Network Reputation-based Quality Optimization of Video Delivery in Heterogeneous Wireless Environments

by

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A Dissertation submitted in fulfilment of the requirements for the award of Doctor of Philosophy (Ph.D.)

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December, 2016

Declaration

I hereby certify that this material, which I now submit for assessment on the programme of study leading to the award of Ph.D. is entirely my own work, that I have exercised reasonable care to ensure that the work is original, and does not to the best of my knowledge breach any law of copyright, and has not been taken from the work of others save and to the extent that such work has been cited and acknowledged within the text of my work.

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To my dear parents.

Acknowledgement

First and foremost I would like to thank my dearest parents, who have put great effort to raise me up and give me the opportunity to be where I am today. It is the strongest support from them that enables me to get over the toughest moments in life. It is the most precious and honest love from them that I benefit every day of my life. They have always been and will forever be my beloved ones.

An extra special acknowledgement is given to my supervisor, Dr. Gabriel-Miro Muntean, who has always been extremely kind, helpful and patient during my Ph.D. study period. Without his advice, I would not have completed this difficult path. I feel I have greatly benefited from his experience and his wisdom, especially in the past year. It would have not been possible for me to complete the research and this doctoral thesis without his help and support. I would like to pay my greatest respect and thank Gabi, who has been and will always be my dearest advisor and friend.

Also, I would like to acknowledge sincerely all my advisors during the Ph.D. study period: Dr. Ramona Trestian, who has kindly offered me great help countless of times during the past three years; Dr. Zhenhui Yuan, who has encouraged me to try my best in research; Dr. Bogdan Ciubotaru, with who it has been a pleasure to work; And Dr. Hrishikesh Venkataraman, who gave me a great experience when I joined his project!

Big thanks go to all my colleagues and friends in the DCU Performance Engineering Laboratory (PEL), including Shengyang, Longhao, Jiaping, Ruiqi, Yang, Yi, Irina, Sebastian, Martin, Quang, John, Anderson, Dan, Ronan, Lejla, Faisal, and Arthy. I was very lucky to meet you all and to spend the past years together. Your friendliness and help made me feel like I am at home and have never felt lonely in a different country.

Appreciations are also sent to the great technical staff members from the School of Electronic Engineering, Dublin City University, Ireland, who have been very supportive when different issues have occurred in the lab.

Special thanks to the Irish Research Council, the Irish Software Research Centre (LERO) and Mr. Alan O'Herlihy, the CEO of Everseen Ltd., who generously offered me financial support to complete this degree.

Final thanks are due to the examination team: Prof. Pablo Angueira, the external examiner, Dr. Conor McArdle, the internal examiner and Dr. Pascal Landais, the chairperson, who have made very useful recommendations regarding the thesis which made it much better.

Dublin, January 2017

Ting Bi

List of Publications

[Journals]

- Yuan, Zhenhui, **Ting Bi**, Gabriel-Miro Muntean, and Gheorghita Ghinea. "Perceived synchronization of mulsemedia services." IEEE Transactions on Multimedia 17, no. 7 (2015): 957-966.

[Conferences]

- **Ting Bi**, Ramona Trestian, and Gabriel-Miro Muntean. "Reputation-based network selection solution for improved video delivery quality in heterogeneous wireless network environments." In 2013 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB), pp. 1-8. IEEE, 2013.
- **Ting Bi**, Ramona Trestian, and Gabriel-Miro Muntean. "RLoad: Reputation-based load-balancing network selection strategy for heterogeneous wireless environments." In 2013 21st IEEE International Conference on Network Protocols (ICNP), pp. 1-3. IEEE, 2013.
- **Ting Bi**, Zhenhui Yuan, and Gabriel-Miro Muntean. "Network reputation-based stereoscopic 3D video delivery in heterogeneous networks." In 2014 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting, pp. 1-7. IEEE, 2014.
- **Ting Bi**, Zhenhui Yuan, Ramona Trestian, and Gabriel-Miro Muntean. "Uran: Utility-based reputation-oriented access network selection strategy for hetnets."

In 2015 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting, pp. 1-6. IEEE, 2015.

- **Ting Bi**, R. Trestian, Gabriel-Miro Muntean, "Reputation-based Network Selection Solution in Heterogeneous Wireless Network Environments", ITT Conference, Ireland, May 2013.
- Quentin, Sylvestre, **Ting Bi**, and Gabriel-Miro Muntean. "CLOVER: CLoud HandOVER Protocol for Multi-constrained Server Handover in the Mobile Cloud Context." IT&T (2015): 97.
- H. Venkataraman, **Ting Bi**, Gabriel-Miro Muntean, "DEAR: An Energy-centric Adaptive Region of Interest Mechanism for Wireless Mobile Devices", International Symposium on Wireless Personal Multimedia Communications, Hyderabad, India, Oct. 2015
- **Ting Bi**, Gabriel-Miro Muntean. "Location-aware Network Selection Mechanism in Heterogeneous Wireless Networks", The 2017 IEEE INFOCOM International Workshop on Mobility Management in the Networks of the Future World, Accepted.

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List of Abbreviations

AAC	Advanced Audio Coding
ABC	Always Best Connected
ABE	Always Best Experience
ADDA	Adaptive Data Delivery Algorithm
AHP	Analytic Hierarchy Process
AMPS	Advanced Mobile Phone System
ANP	Analytic network process
ANQP	Access Network Query Protocol
ANS	Access network selection
AP	Access point
AR	Augmented Reality
ATC	Approximately trust computation
BLER	Block error rate
BMSB	Broadband Multimedia Systems and Broadcasting
BSS	Base Station Subsystem
CA	Context Aware
CAGR	Compound annual growth rate
CBR	Constant bit-rate
CDF	Cumulative distribution function

CIR	Carrier to Interference Ratio
CMT	Concurrent Multipath Transfer
CN	Core Network
CommD	Communication Daemons
CTFT	Contrite Tit For Tat
DCCP	Datagram Congestion Control Protocol
DCE	Direct Code Execution
DDS	Data distribution scheduler
DDSIR	Double Decrement/Single Increment Ratio
DEA	Data envelopment analysis
DFF	Data Forwarding Function
DIBR	Depth image-based rendering
DMU	Decision making units
DSSS	Direct-Sequence Spread Spectrum
DTC	Dynamic trust computation
ECN	Explicit Congestion Notification
EDGE	Enhanced Data rates for GSM Evolution
EEP	Equal error protection
EPC	Evolved Packet Core
EPS	Evolved Packet System
EURANE	Enhanced UMTS Radio Access Network Extensions
FAF	Forward Attachment Function

FAHP	Fuzzy analytic hierarchy process
FANP	Fuzzy analytic network process
FDD	Frequency Division Duplexing
FEC	Forward Error Correction
FIFO	First in, first out
GFA	GPRS Foreign Agent
GPRS	General Packet Radio Service
GRC	Grey relational coefficient
GTFT	Generous Tit For Tat
HAFS	Hops Away From Source
HD	High-Definition
HDS	HTTP Dynamic Streaming
HQAM	Hierarchical quadrature amplitude modulation
ICC	International Conference on Communications
ICNP	International Conference on Network Protocols
IFOM	IP-Flow Mobility
IIS	Internet Information Services
IMS	IP Multimedia Subsystem
ISMP	Inter-System Mobility Policy
ISRP	Inter-System Routing Policy
ITU	International Telecommunication Union
JSCC	Joint source channel coding

LNM	Location & Network Monitor
LNS	Location-aware Network Selection
LNSA	Location-aware Network Selection Algorithm
LTE	Long Term Evolution
MAC	Medium Access Control
MANE	Media Aware Network Element
MAS	Multi-agent system
MDDM	Multipath Data Delivery Management
MEW	Multiplicative Exponential Weighting
MICS	Media Independent Command Service
MIES	Media Independent Event Service
MIHF	Media Independent Handover Function
MIHIS	MIH Information Server
MIIS	Media Independent Information Service
MIMO	Multiple Input Multiple Output
MLP	Mobile Location Protocol
MME	Mobility Management Entity
MN	Mobile Node
MO	Management Object
MPD	Media Presentation Description
MS	Mobile Station
MSE	Mean Squared Error

MSS	Microsoft Smooth Streaming
MVC	Multi-view video coding
NPA	Network Profiling Algorithm
NRA	Network Ranking Algorithm
NRQOVD	Network Reputation-based Quality Optimization of Video Delivery
NSS	Network Switching Subsystem
NUA	Network-side User Agent
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple Access
OMQM	Ontology for MPTPCP's QoS-aware Mechanisms
ORP	Optimal retransmission policy
OS	Operation System
OTSO	On-the-spot offloading
PCC	Policy and Charging Control
PCRF	Policy and Charging Rules Function
PDN	Packet Data Networks
PQEM	Path quality estimation model
PSM	Personalized similarity measure
RAN	Radio access network
RANS	Reputation-oriented Access Network Selection
RAT	Radio Access Technologies
RC	Reputation Claims

RDO	Reputation-based Data Offloading
REP	Random Early Probation
RNC	Radio Network Controller
RNS	Reputation-based Network Selection
RNSM	Reputation-based Network Selection Mechanism
RRA	Radio Resource Agent
RRM	Radio resource management
RS	Reputation Sources
RSNS	Received signal strength-based network-selection
RSS	Received Signal Strength
RSSI	Received signal strength indicator
RT	Reputation Targets
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
RTT	Round-trip time
RW	Random weighting
SA	Service Agent
SAE	System Architecture Evolution
SALOME	Situation And LOcation aware MiddlEware
SAP	Service Abstraction Points
SAW	Simple Additive Weighting
SCTP	Stream Control Transmission Protocol

SD	Standard-Definition
SDN	Software-defined network
SGRA	Strengthen Gray Relative Analysis
SINR	Signal to Interference-plus-Noise Ratio
SIR	Signal to Interference Ration
SLA	Service level agreement
SNR	Signal to Noise Ratio
SVC	Scalable video coding
TBNS	Traffic balanced-based network-selection
TCP	Transport Control Protocol
TDD	Time Division Duplexing
TDMA	Time Division Multiple Access
TFT	Tit For Tat
TOPSIS	Technique for Order Preference by Similarity to Ideal Solution
TVM	Trust value measure
UA	User Agent
UDP	User Datagram Protocol
UE	User Equipment
UEP	Unequal error protection
UHD	Ultra High Definition
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network

VHD	Vertical Handoff Decision
VHO	Vertical Handover
VHME	Vertical Handover Management Engine
VoD	Video on Demand
VR	Virtual Reality
WCDMA	Wideband Code Division Multiple Access
WCNC	Wireless Communications and Networking Conference
WPAN	Wireless Personal Area Networks
WS	White Spaces
ANDSF	Access Network Discovery and Selection Function
BER	Bit Error Rate
GRA	Grey Relational Analysis
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
MADM	Multiple Attribute Decision Making
MOS	Mean Opinion Score
PSNR	Peak Signal-to-Noise Ratio
QoE	Quality of Experience
QoS	Quality of Service
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Networks

WWAN Wireless Wide Area Networks

Abstract

Ting Bi

Network Reputation-based Quality Optimization of Video Delivery in Heterogeneous Networks

The mass-market adoption of high-end mobile devices and increasing amount of video traffic has led the mobile operators to adopt various solutions to help them cope with the explosion of mobile broadband data traffic, while ensuring high Quality of Service (QoS) levels to their services. Deploying small-cell base stations within the existing macro-cellular networks and offloading traffic from the large macro-cells to the small cells is seen as a promising solution to increase capacity and improve network performance at low cost. Parallel use of diverse technologies is also employed. The result is a heterogeneous network environment (HetNets), part of the next generation network deployments.

In this context, this thesis makes a step forward towards the "Always Best Experience" paradigm, which considers mobile users seamlessly roaming in the HetNets environment. Supporting ubiquitous connectivity and enabling very good quality of rich mobile services anywhere and anytime is highly challenging, mostly due to the heterogeneity of the selection criteria, such as: application requirements (e.g., voice, video, data, etc.); different device types and with various capabilities (e.g., smartphones, netbooks, laptops, etc.); multiple overlapping networks using diverse technologies (e.g., Wireless Local Area Networks (IEEE 802.11), Cellular Networks Long Term Evolution (LTE), etc.) and different user preferences. In fact, the mobile users are facing a complex decision when they need to dynamically select the best value network to connect to in order to get the "Always Best Experience".

This thesis presents three major contributions to solve the problem described above: 1) The Location-based Network Prediction mechanism in heterogeneous wireless networks (LNP) provides a shortlist of best available networks to the mobile user based on his location, history record and routing plan; 2) Reputation-oriented Access Network Selection mechanism (RANS) selects the best reputation network from the available networks for the mobile user based on the best trade-off between QoS, energy consumptions and monetary cost. The network reputation is defined based on previous user-network interaction, and consequent user experience with the network. 3) Network Reputation-based Quality Optimization of Video Delivery in heterogeneous networks (NRQOVD) makes use of a reputation mechanism to enhance the video content quality via multipath delivery or delivery adaptation.

CHAPTER 1: Introduction

1.1 Research Motivation

There is an increasing desire to enable the “always best connected” paradigm given today’s heterogeneous wireless network environment. However, supporting such a connectivity goal and enabling very good quality of rich media mobile services anywhere and anytime is very difficult, mostly due to the communication system complexity and diversity of technologies.

According to the Cisco Global Mobile Data Traffic Forecast published in Dec 2016 [1], mobile data traffic has grown 4,000-fold over the past 10 years, from the level it was when it accounted for less than 10 gigabytes per month in 2000. It is forecasted that mobile data traffic will grow at a compound annual growth rate (CAGR) of 53 percent from 2015 to 2020, and 75 percent of the world’s mobile data traffic will be video by 2020 [1]. In terms of video delivery over wireless networks, there are three major access network technologies which enable this: broadband, cellular and broadcast. Broadband wireless networks are mostly represented by the IEEE 802.11 family (i.e. including the best known protocols, 802.11 a/g/b/n and the recent IEEE 802.11 ac [2]) and offer high data delivery rates, but have limited range. Cellular networks, best known for their Global System for Mobile Communications (GSM) [3] and Universal Mobile Telecommunications System (UMTS) [4] technologies, support wider signal coverage areas, but lower average data rates when compared to the IEEE 802.11 family. The latest Long-Term-Evolution-Advanced (LTE-A) [5] standard provides support for higher data rates which could reach up to 3 Gbps downlink and 1.5 Gbps uplink [5]. Broadcast networks are mostly used for distributing video in downlink mode to a large number of users and include standards from the Digital Video Broadcasting (DVB) [6] family, in which the most recent and highly efficient is DVB-T2 [7].

The “optimally connected anywhere, anytime” vision was first introduced in June 2003 by ITU in Recommendation ITU-R M.1645 [8]. The aim was to connect different radio access networks via flexible core networks in order to provide seamless, transparent and Quality of Service (QoS) enabled connectivity to the mobile users.

Moreover, in order to ensure QoS and to facilitate handover between different Radio Access Technologies (RAT), the IEEE 802.21 Media Independent Handover (MIH) Working Group [9] (Jan 2009) considered the interoperability aspects between heterogeneous networks and developed the standard referred to as the IEEE 802.21. This standard enables the handover optimization by providing a media-independent framework and associated services. However, the standard offers support for the handover process only without defining a network selection algorithm, which is a major part of the handover process.

In this context, the "Always Best Connected" (ABC) vision emphasizes the scenario of a variety of RATs that work together in order to offer global wireless connectivity. The end users benefit from an optimal service delivery via the most suitable wireless network that satisfies their interests [10] among all the available wireless networks, which realize the ABC and enhance the user of experience. This could be advertised as an “Always Best Experience” (ABE) paradigm.

Furthermore, recent advances in both wireless technologies and mobile devices, fuelled by increased user interest, have driven the latest development of mobile 3D video services. In the last decade, 3D video has been introduced to home through 3DTV, 3D gaming and 3D movies. Alternative codec solutions for the 3D video have been developed including: i) two-view stereo video coding [11], ii) video and depth coding [12] and iii) multi-view video coding [13]. 3D video introduces additional challenge mostly due to it increased bandwidth requirements.

Currently, most mobile devices have access to different networks as they are equipped with multiple radio interfaces. One of the possibilities to increase the bandwidth offered to applications is to make use of more than one such interface at the same time. By implementing the MPTCP protocol [14], the mobile devices can concurrently utilize multiple interfaces. Regarding 3D video delivery, the 3D video stream can be decomposed into different components according to the coding methods employed for the 3D video [15]: left and right views, in the two-view stereo video coding; video and depth streams in video and depth coding; and several views plus depth information in MVC. These different components could be transmitted over different flows, potentially using different interfaces and therefore communication technologies.

1.2 Problem Statement

In current heterogeneous wireless network environments, there is an increasing number of mobile users requesting mainly video-based applications. However, most of the rich-media applications require high data rates, low delays and low loss rates as basic requirements, in order to offer high levels of users Quality of Experience (QoE) to achieve the “Always Best Experience” paradigm.

Because of user mobility within this heterogeneous environment, they regularly require network selection and handover procedures in order to maintain their seamless connectivity to the Internet. Additionally to supporting the best user perceived video quality level for their multimedia-based applications, the selection of the most appropriate network in terms of performance is required. Choosing the right network is not trivial as network characteristics vary widely, not only in time, but also depending on user location within each network. Predicting the performance of candidate networks is very difficult based on a single user device gathering data, a fact that makes the selection of the network with best performance challenging.

As Fig. 1.2 shows, mobile and Wi-Fi data will account for almost 66 percent of total IP traffic by 2019 [1], which almost two times as fixed traffic. As the ratio of mobile data to Wi-Fi data is almost ¼, it is more than obvious that mobile devices will take

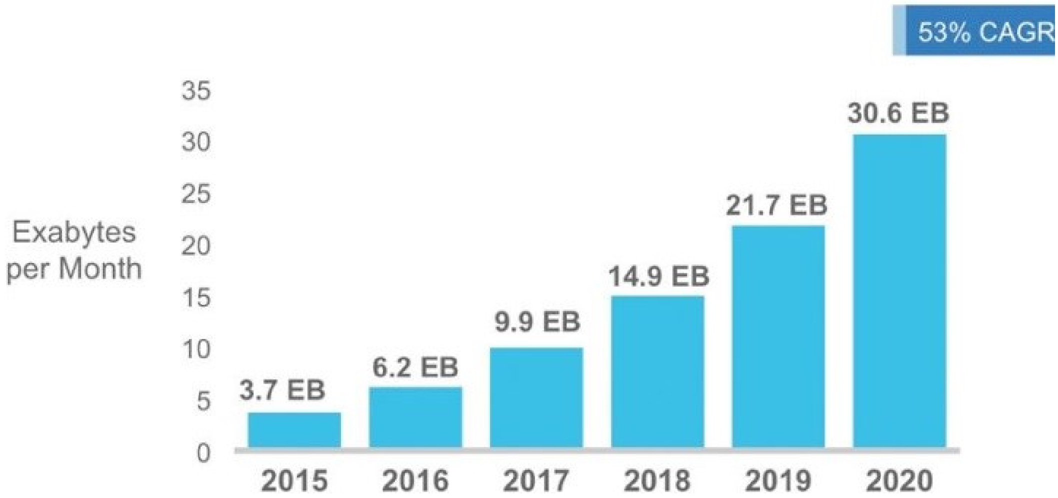

Figure 1-1 Cisco Forecasts 30.6 Exabytes per Month of Mobile Data Traffic by 2020 [1]

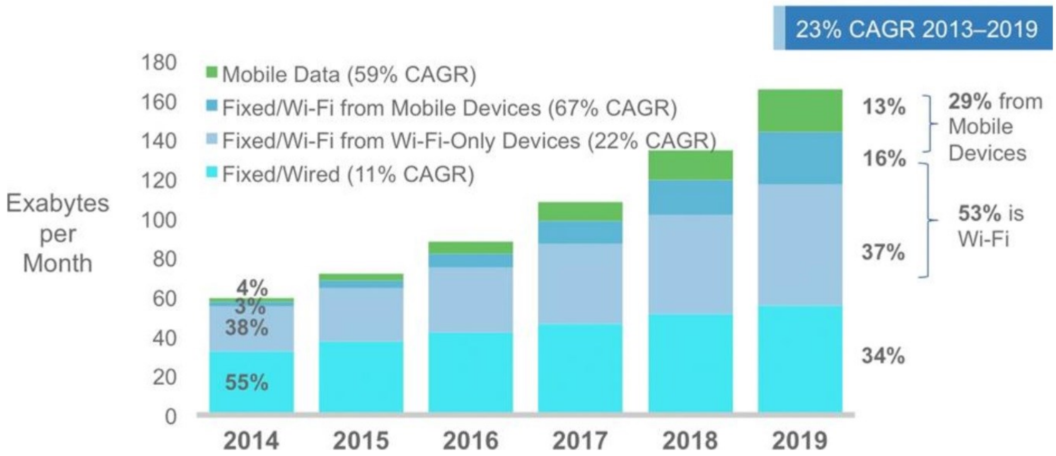

Figure 1-2 IP Traffic by Access Technology [1]

advantage of the Wi-Fi offering. According to a white paper from Mobidia [16], a typical Android user uses approximately 2.9 GB of data per month, where 66% is consumed over Wi-Fi and 34% over cellular networks. Moreover, a typical iPhone user uses nearly 4 GB of data per month where the 82% is transferred over Wi-Fi and only 18% over cellular networks [16]. Hence, for a mobile user and a mobile device, selecting between the cellular and Wi-Fi networks is a major issue (i.e. labeled network selection). Most mobile devices are using the on-the-spot offloading (OTSO) scheme as default [17], which chooses the

Wi-Fi network to offload its data traffic from cellular network whenever possible. But this OTSO scheme is not always desirable, especially when the Wi-Fi network is highly loaded and the deadline is tight [18]. Google project Fi[1] is one of the solutions to enhance the network by using multiple Wi-Fi and cellular networks with smart network selection.

Moreover, achieving ABC combined with network selection decision results in a very complex problem, with the main challenge trading-off different decision criteria (e.g., service class type, user preferences, the mobile device type, battery level, network type, network conditions, time of day, price, etc.). This becomes further complicated by combination of static and dynamic information involved, accuracy of the information available, and effort in collecting all of this information with a battery, memory, and processor limited device.

Furthermore, for 3D video or other new types of video delivery on mobile devices, the limited wireless bandwidth is one of the critical challenges, especially as the 3D content requires higher bandwidth for transmission than the conventional 2D video. In general, a single 3D video stream consists of both color and depth information. This results in the 3D video delivery service requiring higher bandwidth than necessary for the traditional 2D video stream. The emerging LTE-A and 802.11ac standards provide significant improvements in terms of bandwidth and are very good for delivering 3D video sequences. Bandwidth resource allocation for 2D video streams in heterogeneous networks has been extensively studied [19]. However, additional work is needed to propose efficient delivery and scheduling schemes for 3D video.

1.3 Solution Overview and Contribution

To address the above challenges, this thesis proposes a **novel reputation-based mechanism for quality optimization of video delivery in heterogeneous wireless environments**. The proposed solution considers a location-aware network selection

[1] Google Project Fi – https://fi.google.com/about/?u=0

mechanism, multi-user performance information-based network reputation scheme and a novel 3D video delivery scheme. First, the location-aware network selection mechanism provides the available networks for the mobile user based on his location information and historical performance records and routing plan, which will keep achieving the ABC. Then, for a certain mobile user, the reputation-based network selection uses multi-user involvement in the decision process to select the most appropriate network for video delivery to achieve the ABE. A reputation-based network selection mechanism that computes a network reputation value based on the performance information gathered from the currently connected users, such as user location, throughput, signal strength, delay and the network monetary cost is then introduced for 3D video delivery. A novel video delivery scheme makes use of multipath transmissions to transmit different components of stereoscopic 3D video over different wireless links based on the reputation-based mechanism.

In this context, three main contributions of this work are identified as follows:

1) A ***Location-aware Network Selection mechanism in Heterogeneous Networks (LNS)*** is proposed, with the main contributions as follows:
    - Mobile user location information with a timestamp and available network information is regularly monitored and sent to the server. Using the mobile user's previous location information, the next step of movement is predicted.
    - Based on the existing location-related information of the network base station access point and uploaded mobile user's location information together with the available network information, the network that satisfies the requirements of data offloading along the user's routing path is selected from the available networks as the proposed handover target.
    - Simulation results show that the proposed LNS supports the "Always Best Connected" paradigm and in comparison with two state of the art solutions, LNS

can achieve a decrease of up to 60% in the number of handovers and a reduction of up to 59% in the download time.

2) A novel ***Reputation-oriented Access Network Selection mechanism (RANS)*** for Improved Video Delivery Quality over heterogeneous wireless environments is introduced, with the following main contributions:
   - Network performance is regularly monitored and evaluated with the aid of the IEEE 802.21 MIH standard mechanism, which offers support for performance information gathering from the currently connected users over different areas within each network.
   - Based on the existing network performance-related information and mobile user location and speed, the network that offers the best support for video delivery along the user's path is selected as the target network and the handover is triggered.
   - The information is aggregated and disseminated to other mobile users, which can make an informed quality-oriented decision when selecting the candidate network for handover.
   - Simulation results show that RANS supports the "Always Best Experienced" paradigm. Specifically, RANS achieves a considerable reduction in the loss rate during the video streaming. In terms of the average ITU G.1070 quality, RANS scores with up to 144% higher values in compared to two state of the art solutions. Additionally, in comparison with a cellular-only scheme, RANS reduces the energy consumption by 15.8% and the monetary cost by 26.67%.
3) A ***Network Reputation-based Stereoscopic 3D Video Delivery (NRQ-3D)*** in heterogeneous networks is proposed, with the following major contributions:
   - The proposed solution selects the best candidate networks for the smartphone using the network reputation module, which is proposed to report the network

quality based on the quality of service-related parameters (i.e. throughput, signal strength, delay, and loss) and price aspects.

- The Internet Engineering Task Force (IETF) Multipath TCP (MPTCP) protocol is used for delivering the 3D video content to the mobile devices due to the higher throughput provided. Different 3D video components (i.e. color stream and depth stream) are delivered via separate sub-MPTCP flows and synchronized at the receiver.
- Simulation results show significant QoS parameter values and enhanced user perceived quality benefits when using the proposed NRQ-3D in comparison with another state of the art approach: average throughput and two video quality metric values are with up to 5.5%, 20.4% and 53.4% higher, respectively.

1.4 Thesis Structure

This thesis is structured in eight chapters as follows:

- Chapter 1 introduces the research motivation, identifies the problem and presents an overview of the main contributions.
- Chapter 2 explores the technical background for the work presented in this thesis.
- Chapter 3 investigates some of the most important state-of-the-art related works presented in the literature and presents a comprehensive survey of the following topics: network selection strategy in HetNets, reputation mechanisms, multipath transmission, load balancing, and 3D video delivery.
- Chapter 4 introduces the proposed system architecture together with the handover mechanism and data structure.
- Chapter 5 presents the proposed LNS including the algorithms, simulation setups and results.

- Chapter 6 describes the proposed RANS including the reputation mechanism, algorithms, simulation setups and results.
- Chapter 7 presents the proposed NRQ-3D with the 3D video delivery scheme, algorithms, simulation setups and results.
- Chapter 8 draws the conclusions of the thesis and shows possible future work directions.

CHAPTER 2: Technical Background

Abstract

This chapter introduces the technical background of the work presented in this thesis. The chapter provides an overview of the existing background technologies that are particularly relevant for wireless networks and video delivery protocols. Key wireless network technologies related to the work presented in this thesis will be overviewed first. Then, the network selection concept and the reputation mechanism are further described. Furthermore, the main protocols and industry solutions for video delivery relevant to this work are introduced. The chapter is concluded with a short summary.

2.1 Heterogeneous Wireless Network Technologies

2.1.1 Overview

In the context of heterogeneous wireless networks technologies, the existing wireless networks could be grouped into four major categories based on their range: Wireless Wide Area Networks (WWAN), Wireless Metropolitan Area Networks (WMAN), Wireless Local Area Networks (WLAN) and Wireless Personal Area Networks (WPAN). WLAN networks are mostly represented by the IEEE 802.11 family and they offer high data delivery rates, but they have limited transmission range. WWAN provide coverage over extremely large areas, best known for their Global System for Mobile Communications (GSM), General Packet Radio Service (GPRS), Universal Mobile Telecommunications System (UMTS), Enhanced Data rates for GSM Evolution (EDGE) technologies, and the latest Long-Term-Evolution (LTE) protocol. WMANs similar as WWAN also provide long coverage range. Usually, the IEEE 802.16 WiMAX is the best-known technology appertaining to this category. WPANs are the smallest wireless networks used to connect various peripheral devices centered around an individual person's workspace. The wireless technologies used for WPANs include Bluetooth, ZigBee, and Infrared Data Association [20].This thesis employs solutions based on both WLAN and WWAN technologies.

2.1.2 IEEE Standard Wireless Networks

The Institute of Electrical and Electronics Engineers (IEEE) Standards Association Standards Board approved and published the IEEE 802 [21] (IEEE Standard for Local and Metropolitan Area Networks), which is a protocol family for both WLAN and WMAN technologies and also maintained by the IEEE 802 LAN/MAN Standards Committee. In this thesis we focus on a representative here and network range show as follows:

Table 2-1 Features of 6 Key IEEE 802.11 Technologies

IEEE 802.11 standard versions	Frequency (GHz)	Band-width (MHz)	Data Rate (Mbps)	MIMO Streams	Modulation	Coverage (meters)	
						Indoor	Outdoor
a	5	20	6, 9, 12, 18, 24, 36, 48, 54	N/A	OFDM	35	120
b	2.4	22	1, 2, 5.5, 11	N/A	DSSS	35	140
g	2.4	20	6, 9, 12, 18, 24, 36, 48, 54	N/A	OFDM, DSSS	38	140
n	2.4, 5	20, 40	≤150	4	OFDM	70	250
ac	5	20, 40, 80, 160	≤866.7	8	OFDM	35	-
ad	60	2, 160	≤6912	N/A	OFDM	60	100

2.1.2.1 WLAN: IEEE 802.11

WLAN networks normally referred to as WiFi (Wireless Fidelity) are represented by the IEEE 802.11 standard. The original version of the standard was introduced in 1997 providing 1 or 2 Mbps bit rates and operating in the 2.4 GHz band [22]. Two standards released in 1999 as part of the IEEE 802.11 family are 802.11a [23] and 802.11b [24]. As table 2.1 shows, IEEE 802.11a is operating in the 5 GHz band increasing the data rate up to 54 Mbps making use of the Orthogonal Frequency Division Multiplexing (OFDM) modulation. The standard provides eight theoretical data rates: 6, 9, 12, 18, 24, 36, 48, and 54Mbps. The IEEE 802.11b is operating in 2.4 GHz band increasing the data rate up to 11 Mbps and making use of the Direct-Sequence Spread Spectrum (DSSS) modulation. The standard provides four theoretical data rates: 1, 2, 5.5, and 11Mbps. In order to provide higher data rates, a new standard IEEE 802.11g [25] was released in 2003. The standard extended IEEE 802.11b, operating in the 2.4 GHz band, but offering a maximum data rate of up to 54 Mbps using both OFDM and DSSS modulations. IEEE 802.11g provides multiple data rates, such as: 1, 2, 5.5, 6, 9, 11, 12, 18, 24, 36, 48, and 54Mbps. The use of Multiple Input Multiple Output (MIMO) was introduced in the next version of the standard IEEE 802.11n [26] in 2009. Both the 2.4 GHz band and the 5 GHz band are supported in IEEE 802.11n, and the maximum theoretical data rate goes up to 150 Mbps. The latest version of the standard, IEEE 802.11ac [27] was standardized in 2013. IEEE 802.11ac is operating in the 5 GHz band and the maximum theoretical data rate it can reach is 866.7 Mbps. IEEE 802.11ad [28] was standardized in 2012. IEEE 802.11ad is operating in the 60 GHz band which enables support for multi-gigabit transmission speeds between wireless devices and the maximum theoretical data rate it can reach is 7 Gbps.

2.1.2.2 WMAN: IEEE 802.16

A representative of WMAN is the Worldwide Interoperability for Microwave Access (WiMAX) [29] standardized as IEEE 802.16. The original IEEE 802.16 standard was published in 2001, offering point to multipoint capability in the 10–66 GHz band.

New versions of the IEEE 802.16 standard appeared later. Based on the original version, the IEEE 802.16d standard referred to as Fixed WiMAX delivered a system profile for the 2–11 GHz band, providing broadband wireless connectivity to fixed users with data rates up to 75Mbps. IEEE 802.16e referred to as Mobile WiMAX uses the Orthogonal Frequency-Division Multiple Access (OFDMA) scheme, offering mobility support and data rates up to 30Mbps. IEEE 802.16m is the latest version, which achieves data rates up to 1Gbps for stationary usage and 100Mbps for mobile users.

2.1.3 WWAN: Mobile and Cellular Networks

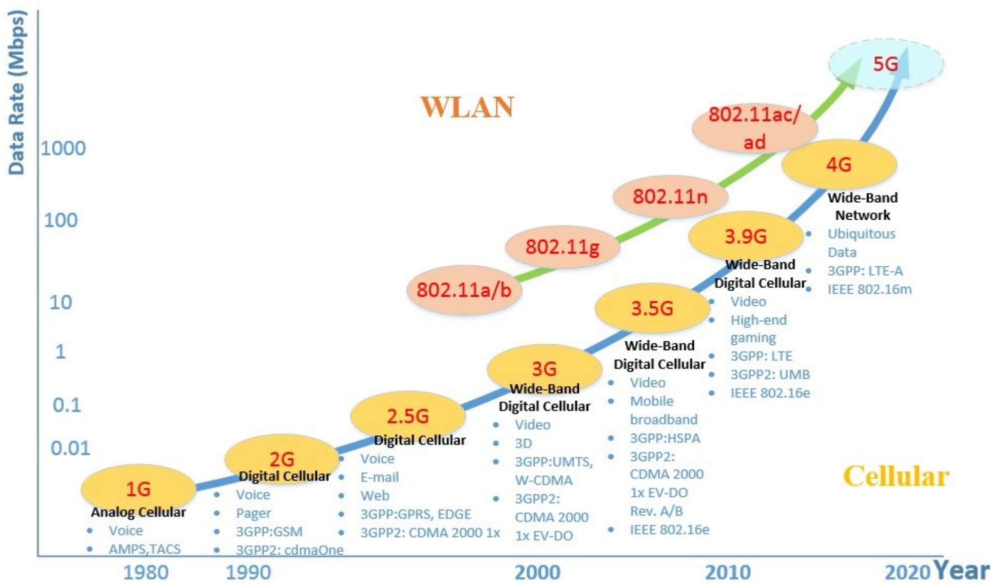

Figure 2-1 Wireless Networks Roadmap

Fig. 2.1 illustrates the evolution of the wireless networks technologies on both WLAN and cellular communication. For the cellular communication area, from the first generation (1G) to the fourth generation (4G) mobile networking and finally towards the fifth generation (5G) network together by interworking with WLAN networks.

From the 1980s, the first Analog Cellular network system (referred to as 1G), Advanced Mobile Phone System (AMPS) operating within the 800 to 900 MHz frequency ranges and widely was introduced to North America, Pakistan and Australia [30].

With the innovation on the digital transmission for speech service, the second generation (2G) widely deployed on the whole world from the 1990s. The most famous and most used system was Global System for Mobile Communication (GSM), which still accounts for over 80% of all 2G subscribers over the world [31]. The GSM network is decentralized and consists of three separate subsystems [32]: Mobile Station (MS), Base Station Subsystem (BSS), and Network Switching Subsystem (NSS). It is operating in the 900MHz frequency range and based on Time Division Multiple Access (TDMA). Some other solutions for 2G are also deployed: Code Division Multiple Access one (cdmaOne) was published by Qualcomm in 1995 [33].

The 3rd Generation Partnership Project (3GPP) was founded by the European Telecommunications Standards Institute (ETSI) [32]. This project aims to make a globally applicable 3rd generation mobile network (3G) based on GSM. By collaboration between different telecommunication research institutes and industry partners, dozens of protocols and standards were released by 3GPP from 1996 till now, increasing 2G, 2.5G, 3G, 3.5G, 3.9G and 4G technologies [34]. The most important 3G cellular system is Universal Mobile Telecommunications Systems (UMTS) which uses Wideband Code Division Multiple Access (WCDMA) for the air interface. The UMTS architecture is composed of three main domains [32]: User Equipment (UE), UMTS Terrestrial Radio Access Network (UTRAN), and the Core Network (CN). There is also another project named the 3rd Generation Partnership Project 2 (3GPP2) which released the 3G technology CDMA2000 with various versions.

Progressing from 3G towards 4G, The Long Term Evolution (LTE)/LTE Advanced (LTE-A) were released by the 3GPP working group. The latest LTE technology defined by the 3GPP provides support for higher data rates which could reach up to 3 Gbps downlink, 1.5 Gbps uplink [5] and less than 10 ms of radio access network (RAN) round-trip time (RTT). In LTE, the downlink uses the Orthogonal Frequency-Division Multiple Access (OFDMA) scheme and the uplink uses the Single Carrier FDMA (SC-FDMA) transmission schemes. Both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) are supported in LTE. The scalable carrier bandwidths: 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz are standardized in LTE. The LTE system architecture consists of three main domains [32]: User Equipment (UE), Evolved-UTRAN (E-UTRAN), and Evolved Packet Core (EPC). EPC consists of three main entities: the Mobility Management Entity (MME), Serving Gateway (SGW), and Packet Data Network Gateway (PGW).

2.2 Network Selection and Reputation Mechanisms

2.2.1 Overview

The "optimally connected anywhere, anytime" vision was first introduced in June 2003 by ITU in Recommendation ITU-R M.1645 [35]. The aim was to connect different radio access networks (RAN) via flexible core networks in order to provide seamless, transparent and Quality of Service (QoS)-enabled connectivity to the mobile users. In this context, the mobile users will be facing a complex decision when selecting from a number of RANs that differ in technology, coverage, pricing scheme, bandwidth, latency, etc., belonging to the same or different service providers. Choosing the right network is not trivial as network characteristics may vary widely, not only in time, but also depending on the user profile and preferences, such as the applications requirements (e.g., voice, video, data, etc.), device types (e.g., smart phones, notebooks, laptops, etc.) with various capabilities, location, device battery lifetime, credit balance, etc. Ideally, the user mobile device should automatically detect and select the best available network dynamically dependent on current user and service needs. However existing multi-user multi-terminal multi-network multi-application heterogeneous environments requires developing new

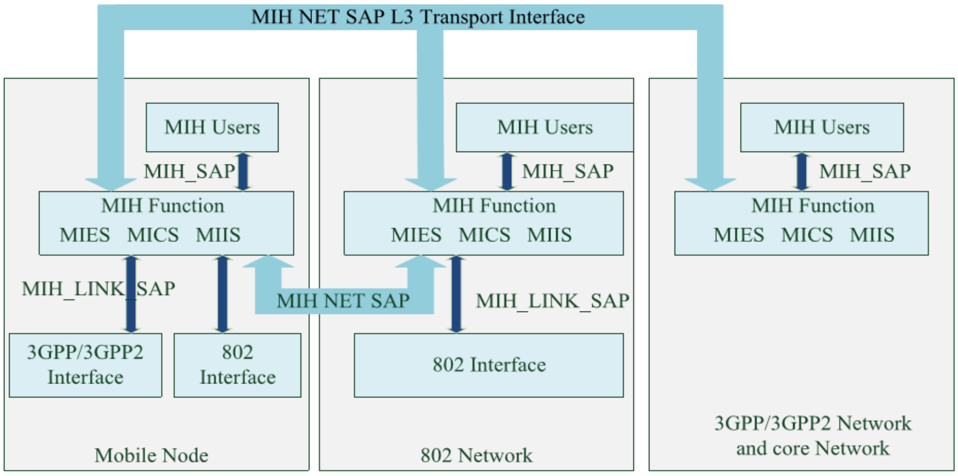

Figure 2-2 MIH General Model [108]

technologies and/or standards that can offer such a dynamic automatic network selection mechanism.

2.2.2 Network Selection Standards

Both IEEE and 3GPP have provided their own network selection-related standards or solutions. The two main options are introduced next:

2.2.2.1 IEEE 802.21

In order to improve the Quality of Service (QoS) and enable seamless handover between heterogeneous wireless networks, IEEE has developed the 802.21 Media Independent Handover (MIH) standard [36]. This standard enables the optimization of handover between heterogeneous IEEE 802 networks and facilitates handover between IEEE 802 and cellular networks by providing a media-independent framework and associated services. The MIH framework defines a cross-layer MIH function (MIHF) as a logical component between the network layer and link layer. MIHF provides three independent services:

- Media Independent Event Service (MIES) – triggered or predicted when changes occur at the physical, data link, and logical link layer (i.e., link parameters change, new networks available, interrupted/established session);
- Media Independent Command Service (MICS) – enables the higher layers to control the link layer by re-configuring or selecting an appropriate link;
- Media Independent Information Service (MIIS) – provides an interface for the handover policy in order to gather information about available networks in range while using the currently active access network.

The MIH framework also defines the MIH Information Server, which uses MIES via the MIHF interface to exchange information about various networks and mobile nodes. The MIH Information Server itself does not provide any network selection algorithm; however, it offers the support for the mobile nodes to perform network selection and seamless handover.

IEEE 802.21 MIH provides mechanisms for gathering and exchanging information between various candidate networks, the MIH Information Server and the Mobile Node (MN). As Fig 2.2 shows, each of the MIH-enabled entities contains a cross-layer MIHF. This function provides Service Abstraction Points (SAP) acting as an abstract interface between a service provider and a user entity. The higher-layer user entities employ the MIH-SAP to control or monitor the link-layer entity, and MIHF uses the MIH-LINK-SAP as an interface together with the link layer to translate the comment received from the MIH-SAP. The remote MIHF entities use the MIH-NET-SAP interface to exchange the information with the MIHF.

However, IEEE 802.21 only facilitates handover and does not specify the network selection algorithm, which means it allows researchers propose innovative solutions. As a result, many diverse proprietary algorithms exist.

2.2.2.2 Access Network Discovery and Selection Function

From Release 8 to Release 12, 3GPP proposed and defined the Access Network Discovery and Selection Function (ANDSF) [37] as a key network element for access network discovery and selection in the Evolved Packet Core (EPC) of System Architecture Evolution (SAE) in 3GPP 4G/beyond 4G [38].

ANDSF enables interworking between the 3GPP network and non-3GPP access networks (e.g., Wi-Fi, WiMAX, and CDMA2000). The architecture for ANDSF is a simple client-server topology. UE as the client contacts an ANDSF server via the S14 interface to request policy information and acts on the policy from the ANDSF server. Two categories of rule-based network selection policies are defined in ANDSF: Inter-System Mobility Policy (ISMP), which guides the selection decision for devices with one active access network, and the Inter-System Routing Policy (ISRP), which directs the distribution of traffic for devices with multiple simultaneous active access networks. ANDSF also provides "A Discovery Information" service to UE, which sends the handover relevant information about the neighboring available access networks to assist the UE in handover. But, ANDSF requires that UE contains the Open Mobile Alliance Device Management (OMA-DM) [39] Management Object (MO) to performs the synchronization [37].

2.2.3 Reputation Mechanism for Network Selection

In order to solve the network selection problem, two major approach types are proposed in both academia and industry: centralized and decentralized. In both centralized and decentralized approaches, different information is required in different forms and with various accuracy levels from operator, user device and service provider. This information presents a statement of the network performance and/or requirements of the application on

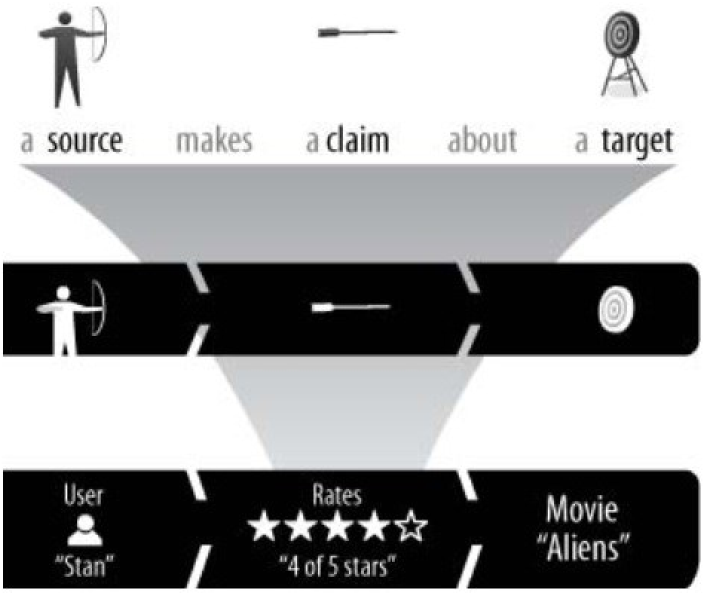

Figure 2-3 Logic of the Reputation System [40]

the UE. The network selection involves making a decision to choose one or more networks as a target based on the data of various networks. For comparison and decision making, a mechanism to associate a value to each network is needed. As reputation is the information used to make a value judgment about an object or a person [40], we consider involving reputation in this work.

In a reputation-based system, there are three major components: Reputation Sources (RS), Reputation Claims (RC), and Reputation Targets (RT). RS is an entity that has made an RC, which normally is the user in the reputation system. RC refers to the numeric (quantitative) or qualitative scores, which are determined by the ***input information*** (e.g., RT statement, RS preference). RT is an entity that is the target of RC. Fig 2.3 illustrates the simplest logic for a reputation system: an RS makes an RC about an RT. In order to gather more accurate judgments for the RT, the reputation system needs to combine or aggregate multiple RCs from different RSs, and this implies requirements on the ***scalability or normalization*** of the various numeric (quantitative) or qualitative scores in the RC and the ***combination or aggregation method*** for multiple RCs to generate the final RC (reputation).

In this context, in order to involve the reputation system into the network selection function, there are three issues need to consider to building the reputation system:

1) ***Input Information -*** the input information that may be used in the network selection process, like network ***QoS*** *parameter*, *user preferences*, *device information*, and *application requirements*.

2) ***Scalability or Normalization –*** as the input information presents different ranges and units of measurement, in order to bring all of them into dimensionless units within [0, 1] for instance and make them comparable, a ***utility function*** is designed to make the input information be scalable and normalized.

3) ***Combination or Aggregation Method –*** diverse solutions are possible for combination of data. ***Multiple Attribute Decision Making (MADM)*** is an algorithmic way of suitably realizing network selection [41].

2.2.3.1 Input Information

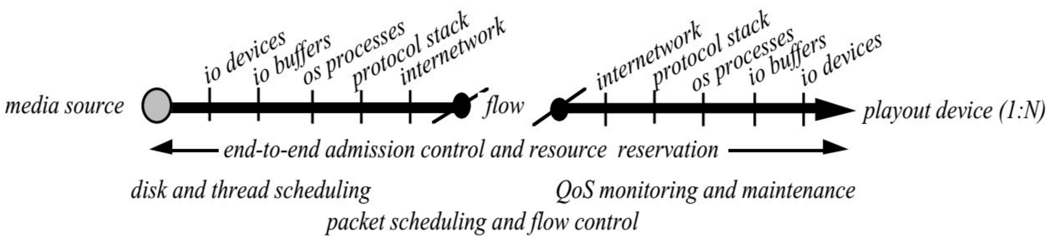

Figure 2-4 End-to-End QoS Scenario [43]

Network parameters were first introduced in the context of **Quality of Service (QoS)** by ITU-T E.800 in 1994 [42] in which QoS was described as “the totality of characteristics of a telecommunications service that bear on its ability to satisfy stated and implied needs of the user of the service". Fig. 2.4 illustrated an end to end QoS scenario that shows how distributed multimedia systems perform active monitoring and maintenance of the delivered QoS by various components working in unison [43]. As Fig. 2.4 shows, QoS comprises of both network performance and non-network-related performance components. Regarding the network performance, network QoS parameters refer to a set of metrics which include delay, jitter, Bit Error Rate (BER), packet loss and throughput.

Delay

Generally, the delay in multimedia communication comprises application delay and end-to-end delay. Application delay represents the time difference between the data arrival time and drain time of the media content. Most of the time the delay is dependent on the hardware/software performance, which is influenced by the CPU/GPU performance and multimedia encoding/decoding solutions. End-to-end delay refers to the duration a packet travels across a network from the source to the destination. End-to-end delay is a summation of the four components as shown in eq (2.1): processing delay, queueing delay, transmission delay, and propagation delay.

$$\textit{end-to-end delay} = \textit{processing delay} + \textit{queueing delay} + \textit{transmission delay} + \textit{propagation delay} \quad (2.1)$$

Where the processing delay is the summation of the time taken to process a packet to determine output link and check the bit errors, queueing delay refers to the time waiting on the output link for transmission, transmission delay is the time taken to transmit a packet on a link, and propagation delay. refers to the time to deliver a bit over the transmission medium.

Jitter

Jitter is the difference between the current packet delay and the delay of the reference packet which represents the delay variation caused by network condition dynamics. In multimedia transmission, high jitter may result in distorted or jerky videos, which seriously affect user perceived quality. To avoid being affected by high jitter, a buffer is implemented at the receiver for video applications.

BER

BER is a key characteristic of the network channel condition, and counts the number of bit errors per unit time. BER is affected by the channel noise, interference, distortion, wireless multipath fading or other transmission related reasons.

Packet loss

Packet loss is the ratio of the packet dropped to the number of the packet transmitted during the transmission through the network. During a transmission session, a packet might be lost due to various factors such as: network congestion, buffer overflow, network connection failure, channel contentions, and collisions. For multimedia content delivery, packet loss can greatly impair user perceived multimedia quality.

Throughput

Throughput is the rate of successful information delivery over a communication channel. Throughput is usually measured in bits per second or packets per second/time slots. Throughput is one of the most important parameters that can determine the network performance at the user side. Low throughput can cause long transmission times and low user perceived quality, especially for real-time services.

Table 2-2 Y.1541 IP Network Performance Requirements for Different Applications [44]

Parameters	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5
Delay (IPTD)	100ms	400ms	100ms	400ms	1s	U
Jitter (IPDV)	50ms	50ms	U	U	U	U
Packet loss ratio (IPLR)	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	1×10^{-3}	U
Applications	Real-time, Highly interactive, Delay variation sensitive (VoIP, video conference)	Real-time, Interactive, Delay variation sensitive (VoIP)	Transaction Data, highly interactive (signaling)	Transaction data, interactive	Low loss only (short transactions, bulk data)	Traditional application of default IP networks
"U" means "unspecified" or "unbounded".						

Other three input information also affect the reputation of a system.

a) **Application requirements** normally include information about the requirements of the application needed in order to provide good quality of service to the end-user: delay, jitter, packet loss, required throughput, BER, etc. Different applications have different demands on QoS, and the International Telecommunication Union (ITU) released recommendation Y.1541 [44] on network performance objectives for IP-based services. To help in this regard, Table 2.2 illustrates the network performance requirements for different applications as then in ITU-T Rec Y.1541.

b) **Device information** includes information about the characteristics of end-users' terminal device, like network technology and interfaces, screen-size and resolution, operation system, battery lifetime, location information, timestamps, etc.

c) **User preferences** include information related to the end-users' classification and satisfaction, data plan ("bill pay" user, or "Pay as you go" user), remaining data balance, service expectations ("always connected" first or quality first), budget (willingness to pay the excess data), energy conservation needs, etc. In the reputation system, user preferences play an important role that sometimes is used to *weigh* the other parameters involved.

2.2.3.2 Utility function

Utility function is widely used in the choice theory to model users' motivations or behavior. In this work, utility functions are defined to describe the user satisfaction with certain QoS parameters. Some popular utility function shapes are those defined by Rakocevic et al. in [45].

As illustrated in Fig. 2.5, the utility function can be classified according to different applications into three types:

Inelastic - see Fig. 2.5 (a). Some real-time applications (e.g. VoIP, video conferencing, video telephony, telemedicine, highly secure data transactions) with strict performance requirements are inelastic with respect to the available bandwidth or other

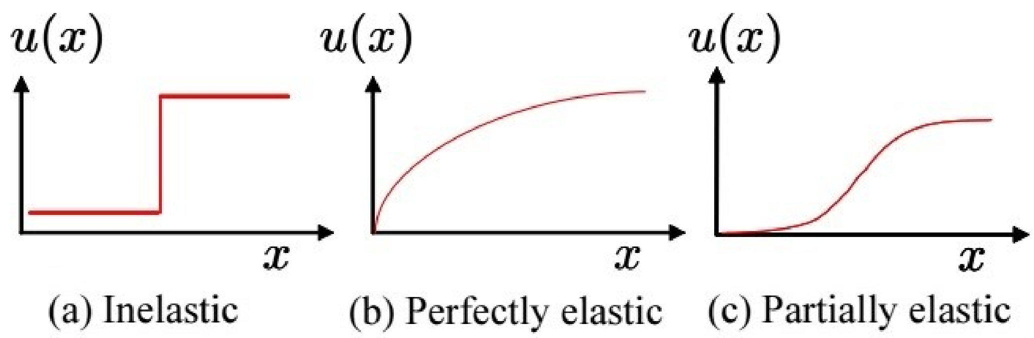

Figure 2-5 Utility Functions vs. Application Service's Elastic [45]

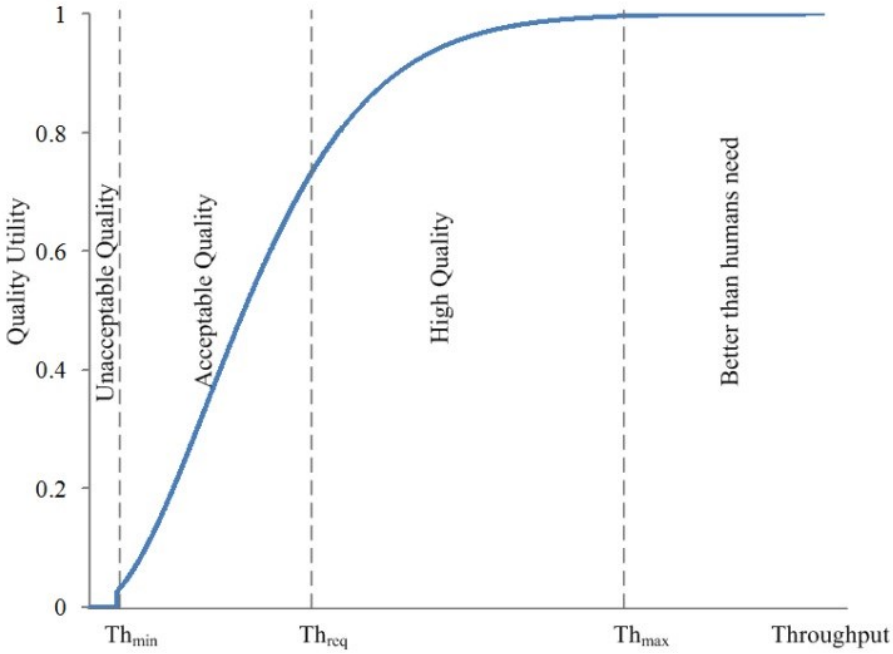

Figure 2-6 Zone-based quality utility function [50]

QoS performance factors. The form of their utility is modeled in general as a step function with only two values, either satisfied or unsatisfied (0 or 1).

Perfectly Elastic - see Fig. 2.5 (b). The non-real-time applications (i.e. Multimedia streaming) usually are perfectly elastic; those types of applications have loose response time requirements and they do not need a minimum level of bandwidth requirement. The form of their utility function is concave.

Partially Elastic - see Fig. 2.5 (c). Some real-time applications that are adaptable to the network conditions (e.g. real-time adaptive video streaming) that require a minimum level of network performance guarantee. Their utility has an "S" shape. In addition, the partially elastic utility function is suitable to model the behavior of users [46], non-real time data transfers [47] and network selection.

Fig 2.6 [50] shows an example of the partially elastic utility function used to model the relation between throughput and video quality.

Some well-known partially elastic forms of utility functions and their characteristics are illustrated in Table 2.3.

Table 2-3 Summary of Existing Utility Functions [45]

Utility Forms	Generalized Mathematical Formula	Increasing & Differentiability	Concavity	Convexity
Linear piecewise	$u(x) = \begin{cases} 0 & x < x_{min} \\ \frac{x-x_{min}}{x-x_{max}} & x_{min} < x < x_{max} \\ 1 & otherwise \end{cases}$	Yes	No	No
Logarithm	$u(x) = \ln(x)$ or $u(x) = \ln(1 + ax)$	Yes	Yes	No
Exponential	$u(x) = e^{x-m} \quad x \in [0, m]$	Yes	No	Yes
Exponential	$u(x) = 1 - e^{-ax} \quad a > 0$	Yes	Yes	No
Sigmoid	$u(x) = \frac{1}{1 + e^{a(x_m - x)}} \quad (a, x_m > 0)$	Yes	Yes	Yes
Sigmoid	$u(x) = \frac{(x/x_m)}{1 + (x/x_m)} \quad (a > 0, x_m > 2)$	Yes	Yes	Yes

As already stated, utility functions are used for normalization. As the sigmoid utility function is the only function that has the three characteristics presented in table 2.3, the sigmoid utility function is used in this work. Fig. 2.7 shows the transfer utility function used to normalize 5-level scale results to [0, 1] interval values employed in this work, as given in eq. (2.2):

$$f(x) = \frac{1}{1 + e^{-(2x-6)}} \tag{2.2}$$

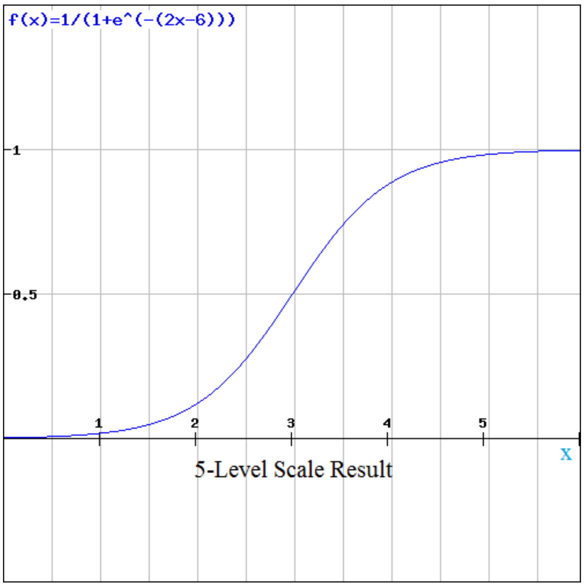

Figure 2-7 Transfer Utility Function

2.2.3.3 MADM

As previously mentioned about the network selection approach, most centralized approaches are network-centric and ruled by the network operator-controlled policy so that the decision is made by the network operator, (e.g. like the ANDSF-based solutions). For the decentralized approach, the decision is made at the user side based on the cooperation from user, service provider and network operator, (e.g. like the MIH-based solutions). The most suitable case for decentralized user-centric approaches is that the mobile users who are not only connected to the cellular network, but also connect to the WLAN network (which is not controlled by the cellular network operator) wish their device to choose the most suitable available network. Some other cases are also working for decentralized approaches, like when the user has not solely subscribed to one network, but instead subscribed to a virtual network operator who can use multiple networks or when the user has subscribed to multiple network operators.

In this work, a decentralized network selection approach is considered. Then the network selection problem becomes very complex and difficult, mostly due to the diversity of technologies involved, conflicting parameters and multiple mixes of static and dynamic aspects into the process. Four well known and widely used mathematical functions on MADM for the aggregation of multi-user multi-parameter multi-reputation claims are described below.

a) Simple Additive Weighting Method (SAW)

The SAW method [48] is one of the most widely used MADM methods in the network selection literature. SAW involves obtaining a weighted sum of the normalized form of each parameter over all candidate networks. Depending on the situation of the network, the highest/lowest score network is selected as the target network. In the case when there is a list of candidate networks to be selected from and for each network, there is a list of n parameters, then for each candidate network i a SAW score is obtained by using eq. (2.3).

$$SAW_i = \sum_{j=1}^{n} w_j r_{ij} \tag{2.3}$$

where r_{ij} is defined as the normalized performance rating of parameter j on network i, and w_j is the weight of parameter j. Usually, the greater the score value, the more preferred the candidate network.

b) Technique for Order Preference by Similarity to Ideal Solution (TOPSIS)

The TOPSIS method [48] is based on the concept that the selected candidate network has the smallest gap to the positive ideal solution and the biggest gap from the negative ideal solution. Normalization is also required in TOPSIS. For each candidate network i, a TOPSIS score is obtained by using eq. (2.4). The higher the score value, the more preferred the candidate network.

$$TOPSIS_i = \frac{D_{w,i}}{D_{w,i} + D_{b,i}} \tag{2.4}$$

where $D_{w,i}$ and $D_{b,i}$ are given in eq. (2.5) and (2.6), respectively, and they represent the Euclidian distance of a network i from the worst and from the best reference network, respectively.

$$D_{w,i} = \sqrt{\sum_{j=1}^{n} w_j^2 (r_{ij} - r_j^w)^2} \tag{2.5}$$

$$D_{b,i} = \sqrt{\sum_{j=1}^{n} w_j^2 (r_{ij} - r_j^b)^2} \tag{2.6}$$

where r_{ij} is defined as the normalized performance rating of parameter j on network i, r_{jw} and r_{jb} are the worst and the best normalized ratings of parameter j within the candidate networks, respectively.

c) Multiplicative Exponential Weighting Method (MEW)

The MEW method [48] uses multiplication for aggregating the multiple network parameters ratings. For each candidate network i a MEW score is obtained by using eq. (2.7).

$$MEW_i = \prod_{j=1}^{n} r_{ij}^{w_j} \tag{2.7}$$

where r_{ij} is defined as the normalized performance rating of parameter j on network i, and w_j is the weight of parameter j. The greater the score value the more preferred the candidate network.

d) Analytic Hierarchy Process (AHP)

The idea behind AHP is to decompose a complicated problem into a hierarchy of simple and easy to solve sub-problems. According to [49], there are four steps involved in the process:

(1) decomposition – the problem is structured as a hierarchy of multiple criteria, where the top level is the problem to be resolved, the subsequent levels are the *decision factors, and the solution alternatives are located at the lowest level;*

(2) pairwise comparison – at each level the elements within the same parent are compared to each other, the results are translated into numerical values on a scale from 1 to 9 and presented in a square matrix, referred to as the AHP matrix;

(3) local weight calculation – the weights of the decision factors are computed by calculating the eigenvector of the AHP matrix;

(4) weight synthesis – the overall weights of the decision factors are computed by multiplying the local weights from each level.

e) Grey Relational Analysis (GRA)

The *GRA* method is used to rank candidate networks and select the one which has the highest rank. There are three steps involved in the process:

(1) *normalization of data* – is performed considering three situations: larger-the-better, smaller-the-better, and nominal-the-best;

(2) *definition of the ideal sequence* – the ideal sequence will contain the upper bound, lower bound and moderate bound respectively in the three considered situations;

(3) *computing the grey relational coefficient (GRC)* as given in eq. (2.8) – the larger the GRC is, the more preferable the sequence is.

$$GRC_i = \frac{1}{\sum_{j=1}^{n} w_j \left| r_{ij} - R_j \right| + 1} \quad (2.8)$$

where r_{ij} is defined as the normalized performance rating of parameter j on network i, w_j is the weight of parameter j, and R_j represents the ideal value of parameter j.

In Fig. 2.7, the numeric score results of each ranking method are illustrated with the scenario that a varying quality weight (w_q) for a choice of different networks (e.g., WLAN1, WLAN2, WLAN3, and WLAN4) at the same quality level. The result has shown that comparison with GRA, SAW, and TOPSIS, MEW is the best way to model the network selection. The main advantages of MEW over the other methods are that it

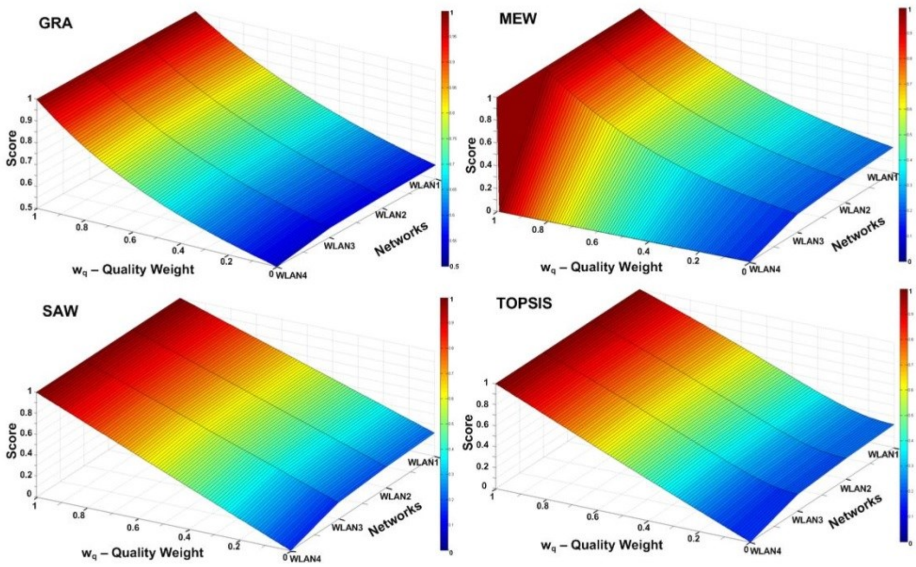

Figure 2-8 Ranking Methods Comparison with Varying Quality Weight [50]

provides a clear difference between the score results of each option and that MEW penalizes alternatives/options with poor parameters/criteria values more heavily [50].

2.3 Multimedia Content Delivery

2.3.1 Overview

The wireless networks technologies evolution, illustrated in Fig. 2.1, shows that current and future wireless environments are and will still be heterogeneous, with the multi-network access technology multi-application multi-terminal device and multi-user. As high-quality video streaming has become core to the latest rich media-services, multimedia content delivery in the heterogeneous wireless network environment is highly challenging. The main focus is to provide good QoS levels in terms of multimedia throughput, delay, and jitter.

2.3.2 Transport Layer Protocols

The multimedia streaming services are built at application layer with support from various transport layer protocols. Transport layer protocols enable end-to-end data transmission between the source and destination hosts. Two fundamental transport layer protocols were designed and are widely deployed in the network environment: Transport Control Protocol (TCP)[2] and User Datagram Protocol (UDP)[3]. TCP supports congestion control, retransmission and flow control functions to provide reliable and in-sequence data delivery. UDP does not support reliable transmission, is message-oriented and is preferred for multimedia delivery. Some other transport layer protocols will be discussed in details in next.

[2] Transmission Control Protocol - https://tools.ietf.org/html/rfc793

[3] User Datagram Protocol - https://www.ietf.org/rfc/rfc768.txt

2.3.2.1 Datagram Congestion Control Protocol

The Datagram Congestion Control Protocol (DCCP) [51] is a message-oriented transport layer protocol that provides unreliable data delivery to achieve timely transmission. It involves two optional congestion control mechanisms: a TCP-like congestion control [52] and a TCP-Friendly Rate Control (TFRC) [53]. Compared to UDP, DCCP has session and congestion control. Compared to TCP, DCCP does not provide reliability and retransmission.

2.3.2.2 Stream Control Transmission Protocol

The Stream Control Transmission Protocol (SCTP)[4] is a reliable, message-oriented transport protocol for IP network data communications. Compared to UDP, SCTP provides reliability and congestion control. Compared to TCP, SCTP provides 'multi-streaming' and 'multi-homing'. SCTP supports multi-homing by exchanging and maintaining lists of IP addresses for each SCTP endpoints which are associated with each other and supports multi-streaming by managing several separated data streams which can transmit data via independent sequence deliveries. For each SCTP association, the end host has one primary path and one or more backup paths. The end hosts with multiple network interfaces can connect to several separate networks concurrently using STCP, which makes it suitable for mobility. Moreover, the end-hosts can change their primary communication path to a new path before the breakdown of the current path triggered by a handover decision. This provides the possibility for seamless handover under SCTP. A TCP-like congestion control mechanism is employed in SCTP at the association level.

[4] Stream Control Transmission Protocol - https://tools.ietf.org/html/rfc4960

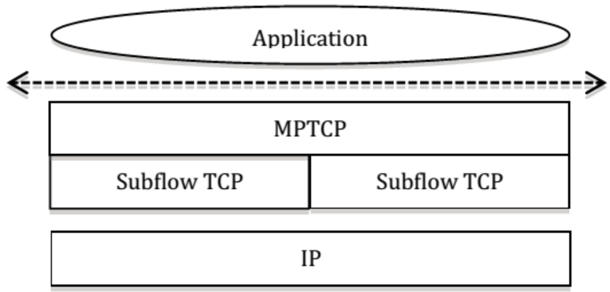

Figure 2-9 MPTCP Architecture [56]

2.3.2.3 Multi-Path Transmission Control Protocol

The Multi-Path Transmission Control Protocol (MPTCP) represents an extension of the classic legacy TCP [54][55] that was designed to be transparent to both applications and network. It allows multiple sub-flows to be set up for a single connection session between two hosts [14]. MPTCP is connection-oriented and allows mobile devices to concurrently utilize multiple interfaces and network access technologies to improve both network delivery performance and QoS, especially in heterogeneous wireless network environments.

Fig. 2.8 illustrates MPTCP architecture which consists of two levels [56]: the “MPTCP level” is an application-oriented level, which gathers the semantic from the application. The “Sub-flow TCP level” is the network-oriented level and helps with protocol’s reuse of the TCP architecture, which ensures the MPTCP packets are not blocked by middle-boxes.

A summary of the five key transfer protocols is listed in Table 2.4

Table 2-4 Features of Transfer Protocols[5]

Feature Name	TCP	UDP	DCCP	SCTP	MPTCP
Packet Header Size (Bytes)	20–60	8	12 or 16	12	50–90
Transport Layer Packet Entity	Segment	Datagram	Datagram	Datagram	Segment
Connection Oriented	Yes	No	Yes	Yes	Yes
Reliable Transport	Yes	No	No	Yes	Yes
Unreliable Transport	No	Yes	Yes	Yes	No
Preserve Message Boundary	No	Yes	Yes	Yes	No
Ordered Delivery	Yes	No	No	Yes	Yes
Unordered Delivery	No	Yes	Yes	Yes	No
Data Checksum	Yes	Optional	Yes	Yes	Yes
Checksum Size (Bits)	16	16	16	32	16
Partial Checksum	No	No	Yes	No	No
Path Mtu	Yes	No	Yes	Yes	Yes
Flow Control	Yes	No	No	Yes	Yes
Congestion Control	Yes	No	Yes	Yes	Yes
Explicit Congestion Notification	Yes	No	Yes	Yes	Yes
Multiple Streams	No	No	No	Yes	Yes
Multi-Homing	No	No	No	Yes	Yes
Bundling / Nagle	Yes	No	No	Yes	Yes

2.3.3 Adaptive Bitrate Streaming Standards

Adaptive bitrate (ABR) streaming uses either a source video format that is encoded at multiple bitrates or performs transcoding from an original rate to the desired one on the fly. ABR works by detecting the delivery conditions (e.g. network bandwidth at the end device and/or CPU capacity, energy level, etc. in real time) and by adjusting the quality of the transmitted video stream accordingly. Traditional adaptive bitrate streaming protocols

[5] Comparison of transport layer protocols - https://en.wikipedia.org/wiki/Transport_layer

are associated with classic standards such as the Real-Time Transport Protocol (RTP)[6], the Real-Time Streaming Protocol (RTSP)[7], and Hypertext Transfer Protocol (HTTP)[8]. RTP is designed for real-time streaming between end-to-end devices and was mostly performed over UDP, which has no QoS guarantee, transmission control and data protection. RTSP is used to provide remote multimedia playback control support such as play/pause commands from the end user devices. Synchronization between media streams is handled by the control protocol Real-Time Control Protocol (RTCP)[9] which also includes network QoS-related information such as loss, delay, and jitter. Most RTSP servers use the RTP in conjunction with RTCP for media stream delivery. There are also other protocols such as RTMP, etc. but they are not widely deployed.

The main disadvantage of using RTP/UDP is that it cannot traverse Internet firewalls and NAT devices as most of them are configured to restrict UDP traffic. Lately, in order to overcome this problem, HTTP is widely used, as it is allowed by the majority of firewalls. HTTP uses TCP or MPTCP as underlying transport protocols. This is the main reason for which the majority of the deployed adaptive multimedia solutions are based on HTTP, and hence either TCP or MPTCP.

HTTP-based adaptive bitrate streaming is client-driven and the adaptation logic resides at the client slide, which can reduce the requirement of persistent connections between client and server. This architecture also increases scalability by removing the session maintenance from the server side, and was seamlessly adopted by the existing HTTP delivery infrastructure (e.g. HTTP caches and servers).

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6 Real-time Transport Protocol – https://www.ietf.org/rfc/rfc3550.txt

7 Real-time Streaming Protocol – https://www.ietf.org/rfc/rfc2326.txt

8 Hypertext Transport Protocol - http://www.ietf.org/rfc/rfc2616.txt

9 RTP Control Protocol - http://tools.ietf.org/html/rfc4961

2.3.3.1 HTTP-based Adaptive Streaming Standards

The first international standard on the adaptive bit-rate HTTP-based streaming solution is the Dynamic Adaptive Streaming over HTTP (DASH) [57] referred to as MPEG-DASH, which started in 2010, had a draft in January 2011 and was released as a final Standard in November 2011. DASH is based on Adaptive HTTP streaming (AHS) [58] in 3GPP Release 9 and on HTTP Adaptive Streaming (HAS) [59] in Open IPTV Forum Release 2.

MPEG-DASH is an adaptive bitrate streaming technique that enables high-quality streaming of the media content over the Internet. The video content is partitioned into one or more segments and delivered from conventional HTTP web servers to the client using HTTP. DASH consists of two main components [60]: Media Presentation and Media Presentation Description (MPD). Media Presentation is a sequence of one or more segments that incorporate periods, adaptation sets, and representations, which break up the video from start to finish. MPD is like a manifest file and is an eXtensible Markup Language (XML) document that identifies the various content components and the location of all alternative segments, providing the relationship between them.

In addition to the standards, some other HTTP-based adaptive streaming solutions are adopted by the key industry players (e.g. Microsoft, Apple, and Adobe).

Microsoft Smooth Streaming (MSS)[10], referred to as Smooth Streaming, was introduced from a patent "Seamless Switching of Scalable Video Bitstreams" [61] from Microsoft, and is an Internet Information Services (IIS) Media Services extension. MSS switches between streams of different quality levels according to the network's available bandwidth.

10 Smooth Streaming https://www.iis.net/downloads/microsoft/smooth-streaming

Apple HTTP Live Streaming (HLV)[11] is a client-side adaptive HTTP streaming solution as part of QuickTime X and iOS, which supports both live and video on demand content. HLV uses its own segmenter to divide the stream/video content into small MPEG2-TS files as video chunks with different duration and bitrate.

Adobe HTTP Dynamic Streaming (HDS)[12] enables on-demand and live adaptive bitrate video delivery of standard-based MP4 media over regular HTTP connections. HDS is deployed on the Adobe Flash media delivery platform [62], which means it is available on any device running a browser with Adobe Flash plug-in.

Compared to MSS, HLS, and HDS, MEPG-DASH not only that supports all the features from other three solutions, but also have some special features such as HTML5 support, definition of quality metrics, multiple video views, etc. 3GPP Release 10 has adopted MEPG-DASH for use over wireless networks [63].

2.3.4 Challenging Multimedia Content

Currently, the most commonly used video codecs in HTTP-based Adaptive Streaming are H.264/AVC[13] and H.265/HEVC[14]. The most common used audio codec is AAC[15]. From Standard-Definition (SD) to High-Definition (HD) and Ultra High Definition (UHD), the MEPG-DASH works well to adapt and stream video content. However, for some challenging multimedia content, there is still need for more investigations. This content includes 3D video, Virtual Reality (VR), and mulsemedia content and discussed next.

11 HTTP Live Streaming https://tools.ietf.org/html/draft-pantos-http-live-streaming-19
12 HTTP Dynamic Streaming http://www.adobe.com/products/hds-dynamic-streaming.html
13 H.264 : Advanced video coding https://www.itu.int/rec/T-REC-H.264
14 H.265 : High efficiency video coding https://www.itu.int/rec/T-REC-H.265
15 Advanced Audio Coding (AAC) http://www.iso.org/iso/iso_catalogue/catalogue_ics/catalogue_detail_ics.htm?csnumber=43345

2.3.4.1 3D Video

Currently, there are two categories of 3D video technologies: stereoscopic 3D, as the first generation 3D video technology and multi-view 3D video as the second generation. Table 2.5 shows the comparison of the two technologies from different points of view.

Table 2-5 Comparison of Two 3D Video Technologies [64]

	Stereoscopic 3D	Multi-view 3D
Idea	1) Creates or enhances the illusion of depth in an image and presents two offset images separately to the left and right eyes. The two images are perceived by humans as 3D depth enhanced. 2) uses different input layouts: side by side, top/down, alternating rows, etc.	1) Simultaneously encodes sequences captured by multiple cameras using a single video stream. 2) uses as input layout: multiple view streams
Strength	1) Compatible with conventional 2D video 2) Saves bandwidth and storage in comparison to Multi-view 3D 3) Good for broadcasting	1) Experiences natural depth perception 2) No glasses 3) Multiple angles
Weakness	1) Resolution of individual view is lower compared to 2D 2) Glasses needed in most cases 3) Lenticular sheet technology can avoid using glasses, but currently provides narrow spots.	It is challenging for broadcasting due to limited bitrate channel
Adaptation	Bitrate scaling e.g. 1) assign lower bitrate for chrominance than for luminance component; 2) reduce bitrate by discarding enhancement layers for either/both left and right eye(s).	View scaling e.g. 1) Discard certain views which might be outside of the user's field of view. 2) Depth based rendering is always adopted to enhance the experience with low added bitrate.

Codec	MPEG4/H.264 AVC for 2D+MPEG4/H.264 for depth; MPEG4/H.264 AVC for 2D+MVC for depth as enhancement; Multi-view Video Coding (MVC)	Multi-view Video Coding (MVC)
Delivery	MPEG-2 transport stream, e.g. Blue-ray disc IETF RTP, e.g. real-time transport via IP ISO base media file format, e.g. progressive download in video-on-demand, HTTP streaming	

2.3.4.2 VR, AR, 360-degree Videos and Mulsemedia Content

The 360-degree video is not VR. These are videos in which a view in every direction is recorded at the same time using either a special rig of multiple cameras (e.g. omnidirectional camera) or a dedicated VR camera. The viewer can control the viewing direction during playback.

Virtual reality (VR) is a realistic and immersive simulation of a three-dimensional environment. VR video used 360-degree video [65] and relies on the mechanism that our brain achieves stereo vision i.e. by fusing two images from our eyes, in which nearby objects have greater disparity than far away objects. The recommendation about the minimum resolution for VR video from YouTube is 5120x5120 [66], which is much higher than the 4K requirements. The VR video needs other features to support the omni-directional stereo vision.

By contrast, Augmented Reality (AR) essentially inserts virtual objects into the real-world view, which the virtual object elements are augmented by computer-generated sensory input such as sound, video, graphics data.

Mulsemedia [67] or multi-sensorial media consists of other media types from human senses (i.e., haptic, olfaction, taste, etc.) in addition to audio and visual content. Unlike traditional multimedia, mulsemedia aims to provide immersive communications and enhances user QoE. Mulsemedia services may include any combination of traditional media objects such as text, graphical images, and video, as well as non-traditional media such as olfactory, haptic and skin-sensorial data. As mulsemedia is essentially about using these multiple media objects to communicate information to users, achieving synchronization between the component media objects that make up the mulsemedia is essential to the success of these systems.

At the moment, among the 3D video, 360-degree video, AR, VR, and Mulsemedia, there are only a limited number of mobile devices that support stereoscopic 3D due to the specific hardware requirements (i.e. 3D-enabled screen, advanced GPU, etc.). So, in this

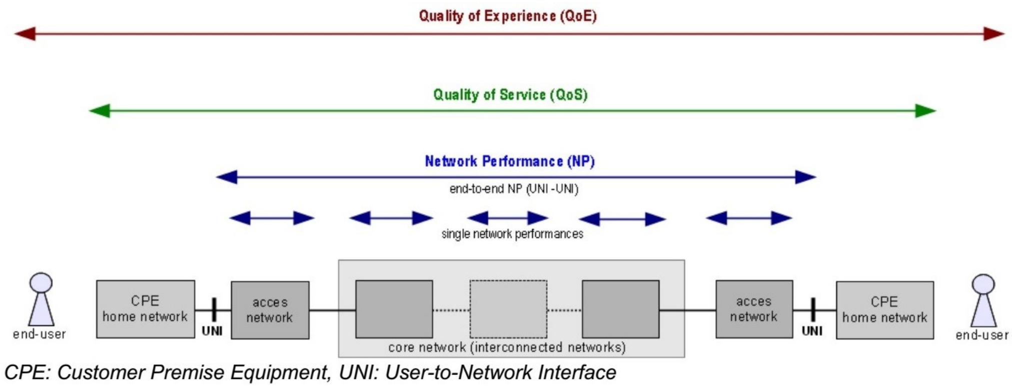


Figure 2-10 Network Performance, QoS and QoE [67]

work, we consider the use of stereoscopic 3D as the multimedia content rather than Multi-view 3D.

2.3.5 Video Content Delivery Quality Measurement

For measuring the user's experience on the service quality levels, two important concepts have been introduced: Quality of Service (QoS) and Quality of Experience (QoE). As already mentioned in section 2.2.3.1, QoS is the overall performance of a telecommunications service that bears on its ability to satisfy stated and implied needs of the user of the service. QoE represents the overall acceptability of a service as subjectively perceived by an end user [68]. The concepts, scopes of application and differences between QoS and QoE in the context of network services are illustrated in Fig 2.9. QoE includes the complete QoS-based effects and also can be influenced by additional psychological factors of end-user perception in a different environment with different types of applications services.

QoE is one of the most important factors when measuring the quality of a service, and is focused on understanding the overall human quality requirements. It involves various research fields such as for example social psychology, cognitive science, economics, and engineering science.

QoS and QoE can be measured in the context of a certain service application. This thesis focuses on assessing video perceived quality level following its network delivery.

2.3.5.1 Subjective and Objective Quality Assessment

In order to assess end-user perceived quality regarding the delivered video content, various methodologies were developed to quantify the received video quality. Two major approaches exist: subjective methods and objective methods.

Subjective methods require direct human exposure by which running subjective tests and ask directly the test participants to score their perceived quality of the video in the experiments. This involves large monetary and time costs and does not work for prediction or real-time quality assessment.

Objective methods are based on the use of metrics and the calculation processes are performed by algorithms. These algorithms can be classified into three main subgroups [69]:

A) *Full reference methods* - based on the comparison between the original video (before transmission) and the received video;

B) *Reduced reference methods* - require a feature vector derived from the statistical model of the reference video for quality evaluation;

C) *No reference methods* - use no-reference models which are based on the network-related or application-specific characteristics information and the received video only.

In this work**,** the Mean Opinion Score (MOS) [70] is one of the metrics adopted for subjective video quality estimation and comparison. MOS is one of the most commonly used metric in assessing the video quality. Five quality levels are defined in the MOS to measure the human quality of the video: from 1 representing "bad" quality to 5 representing "excellent" quality, as shown in Table 2.6.

Table 2-6 ITU MOS Quality and Impairment Scale [70]

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

Another metric is the Peak Signal-to-Noise Ratio (PSNR). PSNR is a full reference objective metric commonly and widely used for assessing video quality. The formula for PSNR is shown in eq. (2.9):

$$PSNR_{dB} = 20\log_{10}\frac{MAX_I}{\sqrt{MSE}} \tag{2.9}$$

MAX_I is the maximum possible pixel value of the image (e.g. 255), Mean Squared Error (MSE), represents the difference between the original video and the received one and can be calculated by eq. (2.10):

$$MSE = \frac{1}{MN}\sum_{m=1}^{M}\sum_{n=1}^{N}[x(n,m) - y(n,m)]^2 \tag{2.10}$$

$x(n,m)$ is the sample of the original source signal, y(n, m) is the distorted video signal. Various different approaches in defining PSNR for different purposes appear in the literature. In this work, we use the PSNR metric proposed by Lee in [71], which estimates the video quality subject to its network transmission. The computation is shown in eq. (2.11):

$$PSNR = 20\log_{10}\left(\frac{MAX\_Bitrate}{\sqrt{(EXP_{Thr} - CRT_{Thr})^2}}\right) \tag{2.11}$$

MAX_Bitrate represents the bit rate of the multimedia stream after the encoding process. EXP_{Thr} represents the average throughput expected to be obtained. CRT_{Thr} is the actual measured throughput during the transmission.

The PSNR metric values are mapped to the MOS scale in [72] by using a conversion table as in Table 2.7.

Table 2-7 PSNR to MOS Conversion [72]

PSNR [dB]	MOS Scale	Quality	Impairment
>37	1	Excellent	Imperceptible
31-37	2	Good	Perceptible but not annoying
25-31	3	Fair	Slightly annoying
20-25	4	Poor	Annoying
<20	5	Bad	Very annoying

2.4 Chapter Summary

This chapter presents an overview of the main background. Technologies involved in this thesis, including wireless networks (i.e. Wi-Fi and cellular access network technologies), network selection and reputation mechanism, multimedia content delivery, including adaptive streaming standards and relevant protocols and video quality assessment methods.

CHAPTER 3: Related Works

Abstract

This chapter discusses works related to the research performed in this thesis, divided into four topics: 1) network selection strategy in heterogeneous wireless networks (HetNets), 2) reputation mechanisms, 3) multipath transmission and load balancing, and 4) 3D video delivery. Existing solutions, standards and approaches in the research literature are presented, analyzed and compared. Building on the limitation of these related works, the architecture of the reputation-based quality optimization of video delivery over heterogeneous networks and the progress beyond the state-of-the-art will be introduced later on in the thesis.

3.1 Network Selection Strategy in HetNets

In this section, the state of the art in terms of network selection strategy is discussed. These works have been split into three major areas: 1) network selection protocol/function defined by different standard organizations and some extended framework based on these protocols, 2) Multiple Attribute Decision Making (MADM)-Based Network Selection which use MADM methods and applications to generate a representative value associated with each network for network selection decisions, 3) Other aspects including fundamentals of network selection, parameter conditions for network selection and network selection mechanisms.

3.1.1 Network Selection Protocol/Function

In order to improve the seamless handover between either the interworking networks or intra-system networks, several protocol/functions have been defined by different standard organizations. Access Network Discovery and Selection Function (ANDSF) was introduced in Release 8 of the 3rd Generation Partnership Project (3GPP) as a key network element for access network discovery and selection. The IEEE 802.21 Protocol Media

Independent Handover (MIH) framework released to provide support for vertical handover between different radio access networks. Access Network Query Protocol (ANQP) [16] described in the IEEE 802.11u standard amendment provides information for the mobile devices connecting to WLANs. Various investigations have studied the integration of these protocols/functions within the heterogeneous wireless environment in order to provide seamless vertical handover.

Hagos et al. [73] studied the approach of using WiFi offloading mechanisms in Long Term Evolution (LTE) cellular networks to move some of the data traffic from the core network to the WiFi network at peak times. The authors explored the use of three different ANDSF offloading algorithms, such as: (1) ANDSF model based on Cell-ID – in which case the mobile user will connect to the WiFi AP with the SNR (Signal to Noise Ratio) value greater than zero; (2) ANDSF mode based on position – where the mobile device will connect to the closest WiFi AP in terms of location and (3) ANDSF model based on Cell-ID and position – which combines the previous two methods. The performance of the three solutions was evaluated by simulations and compared with another two solutions, namely: *WiFi if Coverage* which selects a WiFi network if available and *Fixed SNR Threshold solution* which selects a WiFi network if the SNR is above a fixed threshold. The simulations were performed using the Monte-Carlo static MATLAB-based multi-cell radio access network simulator model and employed the *Equal Buffer traffic model*. The results show that the ANDSF models reduce the average number of APs scanned by the UE from the list of available WiFi access networks. Moreover, by changing the network discovery distance threshold, the percentage of the users connected to the WiFi also varied accordingly, making possible controlling the amount of WiFi offload. Thus the proposed algorithm can achieve high energy savings and reduce the cost in terms of time. The authors considered a scenario in which some of the traffic from an LTE network was

[16] IEEE 802.11u-2011 - http://standards.ieee.org/findstds/standard/802.11u-2011.html

offloaded to WiFi networks based on the SNR only in order to extend the capacity of the LTE access. However, SNR alone might not be enough for the decision making and other performance metrics should also be considered. For example, selecting the WiFi is not always the best alternative especially when it is already heavily loaded; then the mobile user is better off connected to the LTE network.

Kwon et al. [74] enhanced the functionality of ANDSF by extending the original messages format and modifying the procedures to control the congestion in the heterogeneous wireless environments. For this proposed architecture, by employing the Explicit Congestion Notification (ECN) protocol, UEs can inform ANDSF about the congestion condition of various access networks, and ANDSF can recognize the condition and update its ISMP (inter-system mobility policy) corresponding to the access crowded network. Based on the updated ISMPs, the proposed congestion control procedure allows UEs to choose a less crowded access network within the heterogeneous networks. The authors introduce the use of two traffic offloading techniques, namely: Multi-Access PDN (Packet Data Networks) CONnectivity (MAPCON) and the IP-Flow Mobility (IFOM). However, the performance evaluation of the proposed framework is considered to be part of the future work.

GhasemiNajm et al. [75] looked at the integration of IP Multimedia Subsystem (IMS) and LTE and proposed the use of ANDSF as a location enabler. The paper introduced the relationship between Evolved Packet System (EPS) and IMS as a platform to deliver the location based services, and the integration of IMS and LTE as an effective procedure to deliver high quality of services. In the proposed architecture, an interface to handle the Mobile Location Protocol (MLP) is proposed to transmit the location information from ANDSF to the service provider. This procedure used ANDSF as a location enabler in combination with LTE and IMS in order to ensure the delivery of higher bandwidth and lower latency location based services. Further investigations are needed to demonstrate the potential benefits of this framework.

Kim et al. [76] proposed an ANDSF-assisted Wi-Fi control method by taking into consideration user motion states to avoid unnecessary Wi-Fi scanning and connections. The authors propose a new method of detecting motion states to recognize whether the user is stationary or not in a short time period avoiding in this way unnecessary Wi-Fi scanning and unnecessary Wi-Fi connections. The proposed method employs ANDSF to retrieve the available networks based on location information. *User's motion state* is detected based on the 3G received signal strength indicator (RSSI) / base station ID (BSID) variations, which turn on the Wi-Fi interface automatically when a user is stationary only. Consequently, the *WiFi selection* is based on the WiFi RSSI. The performance evaluation of the proposed solutions is done in a real environment and compared against a classic Wi-Fi offload solution. The results show that by using the ANDSF-assisted Wi-Fi control method which considers the motion states, the time cost and the number of WiFi scans can be greatly reduced.

Triantafyllopoulou et al. [77] proposed an ANDSF-assisted network discovery algorithm that exploits both the user location and available access networks information to decide when network scanning should be executed, in order to avoid unnecessary energy consumption for network scanning and misdetection of available access networks. The performance evaluation of the proposed solution is performed through simulations and compared against an algorithm that performs periodic network scanning with a fixed period, without taking into consideration the user or network context information. The simulation model is constructed in C++. Based on the simulation results, the proposed algorithm allows a user to reduce its energy consumption and minimizes the delay of detecting available neighboring networks with no loss in the network detection.

Park et al. [78] proposed a new gateway service approach to integrating heterogeneous networks using different interworking solutions, such as MIH, ANQP and ANDSF. The reserved Service Identifier (SID) of MIH header value is used for the new gateway services. The gateway services encapsulate the interworking messages are encapsulated in the MIH header at the gateway services function of general network entities and sent to the corresponding network entity. Two cases of these gateway services have been introduced: (1) when the MIH Information Server (MIHIS) that support ANQP messages and gateway service, the gateway services provide simple encapsulation function in the integration between MIH and ANQP; (2) when the MIHIS that does not support ANQP messages and gateway service, the integration between ANQP and MIH needed other interworking solutions such as ANDSF to convert ANQP messages into other interworking message on an intermediate network entity which support ANQP messages and gateway service. This gateway service can make the interworking between heterogeneous networks be realized cost-effectively by reducing the complexity of network entities. Further investigations and implementation are needed to demonstrate the potential benefits of this framework.

Song et al. [79] proposed a Data Forwarding Function (DFF) based Forward Attachment Function (FAF) scheme that eliminates the data loss during Vertical Handover (VHO) execution between Mobile WiMAX and 3GPP and resolves the problem of abrupt disconnection to the source network. The DFF is a base station (BS)-level logical entity that is located in the source access network and communicates with the UE via IP tunneling. The FAF is also a BS-level entity, but is located in the target access network. It supports authentication of the UE before the execution of VHO through the IP tunnel. In the case of VHO from mobile WiMAX to 3GPP with the assistance of the ANDSF, the DFF emulates the mobile WiMAX BS and the FAF performs as a Universal Terrestrial Radio Access Network (UTRAN) Radio Network Controller (RNC). Two types of real-time video traffic simulation experiments have been conducted: 1) 80 multiplexed real time H.263 video streams; 2) 40 multiplexed MPEG-4 video streams. Based on the simulation results, the DFF-FAF scheme reduces by 20–99% packet loss probability compared to the FAF-only scheme in different delay bound ranges.

Nahas et al. [80] introduced Context Aware-Access Network Discovery and Selection Function (CA-ANDSF). The authors extended the functionality of ANDSF by involving dynamic and critical parameters into the decision policy to enhance the handover mechanism and modify the communication protocol to convey the inserted parameters between the ANDSF entities. Two types of parameters are inserted into the decision policy: (1) User Related: terminal type, application type, battery power and UE velocity; (2) Network Related: Received Signal Strength (RSS), delay, and jitter and data rate. The weighting criterion of inserted parameter is based on the Analytic Hierarchy Process (AHP). The communication protocol reuses the S14 interface which is defined in the 3GPP standards and adds "PUSH" and "PULL" operation modes to transfer the information between ANDSF entities. Simulation results show that CA-ANDSF improves the performance of successful handover decisions compared to traditional ANDSF and reduces the energy consumption when the battery power of UE reaches a minimum threshold.

Frei et al. [81] proposed a solution that combines MIH and ANDSF for improving the inter system handover behavior. Most mechanisms of two frameworks are different from each other and therefore could complement one another well if these frameworks were both deployed through the networks. ANDSF can provide the important Inter-System Mobility Policy (ISMP) as well as the access network discovery information and if available the inter-system routing information (ISRP) to the Mobile Nodes (MN)/Mobile Station (MS). From the Media Independent Handover Function (MIHF), the MN/MS is informed about the surrounding networks through the Media Independent Information Service (MIIS), can execute commands through the Media Independent Command Service

(MICS), and can send events through the Media Independent Event Service (MIES). The handover procedure of an MN from a WiMAX access network towards a 3GPP LTE network shows that the proposed solution ensures (1) improvement of the handover through the ANDSF; (2) elimination of the data loss during handover; (3) improvement of the resource release. Further investigations and implementation are needed to demonstrate the potential benefits of this framework.

Doppler et al. [82] extended the functionality of ANDSF by involving an energy efficient idle scanning method to reduce the energy consumption. A novel light-weight scanning assistance is proposed to enhance the information retrieved from the ANDSF server by the operating band/channel used by the networks. Then, a push mode proposed for UE to provide information about the access point density for its location and other possible visiting areas. Based on this the UE forms a scanning strategy and can avoid draining the battery in areas with very low access point density. Three different network scanning strategies have been investigated in a residential area: (1) Unassisted scanning with a fixed periodicity; (2) Unassisted scanning only if moved; (3) The assisted idle mode scanning strategy providing each UE with the number of accessible access points on each band or channel. LTE femtocell, WiFi frequency variant for US TV White Spaces (WS) and 5GHz are considered in the tests. Based on the test results, the authors show how the proposed scanning assistance method can reduce with 66% of the energy consumption compared with the unassisted case.

Hu et al. [83] proposed an MIH and software-defined network (SDN)-based framework for network selection in 5G heterogeneous network. In this framework, SDN is used to initiate a pre-selection scheme, which is used to generate the network ranking by an SAW-based QoS-oriented MADM method and eliminate unqualified networks that are associated with performance below the QoS score threshold. By using the MIHF to exchange the information, the best two candidate networks are sent to the selection decision module, and the final decision is made by a two-dimensional cost function. The simulation result shows how this framework can reduce both the handover latency and backhaul energy consumption with an accurate selection of the optimal network.

Table 3.1 summarizes the works discussed in the context of the network selection protocol in terms of description, parameters and details.

Table 3-1 Feature of Network Selection Protocol

Solution	RAT	Description	Parameters	Details
Performance-centric offloading strategies for LTE networks [73]	WiFi-LTE	An optimized SNR-threshold based handover solution and ANDSF framework for WiFi offloading	SNR; SINR; Bitrate; Throughput	algorithm: WiFi if Coverage; Fixed SNR Threshold; ANDSF Models
ANDSF -based congestion control procedure in heterogeneous networks [74]	HetNets	Messages and procedures for ANDSF to collect the congestion condition	N/A	ANDSF-MO(management object), ISMP; explicit congestion notification (ECN) protocol
Combination of LTE and IMS to deliver location based services [75]	LTE	ANDSF as a location enabler for LTE network to deliver location based services	N/A	MLP (Mobile Location protocol)
Efficient ANDSF-assisted Wi-Fi control for mobile data offloading [76]	WiFi-LTE	ANDSF-assisted Wi-Fi control method based on user high-level motion state to avoid unnecessary Wi-Fi scanning and connections	Traffic (bytes), RSSI, BSID, time cost (seconds)	ANDSF-assisted Wi-Fi control method
Energy efficient ANDSF-assisted network discovery for non-3GPP access networks. [77]	LTE	An ANDSF assisted network discovery algorithm that exploits information on the user location and on the location of available networks	Energy consumption per user, average network detection delay and average network detection rate	ANDSF-assisted Network Discovery Algorithm
Gateway service in 802.21c draft [78]	HetNets	Gateway service that supports cost-effective integration of heterogeneous networks using different interworking solutions	SID (Service Identifier), AID (Action Identifier).	MIH; ANQP; ANDSF.

DFF(data forwarding function)-FAF(forward attachment function) [79]	WiMAX -3GPP	The data forwarding function (DFF) that eliminates data loss during VHO execution and resolves the problem of abrupt disconnection to the source network.	Packet loss probability; Delay; Interruption time	Data forwarding function (DFF); forward attachment function (FAF);
CA-ANDSF [80]	WiFi-LTE	Context Aware (CA) decision algorithm is integrated within the ANDSF server so that the inserted measurements are expeditiously invested to improve the handover decision policy.	Terminal type, Application type, Battery power and UE velocity, RSS, Delay, Jitter and Data rate	Weighting criterion: Analytic Hierarchy Process (AHP); communication protocol: S 14 interface
Inter System Handover in the EPC Environment [81]	WiMAX - LTE	A solution that combines MIH and ANDSF is proposed for improving the inter system handover behavior.	N/A	Policy and Charging Control (PCC); Policy and Charging Rules Function (PCRF); Gxa interface
Scanning Assistance in ANDSF [82]	LTE-WiFi	A novel light-weight scanning assistance that in ANDSF by involving an energy efficient idle scanning method to reduce the energy consumption.	Energy consumption (J); Scanning period; SNR.	ANDSF
MIH and SDN based network selection [83]	5G	An intelligent vertical handover framework that combines MIH and SDN for network selection to reduce the backhaul energy consumption, handover latency and frequency.	Received signal strength (RSS), delay, bandwidth, application type-based priority, block error rate (BLER) and jitter	MIH, SDN based pre-selection scheme

3.1.2 MADM-Based Network Selection

MADM methods are a widely used set of mathematical functions that are employed for aggregation of individual values and generate an overall representative value. MADM are in general associated with various network parameters and are used for network selection decisions. A survey [84] on vertical handover decision solutions in heterogeneous wireless networks shows that the network selection in vertical handover decision mainly concentrates on MADM-based approaches.

Sgora et al. [85] proposed an effective access network selection algorithm for heterogeneous wireless networks that combines two MADM methods. The Analytic Hierarchy Process (AHP) method is used to establish weights for the criteria which can reflect the hierarchy of the criteria. The Total Order Preference by Similarity to the Ideal Solution (TOPSIS) method is used to obtain the final access network ranking and minimizes the computational complexity. In this solution, the decision for the network selection is influenced by the requested application indicated by the user. Throughput, delay, jitters, packet loss, cost and security are the parameters used as the criteria. The simulations consider a Universal Mobile Telecommunications System (UMTS) network, a Worldwide Interoperability for Microwave Access (WiMAX) network and two Wireless Local Area Networks (WLAN), employing the IEEE 802.ll b (WLAN1) and IEEE 802.ll g (WLAN2) technologies. VoIP, media streaming and web browsing are the applications analyzed using numerical results. The results show that the proposed solutions can be effective for the selection of the optimal access network according to user application requirements, but no comparison with other schemes or solutions is provided.

Lahby et al. [86] presented a network selection algorithm based on two MADM methods: Diff-AHP (diff-analysis hierarchy process) and TOPSIS. Diff-AHP is an expansion of the AHP method, which is used to compute differentiated weights for the available networks using the following parameters: cost per byte, security, packet delay, available bandwidth, packet jitter and packet loss. By utilizing these QoS, cost and security parameters, the authors used MATLAB to simulate scenarios in a Wi-Fi, WiMAX and UMTS network environment using two different algorithms. The results show that the proposed algorithm can reduce the ranking abnormality problem (i.e. The ranking abnormality problem means that the ranking of candidate networks change can potentially decrease the quality of the results by causing handovers whenever low ranking alternatives are removed from the candidate list.) and provides the best performance regarding the number of handoffs.

Lahby et al. [87] compared five weighting algorithms: analytic hierarchy process (AHP), fuzzy analytic hierarchy process (FAHP), analytic network process (ANP), fuzzy analytic network process (FANP) and random weighting (RW) algorithms in network selection to their effect on the ranking abnormality of the TOPSIS methods. Simulations are used involving MATLAB, and a heterogeneous environment with UMTS, WIFI and WiMAX networks. Six parameters are weighted for those networks: cost per byte, available bandwidth, security, packet delay, packet jitter and packet loss, and four application classes were considered: conversational traffic, streaming traffic, interactive traffic and background traffic. The simulation results show that the TOPSIS algorithm based on ANP method is able to reduce the ranking abnormality for background, conversational and interactive scenarios when compared to other algorithms. In the streaming traffic scenarios, the TOPSIS algorithm based on the AHP method provides the best results.

Wang et al. [88] studied the four MADM-based methods in the context of network selection: simple additive weighting (SAW), multiplicative exponential weighting (MEW), technique for order preference by similarity to ideal solution (TOPSIS) and grey relational analysis (GRA). Network ranking can be generated based on those MADM algorithms for both the adjusting module with the criteria of dynamic/static attributes and weighting module with the criteria of operator policies, terminal properties, customer preferences and application QoS levels. The four algorithms are simulated in a heterogeneous environment which is composed of WWAN, WMAN, WLAN, and WPAN, and two device properties (i.e. power level and velocity) with four QoS levels (i.e. conversational, streaming, interactive and background) are considered. Based on the simulations, the authors proposed an integrated strategy for MADM-based network selection. The proposed strategy is designed to address four issues: efficient subjective weighting, mobility-based network selection, VHO tradeoff scheme and load balancing. Further investigations and implementation are needed in order to demonstrate the potential benefits of this framework.

Godor et al. [89] presented a network selection model which is based on the AHP and GRA methods. This model does not need any user interaction, but uses the user profiles and available networks' QoS parameters to determine the best radio interface to be employed automatically. The model consists of three elements: user equipment (UE), access networks and trusted third party. UE gives the service requirement; access networks will provide QoS parameters such as throughput, jitter, delay and bit error rate and trusted third party stores the information from UE and access network and authenticates the subscriber. The AHP method is used to calculate the global weight from the product of the user profile and the service's detailed parameters, and the GRA method is used to enable network selection. The simulation environment is implemented in MATLAB and considers a heterogeneous network scenario consisting of GSM, WLAN and WiMAX networks. The results show that the algorithm works fine in a realistic network environment, but no comparison with other solutions is provided.

Bakmaz et al. in [90] proposed an optimal network selection algorithm based on the TOPSIS method. In this algorithm, available Radio Access Networks (RAN) that satisfy the RSS condition (greater than a defined threshold) are ranked by four parameters:

available bandwidth, QoS level, security, and cost. For each of the parameters, the weight coefficients are adjusted depending on user and application demands, in a method which uses the entropy. The proposed algorithm is tested based on a C++ software application with three scenarios considering different user demands. Test results show that the proposed model can find simply and effectively the network which best balances between the service demands, user demands, and network conditions.

In [91], the authors present a multi-criteria network selection algorithm as a part of a wider middleware platform SALOME (Situation And LOcation aware MiddlEware). The platform is a multi-agent system (MAS) which consists of four parts: User Agent (UA), Network-side User Agent (NUA), Radio Resource Agent (RRA) and Service Agent (SA). UA sends candidate network and requested service information to NUA, then NUA contacts the SA to gather the economic cost and security level of the available networks. In the meantime, the RRA sends the value of allocable bandwidth, coverage level offered to the user, and network unreliability level to NUA, and NUA runs the network selection algorithm based on these parameters. The simulation scenario considers three user behaviors: private user, working user and businessman. The test results show that the proposed algorithm can select the access network dynamically adapted based on user preferences and profile. However, the authors do not provide any comparison with other schemes.

Song et al. [92][93] proposed a network selection algorithm based on two MADM methods: GRA and AHP. This scheme is designed to provide the user with the best available QoS at any time in an integrated wireless LAN and UMTS environment. In this scheme, the network is selected on behalf of the user by considering the network condition, service application, and user preference for QoS as decision factors. The user preferences and service requirements for QoS are assessed by the AHP, and the performances of network alternatives are ranked by the GRA. The quality of the network in this scheme is associated based on six components: availability, throughput, timeliness, reliability, security, and cost. Timeliness involves delay, response time and jitter, and reliability is based on bit error rate (BER), burst error, average number of retransmissions per packet and packet loss ratio. In order to avoid frequent handoffs for high-speed users, this scheme adopts the received signal strength (RSS) and coverage area as sub-factors determining availability. Simulations are run based on a scenario which considers office, home and airport areas in the network selection between WLAN and UMTS. The results demonstrate that the proposed network selection scheme can efficiently decide the trade-off among user preference, service application, and network condition. However, the authors do not provide any comparison with other solutions from the literature.

Table 3.2 summarizes the works discussed in the context of MADM-based network selection in terms of description, parameters and functions.

Table 3-2 Overview of MADM-based Network Selection

Solution	Networks	Description	Parameters	Functions
AHP & TOPSIS [85]	HetNets: WLAN, WiMAX and UMTS	An effective access network selection algorithm for heterogeneous wireless networks that combines AHP and TOPSIS methods.	Cost, throughput, security, packet dDelay, packet jitter and packet loss.	AHP, TOPSIS
Diff-AHP & TOPSIS [86]	HetNets: WLAN, WiMAX and UMTS	A network selection algorithm based on two MADM methods: Diff-AHP (diff-analysis hierarchy process) and TOPSIS	Ranking abnormality, cost per byte, available bandwidth, security, packet delay, packet jitter and packet loss.	Diff-AHP, TOPSIS
TOPSIS based ANP/AHP [87]	HetNets: WLAN, WiMAX and UMTS	Compared five weighting algorithms: AHP, ANP, FAHP, FANP and random weighting (RW) in network selection to their effect on the ranking abnormality of the TOPSIS methods.	Ranking abnormality, cost per byte, available bandwidth, security, packet delay, packet jitter and packet loss.	MADM: TOPSIS, AHP, ANP, FAHP, FANP.
Integrated strategy for MADM [88]	HetNets: WPAN, WLAN, WMAN and WWAN	A four-step integrated strategy for MADM-based network selection	N/A	SAW, MEW, GRA, TOPSIS
AHP &GRA [89]	HetNets: WLAN, WiMAX, GSM and UMTS	Using the user profiles and the available networks' QoS parameters to determine the best radio interface based on AHP &GRA automatically	Throughput; delay; response time; jitter; BER; burst error; packet loss; security; cost.	AHP; GRA
TOPSIS [90]	HetNets	Networks are ranked based on TOPSIS	Available bandwidth, QoS level, security, and cost.	TOPSIS
SALOME [91]	HetNets: WLAN and UMTS	A network is selected by associating to each cost parameter in the selection function with a weight dynamically adapted to user preferences and profile.	Monetary cost; power consumption; network capacity; security level; network unreliability; allocated bandwidth	Multi-criteria weighting function
GRA&AHP [92], [93]	HetNets: WLAN and UMTS	An integrated AHP and GRA algorithm for network selection in a heterogeneous system	Throughput; RSS; coverage area; delay; response time; jitter; BER; burst error; average number of retransmissions per packet;	GRA; AHP

3.1.3 Other Network Selection Related Solutions

To date there has been extensive academic research related to the Always Best Connected paradigm and various other solutions have been proposed in order to address this problem of ensuring always best connectivity and always best experience in a multi-user multi-terminal multi-network wireless environment. For example, Sharma et al. [94] proposed a vertical handover system (OmniCon) which enables handover between WLAN and GPRS links by using an extension of the Mobile IP protocol. The solution is transparent to the end-user and the switch between WLAN and GPRS is done based on the WLAN availability. In the OmniCon architecture illustrated in Fig. 3.1. The OmniCon Communication Daemons (CommD) establish a TCP connection between the GPRS Foreign Agent (GFA) and Mobile Node (MN) over GPRS link. A virtual interface on the MN is used to transfer the packet over the TCP connection between CommD on MN and

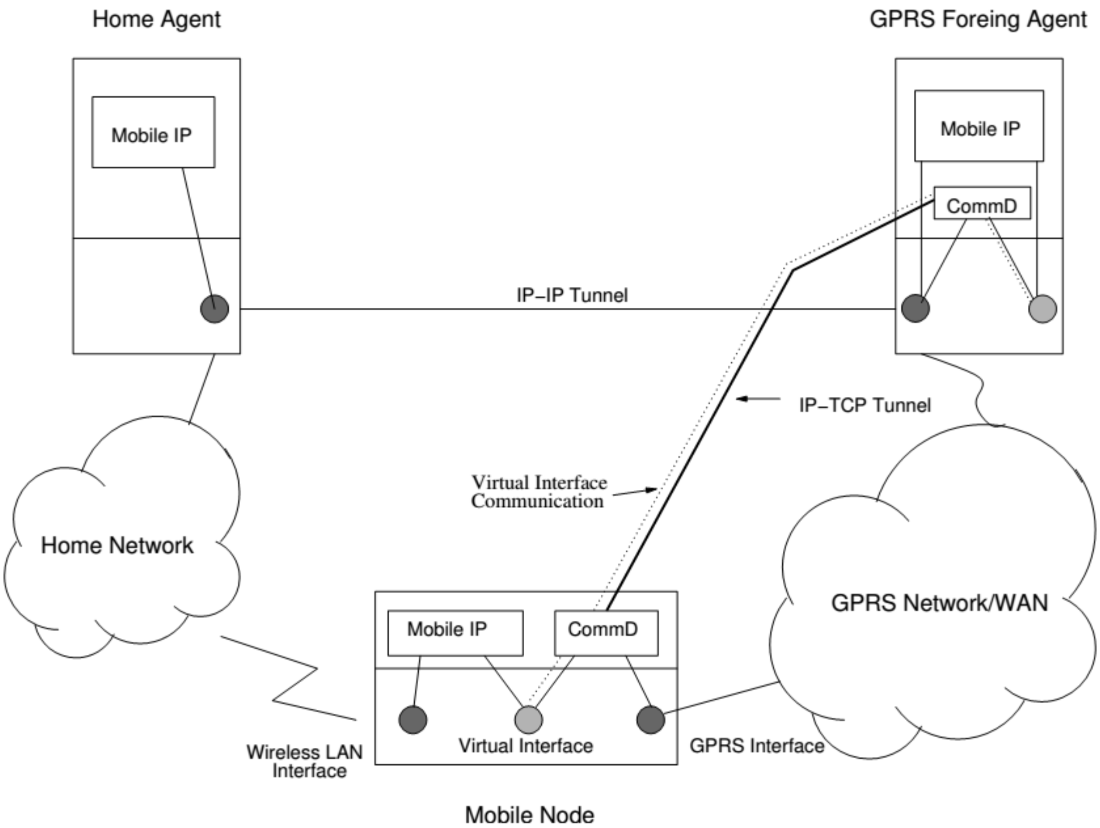


Figure 3-1 OmniCon Architecture [94]

GFA. The decision module relies on CommD monitoring the WLAN link states to decide whether to use WLAN or GPRS and interacts with the CommD on the mobile node for triggering the vertical handover.

Jabban et al. [95] proposed a Signal to Interference-plus-Noise Ratio (SINR)-based network selection strategy which allows users to select the highest SINR value network from a number of available networks. This approach assumes the mobile device is moving according to a 2-D fluid flow mobility model and analyzes the proposed methods in terms of blocking probabilities of calls. When compared with the traffic balanced-based network-selection (TBNS) strategy and the received signal strength-based network-selection (RSNS) strategy, the proposed solution shows a better performance than RSNS strategy, but does not reach the performance of the TBNS strategy because it does not consider the traffic load between available networks.

Andreev et al. [96] investigated the interworking within radio access networks. Based on the 3GPP Release-12 of LTE, the authors proposed a solution for intelligent access network selection, which selects an operating interface for a multi-radio enabled device in multi-radio heterogeneous networks (e.g. LTE and WiFi). This RAN-assisted solution contains a load-aware user-centric scheme, which is based on SNR measurements and network load information. A novel analytical space-time methodology is also proposed to capture user traffic dynamics for assisted network selection. Compared to conventional WiFi-preferred solutions based on the minimum SNR threshold methods, RAN-assisted solution is better both in terms of CDF and throughput.

Cheung et. al. [18] investigated WiFi offloading in a single-user single-flow decision scenario. A Delay-Aware Wi-Fi Offloading and Network Selection (DAWN) algorithm was proposed which uses the minimum cost to deliver a file when the time is not a critical factor, and achieves a higher probability of completing the file transfer under a stringent deadline. A Markovian mobility model that is derived based on the past mobility pattern of the mobile user is used to predict the possible location of the user movement. The

DAWN trade-off between the cost and time to trigger network selection among LTE, WiFi or idle is performed.

A study of access network selection (ANS) mechanisms is presented by **Louta et al.** [97]. The authors surveyed a set of representative vertical handover schemes proposed in the related research literature, compared them with the design of the ANS mechanism, and discussed their distinct features, relative merits and weaknesses by contrasting their objectives, control unit and methodology. The schemes surveyed could be categorized based on objectives as follows:

1) ***Network centric schemes***, which address the ANS problem from the network operators' perspective, aiming mainly at efficiently managing network resources and fulfilling current users' requests, while maximizing their revenue.

2) ***User-centric schemes***, which address the ANS problem from the users' point of view, aiming at assisting and enabling users to find and associate with the most appropriate access network for service provisioning, focusing on satisfying service requirements, user preferences and constraints, without considering the network operation efficiency.

Based on the handover mechanism employed by compare by the control unit, the scheme can also be classified as:

i) ***Network Controlled/Mobile Assisted*** (NCHO) when a network related entity is responsible for controlling and conducting the handover, exploiting information and measurements gathered from the mobile terminal and
ii) ***Mobile Controlled/Network Assisted*** (MCHO) when the mobile terminal has the primary control over the handover exploiting the information provided by the network.

The authors also mention that ANS is a multi-criteria decision problem, which could be solved by adopting multi-objective (MODM) and/or multi-attribute (MADM) related methodologies and algorithms. The characteristics used in those methodologies for ANS can be grouped in:

a) ***Link quality***, evaluated considering indicators such as RSS, Carrier to Interference Ratio (CIR), Signal to Interference Ration (SIR), Signal to Noise and Interference Ration (SNIR);

b) ***Network availability***, considering coverage, bandwidth availability & call blocking probability;

c) ***QoS related aspects***, considering throughput, delay, latency, jitter Bit Error Rate (BER), packet loss ratio, average number of retransmissions per packet;

d) ***Network reliability***, considering call dropping probability and handover execution failure probability;

e) ***Contextual information***, comprising current network load conditions, terminal velocity, terminal location, and remaining battery lifetime.

3.1.4 Discussions

Table 3.1 and Table 3.2 summarize the features employed for network selection protocols/functions and MADM-based network selection, respectively. Also, the parameters employed and functions utilized are described in these tables.

When assessing the related works, it could be concluded that there are still several limitations of these works:

i) most of the related works on network selection protocol/function are proposed as frameworks, but they are not deployed even in simulations;

ii) the research on MADM-based network selection focus on the MADM methods, but do not mention where the data of the parameters used in MADM methods comes from and do not employ any industry standard;

iii) all of the network selection works are based on single user case, and no multi-user cooperation is considered.

In this context, our solutions utilize the network selection protocol/function with multi-user involvement in the parameter information gathering, and using an MADM method to provide reputation value for network selection.

3.2 Reputation Mechanisms

Reputation mechanisms have been introduced for many years in various areas. In this section, the state of the art in relation to reputation mechanisms in wireless networking area is discussed. The discussion is divided in two major directions: 1) reputation mechanisms in Ad-hoc networks, and 2) reputation mechanisms in HetNets.

3.2.1 Reputation Mechanism in Ad-hoc Networks

Jaramillo et al. [98][99] proposed a distributed and adaptive reputation mechanism (DARWIN) for wireless ad-hoc networks. By using the game theory, the DARWIN mechanism is collusion-resistant, is robust to incomplete measurements and is able to accomplish full cooperation among nodes. In this scheme, nodes share the perceived dropping probability with other nodes. A pair of nodes is isolated to facilitate the theoretical analysis. The contrite Tit For Tat (CTFT) strategy is used to avoid retaliation situations when a node is falsely perceived as selfish, which helps to restore cooperation quickly. Performance simulations are conducted using the Network Simulator 2 (NS-2) and the proposed solution is compared with the Generous Tit For Tat (GTFT), Tit For Tat (TFT) and Trigger strategies. The results show that DARWIN can get a higher throughput and lower overhead than the other strategies. The impact of liars on the nodes needs to be considered during the implementation of this scheme.

Refaei et al. [100] proposed a reputation-based mechanism to detect and fast isolate selfish nodes in ad-hoc networks. The reputation is evaluated based on neighbor nodes which send and detect completion of the requested services. Each node maintains reputation table which stores the reputation value of each immediate neighbor. Based on the successful or failed packet delivery event counted, the reputation index of the node will change accordingly, and once the reputation index is lower than the threshold, the node will be isolated as a selfish node. Performance simulations are conducted in Network Simulator 2 (ns-2), and the scenario compares the performance of the proposed solution with three reputation function schemes over different network protocols: Double Decrement/Single Increment Ratio (DDSIR), Hops Away From Source (HAFS) and Random Early Probation (REP). The results show that all of these mechanisms succeed to fast isolate selfish nodes and maintain the false positives at a reasonably low level.

Buchegge et al. [101][102][103] proposed a reputation solution for both Mobile Ad Hoc Networks (MANETs) and P2P networks. The solution uses Bayesian estimation to specifically address the misbehaving nodes by eliminating the effect of incompatible recommendation on reputation. In this algorithm, the first-hand information about another node is periodically published lay each node and exchanged with others and the second hand information which is incompatible with first-hand information is used to update the reputation. Simulations have run based on the GloMoSim simulator and the results show that detection of misbehaving nodes increases significantly with the use of selected reputation information from other nodes.

Wang et al. in [104] proposed an adaptive and robust reputation (ARRep) mechanism in P2P networks. The ARRep mechanism uses the confidence factor to integrate direct trust value and recommended trust value to calculate the trust value of nodes which represent the reputation. This mechanism contains three parts: the transaction decay factor function, recommendation credibility algorithm and transaction validation protocol. The direct trust and recommended trust according to the experience of the requester are balanced by the ARRep mechanism, and the trust value calculation takes into account the transaction volume, size of a common set and decay factor to improve the value calculation metric accurate and timely. Simulation results show that ARRep mechanism is highly effective in the on-off, bad mouthing and collusive cheating attacks.

Jurca et al. [105] proposed a reputation mechanism based on averaging feedback in a service oriented environment. The proposed scheme involves service providers (seen as sellers) repeatedly offer the same service to the interested clients (considered as buyers). Sellers share a service level agreement (SLA) with a promised quality level and price of the service for each day. Buyers give feedback in terms like satisfactory or non-satisfactory about the quality of service of different sellers. This service oriented architecture can use reputation information to forecast the service quality level and dynamically scale the service price. Further investigations are needed to demonstrate the potential benefits of this mechanism.

Xiong et al. [106] proposed a reputation based trust supporting framework (PeerTrust) in P2P networks. In this scheme, three trust parameters are used: feedback in terms of the amount of satisfaction, number of transactions and credibility of feedback and are combined using two trust metrics: PeerTrust TVM (trust value measure) and PeerTrust PSM (personalized similarity measure). Those two trust metrics are based on two content factors: transaction factor (incorporating transaction contexts) and community context factor (providing incentives to rate). Finally, the overall trust value is computed via dynamic trust computation (DTC) or approximately trust computation (ATC) strategies. Performance simulations are conducted using Mathematica 4.0. The results show that the proposed solution can effectively evaluate the trustworthiness of peers and reduce the impact of dishonest feedback and effect of lack of incentives.

Table 3.3 summarizes the works discussed in the context of the reputation mechanisms in Ad-hoc networks in terms of description, parameters and functions.

Table 3-3 Reputation Mechanisms in Ad-hoc Networks

Solution	Networks	Description	Parameters	Functions
DARWIN [98][99]	Ad-hoc Network	By using the game theory, DARWIN mechanism is collusion-resistant, robust to incomplete measurements and supports full cooperation among nodes.	Dropping probability; Forwarded probability.	Game theory: Contrite Tit For Tat
Reputation-based mechanism for isolating selfish nodes [100]	Ad-hoc Network	A reputation-based mechanism to detect and fast isolates the selfish nodes in ad-hoc networks.	Reputation index; failed/ Successful packet delivery times;	DDSIR; HAFS; REP
MANETs [101][102][103]	Ad-hoc and P2P Network	A Bayesian Approach to Reputation Systems	Reputation value; Trust Value; First-hand information;	Standard Bayesian Approach; Reputation Fading Mechanism
ARRep [104]	P2P Network	An adaptive and robust reputation mechanism in P2P networks	Computed Trust Value; Fault Download Rate	Transaction decay function; recommended trust value; Transaction Validation Protocol
Feedback reputation mechanism [105]	P2P Network	A reputation mechanism based on averaging feedback in a service oriented environment	Risk-seeking price; production quality level	Price and cost functions; SLA
PeerTrust [106]	P2P Network	A reputation mechanism based trust supporting framework in P2P networks	Computed Trust Value; Trust computation overhead	TVM; PSM; DTC; ATC

3.2.2 Reputation Mechanisms in HetNets

Zekri et al. in [107] proposed a VHO management solution combining the use of reputation as a Quality of Experience (QoE) indicator for fast decision-making. This solution collects individual user experience and QoS value from the user and mobile device, respectively. Then, a cooperative indicator aggregates the individual score and computes a reputation value for Wi-Fi, WiMAX and UMTS networks. The performance results show that this solution provides better handover latency and throughput than other solutions. Later on, in [108] the authors proposed an enhanced IEEE 802.21 MIH based framework that integrates a Vertical Handover Management Engine (VHME) for vertical handover decision-making based on networks reputation. The authors make use of a large set of parameters that map the QoS and QoE to a network reputation value. The performance results show that the proposed framework can reduce decision delay and battery consumption compared to other existing VHO solutions.

Giacomini et al. in [109][110] proposed a reputation based VHO decision rating system by proposing the use of the grey model first order one variable (GM (1, 1)). Their proposed solution provides a quick and efficient prediction of reputation score for a target network in the handover decision making progress. QoS parameters like Bit Error Rate (BER), delay, jitter and bandwidth are used to calculate the reputation value for UMTS, WiMAX and WLAN networks. The proposed solution was evaluated through simulations using the network simulator NS2. WiMAX entities with the 802.21 protocol are based on the 'Seamless and Secure Mobility' project and UMTS entities via the Enhanced UMTS Radio Access Network Extensions (EURANE) module developed as part of the 'SEACORN' project. The results show that the reputation-based system can provide the mobile node time in advance to make a successful handover and thus experience higher QoS.

Trestian et al. [111] proposed a reputation-based network selection mechanism using game theory. The user-network interaction is modeled as a repeated cooperative game and the reputation of the network is computed based on user payoff. The proposed solution is based on individual user experience and the mechanism is integrated into an extended version of the IEEE 802.21 model.

Table 3.4 summarizes the works discussed in the context of the reputation mechanisms in HetNets in terms of description, parameters and functions.

Table 3-4 Feature of Reputation Mechanism in HetNets

Solution	Networks	Description	Parameters	Function
Reputation-VHO [108]	WLAN, WiMAX, UMTS	A VHO management solution combining the use of reputation as a Quality of Experience (QoE) indicator for fast decision-making.	Reputation Values, Handover decision delay, throughput, packets loss	Vertical handover initiation, Network Selection, Vertical handover execution
VHME [108]	Wi-Fi, UMTS	Vertical handover decision-making based on network reputation under MIH framework.	Handover decision delay, Power consumption	VHME: The policies repository, Reputation management block, Decision making block
GM (1, 1)-VHO [109][110]	UMTS, WiMAX, WLAN	Reputation based VHO decision rating system interaction with grey model first order one variable (GM (1, 1)).	Aggregated reputation score	GM(1,1) model
Game theory-Reputation [111]	WLAN	A reputation-based network selection mechanism using game theory and integrated into an extended version of the IEEE 802.21 model.	Throughput, energy consumption, quality utility, energy utility, average revenue	Weighted multiplicative method, Network Detection Manager, Profile Manager module, utility functions, cooperative game formulation

3.2.3 Discussions

Table 3.3 and Table 3.4 summarize major aspects of the reputation mechanisms discussed in the context of ad-hoc networks and HetNets, respectively, including the parameters employed and functions utilized.

When assessing the solutions proposed in the literature, it can be noted that among the limitations of these works are: 1) most of the related works on reputation mechanisms in Ad-hoc networks are simple judged based on trust and cannot fully be associated with the QoS; 2) the reputation mechanisms in HetNets are used in the decision making for the vertical handover; however it is not mentioned how the data of the parameters used in these mechanisms is collected and how the solutions cooperate with industry standards; 3) all reputation mechanisms are based on a single user case, and multi-user cooperation and reputation information exchange is not considered. In this context, our solutions propose a reputation mechanism that involves multiple users in parameter information gathering in order to provide better reputation values.

3.3 Multipath Transmission and Load Balancing

Another related direction consists of multi-path transmissions and load balancing solutions. Multi-path refers to multiple network paths and is a new approach in which some or all available device network interfaces are used concurrently to improve data throughput with various multipath supported transport protocols. Load balancing is an optimized method to balance traffic across multipath that can increase capacity and reliability of transmissions. Starting from the most recent multipath transmission protocol MPTCP which was published in 2013, this section discusses the state of the art in multipath transmission in wireless networks. Two major areas are considered: MPTCP-based solutions and other multipath solutions. Load balancing, an inevitable problem in multipath transmission, will also be discussed.

3.3.1 MPTCP-Based Solutions

Bonavanture et al. [14] gave an overview of the MPTCP protocol. Firstly, they point out that the TCP protocol developed more than 20 years ago is not suitable in current network environment, especially for traffic to/from mobile devices that contain multi-interfaces and data centers with redundant paths between servers. Secondly, in order to make use of the TCP infrastructure, MPTCP needs to operate over any internet path where TCP operates. In this context, the authors describe how MPTCP achieves the same ability as TCP operation. MP_CAPABLE option in SYN+ACK segment and MP_JOINTCP option in SYN segment shows how to create and add an MPTCP sub-flow connection. The sub-flow sequence number and data sequence number are involved to identify the packet delivery on different paths, and avoid the reacting from transmission through the middle-boxes. Thirdly, the congestion control of MPTCP is designed with three benefits: enables fairness like TCP congestion control, offers better performance than regular TCP, and utilizes the efficient paths to transmit more traffic. Finally, the paper presented an implementation of mobile device handover between WiFi and 3G network and MPTCP and application layer handover options were compared. The results show that MPTCP can get better performance with a smooth handover and always connects the network services. Additionally, testing results involving a data center show that MPTCP can gain better throughput than regular TCP.

Chen et al. [112] proposed an Energy-aware Multipath-TCP-based Content Delivery Scheme (eMTCP), eMTCP offloads the data stream between two network interfaces to balance the QoS and energy consumption on the mobile device in a heterogeneous wireless environment. Two components exist in this scheme: a Sub-flow Interface State Detector and a Decision Maker located at the upper transport layer of the mobile device. Simulations are based on Network Simulator 3 (NS-3) version 15 with LTE and WLAN scenarios, and the results show that eMTCP achieves 14% more energy efficiency in comparison with MPTCP and 66% higher quality in comparison with single-path TCP.

Rahmati et al. [113] proposed a solution to provide seamless TCP migration on the smartphone device without the access network supported. The solution consists of two components: Wait-n-Migrate and Resumption Agent. Wait-n-Migrate utilizes a new feature of mobile devices that are able to connect to multiple networks simultaneously. This mechanism is using monitoring flows to select the primary network interface, and when terminating individual TCP flows network interfaces are disabled The Resumption Agent is used to reduce the influence of network switching when Wait-n-Migrate terminates a flow for migration. By combined those two functions a new policy in deployed called AutoSwitch which attempts to handover the TCP flows from Wi-Fi to cellular when Wi-Fi coverage is dropped or becomes unreliable, and migrates the TCP flow back to Wi-Fi, when a reliable Wi-Fi connection becomes available. Based on the trace-based evaluation and iPhone-based implementation with real tests, the proposed solution can reduce user disruptions significantly.

Dugue et al. [56] proposed an MPTCP-based autonomic transport protocol. In this protocol, a priority driven load-balancing mechanism is estimated to give different priority to different kind of frames/packets of a video stream. The highest priority frame will be delivery by the highest throughput link. A dynamically plugging mechanism is built to monitor the network state for the decision module and decide which priority driven load-balancing mechanism needs to use. Simulation based on NS-2 shows the MPTCP using Priority-driven Load Balancing mechanism (MPTCP-LB) can gather better PSNR than the classic MPTCP when the network state is poor and varies dynamically.

Based on this priority driven load-balancing mechanism for MPTCP, **Diop et al.** [114] extended it to propose the Ontology for MPTPCP's QoS-aware Mechanisms (OMQM). Two new mechanisms are introduced in this ontology system: Selective Discarding Mechanism, which works at the sending side for adaptation driven by data priority are;

Time-Constraint Partial Reliability Mechanism, which works at the receiving side for the adaptation driven by the time constraint.

Wu et al. [115] proposed a quAlity-Driven MultIpath TCP (ADMIT) scheme to stream high-quality mobile video in heterogeneous wireless networks. A utility maximization based Forward Error Correction (FEC) coding solution is used in this scheme to minimize the end-to-end video distortion, and an analytical framework to model video delivery quality over MPTCP and assist the reliability-aware flow rate allocation algorithm on multipath. H.264 video streaming-based emulations show how ADMIT offers benefits in terms of PSNR, delay and throughput.

Based on ADMIT, an Energy Distortion Aware MPTCP (EDAM) solution was proposed by **Wu et al.** in [116]. EDAM aims to enable the energy-efficient and quality guaranteed video streaming via MPTCP in HetNets. The energy-distortion can be captured by an analytical model in EDAM. Based on the utility theory, a novel video flow rate allocation algorithm used to minimize the energy consumption with certain video quality was proposed. Emulation results show that compared to other MPTCP based solutions, EDAM can reduce energy consumption and improve video PSNR.

Corbillon et al. [117] presented a video-content aware scheduler over MPTCP at the application layer. Two theoretical models are proposed to be used in this scheduler. The Omniscient Optimal model computes the optimal transmission of a given video by exploiting multiple paths. The Integer Linear Program model decides if and when a video unit is given to the transport layer. The main idea for this scheduler is to cancel the transmission of low-importance video units, and prioritize the transport of the most important ones. A trace file based MPTCP emulator evaluates the performance of the cross-layer schedulers in realistic configurations. The emulator results show that the cross-layer schedulers can offer benefits in terms of PSNR and Multiscale-Structural Similarity (MS-SSIM) metric compared in comparison with a “first in, first out” (FIFO) scheduler.

Deng et al. [118] performed a measurement study with the Multipath TCP implementation in the Linux kernel based on comparison with single-path TCP over LTE and WiFi. This study analyzed the traffic patterns and categorized applications as either short-flow dominated or long-flow dominated. The results show that for short flows, there is no significant benefit of using MPTCP, and selecting the right access network technology for the primary sub-flow is more crucial for high performance. In terms of long flows, using Multipath TCP increases the volume of data transferred, and the congestion control algorithm enhances this benefit.

Table 3.5 summarizes the works discussed in the context of the MPTCP-Based Solutions in terms of description, parameters and functions.

Table 3-5 Overview of MPTCP-Based Solutions

Solution	Networks	Description	Parameters	Functions
eMTCP [112]	LTE, WLAN	Energy-aware Multipath-TCP-based Content Delivery Scheme	Remaining energy, average throughput, energy efficiency, estimated battery lifespan, PSNR	Sub-flow Interface State Detector
Seamless TCP migration [113]	WLAN, 3G	Seamless TCP migration on smartphone without network support	Probability density function (PDF), cumulative distribution function (CDF)	Wait-n-Migrate, Resumption Agent, AutoSwitch policy
MPTCP-LB [56][114]	HetNets	MPTCP using Priority-driven Load Balancing mechanism	PSNR	Priority-driven Load Balancing mechanism; dynamically plugging mechanism; OMQM
ADMIT[115]	HetNets: WLAN, WiMAX, Cellular	Forward Error Correction (FEC) coding and flow rate allocation on MPTCP	PSNR, delay, throughput	FEC Redundancy Adaption, FEC Packet size Adaption, Reliability-Aware Flow Rate Allocation
EDAM[116]	HetNets: WLAN, Cellular	Energy Distortion Aware MPTCP solutions	Energy Consumption (power), PSNR	Utility-based Flow Rate Allocation, Congestion and retransmission control algorithm
Scheduler over MPTCP[117]	HetNets: WLAN, 3G	Video-content aware scheduler over MPTCP	PSNR, MS-SSIM	Omniscient Optimal model, Integer Linear Program model

3.3.2 Other Multipath Solutions

Rodriguez et al. in [119] proposed a commuter mobile access router infrastructure (MAR) which makes use of striping techniques in order to multiplex the data across multiple interfaces. Two main components make the MAR system work: a MAR router used in the wireless environment to utilize the diversity of different network providers and different network technologies, and a MAR server proxy set in the wired environment to enhance the upper layer network protocol transmission. Based on real tests with different scenarios in 3 different network providers and 3 network technologies (GPRS, 3G, WLAN) with both TCP and UDP protocols, the proposed solution can provide a several-fold improvement in comparison with a single network interface scenario.

Another protocol Stream Control Transmission Protocol (SCTP) was also published for multipath data transfer on multiple network interfaces devices. **Xu et al.** [120] proposed Quality-Aware adaptive Concurrent Multipath Transfer solution (CMT-QA), a quality-aware adaptive concurrent multipath transfer solution, which consists of three module: path quality estimation model (PQEM), data distribution scheduler (DDS), and optimal retransmission policy (ORP). Following the CMT-QA architecture illustrated in Fig.3.2, each path's data handling capability is monitored and analysed regularly by PQEM, DDS selects the qualified paths for concurrent data transfer, and ORP improves the efficiency

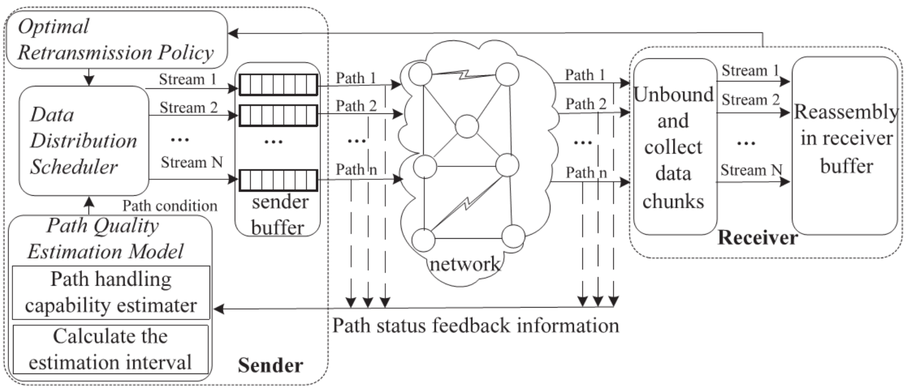


Figure 3-2 CMT-QA Architecture [120]

of retransmission packet by selecting the minimum transfer delay path. Simulations based on NS-2 involving both the FTP data transmission and real-time video delivery are consisted involving a heterogeneous wireless network scenario consisting of a 3G, a WLAN and a WiMax network. The results show that CMT-QA achieves better performance than classic CMT and CMT-PF (CMT during Path Failure).

3.3.3 Load Balancing

Lee et al. [121] proposed a route-selection algorithm for forwarding packets in the ad-hoc mode and a Vertical Handoff Decision (VHD) algorithm with applicability to 3/4G, WLAN or VANET heterogeneous wireless networks. The VHD algorithm enables balancing the overall load among all base stations and access points, and aims at maximizing the collective battery lifetime of the mobile nodes.

Luo et al. [122] introduced a Strengthen Gray Relative Analysis (SGRA) network selection algorithm based on load balancing. In order to provide effective load balancing within a heterogeneous wireless environment, the proposed algorithm uses the load state as the strengthen factor to resolution coefficient of GRA. Gray relational degree is calculated based on position, delay, band, and load state. Performance simulations are conducted in MATLAB considering UMTS and WLAN heterogeneous network scenarios. When compared with the normal GRA and a Multi-radio access selection & Load Balancing (MLB) algorithms, the simulation results show that the SGRA algorithm has an obvious effect on load balancing and lower blocking probability.

Xue et al. [123] proposed a suboptimal radio resource management (RRM) algorithm. Based on the previous optimal RRM algorithm, by analyzing and studying the RRM problem in the downlink of OFDMA involved LTE-WLAN heterogeneous environment, the proposed algorithm tries to maximize the system sum-rate under the proportional user rate constraint. The RRM simulation results are compared with the minimum normalized user rate, fairness index and sum-throughput from six different algorithms. The suboptimal

RRM algorithm achieves higher performance compared to the other RRM strategies and has much lower complexity.

Li et al. [125] proposed an improved Talmud algorithm with data envelopment analysis (DEA) models in real-time water allocation system. The Talmud algorithm provides an economics efficiency evaluation, and then DEA employs decision making units (DMU) to evaluate the efficiency and the demand of different economic sectors. Finally, the water allocation scheme makes a real time water resource allocation. The water use efficiency and the returns to scale classification are the main parameters used in the scheme. Further investigations are needed to demonstrate the potential benefits of this mechanism.

Balasubramanian et. al. [134] proposed a prediction-based offloading scheme labelled Wiffler to combine 3G networks and WiFi for reducing the total cost of data transfer. A history-based predictor estimates the amount of data that can be transferred using Wi-Fi by the deadline and the MN connects to the cellular network only when the Wi-Fi AP cannot transfer all the data within the application's tolerance threshold. The authors use several traces of driving around three cities using their own vehicles which have an 802.11b radio, a 3G data modem, and a GPS unit as part of the testbed. The proposed scheme is compared to OTSO, cellular-only and Oracle strategy solutions and the results show that Wiffler can significantly reduce the cellular usage up to 45% for a delay tolerance of 60 seconds.

Table 3-6 Overview of Other Multipath Solutions

Solution	Networks	Description	Parameters	Function
MAR [119]	GPRS, 3G, WLAN	Commuter mobile access router infrastructure	Bandwidth, Throughput	MAR session protocol, MAR proxy services
CMT-QA [120]	3G, WiMax, WLAN,	Quality-aware adaptive concurrent multipath transfer solution for SCTP-based data delivery over HetNets	RTT, Loss rate, Throughput, PSNR, VQM, SSIM,	PQEM, DDS, ORP

Table 3-7 Overview of Load Balancing Solutions

Solution	Networks	Description	Parameters	Function
VHD [121]	VANET, 3/4G, WLAN, Ad-Hoc	A Vertical Handoff Decision (VHD) algorithm	Bandwidth, power consumption, RSS, etc.	VHD
SGRA [122]	UMTS, WLAN	A Strengthen Gray Relative Analysis (SGRA) network selection algorithm based on the load balance.	Position, delay, band, and using load state	SGRA
Suboptimal RRM [123]	LTE, WLAN	Suboptimal radio resource management (RRM) algorithms.	Normalized user rate, fairness index and sum-throughput.	Suboptimal RRM
Talmud Algorithm [125]	Water resource allocation	An improved Talmud algorithm and data envelopment analysis (DEA) models combined water allocation scheme.	Water uses efficiency, returns to scale classification.	Talmud algorithm, DEA models

3.3.4 Discussions

Table 3.5 and Table 3.6 summarize features of the multipath transmission involving MPTCP and other mechanisms, respectively. The parameters employed, network scenarios and functions utilized are described in the table. When analyzing existing work, it can be observed that there are some limitations of these works including most of the related works on MPTCP-based solutions do not consider the handover possibility on each sub-TCP connection, and most of the related works on MPTCP-based solutions are not deployed even in simulations. A survey [124] on network-layer multipath over 260 solutions shows that only 1% of them have performed implementations in the real world.

Table 3.7 summarizes the features of load balancing solutions increasing parameters employ and function utilized. Four of these load balancing mechanisms are complex to build in a real environment. In this context, our solution tries to utilize each interface with an individual handover process and uses a simple load balancing mechanism for multipath transmissions.

3.4 Adaptive 3D Video Delivery

The previous works on stereoscopic 3D adaptive video transmission in dynamic network environments focus on two major mechanisms: one involving adjusting transmission process to match network conditions use innovative methods to drop not-so-important packets from one view or both views of stereoscopic 3D video to adapt the transmission data rate, the other gives different protection levels to different content from the stereoscopic 3D video to maintain the transmit result at high quality.

Gurler et al. [126] proposed a cross-layer adaptive stereoscopic 3D video streaming framework based on simulcast scalable video coding (SVC) combined with asymmetry coding. This framework reduces the loss rate for key packets by adapting the stereo video rate to the available network rate, in order to achieve the better perceived 3D quality with best scaling option depending on the bitrate/PSNR. A rate estimation method based on periodic packet loss feedback from the client is also present in the paper and is used with the TCP/DCCP protocols. Emulation results show that scalable coding of only one view achieves better compression efficiency than scaling both views, that means by employed the asymmetric encoding the proposed simulcast solution can gather better perceived visual quality.

In [127] the authors differentiated from the mechanism introduced in [126] by extending the use of the 2D SVC adaption framework, and proposing a transparent user-space module-Media Aware Network Element (MANE). MANE runs as a transparent proxy for low delay filtering of scalable video streams, and could work on any existing topology. This mechanism of MANE is to select the packet of enhancement information from both views to drop in order to adapt the transmission data rate to network conditions. Double Exponential Smoothing method is used to forecast the transmission data rate. The performance evaluation involving the RTP protocol shows that the MANE scheme can gain 10 dB PSNR in comparison with no MANE scheme.

The work in [128] focused on the color plus depth 3-D video and proposed a joint source channel coding scheme (JSCC) for depth image-based rendering (DIBR) based 3D video coding. The proposed scheme works in a WiMAX-based communication channel and by investigating the optimum coding performance from various sources and channel coding rates, the optimum bit allocation combination for color plus depth stream sequences can be found. The simulation results show that the quality of a 3D video is dominated by the quality of the color stream.

In [129], authors proposed an unequal error protection scheme (UEP) based on hierarchical quadrature amplitude modulation (HQAM) for 3-D video transmission. The proposed UEP scheme follows the approach of [128] as. In order to achieve high quality 3D video, more protection is assigned to the color sequence rather than to the depth component. When compared with the conventional equal error protection scheme (EEP), the simulation results show that UEP can gain up to 5dB PSNR benefit.

3.4.1 Discussions

Table 3.8 summarizes major aspects of adaptive 3D video delivery solutions, including parameters employed and function utilized. All the transmission process of stereoscopic 3D video delivery solutions was using a single path and the 3D video quality is evaluated by PSNR. Our work considers multipath transmissions and involves a 3D-specific quality metric to evaluate the 3D video stream quality level.

Table 3-8 Overview of Adaptive 3D Video Delivery

Solution	Networks	Description	Parameters	Function
Adaptive stereoscopic 3D [126]	LAN	A cross-layer adaptive stereoscopic 3D video streaming framework based on SVC combined with asymmetry coding	PSNR, bitrate,	SVC, asymmetry coding
MANE [127]	LAN	A transparent user-space module-Media Aware Network Element (MANE) to increase the PSNR on both view of stereoscopic 3D video	PSNR, bitrate, throughput, delay	MANE, SVC, The Double Exponential Smoothing mechanism
JSCC-3D [128]	WiMAX	A joint source channel coding scheme (JSCC) for depth image-based rendering (DIBR) based 3D video coding.	PSNR	DIBR, JSCC
UEP [129]	Hierarchical 16-QAM	An unequal error protection scheme (UEP) based on hierarchical quadrature amplitude modulation (HQAM) for 3-D video transmission	PSNR	DIBR, UEP

3.5 Chapter Summary

This chapter presented the state of the art in the area of network selection strategy in HetNets, reputation mechanisms, multipath transmission and load balancing, and adaptive 3D video delivery techniques with main emphasis on network selection and multipath transmission solutions.

The chapter included a comprehensive survey of current research on network selection strategies which were categorized into: standards-based vertical handover solutions, MADM-based network selection solutions, and other solutions. The existing reputation-based solutions for Ad-hoc and heterogeneous wireless environments are also discussed. In terms of multipath data transmission the chapter introduces a classification of the network multipath approaches in two main categories: MPTCP-based and other multipath solutions, respectively. Load balancing for multipath transmission is also discussed and four different solutions are presented. At last, current adaptive 3D video delivery techniques were introduced in this chapter.

However, when assessing the related works on network selection, it could be concluded that there are still several limitations of these works. They can be summarized as follows:

- Most of the related works on network selection are proposed as frameworks based on either industry protocols/standards or MADM methods, but do not utilize both approaches in conjunction.

- Most of the related works on reputation mechanisms in network selection are simple assessed based on trust and not on measurable metrics which can be easily associated with QoS.

- In majority of the related works on network selection it is not mentioned how the data of the parameters used in the proposed mechanisms is collected and how the solutions employ industry standards.

- All of the network selection works are based on single user case, and no multi-user cooperation is considered.

In this context, our proposed solutions LNS and RANS utilize the industry protocols/standards and a MADM method to provide better reputation value for network selection. They consider parameter information gathering from both users and network operators and employ a multi-user approach.

By analysing existing work on multi-path transmissions, it can be observed that there are some limitations of these works including the facts that the related works on MPTCP-based solutions do not consider the handover possibility on each sub-TCP connection, and most of them are not deployed even in simulations. Furthermore, when assessing the related works on adaptive 3D video delivery solutions, all the related works on stereoscopic 3D video delivery solutions were using a single path and the received 3D video quality evaluation is performed based on PSNR (which has known accuracy limitations) only.

In this context, in our solution NRQ-3D the MPTCP protocol is used for transmitting 3D video content and different 3D video components are delivered via separate sub-MPTCP flows and are synchronized at the receiver. This solution utilizes each interface subject to individual handover processes and uses a simple load balancing mechanism for multipath transmissions. The No-reference Video Quality Metric for 3D video quality assessment (NVQM) is used in conjunction with PSNR to evaluate the 3D video quality level.

The related works for this thesis are included in this chapter, motivating the design of the proposed system architecture which is presented in the following chapter.

CHAPTER 4: Proposed System Architecture

Abstract

In this chapter, the system architecture and details of the proposed algorithms are presented. The general idea and an overview of the overall solution based on the three major contributions to enhancing state of the art are described first. Next, based on current research works, the motivation for the three main standards used is presented. Next, the detailed description of the three major contributions of this thesis is offered: (1) Location-aware Network Selection mechanism in heterogeneous wireless networks (LNS) which eliminate the low feasibility network and track the high feasibility network for the mobile user based on the existing network performance-related information and mobile user location and speed; (2) Reputation-oriented Access Network Selection mechanism (RANS) which selects the best reputation network for the mobile user in terms of the trade-off between QoS, energy consumption and monetary cost; (3) Network Reputation-based Quality Optimization of Video Delivery in heterogeneous networks (NRQOVD) which uses the reputation mechanism to enhance the video content quality by employing multipath delivery or adaptation, in this thesis we choose the stereoscopic 3D video as specific video content to demonstrate this NRQOVD solution and present it as Network Reputation-based Stereoscopic 3D Video Delivery (NRQ-3D).

4.1 Introduction

The mass-market adoption of high-end mobile devices and increasing amount of video traffic has led the mobile operators to adopt various solutions to help them cope with the outstanding increase in mobile broadband data traffic, while also ensuring high Quality of Service (QoS) levels to their services. Deploying small-cell base stations within the existing macro-cellular networks and offloading traffic from the large macro-cells to the small cells is seen as a promising solution to increase capacity and improve network performance at low cost. Parallel use of diverse technologies is also employed. The result is a heterogeneous network environment (HetNets), part of the next generation network deployments

Due to innovation and advances in wireless network and mobile device technologies, the high performance CPU/GPU and high resolution screens of mobile devices could support high-definition video display and transmission via high transmission capacity wireless networks. Meanwhile, expected user experience on video service quality of wireless networks is increasing and there is an increasing desire to realize the "Always Best Connected" paradigm in todays' heterogeneous wireless network environment.

In this context, this thesis makes a step forward and advances towards the "Always Best Experience" paradigm. This paradigm considers mobile users seamlessly roaming in the HetNets environment, illustrated in Fig. 4.1, which supports ubiquitous connectivity and enables very good quality of rich mobile services anywhere and anytime. However this is highly challenging, mostly due to the heterogeneity of the selection criteria, such as: application requirements (e.g., voice, video, data), different device types and with various capabilities (e.g., smartphones, netbooks, laptops), multiple overlapping networks using diverse technologies (e.g., Wireless Local Area Networks (IEEE 802.11), Cellular Networks Long Term Evolution (LTE)) and different user preferences. In fact the mobile users are facing a complex decision when they need to dynamically select the best value network to connect to in order to get the expected "Always Best Experience".

According to Cisco, by 2019, 97% of the total mobile data traffic will be generated by the mobile-connected devices and by 2016, more than half of this mobile traffic will be offloaded from the cellular network to Wi-Fi and femtocells [1]. In this way, by transferring some of the traffic from the core cellular network to Wi-Fi or femtocells at peak times or key locations (e.g., home, office, public HotSpots) the mobile operators can accommodate more mobile users and the users can avail of a wider service offering.

At the mobile user side, the mobile devices have become affordable and powerful with improved CPU, graphics and display contributing to the increase in user demands. Due to the growth of the video content usage, such as IPTV, video on demand (VoD) and 3DTV, which is estimated to reach 72% of the world's mobile data traffic by 2019 [1], ensuring a seamless experience at high quality levels to the end-user has become a challenge. Furthermore, it is known that video-based applications have strict QoS requirements representing the most power-hungry applications. One of the main impediments to progress is the battery lifetime of the mobile device as the battery life has not evolved in-line with the processor and memory advances, becoming a limiting factor.

Figure 4-1 Heterogeneous Wireless Environment – Example Scenario

In this context, mobile users are accessing multimedia content on the move and via heterogeneous networks, they regularly require network selection and handover procedures in order to maintain their seamless connectivity to the Internet. Additionally to supporting the best user perceived video quality level and maintain the battery life longer, the selection of the most appropriate network with best performance is required.

This thesis presents three major contributions to solve the problems described above: 1) Location-aware Network Selection mechanism in heterogeneous wireless networks (LNS) eliminate the low feasibility network and tracking the high feasibility network for the mobile user based on the existing network performance-related information and mobile user location and speed; 2) Reputation-oriented Access Network Selection mechanism (RANS) which selects the best reputation network from the available networks for the mobile user which provides the best trade-off between QoS, energy consumptions and monetary cost. The focus is on the user-network interaction, where we define a network reputation factor obtained as a result of the user's previous experience with the networks. The network reputation factor is then integrated into the network selection decision in order to sustain cooperation between the user and the network; 3) Network Reputation-based Quality Optimization of Video Delivery in heterogeneous networks (NRQOVD) which uses the reputation mechanism to enhance the video content quality via multipath delivery or adaptation.

In this chapter, the system architecture upon which the three contributions are built is described. The detailed descriptions and algorithms of each contribution are presented in the following sections.

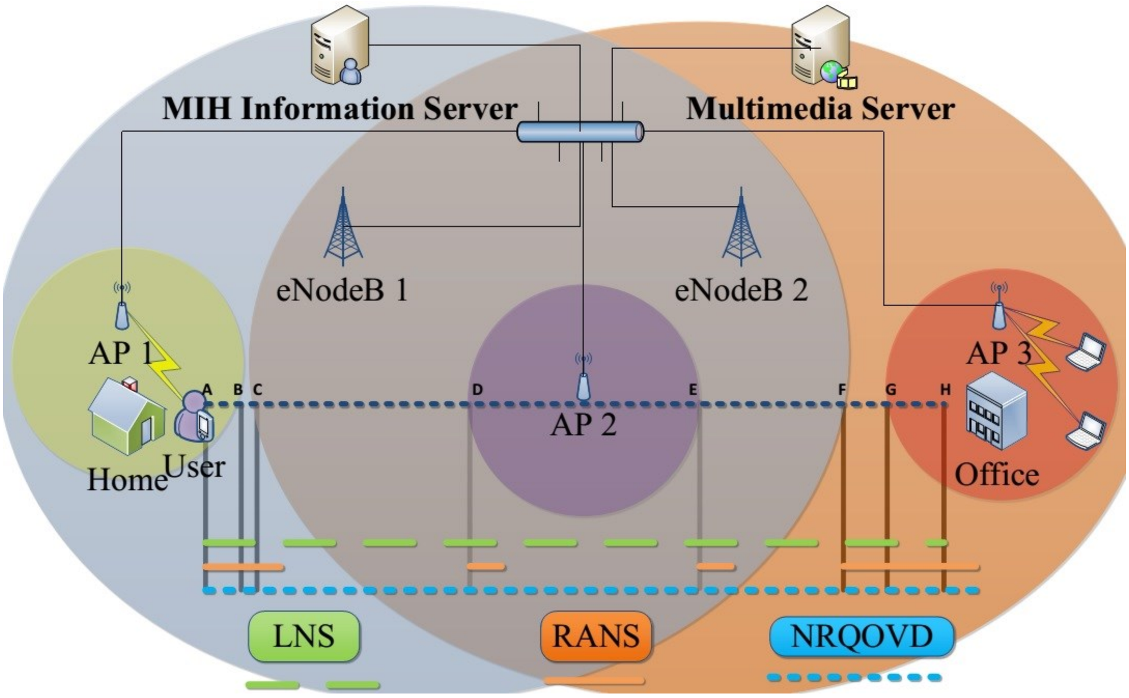


Figure 4-2 Heterogeneous Wireless Network Environment Scenario

4.2 Proposed System Architecture

Fig.4.2 shows a scenario inspired by the daily life of a mobile user, who is going from home (point A) to office (point H), wants to access multimedia services (e.g., watching the news, music video clips). She/he has access to a number of available wireless networks of different types (e.g. WiFi, LTE) via his multi-interface mobile device. While on the move, the user passes through the coverage area of several different radio access technologies which trace the location and network information by employing LNS and executing adaptation/offloading by using NRQOVD. The network handover decision has to be made at the following points: B, C, D, E, F and G and the proposed RANS function is used.

The proposed system block-level architecture which deploys support for LNS RANS & NRQOVD is illustrated in Fig 4.3. The system is distributed and consists of three main components: Mobile Nodes (MN), a MIH Information Server and one or more Multimedia Servers. The architecture is built on top of the IEEE 802.21 MIH standard,

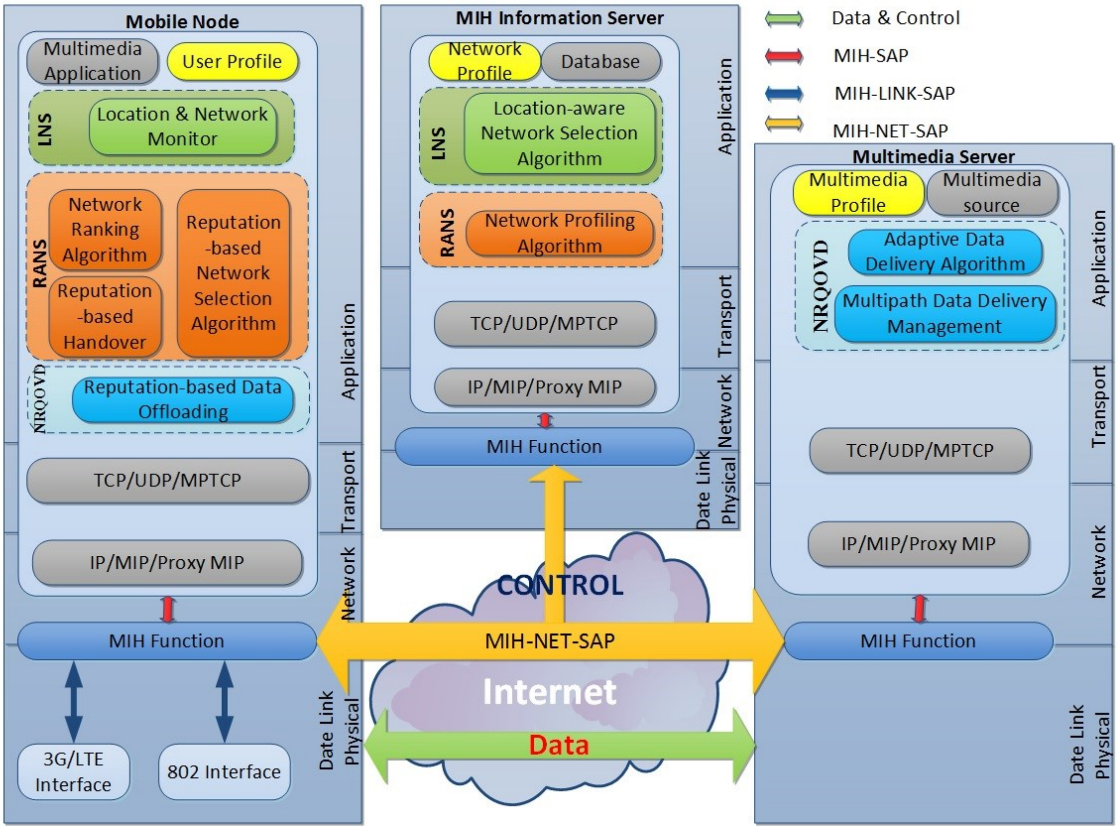


Figure 4-3 System Architecture

thus system components are MIH-enabled entities. The detail description of each of these components is presented next. In this description, a single multimedia server was considered, but equally a server farm a cloud could be envisaged.

In order to perform network selection, the MN needs the list of candidate networks and also their associated quality levels. IEEE 802.21 MIH provides a mechanism to support gathering and exchanging information between various candidate networks, the MIH Information Server and the MN. The MIH framework defines a cross-layer MIH function (MIHF) as a logical component between the network layer and the link layer [8]. Each of the MIH-enabled entities contains a cross-layer MIHF. This function provides Service Abstraction Points (SAP) acting as an abstract interface between a service provider and a user entity. User entities at higher layers employ the MIH-SAP to control or to monitor the link-layer entity and the MIHF uses the MIH-LINK-SAP as an interface together with the link layer to translate the information received from the MIH-SAP. The remote MIHF entities use the MIH-NET-SAP to exchange the information with the MIHF [9].

At the client side, MN integrates components related to LNS, RANS and NRQOVD. The Location & Network Monitor (LNM) employed in LNS gathers the user location information, destination information and network basic information. Network Ranking Algorithm (NRA) applied in RANS computes the network ranking value by employing a utility function and based on the QoS parameters, which are related to the network performance. The reputation-based Network Selection mechanism (RNS) used in RANS, stores the list of the candidate networks together with their reputation values. Reputation-based Handover (RH) utilized in RANS selects the network with the best reputation from the candidate networks and executes the handover via the MIH protocol. Reputation-based Data Offloading (RDO) executed in NRQOVD offloads the data from the low-reputation network path to high-reputation network path during the mobile device using the multipath protocol.

The MIH Information Server integrates sub-functions from both LNS and RANS. The Network Profiling Algorithm (NPA) working in RANS stores information about the network performance, and is based on the joint collaboration of the users within the network, thus the MIH Information Server gathers the performance information feedback from multiple users within the network and computes the performance reputation factor for that particular network. A set of suitable target networks is provided by Location-aware Network Selection Algorithm (LNSA) via LNS based on user's location, destination and route prediction information.

The Multimedia Server is an entity which delivers media data to MNs and receives reports from the MIH Information Server about the state of the MN and networks. The Multimedia Server side contains two components related to NRQOVD. Adaptive Data

Delivery Algorithm (ADDA) running some adaptation schemes to optimize the quality during the data delivery for various network conditions. Multipath Data Delivery Management (MDDM) works as a path controller for the reputation-based multipath data delivery. In this system, Quality-Oriented Adaptation Scheme (QOAS) [130] is employed as the part for high quality Adaptive Data Delivery Algorithm (ADDA). The control information exchange from MN to the Multimedia Server follows the 802.21 MIH protocols. The packet structure includes an additional field with the score which grades the video's quality of delivery. The Multipath Data Delivery Management (MDDM) runs to gather the feedback from the MN with network reputation information and handover/adaptation/offloading message to change the network path for handover/offloading or running the ADDA for adaptation.

4.3 Handover Mechanism and Data Structure

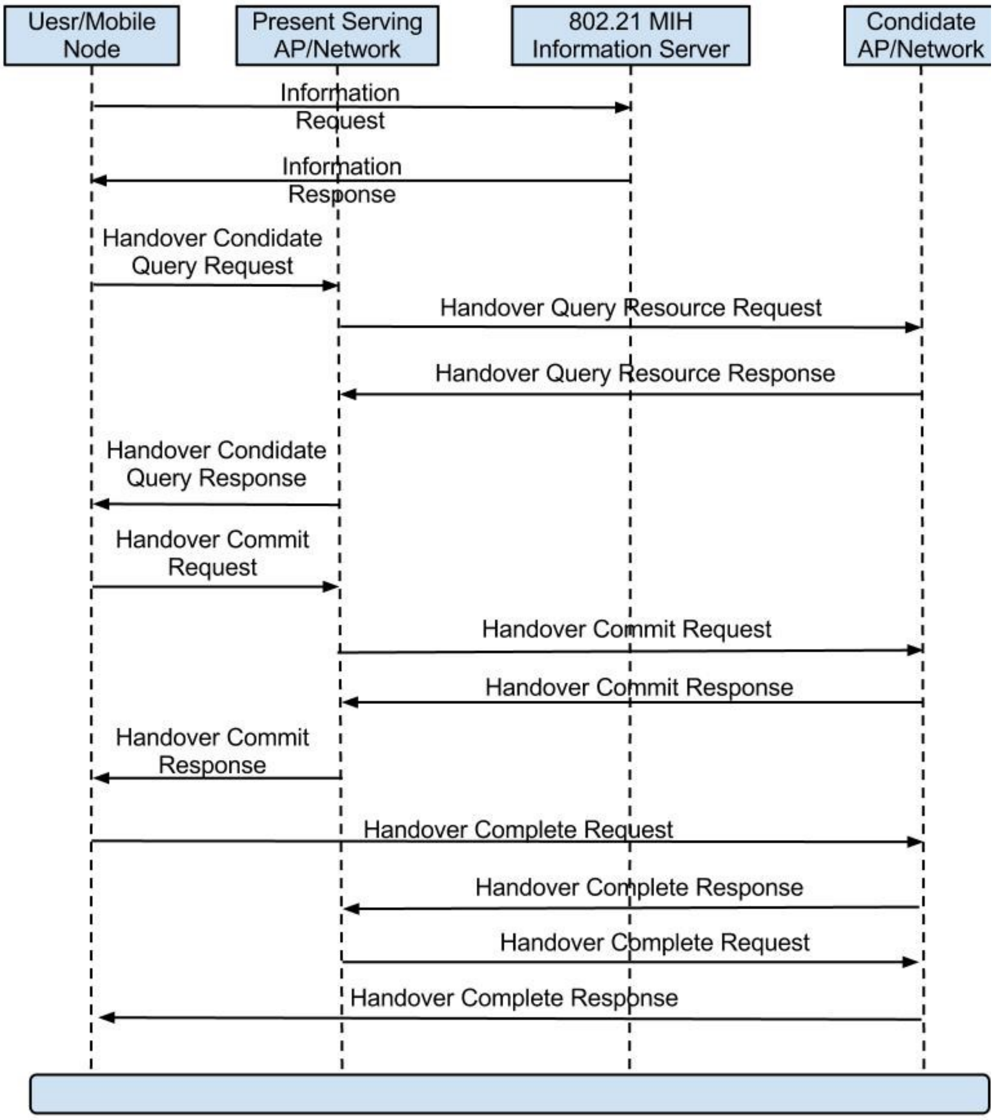


Figure 4-4 MIH Handover Mechanism

In the proposed solution, MN uses the MIH-NET-SAP to send an information request or to report to the MIH Information Server via the current serving network. Then the MIH Information Server sends a response back to the MN. In the meanwhile, the MIH Information Server sends a report to the multimedia server. Based on the information contained in the response, MN executes the network selection algorithm, chooses the best candidate network and executes handover. Fig.4.4 shows the detailed handover process.

In the meanwhile, the Multimedia Server receives a user report from the MIH Information Server and performs adaptive data delivery according to the report. Once the handover process is complete, MN receives video data from the Multimedia Server via the new network. The detailed description of the data structure of each block and function is presented below:

4.3.1. Mobile Node

MN is an entity requesting and receiving multimedia data capable of making network selection decision and executing handover/offloading. In the context of RANS and LNS, MN is involved in a dual request-report process described next:

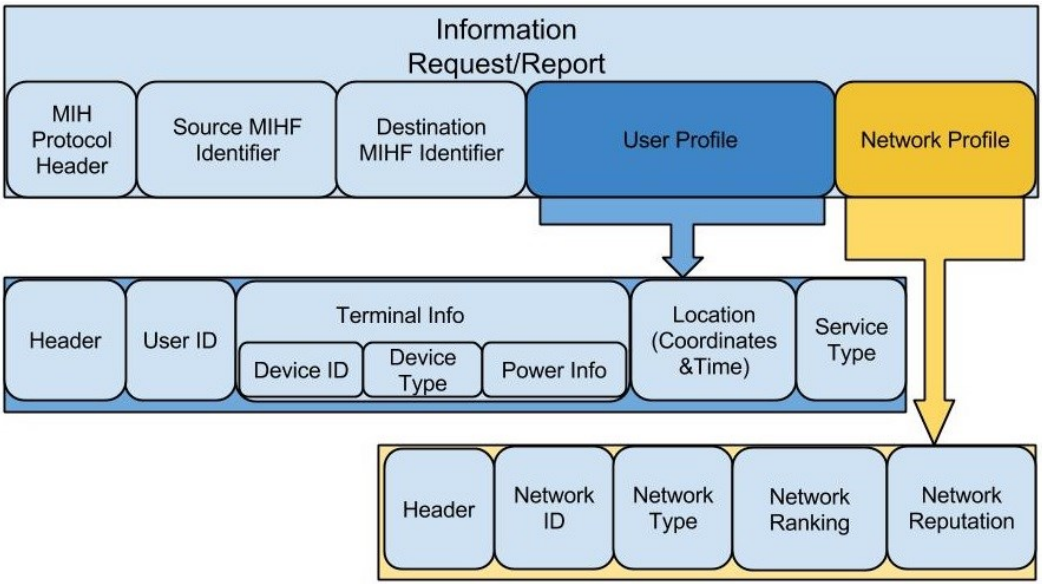


Figure 4-5 Structure of the Information Request/Report

1. Requesting Process

For the purpose of network selection, MN sends an information request to the current serving attachment point when it initiates a connection with the current serving network. The current serving network forwards this information request to the MIH information server.

The information request follows the 802.21 MIH protocol packet structure, and contains at least three fields: the MIH Protocol header, Source MIHF Identifier and Destination MIHF Identifier. In this thesis, two extra specific fields are added as illustrated in Fig.4.5: User Profile and Network Profile to describe both user and network characteristics.

The User Profile field consists of five data structure components:

- Header structures identify that the information of this field is about user profile;
- User ID structures identifies the user by a unique user ID number;
- Terminal Info contains the information of terminal device, using a unique device ID number in Device ID to discriminate with other device, a classification information about the device in the Device type to classify the mass device and the remaining energy information in Power info;
- Location contains a coordinates information with timestamp;
- Service Type contains classification information about the current service.

The Network Profile field consists by five data structure components:

- Header structures identifies that the information in this field is about network profile;
- Network ID identifies the network by a unique network ID number;
- Network Type contains classification information about the current network type.
- Network Ranking contains the ranking value for this network.
- Network Reputation contains the reputation score for this network.

2. Reporting Process

In the context of NRA and LNM, MN sends user reports together with every information request. Additionally, user reports are generated and sent to the MIH

Information Server regularly (the frequency *i*, determined by the scenario). The user report has the same structure as the information request except that in the User Profile field, the terminal information is not included.

4.3.2. MIH Information Server

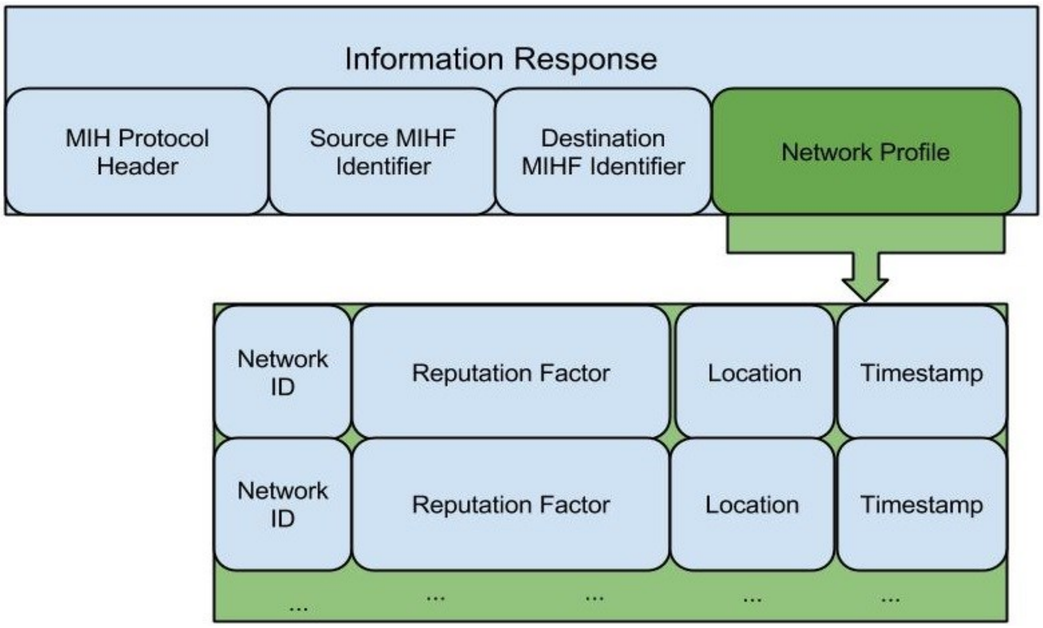


Figure 4-6 Structure of the Information Response

The MIH Information Server receives information requests, and user reports from MNs and network reports from the candidate networks using MIH-NET-SAP. On receiving any information, the MIH Information Server sends it from MIHF to the upper-layers in charge of network selection-related data storage and processing, and immediately responds to the MN. The information response extends the 802.21 MIH protocol with one additional field Network Profile as illustrated in Fig.4.6. This field lists a subset of candidate networks along with values representing their reputation and time instances at which user localization prediction is fulfilled. This network profile field consists of four data components as follows:

- Network ID identifies the network by a unique network ID number;

- Reputation Factor contains the *Reputation Utility* and *Stabilization Utility* score for the candidate networks;
- Location contains the coordinate information of base station;
- Timestamp contains time information about the response time.

Once the MIH server receives any request or report from user, the NPA and LNP will gather, produce and update the user and network information. All the information about users and networks is stored in a specific database. The MIH Information Server data structure is shown in Fig.4.7, Fig.4.8, and Fig.4.9.

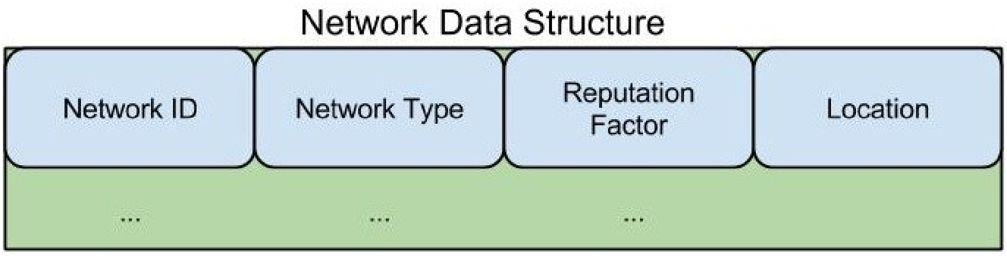


Figure 4-7 Structure of Network Data Saved in MIH Information Server

The network data consists of five data components:

- Network ID identifies the network by the unique network ID number;
- Network Type contains classification information about the network;
- Reputation Value contains the reputation value of the network;
- Location contains the coordinate information of base station;

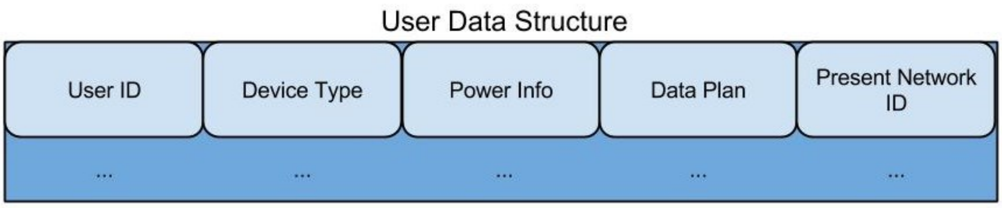


Figure 4-8 Structure of User Data Saved in MIH Information Server

The user data consists of five data components:

- User ID identifies the user by a unique user ID number;
- Device Type contains classification information about the device;
- Power Info contains remaining energy information of device;
- Data Plan contains data plan information and the network operator charge information;
- Present Network ID identifies the current serving network by its unique network ID number.

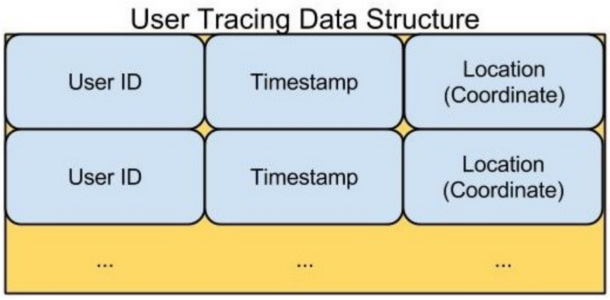


Figure 4-9 Structure of User Tracing Data Saved in MIH Information Server

User tracing data consists of three data components:

- User ID identifies the user by a unique user ID number;
- Location contains the coordinate information when user sends the request or report;
- Timestamp contains time information when user sends the request or report.

4.4 Chapter Summary

This chapter presents the system architecture on which the three proposed solutions are deployed: the Location-aware Network Selection mechanism (LNS), Reputation-oriented Access Network Selection mechanism (RANS) and Network Reputation-based Quality Optimization of Video Delivery scheme (NRQOVD). The placement and the principle of each contribution with respect to the MIH handover mechanism with the detailed data structures are then described. Details of each contribution and test result analysis will be described in the following chapters.

CHAPTER 5: Location-aware Network Selection Mechanism in HetNets

Abstract

In this chapter, the Location-aware Network Selection Mechanism in Heterogeneous Networks (LNS) as the first thesis contribution is described. The heterogeneity of the wireless network environment offers the possibility to the mobile user to select between several available radio access network technologies. However, selecting the network that enables the best connectivity is not trivial given that in general the network characteristics vary widely, not only in time, but also depending on the user location within each network. In this context, this chapter proposes a user location-aware network selection solution which aims at improving content delivery in such a heterogeneous wireless network environment by selecting the best network. Based on the existing network performance-related information and mobile user location and speed, the network that offers the best support for content delivery along the user path is selected as the target network and the handover is triggered. Simulation results show that the proposed solution improves the content delivery quality in comparison with the case when other network selection mechanisms were employed.

5.1 Introduction

The ABC vision was introduced by ITU and its aim was to connect different radio access networks via flexible core networks in order to provide seamless, transparent and QoS enabled connectivity to the mobile users. This vision includes the scenario with a variety of Radio Access Technologies (RAT) that work together in order to provide a global wireless infrastructure in which the end users will benefit from an optimal service delivery via the most suitable wireless network that satisfies their interests [10]. Moreover, the IEEE 802.21 MIH standard is also released as a media-independent framework to ensure QoS and to facilitate handover between different RATs. However, the standard only offers support for the handover process without defining a network selection algorithm, which is a major part of the handover process.

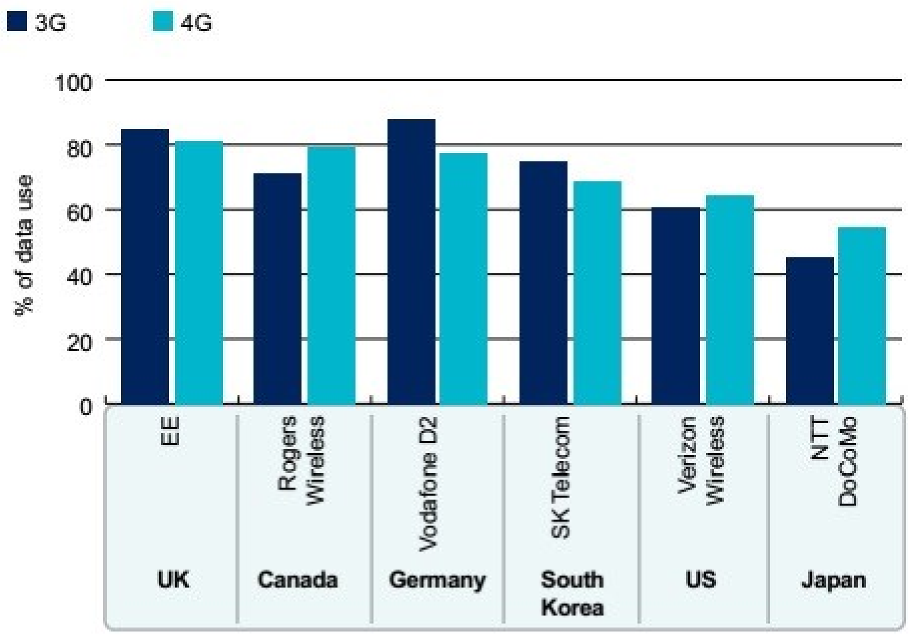


Figure 5-1 Percentage of Data Use via Wi-Fi for 3G and 4G Android Smartphone Users, Selected Operators, Apr-13 [131]

From a study on smartphone user behaviors by analysis of data-usage patterns [131], we know that the most important network selection for the mobile user is between Wi-Fi and cellular networks. Fig.5.1 illustrates that more than 50% of total traffic on Android smartphones is transmitted via Wi-Fi networks. A cisco report estimates that Wi-Fi networks will handle more than 53% of the whole network traffic in the next five years [1].

In order to cope with the latest significant increase in data traffic, mobile data offloading from cellular networks to Wi-Fi emerges as an important solution for network operators in the current heterogeneous wireless networks environment. Major inter-networking offloading approaches are illustrated in Fig.5.2.

The on-the-spot offloading (OTSO) scheme [132], as the most deployed mobile data

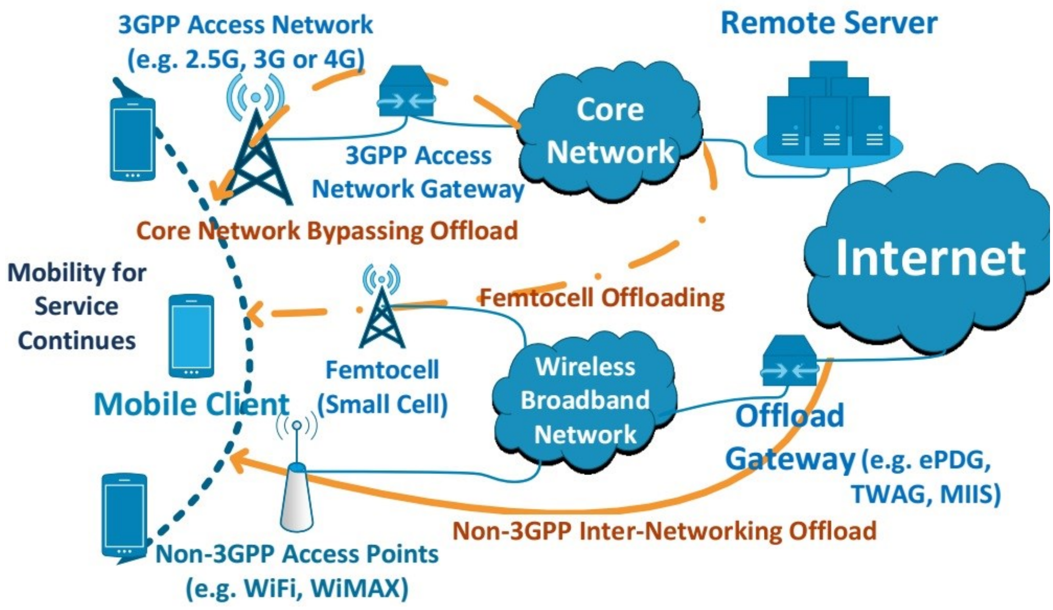


Figure 5-2 Data Offloading Scenario in Heterogeneous Wireless Networks

offloading solution on a mobile device, is simple but inefficient. It chooses the Wi-Fi network to offload the data traffic from cellular network whenever possible, even when the Wi-Fi network is highly loaded. So, there is a need to propose an efficient network selection mechanism for mobile data offloading which considers user mobility, cost and QoS, contributing towards achieving the ABC paradigm.

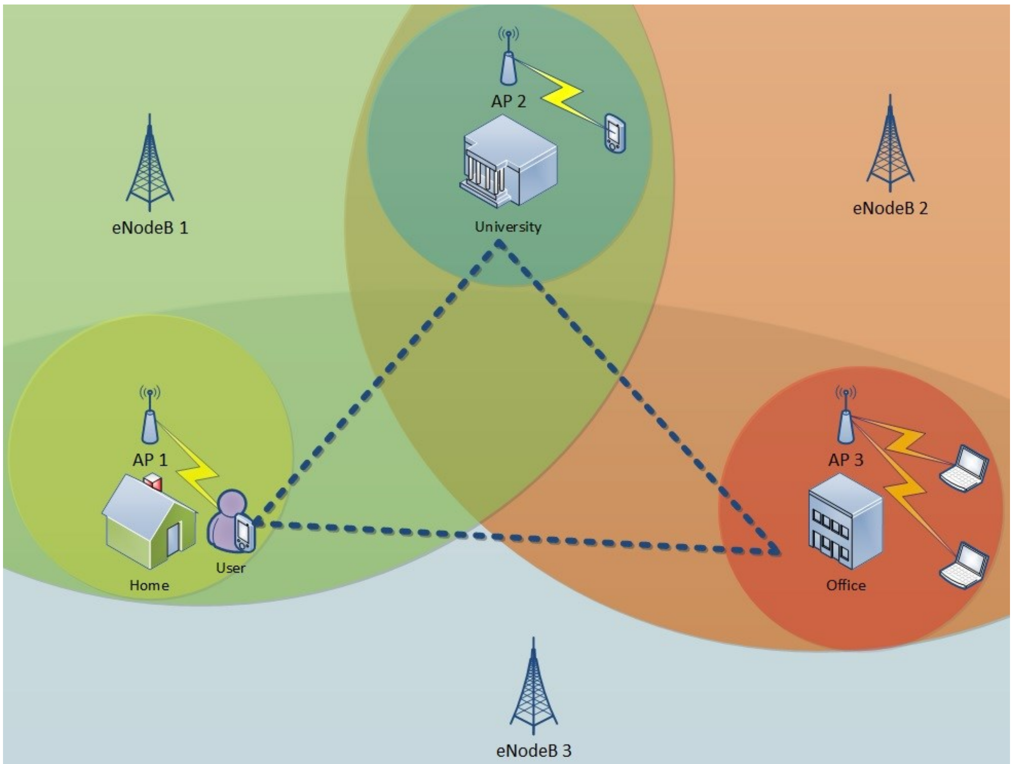


Figure 5-3 Location-aware Network Selection Scenario

Considering the case a mobile user uses a smart phone, which supports various radio access network technologies (e.g. WLAN, UMTS, LTE, etc.) and multi-interfaces. As Fig.5.3 shows, in his/her path from home to office/university, the mobile user wants to know the best suitable networks in the heterogeneous wireless networks environment, which includes WLAN and LTE networks, in order to keep multimedia services running. In this environment, for example, a Mobile Node (MN) (e.g. a smart device) can be located in a home Wireless Local Area Network (WLAN) coverage area. Following user mobility, MN faces the choice of selecting between several WLAN and LTE access networks and in different destination locations (e.g. Office or University) the most suitable networks vary. The MN has to select the most appropriate network in order to continue to receive the services at high quality level

In this context, in order to enable high quality data delivery independent from the network attached to, we propose the Location-aware Network Selection mechanism for

MN. The proposed Location-aware Network Selection mechanism selects the best network among various heterogeneous wireless networks to ensure high quality data delivery in the heterogeneous wireless network environment.

5.2 LNS Architecture

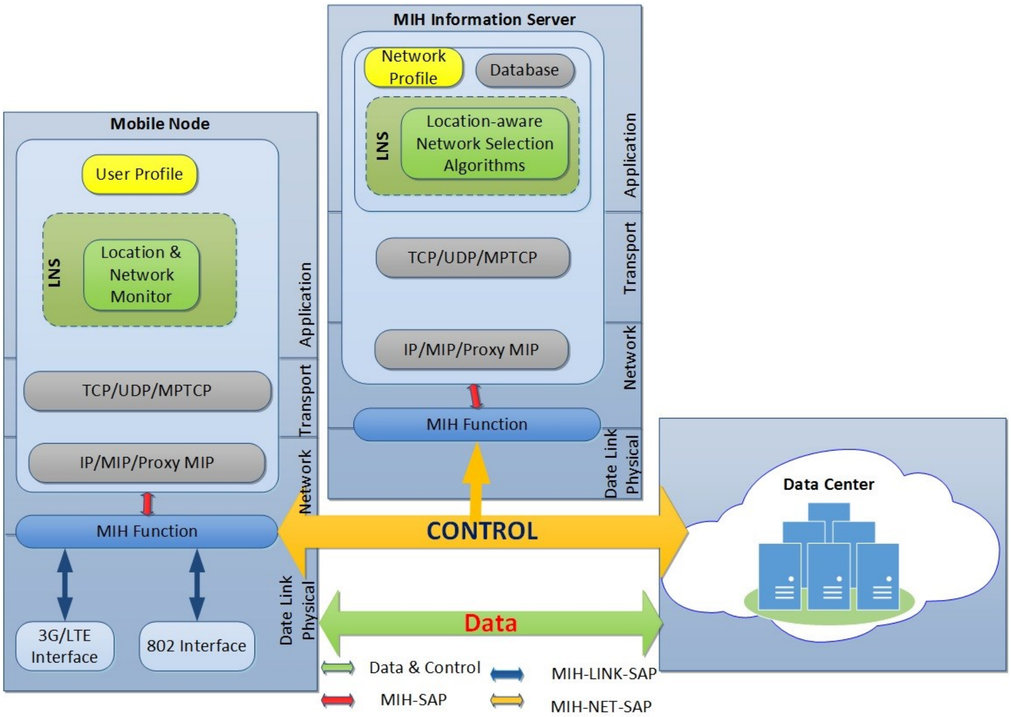


Figure 5-4 System Level Architecture

Fig.5.4 illustrates the proposed LNS system level architecture based on the TCP/IP protocol stack model. LNS is distributed and consists of a server side component (MIH Information Server) and two client side components (Mobile Node and Data Center). The LNS architecture is built on top of the IEEE 802.21 MIH standard, thus all three system components are MIH-enabled entities.

Mobile user location information with a timestamp and available network information is regularly monitored by the Location & Network Monitor (LNM), and sent it by the information request/report to MIH Information Server (MIHIS) via the current serving network. Some novel Location-aware Network Selection Algorithms (LNSA) is introduced which uses the previous location information and their different timestamps to predict the MN mobility at the MIH Information Server. Based on the existing network base station location-related information stored in the MIH Information Server and uploaded mobile user's location information and available network information, the network that offers the suitable support for data delivery along the user's path is selected in the available network list as the proposed handover target by LNSA. The detailed sequence diagram of this location-aware network selection mechanism is shown in Fig 5.5.

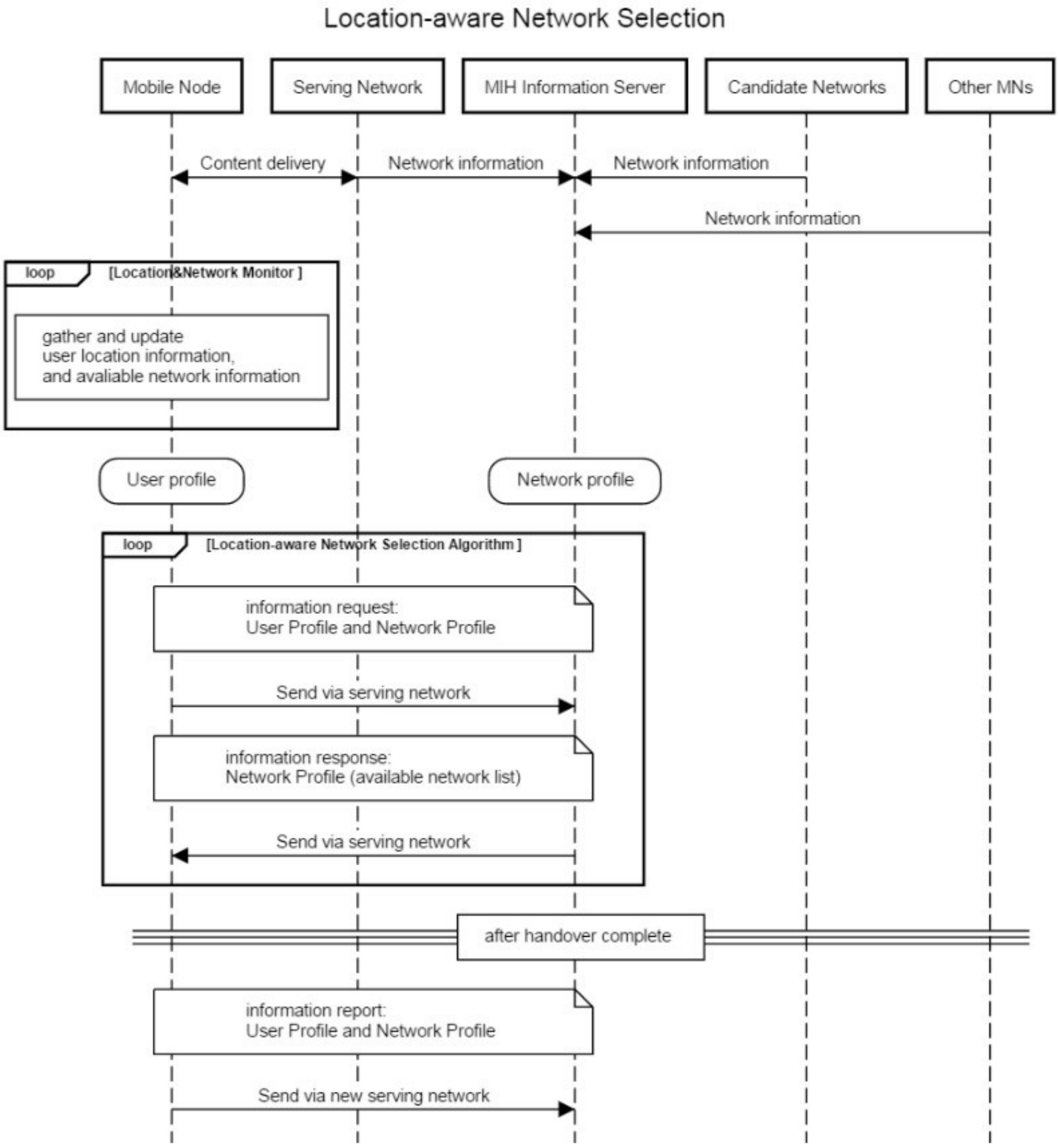


Figure 5-5 Location-aware Network Selection Sequence Diagram

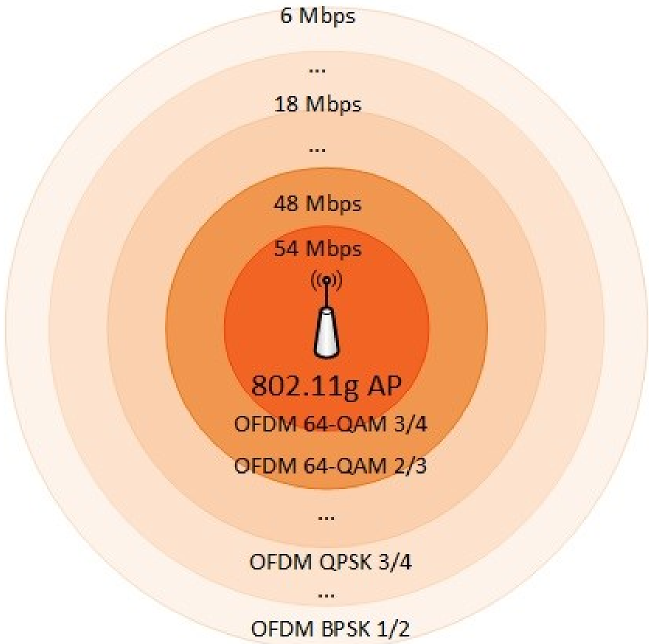


Figure 5-6 Dynamic Rate Variation as a Function of Range for 802.11g

5.3 LNS Algorithms

Compared to cellular networks, an infrastructure-based WLAN gets narrower on coverage, and higher data rate, but the latter is significantly influenced by the distance between the access point (AP) and the mobile node (MN). Fig 5.6 illustrates the dynamic data rate variation as a function of range for the 802.11g network. In this context, the distance between AP and MN must be considered in the network selection decision.

MN location coordinates (*Xmn, Ymn*) and AP location coordinates (*Xap, Yap*) are obtained via LNM in MN and information response from MIHIS, respectively. The distance can be calculated as in eq. (5.1), which can be simplified into eq. (5.2) [133]:

$$D = 2R \times \frac{\pi}{180°} \times \sin^{-1}\sqrt{\sin^2\frac{|Y_{ap} - Y_{mn}|}{2} + \sin^2\frac{|X_{ap} - X_{mn}|}{2} cosY_{ap}cosY_{mn}} \tag{5.1}$$

$$d = 222.39 \sin^{-1}\sqrt{\sin^2\frac{|Y_{ap} - Y_{mn}|}{2} + \sin^2\frac{|X_{ap} - X_{mn}|}{2} cosY_{ap}cosY_{mn}} \tag{5.2}$$

Where: D is the distance between two points measured in kilometers; d is the distance between two points measured in meters; R is the Radius of the Earth, measured in kilometers; X_{ap} is the longitude of the AP; X_{mn} is the longitude of the MN; Y_{ap} is the latitude of the AP and Y_{mn} is the latitude of the MN.

The proposed Location-aware Network Selection Mechanism consists of several algorithms. Some information is needed to initiate the LNS Mechanism, and in this work, we assume the MIHIS knows the whole topology information (e.g. a location information (X_i, Y_i)) about the heterogeneous wireless network environment in which MN is roaming. It is assumed that MIHIS can gather network information and estimate theoretical transmission range for each network by using various approaches and services (e.g. network operator, mobile user report, etc.). In order to simplify the model, in this work, we assume the transmission range of WLAN AP is r. Also the trajectory of MN is assumed to be a straight line between two recorded location information on a flat plane. To identify the topology of the heterogeneous wireless network environment, a classification filter can by run as described in eq. (5.3):

$$\begin{cases} \min(\mathrm{d}_{ap^i \to ap^j}) \geq 2r \ => \ \text{Sparse Network Distribution} \\ \min(\mathrm{d}_{ap^i \to ap^j}) < 2r \ => \ \text{Dense Network Distribution} \end{cases} \quad i, j = 1, 2, 3 \ldots n. \tag{5.3}$$

Where $\min(\mathrm{d}_{ap^i \to ap^j})$ is the minimum distance between any two APs (AP_i, AP_j) in the heterogeneous wireless network environment. Note *Sparse* and *Dense* environment are considered only.

The detailed description and analysis of each algorithm follow:

6.3.1 "Nearest" Network Algorithm

The "Nearest" network algorithm is designed to select the networks which have a high probability to handover for each network interface (i.e. WLAN and cellular), except the current serving network. These networks are automatically added to the available network list that MIHIS maintains. Next, the WLAN interface is considered for an example.

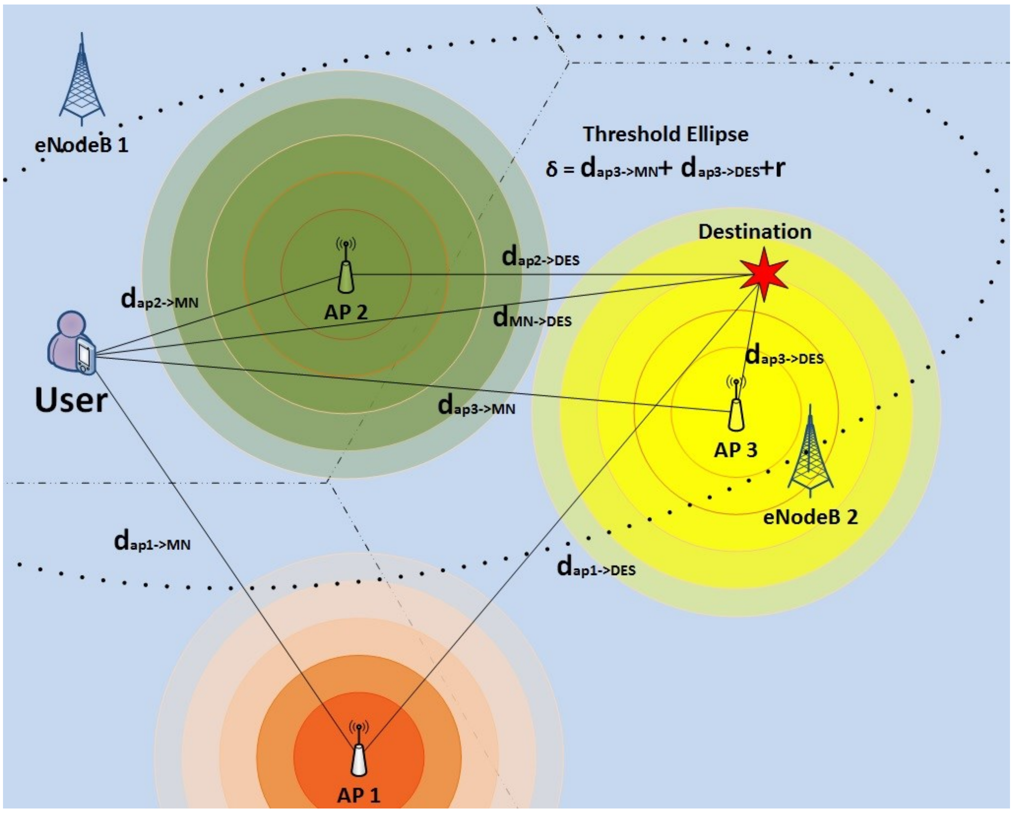


Figure 5-7 Location-aware Network Selection in Sparse Network Distribution Scenario

Based on the distances $\mathrm{d}_{ap^i \to MN}$ between various APs and MN and the distances $\mathrm{d}_{ap^i \to DES}$ between various APs and the destination, the total distance d_{ap^i} for AP_i can be calculated by eq. (5.4):

$$\mathrm{d}_{ap^i} = \mathrm{d}_{ap^i \to MN} + \mathrm{d}_{ap^i \to DES}, i = 1, 2, 3 \dots n. \tag{5.4}$$

The simplest selection action is to choose the "nearest" AP (with the lowest d_{ap^i}) as the first priority handover target to offload the data from the cellular network when MN detects that AP. As the LANSM generates the available network list for MN, several APs will be added into the "information response" and sent back to MN. The selected methods should meet the condition that d_{ap^i} is equal or smaller than a threshold δ, and be sorted according to the value of $\mathrm{d}_{ap^i \to MN}$ in ascending order. The threshold δ is generated via eq. (5.5):

$$\exists \min(\mathrm{d}_{ap^i \to DES}) = \mathrm{d}_{ap^j \to DES}$$
$$\delta = \mathrm{d}_{ap^j} + r, \qquad j = 1, 2, 3 \ldots i. \tag{5.5}$$

Where $\min(\mathrm{d}_{ap^i \to DES})$ is the distances between the destination and the closest AP_j, minimum $\delta = \mathrm{d}_{MN \to DES} + r$, when considering the closest AP_j on the line between MN and destination. The threshold δ value on a flat plane could be illustrated as an ellipse, and is dynamically changing when MN is moving.

When this distribution of APs is *sparse*, this "nearest" network algorithm works fine, like the scenario shown in Fig 5.7: AP2 and AP3 will be added to the available network list and AP2 is the first priority handover target. AP1, which is out of the threshold ellipse, will not be considered. From the description above, a lemma can be formulated:

Lemma 1:

> $\exists$ *any AP_i inside or on the* threshold δ ellipse => $\mathrm{d}_{ap^i} \leq \delta$ =>
>
> AP_i belongs to the available network list.

Proven:

For a point *i* inside or on the ellipse, the sum of distance from point *i* to the two elliptic focus points $\mathrm{d}_{point\ i}$ and the sum of the distance from any point *j* on the ellipse to the two elliptic focus points $\mathrm{d}_{point\ j}$: $\mathrm{d}_{point\ i} \leq \mathrm{d}_{point\ j}$;

For the threshold δ ellipse, the two elliptic focus points are MN and Destination, and the $\mathrm{d}_{point\ j} = \delta$;

So, $\mathrm{d}_{point\ i} \leq \delta$.

The "Nearest" network algorithm is summarized below:

Algorithm 5.1: **"Nearest" network algorithm (WLAN interface)**

1 MN, n APs, $n > 1, i = 1, 2, 3 \dots n.$

if ∃ Destination then

Compute $\mathrm{d}_{ap^i \to MN}, \mathrm{d}_{ap^i \to DES}, \mathrm{d}_{ap^i}, \delta$;

for $\mathrm{d}_{ap^i} \leq \delta$

add AP_i to the available WLAN network list;

end for

sort list {available WLAN network list} on the value of $\mathrm{d}_{ap^i \to MN}$ for each AP_i in ascending order;

select the AP with $\min(\mathrm{d}_{ap^i})$ for handover target;

end if

Algorithm 5.1 presents in details the pseudo-code of this "Nearest" network algorithm on WLAN interface. The complexity of the "Nearest" network algorithm is $O(n \cdot \log(n))$. The "Nearest" network algorithm for the cellular networks is similar, but considers cellular networks only, instead of WLANs.

However, there is another situation like the one illustrated in Fig 5.8 in which two or more APs are densely distributed and there is not a prepared choice. When the distances between two APs and MN are approximately equal, then the "nearest" network algorithm is ineffective. Hence, a trajectory based algorithm is proposed to handle the dense network distribution scenario.

5.3.2 Trajectory-based algorithm

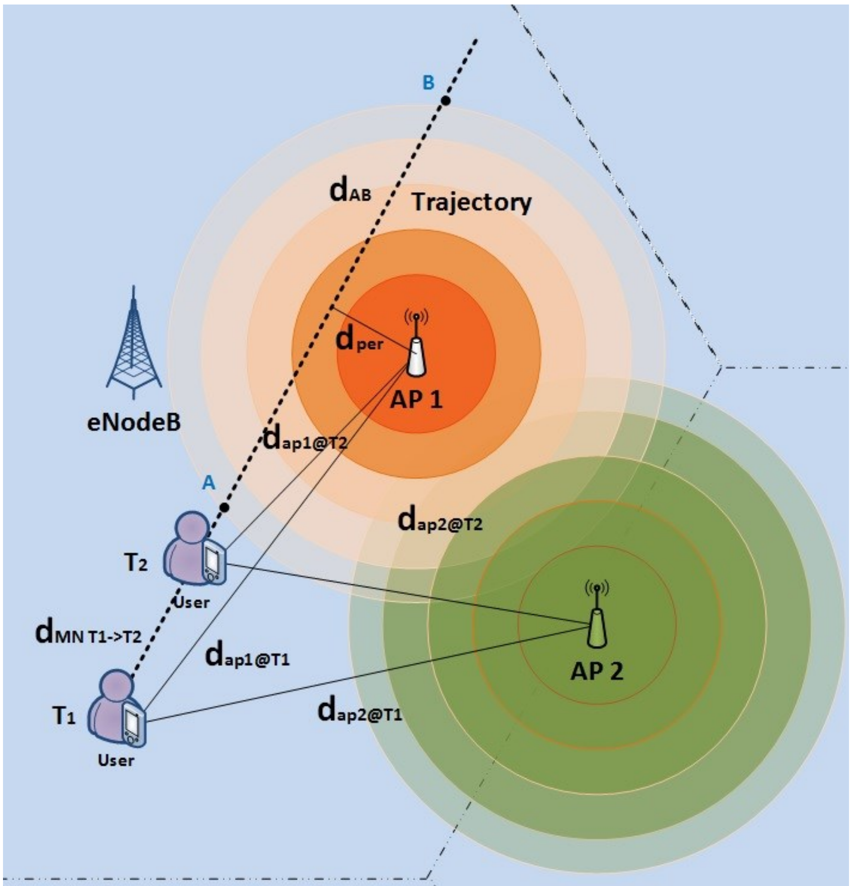


Figure 5-8 Location-aware Network Selection in Tight Network Distribution Scenario

When MN is moving, LNM always monitors MN's location together with timestamps and sends it by information request/reports to MIHIS. With more than one historical record on MN's location information, the trajectory can be generated and used for network selection. The algorithm for this approach is referred to as the trajectory-based algorithm.

As Fig 5.8 illustrates, at time T_1, the distance between MN and AP1 and AP2 is $d_{ap^1@T_1}$ and $d_{ap^2@T_1}$, respectively. After Δt_1 seconds, at time T_2, the two distances are $d_{ap^1@T_2}$ and $d_{ap^2@T_2}$, respectively. The distance between MN at T_1 and T_2 is $d_{MN\ T_1 \to T_2}$. Considering these, the period Δt_m and MN's speed V_m can be generated via the following eq. (5.6) and (5.7):

$$\Delta t_m = T_{m+1} - T_m \tag{5.6}$$

$$V_m = \frac{d_{MN\, T_m \to T_{m+1}}}{\Delta t_m} \tag{5.7}$$

Where V_m is the average velocity of the MN in the period between T_m and T_{m+1}.

Based on the location information for MN (x_{T_1}, y_{T_1}) at T_1 and (x_{T_2}, y_{T_2}) at T_2, the slope k_l of the trajectory line could be generate by eq. (5.8):

$$k_m = \frac{y_{T_{m+1}} - y_{T_m}}{x_{T_{m+1}} - x_{T_m}} \tag{5.8}$$

By using the location information of AP_i (x_{AP_i}, y_{AP_i}), the shortest distance between MN and AP_i d_{per^i} can be calculate via eq. (5.9):

$$d_{per^i} = \frac{\left|-k_m \cdot x_{AP_i} + y_{T_m} + k_m \cdot x_{T_1} - y_{T_1}\right|}{\sqrt{(-k_m)^2 + 1}} \tag{5.9}$$

Next, by comparing d_{per^i} and the AP_i's transmission range *r*, we identify whether MN will be in or not in AP_i's transmission range as in eq. (5.10):

$$\begin{cases} d_{per^i} \geq r, & MN\ will\ not\ enter\ the\ transmission\ \text{coverage}\ of\ AP_i \\ d_{per^i} < r, & MN\ will\ enter\ the\ transmission\ \text{coverage}\ of\ AP_i \end{cases} \tag{5.10}$$

If $d_{per^i} < r$, MN enters the coverage of AP_i. For instance in Fig 5.8 scenarios, d_{AB} is the distance MN passes through transmission coverage of AP_i. In general, eq. (5.11) indicates that $d^i{}_{chord}$ can be generated via Pythagoras's theorem with AP_i's transmission range *r* and the shortest distance between MN and AP_i d_{per^i}, which represents the distance MN passes through transmission coverage of AP_i:

$$d^i{}_{chord} = 2 \cdot \sqrt{r^2 - {d_{per^i}}^2} \tag{5.11}$$

By utilizing equations (5.7) and (5.11), MIHIS estimates the maximum serving time t^i_{max} of MN by AP_i as presented in eq. (5.12):

$$\mathrm{t}^{i}{}_{\max} = \frac{d^{i}{}_{chord}}{V_m} \tag{5.12}$$

Additionally, the location information of the predicted points of MN entry and exit AP_i's transmission range A (x_a, y_a) and B (x_b, y_b), respectively, are determined by equations (5.13) and (5.14):

$$x_i = \frac{-E_b \pm \sqrt{{E_b}^2 - 4E_aE_c}}{2E_a}$$
$$where\ E_a = {k_m}^2 + 1, \tag{5.13}$$
$$E_b = 2\left(k_m\left(y_{T_1} - x_{T_1}\cdot k_m - y_{AP_i}\right) - x_{AP_i}\right)\ and$$
$$E_c = \left(y_{T_1} - x_{T_1}\cdot k_m - y_{AP_i}\right)^2 - {x_{AP_i}}^2 - r^2$$

$$\mathrm{y}_i = k_m\cdot x_i + y_{T_1} - k_m\cdot x_{T_1} \tag{5.14}$$

Where when $x_{T_1} > x_{AP_i}, then\ x_a = \frac{-E_b + \sqrt{{E_b}^2 - 4E_aE_c}}{2E_a}$, else $x_a = \frac{-E_b - \sqrt{{E_b}^2 - 4E_aE_c}}{2E_a}$.

Finally, MIHIS estimates the period ($T^{\,i}_{A}, \mathrm{T}^{i}_{B}$) that MN goes through the AP_i's transmission coverage area by equations (5.15) and (5.16):

$$T^{\,i}_{A} = \frac{\mathrm{d}_{MN\to A^i @ T_{m+1}}}{V_m} + T_{m+1} \tag{5.15}$$

$$\mathrm{T}^{i}_{B} = T^{\,i}_{A} + \mathrm{t}^{i}{}_{\max} \tag{5.16}$$

Where $T^{\,i}_{A}$ is the time that MN enters the AP_i's transmission coverage area, T^{i}_{B} is the time that MN exits the area, and $\mathrm{d}_{MN\to A^i @ T_{m+1}}$ is the distance between MN and the entry point A.

Based on the previous equations, the proposed trajectory-based algorithm selects from the available WLANs according to the comparison between r and d_{per^i}. The maximum serving time $\mathrm{t}^i{}_{\max}$ is also considered in order to eliminate some APs with very short serving time ($\mathrm{t}^i{}_{\max} < t_{threshold}$, where $t_{threshold}$ is the minimum serving time acceptable for a service). Finally the candidate network list is ordered according to the distance between MN and enter point A $\mathrm{d}_{MN\to A^i@T_{m+1}}$ at time T_{m+1}. The Trajectory-based algorithm for WLAN is summarized in Algorithm 5.2 as follows:

Algorithm 5.2: **Trajectory-based algorithm (WLAN interface)**

1 MN, n APs, m records, $m, n > 1, i = 1, 2, 3 \ldots n.$

Compute $\Delta\mathrm{t}_m, \mathrm{d}_{MN\,T_m\to T_{m+1}}, V_m, k_m, d_{per^i}, \mathrm{d}_{ap^i@T_{m+1}}$;

for $d_{per^i} < r$

  Compute $d^i{}_{chord}, \mathrm{t}^i{}_{\max}, \mathrm{d}_{MN\to A^i@T_{m+1}}$;

  If $\mathrm{t}^i{}_{\max} \geq t_{threshold}$

  add $\mathrm{AP_i}$ in to list {available WLAN network pools};

  end if

end for

sort the available WLAN network list based on $\mathrm{d}_{MN\to A^i@T_{m+1}}$ in ascending order;

select the AP with minimum $\mathrm{d}_{MN\to A^i@T_{m+1}}$ as the handover target;

The complexity of the Trajectory-based algorithm is $O(n \cdot \log(n))$. The trajectory algorithm for the cellular networks is similar.

5.3.3 Mobility Prediction-based Update Scheme

As already mentioned at the beginning of section 5.3, we assume that MN is moving in a straight line between two recorded locations on a flat plane. This means the prediction value based on MN location information records at time T_m and time T_{m+1} may not work when MN changes the speed or direction and moves to another position at T_{m+2}, like Fig 5.9 illustrates. Hence, an update scheme based on mobility prediction is proposed.

In the scenario illustrated in Fig 5.9, consider MN at T_1 moving from position

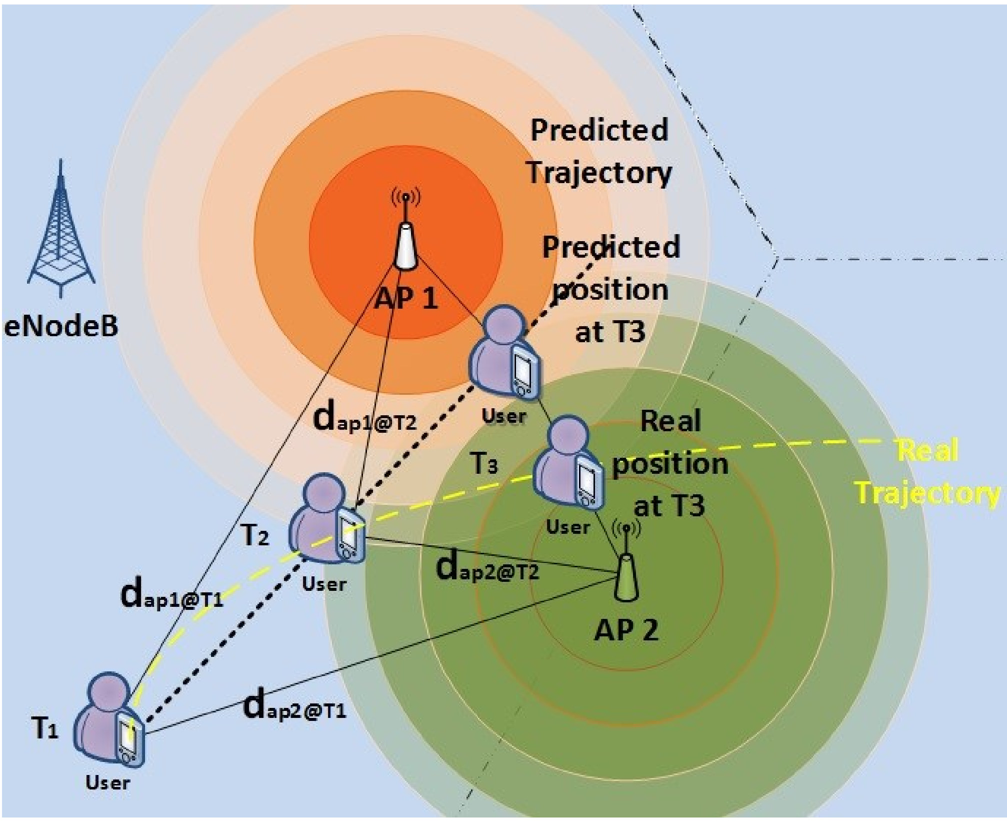


Figure 5.5-9 Prediction and Real Position

(x_{T_1}, y_{T_1}) to position (x_{T_2}, y_{T_2}) at T_2, then the trajectory-based algorithm predicted that MN will moving to position (x_p, y_p) at T_3, but MN moved to position (x_{T_3}, y_{T_3}) as the trajectory is not a straight line, but a curve that more realistic in real life.

Based on Pythagoras's theorem, the predicted position (x_p, y_p) of MN at T3 is determined by equations (5.17) and (5.18):

$$x_p = \frac{V_m(T_{m+2} - T_{m+1})(x_{T_{m+1}} - x_{T_m})}{\sqrt{(x_{T_{m+1}} - x_{T_m})^2 + (y_{T_{m+1}} - y_{T_m})^2}} + x_{T_{m+1}} \quad (5.17)$$

$$y_p = \frac{(y_{T_{m+1}} - y_{T_m})(x_p - x_{T_{m+1}})}{(x_{T_{m+1}} - x_{T_m})} + y_{T_{m+1}} \tag{5.18}$$

Next, the distance Δd between the prediction position and real position of MN at T3 can be calculated via eq. (5.2), written with the latest notations with eq. (5.2) as in eq. (5.19). Considering the limited accuracy of the localization solutions, a tolerance of θ is introduced comparing with distance Δd:

$$\begin{cases} \Delta d > \theta & update\ the\ list\ by\ running\ the\ trajectory-based\ algorithm \\ \Delta d \leq \theta\ , & do\ nothing \end{cases} \tag{5.19}$$

Finally, the proposed LNS Mechanism is designed to select the network that offers the best support for contents delivery along the user's path from the available network list for each independent network interface on MN. Note network utilization information also needs to be considered, and MIHIS can gather this information. So each network *i* has associated with utilization ρ_i:

$$\begin{cases} \rho_i \geq \rho_{max}^i & ,delete\ network\ i\ from\ the\ \text{available WLAN network list} \\ \rho_i < \rho_{max}^i & ,do\ nothing \end{cases} \tag{5.20}$$

Where the ρ_{max}^i is the maximum utilization of network *i*.

LNS Mechanism is summarized in Algorithm 5.3 in pseudo-code:

Algorithm 5.3: **LNS Mechanism**

1 MN, n APs, $n > 1, i = 1, 2, 3 \dots n$.
if $\exists$ destination then
    Running *"Nearest" network algorithm* to eliminate the AP/eNodeBs outside the threshold δ ellipse;
end if
for all network *i* in available network list
    if $\rho_i \geq \rho_{max}^i$ then
    Delete network *i* from the list
    end if

end for
if the number of items in the list > 1 then
    Run *trajectory-based algorithm* for all AP/eNodeBs in the list;
end if
for new information records received
    Compute Δd;
    if $\Delta d > \theta$
        Running *trajectory algorithm* with the rest APs/eNodeBs;
    end if
end for

The complexity of the LNS Mechanism is $O(n \cdot \log(n))$.

5.4 Simulation-based Testing and Result Analysis

The performance of the proposed Location-aware Network Selection Mechanism (LNS) is evaluated through detailed packet-level simulations using Network Simulator V.3 (NS3), which is widely used in networking research. As NS3 is open source, our own module which deployed LNS was added to the NS3 models and simulations were performed.

LNS is compared against both the classic OTSO and innovative Wiffler [134] schemes. The OTSO scheme chooses the Wi-Fi network to offload its data traffic from cellular network whenever it is possible, and connect to the cellular network immediately when Wi-Fi is not available. The Wiffler scheme is a prediction-based offloading scheme [134]. A history-based predictor estimates the amount of data that can be transferred using Wi-Fi by the deadline and the MN connects to the cellular network only when the Wi-Fi AP cannot transfer all the data. As LNS only selects the network from the available network list for both Wi-Fi and LTE interfaces, considering the monetary cost, in these

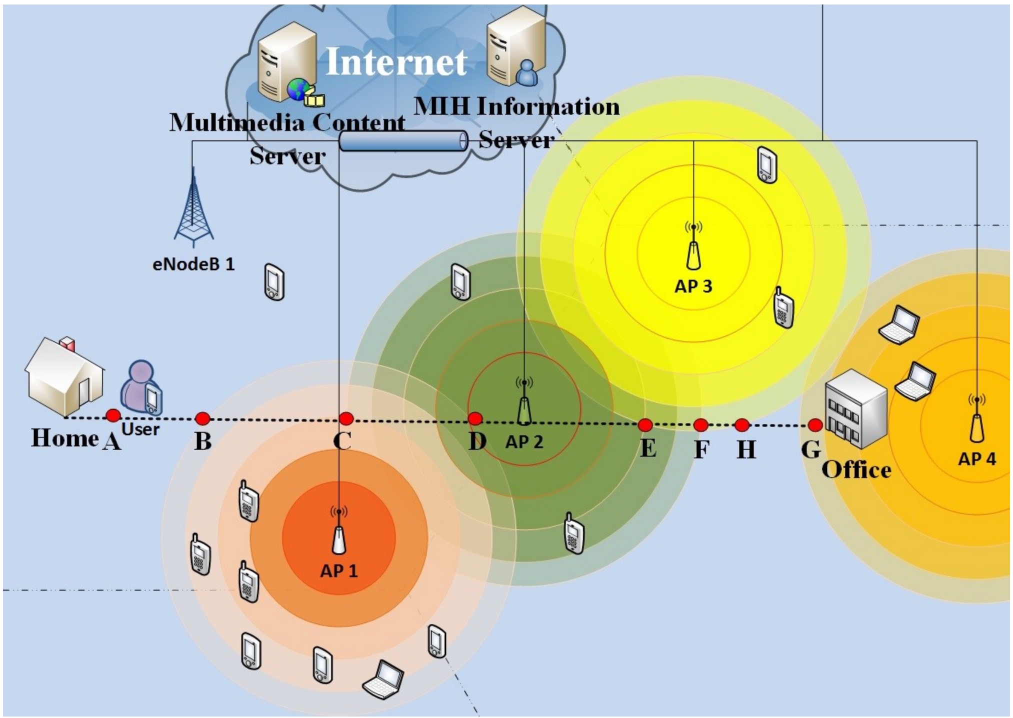


Figure 5-10 Network Topology Used in Simulation

simulations, the Wi-Fi APs get higher priority than LTE eNodeBs, and the handover is executed according to this priority.

The proposed algorithm was analyzed using a scenario from a typical day in a business professional's life, when traveling from home (point A) to his office (point G) as illustrated in Fig. 5.10. On his way to the office the user accesses video download services through his multi-interface mobile device (e.g., LTE and WLAN) from a multimedia server. While on the move, the user passes through the coverage area of several different radio access networks. First, the user is connected to the LTE network via eNodeB 1 which has the widest range (point A). As he passes through the areas with other available networks (e.g., AP 1, AP 2, AP 3 and AP 4), a network selection decision has to be made at the following points: B, C, D, E, F and H. AP 1, AP 2, AP 3 and AP 4 are assumed to be free hot spots, where only AP 1 is fully loaded with other seven extra users generating background traffic among 1Mbps to 20 Mbps, other APs are loaded with 2 users with middle network utilization. Furthermore, it is assumed that MN moves from A to G with a constant speed of 1m/s and the time MN arrives at the office is at the 100 second.

The performance of the LNS in heterogeneous networks was assessed in terms of average throughput, total cost, and time. The simulation parameters listed in Table 5.1.

Table 5-1 Simulation Setup

Parameters	Value
Duration of the simulation	100-350 seconds
Number of UEs	Total 16 UE, 1 UE with mobility, others 15 UEs are static
UE mobility	Direction; Speed = 1m/s
LTE eNodeB Antenna Model	Isotropic Antenna Model
Wi-Fi standard	802.11 ac (40MHz, MCS 9)
Traffic Model	CBR
Path Loss Model	Log-distance Propagation model
Download Content Size	691.5 MB (Big Buck Bunny [17]1080P H.264)

Specifically, in this simulation, the mean LTE data rate is 65Mbps and the mean Wi-Fi data rate is 45Mbps, where the maximum download throughput for Mobile Node (MN) in WiFi network is 20 Mbps.

[17] Big Buck Bunny - https://peach.blender.org/download/

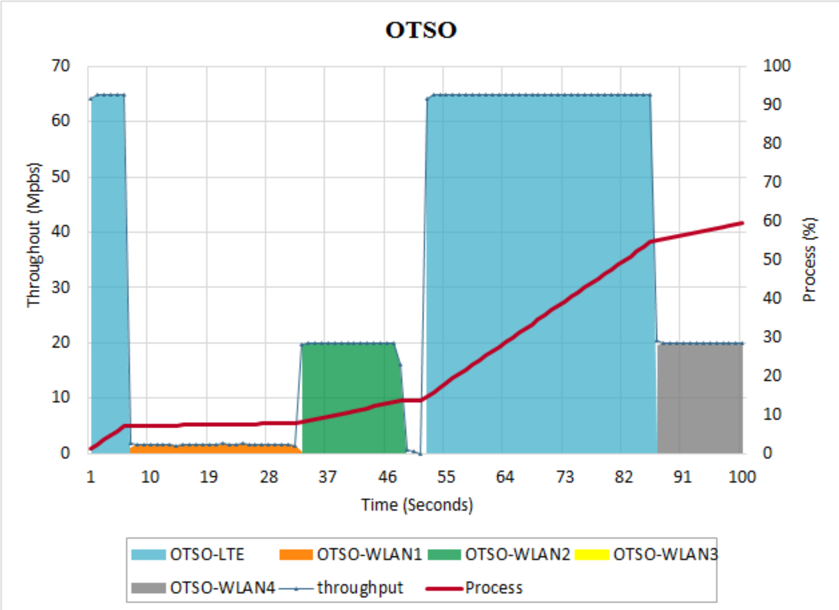


Figure 5-11 OTSO Throughput and Process

Fig. 5.11 illustrates the throughput and percentage of download process under OTSO scheme, the detailed network choice and sub-throughput for each network also displayed. The MN under OTSO at point A will automatically connect to LTE network with 65Mbps when MN moves to point B where the AP 1 is available and MN handover to AP 1 with very low throughput as AP 1 is highly loaded. During the process of moving, MN continues using AP 1 till the edge of AP 1 coverage at Point D, then MN will handover to AP 2 with high throughput up to 20 Mbps, following with the connection with AP 3 even though it is a very short period. At point H, MN will connect back to LTE until the device detects the AP 4, MN will handover to AP 4 and finish the content download after the arrival at the office. Fig. 5.12 shows the throughput and percentage of download process under Wiffler scheme, which is similar to OTSO but never connect to LTE network to avoid the cost.

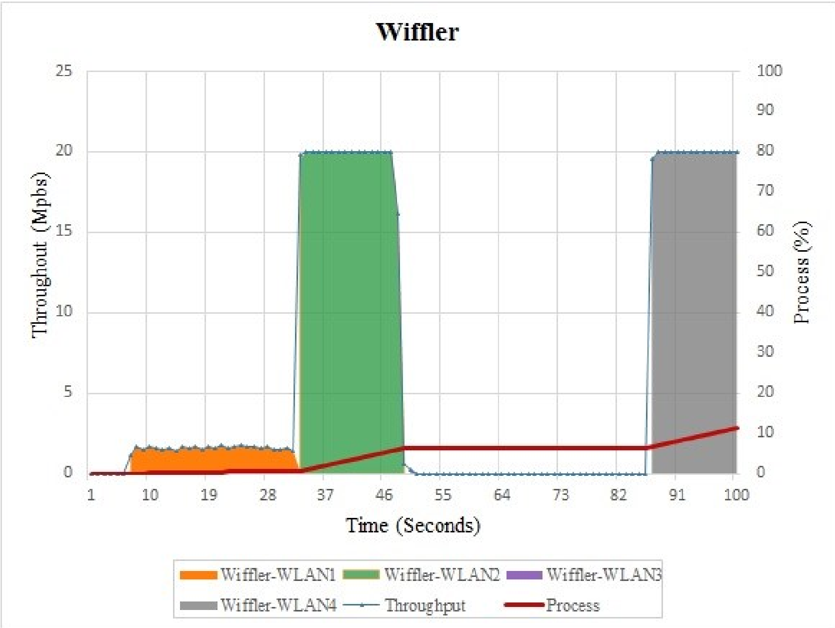


Figure 5-12 Wiffler Throughput and Process

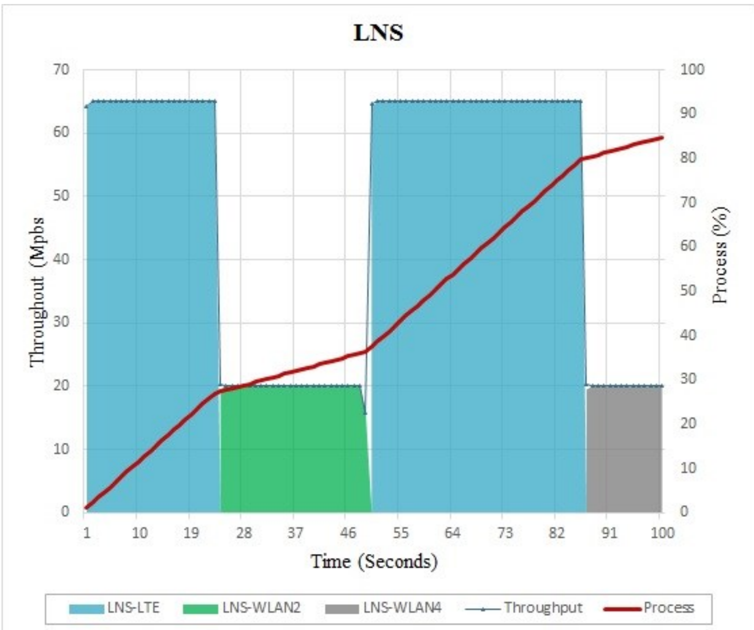


Figure 5-13 LNS Throughput and Process

The proposed LNS scheme shows the throughput and percentage of download process in Fig. 5.13, which eliminates the AP 1 and AP 3 and chooses the AP 2 and AP 4 as the handover target. The cellular-only scheme for MN only using LTE network to download contents is also displayed in Fig. 5.14 as a baseline, which finishes the download before MN reaches the office.

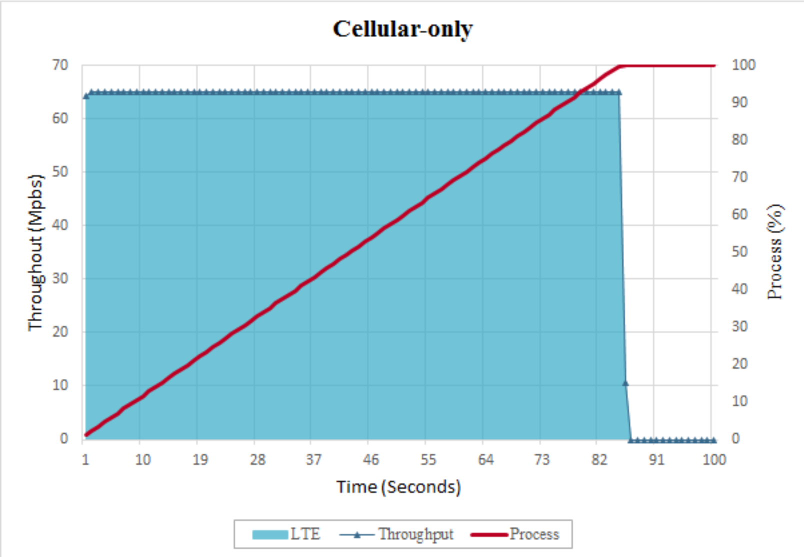


Figure 5-14 Cellular-only Throughput and Process

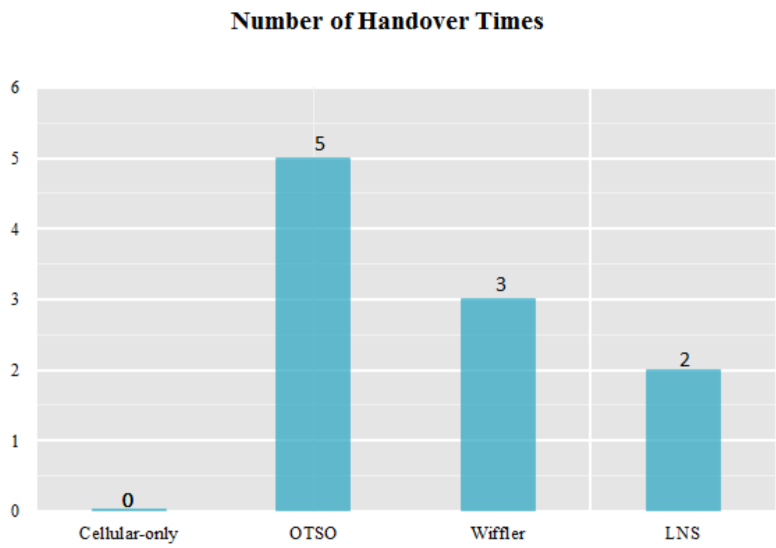


Figure 5-15 Number of Handover Times

Fig. 5.15 shows that by eliminating the low feasibility network, LNS can reduce the number of handover times to twice (5 in OTSO and 3 in Wiffler). And Fig. 5.16 illustrates that LNS enables to reduce the downloading time as much as 33% in OTSO scheme and 58.8% in Wiffler scheme.

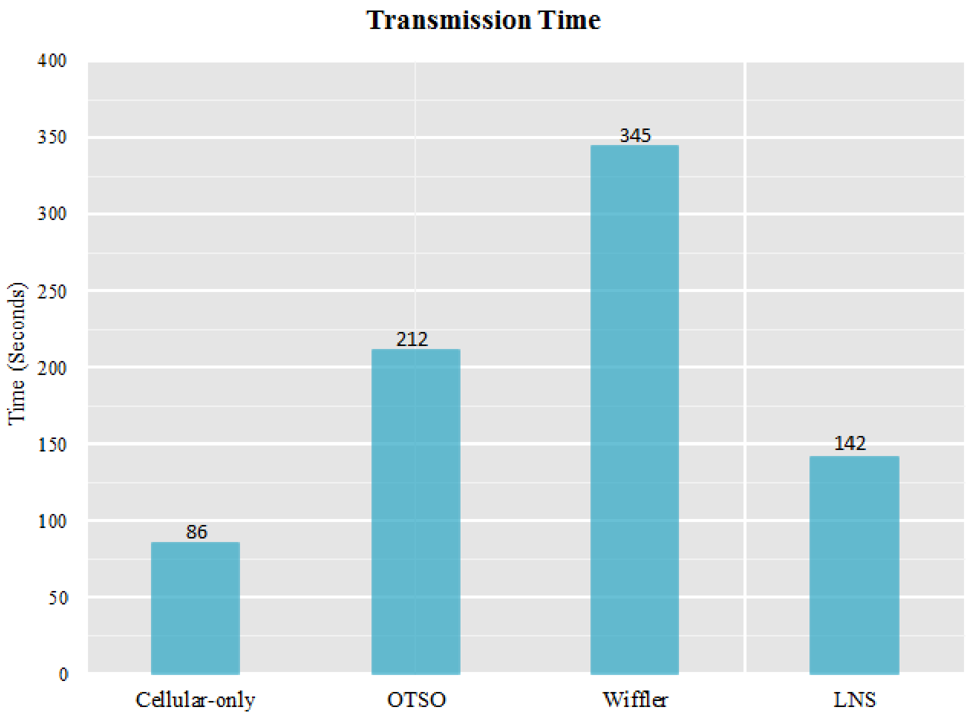


Figure 5-16 Total Downloading Time

For the monetary cost, Fig. 5.17 shows that LNS can reduce 29.5% by comparison with Cellular-only scheme. However, this monetary amount in LNS scheme is 3.08 Euro higher than that in OTSO scheme only when MN's Data Plan is finished. (i.e. for Mentor Ireland[18], Out of plan charges for mobile data is 2c per MB) .

[18] Mentor Ireland - https://www.meteor.ie/bill-pay/other-charges/

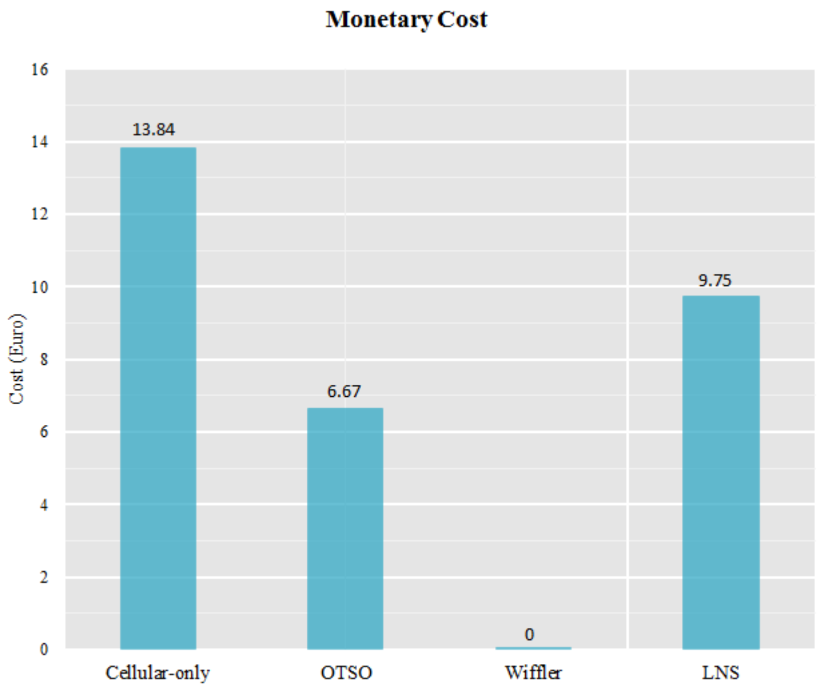


Figure 5-17 Monetary Cost for Download

5.5 Chapter Summary

This chapter proposed a new LNS Mechanism in heterogeneous networks. LNS selects the network that offers the best support for content delivery along the user's path from the available network list for each independent network interface on MN. The LNS employs algorithms: *"Nearest" network algorithm, trajectory-based algorithm* and *mobility prediction based update scheme, which were also described in this chapter*.

Simulation results using network simulation (NS3) show that by comparison with OTSO and Wiffler, proposed LNS can achieve the decrease up to 60% on the number of handover times and 58.8% in downloading time respectively.

To sum up, the LNS Mechanism can eliminate the low feasibility network with the corresponding impact on the decrease of the number of handover times and selects the highest possibility serving network to keep the network service with "Always Best Connected".

CHAPTER 6: Reputation-oriented Access Network Selection Mechanism in HetNets

Abstract

In this chapter, the second thesis contribution: Reputation-oriented Access Network Selection mechanism (RANS) in Heterogeneous Networks is described. The small cell deployment is seen as a promising solution for the network operators to help them cope with the increasing number of mobile broadband data subscribers and their bandwidth-intensive application demands. The result is a HetNets, heterogeneous network environment with a combination of macro-cells and small cells to spread the traffic load, increase the bitrates and maintain the service quality. In this context, network selection mechanisms will be required to keep the mobile users always best experienced. In this paper, we propose a theoretical framework RANS, for combining utility-based network selection mechanism with reputation-based systems. RANS makes use of the user preferences and service requirements to define a network reputation factor which reflects the user satisfaction on the network's previous service guarantee to the mobile user.

6.1 Introduction

The mass-market adoption of high-end mobile devices, as well as the increasing amount of video traffic, has determined mobile operators to adopt various solutions to help them cope with this explosion of mobile broadband data traffic while ensuring good Quality of Service (QoS) levels to the mobile user services. Deploying small-cell base stations within the existing macro-cellular networks, especially in the 3GPP Release-10 [135], is seen as a promising solution to increase capacity and improve the network performance at low cost by offloading the traffic from the large macro-cells. In this way, by transferring some of the traffic from the core cellular network to Wi-Fi or femtocells at peak times or key locations (e.g., home, office, public HotSpots, etc.) the mobile operators can accommodate more mobile users and users can avail of wider service offerings.

According to Cisco, more than half of the mobile traffic will be offloaded from the cellular network to Wi-Fi and femtocells by 2016 [136]. The small cells environment is also referred to as Heterogeneous Networks (HetNets) and is seen as part of the existing and next generation network deployments. In this context, the Always Best Experience (ABE) vision emphasizes the scenario of a mobile user seamlessly roaming in a HetNets environment. Due to the heterogeneity of the selection criteria, such as: the applications requirements (e.g., voice, video, data, etc.); different device types (e.g., smartphones, netbooks, laptops, etc.) with various capabilities; multiple overlapping network technologies (e.g., Wireless Local Area Networks (WLAN), Long Term Evolution (LTE)) and different user preferences, the mobile users will be facing a complex decision when selecting the best value network to connect to.

At the mobile user side, the mobile devices have become affordable and powerful with improved CPU, graphics and display contributing to the increase in user demands. For the mobile communication market, Ericsson mobility report shows that mobile video traffic is increasingly dominant [137], which accounts for around 50% of mobile data

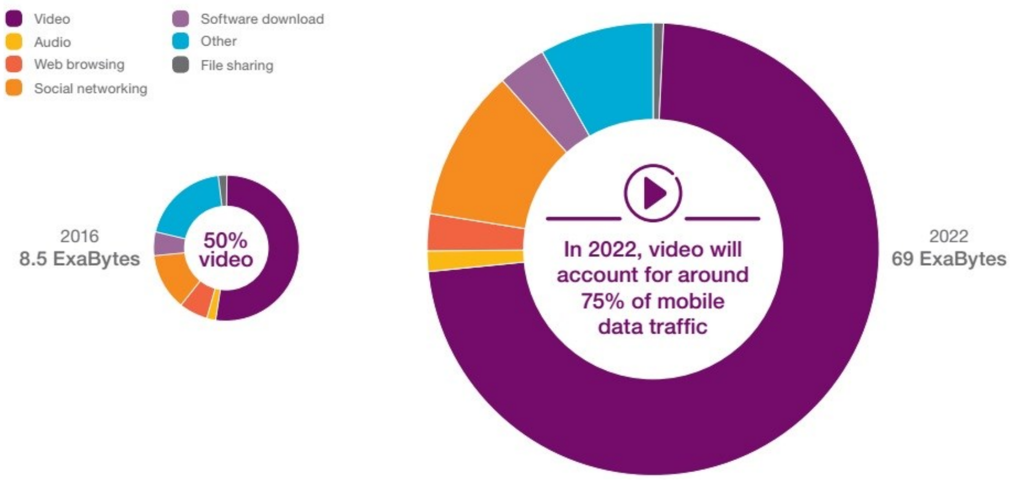


Figure 6-1 Mobile Traffic by Application Category per Month (Exabyte) [137]

traffic in 2016, and as Fig 6.1 shows that it will reach to 75% in 2022 with the compound annual growth rate (CAGR) up to 50% every year. Due to the growth of the video content usage, ensuring a seamless experience at high quality levels to the end-user has become a challenge. Furthermore, it is known that video-based applications have strict QoS requirements representing the most power-hungry applications. In this context, one of the main impediments to progress is the battery lifetime of the mobile device as the battery life has not evolved in-line with the processor and memory advances, becoming a limiting factor.

In this work, we propose a Reputation-oriented Access Network Selection mechanism (RANS), which combines the utility-based network selection mechanism with the reputation-based systems. The focus is on the user-network interaction, where we define a network reputation factor obtained as a result of the user's previous experience with the network. The network reputation factor is then integrated into the network selection decision in order to sustain cooperation between the user and the network.

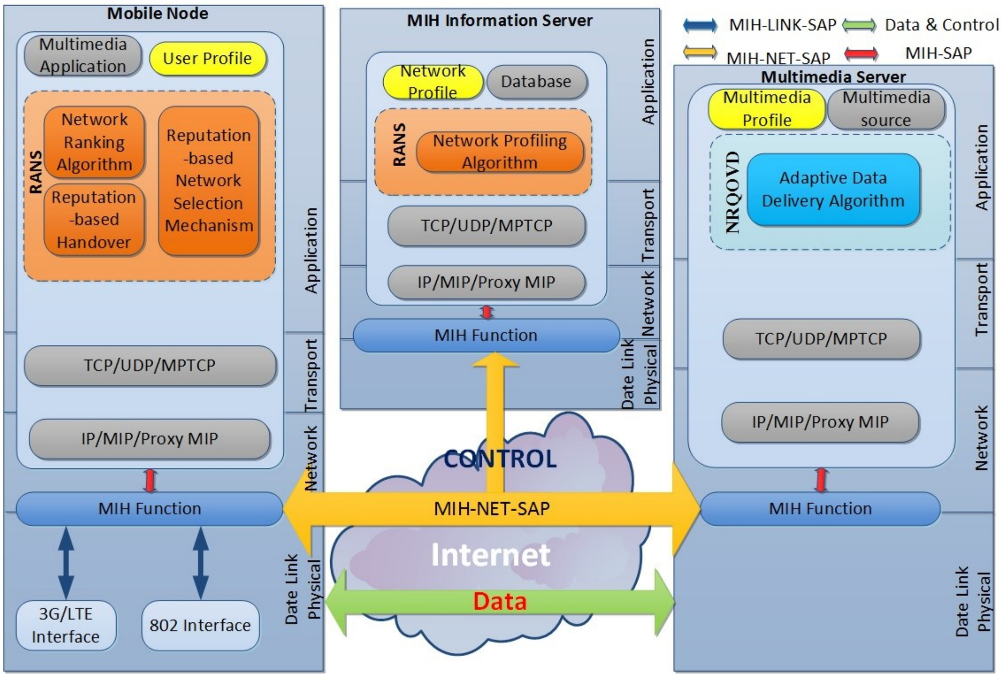


Figure 6-2 System Level Architecture

6.2 RANS Architecture

The proposed reputation-oriented access network selection mechanism, RANS, aims at building a reputation-based system between the users and the networks they are visiting. As illustrated in Fig. 6.2, RANS framework block-level architecture is distributed and consists of a server side component, referred to as RANS MIH Information Server (MIHIS), which integrates the Network Profiling Algorithm (NPA) and a client side component referred to as RANS Mobile Node (MN), consisting of the Network Ranking Algorithm (NRA), the Reputation-based Network Selection Mechanism (RNSM) and Reputation-based Handover (RH). RANS is built on top of the IEEE 802.21 MIH standard, thus both system components are MIH-enabled entities. Network Reputation-based Quality Optimization of Video Delivery (NRQOVD) solutions on Multimedia Servers is co-operated with RNS to deliver the multimedia content based on the network ranking value or network reputation score via Adaptive Data Delivery Algorithm (ADDA).

At the client side, the Mobile Node integrates three modules: Network Ranking Algorithm (NRA) compute the network ranking value by defining a utility function and based on the QoS parameters, which present the network performance. Reputation-based Network Selection Mechanism (RNSM), which stores the list of the candidate networks together with their reputation values. Reputation-based Handover (RH) will select the network with the best reputation from the candidate networks and execute the handover via MIH protocol.

The Network Profiling Algorithm (NPA) at the MIH Information Server (MIHIS) stores information about the network performance, and is based on the joint collaboration of the users within the network, thus MIHIS gathers the performance information feedback from multiple users within the network and computes the performance reputation factor for that particular network. The detailed sequence diagram of this location-aware network selection mechanism is shown in Fig. 6.3.

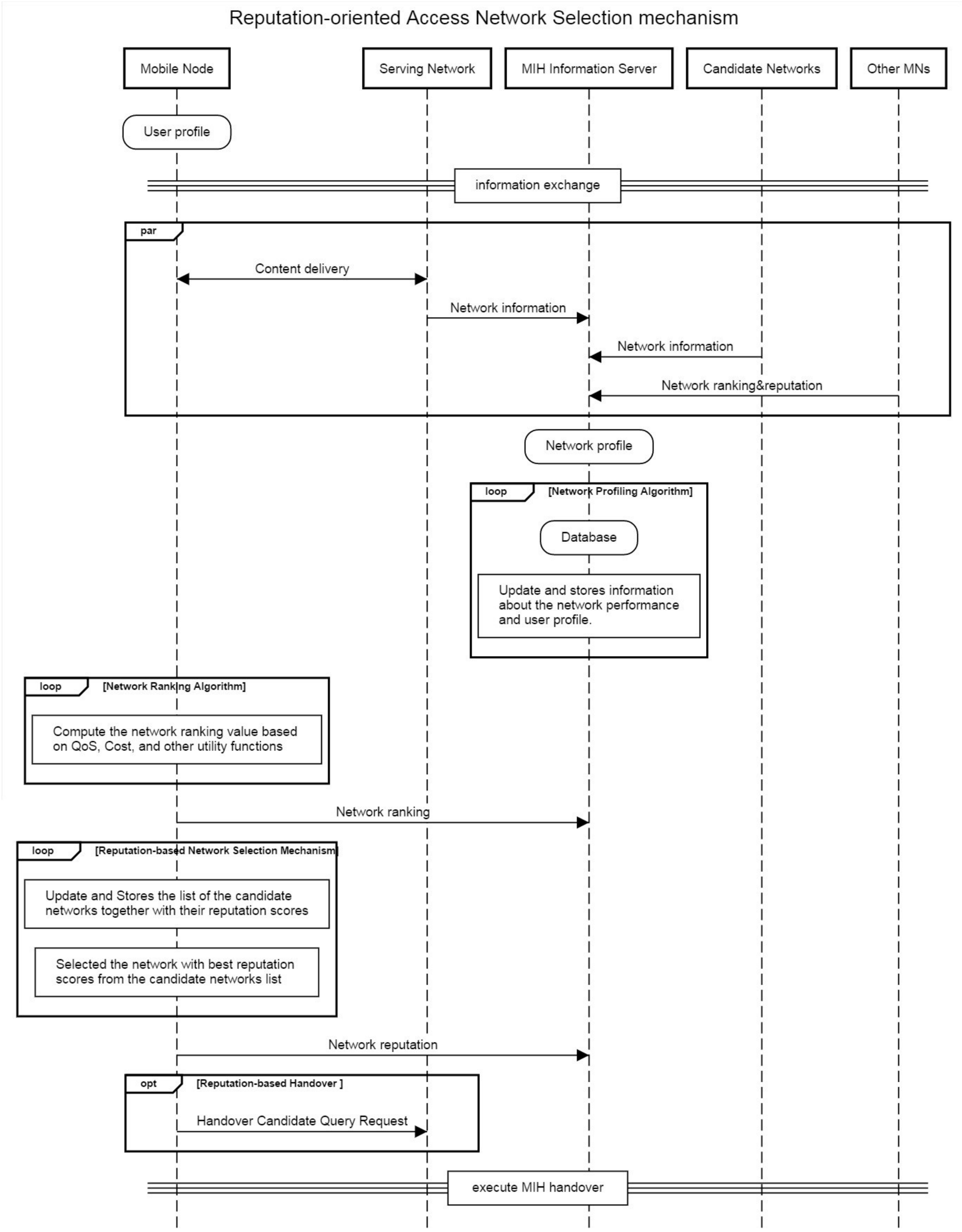


Figure 6-3 Reputation-oriented Access Network Selection Sequence Diagram

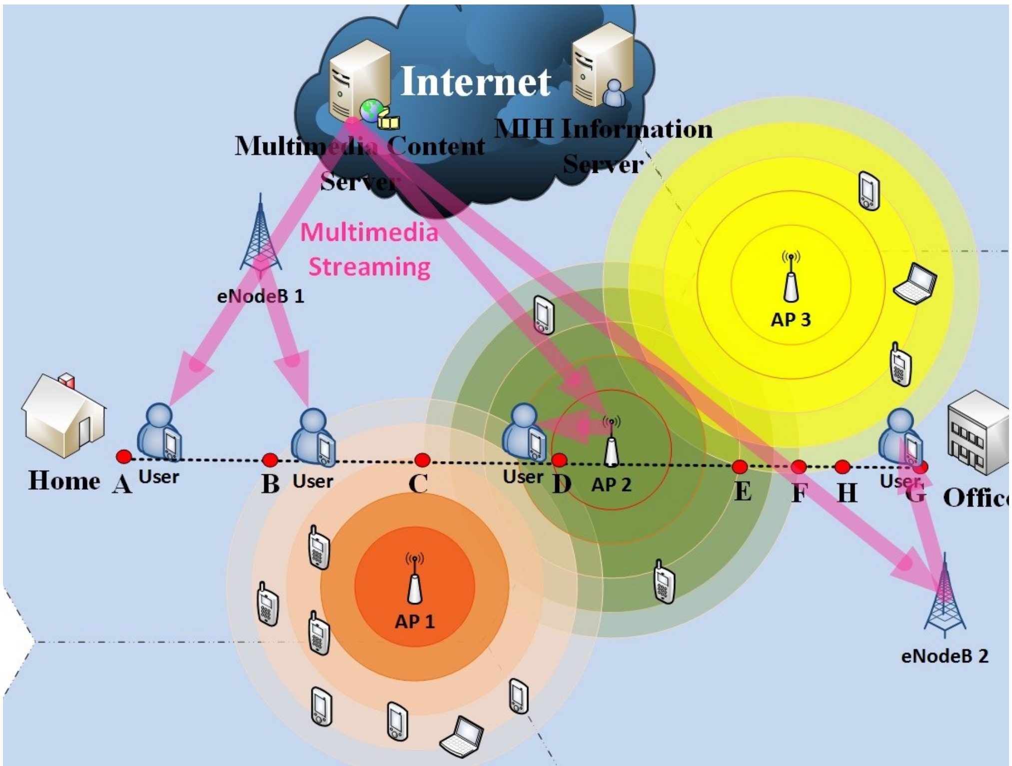


Figure 6-4 HetNets Environment – Example Scenario of a Mobile User Daily Routine

6.3 Algorithms and Decision Process

6.3.1. RANS Functional Principle

The RANS functionality considers a scenario inspired by the daily life of a mobile user, who while going from home to office, wants to access multimedia services (e.g., watching the news, music video clips, etc.) via a number of available wireless networks, as seen in Fig. 6.4. As the mobile user is taking the same path every day they will be crossing the same networks, making it possible to build a timeline/history of the user interaction with different networks. In this context, RANS, a reputation-oriented network selection mechanism is proposed. The idea behind RANS is that each user can have different experiences with different network operators, depending on the user preferences and the service requirements. As a result of this user-network interaction, a reputation factor can be computed for that particular network. For example, if the user was satisfied with the offered services, the network will receive a higher reputation value reflecting the user satisfaction.

The proposed RANS solution combines the utility theory with the reputation theory to build a reputation-based system between the users and networks. Within the HetNets environment, MN have the list of available wireless networks for each interface with their characteristics. The NRA located at the MN will compute a network ranking list based on three criteria, such as energy consumption of the mobile device when running real-time applications, the monetary cost of each network, and the estimated quality of the multimedia stream. Then, the RANS based MN will send an information report to the MIHIS. So, the NPA on MIHIS can make use of utility functions [45] to compute *Reputation Utility* and *Stabilization Utility* score for each network. A list of networks along with their expected utility scores is then sent to the MN. The RNSM on MN will integrate both of *Reputation Utility* and *Stabilization Utility* scores together with the network ranking value to get the *Overall Reputation Score*. Finally, at the end user side, in the first instance, the RNSM will select the best reputation network from the overall reputation score list among various network interface and the RH will execute the handover. After the user connects to the target network, an user-network interaction session starts where the service quality is monitored. During the user-network interaction, a new network ranking value is computed based on the experienced utilities in each specific period, and gather by MIHIS to run the NPA to update *Reputation Utility* and *Stabilization Utility* score for each network. This updated score will impact the *Overall Reputation Score* of each network next time the network selection takes place.

6.3.2. Proposed Utility-based Network Ranking Function

The use of utility function together with the Multiplicative Exponential Weighted (MEW) method in the decision making mechanisms has been shown to be useful in [45]. A generic model of the network ranking function is given in eq. (6.1):

$$U_i = u_{e_i}^{w_e} \cdot u_{q_i}^{w_q} \cdot u_{c_i}^{w_c} \tag{6.1}$$

where i represents the candidate network, U_i is the overall utility for network ranking and u_e, u_q, u_c are the utility functions defined for energy, quality and monetary cost for network i, the value of all three utility functions belong to [0,1], w_e, w_q, w_c are the weights for the three considered criteria: energy, quality and monetary cost, respectively and $w_e + w_q + w_c = 1$. The network ranking function is computed for each of the candidate networks and a ranked list is sent to the URAN mobile node. The utility functions used were previously proved to be efficient in a wireless multimedia heterogeneous environment [138]. As utility functions together with the MEW method are used for normalization in the decision making mechanisms, a ranking abnormality may occur and effect needs to be fixed. Ranking abnormality (or reversal) means that the ranking of several alternative networks changes after removing of them from the list (i.e. when not available anymore) [139]. The effect is that a lower ranked network becomes the highest ranked temporarily. In the handover context, ranking-abnormality will lead to a ping-pong effect which make useless handover and reduces the performance of network communications. In order to reduce possible ranking-abnormality effects, the Enhanced Max-Min [139] method is suggested as normalization procedure for the MAMD method [140].

a) Energy Utility – u_e

The estimated energy consumption for a real-time application is computed using eq. (6.2) as defined in [141]:

$$E = t\left(r_t + Th_{req} r_d\right) \tag{6.2}$$

where t represents the transaction time (s) which can be estimated from the duration of the video stream, r_t is the mobile device's energy consumption per unit of time (W),

Th_{req} is the required throughput (kbps), r_d is energy consumption rate for data/received stream (J/Kbyte), and E is the total energy consumed (J).

Based on the estimated energy consumption E, the utility for the energy criteria u_e is computed using eq. (6.3) via Enhanced Max-Min method:

$$u_e = \begin{cases} 1 & , E \leq E_{min} \\ \dfrac{E_{max} - E}{E_{max} - E_{min}} & , E_{min} < E \leq E_{max} \end{cases} \tag{6.3}$$

Where E is the energy consumption for the current network (Joule), and E_{min} and E_{max} are the minimum and maximum energy consumptions needed for the current video streaming application to run until completion, being calculated using eq. (6.2) for TH_{min} and TH_{max}, respectively.

The weights function for the three considered criteria can be defined by the user profile which depends on the device type or user environment. The default value of the weights function w_e for the energy criteria will depend on the ratio θ_E for MN's remain energy to MN's maximum energy via eq. (6.4):

$$w_e = \begin{cases} 0 & , \theta_E \geq 0{,}5 \\ 1 - \theta_E & , \theta_E < 0.5 \end{cases} \tag{6.4}$$

Where for the MN with more than 50% battery, the energy consumption will not significantly impact the mobile user's ability and behavior on the network using, in contrast to the scenario when battery is less than 50%.

b) Cost Utility - u_c

The cost utility is important as there is a natural human tendency to reduce the monetary cost. The mathematical definition of the cost utility is given in eq. (6.5) via Enhanced Max-Min method:

$$u_c = \begin{cases} 1 & , C = 0 \\ \dfrac{C_{max} - C}{C_{max}} & , 0 < C \leq C_{max} \end{cases} \tag{6.5}$$

Where C is the monetary cost for the current network, $C_{min}(C_{min} = 0)$ and C_{max} are the minimum and the maximum costs that the user is willing to pay.

The same as weights function w_e, the default value of the weights function w_c for the monetary cost criteria will depend on the two parameters, the remain data balance and the ratio $\theta_€$ remain account credit to the maximum account credit via eq. (6.6):

$$w_C = \begin{cases} 0, & \text{remain data balance} > 0 \\ 1 - \theta_€, & else \end{cases} \tag{6.6}$$

Where for the mobile user who still has the data balance in the account, the monetary cost is 0.

c) Quality Utility – u_q

Network quality is the overall performance of a network service that bears on its ability to satisfy stated and implied needs of the user of the service, which could be presented as QoS. So, by using various QoS parameters, the network quality could be measured, similar to network ranking function in eq (6.1), the quality utility could also be given in eq. (6.7):

$$u_{q_i} = u_{th_i}^{w_{th}} \cdot u_{d_i}^{w_d} \cdot u_{j_i}^{w_j} \cdot u_{l_i}^{w_l} \tag{6.7}$$

Where i represents the candidate network, u_{q_i} is the quality utility for network ranking and u_{th}, u_d, u_j, u_l are the utility functions defined for throughput, delay, jitter and loss rate for network *i,* the value of all four utility functions are belong [0, 1], w_{th}, w_d, w_j, w_l are the weights for the four considered criteria: throughput, delay, jitter and loss rate, respectively and $w_{th} + w_d + w_j + w_l = 1$.

In this work, we focus on video delivery service, the study presented in [142] shows that the quality of the received videos over IP networks is significantly influenced by packet loss and video bitrate. In order to simplify the model, we assume that all other parameters for the network are perfect; hence the utility scores are 1. The two parameters we involved is throughput representing the video bitrate received at the MN and network packet loss rate, so the eq. (6.8) can be generated:

$$u_{q_i} = {u_{th_i}}^{w_{th}} \cdot {u_{l_i}}^{w_l} \tag{6.8}$$

A zone-based quality sigmoid utility function is used to map the received bandwidth to user satisfaction [143]. The mathematical formulation of the utility function that maps the quality of the multimedia application effected by throughput is given in eq. (6.9):

$$u_{th} = \begin{cases} 0 & ,Th < Th_{min} \\ 1 - e^{\frac{-\alpha \cdot Th^2}{\beta + Th}} & ,Th_{min} \leq Th < Th_{max} \end{cases} \tag{6.9}$$

Where α and β are two positive parameters which determine the shape of the utility function and *Th* is the predicted average throughput for each of the candidate networks. The minimum throughput (Th_{min}) is a threshold to maintain the multimedia service at a minimum acceptable quality level values below this threshold result in unacceptable quality levels. Whereas values above the maximum throughput (Th_{max}) threshold will not add any noticeable improvements in the user perceived quality. The values for α and β used in this study are 5.72 and 2.66 [143], respectively.

In [144], ITU-T G.1070 standardized a user opinion model for video-telephony applications. It estimates the video quality in telephony applications by considering the network packet loss and encoding parameters (i.e. codec type, video format, key frame interval) and video display size. The video quality is evaluated by eq. (6.10):

$$V_q = 1 + I_{coding} \exp(-\frac{P_{plv}}{D_{P_{plv}}}) \tag{6.10}$$

where I_{coding} represents the basic video quality affected by the coding distortion under a combination of video bit rate [kbit/s] and video frame rate [fps], and the packet loss robustness factor D_{Pply} expresses the degree of video quality robustness due to packet loss where P_{plv} [%] represents the packet-loss rate.

For the specific video bit rate and frame rate, I_{coding} is a constant parameter, which could be presented as a positive parameter φ, the same for the D_{Pply}, a positive parameter ω can be used to represent the value for specific video bit rate and frame rate. The mathematical formula of the utility function that maps the quality of the multimedia application affected by packet loss rate is given in eq. (6.11):

$$u_l = \begin{cases} 0 & , P_{plv} \geq P_{plv_{max}} \\ \dfrac{1 + \varphi \cdot e^{-\frac{P_{plv}}{\omega}}}{1 + \varphi} & , 0 \leq P_{plv} < P_{plv_{max}} \end{cases} \tag{6.11}$$

Where the $P_{plv_{max}}$ is the maximum packet loss rate for video streaming, according to [145], when the packet loss rate goes above 5%, the quality of the video stream becomes unacceptable. The values for φ and ω used in this study are 3.5 and 2.74 [144], respectively.

6.3.3. Proposed Utility-based Reputation Function

The network ranking function provides a list of ranked networks based on the overall scores obtained using the utility function defined in eq. (6.1). These scores are representing utilities that the users will receive once connecting to a particular network. However, during the connectivity session with the target network, network conditions might change thus the utility received by the user might be different from the initially expected utility. In order to reflect this in the network selection process, at the end of every user-network interaction, *Reputation Utility* and *Stabilization Utility* score are computed. Thus, a new utility-based reputation function is given in eq. (6.12):

$$R_i = (u_{r_i}^{w^r} \cdot u_{s_i}^{w^s})U_i, \tag{6.12}$$

Where: i represents the candidate network, R_i is the overall reputation score for network performance and u_r, u_s are the utility functions defined for network reputation and network stabilization for network i. w_r, w_s are utility function weights associated with network reputation and network stabilization, respectively. $w_r + w_s = 1$, and the default values of w_r and w_s are 0.5 which means that both network reputation and network stabilization are equally important for the overall reputation value. In the simulation of this work, as the number of users is very small (and the relevancy of stabilization is limited), the settings of the weight values are $w_r = 1$ and $w_s = 0$.

Network overall reputation score is computed for each of the selected candidate networks and the network with the highest score is selected as the target network.

a) Reputation Utility – u_r

In order to keep track of the past experience with a particular network and strengthen the cooperation between users and networks, and the reputation of the network are build up in past, the network ranking value represents the network performance decreased by time, and the formula gives the eq. (6.13) based on score calculation:

$$r_i = \sum_{j=1}^{N} \frac{U_{i_j}}{T^G} \tag{6.13}$$

Where j is the sequence number of MIHIS received network ranking value, T is duration between current times to the time that MIHIS received network i ranking value U_{ij}, G is a constant, N is the total number of ranking value that MIH information server received for network i.

The duration T can be estimated by the storage time of ranking value U_{ij}, and the number N can be counted by MIH information server. The reputation utility u_r is given in eq. (6.14):

$$u_r = \frac{r_i}{r_{i_{MAX}}} = \frac{\sum_{j=1}^{N} \frac{U_{i_j}}{T^G}}{\sum_{j=1}^{N} \frac{U_{i_{j_{MAX}}}}{T^G}} = \frac{\sum_{j=1}^{N} \frac{U_{i_j}}{T^G}}{\sum_{j=1}^{N} \frac{1}{T^G}} \tag{6.14}$$

Where r_{imax} is the maximum value of network reputation, which only happen when all the U_{ij} are equal to 1.

b) Stabilization Utility – u_s

Theil index[19] used to measure inequality, the more variance between various U_{ij} higher, the lower Theil value is, and is given in eq. (6.15):

$$Theil = \sum_{j=1}^{N} \frac{U_{i_j}}{\sum_{j=1}^{N} U_{i_j}} \log\left(\frac{\sum_{j=1}^{N} U_{i_j}}{U_{i_j}}\right) \tag{6.15}$$

Where j is the sequence number of MIH information server received network ranking value U_{ij} and N is the total number of ranking value U_{ij} that MIH information server received for network i. The maximum Theil equals to $logN$ when all U_{ij} equals together.

So, stabilization utility u_s is given in eq. (6.16):

$$u_s = \frac{\sum_{j=1}^{N} \frac{U_{i_j}}{\sum_{j=1}^{N} U_{i_j}} \log\left(\frac{\sum_{j=1}^{N} U_{i_j}}{U_{i_j}}\right)}{\log N} \tag{6.16}$$

When N equals to 1, u_s equal to 1 and u_r is equaled to U_i.

[19] Theil index - https://en.wikipedia.org/wiki/Theil_index

6.4 Simulation-based Testing and Result Analysis

The performance of the proposed RANS was evaluated by using NS version 3.22. The performance of the video delivery in heterogeneous networks was assessed in terms of average throughput, delay, Peak Signal-to-Noise Ratio (PSNR).

RANS is compared against three other solutions: OTSO [132], PoFans [149] and an always LTE cellular scheme [151]. PoFans trades-off between energy consumption, monetary cost and network load to select the best utility value network for both Wi-Fi and LTE interfaces. Since OTSO considers the monetary cost, the Wi-Fi APs gets higher priority than the LTE eNodeB, and the Wi-Fi WLANs will be used to connect to until the current serving network is not available anymore. The cellular only scheme prefers LTE to any other network for connectivity.

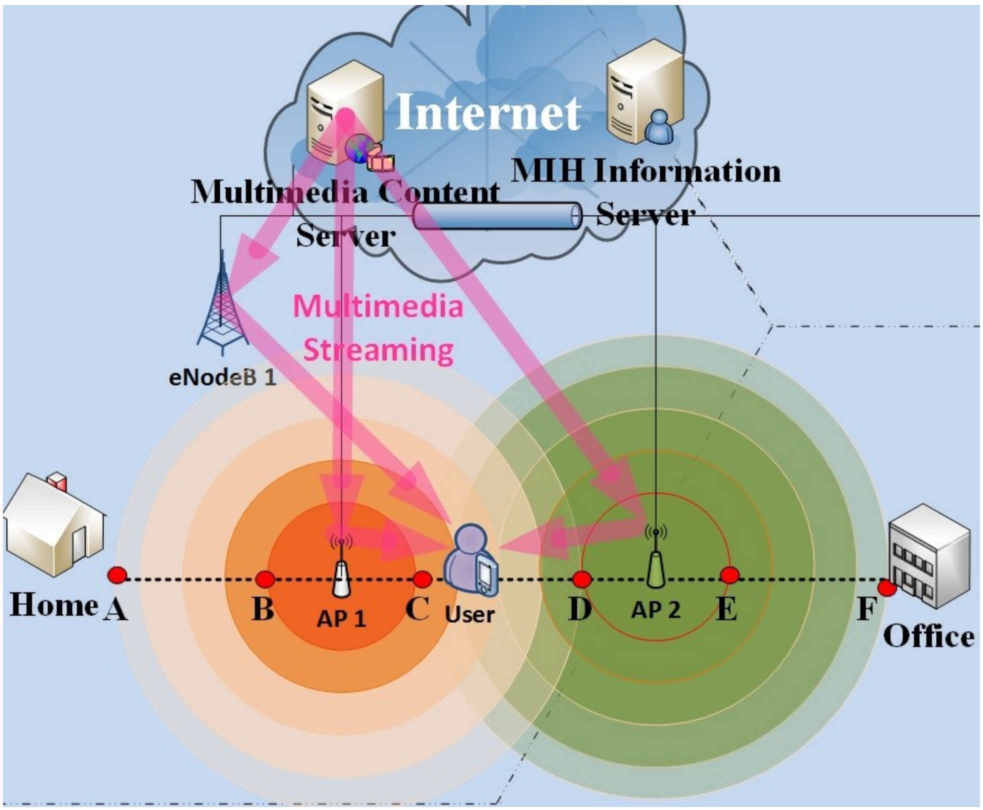


Figure 6-5 Network Topology Used in Simulation

RANS, the proposed scheme is analyzed using a scenario from a typical day in a business professional life, when traveling from home (point A) to office (point F), as illustrated in Fig. 6.5. On the way to the office, the user accesses interactive multimedia streaming services through a multi-interface mobile device (e.g., LTE and WLAN) from a multimedia server. While on the path from home to the office, the user passes through the coverage area of several different radio access technologies. As Mobile Node (MN) passes through the areas with a number of other available networks (e.g., AP 1 and AP 2), network selection decisions are made. The distance between A and H is 360 meters.

Table 6-1 YouTube Recommended Video Bitrates [146]

	1080p	**720p**	**480p**	**360p**	**240p**	**144p**
Resolutions	1920×1080	1280×720	854×480	640×360	426×240	256×144
Encoding Bitrates	8Mbps	5Mbps	2.5Mbps	1Mbps	0.543Mbps	0.4Mbps
Encoding Format	H.264/MPEG-4 AVC Baseline Profile					
Frame Rate	30fps					

In the proposed RANS, the user profile is used for the network ranking mechanism and includes the following settings: preferences for energy, quality and monetary cost and the minimum and maximum cost the user are willing to spend for multimedia services. The costs for each of the three networks considered in Fig. 6.5 are set to: WLAN A and WLAN B are free hot-spots, and LTE – 0.2 cents per MByte. The user is running a 360 seconds long MPEG-4 multimedia stream, and it is assumed that the Multimedia Server stores six different quality levels of the multimedia stream with the encoding settings presented in Table 6.1 (i.e. YouTube recommended video bitrates [146]). Furthermore, it is assumed that the MN moves from A to H with a constant speed of 1m/s (e.g. typical for a walking user).

As the minimum video bitrate is 0.4Mbps and maximum video bitrate is 8Mbps in table 6.1, following the definition of eq. (6.9), Th_{min} = 0.4Mbps, Th_{max}= 8Mbps, and Th_{req} = Th_{min} =0.4Mbps. In terms of energy consumption of the mobile device, the values for the energy consumption rate per unit time (r_t) and the energy consumption rate for data/received stream (r_d) under various network conditions are listed in Table 6.2.

Table 6-2 Energy Consumption Rate for Two Interfaces

Interface	r_t (mWatt)	r_d (mJoule/kbps)
LTE	1403.77	0.4714
WLAN (Wi-Fi)	570.46	0.4101

The value of r_t and r_d are based on a real experimental test-bed illustrated in Fig. 6.6 [147], which consists of a Smartphone (Samsung Galaxy S4 Mini LTE i9195[20]) with Android Operation System (OS) and communication using LTE cat 3 and IEEE 802.11g networks. The LTE network operator is Meteor Ireland[21], and the signal strength during the tests is between -93dbm to -94dbm. The IEEE 802.11g network is supported by the Belkin N wireless router[22] on Channel 13 (2.472GHz) in IEEE 802.11g mode, and the signal strength during the tests is between -45dbm to -56dbm. The mobile device power consumption measured by an Arduino Duemilance Board[23], and the measurement data is processed by an open-source monitoring platform[24] in order to compute the energy consumption. The detailed circuit description and configuration can be found in [148].

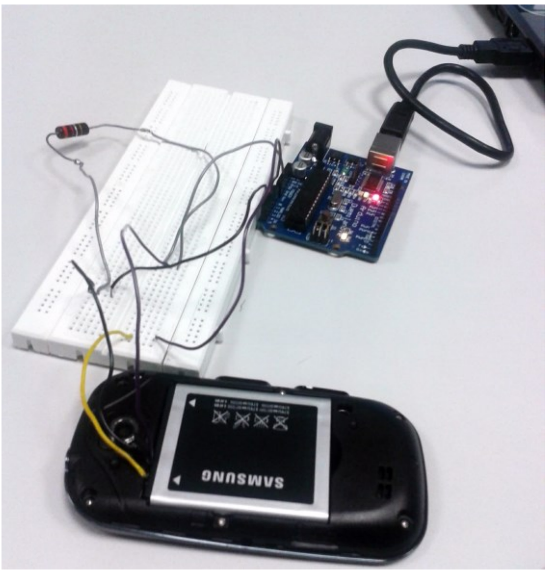
[20] Galaxy S4 mini - http://www.samsung.com/uk/support/model/GT-I9195ZKABTU

[21] Meteor Ireland Netwroks - https://www.meteor.ie/ournetwork/

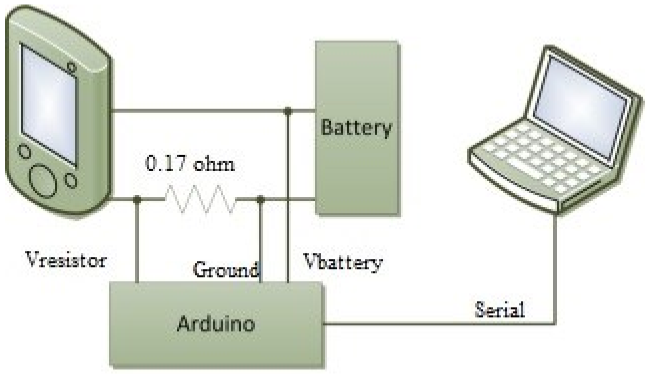
[22] Belkin N Wireless Router - http://www.belkin.com/us/support-product?pid=01t80000001JNW5AAO

[23] Arduino Duemilance Board - https://www.arduino.cc/en/Main/ArduinoBoardDuemilanove

[24] Smartphone_PowerMonitor - https://github.com/allengzmm/Smartphone_PowerMonitor



(a) Arduino-based Testing Platform Measuring Energy Consumption in Mobile Device



(b) Schematic Representations for Measuring Energy Consumption Test-bed

Figure 6-6 Experimental Test-bed Setup

The values of r_t and r_d for each interface is used to compute energy E by using eq. (6.2) for each video bitrates with 360 seconds duration, the results are listed in Table 6.3. The values for E_{max} and E_{min} are 1862.989 Joules and 264.42 Joules, respectively.

Table 6-3 Computed Energy [Joule]

Video resolution	1080p	720p	480p	360p	240p	144p
LTE	1862.989	1353.877	931.617	675.061	597.507	573.239
WLAN	1386.454	943.546	574.456	353.002	285.532	264.420

In order to study the impact of user preferences, which expressed in terms of weight values, on network selection where the mobile user has a choice of two networks: LTE and WLAN (Wi-Fi). Three case studies are considered: (a) *balanced user* with w_e=0.4, w_q=0.4, w_c=0.2, where the user is willing to pay a certain amount while maintaining a balance between the quality level and the energy consumption; (b) *equal interest user* with w_e=0.33, w_q=0.33, w_c=0.33, where the user equally cares about the three criteria energy, quality, and cost; and (c) *quality-oriented user* with w_e=0.1, w_q=0.8, w_c=0.1, where the user is quality aware and has a strict quality requirement. As neither of the networks is connected, the packet loss is not considered in this scenario, then the overall ranking function values computed with eq. (6.1) for all three case studies are listed in Table 6.4.

Table 6-4 Overall Ranking Results

		1080p	720p	480p	360p	240p	144p
Balanced User	**LTE**	0.025119	0.427411	0.6934	**0.803161**	0.740663	0.68595
	WLAN	0.616234	0.801526	0.91719	**0.932551**	0.831978	**0.762945**
Equal Interest User	**LTE**	0.047863	0.495961	0.73916	**0.802809**	0.67365	0.586136
	WLAN	0.670718	0.833167	**0.93102**	0.908093	**0.741463**	0.639903
Quality-Oriented User	**LTE**	0.398107	0.808558	**0.91186**	0.793616	0.474771	0.329931
	WLAN	0.886005	0.946192	**0.9779**	**0.823812**	0.488773	0.338823

The results show that for the equal interest and quality-oriented users, when the WLAN network is available, the multimedia server delivering the 480p streaming to the user will have the best reputation. For the balanced user, 360p streaming gets the best reputation on both LTE and WLAN networks. When WLAN can only support the 360p streaming, the quality-oriented user will handover to the LTE network with 480p streaming

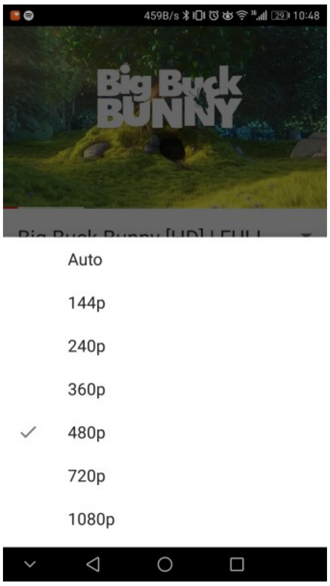


Figure 6-7 Video Quality Switch Interface in YouTube App on Android Phone

in order to achieve a better reputation. AT that time, the 360p streaming under WLAN network is still the best choice for the equal interest user compared to any video streaming via LTE network, but when the WLAN network only supports 240p streaming, the equal interest user will handover to the LTE network with 360p streaming in order to gain a better reputation. Simultaneously, multimedia server delivering 240p streaming through WLAN to the balanced user could provide superior reputation which exceeds the reputation achieved from LTE network, until the WLAN network performance decreases and can only support the 144p streaming, the balanced user will handover to the LTE network with 360p streaming.

This result is consistent with the study on YouTube, in which the largest majority of resolution switches are between 360p to 480p [150]. Fig. 6.7 shows the interface of the video quality switch in the YouTube App[25] on an Android phone.

In this context, a 480p video with quality-oriented user preference is employed in the simulation for performance evaluation. The detailed simulation parameters are listed in Table 6.5.

[25] YouTube - https://play.google.com/store/apps/details?id=com.google.android.youtube&hl=en

Table 6-5 Simulation Setup

Parameters	Value
Duration of the simulation	360 seconds
Number of UEs	1 UE with mobility
UE mobility	Direction; Speed = 1m/s
LTE eNodeB Antenna Model	Isotropic Antenna Model
Wi-Fi standard	802.11 g
Traffic Model	CBR
Path Loss Model	Nist Error Rate Model
Streaming Bitrate	2.5 Mbps

The cellular-only scheme [151] is used for baseline comparison. The location information of the APs, home and office are shown as yellow points in Fig. 6.9, and the position of handover performed by different solutions are marked with red points.

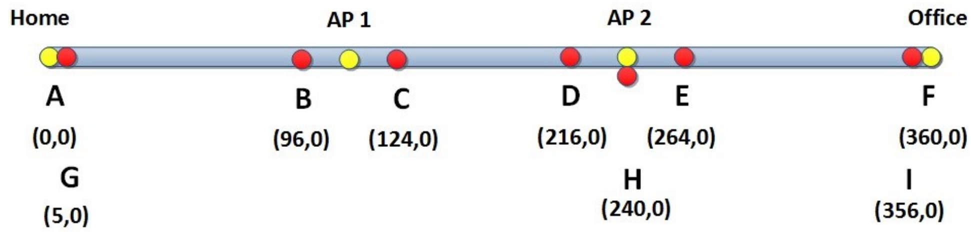


Figure 6-9 Handover Position

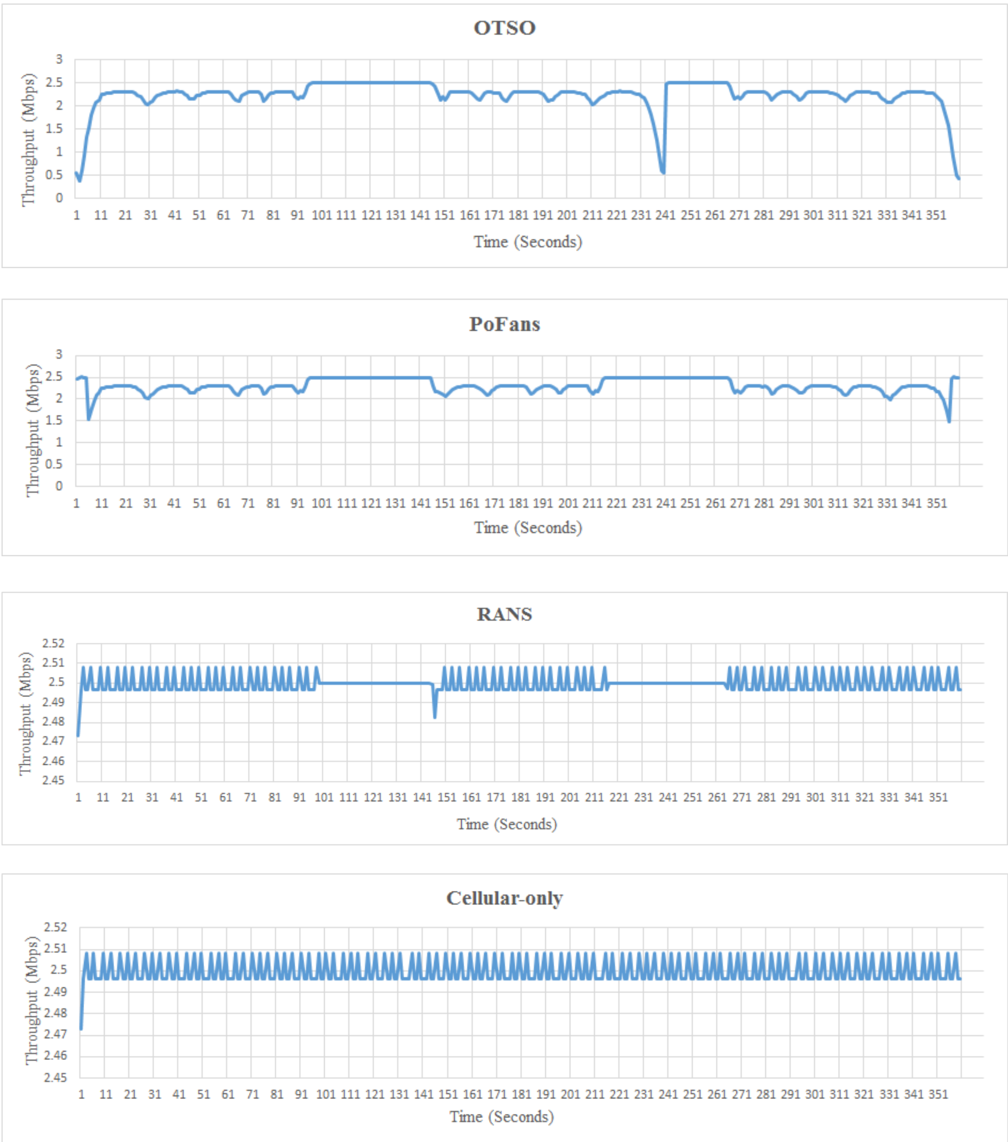


Figure 6-8 Overall Throughput for Four Solutions

Fig. 6.8 shows the throughput when four different solutions are employed OTSO, PoFans, RANS and LTE cellular-only. MN deploying OTSO scheme uses the WLAN 1 from point A to Point H, then handovers to WLAN 2 to the end.

MN using PoFans scheme switches to different networks as it always selects the network with the best PoFans utility value. At point A, MN connects to LTE and handovers to WLAN 1 at point G. WLAN 1 will be used until MN moves to point D where the PoFans utility value of WLAN 2 becomes is higher than that for WLAN 1, then MN handovers to WLAN 2 and continues its movement to point I, where the PoFans utility value of LTE is higher than that of WLAN 2, so MN handovers to LTE and uses it to the end.

Similarly, MN using RANS always handovers to the network with the highest reputation score. Therefore, MN connects to LTE at point A and LTE will be used until MN arrives at point B, where the WLAN 1's reputation score is higher than that of LTE, and MN handovers to WLAN 1. Later on, at point C, since WLAN 1's reputation score decreases and LTE has the best reputation, MN handovers to LTE until MN moves to point D. At point D a similar situation with the one at point B happens and therefore MN handovers to WLAN2 and uses it until it handovers back to LTE at point E. As expected, MN using the cellular-only approach does not handover to other networks and keeps using LTE.

The PoFans utility value and RANS reputation score of three networks are illustrated in Fig. 6.10 and Fig. 6.11, respectively.

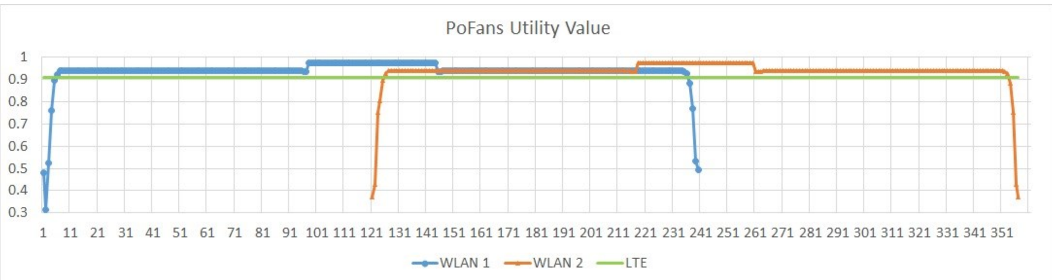


Figure 6-10 PoFans Utility Value for WLAN 1&2 and LTE

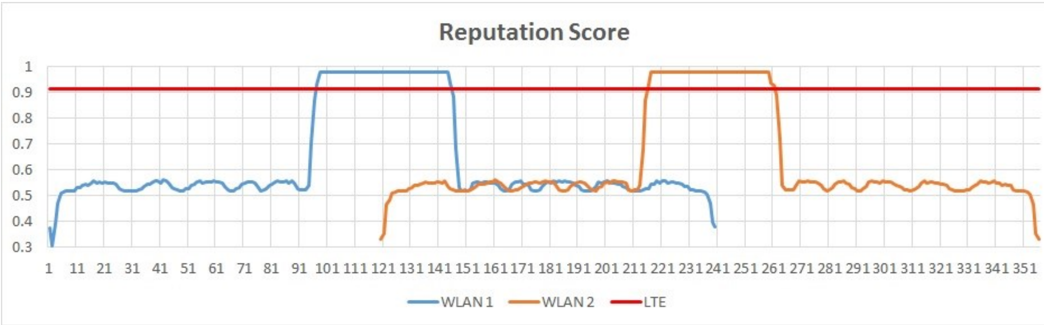


Figure 6-12 Reputation Score for WLAN 1&2 and LTE

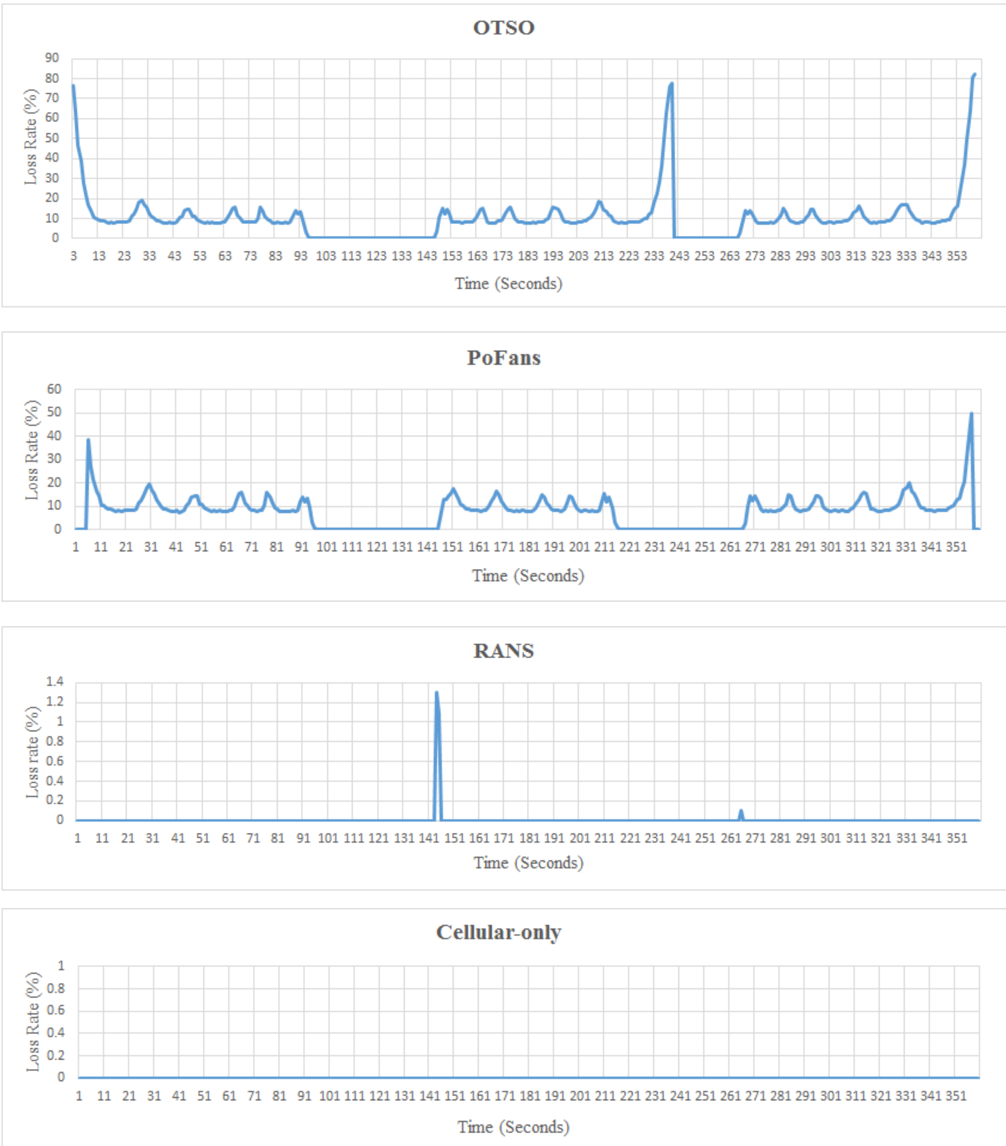


Figure 6-11 Loss Rate for the Four Solutions

The loss rate when employing the four solutions in turn is displayed in Fig. 6.12.

Compared to OTSO and PoFans, RANS gets a significant decrease of the loss rate during the video streaming. In this work, in order to evaluate the perceived video experience, ITU G.1070 is used via eq. (6.10) and the result is shown in Fig 6.13.

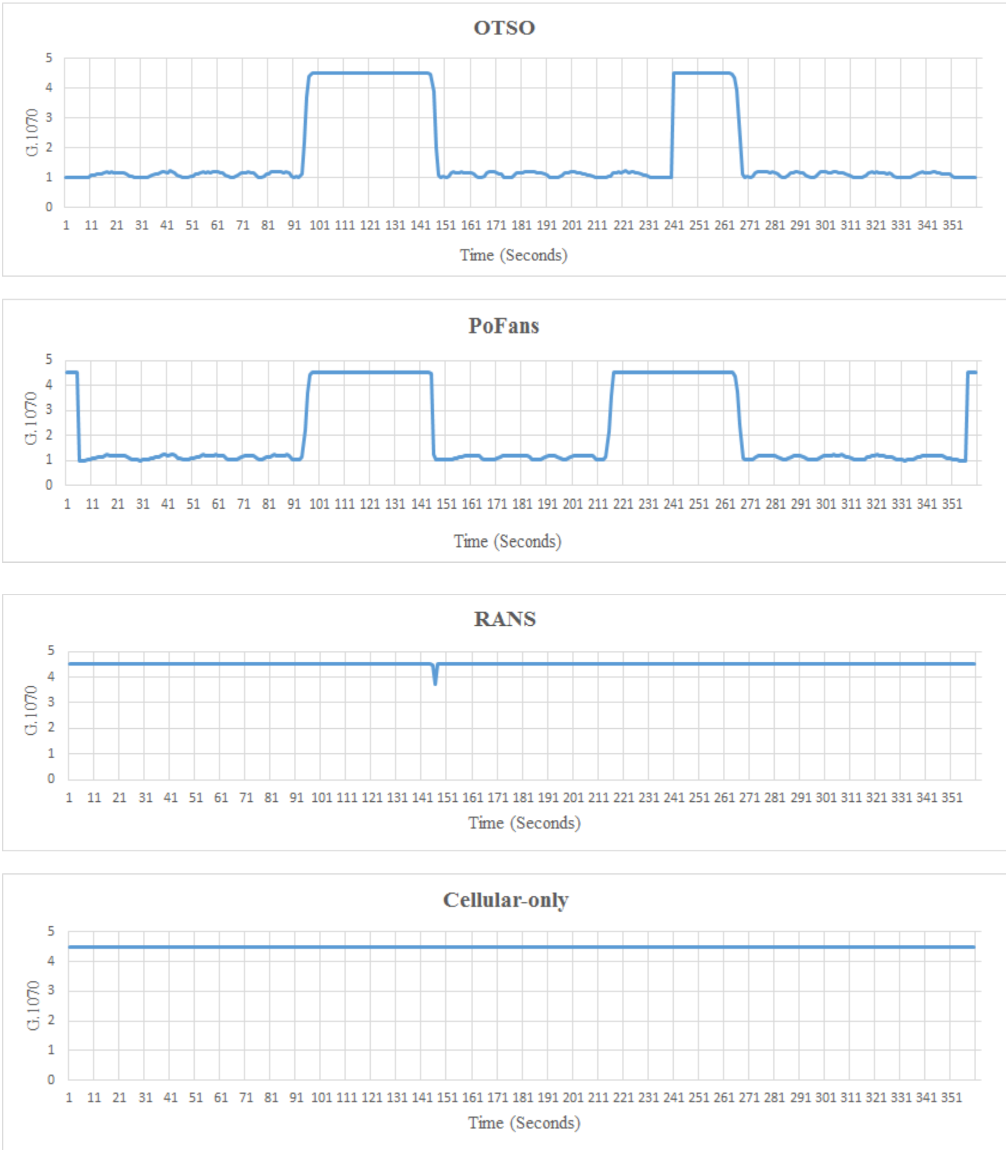


Figure 6-13 G.1070 under Four Solutions

The average G.1070 quality value is illustrated in Fig. 6.14. Compared to OTSO, PoFans and cellular-only, RANS value achieves an increase of over 144% and 109% and a decrease lower than 0.01%, respectively.

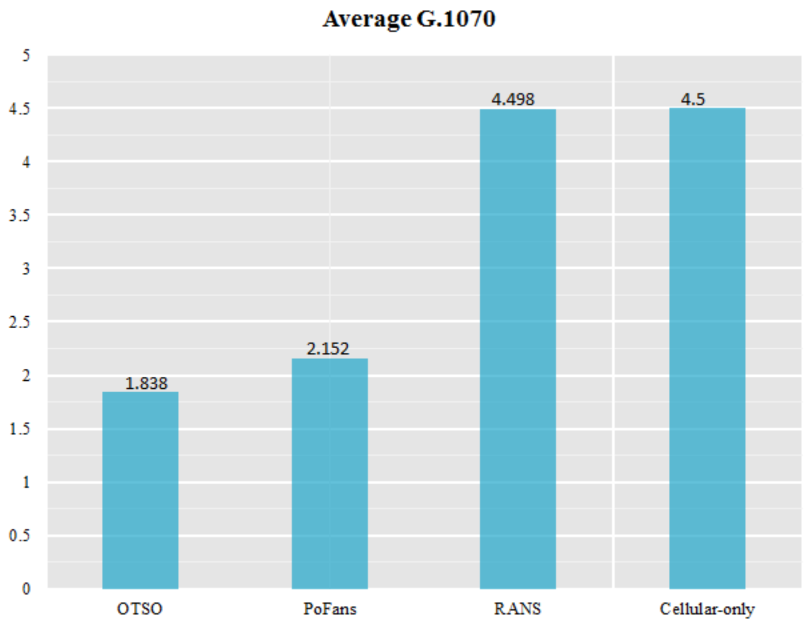


Figure 6-14 Average G.1070 Value under Four Solutions

Finally, the total monetary cost and energy consumption are illustrated in Fig. 6.15 and Fig. 6.16, respectively.

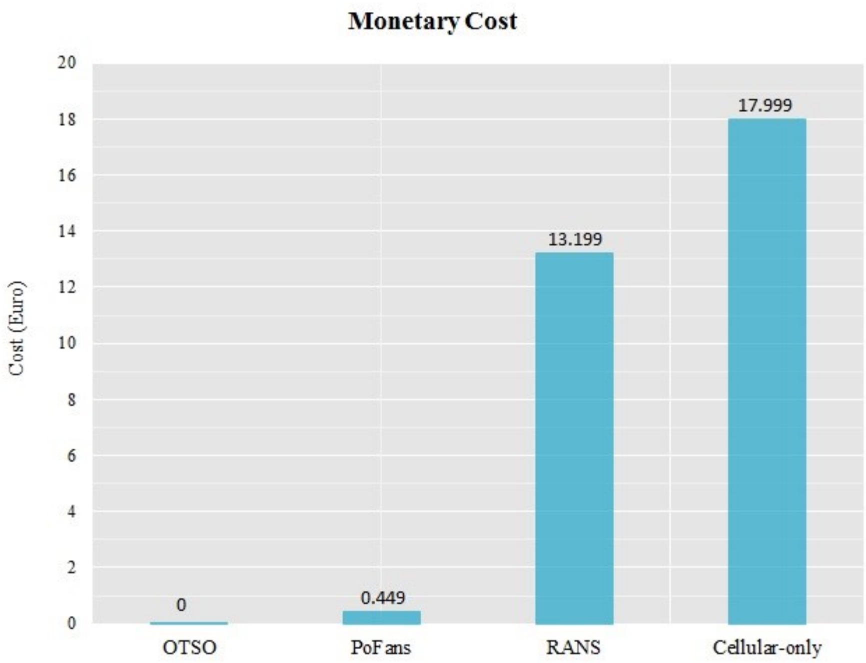


Figure 6-15 Monetary Cost under Four Solutions

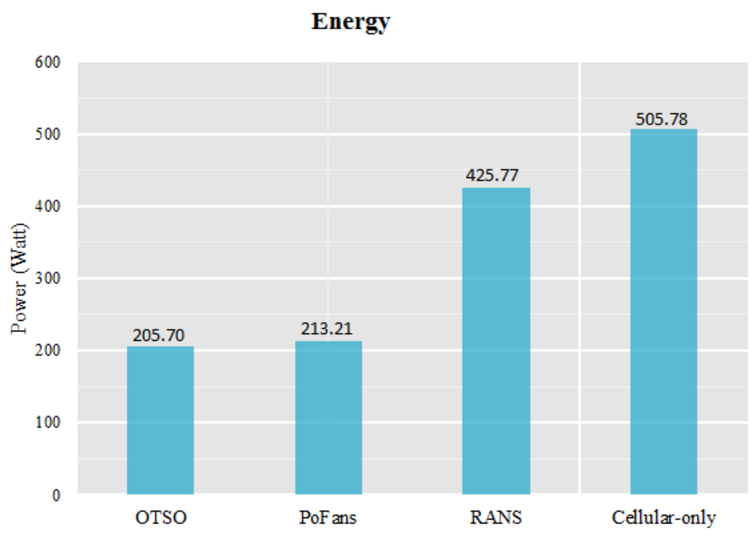


Figure 6-16 Energy Consumption under Four Solutions

In comparison with the cellular-only scheme, RANS reduces the energy consumption by 15.8% and monetary cost by 26.67%.

6.5 Multi-user RANS Testing

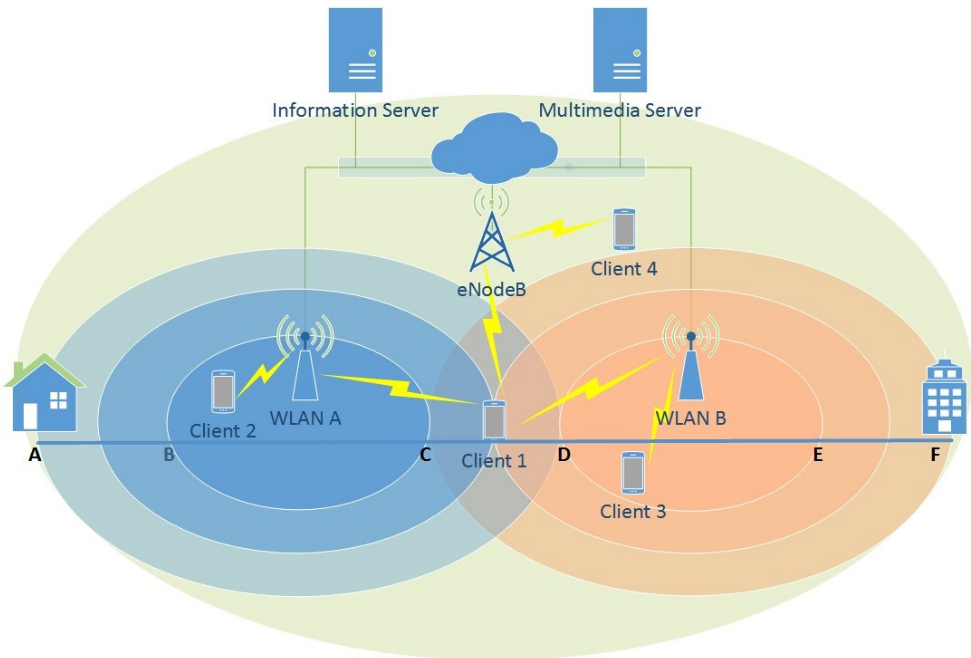


Figure 6-17 Multi-user Simulation Scenarios

Consider a scenario with multiple users in a wireless heterogeneous network environment illustrated in Fig. 6.17. The environment consists of three networks: WLAN A, WLAN B and LTE and their details are the same as described in Section 6.4. Three additional clients were added to the scenario, one in the coverage area of each these networks as follows: Client 2 in WLAN A, Client 3 in WLAN B and Client 4 in LTE, respectively. Client 1 is mobile and is the client of focus in this scenario. He/she moves from point A to point F, the same as in the previous scenario introduced in Section 6.4. Clients 2, 3 and 4 are static and will not execute any handover during the whole simulation. However they will update the network ranking value every 60 seconds. Client 1 also updates the network ranking value each 60 seconds, but being mobile executes handover based on RANS when necessary.

Table 6-6 Network Utility Ranking Value

Time	60	96	120	144	180	216	240	264	300	356	360
WLAN I											
Client I	0.556956	0.930764	0.978286	0.934508	0.551757	0.527751	0.380438				
Client 2	0.978286		0.978286		0.978286		0.978286		0.978286		0.978286
WLAN 2											
Client I			0.331561	0.536453	0.554752	0.930767	0.978286	0.934511	0.55653	0.331561	
Client 3	0.978286		0.978286		0.978286		0.978286		0.978286		0.978286
LTE											
Client I	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186
Client 4	0.91186		0.91186		0.91186		0.91186		0.91186		0.91186

Table 6.6 lists all the network ranking values received from four clients. Then, by using the network ranking value and reputation utility described in equation (6.13) and equation (6.14), the network reputation utility factor is computed and listed in Table 6.7. Finally, by using the utility-based reputation function described in equation (6.12) with network reputation weight w_r = 1 and network stabilization weight w_s = 0, the network reputation values are calculated and listed in the Table 6.7.

Table 6-7 Network Reputation Utility Factor & Network Reputation Value

	60	96	120	144	180	216	240	264	300	356	360
WLAN1											
ur	1	0.967615	0.991428	0.97641	0.986808	0.986808	0.990877	0.990877	0.993095	0.993095	0.994491
Ri	0.556956	0.930764	0.946605	0.926497	0.538741	0.520789	0.375419				
WLAN 2											
ur	1	1	1	1	1	0.976685	0.992603	0.979343	0.988204	0.988204	0.991701
Ri			0.331561	0.536453	0.554752	0.930767	0.955477	0.927598	0.545034	0.32765	
LTE											
ur	1	1	1	1	1	1	1	1	1	1	1
Ri	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186	0.91186

By using the values from Table 6.7, it can be seen how Client 1-deployed RANS determines the selection of the best reputation network as it progresses from Point A to Point F. In this context Client 1 will connect to the LTE network until it arrives in Point B at 96 seconds and handover to WLAN A. After 144 seconds, when it passes Point C, Client 1 will handover back to the LTE network. Later on, Client 1 will handover to WLAN B from Point D to Point E when time reaches 216 seconds until 264 seconds and finally will connect back to the LTE network until Point F.

6.6 Chapter Summary

This paper proposes RANS, a Reputation-oriented Access Network selection strategy for HetNets. RANS combines the utility theory with the reputation theory to build a reputation-based system between users and networks in a HetNets environment. RANS takes into consideration user preferences, energy consumption of the mobile device, the quality of the multimedia applications, and the monetary cost of the network to select the best value network that satisfies the users' needs and provides incentives for the user-network interaction to maintain cooperation in the long term by integrating a reputation-based system.

Simulation results show that RANS supports the "Always Best Experienced" paradigm. Specifically, RANS achieves a considerable reduction of the loss rate during the video streaming. In terms of the average G.1070 value, compared to OTSO and PoFans, RANS value increases up to 144%. Besides, in comparison with the cellular-only scheme, RANS reduces the energy consumption by 15.8% and the monetary cost by 26.67%.

CHAPTER 7: Network Reputation-based Stereoscopic 3D Video Delivery in HetNets

Abstract

In this chapter, the third thesis contribution: the Network Reputation-based Stereoscopic 3D Video Delivery in Heterogeneous Networks (NRQ-3D) is described as the instance for stereoscopic 3D video delivery in Network Reputation-based Quality Optimization of Video Delivery (NRQOVD) solutions. The recent advances in both wireless technologies and mobile devices, fuelled by increased user interest, have driven the latest development of mobile 3D video services. However, limited wireless bandwidth is one of the critical challenges for mobile 3D video delivery, especially as the 3D content requires higher bandwidth than the conventional 2D video. This chapter presents in details the proposed network reputation-based stereoscopic 3D video quality enhancement scheme in heterogeneous networks. It employs a network reputation module which reports network quality based on the quality of service-related parameters (i.e. throughput, signal strength, delay, and loss) and price aspects. The proposed solution selects the best candidate networks for the device using the info from the network reputation module. The IETF Multipath TCP (MPTCP) protocol is used for delivering the 3D video content to the mobile devices due to the higher throughput supported. Different 3D video components (i.e. color stream and depth stream) are delivered via separate sub-MPTCP flows and synchronized at the receiver. Simulation-based testing shows how the proposed solution

improves the throughput, delay and estimated 3D video quality in different delivery situations.

7.1 Introduction

In the last decade, 3D video has been introduced to home through 3DTV, 3D gaming and 3D movies. Diverse codec solutions for the 3D video have been developed including: i) two-view stereo video coding (Color Anaglyph 3D) [11], ii) video and depth coding (Depth Stereoscopic 3D) [12] and iii) multi-view video coding (MVC) [13]. In general, a single Stereoscopic 3D video stream consists of both color and depth information. This results in the 3D video delivery service require higher bandwidth than necessary for the traditional 2D video stream. The emerging LTE-A [5] and 802.11ac [2] standards provide significant improvements in terms of bandwidth and are very good for delivering 3D video sequences. Bandwidth resource allocation for 2D video streams in heterogeneous networks has been extensively studied [19], however, additional work is needed to propose efficient scheduling schemes for 3D video.

Currently, most mobile devices have access to different networks as they are equipped with multiple radio interfaces. By employing the MPTCP [14] protocol, the mobile devices can concurrently use multiple streams to transport content, including utilizing multiple interfaces as Fig 7.1 illustrates. In the context of 3D video delivery, the 3D video stream can be decomposed into different components according to the coding methods employed for the 3D video [8]: left and right views, in the two-view stereo video coding; video and depth streams in video and depth coding; and several views plus depth information in MVC. Different streams can be used to transfer these different components.

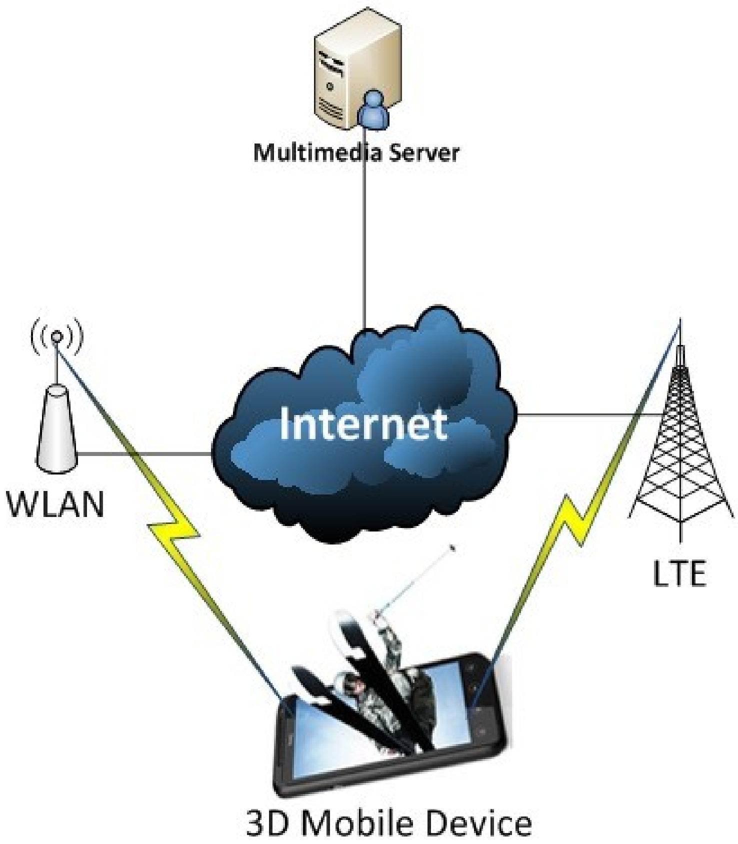


Figure 7-1 3D Video Delivery Using LTE and WiFi

In our previous works [20] [152], a network reputation mechanism was introduced to help enhance the content quality across various unscalable wireless networks. In this chapter, we make use of the reputation to select the most appropriate set of networks. A Network Reputation-based Quality-aware 3D video delivery (NRQ-3D) scheme is proposed that makes use of the MPTCP protocol in order to balance the traffic among a set of networks and trades-off QoS and monetary cost.

The structure of this chapter is as follows: Section 7.2 discusses the detailed information about the 3D video delivery scheme and system architecture. The algorithms are described in section 7.3 Section 7.4 introduces the simulation scenarios and the analysis of results. Section 7.5 presents the chapter summary.

7.2 NRQ-3D System Architecture

7.2.1 3D Video Delivery using MPTCP

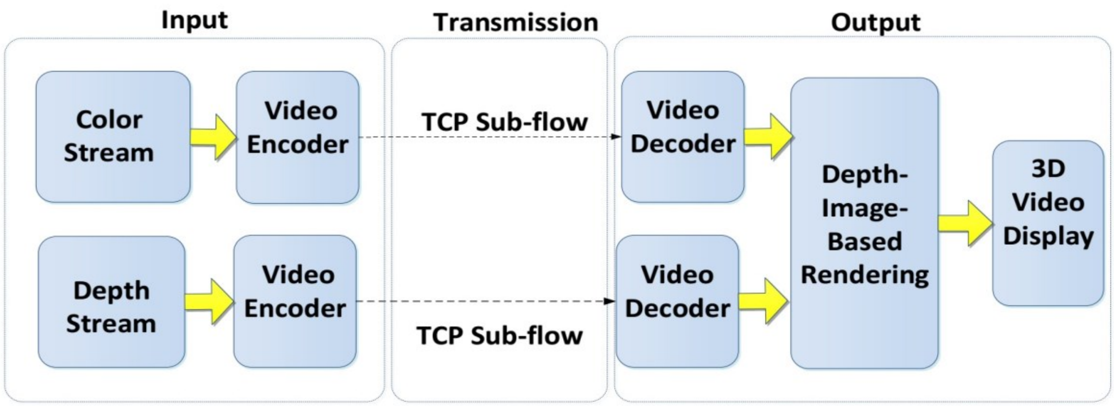


Figure 7-2 Architecture of 3D Video Delivery Scheme

Fig. 7.2 shows the overview architecture of 3D Video Delivery Scheme. It uses the depth-enhanced based 3D video representation, which reproduces the stereo video (left view and right view) from video plus depth data format [153][154]. The input data consists of two components: the color stream and the depth stream. Both color stream and depth stream are encoded by using a standard video encoder (i.e. H.264/MPEG-4 AVC). During the transmission, each stream will be transmitted via separate MPTCP sub-flows. At the receiver, the two streams are sent to separate video decoders and are processed using the Depth-Image-Based Rendering (DIBR) methodology, which use the per-pixel depth information to warp the original image points into the desired novel view [154] [155]. Finally, the 3D video is generated and displayed on the 3D mobile device screen.

7.2.2 Reputation-based 3D Video Delivery

The block architecture of the proposed NRQ-3D is illustrated in Fig. 7.3. Two wireless network technologies, LTE and WLAN, are considered. The proposed scheme is deployed at the application layer providing high flexibility to the existing video delivery protocols and mobile devices. The NRQ-3D is distributed and consists of a server side

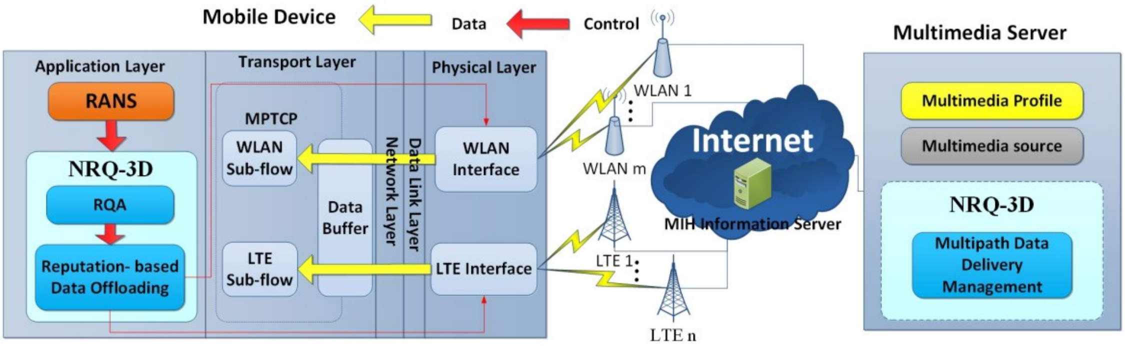


Figure 7-3 NRQ-3D Block Structure

component (Multimedia Server) and a client side component (Mobile Device). The LNS architecture is built on top of the IEEE 802.21 MIH standard, thus both system components are MIH-enabled entities and exchange information via MIH Information Server. The NRQ-3D requires both system components support multi-path transmission protocol on the transport layer, in this thesis, MPTCP is specified.

The NRQ-3D on Multimedia Server is a Multipath Data Delivery Management module (MDDM) that active the transmission require from Mobile Device, and manage the content delivered of each sub-link which give different sequence list of the different content packets to different sub-link.

The NRQ-3D on Mobile Device consists of two main modules in the application layer: Reputation-based 3D Video Delivery Quality Enhancement Algorithm (RQA) and Reputation-based Data Offloading (RDO).

1) RQA cooperates with Reputation-oriented Access Network Selection mechanism (RANS), which is mentioned in Chapter 6 and driven the RDO to execute commands. The principles behind the functionality of RQA and the corporation with RANS are described next.

- RANS's goal is to generate the candidate network list and select the network with the highest reputation for each interface based on historical and updated reputation data. Where the reputation represents the network performance based on current network QoS and user's QoE.

- RQA compares the reputations of the two interfaces from RANS and selects the interface with the highest reputation to transmit the higher bitrate stream (i.e. color stream). And the other one has the transmission of the second stream.
- During the transmission, RANS continuously monitors the QoS parameters (i.e. throughput, signal strength, delay) for each active network.
- RQA sends control commands (i.e. *idle, handover, transmit, exchange)* to RDO. *idle* command to the interface will stop the transmission, *handover* command to the interface will execute a handover, *transmit* command to the interface will start a transmission, and *exchange* command to exchange the transmission content between two or more interfaces.

2) RDO's major functionality steps are as follows.

- Sends the *idle* command to either the LTE or WLAN interfaces when the reputation is lower than the threshold.
- Sends the *handover* command and alterative network information to the WLAN or LTE interface.
- Sends the *transmit* command to both interfaces to transmit content when RQA sets or changes the transmission link.
- Send the *exchange* command to both interfaces when the reputation of the one interface is lower than another interface.

Once the system is initialized, RANS starts to monitor the available networks and RQA selects the best candidate network for each interface, by sending the *transmit* command through RDO to connect to the multimedia server. The interfaces with higher and lower reputation networks are used to transmit the color and the depth streams, respectively. The color and depth streams received from their respective network interfaces will be sent to the data buffer module in the MPTCP structure. Both streams will be synchronized using the DIBR module at the receiver and displayed at the 3D-enabled device.

During the transmission, RANS keeps monitoring in the available networks, including the currently used networks and computes the network reputation levels. If the reputation of the LTE network is lower than the minimum threshold, the RQA sends an idle command through RDO to stop the multipath transmission. If RANS detected that the reputation of the available WLAN network is higher than the differential threshold plus the reputation of the current working WLAN network, then RQA sends a handover command through RDO to the WLAN interface. Exchange command will be sent through RDO to both interfaces when the reputation of the WLAN network is lower than that of the LTE network. The same principle can be applied to multiple networks from N available. The major issues are the availability of network interfaces (M) and associated increase in energy consumption which eventually reduces the benefit of using multiple streams of a content delivery.

7.3 Algorithms and Decision Process

As the QoS requirements for 3D video delivery applications are much higher than for normal multimedia applications [156], the minimum throughput and cost of the available network need to be considered. So RANS includes these two parameters.

At initialization, RANS generates the candidate network list with the reputation score and selects the best reputation network for each network interface. Once RQA gets the candidate network list and the network with the highest reputation for each interface from RANS, RQA starts aiming to enhance the quality of 3D video delivery. The pseudo-code of the decision process handled by RQA is described below:

Algorithm: Reputation-based 3D Video Delivery Quality Enhancement Algorithm (RQA)

Input:

$[Cl_i]$- candidate network list for LTE interface;

$[Cw_i]$- candidate network list for WiFi interface;

Rl_{max}- highest reputation in LTE networks;

Rw_{max}- highest reputation in WiFi networks;

N_l - number of candidate network in list $[Cl_i]$;

N_w - number of candidate network in list $[Cw_i]$;

f – frequency of schedule to update reputation;

t – current time;

Procedure:

```
t=0;
while (t = 0 or t%f = 0 ) do
{
  if(Nl != 0 and Nw != 0)
  {
     if(Rwmax ≥ Rlmax )
     {
         set Linkv =0; // color link using WiFi
         set Linkd =1;// depth link using LTE
     }else{
            set Linkv =1;// color link using LTE
            set Linkd =0; // depth link using WiFi
            }end if
  }else{
         if(Nl = 0 and Nw = 0)
        {
           wait f;
        }else{
              if(Nl = 0)
              {
                 set idle = 0;// idle the LTE interface
              }else{
                 set idle = 1;// idle the WiFi interface
```

```
                }end if
            } end if
        }
    run RANS;
}
```

The complexity of the RQA is $O(n)$.

7.4 Simulation-based Testing and Result Analysis

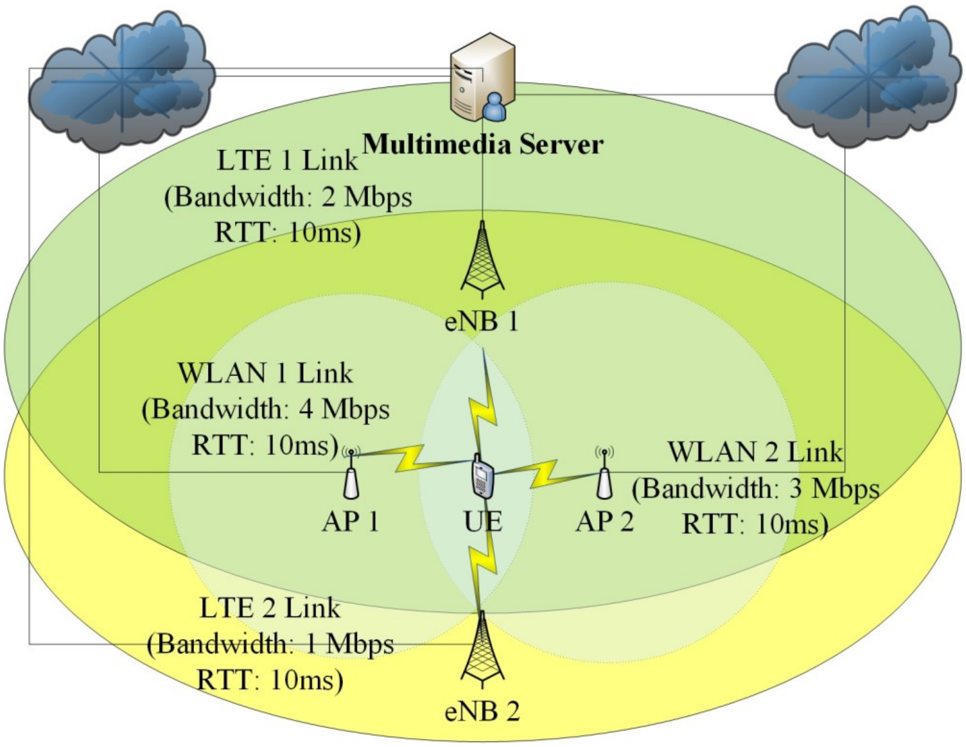


Figure 7-4 Network Topology Used in Simulation

The performance of the proposed NRQ-3D was evaluated using NS version 3.17 with the Direct Code Execution (DCE) [157] package. The performance of the 3D video delivery in heterogeneous networks was assessed in terms of average throughput, delay, Peak Signal-to-Noise Ratio (PSNR) and No reference objective Video Quality Metric (NVQM) [158].

The test-bed is based on NS-3.17 and DCE provides the possibility to directly execute real applications running over the NS-3 with actual network protocols [157]. MPTCP used is released by the Linux kernel Multi-Path TCP project [159] and is based on Linux 3.5.7 version of the kernel.

7.4.1 Simulation Test-bed Setup

The network topology used in this simulation is presented in Fig. 7.4. It involved six nodes: one node is used as the multimedia server, two are LTE eNB nodes (labeled eNB 1 and eNB 2), two are WLAN access points (labeled AP 1 and AP 2) and the last one is a wireless user equipment (UE). The distance between UE and eNB 1 node was 3000 meters, and the link denoted LTE 1 was set to 2Mbps bandwidth and RTT of 10ms. The distance between UE and eNB 2 nodes was 4000 meters, and the link denoted LTE 2 was set to 1Mbps bandwidth and RTT of 10ms. The distance between UE and AP 1 was 10 meters, and the link denoted WLAN 1 was set to 4Mbps bandwidth and RTT of 10ms. The distance between UE and AP 2 was 50 meters, and the link named WLAN 2 was set to 3Mbps bandwidth and RTT of 10ms.

Based on this topology, four test cases were considered with different available communication links between UE and the multimedia server, as follows:

- Case 1: LTE 1 and WLAN 1 links were established.
- Case 2: LTE 1 and WLAN 2 links were established.
- Case 3: LTE 2 and WLAN 2 links were established.
- Case 4: LTE 2 and WLAN 1 links were established.

Whenever two links established between UE and multimedia server, other two links will be idle (e.g. LTE 1 and WLAN 2 will be idle during the simulation in case 4).

7.4.2 Test Scenarios

Two scenarios were designed in this simulation for NRQ-3D performance assessment:

Scenario 1: Both LTE and WLAN links were established between UE and multimedia server. One constant bit-rate (CBR) 4 Mbps data stream, which aggregates all 3D video components, was sent from the multimedia server to UE using MPTCP as the transport protocol. Simulation considered all four cases indicated in the previous section and lasted 50 seconds.

Scenario 2: Two CBR data streams were sent from the multimedia server to the UE: a 3.2 Mbps color stream and a 0.8 Mbps depth stream. The simulations consider cases 2 and 3 only, as case 1 achieves very good quality and little additional improvement can be obtained and case 4 is similar to case 2. The other settings were the same as in Scenario 1.

The ratio between the bitrates of two streams in scenario 2 is 4:1. This ratio was computed according to [128], which shows that for the transmission of the depth map approximately 20% of the total source coding bit rate in H.264/AVC format is required. Big Bunk Bunny[26] is a cartoons video and is typically used for research proposes. For testing, the bitrate of the stereoscopic 3D video version was 6 Mbps, the video-depth version was 3 Mbps and the depth stream was approximately 0.75 Mbps. The framerate was 30 fps.

7.4.3 Quality Assessment

[26] Big Bunk Bunny http://bbb3d.renderfarming.net/download.html

To assess the user-perceived quality of the videos, this paper uses an estimation of PSNR: based on throughput and loss, as shown in eq. (7.1) [71].

$$PSNR = 20\log_{10}\left(\frac{Max\_Bitrate}{\sqrt{(Exp\_Thr - Crt\_Thr)^2}}\right) \quad (7.1)$$

In eq. (7.1), *Max_Bitrate* is the maximum data rate of the transmitted stream, *Exp_Thr* is the expected throughput and *Crt_Thr* is the actual average throughput.

Additionally, as the 3D video might be associated with different human perception in terms of quality than the 2D video, No reference Video Quality Metric for 3D video quality assessment (NVQM) was also used. NVQM, proposed in [158] and whose formula is shown in eq. (2), is an objective metric to estimate the 3D video quality by considering the bitrate and packet loss

$$V_{3D_q} = a1 + a2 * e^{\left(-\frac{Ppl_v}{a3 + a4 * e^{-\frac{Br_v}{a5}}}\right)} \quad (7.2)$$

In eq. (7.2), *Pplv* is the packet loss ratio, and *Brv* is the stream bitrate. *a1, a2, a3, a4 and a5* are constant coefficients determined from subjective testing, with default value recommend in [158].

7.4.4 Result Analysis

7.4.4.1. Scenario 1 Environment

Scenario 1 investigated the effect of using RANS for data transmission with MPTCP protocol with Opportunistic Retransmission Mechanism (MPTCP-ORM) [160]. Based on RANS, by computing the utilities of the four networks in the given network topology, the reputation of four networks were generated in Table 7.1 and listed like this: $R_{WLAN1} > R_{WLAN2} > R_{LTE1} > R_{LTE2}$.

Table 7-1 Reputation Score of Four Links

	R_{WLAN1}	R_{WLAN2}	R_{LTE1}	R_{LTE2}
Reputation Score	**0.960387**	0.960301	0.886619	0.746022

The best WiFi and the best LTE were selected following the results of performance from the table 7.1, then for the four possible cases: case 1 (WLAN1+LTE1), case 2 (WLAN1+LTE2), case 3 (WLAN2+LTE2), case 4 (WLAN2+LTE1), by running RANS, case 1 will be selected for the mobile device to receive the video traffic.

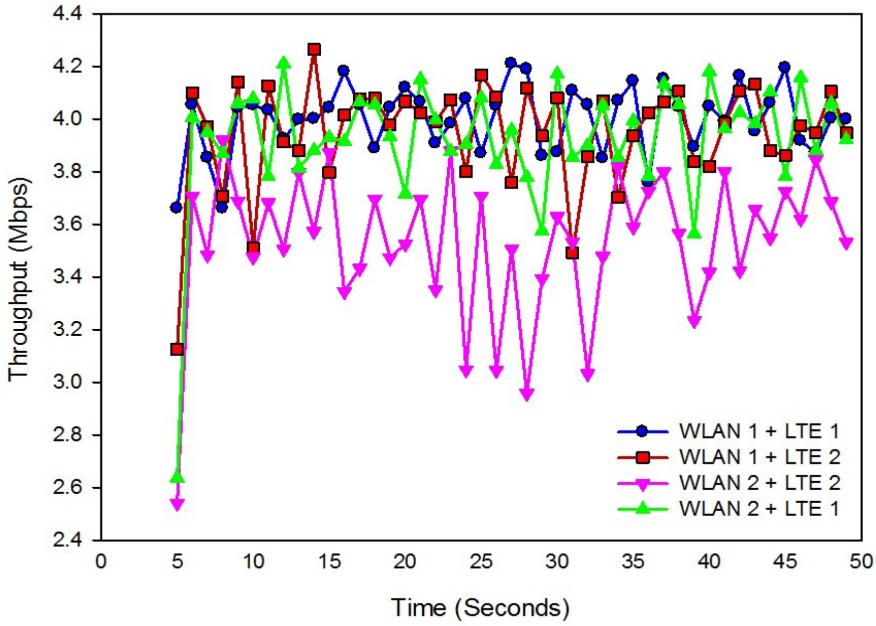


Figure 7-5 Throughput in the Four Cases in Scenario 1

Fig. 7.5 illustrated the throughput in the four cases, WLAN 2 + LTE 2 case with both two lower reputation score interface get the worst throughput compared to other 3 cases. Fig. 7.6 illustrated the delay in the four cases, the delay of WLAN 2 + LTE 2 case is the higher than WLAN 2 + LTE 1 case, which is already higher than other two cases.

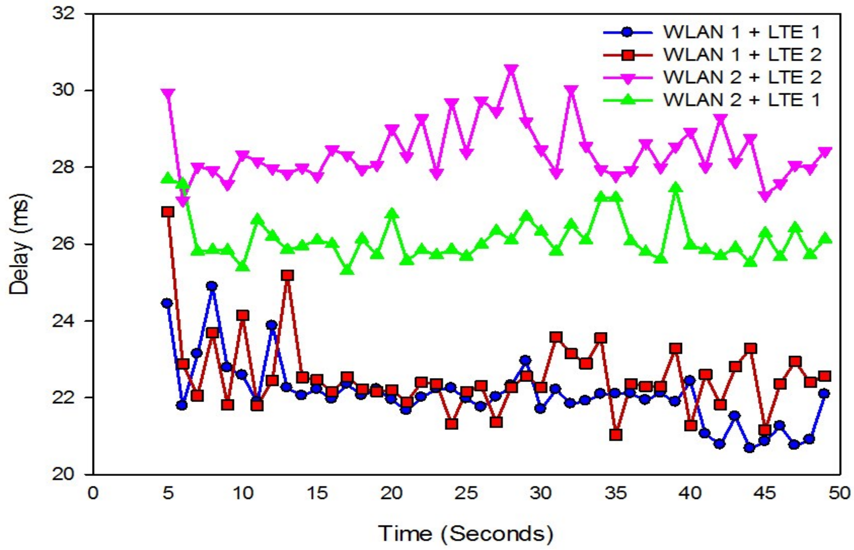


Figure 7-6 Delay in the Four Cases in Scenario 1

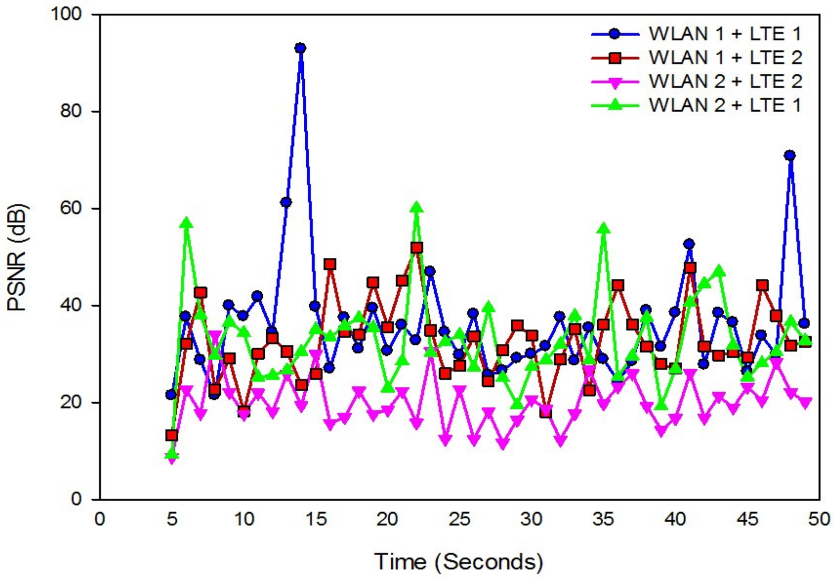


Figure 7-7 Estimated PSNR of Four Cases with Scenario 1

As shown in Fig. 7.7, the case 3 with WLAN 2 and LTE 2 interface present the worst PSNR value among all 4 cases. The NVQM in the four cases 1 are displayed in Fig. 7.8, case 1 WLAN 1 + LTE 1 achieves the highest NVQM value 3.66, which represent good video quality. The NVQM value for the other 3 cases shows that the 3D video stream they deliver have poor or even bad quality levels.

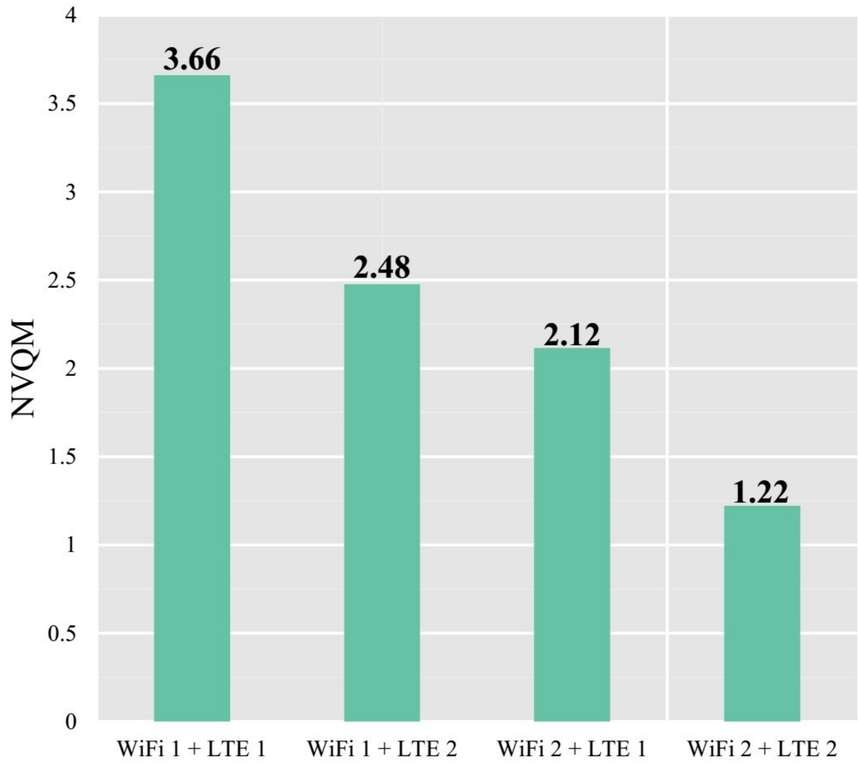


Figure 7-8 NVQM of Four Cases with Scenario 1

Table 7-2 Average Throughput, Delay and PSNR in Scenario 1

	WLAN 1 + LTE 1	WLAN 1 + LTE 2	WLAN 2 + LTE 2	WLAN 2 + LTE 1
Average Throughput (Mbps)	**3.999**	3.942	3.532	3.921
Average Delay (ms)	**22.08**	22.57	28.42	26.13
Average PSNR	**36.13**	32.54	20.10	32.79

According to Table 7.2 and Fig. 7.8, case 1 WLAN 1+ LTE 1 with RANS achieves the best result among four cases. The average throughput in case 1 is with 1.3%, 13.2% and 2% better than in case 2, case 3 and case 4, respectively. The average delay in case 1 is with 2.2%, 22.3% and 15.4% lower than that in case 2, case 3 and case 4, respectively. PSNR in case 1 is 11% better than that in case with 2, 79.7% higher than that obtained case3 and 10.1% better than the value in case 4. NVQM in case 1 is with 1.2 points higher than in case 2, with 2.4 points higher than in case3 and with 1.5 points higher than in case 4.

In this context, by running RANS to choose the best reputation score interface, 3D video transmission via MPTCP-ORM will result in best video quality.

7.4.4.2. Scenario 2 Environment

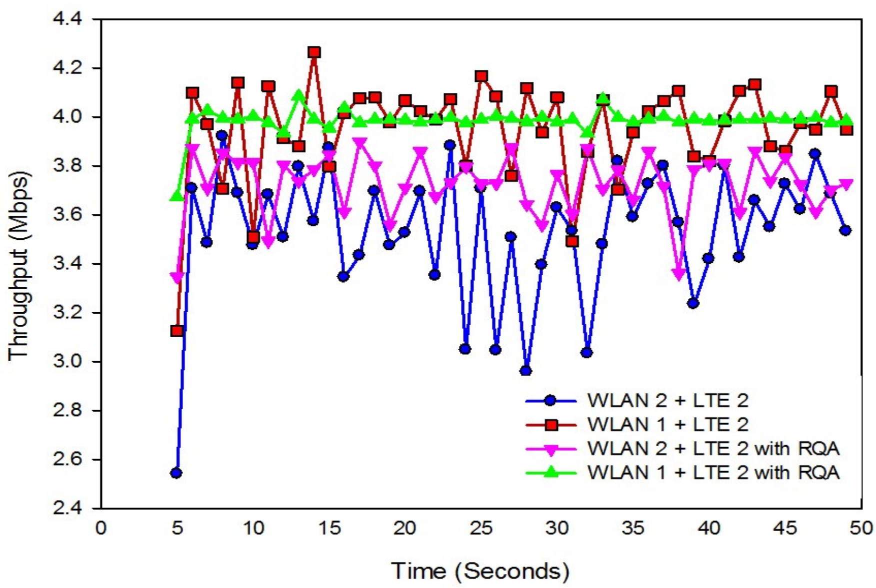


Figure 7-9 Overall Throughput in Case 2 and Case 3 on both Scenario 1 and Scenario 2

Scenario 2 investigated the effect of RQA when the 3D video is transmitted divided into color and depth streams via MPTCP-PRM by using RANS. As the reputation of WLAN link is better than that of the LTE link, from RQA, the color stream with higher bitrate will be transmitted via WLAN and the depth stream via LTE.

The performance of case 2 and case 3 is compared to that in Scenario 1.

Fig. 7.9 illustrated the throughput in case 2 and case 3 on both Scenario 1 and 2, case 2 WLAN 1 + LTE 2 and case 3 WLAN 2 + LTE 2 with RQA gain significant benefit compared to the two cases without RQA. Fig. 7.10 illustrates the delay in case 2 and case 3 on both Scenario 1 and Scenario 2. By using RQA the delay of case 3 WLAN 2 + LTE 2 decreased, but for case 2, the delay increased 2-4 ms after using RQA.

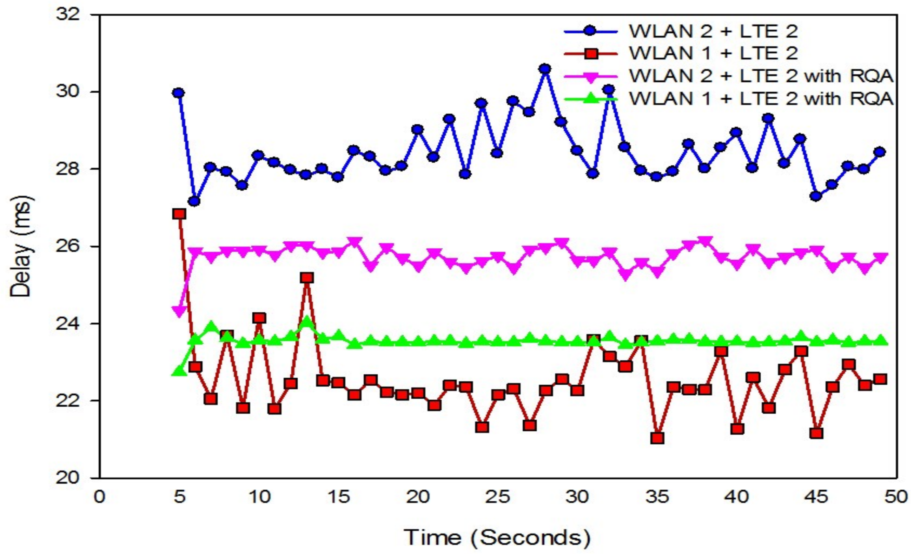


Figure 7-10 Delay in Case 2 and Case 3 on both Scenario 1 and Scenario 2

In terms of user perceived quality, Fig. 7.11 shows the PSNR in case 2 and case 3 on both Scenario 1 and Scenario 2. In case 2 WLAN 1 + LTE 2 with RQA significant benefit results compared to the case 2 without RQA. Fig. 7.12 illustrates the NVQM in case 2 and case 3 on both Scenario 1 and Scenario 2. By using RQA, NVQM of the two cases increases, but for case 3 with WLAN 2 + LTE 2, there is only 0.07 improvement, and still result in bad video quality.

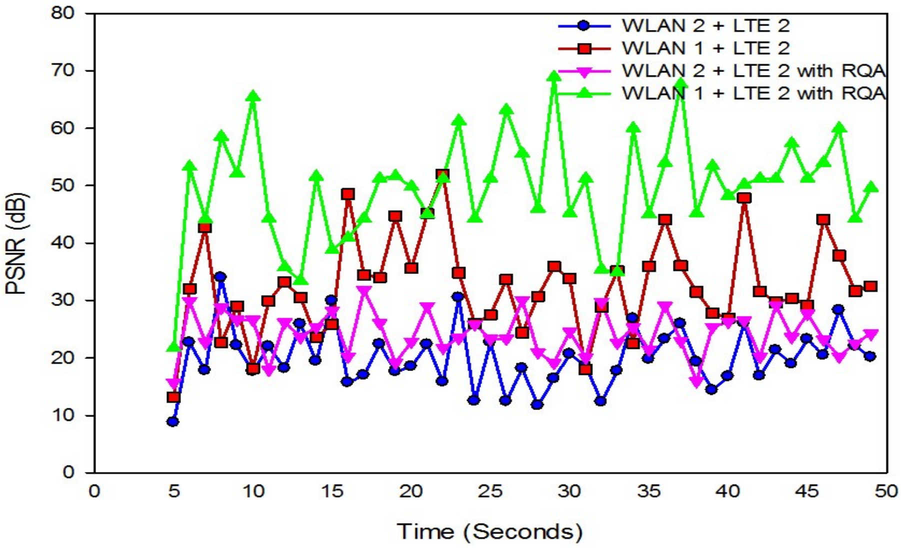


Figure 7-11 Estimated PSNR in Case 2 and Case 3 on both Scenario 1 and Scenario 2

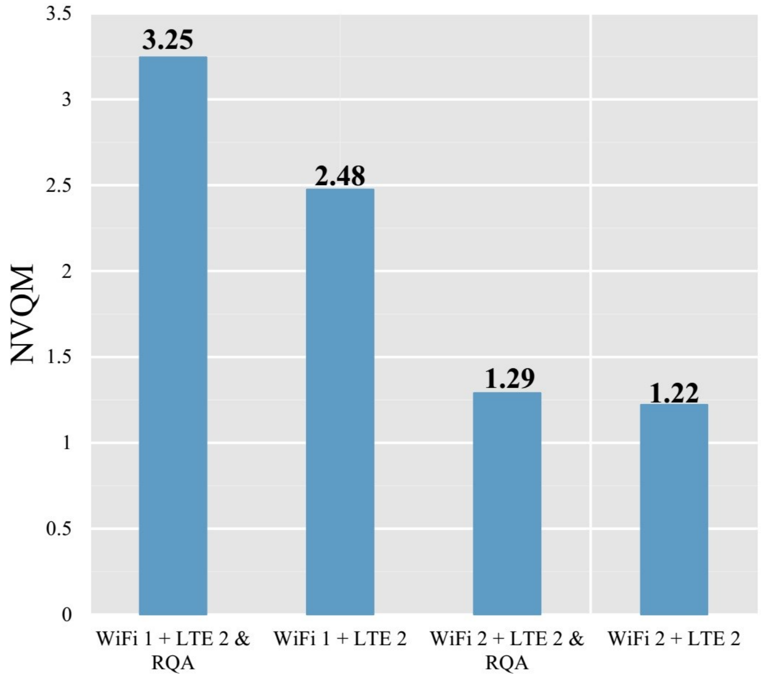


Figure 7-12 NVQM in Case 2 and Case 3 on both Scenario 1 and Scenario 2

Table 7-3 Average Throughput, Delay and PSNR in Case 2 and Case 3 on Both Scenario 1 and Scenario 2

	WLAN 1 + LTE 2 with RQA	WLAN 1 + LTE 2	WLAN 2 + LTE 2 with RQA	WLAN 2 + LTE 2
Average Throughput (Mbps)	3.984	3.947	3.728	3.532
Average Delay (ms)	23.54	22.57	25.73	28.42
Average PSNR	49.64	32.54	24.22	20.10

According to Table 7.3, the average throughput of case 3 is with 5.5% higher when RQA is employed than without RQA. The average throughput of case 2 is also 0.9% higher when RQA is used than otherwise. The average delay in case 3 with RQA is 9.3% lower than without, but at the same time, the average delay of case 2 with RQA is 4.3% higher than without RQA. In case 3 PSNR with RQA is 20.4% higher than without RQA and in case 2 is 2.5% higher with RQA than without RQA. Fig.7.12 shows a similar trend for NVQM which is 5.6% higher in case 3 with RQA than when no RQA is employed, and 53.4% higher in case 2 with RQA than without RQA.

In this context, when the 3D video is transmitted divided into color and depth streams and using RANS with RQA via MPTCP protocol with Opportunistic Retransmission Mechanism significant improvements are obtained in terms of throughout and estimated perceived video quality levels.

7.5 Chapter Summary

This chapter proposed NRQ-3D in heterogeneous networks. NRQ-3D selects the best candidate networks based on a Reputation-based 3D Video Delivery Quality Enhancement Algorithm corporate by RANS, which is proposed to report the network quality based on quality of service-related parameters and monetary cost aspects, for the smartphone to receive the 3D video content from the multimedia server by two or more linked flows via MPTCP protocol.

Different 3D video components are also delivered via separate sub-MPTCP flows based on the different reputation value and synchronized at the receiver. Simulation results show the significant quality of service and enhanced user perceived quality benefits when using the proposed solution in comparison with MPTCP-ORM approaches: the average throughput, PSNR and NVQM are increased up to 5.5%, 20.4% and 53.4%, respectively.

CHAPTER 8: Conclusions and Future Work

Abstract

In this chapter, the conclusions of this thesis are presented with regard to the three main contributions along with their corresponding results. Finally, several directions for future research work are indicated.

8.1 Conclusions

8.1.1 Overview

In recent years, smart mobile devices have become one of the basic necessities in people's life. Rich multimedia content delivery-based services over wireless networks have become attracted very much user interest and the high QoE requirement have made researchers focus on both new wireless technology innovations and various delivery solutions in the current heterogeneous wireless environments deployed in the most urbanised areas. People enjoy the convenience of being connected to the Internet all the time everywhere, when stationary or when moving, so meeting these high quality multimedia content transmission requirements in the complex heterogeneous wireless environments is a major challenge.

8.1.2 Contributions

In this context, in order to provide the "Always Best Experience" for mobile users, there are two main limitations: 1) the mobile devices are not smart enough to select the best network in these complex heterogeneous wireless environments. Given the diversity of services, users, devices, etc.; 2) the video delivery system is not efficiently using the existing multi-interface in its interaction with the heterogeneous wireless network environments. This thesis has presented the research work and outcomes of designing a reputation based mechanism for quality optimization of video delivery in such heterogeneous wireless network environments. The following contributions are provided to the advancement of the current state of the art:

- Proposal of a Location-aware Network Selection (LNS) mechanism for heterogeneous wireless networks that enables the mobile device to select the network that offers the best support for content delivery along the user's path from the available network list for each independent network interface on MN. Using network simulation 3 (NS3), simulation results show that the LNS mechanism can eliminate the low feasibility networks and select the highest possibility serving network to keep the network service with "Always Best Connected". Particularly, by comparison with OTSO and Wiffler, the proposed LNS can achieve the decrease up to 60% on the number of handover times and the maximum reduction of 58.8% in downloading time.
- Proposal of a novel Reputation-oriented Access Network Selection mechanism (RANS) for improved video delivery quality in heterogeneous wireless network environments that enables the selection of the network with the best performance for multimedia transmissions. RANS, based on the IEEE 802.21 MIH standard mechanism, supports gathering of delivery performance information from the currently connected users from different areas within each network. The information is aggregated and disseminated to other mobile users, which can make an informed quality-oriented decision when selecting the candidate network for handover.

  Simulation results show that RANS supports the "Always Best Experienced" paradigm. Specifically, RANS achieves a considerable reduction of the loss rate during the video streaming. In terms of the average G.1070 value, compared to OTSO and PoFans, RANS value increases up to 144%. Besides, in comparison with the cellular-only scheme, RANS reduces the energy consumption by 15.8% and the monetary cost by 26.67%.

- Proposal of a Network Reputation-based Quality-aware 3D video delivery scheme (NRQ-3D) which makes use of reputation to select the most appropriate set of networks to deliver the content over multiple simultaneous flows. A network reputation module is proposed to report the network quality based on the quality of service-related parameters (i.e. throughput, signal strength, delay, and loss) and price aspects. IETF Multipath TCP (MPTCP) protocol is used for delivering the 3D video content to mobile devices. Different 3D video components (i.e. color stream and depth stream) are delivered via separate sub-MPTCP flows and are synchronized at the receiver. A reputation adaptive mechanism is proposed to balance the traffic among the schemed set of networks and find the best trade-off between QoS and monetary cost.

  Simulation results show the significant quality of service and enhanced user perceived quality benefits when using the proposed solution in comparison with MPTCP-ORM approaches: the average throughput, PSNR and NVQM are with up to 5.5%, 20.4% and 53.4% higher, respectively.

8.2 Future Work

All three solutions proposed are heuristic-designed based mainly on modeling and simulation involving wireless network topologies. In order to make the next steps towards users "Always Best Experience" in relation to multimedia streaming services in future wireless networks, the realist in LNS/RANS/NRQ-3D real testbed deployments need to be done in the near future. These would enable perceptional tests to be performed and results to be compared with those from simulations. Finally, an improved quality metric for the network reputation mechanism could be proposed.

The current contributions are validated by simulations; however, the simulations are limited to using theoretic values and considering reduced number of elements. So, for complete research outcome assessment, a real test bed for video-based multipath transmission in heterogeneous wireless environments is needed.

First, a MPTCP-kernel Linux server and a MPTCP-kernel Android mobile device should be prepared, and multipath video transmission between the server and client should be performed. Once the video transmission test bed has been built, the subjective tests will take place.

Second, a 3D MPEG-DASH streaming server should be prepared and 3D adaptive streaming quality assessment via subjective tests should take place. Currently, left-right view based 3D video is supported only. Work will merge the multipath transmission and the rich video content adaptive streaming (VR/depth 3D).

Third, an information monitor application on Android phone should be developed to gather location-related information with timestamps, signal strength, and video streaming based QoS information (e.g. throughput, delay, and jitter). The current version has limited accuracy and works for cellular networks only. Future work on server side reputation calculation and information storage need to be performed, too.

Fourth, the current network reputation mechanism considers various network QoS parameters only. Future work can consider developing a new quality metric based on the data from different subjective tests. It is hoped that the video delivery scheme based on the newly improved reputation metric which brings a benefit of the overall reputation mechanism, too.

On the one hand, the more parameters involved, the higher complexity of the reputation system in terms of computing time and cost. Machine learning is one of the most powerful methods to solve the decision/recommendation problem and could be utilized to further enhance the reputation-based network selection and video adaptation system solutions proposed in this thesis.

Bibliography

[1] Cisco, (2016)"Cisco visual networking index: Global mobile data traffic forecast update, 2015–2020 white paper," [Online]. Available: http://www.cisco.com/c/en/us/solutions/collateral/service-provider/visual-networking-index-vni/mobile-white-paper-c11-520862.html. Accessed: Dec. 12, 2016.

[2] IEEE P802.11ac. (2011) "Specification framework for TGac", IEEE 802.11-09/0992r21.

[3] Nicopolitidis, P., (2003) "Wireless networks", John Wiley and Sons.

[4] 3GPP TS 21.101, (2016)"Technical Specifications and Technical Reports for a UTRAN-based 3GPP system", [Online]. Available: http://www.tech-invite.com/3m21/tinv-3gpp-21-101.html. Accessed: Dec. 12, 2016

[5] LTE-Advanced Rapporteur, (2009) "RP-090939: 3GPP Submission Package for IMT-Advanced", www.3gpp.org, 3GPP TSG RAN, meeting 45, Seville, Spain.

[6] European Standard ETSI EN 300 744 DVB-T STANDARD V1.5.1 ED. (2004) "Digital Video Broadcasting, Framing Structure, Channel Coding and Modulation for Digital Terrestrial Television", DVB Project.

[7] European Standard ETSI EN 302 755 V1.3.1 (2012). "Digital Video Broadcasting; Frame structure channel coding and modulation for a second generation digital terrestrial television broadcasting system (DVB-T2)", DVB Project.

[8] ITU-R M.1645 (2003) "ITU —Framework and overall objectives of the future development of IMT-2000 and systems beyond IMT-200", Switzerland.

[9] IEEE 802.21-2008, (2009) Standard for Local and Metropolitan Area Networks- Part 21: Media Independent Handover Services, IEEE Computer Society.

[10] Bi, T., Trestian, R., Muntean, G-M., (2013.) Reputation-based Network Selection Solution in Heterogeneous Wireless Network Environments, 12th Information Technology & Telecommunication Conference (IT&T 2013).

[11] Su, G.-M., Lai, Y.-C., Kwasinski, A., Wang, H., (2011) "3D video communications: Challenges and opportunities," International Journal of Communication Systems, vol. 24, no. 10.

[12] Hill, L. and Jacobs, A., (2006). 3-D liquid crystal displays and their applications. Proceedings of the IEEE, 94(3), pp.575-590.

[13] ITU-T and ISO/IEC JTC 1, (2010) "Advanced video coding for generic audiovisual services," ITU-T Recommendation H.264 and ISO/IEC 14496-10(MPEG-4 AVC).

[14] Bonaventure, O., Handley, M., Raiciu, C., (2012) “An Overview of Multipath TCP,” USENIX login, vol. 37, no. 5, pp. 17–23.

[15] Muller, K., Merkle, P. and Wiegand, T., (2011). 3-D video representation using depth maps. Proceedings of the IEEE, 99(4), pp.643-656.

[16] Radicati, F., and Mavrakis, D., (2012) "White papers – Understanding today’s smartphone user: Demystifying data usage trends on cellular and Wi-Fi networks," [Online]. Available: http://www.informatandm.com/wp-content/uploads/2012/02/Mobidia_final.pdf. Accessed: Dec. 12, 2016.

[17] Rayment, S., and Bergstrom, J., (2012) “Achieving carrier-grade Wi-Fi in the 3GPP world,” Ericsson Review., Stockholm, Sweden.

[18] Cheung, M.H., and Huang, J., (2015). Dawn: Delay-aware Wi-Fi offloading and network selection. IEEE Journal on Selected Areas in Communications, 33(6), pp.1214-1223.

[19] Yuan, Z., and Muntean, G.M., (2013). A prioritized adaptive scheme for multimedia services over IEEE 802.11 WLANs. IEEE Transactions on Network and Service Management, 10(4), pp.340-355.

[20] Bi, T., Trestian, R., Muntean, G.M., (2013) “Reputation-based Network Selection Solution for Improved Video Delivery Quality in Heterogeneous Wireless Network Environments” IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB 2013).

[21] IEEE. (2016) IEEE Std 802®-2014: IEEE Standards for Local and Metropolitan Networks: Overview and Architecture [Online]. Available: http://standards.ieee.org/about/get/802/802.html.

[22] ANSI/IEEE STD 802.11, 1999 Edition (2003), ―Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications.

[23] IEEE 802.11a-1999, (1999) IEEE Standard for Local and Metropolitan Area Networks Specific Requirements – Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications High Speed Physical Layer in the 5 GHz Band.

[24] IEEE 802.11b-1999, (1999) IEEE Standard for Local and Metropolitan Area Networks Specific Requirements – Part 11: Wireless LAN Medium Access

Control (MAC) and Physical Layer (PHY) Specifications High Speed Physical Layer Extension in the 2.4 GHz Band, September 16.

[25] IEEE 802.11g-2003, (2003) IEEE Standard for Local and Metropolitan Area Networks Specific Requirements – Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications Amendment 4: Further Higher Data Rate Extension in the 2.4 GHz Band.

[26] IEEE Standard for information technology (2009)—Telecommunications and information exchange between systems—Local and metropolitan area networks—Specific requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications -Amendment 5: Enhancements for Higher Throughput.

[27] IEEE 802.11ac (2013)- IEEE Draft Standard for IT - Telecommunications and Information Exchange Between Systems - LAN/MAN - Specific Requirements - Part 11: Wireless LAN Medium Access Control and Physical Layer Specifications - Amendment 4: Enhancements for Very High Throughput for operation in bands below 6GHz".

[28] IEEE 802.11ad-2012 (2012)- IEEE Standard for Information technology—Telecommunications and information exchange between systems--Local and metropolitan area networks--Specific requirements-- Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications Amendment 3: Enhancements for Very High Throughput in the 60 GHz Band.

[29] Andrews, J.G., Ghosh, A. and Muhamed, R., (2007). Fundamentals of WiMAX: understanding broadband wireless networking. Pearson Education.

[30] Australian Mobile Telecommunications Association (2003) Ten years of GSM in Australia. Available: https://www.accc.gov.au/system/files/4%20AMTA,%20Ten%20years%20of%20GSM%20in%20Australia,%20www.amta.gov.au.pdf [Accessed: 13-OCT-2016].

[31] 3GPP. (2016) About 3gpp home. [Online]. Available: http://www.3gpp.org/about-3gpp/about-3gpp

[32] Sauter, M., (2009) "Beyond 3G - Bringing Networks, Terminals and the Web Together: LTE, WiMAX, IMS, 4G Devices and the Mobile Web 2.0", 1st ed. John Wiley and Sons.

[33] Telecommunications Industry Association, (1995) "Tia/eia/is-95," Tech. Rep., Available: http://tia.nufu.eu/std/TIA%7CEIA%7CIS-95-A

[34] 3GPP. (2016) 3gpp releases. [Online]. Available: http://www.3gpp.org/specifications/releases

[35] ITU, (2003) ―Framework and overall objectives of the future development of IMT-2000 and systems beyond IMT-200, Switzerland, ITU-R M.1645.

[36] IEEE 802.21-2008, (2009) Standard for Local and Metropolitan Area Networks-Part 21: Media Independent Handover Services, IEEE Computer Society.

[37] 3GPP, (2014) "Access network discovery and selection function (ANDSF) management object (mo) (release 12)," Tech. Rep. TS-24.312.

[38] 3GPP, (2014) "Access to the 3gpp evolved packet core (EPC) via non-3gpp access networks; stage 3 (release 12)," Tech. Rep. TS-24.302.

[39] Open Mobile Alliance. (2016) "OMA Device Management V2.0" [Online]. Available: http://technical.openmobilealliance.org/Technical/technical-information/release-program/current-releases/oma-device-management-v2-0

[40] Farmer, R. and Glass, B., (2010). Building web reputation systems. "O'Reilly Media, Inc.".

[41] Zavadskas, E.K., Kaklauskas, A., Turskis, Z. and Tamošaitienė, J., (2009). Multi-attribute decision-making model by applying grey numbers. Informatica, 20(2), pp.305-320.

[42] ITU-T, (2008) "Quality of telecommunication services: concepts, models, objectives and dependability planning - terms and definitions related to the quality of telecommunication services," Tech. Rep. E.800.

[43] Aurrecoechea, C., Campbell, A.T. and Hauw, L., (1998). A survey of QoS architectures. Multimedia systems, 6(3), pp.138-151.

[44] ITU Y.1541 (2006) Network performance objectives for IP-based services - http://www.itu.int/rec/T-REC-Y.1541/en

[45] Nguyen-Vuong, Q.T., Ghamri-Doudane, Y. and Agoulmine, N., (2008), April. On utility models for access network selection in wireless heterogeneous networks. In NOMS 2008-2008 IEEE Network Operations and Management Symposium (pp. 144-151). IEEE.

[46] Norstad, J. (1999) An introduction to utility theory. Updated: Nov. 3, 2011. [Online]. Available: http://www.norstad.org/finance/util.pdf

[47] DaSilva, L. A., (2000) "Pricing for qos-enabled networks: A survey," Communications Surveys Tutorials, IEEE, vol. 3, no. 2, pp. 2-8.

[48] Yeh, C.-H., (2002) “A Problem-based Selection of Multi-attribute Decision-making Methods,” International Transactions in Operational Research, vol. 9, pp. 169-181.

[49] Song, Q. and Jamalipour, A., (2005) “A network selection mechanism for next generation networks,” in IEEE International Conference on Communications, (ICC).

[50] Trestian. R., (2012) “User-Centric Power-Friendly Quality-based Network Selection Strategy for Heterogeneous Wireless Environments”, Ph.D. Thesis. Dublin City University, [Online]. Available: http://doras.dcu.ie/16783/1/Ramona_Trestian_-_PhD_Thesis_Final.PDF

[51] Floyd, S., Handley, M. and Kohler, E., 2006. Datagram congestion control protocol (DCCP).

[52] Floyd, S., and Kohler, E., (2006.) “Profile for datagram congestion control protocol (DCCP) congestion control ID 2: TCP-like congestion control,” IETF RFC 4341.

[53] Floyd, S., Kohler, E and Padhye, J., (2006) “Profile for datagram congestion control protocol (DCCP) congestion control ID 3: TCP-friendly rate control (TFRC),” IETF RFC 4342.

[54] Ford, A., Raiciu, C., Handley, M., Barre, S. and Iyengar, J., (2011). Architectural guidelines for multipath TCP development (No. RFC 6182).

[55] Ford, A., Raiciu, C., Handley, M. and Bonaventure, O., (2013). TCP extensions for multipath operation with multiple addresses (No. RFC 6824).

[56] Dugue, G., Diop, C., Chassot, C. and Exposito, E., (2012), June. Towards autonomic multipath transport for infotainment-like systems. In 2012 IEEE International Conference on Communications (ICC) (pp. 6453-6457). IEEE.

[57] ISO/IEC 23009-1:2014 (2014) - Information technology -- Dynamic adaptive streaming over HTTP (DASH) -- Part 1: Media presentation description and segment formats. [Online] Available: http://www.iso.org/iso/home/store/catalogue_ics/catalogue_detail_ics.htm?csnumber=65274 [Accessed: 13-OCT-2014].

[58] 3GPP, (2014) "Transparent end-to-end packet-switched streaming service (pss) (release 12)," Tech. Rep. TS-26.234..

[59] Open IPTV Forum, (2014) " OIPF Release 2 specification volume 2a - http adaptive streaming [v2.3]," Tech. Rep.

[60] Stockhammer, T., (2011) "Dynamic Adaptive Streaming over HTTP-Standards and Design Principles", ACM Multimedia Systems, San Jose, CA, USA.

[61] Wu, F., Li, S., Sun, X. and Zhang, Y.Q., Microsoft Corporation, (2006). Seamless switching of scalable video bitstreams. U.S. Patent 6,996,173.

[62] Hassoun, D., (2009). Dynamic streaming in flash media server 3.5 part 1: overview of the new capabilities. [Online]. Available: http://www.adobe.com/devnet/adobe-media-server/articles/dynstream advanced pt1.html, [Accessed: 13-DEC-2016].

[63] Varsa, V. and Curcio, I., (2003). Transparent end-to-end packet switched streaming service (pss); rtp usage model (release 5). 3GPP TR26, 937, p.V1.

[64] Bi, T., Yuan, Z. and Muntean, G.M., (2014), June. Network reputation-based stereoscopic 3D video delivery in heterogeneous networks. In 2014 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (pp. 1-7). IEEE.

[65] Google, (2016) "Rendering Omni - directional Stereo Content," [Online]. Available: https://developers.google.com/vr/jump/rendering-ods-content.pdf, [Accessed: 30-NOV-2016].

[66] YouTube, (2016) "Upload virtual reality videos," [Online]. Available: https://support.google.com/youtube/answer/6316263?hl=en&ref_topic=2888648, [Accessed: 30-NOV-2016].

[67] Yuan, Z., Bi, T., Muntean, G.M. and Ghinea, G., (2015). Perceived synchronization of mulsemedia services. IEEE Transactions on Multimedia, 17(7), pp.957-966.

[68] BEREC, (2011) “A framework for Quality of Service in the scope of Net Neutrality”. [Online]. Available: http://berec.europa.eu/eng/document_register/subject_matter/berec/download/0/117-a-framework-for-quality-of-service-in-th_0.pdf, [Accessed: 13-NOV-2016].

[69] Carnec, M., Le Callet, P. and Barba, D., 2003, July. Full reference and reduced reference metrics for image quality assessment. In Signal Processing and Its Applications, 2003. Proceedings. Seventh International Symposium on (Vol. 1, pp. 477-480). IEEE.

[70] ITU-T RECOMMENDATION, P., 1999. Subjective video quality assessment methods for multimedia applications.

[71] Lee, S.B., Muntean, G.M. and Smeaton, A.F., 2009. Performance-aware replication of distributed pre-recorded IPTV content. IEEE Transactions on Broadcasting, 55(2), pp.516-526.

[72] Ke, C.H., Shieh, C.K., Hwang, W.S. and Ziviani, A., (2008). An Evaluation Framework for More Realistic Simulations of MPEG Video Transmission. J. Inf. Sci. Eng., 24(2), pp.425-440.

[73] Hagos, D. H., & Kapitza, R. (2013). Study on performance-centric offload strategies for LTE networks. 6th Joint IFIP Wireless and Mobile Networking Conference (WMNC), 1–10.

[74] Kwon, Y. M., Kim, J. S., Gu, J., & Chung, M. Y. (2013). Andsf-based congestion control procedure in heterogeneous networks. The International Conference on Information Networking 2013 (ICOIN), 547–550.

[75] GhasemiNajm, M., Sibley, M. J. N., & Jafarian, a. (2012). Combination of LTE and IMS to deliver location based services. 2012 20th Telecommunications Forum (TELFOR), 56–59.

[76] Kim, D. S., Noishiki, Y., Kitatsuji, Y., & Yokota, H. (2013). Efficient ANDSF-assisted Wi-Fi control for mobile data offloading. 2013 9th International Wireless Communications and Mobile Computing Conference (IWCMC), 343–348.

[77] Triantafyllopoulou, D., Guo, T., & Moessner, K. (2012). Energy efficient ANDSF-assisted network discovery for non-3GPP access networks. 2012 IEEE 17th International Workshop on Computer Aided Modeling and Design of Communication Links and Networks (CAMAD), 297–301.

[78] Park, H., Lee, H., & Chan, H. (2013). Gateway service for integration of heterogeneous networks using different interworking solutions. Advanced Communication Technology (ICACT), 2013 15th International Conference on, 489–494.

[79] Song, W., Chung, J., & Lee, D. (2009). Improvements to seamless vertical handover between mobile WiMAX and 3GPP UTRAN through the evolved packet core. Communications Magazine, IEEE, vol.47, no.4, 66–73.

[80] Nahas, M., Mjalled, M., Zohbi, Z., Merhi, Z., & Ghantous, M. (2013). Enhancing LTE WiFi Interoperability Using Context Aware Criteria for Handover Decision. Microelectronics (ICM), 2013 25th International Conference on, 1–4.

[81] Frei, S., Fuhrmann, W., Rinkel, a, & Ghita, B. V. (2011). Improvements to Inter System Handover in the EPC Environment. 2011 4th IFIP International Conference on New Technologies, Mobility and Security, 1–5.

[82] Doppler, K., Ribeiro, C. B., & Kneckt, J. (2011). On efficient discovery of next generation local area networks. 2011 IEEE Wireless Communications and Networking Conference, 269–274.

[83] Hu, S., Wang, X., & Shakir, M. Z. (2015). A MIH and SDN-based Framework for network selection in 5G HetNet: Backhaul requirement perspectives. 2015 IEEE International Conference on Communication Workshop, ICCW 2015, (BackNets), 37–43.

[84] Ahmed, A., Boulahia, L. M., & Gaïti, D. (2014). Enabling vertical handover decisions in heterogeneous wireless networks: A state-of-the-art and a classification. IEEE Communications Surveys and Tutorials, 16(2), 776–811.

[85] Sgora, A., Vergados, D. D., & Chatzimisios, P. (2010). An access network selection algorithm for heterogeneous wireless environments. The IEEE Symposium on Computers and Communications, 890–892.

[86] Lahby, M., Cherkaoui, L., & Adib, A. (2012). Network selection algorithm based on Diff-AHP and TOPSIS in heterogeneous wireless networks. Multimedia Computing and Systems (ICMCS), 2012 International Conference on.

[87] Lahby, M., Cherkaoui, L., & Adib, A. (2012). A survey and comparison study on weighting algorithms for access network selection. Wireless On-demand Network Systems and Services (WONS), 2012 9th Annual Conference on, 35–38.

[88] Wang, L., & Binet, D. (2009). MADM-based network selection in heterogeneous wireless networks: A simulation study. Wireless Communication, Vehicular Technology, Information Theory and Aerospace & Electronic Systems Technology, 2009. Wireless VITAE 2009. 1st International Conference on, 559–564.

[89] Godor, G., & Detari, G. (2007). Novel Network Selection Algorithm for Various Wireless Network Interfaces. Mobile and Wireless Communications Summit, 2007. 16th IST, pp. 1–5, 2007.

[90] Bakmaz, B., Bojkovic, Z., & Bakmaz, M. (2007). Network selection algorithm for heterogeneous wireless environment. Personal, Indoor and Mobile Radio Communications, 2007. PIMRC 2007. IEEE 18th International Symposium on, 1–4.

[91] Iera, A., Molinaro, A., Campolo, C., & Amadeo, M. (2006). An Access Network Selection Algorithm Dynamically Adapted to User Needs and Preferences. 2006 IEEE 17th International Symposium on Personal, Indoor and Mobile Radio Communications, 1–5.

[92] Song, Q., & Jamalipour, A. (2005). Network selection in an integrated wireless LAN and UMTS environment using mathematical modeling and computing techniques. Wireless Communications, IEEE, (June), 42–48.

[93] Song, Q., & Jamalipour, A. (2005). A network selection mechanism for next generation networks. Communications, 2005. ICC 2005. 2005 IEEE International Conference on (Volume: 2), 1418–1422.

[94] Sharma S., Baek I., Dodia Y., and Chiueh T.-C., "OmniCon: A mobile IP-based vertical handoff system for wireless LAN and GPRS links," Mobile Wireless Netw. , pp. 330 – 337, 2004.

[95] Jabban, A., Nasser, Y. and Hélard, M., (2012), April. SINR based network selection strategy in integrated heterogeneous networks. In Telecommunications (ICT), 2012 19th International Conference on (pp. 1-6). IEEE.

[96] Andreev, S., Gerasimenko, M., Galinina, O., Koucheryavy, Y., Himayat, N., Yeh, S. P., & Talwar, S. (2014). Intelligent access network selection in converged multi-radio heterogeneous networks. IEEE Wireless Communications, 21(6).

[97] Louta, M., & Zournatzis, P. (2011). Towards realization of the ABC vision: a comparative survey of access network selection. Computers and Communications (ISCC), 2011 IEEE Symposium on, 472–477

[98] Jaramillo, J. J., & Srikant, R. (2010). A game theory based reputation mechanism to incentivize cooperation in wireless ad hoc networks. Ad Hoc Networks, 8(4), 416–429.

[99] Jaramillo, J., & Srikant, R. (2007). DARWIN: distributed and adaptive reputation mechanism for wireless ad-hoc networks. Proc. MobiCom.

[100] Refaei, M., & Srivastava, V. (2005). A reputation-based mechanism for isolating selfish nodes in ad hoc networks. Mobile and Ubiquitous Systems: Networking and Services, 2005. MobiQuitous 2005. The Second Annual International Conference on, 3 – 11.

[101] Buchegger, S., Boudee, J. Le. (2005). Self-policing mobile ad hoc networks by reputation systems. Communications Magazine, IEEE, (July), 101–107.

[102] Buchegger S., Boudee, J. Le. (2014) "A robust reputation system for P2P and mobile ad-hoc networks". In Proceedings of the 2nd Workshop on the Economics of Peer-to-Peer Systems, Cambridge.

[103] Buchegger S., Mundinger, J., & Le Boudec, J. Y. et al., (2008) "Reputation systems for self-organized networks", IEEE Technology and Society Magazine, vol. 27, no. 1, pp. 41 – 47.

[104] Wang, M., Tao, F., Zhang, Y., & Li, G. (2010). An Adaptive and Robust Reputation Mechanism for P2P Network. 2010 IEEE International Conference on Communications, 1–5.

[105] Jurca, R., & Faltings, B. (2005). Reputation-based pricing of P2P services. Proceeding of the 2005 ACM SIGCOMM workshop on Economics of peer-to-peer systems - P2PECON '05, 144.

[106] Xiong, L., & Liu, L. (2004). Peertrust: Supporting reputation-based trust for peer-to-peer electronic communities. Knowledge and Data Engineering, IEEE Transactions on, 16(7), 843–857.

[107] Zekri, M., & Pokhrel, J. (2011). Reputation for Vertical Handover decision making. Communications (APCC), 2011 17th Asia-Pacific Conference on, (October), 318–323.

[108] Zekri, M., Jouaber, B., & Zeghlache, D. (2012). An enhanced media independent handover framework for vertical handover decision making based on networks' reputation. 37th Annual IEEE Conference on Local Computer Networks -- Workshops, 673–678.

[109] Giacomini, D., & Agarwal, A. (2012). Vertical handover decision making using QoS reputation and GM(1,1) prediction. 2012 IEEE International Conference on Communications (ICC), 5655–5659

[110] Giacomini, D., & Agarwal, A. (2013). Optimizing end user QoS in heterogeneous network environments using reputation and prediction. EURASIP Journal on Wireless Communications and Networking, 2013(1), 256.

[111] Trestian, R., Ormond, O., and Muntean G-M. (2011), "Reputation-based network selection mechanism using game theory," Physical Communication, vol. 4, no. 3, pp. 156–171, Sep. 2011.

[112] Chen S., Yuan Z., Muntean G.-M. (2013). An energy-aware multipath-TCP-based content delivery scheme in heterogeneous wireless networks. 2013 IEEE Wireless Communications and Networking Conference (WCNC), 1291–1296.

[113] Rahmati, A., Shepard, C., Tossell, C., & Zhong, L. (2013). Seamless TCP Migration on Smartphones without Network Support, Mobile Computing, IEEE Transactions on, pp. 1–14.

[114] Diop, C.; Gomez-Montalvo, J.; Dugue, G.; Chassot, C.; Exposito, E.. (2012). Towards a semantic and MPTCP-based Autonomic Transport Protocol for Mobile and Multimedia Applications. Multimedia Computing and Systems (ICMCS), 2012 International Conference on, 496 – 501.

[115] Wu, J., Yuen, C., Cheng, B., Wang, M., & Chen, J. (2016). Streaming High-Quality Mobile Video with Multipath TCP in Heterogeneous Wireless Networks. IEEE Transactions on Mobile Computing, 15(9), 2345–2361.

[116] Wu, J., Cheng, B., & Wang, M. (2016). Energy Minimization for Quality-Constrained Video with Multipath TCP over Heterogeneous Wireless Networks. 2016 IEEE 36th International Conference on Distributed Computing Systems (ICDCS), 487–496.

[117] Corbillon, X., Aparicio-Pardo, R., Kuhn, N., Texier, G. and Simon, G., 2016, May. Cross-layer scheduler for video streaming over MPTCP. In Proceedings of the 7th International Conference on Multimedia Systems (p. 7). ACM.

[118] Deng, S., Netravali, R., Sivaraman, A. and Balakrishnan, H., 2014, November. Wifi, lte, or both?: Measuring multi-homed wireless internet performance. In Proceedings of the 2014 Conference on Internet Measurement Conference (pp. 181-194). ACM.

[119] Rodriguez P., Chakravorty R., Chestereld J., Pratt I., and Banerjee S. (2014) "MARS: A Commuter Router Infrastructure for the Mobile Internet". In Proc. MobiSys.

[120] Xu, C., Liu, T., Guan, J., Zhang, H., & Muntean, G.-M. (2013). CMT-QA: Quality-Aware Adaptive Concurrent Multipath Data Transfer in Heterogeneous Wireless Networks. IEEE Transactions on Mobile Computing, 12(11), 2193–2205.

[121] Lee, S., Sriram, K., Kim, K., Kim, Y.H. and Golmie, N., 2009. Vertical handoff decision algorithms for providing optimized performance in heterogeneous wireless networks. IEEE transactions on Vehicular Technology, 58(2), pp.865-881.

[122] Luo, Y., Qi, B., Chen, Y., & Tang, L. (2012). Strengthen-Gray Relative Analysis access selection algorithm based on load balance in heterogeneous wireless networks. 2012 8th International Conference on Natural Computation, (Icnc), 843–847.

[123] Xue, P., Gong, P., Park, J. H., Park, D., & Kim, D. K. (2012). Radio Resource Management with Proportional Rate Constraint in the Heterogeneous Networks. IEEE Transactions on Wireless Communications, 11(3), 1066–1075.

[124] Qadir, J., Ali, A., Yau, K. L. A., Sathiaseelan, A., & Crowcroft, J. (2015). Exploiting the Power of Multiplicity: A Holistic Survey of Network-Layer Multipath. IEEE Communications Surveys and Tutorials, 17(4), 2176–2213.

[125] Li, X., & Cui, J. (2009). Real-time water resources allocation: methodology and mechanism. 2009 IEEE International Conference on Industrial Engineering and Engineering Management, 1637–1641.

[126] Gürler, C.G., Bağcı, K.T. and Tekalp, A.M., 2010, September. Adaptive stereoscopic 3D video streaming. In 2010 IEEE International Conference on Image Processing (pp. 2409-2412). IEEE.

[127] Kordelas, A., Politis, I., Lykourgiotis, A., Dagiuklas, T., & Kotsopoulos, S. (2013). An media aware platform for real-time stereoscopic video streaming adaptation. 2013 IEEE International Conference on Communications Workshops (ICC), 687–691, 2013.

[128] Kamolrat, B., Fernando, W.A.C., Mrak, M. and Kondoz, A., 2008. Joint source and channel coding for 3D video with depth image-based rendering. IEEE Transactions on Consumer Electronics, 54(2), pp.887-894.

[129] Alajel, K.M.; Wei Xiang, (2012) "Color plus depth 3-D video transmission with hierarchical 16-QAM," 3DTV-Conference: The True Vision - Capture, Transmission and Display of 3D Video (3DTV-CON), 2012, vol., no., pp.1-4, 15-17.

[130] Muntean, G.M., Perry, P. and Murphy, L., (2005). Subjective assessment of the quality-oriented adaptive scheme. IEEE Transactions on Broadcasting, 51(3), pp.276-286.

[131] Roberts, M., (2013). Understanding today's smartphone user: An updated and expanded analysis of data-usage patterns in six of the world's most advanced 4G LTE markets. White Paper, June.

[132] Lee, K., Lee, J., Yi, Y., Rhee, I. and Chong, S., 2013. Mobile data offloading: How much can WiFi deliver?. IEEE/ACM Transactions on Networking (TON), 21(2), pp.536-550.

[133] Bi T., (2011) "Application of propagation models to acquired digital terrain information Final report," School of Electronics Engineering, Dublin City University, Master Thesis.

[134] Balasubramanian, A., Mahajan, R. and Venkataramani, A., (2010), June. Augmenting mobile 3G using WiFi. In Proceedings of the 8th international conference on Mobile systems, applications, and services (pp. 209-222). ACM.

[135] Sankaran, C.B., (2012). Data offloading techniques in 3GPP Rel-10 networks: A tutorial. IEEE Communications Magazine, 50(6), pp.46-53.

[136] Cisco, (2014), Cisco Visual Networking Index: Global Mobile Data Traffic Forecast Update 2014–2019. White Paper c11-520862. Available on http://www.cisco. com/c/en/us/solutions/collateral/serviceprovider/visual-networking-index-vni/white_paper_c11-520862. html. [Accessed: 05-May-2015].

[137] Ericsson, (2016). "Ericsson mobility report – Ericsson," Ericsson.com, [Online]. Available: https://www.ericsson.com/mobility-report. Accessed: Dec. 19, 2016.

[138] Trestian, R., Ormond, O. and Muntean, G.M., (2015). Performance evaluation of MADM-based methods for network selection in a multimedia wireless environment. Wireless Networks, 21(5), pp.1745-1763.

[139] Huszak, A. and Imre, S., (2010), May. Eliminating rank reversal phenomenon in GRA-based network selection method. In Communications (ICC), 2010 IEEE International Conference on (pp. 1-6). IEEE.

[140] Bikmukhamedov, R., Yeryomin, Y. and Seitz, J., (2016), July. Evaluation of MCDA-based handover algorithms for mobile networks. In Ubiquitous and Future Networks (ICUFN), 2016 Eighth International Conference on (pp. 810-815). IEEE.

[141] Mahmud, K., Inoue, M., Murakami, H., Hasegawa, M. and Morikawa, H., (2004), September. Measurement and usage of power consumption parameters of wireless interfaces in energy-aware multi-service mobile terminals. In Personal, Indoor and Mobile Radio Communications, 2004. PIMRC 2004. 15th IEEE International Symposium on (Vol. 2, pp. 1090-1094). IEEE.

[142] Gustafsson, J., Heikkila, G. and Pettersson, M., (2008), October. Measuring multimedia quality in mobile networks with an objective parametric model. In 2008 15th IEEE International Conference on Image Processing (pp. 405-408). IEEE.

[143] Trestian, R., Moldovan, A.N., Muntean, C.H., Ormond, O. and Muntean, G.-M., (2012), June. Quality Utility modelling for multimedia applications for Android Mobile devices. In IEEE international Symposium on Broadband Multimedia Systems and Broadcasting (pp. 1-6). IEEE.

[144] ITU-T Rec. G.1070, (2012). “. Opinion model for video telephony applications,”.

[145] Mwela, J.S., (2010). Impact of Packet Loss on the Quality of Video Stream Transmission (Doctoral dissertation, Blekinge Institute of Technology).

[146] Google, “Recommended encoding setting”, available: https://support.google.com/youtube/answer/1722171?hl=en.

[147] Venkataraman, H., Bi, T., Muntean, G.-M., (2015) “DEAR: An Energy-centric Adaptive Region of Interest Mechanism for Wireless Mobile Devices”, International Symposium on Wireless Personal Multimedia Communications, Hyderabad, India.

[148] Zou, L., Trestian, R. and Muntean, G.-M., (2014), April. eDOAS: Energy-aware device-oriented adaptive multimedia scheme for Wi-Fi offload. In 2014 IEEE Wireless Communications and Networking Conference (WCNC) (pp. 2916-2921). IEEE.

[149] Trestian, R., Ormond, O. and Muntean, G.M., (2010), March. Power-friendly access network selection strategy for heterogeneous wireless multimedia networks. In 2010 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB) (pp. 1-5). IEEE.

[150] Finamore, A., Mellia, M., Munafò, M.M., Torres, R. and Rao, S.G., (2011), November. Youtube everywhere: Impact of device and infrastructure synergies on user experience. In Proceedings of the 2011 ACM SIGCOMM conference on Internet measurement conference (pp. 345-360). ACM.

[151] Zou, L., Trestian, R. and Muntean, G.M., 2013, June. A utility-based priority scheduling scheme for multimedia delivery over LTE networks. In 2013 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB) (pp. 1-7). IEEE.

[152] Bi, T., Trestian, R. and Muntean, G.M., (2013), October. RLoad: Reputation-based load-balancing network selection strategy for heterogeneous wireless environments. In 2013 21st IEEE International Conference on Network Protocols (ICNP) (pp. 1-3). IEEE.

[153] Fehn, C., Kauff, P., De Beeck, M.O., Ernst, F., Ijsselsteijn, W., Pollefeys, M., Van Gool, L., Ofek, E. and Sexton, I., (2002), September. An evolutionary and optimised approach on 3D-TV. In Proc. of IBC (Vol. 2, pp. 357-365). Vancouver

[154] Dufaux, F., Pesquet-Popescu, B. and Cagnazzo, M., (2013). Emerging technologies for 3D video: creation, coding, transmission and rendering. John Wiley & Sons.

[155] Vetro, A., Wiegand, T. and Sullivan, G.J., (2011). Overview of the stereo and multiview video coding extensions of the H. 264/MPEG-4 AVC standard. Proceedings of the IEEE, 99(4), pp.626-642.

[156] Netflix, Internet Connection Speed Recommendations [online]. Available: https://help.netflix.com/en/node/306 access on: 25/04/2014

[157] NS-3 Direct Code Execution [online]. Available: https://www.nsnam.org/overview/projects/direct-code-execution/ access on: 25/04/2016

[158] Han, Y., Yuan, Z. and Muntean, G.M., (2014) "An Objective Quality Metric for Stereosecopic 3D Video," IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB 2014), IEEE.

[159] The Linux kernel MultiPath TCP project [online]. Available: http://multipath-tcp.org/pmwiki.php?n=Main.HomePage, access on: 20/01/2016

[160] Raiciu, C., Paasch, C., Barre, S., Ford, A., Honda, M., Duchene, F., Bonaventure, O. and Handley, M., 2012, April. How hard can it be? designing and implementing a deployable multipath TCP. In Proceedings of the 9th USENIX conference on Networked Systems Design and Implementation (pp. 29-29). USENIX Association.