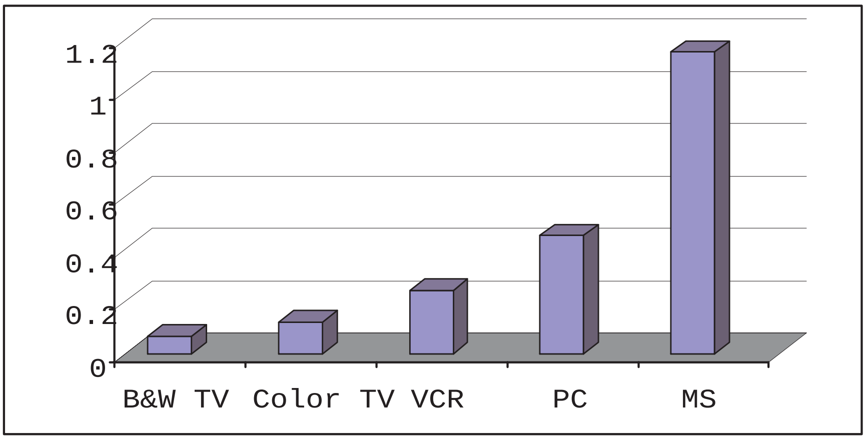


Figure 1.6: Comparing the mobile handset (MS) market of 1992 (scaled down by factor 20) with the market of other technologies after 6 years of commercial introduction. The ordinate is measured in millions of devices. Part of figure obtained from [9].

3. Design and modelling of DQoS enabled schemes to support traffic of heterogeneous applications.

4. Handset limitations such as limited battery power, small display, memory size and processing power. These limit the complexity of protocols and applications for mobile wireless Internet.

5. Dynamically changing user/network interface (UNI) due to mobility. Thus resources needed to meet an application's QoS requirements in a cell may not be available in the future cells the mobile handset migrates to. A proposed protocol to tackle this issue is the MRSVP.

6. Limited and/or high cost of RF spectrum acquisition.

1.5 Terminology

This thesis adopts the following terminology in addition to the terminologies established in the communications discipline. A converged fixed Internet and mobile wireless network comprises an IP based radio access network (IP-RAN) and an IP based core (backbone) network. In such a converged network, a conventional base station (BS) and a mobile station (MS) become Internet Protocol (IP) capable. These network nodes are thus referred to in the thesis as IBS and IMS, respectively. Since, in essence, they function as wireless routers (WR), we also refer to an IBS as a fixed WR (FWR), and to an IMS as a mobile WR, or MWR, or mobile host (MH). The routing functionality of the mobile handset is more critical in multihop, infrastructure-less (i.e. ad hoc) wireless networking with dynamically changing topology. The term *radio node* is used to refer to both MWR and FWR. In some systems, notably the IEEE 802.11 standard, BS is referred to as an access point/port (AP) or point coordinator (PC), while an MS is called a station (STA).

Mobile networks using IP philosophy, and hence IBSs and IMSs, are referred to as IP based mobile wireless networks (MOWINT), wireless Internet, mobile computing, wireless IP network, or packetized wireless networks. Networks which offer differentiated services to applications traffic for one reason or another are referred to as QoS enabled networks. Hence, the thesis title, "QoS Enabled IP Based Wireless Networks," is about terrestrial wireless networks using the packet based IP philosophy, and with network elements which offer differentiated treatment to the traffic they handle.

Network traffic is classified into *generic streams* (GeSs). A GeS can be a stream of packets belonging to a single application, or in the Internet parlance, a single *flow,* as considered in the IETF integrated services standards [13, 14]. A GeS can also refer to packets of aggregated flow(s) (or traffic class), as viewed in the IETF differentiated services (DiffServ) standards [10, 50, 53]. According to the IPv6 standard [22], a flow is a sequence of packets transferred between a particular source and a particular unicast or multicast destination node(s) for which the source desires a special handling by routers along the flow's path. A flow is identified by source and destination nodes' IP addresses, flow label, and a priority/QoS class. A GeS can also map onto users instead of being traffic centric. Other terminologies used are defined accordingly in the body of the thesis.

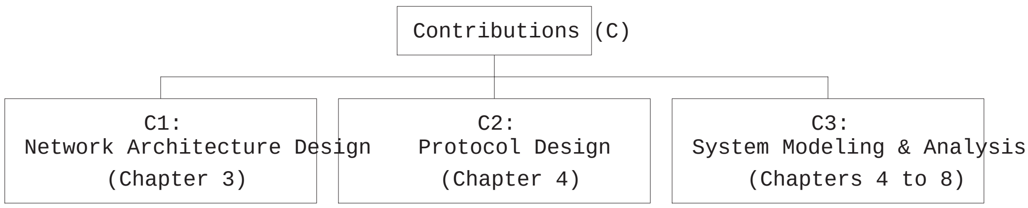


Figure 1.7: Original contributions of the thesis and corresponding chapters.

1.6 Original Contributions of this Thesis

The contributions of the thesis can be put into three categories, as illustrated in Fig. 1.7. Below, each of the contributions is discussed.

1.6.1 Contribution 1: QoS Enabled Wireless Internet Architecture

Integrated networks supporting voice and data traffic on the same carrier is known to show the following drawbacks due to the fact that these services are fundamentally different and thus require different network services:

- Voice service is highly delay sensitive and offer equal service to all users, irrespective of their location in the cell, resulting in power control schemes (especially in CDMA systems), where advantaged users receive less power than disadvantaged users.

- Integrated networks need to make compromises in their design (e.g. packet sizes, delay bounds and signalling) in order to accommodate integrated services. For instance data service requires long packet/frame sizes for efficient transmission, while voice services require short packets for low transmission delay. The same argument behind ATM's fixed size cells of 53 bytes still holds today.

- There is the tricky task of balancing system load, i.e. how to share transmission capacity and transmission order (priority) among data and voice traffic.
- Even high-quality voice services can be achieved with a modest data rate, making high-rate services almost irrelevant.

Due to these issues a network architecture delineating voice and data services is proposed in [34]. This architecture has an added advantage of leveraging existing networking infrastructure to reduce network infrastructure costs, which potentially results in cheaper service offering to customers. In Chapter 3, I propose a QoS-enabled wireless Internet access architecture, which leverages the micromobility in wireless standards to reduce mobile IP weaknesses, such as long handoff delay, to achieve effective interworking between mobile wireless networks and the global, fixed Internet. Although the idea here is applicable to any wireless standard, the design example in this thesis is based on the IEEE 802.11 wireless standard, which is commonly referred to as *dot11*. The dot11 standard, however, specifies only the first two ISO/OSI layers. I extend these layers with layer 3 (Mobile IP) to achieve a wireless Internet access architecture. QoS is introduced into IEEE 802.11 network via the Integrated Services (IntServ) architecture. Layer 2 services of dot11 are mapped onto IntServ QoS classes. A possible mapping of 802.11 services into Differentiated Services model is also given. IEEE 802.11 is the standard for WLANs, and it is finding applications in multihop ad hoc networking recently.

1.6.2 Contribution 2: Wireless Channel State Dependent Packet Scheduling

This part of the thesis proposes a framework for a class of packet scheduling schemes, which accounts for the (a) QoS requirements of the applications' traffic; (b) the wireless channel state (reflected in instantaneous data rate or noise level); and (c) optimises the usage of the expensive wireless resource. The operation of the QoS-enabled, channel state-dependent packet scheduler is analysed using optimisation theory. I believe that a packet scheduler for wireless Internet systems should be aware of both wireless channel condition and traffic QoS requirements.

The packet scheduling scheme is referred to as Best-Link, Lowest-Loss, Lowest-Delay First (BL^{4}DF) scheduler, since in the ideal case, a mobile with the best wireless link quality which is backlogged with the highest QoS (priority) traffic is given the highest scheduling priority. The BL^{4}DF scheduler is presented in Section 4.2, analysed mathematically in Section 4.3 and analysed experimentally in Section 4.4. Other variants of the BL^{4}DF scheme are also discussed. These include the delay-oriented BL^{2}DF and the service premium-oriented BL^{2}PF schemes discussed in Sections 4.4 and 4.5, respectively.

1.6.3 Contribution 3: Analytical Model and Performance Analysis of DQoS Schemes for Packetized Wireless Networks

This part of the thesis investigates the effects of wireless channel properties on differentiated QoS (DQoS) schemes, using two-dimensional, channel-state-dependent queuing theory [36, 37, 38, 39]. Basically, there are three complementary types of system analysis: analytic modelling, simulation modelling, and experimental modelling (i.e., measurement & testing) of real/prototype system, in increasing order of complexity and accuracy, as illustrated in Fig. 1.8 [75]. This thesis, however, focuses on analytic analysis, which is enlightened with numerical examples. Analytic modelling estimates protocol/system behaviour through, possibly, a closed-form mathematical relation, which relates interesting performance (QoS) measures with system design parameters. Performance metrics of interest in this thesis are per-packet expected queuing delay, per-packet loss probability, throughput, and the average scheduling rate received by a traffic stream. Analytical model of DQoS schemes, especially models accounting for wireless channel properties, such as fading, spatio-temporarily varying link quality, user mobility and speed of motion, and low link rate, is not properly covered in the open literature, and hence the motivation for this work.

The wireless channel is discretized into discrete-time Markovian states based

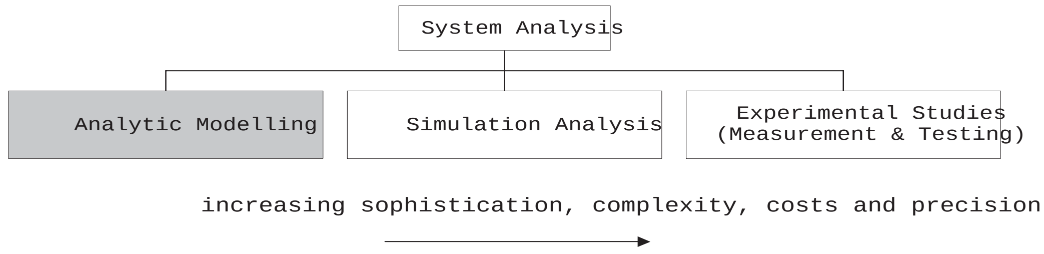


Figure 1.8: Basic types of protocol and system analysis.

on the received signal-to-noise plus interference ratio (SNIR), which also reflects on the instantaneous link quality. The link quality, in turn, influences the QoS experienced by the transported applications sitting on top of the ISO/OSI protocol hierarchy. The parameters the Markovian states are evaluated using realistic physical channel noise models and transceiver properties, such as modem. Source traffic models are used in the analysis. Three different wireless channel cases are considered: fat, slow fading case; flat, fast fading case; and flat slow and fast fading case. These analytical DQoS models are presented in the latter part of the thesis, starting from Chapter 5 through Chapter 8.

1.7 Outline of the Thesis

The rest of the thesis is structured as follows. It begins with a detailed background material on mobile wireless Internet in Chapter 2. Covered in this chapter includes wireless channel models, error control schemes, multiple access and mobility issues, potential QoS models for wireless Internet, wireless internet accounting models, traffic models, packet schedulers and buffer management schemes. Following this background, we begin the main thesis material with a high level IP based wireless network design in Chapter 3. This architecture is supposed to offer a cheap wireless access to the global Internet, and at the same time support QoS in a mobile environment.

The mathematical sections begin in Chapter 4, where some differentiated QoS schemes are proposed for wireless Internet. Section 4.3 analyses one of the DQoS schemes proposed in the previous sections of the chapter, namely the BL^{4}DF scheduling. The subsequent chapters of the thesis, starting from Chapter 5 through Chapter 8, uses queuing theory and matrix analytic methods to stochastic modelling to analyse the effects of wireless channel shortcomings, such as limited bandwidth (data rate) and spatio-temporarily varying quality, on DQoS schemes. Finally, the thesis concludes with future directions in Chapter 9.

Chapter 2

Background and Related Work

This chapter presents the background material needed for the subsequent chapters of the thesis. Review of wireless radio channel models is the topic of Sect. 2.1. Sect. 2.2 covers the background material on error control coding for wireless channels. Multiple access schemes are reviewed in Sect. 2.3. Sections 2.4 and 2.5 review briefly mobility management and Internet QoS architectures, respectively. Sect. 2.6 categorizes applications' traffic into classes based on their QoS requirements in accordance with third generation (3G) mobile standards.

Internet charging models which may be applicable to MOWINT services are discussed in Sect. 2.7. Sect. 5.5.1 reviews traffic source models. Three of the most fundamental QoS mechanisms—traffic admission control, packet scheduling and buffer management—are reviewed in Sections 2.9, 2.10 and 2.11, respectively. A sample of related literature to the thesis is reviewed in Sect. 2.12. Finally, the chapter concludes with a summary in Sect. 2.13.

2.1 Wireless Channels

This section briefly classifies wireless radio channels, discusses wireless channel noise models, and finite-state, discrete-time models used for mathematical analysis of wireless communications systems.

2.1.1 Taxonomy of Wireless Channels

Wireless radio channels can be grouped into *memoryless channels, burst-error channels* of which some have memory, or combinations thereof to yield *hybrid channels.* In contrast to bursty channels, impairments (e.g. noise) on memoryless channels affect the transmitted symbols independently, causing random errors. Error control codes designed to mitigate random errors are called random-error-correcting codes [64]. Multipath signal propagation can cause deep fades in the amplitude of radio signals which result in burst errors. Burst-error-correcting codes are used to mitigate the effects of bursty channels on information-bearing signals. Burst- and random-error-correcting codes are applied to hybrid or compound [64] channels that cause both types of basic channel errors: random and burst errors. As error control schemes work best on random-error channels, usually, the information symbols are passed through an interleaver at the transmitter to interleave the symbols so as to mitigate burst noise effects. Hence, interleavers proactively de-burst the wireless channel and hence mitigate possible burst errors beforehand. The above channel classification is also applicable to wired channels.

2.1.2 Wireless Channel Noise Models

The common impairments on wireless communications channels are *thermal noise* in transceiver electronics circuits, which is usually modelled as an additive *white* Guassian noise (AWGN), and *fading.* AWGN is referred to as *white* as its spectral density is broad and uniform over a wide frequency range analogous to *white light.* The AWGN is usually assumed to be independent of the fading, and it is modelled by its two-sided (one-sided) power spectral density $N_0/2$ (N_0) in units of Watts per Hertz (W/Hz).

Fading can spread the frequency range of information-bearing symbols (signals) and cause them to overlap in time, a phenomenon referred to as *inter-symbol interference* (ISI). ISI can be counteracted by using Nyquist pulses, such as root-raised and raised cosine pulse shaping filters [35]. Fading can be said to be *frequency-selective* or *non-frequency-selective* (i.e. flat fading). In contrast to frequency selective fading, flat fading impairs all the spectral components of a transmitted signal similarly. Flat fading is common in narrowband communications systems where the transmitted signal bandwidth ($\mathcal{B}_s$) is much smaller than the channel's *coherence bandwidth*[1] ($\mathcal{B}_c$). Frequency-selective fading impairs the transmitted signal's spectral components with different gains and phase shifts. Wideband systems with transmitted signal's bandwidth larger than the channel's coherence bandwidth suffer from frequency-selective fading. Fading can be *slow fading* or *fast fading.* On slow fading channels the symbol transmission time is smaller than the *channel's coherence time,* i.e., the period of time over which the fading process is correlated. Hence, more symbols are degraded similarly, causing *burst errors.* Fast fading radio channels, on the other hand, affect the transmitted symbols independently, and hence causing *random errors.* Both fast and slow fading can be mitigated by *diversity techniques.*

All the communications system performance analysis studied in this thesis assume flat fading. Hence, frequency-selective fading is not further explored in this monograph. As the exact characterizations of these wireless channel impairments are rather complicated, or even unknown, statistical models are often used to represent them when designing and analysing the performance of communications systems.

Flat, slow shadow fading is caused by shadows of terrain, buildings and foliage along the signal's propagation path, and it is usually modelled by lognormal distribution. Slow fading causes long-term fluctuations in signal's level. On a lognormally distributed shadowing channel, the probability density function (pdf) of the path signal-to-noise ratio (SNR), γ, can be expressed as [97, 38]

$$f_\gamma(\gamma) = \frac{10}{\log_e 10\sqrt{2\pi}\sigma\gamma} e^{-(10\log_{10}\gamma-\mu)^2/2\sigma^2}, \qquad (2.1)$$

[1]The coherence bandwidth ($\mathcal{B}_c$) indicates the frequency range over which the fading process is correlated. It is defined in [55, 76] as $\mathcal{B}_c = \frac{1}{2\pi\mathcal{S}}$ where $\mathcal{S}$ is the rms delay spread, which is defined as $\mathcal{S} = \sqrt{\frac{\int_0^\infty (\tau-D)^2 P(\tau)d\tau}{\int_0^\infty P(\tau)d\tau}}$ [82]. $P(\tau)$ is the power delay profile, and D is the average delay defined as $D = \frac{\int_0^\infty \tau P(\tau)d\tau}{\int_0^\infty P(\tau)d\tau}$. The coherence bandwidth can also be defined in terms of the maximum delay spread. The *coherence time* is the reciprocal of the coherence bandwidth.

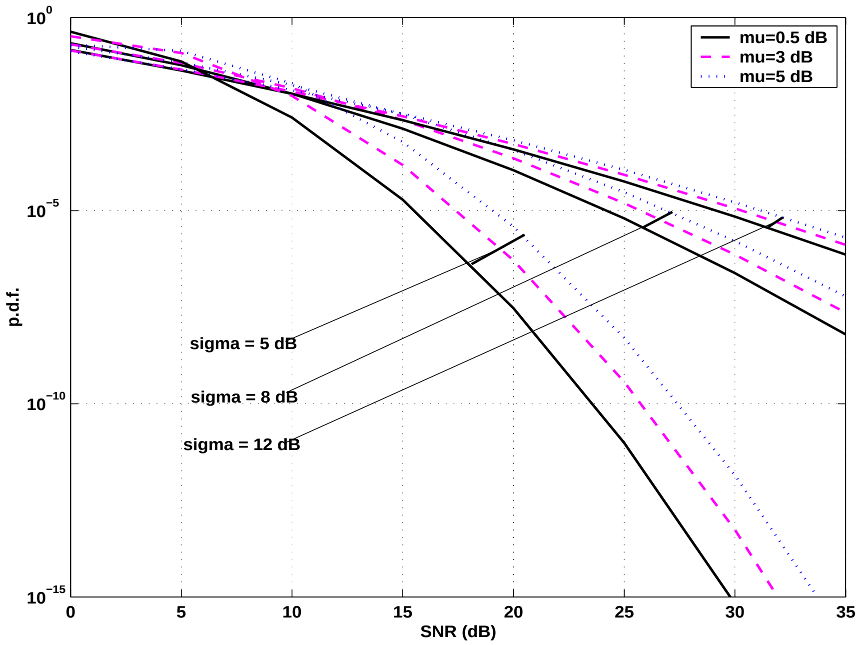


Figure 2.1: Probability density function (pdf) of lognormal slow fading vs. signal-to-noise ratio (SNR) in dB for various σ (sigma) and μ (mu).

where μ (dB) and σ (dB) are the mean and the spread of $10\log_{10}\gamma$, respectively. Fig. 2.1 illustrates the lognormal pdf for various σ (sigma) and μ (mu). Equation (2.1) can be derived from a zero-mean, unit-variance real Guassian noise process [84].

Multipath signal propagation, is caused by destructive and constructive combinations of reflected, scattered, diffracted and delayed components of a transmitted signal, and it can be flat or frequency-selective, slow or fast. Multipath fading causes short-term variations of the signal's amplitude. On a slowly varying multipath fading channel, the transmitted signal's amplitude remains constant over a symbol transmission time. Flat, multipath fading can be mitigated by efficient *micro-diversity techniques.* Common statistical models used for flat multipath fading are Rician with line-of-sight (LOS) path component, Rayleigh with non-LOS path component, and Nakagami-m models. This thesis, however, uses only the latter two models. The pdf of the SNR, γ, on a Nakagami$-m$ distributed fading channel can be written as [73, 36]

$$f_\gamma(\gamma) = \frac{m^m \gamma^{m-1}}{\rho^m \Gamma(m)} e^{-\frac{m\gamma}{\rho}}, \quad m \geq \frac{1}{2}, \ \gamma \geq 0, \tag{2.2}$$

where $\rho = E\{\gamma\}$ is the average power, $\Gamma(z) = \int_0^\infty t^{z-1} e^{-t} dt$, $\mathcal{R}e\{z\} > 0$ is the gamma function, and m is the fading severity figure. The Nakagami distribution fits land-mobile [99, 95], indoor-mobile [94, 95] and ionospheric radio [7, 95] multipath propagation scenarios. It degenerates to the one-sided Guassian distribution for $m = 1/2$ (worst-case fading), the Rayleigh distribution for $m = 1$, and the pure AWGN channel (i.e., no fading) for $m \to \infty$.

Radio channels impaired by AWGN, shadowing and multipath propagation can be modelled by a composite multipath/shadowing pdf. A common composite model is the so called Nakagami-lognormal (NLN) channel model [101], which for the SNR, γ, is given by [95, 73, 42]

$$f_\gamma(\gamma) = \frac{10 m^m \gamma^{m-1}}{\sqrt{2\pi}\sigma \log_e 10 \Gamma(m)} \int_0^\infty \frac{1}{z^{m+1}} \exp\left[-\frac{m\gamma}{z} - \frac{(10\log_{10} z - \mu)^2}{2\sigma^2} \right] dz \tag{2.3}$$

2.1.3 Finite-State, Discrete-Time Channel Models

This section discusses *finite-state* wireless channel models used in this thesis. Here, finite-state refers to a visualization of the wireless channel (including transceiver elements in the end-to-end signal transmission path) as being in one of several identifiable "states" or "conditions" at any time instant. The state transitions are governed by a statistical rule, which is part of the model [57]. Finite-state channel models are probabilistic, rather than waveform-level models. Finite state models are used for both bursty channels with correlated error patterns, as well as memoryless (i.e., random-error) channels with uncorrelated errors. Finite-state, memoryless channels have only one state. Finite-state, burst-error channels, however, are modelled by a discrete-time, finite-state Markov (FSM) process in which a state model is used to characterize the various states of the channel and the state transitions are captured by a set of transition rates or probabilities.

Binary Symmetric Channel

A common example of finite-state, discrete-time and memoryless wireless channels is the *binary symmetric channel* (BSC). A BSC is illustrated in Fig. 2.2. On a BSC a transmitted binary digit $d \in \{0,\ 1\}$ is received erroneously with a probability P_{e}, and correctly with a probability $1 - P_{\mathrm{e}}$. The error probability, P_{e}, is also referred to as *crossover probability.*

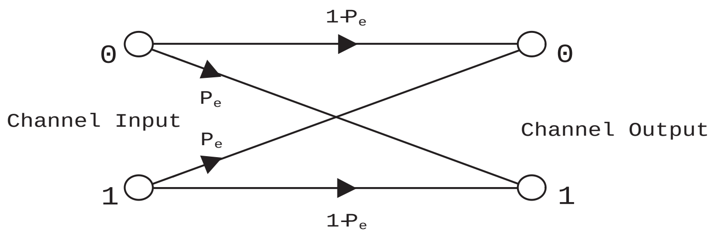


Figure 2.2: Finite-state memoryless wireless channel: the single-state BSC.

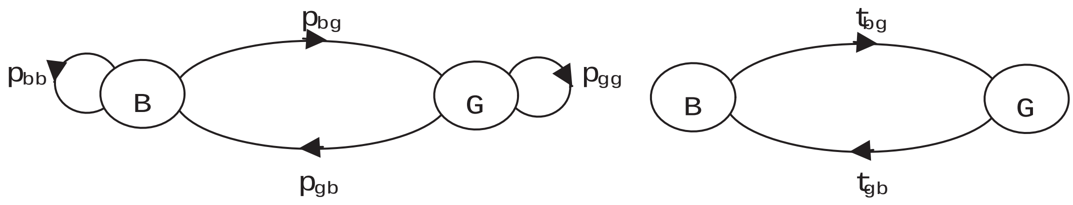
The discrete-time, finite-state Markov model (DT-FSMM) is a popular model for burst-error wireless channels for the following reasons [57]:

- it is analytically tractable,
- it has well established theory, and
- efficient techniques to estimate the model parameters of the Markovian sequences from measured or simulated error patterns are available.

The following subsections review some commonest finite-state models for burst-error wireless channels.

Two-State Markov-Modulated Channel Model

One of the commonest DT-FSM models is the two-state model, which is widely known as the *Gilbert-Elliot channel* (GEC) [30, 24]. The GEC model is illustrated in Fig. 2.3. Fig. 2.3(a) shows the state transition *probability* diagram with the transition probabilities from state G (good state) to state B (i.e., bad state) given by p_{gb}, and from state B to state G given by p_{bg}. Fig. 2.3(b),



(a) state transition probability diagram (b) state transition rate diagram

Figure 2.3: Transition diagram of the two-state DT-FSMM of a bursty wireless channel.

on the other hand, shows the state transition *rate* diagram with the transition rates[2] from state G to state B given by t_{gb}, and from state B to state G given by t_{bg}. The probabilities of the channel being in states B and G at any time are denoted as p_{b} and p_{g}, respectively. It is assumed that in the good state the received signal level is strong enough that the error (crossover) probability, $P_{\mathrm{g,e}}$, is almost zero. In the bad state, on the other hand, the channel quality is so bad that the error rate, $P_{\mathrm{b,e}}$, is much larger than zero. In the extreme case (which is the original definition in [30, 24]), it is assumed that $P_{\mathrm{g,e}} = 0$ (i.e., ideal, noiseless channel) and $P_{\mathrm{b,e}} = 0.5$ (i.e., worst-case error pattern on totally noisy channel).

The state dwell time and the state transition rates depend on temporal correlation of the fading process. Each particular state of the GEC corresponds to a unique BSC. The state transitions require a reference time, and are in many cases measured in increments of the transmission system's symbol or bit durations. A DT-FSM channel model can be represented by its state *transition rate matrix,* $\mathbf{T} = (t_{ij})$, or its state *transition probability matrix,* $\mathbf{P} = (p_{ij})$. $\mathbf{T}$ is also referred to as *infinitesimal generator matrix* [63]. The columns of $\mathbf{T}$ sum to a zero column vector (i.e., has a zero eigenvalue), while the elements of each row of $\mathbf{P}$ sum to unity. These matrices are related by

$$\mathbf{P} = \mathbf{I} + t_0^{-1}\mathbf{T} \tag{2.4}$$

[2]In this thesis, the state transition rate from state n_1 to state n_2 is denoted as t_{n_1,n_2}.

where $t_0 = \max_{\forall i}\{|r_{ii}|\}$, $\mathbf{I}$ is an identity matrix of appropriate dimension,

$$\mathbf{T} = (t_{i,j}) = \begin{pmatrix} -t_{\text{bg}} & t_{\text{bg}} \\ t_{\text{gb}} & -t_{\text{gb}} \end{pmatrix}$$

and

$$\mathbf{P} = (p_{i,j}) = \begin{pmatrix} p_{\text{bb}} & p_{\text{bg}} \\ p_{\text{gb}} & p_{\text{gg}} \end{pmatrix}.$$

Noting that $\sum_{\forall k} p_{i,k} = 1,\ \forall i$ yields $p_{\text{bg}} = 1 - p_{\text{bb}}$ and $p_{\text{gb}} = 1 - p_{\text{gg}}$. The relation (2.4) is referred to as *randomization* [56], Jensen's method, or *uniformization* [63]. Since (2.4) is used repeatedly in the thesis, we present its proof next as a theorem.

Theorem 2.1 If $\mathbf{T} = (t_{k,l})$ and $\mathbf{P} = (p_{k,l})$ are the infinitesimal generator and the state transition probability matrices of a stochastic process, and $\mathbf{T}$ fulfils the system of *global balance equations*

$$\mathbf{p}_\infty \mathbf{T} = \mathbf{0} \quad \text{and} \quad \mathbf{p}_\infty \cdot \mathbf{u} = 1, \tag{2.5}$$

then the following also holds true

$$\mathbf{p}_\infty \mathbf{P} = \mathbf{p}_\infty \quad \text{and} \quad \mathbf{p}_\infty \cdot \mathbf{u} = 1, \tag{2.6}$$

where $\mathbf{u}$ is a vector of 1's with appropriate dimension, and $\mathbf{p}_\infty$ is the vector of steady-state probabilities.

Theorem 2.1 forms the basis of transient queuing analysis, which is reviewed in Section 2.1.4.

Fritchman Finite-State Channel Model

The Fritchman model [28, 57] is suitable for modelling bursty mobile radio channels, and it is relatively easy to estimate the model parameters [57]. For a binary wireless channel, the Fritchman model divides the discrete channel states into n 'good' states and $N - n$ 'bad' states. The $N - n$ 'bad' states have different levels of degradations ('badness'). Like the GEC model, the good states are error-free, while the bad states have a finite probability of transmission errors. There are no transitions between the set of good states, just as there are

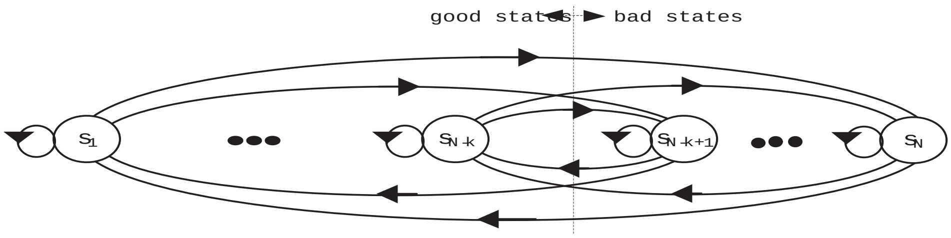


Figure 2.4: Fritchman model with k bad states and $N-k$ good states.

none between the set of bad states (Fig. 2.4). This is so since such transitions do not change the error patterns observable at the channel output. Figure 2.4 illustrates a simplified $N-$state Fritchman model having two bad states, s_{N-1} and s_N. However, commonly used Fritchman model has only one bad state.

The state transition probability matrix of the Fritchman model is often written as

$$\mathbf{P} = (p_{i,j}) = \begin{pmatrix} \mathbf{P}_{\rm gg} & \mathbf{P}_{\rm gb} \\ \mathbf{P}_{\rm bg} & \mathbf{P}_{\rm bb} \end{pmatrix},$$

where the submatrices contain the transition probabilities within and between respective states. The Fritchman model is not unique in situations where more than one bad channel state is needed, and it may not adequately model very complicated burst error patterns requiring more than one error state in the model [57].

If $(0^k|1)$ $((1^k|0))$ denotes the event of k or more consecutive error-free (erroneous) transmissions prior to the occurrence of an error (error-free) event, then their corresponding probabilities are given by [28, 57]

$$Pr\{0^k|1\} = \sum_{j=1}^{n} g_j \chi_j^{k-1} \quad \text{and} \quad Pr\{1^k|0\} = \sum_{j=n+1}^{N} g_j \chi_j^{k-1},$$

where $(\chi_1, \chi_2, \cdots, \chi_n)$ and $(\chi_{n+1}, \chi_{n+2}, \cdots, \chi_N)$ are the eigenvalues of $\mathbf{P}_{\rm gg}$ and $\mathbf{P}_{\rm bb}$, respectively, and the g_j's are functions of the p_{ij}'s.

N-State Channel Model

It is observed that the two-state GEC model of a burst-error channel may not be adequate when the wireless channel's spatio-temporal variations are dramatic

[110]. This motivated multi-state channel models, which have more "bad" and more "good" states, each of which represents a unique channel quality or a BSC. The channel quality is often characterized by the level of the received signal-to-noise-plus-interference-ratio (SNIR), as illustrated in Fig. 2.5. Consider a case where the received SNIR is partitioned into N quality intervals with thresholds, $\gamma_1, \ \gamma_2, \ \cdots, \ \gamma_{N+1}$, and map the channel to state s_n if the received SNIR falls in the range $(\gamma_n, \gamma_{n+1}]$. The resulting $N-$state FSMM model has only nearest-neighbour state transitions.

Denote the finite set of channel states by $\mathcal{S}= \{s_1, \ s_2, \ \cdots, \ s_N\}$, and define a stationary first-order Markovian stochastic sequence $\{S_t\}_{t=1,2, \ ...}$, where t is a time variable. It is often assumed that the Markov process operates with a transition rate in the order of (or equal to) the signalling rate (symbol or bit rate) of the communications system under study. Let p_n be the probability of the channel being in state $s_n \in\mathcal{S}$ at an arbitrary time t, and let p_{ik} be the transition probability from state s_i to state s_k, $s_i, s_k \in\mathcal{S}$. Then we have

$$p_n = Pr\{S_t = s_n\}, \quad \forall t \text{ and } s_n \in \mathcal{S}, \tag{2.7}$$

$$p_{ik} = Pr\{S_{t+1} = s_k | S_t = s_i\}, \quad \forall t \text{ and } s_i, s_k \in \mathcal{S}, \tag{2.8}$$

and the birth-and-death condition

$$p_{ik} = 0, \quad \forall \ |i-k| > 1 \text{ and } s_i, s_k \in \mathcal{S}. \tag{2.9}$$

Some literature refer to this FSMM as a *hidden Markov model* (HMM), as only the input and the output of the channel but not its internal sequence of states are observable externally. The state transition probabilities for a slowly fading radio channel[3] can be estimated as [110]

$$p_{ij} = \begin{cases} \frac{L_j}{p_i \times R_S}, & j = i+1, \ i = 1, 2, \ \cdots, \ N-1 \\ \frac{L_i}{p_i \times R_S}, & j = i-1, \ i = 2, 3, \ \cdots, \ N \\ 1 - p_{i,i+1} - p_{i,i-1}, & j = i, \ i = 2, 3, \ \cdots, \ N-1 \end{cases} \tag{2.10}$$

[3]In the case of fast fading radio channel we may have $L_j > p_i \times R_s$. In which case, the reciprocals of the expressions in (2.10) apply so that $0 \leq p_{ij} \leq 1$ is fulfilled. This is the rationale behind Section 5.3.2.

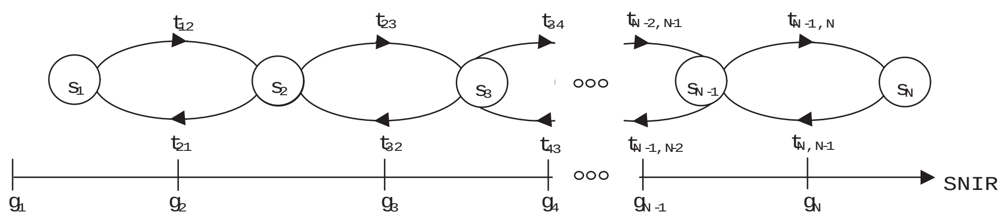


Figure 2.5: State transition rate diagram of the N-state DT-FSMM mapping discrete states onto received SNIR intervals.

and $p_{11} = 1 - p_{12}$, $p_{NN} = 1 - p_{N,N-1}$, where R_S is the symbol rate, and L_n is the average number of times per second the received SNIR (γ) crosses downward the threshold γ_n. L_n is also referred to as the *level crossing rate.* In the subsequent chapters, L_n and p_n are evaluated using realistic wireless channel noise models and transceiver properties.

2.1.4 Transient Analysis

For transient, continuous-time analysis, the time-dependent state occupancy probabilities $p_n(\tau)$, $n = 1, 2, \cdots, N$ are used. These parameters relate the generator matrix $\mathbf{T}$ according to the system of linear differential equations $\frac{d}{dt}\mathbf{p}(\tau) = \mathbf{p}(\tau)\mathbf{T}$, where $\mathbf{p}(\tau) = [p_1(\tau),\ p_2(\tau),\ \cdots,\ p_N(\tau)] \in \Re_+^N$ and N is the number of states of the Markov chain. The general solution to this equation is $\mathbf{p}(\tau) = \mathbf{p}_0 \exp(\mathbf{T}\tau)$ where $\mathbf{p}_0$ is the vector of initial state occupancy probabilities. Reference [72] presents several ways of evaluating such matrix exponentials.

2.2 Error Control for Wireless Channels

As the wireless channel medium used for information transfer is less than ideal, error control coding via redundancy is needed for reliable data transfer. The process of adding redundant bits to message bits at the transmitter to offset possible effects of channel noise is referred to as *channel coding.* There are two basic types of channel coding for communications channels: block codes and

Figure 2.6: Block diagram of a linear code

convolutional codes [64]. A code can also be linear or non-linear. Basically, an efficient encoder maps a k-bit message vector $\mathbf{m}$ in its input onto a unique ν-bit code vector $\mathbf{w}$ in its output, as shown in Figure 2.6.

A coding scheme can have *memory* or be *memoryless*. Rate $r = k/\nu$ block codes, denoted as (ν, k), are memoryless since their instantaneous output codeword $\mathbf{w}$ depends only on the corresponding input message word $\mathbf{m}$. Rate $r = k/\nu$ convolutional codes, denoted as (ν, k, d) code, however, contain memory which is reflected in the memory order (or delay) d. Redundancy is introduced via d which indicates the number of previous message blocks/vectors that are used in computing the current codeword of length ν. The choice of d to design efficient codes that are robust to wireless channel imperfections is a key issue in convolutional encoding. This thesis, however, applies only memoryless codes to analyse the performance of wireless communications networks.

There are two basic ways of controlling errors on wireless channels: forward-error correction (FEC) and automatic-repeat-request (ARQ). Ideally, FEC schemes detect and correct the received vector, whilst ARQ schemes only detect errors and query the transmitter to repeat the codeword. Obviously, ARQ works only on duplex but not on simplex channels and is simpler than FEC. Table 2.1 compares the basic features of ARQ with FEC schemes. In Table 2.1, the symbols '+' and '-' indicate advantage and disadvantage, respectively. ARQ schemes can be organised into stop-and-wait ARQ (SAW-ARQ) and continuous ARQ (C-ARQ). C-ARQ, in turn, has two variants: selective-repeat ARQ (SR-ARQ) and go-back-N ARQ (GBN-ARQ). With SAW-ARQ, the transmitter generates a codeword, sends it to the receiver and wait for acknowledgement, either positive (ACK) or negative (NACK), from the receiver before sending the next codeword.

With continuous ARQ, on the other hand, codewords are generated and sent by the transmitter to the receiver without waiting for ACKs/NACKs. If a NACK is received, GBN-ARQ transmitter repeats the erroneous codeword and all subsequent $N-1$ codewords, whereas SR-ARQ encoder repeats only the erroneous codeword. Obviously, SR-ARQ appears much more efficient than GBN-ARQ, just as GBN-ARQ is more efficient than SAW-ARQ. The issue is that the more efficient the codec is, the more complex and costly (in terms of buffering and logic) it is. GBN-ARQ is interesting for long round-trip channels with high data rate (bandwidth) and it is popular in duplex systems. SAW-ARQ, on the other hand, proves more efficient when codeword transmission time compares with the link round-trip time. SAW-ARQ schemes find applications in systems with half-duplex links [64]. For NRT applications traffic without strict timing constraint, ARQ can retransmit an erroneous codeword until successfully received. The efficiency of channel encoders is measured by metrics such as *complexity*, *throughput* and *reliability*, and low complexity codes with low error rates (i.e. highly reliable) and high throughput are somewhat contradictory but crucial code design criteria.

Hybrid ARQ schemes. For highly error-prone wireless channels, hybrid ARQ schemes comprising both FEC and ARQ codes have been proposed [64]. The hybrid schemes improve the delay and throughput drawbacks in pure ARQ whilst enhancing the reliability and complexity issues in pure FEC. In this way, FEC codes are designed to combat most frequently occurring non-zero syndromes (i.e. error patterns), whilst ARQ schemes are designed to deal with less frequent syndromes. There are two types of hybrid ARQ schemes: type-I and type-II hybrid ARQ [64]. Hybrid type-I ARQ schemes, denoted in this thesis as $(\nu, k, \tau, \hat{r}_{\mathrm{arq}})$, has a FEC component that can detect and correct up to τ errors and an ARQ component with an infinite or finite maximum number of retransmissions $\hat{r}_{\mathrm{arq}}$ that can detect a given number of error patterns. A hybrid type-II ARQ scheme has a high-rate code to detect errors only and an invertible half-rate code for error correction. The parity check bits of this type of code is sent to the receiver only on demand [64]. There are various other types of error control codes. For example, *turbo codes* achieving performance close to Shannon's information theoretic capacity limit have been introduced since 1993. However, due to their complexity and high decoding delays, they are not the best codes for time-critical applications traffic. Adaptive encoders with code redundancy depending on instantaneous radio fading level have also been introduced recently.

Let P_{a} be the probability that the decoder acknowledges a received vector as a codeword. This happens when either the received vector is error-free or there is an undetectable error in it. Thus the likelihood that the receiver queries the sender to repeat the received vector, i.e. the retransmission probability, is $P_{\text{r}} = 1 - P_{\text{a}}$. Let the bit error probability of the wireless channel achievable by a given modulator/demodulator (modem) be given by P_{b}. Then for a hybrid type-I ARQ linear code, $(\nu, k, \tau, \hat{r}_{\text{arq}})$, P_{a} in (2.13), (2.14) and (2.15) on a *random-error* channel (or perfectly interleaved/deinterleaved bursty channel) is given by [36, 37, 38, 39]

$$P_{\text{a}} = \textit{no errors} \; + \; \textit{undetected errors} \; = \; (1 - P_{\text{b}})^{\nu} + \sum_{i=\tau+1}^{\nu} \binom{\nu}{i} P_{\text{b}}^{i} (1 - P_{\text{b}})^{\nu - i} \tag{2.11}$$

For such a linear code the average throughput efficiency is given by

$$\bar{S}_{\text{arq}} = k / (\nu \, \bar{r}_{\text{arq}}) \tag{2.12}$$

where $\bar{r}_{\text{arq}}$ is the mean number of retransmissions (including the original transmission) for the corresponding ARQ scheme. Assume that the number of ARQ retransmissions, including the first original transmission, is bounded to $\hat{r}_{\text{arq}}$ due to delay and throughput constraints. As N code vectors (equivalent to GBN-ARQ round-trip delay) are retransmitted for each detected erroneous vector, the mean number of retransmissions $\bar{r}_{\text{arq}}$ for the GBN-ARQ scheme is

$$\bar{r}_{\text{gb}n} = \sum_{i=1}^{\hat{r}_{\text{arq}}} [(i-1)N + 1] P_{\text{a}} (1 - P_{\text{a}})^{i-1}$$

which, for $P_{\text{a}} \neq 0$, simplifies to [37]

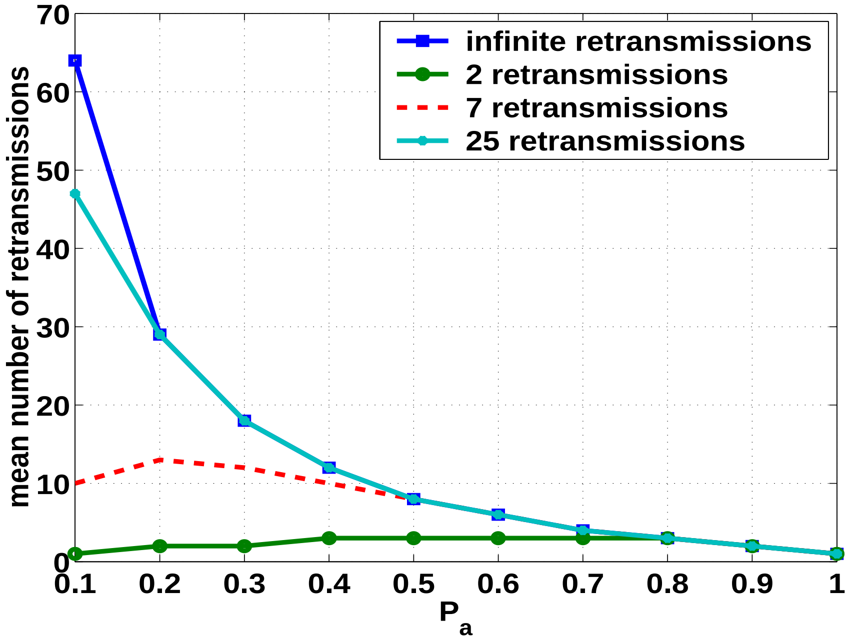


Figure 2.7: GB7-ARQ: $\bar{r}_{\text{gbn}}$ versus P_{a}

$$\bar{r}_{\text{gb}n} = \begin{cases} \lceil 1 + N/P_{\text{a}} - N \rceil, & \hat{r}_{\text{arq}} \to \infty \\ \lceil 1 - (1 + N\hat{r}_{\text{arq}})(1-P_{\text{a}})^{\hat{r}_{\text{arq}}} \; + \; N(1-P_{\text{a}}) & \\ \cdot\, (1-(1-P_{\text{a}})^{\hat{r}_{\text{arq}}})/P_{\text{a}} \rceil, & \text{for finite } \hat{r}_{\text{arq}} \end{cases} \tag{2.13}$$

where $\lceil x \rceil \geq x$ is an integer. Figure 2.7 plots $\bar{r}_{\text{gbn}}$ versus P_{a}, where it can be observed that $\bar{r}_{\text{gbn}}(\hat{r}_{\text{arq}} \to \infty)$ turns to the case for $\hat{r}_{\text{arq}} = \infty$. Also $\bar{r}_{gbn}$ converges to unity irrespective of the value of $\hat{r}_{arq}$ as P_a approaches unity.

The expected number of retransmissions for an SR-ARQ linear code, on the other hand, is

$$\bar{r}_{\text{sr}} = \sum_{i=1}^{\hat{r}_{\text{arq}}} iP_{\text{a}}(1-P_{\text{a}})^{i-1}$$

which, for $P_{\text{a}} \neq 0$, simplifies to [36]

$$\bar{r}_{\text{sr}} = \begin{cases} \lceil 1/P_{\text{a}} \rceil, & \hat{r}_{\text{arq}} \to \infty \\ \lceil [1-(1-P_{\text{a}})^{\hat{r}_{\text{arq}}}(1+\hat{r}_{\text{arq}}P_{\text{a}})]/P_{\text{a}} \rceil, & \text{for finite } \hat{r}_{\text{arq}} \end{cases} \tag{2.14}$$

Let $r_{\rm t}$ be the transmitter messaging bit rate and $t_{\rm idle}$ be the inter-code vector transmission (i.e. idle) time of the SAW-ARQ code. Then the maximum number of code digits transferable per round-trip time if the transmissions were continuous is $N_{\rm d} = \nu + r_{\rm t}t_{\rm idle}$. Hence the expected number of code digits transferable per round-trip delay if there were no idling periods is

$$\bar{D}_{\rm saw} = N_{\rm d}P_{\rm a}\sum_{i=1}^{\hat{r}_{\rm arq}} i(1-P_{\rm a})^{i-1}$$

which, for $P_{\rm a} \neq 0$, simplifies to

$$\bar{D}_{\rm saw} = \begin{cases} N_{\rm d}/P_{\rm a}, & \hat{r}_{\rm arq} \to \infty \\ (1-(1-P_{\rm a})^{\hat{r}_{\rm arq}}(1+\hat{r}_{\rm arq}P_{\rm a}))N_{\rm d}/P_{\rm a}, & \text{for finite } \hat{r}_{\rm arq} \end{cases} \tag{2.15}$$

In order to apply (2.12) to obtain SAW-ARQ throughput, the mean number of 'retransmissions' should be modified as $\bar{r}_{\rm saw} = \lceil \bar{D}_{\rm saw}/\nu \rceil$.

To recap, a GBN-ARQ needs large enough memory at the transmitter to buffer the N codewords it needs to retransmit in case of error, while SR-ARQ has to buffer the correctly received codewords subsequent to erroneous one for orderly delivery. While SR-ARQ avoids retransmission of correctly received codewords, GBN-ARQ retransmits some of them making it inefficient on high rate wireless links with large round trip times (RTTs). SAW-ARQ's inefficiency lies largely in its idle times. In all, SR-ARQ is the most efficient ARQ scheme, but achieves that at the cost of being the most complex ARQ scheme.

2.3 Multiaccess Schemes

Multiple access schemes (MAS) are used by the Media Access Control (MAC) layer to allow as many mobiles as possible to access the radio base stations. The basic MAS schemes are code-division multiple access (CDMA), time-division multiple access (TDMA), frequency division multiple access (FDMA), and its variant, orthogonal FDMA (OFDMA). Various variants, or hybrids, of these basic MASs and are also available. Since high data rates result in small TDMA bit duration (since the smaller the bit duration, the worse the signal immunity against multipath effects) and high CDMA code chip rates, 4G mobile

Table 2.1: Comparison between FEC and ARQ error control.

ARQ encoder	**FEC encoder**
(+) adaptive (no errors, no retransmission)	(-) constant coding overhead
(+) simpler (it detects errors only)	(-) complex (detects and corrects errors)
(-) channel-state-dependent throughput	(+) constant throughput equals code rate
(-) retransms increase codeword delay	(-) Code complexity can increase decoding delay
(+) highly reliable	(-) high reliability requires long code which increases code complexity

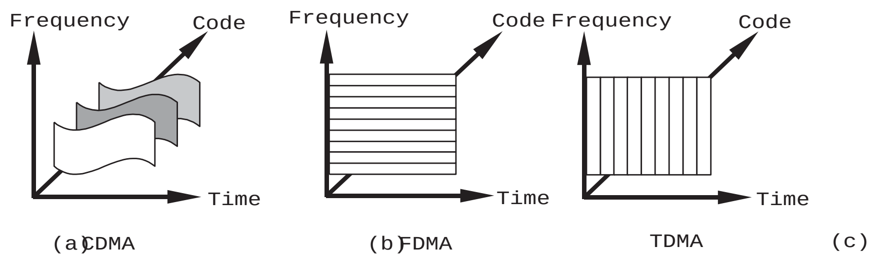


Figure 2.8: Fundamental multiple access schemes (guard times for TDMA and guard bands for FDMA suppressed).

systems are envisaged to deploy Orthogonal Frequency Division Multiplexing [34]. Figure 2.8 illustrates the fundamental multiaccess schemes used in cellular networks. In the downlink (forward link), these multiaccess schemes become multiplexing, rather than multiple access, and are thus referred to as CDM, FDM and TDM, respectively. Detailed treatments of TDM/TDMA technology can be found in [31], CDMA in [109, 76, 31], while OFDMA is treated in [107]. Besides first generation of mobile cellular communications systems, FDM is used in CATV systems and FM radio broadcasting.

2.4 Mobility Management

Mobility management (MM) [1] procedures include location management, handoff management, authentication and identification, access rights checking, and encryption. Handoff management mechanisms are triggered by either wireless link deterioration or user mobility. User mobility always results in a change of FWR, but changing only a radio bearer (intra-cell handoff) can solve a link deterioration problem. Mobility can be micromobility or macromobility.

Handoff management (HM) mechanisms aid the network to maintain connections with mobile terminals as they change their point of attachment to the network/FWR. HM consists of handoff initiation, new connection generation and data flow control [1]. Prior to handoff initiation, user movement must be detected and network conditions be monitored. New connection generation involves resource allocation and connection routing. Data flow control utilizes buffering /sequencing and multicasting [1].

2.5 Internet QoS and Signalling Architectures

To date the wireline Internet supports only a single service based on the so-called *best effort* service model, as it does not differentiate traffic or packets injected into it, nor guarantee explicit delivery of packets. However, with the advent of future Internet (including wireless Internet) poised to support multimedia traffic with different sensitivities, traffic differentiation and service guarantees (i.e., QoS) are more of utilities than luxury. Scalable schemes that can provide differentiated quality-of-service (QoS) to applications traffic is very crucial to the successful uptake of wireless Internet. Although QoS is a much talked-about topic in networking in recent years, there seems to be no consensus on its exact definition to date. However, the following definitions are currently being discussed on the mailing list of the Internet Engineering Task Force (IETF) Next Steps in QoS Signalling (NSIS) [39]. These definitions are widely compliant with QoS definitions of the International Telecommunications Union (ITU). Here, a *flow* is to be understood in the context of IETF standards, i.e., a stream of packets identified by a combination of the sender and receiver sockets. (A socket is the 2-tuple of IP address and the TCP port number.)

QoS is a set of attributes and their values that, taken together, characterize the performance experienced by a flow. Thus, quantification of a flow's QoS is non-trivial in QoS schemes based on aggregated flows, such as DiffServ [10]. End-to-end QoS is a set of attributes and their values that, taken together, characterize the performance experienced by an end-to-end flow, i.e., as observed by protocols above the IP layer. Edge-to-edge QoS is a set of attributes and their values, that, taken together, characterize the performance experienced by a flow from the point where it enters a domain (or a concatenation of domains) to the point(s) where it leaves that (those) domain(s). Agreed QoS is a set of attributes and their values that have been agreed between a network (or concatenation of networks) and two (or more) hosts. This set characterizes the performance the network(s) agree that a flow will experience for a defined period. The agreement can be reached through signalling (i.e., on-the-air or dynamically) or administratively (i.e., off-the-air or statically). Finally, actual QoS is the observed/measured values of a set of QoS attributes, such as delay, loss ratio and throughput.

Traditionally, the Internet was envisaged for fixed, immobile terminals. With the desire for portable network connectivity the Mobile IP [85, 86] concept was borne within the Internet Engineering Task Force (IETF). However, Mobile IP solves the connectivity issue concomitant with portable terminals, but creates another technically interesting issue due to its long handoff delay that can degrade applications quality if the transmission control protocol (TCP) [88] is used as a transport protocol [19]. Furthermore, the Internet traditional design did not perceive support of real-time applications. However, with the advent of multimedia communications, QoS-enabled architectures have been designed for the Internet such as the Integrated Services (IntServ) [13], the RSVP [14], the Mobile RSVP (MRSVP) [100], and the Differentiated Services (DiffServ) [10]. All these QoS models operate at the IP (i.e. network) or application layers.

QoS provisioning requires differentiating between traffic and assuring required network performance to support the individual sessions based on their characteristics. The current Internet or IP transport does not support service differentiation and neither does it offer network performance assurance to its traffic. Packet differentiation is needed since some packets may be more QoS sensitive than others. Since no resource setup prior to payload transmission is used, the IP transport does not use signalling. Packets injected into the Internet are treated the same way and are served with network resources (e.g. buffer space and bandwidth) on a first-come, first-served queuing principle. The main reason for designing the Internet as it is was to make it scalable and robust (i.e., avoid single point of failure) [111]. There are also no proactive mechanisms to control network congestion due to the lack of packet admission control functionality in the Internet protocols, allowing sources to release packets into the network at any rate. Therefore, adopting the current IP technology, as it is, in RANs for future IP based mobile networks that are to support multiservice traffic with diverse QoS demands is not feasible. Hence investigating ways of providing appropriate QoS assurance for applications running over IP-RANs is warranted.

To provide different levels of QoS with efficient usage of network resources as a constraint, an IP-RAN needs to implement radio resource management (RRM) functions including:

- packet differentiation,
- resource allocation,
- packet scheduling, and
- packet admission control.

In the following subsections, the key service models and mechanisms being discussed recently to provide QoS in IP enabled mobile wireless networks are briefly reviewed. The schemes include: integrated services QoS architecture (IntServ), differentiated services QoS architecture (DiffServ) and the fat-dumb-pipe (FDP) architecture. The FDP model overprovisions the network to avoid congestion, and hence queuing delays and packet losses. The issue with this scheme is uneconomical usage of network resources, as it is an extremely inefficient scheme. Other interesting protocols which can be used with any QoS model is MPLS (multiprotocol label switching) [54] and NSIS (next steps in signalling). MPLS is a packet forwarding mechanism which can be used with any Layer 3 protocol, while NSIS is a signalling protocol. MPLS and NSIS are thus not further discussed in this thesis.

2.5.1 IntServ QoS Architecture

Intserv (IS) [13] is the first non-best-effort Internet QoS architecture, and it deviates significantly from the best-effort forwarding (BEF) scheme. IS model defines two services, guaranteed service (GS) and controlled load (CL). However, IS's definition of a *guaranteed* service is interpretated as a *predictable* service, rather than absolute [26] service level. The main QoS parameter of interest in the IS model is the worst-case per-packet delay. IS comprises four key components:

1. an admission controller,
2. an explicit signalling scheme for resource reservation,
3. a QoS-aware routing,
4. a scheduling algorithm.

IntServ uses explicit signalling protocol, such as RSVP, to reserve resources in all routers on the path from the packet sender to its receiver through the network on a per-flow basis, which raises scalability concerns for medium size to large networks with many simultaneous flows. Another noted flaw in IntServ is the use of bandwidth wasteful regular state refresh. Recapping, the drawbacks of the IntServ QoS architecture include scalability, manageability, need of new application/network interfaces for proper functioning, and lastly, all routers in the end-to-end path of a traffic stream need to be IntServ capable for proper functioning of IntServ.

2.5.2 Resource Reservation Protocol (RSVP)

The RSVP [14] is the signalling protocol recommended for use with IntServ. As it aims to support multicasting, dynamic group membership, and diverse

Table 2.2: RSVP's key messages and meanings.

Message	Action	Orientation	Content/Remarks
Path	installs path state	downstream	previous hop, TSpecAd-spec, template
Resv	reservation message	upstream	flow descriptor reservation style
PathErr	unsuccessful Path	upstream	doesn't modify path state
ResvErr	unsuccessful Resv	upstream	doesn't modify path state
PathTear	deletes path and resv state	downstream	explicit by senders or by timeout
ResvTear	deletes resv state	upstream	generated explicitly by receivers
ResvConf	confirms reservation	downstream	generated explicitly by all transit path nodes

receiver capabilities and requirements Roughly speaking, its reservation is performed by the data receiver(s).

The RSVP operates as follows. An application running on an IS-enabled router first specifies its QoS requirements to meet its flow characteristics. This is referred to as *flow specification.* Based on the flow specification, RSVP then sets up the flow state, i.e. reserves resources for the flow in the *unidirectional* path created by a routing protocol. The reservations are timed, and are deleted upon timeout. Thus periodic state refresh is required to maintain the per-flow states. This *soft state* of RSVP is one of the headaches posed in IS.

Table 2.2 illustrates the key RSVP messages [111]. These messages are transmitted as raw IP packets with the protocol ID of 46, or encapsulated in UDP packets. The flow descriptor comprises the *flowspec* (flow specification) and the *filter spec* objects. The requested QoS and packet scheduling parameters are contained in the flowspec object.

Table 2.3: Mobile RSVP versus RSVP signalling [34].

Feature	**MRSVP**	**RSVP**
Support of active resource reservation?	yes	yes
Support of passive resource reservation?	yes	no
Support of resource reservations in RSVP-capable routers en route an IP tunnel?	yes	no
Support of advance resource reservations in potential future locations of mobile hosts?	yes	no
Resv message forwarding for different reservation styles in case of multiple-source Path messages	yes	no
Per-flow reservation	yes	yes
Explicit signalling?	yes	yes

2.5.3 Mobile RSVP

The mobile RSVP (MRSVP) [100] is a modified form of RSVP that is tailored for wireless environments. It was invented in response to the weakness in IntServ regarding resource renegotiation after every handoff (mobility) that wastes network resources (bandwidth). As with its parent protocol, RSVP, MRSVP reserves resources for applications traffic on a per-flow basis. However, unlike RSVP, MRSVP has two types of reservations: active and passive. Table 2.3 contrasts the features of MRSVP with those of its parent protocol, RSVP.

Two types of QoS guarantees to mobile applications are identifiable: mobility dependent and mobility independent [6, 41]. The latter is accomplished by making spatial resource reservations in all possible next hop cells the mobile terminal may migrate to during an active session/connection. The flaw in this technique is that at best only resources reserved in one of the anticipated cells is actually utilised while reserved resources in other cells go wasted, degrading bandwidth utilization. The MRSVP tackles this problem by allowing the passively reserved resources to be used by other applications when the intended user is not available to occupy them.

The mobility dependent QoS scheme does not make any prior spatial resource reservations and hence conserves resources. However, the service quality of applications can be degraded if the cell an active terminal roams to is congested. Hence further research for, notably, hybrid schemes to optimise above flaws and advantages is warranted. QoS can also be investigated on the bit level (e.g. bit error rate (BER) and bit rate), packet level (e.g. packet error rate, inter-packet delay variation (jitter) and data rate (bandwidth)) or connection/session level (e.g. connection setup time and connection handoff dropping).

2.5.4 DiffServ QoS Architecture

Diffserv (DS) [10] offers a relative, better than best-effort forwarding, but no fine-grained, per-flow QoS as in Intserv. It divides traffic into a few number of forwarding classes (FC) called Behavior Aggregates (BA). The FC class of a packet is encoded in the 6-bit DSCP (differentiated services codepoint) in its IP header, and the FC pre-determines the forwarding treatment the packet receives in Diffserv-enabled routers. The forwarding treatment of a FC in a router is referred to as the Per-Hop Behavior (PHB), and reflects the resources provided to the FC and drop priorities in case of congestion. The DSCP bits may be set within the protocol stack by, e.g., a QoS manager, or downstream by a router. However, endpoint applications in the user terminals which know exactly the QoS requirements of their traffic should, in principle, be enabled to handle the DSCP settings via their programming interfaces (APIs).

The DS model has two types of routers—edge (boundary) and core (interior) routers—and the edge routers perform most of the DS-related tasks. They perform packet classification (i.e. map packets to respective forwarding classes) and traffic conditioning (i.e. metering, marking, shaping, dropping, etc.) to confine the traffic into agreed-on service level specifications (SLS). Core routers forward packets based only on their DSCPs. As the PHB defines only local treatment of individual packets in a router but does not provide any end-to-end service, a Per-Domain Behavior (PDB) has been defined recently [74]. Table 2.4 contrasts the features of Intserv with Diffserv (see also [58], Table 12.1).

DiffServ overcomes the scalability flaw in IntServ by treating packets as flow

Table 2.4: Diffserv versus Intserv QoS models.

Feature	Diffserv	Intserv
Qos	per-class, relative, no per-flow protection	per-flow, absolute
Per-flow qos	prioritisation, provisioning	reservation, guaranteed
Packet classification	unified (DSCP)	multifield
Resource reservation	per-class	per-flow
State maintenance	edge routers, per class	all routers, per flow
Admission control	edge routers	all routers
Service categories	aggregated, EF & AF	individual, GS & CL
Signalling	no explicit required	any, RSVP recommended
Scalability	good	bottleneck
Scope of protocol	local packet handling, PHB	end-to-end service
Per-packet handling	yes	no
Protocol type	data plane	control plane
Adaptation of qos demand	no, due to static SLA	yes, flow oriented qos
Business model embedded into network technology	no	yes

aggregates. It defines PHB for these aggregated flows and solves the QoS issue through resource provisioning and prioritisation, rather than hard QoS guarantees used in the IntServ model. It does not therefore use an explicit signalling. Besides the best effort service model, two QoS enabled PHB forwarding classes have been standardised for DiffServ networks: expedited forwarding (EF) [53] and assured forwarding (AF) [50].

While EF packets are marked with a single DiffServ codepoint, AF has twelve DSCPs for its packets as it has four subclasses and three drop priorities per subclass. The EF service is designed to support mission-critical traffic such as voice over IP (VoIP) and network control messages requiring low delay and low jitter, whereas the AF PHB provides different levels of forwarding assurances to IP packets. Just as Intserv, Diffserv is link-layer independent. This makes it widely applicable. However, as the wireless environment is quite different than the wired, *wireless Diffserv* needs be properly studied.

2.5.5 Mobile IP

The traditional Internet Protocol (IP) does not support host mobility, since it assumes a fixed host point of attachment (POA) to a network [85]. Thus it requires a mobile terminal to change its IP address whenever it changes its network POA, or else it loses packet routability. However, a terminal loses network connectivity if it changes its IP address upon changing POA, since higher layer protocols (e.g. TCP) cannot identify the host. For example, the packet sender and receiver sockets identify a TCP connection. (A TCP socket comprises the IP address and the TCP port number on the host.) Hence modifying any of the sockets results in connection disruption and eventual loss.

In an attempt to mitigate this impediment to mobile data networking, the Internet Engineering Task Force (IETF) designed the Mobile Internet Protocol (MIP) [86]. MIP complements the traditional IP functionality to provide completely automatic reconnection of previous computing activities after changing network POA. However, the MIP mobility management mechanism is not 'seamless' and thus not transparent to transport layer protocols (here TCP). Its handoff delay can be too long to maintain TCP connections, which may result in degradation in TCP throughput performance as discussed below.

For its proper operation, Mobile IP (MIP) defines three types of network entities—a mobile host (MH) or mobile node (MN), a corresponding host (CH) or corresponding node (CN) and the mobility agent (MA) comprising the home agent (HA) and the foreign agent (FA). MIP is designed to support universal mobility across all IP networks. All functionalities required to process and manage mobility information are embedded in these three entities. The MH is the terminal roaming within and between (foreign) subnetworks, while the CH is the device trying to communicate with the MN.

The HA and FA co-operate to enable the MN to roam without changing its *home IP address.* This is achieved by assigning two IP addresses to each mobile node: a 'stable' home IP address with the HA on home network to maintain connectivity and identity, and a 'volatile' care-of address (CoA) on foreign network to maintain routability and locationing. The CoA can be of two types—a collocated or issued by a FA [85]. The home network is the subnet a user subscribes to for service. Location directories are needed to store associations between CoA and home address of MHs. One of the weaknesses of the early version of MIP is its so-called *triangular routing,* which is illustrated in Fig. 2.9. However, this issue has been identified and solved in the recent MIP versions, in which there is an option of direct MN/CN communication without the intervention of a FA. The HA puts a packet from a CH to a roaming mobile host in another IP packet and forwards it to the MN, either directly (in case of colacated CoA) or via a FA (in case of FA CoA). This IP-in-IP encapsulated packet forwarding is referred to as *packet tunneling.*

Although mobile IP aids a mobile host to maintain connectivity and packet routability, it has its own drawbacks. Notable mobile IP shortcomings include:

- long handoff latency,
- heavy signalling overhead due to e.g. periodic beacon broadcasts,
- packet losses due to connection breakage during handoff, and
- TCP trottling mechanisms due to packet losses and delay during long handoff.

The handoff latency comprises the (a) time the MN requires to discover (learn) the IP address of the new subnet; (b) time for MN to establish a new CoA with the new subnet/FA; and (c) time required to complete the binding update (BU), i.e., time to notify the CN/HA of new CoA. The wireless Internet access architecture proposed in Chapter 3 attempts to solve these mobile IP issues by exploiting wireless layer 2 (L2) handoff signalling to speed up mobile IP's L3 handoff.

2.6 QoS Based Applications Classification

According to the UMTS 3G standards, traffic can be categorized into conversational, streaming media, interactive or background, depending on its quality of service requirements [51, 34]. Traffic generated by background applications, such as file transfer and e-mail, require no QoS guarantees and thus are tolerant

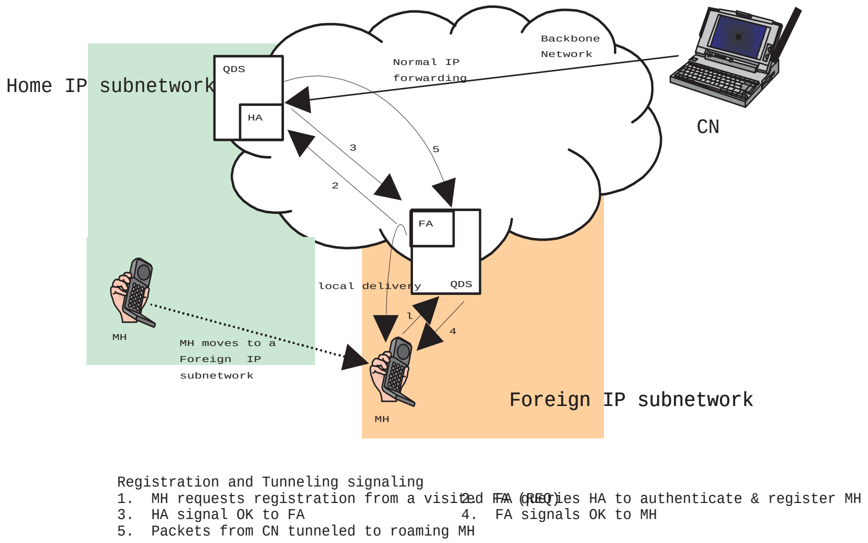


Figure 2.9: Illustration of Mobile IP's triangular routing (Source: [44], ©IEEE).

of poor network conditions. For example, when one sends e-mail one does not usually worry much about its exact delivery time. Traffic of interactive applications, such as telnet, Web browsing and file downloading, require minimum QoS guarantees, and are hence somewhat tolerant of poor network conditions. For example, when downloading a Web page, one usually expects the page to appear within a certain time, after which the user backs off or retries.

Traffic from streaming applications, such as packetized voice & video, have relatively relaxed delay requirements and can thus tolerate some jitter (i.e. per-packet delay variance) by smoothing out the received packets via, for example, buffering (i.e. playback). In contrast to pure NRT traffic which may be fully downloaded before viewing, streaming traffic is viewed/watched while being downloaded. It thus requires a transfer speed at least as fast as it is being displayed on a screen. At the other end of the spectrum are traffic of conversational applications, such as speech and video telephony. These applications demand strict QoS guarantees and do not tolerate any deviation from contracted network service requirements. Although delay sensitivity has been the focus of the above classification, other QoS metrics may count, such as network reliability (e.g. packet error rate), throughput (bandwidth), and jitter, depending on the traffic type [34]. Applications traffic can also be coarsely grouped into real-time (RT) or non-real-time (NRT), and elastic or inelastic. In this case, RT or inelastic traffic, such as video/audio conferencing and gaming, require the network to respect their QoS requirements unconditionally in both spatial and temporal dimensions, whereas elastic or NRT traffic, such as WWW, ftp, e-mail, file transfer and on-line transactions, are somewhat adaptable to degrading network conditions which result in applications' traffic QoS violations.

2.7 Business Models for Wireless Internet

After acquiring expensive infrastructure, coupled with high RF spectrum acquisition costs in some geographical regions, network operators and/or service providers need feasible business models so as to receive enough revenues to pay off their investment. At the same time service charges should not be too expensive so as to attract enough number of users. It has been acknowledged that pricing can be used as a form of traffic or congestion management scheme. This has spawned a number of research interests into pricing-sensitive QoS models.

Some Internet Service Providers (ISPs) have been charging their clients the usage of the best-effort Internet service using the billing model [58]

$$P = c_0 \ + \ c_1 \times T \ + \ c_2 \times V, \tag{2.16}$$

where $c_k,\ k = 0, 1, 2$ depend on the selected tariff, T is the service usage time, and V is the volume of traffic downloaded [58]. Other ISPs invoice users only for the access to the Internet. While these simple models perhaps sufficed for the single-service Internet, the multi-service wireless Internet requires better and fairer charging models.

Some of the accounting models considered for MOWINT services are: (a) Paris metro pricing [77, 78], (b) smart market [67, 68], (c) flat-rate pricing, (d) usage-based pricing, (e) quota-based pricing [11], (f) edge charging/pricing, (g) metered charging, (h) Fair charging and fair allocation, and (i) expected capacity. Most of these accounting models are self-explanatory. In the following, the first two charging models are briefly discussed. A detailed descriptions of pricing models can be found at the sites [104, 105, 106] and in [89]. Figure 2.14 illustrates the components of accounting for wireless Internet services [89].

2.7.1 Paris Metro Pricing Model

The Paris Metro Pricing (PMP) model [77, 78] attempts to provide differentiated QoS in packet networks using *only* pricing differentiation. Inspired by the two-class Paris metro service, it partitions communications network resources into several logically separated networks or channels, and price them differently.

The PMP model advocates stress the simplicity of the best-effort Internet service, as each logical network treats all packets equally on a best effort basis [78]. Thus, there is no service guarantees whatsoever. The rational behind it is to use differential pricing to reduce usage of some of the logical services, thereby reducing or even preventing congestion and providing a differentiated (better) service. In essence, it integrates traffic management with service pricing. The PMP model argues that, if a supposedly first-class logical network (channel) gets congested some of its users will migrate to a lower class service and hence self-regulate the intended service differentiation.

2.7.2 Smart Market

The *smart market* argues that users willing to pay the highest price for a service have the highest value for the service. It sets the price for network accesses at different priorities. If the same access charge is used at different times and/or different priorities then it can happen that users willing to pay more access fees may experience congestion blocking, while users not willing to pay high access fees receive service. This also causes the "tragedy of the commons" problem as every user will compete for the best service in the absence of tariff differentiation. The smart market model defines the two terms:

- *bid*—users willingness to pay for a network access, and
- a predefined *cut-off* bid (b_{th}).

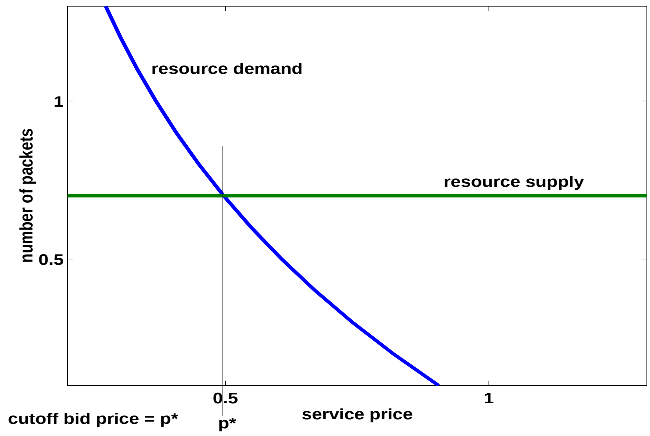


Figure 2.10: Determination of the smart market's threshold bid price.

The bid is set in the packet header, and each router admits a packet if its bid exceeds the threshold b_{th}. Once a packet is admitted, it is charged the cut-off bid price irrespective of its actual bid. Hence, a packet is never charged more than its bid.

Given a network capacity (resources), the cut-off bid price is determined from the intersection of available resources (bandwidth) and the resources in demand, as illustrated in Fig. 2.10. Comparing the smart market with auctioning, the packet bid is seen as a *relative* (not absolute) priority scheme, as the higher the bid, the higher the chance of exceeding the current bid threshold, and hence the higher the chance of getting a service at a router. In the smart market pricing model, users willing to trade off service quality for service charge can select a small bid in their packet headers. The bid can also be set to reflect the time of day, etc.

2.7.3 Hybrid Billing Model

It may be interesting in some situations to consider a pricing model considering as many as the following factors: access charge (A), volume of data transported (received and/or sent) (V), QoS level used (Q), and time of day (T). In this case, a reasonable charging (C) model may be formulated as

$$C = f(A, V, Q, T) = c_0A + c_1V + c_2Q + c_3T \tag{2.17}$$

where $c_i,\ i = 0, 1, 2, 3$, are some coefficients, which can be static or vary with factors, such as time of day, congestion level, type of application communicated, etc. In general, $f(A, V, Q, T)$ can also be a non-linear function of its arguments.

Telcos have been pricing the usage of voice-oriented services based on time of day, call destination (D) (i.e., local, inter-state, or international), and the duration of the call (T_{d}). However, owing to the connectionless and *always-on* nature, wireless IP services cannot use the same pricing models. IP addresses may not tell easily (if at all) the session endpoint; always-on will result in prohibitive invoices; and connectionless transport permits statistical multiplexing of other users' traffic onto the same infrastructure during inactive sessions of a given user, resulting in unfair charging.

2.8 Traffic Models

Since real traffic may not be available during protocol and/or system design, traffic models are used to predict the performance of communications protocols, such as scheduling algorithms. Traffic models are used for analytic modelling of communications systems and protocols, and as input to discrete-event system simulations [29]. This section reviews a sample of source traffic models commonly used for communications networks performance analysis.

2.8.1 Poisson Traffic Characterization

A Poisson process (the oldest traffic model) is a memoryless, pure birth process, i.e., a special class of Markovian stochastic processes, with a constant birth rate, say, λ, and independent increments. With Poisson traffic modelling, consecutive traffic (packet) arrivals are assumed uncorrelated or independent. As a member of birth-death processes, a Poisson process has only nearest-neighbour state transitions. For a Poisson random process, the time between state transitions is exponentially distributed.

Given $Y(t_1, t_2)$ as the number of arrivals in the time interval $[t_1, t_2)$ of a Poisson process $Y(t)$, the following relation holds

$$Pr\{Y(t_1, t_2) = i\} = \frac{[\lambda(t_2 - t_1)]^i e^{-\lambda(t_2 - t_1)}}{i!}, \tag{2.18}$$

where $t, t_1, t_2 \in \Re_+$. If $\tilde{a}$ is the inter-state transition time, then its p.d.f. is exponentially distributed according to

$$f_{\tilde{a}}(x) = \lambda e^{-\lambda x} = \frac{d}{dx} Pr\{\tilde{a} \leq x\}, \quad x \geq 0. \tag{2.19}$$

Poisson-modelled traffic is said to possess a probability distribution with an exponentially-decreasing tail or *light-/short-tailed distributed.* A generalized Poisson model where, instead of a single arrival, a random number of i.i.d. arrivals occur simultaneously is referred to as a *batch Poisson process.*

2.8.2 Markov-Modulated Traffic Characterization

In the Markov-modulated traffic model (MMTM) [49], the traffic source characteristic is quantized into a finite number of irreducible continuous-time Markov chains or states, say N_s. The traffic arrival rate and process are variable, and are controlled (i.e., *modulated*) by the corresponding Markov state. The Markov modulation serves to introduce some correlation into the time evolution of the traffic arrival/generation process. A special case of a Markov-modulated traffic model is the *Markov-modulated Poisson process* (MMPP). In MMPP, the traffic arrival process in each state, say n, is Poisson distributed with rate λ_n, and the state dwell (sojourn) time in each Markov state is exponentially distributed. Hence, an MMPP is said to be a doubly stochastic process. The traffic generation rate depends on the state, the modulated process is Poisson distributed, and the temporal evolution of the modulating process is Markovian.

The fluid version of a Markov-modulated traffic model considers traffic as a continuum of fluid with a given flow (bit) rate with packets seen as the atoms of the fluid and traffic measured in volumes. Fluid models are justified on high-speed links in which packet transmission time becomes an infinitesimal quantity, and the effects of a single packet on the system becomes almost negligible. Fluid models are commonly used to model bursty traffic sources. An on/off traffic source with state transition diagrams as shown in Figs. 2.3 and 2.11 is a common model for a two-state fluid version of the MMTM, where the traffic generation rate in the *off* state is $\lambda_{\text{off}} = 0$, hence referred to as an *interrupted process.* If the dwell times in each state is exponentially distributed, then we have a fluid MMPP traffic model.

Recently, a more interesting type of MMTM is where the statistical properties of the 'on' and 'off' *state dwell times* are modelled by *heavy-tailed* distributions, such as the Weibull or the Pareto (see Eq. 5.16). By superimposing many of such on/off traffic sources, a self-similar traffic can be obtained, which suitably models WWW data traffic. A Weibull distributed random process $Y(t)$ has the p.d.f.

$$f_Y(y) = \frac{\alpha}{\beta^\alpha} y^{\alpha-1} e^{-(\frac{y}{\beta})^\alpha}, \quad \text{for } y \geq 0, \tag{2.20}$$

where $\beta,\ \alpha > 0$ are real numbers referred to as scale and shape parameters, respectively. Note that the exponential distribution is a special case of the Weibull distribution, i.e. when $\alpha = 1$.

2.8.3 Self-Similar Traffic Model

Recent network traffic research widely accepts that packetized data, especially Ethernet LAN, file transfer and VBR video traffic, do not conform to the widely used Poisson process, but it is rather *bursty.* It is shown that such traffic show self-similarity over all time scales [81]. The effects of self-similarity in network traffic on QoS is not yet certain. However, it has been identified that self-similar traffic has a queue length distribution which decays slower than the exponential distribution underlying Poisson traffic. The slower decay of queue length can lead to an increased packet losses in IP based networks.

Self-similar pattern in the traffic can be observed through (a) long-range dependence of traffic, i.e., non-summable autocorrelation function; (b) diverging spectral density of traffic at the origin; and (c) monotonically increasing index of dispersion counts with sample time [81]. A key parameter indicating self-similarity level in traffic is the Hurst parameter (H). Synonymous expressions used to describe traffic exhibiting self-similarity patterns are: (a) heavy-tailed distributed traffic, (b) bursty traffic, and (c) long-range dependent (LRD) traffic. A common p.d.f. used to describe a self-similar traffic is the heavy-tailed Pareto distribution given by

$$f_X(x) = \mu\vartheta^{\mu}x^{-\mu-1}, \quad \text{for} \;\; \mu, \vartheta \geq 0 \text{ and } \vartheta \leq x < \infty, \tag{2.21}$$

where ϑ and μ are the location and the shape parameters of the distribution [83]. Since the mean, m_X, of X is $m_X = \int_{\vartheta}^{\infty} x f_X(x)dx = \vartheta\mu/(\mu-1)$, obviously $\mu \neq 1$. Other forms of heavy-tailed density functions used in the literature are (see e.g. [81] pp. 326–334)

$$f_X(x) = \vartheta\mu(1+\vartheta x)^{-(1+\mu)}, \qquad \vartheta > 0, \;\; 1 < \mu < 2 \text{ and } 0 \leq x < \infty \tag{2.22}$$

and for $A > 0, \;\; 1 < \mu < 2$

$$f_X(x) = \begin{cases} \frac{\mu}{A}e^{-\mu x/A}, & x \leq A \\ \mu(\frac{A}{e})^{\mu}x^{-(1+\mu)}, & x > A. \end{cases} \tag{2.23}$$

We note that a smaller μ yields a heavier tail of the distribution.

An interesting type of self-similarity is the second-order self-similarity. Let $Y(t), \; t \in \mathcal{Z}$ be a discrete-time random process with mean $\mu_y = E[Y(t)]$. $Y(t)$ indicates the traffic volume measured in units of bits, octets, or packets. Assume $Y(t)$ is wide-sense (2^{nd}-order) stationary, and let $Y^m(t)$ be $m-$level non-overlapping aggregates of $Y(t)$. Let $R_y(t_0)$ and $R_y^m(t_0)$ be the autocovariances of $Y(t)$ and $Y^m(t)$, respectively. Then $Y(t)$ is a 2^{nd}-order self-similar traffic if [81]

$$R_y(t_0) = E\Big[\Big(Y(t) - \mu_y\Big)\Big(Y(t-t_0) - \mu_y\Big)\Big] = \frac{\sigma_y^2}{2}[(1+t_0)^{2H} - 2t_0^{2H} + (t_0-1)^{2H}], \tag{2.24}$$

or asymptotically 2^{nd}-order self-similar if

$$Y^m(t) = \frac{1}{m}\sum_{t_1=m(t-1)+1}^{mt} Y(t_1), \quad m \in \mathcal{Z}_+ \tag{2.25}$$

and

$$\lim_{m\to\infty} R_y^m(t_0) = \lim_{m\to\infty} E\Big[\Big(Y^m(t) - \mu_y^m\Big)\Big(Y^m(t-t_0) - \mu_y^m\Big)\Big]$$

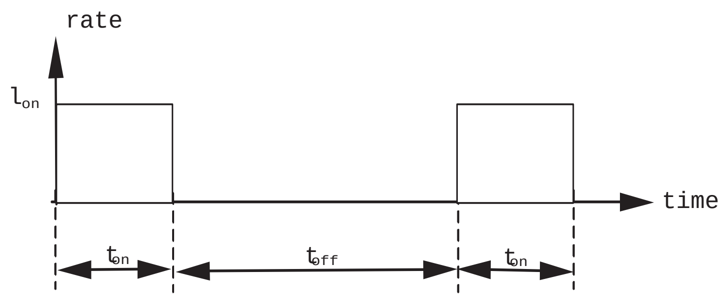


Figure 2.11: Simple illustration of an on/off bursty traffic source.

$$= \frac{\sigma_y^2}{2}[(1+t_0)^{2H} - 2t_0^{2H} + (t_0 - 1)^{2H}], \qquad (2.26)$$

where $\sigma_y^2 = E[(Y(t) - \mu_y)^2]$, $k \geq 1$ and $1/2 < H < 1$ is the Hurst parameter. These two equations indicate that the correlation is exactly (i.e., $R_y(t_0) = R_y^m(t_0)$) or asymptotically (i.e., $R_y(t_0) = \lim_{m\to\infty} R_y^m(t_0)$) preserved under different time scales (or aggregation) [81].

Park and Willinger [81] describe two ways of generating a LRD traffic:

- aggregate c independent on/off sources (see Fig. 2.11), each injecting traffic according to a Poisson process with constant rate of, say, λ_{on}, when on; and let $c \to \infty$.
- sum c independent traffic sources with ACF such as given in (2.24) or (2.26) and with peak rates, say, λ_{on}, and let $\lambda_{on} \downarrow 0$ as $c \uparrow +\infty$.

2.8.4 EBB Traffic Model

Let $A_i(\tau, t)$ be the amount of traffic generated by a traffic source i in the time interval $[\tau, t]$, and let ρ_i, α_i and δ_i be respectively the source's long-term maximum traffic generation rate, decay rate of an exponential decay function, and a constant prefactor. Then, for any τ and t, source i's arrival process $A_i(\tau, t)$ is called $(\rho_i, \alpha_i, \delta_i)$-EBB (exponentially bounded burstiness) process if [113]

$$Pr\{A_i(\tau, t) \geq \rho_i(t - \tau) + y\} \leq \delta_i e^{-\alpha_i y}, \quad \forall y \geq 0 \qquad (2.27)$$

For stability reasons, the sum of ρ_i's of traffic from all sources simultaneously competing for service at a scheduler (server) should not be greater than the maximum scheduler (server) capacity, say, r_{S}, i.e., $\sum_{\forall k} \rho_k < r_{\mathrm{S}}$.

2.8.5 Packet Size Distribution

Besides the arrival traffic process, packet size is another important parameter in network modelling and performance. As noted in Sect. 1.3.2, IP packets over various networks have variable sizes in the range (575, 65 575) bytes. However, for some reasons such as transmission delay, many networks have much smaller maximum transmission unit (MTU) than the allowable 65 575$(= 2^{16} - 1 + 40)$ size. For example, Ethernet LANs have an MTU of 1518 bytes. This, however, necessitates packet fragmentation somewhere within the IP network. We note that, in IPng network, routers are not allowed to fragment packets, requiring IPng sources to learn MTUs of the links they attach to.

Recently, the size of packets transmitted over the Internet (especially, WWW data) is believed to follow the Pareto distribution given in (5.16). Bimodally distributed packet lengths is also acknowledged in the recent literature [2]. A bimodal probability mass function (p.m.f.) used to model the size of IP packets in [2] can be approximated by

$$\begin{aligned} p_X(L_p) \approx\ & 0.13\delta(L_p - 96) + 0.12\delta(L_p - 144) + 0.42\delta(L_p - 192) \\ & +0.01\delta(L_p - 720) + 0.01\delta(L_p - 960) + 0.27\delta(L_p - 1104) \\ & +0.01\delta(L_p - 1248) + 0.03\delta(L_p - 1536), \end{aligned} \tag{2.28}$$

where L_p is the IP packet size in bytes and $\delta(\cdot)$ is the delta function with $\delta(k - l) = 1$ for $k = l$ and zero otherwise.

2.9 Traffic Admission Control

Traffic admission control (TAC) falls into the area of radio resource management (RRM). Basically, RRM manages radio link QoS, optimises or maximises

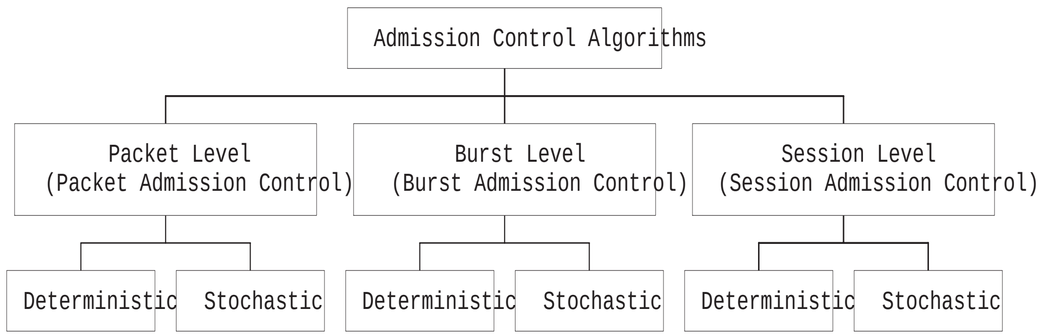


Figure 2.12: Grannularity of admission control algorithms.

network capacity and maintains planned network coverage [51, 41]. TAC algorithms can operate on session (call/connection) level, burst level, or packet (cell/frame) level, each of which can be deterministic or probabilistic, as illustrated in Fig. 2.12. TAC algorithms can also be categorized into measurement based or parameter based [111]. They monitor and measure the instantaneous network resources available, and upon new traffic request, decide whether to reject or accept it, based on some control criteria. The control criteria are mainly used to prevent disruption of committed QoS profiles or service level agreement (SLA) to active sessions and the new access request once admitted into the network. Thus TAC algorithms are used to guarantee certain but differing QoS levels to a mix of different traffic types with different QoS requirements, without sacrificing network throughput and spectral efficiency [41].

By discriminating and restricting network entrance of some traffic, TAC schemes prevent network congestion proactively, which is crucial to QoS provisioning [41]. Network congestion can also be solved by traditional means such as resource overprovisioning and usage of leased or dedicated links. A TAC algorithm can operate in the BTS, for distributed RRM, or in the RNC, for centralised RRM. Irrespective of the type of and where a TAC algorithm actually operates, it has to trade off the following basic criteria before accepting a new traffic request into the network:

- new session request, if accepted, will not deteriorate QoS of all existing connections/calls;
- network has enough surplus resources to meet required QoS of the new call;
- network resources are used efficiently with network running near capacity;
- new call request, if accepted, will not deteriorate planned network coverage; and
- fairness to traffic of the same service class.

Much research has been reported on admission control schemes to provide QoS to applications. However, most of them are based on non-IP protocols such as ATM (asynchronous transfer mode). Other proposed admission control protocols for mobile networks, e.g. [102], do not consider the overlying network layer protocol.

2.10 Packet Schedulers

Packet schedulers are algorithms which assign resources and service order to generic streams (GeSs) contending for service at a common facility, either wired or wireless. Schedulers are one of the very relevant mechanisms operating in communications network nodes to provide QoS. On the networking level, schedulers can be tailored to the connectionless IP (Internet Protocol) philosophy, or to the connection-oriented ATM (Asynchronous Transfer Mode) philosophy. This thesis however, focuses on schedulers for IP based wireless media. Design of packet schedulers requires design choices and trade-offs, such as *work-conserving* or *non-work-conserving.* In contrast with non-work-conserving schedulers, work-conserving schemes never idle if there are packets to be scheduled. Hence, non-work-conserving schemes can waste network bandwidth as they buffer traffic and idle network resources. However, such schemes are needed to reduce jitter (i.e., delay variance) in traffic through smoothing for real-time applications. The buffering also increases the average end-to-end traffic delay.

Scheduling philosophy can be based on *GeS (traffic) isolation,* where resources are reserved for each GeS, or be based on *resource sharing,* where all GeSs share a common resource. Extreme examples of GeS isolation and resource sharing are the circuit-switched (connection oriented) PSTN telephony and the packet-switched (connectionless) fixed Internet, respectively. Schedulers based on GeS isolation have imbedded *GeS protection* as a misbehaving (greedy) GeS cannot degrade the QoS experienced by other GeSs. It can also provide a deterministically bounded delay which is essential for strictly delay sensitive applications' traffic. However, GeS isolation schedulers show degraded resource efficiency (i.e., no *statistical multiplexing),* as resources are mostly underutilized. Resource sharing schedulers do not provide GeS isolation nor protection against misbehaving GeSs, and can provide only *probabilistic delay bounds* (i.e., *soft QoS guarantees*). However, they achieve better *statistical multiplexing.* Schemes combining both philosophies are still being researched, such as the fair-queuing schemes. Scheduling schemes need be fair, but fairness does not necessarily mean that different GeSs are assigned equal network resources. A common fairness policy in use is the *max-min fairness* [111]. Performance of packet scheduling algorithms are evaluated based on metrics such as fairness, computational complexity, tightness of delay bounds, GeS protection, protocol overhead, and achievable network utilization [17].

Examples of packet schedulers commonly used on the fixed Internet are first-come, first-served (FCFS), generalized processor sharing (GPS) [79, 80], which is also known as weighted fair queuing (WFQ), hierarchical round-robin (HRR) [59], stop-and-go [32, 33], and their variants. Examples of wireless channel state dependent schedulers proposed recently include Largest Weighted Delay First (LWDF) and its modification, the Modified LWDF (M-LWDF) [4], the non-preemptive priority with partial assurance (NP^3A), and the BL^4DF proposed in this thesis (see Chapter 4.2). In depth treatments of packet scheduling algorithms can be found in, e.g., [17, 111, 26]. Since the GPS (or WFQ) is a very popular scheme in the literature, it is briefly discussed below. FCFS is also discussed, as it is used in the thesis.

2.10.1 First-Come, First-Served (FCFS)

The FCFS scheme is sometimes *wrongly*[4] referred to as first-in, first-out (FIFO). It is the simplest packet scheduling scheme in use today. FCFS transmits packets in the order of arrival, and drops an arriving packet on full buffer. Strictly speaking, FCFS does no scheduling at all. This scheme is tailored to the best-effort service, as it does not different packet, but not suited for the upcoming multiservice wireless Internet.

2.10.2 Generalized Processor Sharing

Scheduling algorithms can be *traffic centric* or *channel centric.* Traffic-centric schemes concentrate on fulfilling the QoS requirements of applications traffic without considering the wireless channel noise levels, and hence achievable *data rate*[5]. Channel-centric schedulers, on the other hand, optimise the usage of the wireless channel without doing much with the actual traffic QoS requirements.

The GPS is an *ideal* work-conserving scheduler meets the exact *max-min fairness* criteria, in that it shares the entire link data rate to all backlogged traffic streams (GeS) in proportion of their weights or minimum data rate requirements [79, 80, 17]. The GPS is and idealization with the basic assumptions that the link capacity can be split infinitesimally to serve all backlogged sessions instantaneously. However, practically, only a single GeS can be served at a time and link capacity cannot be split beyond the rate required to transmit a packet.

The packet based version of the GPS scheme, a practically realizable version of GPS, is referred to as weighted fair queuing (WFQ) or packetized GPS (PGPS). PGPS attempts to approximate the *ideal* GPS scheme. The GPS scheme operates as follows. First, it defines the parameters [79]:

- c : number of GeSs sharing a (wireless) link with data rate R_b,

[4]I use *wrongly* as the fact that a session, say, A, begins service earlier than another session, say, B, does not necessarily mean that A will complete service earlier than B. It is true only for special cases of a single server or/and equal service times.

[5]In this thesis, we differentiate between channel data rate (r) and channel bandwidth (b) according to the Shannon-Hartley information theoretic law, $r = b * \log_2(1 + SNIR)$, where $SNIR$ is the instantaneously received signal-to-noise-plus interference ratio.

- r_k : normalized minimum data rate requirements of GeS_k,
- $B(\tau)$: the set of backlogged GeSs at a scheduling instant τ,
- $g_k(\tau)$: data rate received by backlogged GeS_k at a scheduling instant τ.

The GPS requires that the minimum data rates (r_k) cumulatively adhere to the admission control condition

$$1 \geq \sum_{k=1}^{c} r_k. \tag{2.29}$$

The instantaneous data rate received by GeS_k is then computed as

$$g_k(\tau) = \frac{r_k}{\sum_{i \in B(\tau)} r_i} R_b \geq r_k R_b, \quad k = 1, 2, \cdots, c. \tag{2.30}$$

2.11 Buffer Management Schemes

Buffer management (BM) is a scheme used to prevent congestion at a buffer via packet discarding. Its decisions include how and when, and which packets to discard in times of congestion [17]. Packet discard can be done proactively upon packet arrival or upon incipient occurrence of network congestion. Packet discard can be performed on a per-packet, per-flow or per-class basis. Like packet schedulers, performance measures of interest include complexity, fairness and efficiency. Common BM schemes include random early detection (RED), drop on full and the simplest of all, tail dropping (TD). In depth treatments of packet scheduling and buffer management algorithms can be found in [17, 111, 26]. TD and RED are briefly discussed next, as they are applied in the performance analysis chapters of this thesis.

2.11.1 Tail Dropping (TD)

TD sets a threshold (Q_{th}) in packets for each queue/buffer, and an incoming packet is dropped upon arrival if the buffer occupancy has reached (Q_{th}), i.e., full queue. Though simple, TD suffers from the *full queue* and *lockout* syndromes. Lockout is the situation where one or few sessions monopolize the transmission buffers, causing drastic packet lost to other sessions. The blocked sessions may then back off simultaneously, resulting in timing effects such as *global synchronization* [27, 17]. Full queue has no chance of absorbing packet burst arrival, which, if discarded, can also cause global synchronization. There are other variants of drop-on-full BM schemes: random drop-on-full and drop-front-on-full. The RED algorithm is designed to avoid full queues and lockout through active buffer management. RED is discussed next.

2.11.2 RED Algorithm

RED [27] is an active buffer management (ABM) scheme which finds wide application in TCP/IP routers. The RED algorithm is illustrated in Fig. 2.13, where d_i is the probability that an arriving packet belonging to a GeS_i is marked when the average buffer occupancy is k, min_i and max_i are the average minimum and maximum queue length thresholds, and $d_{m,i}$ is the maximum packet marking probability. Upon each packet arrival, RED computes k and then $d_i(k)$. Mathematically, $d_i(k)$ is given by

$$d_i(k) = \begin{cases} 0, & k < n_i \\ \frac{k - min_i}{max_i - min_i} d_{m,i}, & min_i \leq k < max_i \\ 1, & k \geq max_i. \end{cases} \tag{2.31}$$

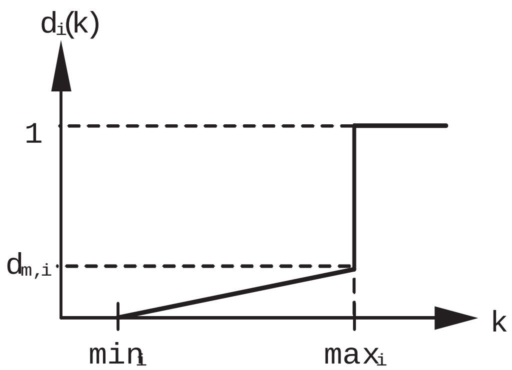


Figure 2.13: Simplified illustration of the RED algorithm.

The RED scheme is scalable, as it does not maintain per-flow state information in the routers. If all marked packets are dropped, RED prevents *full queues* and *lockout* (i.e., monopoly of buffer by one or a few GeSs for a prolonged time). Both full queue and lockout cause global synchronization [27] typical of TCP (i.e., responsive) sources, which, in turn, degrades link/network utilization, as many traffic sources back off simultaneously in response to packet loss (i.e., congestion indication signals).

2.12 Related Works

The wealth of published work in the open literature has provided a firm foundation for this thesis. Owing to limited space, only the very few related works are cited in this section. Other works which helped the design and organization of this thesis are cited in appropriate places in the body of the thesis. Contribution 1 is based on the candidate's own prior work and the works cited in references [34] and [44]. Similar works to the BL^{4}DF packet scheduling scheme and its variants proposed in the second part of this thesis are reported in references [3, 4] and [8]. These works, however, are more user-centric instead of traffic-centric. While the queuing and scheduling at the transmitter are based on users' link quality [3, 4, 8], the scheme proposed in this thesis queues traffic based on traffic QoS requirements, but accounts for user's link quality in deciding the scheduling orders. The packet scheduling schemes (i.e. CSDPS and NP^{3}A) proposed in Chapter 4 can degenerate into the widely known Weighted Fair Queuing (WFQ) scheme [79], which is the packetized version of the celebrated Generalized Processor Sharing (GPS) scheduling proposed by Parekh and Gallager [79, 80]. For Contribution 3, the primary references used in this part of the thesis are the excellent works by Chan and Hong [18]; Anick, Mitra and Sondhi [5]; Mitra [71]; and that of Krunz and Kim [62, 61].

Chan and Hong [18] analysed the assured forwarding (AF) Per-Hop Behavior (PHB) of the IETF's differentiated services (DiffServ) model over a two-state Markovian wireless channel model widely known as Gilbert-Elliot model. The DQoS scheme is based on the RED/RIO buffer management scheme and an

FCFS service order within a queue class. It also accounted for link layer error control. The results focused on average packet loss due to buffer overflow, average packet delay and effective throughput as functions of the wireless channel utilization. My work differs from [18] in many ways, such as, [18] uses a two-state DTMC, does not show how parameters such as the stationary distribution, the transition and the state occupancy probabilities of the Markovian model are computed, and assumed a fixed relation between the service rates in both channel states which may not be necessarily so in practical situations. Unrealistic results can be achieved if the real wireless properties are not considered in the Markov modelling. The two state model does not account for the handoff region (hysteresis) common to mobile systems.

Like [18], some works on wireless IP QoS assume some wireless link-level properties and, based on such assumptions, study network-level QoS, without considering the radio link's minutiae. Other works specialise on a specific layer without a linkage to other layers. However, to understand the interactions between the achievable QoS at the networking layers and the link-level properties, a unified study such as the framework of this thesis is crucial. This is foremost important with the upcoming QoS-aware protocol layering and the blurring between layers (inter-layer communications), as well the convergence between Internet and wireless networks.

Anick, Mitra and Sondhi [5] and Mitra [71] provide stochastic theory of a fluid model of traffic sources, traffic consumers (transmission media) and buffers. Since the eigenanalysis in these works are generally applicable, I applied them to situations where the consuming media contains a wireless component, as in [62, 61]. Unlike [5], which assumes a channel or a switch with a constant output rate, in this thesis the buffer is coupled to a wireless channel with a spatio-temporarily varying output data rate. Kim and Krunz in [62, 61] follows this pattern, and they also utilized the framework of Mitra and Anick's work [71, 5].

Kim and Krunz [62, 61] accounted for the interactions between the link-layer error control scheme, time-varying wireless channel, and the packet-level QoS provisioning using a two-state Markov channel model. The mathematical analysis draws from the works of Mitra, Anick, and Sondhi [71, 5]. In contrast to my work, [62, 61] use a single on/off source as traffic generator and do not consider the transceiver properties such as fading and signal modulation. The effects of mobile user's speed of motion on QoS are also not considered.

The finite-state, discrete-time model used for the wireless channel in this thesis was motivated by the classic work of Wang and Moayeri [110]. Wang and Moayeri's multi-state Markov channel model, however, permits only nearest-neighbour state transitions, and they computed the Markov state parameters using Rayleigh fading model and BPSK modulation. In some parts of this thesis, the nearest-neighbours state transition assumption is not enforced in order to permit transitions between non-adjacent Markov states.

2.13 Summary

This chapter has presented the material that provides the context needed to understand the central theme of the thesis, QoS-enabled mobile wireless Internet systems, with minimal difficulties. The latter section reviewed a sample of the relevant literature. Other literature which helped in the preparation of the thesis are cited in appropriate sections. As can be seen from the wealth of background material, the wireless IP QoS issues are rather diverse and manifold, and hence no claim can be made that all relevant material are covered, or will even be covered in the rest of the thesis. However, the presented material forthwith is representative of wireless IP QoS topical issues.

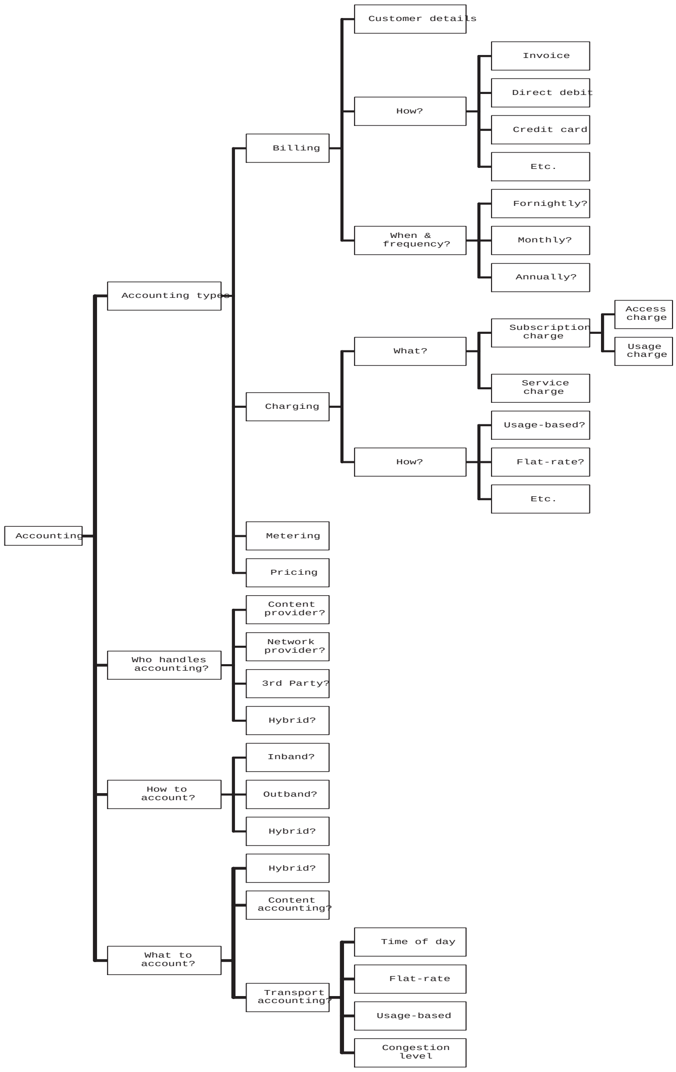


Figure 2.14: Components of mobile wireless Internet accounting.

Chapter 3

MOWINTA—A QoS Enabled Wireless Internet Access Architecture Based on Mobile IP/IEEE 802.11 Interworking

Realistic realization and mass acceptance of mobile data services require networking architectures offering acceptable Quality of Service (QoS) and attractive tariffs. A novel strategy for this goal is maximum integration of popular data networking standards and their infrastructure into wireless networks. This chapter discusses Mobile IP based network architecture to provide IP services in the IEEE 802.11 wireless standard to support multimedia applications. IEEE 802.11 offers micromobility within multicell subnets, whilst Mobile IP supports macromobility between multicell subnets. Incorporating Mobile IP into IEEE 802.11 handoff mechanism in this way extends IEEE 802.11 micromobility with IP macromobility. Also, utilizing fast, seamless IEEE 802.11 handoff management reduces Mobile IP handoff delay to circumvent TCP throughput degradation during handoff and reduce frequency of Mobile IP signaling over the ether to conserve spectral efficiency. This feature seamlessly unifies IEEE 802.11 with the global Internet. Seamless integration of IEEE 802.11 with the Internet is crucial due to the continuing phenomenal popularity of the Internet and wireless communications, and ubiquity of IEEE 802.11 systems. To achieve the above IEEE 802.11/IP interworking efficiently, the architecture introduces a network entity called QoS enabled Distribution System (QDS), which is an extended IEEE 802.11's Distribution System. The IEEE 802.11's Layer 2 services are mapped onto those of the IETF Integrated Services (IntServ) architecture to provide QoS in both the wireless part of an end-to-end communications link and the backbone Internet. Mobile Resource ReSerVation Protocol (MRSVP), an extended RSVP tailored to mobile networking, is adopted to provide the needed signaling in IntServ.

3.1 Introduction

Commercial deployment of mobile multimedia services has been hindered by factors such as low data rate, high service costs, lack of or slow standardization, and the form factor of wireless handsets [34]. The third-generation (3G) mobile technologies, generically known as IMT-2000, provide fast wireless Internet access and are anticipated to reduce the impact of these issues. Device cost and form factor are improved by the realization of Moore's law coupled with the economies of scale. IMT-2000 establishes a common framework of standards that merge the attributes of paging, cordless, cellular and satellite networks to provide high bandwidth so as to support mobile multimedia applications in different mobility scenarios. The current impetus towards convergence of computer (Internet), video and voice networks will speed up the realization of mobile multimedia communications (MMC). Also, with continuing advancement of image and video coding technologies, even high-bandwidth applications may be supportable in future by IMT-2000 technologies at cell fringes and in rural areas experiencing data rates in the range 384kb/s-144kb/s.

Until late 2000, we had witnessed a frenetic and continuing advancement in and demand on the communications industry. However, important questions arising from these developments include: whether costs will allow customers to follow the technological bandwagon; and what percentage of the world's population can afford to utilize the technologies being developed. Such questions render cost reduction an indispensable part of the technological wave. To fulfil users' demand for advanced Telecom services at moderate costs, and to reduce the time-to-market of services, infrastructure of most 2G technologies (notably, GSM) is being leveraged through continuing upgrading. For instance, Table 1.1 shows GSM based evolution of mobile services from low-rate circuit-switched data (CSD) and Short Message Service (SMS) communications to high-resolution fourth-generation (4G) MMC [43]. Also shown in the table are the High Speed Circuit Switched Data (HSCSD), the General Packet Radio Service (GPRS) and the Enhanced Data rates for Global/GSM Evolution (EDGE), their theoretical data rates, and projected commercial rollout schedules. We expect 4G mobile technologies to offer even higher bit rates than IMT-2000 technologies to support *bandwidth-ravenous* applications. Since high data rates result in small TDMA bit duration (since the smaller the bit duration, the worse the signal immunity against multipath effects) and high CDMA code chip rates, 4G mobile systems may deploy Orthogonal Frequency Division Multiplexing.

On the network level, the Internet Protocol suite (TCP/IP) and its QoS architectures have the potential to champion both the access and core networks to yield an all-IP network. Mobile IP and its concomitant technologies have a key role to play in this anticipated IP breakthrough. For mobility, wireless technologies may dominate the access network, while all-optical solutions may continue to penetrate the core network to provide high switching speeds and multiplexing gains.

To reduce costs of wireless multimedia services and promote mass service usage, maximum reuse of popular data networking standards and their infrastructure is paramount. This point is one of the strengths of the architecture discussed in this chapter. In addition, due to the phenomenal popularity of the fixed Internet and wireless networks, integrating them to provide wireless data (wireless Internet) seems a very promising strategy. The number of Internet hosts grew from only 213 in 1979 to about 60 million in 2000 [34] and accelerated to 100 million by Jan 2001 worldwide [34]. Growth in mobile phone usage has been witnessed in various places. For example, 65 of every 100 persons in

Western Europe had a mobile phone subscription by July 2000 (see Ref [4] in [34]). The above figures exclude mobile computer users, and the fact that many users may use a single host is also unconsidered. Figure 1.1 shows a snapshot of the evolution of worldwide mobile cellular subscribers since 1992 [34].

This chapter proposes a wireless Internet access architecture based on the IEEE 802.11 WLAN interface. In accordance with the state-of-art, this architecture supports two types of mobility: micromobility (or intra-subnet mobility) and macromobility (or inter-subnet mobility). Micromobility is handled within the IEEE 802.11 layer 2 (L2) using 802.11 mechanisms, while macromobility is handled at L3 using mobile IP mechanisms. Each subnet consists of one or more radio cells, with each cell served by ab 802.11 access port (AP). L2 services of 802.11 are mapped into IntServ/MRSVP or DiffServ QoS architecture to introduce explicit differentiated QoS into 802.11 systems. The idea underlying this chapter originates from [34], which proposes a faster mobile IP L3 handoff by utilizing L2 wireless handoff signalling.

The rest of this chapter is structured as follows. Section 3.2 presents the background material of the chapter. Section 3.3 provides a detailed review of the IEEE 802.11 WLAN standard. The main topic of the chapter, a QoS enabled Mobile IP/IEEE 802.11 interworking, is presented in Section 3.4. Since the presented architecture requires an extensive material, the mobility management aspects are treated in a separate section, namely, Section 3.5. The chapter concludes with a summary in Section 3.6.

3.2 Background Material

This section discusses the objective and the rationale of the architecture presented subsequently. Micro-mobility management in IEEE 802.11 is similar to Mobile IP macro-mobility management. In IEEE 802.11, the mobile host detects its location change via MAC sublayer beacon broadcasts, just as Mobile IP hosts do via agent advertisement. The agent advertisement is a router advertisement extended with care-of addresses, which are used by roaming terminals for packet routing purposes. This feature is exploited to ease 802.11/Mobile IP internetworking.

3.2.1 TCP QoS Degradation in Wireless Environment

It is known that the throughput performance of Transmission Control Protocol (TCP) [88], the most commonly used transport layer protocol in the global Internet, degrades in a lossy mobile environment during handoff [19, 34]. This results from interruption of the communications link during handoff, undue random handoff latency, or link degradation (bit/packet errors) due to sporadic radio fading and interference. Long handoff latency may cause transport protocol timeout, while link interruption and degradation may cause IP datagram loss. Both events trigger TCP congestion control mechanisms (e.g. *slow start* and *congestion avoidance*), which were traditionally designed for reliable fixed links requiring no handoff/mobility. The problem with TCP is that it was developed for use over fixed links and it is not very well behaved over wireless channels, which may be prone to poor channel error performance and handoff delay due to mobility (see also Section 2.5.5). Traditional TCP assumes that every packet loss or delay is due to congestion in network switches and thus releases its throttling mechanism, which degrades throughput performance. This requires wireless concealment schemes at the ISO/OSI layers below TCP.

Researchers have proposed a number of approaches to circumvent the TCP ill effects related to motion as outlined above. Most of the approaches proposed in the literature fall into two basic categories: a *motion-unaware* TCP approach that hides mobility from TCP, and a *motion-aware* TCP approach that adapts TCP to motion [15, 66]. The motion-aware TCP approaches (e.g. changing TCP retransmission timer resolution [15], delaying TCP congestion management [66], and fast retransmission [15]) usually require modifications to the TCP/IP suite and/or intercommunication between network and transport layer protocols. Increasing TCP retransmission timer resolution may exacerbate TCP degradation if there is a satellite component in the communications link. Network layer/transport layer inter-communication violates traditional networking principles of protocol independence. The result is that one protocol cannot be modified without affecting the others. This makes protocol updating costly and tedious. Moreover, modifying the large installed base of TCP/IP suite is costly and tedious.

On the other hand, the motion-unaware TCP approaches hide mobility from TCP by relying on such features as smooth (seamless) handoff during cell crossings [15]. This requires large enough buffers in the base station to keep data packets of all departing mobiles in the cell, and forward the packets to the new base stations to serve the mobiles. At best, this approach prevents packet losses due to handoff, but not losses due to any other reason such as actual network congestion or poor radio channel. In this chapter, the motion-unaware TCP approach is followed, which circumvents all the issues with the motion-aware approach. We utilize 802.11b's inherent fast handoff to reduce Mobile IP handoff delay to save TCP degradation.

3.2.2 Faster Mobile IP Handoff via IEEE 802.11 Beacon

Movement detection by mobile hosts is an indispensable event for Mobile IP handoff/location update mechanisms. However, the fastest movement detection algorithm commonly utilized by Mobile IP is Eager Cell Switching (ECS) [85], which incurs a minimum movement detection delay of about $t_{ECS=\tau/2}$, where τ is the inter-agent advertisement time. The recommended minimum value for τ is 1 sec [86], yielding a minimum movement detection time of 500 msec for Mobile IP hosts. Besides causing TCP performance degradation, this delay is unacceptable by the time-critical multimedia applications to be handled by third-generation IMT-2000 wireless systems.

As a wireless technology, movement detection in 802.11 can be performed via the MAC layer beacon signal, which is transmitted periodically and relatively faster than the recommended MIP movement detection scheme. The IEEE 802.11 standard recommends inter-beacon transmission time of not longer than 100 ms, in contrast to the 500-ms Mobile IP movement detection delay. Therefore, the overall network layer and link layer handoffs delay can be reduced by a factor of 5 during *external handoff.* We attempt to do this by requiring the databases in the QDS (see Section 3.3) to buffer the latest Mobile IP agent advertisement message and 'embed' it in the WLAN location management signalling. Hence, upon completion of wireless link layer handoff, a MH does not need to await an agent advertisement before updating its network mobility binding.

In this way, IEEE 802.11's fast, seamless handoff mechanism is utilized to reduce Mobile IP handoff delay in order to circumvent TCP throughput degradation by hiding motion from it. This approach reduces the overall handoff delay. The overall network signalling overhead is also reduced considerably, since periodic broadcast of bulky Mobile IP control messages over the ether is avoided, resulting in better spectral efficiency. The basic approach underlying this chapter was conceived in [44]; however, detailed explanation of the approaches was missing. A similar approach based on DECT is reported in [34, 65]. However, in [65], the network waits for the DECT mobility management to complete before forwarding the latest Mobile IP agent advertisement to the mobile station. Hence it is costlier in overall handoff delay and signalling overhead than the approach proposed in this chapter. Moreover, the connection to the old foreign agent is not released after MIP handoff until all IP datagrams are transmitted, which wastes resources.

In the following architecture, a Mobile IP subnetwork (subnet for short) is mapped onto a multicell WLAN subnet, which is managed by a single central controller denoted here as *QoS enabled distribution system* (QDS). Subnet is used to refer to both in this chapter. Because of this subnet mapping, a Mobile IP handoff is executed only if an external handoff (change of QDS) occurs, since all cells in a subnet have the same IP address prefix. Clustering multiple cells under the management of a single QDS also reduces the frequency of external handoff execution in the WLAN and corresponding reduced signalling overhead for radio resource conservation.

3.3 IEEE 802.11 Review

It is envisioned that the future broadband *mowint* will consist of interconnected WLANs based on standards, such as the IEEE 802.11b. The IEEE 802.11 standard has its own terminology. The Access Point (AP) is the base station

(BS). A station (STA) is any 802.11 compatible device, mobile or fixed, such as mobile node (MH). In the case of ad-hoc networking, a STA can perform some of the functions of an AP. Clear to send (CTS) and Ready To Send (RTS) are small control packets from transmitter to receiver or receiver to transmitter, respectively. A message length is called a network allocation vector (NAV). In the following, the IEEE 802.11 (dot11) standard is briefly reviewed.

3.3.1 Reference Architecture

Figure 3.1 shows a three-cell architecture of a wireless LAN (WLAN) based on *infrastructure dot11* standard. A group of stations in a cell is referred to as a Basic Service Set (BSS), which is the basic building block of the 802.11 standard. There are two variants of BSSs: an independent BSS (IBSS) and infrastructure BSS (which we refer to here as infra-BSS). The IBSS has no AP, and it is mainly used for direct, 'line-of-sight', short-range, mobile-to-mobile communications in ad-hoc networking environments. The infra-BSS, however, requires one or more APs and it is used for infrastructure networking (see Fig. 3.1). This chapter concentrates mainly on infrastructure networking.

A set of infra-BSSs can be linked together through a Distribution System (DS) to obtain an Extended Service Set (ESS) to aid roaming/mobility of STAs among BSSs. The DS is the backbone of the WLAN system and performs functions, such as bridging the APs and forwarding of packets between the APs, forwarding of packets between APs and the wired or wireless wide area network. Mobility within an ESS is transparent to all systems and stations outside of it. A functionality referred to as a *portal* interfaces an 802.11 WLAN with other system (LANs). The portal can be integrated into the AP. Table 3.1 compiles some key parameters of the IEEE 802.11 (in short 'dot11') WLAN standard.

3.3.2 Protocol Architecture

The protocol architecture of the IEEE 802.11 WLAN standard is shown in Figure 3.2. In the following, each of the layers is briefly reviewed.

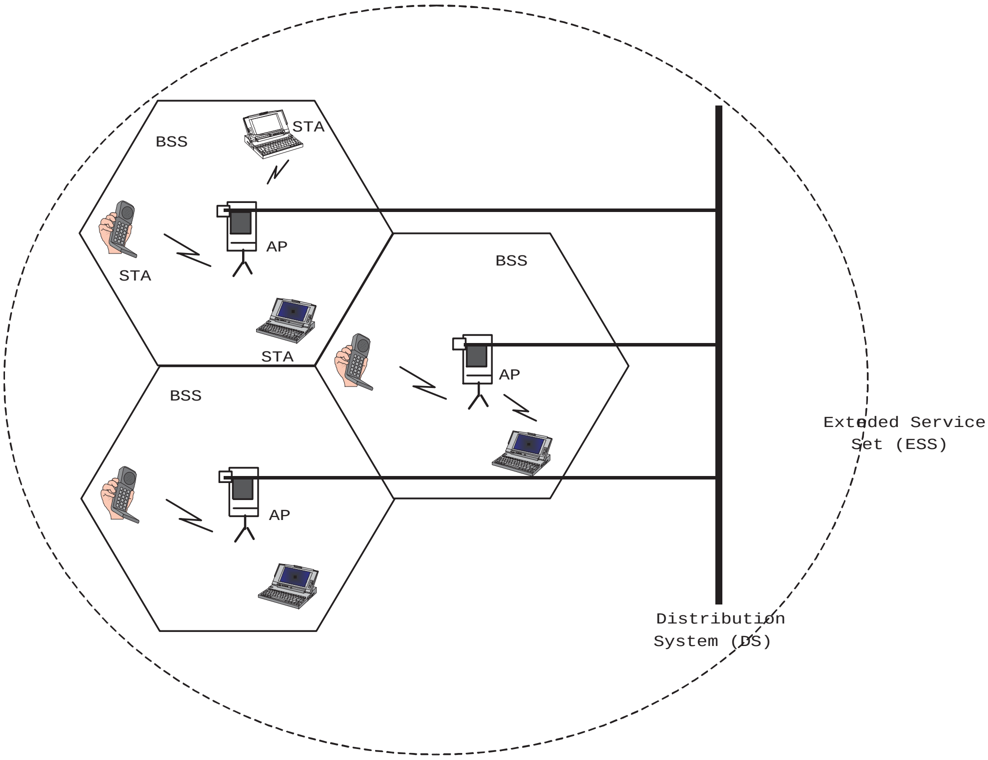


Figure 3.1: An IEEE 802.11 multi-cellular, infrastructure network architecture showing three BSS (cells).

Physical Layer

The physical layer (PHY) is the interface between the MAC sublayer and the radio medium. It checks the status of the radio transmission medium and if it senses a carrier (busy medium) notifies the MAC accordingly to enhance MAC's multiple access mechanism. The PHY modulates data frames onto RF or infrared carrier and transmit over the shared radio channel. The 802.11 standard defines three types of PHY layer: baseband, diffused infrared (IR), radio frequency (RF) with frequency hopping spread spectrum (RF-FHSS) and RF with direct sequence spread spectrum (RF-DSSS), as illustrated in Fig. 3.2 and Table 3.1.

The IR PHY operates in the wavelength range of 850–950 nm, and specifies 4-level or 16-level pulse positioning modulation (PPM) to achieve data rates of

Table 3.1: Some key features of IEEE 802.11b WLAN standard.

Feature	Physical Layer Variants		
	InfraRed	RF_1	RF_2
Spectrum	850- 950 nm	2.4- 2.483 GHz	2.4- 2.483 GHz (ISM band)
Carrier Modulation	4-level Pulse Position Modulation 16-level Pulse Position Modulation	Differential BPSK Differential QPSK	2-4 level Gaussian FSK
Spreading Modulation		High rate DSSS	FHSS
Spreading type		11-bit Barker sequence (process gain: 10.4 dB)	22 hop patterns, 20–400 ms hopping rate, 79 carriers with 1 MHz spacing
Data rates	1 Mb/s (16-PPM), 2 Mb/s (4-PPM)	1, 2, 5.5, 11, 54 Mb/s	1, 2 Mb/s
Max. EIRP	2 W	1 mW < P_t < 100 mW	10 mW < P_t < 100 mW
ACI rejection		~ 35 dB (30 MHz RF carrier spacing)	
Antenna gain	6 dBi		
Max transmit power	1 W (USA), 10 mW (Japan), 10 mW/1 MHz (Europe)		
Specification time	1990-97 (The world's first WLAN standard)		
Operational modes	Ad hoc and client/server		
Coverage	~ 10^2 m		
Handoff	Not defined, MAC-layer transparent roaming		
Layers specified	Layer 1 (Physical) and Layer 2 (MAC and LLC)		
Encryption	Open system authentication and shared key authentication		

2 Mb/s and 1 Mb/s, respectively. Both RF PHY modes are operated in the 2.4 GHz Industrial, Scientific and Medical (ISM) band. The RF-DSSS uses a 11-bit Barker spreading code to achieve a processing gain of 10.4 dB. The RF-FHSS has 22 hopping patterns and 79 RF carriers. It hops through the 79 carriers at a variable rate of 20–400 hops/ms. It has maximum antenna gain of 6 dBi. However, the transmitted powers depend on the location as shown in Table 3.1. Both the FHSS and the DSSS PHY options achieve the 1 and 2 Mb/s data rates, while the DSSS can achieve 5.5 and 11 Mb/s in the high rate mode specified for dot11b by using the complementary code keying (CCK) coding scheme, and 54

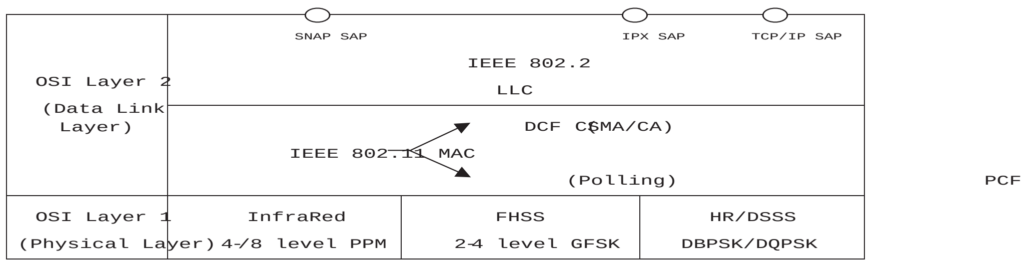


Figure 3.2: IEEE 802.11b protocol architecture showing three of the commonly used service access points (SAP): TCP/IP, IPX and SNAP.

Mb/s through OFDM at 5 GHz.

Medium Access Control Sublayer

The MAC's data units are called *frames,* and there are three types of MAC frames: short control frames (i.e. ACK, RTS, CTS), data frames, and management frames which are used within the 802.11 layers. Only unicast frames are acknowledged, however. To be compatible with other LAN technologies, such as the IEEE 802 LANs, the MAC layer has some functionalities which are commonly handled by higher OSI layers than layer 2. It handles traditionally higher layer functions such as packet fragmentation (see also Sect. 1.3.2), frame retransmission, mobility of stations which should be transparent to upper OSI layers, and acknowledgements.

For its operation, the MAC sublayer defines a couple of time intervals, notably, a SlotTime (TS), the short interframe space (SIFS), the point coordination IFS (PIFS) and the distributed IFS (DIFS), where $SIFS < PIFS < DIFS$. These timing variables are used to provide different network access and scheduling priorities to different STAs/packets. After waiting a SIFS interval only control frames (e.g., ACK and CTS) and data frames responding to polling by the point coordinator (PC) may be transmitted. In a contention-free mode controlled by the PC, frames may be transmitted after waiting for a PIFS interval. The DIFS interval allows asynchronous transmission of frames in the

DCF mode. Typical values of these parameters for the FHSS PHY layer are TS= 50 μs, SIFS=28 μs, PIFS=78 μs and DIFS=128 μs. The MAC provides three main services to the upper OSI layers above it:

- provides security through encryption and authentication
- arbitrates between transmissions of multiple stations in a BSS
- provides *best-effort,* connectionless transfer of packets over radio links.

The MAC sublayer has two modes of operation: the distributed coordination function (DCF) and the point coordination function (PCF) [52].

Point Coordination Function. The PCF is a centralized, synchronous and optional service controlled by a *point coordinator* using *polling.* Hence, it is a contention-free service. A point coordinator can be the AP or any other STA (master), which *polls* the *slave* STAs in a round-robin manner to take service turns. The PCF is not completely independent service, as it builds upon the asynchronous DCF, but it can seize the radio channel and exercise priority over any backlogged STAs while the PCF is in operation. The point coordinator does so using the PIFS. A time interval, referred to as a superframe, is defined in the standard to prevent service starvation of the STAs. The first part of the superframe is used by the point coordinator to poll STAs synchronously, while the latter part is left for asynchronous use by the distributed STAs, if any are backlogged. The network architecture designed in this chapter is based on the DCF protocol.

Distributed Coordination Function. The DCF is an asynchronous, mandatory service which is based on the carrier sense multiple access with collision avoidance (CSMA/CA) protocol. Unlike other 802 LANs, the DCF avoids collision since collision detection is impossible on a radio channel. The 802.11's CSMA/CA works as follows:

1. a backlogged STA ready to send data first checks (senses) the availability of the radio channel. If the channel is free it sets a timer to DIFS, and transmits if channel is still free upon timeout. If channel is sensed busy,

the STA backs off and initiates a *binary exponential backoff (BEB)* process. The BEB process is also executed after a collision or before every retransmission attempt.

2. binary exponential backoff process as a collision avoidance technique:
   - STA defines a contention window (CW) and a SlotTime, $t_s = 2 \cdot RTT_{max}$, RTT is the round trip time in the serving BSS
   - STA computes a random number (n_b) uniformly distributed in the range $(0, CW)$, and then the backoff delay $t_b = n_b \cdot t_s$
   - STA sets the backoff timer to t_b
   - STA decrements backoff timer by t_s each time the channel is sensed free for a time interval DIFS, otherwise it freezes the backoff timer.
   - STA transmits its frame upon backoff timeout
   - In case of collision, STA doubles CW, i.e., $CW \leftarrow 2 \cdot CW$ and restarts the backoff process until $CW = CW_{max}$. Thus, for the kth retransmission, t_b lies in the range $[0, 2^r) \cdot t_s$, $r = min(k, CW_{max})$.

3. Upon successful reception of a frame, the receiving station waits for $SIFS = DIFS - 2 \cdot t_s$, and then acknowledges the frame's reception.

If the ACK is not received in time, STA retransmits the frame until it receives an ACK from the frame receiver, or until the maximum number of retransmissions (i.e. seven) per frame reaches. Thus the sender discards a frame after the seventh unsuccessful retransmissions. The MAC layer retransmissions may be preferred to upper layer retransmissions due to reduced delays. To reduce the amount of collisions, CW is doubled until it reaches its maximum value (CWmax) any time a collision occurs. The backoff process is needed prior to each transmission attempt. To reduce the effects of the hidden terminal problem, 802.11's MAC introduces two short signalling frames: the ready to send (RTS) and the clear to send (CTS). These control frames indicate the residual time (t_r) required to transmit a corresponding frame, as well as the sending and receiving STAs identities so that other stations would backoff. Any station that detects RTS, CTS or ACK signal sets its NAV (network allocation vector) timer, and would defer any transmissions until the NAV times out. The NAV is a virtual carrier sense indicator. However, to optimize link utilization, the RTS and CTS frames are enabled only if the data frame to be transmitted has a length exceeding a given but variable threshold, *RTS threshold.*

Logical Link Control Sublayer

The IEEE 802.11 logical link control (LLC) sublayer is based on the IEEE 802.2 LLC. It interfaces the MAC and the Layer 3, which, in this chapter, is assumed to be IP/Mobile IPv6. The LLC's data units are called PDU (protocol data units). The LLC has three types of services: Type 1 through Type 3. The Type 1 service is a best-effort-like connectionless service without acknowledgment, flow control, or error control. It avoids possible duplication of functions many OSI higher layers provide on the link level.

The Type 2 LLC service, however, is an end-to-end, connection-oriented service, and it provides flow and error control using an ARQ scheme. The connection can be broadcast, multicast or unicast (point-to-point, or P2P). Although the third service is connectionless, the receiving STA should acknowledge a PDU it receives correctly. Here, we refer to these services as CU (connectionless unacknowledged), CA (connectionless acknowledged) and CO (connection-oriented), respectively.

3.4 IEEE 802.11/Mobile IP Internetworking

This section describes the features of the 802.11/Internet interworking architecture we are proposing to support 3G applications. In this architecture, each 802.11 STA is assigned two addresses: an IP address for identification and routing in the Internet, and an IPUI (or TPUI) for use within the 802.11 WLAN. Each STA uses the home DSSP as its default IP router. The STA and QDS run the MIP software and the DiffServ architecture. For data transmission, we adopt both the connection-oriented and connectionless modes of the LLC. All the protocols adopted in the architecture are open standards; hence they

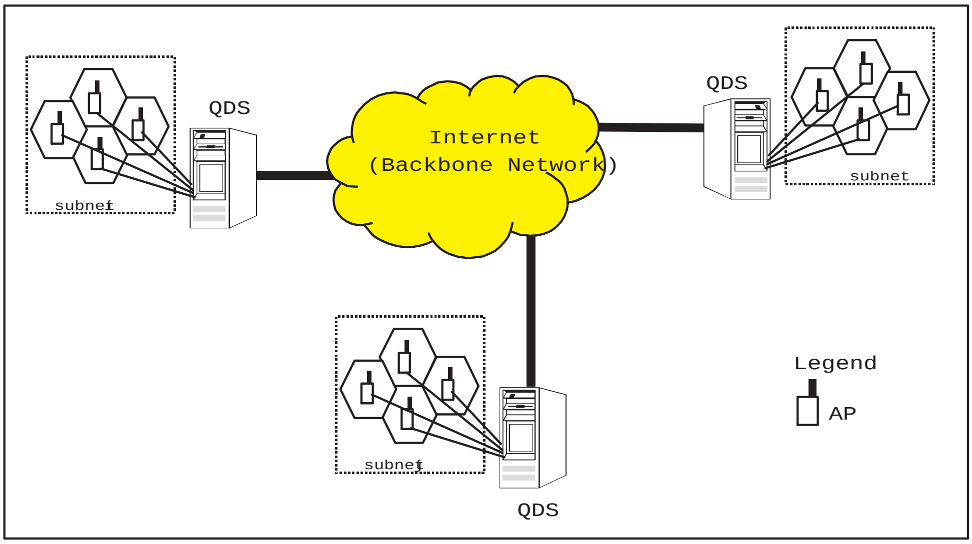


Figure 3.3: System architecture of the IEEE 802.11/Mobile IP interworking showing three ESSs as IP subnets, QDS is the QoS enabled distribution system.

are off-the-shelf and less costly due to economies of scale. The reference architecture for the IEEE 802.11/Mobile IP internetworking is shown in Fig. 3.3. The wireless Internet access architecture described in this chapter is based on infrastructure IEEE 802.11 standard, which is denoted as dot11.

3.4.1 General Descriptions

The overall system architecture is shown in Fig. 3.3. In the architecture, the Internet backbone represents both the traditional Internet itself and future generations of IPv6 based Next Generation Internet (NGI), since the latter will eventually be widely deployed. Basically, the proposed architecture comprises dot11 based multicellular subnets interworking with the Internet via the QDS. In this architecture, subnets managed by different QDSs can be under different administrative domains, or belong to a multisite corporation. In the latter case firewalls can be used to create Intranets based on the multi-QDS subnets with reduced costs. Thus each subnet can function as an autonomous wireless LAN with one or more paging areas. Routing of packets between different subnets is achieved via IP and MIP mechanisms since it is beyond IEEE 802.11 capability and scope.

When a STA initiates a session from a cell in a subnet, the request is routed over dot11 air interface to the serving AP, which forwards it to the DS integrated in the QDS. Upon receiving the connection request, the DS checks the correspondent station's (CN) address in the packet header, starting from the dot11 identity. If the CN is in the same subnet as the MN (i.e. internal session) the packet is handled within the confines of the DS using only dot11 identities. Alternatively, if the MN and the CN are in different subnets (i.e. external session), the DS passes the packets to the IWU/portal, which with the QoS module (QoSM) maps the packet to the corresponding IntServ class or DiffServ PHB. This is done to maintain corresponding QoS for the packet streams. Upon table lookup, the routing module forwards the datagram to the appropriate IP gateway for further switching. This requires each QDS to map IP addresses of STAs to dot11 identities for correct packet forwarding. It is thus crucial to refresh the address resolution table (ART) in the packet forwarding module (PFM) upon each link transfer (handoff/re-association), registration (association) and deregistration (disassociation).

An external session can be a delay-sensitive packetized voice session or a data session. It is the task of the PFM and QoSM module to classify and map the traffic to the corresponding QoS class: either as guaranteed service (or EF PHB) traffic (if VoIP traffic), or controlled-load (or AF PHB), if data session. Within the data session, the PFM finely classifies the traffic into background, interactive, streaming and conversational types. Packetised voice (VoIP) traffic, for example, can be classified as a conversational data session, while file transfer is classified as background session. By this means traditional connection-oriented services are preserved, while, at the same time, differentiated QoS classes are available to IP data traffic.

3.4.2 The QoS Enabled Distribution System (QDS)

The QoS enabled distribution system (QDS), an extended dot11's distribution system (DS), is the heart of the proposed architecture and it functions as an *Internet portal* (access router, AR). The term QDS coined to reflect the fact that it is an enhancement of the 802.11's DS with features, such as QoS via IntServ/MRSVP or Diffserv and location databases for mobility handling. The home network is the QDS subnet a user subscribes to for service. As illustrated in Fig. 3.4, the QDS contains functional entities including:

- IEEE 802.11's Distribution System (DS),
- Interworking Unit (IWU) or portal with IP routing functions,
- Mobile IP Mobility Agent (MA),
- Location Databases (DB),
- Packet Forwarding Module (PFM), and
- QoS Manager/Module (QoSM), either Diffserv or Intserv/MRSVP.

The Distribution System is the usual 802.11's DS, as discussed above. The QoSM, which can be based on IntServ/MRSVP or DiffServ architecture, is discussed in Section 3.4.4.

Interworking Unit/Portal

The interworking function (IWF) is the protocol running in the interworking unit (IWU). It allows 802.11 the flexibility to support a variety of applications and interworks with a variety of local-area and wide-area backbone networks (BBN). On the backbone network/802.11 interface side, the IWU translates between 802.11 and backbone network protocols and signalling, and provides IP routing functions. On the application side, the IWU adapts the applications characteristics to the radio quality. In the data transmission plane, the IWF reduces to packet relaying and the application comes directly over the LLC sublayer since the NWK is passive in this plane. In this architecture, the portal's functionalities, as defined in dot11 standard, are embedded in the IWFs.

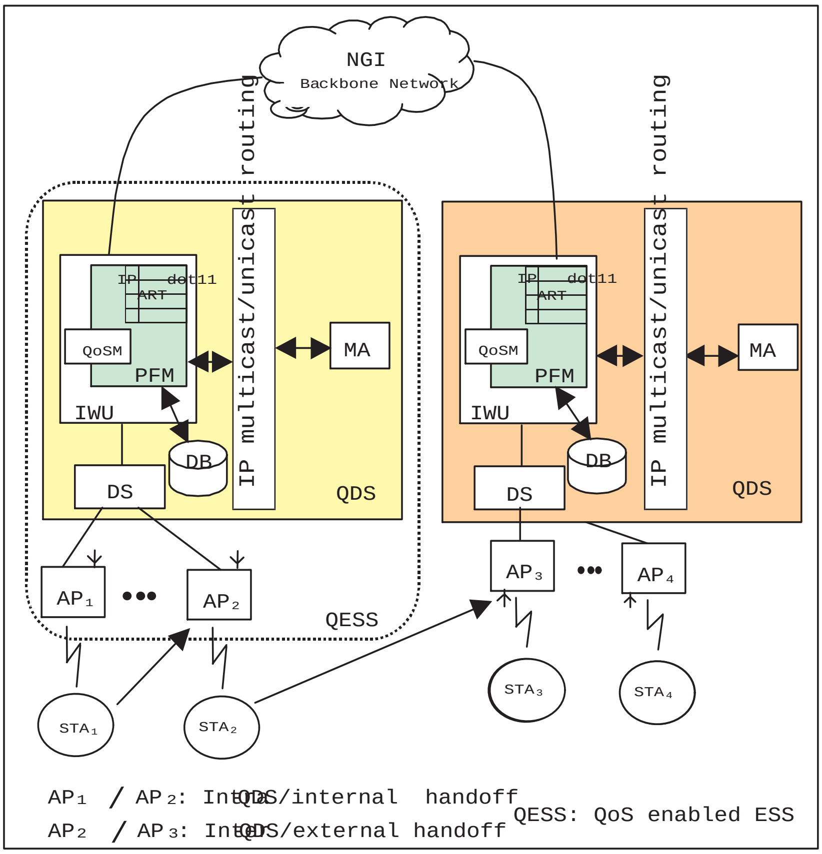


Figure 3.4: Key functional elements of the QDS and handoff scenarios.

Mobility Agent

The MA encompasses the functionalities of the Mobile IP home agent (HA) and foreign agent (FA). These Mobile IP agents cooperate to provide network connectivity and routing functions to roaming portable parts as described in [85, 86].

Location Registers/Databases

The DB contains the location directories needed for managing mobility and comprises the home database (HD) and visitor database (VD). The VD contains the transient data for visiting terminals while the HD contains the static data of home network service subscribers. Together, the HD and VD administer users' subscription profiles and identity. They store associations between CoA and home addresses of STAs. A user profile includes IPUI, IP address and subscription features like mobility scope (intra-802.11 or inter-802.11 roaming support) and QoS parameters. Also, the service activation data needed for authentication, billing and service management are kept in the DB.

Packet Forwarding Module

The PFM comprises a packet classifier and a scheduler. The classifier maps the packets into QoS classes while the scheduler schedules the data packets to the right output interface. The PFM requires an address resolution table (ART) to translate IP addresses to/fro 802.11 identities to service external sessions or handoffs. The traffic scheduler has one input interface but many outbound interfaces: one for each QoS-dependent traffic type. The traffic can be background, interactive, streaming and conversational, according to their respective QoS requirements. By so doing the Internet (IP) traffic is given respective differentiated QoS.

3.4.3 Protocol Architecture

The protocol stack of the three key components of a MOWINTA network is as shown in Fig. 3.5. The 802.11 protocol stack is enhanced with TCP/IP. The TCP/IP/MIP protocol suite is adopted to provide standard-conforming, efficient mobile computing. Nothing in the 802.11 standard is changed, except the Mobile IP control messages needed for mobility management that is buffered and imbedded (piggybacked) in 802.11 signalling, as described above. For instance, the MAC's DCF or PCF manage multiple cells of a subnet (ESS), whilst an AP manages a single cell in a subnet. Since the 802.11 protocol architecture is discussed above, we do not repeat it here.

3.4.4 IEEE 802.11/IP QoS Mapping via IntServ

The current Internet and its protocols provide the same single "flat" best-effort service, characterized with variable queuing delays and packet losses during

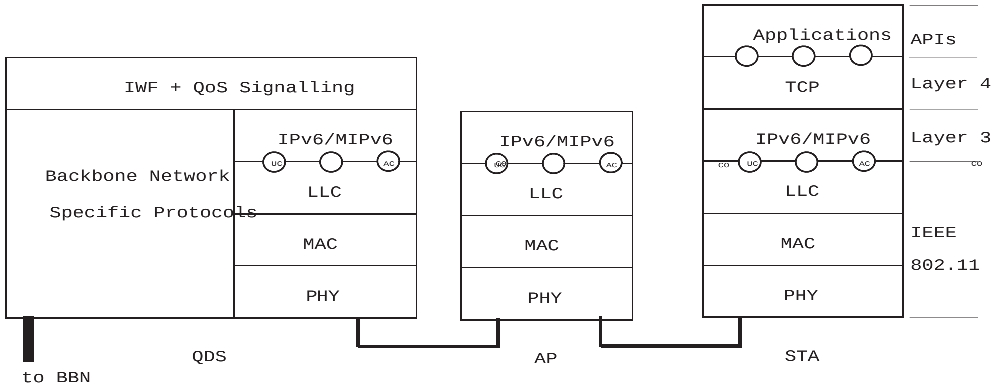


Figure 3.5: Protocol architecture of QDS, AP and STA illustrating the service access points of the three LLC services, viz. UC, AC and CO, and three application programming interfaces (APIs).

network congestion, to all applications [69]. The lack of guaranteed QoS and explicit traffic admission control in the Internet is no more acceptable for emerging real-time multimedia applications. Thus the initiatives towards the development of technologies such as the Differentiated Services (DiffServ) [10], the Integrated Services (IntServ) [13], IP over ATM [103], and the Resource ReSerVation Protocol (RSVP) [14] and its enhancement, Mobile RSVP (MRSVP) [100], to equip IP with the signalling needed for QoS provisioning. In the architecture that is presented in this chapter, we propose running the IntServ and MRSVP protocols on the QDSs and the mobile hosts to support QoS signalling in the Internet. This is needed to maintain QoS offered by 802.11 services for *external* communication between two terminals bridged by the backbone Internet.

IntServ: As mentioned above, the IntServ architecture is designed to use explicit signalling to support QoS in IP networks. In addition to the best-effort service suitable for background applications, it provides two QoS classes as discussed in Sect. 2.5.1. They are controlled-load (CL) or delay and the guaranteed-delay (GS) services. The GS service offers firm end-to-end delay bound with no queuing loss for contract-adherent packets of a flow and strict bandwidth, which is suitable for real-time conversational and near-real-time streaming applications. The CL service, on the other hand, offers several QoS levels from which an interactive application requiring a minimum QoS can adapt. Basically, the IntServ architecture comprises a signalling protocol, an admission controller, a packet classifier and scheduler, and a flow specification [13, 69].

The flow specification contains parameters indicating the resources needed to satisfy an application's QoS requirements. The admission controller ensures QoS requirements of both the new and existing calls before accepting the new call request. The signalling protocol signals an application's QoS requirements in form of the flow specification to the network routers/switches. The packet classifier parses the header of each packet in a flow in order to map it to the right QoS class. And, finally, the scheduler ensures that packets accepted into the network get their required QoS, and it grants the same QoS to packets of the same QoS class.

RSVP: As noted above, to provide QoS in the traditional best-effort IP based Internet, IntServ requires a signalling protocol. The signalling protocol is needed to create flow-specific path state and reserve resources (e.g. link bandwidth, buffer space, CPU processing time) in the portables, QDS and backbone network routers along the end-to-end routing path set up by the routing protocol prior to data transmission. RSVP is a control protocol that usually runs at the transport layer. With RSVP, unidirectional resource reservation is initiated by the data receiver(s) hence permits receivers to support different QoS in case of data multicasting. Messages of RSVP can be transported in normal IP datagrams by using the protocol number 46. Key messages used by RSVP to accomplish its tasks include PATH, RESV, PathTear, ResvTear, ResvErr, and PathErr.

Basically, the PATH message, from packet sender to receiver(s), informs the receiver(s) about the traffic characteristics of the sender to enable the receiver(s) to reserve appropriate resources for the packets with the RESV message. If any errors are detected in the PATH message by an RSVP-capable router, the router sends a PathErr message to the sender and deletes the corresponding PATH message. The PathTear message is used to explicitly delete a path state by releasing reserved resources and path entries in the routers along the corresponding path. PathTear is used in addition to the automatic timeout of path state after the default time of 157.5 s [14] (i.e. soft state mechanism in RSVP). A router sends the ResvErr message downstream upon denying a resource reservation request. To explicitly tear down a resource reservation state, a router sends the ResvTear message upstream to the source.

MRSVP: Signalling delay to re-setup the RSVP reservation path after a change of network POA has been shown to be significant [100] leading to bad performance when RSVP is used with MIP. Hence, the need for Mobile RSVP (MRSVP), an extended RSVP. The key differences between MSRVP and RSVP are summarized in Table 2.3. MRSVP attempts to maintain QoS to mobile hosts by reserving resources in advance in potential locations the mobile may roam to during active sessions. MRSVP creates two types of reservations called active and passive reservations. Active reservations serve active sessions while passive reservations are meant for possible future usage. In our architecture, MSRVP is adopted to provide signalling in IntServ. For detailed presentation of MRSVP see e.g. [100] MRSVP is adopted in this chapter since it tailors RSVP to mobile environments and RSVP appears a natural choice to provide signalling in IntServ architecture. The choice of IntServ for the IP signalling is based on the following reasons:

- IntServ service classes can be mapped onto IEEE 802.11 protocol.
- IntServ is a well-known, mature technology.
- IntServ requires only minor changes to the ubiquitous best-effort service and its applications (i.e. subdivision of best-effort service into interactive burst (e.g. Telnet), interactive bulk (e.g. FTP) and asynchronous bulk transfer (e.g., e-mail and fax).

IntServ scalability and manageability problems due to per-flow QoS guarantee may be overcome with MRSVP, or by aggregating flows when reserving resources, or by integrating DiffServ into the architecture.

As shown in Table 3.2, we map the three IntServ service classes onto the 802.11 logical link layer (LLC) services as follows. The LLC's unacknowledged

Table 3.2: Mapping IEEE 802.11 services onto IntServ/DiffServ QoS model.

IEEE 802.11 LLC Service	**IntServ QoS Class**	**DS PHB**	**3G Traffic Class**
Connection-oriented (CO)	Guaranteed delay (GS)	EF	Conversational
Connection-oriented (CO)	Guaranteed delay (GS)	EF	Streaming
Acknowledged connectionless (AC)	Controlled load (CL)	AF	Interactive
Unacknowledged connectionless (UC)	Best effort (BE)	BEF	Background

connectionless (UC) service is mapped onto the IntServ best-effort service offering no service guarantee to support background traffic. The LLC's acknowledged connectionless (AC) service is mapped onto the IntServ controlled-load service class to support interactive traffic. Lastly, the LLC's connection-oriented (CO) service is mapped onto the IntServ guaranteed-delay service class to support conversational and streaming traffic. Table 3.2 also shows a possible mapping of IntServ and the LLC services into DiffServ. There are three potential scenarios of integrating DiffServ (DS) with IntServ (IS), namely

- Run both DS and IS on all routers, but use IS for mission-critical traffic such as IP telephony and videoconferencing, and DS for less time-sensitive traffic,
- Run both DS and IS on all routers, but activate only IS at low traffic loads and DS at congested times,
- Run only DS on core routers, while edge routers are both IS and DS enabled.

The DiffServ mapping, however, is not expanded in this chapter and it is seen as a future research.

3.4.5 Session Setup and Frame Transmission

Figure 3.9 illustrates the signalling flow sequence and data transmission for an inter-subnet session based on the connection-oriented LLC service. In this figure, it is assumed that both communicating stations remain in their home subnets during the entire session. The dot11 standard has five distribution services: association, disassociation, re-association, distribution and integration.

A STA willing to exchange frames with another STA in an *infrastructure* dot11 network needs to associate (i.e., establish a logical connection) to an AP and synchronize its timer (clock) with that of the AP. STA/AP synchronization is needed for power management in sleep/wake mode, as well as for RF carrier hopping pattern in the case of FHSS PHY. There are two basic ways that an STA can synchronize with an AP: active and passive scanning. In active scanning, the STA solicits timing information from the AP by sending a probing signal. In passive scanning, however, the STA awaits the next periodic MAC layer's beacon transmissions, which also contains system (AP) identities and capabilities. Due to battery savings, passive scanning appears attractive. The beacon signal is part of MAC layer's frame control bits in its header. It is encoded as 001000.

Upon beacon detection, the STA authenticates with the AP (see Fig. 3.9) and then exchange identities and capabilities (i.e., association phase). Following the association, the STA sends the RTS control frame to the corresponding node, who responds with CTS. The frame and ACK transmission phase comes next. After all its frames are successfully transmitted, the STA disassociates (i.e., informs the AP of not requiring its service any more) from the AP.

3.5 Mobility Management

In general, mobility management (MM) [1] includes location management (LM), handoff management (HM), authentication and identification, access rights checking, and encryption. MM in MOWINTA is supported by both dot11 MAC layer and at IP layer. IEEE 802.11's functionalities for MM include active/passive scanning for AP's beacon, authentication, association, re-association and disassociation processes. Handoff management mechanisms are triggered by either wireless link deterioration or user mobility. User mobility always results in a change of AP, but changing only a radio bearer (intra-cell handoff) can solve a link deterioration problem.

As noted above, intra-subnet roaming (micromobility) is supported by inherent dot11 internal intra- and inter-cell handoff mechanisms, while macromobility is supported with IP/mobile IP. IEEE 802.11 beacon broadcast is very crucial for the MM. STAs detect their movement in 802.11 by a change of access port identity in the beacon since cells belonging to the same subnet have the same subnet IP subnet address, but unique link layer identities. An MIP node usually detects a movement via expiration of agent advertisement lifetime, or prefix matching [85] or modifications thereof.

Movement detection invokes either only 802.11 handoff or both dot11and Mobile IP handoff, depending on whether or not there is a change of subnet (external handoff). And agent advertisement is not necessary if there is no external handoff since no MIP handoff mechanism is required. As noted above, 802.11movement detection is faster than MIP due to longer inter-advertisement times and that MIP agent advertisement messages are relatively large. This motivates embedding the latest MIP agent advertisement in a dot11 beacon transmissions (see below). This approach saves bandwidth due to reduced signalling overhead over the ether and reduces handoff latency.

3.5.1 Handoff Management

Handoff management (HM) mechanisms aid the network to maintain connections with mobile terminals as they change their point of attachment to the network/AP. HM consists of handoff initiation, new connection generation and data flow control [1]. Prior to handoff initiation, user movement must be detected and network conditions be monitored. New connection generation involves resource allocation and connection routing. Data flow control utilizes buffering /sequencing and multicasting [1]. In the following, we discuss the internal and external handoff mechanisms, which are controlled by the mobile terminal.

Micromobility

Micromobility or internal handoff is handled by a single QDS. This can be intra-cell handoff (only link layer is changed) or inter-cell handoff (change of AP but

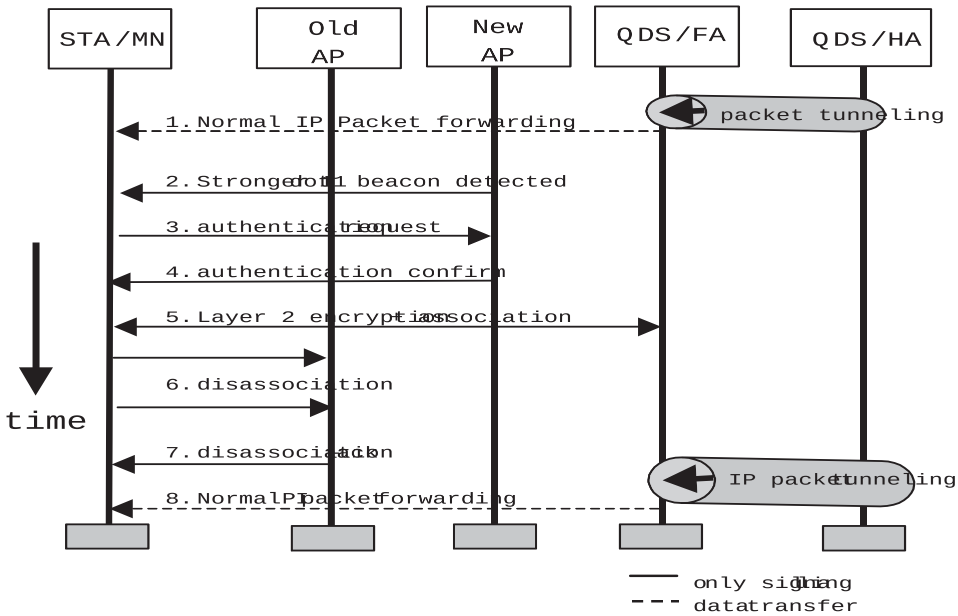


Figure 3.6: Intra-subnet handoff message flow sequence (no L3 handoff). Signalling propagation delays are not considered.

managed by the same QDS). It involves only layer 2 processing and the old IP routing path is left intact. Thus this type of handoff does not involve the network layer and hence completely handled by inherent dot11 micromobility management functions. It maintains the network layer (signalling) association, but changes either the MAC bearer (resulting in bearer handoff) or the MAC connection (resulting in connection handoff) or both. Handoff between AP_1 and AP_2 in Fig. 3.4 is an example of inter-cell internal handoff. Figure 10 illustrates the simplest intra-cell internal handoff requiring only a bearer change. The signalling flows as follows.

In step 1, we have the normal IP forwarding. The STA detects a stronger MAC beacon, which tells the STA all the information it needs to gain access to the target AP in step 2. The STA then initiates authentication in step 3, which the AP responses in step 4. Then follows association and radio bearer encryption in step 5. In steps 6 and 7, the STA disassociates itself with the former AP before resuming frame transmission through the new AP in the last step, step 8. In Fig. 3.6, it is assumed that the MN initiates the handoff after detecting a periodic beacon of the target access port. The MN can also solicits an AP's identity instead of having to await a beacon if , for example, its signal strength (or link quality) falls below a threshold. In which case step 2 in Fig. 3.6 would be replaced by a *probe request* from MN to the AP and a *probe response* in the other direction.

Macromobility

An example of external (inter-subnet) handoff is the handoff between AP_1 and AP_3 in Fig. 3.4. Macromobility or external handoff disrupts the existing IP routing path, thus requiring both re-establishment of the wireless link layer connection in dot11 (i.e. L2 handoff) and Mobile IP care-of address updating (i.e. L3 handoff). Thus, external handoff is outside dot11's scope. It uses the mobility management functions of dot11 MAC layer as well as the (Mobile) IP functions. Thus requiring IP and dot11 address translation.

A signalling flow diagram for an external handoff is illustrated in Fig. 3.7. For simplicity, signalling propagation delays are not shown in the figure, and MRSVP/RSVP messages are also not shown (see a similar signalling in e.g., [91]). The dot11 LLC layer CO service is used in this diagram. In step 1, we have normal HA to FA Mobile IP tunneling and FA to STA packet forwarding. In step 2, the STA detects the beacon broadcast of a new AP having a better signal quality, and requests authentication with the new AP in steps 3 and 4. Upon receiving authentication confirmation from the new AP in step 4, the STA requests a dot11 handoff (re-association) to the target QDS in step 5. The new QDS piggybacks the latest MIP agent advertisement in its memory on the re-association message to the STA in step 6. Ciphering takes place in step 7.

Buffering and embedding Mobile IP control messages in dot11 MM messages in this way allows the STA access to the information required to initiate Mobile IP handoff. Consequently, the STA does not need to await the next Mobile IP agent advertisement before initiating Mobile IP handoff, hence reducing handoff latency. Hence, the overall handoff delay is limited by that of dot11, which is shorter than handoff delay of Mobile IP. This hides the handoff mechanism from

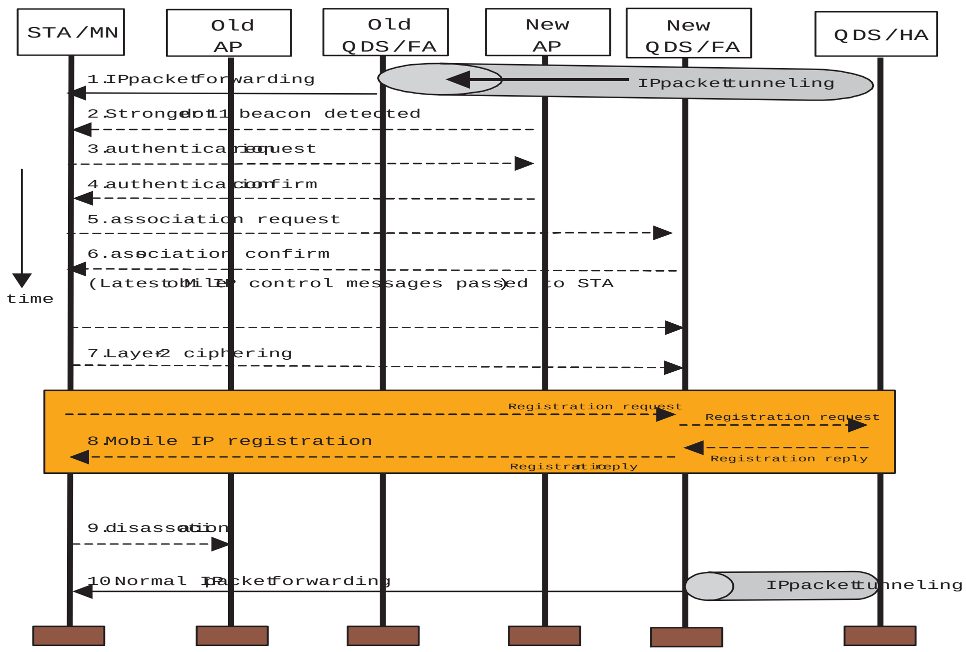


Figure 3.7: External handoff message flow sequence with MIP agent advertisement embedded in 802.11 signalling.

TCP without modifying TCP or Mobile IP, and at the same time circumvents TCP throttling mechanisms, which are damaging to throughput.

Having the necessary Mobile IP control messages the STA updates its mobility binding with the new foreign agent/DSSP in step 8. Upon successful mobility binding update, the STA releases its association with the old AP in step 9. Upon successful handoff completion, IP packet transmission resumes in step 10. It is noted in all the mobility management figures that the handoff is seamless, i.e., the new link layer bearer and connection are set up before the old ones are released. This reduces packet losses during handoff and corresponding improvement in TCP degradation during handoff. Buffering and releasing the latest Mobile IP agent advertisement to the STA for network level mobility binding update immediately dot11 handoff completes also reduces TCP performance degradation.

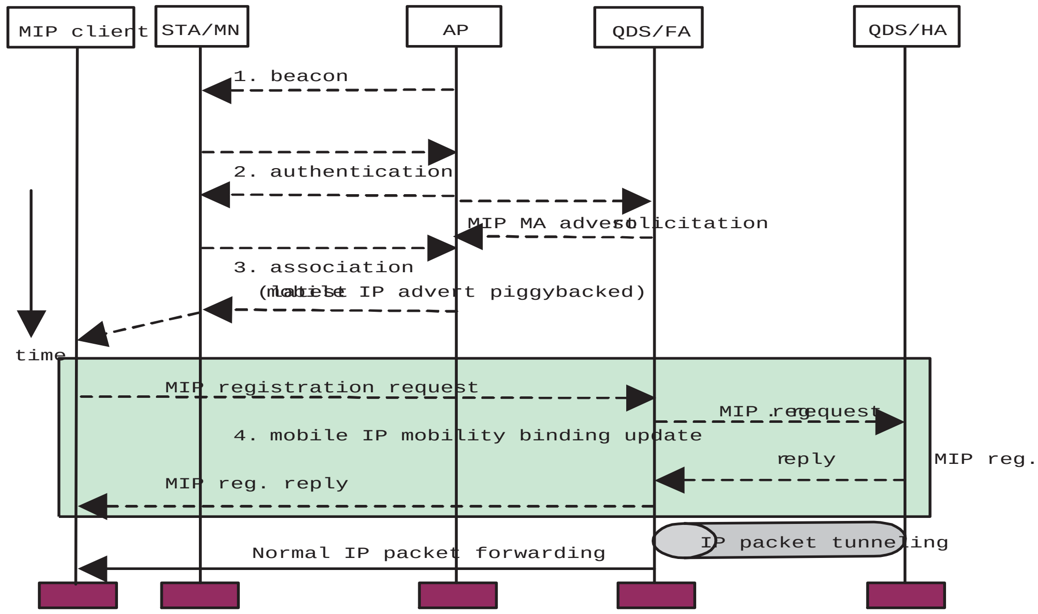


Figure 3.8: Location registration signalling flow with mobile IP agent advertisement message embedded in 802.11 signalling. (Signalling propagation delays are assumed negligible.

3.5.2 Location Management

In general, location management (LM) mechanisms locate roaming mobiles for call delivery [1]. LM in dot11 embodies beacon detection, authentication and association. A STA indicates to the network its availability or unavailability to receive an incoming call using the association procedures, as it cannot communicate without being attached to an AP. Location registration is initiated by the STA upon detecting change in location via the strongest beacon signal. To update the location of a STA in the database (VLR/HLR), the network/AP has to authenticate the STA. Prior to call delivery to the STA, the network has to query the location management databases for exact location area of the STA before paging it to reduce paging area and save limited radio resources.

Figure 3.8 depicts a timing diagram for both dot11 and MIP registration, and IP packets delivery. As usual, dot11 registration precedes that of MIP. As noted above, we have 'embedded' the MIP agent advertisement messages in dot11

¡association-ack¿ messages to save bandwidth and reduce registration delay. Reduction in registration delay reduces overall handoff delay, which improves TCP throughput degradation. The location updating signalling flows as follows.

In step 1, the STA detects the beacon broadcast messages from the strongest AP and requests authentication and association with the AP in steps 2 and 3, respectively. In between the authentication and association process, the serving AP queries the serving QDS for the latest mobile IP agent advertisement and pass it on to the STA as soon as the association process completes. Thus, in step 4, mobile IP location updating occurs without the need to await the next MIP agent advertisement, reducing the overall dot11/Mobile IP handoff remarkably by at least factor 5.

3.6 Summary

A network architecture is proposed which permits users to roam without changing the home IP address. Since the architecture is based on IEEE 802.11 radio interface, a data rate in excess of 11 Mb/s can be available to an application. Thus, except in highly vehicular environments, the proposed architecture is capable of supporting some 3G applications. With continuing advancement in image and video coding technologies, most applications may be supportable in future by a 11 Mb/s interface.

Since 802.11 operates in the unlicensed ISM radio spectrum, it is not necessary to acquire deep pockets for RF spectrum auctions. Moreover, no time consuming frequency planning is needed, especially in ad hoc networking, which reduces costs and time-to-market. Other key strengths of 802.11 include high data rate and proven standard, which is supported by the IEEE and many industries. Together these make the proposed architecture technically and economically feasible.

If scalability and manageability of the IntServ architecture due to its per-flow QoS provisioning become significant issues when the population of network users grows (even with MRSVP), aggregating multiple flows or incorporating DiffServ can be considered. This is something that requires further study. Furthermore, the fact that 802.11 is a WLAN technology may require more handoffs of active connections depending on user mobility scenarios, which, if not seamless, will disturb communications. Thus, a testbed may be needed to examine the extent to which 802.11 signalling can be exploited to make the overall (802.11 + Mobile IP) handoffs seamless in order to prevent TCP throttling mechanisms as discussed above.

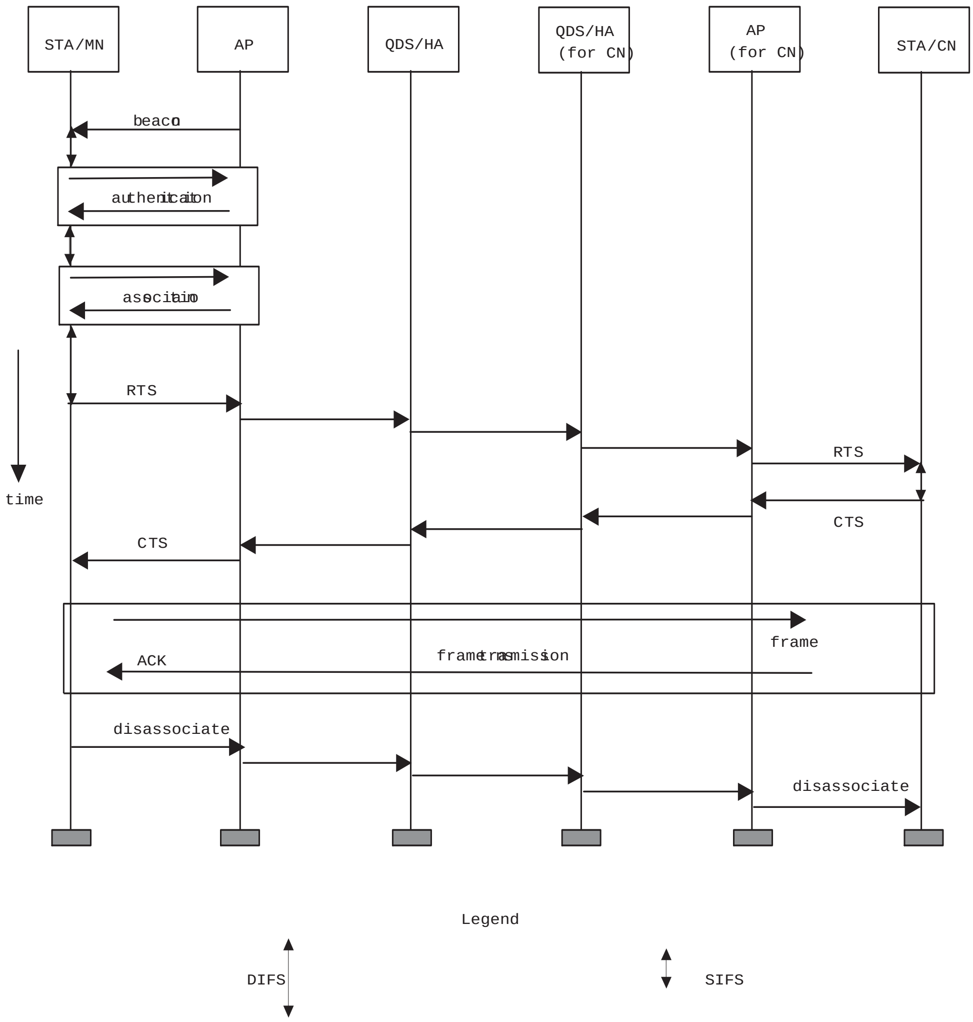


Figure 3.9: Beacon detection, association, data transmission and disassociation signalling flow sequence for an inter-subnet session based on the connection-oriented LLC service. It is assumed both communicating stations remain in their home subnets during the entire session.

Chapter 4

Schemes for Differentiated QoS Provisioning in Packetized Wireless Networks

4.1 Introduction

This chapter presents some simple and yet efficient packet scheduling and buffer management schemes to provide differentiated QoS (DQoS) in wireless IP networks. With the convergence of wireless mobility and Internet Protocol based content into mobile wireless Internet (MOWINT), mobile and base stations become Internet portals and hence need to function as wireless routers (WR), either fixed (FWR) or mobile (MWR). MWR functionality is more critical in infrastructure-less (i.e. ad hoc) networking environments. MOWINT systems are poised to be multiservice communications networks to support multimedia traffic with different characteristics, and hence different service requirements.

As MOWINT systems rely on radio transmissions, the user-network interface is dynamically changing, both in time and space (i.e., spatio-temporal) due to mobility, noise and interference, etc. This spatio-temporal channel property can be exploited to optimise the spectral efficiency. As the ether is getting congested or even scarce[1], radio spectrum has become an expensive commodity in some

[1]Due to scarcity of RF spectrum, in some geographical regions (notably US) of the world

geographical regions of the world, notably UK and France. Unlike fixed links, the issue of wireless capacity optimisation is not driven by only radio frequency spectrum acquisition costs, but, above all, on its availability. Hence, packet-based differentiated QoS (DQoS) schemes optimising radio resource usage while considering traffic QoS requirements are very valuable. DQoS is crucial to the successful uptake of next generation multiservice wireless networks poised to support multimedia traffic with diverse characteristics.

Many scheduling schemes proposed in the literature for wireless networks are user-centric in that the MAC (medium access control) layer schedules traffic based on user's instantaneous wireless link condition without differentiating the QoS requirements of the user's traffic. Thus a user with QoS-sensitive traffic but in poorer radio environment tends to receive poorer QoS all the time than a user with best-effort traffic but in a better radio environment. Other scheduling schemes are applications' centric, as they focus only on the applications QoS without optimizing the usage of the scarce radio resources. As will be evident shortly, the BL^{4}DF scheme is a hybrid scheme.

Next, we present the BL^{4}DF packet scheduling in Section 4.2 and its mathematical analysis in Section 4.3. A special form of the BL^{4}DF scheme, namely, the BL^{2}DF scheduler focusing on timing, is discussed in Section 4.4. Another form of the BL^{4}DF scheme embedding a business model into the QoS scheme is presented in Section 4.5. Section 4.6 presents the Non-Pre-emptive Priority with Partial Assurance (NP^{3}A) scheduler. The wireless Generalized Processor Sharing (WGPS) and the Channel State Dependent rate Proportional Scheduling (CSDPS) schemes are discussed in Sections 4.7 and 4.8, respectively. Unlike the previous sections, Section 4.9 presents a buffer management scheme called Proportional Buffer Sharing with Pushout (PBSP). The chapter concludes with a summary in Section 4.10.

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some systems are forced to release some or even entire frequency spectrum for the usage of another, perhaps newer, system/technology.

4.2 BL^{4}DF—Best Link Lowest Loss Lowest Delay First Scheduler

In this section, we will discuss an efficient packet based scheduling scheme—the best link, lowest loss, lowest delay first (BL^{4}DF) scheme—which accounts for the QoS requirements of user traffic via DQoS, and exploits the spatio-temporal wireless link state of the mobile user in a cell of a wireless cellular system. The BL^{4}DF scheduling is a *framework* for a new class of wireless channel state dependent packet schedulers to provide QoS in packetized wireless networks. The idea of exploiting the wireless channel spatio-temporality to optimise the spectral efficiency and mobile battery power has also been explored in [3, 8].

Under the BL^{4}DF scheme, applications traffic is categorized into QoS classes; and at each scheduling instant a packet of the highest backlogged QoS class destined to the mobile with the best radio link quality, and hence data rate, is scheduled. Thus, the *scheduling algorithm* can be formulated as an *inequality-constrained optimisation problem,* in which an objective (or cost) function is to be optimised subjected to some constraints. The applications QoS requirements are embedded in the QoS constraints. The scheme is applicable to multiservice mobile wireless networks such as Qualcomm's HDR (High Data Rate) or other CDMA systems.

4.2.1 Problem Formulation

Assume that there are M active MWRs (indexed by m) in a cell of a cellular network, and that each of the M MWRs can have one or more different applications active simultaneously. Here, the focus is laid on the forward link (i.e., FWR to MWR direction). The FWR schedules a packet to only one of the M active MWRs in each scheduling interval (or time slot). Assume that the applications traffic is categorized into c traffic classes (TC) (indexed by i) based on QoS requirements. The criteria for mapping traffic onto QoS classes can be based on technical issues, such as traffic characteristics, or on business issues, such as tariff, or some other non-technical policies. Traffic belonging to the same QoS TC but to be scheduled to different MWRs is queued in the

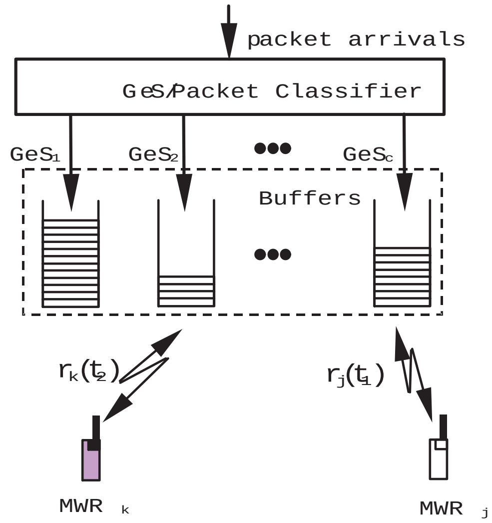

Figure 4.1: A simple queuing situation at a cell with two active MWRs

same logical queue, as shown in Fig. 4.3. Figure 4.3 illustrates an example of a queuing situation in the downlink direction at a cell with two active MWRs (k and j) each with one active application belonging to different QoS generic streams (or traffic classes), GeS_2 and GeS_c, respectively.

In the example shown, a packet of TC_c is scheduled to MWR_j at a scheduling instant t_1, while at another instant t_2, a packet of TC_2 is scheduled to MWR_k. To improve link utilization, the FWR is allowed to do some buffering even if a flow controlled transport protocol such as TCP (Transmission Control Protocol) is in use. This class-centric queuing ensures that each QoS class is guaranteed the appropriate relative QoS. However, the scheduling is not based on only the traffic QoS class, but also on the instantaneous wireless link state (fading level) of each of the MWRs. The latter constraint ensures that the scare wireless link bandwidth is used efficiently, as the wireless link state and hence the useful data rate changes randomly and asynchronously for different users [3]. Before proceeding to the details of the BL^4DF scheduling algorithm, let us note the following definitions.

Definition 4.1 **A backlogged traffic class** i is a traffic class with at least a single packet in its queue at the beginning of a scheduling period.

Definition 4.2 **Scheduler busy period** is the maximum interval of time during which it is continuously busy scheduling packets.

Definition 4.3 **Continuously differentiable functions** C^n : If $D^n y : R_1 \to R_2$ is a continuous function, then y is an n-th order continuously differentiable function on R_1, i.e., $y \in C^n$, where R_1 and R_2 are some given sets.

Definition 4.4 **Feasible solution** is a point in the domain of a function which fulfils a given set of constraints. The set of all such points is called feasible set.

4.2.2 Intuitive Illustration of Algorithmic Performance

For illustrative reasons, assume that there are two active mobiles (or MWR) at a cell of a mobile cellular network served by a single FWR. Also, for simplicity of illustration, assume that each of the two mobiles has a single active traffic, and that the FWR schedules traffic to only one of the mobiles at a time. Denote the traffic to be scheduled to MWR_1 by TC_1, and that to MWR_2 by TC_2. Assume that both TC_1 and TC_2 are backlogged for the entire period of time under study, and that TC_1 requires better QoS than TC_2. Thus, a strict priority based scheduler would never schedule TC_2 to MWR_2 so long as both are backlogged, although MWR_2 has the best fading situation (i.e., highest data rate) at the time instants t_1, t_2 and t_3. However, at time instant t_4, although $r_2(t_4) > r_1(t_4)$, TC_1 is scheduled as $r_2(t_4)$ is not larger enough than $r_1(t_4)$ to fulfil $f_1(r_2(t_4), d_2, l_2; t_4) > f_1(r_1(t_4), d_1, l_1; t_4)$. Hence the $\mathrm{BL}^4\mathrm{DF}$ scheduler saves TC_2 from complete service starvation whilst optimizing the wireless link's usage. Figure 4.2 illustrates the above points.

For this simple case of two mobiles, each with a single active traffic type (class), the scheduling order can be decided by intuition. However, for a more complicated situation where traffic must be scheduled to many mobiles each with more than one type (class) of traffic active simultaneously, a rigorous optimisation analysis is required for proper functioning of the scheduler. This argument is the basis of the subsequent sections.

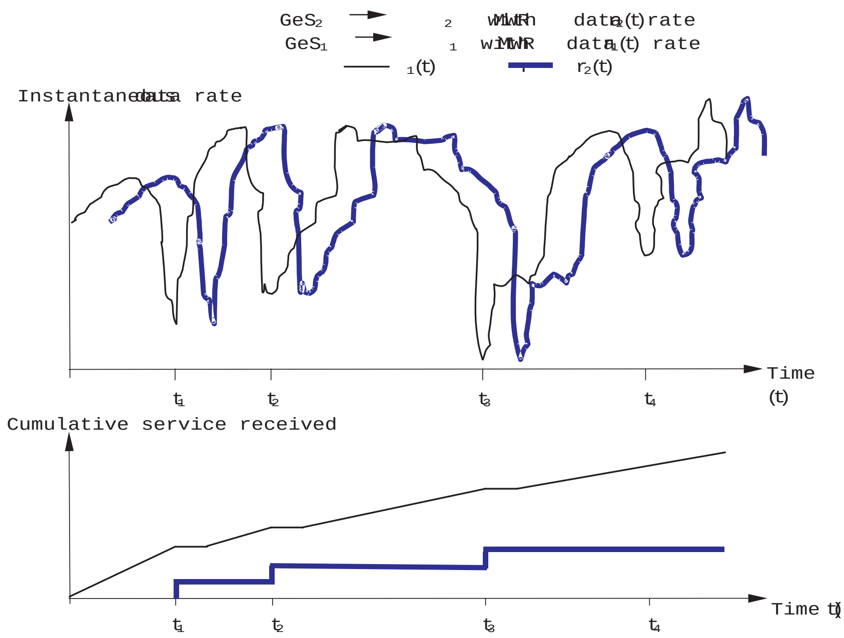

Figure 4.2: A simple illustrative example of two mobiles, each backlogged with a single traffic type (class). Throughput is normalized with respect to the amount of scheduling units received by a corresponding GeS.

4.3 Analysis of the BL^{4}DF Scheduler

This section analyses the BL^{4}DF packet scheduling scheme presented in Section 4.2.

4.3.1 Motivation

This subsection analyses a wireless channel state dependent packet scheduling scheme—the best link, lowest loss, lowest delay first (BL^{4}DF) scheme— to provide DQoS in wireless networks. DQoS is crucial to the successful uptake of next generation multiservice wireless networks poised to support multimedia traffic with diverse characteristics. The proposed scheduling scheme optimizes the usage of the scarce radio resource while meeting applications QoS constraints. Under this scheme, applications traffic is categorized into QoS classes; and at each scheduling instant a packet of the highest backlogged QoS class destined to the mobile with the best radio link quality, and hence data rate, is scheduled. The BL^{4}DF scheme deviates from the philosophy of simultaneous, uncoordinated multi-user transmissions common in conventional mobile systems by using channel state and traffic QoS to transmit to a single mobile at a time. This aids in optimizing the usage of mobile battery power and spectral efficiency. The scheme is applicable to multiservice mobile wireless networks such as Qualcomm's HDR (High Data Rate).

As the best-effort Internet has evolved to 'involuntarily' support heterogeneous traffic with differing quality-of-service (QoS) sensitivities and user expectations, there is a recent realization of the exigency of differentiated quality of service (DQoS) schemes. For example, voice traffic requires end-to-end delay of less than 100 ms to avoid echoing effects but can tolerate some packet losses, longer connection setup time, and bit error rates (BER) of order 10^{-3}. Data traffic, on the other hand, requires no packet losses, BER $< 10^{-5}$, fast session setup, but tolerates delays in the order of 10 s. DQoS also allows differentiated pricing of network services, which prevents the *tragedy of the commons,* which motivates every user to request the highest QoS due to lack of pricing differentiation (see Paris Metro Pricing principle [77, 78]).

Supporting different traffic types with the same QoS may:

1. not scale (in case of the fat-dump-pipe model if the highest QoS class is used as the basis of network overprovisioning);

2. degrade network utilization (not all applications need high QoS);

3. violate some flows' requirements (if the least QoS class is used as the basis of network provisioning); or

4. charge users unfairly as some pay for the QoS their traffic do not need.

The reason is that no single QoS level can be optimized for different traffic types. Also, with the convergence of wireless mobility and Internet Protocol based content, mobile and base stations become Internet portals and hence need to function as wireless routers (WR), either fixed (FWR) or mobile (MWR). (The MWR and the FWR are also referred to as IMS and IBS, respectively, in some parts of this thesis.) MWR functionality is more critical in infrastructure-less (i.e. ad hoc) networking environments. As the ether is getting congested, radio spectrum is now an expensive commodity in many places, often tagged with very high auction prices. Unlike fixed links, the issue of wireless capacity optimisation is not driven by only radio frequency spectrum acquisition costs, but above all on its availability. Hence, packet-based DQoS schemes optimising radio resource usage while considering traffic QoS requirements are very valuable. This section discusses the BL^4DF scheduling scheme, which accounts for the QoS requirements of user traffic and exploits the spatio-temporal wireless link state of the mobile user in a cell of a cellular system.

The rest of this section is structured as follows. Subsect. 4.3.2 reviews some related work. Subsect. 4.2.1 presents the system model and the model assumptions. The BL^4DF scheduling algorithm is described in Subsect. 4.3.3. In Subsection 4.2.2, an intuitive example of the efficiency of the presented scheme is given. Lastly, Subsection 4.3.4 presents some insights into the mathematical analysis associated with the scheduling algorithm.

4.3.2 Related Previous Work

This section reviews concisely some recently proposed scheduling schemes for wireless networks. Most scheduling schemes proposed for Qualcomm's HDR are user-centric in that the MAC (medium access control) layer schedules traffic based on user's instantaneous wireless link condition (i.e., fading level) without differentiating the QoS requirements of the user's traffic. Thus a user with QoS-sensitive traffic but in poorer radio environment tends to receive poorer QoS all the time than a user with best-effort traffic but in a better radio environment. The scheduling scheme in [3] maximizes the wireless channel capacity in a multi-user environment, but does not focus on traffic differentiation as all traffic is treated as real-time. The scheme discussed in [8] maximizes the achievable data rate per cell over all active users based on the user data rates in subsequent timeslots fed back to the FWR, under the same or disparate latency requirements of users.

4.3.3 BL^{4}DF Algorithmic Description

As discussed in Sect.2.6, different applications traffic have different characteristics which poses different QoS constraints. For packets of a generic stream (GeS) i (GeS$_i$), let D_i, α_i, d_i, and δ_i be, respectively, the packet delay, the weight for the delay, the worst-case delay threshold, and the maximum probability of the delay exceeding the delay threshold. Also, let L_i, β_i, l_i and ϵ_i be the corresponding parameters for the buffer (queue) occupancy to reflect the packet loss probability. The loss and delay parameters can reflect the instantaneous (transient), average, or steady-state (long-term) QoS experienced by corresponding traffic class. For the time being, we leave this choice open in order to generalise the algorithm. Obviously, we require that $\delta_i,\ \epsilon_i \in \Re_+$ and $d_i, l_i \in \Re_{++}$, where $\Re^n_+ = \{x \in \Re^n | x \geq 0, n \geq 1\}$ and $\Re^n_{++} = \{x \in \Re^n | x > 0, n \geq 1\}$ are nonnegative and strictly positive orthants of $\Re^n$, respectively.

With these parameters, define the following *probabilistic QoS inequality constraints*, in terms of packet delay and packet loss ratio, for TC$_i$

$$Pr\{D_i > d_i\} \leq \delta_i, \quad i = 1, 2, ..., c \tag{4.1}$$

$$Pr\{L_i > l_i\} \leq \epsilon_i, \quad i = 1, 2, ..., c \tag{4.2}$$

For the purpose of analytical tractability, we can also impose a realistic constraint such as

$$r_m(t) \geq \varsigma_m(t), \quad m = 1, 2, ..., M \tag{4.3}$$

on the data rate, $r_m(t)$, where $\varsigma_m(t)$ is any small positive number. Equation (4.3) can be interpreted as assured minimum service rate to mobile user m. Although probabilistic QoS constraints yield only *soft* rather than *hard* or *deterministic QoS guarantees,* they are attractive as they yield better network resource utilization. Deterministic schemes hardly achieve their targets without network overprovisioning, which penalises network resource utilization. It can also be argued out that some applications can tolerate some packet delays or even losses. Also, define the following instantaneous channel state-dependent $objective\ function(s)$

$$f_1(r_m(t), d_i, l_i; t) = (\alpha_i/d_i + \beta_i/l_i) r_m(t), \quad i = 1, 2, ..., c;\ m = 1, 2, ..., M \tag{4.4}$$

$$f_2(r_m(t), d_i, l_i; t) = (\alpha_i d_i + \beta_i l_i)/r_m(t), \quad i = 1, 2, ..., c; \; m = 1, 2, ..., M \quad (4.5)$$

As discussed above, a key feature of this scheduling scheme is that it depends on both the instantaneous link condition of the mobile (and hence the mobile's current location in a cell), as well as the QoS requirements of the mobile's backlogged packets. Thus, the BL^4DF scheduler ensures that a packet belonging to the highest QoS traffic class i to be scheduled to the MWR_m with the best wireless link during the scheduling timeslot beginning at time t is given the highest scheduling priority. The scheduling algorithm works as follows. For each scheduling interval of length t_{S} beginning at time t (i.e. scheduling period $[t, t + t_{\text{S}})$), schedule a packet in queue i to MWR_m for which the objective function (4.4) is maximized, or (4.5) is minimized under the probabilistic QoS inequality constraints (4.1) and (4.2). $r_m(t) \in \Re_{++}$ is the received channel state dependent usable data rate for the link between the serving FWR and MWR_m, and it is measured from the downlink pilot signals in preceding timeslots and fed back to the FWR on the uplink. While α_i has the dimension of time, β_i is dimensionless. If the optimisation (i.e., the *basic scheduling* rule) fails to deliver a unique solution, i.e., more than one set of mobile/traffic class pairs are in the *feasible set*, then any of the following rules may be applied:

- arrival order (i.e., first-come, first-served, or FCFS) or random order (e.g., tossing of a coin),
- mobile with the smallest cumulative scheduling rate,
- business factors such as service tariff, or
- organizational policy (e.g. traffic sender's status).

The *basic scheduling* rule is a function of user traffic QoS requirements and user's current wireless link data rate (or quality which is reflected in the received signal-to-noise ratio, or SNR).

Remark 4.2. The weights α_i and β_i depend on the delay and loss sensitivities of TC_i. Hence they are *shape factors* (or tuning knobs) [23] as they can be used to shape the QoS profile of TC_i. For example, a streaming media or a conversational traffic with strict delay sensitivity may require $\alpha_i \to 1$, while $\alpha_i \to 0$ & $\beta_i \neq 0$ will do for a best-effort traffic like e-mail. Therefore, α_i and β_i can take values in the set $\{0, 1\}$. It is assumed that the serving FWR has a perfect knowledge of $r_m(t), m = 1, 2, ..., M$, at the beginning of the scheduling period $[t, t + t_{\mathrm{S}})$. $\square$

Remark 4.3. Consider two users with instantaneous link rates $r_1(t)$ and $r_2(t)$ with backlogged traffic of QoS classes TC_k and TC_l, respectively, where TC_l requires a better QoS than TC_k. Due to the inherent randomness in the wireless channel condition (state), there is a non-zero probability that TC_k is scheduled in a time slot $[t, t+t_{\mathrm{S}})$ if $r_1(t) \gg r_2(t)$ although TC_l is also backlogged. This is not a mishap as it protects any of the queues against complete service starvation common to strict priority scheduling schemes, and yet optimizes the radio resource usage and provides average relative differentiated QoS. Strict priority scheduling schemes violates the fundamental issue of *fairness,* as some queues can be completely starved of service. $\square$

4.3.4 Mathematical Analysis: Constrained Optimisation

For a given set of packet delay and packet loss distributions, let us suppose that the following functions are obtained after evaluating the respective inequality constraints (4.1)–(4.3):

$$h_1(d_i, \tau_i) \geq 0, \quad h_2(l_i, \epsilon_i) \geq 0, \quad \text{and} \quad h_3(r_m(t), \varsigma_m) \geq 0.$$

Also, define the nonnegative vector $\mathbf{x}_{i,m} = (r_m(t), d_i, l_i) \in \Re^3_{++}$, and let $\mathcal{D}(\theta) = \{\mathbf{x}_{i,m} | r_m(t) > \varsigma_m, h_1(d_i, \tau_i) \geq 0, h_2(l_i, \epsilon_i) \geq 0\} \in \Re^3_{++}$ be the set of feasible solutions of (4.4) or (4.5), where θ is a vector of traffic distribution parameters. This permits us to transform the given problem into a *parametric optimisation problem* of the form

$$\mathcal{D}_1^*(\theta) = arg \max_{\{i,m\}} \{f_1(\mathbf{x}_{i,m}, \theta) | \mathbf{x}_{i,m} \in \mathcal{D}(\theta)\} \tag{4.6}$$

$$\mathcal{D}_2^*(\theta) = arg \min_{\{i,m\}} \{f_2(\mathbf{x}_{i,m}, \theta) | \mathbf{x}_{i,m} \in \mathcal{D}(\theta)\} \tag{4.7}$$

Claim 4.1—Existence of (global) optimum solution. We claim that (4.6) and (4.7) have optimal solutions.

Proof. As $\mathcal{D} \in \Re^3_+$, $\mathcal{D}$ is a closed set. Also, from (4.1) and (4.2) and noting the physical constraint imposed on the data rate, i.e., $r_m(t) \neq \{0, \infty\}$, it can be proved that $\mathcal{D}$ is bounded. Thus, $\mathcal{D}$ is a *compact* constraint set. From the definition of the constraint set $\mathcal{D}$, it is clear that $f_1(\cdot) : \mathcal{D} \to \Re$ and $f_2(\cdot) : \mathcal{D} \to \Re$ are continuous. Hence, from *Weierstraß Theorem* ([98], pp. 90), $f_1(\cdot)$ ($f_2(\cdot)$) attains a maximum (minimum) on $\mathcal{D}$, i.e.

$$\exists \mathbf{y}_1, \mathbf{y}_2 \in \mathcal{D} \quad \text{s.t.} \quad f_1(\mathbf{y}_1) \geq f_1(\mathbf{y}) \geq f_1(\mathbf{y}_2), \quad \mathbf{y} \in \mathcal{D}$$

and

$$\exists \mathbf{y}_1, \mathbf{y}_2 \in \mathcal{D} \quad \text{s.t.} \quad f_2(\mathbf{y}_1) \leq f_2(\mathbf{y}) \leq f_2(\mathbf{y}_2), \quad \mathbf{y} \in \mathcal{D}.$$

□

After ensuring the existence of a solution to the inequality-constrained optimisation problem, we recast (or summarize) the problem as follows:

$$f_1^*(\theta) = sup \; \{f_1(\mathbf{x}_{i,m}, \theta) \mid \mathbf{x}_{i,m} \in \mathcal{D}(\theta)\} \tag{4.8}$$

$$f_2^*(\theta) = min \; \{f_2(\mathbf{x}_{i,m}, \theta) | \mathbf{x}_{i,m} \in \mathcal{D}(\theta)\} \tag{4.9}$$

$$\mathcal{D}_1^*(\theta) = arg \max_{\{i,m\}} \{f_1(\mathbf{x}_{i,m}, \theta) \mid \in \mathcal{D}(\theta)\}$$
$$\mathcal{D}_2^*(\theta) = arg \min_{\{i,m\}} \{f_2(\mathbf{x}_{i,m}, \theta) \mid \in \mathcal{D}(\theta)\}$$

where

$$\mathcal{D}(\theta) = \{\mathbf{x}_{i,m} \in \Re^3_+ | h_1(d_i, \tau_i) \geq 0, h_2(l_i, \epsilon_i) \geq 0, r_m(t) \geq \varsigma_m\}$$

and evaluate it using the *Theorem of Kuhn and Tucker* (KT) and *Lagrange multipliers* [98], which, for completeness, is replicated below without proof:

Kuhn – Tucker (KT) Theorem Let $g(\cdot) : \Re^n \to \Re$ and $h_k(\cdot) : \Re^n \to \Re$ be $\mathcal{C}^1$ functions, $k = 1, 2, ..., K$. Suppose $\mathbf{y}^*$ is a local maximum of $g(\cdot)$ on

$$\mathcal{D} = \mathcal{U} \cap \{\mathbf{y} \in \Re^n | h_k(\mathbf{y}) \geq 0, k = 1, 2, ..., K\},$$

where $\mathcal{U} \subseteq \Re^n$. is an open set. Let $E \subset \{1, 2, ..., K\}$ be the set of *effective constraints*[2] at $\mathbf{y}^*$, and let $h_E = (h_k)_{k \in E}$. Assume the validity of the rank condition (i.e., *constraint qualification* [98]) $\rho(Dh_E(\mathbf{y}^*)) = |E|$, where $|\mathcal{S}|$ is the cardinality of a finite set $\mathcal{S}$ and $Df(\cdot)$ is the first order differential of $f(\cdot)$ with respect to its arguments. Then there exists a vector of *Kuhn-Tucker multipliers* $\lambda^* = (\lambda_1^*, \lambda_2^*, ..., \lambda_K^*) \in \Re^K$ such that the following conditions hold:

$$[KT-1] \quad \lambda_k^* \geq 0 \quad and \quad \lambda_k^* h_k(\mathbf{y}^*) = 0, \quad \forall k = 1, 2, ..., K.$$

$$[KT-2] \quad Dg(\mathbf{y}^*) + \sum_{k=1}^{K} \lambda_k^* Dh_k(\mathbf{y}^*) = 0. \qquad \diamond$$

Condition KT-1 is referred as the *condition of complementary slackness* [98]. The KT theorem, which states the necessary conditions on $(\lambda^*, \mathbf{y}^*)$ to be a local optimum of $g(\cdot)$, can be easily amended for the case of a local minimum.

Corollary Since $h_k(\mathbf{y}^*) \geq 0$, KT-1 has two consequences, i.e.,

$$\text{if} \quad \lambda_k^* = 0 \Rightarrow h_k(\mathbf{y}^*) > 0$$

$$\text{if} \quad \lambda_k^* > 0 \Rightarrow h_k(\mathbf{y}^*) = 0. \qquad \diamond$$

We are now in a position to state the solution steps of the Kuhn-Tucker procedure for solving our inequality-constrained problem:

Application of Kuhn – Tucker Theorem Applying the Kuhn and Tucker method described above to our problem, the following steps results:

1. Compose the Lagrangian[3]

$$\mathcal{L}(\mathbf{x}_{i,m}, \lambda) = f_j(\mathbf{x}_{i,m}, \theta) \pm \lambda_1 h_1(d_i, \tau_i) \pm \lambda_2 h_2(l_i, \epsilon_i) \pm \lambda_3 h_3(r_m(t), \varsigma_m), \ j = 1, 2$$

[2]An inequality constraint $p(t) \geq 0$ is *effective* at a point t^* within its domain if the constraint holds with equality at t^*, i.e., $p(t^*) = 0$ [98].

[3]Formulating a constrained optimisation problem as unconstrained problem in this manner is attributed to Lagrange, who first proposed it in 1760.

2. Find all solutions $(\mathbf{x}_{i,m}, \lambda)$ to the system of equations

$$\frac{\partial \mathcal{L}(\mathbf{x}_{i,m}, \lambda)}{\partial x_k} = 0, \quad k = 1, 2, 3$$

$$\lambda_k \geq 0, \quad \frac{\partial \mathcal{L}(\mathbf{x}_{i,m}, \lambda)}{\partial \lambda_k} \geq 0, \quad \lambda_k \frac{\partial \mathcal{L}(\mathbf{x}_{i,m}, \lambda)}{\partial \lambda_k} = 0, \quad k = 1, 2, 3$$

3. Let $\mathcal{M}$ be the set of *critical points*[4] of $\mathcal{L}(\cdot)$ and evaluate the value of $f_j(\cdot)$, $j = 1, 2$, at each $\mathbf{x}_{i,m}$ in the set $\{\mathbf{x}_{i,m} \in \mathcal{D} \mid \exists \lambda \; s.t.^5 \; (\mathbf{x}_{i,m}, \lambda) \in \mathcal{M}\}$.

We note in above Lagrangian that '+' and '-' signs refer to maximization and minimization, respectively. We assume that $j = 1$ for the case of maximization, while, for the minimization case, $j = 2$. The following proposition aids in establishing a (global) maximum from the above analysis.

Proposition (**Existence of Global Optimum**) There exists a vector λ^* such that $(\mathbf{x}^*_{i,m}, \lambda^*)$ is a critical point of the Lagrangian $\mathcal{L}(\cdot)$ iff the following twin conditions hold:

1. A global maximum $\mathbf{x}^*_{i,m}$ exists to the given inequality-constrained optimisation problem.

2. $\mathbf{x}^*_{i,m}$ fulfils all the given constraints, i.e., $h_k(\mathbf{x}^*_{i,m}) \geq 0, k = 1, 2 \quad \diamond$.

Point (a) of the above proposition can be ascertained by Weierstraß Theorem.

4.4 BL^{2}DF—Best Link Lowest Delay First Scheduler

This section studies a special case of the BL^{4}DF packet scheduler in which only the delay component is considered, i.e., $\beta_i = 0$ and $\alpha_i = 1$ in (4.4) and (4.5).

[4]Any solution $(\mathbf{x}_{i,m}, \lambda)$ to the above system of equations in point (2) is referred to as a *critical point* of the Lagrangian $\mathcal{L}(\cdot)$, where $\mathcal{M} \subseteq \mathcal{D}$

[5]s.t. means "such that"

4.4.1 Problem Formulation

Assume that the wireless channel is discretized into N states indexed by n with corresponding bandwidth/coding efficiency β_n as given in (5.13). Also, assume a multiple access scheme where M mobile users share a wireless interface with 'raw' rate R_p packets/sec at a cell of a cellular network. The scheduling time is slotted in intervals of duration T_s. At the beginning of each scheduling slot the link of each of the active mobiles is in one of the N discrete channel states. Thus the effective total link rate at the scheduling slot beginning at time t is

$$R_n(t) = \beta_n(t)R_p, \quad n = 1, 2, \cdots, N, \; t \in \{0, \infty). \tag{4.10}$$

Assume that traffic is classified into c generic traffic streams (GeS) based on traffic characteristics, QoS requirements or a form of policy, and that each of the M mobiles can have any number of GeSs active simultaneously. Let λ_i in packets per second be the mean traffic rate generated from GeS_i, $i = 1, 2, \cdots, c$ and let an GeS_i be associated with QoS-related parameter d_i. For instance, d_i can be mapped onto the flow identifier field in an IPv6 header. Suppose, by convention, that if $d_i < d_j$ then GeS_i receives (or at least requires) a better scheduling service than GeS_j at each server (radio port). For instance, if d_i is interpreted as delay bound or tolerance, then GeS_i is said to be more timing sensitive than GeS_j. Let $r_m(t)$ be the effective data rate of the link between mobile user m, $m = 1, 2, \cdots, M$ and the serving FWR in the scheduling slot beginning at a time instant t. Note that $r_m(t) \in \{R_n(t) | n = 1, 2, \cdots, N\}$. Hence, the link of each active mobile in the cell is in one of the N discrete channel states at each scheduling instant.

The packet scheduling scheme being described belongs to a new class of *channel state-dependent packet schedulers* (CSDPS), as its functionality depends on the instantaneous wireless link quality (state) of the mobile's link to the corresponding serving access port (FWR). The architecture of the scheduling scheme is illustrated pictorially in Fig. 4.3 for the case of three active mobiles in a cell. The scheduler schedules the c queues using a simple but efficient scheduling rule, which is: for each scheduling slot beginning at time t, select and schedule mobile m^* with link rate $r_{m^*}(t)$ and backlogged with traffic belonging

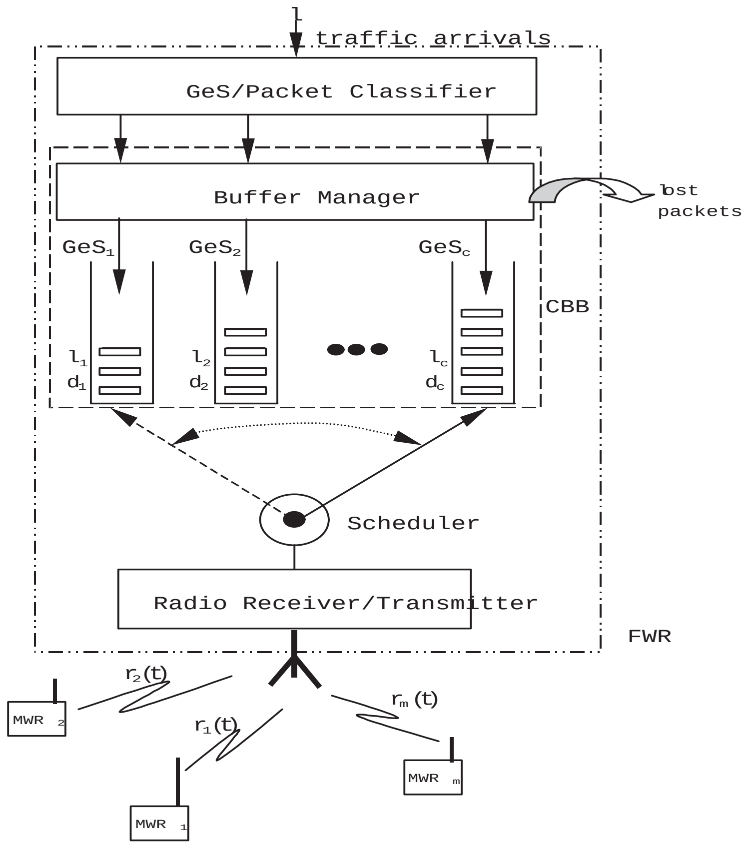

Figure 4.3: Schematic operations of the integrated generic stream (GeS) classifier, class-based buffering (CBB), and the channel state dependent packet scheduler.

to GeS_{i^*} for which[6]

$$(i^*,\ m^*) = \arg\ \max_{i,m}\Big\{r_m(t)/d_i\Big\} \tag{4.11}$$

is fulfilled. For the obvious reason, we refer to this service scheme as *Best Link, Lowest Delay First* scheduler, or simply BL^{2}DF. The BL^{2}DF scheduler has the complexity of only $O(M)$ as for each scheduling slot it performs M divisions and $M-1$ comparisons. The key features of the BL^{2}DF scheme are:

[6]Note that there is a possibility of a mobile with non-head-of-line packet being selected for scheduling if its link quality is much better than mobiles with packets preceding it in the same queue.

- It makes efficient use of the expensive radio resource by optimizing channel utilization.
- It accounts for QoS (delay) constraints of user traffic.
- It provides relative QoS to traffic, as absolute QoS provisioning on a dynamically changing wireless link is perhaps a fantasy.
- Arriving packets are classified and queued with respect to the generic stream to which they belong.
- Usually packets in a given logical queue are served in an first-come-first-served manner.
- It prevents service and buffer starvation to any of the c queues.
- Only a single GeS/queue is scheduled at a time (cf. TDMA multiple access).
- No compensation is awarded to any lagging traffic stream.

If the scheduling interval (or slot duration) T_s is shorter than the wireless channel state dwell time or fade duration (which can be ascertained using Eq. (5.7) and/or Table 5.1), then a given mobile may be scheduled in more than one consecutive scheduling slots. This aids in optimizing the usage of mobile battery power as it reduces the frequency of switching between the active and sleep modes. It is noted that the scheduling rates are time-dependent and varies as a function of the channel BER (bit error rate) and the efficiency of the error control scheme as reflected in β_n [see (5.13)]. The BER, in turn, depends on the modulation format used, as well as the channel noise (fading) level.

4.4.2 Illustration of BL^{2}DF Performance

A simple experiment to illustrate the effectiveness of the BL^{2}DF scheduler is depicted in Fig. 4.4 for the case where only two generic streams (GeSs) are differentiated. For simplicity of illustration, it is assumed that there are two active mobiles (MWR$_1$ and MWR$_2$) in a cell of a cellular network, and that

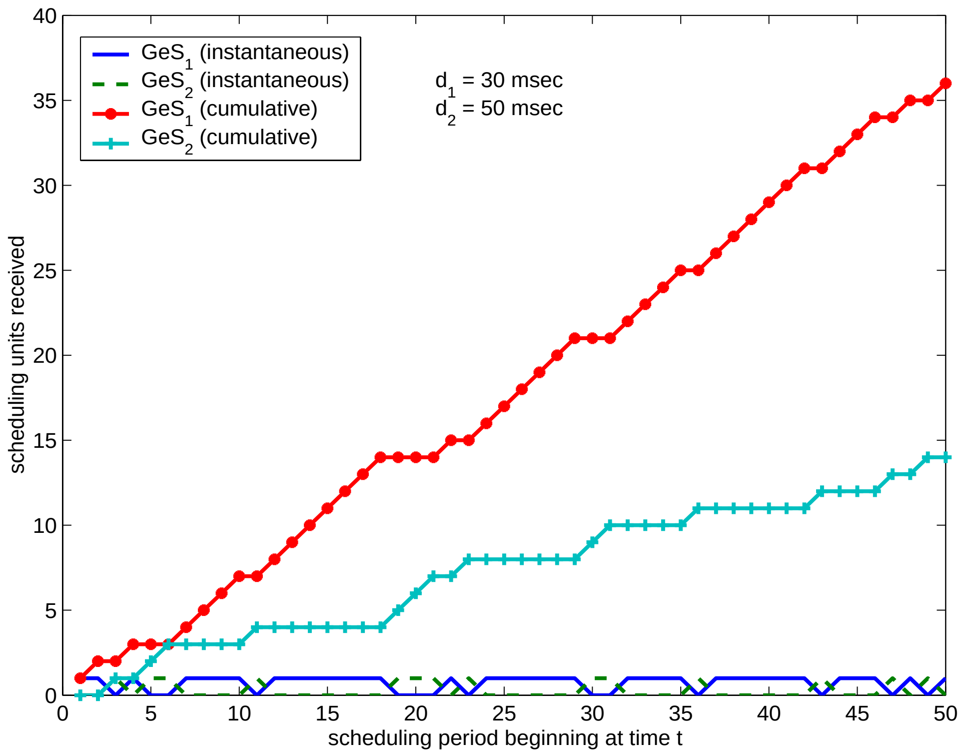

Figure 4.4: Illustration of the efficiency of the BL^{2}DF scheduler for the case of two different GeSs and two mobiles.

MWR$_1$ is communicating packets belonging to GeS$_1$, while packets to/from MWR$_2$ belong to GeS$_2$. Here, it is assumed that GeS$_1$ and GeS$_2$ have the respective delay constraints of 30 msec and 50 msec. Hence, GeS$_1$ is more delay sensitive than GeS$_2$. In contrast to strict priority scheduling, it is noted from Fig. 4.4 that the BL^{2}DF scheduler does not completely starve the queue for GeS$_2$ while at the same time given a better service to GeS$_1$ (on the average) and optimizing the wireless bandwidth usage. The reason is that the scheduling rule (4.11) permits the scheduling of GeS$_2$ even if GeS$_1$ is also backlogged. This happens if the wireless link of MWR$_2$ to the serving FWR is better than that of MWR$_1$ so that $r_2(t^*)/d_2 > r_1(t^*)/d_1$ at time t^*.

Fig. 4.4 was obtained as follows:

- At each scheduling instant t generate two independent random numbers b_1 and b_2 in the range $[1, N]$, where N is the number of channel states (see

Section 5.3);

- Set $r_1(t) = R_p \times \beta(b_1)$ and $r_2(t) = R_p \times \beta(b_2)$;
- Schedule a packet belonging to GeS_1 to/from MWR_1 if $r_1(t)/d_1 \geq r_2(t)/d_2$, otherwise schedule a packet belonging to GeS_2 to/from MWR_2;

where $\beta = (\beta_1, \beta_2, \cdots, \beta_N)$. $\beta_n,\ n = 1, 2, \cdots, N$ is obtained from the wireless link model described in Section 5.3.3 (see Eq. 5.13) except that Nakagami$-m$ fading model (see Eq. 2.2) with $m = 0.5$ is used instead of the Rayleigh case where $m = 1$. Also, a four-state (i.e. $N = 4$) channel model such as shown in Fig. 2.5 is used to obtain the results shown in Figs. 4.4 and 4.5. The other parameters are set to the values: $\rho = 8$ dB, $0 \leq \gamma \leq 12$ dB, $\hat{r} = 5$, $\tau = 2$, and $\kappa = 12$ bytes. IP packets of variable size L_p is generated from the probability distribution (2.28) in Section 5.7.

4.5 BL^{2}PF—Best Link Largest Premium First Scheduler

This section considers a variant of the BL^{2}DF scheme of Section 4.4 in which the QoS parameters are imbedded into a single service premium p. Hence, such a QoS scheme integrates a business model into the channel state dependent scheduling scheme. Such a scheme is attractive as the user willing to pay the largest premium can be said to be the user who values or needs the service most, and hence may be given a scheduling priority.

In the following let p_i be the service premium, measured in price per unit of data transported, for GeS_i's traffic. The reference unit of service charge can be bits, packets, bytes, etc. With this, the scheduling rule (4.11) can be amended as

$$(i^*,\ m^*) = \arg\ \max_{i,m}\Big\{r_m(t) \times p_i\Big\}, \tag{4.12}$$

where all other parameters have the same meaning as used in (4.11). The efficiency of the BL^{2}PF scheme is illustrated experimentally in Fig. 4.5.

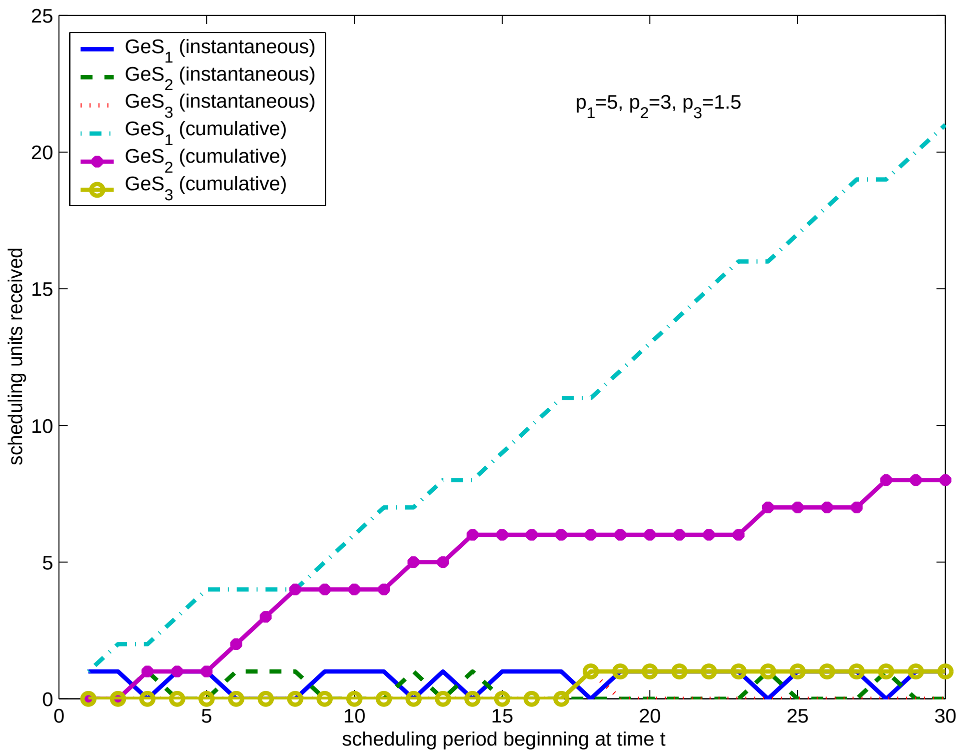

Figure 4.5: Illustration of the efficiency of the BL^{2}PF scheduler for the case of three different GeSs.

4.6 NP^{3}A—Non-Pre-emptive Priority with Partial Assurance Scheduling

In contrast to the BL^{4}DF scheme, the NP^{3}A scheduling scheme assigns scheduling times to traffic classes based solely on their QoS requirements without considering the instantaneous channel quality of the mobiles. As such, it is simpler than the BL^{4}DF scheme, but fails to optimise the wireless bandwidth. Also, this scheme focuses on the long-term (i.e., stationary) cumulative service times rather than instantaneous scheduling priority.

Assume that the wireless channel is partitioned into N discrete states (indexed by n, and that there are M mobiles (indexed by m in the cell of an FWR. Assume that applications traffic is classified into c QoS classes. Without loss of generality, here, we focus our attention on the case where $c = 3$. This permits a one-to-one mapping onto the IETF Diffserv architecture discussed in Section 2.5.4, where TC_1, TC_2 and TC_3 map onto best-effort, assured service and expedited service classes, respectively. The same rationale holds for the PBSP buffer sharing scheme discussed in Section 4.9. Assume that TC_j has non-pre-emptive priority over TC_i, $\forall j > i$ and that $TC_i, i = 2, 3$ is assured at least a fraction f_i of the available data rate so long as it is backlogged. Backlogged TC_1, on the other hand, is served iff $TC_i, i \neq 1$ are not backlogged, albeit with entire bandwidth. Within a queue class, packets are served in a FIFO (first-come, first-served) order. Note, however, that any of the c traffic classes, say, TC_i, can be further subdivided into independent classes with assured fraction f_{ij} of its resources to subclass TC_{ij}. Under the above assumptions, we obtain the following average scheduling rates for mobile user m's TC_i in wireless link state $n, n = 1, 2, ..., N$

$$r_{n,i} = \begin{cases} p_i\bar{p}_2\bar{p}_3 r_n, & \text{if } i = 1 \\ (\bar{p}_1\bar{p}_i\bar{p}_3 f_2 + p_i p_3 f_2 + p_i\bar{p}_3) r_n, & \text{if } i = 2 \\ (\bar{p}_1\bar{p}_2\bar{p}_i\bar{f}_2 + p_2 p_i\bar{f}_2 + p_i\bar{p}_2) r_n, & \text{if } i = 3 \end{cases} \tag{4.13}$$

Note that, in general, $r_n = \beta_n R_p$ can be different for different i's as the mobiles may be liable to different channel conditions, and hence qualities and data rates. Here, R_p is the nominal link transmission rate in packets/second and the link efficiency, β_n, can be evaluated as in (5.13). The weights for the r_n's should sum up to unity, as they reflect the proportion of times each traffic class receives service.

Remark 4.1 As $f_j > f_i, \forall j > i$, on an ideal link TC_j should have low delay and low loss than TC_i. However, this is not guaranteed on a dynamic wireless link. f_i can be computed dynamically to reflect run-time (i.e. active) resource sharing between traffic classes or dimensioned statically.

4.7 Wireless GPS Scheduling

The GPS scheduling is reviewed in Section 2.10.2. In this section, we adapt its definition to wireless channels with spatio-temporarily changing data rate.

With the wireless GPS (WGPS), the channel state dependent instantaneous data rate received by GeS_k is computed as

$$g_k(\tau) = \frac{r_k}{\sum_{i\in B(\tau)} r_i}\beta_n R_p \geq r_k\beta_n R_p, \quad k = 1, 2, \cdots, c \text{ and } n = 1, 2, \cdots, N, \tag{4.14}$$

where β_n is computed similarly to (5.13) or (7.10).

4.8 CSDPS—Channel State-Dependent Rate Proportional Scheduling

Assume that there are c stream of traffic (GeS) indexed by i sharing a wireless interface with 'raw' rate R_b. Assume that traffic stream GeS_i is given a resource-sharing-related weight w_i. Assume that the wireless channel is discretized into N states indexed by n, and that the instantaneous state of the link depends on the received signal-to-noise ratio (SNR). Let β_n be the wireless link efficiency when the link is in a discrete-time state n. Then the *effective* rate of the *channel state-dependent proportional scheduler* (CSDPS) for GeS_i's traffic is

$$r_{n,i} = w_i\beta_n R_b, \quad i = 1, 2, \cdots, c \text{ and } n = 1, 2, \cdots, N \tag{4.15}$$

Note that, if we set $c = 3$ and interpret r_n as $\beta_n R_b$, then (4.15) reduces to (4.13), where the weights w_i, $\forall i$ are given by $w_1 = p_1\bar{p}_2\bar{p}_3$, $w_2 = \bar{p}_1\bar{p}_2\bar{p}_3 f_2 + p_2 p_3 f_2 + p_2\bar{p}_3$ and $w_3 = \bar{p}_1\bar{p}_2\bar{p}_3\bar{f}_2 + p_2 p_3\bar{f}_2 + \bar{p}_2 p_3$. Also, the CSDPS scheduler reduces to the WGPS scheduler in (4.14) if we set $w_k = r_k/\sum_{i\in B(\tau)} r_i$, for $k = 1, 2, \cdots, c$. Hence, the CSDPS scheduling is used in this thesis as a generic scheduler which can degenerate to the WGPS or the $\mathrm{NP}^3\mathrm{A}$, depending on how we dimension the weights.

4.9 PBSP—Proportional Buffer Sharing with Pushout

Let the router's buffer of size B be dynamically partitioned into c queues of sizes $B_i = f_i B$, where $f_i < f_j, \forall i < j$. Dynamic buffer partitioning avoids packet loss until the entire router's buffer is full, and absorbs dynamics in load distribution. An arriving TC_1 packet is dropped if its queue is full. However, if a TC_3 packet meets a full queue, it pushes out the TC_1 packet nearest to the head-of-the-line (HOL), if any. Pushing out the HOL TC_1 packet reduces the average queuing delays of its packets. Thus a TC_3 packet is dropped iff there are $B_1 + B_2$ packets of its kind in the system. These assumptions lead to the actual aggregate packet arrival rate to the buffer as

$$\lambda = \sum_{i=1}^{c} \lambda_i I\{n_i < B_i\} \tag{4.16}$$

where

$$I\{y\} = \begin{cases} 1, & \text{if } y \text{ is true} \\ 0, & \text{otherwise} \end{cases}$$

is the indicator function and n_i is the amount of TC_i's load in the queue.

4.10 Summary and Discussions

Without lack of generality, we can set the scheduling interval t_s as integer multiples of the frame duration of the wireless multiple access scheme being used. To evaluate the above optimisation problem, we first need to assume some packet delay and packet loss distributions to analyse the QoS constraints (4.1) and (4.2). Recent research widely beliefs that packet inter-arrival times are *self-similar* and this self-similarity can be modelled by a heavy-tailed distribution such as the Pareto distribution [83] with the p.d.f. given in (5.16). Hence such a distribution can be used for the packet delays. Since different traffic has different loss and delay distributions, it is not reasonable to evaluate the optimum scheduler which is generally applicable. Hence, this task is left for further study.

This chapter presented simple scheduling and buffer management schemes which can aid in the provisioning of differentiated service quality in Internet Protocol based wireless networks. In the following chapters, these schemes are used to model and analyse the performance of communications systems using optimisation methods and elementary queuing theory. The NP^{3}A, CSDPS and the WGPS schemes are used in the QoS analysis in Chapter 5, while Chapter 8 explores NP^{3}A and an abridged version of the PBSP schemes. Section 4.4 analysed a special case of the BL^{4}DF scheme. A key difference between the BL^{2}DF scheme presented in Section 4.4 and the works in references [3, 8] is the design of the scheduling rule. As a brief recap, a framework of wireless channel-state-dependent packet scheduling schemes is proposed in this chapter which offer many advantages including:

- It prevents absolute service starvation to any of the queues, due to the inherent randomness in the scheduling rules.
- It prevents buffer starvation to any of the queues, as an empty queue is not scheduled.
- It optimizes the 'expensive' wireless bandwidth resource.
- It exploits wireless spatio-temporarily varying properties; and lastly,
- It provides differentiated QoS based on actual traffic characteristics and the wireless link quality of a wireless/mobile user.

Science can purify religion from error and superstition. Religion can purify science from idolatry and false absolutes – Pope John Paul II.

Chapter 5

Fluid Analysis of a QoS Scheme for IP Networks over Fading Wireless Channels

This chapter[1] uses simple analytic modelling to investigate the effects of inherent wireless properties on traffic quality of service (QoS). The QoS metrics considered are the mean packet queueing delay at a server (e.g. radio base station in a cellular network) and the server's buffer overflow probabilities. Wireless properties accounted for are the user mobility, radio fading level in form of the received mean signal-to-noise ratio, and error control overhead (i.e., number of allowable ARQ retransmissions and FEC parity check bits). It is hoped that the study will complement the many simulation studies reported on similar issues. Many of the results are rather intuitive. However, it has still proved valuable to analyse them analytically. The well known results that delay increases with buffer capacity, but the packet loss decreases with increasing buffer capacity are also verified thereby, and most importantly, used as a check on the validity of the analytical modelling. It is observed that the packet losses increase with increasing user mobility, decrease with increasing received SNR, decrease with increasing FEC redundancy, and increase with increasing maximum number of

[1] An abridged version of this chapter is submitted to the IEEE Trans. on Netw. on Jan 21, 2003.

ARQ retransmissions, when all other parameters are held the same. The packet queueing delay, on the other hand, decreases with increasing user mobility but increases with increasing received SNR.

5.1 Introduction

With the convergence of wireless mobility and Internet Protocol based rich content into mobile wireless Internet (MOWINT), mobile station (MS) and base station (BS) evolve to Internet portals and hence need to function as wireless routers (WR), either Fixed Wireless Router (FWR) or Mobile Wireless Router(MWR). MWR functionality is more critical in infrastructure-less (ad hoc) networking environments.

To date the wireline Internet supports only a single service based on the so-called best effort service model, which does not differentiate the traffic transported by it, nor guarantee explicitly even the successful delivery of packets [34]. However, MOWINT systems are poised to be multiservice communications platforms to support multimedia traffic with different characteristics, and hence different service requirements. Hence, traffic differentiation and service guarantees (QoS) are more of utilities than supplementary features in upcoming MOWINT systems. Scalable schemes that can provide appropriate quality-of-service (QoS) to traffic of various types of applications is very crucial to the successful uptake of wireless Internet.

As MOWINT systems rely on radio transmissions, the user-network interface is dynamically changing, both in time and space (i.e., spatio-temporarily varying state) due to user mobility, radio fading, noise and interference, etc. This spatio-temporarily changing channel state property can, however, be exploited to optimise the spectral efficiency. On the other hand, the spatio-temporality feature of wireless links makes QoS provisioning in MOWINT systems a non-trivial endeavour.

A packet scheduling scheme is a common mechanism used to control the service order and delay of packets competing for service at a common server (e.g., base station). Many scheduling schemes proposed in the literature for wireless networks are user-centric in that the MAC (medium access control) layer schedules traffic based on user's instantaneous wireless link condition without differentiating the QoS requirements of the user's traffic scheduled. Thus a user with QoS-sensitive traffic but in poorer radio environment tends to receive poorer QoS all the time than a user with best-effort traffic but in a better radio environment. Other scheduling schemes are applications' centric, as they focus only on the applications QoS requirements without optimizing the usage of the scarce radio resources.

This work analyses a simple QoS scheme for packetized mobile wireless networks; specifically, networks advocating to the packet based transport philosophy of the Internet Protocol (IP). The scheme is based on traffic classification and wireless channel state dependent packet scheduling. The interactions between error control used to mitigate wireless impairments at the link layer and the packet scheduling scheme (in terms of packet transfer delay and packet loss probabilities) are examined. Also, our model integrates physical layer properties such as fading and modulation into higher layer QoS analysis. The validity of the scheme is analysed on a three-state wireless link model, accounting for fast multipath fading and additive white noise, modulation format, and error control, under bursty traffic sources. Although many works on QoS are reported in the open literature, analytic studies investigating the effects on traffic QoS of wireless inherent properties such as user mobility, radio fading level (via received signal-to-noise ratio) and error control scheme are uncommon. Hence, the motivation for this study. The analytic studies presented in this chapter aims to complement the numerous QoS simulation analysis reported in the open literature.

The rest of the chapter is structured as follows. It comprises two parts. Part I uses a single on/off source to model the traffic generated by each traffic type (see below). Part II, however, models traffic from each traffic type using multiple on/off sources, where the number of the sources in the 'on' state at each instant is a random process. The first part begins with a discussion of the background material needed to follow the analysis. This comprises a general description of the model assumptions used for the analysis in Section 5.2.1, and a brief review of the related previous works in Section 5.2.2. Following this, the detailed system model is described in Section 5.3. The system model described include the wireless channel model used (Sections 5.3.1 and 5.3.2) and the error control scheme at Layer 2 (Section 5.3.3). A simple wireless channel state dependent packet scheduling scheme is discussed in Section 5.4. The single-source traffic model appears in Subsection 5.5.1, while its corresponding mathematical analysis is presented in Sections 5.5.2 and 5.6.3. Here, the QoS metrics of interest–the expected packet queuing delay and loss probability–are evaluated. Part II of this chapter comprises Section 5.6, which is subdivided into Section 5.6.1 (multiple-source traffic model) and its corresponding mathematical analysis in Sections 5.6.2 and 5.6.3. Numerical investigations, results and discussions are the objects of Section 5.7. The chapter concludes with a summary in Section 5.8.

5.2 Background and Related Works

The background material and the previous works related to this chapter are briefly reviewed here.

5.2.1 General System Description

This chapter provides a unified model and quality of service analysis of the first three ISO/OSI protocol layers. Based on their characteristics (QoS requirements) or a form of policy, traffic is classified into c generic streams (GeS) and indexed by i. The different GeSs are given differentiated treatment (service) in the network elements. Here, a simple differentiated QoS (DQoS) scheme is used (Fig. 5.1). This DQoS scheme consists of a buffer manager (BM) which is coupled to a packet scheduler (PS). The BM controls packet losses due to buffer overflow, while the PS controls packet transmission delays. The BM and the PS are indirectly dependent, as a slower PS would cause faster buffer overflow. Although any combination of BM and PS schemes may be used, here, for simplicity, we use the threshold dropper (TD) [17] BM scheme and a simple wireless channel state dependent PS scheme which we shall refer to as Non-Pre-

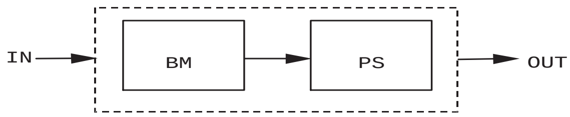

Figure 5.1: A generic QoS scheme comprising a buffer manager coupled with a packet scheduler.

emptive Priority with Partial Assurance (NP^{3}A). The NP^{3}A scheme is explored in Section 5.4.

Each GeS is modelled by one or more on/off traffic sources, and each on/off source toggles between the on and the off states independently and randomly. Moreover, all the on/off sources have identical probability distributions, i.e., i.i.d. sources. In accordance with recent networking research findings, IP traffic inhibits bursty (i.e., correlated) patterns [81]. We thus model the bursty traffic with on/off sources, in which the dwell (sojourn) times in the 'on' and 'off' states are assumed to be Pareto and Weibull distributed, respectively. These on/off traffic sources feed the buffer of a radio node (mobile or base station) which is coupled to a wireless channel with a spatio-temporarily varying condition (state) and hence output data (service) rate, as described below. This time varying channel rate is reflected in the bandwidth (coding) efficiency β_n as evaluated in (5.13)

The model assumptions used for these layers are as illustrated in Fig. 5.2. On the physical layer (L1), it is assumed that the wireless channel is impaired mainly by additive background noise due to transceiver electronics scintillations, and a fast, flat fading due to multipath radio signal propagation. The additive noise is assumed to possess a flat spectrum (i.e. white noise) and Gaussian distributed, while the fading is assumed to be flat (i.e. non-frequency-selective) and without a direct line-of-sight (LOS) path component. Hence, we use the Rayleigh distribution to model the multipath fading. Multipath fading is caused by destructive and constructive mixing of the components of the same signal that traverse different paths due to random delays, scattering, reflection and diffraction [95]. The spatio-temporarily varying nature of this noisy channel is captured in a three-state (i.e. $N_c = 3$), discrete-time model with the state

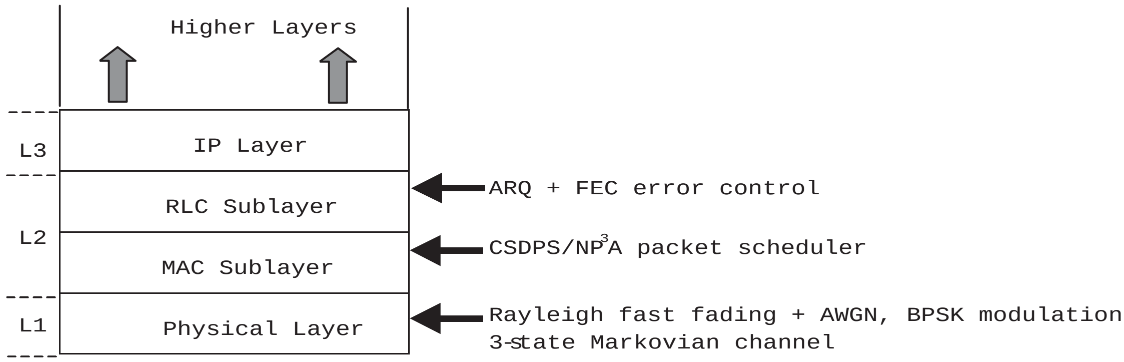

Figure 5.2: Model assumptions at each ISO/OSI protocol layer considered.

space $S_c = \{s_1, s_2, s_3\}$. The parameters of the discrete-time model, such as the state occupancy distributions, $p_n,\ n = 1, 2, 3$, and the state transition rates, $t_{i,j},\ i, j \in S_c$, are evaluated using a binary phase-shift keying (BPSK) modulator/demodulator. BPSK modem is chosen as it is used in many modern wireless networking standards such as IEEE 802.11b due to its robustness. It is also simple, enhancing analytical tractability.

The MAC (medium access control) sublayer (L2) utilizes the proposed NP^{3}A scheme to ascertain the scheduling order to the packets it services. Figure 5.3 illustrates the scheduling scheme operating in a fixed wireless router (FWR) or an IP enabled base station (IBS) serving a cell of a cellular network. The scheduling rate and order each mobile receives depends on the state of its wireless link with the serving FWR, and the type of traffic it is communicating.

In Fig. 5.3, three mobiles (MWRs) are in the range of an FWR, where the wireless links of MWR$_1$, MWR$_2$ and MWR$_3$ are in states (condition) s_2, s_3 and s_1, respectively. MWR$_1$ has traffic of class 1 (i.e. GeS$_1$) active, while both mobiles MWR$_2$ and MWR$_3$ are backlogged with traffic of class 2 (i.e. GeS$_2$)[2]. The details of the scheduling scheme is discussed in Section 5.4. The RLC (radio link control) sublayer (L2) uses a hybrid type-I ARQ scheme, comprising

[2]Although a mobile may possibly have more than one type of traffic active simultaneously, for simplicity of analysis, we assume herein that each active mobile communicates a single traffic type at a time.

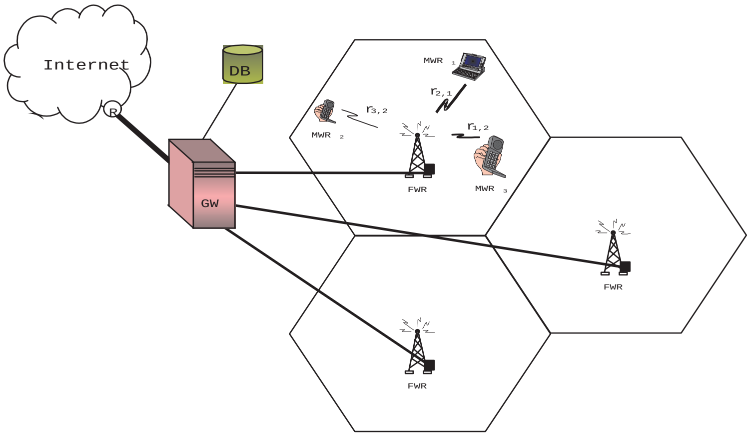

Figure 5.3: Three active mobile wireless routers (MWR) in a cell of a mobile cellular network being served by a fixed wireless router (FWR) and connected to the backbone Internet via a gateway (GW).

both an FEC (forward error correction) and an ARQ (automatic repeat request) components to optimise bandwidth inefficiency and transmission delays.

5.2.2 Related Works

The primary references in this chapter are the excellent work by Mitra [71], Anick, Mitra and Sondhi [5], and Krunz and Kim in [62, 61]. Anick, Mitra and Sondhi in [5] and Mitra [71] provide stochastic theory of a fluid model of traffic sources, traffic consumers (transmission media) and buffers. Since the eigenanalysis in these works are generally applicable, I applied them to situations where the consuming media contains a wireless component, as in [62, 61]. Unlike [5], which assumes a channel or a switch with a constant output rate, in this study the buffer is coupled to a wireless channel with a spatio-temporarily varying output data rate. Kim and Krunz in [62, 61] also follows this pattern, as they also utilized the framework in [71, 5].

Kim and Krunz in [62, 61] accounted for the interactions between the link-layer error control scheme, time-varying wireless channel, and the packet-level QoS provisioning using a two-state Markov channel model. The mathematical analysis draws from the works of Mitra, Anick, and Sondhi in [71, 5]. In contrast to my work, [62, 61] use a single on/off source as traffic generator and do not consider the transceiver properties such as fading and signal modulation. The effects of mobile user's speed of motion on QoS are also not considered. The finite-state, discrete-time model used for the wireless channel in this study is motivated by the classic work of Wang and Moayeri in [110]. Wang and Moayeri's multi-state Markov channel model, however, permits only nearest-neighbour state transitions, and they computed the Markov state parameters using Rayleigh fading model and BPSK modulation. In this study, the nearest-neighbour state transition assumption is not enforced in order to permit transitions between non-adjacent Markov states.

Next, we discuss the detailed system modelling.

5.3 Link Level Model

This section presents the wireless link models and the error control schemes used in the subsequent analytical analysis.

5.3.1 Wireless Channel Model 1

We assume that the wireless channel impairments are dominated by non-frequency-selective (flat), fast, multipath fading and additive noise (AWGN). It assumed that other possible impairments, such as slow fading is mitigated by techniques like interleaving. Such a fading scenario without a line-of-sight (LOS) path can be modelled with the Rayleigh density function, which is given by

$$f_\gamma(\gamma) = \frac{1}{\rho} e^{-\gamma/\rho} u(\gamma), \tag{5.1}$$

where γ is the received signal-to-noise plus interference ratio (SNIR), $\rho = E\{\gamma\}$, and $u(\cdot)$ is the unit step function. The wireless channel with above noise model is modelled by a three-state discrete-time model as illustrated in Figures 5.4 and 5.5. The three Markov states in the model are interpreted as follows. State 3 represents a good-quality link with negligibly low error rate; state 2 represents a degrading link where session handoff is in progress, while state 1 is a poor-quality link in which communications quality is severely impaired. If a better

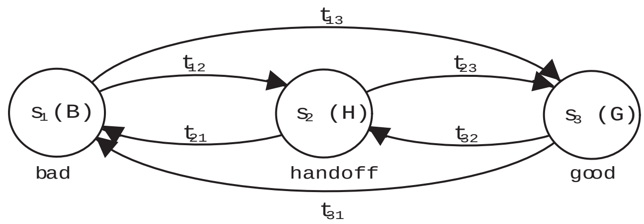

Figure 5.4: State transition rate diagram of a three-state, discrete-time model of a fading wireless channel with time-varying condition.

link is available then mobiles avoid using state 1 links, and that mobiles initiate the handoff process as their link quality falls in state s_2.

This channel model differs from a three-state Fritchman model [28] with two good and one bad states in the sense that state transitions are allowed between the two best link states. Also, the chosen discrete-time model differs from the model in [110] with three states in that not only nearest-neighbour transitions are permitted. The rationale behind the chosen model in this study is that a deep fade can move the link from state s_3 (the good state, or G) to state s_1 (the bad state, or B) so fast that the dwell time in state s_2 (the handoff region, or H) is negligibly small. Similarly, a mobile in outage (with link in state s_1) may obtain a better link (say, state s_3 link) without even the need to hand off to another cell/channel. This may occur, for example, if an obstructing object moves away from the mobile's radio signal propagation path.

The channel state partitions are based on the received SNIR or the received signal strength indicator (RSSI), as illustrated in Fig. 5.5. The received SNIR γ is partitioned into $N_c = 3$ intervals with the thresholds: $0 \leq \gamma_1 < \cdots < \gamma_{N_c+1} < \infty$, and the channel is said to be in state s_n for $n = 1, 2, \cdots, N_c$ if $\gamma \in [\gamma_n, \gamma_{n+1})$. The state space of the three-state channel model is thus $S_c = \{s_1, s_2, \cdots, s_{N_c}\}$. The steady-state probability that the wireless channel is in fading-level dependent state s_n, for $n = 1, 2, \cdots, N_c$ is obtained as

$$p_n = Pr[\gamma_n \leq \gamma < \gamma_{n+1}] = \int_{\gamma_n}^{\gamma_{n+1}} f_\gamma(\gamma) d\gamma = e^{-\gamma_n/\rho} - e^{-\gamma_{n+1}/\rho} \tag{5.2}$$

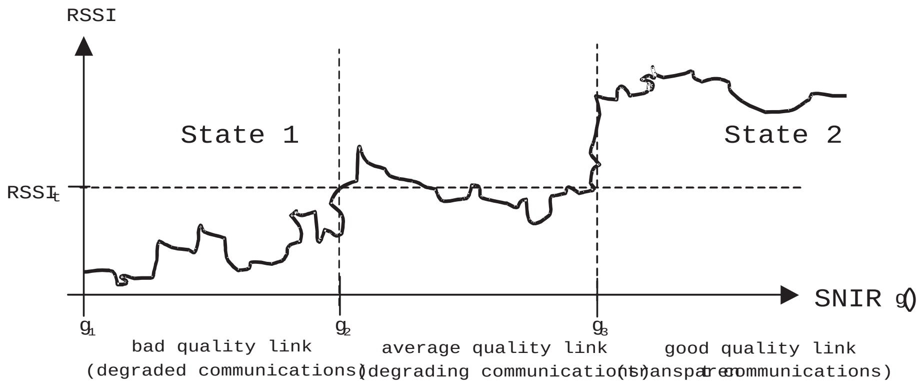

Figure 5.5: Mapping of the three-state Markov wireless channel model onto the received signal strength indicators (RSSI) and the received signal-to-noise plus interference ratios (SNIRs).

Crossover probabilities. We assume that each of the N_c discrete-time states of the channel represents a binary symmetric channel (BSC) which is identifiable with a bit error (crossover) probability $p_{e,n}$. Assuming that the Rayleigh fading is independent of the AWGN, we can obtain the error probability in state s_n as

$$p_{e,n} = \frac{1}{p_n}\int_{\gamma_n}^{\gamma_{n+1}} p_b(\gamma) f_\gamma(\gamma) d\gamma, \quad n = 1, 2, \cdots, N_c \tag{5.3}$$

where $p_b(\gamma) = \frac{1}{2}erfc(\sqrt{\gamma})$ is the bit error rate (BER) of a BPSK modulator/demodulator on AWGN channels. Substituting (5.1) in (5.3) and combining integration by parts[3] with the Leibniz rule[4] for differentiating an integral, we obtain

$$p_{e,n} = \frac{1}{2p_n\rho}\int_{\gamma_n}^{\gamma_{n+1}} erfc(\sqrt{\gamma})e^{-\gamma/\rho} d\gamma = \frac{1}{2p_n}(\eta_n - \eta_{n+1}), \quad n = 1, 2, \cdots, N_c \tag{5.4}$$

where $\eta_n = e^{-\gamma_n/\rho} erfc(\sqrt{\gamma_n}) - \sqrt{\frac{\rho}{1+\rho}} erfc(\sqrt{\frac{1+\rho}{\rho}\gamma_n})$.

[3] Integration by parts states that $\int_a^b udv = uv|_a^b - \int_a^b vdu$.

[4] The Leibniz rule for differentiating an integral states that $\frac{d}{dt}\int_{g(t)}^{h(t)} f(x;t)dx = \int_{g(t)}^{h(t)} \frac{\partial f(\cdot)}{\partial t}dx + f(h(t);t)\frac{dh}{dt} - f(g(t);t)\frac{dg}{dt}$.

State transition rates. The state transition rates are needed to capture the progression of the wireless channel between the allowable states. Let the average number of times per time unit (i.e. level crossing rate, LCR) that the received SNIR passes downward across a given SNIR level γ be $N(\gamma)$. Then $N(\gamma)$ can be estimated as [55]

$$N(\gamma) = \int_0^\infty \dot{\gamma} f_{\gamma,\dot{\gamma}}(\gamma, \dot{\gamma}) d\dot{\gamma}. \tag{5.5}$$

Substituting (5.1) in (5.5) and following the derivations in [55, 101] we obtain the approximated transition rate per second from channel state s_k to state s_l, $k, l \in \{1, 2, \cdots, N_c\}$ from the LCR as [110]

$$t_{k,l} \approx \begin{cases} \sqrt{\frac{2\pi\gamma_l}{\rho}} \frac{v_m f_c}{v_0} e^{-\gamma_l/\rho}, & k < l \\ \sqrt{\frac{2\pi\gamma_k}{\rho}} \frac{v_m f_c}{v_0} e^{-\gamma_k/\rho}, & k > l, \\ 0, & |k-l| > 2 \end{cases} \tag{5.6}$$

where v_m is the mobile user's speed of motion, $v_0 \approx 3 \cdot 10^8$ m/s is the speed of light in vacuum, and f_c is the operating radio frequency. Hence, the state transition rate matrix of the three-state, discrete-time wireless channel model with elements in (5.6) can be given as[5]

$$\mathbf{T}_c = \begin{bmatrix} t_{1,1} & t_{1,2} & t_{1,3} \\ t_{2,1} & t_{2,2} & t_{2,3} \\ t_{3,1} & t_{3,2} & t_{3,3} \end{bmatrix}^T,$$

where $t_{k,l} = t_{l,k} \forall k \neq l,\ t_{k,k} = -\sum_{\forall j, j \neq k} t_{k,j}$. For $\mathbf{T}_c$ to be a valid infinitesimal generator matrix of a stochastic process its columns must sum to a column vector of zeros.

Another interesting parameter concomitant of radio fading channels is the average fade duration (AFD), which is the average time that the fading envelope remains below a specified level after crossing that level. The AFD for a single-branch selection combiner micro-diversity on Rayleigh fading channels can be obtained as

[5] In this study, we denote the transposition of the vector/matrix $\mathbf{h}$ as $\mathbf{h}^T$.

$$T(\gamma_k) = \frac{v_0}{\sqrt{2\pi} v f_c} \sqrt{\frac{\rho}{\gamma_k}} \exp\left(-\frac{\gamma_k}{\rho}\right) \left[1 - \exp\left(-\frac{\gamma_k}{\rho}\right)\right]. \tag{5.7}$$

Equation (5.7) is obtained by assuming that the fading envelope is given by $X(t) = |x_I(t) + jx_Q(t)|$, where both the inphase component, $x_I(t)$, and the quadrature component, $x_Q(t)$, are narrowband, identically distributed zero-mean Gaussian random processes, and that the multipath waves arrive from all directions with equal strengths and equal probabilities. Table 5.1 shows the LCR (5.5) and corresponding AFD (5.7) for different mobile velocities. For the same received SNIR, it is observed that the higher the mobile speed, the higher the LCR but the lower the AFD. The AFD increases with increasing received SNIR, while, in most cases, the LCR decreases with increasing received SNIR.

5.3.2 Wireless Channel Model 2

Wireless channels are commonly impaired by burst errors which are caused by multipath radio propagation (fast fading), shadowing, environmental noise and host mobility [93]. A common model used to model such a burst-error wireless channel is the Fritchman model [28].

In this section, we use a three-state (i.e. $N_c = 3$) Fritchman model with a single bad (erasure) state (s_1) and two good states (s_2, s_3) to model a bursty wireless channel, as illustrated in Fig. 5.6. In this model transitions between the good (or bad) states are forbidden. The reason being that there may not be any remarkable difference between the observable channel quality if a transition occurs between two bad (or good) states. The Fritchman model is applicable to discrete channels with simple burst error patterns [57]. The two good states have different levels of "goodness". The states of this model can be mapped one-to-one onto the three-state model illustrated in Fig. 5.4. The state transition (probability) matrix of the stochastic process underlying the three-state Fritchman channel model shown in Fig. 5.6 can be given as

$$\mathbf{P}_c = \begin{bmatrix} p_{11} & p_{12} & p_{13} \\ p_{21} & p_{22} & 0 \\ p_{31} & 0 & p_{33} \end{bmatrix}^T ,$$

Table 5.1: The LCRs per second and concomitant AFDs in seconds for $f_c = 2$ GHz, $N = 12$ discrete channel states, $\rho = 8$ dB, and $v_0 = 3 \cdot 10^8$ m/s. The mobile velocities chosen correspond to residential area, city CBD and highway speed limits in Australia.

		$v = 45$ km/hr		$v = 60$ km/hr		$v = 110$ km/hr	
s_n	γ range [dB]	AFD	LCR	AFD	LCR	AFD	LCR
s_1	0-5.5046	0.0021	70.9705	0.0015	94.6273	0.0008	173.4834
s_2	5.5046-7.8560	0.0048	89.2603	0.0036	119.0137	0.0020	218.1917
s_3	7.8560-9.3730	0.0079	78.0869	0.0060	104.1158	0.0032	190.8790
s_4	9.3730-10.4951	0.0120	62.0554	0.0090	82.7405	0.0049	151.6909
s_5	10.4951-11.3860	0.0176	47.1232	0.0132	62.8309	0.0072	115.1900
s_6	11.3860-12.1249	0.0255	34.8442	0.0191	46.4589	0.0104	85.1746
s_7	12.1249-12.7562	0.0365	25.3177	0.0274	33.7569	0.0149	61.8877
s_8	12.7562-13.3072	0.0523	18.1693	0.0392	24.2257	0.0214	44.4138
s_9	13.3072-13.7962	0.0748	12.9193	0.0561	17.2258	0.0306	31.5806
s_{10}	13.7962-14.2356	0.1072	9.1209	0.0804	12.1612	0.0438	22.2955
s_{11}	14.2356-14.6346	0.1539	6.4026	0.1154	8.5368	0.0629	15.6509
s_{12}	14.6346-15.0000	0.2213	4.4736	0.1660	5.9648	0.0905	10.9355

where $\sum_k p_{k,l} = 1$. We then evaluate the generator matrix from $\mathbf{P}_c$ using the *uniformization theorem* (see [21], p. 84), i.e.,

$$\mathbf{T}_c = C \cdot [\mathbf{P}_c - \mathbf{I}(N_c)] = C \cdot \begin{bmatrix} p_{11} - 1 & p_{12} & p_{13} \\ p_{21} & -p_{21} & 0 \\ p_{31} & 0 & -p_{31} \end{bmatrix}^T = (t_{k,l}), \qquad (5.8)$$

where C is a normalizing constant factor and $\mathbf{I}(k)$ is an identity matrix of order k.

From the state transition rates giving in (5.6), we can evaluate the transition

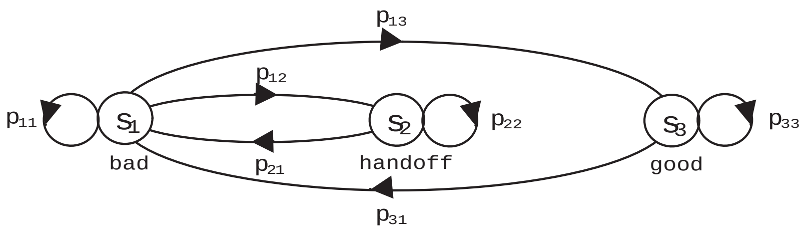

Figure 5.6: State transition rate diagram of a three-state Fritchman model of a time-varying wireless channel.

probabilities $p_{k,l}$ as follows. If the wireless channel has a transmission rate of R_p packets/time unit, then the number of packets transmitted per time unit during the dwell time in state s_k is $R_p(k) = R_p \times p_k$. Due to the fast fading assumption made above, we expect $R_p(k)$ to be smaller than the level crossing rate (5.5). Hence, we can estimate the channel state transition probabilities as [6]

$$p_{k,l} = Pr[S_c(n+1) = s_l | S_c(n) = s_k] \approx (R_p \times p_k)/t_{k,l}. \tag{5.9}$$

Now define the probabilities[7]

$$p_c(t;n) := Pr[\text{wireless channel is in state } s_n \text{ at time } t, \quad t \geq 0,\ n = 1, 2, \cdots, N_c]$$

and arrange these values lexicographically as $\mathbf{p}_c(t) := \{p_c(t;1)p_c(t;2)\cdots p_c(t;N_c)\}^T$. These channel distributions fulfil the first-order system of differential equations

$$d\mathbf{p}_c(t)/dt = \mathbf{T}_c\mathbf{p}_c(t). \tag{5.10}$$

From $\mathbf{T}_c\mathbf{p}_c(\infty) = 0$ and $\sum_{n=1}^{N_c} p_c(\infty;n) = 1$, the equilibrium probabilities of the wireless channel state occupancies can be obtained as

$$\mathbf{p}_c(\infty) = [p_{21}p_{31}\ p_{12}p_{31}\ p_{13}p_{21}]^T/[p_{12}p_{31} + p_{21}(p_{13} + p_{31})].$$

[6]It is assumed that the wireless channel state transition probabilities are independent of the time index n.

[7]In this thesis, the symbol ":=" is used to denote a definition.

Matrices such as $\mathbf{T}_c$ is referred to as *essentially nonnegative matrix* since its elements off the main diagonal are nonnegative. Most of the matrices appearing in this work belongs to this group.

5.3.3 Error Control

Usually, the radio link control (RLC) enhances the network-level QoS via error control since the latter is often unaware of wireless link's imperfections and the wireless channel is barely error-free. Commonly used error control schemes by the RLC sublayer to correct transmissions errors on wireless links are automatic repeat request (ARQ) for closed-loop error control and forward error correction (FEC) for open-loop error control. FEC attempts to correct transmission errors by introducing more redundant bits, while ARQ introduces relatively few redundant bits to only detect possible transmission errors and invoke retransmissions. As such, FEC uses the link capacity (bandwidth) inefficiently whilst ARQ is not well suited for time-critical applications traffic. To trade off between inefficient bandwidth usage due to FEC encoding and a long transmission delay due to ARQ retransmissions, a hybrid FEC/ARQ error control schemes have attracted interest recently. In this case the FEC can be designed for the most frequently occurring error patterns, while the ARQ attempts to correct the less frequent error patterns and the errors that the FEC encoding is unable to correct.

In this chapter, a type-I hybrid automatic-repeat request (ARQ) [64] scheme comprising a FEC and a selective-repeat ARQ (SR-ARQ) codes is used, due to its throughput efficiency. Such a hybrid scheme, especially useful for highly error-prone wireless links, improves the throughput and delay drawbacks in pure ARQ, and the complexity and reliability issues in pure FEC. Analysing the details of the hybrid error control scheme would render the analysis intractable. Hence, for simplicity of analysis, only the mean output efficiency of the hybrid error-control encoder is considered. Assume that an IP packet of variable size L_p is fragmented into N_f blocks of fixed size κ at the RLC and error-control encoded with an L2 header of size $L2hdr$. Hence the L2 frames have the size of $\nu = L2hdr + \kappa$ as illustrated in Fig. 5.7.

Let $(\nu, \kappa, \tau, \hat{r})$ denote a rate κ/ν linear block type-I hybrid ARQ code with

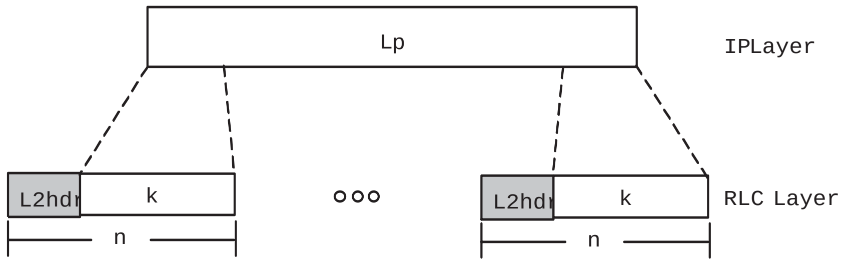

Figure 5.7: Processing of IP packets at the RLC.

a FEC component that can correct up to τ bit errors and an ARQ component which has a maximum of $\hat{r}$ transmissions [37]. Hence, an L2 frame is discarded (lost) if it is still erroneous after $\hat{r}-1$ unsuccessful retransmission attempts. As we have assumed a fast fading wireless channel, the channel causes *random errors* and hence the bit errors can be assumed to occur independently. Thus, L2 frame error rate after the FEC block of the hybrid ARQ code on a random-error channel in state s_n is

$$P_{e,n} = Pr\{\#\text{ bit errors per L2 frame} > \tau | S_c = s_n\} = \sum_{k=\tau+1}^{\nu} \binom{\nu}{k} p_{e,n}^k (1-p_{e,n})^{\nu-k}, \tag{5.11}$$

where $p_{e,n}$ is given in (5.4). Assuming that acknowledgement frames, either positive (ACK) or negative (NACK), are never erroneous, the expected number of SR-ARQ transmissions required to successfully transmit an RLC block when the wireless channel is in state s_n is [36]

$$\bar{N}_n = (1-P_{e,n}) \sum_{k=1}^{\hat{r}} k P_{e,n}^{k-1} = \begin{cases} \lceil 1/(1-P_{e,n}) \rceil, & \hat{r} \to \infty \\ \lceil [1-P_{e,n}^{\hat{r}} - \hat{r}(1-P_{e,n})P_{e,n}^{\hat{r}}]/(1-P_{e,n}) \rceil, & \text{for finite } \hat{r} \end{cases} \tag{5.12}$$

Assuming that the L2 frames are transmitted independently, we obtain the average *coding efficiency* per IP packet of the hybrid ARQ codec when the wireless link is in state s_n as [38]

$$\beta_n(\kappa) := [\kappa/(\bar{N}_n \nu)]^{N_f} = \left[\frac{\kappa}{(\kappa + L2hdr)\bar{N}_n}\right]^{N_f}, \quad n = 1, 2, \cdots, N_c, \tag{5.13}$$

where $\lceil N_f = L_p/\kappa \rceil$ is the number of RLC blocks per IP packet. The optimal RLC block size (κ_{opt}) required for the best $\beta_n(\kappa)$ can be obtained by solving the nonlinear relation $\frac{\partial}{\partial\kappa}\beta_n(\kappa)|_{\kappa=\kappa_{opt}} = 0$.

5.4 NP^{3}A—Channel State Dependent Scheduling

Assume a multiple access scheme where traffic sharing a wireless bandwidth with 'raw' rate R_b bits/sec (or R_p packets/sec) at a cell of a cellular network are classified into c generic traffic streams (GeS) based on e.g. QoS requirements or a form of policy. Let the GeSs be indexed by i, and an GeS$_i$ be associated with a scheduling weight w_i. Assume by convention that if $w_i > w_j$ then GeS$_i$ receives a better scheduling service than GeS$_j$ at the server. The packet scheduling scheme being described is a kind of *channel state-dependent rate proportional scheduling* (CSDPS), as its functionality depends on the wireless link quality (state) of the mobile's link to the corresponding serving access port (FWR), and the amount of service (scheduling rate) traffic of an GeS$_i$ receives is proportional to its scheduling weight w_i. The operation of the NP^{3}A scheme is illustrated pictorially in Fig. 5.8.

Assume that the wireless channel is discretized into N_c states indexed by n as described above. Based on the foregoing discussions, the *effective* rate of the CSDPS for GeS$_i$'s traffic can be written as

$$r_{n,i} = w_i \beta_n R_b, \quad i = 1, 2, \cdots, c \text{ and } n = 1, 2, \cdots, N_c \tag{5.14}$$

Hence, the scheduling rates are time-dependent and varies as a function of the channel BER (bit error rate) and the efficiency of the error control scheme as reflected in β_n [see (5.13)]. The BER, in turn, depends on the modulation format used, as well as the channel noise (fading) level. If we denote $r_n = \beta_n R_b$, then the weights w_i can be interpreted or designed in various ways to yield various schedulers. The next subsection describes how the w_i's for the NP^3A scheme are evaluated.

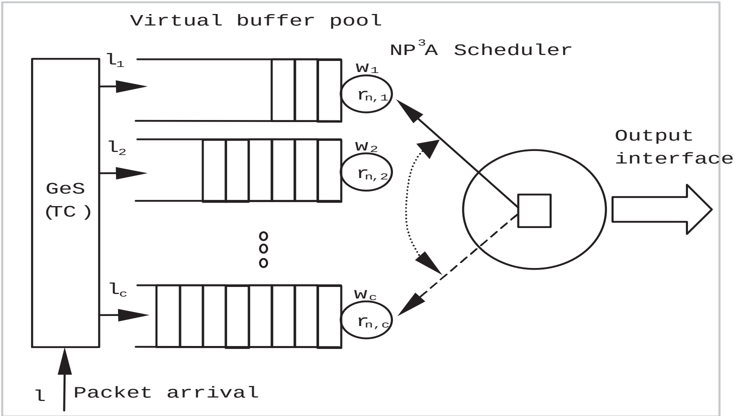

Figure 5.8: Schematic operations of the integrated traffic classifier (TC), class-based buffering, and the channel state dependent rate proportional scheduler (NP^{3}A).

5.4.1 NP^{3}A Scheduling Weights

This scheduling scheme assigns scheduling weights to packets of different GeSs based on their QoS requirements, and it focuses on the long-term (i.e., stationary) cumulative service times rather than instantaneous scheduling priority. The NP^{3}A scheme is described in Section 4.6.

Assume that applications traffic is classified into c QoS groups as underscored above. Without loss of generality, here, we focus our attention on the case where $c = 3$. This permits a one-to-one mapping onto the IETF Diffserv architecture [10, 53, 50], where GeS$_1$, GeS$_2$ and GeS$_3$ map onto best-effort, assured service and expedited service classes, respectively. Assume that GeS$_j$ has non-preemptive priority over GeS$_i$, $\forall j > i$ and that GeS$_i, i = 2, 3$ is assured at least a fraction f_i of the available link bandwidth so long as it is backlogged. Backlogged GeS$_1$, on the other hand, is served iff (if and only if) GeS$_i, i \neq 1$ are not backlogged, albeit with entire bandwidth. Within a queue class, packets are served in a FIFO (first-come, first-served) order. Note, however, that any of the c GeSs, say, GeS$_i$, can be further subdivided into independent classes with assured fraction f_{ij} of its resources dedicated to its subclass GeS$_{ij}$. Under the above assumptions, the weights for the NP^3A scheme can be computed as

$$w_i(NP^3A) = \begin{cases} q_i\tilde{q}_2\tilde{q}_3, & \text{if } i = 1 \\ \tilde{q}_1\tilde{q}_i\tilde{q}_3 f_2 + q_i q_3 f_2 + q_i\tilde{q}_3, & \text{if } i = 2 \\ \tilde{q}_1\tilde{q}_2\tilde{q}_i\tilde{f}_2 + q_2 q_i\tilde{f}_2 + q_i\tilde{q}_2, & \text{if } i = 3 \end{cases} \tag{5.15}$$

where q_i is the fraction of time GeS$_i$ is backlogged, $\tilde{\kappa} = 1 - \kappa$, and f_i is the minimum fraction of resources guaranteed (assured) to GeS$_i$ if backlogged. Hence, the CSDPS scheduling is used in this chapter as a generic scheduler which can degerate to e.g. the NP^{3}A, depending on how we dimension and interprete the weights. Note that the weights can also be made deterministic in contrary to our assumption here.

5.5 Single Source Case

In this section the DQoS schemes presented in the preceding sections are analysed using a single on/off source to model traffic generated by each traffic type.

5.5.1 Traffic Model: Single Source Case

Recent network traffic research widely agree that IP traffic, especially Ethernet LAN, file transfer and VBR video traffic, inhibit burstiness patterns, contradicting the classical Poisson traffic model based on uncorrelated traffic [83]. It is shown that such IP traffic show self-similarity over all time scales [81]. The effects of self-similarity in IP based network traffic on QoS is not yet certain. However, it has been identified that self-similar traffic has a queue length distribution which decays slower than the exponential distribution underlying Poissonian traffic model. The slower decay of queue length can lead to an increased packet losses in IP based networks.

Self-similar pattern in the traffic can be observed through (a) long-range dependence (correlation) of traffic, i.e., non-summable autocorrelation function; (b) diverging spectral density of traffic at the origin; and (c) monotonically increasing index of dispersion counts with sample time [81]. A key parameter indicating self-similarity level in traffic is the Hurst parameter (H). Synonymous expressions used to describe traffic exhibiting self-similarity patterns are: (a)

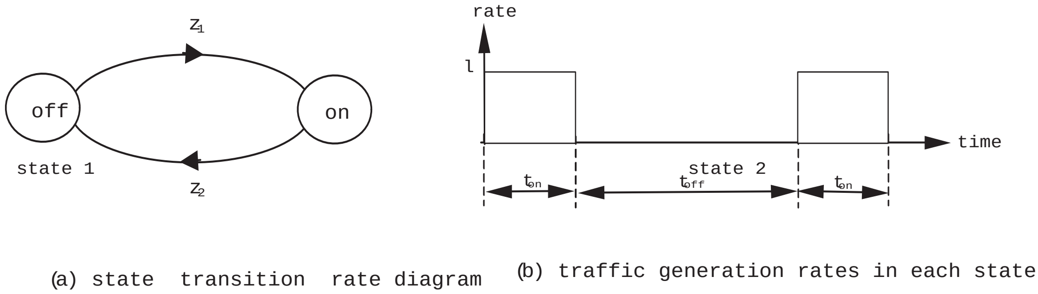

Figure 5.9: A fluid on/off traffic source.

heavy-tailed distributed traffic, (b) bursty traffic, and (c) long-range dependent (LRD) traffic. A common p.d.f. used to describe a self-similar traffic is the heavy-tailed Pareto distribution, which is given by ([81], p. 22)

$$f_X(x) = \mu\vartheta^{\mu}x^{-\mu-1}, \quad \text{for} \ \ \mu, \vartheta \geq 0 \text{ and } \vartheta \leq x < \infty, \tag{5.16}$$

where ϑ and $0 < \mu < 2$ are the location and the shape parameters of the distribution [83]. The Hurst (self-similarity) parameter (H) and μ are related by $H = (1 + \mu)/2$.

Another heavy-tailed distribution used widely in the open literature is the Weibull distribution. A Weibull distributed random process $Y(t)$ has the p.d.f. ([81], p. 234)

$$f_Y(y) = \frac{\alpha}{\beta^{\alpha}} y^{\alpha-1} e^{-(\frac{y}{\beta})^{\alpha}}, \quad \text{for } y \geq 0, \tag{5.17}$$

where $\beta > 0$, $0 < \alpha < 1$ are real numbers referred to as scale and shape parameters, respectively. Note that the exponential distribution is a special case of the Weibull distribution, i.e. when $\alpha = 1$.

In this study, we model the burstiness in IP traffic with an on/off source in which the sojourn times in the on and the off states are assumed to be Pareto and Weibull distributed, respectively. In the on state, the source generates traffic at a constant peak rate of λ packets/unit time, while in the off state the source is idle. The state transition rate diagram and the traffic generation process of the on/off traffic source is illustrated in Fig. 5.9. For a single on/off source with features as illustrated in Fig. 5.9, the state transition rate matrix of the random process underlying its traffic generation is given by

$$\mathbf{T}_s = \begin{bmatrix} -z_1 & z_2 \\ z_1 & -z_2 \end{bmatrix}.$$

Throughout this work we write matrices and vectors in boldfaced font type. The expected dwell times in the on and off states are given by $\bar{t}_{off} = E[t_{off}] = \frac{\beta\alpha}{\alpha-1}, \alpha \neq 1$ (or β for $\alpha = 1$) and $\bar{t}_{on} = E[t_{on}] = \frac{\mu\vartheta}{\mu-1},\ \mu \neq 1$, respectively. Hence, the average traffic generated during an on and an off periods are $A(\bar{t}_{on}) = \lambda\bar{t}_{on} = \frac{\mu\vartheta}{\mu-1}\lambda,\ \mu \neq 1$ and $A(\bar{t}_{off}) = 0\cdot\bar{t}_{off} = 0$. Denote the time period $\bar{t}_p = \bar{t}_{on} + \bar{t}_{off}$ as the on/off *source's cycle time.* Defining the expected fraction of time the source is on (off) as $p_{on} = Pr[\text{source is on}] = \bar{t}_{on}/(\bar{t}_{on} + \bar{t}_{off})$ ($p_{off} = Pr[\text{source is off}] = \bar{t}_{off}/(\bar{t}_{on} + \bar{t}_{off})$), we obtain the total traffic generated in a cycle time as

$$A(\bar{t}_p) = p_{on}A(\bar{t}_{on}) + p_{off}A(\bar{t}_{off}) = \lambda\frac{\mu\vartheta}{(\mu-1)(1+\frac{\mu-1}{\alpha-1}\frac{\beta\alpha}{\mu\vartheta})},\ \alpha, \mu \neq 1. \quad (5.18)$$

Hence, the traffic intensity over a cycle time when the wireless channel is in state s_n for GeS_i[8] is given by

$$\rho_1^{(i)}(n) = \frac{\lambda}{r_{n,i}}\Big[1 + \frac{\mu_i - 1}{\alpha_i - 1}\cdot\frac{\beta_i^{\alpha_i}}{\mu_i\vartheta_i}\Big]^{-2} < 1, i = 1, 2, \cdots, c \text{ and } n = 1, 2, \cdots, N_c. \quad (5.19)$$

The right-hand side of the inequality in (5.19) is required for system stability. To prevent the case of having empty buffers continuously, we may require that $\lambda > r_{n,i}$. In the following, without the abuse of analysis, we require the state transition rates of the on/off source for GeS_i to be

$$z_{k,i} = \frac{1}{E[t_{k,i}]} = \begin{cases} (\alpha_i - 1)\beta_i^{-\alpha_i}, & \text{if } k = 1 \\ (\mu_i - 1)/(\mu_i\vartheta_i), & \text{if } k = 2 \end{cases} \quad (5.20)$$

where the off and the on states are encoded respectively as 1 (state 1) and 2 (state 2), as shown in Fig. 5.9.

[8]It is noted that the index, i, differentiating the traffic classes/types has been sofar suppressed for better clarity of presentation and without ambiguity. Henceforth, it is introduced wherever necessary.

5.5.2 DQoS Analysis: Single Source Case

Given the generator matrices of the random processes underlying the wireless channel variations ($\mathbf{T}_c$) and that of the traffic generation process ($\mathbf{T}_{s,i}$), we can compute the composite generator matrix ($\mathbf{T}_i$) of the two separable stochastic processes of traffic generation and wireless channel state dynamics as

$$\mathbf{T}_i = \mathbf{T}_c \otimes \mathbf{I}(K) + \mathbf{I}(N_c) \otimes \mathbf{T}_{s,i}, \tag{5.21}$$

where $\mathbf{A} \otimes \mathbf{B}$ is the Kronecker tensor product of the matrices $\mathbf{A}$ and $\mathbf{B}$, $\mathbf{I}(k)$ is an identity matrix of order k, and K and N_c are the orders of $\mathbf{T}_{s,i}$ and $\mathbf{T}_c$, respectively. For an $m \times n$ matrix $\mathbf{A}$ and a $p \times q$ matrix $\mathbf{B}$, their Kronecker product is $\mathbf{A} \otimes \mathbf{B} = \{\mathbf{A}(k, l)\mathbf{B}\}$, which has the dimension $mp \times nq$. Hence, (5.21) becomes

$$\mathbf{T}_i = \begin{bmatrix} \mathbf{T}_{1,i} & t_{1,2}\mathbf{I}(2) & t_{1,3}\mathbf{I}(2) \\ t_{2,1}\mathbf{I}(2) & \mathbf{T}_{2,i} & t_{2,3}\mathbf{I}(2) \\ t_{3,1}\mathbf{I}(2) & t_{3,2}\mathbf{I}(2) & \mathbf{T}_{3,i} \end{bmatrix}^T,$$

where $\mathbf{T}_{n,i} = \begin{bmatrix} t_{n,n} - z_{1,i} & z_{1,i} \\ z_{2,i} & t_{n,n} - z_{2,i} \end{bmatrix}$, $\forall\ n = 1, 2, 3$. The two-dimensional stochastic process with the generator matrix $\mathbf{T}_i$ has the vector state space

$$\mathbf{Z} = \{\mathbf{z} | \mathbf{z} = (k, s_n) :\ k = 0, 1, \cdots, K;\ n = 1, 2, \cdots, N_c\},$$

where the first index indicates the number of on/off traffic sources in the 'on' state, while the second index indicates the wireless channel condition (state). For $K = 1,\ N_c = 3,$, we obtain $\mathbf{Z} = \{(0, s_1), (0, s_2), (0, s_3), (1, s_1), (1, s_2), (1, s_3)\}$.

Following [5, 62], let us define the time-independent, steady-state probability distribution for GeS_i as

$$F_{\mathbf{z}}^{(i)}(y) := Pr[\text{buffer content for GeS}_i \le y \mid \text{system is in state } \mathbf{z}, \mathbf{z} \in \mathbf{Z}] \tag{5.22}$$

By defining the column vector of steady-state distributions as

$$\mathbf{F}^{(i)}(y) = \{F_{0,s_1}^{(i)}(y), F_{0,s_2}^{(i)}(y), F_{0,s_3}^{(i)}(y), F_{1,s_1}^{(i)}(y), F_{1,s_2}^{(i)}(y), F_{1,s_3}^{(i)}(y)\}^T,$$

we can evaluate $\mathbf{F}^{(i)}(y)$ via the matrix/vector first-order, differential equation [5]

$$\mathbf{D}_i \frac{d}{dy}\mathbf{F}^{(i)}(y) = \mathbf{T}_i\mathbf{F}^{(i)}(y), \quad y \geq 0 \text{ and } i = 1, 2, \cdots, c, \tag{5.23}$$

where the drift matrix $\mathbf{D}_i$ for GeS_i is a diagonal matrix[9], which is given by

$$\mathbf{D}_i = \mathrm{diag}(-r_{1,i}, -r_{2,i}, -r_{3,i}, \lambda - r_{1,i}, \lambda - r_{2,i}, \lambda - r_{3,i}).$$

The drift matrix $\mathbf{D}_i$ can also be written in its Kronecker form as

$$\mathbf{D}_i = \mathbf{A}_i \otimes \mathbf{I}(N_c) - \mathbf{I}(K) \otimes \mathbf{B}_i, \tag{5.24}$$

where $\mathbf{A}_i = \begin{bmatrix} 0 & 0 \\ 0 & \lambda \end{bmatrix}$ and $\mathbf{B}_i = \begin{bmatrix} r_{1,i} & 0 & 0 \\ 0 & r_{2,i} & 0 \\ 0 & 0 & r_{3,i} \end{bmatrix}$. The *stable solution* to (5.23) is of the form [5]

$$\mathbf{F}^{(i)}(y) = \sum_{k,\ Re\{a_{k,i}\}<0} b_{k,i} e^{a_{k,i}y} \mathbf{v}_{k,i} + b_{\infty,i}\mathbf{F}^{(i)}(\infty), \quad y \geq 0 \text{ and } i = 1, 2, \cdots, c, \tag{5.25}$$

where $\{a_{k,i}, \mathbf{u}_{k,i}, \mathbf{v}_{k,i}\}$ is the set of eigenvalue, left eigenvector and right eigenvector of the system of differential equations (5.23), respectively. That is $a\mathbf{u}_i^T\mathbf{D}_i = \mathbf{u}_i^T\mathbf{T}_i$ and $a\mathbf{D}_i\mathbf{v}_i = \mathbf{T}_i\mathbf{v}_i$, $i = 1, 2, \cdots, c$. The asymptotic solution, $\mathbf{F}^{(i)}(\infty)$, is obtained from the system of linear equations $\mathbf{F}^{(i)}(\infty)\mathbf{T}_i = \mathbf{0}$ and $\left\langle \mathbf{F}^{(i)}(\infty), \mathbf{1} \right\rangle$, where $\mathbf{1}$ is the vector with all elements being 1 and $\langle \mathbf{x}, \mathbf{y} \rangle$ is the scalar product of the two vectors $\mathbf{x}$ and $\mathbf{y}$. The constant coefficients b_k are evaluated in Section 5.6.3.

If over time the cumulative scheduling rate to the queue of GeS_i falls below the cumulative traffic generation rate fed to the buffer, the radio node stores the excess packets. Since the radio nodes' buffer has a finite capacity, the queue length can exceed the buffer capacity, leading to packet losses. The steady-state packet loss rate due to buffer overflow, $L_o^{(i)}(y)$, can be obtained as

[9] We denote a diagonal matrix whose (main) diagonal elements are the elements of the vector, say, $\mathbf{u}$ as $\mathrm{diag}(\mathbf{u})$.

$$L_o^{(i)}(y) = Pr\{\text{buffer content for GeS}_i > y\} = 1 - \left\langle \mathbf{F}^{(i)}(y), \mathbf{1} \right\rangle, \; i = 1, 2, \cdots, c. \tag{5.26}$$

If we use the simple FCFS scheduling order (see Section 2.10.1) within a logical queue, then we obtain the average packet queuing delay at the transmitter for GeS$_i$'s traffic as

$$D_i(y) = (y+1) \sum_{\forall \mathbf{Z},\ \mathbf{Z} \in \mathbf{Z}} F_{\mathbf{Z}}^{(i)}(y)/r_{\mathbf{Z},i}, \; i = 1, 2, \cdots, c, \tag{5.27}$$

5.5.3 Spectral Analysis

To solve (5.23), let the set $\{a, \mathbf{u}_i, \mathbf{v}_i\}$ be the eigenvalue and the eigenvectors as defined in (5.25), respectively. It then follows that

$$a\mathbf{u}_i^T \mathbf{D}_i = \mathbf{u}_i^T \mathbf{T}_i, \; i = 1, 2, \cdots, c. \tag{5.28}$$

$$a\mathbf{D}_i \mathbf{v}_i = \mathbf{T}_i \mathbf{v}_i, \; i = 1, 2, \cdots, c. \tag{5.29}$$

From (5.28), the eigenvalues (and then the left eigenvectors) can be obtained from the zeros of the function

$$f(a) = det[a\mathbf{I} - \mathbf{Q}_i], \tag{5.30}$$

where $\mathbf{Q}_i = \mathbf{D}_i^{-1}\mathbf{T}_i$. By noting that the inverse of a diagonal matrix is also a diagonal matrix with each element inverted, we obtain

$$\mathbf{D}_i^{-1} = \text{diag}\Big[-\frac{1}{r_{1,i}}, -\frac{1}{r_{2,i}}, -\frac{1}{r_{3,i}}, \frac{1}{\lambda - r_{1,i}}, \frac{1}{\lambda - r_{2,i}}, \frac{1}{\lambda - r_{3,i}}\Big].$$

Hence, we obtain

$$\mathbf{Q}_i = \begin{bmatrix} -\frac{t_{11}-z_1}{r_{1,i}} & -\frac{z_2}{r_{1,i}} & -\frac{t_{21}}{r_{1,i}} & 0 & -\frac{t_{31}}{r_{1,i}} & 0 \\ -\frac{z_1}{r_{2,i}} & -\frac{t_{11}-z_2}{r_{2,i}} & 0 & -\frac{t_{21}}{r_{2,i}} & 0 & -\frac{t_{31}}{r_{2,i}} \\ -\frac{t_{12}}{r_{3,i}} & 0 & -\frac{t_{22}-z_1}{r_{3,i}} & -\frac{z_2}{r_{3,i}} & -\frac{t_{32}}{r_{3,i}} & 0 \\ 0 & \frac{t_{12}}{\lambda-r_{1,i}} & \frac{z_1}{\lambda-r_{1,i}} & \frac{t_{22}-z_2}{\lambda-r_{1,i}} & 0 & \frac{t_{32}}{\lambda-r_{1,i}} \\ \frac{t_{13}}{\lambda-r_{2,i}} & 0 & \frac{t_{23}}{\lambda-r_{2,i}} & 0 & \frac{t_{33}-z_1}{\lambda-r_{2,i}} & \frac{z_2}{\lambda-r_{2,i}} \\ 0 & \frac{t_{13}}{\lambda-r_{3,i}} & 0 & \frac{t_{23}}{\lambda-r_{3,i}} & \frac{z_1}{\lambda-r_{3,i}} & \frac{t_{33}-z_2}{\lambda-r_{3,i}} \end{bmatrix}.$$

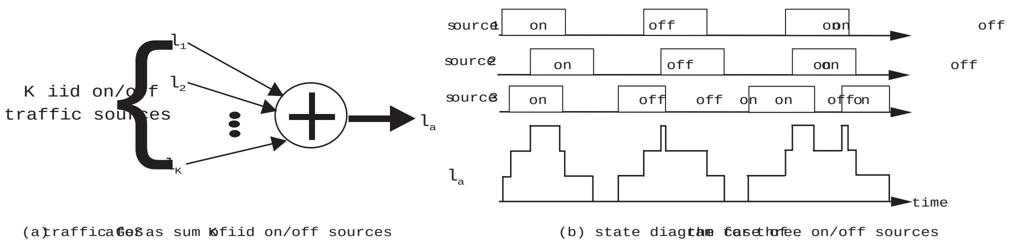

Figure 5.10: Bursty traffic arrivals modelled as aggregates of K iid on/off sources.

Plugging $\mathbf{Q}_i$ into (5.30) and (5.29) the eigenvalues and the eigenvectors can be evaluated using methods such as given in, for example, [71, 61].

5.6 Multiple Source Case

Suppose c generic traffic streams (GeSs), $1, 2, \cdots, c$, share a wireless channel with data rate R_b in a cell of a mobile cellular network. Assume that the wireless bandwidth is shared among the c GeS in a cell and that the MAC layer arbitrates between the transmissions of multiple users using the CSDPS scheduling scheme described in Section 5.4. The CSDPS scheduler is coupled to a buffer manager, which controls the packet losses. Here, we consider the simple tail dropping (TD) buffer management scheme.

5.6.1 Traffic Model: Multiple Source Case

We model traffic arrivals from GeS$_i$ as aggregates of K ON/OFF sources with statistical properties as described in Section 5.5.1. In the OFF state no traffic is generated. In the ON state, source j of GeS$_i$ generates traffic according to a given random process with mean rate λ. We assume that the sources of each GeS generate traffic independently, but obey the same statistical distributions, i.e., i.i.d. sources (see Section 5.5.1). Hence, traffic from GeS$_i$ has the mean rate of $Ns\lambda$, as shown in Fig. 5.10. We thus attempt to model each GeS$_i$ as a bursty traffic with maximum burst size $Ns\lambda$.

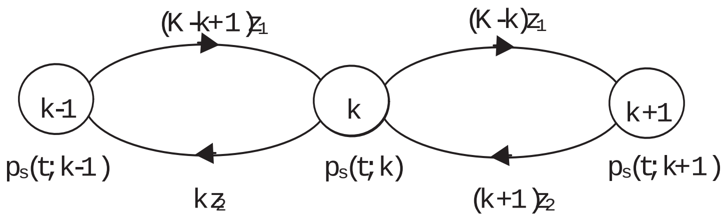

Figure 5.11: State transition rate diagram of the number of "on" sources.

Also, define the probability

$$p_s(t;k) := Pr[k \text{ sources are on simultaneously at time } t],\ k = 0, 1, \cdots,\ K$$

and arrange these quantities lexicographically into the vector

$$\mathbf{p}_s(t) := \{p_s(t;0)\ p_s(t;1)\ \cdots\ p_s(t;K)\}^T.$$

Following the definitions in Section 5.5.1, we can obtain the steady-state probability that k out of the K i.i.d. on/off sources of GeS$_i$ is on as

$$P_s(k) := \lim_{t\to\infty} p_s(t;k) = \frac{1}{(1+\eta_s)^K}\binom{K}{k}\eta_s^k, \quad 0 \le k \le K, \tag{5.31}$$

and compile these probabilities into the vector $\mathbf{P}_s := \{P_s(0)\ P_s(1)\ \cdots\ P_s(K)\}^T$, where $\eta_s = \bar{t}_{on}/\bar{t}_{off}$. The instantaneous traffic intensity in the multiple source case is then $\rho_k^{(i)}(n) = k\lambda P_s(k)/r_{n,i}$. Note that $\mathbf{P}_s$ is normalized, i.e., $\langle \mathbf{P}_s, \mathbf{1}\rangle = 1$. Now, let $\mathbf{T}_s$ denote the generator matrix of the stochastic process underlying the number of sources that are on simultaneously. Figure 5.11 (see also [5, 71]) illustrates the number of simultaneous on sources as a Markov chain with only nearest-neighbour transitions. From this figure we can formulate the balance system of equations

$$(K-i+1)z_1p_s(t;k-1)-[kz_2+(K-k)z_1]p_s(t;k)+(k+1)z_2p_s(t;k+1) = 0, \tag{5.32}$$

where $0 \le k \le K,\ t \ge 0$, and by convention $p_s(t;m) = 0,\ \forall m \notin \{0, 1, \cdots, K\}$. Equation (5.32) allows us to obtain the generator matrix underlying the traffic generation process as

$$\mathbf{T}_s = \begin{bmatrix} -N_s z_1 & N_s z_1 & & & & & \\ z_2 & -[z_2+(K-1)z_1] & (K-1)z_1 & & & & \\ & 2z_2 & -[2z_2+(K-2)z_1] & (K-2)z_1 & & & \\ & & \ddots & \ddots & \ddots & & \\ & & & (K-1)z_2 & -[(K-1)z_2+z_1] & z_1 \\ & & & & N_s z_2 & -N_s z_2 \end{bmatrix}^T .$$

The matrix $\mathbf{T}_s$ relates the vector $\mathbf{p}_s(t)$ according to

$$\frac{d}{dt}\mathbf{p}_s(t) = \mathbf{T}_s \mathbf{p}_s(t). \tag{5.33}$$

5.6.2 Mathematical Analysis: Multiple Source Case

Here, we analyse the unified system accounting for buffer dynamics, wireless channel state (condition) variations and the traffic source dynamics. The vector state space listings for the multiple source case is

$$\mathbf{Z} := \{\mathbf{z}\} = \{(0,s_1),(0,s_2),(0,s_3),(1,s_1),(1,s_2),(1,s_3),\cdots,(K,s_1),(K,s_2),(K,s_3)\}.$$

Let $p_{\mathbf{Z}}(t;y)$ be the joint probability that at a given time instant t the wireless channel is in state s_n, when there are k traffic sources on simultaneously and the buffer occupancy does not exceed y, $0 \le y \le Y$, where Y is the buffer capacity. Now, let the vector $\mathbf{p}_{\mathbf{Z}}(t;y)$ be the lexicographic arrangements of the $p_{\mathbf{Z}}(t;y)$'s, i.e.

$$\mathbf{p}_{\mathbf{Z}}(t;y) = \{p_{0,s_1}(t;y)\; p_{0,s_2}(t;y)\; p_{0,s_3}(t;y)\; p_{1,s_1}(t;y)\; \cdots\; p_{K,s_2}(t;y)\; p_{K,s_3}(t;y)\}^T,$$

and define the stationary version $\mathbf{F}(y) = \lim_{t\to\infty} \mathbf{p}_{\mathbf{Z}}(t;y)$ which means $F_{\mathbf{Z}}(y) = \lim_{t\to\infty} p_{\mathbf{Z}}(t;y)$. Then, the following time-dependent differential equation holds

$$\frac{\delta}{\delta t}\mathbf{p}_{\mathbf{Z}}(t;y) + \mathbf{D}\frac{\delta}{\delta y}\mathbf{p}_{\mathbf{Z}}(t;y) = \mathbf{T}\mathbf{p}_{\mathbf{Z}}(t;y), \quad t \ge 0,\ 0 \le y \le Y, \tag{5.34}$$

where $\mathbf{T}$ is obtained as

$$\mathbf{T} = \mathbf{T}_s \otimes \mathbf{I}(N_c) + \mathbf{I}(K+1) \otimes \mathbf{T}_c \tag{5.35}$$

yielding

$$\mathbf{T} = \begin{bmatrix} \mathbf{A}_1 & N_s z_1\mathbf{I} & & & & & \\ z_2\mathbf{I} & \mathbf{A}_2 & (K-1)z_1\mathbf{I} & & & & \\ & 2z_2\mathbf{I} & \mathbf{A}_3 & (K-2)z_1\mathbf{I} & & & \\ & & \ddots & \ddots & \ddots & & \\ & & & (K-1)z_2\mathbf{I} & \mathbf{A}_K & z_1\mathbf{I}) & \\ & & & & N_s z_2\mathbf{I} & \mathbf{A}_{K+1} & \end{bmatrix}^T,$$

where $\mathbf{I} = \{\delta_{kl}\}_{k,l=1,2,\cdots}$ is an identity matrix of the same order as $\mathbf{T}_c$, δ_{kl} is the Kronecker product, and $\mathbf{A}_j = \mathbf{T}_c - [(K+1-j)z_1 + (j-1)z_2]\mathbf{I}$. The square drift matrix [71] of order $(K+1)N_c$ has a diagonal structure with the diagonal elements

$$D_{kN_c+n} = k \cdot \lambda - r_n, \quad 0 \le k \le K,\ n = 1, 2, \cdots, N_c(=3), \tag{5.36}$$

which denotes the change in buffer content (however infinitesimal it is) when k of the traffic sources are on and the wireless channel is in state s_n. From the partial differential equations (5.34) we obtain the steady-state solution from the following total differential equations [see also (5.23)]

$$\mathbf{D}\frac{d}{dy}\mathbf{F}(y) = \mathbf{TF}(y), \quad 0 \le y \le Y. \tag{5.37}$$

Similarly to (5.23), the *stable solution* to (5.37) can be formulated as

$$\mathbf{F}(y) = \sum_{k,\ \Re\{a_k\}<0} b_k e^{a_k y}\mathbf{v}_k + b_\infty \mathbf{F}(\infty), \quad 0 \le y \le Y < \infty, \tag{5.38}$$

where $\mathbf{v}_k$ is the right eigenfunction corresponding to the eigenvalue[10] $a_k \in \sigma(\mathbf{TD}^{-1})$, and the constant coefficients b_k are evaluated in Section 5.6.3. Again, the asymptotic solution $\mathbf{F}(\infty)$ is obtained from the set of equations $\mathbf{TF}(\infty) = \mathbf{0}$ and $\langle \mathbf{F}(\infty), \mathbf{1} \rangle$, where $\mathbf{1}$ is a vector with all elements being 1. Note that $\mathbf{F}(\infty)$ is the same as $\mathbf{P}_s$ in (5.31). For the spectral analysis, methods such as those presented in e.g. [71, 5] can be used.

[10] The set of eigenvalues of a matrix $\mathbf{A}$, denoted $\sigma(\mathbf{A})$, is referred to as the spectrum of $\mathbf{A}$.

From the preceding argument, the packet loss probability due to buffer overflow at the transmitting node is given by

$$L_o(y) = 1 - \langle \mathbf{F}(y), \mathbf{1} \rangle, \quad 0 \le y \le Y < \infty. \tag{5.39}$$

If we use FCFS scheduling order within a logical queue, then we obtain the average packet queuing delay at the transmitter for GeS$_i$'s traffic as

$$D(y) = (1+y) \sum_{\forall \mathbf{z},\ \mathbf{z} \in \mathbf{Z}} F_{\mathbf{z}}(y)/r_{\mathbf{z}}, \quad 0 \le y \le Y < \infty, \tag{5.40}$$

5.6.3 Evaluation of b_k in (5.25) and (5.38)

In this work we focus our attention only on the stable solutions to (5.23) and (5.37) as given in (5.25) and (5.38). Let L be the number of eigenvalues of $\mathbf{D}_i^{-1}\mathbf{T}_i$ (or simply, $\mathbf{D}^{-1}\mathbf{T}$)[11] with negative real parts, and $v_l(k, s_n)$ be the (k, s_n)th element of the eigenvector $\mathbf{v}_l$ corresponding to the eigenvalue a_l, $l = 1, 2, \cdots, L$. We remind ourselves that $(k, s_n) \in \mathbf{Z}$ as defined in Section 5.5.2.

Let the b_k's in (5.25) and (5.38) be arranged in the vector form $\mathbf{b} = (b_1, b_2, \cdots, b_L)^T$. For the infinite buffer capacity case the coefficient corresponding to the zero eigenvalue ($a_l = 0$) is $b_\infty = 1$ as $\lim_{y\to\infty} \mathbf{F}^{(i)}(y) \to \mathbf{F}^i(\infty)$. Following Propositions 6.1 and 6.2 in [71], $\mathbf{b}$ can be obtained from the relation[12]

$$\mathbf{b} = -(\mathbf{V}_m^T \mathbf{V}_m)^{-1} \mathbf{V}_m^T \mathbf{F}_m(\infty), \tag{5.41}$$

where $\mathbf{F}_m(\infty) \in \Re^{(K+1-k_+)N_c \times 1}$ contains the part of the vector $\mathbf{F}(\infty)$ for which the buffer drift is positive, i.e., $D_{k_+N_c+n} = k_+\lambda - r_n > 0$[13] and $\mathbf{V}_m \in \Re^{(K+1-k_+)N_c \times L}$ is a matrix of eigenvectors corresponding to the stable eigenvalues a_l. For our case where $N_c = 3$ we obtain

$$\mathbf{F}_m(\infty) = [F_{k_+,s_1}(\infty)\ F_{k_+,s_2}(\infty)\ F_{k_+,s_3}(\infty)\ F_{k_++1,s_1}(\infty)\ \cdots\ F_{K,s_1}(\infty)\ F_{K,s_2}(\infty)\ F_{K,s_3}(\infty)]^T$$

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[11] In some part of this work the index i differentiating the c traffic classes is omitted to reduce the symbology if there is no ambiguity.

[12] Note that, for a non-square matrix $\mathbf{V}$, the equation $\mathbf{Vb} = \mathbf{d}$ can be solved via the modified equation $\mathbf{V}^T\mathbf{Vb} = \mathbf{V}^T\mathbf{d}$.

[13] This means that the buffer drift is positive so long as k_+ of the K on/off traffic sources are 'on'. For the single source case $k_+ = 1$ in order to prevent empty buffers continuously.

and

$$\mathbf{V}_m = [v_{ij}]_{(K+1-k_+)N_c \times L} = \begin{bmatrix} v_1(k_+, s_1) & v_2(k_+, s_1) & \cdots & v_L(k_+, s_1) \\ v_1(k_+, s_2) & v_2(k_+, s_2) & \cdots & v_L(k_+, s_2) \\ v_1(k_+, s_3) & v_2(k_+, s_3) & \cdots & v_L(k_+, s_3) \\ \vdots & \vdots & & \vdots \\ v_1(K, s_1) & v_2(K, s_1) & \cdots & v_L(K, s_1) \\ v_1(K, s_2) & v_2(K, s_2) & \cdots & v_L(K, s_2) \\ v_1(K, s_3) & v_2(K, s_3) & \cdots & v_L(K, s_3) \end{bmatrix}.$$

The coefficients $\mathbf{b}$ as computed in (5.41) are obtained by noting that the stationary buffer content cannot be zero for a positive drift, i.e.

$$D_{kN_c+n} = k\lambda - r_n > 0 \Rightarrow F_{k.s_n}(y)|_{y=0} = 0.$$

Plugging these conditions into (5.25) or (5.38) yields the system of L linear equations

$$\mathbf{0} = b_\infty \mathbf{F}_m(\infty) + \sum_{l=1}^{L} b_l \mathbf{v}_{m,l}$$

from which $\mathbf{b}$ can be obtained as in (5.41), where the $\mathbf{v}_{m,l}$'s are the column vectors of $\mathbf{V}_m$ in (5.41).

5.7 Results and Discussions

Packet sizes (L_p) used for the numerical examples are assumed to be variable and are estimated from the probability mass function in (2.28). Layer 3 packets generated from the distribution in (2.28) are error control encoded at Layer 2. A BCH code with maximum parity check bits (redundancy) is used for the FEC component of the hybrid ARQ encoder. Hence, we obtain the codeword of size $\nu = L_p + L2hdr = 2^l - 1$ with the data part of size $\kappa = L_p = \nu - l \cdot \tau$, where l is the number bits per codeword. Given the wireless link bit rate R_b b/sec and packets of variable length L_p bytes, the packet transmission rate is $R_p = R_b/(8L_p)$ packets/sec. Unless otherwise stated, the values of the system parameters used for the numerical investigations are compiled in Table 5.2.

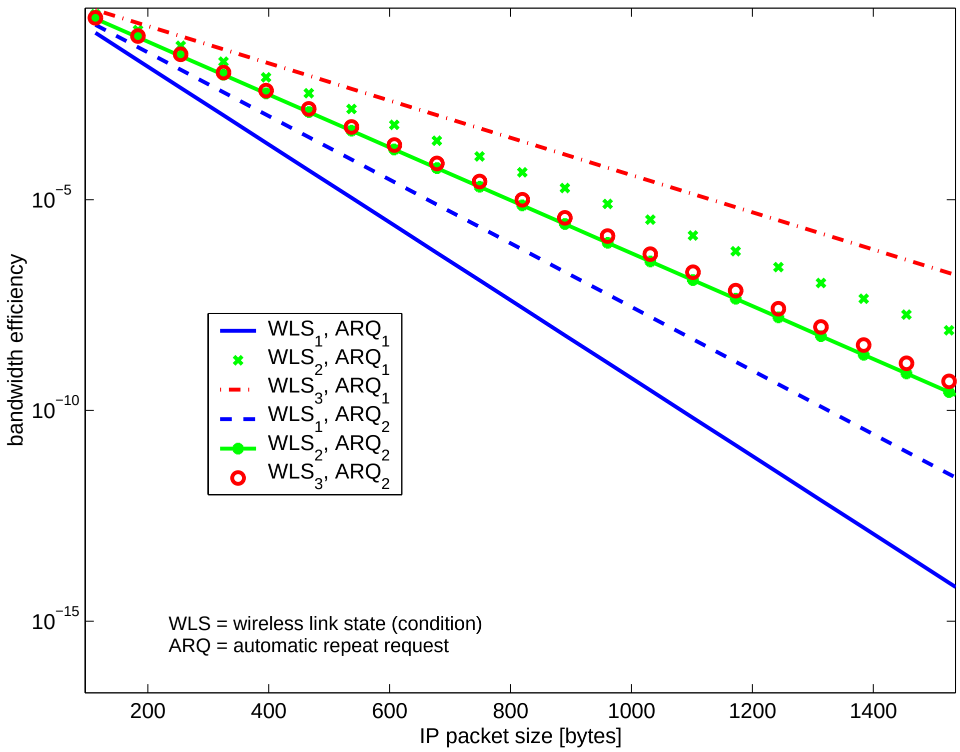

Figure 5.12: Bandwidth efficiency as a function of IP packet size with the wireless link state (WLS), τ and $\hat{r}$ as parameters; for ARQ$_1$: $\tau = 2$, $\hat{r} = 5$; ARQ$_2$: $\tau = 3$, $\hat{r} = 5$ [$\kappa = 12$ bytes, $L2hdr = \tau log_2(\nu + 1)$, $\rho = 8$ dB, and $\gamma \in \{0, 4\}$ dB].

Figure 5.12 shows the bandwidth (coding) efficiency β as a function of the IP packet size. The following intuitively appealing observations can be observed in the figure: (1) β decreases with the IP packet size, as the longer the packet the higher the chance that transmission bit errors occur; (2) A better channel quality yields a higher β; (3) For a bad wireless channel, FEC with a higher number of parity check bits outperforms a FEC with fewer redundant bits; this can be observed in the solid and the dashed curves; however, a FEC with lower coding overhead offers a better bandwidth utilization than that with larger redundancy on a good wireless channel.

The expected per-packet loss probabilities as functions of the buffer capacity of the server (e.g. base station) are plotted in Figures 5.13, 5.14, 5.15, 5.16 and

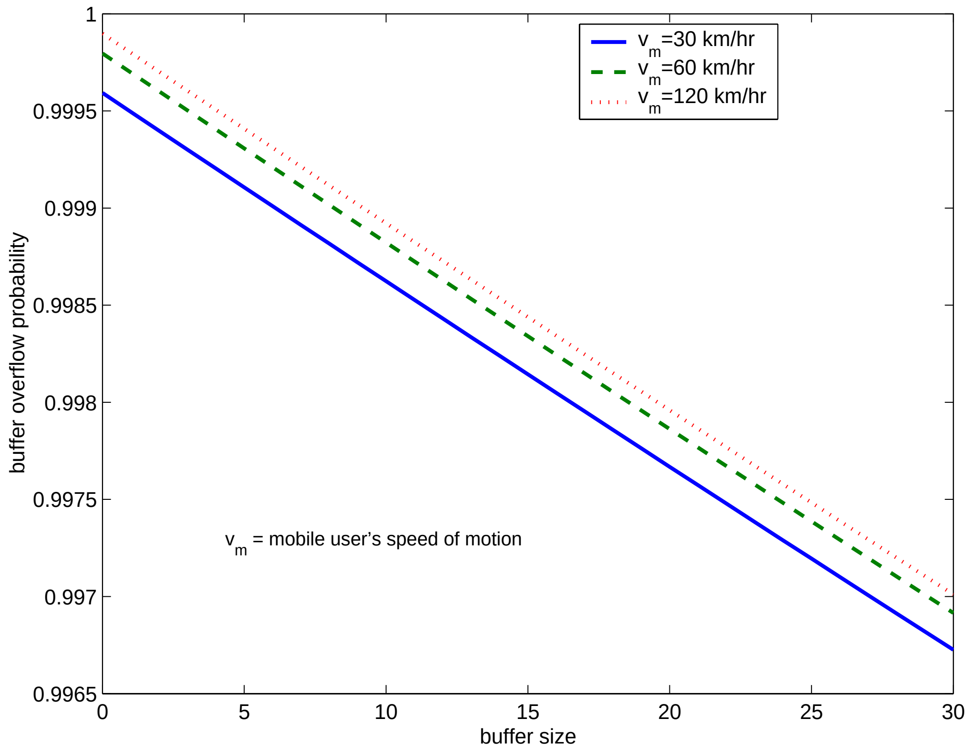

Figure 5.13: Packet loss probability versus buffer capacity: effects of user mobility on traffic QoS.

5.17. Figure 5.13 uses the speed of motion of the service user as a parameter. It can be observed that the loss rate degrades with the speed of motion. Hence, a slowly moving user may experience a degraded communications quality if he increases his mobility scenario. The wireless related parameter examined in Fig. 5.14 is the received signal-to-noise ratio (SNR), which depends on the radio link's noise (fading) level. It is observed that the packet overflow rate decreases with increasing SNR.

Figures 5.15 and 5.16 examine the effects of Layer 2 error control scheme on packet low rate. It is observable in Fig. 5.15 that the loss rate degrades with decreasing number of FEC parity check bits. This may be interpreted as follows: fewer parity check bits means that fewer bit error per packet may be correctable by the FEC code. Hence, more erroneous packets need to be retransmitted by the ARQ component of the hybrid ARQ code. As more blocks

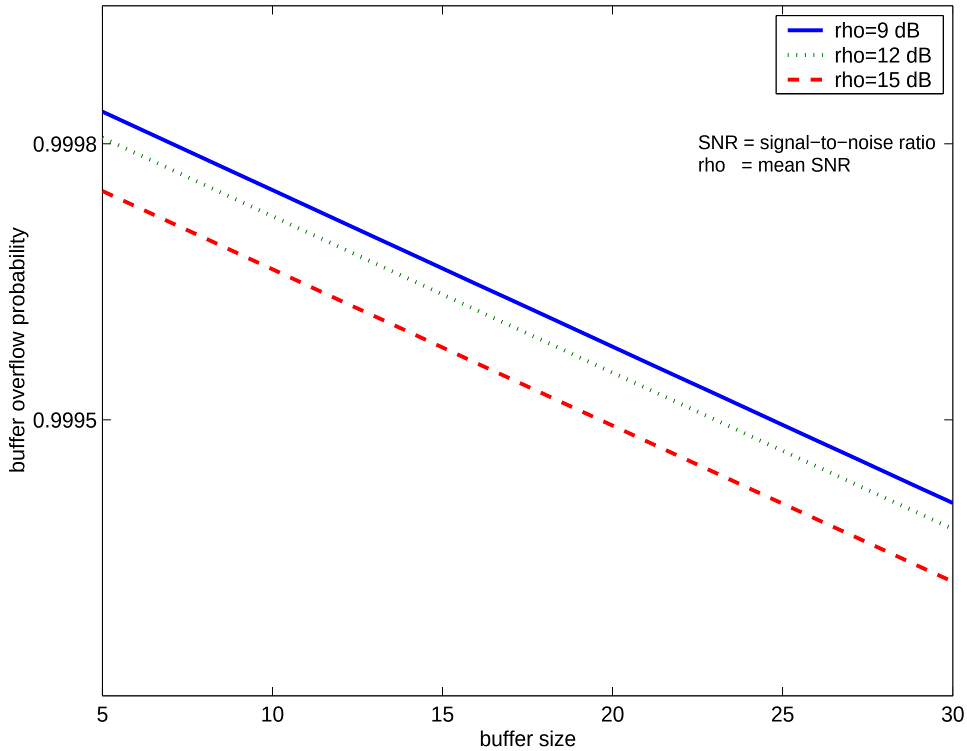

Figure 5.14: Packet loss probability versus buffer capacity: effects of radio fading level on traffic QoS.

require retransmissions, the probability of the buffer filling up increases resulting in higher packet losses. Fig. 5.16 indicates that the packet loss rate increases with increasing number of ARQ retransmissions. This follows from the above discussion, where it is indicated that more retransmission per RLC block (which is part of an IP packet) results in higher chance of buffer overflow.

In Figs. 5.17 and Fig. 5.18 we show the performance of the NP^{3}A scheduler, which differentiates between the traffic from three generic streams (i.e., $c = 3$). By convention we assume in the entire study that GeS$_i$ has or requires a better service (QoS) than GeS$_j$ where $i > j$. Hence, GeS$_3$ is the most delay sensitive class (see Fig. 5.18), while GeS$_1$ can be classified as the best-effort class. It is clear that GeS$_2$ compares well with an assured service since it is always guaranteed a bandwidth portion f_2 so long as it is backlogged. The argument behind Figs. 5.17 and Fig. 5.18 is that traffic can be either delay sensitive or

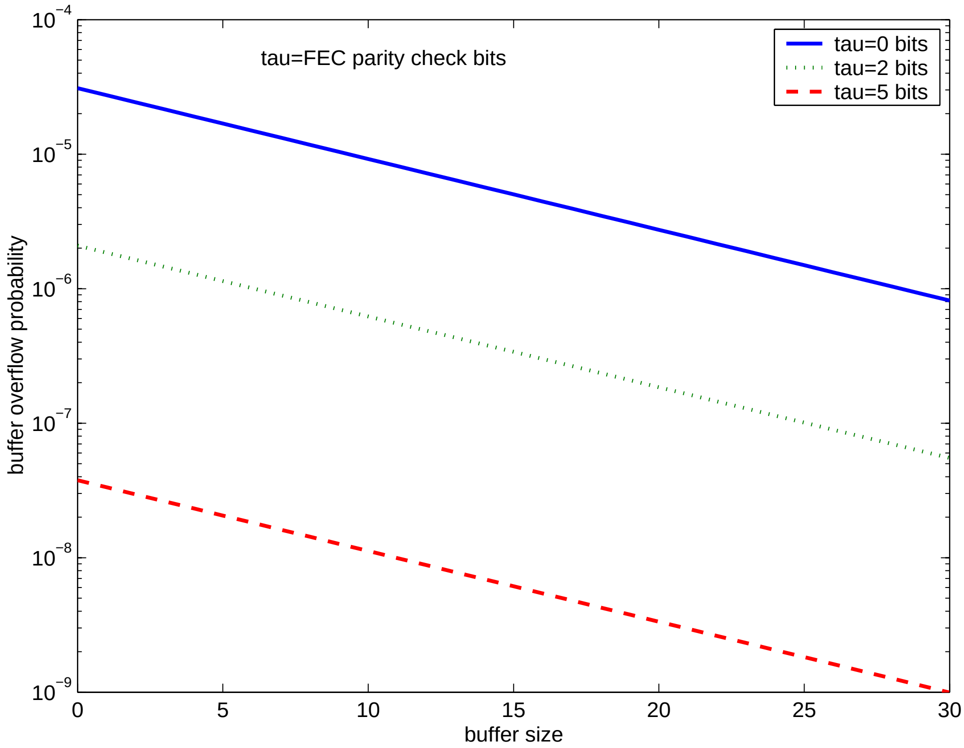

Figure 5.15: Average packet loss probability versus buffer capacity: effects of error control coding (FEC redundancy) on QoS.

loss sensitive. Hence, traffic belonging to GeS_3 experiences the lowest queuing delay but the highest packet loss ratio, while GeS_1 undergoes the best loss rate but the least scheduling priority. GeS_3 can be traffic from strictly delay intolerant applications such as video telephony, VoIP, videoconferencing, control and signalling messages and e-banking (mission critical) traffic. GeS_1 can serve the conventional best-effort forwarding traffic of today's Internet, which includes background applications such as email and file transfers (ftp) [34]. Traffic belonging to GeS_2, on the other hand, is less time-critical but requires an assured minimum service rate.

Figures 5.19 and 5.20 investigate the delay versus the buffer capacity whilst using the mobile user's speed (v_m) and the mean SNR as parameters. Fig. 5.19 indicates the user mobility enhances the packet delay, contrasting the observed effects of mobility on buffer overflow. Thus the higher the user speed of motion,

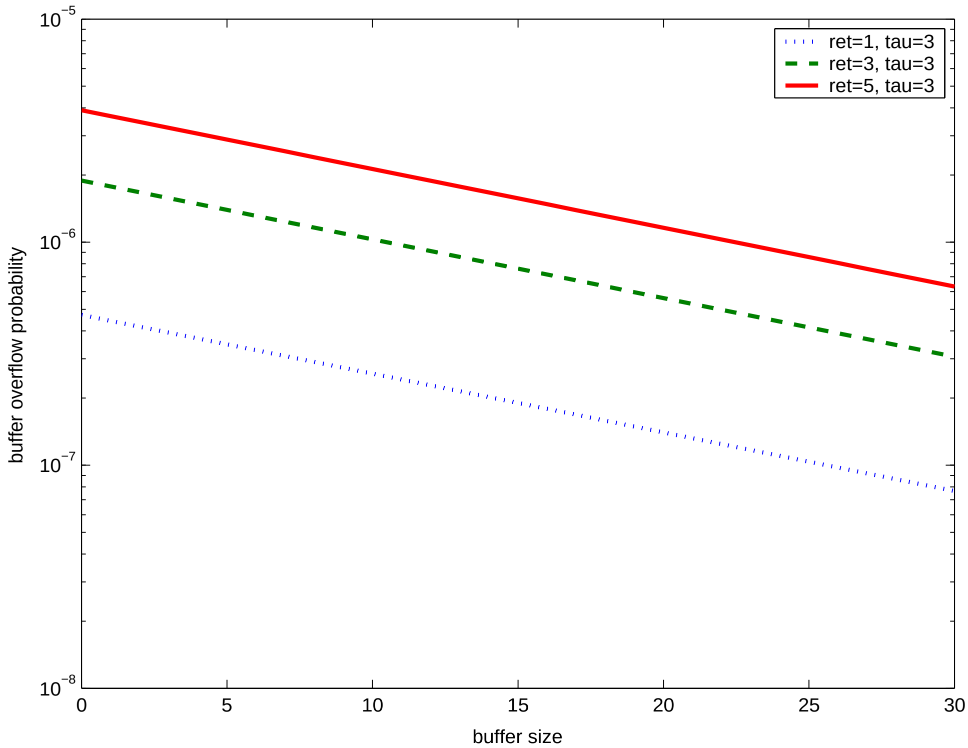

Figure 5.16: Packet loss probability versus buffer capacity: effects of error control coding (RLC retransmissions) on QoS [ret=max. number of ARQ retransmissions, tau (τ)=FEC parity check bits].

the lower the observable queuing delay. The SNR ratio also shows a remarkable thing, as low SNR offers better delay, which contrasts its dependence on the packet loss.

5.8 Summary

This study uses analytic modelling to investigate the effects of inherent wireless properties on traffic quality of service (QoS). The QoS metrics considered are the mean packet queuing delay at a server (e.g. radio base station in a cellular network) and the server's buffer overflow probabilities. Wireless properties accounted for are the user mobility, radio fading level in form of the received mean signal-to-noise ratio, and error control overhead (i.e., number of allowable

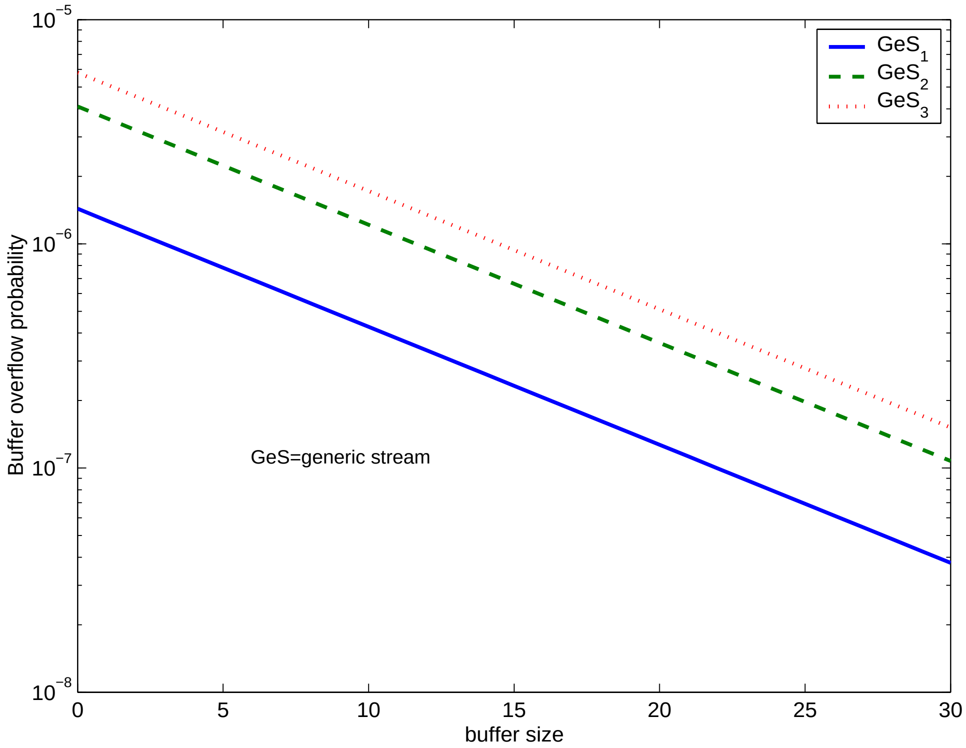

Figure 5.17: Expected packet loss probability versus buffer capacity for packets of three different GeSs.

ARQ retransmissions and FEC parity check bits). This study will complement the many simulation studies reported on similar issues in the open literature. For instance, the loss profile can aid in the dimensioning of buffers of network nodes.

Some of the results confirm informed intuition. However, it has still proved valuable to analyse them analytically, as such analytic studies are not common in the literature. The well known results that delay increases with buffer capacity, but the packet loss decreases with increasing buffer capacity are also verified thereby, and most importantly, used as a check on the validity of the analytical modelling. It is observed that the packet losses

- increase with increasing user mobility,
- decrease with increasing received SNR,

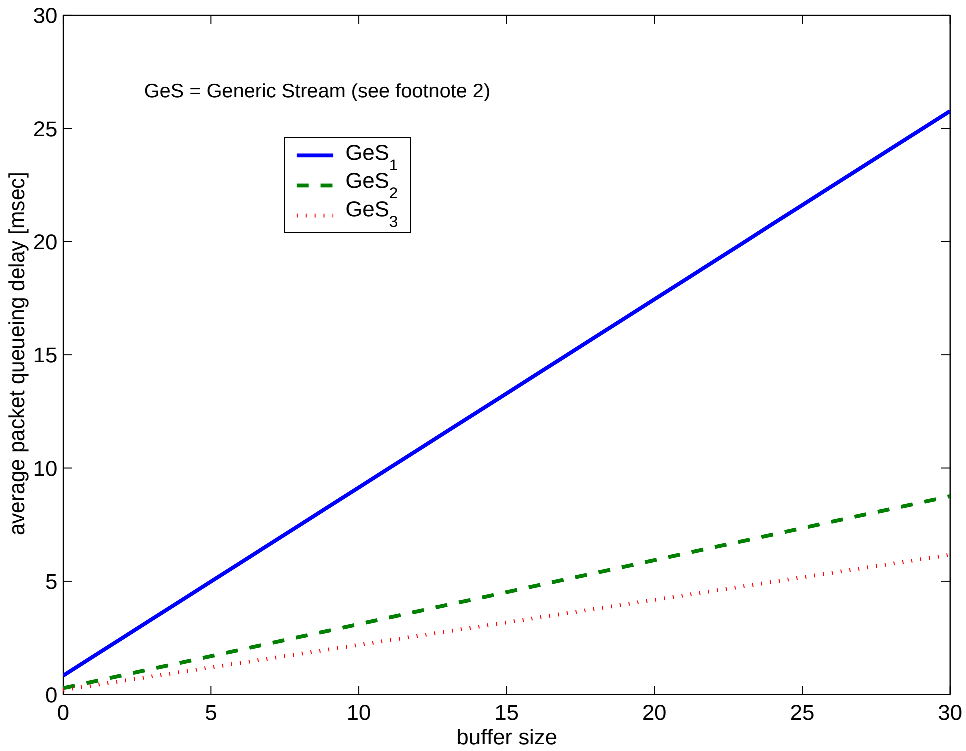

Figure 5.18: Expected packet queuing delay versus buffer capacity for packets of three different GeSs.

- decrease with increasing FEC redundancy, and
- increase with increasing maximum number of ARQ retransmissions

when all other parameters are held the same. The packet queuing delay, on the other hand, decreases with increasing user mobility but increases with increasing received SNR.

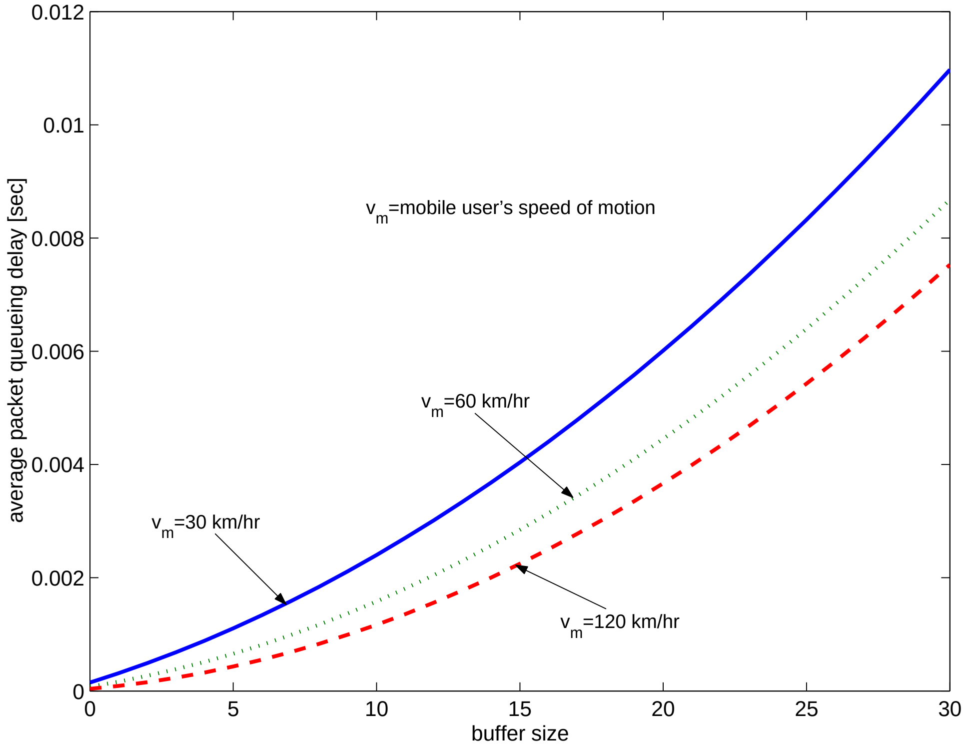

Figure 5.19: Expected packet queuing delay versus buffer capacity: effects of user mobility on traffic QoS.

Table 5.2: Parameters for the numerical investigations.

Parameter	Value
Link level parameters	
speed of light in vacuum	$v_0 = 3 \times 10^8$ m/s
speed of mobile user	$v_m = 30, 60, 120$ km/hr
operating radio frequency	$f_c = 2$ GHz
mean SNIR	$\rho = 9,\ 12\ \&\ 15$ dB
received SNIR	$0 \leq \gamma \leq 6$ dB
wireless interface bit rate	$R_b \geq 144$ kb/s
RLC block size	$\kappa = 12$ bytes
number of FEC parity check bits	$\tau = 0 - 5$ bits
max. # of ARQ retransmissions	$\hat{r} = 0 - 5$
Traffic parameters	
IP packet size [variable, see Eq. (2.28)]	$96 \leq L_p \leq 1536$ bytes
number of GeS (traffic classes)	$c = 3$
fraction of available bandwidth guaranteed to GeS_2	$f_2 \geq 0.3$
shape parameter of Pareto distribution	$\mu = 1.2$
location parameter of Pareto distribution	$\eta = 1.5$
shape parameter of Weibull distribution	$\alpha = 0.5$
scale of Weibull distribution	$\beta = 2$
traffic generation rate per on/off source	$\lambda = 20 \times R_b/L_p$

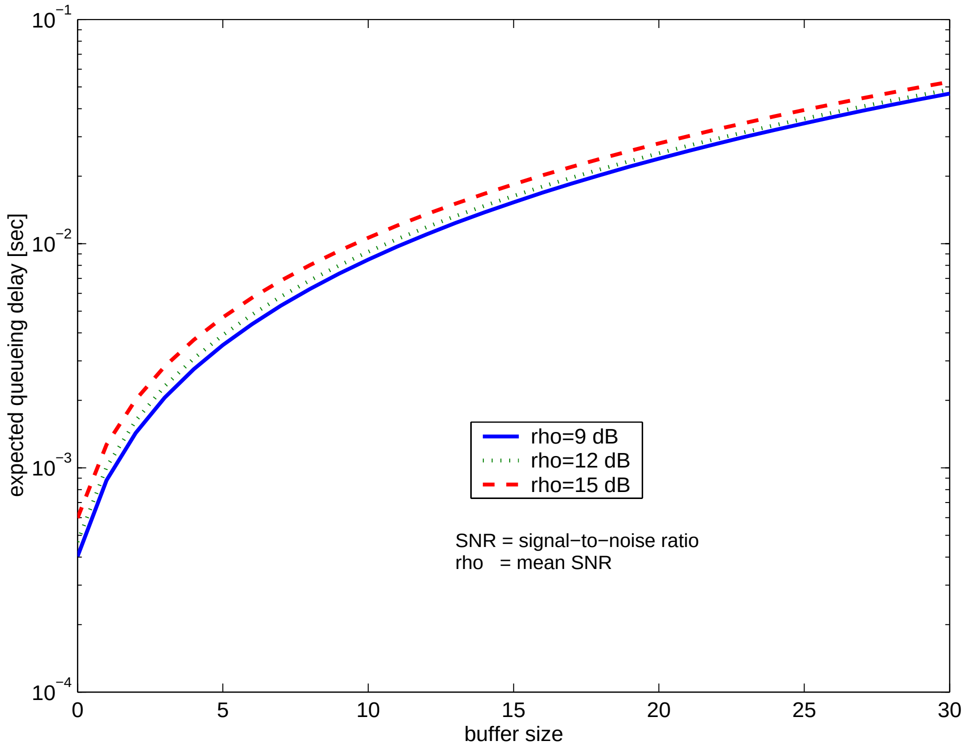

Figure 5.20: Expected packet queuing delay versus buffer capacity: effects of radio fading (received SNR) on traffic QoS.

Chapter 6

Analysis of RED based DQoS Scheme over Fading Channels

6.1 Introduction

Wireless Internet is the offshoot of the symbiotic relationship between mobile wireless communications and data/Internet communications. This paradigm shift of convergence is referred to by various other names, such as Wireless IP/Data, Mobile Internet/Data or Mobile Computing, depending on the background of the proponent. Traditionally, these disciplines had been separated in philosophy and application: the Internet is based mainly on high-speed, fixed links for data transfer, while mobile communications (MCOM) is based on low-speed, wireless links for speech transfer. Thus both technologies have been principally single-service networks. Traditionally, the Internet is based on connectionless packet switching (PS), while MCOM is based on connection-oriented circuit switching (CS). The connectionless trait of the Internet is due to its cornerstone, the Internet Protocol (IP). It seems easier to provide quality of service (QoS) via CS than PS, but PS provides better network resource utilization than CS via multiplexing.

Wireless IP aims to exploit the virtues of both parent technologies but discard their downsides to support multi-applications traffic on high-speed, multiservice (i.e. integrated video, voice and data) platforms. An all-IP Wireless

Internet comprises an IP based radio access network (IPRAN) and an IP based core network, resulting in IP radio base stations (IBS) and IP mobile stations (IMS). This avoids the need for QoS mapping between the Internet and the RAN. By its nature IP/Internet does not provide QoS to applications' traffic as it is based on the single-suit-fit-all paradigm. However, as Wireless IP is poised to support heterogeneous traffic with heterogeneous characteristics, it needs to support differentiated QoS. Network engineers expect the robustness, flexibility and efficiency of IP transport retained, just as network users need the QoS and the mobility of MCOM. Also, the QoS control protocols for Wireless IP should be fair, power efficient and scalable, as network growth predictions may not be reliable at rollout time.

This work analyses a simple, differentiated QoS scheme for wireless Internet. The scheme is based on traffic classification, random early detection (RED) active queue management scheme (see Section 2.11), and weighted first-come-first-served (FCFS) scheduling, in which the scheduling rate is proportional to a RED parameter, the maximum packet dropping (marking) probability. The validity of the scheme is analysed on a wireless channel degraded by Nakagami-m distributed fast multipath fading, lognormal distributed slow shadow fading, and additive white Gaussian noise (AWGN). Such a noisy channel is referred to as Nakagami-Lognormal (NLN) wireless channel [101]. The dynamics of this noisy wireless channel is modelled by a three-state Markov model. The parameters of the Markov model are evaluated using QPSK digital modulation format. As in the previous chapter, the interactions between link level error control and the higher level QoS scheme are analysed through a hybrid type-I ARQ error control schemes comprising a FEC and SR-ARQ. Both Poisson and bursty traffic sources are considered in the modelling. The presented scheme offers low loss to loss-sensitive applications traffic, and low queuing delay to delay-sensitive applications traffic.

The rest of the chapter is structured as follows. Section 6.2 presents the wireless link model and the scheduling algorithm. It computes the link parameters, such as the state transition rates, the bit error rates, and the state occupancy probabilities. Section 6.3 evaluates the steady-state queuing parameters, starting with the steady-state distribution and ending with the QoS metrics: packet loss ratio, expected per-packet queuing delay and throughput. Numerical results and discussions are presented in Section 6.4. The chapter ends with a summary in Section 6.5.

6.2 Wireless Channel Model

A three-state Markovian radio channel model is assumed with the following interpretation. State 1 represents a good-quality link with negligibly low error rate; state 2 represents a degrading link where session handoff is usually processed, while state 3 is a poor-quality link in which communications quality is severely impaired (see Fig. 5.5). If a better link is available then mobiles avoid using state 3 links. In this chapter, we consider Markov model with only nearest-neighbor state transitions, i.e. a quasi-birth-and-death (QBD) stochastic process, which is suited for random-error channels. The received instantaneous signal-to-noise ratio (SNR) γ is partitioned into three intervals with the thresholds: $0 \geq \gamma_1 < \cdots < \gamma_4 < \infty$, and the channel is in Markov state $s_n,\ n = 1, 2, 3(= N)$ if $\gamma \in [\gamma_n, \gamma_{n+1})$, as illustrated in Fig. 5.5.

Assume a wireless link perturbed by fast multipath fading, slow lognormal shadowing and additive noise. Such a wireless link is referred to as Nakagami-lognormal channels (NLN) [101]. The SNR γ on an NLN channel can be described by the probability density function (pdf) [95]

$$f_\gamma(\gamma) = \frac{10 m^m \gamma^{m-1}}{\sqrt{2\pi}\sigma \log_e(10)\Gamma(m)} \int_0^\infty z^{-(m+1)} e^{-m\gamma/z} e^{(z-\mu)^2/2\sigma^2} dz, \tag{6.1}$$

where μ and σ are the mean and spread of the lognormal shadow fading in decibels, $m \geq 1/2$ is the Nakagami-m fading figure, and $\Gamma(\cdot)$ is the gamma function. Let $S = \{s_1, s_2, s_3\}$ be the channel state space. Then, with (6.1), the probability that the channel is in fading-dependent state s_n is

$$p_n = \frac{10}{\sqrt{2\pi}\sigma \log_e(10)} \int_{z=0}^\infty \frac{P(m\gamma_{n+1}/z, m) - P(m\gamma_n/z, m)}{z} e^{-(z-\mu)^2/2\sigma^2} dz, \tag{6.2}$$

where $P(x,z) = \frac{1}{\Gamma(z)} \int_0^x t^{z-1} e^{-t} dt,\ \Re\{z\} \geq 0$ is the incomplete gamma function. It is easy to verify that $\lim_{\gamma_n \to 0\ \wedge\ \gamma_{n+1} \to \infty} p_n \to 1$.

State transition rates. Following the argument in Section 5.3.1 and noting (6.1), the transition rates between the three wireless channel states can be evaluated as [42]

$$t_{n,j} \approx \frac{10 \log_{10}(e)(m\gamma_j)^{m-1/2}}{\pi\sigma\Gamma(m)\gamma_0^m} \int_0^\infty \frac{\sqrt{mah^2\sigma^2\gamma_j + bx}}{x^{m+1}} e^{-(x-\mu)^2/2\sigma^2 - m\gamma_j/\gamma_0 x} dx,\ |n-j| = 1 \tag{6.3}$$

where $h = \log_{10} 2/20$, $a = 2\pi^2 f_{3\ dB}^2$, $b = \frac{\Omega}{4} \frac{d^2\Psi(t)}{dt^2}|_{t=0}$ and $f_{3\ dB}$ is the 3 dB cut-off frequency of the low-pass filter used to derive the shadow fading process. γ_0 and $\Psi(t)$ are the mean power and the normalized autocorrelation function of the Nakagami-m fading process, respectively.

The crossover probability (BER) in discrete-time link state s_n using QPSK modulation on an NLN channel is computed in [39] as

$$P_{b,n} = \frac{5m^m}{\sqrt{2\pi}\sigma \log_{10} e\Gamma(m) p_n} \int_0^\infty \int_{\gamma_n}^{\gamma_{n+1}} \frac{\gamma^{m-1}}{z^{m+1}} e^{\frac{-m\gamma}{z} \frac{-(z-\mu)^2}{2\sigma^2}} [erfc(\sqrt{\gamma}) - erfc^2(\sqrt{\gamma})] d\gamma dz \tag{6.4}$$

We will evaluate (6.2)–(6.4) in the numerical results section using Monte Carlo simulations.

6.2.1 Wireless Channel State Dependent Scheduling

We use an integrate RED buffer management scheme (see Section 2.11) and the CSDPS scheduling (see Section 4.8), as illustrated in Fig. 6.1, to provide differentiated QoS to IP traffic. As in Section 4.8, let the RED parameter set for GeS$_i$'s traffic be $\{d_{m,i}, max_i, min_i\}$, where $d_{m,i}$ is the maximum marking (dropping) probability.

We require that the scheduling rate a GeS i with RED parameter set $\{d_{m,i}, max_i, min_i\}$ receives is proportional to $d_{m,i}$, and hence the name RED drop threshold proportional scheduling (RDTPS). The RDTPS is a variant of the CSDPS scheme discussed in Section 4.8 and in (8.2) with the weights

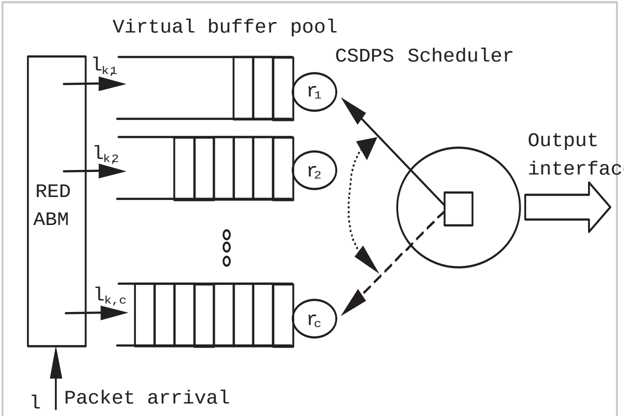

Figure 6.1: Schematic operations of the integrated RED buffer manager and weighted FCFS scheduler. (From [39] ©IEEE)

$$w_i = \frac{d_{m,i}}{\sum_{k \in \varphi_a} d_{m,k}}, \quad i = 1, 2, \cdots, c, \tag{6.5}$$

where φ_a is the set of active GeSs at the corresponding scheduling instant. The bandwidth efficiency in (8.2), β_n, is computed as in (5.11)–(5.13), but with $p_{e,n}$ replaced by $p_{b,n}$ in (6.4).

6.3 Queuing Model and Analysis

Assume that packets belonging to c types of traffic streams (GeS) arrive at a wireless router (IP based mobile or base station) according to independent but arbitrary distribution with aggregate mean arrival rate λ (Fig. 6.1). Let k be the average occupancy of the router's buffer of capacity K upon packet arrival. Assume that a packet burst of maximum length equal to the router's residual buffer capacity of $K - k$ is admissible. However, packets in a given logical queue are served sequentially and in an FCFS order. These assumptions lead to the two-dimensional (i.e. bivariate) state transition rate diagram of Fig. 6.2. In the Fig. 6.2, $r_{n,i}$ is the scheduling rate for packets belonging to GeS$_i$ when the wireless link is in state s_n, and λ_k is the aggregate effective packet arrival rate when the average buffer occupancy is k traffic units.

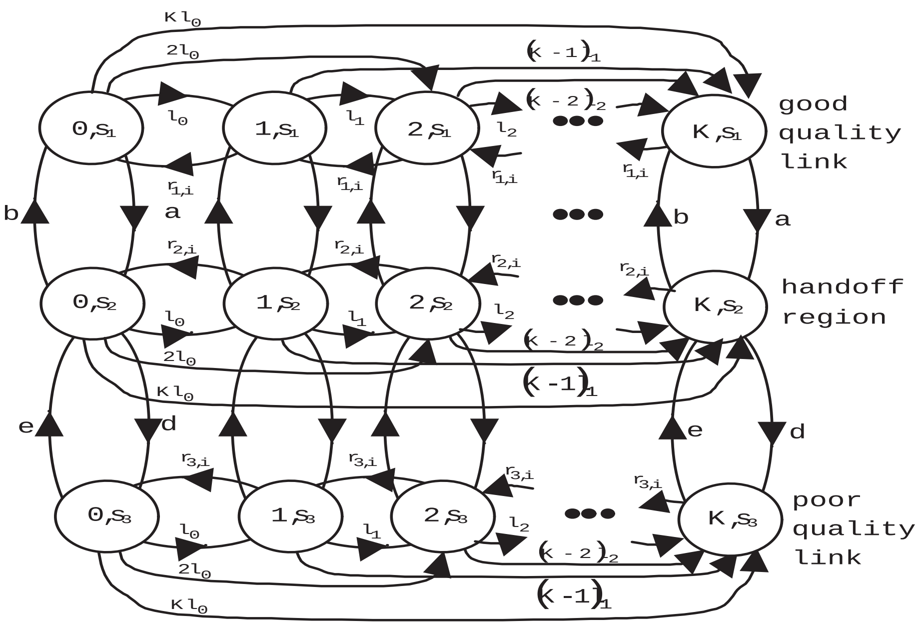

Figure 6.2: Two-dimensional state transition rate diagram with bursty traffic and three-state wireless link model.(From [39] ©IEEE)

6.3.1 Multiclass RED

Random early detection (RED) [27] is an active buffer management (ABM) scheme that is based on random packet discarding. RED prevents lock-out (monopoly of buffer space by a single or few flows) and full queue to absorb packet burst arrival and hence prevent global synchronization [27]. Global synchronization degrades link utilization since it throttles many responsive traffic sources simultaneously as they endure packet dropping. The RIO [20] queue management scheme is an extension of RED where the packets to be operated on are categorized into two classes: in-profile (IN) and out-of-profile (OUT) packets. During network congestion the OUT packets are preferentially discarded, as they are traffic in excess of the guaranteed rate.

This chapter uses a multi-class RED scheme with c traffic classes; each traffic class may comprise a number of flows. This allows easier mapping onto any QoS based application classification, such as the standard DiffServ [10] and the UMTS/3G application types which uses $c = 4$ [34]. Note that this work's interpretation of the traffic types is different from that of RIO. RIO is based on service level agreement (SLA) and instantaneous behaviour of flows, rather than QoS requirements.

Under the RED mechanisms, the actual packet arrival rate to the router's buffer with occupancy of k packets is

$$\lambda_k = \sum_{i=1}^{c} p_i \lambda_{k,i} = \lambda \sum_{i=1}^{c} p_i \alpha_i(k), \tag{6.6}$$

where $\alpha_i(k)$ is equivalent to $d_i(k)$ in (2.31). Thus, each traffic class has a unique set of RED parameters in order to provide a differentiated QoS.

Let N_t be the number of packets awaiting service in the radio node's buffer and S_t be the channel state at a scheduling instant t. Then $\{(N_t, S_t),\ t \geq 0\}$ is a bivariate, semi-Markov process on the state space $\{p(k,n)\}$ with $p(k,n) = Pr\{N_t = k | S_t = s_n\}$. We can thus define the state vector of order $N(K+1)$ as $\mathbf{p} = [p(0,1), p(1,1), \cdots, p(K,1), \cdots, p(0,3), p(1,3), \cdots, p(K,3)]$. Applying the flow conservation principle to each state in Fig. 6.2, we obtain the global balance equations from which the steady state vector $\mathbf{p}_\infty$ can be evaluated as

$$\mathbf{p}_\infty \mathbf{T} = \mathbf{0} \quad \text{and} \quad \mathbf{p}_\infty \mathbf{u}^t, \tag{6.7}$$

where $\mathbf{u}$ is a row vector of all 1's. The latter part of (6.7), the normalization relation, is needed since the rows of the state transition rate matrix $\mathbf{T}$ are linearly dependent. $\mathbf{T}$ is a block tridiagonal square matrix of order $N(K+1)$ and it is obtained from Fig. 6.2, as

$$\mathbf{T} = \begin{bmatrix} \mathbf{A}_1 & a\mathbf{I} & \\ b\mathbf{I} & \mathbf{A}_2 & d\mathbf{I} \\ & e\mathbf{I} & \mathbf{A}_3 \end{bmatrix},$$

where $a = t_{1,2},\ b = t_{2,1},\ d = t_{2,3}$ and $e = t_{3,2}$. All the submatrices of $\mathbf{T}$ are square of order $K+1$, $\mathbf{I}$ is an identity matrix of appropriate order, $c_l = \sum_{m=1}^{K-l} m = (K-l)(K-l+1)/2$, $t_1 = a,\ t_3 = e$, and the $\mathbf{A}_n$'s are given by

$$\mathbf{A}_n = \begin{bmatrix} -t_n - c_0\lambda_0 & \lambda_0 & 2\lambda_0 & 3\lambda_0 & \cdots & K\lambda_0 \\ r_{n,i} & -\xi_{n,i}(1) & \lambda_1 & 2\lambda_1 & \cdots & (K-1)\lambda_1 \\ & r_{n,i} & -\xi_{n,i}(2) & \lambda_2 & \cdots & (K-2)\lambda_2 \\ & & \ddots & \ddots & \ddots & \vdots \\ & & & r_{n,i} & -\xi_{n,i}(K-1) & \lambda_{K-1} \\ & & & & r_{n,i} & -t_n - r_{n,i} \end{bmatrix}, \; n = 1, 3,$$

$$\mathbf{A}_n = \begin{bmatrix} -b - d - c_0\lambda_0 & \lambda_0 & 2\lambda_0 & 3\lambda_0 & \cdots & K\lambda_0 \\ r_{n,i} & -\varsigma_{n,i}(1) & \lambda_1 & 2\lambda_1 & \cdots & (K-1)\lambda_1 \\ & r_{n,i} & -\varsigma_{n,i}(2) & \lambda_2 & \cdots & (K-2)\lambda_2 \\ & & \ddots & \ddots & \ddots & \vdots \\ & & & r_{n,i} & -\varsigma_{n,i}(K-1) & \lambda_{K-1} \\ & & & & r_{n,i} & -b - d - r_{n,i} \end{bmatrix}, \; n = 2,$$

where $\xi_{n,i}(k) = t_n + r_{n,i} + c_k\lambda_k$ and $\varsigma_{n,i}(k) = b + d + r_{n,i} + c_k\lambda_k$.

6.3.2 QoS Analysis

This subsection evaluates the QoS metrics: average packet loss ratio, effective link throughput and average per-packet queuing delay. Based on the ASTA (Arrivals See Time Averages) [70] principle in queuing theory, these metrics can be given for class i packets as follows [18]. The loss probability for GeS$_i$' s traffic can be obtained as

$$L_i = 1 - \sum_{k=0}^{K}\sum_{n=1}^{N} \alpha_i(k) p_\infty(k,n) = 1 - \mathbf{p}_\infty \mathbf{a}_i^t, \quad i = 1, 2, \cdots, c, \tag{6.8}$$

where $\mathbf{a}_i = [\mathbf{b}_i\mathbf{b}_i\cdots\mathbf{b}_i]$, with $\mathbf{b}_i = [\alpha_i(0)\alpha_i(1)\cdots\alpha_i(K)]$, is a row vector of length $N(K+1)$. The corresponding effective throughput for packets of GeS$_i$ is

$$T_i = 1 - L_i, \quad i = 1, 2, \cdots, c. \tag{6.9}$$

Similarly, by defining the vectors $\hat{\mathbf{a}}_i = [\alpha_i(0)\ 2\alpha_i(1)\ 3\alpha_i(2)\ \cdots\ K\alpha_i(K-1)]$, $\hat{\mathbf{b}}_i = [\sum_n \frac{p_\infty(0,n)}{r_{n,i}}\ \sum_n \frac{p_\infty(1,n)}{r_{n,i}}\ \cdots\ \sum_n \frac{p_\infty(K-1,n)}{r_{n,i}}]$, the expected queuing delay using a simple first-come, first-served scheduling within the logical queue of GeS_i is given by

$$D_i = \sum_{k=0}^{K-1} \sum_{n=1}^{N} (k+1)\alpha_i(k) p_\infty(k,n)/r_{n,i} = \hat{\mathbf{a}}_i \hat{\mathbf{b}}_i^t, \quad i = 1, 2, \cdots, c. \tag{6.10}$$

The QoS metrics evaluated in this section are numerically tested in the next section.

6.4 Results and Discussions

Unless otherwise stated differently elsewhere in the thesis, the parameters for the numerical investigations are compiled into Tables 6.1 and 6.2. The results are illustrated in Figures 6.3–6.6. In Figure 6.3, we observe that the packet loss ratios are higher for Poisson traffic than for bursty traffic under the same average traffic arrival rate. However, in contrast to Poisson traffic, bursty class 1 traffic has higher loss probability than class 2 traffic. Figure 6.4 shows that bursty traffic has higher expected queuing delays than Poisson traffic under the same average traffic arrival rate. Figures 6.5 and 6.6 analyse the loss probability and the queuing delay as a function of wireless link state (WLS) under different traffic sources. Again, the expected queuing delays are higher for bursty traffic than Poisson traffic, but the packet loss probabilities show opposite scenarios.

It is observed in Figure 6.6 that wireless link state 2 has worse queuing performance than state 1 under the same nature of traffic and average arrival rates. However, the loss ratios are better for wireless link state 2 than for state 1, all other parameters being the same. In this work, wireless link state 2 has less degradation and thus better effective bandwidth (bit rate) than wireless link state 1. Thus better link quality offers better loss performance, but poorer expected queuing delays. This can be explained as follows. Better quality links

Table 6.1: Link level parameters for the numerical examples.

Parameter	**Value**
Nakagami-m fading figure	$m = 0.75$
standard deviation of lognormal fading	$\sigma = 6$ dB
mean of lognormal fading process	$\mu = 0.5$ dB
received SNIR	$0 \leq \gamma \leq 12$ dB
3-dB cut-off frequency	$\mathrm{f}_{3\ dB} = 50Hz$
mean power of Nakagami fading	$\gamma_0 = 2$ dB
maximum link bandwidth	$R_b = 144$ kb/s
max. number of ARQ retransmissions	$n_{arq} = 6$
layer 2 packet length (fixed)	$\nu = 63$ bits
number of correctable bits in FEC	$\tau = 2$

cause fewer dropping of packets than links in poorer state/quality and thus more packets are queued in the wireless routers, resulting in longer queues and longer packet waiting times in the buffers.

6.5 Summary

With the advent of mobile wireless Internet poised to support multi-applications' traffic with different sensitivities; service differentiation and service guarantee

Table 6.2: Queuing parameters for the numerical examples.

Parameter	**Value**
buffer capacity	$K = 20$ packets
number of traffic classes	$c = 2$
	$\mathbf{d}_m = [0.01\ 0.05]$
RED parameters	$\mathbf{min} = [0.4K\ 0.3K]$
	$\mathbf{max} = 2 \times \mathbf{min}$
arrival ratio of traffic types	$1/c$

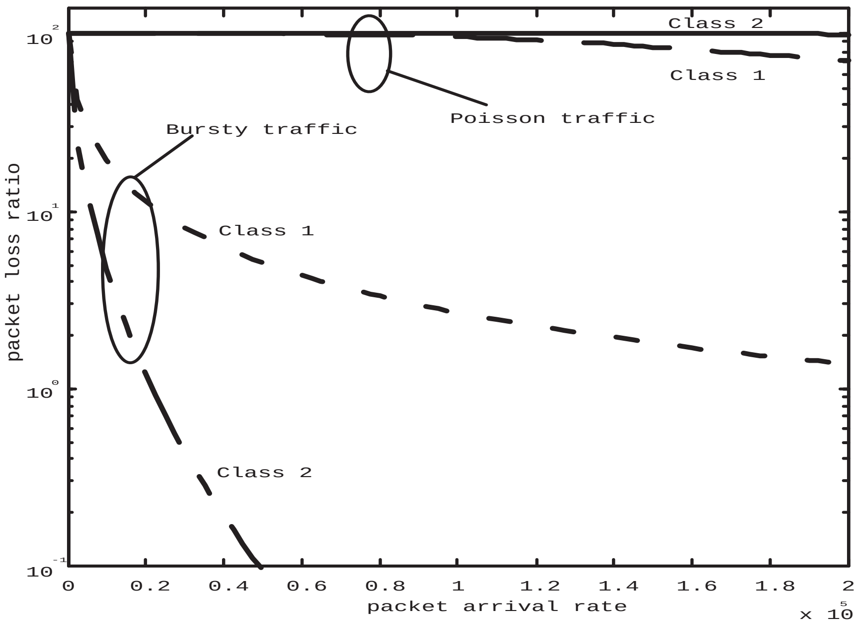

Figure 6.3: Packet loss probability versus packet arrival rate for two different traffic classes and two types of source traffic: Poisson and bursty source.(From [39] ©IEEE)

are catching more attention. This work analyses the steady-state queuing performance of a differentiated quality-of-service scheme over wireless links. The scheme is based on traffic classification, random early detection (RED) queue management and weighted first-come-first-served (FCFS) scheduling, in which the scheduling rate is proportional to a RED parameter, the maximum packet dropping (marking) probability. The validity of the scheme is analysed on a three-state wireless link model, accounting for fast multipath fading, slow shadowing fading, additive noise, modulation format, and error control, under bursty traffic sources.

The presented scheme offers low loss to loss-sensitive applications traffic, and low queuing delay to delay-sensitive applications traffic. Dependence of QoS metrics on wireless link state/quality is also examined, under both Poisson

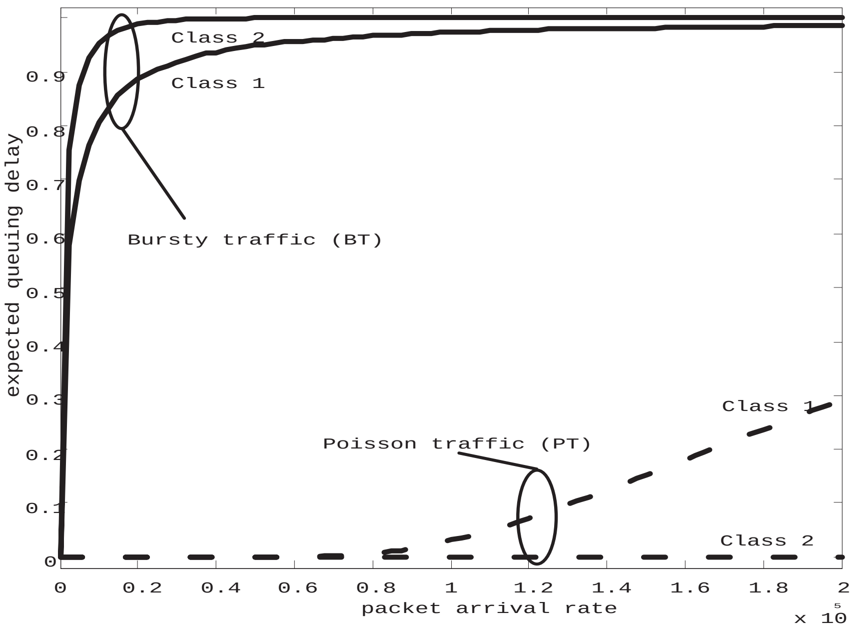

Figure 6.4: Expected packet queuing delay versus packet arrival rate for two different traffic classes and two types of source traffic: Poisson and bursty sources. (From [39] ©IEEE)

and bursty traffic environments. It is observed that bursty traffic has longer expected queuing delays than Poisson traffic, but less loss ratio than Poisson traffic, if all other conditions are held the same for both source traffic types. The parameter used for above conclusion is the aggregate packet arrival rates. Thus the same aggregate traffic arrival rates are used for both traffic types.

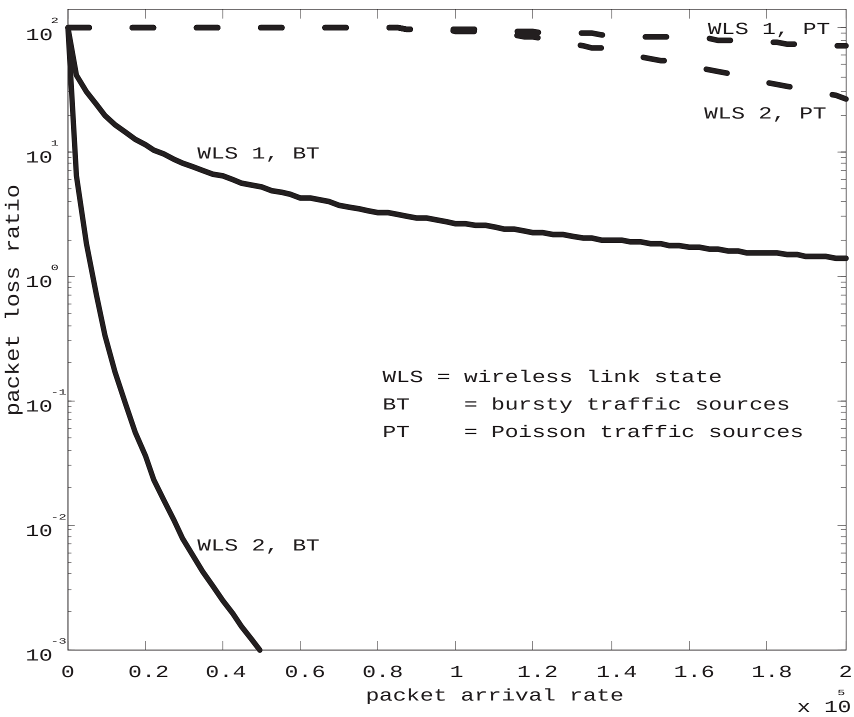

Figure 6.5: Packet loss probability versus packet arrival rate with wireless link state and source traffic type as parameters. (From [39] ©IEEE)

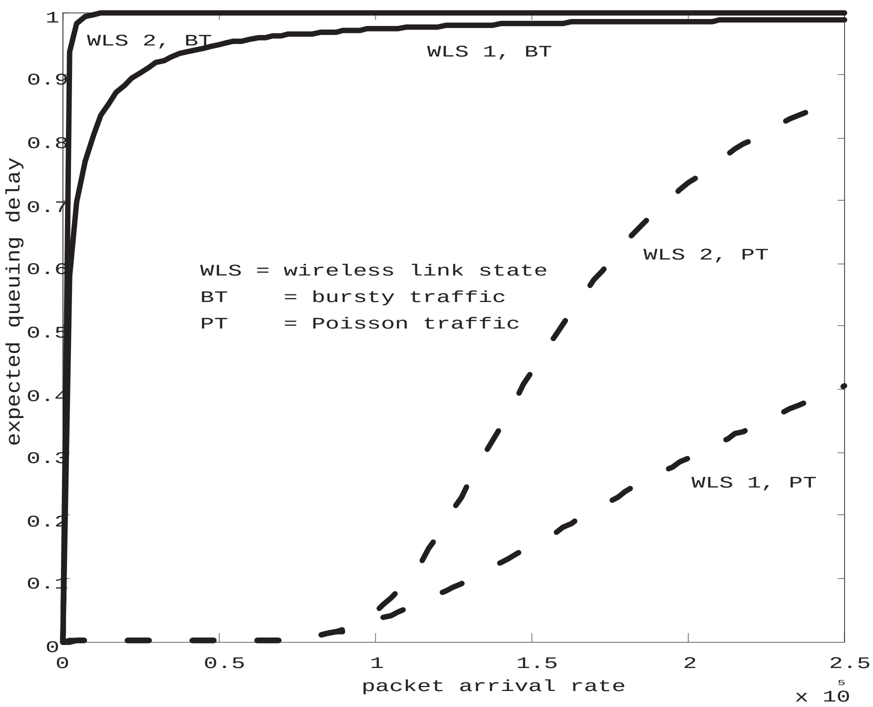

Figure 6.6: Expected packet queuing delay versus packet arrival rate with wireless link state and source traffic type as parameters. (From [39] ©IEEE)

Chapter 7

Analytical Model of a DQoS Scheme: Slow Fading Case

7.1 Introduction

The global Internet has so far gotten by with its single service performance category called *best effort delivery* , which does not guarantee any service, let alone service quality. However, over the years many QoS-hungry applications and services have evolved which were unthinkable at the onset of the Internet and its supporting protocols. Moreover, the protocols and applications for the internet were developed for fixed networks with little regard for the challenges of the wireless environment. A challenge in multiservice networks aiming to integrate voice, data and video services is the fact that no single network service can be optimized to support different traffic types. Hence, the impetus towards the introduction of differentiated quality-of-service (DQoS) support into the Internet.

An alternative approach adopted in Qualcomm's High Data Rate (HDR) technology is the usage of different radio frequency carriers for voice and data, and the overlay of data-oriented HDR over voice-oriented IS-95/1X system. Of similar uptake growth to the Internet services is mobile communications (see Fig. 1.1), as people need wireless connectivity for everywhere, every time communications. A recent paradigm shift is the marriage of these two traditionally separated technologies: data-oriented Internet based communications, and voice-oriented mobile wireless communications, resulting in Mobile Wireless Internet (MOWINT). Hence, mobile and base stations will evolve to Internet portals as they 'speak' the language of the Internet, the Internet Protocol (IP).

This chapter analyses the queuing performance of a differentiated QoS scheme over slowly fading wireless channels. The scheme is compatible with the 'wired' DiffServ standard of IETF. The differentiated service is provisioned via traffic classification, deterministic buffer sharing, packet admission control (PAC), and buffer share proportional scheduling rates over the wireless links. Deterministic service rate guarantees bandwidth to each traffic class and thus controls the queuing delay and loss rate of a given traffic class. Although most part of the analysis is generally valid, of particular interest is the case where traffic is differentiated into four QoS classes, as it maps one-to-one onto the traffic classes specified for third generation (UMTS) mobile systems [34, 25]. We concentrate on the wireless access part of a MOWINT system, since core routers in the Internet have relatively high processing speeds and thus packet queuing and processing delays at such nodes may be relatively small.

This chapter is organized as follows. General system descriptions appear in Section 7.2.1. Section 7.2 presents the system model and the model assumptions. Steady-state queuing analysis is the topic of Section 7.3, where the QoS metrics are also evaluated. Section 7.3.3 presents numerical examples and discussions for the steady-state analysis. Transient queuing analysis is presented in Section 7.4. Finally, Section 7.5 summarizes the chapter with conclusions.

7.2 System Model

This section begins with a description of the general system modelling features in Subsection 7.2.1. The traffic arrival process is described in Section 7.2.2, followed by the link level parameters in Sect. 7.2.3. Sections 7.2.4 and 7.2.5 evaluate the state transition rates and MAC scheduling rates, respectively.

7.2.1 General System Description

Traffic is classified into GeSs, and the radio node buffer is shared (either statically or dynamically) among the differentiated GeSs. It is assumed that each radio node in the MOWINT system is equipped with a differentiated QoS (DQoS) functionality. The DQoS scheme comprises a packet admission controller (PAC) (see Fig. 7.2), a class-based buffering (CBB) which queues packets belonging to the same GeS in the same buffer, and a packet scheduler (PS) (Fig. 7.1). The PAC, in turn, comprises a packet classifier which sorts packets into corresponding GeS, and a threshold dropper (TD). The TD operates as a buffer admission controller (BAC), and drops an incoming packet if upon arrival its buffer share is used up. The operational details of the PAC in Fig. 7.1 is explicated in Fig. 7.2. In Fig. 7.2, P_i^k denotes the $k-$th arriving packet of GeS$_i$.

It is assumed that the wireless link impairment is dominated by an additive noise (AWGN) and a slow fading with lognormal distribution. Owing to the slow fading assumption, the average time between the state transitions of the semi-Markov-modulated wireless channel is larger than a single packet transmission time. The wireless channel is modelled by a two-state, discrete-time Gilbert-Elliot channel (GEC) model. The GEC model parameters, viz state transition and state occupancy probabilities, are evaluated using the channel noise model and a BPSK modulation. For simplicity of analysis, it is also assumed that none of the events, packet arrival, wireless link state change and service completion occur at the same time. This assumption is made void in some parts of the thesis, where generalized cases are examined (see e.g., Chapter 5).

7.2.2 Traffic Arrival Process

Recently, it has become a widely accepted notion that packetized data traffic, especially over LAN or the Internet, inhibit burstiness or self-similarity pattern with heavy-tailed distribution. However, the effects of heavy-tailed distributed traffic on QoS analysis is not as certain as the traffic distribution. In this chapter, we assume Poisson distributed (i.e. light-tailed) traffic, for simplicity of analysis. In the subsequent chapters, however, bursty traffic sources will be

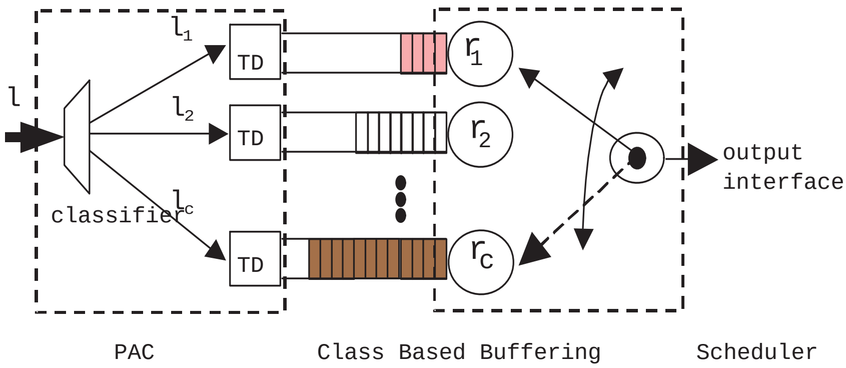

Figure 7.1: Block diagram of the DQoS scheme in the radio nodes. (From [38] ©IEEE)

considered.

Assume packets belonging to c traffic classes arrive from/at an Internet router to/from an FWR according to an independent Poisson processes with respective mean arrival rates $\lambda_1,\ \lambda_2,\ \cdots,\ \lambda_c$ packets per time units (Fig. 7.1). These packets are rescheduled and forwarded to an MWR in the same cell or through the fixed backbone Internet to another cell. Assume the FWR has a shared buffer of capacity B packets, which is logically partitioned into c queues of sizes $B_1,\ B_2,\ \cdots,\ B_c$, where $B_i > B_j,\ \forall\ i < j$ and $\sum_{k=1}^{c} B_k = B$. The buffer sharing can be done statically or dynamically on a packet by packet basis. This, however, requires a trade-off between algorithmic complexity and resource utilization efficiency. Within a logical queue packets are served in the order of arrival (i.e., FCFS). An arriving packet of class i is dropped if it meets B_i packets in its respective queue (Fig. 7.2). Thus the net aggregate packet arrival rate to the buffer is

$$\lambda = \sum_{i=1}^{c} \lambda_i I\Big\{ N_i(t) < B_i \Big\}, \tag{7.1}$$

where $N_i(t)$, with $\lim_{t\to\infty} N_i(t) = N_i$, is the number of GeS$_i$'s packets in the system at time t, and I is the indicator function, which is defined as

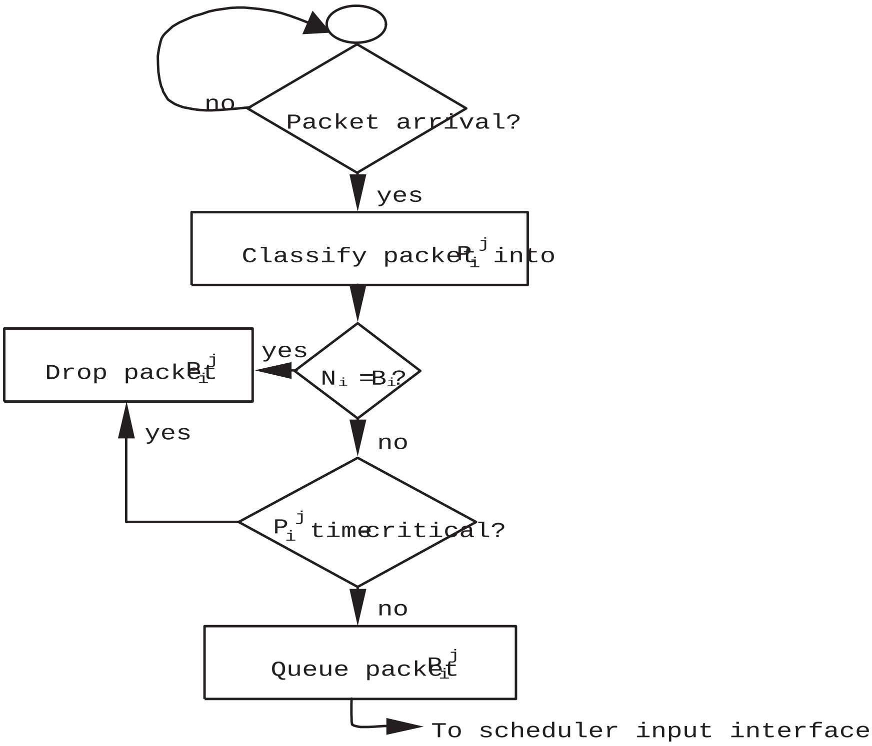

Figure 7.2: Operation of the packet admission controller in Fig. 7.1 (From [38] ©IEEE)

$$I\{\vartheta\} = \begin{cases} 1, & \text{if } \vartheta \text{ is true} \\ 0, & \text{otherwise} \end{cases} \tag{7.2}$$

7.2.3 Link Level Parameters

Assume that a possible fast fading impairments on the wireless channel is eliminated by either an efficient averaging or micro-diversity technique. Hence, only slow fading and additive noise (AWGN) are modelled in this chapter[1]. Assume the wireless channel is modelled by a two-state (i.e. $N = 2$), discrete-time Gilbert-Elliot model (see Section 2.1.3) with mean state residence (dwell) times of d_1 and d_2, respectively. Hence, the respective state transition rates[2] are

[1] This assumption is realistic for slow moving contexts and/or perfect diversity techniques.

[2] In this thesis, we denote the transition rate from state n_1 to state n_2 as t_{n_1,n_2}.

$t_{12} = 1/d_1$ and $t_{21} = 1/d_2$. The Gilbert-Elliot channel model is usually used for slowly fading wireless links, as discussed in details in Section 2.1.3.

Assume that the remaining wireless link's degradation—slow fading on AWGN channel—is modelled with the lognormal statistical distribution given in (2.1). Terrain, large foliage and buildings along the signal propagation path between the MWR and the FWR cause the shadowing and the AWGN is due to transceiver thermal noise. The received signal-to-noise ratio γ is partitioned into $N = 2$ intervals and the link is assumed to be in state $s_n,\ n = 1, 2(= N)$ if $\gamma \in (\gamma_n, \gamma_{n+1}]$. Hence, using the p.d.f. in (2.1), the probability that the channel is in discrete state s_n is given by $Pr\{\gamma_n \leq \gamma < \gamma_{n+1}\}$, i.e.,

$$p_n = \int_{\gamma_n}^{\gamma_{n+1}} f_\gamma(\gamma)d\gamma = \frac{1}{2}\left\{erfc\Big(\frac{10\log_{10}\gamma_n - \mu}{\sqrt{2}\sigma}\Big) - erfc\Big(\frac{10\log_{10}\gamma_{n+1} - \mu}{\sqrt{2}\sigma}\Big)\right\} \tag{7.3}$$

In this chapter, we set $p_n = 1/N,\ n = 1,\ 2$ and use simulations to compute the corresponding SNIR thresholds $\gamma_n,\ n = 1, 2, 3(= N + 1)$. This gives the most randomly varying (worst-case) two-state wireless channel.

For simplicity, we assume that the communications system employs a binary phase-shift keying (BPSK) modulation/demodulation (modem), and evaluate the crossover probabilities of the binary symmetric channel associated with each of the two states. The bit-error probability (BEP) of a BPSK signalling on AWGN channel is given in [97] as $P_{\mathrm{b}} = erfc(\sqrt{\gamma})/2$. Hence, assuming that the slow fading is independent of the AWGN, the BEP in a Markov state s_n can be evaluated as

$$p_{\mathrm{b},n} = \frac{1}{p_n}\int_{\gamma_n}^{\gamma_{n+1}} P_{\mathrm{b}}(\gamma)f_\gamma(\gamma)d\gamma,$$

which yields

$$p_{\mathrm{b},n} = \frac{5}{\sqrt{2\pi}\sigma p_n \log_e(10)}\int_{\gamma_n}^{\gamma_{n+1}} \frac{1}{\gamma} erfc(\sqrt{\gamma})e^{-(10\log_{10}\gamma - \mu)^2/2\sigma^2}d\gamma. \tag{7.4}$$

The integrals in (7.3) and (7.4) are evaluated using Monte Carlo simulations in Sections 7.3.3 and 7.4 by transforming them into the suitable form $\int_0^1 g(x)dx$.

7.2.4 Channel State Transition Rate

The channel or link state transition rates (i.e. level crossing rates, LCR) capture the time progression of the wireless link. The LCR is the average rate at which the signal envelope crosses a specified SNIR level in the positive direction [55]. The LCR at the signal level γ_j (at the boundary between link state n and j) can be given as [55]

$$t_{nj} = \int_0^\infty \dot{\gamma} f(\gamma_j, \dot{\gamma}) d\dot{\gamma}, \ |n-j| \leq 1, \tag{7.5}$$

where $\dot{\gamma} = d\gamma/dt$ and $f(\gamma_j, \dot{\gamma})$ is the joint p.d.f. of γ and $\dot{\gamma}$. Following ([84], (38) and (39)) and noting the variable transformation $x = (10\log_{10}\gamma - \mu)/\sigma$, $f(\gamma_j, \dot{\gamma})$ for the lognormal p.d.f. in (2.1) can be derived as

$$f(\gamma_j, \dot{\gamma}) = \frac{100\exp\Big(-(10\log_{10}\gamma_j - \mu)^2/2\sigma^2\Big)}{2\pi\sqrt{b_0}\ln^2(10)\sigma^2\gamma_j^2} \cdot \exp\Bigg(-\frac{(10\log 10e)^2}{2b_0\sigma^2\gamma_j^2}\dot{\gamma}^2\Bigg), \tag{7.6}$$

where $b_0 = 2\pi^2 f_C^2/\ln(2)$ and f_C is the 3-dB cut-off frequency of the low-pass filter which generated the Guassian process used to derive (2.1) and (7.6) [84]. Substituting (7.6) into (7.5) yields the wireless channel state transition rates (measured in system symbol rate, R_b/L_p, where R_b and L_p are the channel bit rate and packet length, respectively,)

$$t_{nj} = \frac{5}{\sqrt{2\pi}\ln(10)\sigma\gamma_j}\frac{R_b}{L_p}\exp\Big(-(10\log_{10}\gamma_j - \mu)^2/2\sigma^2\Big), \quad |n-j| \leq 1. \tag{7.7}$$

7.2.5 Channel State-Dependent Proportional Scheduling

An error control scheme combining an ARQ and an FEC schemes is becoming attractive for wireless channels [64]. This approach allows trade-off between link inefficiency due to FEC encoding overhead and long delay due to ARQ retransmissions. The FEC can be designed to correct the most rampantly occurring error patterns. Let (ν, κ, τ, $\hat{r}_{\text{arq}}$) denote a rate κ/ν hybrid type-I linear code with τ-error-correcting FEC component and a stop-and-wait ARQ (SAW-ARQ) component with maximum of $\hat{r}_{\text{arq}}$ transmissions (see Section 2.2).

Let P_n be the probability that the receiver acknowledges a received vector as a codeword when the link is in state s_n. This happens if either the received vector is error-free or contains undetectable errors. Hence [36, 37],

$$P_n \leq (1-p_{b,n})^{\nu} + \sum_{j \geq 2\tau+1}^{\nu} \binom{\nu}{j} p_{b,n}^{j} (1-p_{b,n})^{\nu-j}, \tag{7.8}$$

as some error patterns with Hamming weight $\geq 2\tau+1$ (i.e. code's minimum distance) may be detectable. Let r_{b} be the transmitter messaging bit rate and t_{idle} be the inter-code vector transmission (i.e. idle) time of the SAW-ARQ scheme. Hence, the average number of 'transmissions' in link state s_n can be obtained as $\bar{N}_n = (1 + r_b t_{\text{idle}}/\nu) P_n \sum_{j=1}^{\hat{r}_{\text{arq}}} j(1-P_n)^{j-1}$, resulting [38]

$$\bar{N}_n = \begin{cases} 1 + \frac{r_b}{\nu} t_{\text{idle}})/P_n, & \text{if } \hat{r}_{\text{arq}} \to \infty \\ (1 + \frac{r_b}{\nu} t_{\text{idle}}) \Big(1 - (1-P_n)^{\hat{r}_{\text{arq}}} (1 + \hat{r}_{\text{arq}} P_n)\Big)/P_n, & \text{if } \hat{r}_{\text{arq}} \neq \infty \end{cases} \tag{7.9}$$

Assuming an IP packet size of L_p bits, there are L_p/κ link layer packets (blocks) per IP packet. Hence the link state dependent *bandwidth efficiency* or *bandwidth degradation factor* [42] is

$$\beta_n = \kappa / (\nu \bar{N}_n \lceil L_p/\kappa \rceil) < 1, \;\; n = 1, 2. \tag{7.10}$$

Now, assume that the wireless link interface bandwidth is R_b bits/sec, and that different traffic classes receive differentiated services via allocated buffer share proportional scheduling rates. Hence, the *effective rate* of the CSDPS (see Section 4.8) for GeS$_i$'s traffic is

$$r_{n,i} = \frac{B_i}{B} \beta_n R_b, \quad i = 1, 2, \cdots, c \text{ and } n = 1, 2 \tag{7.11}$$

The ratio B_i/B may be proportional to the percentage of codes, timeslots or frequencies the medium access control (MAC) layer allocates to class i packets in a CDMA, TDMA or OFDMA multi-access systems, respectively. Note that if $B_i \neq 0$ then $r_{n,i} \neq 0$, i.e. each traffic class is guaranteed a scheduling rate and hence none of the queues is starved of service as in strict priority queuing. If B_i is interpreted as a token bucket depth, then $r_{n,i}$ is the corresponding token

Table 7.1: State transitions and corresponding rates.

From state	To state	Rate
$(n_i;\ s_n)$	$(n_i - 1;\ s_n)$	$r_{n,i};\ n = 1, 2$ and $i = 1, 2, \cdots, c$
$(n_i;\ s_n)$	$(n_i + 1;\ s_n)$	$\lambda_i;\ i = 1, 2, \cdots, c$
$(n_i;\ s_n)$	$(n_i;\ s_k)$	$t_{nk};\ n, k = 1, 2$
$(n_i;\ s_n)$	$(n_i - 1;\ s_k)$	$r_{n,i} + t_{nk};\ n, k = 1, 2;\ i = 1, 2, \cdots, c$
$(n_i;\ s_n)$	$(n_i + 1;\ s_k)$	$\lambda_i + t_{nk};\ n, k = 1, 2;\ i = 1, 2, \cdots, c$

rate. The novelty in the presented DQoS scheme lies largely on (7.11), and the efficient sharing of the available buffer capacity among the generic streams differentiated.

7.3 Steady-State Queuing Analysis

Assume that none of the events, service completion, packet arrival and link state change, occur simultaneously. Hence, the system can be described as a $(c+1)-$st dimensional embedded Markov process $\Big\{\Big(n_1(t), n_2(t), \cdots, n_c(t); S(t)\Big),\ t \in \Re_+\Big\}$ on the state space $\Phi = \Big\{(n_1, n_2, \cdots, n_c; S) : n_i = 1, 2, \cdots, B_i,\ i = 1, 2, \cdots, c;\ S = 1, 2\Big\}$, where S is the wireless channel state variable. The possible state transitions and instantaneous rates of this 2-dimensional queuing system are given in Table 7.1.

Define the probability

$$p_{ij\cdots k|n} = Pr\Big\{n_1 = i,\ n_2 = j,\ \cdots,\ n_c = k|S = n\Big\} \tag{7.12}$$

and let $\mathbf{p}$ be the lexicographic arrangement of the $p_{ij\cdots k|n}$'s, i.e. in increasing value of n_1, and for a given n_1, in increasing value of n_2, etc. In the following, we evaluate the square infinitesimal generator matrix, $\mathbf{T} = (t_{kl})$, of order $N_{\text{T}} = N \cdot \prod_{i=1}^{c}(1 + B_i)$ of the above stochastic system using Table 7.1, where t_{kl} is the state transition rate from state k to state l for $k \neq l$ and $t_{kk} = -\sum_{\forall l \neq k} t_{kl}$. It is worthy of note that, $\mathbf{T}$, as computed here, is valid for both the steady-state and the transient analyses. Next, the steady-state distribution vector $\mathbf{p}_\infty$ of order N_T is evaluated using the *global balance* and *probability conservation* system of equations

$$\mathbf{p}_\infty \mathbf{T} = \mathbf{0} \quad \text{and} \quad \mathbf{p}_\infty \cdot \mathbf{u} = 1, \tag{7.13}$$

where $\mathbf{u}$ is a column vector of 1's. We then use $\mathbf{p}_\infty$ to analyse the average packet queuing delay and the packet loss probability for packets of a given traffic class, say i, in Section 7.3.3.

Let $\mathbf{I}_1$, $\mathbf{I}_2$ and $\mathbf{I}_3$ be identity matrices of respective orders $\prod_{i=1}^{c}(1+B_i)$, $\prod_{i=2}^{c}(1+B_i)$ and $\prod_{i=3}^{c}(1+B_i)$. Using the data in Table 7.1 we obtain

$$\mathbf{T} = \begin{bmatrix} \mathbf{T}_1 & t_{12}\mathbf{I}_1 \\ t_{21}\mathbf{I}_1 & \mathbf{T}_2 \end{bmatrix},$$

where for $n = 1, 2$

$$\mathbf{T}_n = \begin{bmatrix} \mathbf{A}_n & \lambda_1\mathbf{I}_2 & 0 & \cdots & \cdots \\ r_{n,1}\mathbf{I}_2 & \mathbf{B}_n & \lambda_1\mathbf{I}_2 & 0 & \cdots \\ & \ddots & \ddots & \ddots & \\ & & r_{n,1}\mathbf{I}_2 & \mathbf{B}_n & \lambda_1\mathbf{I}_2 \\ & & & r_{n,1}\mathbf{I}_2 & \mathbf{C}_n \end{bmatrix},$$

$$\mathbf{A}_n = \begin{bmatrix} \mathbf{A}_{1n} & \lambda_2\mathbf{I}_3 & 0 & \cdots & \cdots \\ r_{n,2}\mathbf{I}_2 & \mathbf{A}_{2n} & \lambda_2\mathbf{I}_3 & 0 & \cdots \\ & \ddots & \ddots & \ddots & \\ & & r_{n,2}\mathbf{I}_3 & \mathbf{A}_{2n} & \lambda_2\mathbf{I}_3 \\ & & & r_{n,2}\mathbf{I}_3 & \mathbf{A}_{3n} \end{bmatrix},$$

$\mathbf{B}_n = \mathbf{A}_n - r_{n,1}\mathbf{I}_2$ and $\mathbf{C}_n = \mathbf{B}_n + \lambda_1\mathbf{I}_2$. The submatrices $\mathbf{A}_{kn},\ k = 1, 2, 3$ of $\mathbf{A}_n$ are given by $\mathbf{A}_{2n} = \mathbf{A}_{1n} - r_{n,2}\mathbf{I}_3$, $\mathbf{A}_{3n} = \mathbf{A}_{2n} - \lambda_2\mathbf{I}_3$, and[3]

[3]Note that in the following matrix $\mathbf{A}_{1n}$ we write r_i^n instead of $r_{n,i}$ just to conserve space.

$$
\mathbf{A}_{1n} = \left[\begin{array}{cccccccccccccc}
-a_1^n & \lambda_4 & 0 & \cdots & 0 & \lambda_3 & & & & & & & & \\
r_4^n & -a_2^n & \ddots & & & & \lambda_3 & & & & & & & \\
0 & \ddots & -a_2^n & \lambda_4 & & & & \ddots & & & & & & \\
\vdots & & r_4^n & \ddots & 0 & & & & \lambda_3 & & & & & \\
0 & & & 0 & -a_2^n & \lambda_4 & & & & \lambda_3 & & & & \\
r_3^n & & & & r_4^n & -a_3^n & \lambda_4 & & & & \lambda_3 & & & \\
 & r_3^n & & & & \ddots & \ddots & \ddots & & & & \ddots & & \\
 & & \ddots & & & & r_4^n & -a_3^n & \lambda_4 & & & & \lambda_3 & \\
 & & & r_3^n & & & & r_4^n & -a_3^n & \lambda_4 & & & & \lambda_3 \\
 & & & & r_3^n & & & & r_4^n & -a_4^n & 0 & & & 0 \\
 & & & & & r_3^n & & & & 0 & -a_4^n & \lambda_4 & & \vdots \\
 & & & & & & \ddots & & & & r_4^n & \ddots & \ddots & 0 \\
 & & & & & & & r_3^n & & & & \ddots & -a_4^n & \lambda_4 \\
 & & & & & & & & r_3^n & 0 & \cdots & 0 & r_4^n & -a_5^n
\end{array}\right],
$$

where $a_1^n = \sum_{i=1}^{c} \lambda_i + 1/d_n$, $d_n = 1/t_{n,k}, k, n = 1, 2$, $a_2^n = a_1^n + r_{n,c}$, $a_3^n = a_2^n + r_{n,c-1}$, $a_4^n = a_3^n - \lambda_{c-1}$, and finally, $a_5^n = a_4^n - \lambda_c$. Few words about the structure of the matrix $\mathbf{A}_{1n}$ is in order. There are $(B_4 - 1)$ zeros between λ_4 and λ_3, just as there are $(B_4 - 1)$ zeros between $r_{n,4}$ (r_4^n) and $r_{n,3}$ (r_3^n) horizontally and vertically. Along the immediate superdiagonal (subdiagonal), the vectors $\lambda_4 \mathbf{u}_4$ $(r_{n,4}\mathbf{u}_4)$ are separated by a single '0', where $\mathbf{u}_4$ is a length-B_4 vector of 1's.

7.3.1 Steady-State Distribution

From (7.13) one may be tempted to compute the stationary probability vector, $\mathbf{p}_\infty$, via $\mathbf{p}_\infty = \mathbf{i}_k \mathbf{T}_{\mathrm{m}}^{-1}$, where $\mathbf{T}_{\mathrm{m}}^{-1}$ is the inverse of $\mathbf{T}$ with a column vector of 1's substituted for, say, its kth column and $\mathbf{i}_k$ is a unit row vector with the only non-zero element in its kth column. However, since $\mathbf{T}_{\mathrm{m}}$ is a slight perturbation of the singular matrix $\mathbf{T} = (t_{k,l})$, this approach is not attractive due to:

- possible system underflows as some of the probabilities may be very small,
- computational inaccuracy, and
- computation time and memory requirements if matrix $\mathbf{T}$ is large.

This chapter uses Theorem 2.1 and the algorithm illustrated in the flow diagram shown in Fig. 7.7 to evaluate $\mathbf{p}_\infty$ in (7.13). Theorem 2.1, referred to as uniformization (see Eq. 2.4), is adapted from [56, 63], while Fig. 7.7 is adapted from the linear level reduction algorithm in ([63], Ch. 10 and 12). The matrix $\mathbf{P}$ in Theorem 2.1 is obtained as

$$\mathbf{P} = \mathbf{I} + \mathbf{T}/t_0 \equiv \begin{bmatrix} \mathbf{P}_1^{(1)} & \mathbf{P}_0^{(1)} \\ \mathbf{P}_2^{(2)} & \mathbf{P}_1^{(2)} \end{bmatrix},$$

$t_0 \geq \max_{\forall k}\{|t_{kk}|\}$, $\mathbf{P}_0^{(1)} = \frac{t_{12}}{t_0}\mathbf{I}_1$, $\mathbf{P}_2^{(2)} = \frac{t_{21}}{t_0}\mathbf{I}_1$, and

$$\mathbf{P}_1^{(n)} = \begin{bmatrix} \mathbf{I}_2 + \frac{1}{t_0}\mathbf{A}_n & \frac{\lambda_1}{t_0}\mathbf{I}_2 & & & \\ \frac{r_{n,1}}{t_0}\mathbf{I}_2 & \mathbf{I}_2 + \frac{1}{t_0}\mathbf{B}_n & \frac{\lambda_1}{t_0}\mathbf{I}_2 & & \\ & \ddots & \ddots & \ddots & \\ & & \frac{r_{n,1}}{t_0}\mathbf{I}_2 & \mathbf{I}_2 + \frac{1}{t_0}\mathbf{B}_n & \frac{\lambda_1}{t_0}\mathbf{I}_2 \\ & & & \frac{r_{n,1}}{t_0}\mathbf{I}_2 & \mathbf{I}_2 + \frac{1}{t_0}\mathbf{C}_n \end{bmatrix}, \; n = 1, 2.$$

7.3.2 QoS Metrics

Here, we apply the simple tail dropping (or drop-on-full) buffer manager discussed in Section 2.11.1 and the channel state-dependent proportional scheduler (CSDPS) in (7.11) to DQoS analysis. Following the preceding argument, the proportion of traffic class i's packets that are dropped on arriving at the full buffer[4] is

$$L_i = \frac{1}{N} \cdot \sum_{y=\Phi} \mathbf{p}_\infty(y) \cdot I(y_i = B_i), \; i = 1, 2, \cdots, c, \tag{7.14}$$

where I is the indicator function in (7.2). Similarly, the corresponding normalized throughput is given by $T_i = 1 - L_i$, $i = 1, 2, \cdots, c$. Assuming a simple

[4] Here, we consider only full buffer related packet losses as wireless channel state dependent losses have already been accounted for in the effective scheduling rate (7.11).

first-come-first-served (FCFS) scheduling within a logical queue, the average queuing delay for packets of GeS_i is

$$D_i = \sum_{\mathbf{y}=\Phi} \frac{L_{\text{p}}}{N r_{y_{c+1},i}} \mathbf{p}_\infty(\mathbf{y})(n_i + 1) \cdot I(y_i = B_i), \; i = 1, 2, \cdots, c, \tag{7.15}$$

where y_{c+1} is the $(c+1)-$st element of the vector $\mathbf{y}$, which indicates the condition of the wireless channel, and L_{p} is the packet size in bits. These QoS parameters are the basis of the numerical examples in the next section.

7.3.3 Results and Discussions

The following values are used in the examples: $\sigma = 12$ dB, $\mu = 0.5$ dB, $0 \leq \gamma \leq 6$ dB, (7.3) and 7.4 are estimated using 500 runs of Monte Carlo simulations. To check the validity of the scheme the average packet arrival rates for all traffic classes are set to the same value, i.e. $\lambda_i = 0.5, \; i = 1, 2, 3, 4$. For the hybrid SAW-ARQ scheme (see Section 2.2), the idle time of the SAW-ARQ is set equal to a codeword transmission time. The SAW-ARQ is based on a BCH code with maximum FEC parity check bits and $\hat{r}_{\text{arq}} = 3$, i.e., $(\nu, \kappa, \tau, \hat{r}_{\text{arq}})$= $(2^l - 1, 2^l - 1 - l\tau, \tau = 2, 3)$ with $l = 6$. IP packet size for packets of all traffic classes is set to $L = 53$ bytes. Four traffic classes (i.e. $c = 4$) are differentiated and the available buffer capacity is shared in the ratio $(10 : 5 : 3 : 2)/20$. Two sets of numerical investigations are performed. In the first set of analysis the wireless link rate is set to $R_b = 144$ kb/s while the buffer capacity is varied between 20 and 30 packets. The results are shown in Fig. 7.3. In the second set, the dependence of the loss and queuing delay on the wireless link bandwidth is investigated at a fixed buffer capacity of 20 packets. The results are depicted in Figs. 7.4 and 7.5.

As expected, the loss and delay decrease with increasing link bandwidth, and the loss decreases with increasing buffer capacity. In general, the queuing delays are expected to increase with the buffer capacity. However, as Fig. 7.3 shows, this is not the case in the presented scheme, especially for traffic classes 1 and 2. This is due to the fact that buffer sharing and bandwidth sharing are intertwined in such a way that, as buffer capacity for a traffic class increases

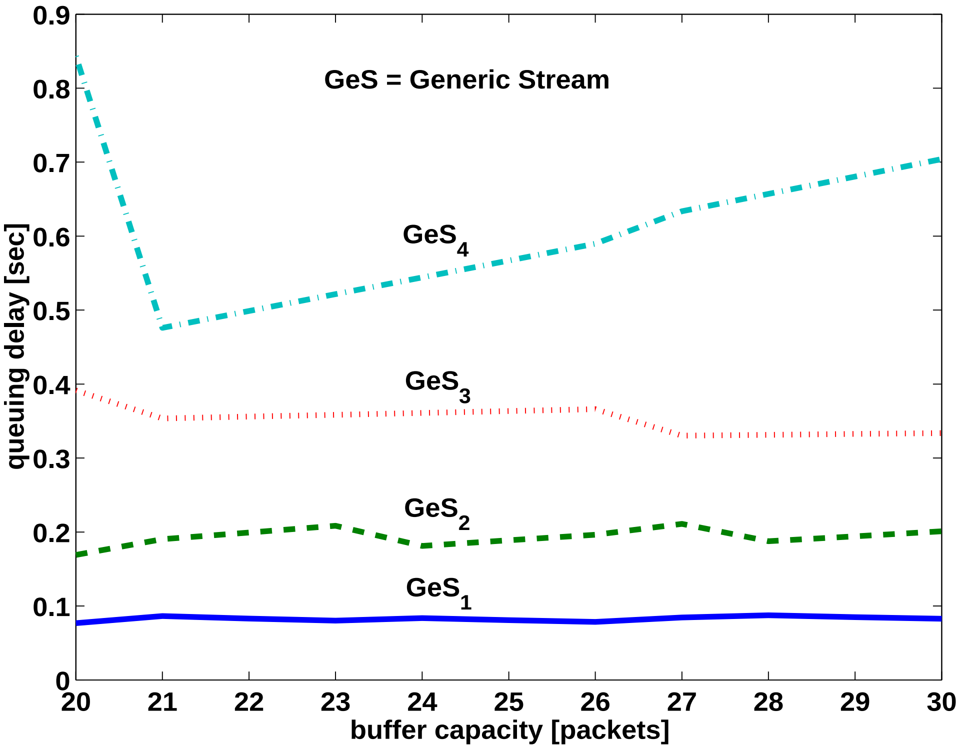

Figure 7.3: Steady-state analysis: expected packet queuing delay versus buffer capacity.

to reduce packet losses, its share of bandwidth also increases to offset buffering delays (see eq. 7.11). It is also observed from the figures that the fluctuation in QoS of traffic classes 1 and 2 are relatively small, as required by QoS-stringent applications traffic. As expected, packets belonging to traffic class i receive better QoS than packets of traffic class $i+1, i = 1, 2, \cdots, c-1$. An exception is the delay experienced by packets of traffic class 3, which is certainly an outlier.

7.4 Transient QoS Analysis

Following the argument in Section 2.1.4, we obtain the time-dependent state probabilities vector as

$$\mathbf{p}(t) = \mathbf{p}(0)e^{\mathbf{T}t}, \quad t \in \Re_+. \tag{7.16}$$

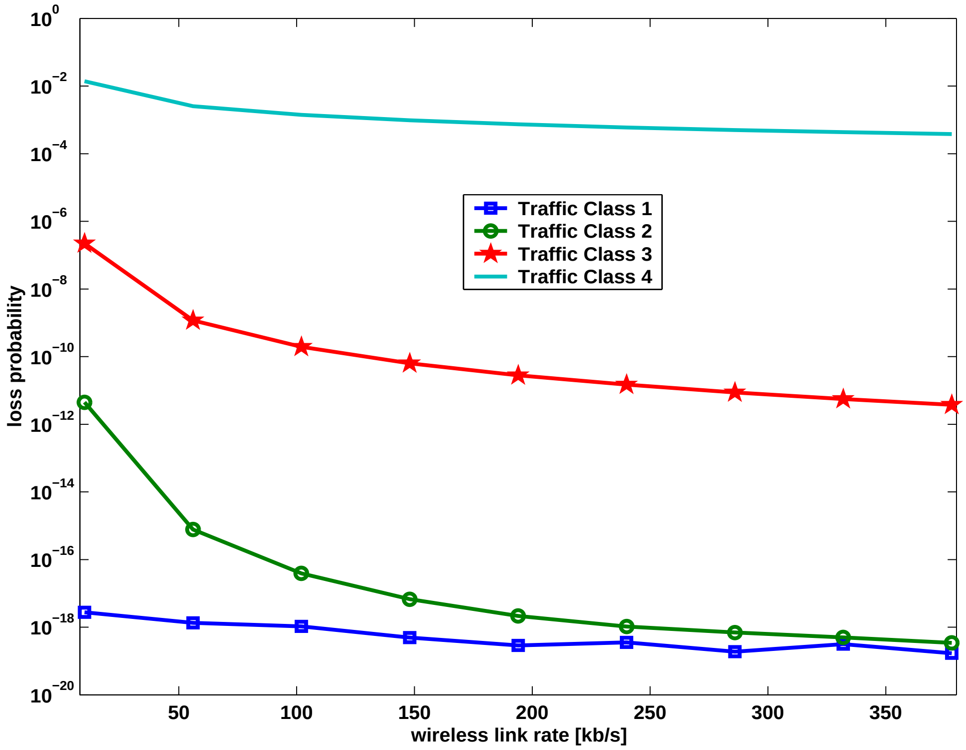

Figure 7.4: Steady-state analysis: loss probability versus 'raw' wireless link rate.

We assume here that the buffers are empty at the reference time $t = 0$, and that the wireless channel has equal probabilities of being in any of the $N = 2$ states at the reference time. Hence, with $c = 4$ and $n = 1, 2 (= N)$, we obtain the elements of $\mathbf{p}(0)$ as

$$p_{jklm|n}(0) = \begin{cases} \frac{1}{N} = \frac{1}{2}, & \text{if } \; j = k = l = m = 0 \\ 0, & \text{otherwise.} \end{cases}$$

Figure 7.6 shows the evolution of the transient delay for two traffic types: real time (GeS_1) and non-real-time (GeS_2). Again, as desired, it can be observed from Fig. 7.6 that the highest QoS traffic (GeS_1) undergoes lower queuing delays than the less time critical traffic GeS_2 over all the time scales shown. As the transient solution depends on the initial state vector $\mathbf{p}_0$'s, different $\mathbf{p}_0$'s lead to different results.

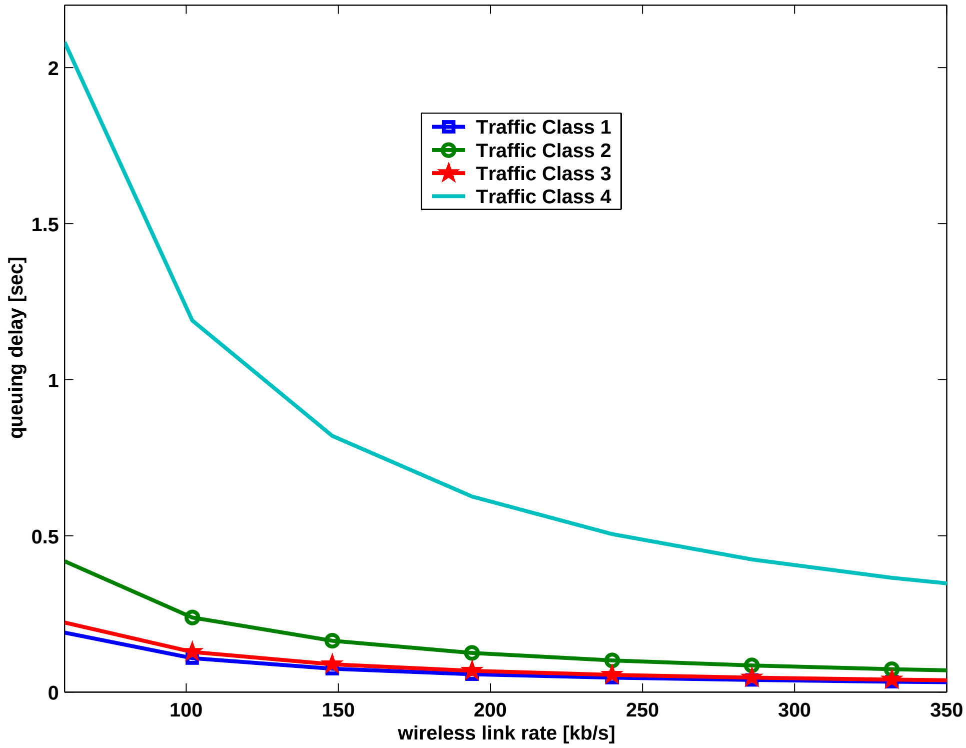

Figure 7.5: Steady-state analysis: average queuing delay versus 'raw' wireless link rate.

7.5 Summary

With the advent of multiservice mobile wireless Internet systems aiming to support integrated voice, data and video services, there is a recent need in the design, modelling and analysis of architectures that can support differing network traffic. This work analyses the performance of a relative differentiated QoS (DQoS) scheme applicable in Internet Protocol based wireless networks. The model accounts for wireless link's features, such as slow fading and error control. Network traffic is classified into QoS classes and given differentiated services based on buffer sharing and buffer share proportional scheduling rates. The resulting queuing system is analysed using matrix analytic methods. Numerical investigations show that the model can guarantee low loss and low queuing delays simultaneously to packets of a given traffic class via proper buffer sharing

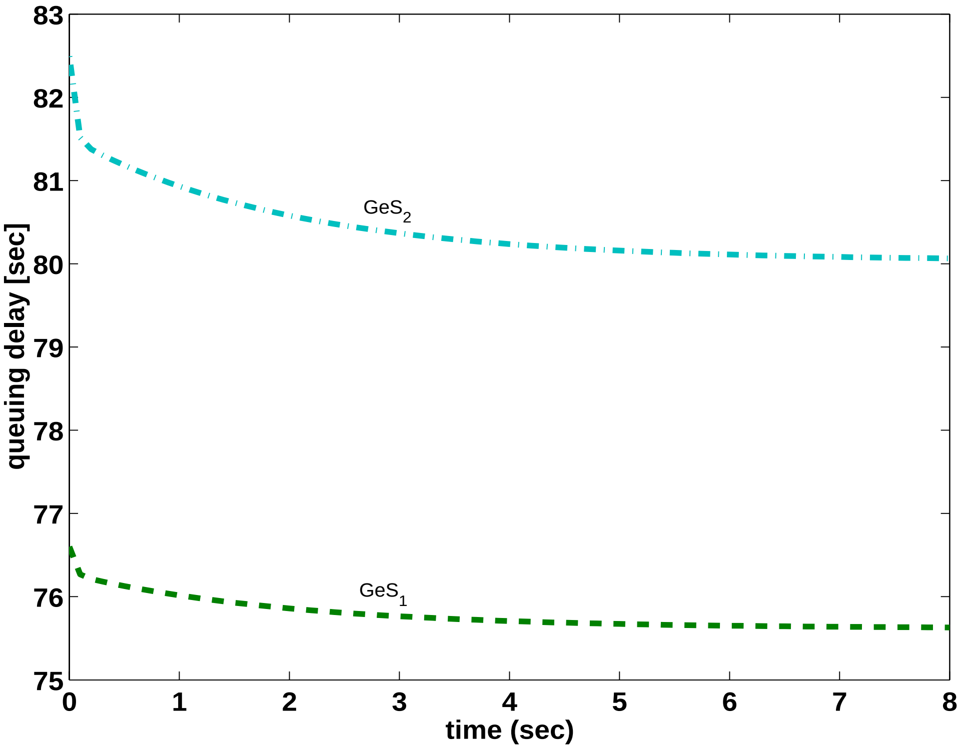

Figure 7.6: Transient delay evolution for two traffic streams: $R_b = 144kb/s$, $B = 8$ packets, buffer sharing ratio $= 3 : 1$.

policy.

The analysed four GeSs (traffic classes) can be mapped onto the assured forwarding (AF) PHB [50] of DiffServ, where GeS_1 through GeS_4 map, respectively, onto AF class 1 (AF_1) through AF class 4 (AF_4) to provide differentiated QoS levels. Although AF assures only the forwarding rate, but not the delay or jitter, the proposed scheme offers quantifiable loss and delay, given the buffer share or link bandwidth.

In the transient case, however, only two types of traffic are differentiated. These traffic can be categorized into real-time (GeS_1) or non-real-time (GeS_2), based on the level of timing sensitivities.

An inventor is a person who makes an ingenious arrangement of wheels, levers and springs, and believes in civilization – Ambrose Bierce

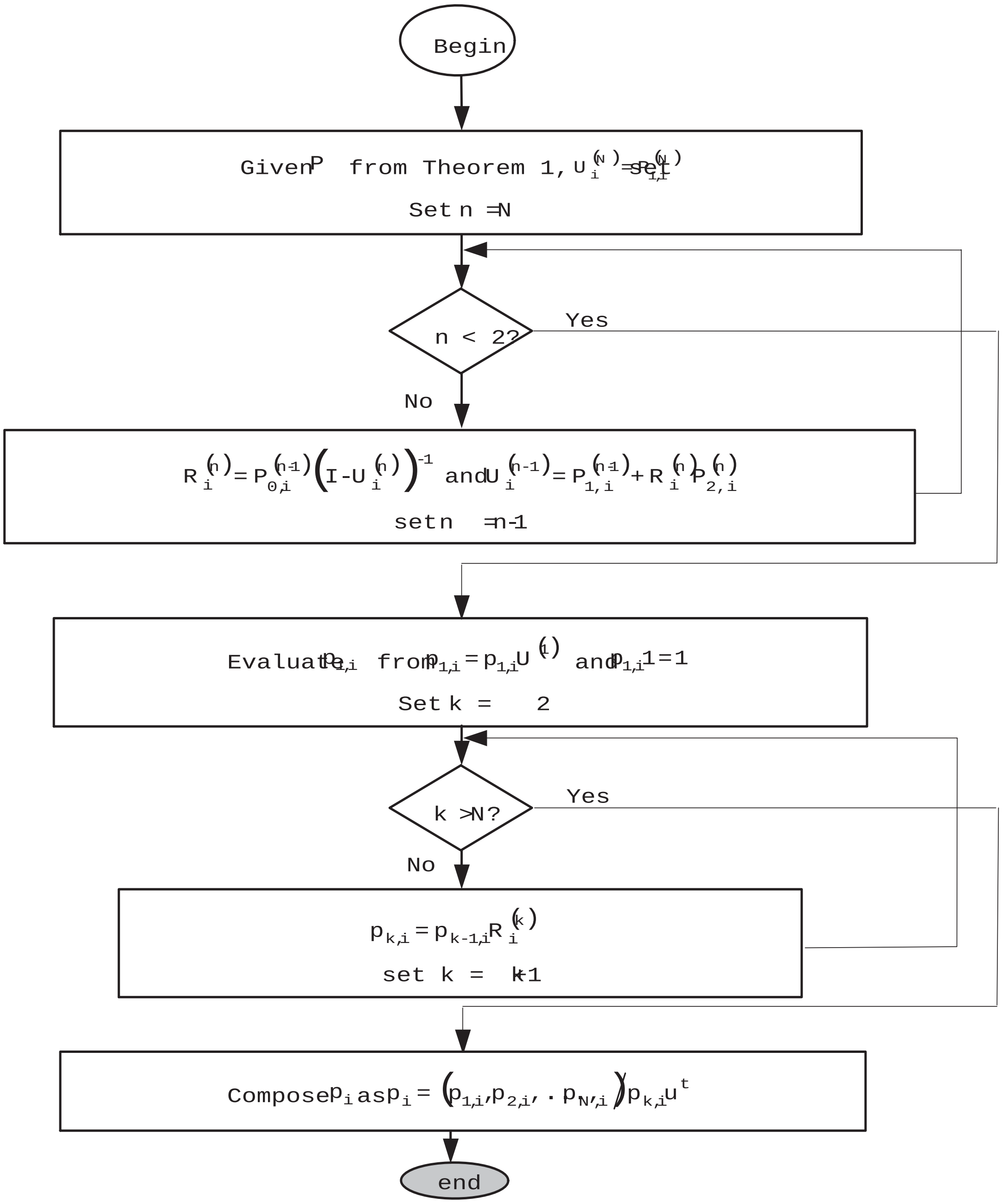

Figure 7.7: The linear level reduction algorithm (LLRA), $\mathbf{u}^t$ is the transpose of a vector of 1's.

Chapter 8

Analytical Modelling of a DQoS Scheme over Wireless Channels

8.1 Introduction

As the Internet has evolved to support heterogeneous services and applications requiring differing network treatment, there is a recent concern for its lack of quality of service (QoS). Also, with the advent of wireless Internet systems, mobile and base stations, as Internet portals, need to 'speak' the language of the Internet, the Internet Protocol (IP). Hence, modelling of simple and thus scalable wireless QoS schemes compatible with standard Internet QoS models is crucial. One approach to QoS support is to differentiate the traffic transferred over the Internet and assign appropriate QoS. Supporting different traffic types with the same QoS may not scale, may degrade network utilization, or may violate some flows' requirements, as a single service support cannot be optimized for different traffic types. Differentiated Services (DS) [10], designed for IP based networks, adopts this philosophy. DS enabled networks allocate network resources to aggregated flows and devolve maintenance of state information on individual flows in the aggregate to the access (edge) nodes, where the aggregation actually occurs, to achieve scalability.

Only few analytic models of DS exist to date, especially models accounting for wireless link features [18], hence the motivation for this chapter. This chapter analyses the queuing performance of a differentiated service model for IP based wireless networks. The differentiated service is provisioned via traffic classification, threshold dropping, and allocated buffer share proportional service rates. Deterministic bandwidth sharing schemes guarantee service to each traffic class to avoid 'knock-out', and to control both delay and loss. The two-state Gilbert-Elliot wireless link model is adopted. The steady-state properties of the resulting queues, accounting for both link state and buffer dynamics, are analysed using matrix analytic methods. Numerical investigations show that the model can guarantee both low loss rate and low expected queuing delays to a traffic class, by increasing its share of link bandwidth or buffer capacity in the network elements. We concentrate on the wireless access part of the architecture shown in Fig. 8.1, as core routers in the Internet have relatively high processing speeds and hence packet queuing and processing delays at such nodes may be relatively small. A related analysis with RIO buffer management and exponential Markov state dwell time is reported in [18]. Section 8.2 presents the system model, while the queuing analysis appears in Section 8.3. Finally, results and discussions appear in Section 8.4.

8.2 System Model

Assume a two-state Gilbert-Elliot channel (GEC) model with Markov state transition rates t_{ij}, $|i - j \leq 1$ and mean state 'dwell times' $d_1 = 1/t_{12}$ and $d_2 = 1/t_{21}$ sec, respectively (Fig. 8.1). The GEC model is usually used for slowly fading wireless links. Assume packets of c traffic classes arrive from an Internet server/router to an IP base station (IBS) according to an independent Poisson processes with respective mean arrival rates $\lambda_1, \lambda_2, \cdots, \lambda_c$. These packets are rescheduled to[1] IP mobile stations (IMS) in the cell of the particular IBS (Fig. 8.1). Let the IBS buffer of capacity B packets be virtually partitioned into c queues of sizes $B_1, B_2, \cdots, B_c$, where $B_I > B_j$, $\forall i < j$ and $\sum_{k=1}^{c} B_k = B$.

[1]The presented model and analysis is also adaptable to the uplink, as IMS will be sophisticated enough to support multi-applications traffic simultaneously. Also, applications like videoconferencing require symmetric links.

Within a queue class packets are served in the order of arrival (i.e. FIFO). A packet of class i is dropped if, upon arrival, there are Bi packets in its respective queue. Thus the actual aggregate packet arrival rate to the buffer is

$$\lambda = \sum_{k=1}^{c} \lambda_k I\{N_i(t) < B_i\}, \tag{8.1}$$

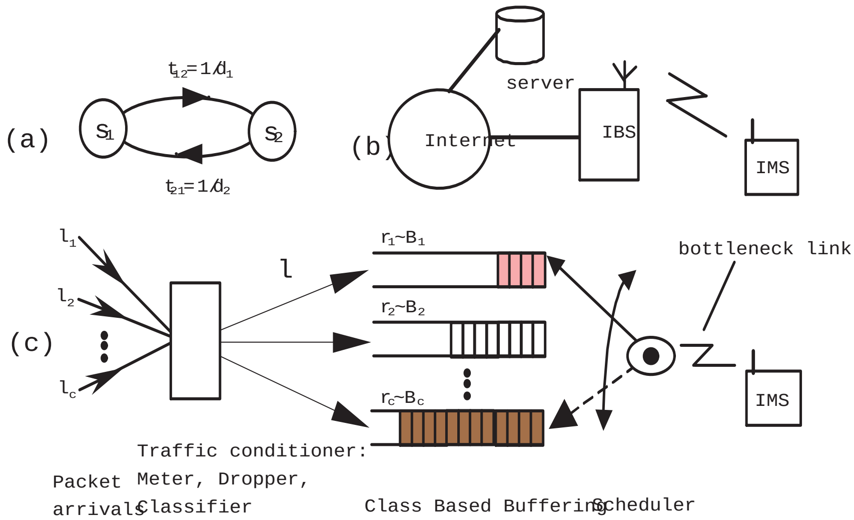

Figure 8.1: System model: (a) Gilbert-Elliot wireless link model, (b) overall network architecture, and (c) packet processing in the wireless router or IBS.

where $N_i(t)$ is the number of class i packets in the system at the scheduling instant t with $\lim_{t\to\infty} N_i(t) \to N_i$, and $I(\cdot)$ is the indicator function.

8.2.1 Packet Scheduling

Here we use the NP^{3}A scheduling to provide prioritised (differentiated) QoS to three types of traffic. We note from the discussions in Section 5.3.3 that the NP^{3}A scheduling is a special case of the CSDPS discussed in Section 4.8 when the weights (w_i) in the scheduling rates

$$r_{n,i} = w_i \beta_n R_b, \quad i = 1, 2, \cdots, c \text{ and } n = 1, 2, \cdots, N_c \tag{8.2}$$

are given by

$$w_i(NP^3A) = \begin{cases} q_i\tilde{q}_2\tilde{q}_3, & \text{if } i = 1 \\ \tilde{q}_1\tilde{q}_i\tilde{q}_3 f_2 + q_i q_3 f_2 + q_i\tilde{q}_3, & \text{if } i = 2 \\ \tilde{q}_1\tilde{q}_2\tilde{q}_i\tilde{f}_2 + q_2 q_i\tilde{f}_2 + q_i\tilde{q}_2, & \text{if } i = 3 \end{cases} \tag{8.3}$$

Without loss of generality, we set $w_i(NP^3A) = B_i/B$ in this chapter. The ratio B_i/B may be proportional to the percentage of MAC layer resources (e.g. CDMA codes) allocated to class i packets. We note that if $B_i > 0$ then $r_{n,i} \neq 0$, so each traffic class is guaranteed a positive rate and hence none of the queues is starved of service, as in strict priority queuing.

8.3 Queuing Analysis

In the following, assume that none of the events: service completion, packet arrival and link state change occur simultaneously. Then, the system can be described as a $(c+1)$-st dimensional embedded Markov process $\{(N_1(t), N_2(t), \cdots, N_c(t);\ S(t)),\ t \in \Re_+\}$ on the state space $Z = \{(N_1, N_2, \cdots, N_c;\ S) : N_i = 1, 2, \cdots, B_i,\ i = 1, 2, \cdots, c;\ S = 1, 2\}$. For example, the possible state transitions and the instantaneous rates for this queuing model are given in Table 8.1 for $c = 3$. Define $p_{ij\cdots k|n} = Pr[N_1 = i, N_2 = j, \cdots, N_c = k | S = n]$ and let $\mathbf{p}_\infty$ be the lexicographic arrangement of $p_{ij|n}$, i.e. in increasing value of N_1, and for a given N_1, in increasing value of N_2, etc.

This section evaluates the square state transition rate matrix $\mathbf{Q} = (q_{kl})$ of order $N_Q = 2\times$ using Table 8.1, where q_{kl} is the instantaneous transition rate from system state k to l for $k \neq l$ and $q_{kk} = -\sum_{\forall k \neq l} q_{kl}$. (Note that q_{kk} is negative, as the rows of $\mathbf{Q}$ should sum to zero.) Then, we evaluate the steady-state probability vector $\mathbf{p}_\infty$ of length N_Q via the global balance and probability conservation equations (6.7), with $\mathbf{T}$ replaced by $\mathbf{Q}$. We then use $\mathbf{p}_\infty$ to analyse the average per-packet queuing delays D_i and the packet loss probability L_i for traffic class i, $i = 1, 2, \cdots, c$.

Let the identity matrices of respective orders $N_Q/2$, $\prod_{i=2}^{c}(1+B_i)$ and $\prod_{i=3}^{c}(1+B_i)$ be defined by $\mathbf{I}_0$, $\mathbf{I}$ and $\mathbf{I}_1$. Then, from Table 8.1, we have

Table 8.1: State transitions and corresponding rates for the queue of Fig. 8.1.

From state	**To state**	**Rate**	**Remarks**
$(b_1, \cdots, b_i, \cdots, b_c;\ n)$	$(b_1, \cdots, b_i - 1, \cdots, b_c;\ n)$	$r_{n,i}$	service completion
$(b_1, \cdots, b_i, \cdots, b_c;\ n)$	$(b_1, \cdots, b_i + 1, \cdots, b_c;\ n)$	λ_i	packet arrival
$(b_1, \cdots, b_i, \cdots, b_c;\ n)$	$(b_1, \cdots, b_i, \cdots, b_c;\ k)$	$t_{n,k}$	link state change

$$\mathbf{Q} = \begin{bmatrix} \mathbf{Q}_{11} & \mathbf{Q}_{12} \\ \mathbf{Q}_{21} & \mathbf{Q}_{22} \end{bmatrix},\ \mathbf{Q}_{12} = \frac{1}{d_1}\mathbf{I}_0,\ \mathbf{Q}_{21} = \frac{1}{d_2}\mathbf{I}_0,\ \mathbf{Q}_{nn} = \begin{bmatrix} \mathbf{A}_n & \lambda_1\mathbf{I} & & & \\ r_{n,1}\mathbf{I} & \mathbf{B}_n & \lambda_1\mathbf{I} & & \\ & \ddots & \ddots & \ddots & \\ & & r_{n,1}\mathbf{I} & \mathbf{B}_n & \lambda_1\mathbf{I} \\ & & & r_{n,1}\mathbf{I} & \mathbf{C}_n \end{bmatrix}.$$

The submatrices of $\mathbf{Q}$ are square, band and block-tridiagonal of order $\prod_{i=2}^{c}(1 + B_i)$. For $n = 1, 2$ and $j = 1, 2, 3$ they are given by

$$\mathbf{A}_n = \begin{bmatrix} \mathbf{A}_{1n} & \lambda_2\mathbf{I}_1 & & & \\ r_{n,2}\mathbf{I}_1 & \mathbf{A}_{2n} & \lambda_2\mathbf{I}_1 & & \\ & \ddots & \ddots & \ddots & \\ & & r_{n,2}\mathbf{I}_1 & \mathbf{A}_{2n} & \lambda_2\mathbf{I}_1 \\ & & & r_{n,2}\mathbf{I}_1 & \mathbf{A}_{3n} \end{bmatrix},$$

$$\mathbf{A}_{jn} = \begin{bmatrix} -a_{j1}(n) & \lambda_3 & & & \\ r_{n,3} & -a_{j2}(n) & \lambda_3 & & \\ & \ddots & \ddots & \ddots & \\ & & r_{n,3} & -a_{j2}(n) & \lambda_3 \\ & & & r_{n,3} & -a_{j3}(n) \end{bmatrix}.$$

Replacing $\mathbf{A}_{jn}$ and $a_{jk}(n)$ in $\mathbf{A}_n$ with $\mathbf{B}_{jn}$ and $b_{jk}(n)$ or $\mathbf{C}_{jn}$ and $c_{jk}(n)$ respectively yields $\mathbf{B}_n$ or $\mathbf{C}_n$. The diagonal elements of $\mathbf{A}_{jn}$, $\mathbf{B}_{jn}$ and $\mathbf{C}_{jn}$ are given by $a_{11}(n) = 1/d_n + \sum_{i=1}^{c} \lambda_i$, $a_{12}(n) = r_{n,3} + a_{11}(n)$, $a_{13}(n) = a_{12}(n) - \lambda_3$, $a_{21}(n) = r_{n,2} + a_{11}(n)$, $a_{22}(n) = r_{n,3} + a_{21}(n)$, $a_{22}(n) = r_{n,2} + a_{13}(n)$, $a_{31}(n) = a_{21}(n) - \lambda_2$, $a_{32}(n) = a_{31}(n) + r_{n,3}$, $a_{33}(n) = a_{32}(n) - \lambda_3$, $b_{11}(n) = a_{11}(n) + r_{n,1}$,

$b_{12}(n) = b_{11}(n) + r_{n,3}$, $b_{13}(n) = b_{12}(n) - \lambda_3$, $b_{21}(n) = b_{11}(n) + r_{n,2}$, $b_{22}(n) = b_{12}(n) + r_{n,2}$, $b_{23}(n) = b_{22}(n) - \lambda_3$, $b_{31}(n) = b_{21}(n) - \lambda_2$, $b_{32}(n) = b_{31}(n) + r_{n,3}$, $b_{33}(n) = b_{32}(n) - \lambda_3$, $c_{11}(n) = a_{11}(n) - \lambda_1$, $c_{12}(n) = c_{11}(n) + r_{n,3}$, $c_{13}(n) = c_{12}(n) - \lambda_3$, $c_{21}(n) = c_{11}(n) + r_{n,2}$, $c_{22}(n) = c_{21}(n) + r_{n,3}$, $c_{23}(n) = c_{22}(n) - \lambda_3$, $c_{31}(n) = c_{21}(n) - \lambda_2$, $c_{32}(n) = c_{31}(n) + r_{n,3}$, and finally $c_{33}(n) = c_{32}(n) - \lambda_3$.

Given the matrix $\mathbf{Q}$, we evaluate the stationary vector $\mathbf{p}_\infty$ and the QoS metrics of interest using the algorithms presented in Chapter 5 or the LLRA presented in Fig. 7.7 of Chapter 7.

8.3.1 DQoS Metrics

Following the preceding analysis, the probability that a packet of traffic class i arriving at the buffer is dropped (lost) is

$$L_i = \sum_{\mathbf{X} \in Z} p_\infty(\mathbf{x}) I\{x_i = B_i\}, \quad i = 1, 2, \cdots, c. \tag{8.4}$$

Similarly, the expected queuing delay using FIFO scheduling is

$$D_i = \sum_{\mathbf{X} \in Z} p_\infty(\mathbf{x}) I\{x_i = B_i\}(N_i + 1) n_p / r_i^{x_{c+1}}, \quad i = 1, 2, \cdots, c. \tag{8.5}$$

where n_p is the packet size in bits.

8.4 Numerical Investigations and Discussions

The following parameters are used in the numerical examples: $c = 3$, $d_1 = 41.97$ *sec*, $d_2 = 6.95\,\text{sec}$, $\beta_1 = 0.2$, $\beta_2 = 0.8$, $\lambda_1 = \lambda_2 = \lambda_3 = 0.7$ and $n_p = 100$. Two sets of examples, based on the analytical model, are investigated. In one case, the dependence of the delay and loss on the buffer size is examined at fixed link bandwidth (Figs. 8.2 and 8.3). In the other case, the dependence of the delay and loss on link bandwidth at fixed buffer size is examined (Figs. 8.4 and 8.5). In both cases the available buffer size is shared in the ratio $1/2 : 1/3 : 1/6$. The values of β_n and d_n can be obtained from a wireless link model such as used in [39].

It is observed in all the figures that traffic class 1 packets experience the best QoS in the QoS metrics analysed, while class 3 packets undergo the worst QoS, as expected. As expected, the loss probability curves decrease with increasing buffer size (Fig. 8.3) and link bandwidth (Fig. 8.5), but flatten above a certain buffer capacity (B_{th}). Above (B_{th}), further increase in the buffer capacity does not remarkably enhance the loss level. Appropriate loss and delay profiles can thus be obtained by manipulating a single buffer parameter, or the link bandwidth. In general, the queuing delays are expected to increase with increasing buffer capacity. However, almost the opposite effect is observed in the presented model, since the service rate 8.2 is proportional to the allocated buffer share of a traffic class. This guarantees that the loss probability is minimized without penalizing the queuing delays. Note that, for $B < 5$, all arriving packets of class 3 are dropped (hence zero queuing delay), as it obtains no share of the buffer capacity. Scalability is not an issue as the presented model is simple (no packet reordering as in WFQ and CBQ) and operates at the IBS in the *last mile* links where traffic load and link speed are relatively small, compared to high-speed backbone links.

The presented model can e.g. be used to provide service differentiation in a forwarding class of an assured forwarding Per-Hop Behavior (PHB) of DS. It can also be mapped onto standard DS as: class 3 onto best-effort forwarding (BEF), class 2 onto assured forwarding (AF) and class 1 onto expedited forwarding (EF) PHBs. These in turn can be mapped onto the 3G/UMTS traffic classes as: conversational and streaming onto EF, interactive onto AF and background onto BEF. Note that few traffic classification is attractive for scalable QoS schemes.

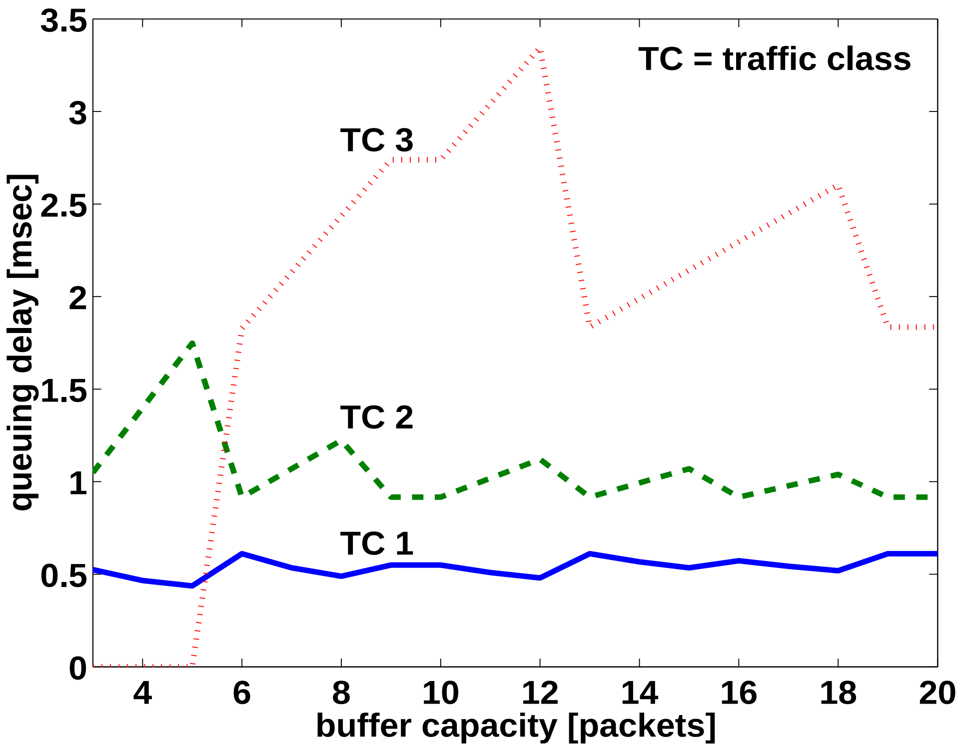

Figure 8.2: Expected packet queuing delay vs. buffer size (B) at $R_b = 10$ kb/s.

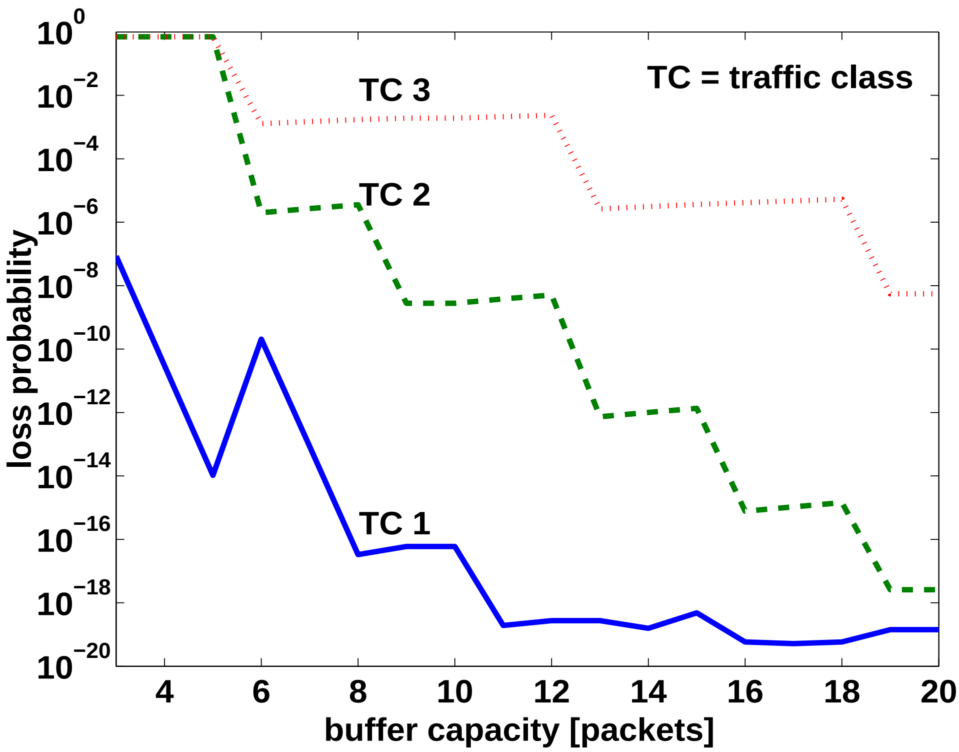

Figure 8.3: Packet loss probability vs. buffer size (B) at $R_b = 10$ kb/s.

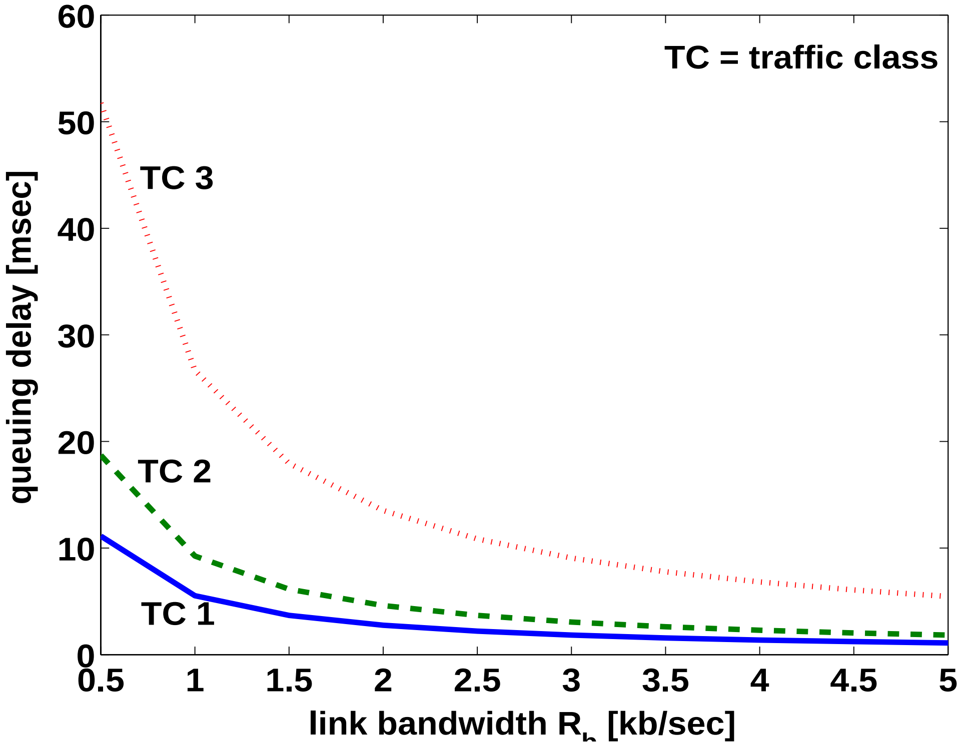

Figure 8.4: Expected queuing delay vs. link bandwidth (R_b) at $B = 10$ packets.

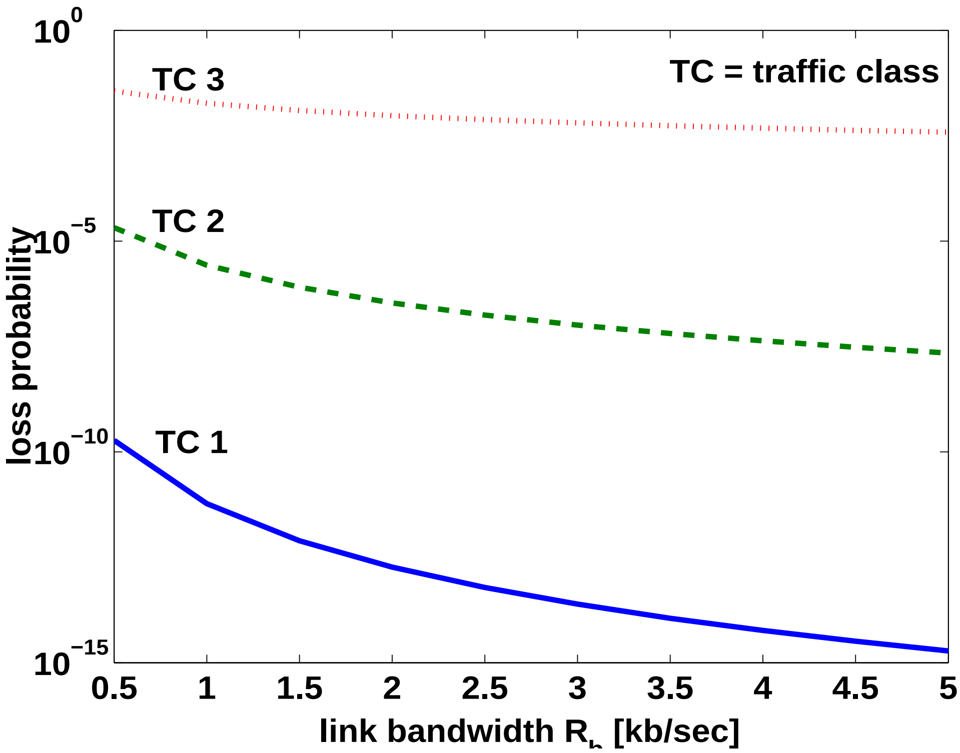

Figure 8.5: Packet loss probability vs. link bandwidth (Rb) at $B = 10$ packets.

Chapter 9

Conclusions & Future Work

Quality of service differentiation has never achieved much attention and relevance until the advent of the convergence of mobile wireless network and the fixed Internet, that is, Internet Protocol (IP) based mobile wireless networks, or wireless Internet. These networks are poised to support multimedia applications' traffic with diverse QoS sensitivities. To date, most traffic transferred over the Internet still undergo best-effort forwarding, which does not differentiate the traffic it handles, nor guarantees the successful transfer of packets from the source to the intended destinations, let alone guaranteeing loss and timing bounds.

This thesis presents a design of an IP based wireless network architecture supporting IP QoS, designs differentiated QoS schemes for IP networks, and performed performance analysis using queuing theory, matrix analytic methods to stochastic modelling, linear algebra, and optimisation theory. With the vastness of the field of mobile wireless Internet and its complexity, it is not even thinkable to claim a complete treatment of the field. However, an attempt is made to do a representative coverage. The major contributions of this thesis is three-fold:

1. First, the thesis proposes a QoS-enabled wireless Internet access architecture, which leverages the micromobility in wireless standards to reduce mobile IP weaknesses, such as long handoff delay, to achieve effective interworking between mobile wireless networks and the global, fixed In-

ternet. Although the idea here is applicable to any wireless standard, the design example in this thesis is based on the IEEE 802.11b wireless LAN standard, which is commonly referred to as *dot11*.

2. Second, it proposes a framework for a class of packet scheduling schemes, which consider the (a) QoS requirements of the applications' traffic; (b) the wireless channel state (reflected in instantaneous data rate or noise level); and (c) optimises the usage of the expensive wireless resource. The operations of the QoS-enabled, channel state-dependent packet scheduling schemes are analysed using optimisation theory, linear algebra and stochastic modelling.

3. Third, the thesis analyses the effects of wireless channel properties on differentiated QoS (DQoS) schemes, using two-dimensional, channel-state-dependent queuing theory. Analytical model of DQoS schemes, especially models accounting for wireless channel properties, such as fading, spatio-temporally varying quality and low rate, is not properly covered in the open literature, and hence the motivation for this work. The wireless channel is discretized into discrete-time Markovian states based on the received signal-to-noise plus interference ratio (SNIR), which also reflects on the instantaneous link quality. The link quality, in turn, influences the QoS experienced by the transported applications sitting on top of the ISO/OSI protocol hierarchy. The parameters the Markovian states are evaluated using realistic physical channel noise models and transceiver properties, such as modem. Source traffic models are used in the analysis.

4. Lastly, the thesis provides an extensive introduction to, and provides a detailed background material for the new area of mobile wireless Internet systems, upon which a considerable future research can be based.

In conclusion the following paragraph discusses some interesting future research directions. The wireless channel state dependent packet schedulers considered in Chapter 4 is an interesting area for upcoming IP based multi-service networks, and hence requires a deeper exploration. Statistical behaviour of the error control schemes were also not accounted for. Instead, their output efficiencies were used in the DQoS modelling. Analysis considering the detailed operation of these schemes may be interesting too.

Another interesting research direction is the development of new queuing theory based on the recent discovery of long-range dependence (or self-similarity) of IP traffic. Today's queuing theory is solidly developed, but under the traditional assumption that traffic inter-arrival times are exponentially distributed, leading to Poisson arrival process. Hence, application of today's queuing theory to the analysis of modern IP based communications network may be heavily limited, or even flawed. Notwithstanding, few parts of this thesis uses the Poisson traffic assumption for the only reason of analytical beauty. Extensive investigation, especially via in-depth simulation modelling, to clarify the credibility or incredulity of modelling non-Poisson traffic with Markov chains is also interesting. Traditionally, queuing theory is link-layer independent. This thesis, however, embedded link layer properties into queuing analysis. Queuing theory accounting for spatio-temporarily varying nature of wireless channels is quite useful for a unified performance analysis of protocols for wireless Internet systems.

The scale, properly speaking, does not permit the measure of the intelligence, because intellectual qualities are not superposable, and therefore cannot be measured as linear surfaces are measured $-$ Alfred Binet.

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What I am going to tell you about is what we teach our physics students in the third or fourth year of graduate school... It is my task to convince you not to turn away because you don't understand it. You see my physics students don't understand it... That is because I don't understand it. Nobody does. – Richard P Feynman.

Glossary

Ad hoc network is a mobile wireless network without fixed infrastructure such as base stations or routers. As its name says, it is formed without any prior planning, and it is most useful during emergencies. Ad hoc networks are mostly discussed under the acronym *MANET*, meaning Mobile Ad hoc NETworking.

Classifier is an algorithm that organizes packets into groups based on the content of their headers according to defined rules.

Datagram used to be the name for the ISO/OSI Layer 3 protocol data unit. With the advent of IPv6, however, *packet* is used instead of *datagram*.

Generic Stream (GeS) can be a stream of packets belonging to a single application, or in the Internet parlance, a single flow, as considered in the IETF integrated services standards. A GeS can also refer to packets of aggregated flow(s) (or a traffic class), as viewed in the IETF differentiated services (DiffServ) standards. According to the IPv6 standard [22], a flow is a sequence of packets transferred between a particular source and a particular unicast or multicast destination node(s) for which the source desires a special handling by routers along the flow's path. A flow is identified by source and destination nodes' IP addresses, flow label, and a priority/QoS class. A GeS can also be user-centric (i.e. map onto users) instead of being traffic-centric.

Internet Protocol (IP) is a connectionless packet based Layer 3 protocol which internetworks multiple packet-switched networks. It is the backbone protocol of the global Internet.

Internetworking is a collection of packet-switched and broadcast networks which are interconnected by routers.

Killer apps are applications that use most of the available network resources and perhaps generate most of service providers' revenue.

Local Area Network (LAN) is the interconnection of data communications devices within a small area. The scope of a LAN is larger than that of a PAN but smaller than a MAN.

Metropolitan Area Network (MAN) is similar to a LAN except that its size and scope are larger than those of a LAN.

Mobile Computing offers the possibility of using the same computer in different locations in contrast to traditional fixed desk-top PCs.

Mobile IP (MIP is a secure and robust standard for Internet mobility that allows nodes to move in the Internet.

Mobile Network is a wireless network which allows its users to move about whilst maintaining network connectivity. During active communications such networks allow users to change their point of attachment (interface) to the network.

Mobility is the ability to maintain connectivity to a network service whilst changing points of attachment to the netwok.

Packet is in general an ISO/OSI Layer 3 protocol data unit, i.e., a bunch of bits comprising data (payload) and control and signalling information.

Packet scheduler is an algorithm that decides the order in which packets are served at a server.

Packet Switching is a way of packaging long messages into short packets, appending headers to them, transmitting them independently, and assembling them at the destination. Packets belonging to the same message may traverse different paths, and hence are liable to different impairments such as transmission delays.

PHB is the externally observable forwarding treatment of aggregate traffic at a DiffServ capable node.

Personal Area Network (PAN) is a body networking in which small interconnected devices are attached to the human body. Its scope is the smallest among all the area networking scenarios. Short range radio standards such as Bluetooth is a potential protocol for PANs. The question whether or not PANs are healthy to use is sensitive.

Pervasive Computing is a new paradigm of anytime, anywhere communications and it is also often referred to as ubiquitous computing. Pervasive Computing can be realised using ad hoc network protocols.

Portability is the ability to connect to a network after establishing a link at a new point of attachment.

QoS is a set of attributes and their values that, taken together, characterize the performance experienced by a flow. See also Section 2.5.

QoS Enabled Network is a network whose elements differentiate the traffic they process and offer differentiated treatment such as service order and buffer share.

RED, meaning random early discard, is a congestion avoidance active buffer management scheme which drops a fraction of packets when a queue size exceeds a given threshold.

Router is a Layer 3 network element which internetworks two data (computer) networks by forwarding packets not addressed to itself between them. A router assumes that all devices attached to the network uses the same communications protocols and architecture.

Spectrum is a range of radio frequencies usable for information transmission wirelessly.

Wide Area Network (WAN) is similar to a MAN except its size and scope which is larger.

Wireless DiffServ (WDS) The IETF DS standard is in general link layer independent. WDS is DS which is tailored to link layer (especially, wireless link) features.

Wireless IP (WIP) is a wireless network using the Internet Protocol as layer 3 protocol. Such networks use packet switching, contrasting conventional wireless networks based on circuit switching.

Wireless Network is a network with cordless user/network interfaces. Wireless networks have two main flavours: terrestrial and satellite networks.

Wireless Router (WR) is a router with a wireless interface.

Well-being and happiness never appeared to me as an absolute aim. I am even inclined to compare such moral aims to the ambitions of a pig – Albert Einstein.

Autobiography

Amoakoh Gyasi-Agyei had been a PhD scholar in the department of Electrical & Electronic Engineering of Adelaide University, South Australia, until February 2003, where he completed his PhD with thesis entitled "QOS Enabled IP Based Wireless Networking: Architectures, Protocols and Performance Analysis." From February 2003 Amoakoh is a Lecturer at the Central Queensland University (CQU), QLD, Australia. He obtained M.Sc. degree in Digital Communications Engineering from Chalmers University of Technology (CTH), Gothenburg, Sweden, and a B.Sc. degree in Electrical Engineering from Hamburg-Harburg University of Technology (TUHH), Germany. Amoakoh is alumnus of Prempeh College, Ghana.

During his undergraduate degree program in Hamburg at TUHH, Gyasi-Agyei worked for over 4 years as part-time Engineering Assistant (HIWI) at the same university. Between 1997 and 2001, he worked at many places, both in renowned academic institutions and in the Telecom industry. Below are some of the places he's worked before. He was a Visiting Research Scientist at RAMO Lab, Information and Communications University, Daejeon, South Korea in mid-2002. Between October 2000 and March 2001, he was a Visiting R&D Engineer at the Institute for Communications Research (ICR), National University of Singapore (NUS). From April 1999 to December 2000, he was with the Communications Laboratory (ComLab) as a Research Engineer, and from February 1998 to April 1999 a Research Scientist with the Networking Laboratory, both at Helsinki University of Technology (HUT), Finland. While at HUT's ComLab, he also organized graduate level courses with Prof Seppo J. Halme and some of the course materials were published into a book under the ISBN number 951-22-4982-0. Prior to joining HUT, he worked at Philips Semiconductors in Hamburg, Germany around 1992/3, and at Ericsson Mobile Data Design AB in Gothenburg, Sweden, around 1997.

Gyasi-Agyei's current research interests centre around QoS provisioning in Internet Protocol based Wireless Networks, pervasive networking protocols, Packet Schedulers, design of network solutions and architectures, and network performance analysis. Gyasi-Agyei is a member of both the IEEE and the IEE. While studying in Hamburg, he became a Students' Representative of the students' council (ASTA) in 1994/95, and also held a Postgraduate Students' Representative position at Adelaide University's Student Council. Amoakoh has one wife, two daughters and seven sieblings, many friends around the world, and he enjoys knowing many more people. Interested in joining the informal global friendship alliance?

This Thesis was typed and formatted by Amoakoh Gyasi-Agyei