Mobility management and multihoming in heterogeneous networks

Nguyen, Dang Duc

2010

Nguyen, D. D. (2010). Mobility management and multihoming in heterogeneous networks. Master's thesis, Nanyang Technological University, Singapore.

<https://hdl.handle.net/10356/47493>

<https://doi.org/10.32657/10356/47493>

Nanyang Technological University

Downloaded on 26 Apr 2023 17:55:51 SGT

MOBILITY MANAGEMENT AND MULTIHOMING IN HETEROGENEOUS NETWORKS

DANG DUC NGUYEN

School of Computer Engineering

A thesis submitted to the Nanyang Technological University
in partial fulfillment of the requirement for the degree of
Master of Engineering

2009

Acknowledgements

I would like to thank my supervisor, Associate Professor Yeo Chai Kiat, for her guidance, support, and encouragement during this research project.

I would like to thank my co-supervisor, Associate Professor Lee Bu Sung Francis, for his support and encouragement.

I would like to express my gratitude to Mr. Xia Yang, Dr. Lim Teck Meng and Dr. Juki Wirawan Tantra and Mr. Mai Ngoc Son for giving me useful guidance, support and discussions.

I would like to thank my family and my friends for the encouragement and continuous support.

Finally, my appreciation goes to the laboratory technicians at CeMNet and the other research officers at this laboratory.

Dang Duc Nguyen

Nanyang Technological University

August 2009

Abstract

Heterogeneous networks become popular nowadays with wireless technologies such as WiMax, 3G/UMTS, WiFi, in which all could be deployed in a single device. The combination of multiple interfaces enables mobile devices to use either one or all of the interfaces at the same time. However, the support for mobility raises new issues not only in handoff management but also in the quality of service (QoS). Various research works have focused on how to manage QoS requirements in heterogeneous networks where delay sensitive real-time applications need low delay-tolerance and high bandwidth links.

A novel framework to support mobility in heterogeneous network for multihomed devices with QoS support is proposed. Based on a cross-layer handoff approach, we make use of SIP protocol to ensure seamless connection during handover across networks. The solution employs multiple interfaces to take advantage of the multihoming capability of the device thus enhances QoS support for data transfer, especially in multimedia streaming. Unlike many current proposals, the framework can support both UDP and TCP transmissions where UDP is less reliable while TCP incurs extra delay and overhead. The framework has been implemented in a testbed. Experimental results collected from real-time streaming show that the framework can provide seamless mobility support and optimized flow distribution for heterogeneous multihomed mobile nodes.

The seamless mobility idea is further extended. An application framework to support seamless handoff between 3G/UMTS and WLAN for VoIP applications on the Symbian

platform is proposed. VoIP applications require high quality and low latency transmissions. Unlike other approaches, the proposed framework lies in the application layer of the mobile devices and hence does not require a change in network infrastructure. Neither is modification required in the kernel layer. Hence it is portable among mobile devices and is easily customized for different VoIP systems.

We further apply the multihoming concept to Delay Tolerant Networks (DTN) and modify the DTN routing protocol MaxProp to exploit multiple network interfaces. Simulation results show that the multihomed MaxProp improves delivery rate and reduces message latency compared to the original MaxProp. The improvement is even more prominent when buffer space is limited in the DTN node. The DTN simulator, ONE (Opportunistic Network Environment) has also been modified to support the simulation of DTN with multiple interfaces.

Contents

Acknowledgements	ii
Abstract	iii
List of Tables	ix
List of Figures	x
List of Acronyms	xii
Chapter 1 Introduction	1
1.1 Motivation and objectives	1
1.2 Major Contributions of the Dissertation	3
1.2.1 Multihomed Mobility Framework	3
1.2.2 Mobility for Mobile Devices	3
1.2.3 Multihomed Node for Delay Tolerance Networks	4
1.3 Organization of Dissertation	4
Chapter 2 Literature Review	6
2.1 Current Trends in Mobility Management	6
2.2 QoS in Heterogeneous Networks	8

2.2.1	Definition of Heterogeneous Networks	8
2.2.2	Integration of Access Networks for Multiple-Interface Devices	10
2.2.3	Mobility in Heterogeneous Networks	11
2.2.4	QoS in Heterogeneous Networks	12
2.3	Delay Tolerant Networks	13
Chapter 3 Multihoming in UDP		17
3.1	Introduction	17
3.2	System Architecture	19
3.2.1	Multihoming and Mobility	19
3.2.2	Address Translation	21
3.2.3	QoS Support and IKPR Service	24
3.3	Performance Evaluation	25
3.3.1	Experimental Results	25
3.3.2	Discussions	29
3.4	Summary	30
Chapter 4 Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices		32
4.1	Introduction	32
4.2	Design Considerations and Challenges	33
4.2.1	Requirements for VoIP service	34
4.2.2	Design Considerations	34
4.2.3	Design Challenges	36
4.3	System architecture	36
4.3.1	Network Manager	38
4.3.2	VoIP Session Manager and Streamer Module	41

4.4	Performance Analysis	43
4.5	Conclusion	47
Chapter 5 Enhancement of Seamless Multi-homing Connectivity in Het-		
erogeneous Networks for TCP Communication		48
5.1	Introduction	48
5.2	Motivation, Challenges and Approaches	49
5.3	System Architecture	50
5.3.1	IP Swapping Module	51
5.3.2	IP and Port Mapping Module	53
5.3.3	Packet Redirecting Module	54
5.3.4	Communication Module	55
5.3.5	Intelligent Software Agent	55
5.4	Implementation	56
5.4.1	Experiment Settings	56
5.4.2	Experimental Results	58
5.5	Discussion	63
5.5.1	Aggregate Link and Seamless Connectivity	63
5.5.2	Routing based vs Tunnel based	63
5.5.3	Network Detection and Handoff Algorithm	63
5.5.4	In-kernel Re-route Service	64
5.5.5	Signaling among Different Networks	64
5.5.6	Out-of-Sequence or Missing Packets	65
5.6	Conclusion	66
Chapter 6 Multihoming in Delay Tolerant Network		67
6.1	Introduction	67

6.2	Modification of ONE Simulator	68
6.2.1	Overview	68
6.2.2	Modifications of ONE Simulator	69
6.3	Modification of DTN Routing Protocol - MaxProp	71
6.3.1	Utilization of Interfaces	72
6.3.2	Buffer Management Policy	72
6.4	Estimation of Delivery Likelihood	72
6.5	Experimental results	73
6.5.1	Multiple Interfaces with Same Type	73
6.5.2	Multiple Interfaces of Different Types	74
6.6	Discussion and Conclusion	77
Chapter 7 Conclusion and Future Works		78
7.1	Conclusion	78
7.2	Future works	79
List of Publications		81
References		82

List of Tables

4.1	Feature comparison among different products	46
4.2	Mean opinion score (MOS)	46

List of Figures

2.1	Illustration of a mobility scenario for multihomed mobile node communicating with CN via multiple heterogeneous links	7
2.2	Heterogeneous networks	9
2.3	DTN operating network	14
2.4	DTN architecture	16
3.1	Illustration of a mobility scenario for multihomed mobile node communicating with CN via multiple heterogeneous links	19
3.2	Illustration of SIP message flow for session setup and update	20
3.3	Illustration of system architecture	22
3.4	Illustration of network address translation process	23
3.5	Throughput of single UDP flow from CN to MMN via WiMAX link	26
3.6	Throughput of single UDP flow from CN to MMN via WIFI link 1	27
3.7	Throughput of single UDP flow from CN to MMN via WIFI link 2	27
3.8	Aggregated throughput of single UDP flow from CN to MMN via all 3 links	28
3.9	Delay improvement when switching from single WiMAX link to two WIFI links	28
3.10	Improvement on the number of out-of-sequence packets by using IKPR. MaxD is the maximum displacement in sequence.	29

4.1 3G-WLAN network architecture 37

4.2 System Architecture 39

4.3 System protocol 42

4.4 Delay during handoff 44

5.1 Network architecture 50

5.2 System architecture 51

5.3 Network architecture 56

5.4 End-to-End delay 59

5.5 Packet delivery 61

5.6 Packet delivery from $t = 10$ s to $t = 20$ s 62

6.1 ONE simulator screen shots 70

6.2 ONE simulator architecture 71

6.3 Delivery rate (two interfaces of the same type) 74

6.4 Mean latency (two interfaces of the same type) 75

6.5 Delivery rate (two interfaces of different types) 76

6.6 Mean latency (two interfaces of different types) 76

List of Acronyms

3G	Third Generation
3GPP	3rd Generation Partnership Project
AP	Access Point
BU	Binding Update
BS	Base Station
CDMA	Code Division Multiple Access
CN	Correspondence Node
CoA	Care of Address
DCCP	Datagram Congestion Control Protocol
DTN	Delay Tolerant Network
FA	Foreign Agent
GUI	Graphical User Interface
GPRS	General Packet Radio Service
HA	Home Agent

HO	Handoff
HoA	Home Address
IKPR	In-Kernel Packet Re-ordering
IP	Internet Protocol
ISP	Internet Service Provider
MAC	Media Access Control
MIP	Mobile IP
MN	Mobile Node
MMN	Multihomed Mobile Node
MOS	Mean Oppinion Score
NAT	Network Address Translation
ONE	The Opportunistic Network Environment
PAN	Bluetooth Personal Area Network
QoS	Quality of Service
RO	Routing Optimization
RSS	Relative Signal Strength
SCTP	Stream Control Transmission Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol

TDM	Traffic Distribution Module
UDP	User Datagram Protocol
UMS	Unified Mobility Scheme
UMTS	Universal Mobile Telecommunications System
VoIP	Voice over IP
WiMax	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network

Chapter 1

Introduction

During the last decade, technology has evolved so rapidly that one can never imagine what it was like back in the old days. More and more people have become active users of wireless networks and these networks are changing the way people live and get connected to the world. Now, we are talking about the mobile life where consumers can communicate, entertain themselves, study, etc. with just a small device. A variety of wireless access technologies are available nowadays. The infrastructure and the convenience of mobile devices such as PDAs, netbooks and smart phones with multiple access technologies spawn a new trend for consumers to gain access to the world. These include Wireless Local Area Network (WLAN), Worldwide Interoperability for Microwave Access (WiMAX), General Packet Radio Service (GPRS), Bluetooth Personal Area Network (PAN), etc.

1.1 Motivation and objectives

Heterogeneous network enables devices to connect to different networks at once. In fact, in a common scenario, where different networks overlap such as 3G/UMTS network, home or public WLAN, an Ethernet backbone network, the roaming devices can easily

1.1. Motivation and objectives

gain access to all of the available networks. However, those technologies do not share the same standard protocols. Hence there is a need to establish a common layer to adapt the different technologies into one. The current issues are how to seamlessly handover in heterogeneous networks and how to provide efficient services in terms of Quality of Service (QoS) especially for real-time applications.

Therefore, mobility in heterogeneous networks for multihomed devices is a emerging topic where each interface uses a different protocol to gain access to the network. Such vertical handover is much more complicated and difficult than horizontal handover where the mobile node moves between same type of access networks.

For multihomed devices, additional network resources can be used to improve the performance and total throughput over all the links. One of the many issues to address will be whether flow or packet distribution should be adopted for the multiple links. With each flow being redirected to multiple links at any one time, a policy scheme is needed to control the flow for reliable and connection-oriented transmission (i.e. TCP) where out-of-order or missing packet problems can degrade the performance significantly. This is especially important to latency sensitive, interactive, real-time applications such as video streaming for conferences, video phone, mobile TV or VoIP. Mobility for multihomed devices also raises issues on QoS control when the new network provides a different bandwidth link from the previous network.

There are various research studies in this field and in this dissertation, the author only focuses on terminal based mobility framework to provide a novel technique to support horizontal handover and link aggregation for multihomed devices in heterogeneous network. End-to-end QoS provisioning is supported at the network layer to accommodate seamless and smooth handover across the heterogeneous networks.

1.2 Major Contributions of the Dissertation

The main contributions of the dissertation are as follows:

1.2.1 Multihomed Mobility Framework

A novel framework to support mobility in heterogeneous network for multihomed devices with QoS support is proposed as an alternative to Mobile IP (MIP). Using a cross-layer handoff approach, we make use of SIP signaling SIP protocol to ensure seamless connection during handover across networks. The framework is based on terminal mobility where each node handles its connection and seamless connection is maintained in the network layer. The application is transparent to the node movement and the location addresses. This new framework is different from tunnel-based methods such as MIP as no tunnel is required. It exploits the multi-homing capability of the device thus enhancing QoS support for data transfer, especially in multimedia streaming. Unlike many current proposals, the framework can support both UDP and TCP transmissions. It has been implemented in a testbed. Experimental results collected from real-time streaming show that the framework can provide seamless mobility support and optimized flow distribution for heterogeneous multihomed mobile nodes.

1.2.2 Mobility for Mobile Devices

The seamless mobility idea is extended to mobile devices running VoIP services on a Symbian platform [1]. An application framework is developed to empower a mobile phone to provide seamless handoff between 3G/UMTS and WLAN for VoIP services. With more stringent requirements on quality and low latency, VoIP services need a sensitive link and the trade-off between cost and quality is always an issue. The framework exploits both WLAN and 3G/UMTS, to provide cheap, high quality but short-range channel whenever WLAN is available to complement the more costly, low bandwidth but long-range

3G/UMTS. It has been shown to be feasible and efficient for the mobile industry and has been packaged for any Symbian OS based phones which command about 50% of existing mobile handset market [2]. Unlike other approaches, the proposed framework lies in the application layer of the mobile devices and hence does not require a change in network infrastructure. Neither is modification required in the kernel layer. Hence it is not only portable among mobile devices and is also easily customized for different VoIP systems.

1.2.3 Multihomed Node for Delay Tolerance Networks

Delay tolerant network (DTN) [3] is designed to support long-delay and intermittent transmissions in challenging scenarios. The purpose of multihomed devices are to provide extra links to give a better performance in terms of packet loss and reliability. Data exchange protocols such as Max-Prop has been modified to exploit the multihoming capability. Simulation results show that the multihomed Max-Prop improves the probability of packet delivery and reduces message latency. The improvement in performance is even more prominent when buffer space is limited in the DTN node.

The Opportunistic Network Environment (ONE) simulator [4] which supports simulations with different routing protocols and measurements under different scenarios with customizable graphical user interface (GUI) maps, has been modified to support simulation of multihoming routing protocols.

1.3 Organization of Dissertation

This dissertation is organized as follows: Chapter 2 briefly describes some of the related work in mobility management as well as multi-interface management in heterogeneous networks.

Chapter 3 extends the work on terminal mobility support which proposes a cross-layer approach to improve the performance of horizontal handoffs for heterogeneous network.

Chapter 1. Introduction

[5, 6]. We exploit the multihoming capability of mobile nodes to achieve high bandwidth aggregation and QoS requirement for interactive multimedia applications. The proposed scheme uses SIP-based host mobility support to provide seamless mobility for multihomed mobile nodes with QoS awareness for data distribution. It supports both UDP and TCP transmissions. Moreover, the scheme utilizes all the current available interfaces simultaneously.

Seamless mobility support for TCP protocol is detailed in chapter 5. TCP's complexity poses more challenging and interesting problems than those of UDP connection. For TCP connections, overhead and "slow start" problems are trade-offs to reliability, which leads to degradation of quality and complexity in the routing packets in multiple links. Some proposed approaches to overcome the issues are described in the chapter.

Chapter 4 is an extension of the terminal mobility solution for Symbian mobile devices to support VoIP applications where delay constraint is important. The proposed application framework focuses on utilizing both WLAN and 3G/UMTS interfaces to enable seamless mobility for VoIP applications while still maintaining high QoS requirements such as delay sensitivity and play-out buffering.

Chapter 6 describes the development of a simulation environment for delay tolerant networks (DTN). The new simulation platform is based on ONE simulator. The popular Max-Prop routing algorithm has also been adapted to support multihomed DTN nodes.

Finally, chapter 7 concludes the dissertation with a discussion on the potential future works.

Chapter 2

Literature Review

2.1 Current Trends in Mobility Management

Mobile IP(MIP) [7] is one of the most popular mobility schemes proposed. In MIP scheme, each Mobile Node (MN) is located by a Care-of Address (CoA). A Home Agent (HA) is used in MN's home network for the packet redirection. When a Mobile Node (MN) moves away from its home network to a foreign network, it obtains a new CoA from the foreign router and sends a binding update (BU) to its HA in its home network. Upon receiving the BU, the HA will help to tunnel all the traffic to MN's current CoA. This is commonly known as triangular routing. Mobile IP schemes have been proposed for both IPv6 and IPv4, namely Mobile IPv6(MIPv6) [8] and Mobile IPv4(MIPv4) [9]. Mobile IPv4 suffers from a lack of security constructs for authorization, authentication and accounting, as well as for source routing. Routing Optimization (RO) has been proposed for MIPv6 to remove the triangular routing. When RO is applied, if MN moves from one network to another network, MN will send BU to both its HA and any communicating correspondent nodes (CNs). Then the packets from CN will be sent directly to the current CoA of MN instead of being routed to MN's HA. This effectively reduces the network overhead and

Chapter 2. Literature Review

Figure 2.1: Illustration of a mobility scenario for multihomed mobile node communicating with CN via multiple heterogeneous links

delay. Mobile IPv6 includes embedded information to help to process the MN's address changes. The newer version of MobileIPv6 even includes authentication header processing to provide validation of mobile nodes. An improvement of MobileIPv6 is Hierarchical Mobile IPv6 [10] which minimizes the amount of signaling between the CN and the HA.

In [11], MIPv6 is extended to provide multihoming support and flow distribution over heterogeneous links. In this scheme, to enable multihoming support, the BU message is extended to carry some new flags, namely M flag and S flag. If a BU message with M flag set reaches the HA, HA will know that the CoA contained in this BU message should be appended to the current CoA list without removing the previous CoAs. The S flag is to indicate the default CoA for the MN. In this way, the modified BU message allows the HA to keep track of the multiple CoAs and the respective tunnel flows to the different CoAs.

Perhaps SHIM6[12] is a more well known proposal to address the multihoming issue for IPv6 network. A number of mobility proposals for multihomed nodes, [13], [14] and [15], have employed it to address the multihoming issue. SHIM6 attempts to decouple

the node identifier from the node locator. It uses a constant IP address as an identifier for connections of the upper layers. Multiple IP addresses are used as the node locator for routing packets. By separating the node identifier and node locator, the change in IP address is made transparent to upper layers. [16] also attempts to separate the node identifier and locator, but via a different approach. It redefines the IPv6 header and instead of using the 128-bit source and destination IPv6 address fields, source and destination identifier and locator are used in the header. Although this scheme helps to address the multihoming issues, it requires modification to existing IPv6 protocol architecture, which could be difficult to deploy.

Many schemes and algorithms have been proposed for load balancing and flow distribution. Schemes like [17] and [18] perform flow distribution at the transport layer. Other schemes, as in [19], work at the network layer. In [19], a Weighted-Round-Robin method is proposed for link selection. It utilizes the information on the bandwidths of each link to compute the weights for flow distribution. In this manner, more packets are transmitted over faster links and congestion is avoided. It also proposes a “Jump-Ahead” packet scheduling method to reduce the number of out-of-sequence packets. In “Jump-Ahead” packet scheduling, packets are not transmitted in sequence. Instead, a trunk of packets are reserved for each link. The size of each trunk is computed based on the weight of each link. Packets in each trunk are transmitted by a single link in sequence. In this manner, the packets are more likely to arrive in sequence at the receiver and the number of out-of-sequence packets is reduced.

2.2 QoS in Heterogeneous Networks

2.2.1 Definition of Heterogeneous Networks

Heterogeneous network is a network that comprises devices (PC, servers, gateways) that use a variety of different data-link protocols to connect to the networks. These proto-

Chapter 2. Literature Review

Figure 2.2: Heterogeneous networks

cols could include a WLAN/ UMTS/UHF/WiMAX/LAN-cable networks. On top of the data-link layer, each node in the network uses TCP-IP for connection with other networks. Different approaches to integrate these technologies are needed at the node level or the network level.

As wireless networks do not commonly overlap with each other, they are inter-connected by gateways and intermediate routers. Packets are routed through this gateway and networks to reach the final destination. Different technologies could complement each other in terms of frequency, range, bandwidth, cost and power rate. As different technologies are developed in different ways, research has been ongoing to seek some standard integration solution to this situation.

2.2.2 Integration of Access Networks for Multiple-Interface Devices

M. Buddihikot et. al. [20] discuss the integration between WLAN-3G and propose an approach that consists of interworked gateway in a loosely coupled network. The gateway helps mobile host to seamlessly handover between 3G/WLAN. However, this approach needs the reconstruction of the network architecture and cooperation from telecommunication companies.

Mobile IP (MIP) [7] is one of the most popular mobility schemes proposed. Mobile IP schemes have been proposed for both IPv6 and IPv4, namely Mobile IPv6 (MIPv6) [8] and Mobile IPv4 (MIPv4) [9]. In [11], MIPv6 is extended to provide multihoming support and flow distribution over heterogeneous links. However, for cellular network, it entails significant infrastructure modification to support both Home Agent and Foreign Agent in the current existing network.

Session Initiation Protocol (SIP) [21] has been used to support application layer multimedia signaling protocol. SIP allows users to be located at different terminals and be referenced by the same logical SIP address. A SIP-based VoIP request scheme [22] is used by mobile host to exchange data and update addresses between WLAN and 3G networks.

Andrés Navarro Cadavid et. al. [23] propose a network re-selection system at mobile host level, which allows users to auto-select and switch between GSM/GPRS and WiFi. However, the application does not guarantee seamless connectivity and Quality-of-Service for real-time applications.

In this dissertation, the author employs a cross-layer solution by using layer 2 information (link layer info) to help to perform handoff in layer 3 (network layer). By using kernel core function to pass signal strength reports and movement detection, the system can perform a “make-before-break” handoff with multiple interfaces and provide seamless connectivity while supporting QoS for different access networks.

Chapter 2. Literature Review

2.2.3 Mobility in Heterogeneous Networks

Mobility support in heterogeneous network is performed at different layers. There are recent research studies in the topic relating to the different approaches. These approaches could be listed according to the respective layers of operation:

A. Network Layer Approaches

The gateway or intermediate routers or agents are responsible for routing data to different access networks including WiMax/UMTS or WLAN. Mobile IP is an example that is based on this approach. Here, HA and FA are used to forward packets to the MN when it moves to a new network. HA and FA could be routers and gateways of each network and they signal to each other to inform about the MN's location and IP changes. Information of current CoAs is stored at the base station of the current access networks. When a new node joins the network, its information is added into the database and when it leaves for a new network, the node is removed and the database updated to reflect the other base station. Therefore, the status of MN is maintained during the handover. Tunneling technique is used to forward data from HA to FA so that MN does not need to store all the routing information and data transfer is uninterrupted during handover.

These approaches either require the MN to register with a server or the presence of a tunnel from HA to FA to redirect traffic when MN moves to a new network. Extra delay is incurred in the end-to-end delay as well as signaling overhead when data is transferred through the tunnel.

B. Transport Layer Approaches

The MN need to detect and configure itself at the transport layer to maintain a seamless connection during handover. Recent transport-level handoff management protocols are Stream Control Transmission Protocol (SCTP) [24] and Datagram Congestion Control Protocol (DCCP) [25]. Transport layer provides services to handle congestion control, therefore, it can detect the address changes to adjust the data rate according to the new

2.2. QoS in Heterogeneous Networks

bandwidth. Furthermore, there is no further need to have intermediate active agents to maintain the state information of the mobile node as location management is handled by DNS, which means that the new IP address is known and updated to CN and subsequent data flows are routed to the new CoA. SCTP also supports multi-homing capability as it allows an association to maintain multiple IP addresses. An extension to SCTP called mSCTP (mobile SCTP) allows dynamic adding and removal of IP addresses, e.g. to provide seamless connection while moving from network to network.

However, the transport layer handoff approach depends on other layers for location management. Network detection is triggered by lower layers and each of the protocol requires security from spoofing. It means that we need to rebuild the structure of other layers and a new authentication scheme.

C. Application Handoff Approaches

The application-layer solution takes advantage of using SIP protocol for signaling. It requires the presence of a SIP server and redirect server. When MN moves to a new network it sends a re-invite SIP message to update the new address of location to CN. Once the CN gets the updated information about the MN, the data flow is redirected to a new address. The terminal mobility assures that the application is aware of the changes and has to re-configure itself to control the traffic to a new address. It means that each application running in the MN needs to be modified to support the mobility technique, which may not favor ease of deployment.

2.2.4 QoS in Heterogeneous Networks

In this dissertation, the author only focuses on end-to-end QoS over heterogeneous networks for real-time sensitive applications such as VoIP or video streaming service. To support the QoS entails the usage of network resource, bandwidth allocation and management, jitter control, and latency management, data rate as well as reliability commitment.

Chapter 2. Literature Review

The challenges in heterogeneous network is the difference in QoS support in each network. End-to-end QoS could be supported in access network, backbone network or at a centralized QoS management unit (clearinghouse) [26]. However, the difference in QoS across domains would prevent service providers from exercising policy control and accounting which would make the services very costly. The author uses QoS support at the node level within the OSI layers in order to provide top-to-bottom and end-to-end QoS between sender and receiver.

2.3 Delay Tolerant Networks

A delay tolerant network (DTN) [27] is a network that operates over long distances such as in space communications or interplanetary scale. Long latency in data transmission, even hours or days is inevitable. It involves operation in challenging scenarios, such as in disaster or war zone, where normal operational infrastructure networks are destroyed or inhibited by extreme interference. Some of the current standards can be seen with the recent publication of two Internet RFCs (RFC 4838 [3] and RFC 5050 [3]). Several projects on DTN, both in simulation or real devices such as [28], [4] and [29], presently exist, serving different purposes for academia, government or industry.

Delay Tolerant Networks has since been generalized to refer to networks without continuous connectivity. A cause of the intermittent connectivity is a disruption which may be caused by the limits of wireless radio range, sparseness of mobile nodes, energy resource constraints, attacks or noises. Networks which can withstand disruption is also called Disruption Tolerant Networks, which is thus a subset of Delay Tolerant Network. A delay tolerant network will require hardware that can store large amounts of data. Such media must be able to survive extended power loss and system restarts. It must also be immediately accessible at any time. High-volume hard drives or flash memory storage devices are preferable in this case. The data stored on these media must be organized and prioritized

2.3. Delay Tolerant Networks

Figure 2.3: DTN operating network

by software that ensures accurate and reliable store-and-forward functionality.

In a delay tolerant network, traffic can be classified in three ways, called expedited, normal and bulk in order of decreasing priority [3]. Expedited packets are always transmitted, reassembled and verified before data from any other classes are serviced. Normal traffic is sent after all expedited packets have been successfully assembled at their intended destination. Bulk traffic is not dealt with until all packets of other classes from the same source and bound for the same destination have been successfully transmitted and reassembled.

Fig. 2.3 shows a scenario where DTN is used to pass a message from remote villages to each other or to the centralized server [30]. Each remote village has a short range wireless network and could not reach each other. A reliable end-to-end path could not be established from one node to the centralized server or to other peers. The mobile router allocated in a bus travels to different village and has a relatively short contact time with each of the kiosk's computer. Messages are stored in the bus until it reaches the centralized server in the downtown area. After that, data is uploaded and forwarded to the destination

Chapter 2. Literature Review

addresses.

Delay tolerant network or disruption tolerant networking is an approach to computer network architecture that seeks to address the technical issues in heterogeneous networks that may lack continuous network connectivity. DTN architecture provides interoperable communications between and among challenged networks. Delay tolerant Networking has been applied to various challenging networking environments, including space communications, sparse sensor network and opportunistic mobile ad-hoc networks. The DTN architecture is shown in Fig. 2.4. It proposes a bundle layer to span different (inter)networks. Messages, stored in bundles, are forwarded hop-by-hop between DTN nodes (bundle routers) as best-effort traffic. Custody transfer provides reliable transmission through persistent hop-by-hop forwarding. Control messages are used to inform a sender or custody router about message forwarding progress, end-to-end delivery, or failures. Various research topics focus on the feasibility of DTN in supporting communication in challenging environments where there are intermittent connectivity, long or variable delay with asymmetric data rates and high error rates. The performance of multihop wireless communication depends on the density of the node distribution, the routing and forwarding algorithms in use. There are various study on the improvement of performance of routing protocols used in delay tolerant networks, including Prophet[31], MaxProp [32] or Spray and Wait [33]. However, they do not concern the mobile node with multi-homed capability in ad-hoc heterogeneous networks. With multiple interfaces, the probability of successful delivery is improved and that affect mechanism and performance of all the current existing routing protocols.

Figure 2.4: DTN architecture

Chapter 3

Multihoming in UDP

In this chapter, the author describes the work that he has done in developing a framework that supports multihoming for UDP transmission for multiple-interface devices.

3.1 Introduction

In this chapter, the author describes the Unified Mobility Scheme (UMS) for multihomed mobile node (MMN) [34] ¹. UMS addresses the multihoming issues by using SIP. In UMS, each MMN is identified by a unique SIP address. Each SIP address is associated with multiple Care-of Addresses (CoA) of MMN. The association is managed by a SIP server. Thus the resolution of a SIP address can be done by querying the SIP server. UMS enables terminal mobility for MMN and provides support for both horizontal handover and vertical handover. When a Correspondent Node (CN) wants to communicate with MMN, it can query the SIP server using MMN's SIP address. The SIP server will reply with a list of CoAs of MMN. Then CN will choose one or more addresses from the list for the distribution of the packets. When MMN detects that one of its interface is about to move

¹This work is supported by the Agency for Science, Technology and Research(A*STAR) of Singapore. (Grant number: 0721010028 - M47020034)

3.2. System Architecture

to a different network, it will send a Binding Update (BU) to CN, informing it about the change in the list of CoAs. CN will then adjust the distribution policy and redirect the packets to other available addresses.

UMS framework supports QoS management and adjusts QoS policy according to the QoS requirements specified by the applications. UMS is able to perform optimal path selection satisfying the QoS requirements. Depending on the QoS policy, the network traffic can be distributed across multiple links at flow level or packet level. Being able to perform flow distribution at packet level is very important, especially when the bandwidth requirement is too high to be fulfilled by a single link. By aggregating multiple links, UMS enables bandwidth demanding applications to survive under low speed network environment. The out-of-sequence problem surfaces when packets from one single flow are routed to the destination via different paths. This problem is usually caused by the different end-to-end delays of the different paths used. Some applications such as video conferencing are very sensitive to the out-of-sequence problem.

To minimize the negative effect of the out-of-sequence problem, UMS includes an In-Kernel Packet Re-ordering (IKPR) service to reduce the number of out-of-sequence packets when a flow spans across multiple links. IKPR makes use of a small buffer in the kernel space to re-order the incoming packets without violating the delay constraint. This is especially useful for UDP applications that are sensitive to out-of-sequence problem.

UMS has been implemented on a Linux testbed. Experimental results show that this scheme can provide seamless mobility to multihomed mobile nodes without triangular routing and the network traffic overhead is kept small. The results also show that the UMS can effectively aggregate multiple links and increase the maximum throughput.

Chapter 3. Multihoming in UDP

Figure 3.1: Illustration of a mobility scenario for multihomed mobile node communicating with CN via multiple heterogeneous links

3.2 System Architecture

With reference to the mobility scenario shown in Fig. 3.1, MMN is multihomed in three heterogeneous wireless networks including one WiMAX wireless network and two 802.11b wireless networks. With UMS, MMN is able to communicate with CN and enjoy seamless mobility, QoS-aware path selection and flow distribution across multiple heterogeneous links.

3.2.1 Multihoming and Mobility

In UMS, SIP protocol is employed to address the multihoming issue. Each multihomed mobile node is identified by a unique SIP address. A SIP server is used to manage the association between mobile node's SIP address and its Care-of Addresses. The signaling procedure is shown in Fig. 3.2. When a MMN joins the network, it will publish its SIP address and the list of CoAs using REGISTER message with the SIP server. CN can then

3.2. System Architecture

Figure 3.2: Illustration of SIP message flow for session setup and update

Chapter 3. Multihoming in UDP

obtain the list of CoAs of MMN from the SIP server by using QUERY message. When CN wants to transfer data to MMN, it will first attempt to set up a session with MMN using INVITE message. When MMN receives the INVITE message, it will accept and reply with 200 OK message with a list of preferred CoAs. This list of CoAs is a subset of MMN's interface addresses which have better signal strength or lower traffic. This implies that the list of CoAs used in different sessions may be different and this makes it possible to balance the traffic load among multiple links. The session is established after CN replies with an acknowledgment.

When the session is established, both MMN and CN will record each other's address as reference peer address of this session and the network port used for data transfer. This peer IP address is the address used by the transport layer connection and the network address translation process. The network ports are used to identify data flows. Then the data transmission will start and CN can distribute the traffic among the list of CoAs of MMN based on QoS requirements.

UMS supports horizontal and vertical handover both proactively and reactively. When MMN senses that one of its wireless interfaces is seeing lower signal strength, and it is picking up stronger signals from a new network, or it suddenly drops out of the network, it will send a Binding Update(BU) to SIP server and any communicating CNs to update its CoA list. The BU message will also include information on the traffic condition for each CoA. CN can then re-distribute the flow based on the new CoA list.

3.2.2 Address Translation

The system architecture is shown in Fig. 3.3. A virtual interface is created in the system to hide the heterogeneity of the physical interfaces. The virtual interface essentially decouples the node identifier and node locators. A constant IP address from unused private prefix is arbitrarily assigned to the virtual interface as the permanent address. This

3.2. System Architecture

Figure 3.3: Illustration of system architecture

permanent address is used by upper layers and it can be seen as the node identifier which is only valid within the node itself. The CoAs of the interfaces are the node locators for routing packets. The virtual interface makes the changes of CoAs transparent to upper layers and enables ongoing sessions to survive when physical interface in use are brought down.

The network address translation process is illustrated in Fig.3.4. When outgoing packets enter the virtual interface, they are directed to the Traffic Distribution Module (TDM). TDM manages the flow distribution of outgoing traffic based on the QoS policy. By rewriting the source addresses in the IP header of the packets with the CoA of an interface, packets are marked to be transmitted via corresponding physical interface. As shown in Fig.3.4, the source address of packets are modified by the TDM from "dummy_src" to "src_CoA_x" and "src_CoA_y", where "dummy_src" is the permanent IP address of the virtual interface and "src_CoA_x" and "src_CoA_y" are the CoAs of physical interfaces. The Network Address Translation(NAT) module will check the destination address of the IP header. If it matches any of the reference peer address of the ongoing session, it will be

Figure 3.4: Illustration of network address translation process

3.2. System Architecture

replaced by the corresponding CoA. In Fig. 3.4, the destination address is modified from "ref_dst" to "dst_CoA_m" and "dst_CoA_n" which are the current CoAs of destination node. When packets arrive at the destination node, the NAT module will perform the corresponding address translation for matched packets. For each matched packet, its source address is replaced by the reference peer address and the destination address is replaced by the permanent address of the virtual interface, which is "dummy_dst" in Fig. 3.4. In this way, seamless mobility support is provided and no triangular routing is required.

3.2.3 QoS Support and IKPR Service

UMS allows the application to specify the QoS requirements. The policy for QoS support and flow distribution can be adjusted by application through the QoS support module in user space. Given the delay requirement, UMS can detect and select a path with satisfactory delay. UMS allows different algorithms to be used for path delay evaluation. One simple method is to use a periodic probe packet to sense the end-to-end delay for each path and then choose a suitable one from all the paths. If the application requires very high throughput and is immune to out-of-sequence problem, such as file transfer application, UMS can provide link aggregation to increase the total throughput. On the other hand, if the application requires high throughput but is also sensitive to out-of-sequence problem, UMS provides In-Kernel Packet Re-ordering service (IKPR) to reduce the number of out-of-sequence packets. If the application activates the IKPR service, outgoing packets from the flows of this application will be tagged. The tagging is done by inserting a one-byte tag into the beginning of the payload. This tag contains the sequence in which packets are transmitted. At the receiver end, IKPR will extract and remove the tag from the packets. The tag enables the IKPR service to know the actual sequence in which packets are transmitted. Out-of-sequence packets are stored temporarily in a buffer in the kernel. They are released and passed to the higher layers when the next expected packet arrives

Chapter 3. Multihoming in UDP

or the delay constraint of the stored packets is about to be violated. IKPR is different from application layer buffering. It does not aim at re-ordering all out-of-sequence packets which may require a large buffer and the delay constraint may be violated. Instead, it attempts to re-order packets with small displacement in sequence while not violating the delay constraint. While algorithms like Weighted Round-Robin link selection and Jump-Ahead scheduling [19] are to minimize the out-of-sequence problem at the transmitter end, the packets may still arrive out of sequence. In this case, most of the out-of-sequence packets will probably have smaller displacement in arrival sequence. With IKPR service at the receiver end, it helps to iron out the majority of out-of-sequence packets which have small displacements.

3.3 Performance Evaluation

3.3.1 Experimental Results

UMS is implemented for Linux kernel 2.6.17 on Ubuntu Linux platform. A testbed similar to Fig. 3.1 is set up for data collection. A MMN is connected to CN via three heterogeneous wireless links including one WiMAX and two WIFI links. A single UDP flow is sent from CN to MMN at the rate of 948 kbps. The changes in packet distribution are shown in Fig. 3.5-Fig. 3.8 to demonstrate mobility support and flow distribution capability of UMS. Fig. 3.5, Fig. 3.6 and Fig. 3.7 show the change in throughput measured on the WiMAX link and WIFI links 1 and 2 respectively.

At $t = 0$ s, all packets are directed to the WiMAX link. At $t = 135$ s, WIFI interface 1 is up. Part of the traffic goes to WIFI link 1 and the throughput measured at the WiMAX link drops. At $t = 234$ s, WIFI 2 interface is brought up. The traffic is further distributed among all three links. When WIFI 2 is brought down at $t = 334$ s, the traffic is redistributed to the two active links. The average measured aggregated throughput from all links is 945.8 kbps. As shown in Fig. 3.8, at any point in time, the aggregated throughput

3.3. Performance Evaluation

Figure 3.5: Throughput of single UDP flow from CN to MMN via WiMAX link

approximately equals to the data rate of the flow. This suggests that the packet loss during each handover is very small. This proves that UMS can provide seamless mobility and flow distribution for multiple heterogeneous wireless links.

In Fig. 3.9, the traffic is initially directed to single WiMAX interface. The average end-to-end delay is 270.8 ms. At $t = 49.27$ s, the two WIFI links are up and UMS senses that the WIFI links have lower end-to-end delay of 74.3 ms. Hence, UMS redistributes the packets from the WiMAX link to two WIFI links, resulting in much lower delay.

The improvement on out-of-sequence packets by IKPR service is shown in Fig. 3.10. Assume the displacement in the packet arrival sequence is uniformly distributed from 0 to MaxD and the delay constraint of packets is always met. The improvement is almost 100% when the displacement in the arrival sequence is smaller than the buffer size. However, it performs poorly when the displacement is greater than the buffer size. Hence the larger the buffer size, the more out-of-sequence packets can be recovered. If the scheduling algorithm is good enough to make the sequence displacement small, then a small buffer is sufficient.

Chapter 3. Multihoming in UDP

Figure 3.6: Throughput of single UDP flow from CN to MMN via WIFI link 1

Figure 3.7: Throughput of single UDP flow from CN to MMN via WIFI link 2

3.3. Performance Evaluation

Figure 3.8: Aggregated throughput of single UDP flow from CN to MMN via all 3 links

Figure 3.9: Delay improvement when switching from single WiMAX link to two WIFI links

Chapter 3. Multihoming in UDP

Figure 3.10: Improvement on the number of out-of-sequence packets by using IKPR. MaxD is the maximum displacement in sequence.

3.3.2 Discussions

The proposed UMS framework utilizes SIP protocol for session management and the localized network address translation approach helps to minimize the network traffic overhead. Compared to MIPv6 with route optimization, this approach has much lower overhead. The details of the overhead comparison has been discussed in [5]. Thus, UMS also has less traffic overhead than proposals such as [14] which is based on mobile IP. Since triangular routing is not used by UMS, the traffic overhead generated by UMS is thus also less compared to proposals such as [13] and [15], which need triangular routing for packet redirection. By using the virtual interface and the permanent address, UMS completely hides the heterogeneity of the physical interfaces and decouples the node identifier and node locator. This not only enables transparent change of interface addresses but also allows the upper-layer connections to survive the breaking down of the physical interface in use. The approach taken by UMS does not require heavy modifications to existing protocol architecture which is very different from other multihoming proposals like [16].

3.4. Summary

The ability to distribute data flows freely at either flow level or packet level is another distinguishing feature of UMS. When a bandwidth demanding application is requesting for large bandwidth which cannot be fulfilled by any single link, conventional flow based distribution algorithms are not able to satisfy the requirement. However, with UMS scheme, multiple low speed links can be aggregated together to provide higher throughput and it is able to serve applications with much higher bandwidth requirements. Distributing data flows at packet level across multiple links usually causes out-of-sequence problem. While a lot of methods have been proposed for the transmitting node to solve the out-of-sequence problem, the proposed IKPR service takes a different approach. IKPR works at the receiver end and it can provide complementary enhancement to transmitter-end methods. The result has shown that it performs very well when the buffer size is larger than the sequence displacement. When IKPR is used together with transmitter end algorithms, if the transmitter-end algorithms can reduce the sequence displacement, IKPR will be able to iron out the out-of-sequence packets on the fly with very small buffer size. The smaller the sequence displacement is, the smaller the buffer is required. Thus, UMS is a viable framework to provide host mobility and QoS aware flow distribution for next generation wireless network.

3.4 Summary

In this chapter, we have proposed UMS, a mobility management scheme for multi-homed mobile node with QoS support. The three distinguishing features of UMS are the SIP based host mobility support, virtual interface enabled multihoming and QoS-aware flow distribution. UMS allows both vertical handover and horizontal handover. Compared to MIP, UMS does not require triangular routing and hence has very small overhead. By using the virtual interface, the node identifier and the locators are decoupled and the change of locator is transparent to upper layers. This transparency does not require modifications

Chapter 3. Multihoming in UDP

to existing network layer design and therefore is easier to deploy. The flow distribution capability helps to increase the observable bandwidth and the IKPR service helps to reduce the number of out-of-sequence packets. Experimental results have shown that UMS can provide seamless handover and QoS-aware flow distribution for WiMAX and WIFI links and the traffic overhead is low. The results also show that UMS can perform path selection intelligently and IKPR service can help to iron out the out-of-sequence packets. Therefore, UMS is a feasible framework to provide seamless mobility support and optimized flow distribution for heterogeneous multihomed mobile nodes

Chapter 4

Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

4.1 Introduction

It has been several years since the day that mobile devices with WLAN capabilities were launched to curious and anxious customers. Nowadays, WLAN has become a necessary basic feature of a mobile phone. In addition, the data rate has improved due to the advances in telecommunication infrastructure and technology, allowing users to browse the Internet more conveniently than before. However, the charges are still costly for interactive or streaming applications. Along with the prevalence of many available and open wireless access points in all big cities in the world, mobile users have ample opportunities to gain access to the Internet anytime, anywhere.

Voice over IP (VoIP) is the transmission technologies of voice communication over IP networks. It employs session control protocols to handle session establishment and

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

termination as well as audio codecs, thus allowing voice to be transmitted over IP networks as digital data. VoIP service is gaining huge popularity with the prevalence of broadband Internet access services. One of the challenging issues to address in VoIP is seamless handoff over heterogeneous networks, namely WLAN and 3G. There is a lack of common standard in handset hardware and software which renders seamless handoff even more challenging.¹

In this chapter, we introduce an application framework that supports seamless connectivity for VoIP application in Symbian phone over 3G and WLAN networks and which satisfies mobility and QoS issues in heterogeneous networks. The application performance is also analyzed and compared with other approaches.

We focus on Symbian-based handset devices and provide seamless connectivity for VoIP services. Comparing with other solutions and given the network complexity, the workload is moved from the interwork gateway to the mobile host. The system framework is deployed at the application level of the mobile host and it can adapt to the existing network and requires less overhead when deploying on the current generation of mobile devices. Note that authentication, authorization and accounting are not within the scope of this dissertation.

4.2 Design Considerations and Challenges

The application framework is to be developed in Symbian OS which is a proprietary operating system for mobile devices, running on almost 50% of the current handsets including Nokia, Samsung, Ericsson. The framework is designed to satisfy the requirements of VoIP services. The following section provides a brief summary of VoIP service requirements.

¹The project was sponsored by StarHub Limited, Singapore, in collaboration with Centre of Multimedia and Networking, School of Computer Engineering, Nanyang Technological University.

4.2. Design Considerations and Challenges

4.2.1 Requirements for VoIP service

According to [35], VoIP applications vary from system to system but they do share some common points:

- Delay sensitivity: Since it is a real-time application, delay affects the quality of service and user satisfaction. With the different types of networks available, delay accumulates and becomes substantial when users move from one network to another.
- Play-out buffering: When the packet loss is high due to network problem, or buffer overflows at the receiving end due to insufficient buffer size needed by the different codecs at the sending end, the quality of the voice call is severely affected. However, for real-time applications, due to limited power and resources in the device, a proper buffer management mechanism is needed.
- Ping-pong effect due to frequent handover.
- Congestion: When user moves to a network with a lower bandwidth, it results in packet loss and service degradation.
- Server disruption due to long handover time.

4.2.2 Design Considerations

The proposed framework takes the following into account in its design:

- Commercial VoIP products vary from IP phones to software in PC, PDA or mobile phones. The proposed framework must be able to handle different mobile devices and environment platforms. For this dissertation, Symbian phones are used in demonstrating the feasibility and performance of the proposed framework.
- The presence of any wireless networks with open key must be detected and stored in the memory.

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

- The framework must be able to optimize and select the best available wireless network to use.
- Common solutions for WLAN/3G integration are to handle both 802.11 and 3G wireless data services at network level. However, complications arise as such integration involves 3G operators and authentication issues to access the different networks. Hence, our design shall focus on terminal mobility thus moving the workload to the mobile host. The system is built as a local proxy server to provide seamless connectivity to 802.11 and 3G networks.
- Mobility in heterogeneous network is a challenging topic due to the increasing number of different access networks and rapid development of end-user terminal devices. The framework is designed to provide seamless connectivity to 802.11 and 3G networks for Symbian mobile devices. The mobile device has access to 3G network provided by phone companies. Only when users are completely out of range of any WLAN then will the framework automatically switch back to 3G network. The reasons for adopting this approach are:
 - Mobile users could use existing WLAN networks for VoIP service. They are widely deployed in public places, supported in various current phone devices and save on cost for short range communication.
 - Non-overlapped WLAN access networks result in connection disruption which degrades user's experience in using VoIP service.
- The framework is designed to meet the delay sensitivity and buffering requirements of VoIP services which will also minimize server disruption caused by long handover. The ping pong effect due to frequent handover is alleviated by an optimized algorithm used in the selection of new access network to maintain a stable and uninterrupted connection so long as the current wireless link is still good enough. Addressing the

4.3. System architecture

loss due to congestion is beyond the scope of the proposed application framework.

The novelty of our proposed framework is its easy deployment without any modification in the network infrastructure or existing protocols. From the commercial perspective, as it is built in the application layer, it can adapt to all Symbian-based mobile devices without much, if any customization.

4.2.3 Design Challenges

Phone companies such as Nokia, Samsungs, etc provide VoIP client applications with their API libraries hidden inside the core of Symbian OS. This poses the following development constraints:

- The application needs to acquire security permission to gain access to the network control capability.
- There are no resources and development access to the kernel as Symbian OS limit access to the kernel API by non-certified third party developers. Access to stream buffers is thus blocked. A workaround is proposed to provide extra buffers at the user level to satisfy the delay sensitivity and buffering requirements for the VoIP services.

4.3 System architecture

The network architecture, upon which the proposed application framework operates, is shown in Fig. 4.1. The mobile node running the application can access different WLAN networks and establish a voice call through the ISP gateway to the subscriber server. The server handles SIP request and redirects RTP data from the mobile node (MN) to the correspondent node (CN). This centralized system allows MN and CN to be isolated from each other and solve the NAT/Firewall issues. The SIP server is equipped with a STUN

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

Figure 4.1: 3G-WLAN network architecture

server [36] to help establish connections between different access networks. MN has access to all the available WLAN hotspots.

At the initiation stage, both MN and CN are provided with a unique SIP account. Users input username and password for the SIP account when they start their application. CN registers with the SIP server and inform the server of CN's IP address. CN's CoAs are stored in the SIP server's database. MN registers with the SIP server by sending REQUEST message with its unique SIP address. The SIP server accepts the request and also stores all three CoAs in the database. At this moment, MN's id is attached to three

4.3. System architecture

CoAs. MN requests INVITE message to connect to CN. The SIP server accepts and sends back CN's CoAs to MN. The connection between CN and MN is established. The voice stream is set up, from the MN to CN. When MN moves, it switches from one WLAN to another WLAN. Every time the IP changes, a binding update message is sent to notify SIP server. When MN detects changes in the current WLAN network, it switches to 3G/UMTS network, and sends a binding update (BU) message to the SIP server to notify about the changes. The connection between SIP server and MN is disconnected and re-established while the communication link between CN and SIP server is maintained during the handoff.

There are similar systems [23] that can auto-select the best wireless network and provide seamless connectivity for mobile devices. We now propose a new application framework that uses simple but effective methods to solve development complications that other people have encountered while working with Symbian OS. From [5], we adopt similar concepts of replacing IP addresses with new IP addresses when the client roams and the initial IP addresses are changed. This is transparent to the VoIP client application.

We divide the system into two main modules shown in Fig. 4.2:

(i) Connectivity module, which is called Network Manager, controls all outgoing or incoming connections to and from the device.

(ii) VoIP Session module, which is responsible for maintaining application session and handling the handover of the network connection including optimizing, selection, and mapping network connection with different ports. It consists of the VoIP Session Manager and Streamer module. Each Streamer module attaches to a VoIP session and control data transfer at both sides, and they are handled by the VoIP Session Manager.

4.3.1 Network Manager

The network manager module performs the following tasks:

- Detect any new networks available and retrieve network information.

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

Figure 4.2: System Architecture

4.3. System architecture

- Optimize and select the best networks when the current network is below the threshold.
- Monitor all current connections
- Perform handoff.

As shown in Fig. 4.2, the module works directly with the kernel level functions, where the Symbian network controls manage the 3G/UMTS or WLAN interfaces. As it is accommodated at the user level, the Network Manager has to use the Symbian functions to detect or scan wireless network in the proximity. As mentioned earlier, authentication, authorization and accountability are outside the scope of this dissertation. Authorization data such as WLAN security key or account is stored in different profile from the VoIP user client profile. The profiles have to be stored in the system profile managers which is not under control of the VoIP application. The Network Manager manages all current connections while detecting the current wireless signal strength. [20] introduces an algorithm which considers the wireless signal strength, priority for each link, low threshold and high threshold. The algorithm is adapted and applied to keep the connection stable and uninterrupted if the current wireless link is still good enough. Let us denote w_i , s_i , p_i , L_i , H_i as the weight, signal strength, priority, low threshold and high threshold where $s_i, L_i, H_i \in [0, 100]$ and $p_i \in \{1, 2, 3, \dots\}$. The constant c is to maintain a balance between the priority (the known factor) and the signal strength (the unknown factor).

$$w_i = 2(s_i - L_i) + c * p_i \text{ if } i \text{ is the current link}$$

$$w_i = 2(s_i - H_i) + c * p_i \text{ if } i \text{ is not the current link}$$

The system will select the link that has a better weight than the current link to establish the new connection. The two thresholds ensure that the link could maintain a sufficient bandwidth. The priority is a factor in the algorithm as we assume that the mobile nodes would move around familiar areas where known wireless hotspots could be

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

easily found. If the current link drops slightly lower than L_i or a new access point has signal strength which is slightly better than H_i , the system will not switch to the new network. This prevents the application from switching back and forth between wireless networks.

4.3.2 VoIP Session Manager and Streamer Module

By using the VoIP session establishment through SIP protocol, the mobile node can update if there are any changes of the IP addresses. Fig. 4.3 explains the message exchange between the client application and our system.

At the initialization procedure, the client application sends request to the Session Manager to open a new connection. The final destination IP address, local port, and destination port are packaged in the message to the Session Manager to set up a new connection. An instance of Streamer is initiated and it starts listening for request. The local port is sent back in the ACK message to the requesting client application. From that point, the client application would automatically replace the final destination IP with the local IP, the destination port with the local port to the Streamer. All packets are forwarded through this proxy-like module to the media server including SIP registration as well as RTP media data.

During the handoff procedure, the Network Manager, which is operating independently from other modules, notifies the Session Manager about changes in the network. The Session Manager passes down the message to the Streamer instances (if there is more than one). Local connections between VoIP client applications and the Streamer are maintained while the outgoing connections are disconnected. After Network Manager manages to associate with a new network, outgoing connection is re-established by using the user information that the Streamer stores at the initialization stage. The handoff procedure is transparent to the VoIP client application. Packets that are sent from the VoIP client

4.3. System architecture

Figure 4.3: System protocol

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

application during the handoff procedure are stored in a buffer in Streamer. Depending on the size of the buffer and the dropping policy, packets are dropped silently or alternately to achieve a certain level of quality of service.

4.4 Performance Analysis

The experiment setup consists of Nokia mobile phone with Symbian OS version 9.1 as a mobile node (MN), a media server to relay data. A Correspondent node device (CN) in the network can be accessible from both 3G/GPRS and WLAN networks. A RTP/UDP data stream is established from the mobile node device (MN) to CN. At first, MN connects to both WLAN and 3G at the same time. Since the WLAN is available, it is used as the main communication link. When MN moves to a new network, the wireless link quality drops and the Network Manager signals to switch all current connections.

Fig. 4.4 shows the delay during handoff from WLAN to 3G and vice-versa. At the beginning, the mobile node uses WLAN and the average end-to-end delay is 172.68 ms. At $t = 1419$ s, the first handoff occurs and the connection is switched to 3G immediately while the application searches for a new wireless network; the average end-to-end delay at the 3G/UMTS network is measured at 997.5 ms. At $t = 1507$ s, after a successful search for the new network, the connection is switched to WLAN network immediately without a break. This approach reduces the time delay incurred in the network re-selection process and takes advantage of 3G as a backup link. During the handoff, there is a sequence of out-of-order packets since the CN still receives from both interfaces. This out-of-order issue is solved at the application level where the VoIP client application has its own buffer to queue the incoming packets. The “make-before-break” handoff method provides a low latency handoff, which is essential for delay-sensitive VoIP applications.

Without the buffer, the author observed that there is a 5% packet loss during the transmission, and with extra buffer; the packet loss rate is reduced to 1%. When the

4.4. Performance Analysis

Figure 4.4: Delay during handoff

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

switching occurs, the buffer will store incoming data from the local application and wait until the signal is flagged from the Network Manager to prevent out-of-order packet sequences during transmission. The buffer size varies according to the different codecs used in the application and is configurable before the application starts. The system has been proved to be able to reduce packet loss rate during handover by using the “make-before-break” handover and with the use of an extra buffer to reduce the number of out-of-order packets.

The author performs a qualitative feature comparison between the proposed application framework and existing VoIP client applications or wireless access network assisted applications. Note that it is difficult to compare with existing works which are identical to our proposed application framework due to the latter’s novelty. Moreover, as we have limited access to the commercial products, we identify a few representative approaches, namely, Devicescape [37], iCall [38] and WeFi [39] which have some similar features to our proposal for comparison. Table 4.1 summarizes the qualitative comparison results.

For commercial products, we evaluate some features amongst available existing software that support similar functions. Linux and Apple iPhone have small market share compared to Symbian. Although Symbian is used in 50% of the mobile devices in the market, Nokia (who recently bought Symbian) still imposes access restriction on third party developers to their core functions. This explains the scarcity of supporting applications in this platform.

Among the products, only iCall application on iPhone platform supports seamless connectivity. However, it only provides switching function from cell phone calls to VoIP phone calls using wireless networks. Such calls are restricted to Canada and US phone numbers only. Notably, iCall does not support WLAN-3G handoff. Hence, no quantitative comparison can be made between our proposal and the products.

A subjective listening test is conducted to measure the Mean Opinion Score (MOS) [40] of the streamed VoIP speech. MOS is expressed as a single number in the range 1

4.4. Performance Analysis

Table 4.1: Feature comparison among different products

	Featured products			
	Devicescape[37]	iCall [38]	WeFi [39]	Our proposed framework application
OS Platform	Symbian, Windows CE, Linux	Apple OS	Linux, Symbian	Symbian
Seamless connectivity	No	Yes	No	Yes
WLAN-3G Handoff	No	No	No	Yes
WLAN-WLAN Handoff	Yes	Yes	Yes	Yes
VoIP support	No	Yes	No	Yes
Overall	Poor	Good	Poor	Good

Table 4.2: Mean opinion score (MOS)

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	4 Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

to 5, where 1 is lowest perceived speech quality, and 5 is the highest perceived speech quality [41]. MOS is generated by averaging the results of the subjective listening tests of a number of listeners. The listeners rate the audio over the communications medium being tested. A listener is required to give the listened audio a rating using the rating scheme in Table 4.2.

For our listening tests, users listen to an audio stream from CN to MN and are then asked to rate the VoIP audio quality throughout the streaming session with handoff. A VoIP stream is sent from CN to MN and handoff occurs from WLAN to 3G and vice verse.

Chapter 4. Seamless VoIP Communication in Heterogeneous Networks for Symbian Devices

G.711 is used for voice encoding. The MN is a Nokia N95 phone while CN is a soft phone in the desktop PC. The average MOS computed from the MOS of 20 listeners is at 4.66, which is excellent for VoIP streaming.

4.5 Conclusion

In this chapter, the author proposes a novel application framework that supports seamless connectivity for VoIP applications in heterogeneous networks. Compared to other approaches, the proposed framework lies in the application layer of the mobile devices and therefore requires no change in the network infrastructure and no implementation in the kernel layer. It is portable on different mobile devices and it can be easily adapted to existing VoIP systems.

The architecture details are discussed in conjunction with VoIP service requirements and development constraints in Symbian OS. The application framework takes advantage of multi-homing capability of Symbian OS to ensure seamless connectivity for delay sensitive VoIP applications. Our experiments show that, with extra buffer, along with the VoIP client application's own buffer, the application can effectively reduce the amount of packet loss and out-of-sequence packets during the handover. The result proves that the proposed framework is effective to support VoIP service requirement and it is a feasible solution for the current generation of mobile devices.

Chapter 5

Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

5.1 Introduction

As discussed in the previous chapter, seamless connectivity and multihoming for UDP have been implemented to support streaming applications and even VoIP application. TCP is more complicated and raises various issues such as control overhead, re-establishment problem or the “slow start” leading to the reduction of window size.

The author proposes a novel network layer multihoming and mobility support system (MMS) for multihomed mobile devices. The system can support TCP multihoming and provide seamless connectivity for multihomed mobile devices. Since it is built on the network layer, the system can support multiple types of interfaces including WiMax, WLAN

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

and UMTS/3G. The system focuses on the following requirements:

1. Seamless terminal mobility amongst different access networks.
2. Bandwidth aggregation for data transfer over multiple links.
3. QoS support at network layer.

5.2 Motivation, Challenges and Approaches

As technology nowadays has empowered mobile devices with multiple accesses to the Internet, mobile users will certainly wish to take advantage of the technology to maximize their benefits when using the Internet, especially for data streaming, such as video streaming or VoIP services. However, different technologies with different access and complicated network protocols prevent applications from effectively exploiting them. In most cases, the application needs to rely on either network-based solution or terminal-based solution due to the following reasons:

1. Unable to handle different interfaces at once.
2. Complicated protocol needs sophisticated solution - how to address the issue of maintaining QoS level while using multihomed solution.
3. A virtual interface is needed everytime, and therefore, Care-of-Addresses (CoAs) and Home addresses (HoAs) mapping are complicated.

Furthermore, TCP provides congestion control. Therefore, it is crucial to notify the transport layer at both ends to adjust the rate of data transfer when the MN moves to a new network. If MN's sending rate is too fast for the new network link, substantial packet loss will result if the transport layer does not reinitialize its congestion control state for the new network path. A "slow start" will occur during handover, if the receiver side senses a link degradation and requests to reduce the window size of the TCP flow, impacting performance especially for real-time applications.

TCP packet is sequenced and sent in order in a normal channel. For multiple links,

Figure 5.1: Network architecture

each packet is redirected to multiple network paths which has an unknown number of hops. This may increase the end-to-end delay of a certain packet. If the delay differences between links are significant, the overall average delay will be substantial and link quality will be degraded. Out-of-order incoming packets could spawn a series of redundant retransmission packets, causing the slowest link to be even more jammed, further compounding the out-of-order problem.

Fig. 5.1 shows a scenario where a Mobile Node (MN) connects to a Correspondence Node (CN) through three different interfaces (GPRS, WLAN and WiMax).

5.3 System Architecture

Fig. 5.2 shows architecture of the proposed system. The system spans from user space to the kernel space, comprising of both kernel modules and application agents. The sys-

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

Figure 5.2: System architecture

tem comprises IP Swapping, IP/Ports Mapping, Packet Redirecting and Communication Modules as well as the Intelligent Software Agent.

As mentioned in chapter 3, addresses of outgoing and incoming packets are altered in the kernel space. The application is not aware of the changes under the data-link layer where different interfaces are used to transfer the packets. Outgoing data is transferred from the application to the kernel space, IP header is swapped and matched to the corresponding CoAs of the destination, then the packet is routed to the interface accordingly. Incoming data packets IP header is compared and swapped with the stored CoAs and HoAs in the Mapping Reference Table (MRT). After that, the packet returns to the user space to the application with the same IPs as previous packets. The application accepts the packets as a normal connection.

5.3.1 IP Swapping Module

IP swapping module is located at the network layer. It gains access to both incoming and out-going packets, allowing it to alter the IP headers. When multihoming packets

5.3. System Architecture

are received, each packet is assigned to a particular interface from the packet redirecting module. The IP swapping module matches the source IP with the corresponding CoAs in the IP mapping table and replace it with HoA. Likewise for the destination IP address. Note that one HoA can correspond to multiple CoAs in the mapping table if that machine (the current device or the communicating party) has multiple interfaces in active mode. It means that all CoAs and HoAs have been registered before at the SIP server. The registration step is discussed in the next section.

At the initialization stage, the HoAs of both parties are recorded together with the local port and the remote port.

For the outgoing packets, the destination address is compared and matched to the corresponding CoAs. There might be more than one, in case the other party is also using multiple interfaces. The source address for each packet is replaced by the kernel, with the corresponding IP address of the interface that it uses to send out the packet.

For the incoming packets, the destination IP address is swapped with the virtual interface's IP address which is a fixed IP address assigned at the start of communication. The source IP address is passed to the IP and port mapping module, to check whether the packet belongs to the current connection. If it does, the source IP address is swapped with the corresponding HoA. After that, the packet is passed to the upper layers. At application level, the running application is not aware of changes in the IP addresses since there are no changes in the source and destination IP addresses from the application's perspective. Therefore, the packet is accepted and processed as per normal one. For UDP packet, there is out-of-sequence packet problem and we have already proposed an In-Kernel-Re-Ordering Service in which packet is marked and buffered at the kernel level. For TCP packet, there is no out-of-sequence packet problem, as this IP swapping module is located at the network layer. The transport layer processes the packet after it is modified at the network layer where the checksum and QoS controls are processed here by the kernel functions. There is a problem with TCP packet when the window size of each packet is reduced during

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

handover as the receiver will perceive that the connection is interrupted due to network congestion. It will therefore send a message to reduce the window size of the subsequent packets. This is called *slow start* and this problem could be solved by a buffer at the kernel level. More details will be given in the following sections.

5.3.2 IP and Port Mapping Module

In this module, the IP address is translated or mapped to multiple CoAs stored in a table. At initialization stage, the destination IP address, source IP address, local port, remote port are all combined into a row entry to represent a connection. Each connection is assigned with a unique index which is generated from the local port and the remote port.

An arbitrary constant IP address from an unused private prefix is assigned to the virtual interface as the permanent address of the local machine. For UDP connection, since there is no relation between two parties, the connection could be broken at any time and is called connectionless. Each connection could be represented with a remote port and destination IP address. For TCP connection, however, each connection is maintained with a session controlled by the transport layer. TCP process creates a sub-connection with a random port to the other party without informing the mapping module. Therefore, we sort it by hashing both local port and remote port to generate a unique index for each active connection. This unique identification number is used to match incoming packets if they belong to an existing connection else a new connection is established.

$$index = port \text{ XOR } peer_port$$

As a peer node may have multiple CoAs, the mapping table is used to direct packets to the respective connections and to lookup the relevant CoAs to represent the IP address. As there is no awareness of multiple CoAs, the IP address is swapped and passed to the application and hence the entire process is transparent to the upper layers. It enables ongoing sessions to survive when multiple interfaces in use are brought down.

5.3.3 Packet Redirecting Module

This module is to redirect outgoing packet to the corresponding interface or incoming packet to the upper level. It is located at the network layer so that outgoing packets have to go through this module to reach the link layer or before incoming packets enter the network layer.

After outgoing packet is examined and altered, the source IP address is still the arbitrary address which is given from the virtual interface. The module is responsible for redirecting packet according to the policy set by the user. Each interface has a different IP address and is assigned to a different route table. The source IP address is replaced by the outgoing interface's IP address and attached to the corresponding routing entry. The routing entry is set up by the intelligent software at the beginning. Each interface has a separate routing table which stores the gateway, next node and the name of the interface. Routing table is updated during switching time when new access point is found and new connection is established.

When a packet is assigned to a certain interface, the module will seek out the routing information in the corresponding routing table and request for the packet to be re-route to the normal network flow. The packet is again returned to the network flow and routed out through the desired path.

The path availability is decided during the handover when a new network is discovered and routing information is gathered from the router. This router can be the nearest router or a router specified by a centralized router, in this case, the SIP server serves this role. The path could be complicated if each domain does not have direct contact with each other. Router information is propagated in routers within the path. Here, we assume that the routing information is retrieved from the centralized SIP server which is known to be accessible to all other nodes.

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

5.3.4 Communication Module

This module is used to communicate between the kernel modules and software agent at the user level. This module uses IOCTL commands so that information and command could be passed from the user level. The information includes active interfaces and policy that user desires to use at the redirecting module.

The module use IOCTL commands [42], which are supported by Linux Ubuntu [43] core kernel to build a virtual point in the network flow. Data commands are passed down from the user space in real time and could immediately affect the process. The communication module also sets up a signaling protocol to inform the user space about interface changes in the MAC layer. Signal strength, lack of beacons or dropping of bit rates are polled by the intelligent software agent. When the link information is available, the agent shall decide whether to tear down and seek out a new connection. Depending on the hardware design, the MAC layer information requires different drivers to access.

5.3.5 Intelligent Software Agent

Intelligent software agent is an application running in user level as a daemon programme. Its responsibility is to control the number of interfaces in use, creating a communication channel with the modules at kernel level. It searches for AP to find the best access point for wireless network or scans the wireless network to check if the current signal quality is lower than required. The signal quality of each link is passed from the kernel through the communication module to this agent software. We apply a simple algorithm to rank each wireless network according to a certain level of acceptance. If the current wireless network is lower than the accepted level, the agent sends request to tear down the link and starts searching for new wireless network. The best access point is selected and a new connection is re-established. After this, the kernel modules would be informed of the new connection and a binding update is sent to the SIP server to inform other nodes so

Figure 5.3: Network architecture

that they could update their IP mapping tables.

5.4 Implementation

5.4.1 Experiment Settings

This section describes the implementation of our proposed framework. Fig. 5.3 shows the test-bed setup to measure the performance of the proposed system. We have one Correspondent Node (CN) communicating with a Mobile Node (MN) which is equipped with three interfaces: UMTS, WLAN and WiMax. The Mobile Node has access to three different networks. As mentioned before, we do not take into account the authentication and authorization, e.g. access to MN is presumed to exist before starting the experiment.

The scenario consists of a setup with three different links to MN with the presence of a SIP server in a public IP. CN transfers data to MN through different links and the

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

MN moves among the networks (including WiMax, WLAN and UMTS). It means that there are periods when MN uses all three links or when there is no network access at all. When MN moves, we run a streaming application using both UDP and TCP connections to connect the two nodes. The video streaming application is fed by a webcam at CN and streams live video to MN. The real time application requires that the video be stable and undistorted during the transmission. QoS is supported at both the application and the kernel levels.

We measure the performance of the proposed system based on three metrics, namely performance delay, packet loss and user satisfaction.

1. Performance delay: or end-to-end delay, measures the time taken for a packet to travel from CN to MN. It is a critical factor in interactive real time application. Even though streaming applications have a playout buffer which stores incoming packets before playing the frames, the end-to-end delay is still very substantial. Our experimental results show that the end-to-end delay cannot be any lower than 2 seconds from CN to MN during handover. The delay is not significant when there are still active links. However, it is a problem when there is no active link available or there is a great drop in bandwidth. For instance, when MN has to use 3G/UMTS which has limited bandwidth to receive data while the other two links are unavailable.

2. Packet loss: packet loss occurs all the time especially during handover period. Since we use multiple interfaces, we inject a buffer in the kernel, which we use to re-order the out-of-order packets. For UDP packet, the scheme improves the performance, as there is no flow control being supported in the protocol. However, for TCP packet, which has its own QoS control in the protocol, the improvement is not prominent.

At the network layer, all incoming packets are captured before entering the transport layer where TCP protocol processes the packets by counting the sequence number. Out-of-order packets occur as the delay in each path for each packet varies. This delay depends on the number of hops between two ends of each path and it can severely affect large

5.4. Implementation

packets which will be split to smaller packets. There are some proposals [44, 45] which solve the issue by injecting Media Independent Handover (MIH) header into the packet. Thus the quality of each link is conveyed to the sender and the sender can select the path intelligently.

3. User satisfaction: this is based on user satisfaction including informal subjective rating of the audio/video quality, especially during the handover period. Since it is often vague to define good or bad quality, it must be decided statistically using user surveys where various parameters are graded. We formed a group of 20 users and let them watch/listen to different types of video/audio streaming. Users grade them in terms of quality and their own judgments using the MOS score of 1 to 5.

5.4.2 Experimental Results

The MN has three links including WiMax, WLAN and UMTS. At first, MN connects to CN through WiMax which has the longest range. Thereafter, when MN moves into the WLAN range, it turns on WLAN to connect to CN. Finally, UMTS is used with the other two links to deliver packets to the CN. We used round-robin to distribute the packets over the available links. The results are shown in Fig. 5.4 and Fig. 5.5.

Fig. 5.4 shows the end-to-end delay of TCP transmission from MN to CN.

At $t = 0$ s, MN uses only WiMax link, which maintains a reliable and high-speed link. It yields an end-to-end delay of $d = 27$ ms on average. At $t = 15$ s, MN connects to the WLAN network, using two links at the same time. As we can observe, the end-to-end delay increases to $d = 70$ ms on average. As the WLAN link has lower range and more hops than WiMAX which is only one hop away, the average end-to-end delay is increased. At $t = 35$ s, MN connects to UMTS network and it uses all three links at the same time to send packets to CN. It is shown that the average end-to-end delay is increased to $d = 300$ ms. Since UMTS has more hops than WLAN networks, the delay is increased correspondingly.

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

Figure 5.4: End-to-End delay

5.4. Implementation

At $t = 55$ s, MN disconnects from the UMTS network, and goes back to using two links (WiMax and WLAN), resulting in the drop of average end-to-end delay to $d = 70$ ms. Finally, when MN moves back to use a single WLAN link, the average delay drops to $d = 27$ ms.

As it is observed, since all network links have different delays, they affect each other and strangle other packets if we use round-robin algorithm for load-balancing. When a packet using a slow link arrives in the CN, it is queued before the transport layer if it is not in order. The transport layer will send the re-transmission request if the timer runs out. The next time, the packet is sent from the faster link and it arrives in the correct order. Therefore, on average, the delay is increased.

Fig. 5.5 shows the packet delivery from $t = 0$ s to $t = 95$ s and Fig. 5.6 zooms in a handover period which is from $t = 10$ s to $t = 20$ s. As TCP is used, there is no packet loss during the handover. However, Fig. 5.6 shows that during the handover period, even though there is no lost packet and all of the packets arrived in the correct order, the interval of arrival has changed and packets are delivered in bundles. This is because before the handover, arriving packets are queued at the receiver end (while the MN is trying to connect to a different link) and during the handover, these packets are grouped and delivered in bundle at the same time to the application level. This proves that the proposed system could provide seamless connectivity during handover in heterogeneous networks.

For our listening tests, users listen to an audio stream from MN to CN and are then asked to rate the audio quality throughout the streaming session with handovers. The MN and CN are both laptops and we use a lower bitrate stream than the previous scenario, to transfer from MN to CN and get the average MOS of 20 listeners which stands at 3.66, which is fair quality for the listeners.

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

Figure 5.5: Packet delivery

Figure 5.6: Packet delivery from $t = 10$ s to $t = 20$ s

5.5 Discussion

In this section, we discuss the strengths and limitations of the proposed framework as well as the possible improvements.

5.5.1 Aggregate Link and Seamless Connectivity

By distributing traffic across multiple links, the system aggregates links to achieve increased total throughput and does not modify the existing network topology or requires any new protocols. When the mobile node moves to a new network, the connectivity is maintained using soft-handoff mechanism. By interfacing to different networks with varied ranges, the mobile node ensures that it is overlapped by different networks during roaming.

5.5.2 Routing based vs Tunnel based

Instead of tunnel based approaches like Mobile IP, where packets are re-routed from the Foreign agent to the Home Agent, the proposed framework uses routing based technique, attaching packets to different routes to the destination. Each packet would take different paths to the final destination. The approach reduces the triangular routing by redirecting packets directly to the destination node. The end-to-end delay is reduced since the time to tunnel to the Foreign agent is not included. However, since it needs routing information in order to direct to a correct route, it increases the header size of each packet. In the case where the new network is behind firewall/NAT [46], extra information of the source address and source port are included in the IP header which increases the size of the packet.

5.5.3 Network Detection and Handoff Algorithm

The Handoff (HO) detection and triggering are essential in wireless mobility as they occur often when mobile node moves among different wireless access networks. HO detection is decided by measurement of Relative Signal Strength (RSS) in the overlapped

or border region of two wireless networks. Each interface uses different technology and manufacturers use different standards to define the signal strength and link quality. To generate an optimized mechanism to distribute packets into different interfaces at the same time, we must normalize all the factors or parameters. Each link has different bandwidth, for instance, WLAN's bandwidth can vary from 5 Mb to 54 Mbps while WiMax can reach 70 Mbps and UMTS's bandwidth is around 50 Kbps. While WLAN covers short ranges, WiMax could reach further up to 50 km. Hence the signal strength and transmission power for WLAN register drastic change when MN leaves the coverage of one WLAN to another but for WiMax or UMTS, the signal strength and transmission power usually remain constant within the small region.

5.5.4 In-kernel Re-route Service

The routing process is entered after the IP header is modified at the network layer; routing information is searched and retrieved at the routing table. A module is designed to help maintain the routing information of each packet so that it can be retrieved easily at the network layer. Routing information is stored in different tables and assigned to different physical interfaces. The usual routing process is changed as the IP header is modified at the network layer. Therefore, the module searches through all the tables to retrieve the corresponding entry and attaches the packet to the correct physical interface.

5.5.5 Signaling among Different Networks

As the mobile host moves to different networks, the signaling mechanism requires that the new location be reported to the centralized SIP server. The signaling protocol requires routing information to be provided before switching to a new network, which increases the latency when the mobile node associates with the new network. While seeking the routing information from the local routers to connect to the centralized SIP server, the mobile

Chapter 5. Enhancement of Seamless Multi-homing Connectivity in Heterogeneous Networks for TCP Communication

host might encounter firewall or NAT issues in the different subnet networks. IP tunneling is one technique used to create a virtual private network (VPN) among the SIP server, the mobile node and other peers. However, this approach is infeasible if the firewall is not a simple symmetric cone type and it reduces the scalability of the system when newly joined hosts need to set up the VPN. A STUN server is required to establish a tunnel to solve NAT issues, which are common in current UMTS networks where different operators possess their own private IP networks.

5.5.6 Out-of-Sequence or Missing Packets

As we applied packet distribution instead of flow distribution to different links, for unreliable channel using UDP, number of out-of-sequence or missing packets are inevitable. There are several approaches to reduce the issues including header injection or flagged packets. Several approaches include buffer usage at the receiving end to re-order the packet.

An In-Kernel Re-Order service is implemented to solve the out-of-sequence packets in UDP transmission. A buffer is used to hold the packets before they enter the network layer. Each packet is marked with a flag field in the UDP header. The sequence is marked so that the receiving end could track down the order of incoming packets. If the number of out-of-sequence packets is larger than the buffer's capacity, the oldest packet in the queue is removed and the new packet stacks up. It has been proved in simulation that the buffer is sufficient in reducing the total number of loss packets in UDP transmission

For reliable TCP transmission, although there is flow control at transport layer, overhead from retransmission packets can reduce the quality of the stream. As there are different paths with unknown end-to-end delays and handoffs may occur along the route from source to destination, the cumulative latency experienced by a packet will lead to an increase in retransmission requests since the transport layer is unaware of the presence of

multiple interfaces. Retransmissions consume valuable bandwidth and degrade link quality, resulting in that the receiving ends having to issue window resize advertisement to immediate routers and sender to reduce the sending packet's TCP window size.

A solution to the problem could be a network detection scheme to detect any changes in the delay of each link. As the sender does not have the ability to sense the number of hops in a certain link, a trial period could be executed when the MN joins a new network. Packets are sent in the link and delays measured at the receiver are compared to those of existing links. The sender is informed of the respective link delays so that the sender could change the load balancing policy in distributing packets to its physical interfaces. A slow link will get less traffic. However, it means that a signaling protocol has to be designed to signal the changes between sender and receivers, which complicates the deployment of the system.

5.6 Conclusion

TCP is a complicated and reliable protocol with congestion control. It is shown that when applying our proposed system for multihomed devices, it increases the overhead packets and the re-establishment problem or the "slow start" leads to the reduction of window size.

Even though, the system is equipped with multihoming support capability, it needs to be reconfigured to support different wireless access links. Although TCP presents challenging issues and unsolved problems, the experiment and implementation show that the proposed MMS meets the user requirements and provides seamless connectivity during handover in heterogeneous networks.

Chapter 6

Multihoming in Delay Tolerant Network

6.1 Introduction

Delay Tolerant Networking (DTN) is an emerging research area that takes a different approach to (inter)networking and allows working in unstable as well as in highly heterogeneous environments. In order to research on the performance of DTN, Opportunistic Network Environment (ONE) simulator [4] is developed to provide an environment to experiment with the different routing protocols and to visualize the simulation interactively in a real-time manner.

This chapter describes the following applied research work performed:

1. Modification of ONE simulator

The ONE simulator is modified to support the simulation of DTN with multiple network interfaces.

A GUI tool has been developed for customizing geographic information data for ONE simulator to create new simulation environment with heterogeneous network where each

mobile node has multiple access to different networks (UHF, UMTS, WLAN). This simulation framework can be customized to work with different kinds of maps and although the physical interfaces are abstractly designed, link quality, coverage range, transmission power can be configurable or dynamically changed during the movement of the nodes. The robustness can be increased by an open API so that different routing protocols can be applied to measure the performance metrics of the protocols and analyze the performance.

2. Modification of DTN routing protocol: MaxProp

Equipped with multiple network interfaces, DTN nodes can perform better in some metrics than nodes with single interface. The DTN routing protocol, MaxProp [32] has been modified to take advantage of the multiple network interfaces so that it can now perform neighbour discovery of the multiple interfaces in the multihomed nodes as well as select the best interface in the multihomed DTN node for data transmission.

6.2 Modification of ONE Simulator

6.2.1 Overview

ONE is a Java-based simulation [4]. It is able to use various trace formats as inputs and interface to other simulator engines (ns2 [47], dtnsim2 [48]). ONE simulator is capable of generating node movement using different movement models; routing messages between nodes with various routing algorithms and with different sender and receiver types; visualizing both mobility and message passing in real time in its graphical user interface as shown in Fig. 6.1.

There are several existing DTN simulators which simulate the mobile node's movements, connections and packet exchanges, for example [49] or [50]. However, most of them do not support multiple-interface capability on the mobile node. ONE simulator proves to be a potential candidate to be incorporated with the multi-interface-capability DTN node due to the ease of development using Java and the strong visualization support.

Chapter 6. Multihoming in Delay Tolerant Network

ONE simulator consists of components such as the mobility and connectivity simulator, the simulation results visualizer and the simulation report generator.

To support the simulation of DTN in a multiple-interface, heterogeneous network environment, the connectivity simulator is the primary module to work on.

Using Java, the simulator offers robust and open API to develop a framework such that different routing algorithms are used. GPS map data provides the scenario with roads or buildings where mobile nodes move along to simulate the real situation. Flexible inputs are configurable to modify the environment including speed, wireless range and random patterns are used to generate the practical movements of mobile nodes.

There are numerous recent research topics in routing in DTN network including Epidemic routing, Prophet routing [31], model based routing, ferry based routing and link state routing. Coding schemes are also proposed to improve the robustness on data delivery. In DTN reference implementation (DTN2), three types of routing schemes are implemented, namely static routing, DTLSR routing [51] and Prophet routing. Fig. 6.1 is an illustration of the DTN trial setup to enable testing of new routing protocols. It provides an interface abstraction to represent a network connectivity containing information such as the type of convergence layer, etc.

6.2.2 Modifications of ONE Simulator

The ONE simulator's architecture is shown in Fig. 6.2. The following are some important modules that are modified to create a new simulator with multihomed capability. DTNHost is the core module responsible for simulating a mobile host's movement and message exchange behaviors.

DTNHost: Handles the mobile node's attributes. It acts as the abstract node in the framework where the different entities such as bus, pedestrians, cars, etc. are created. Each of these entities has its own attributes and behaviors such as storage capability,

6.2. Modification of ONE Simulator

Figure 6.1: ONE simulator screen shots

Chapter 6. Multihoming in Delay Tolerant Network

Figure 6.2: ONE simulator architecture

range, power, speed, etc.

Routing Model: Defines routing protocols to exchange messages when two nodes meet with each other within a certain range.

Connection Listener: Handles connections when two nodes have a chance to contact with each other.

As each node has multihomed capability, the routing model which is dependent on the range and capability of the interface is modified to support multiple interfaces. Connection Listener will detect that two nodes are within the range of each other, e.g. less than the longest possible communication ranges of both of them, to create a valid connection. When the connection is made, data is exchanged as defined in the routing model. If two nodes make more than one connection by using their multiple interfaces, multiple connections are concurrently active at one time and thus support link aggregation.

6.3 Modification of DTN Routing Protocol - MaxProp

The existing DTN routing protocol, MaxProp [32], is modified to exploit the multiple network interfaces. MaxProp is a simple routing protocol which is based on prioritizing the

6.4. Estimation of Delivery Likelihood

packets to facilitate transmitting or dropping the packets from the buffer. An improved MaxProp routing protocol which uses the multiple network interfaces is designed.

6.3.1 Utilization of Interfaces

Each interface concurrently transmits packets from one common shared buffer, and store all incoming packets to that buffer. This means there is no replication of packets and the same packet is sent via two independent links. Alternatively, only one of the interfaces is used to send the packets and the other interface is switched on only when the current one fails.

6.3.2 Buffer Management Policy

All interfaces in a DTNHost share a common buffer. The buffer management policy follows the original MaxProp[32]. It uses the common shared buffer and logically splits the buffer space into two parts. A constant threshold is defined to sort packets into two different parts. The packets whose hop counts are less than the threshold have a higher priority and are stored in one part of the buffer; while those whose hop count is higher than the threshold have a lower priority and are stored in the other part. In this way, maximum use of the buffer is ensured. The list of stored packets is ranked based on a cost assigned to each destination. The next destination is decided based on this list. The cost is an estimate of the delivery likelihood from that node to the requested destination, the details of which are given in the following section.

6.4 Estimation of Delivery Likelihood

In the original MaxProp, each node keeps track of a probability of meeting other peers. So, in MaxProp with two interfaces each node keeps track of two probability values

Chapter 6. Multihoming in Delay Tolerant Network

of meeting another node using two interfaces. For example, a node i has two values of the probability of meeting another node j :

$f(i, j)(1)$ using interface 1,

$f(i, j)(2)$ using interface 2.

For all nodes, $f(i, j)(m)$ is initially set to $\frac{1}{2(|s|-1)}$ where s represents the set of nodes in the network. When node j is encountered using interface m , the value of $f(i, j)(m)$ is incremented by 1 and then all values of f are renormalized. Like the original MaxProp, each time two peers meet, they exchange these values with one another. Each node calculates the cost for each possible path to the destination of the packets, up to n hops long. The path cost between two nodes which have two interfaces is the average meeting probability of two nodes. For example, if the meeting probability of interfaces 1 in peers A and B is p_1 and for interfaces 2 is p_2 , the path cost AB is calculated as:

$$p_{AB} = 1 - (1 - p_1)(1 - p_2)$$

The cost for a destination is the lowest path cost among all possible paths.

6.5 Experimental results

6.5.1 Multiple Interfaces with Same Type

Firstly, we simulated the DTN where each host has two interfaces of the same type (i.e. same wave transmit range, same transmit speed). For comparison, we also simulate the original MaxProp DTN which uses one interface of the same type. A total of 20 DTN nodes are simulated.

Since neither the contact time between two hosts nor the contact chance differs very much, it is expected that the delivery rate and the message latency are almost the same as the performance of the original MaxProp. Fig. 6.3 and Fig. 6.4 show the result of the simulation. As expected, the performance is about the same. This is because at any one time, only one interface is selected to be used in the data transmission by the improved

Figure 6.3: Delivery rate (two interfaces of the same type)

MaxProp. Theoretically, the improved MaxProp will choose the more efficient of the two interfaces for data transmission and hence the delivery rate will be improved. However, in this case, as the two interfaces are of the same type (WLAN) as the single-interface DTN, the performances are thus similar. Another point to note is that with single interface, once it fails, data transmission is terminated whereas with multiple interfaces, the improved MaxProp can still continue with data transmission via the other interface.

6.5.2 Multiple Interfaces of Different Types

We simulate the DTN where each host has two different types of interfaces (i.e. one with a larger wave transmit range but less capacity representing UMTS, the other with a smaller range but more capacity representing WLAN). For comparison, we simulate the original MaxProp which uses only one interface, the one with the larger range and less capacity. Fig. 6.5 and Fig. 6.6 show result from the output of ONE simulation. Since there is more channel to communicate between two nodes, then both the delivery rate and the message latency improve.

Chapter 6. Multihoming in Delay Tolerant Network

Figure 6.4: Mean latency (two interfaces of the same type)

As mentioned above, all of interfaces share a common buffer. Although different interfaces are used, only one interface is used to transfer packets to the destination node at a time (after the best path to destination node is chosen). Technically, with two interfaces, the host node can both transfer and receive data at the same time which could improve the overall performance on delivery rate of the whole network. However, in this simulation, we only consider the delivery rate between two single nodes at any contact time.

Compared with the previous case with same type of interface, the DTN host with different types of interfaces yields a much better delivery rate and mean latency with smaller buffer size for both original and advanced MaxProp.

Original MaxProp with two interfaces of different types yields 63% successful delivery (Fig. 6.5), 730 minute delay (Fig. 6.6) with 0.5 MB buffer versus 28% (Fig. 6.3) and 1550 minute delay (Fig. 6.4) with 5 MB buffer of those with two interfaces of the same type.

Advanced MaxProp with two interfaces of different types yields 80% successful delivery (Fig. 6.5) and 820 minute delay (Fig. 6.6) with 0.5 MB buffer versus 28% (Fig. 6.3) and 1600 minute delay (Fig. 6.4) with 5 MB buffer of those with two interfaces of the same type.

6.5. Experimental results

Figure 6.5: Delivery rate (two interfaces of different types)

Figure 6.6: Mean latency (two interfaces of different types)

Chapter 6. Multihoming in Delay Tolerant Network

The usage of multiple different interfaces gives the advanced MaxProp an advantage in terms of delivery rate where buffer size is less than 3 MB; with larger buffer size, both of the protocols perform almost equally. As the best path is always chosen for data transfer, queuing time for data packets is reduced; therefore the mean latency of advanced MaxProp is lower than the original one as shown in Fig. 6.6. The increased delivery rate also reduces the chances of buffer overrun resulting in better delivery rate as shown in Fig. 6.5.

6.6 Discussion and Conclusion

An improved MaxProp which exploits the presence of multiple network interfaces has been designed. The simulation environment for testing the performance of the DTN consisting of multiple heterogeneous networks has also been designed from the original ONE simulator. Simulation results show improvement in the performance of the modified multi-interface MaxProp compared to the original MaxProp especially when the multiple interfaces are of different access technologies. This is because the improved MaxProp is able to perform neighbour discovery to learn about the multiple interfaces and then selects the most efficient interface for data transmission.

Improvement is more significant when the buffer size is not enough. The improvement is still perceptible even when the additional interface is very much inferior to the primary interface in terms of bandwidth as shown in Section 6.5.2 where the two interfaces comprise the lower capacity, longer range UMTS and the higher capacity, shorter range WLAN as opposed to that in Section 6.5.1 where the two interfaces comprise both higher capacity WLAN. Such is the benefit of having multihomed capability in a DTN node.

Chapter 7

Conclusion and Future Works

7.1 Conclusion

In this dissertation, the author proposes a novel framework that supports seamless mobility for multi-homed devices in heterogeneous networks. UMS system enables multi-homing capability for mobile devices with multiple interfaces where all the interfaces can be utilized to increase the effective bandwidth of the mobile device as well as supports seamless handover between interfaces. Compared with other works, the framework can support UDP (UMS) and TCP transmission and there is no need to incur capital investment on new infrastructure. The main contributions of UMS are the SIP based host mobility support, the virtual interface enabled multi-homing and QoS-aware flow distribution. UMS allows both vertical and horizontal handovers. By using the virtual interface, the node identifier and the locators are decoupled and the change of locator is transparent to upper layers. The use of IKPR service helps to reduce out-of-sequence packets.

A similar idea of supporting mobility is designed for Symbian mobile devices which have more limited resources and computational capability. This application level framework provides seamless handover between 3G/UMTS and WLAN networks while main-

Chapter 7. Conclusion and Future Works

taining QoS requirements for delay-sensitive real-time VoIP applications. Compared to other approaches, the proposed framework lies in the mobile node layer, and therefore, requires less change in the network infrastructure, no implementation in the kernel space and no change in network protocols. It is thus portable across Symbian based mobile devices and easily customized to current VoIP systems. The application framework takes advantage of multi-homing capability of Symbian OS, using 3G-WLAN handover to ensure seamless connectivity for delay sensitive VoIP applications. Experiment results show that, with extra buffer, along with the VoIP client application's own buffer, the application can reduce the amount of packet loss and out-of-sequence packet during the handover. The solution is shown to be effective in supporting VoIP service requirements and achieving seamless mobility for both vertical and horizontal handovers. It will serve as a basic for further study to support other interactive multimedia applications.

We extended multihoming concept to DTN using MaxProp as the routing protocol. MaxProp has been modified to exploit the multihoming capability of DTN nodes to yield better performance in packet loss rate and delivery. Simulation results show that the multihomed MaxProp improves the probability of packet delivery and reduces message latency. The improvement in performance is even more prominent when buffer space is limited in the DTN node. The Opportunistic Network Environment (ONE) simulator which supports simulations with different routing protocols and measurements under different scenarios with customizable GUI maps, has been modified to support simulation of multihoming routing protocols.

7.2 Future works

UDP multihoming has proven to be efficient in providing a reliable connectionless service and at the same time, increasing total bandwidth in the data transfer. We propose future research work in the security aspects of the end applications by encapsulating

7.2. Future works

authentication information in the Internet layer or using a secure channel with Internet key exchange method. As the IP swapping method is based on addresses of locations and ports, they are vulnerable to spoofing attacks where imposters can join the network to get information of other nodes.

As multihomed devices allow traffic to be distributed over different links, load balancing to optimize the bandwidth is another interesting topic for future work. The decision making process is based on various parameters including the end-to-end delay, bandwidth capability and cost of the network paths. However, some of the parameters are not known or hard to retrieve such as end-to-end delay for a non-visited link or unreliability of wireless networks which causes the actual link bandwidth to be very much different from the default values. All these parameters need to be considered in building a load balancing scheme to optimize the traffic distribution.

Experimental results in this dissertation show that the proposed VoIP application framework satisfies most of the requirements of VoIP services. The framework is promising and further research should be conducted to support other real-time streaming applications.

Terminal mobility support can be extended to vehicular delay tolerant network (DTN), in which each mobile node is equipped with multiple interfaces. Mobile nodes are connected in the wireless ad-hoc mode (with store-and-forward capability).

List of Publications

1. Nguyen, D. D.; Xia, Y.; Son, M. N.; Yeo, C. K. & Lee, B. S, "*A Mobility Management Scheme with QoS Support for Heterogeneous Multihomed Mobile Nodes*". IEEE Global Telecommunications Conference (GLOBECOM), 2008.
2. Tantra, J. W.; Son, M. N.; Nguyen, D. D.; Lim, T. M.; Yeo, C. K. & Lee, B. S, "*Seamless Mobility Across Heterogeneous Wireless Domain*". 5th IEEE Consumer Communications and Networking Conference (CCNC), 2008.
3. N.S. Mai, D.N. Dang, Y. Xia, C.K. Yeo, B.S. Lee, "*Seamless Handover Across Heterogeneous Wireless Networks*". NTU InfoComm Cluster Seminar, Singapore, 2008.
4. Nguyen, D. D.; Lim, T. M; Yeo, C. K. & Lee, B. S, "*Seamless Handoff for VoIP Applications in Heterogeneous Networks*". submitted to 7th IEEE Consumer Communications and Networking Conference (CCNC), 2010.

References

- [1] Symbian, “www.symbian.com/.”
- [2] I. Gartner, “Gartner Says Worldwide Smartphone Sales Grew 16 Per Cent in Second Quarter of 2008,” Gartner, august 2008. [Online]. Available: <http://www.gartner.com/it/page.jsp?id=754112>
- [3] V. Cerf, S. Burleigh, A. Hooke, L. Torgerson, R. Durst, K. Scott, K. Fall, and H. Weiss, “Delay-Tolerant Networking Architecture,” RFC 4838 (Informational), Apr. 2007. [Online]. Available: <http://www.ietf.org/rfc/rfc4838.txt>
- [4] ONE, “<http://www.netlab.tkk.fi/tutkimus/dtn/theone/>.”
- [5] B.-S. Lee, T. M. Lim, C. K. Yeo, and Q. V. Le, “A Mobility Scheme for Personal and Terminal Mobility,” *Communications, 2007. ICC '07. IEEE International Conference on*, pp. 1815–1820, 24-28 June 2007.
- [6] F. Q. V. L. Teck Meng Lim; Chai Kiat Yeo; Lee, “TMSP: Terminal Mobility Support Protocol,” *IEEE Transactions on Mobile Computing*, vol. 8, no. 6, pp. 849–863, 2009.
- [7] C. Perkins, “IP Mobility Support,” RFC 2002 (Proposed Standard), Oct. 1996, obsoleted by RFC 3220, updated by RFC 2290. [Online]. Available: <http://www.ietf.org/rfc/rfc2002.txt>
- [8] D. Johnson, C. Perkins, and J. Arkko, “Mobility Support in IPv6,” RFC 3775

REFERENCES

- (Proposed Standard), Jun. 2004. [Online]. Available: <http://www.ietf.org/rfc/rfc3775.txt>
- [9] C. Perkins, "IP Mobility Support for IPv4," RFC 3344 (Proposed Standard), Aug. 2002, updated by RFC 4721. [Online]. Available: <http://www.ietf.org/rfc/rfc3344.txt>
- [10] N. Montavont and T. Noel, "Handover Management for Mobile Nodes in IPv6 Networks," *IEEE Communications Magazine*, vol. 40, pp. 38–43, 2002.
- [11] C. Ahlund, R. Brannstrom, K. Andersson, and O. Tjernstrom, "Port-based Multihomed Mobile IPv6 for Heterogeneous Networks," *Local Computer Networks, Proceedings 2006 31st IEEE Conference on*, pp. 567–568, Nov. 2006.
- [12] E. Nordmark and M. Bagnulo, "Shim6: Level 3 Multihoming Shim Protocol for IPv6," Internet-Draft, draft-ietf-shim6-protocol-10.txt, Feb. 2008.
- [13] T. Chen and L. Wenyu, "A Novel IPv6 Communication Framework: Mobile SHIM6 (M-SHIM6)," *Wireless Communications, Networking and Mobile Computing, 2006. WiCOM 2006. International Conference on*, pp. 1–3, 22–24 Sept. 2006.
- [14] M. Bagnulo, A. Garcia-Martinez, and A. Azcorra, "IPv6 Multihoming Support in The Mobile Internet," *Wireless Communications, IEEE [see also IEEE Personal Communications]*, vol. 14, no. 5, pp. 92–98, October 2007.
- [15] D. Le, J. Lei, and X. Fu, "A New Decentralized Mobility Management Service Architecture for IPv6-based Networks," in *WMuNeP '07: Proceedings of the 3rd ACM workshop on Wireless multimedia networking and performance modeling*. New York, NY, USA: ACM, 2007, pp. 54–61.
- [16] R. Atkinson, S. Bhatti, and S. Hailes, "A Proposal For Unifying Mobility With Multihoming, NAT, & Security," in *MobiWac '07: Proceedings of the 5th ACM international workshop on Mobility management and wireless access*. New York, NY,

REFERENCES

- USA: ACM, 2007, pp. 74–83.
- [17] H.-Y. Hsieh and R. Sivakumar, “pTCP: An End-to-end Transport Layer Protocol for Striped Connections,” *Network Protocols, 2002. Proceedings. 10th IEEE International Conference on*, pp. 24–33, 12-15 Nov. 2002.
 - [18] T. Goff and D. Phatak, “Unified Transport Layer Support for Data Striping and Host Mobility,” *Selected Areas in Communications, IEEE Journal on*, vol. 22, no. 4, pp. 737–746, May 2004.
 - [19] H. Huang, J. Cai, A. Kassler, and C. Fu, “Load-sharing in Wireless Multi-homed Systems,” *Communications, 2005. ICC 2005. 2005 IEEE International Conference on*, vol. 5, pp. 3489–3493 Vol. 5, 16-20 May 2005.
 - [20] M. Buddhikot, G. Chandranmenon, S. Han, Y. Lee, S. Miller, and L. Salgarelli, “Integration of 802.11 and Third-generation Wireless Data Networks,” in *INFOCOM 2003. Twenty-Second Annual Joint Conference of the IEEE Computer and Communications Societies. IEEE*, vol. 1, 30 March-3 April 2003, pp. 503–512vol.1.
 - [21] M. Handley, H. Schulzrinne, E. Schooler, and J. Rosenberg, “SIP: Session Initiation Protocol,” RFC 2543 (Proposed Standard), Mar. 1999, obsoleted by RFCs 3261, 3262, 3263, 3264, 3265. [Online]. Available: <http://www.ietf.org/rfc/rfc2543.txt>
 - [22] K. S. Munasinghe, M. R. Kibria, and A. Jamalipour, “Designing VoIP Session Management over Interworked WLAN-3G Network,” *IEEE Wireless Communication*, 2008.
 - [23] A. N. Cadavid, A. F. Millán, A. García, G. Tamura, and O. W. Jiménez, “Handset Based Automatic Network Re-selection System for GSM/GPRS and WiFi,” *Proc. ACM MobiWac*, 2008.
 - [24] L. Ong and J. Yoakum, “An Introduction to the Stream Control Transmission Protocol (SCTP),” RFC 3286 (Informational), May 2002. [Online]. Available:

REFERENCES

- <http://www.ietf.org/rfc/rfc3286.txt>
- [25] E. Kohler, M. Handley, and S. Floyd, "Datagram Congestion Control Protocol (DCCP)," RFC 4340 (Proposed Standard), Mar. 2006. [Online]. Available: <http://www.ietf.org/rfc/rfc4340.txt>
 - [26] C.-N. Chuah, L. Subramanian, R. Katz, and A. Joseph, "QoS Provisioning Using A Clearing House Architecture," in *Quality of Service, 2000. IWQOS. 2000 Eighth International Workshop on*, 5-7 June 2000, pp. 115–124.
 - [27] K. Fall, "A Delay-Tolerant Network Architecture for Challenged Internets," *Procc. ACM SIGCOMM*, 2003.
 - [28] M. Demmer and DTNRG, "DTN Reference Implementation," Dec 2004, available at <http://www.dtnrg.org>.
 - [29] E. N. Emulation, "<http://www.emulab.net>."
 - [30] X. Zhang, J. Kurose, B. Levine, D. Towsley, and H. Zhang, "Study of a Bus-based Disruption-Tolerant Network: Mobility Modeling and Impact on Routing," *ACM Mobicom*, 2007.
 - [31] J. Xue, J. Li, Y. Cao, and J. Fang, "Advanced PROPHET Routing in Delay Tolerant Network," in *Communication Software and Networks, 2009. ICCSN '09. International Conference on*, 27-28 Feb. 2009, pp. 411–413.
 - [32] J. Burgess, B. Gallagher, D. Jensen, and B. N. Levine, "MaxProp: Routing for Vehicle-Based Disruption-Tolerant Networks," in *INFOCOM 2006. 25th IEEE International Conference on Computer Communications. Proceedings*, April 2006, pp. 1–11.
 - [33] J. Xue, X. Fan, Y. Cao, J. Fang, and J. Li, "Spray and Wait Routing Based on Average Delivery Probability in Delay Tolerant Network," in *Networks Security*,

REFERENCES

-
- Wireless Communications and Trusted Computing, 2009. NSWCTC '09. International Conference on*, vol. 2, 25-26 April 2009, pp. 500–502.
- [34] D. D. Nguyen, Y. Xia, M. N. Son, C. K. Yeo, and B. S. Lee, “A Mobility Management Scheme with QoS Support for Heterogeneous Multihomed Mobile Nodes,” in *Global Telecommunications Conference, 2008. IEEE GLOBECOM 2008. IEEE*, Nov. 30 2008-Dec. 4 2008, pp. 1–6.
 - [35] M. Bernaschi, F. Cacace, G. Iannello, and M. Vellucci, “Mobility Management for VoIP on Heterogeneous Networks: Evaluation of Adaptive Schemes,” *IEEE Transactions on Mobile Computing*, 2007.
 - [36] STUN, “<http://www.voip-info.org/wiki/view/STUN>.”
 - [37] Devicescape, “<http://www.devicescape.com/>.”
 - [38] ICall, “<http://www.icall.com/>.”
 - [39] WeFi, “<http://www.wefi.com/>.”
 - [40] T. S. S. I.-T. S. P. T. T. Q. T. Installations, “Local Line Networks: Methods for objective and subjective assessment of quality Mean Opinion Score (MOS) Terminology,” ITU-T P.800.1, July 2006.
 - [41] M. O. S. Table, “<http://en.wikipedia.org/wiki/meanopinionscore/>.”
 - [42] I. C. to talk to the device, “<http://tldp.org/ldp/lkmpg/2.6/html/x892.html>.”
 - [43] L. Ubuntu, “<http://www.videolan.org/>.”
 - [44] C.-M. Huang, C.-H. Lee, and P.-H. Tseng, “Multihomed SIP-Based Network Mobility Using IEEE 802.21 Media Independent Handover,” in *Communications, 2009. ICC '09. IEEE International Conference on*, 14-18 June 2009, pp. 1–5.
 - [45] F. Iqbal, M. Javed, and G. Shah, “IEEE 802.21 Based Multihoming Approach to Vertical Handover in Heterogeneous Networks,” in *Computational Technologies in*

REFERENCES

- Electrical and Electronics Engineering, 2008. SIBIRCON 2008. IEEE Region 8 International Conference on*, 21-25 July 2008, pp. 449–452.
- [46] K. Egevang and P. Francis, “The IP Network Address Translator (NAT),” RFC 1631 (Informational), May 1994, obsoleted by RFC 3022. [Online]. Available: <http://www.ietf.org/rfc/rfc1631.txt>
- [47] N. S. 2, “<http://www.isi.edu/nsnam/ns/>.”
- [48] DTNSim2, “<http://watwire.uwaterloo.ca/dtn/sim/>.”
- [49] DTNSimulators, “<http://www-scf.usc.edu/apoorvaj/dtnsimulator.html>.”
- [50] K. framework, “<http://www.tzi.de/dku/research-dtn.html>.”
- [51] M. Demmer and K. Fall, “DTLSR: Delay Tolerant Routing for Developing Regions,” *ACM SIGCOMM*, vol. 5, 2007.