Enhancing SIP Signaling System over MANET

Mazin Alshamrani

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University of Surrey

Institute of Communication Systems (ICS)
Faculty of Engineering and Physical Sciences
University of Surrey
Guildford, Surrey GU2 7XH, UK

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Abstract

The implementation of the SIP-based Voice over IP (VoIP) and multimedia over MANET is still a challenging issue as many routing factors affect the SIP signaling performance and the voice Quality of Service (QoS). The nodes' mobility in MANET cause dynamic changes on the route calculations, topology, hop numbers, and the connectivity status between the correspondent nodes. The SIP-based VoIP depends on the caller’s registration, call initiation, and call termination processes. Therefore, the SIP signaling performance has an important role for the overall QoS of SIP-based VoIP applications over IPv4 and IPv6 MANET. The SIP end-to-end performance metrics have been defined in RFC 6076 to provide a standardized method of evaluation for the performance of the SIP signaling system over different platforms. However, no benchmarked values for these metrics have been proposed yet.

In this thesis, an evaluation study for the SIP signaling system performance over the AODV-based and OLSR-based MANET is introduced using various mobility models. The related SIP end-to-end performance metrics were employed for the performance investigations and enhancement efforts for the SIP signaling system. The evaluation study was used to benchmark the related performance metrics for the SIP signaling system in general. In addition, the study evaluated the implementations of the ROHC-based system over IPv6 MANET for SIP-based VoIP. Furthermore, novel Cross-Layer performance enhancement approaches are proposed, implemented, and evaluated to improve the performance of the SIP signaling system over MANET, based on simple dynamic modifications for the routing parameters. The SIP performance metrics reflect the SIP signaling state and the required actions for the routing parameters. The implementation of the Cross-Layer approaches succeeded in reducing the total delays in the SIP processes, enhanced the signaling performance, and increased the utilisation level in the system bandwidth and routing processes.

Key words: SIP, Performance Metrics, QoS, VoIP, MANET

Email: m.alshamrani@surrey.ac.uk

WWW: http://www.surrey.ac.uk/ics
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# Contents

Abstract ................................................................................................................................. ii

Acknowledgments ................................................................................................................ iii

Contents ................................................................................................................................. iv

List of Figures ........................................................................................................................ xi

List of Tables .......................................................................................................................... xv

Glossary of Terms ................................................................................................................... xviii

1 Introduction ......................................................................................................................... 1

  1.1 Problem Statement ........................................................................................................ 2

  1.2 Research Objectives ...................................................................................................... 3

  1.3 Scope of the Study ......................................................................................................... 4

  1.4 Research Methodology ................................................................................................. 4

  1.5 Main Contributions ....................................................................................................... 5

  1.6 Thesis Structure ........................................................................................................... 6

  1.7 List of Publications ....................................................................................................... 7

2 Introduction to SIP Signaling System and MANET ......................................................... 8

  2.1 SIP Signaling System ................................................................................................... 9

  2.2 SIP Components .......................................................................................................... 10

  2.3 SIP Messages ................................................................................................................ 10

  2.4 SIP Implementations .................................................................................................... 12

    2.4.1 Peer-to-Peer SIP System ......................................................................................... 13

    2.4.2 Multiple Servers Based SIP System ..................................................................... 15

    2.4.3 Single Server Based SIP System ......................................................................... 16

    2.4.4 IMS-Based SIP System ......................................................................................... 18
2.5 SIP over the Transport Layer Protocols ................................................................. 19
  2.5.1 SIP over TCP ................................................................................................. 20
  2.5.2 SIP over UDP ............................................................................................... 20
2.6 Supportive Protocols for SIP-based VoIP .............................................................. 21
  2.6.1 Session Description Protocol (SDP) ............................................................... 21
  2.6.2 Real-Time Transport Protocol (RTP) ............................................................ 22
2.7 SIP Implementations with Mobility Scenarios ....................................................... 22
2.8 Queuing Theories for SIP Server Systems ............................................................ 24
2.9 Robust Header Compression (ROHC) .................................................................... 25
  2.9.1 Overview of Robust Header Compression ....................................................... 26
    2.9.1.1 ROHC Compressor State ....................................................................... 26
    2.9.1.2 ROHC Decompressor State ................................................................... 27
    2.9.1.3 Mode of Operation ............................................................................... 27
  2.9.2 ROHC Header Compression Performance ...................................................... 28
2.10 Classification of MANET Routing Protocols ......................................................... 28
2.11 Ad hoc On-Demand Distance Vector (AODV) ................................................... 29
  2.11.1 AODV Characteristics ............................................................................... 30
  2.11.2 AODV Route Discovery (RREQ and RREP) ............................................... 30
  2.11.3 Route Maintenance ...................................................................................... 32
  2.11.4 AODV Performance .................................................................................... 33
2.12 Optimized Link State Routing (OLSR) ................................................................. 33
  2.12.1 Multipoint Relays ....................................................................................... 34
  2.12.2 OLSR Control Messages .......................................................................... 36
  2.12.3 Routing Table Calculation ....................................................................... 37
  2.12.4 OLSR Performance ..................................................................................... 37
2.13 IPv4 and IPv6 MANET ......................................................................................... 38
  2.13.1 Challenges and Issues in IPv4 and IPv6 Mobile Ad hoc Networks ............... 38
  2.13.2 QoS in IP Mobile Ad hoc Network .............................................................. 39
2.14 Mobility Models in Mobile Ad hoc Network ................................................................. 39
  2.14.1 Classification of Mobility Models .............................................................................. 40
  2.14.2 Random Waypoint Mobility Model ........................................................................ 42
2.15 Mobility of Nodes In Real-time Application .............................................................. 43
  2.15.1 Impact of Node Mobility on Real-time Applications .................................................. 45
2.16 Summary .......................................................................................................................... 46

3 SIP over MANET – A Literature Review ........................................................................ 48
  3.1 Peer-to-Peer SIP Signaling Implementations over MANET .............................................. 49
  3.2 Implementations of Multiple SIP Servers over MANET .................................................. 52
  3.3 Implementations of Single SIP Server over MANET ....................................................... 53
  3.4 IMS-based SIP Signaling Implementations over MANET ................................................ 55
  3.5 Performance Enhancement Approaches for SIP-based Applications over MANET ...... 56
  3.6 SIP Implementations with Cross-Layer Approaches ...................................................... 57
  3.7 Supportive Simulation Tools for SIP-based Applications over MANET ............... 59
  3.8 SIP-based Applications over MANET Using Asterisk ................................................... 61
    3.8.1 MANET Implementations in Asterisk ........................................................................ 62
  3.9 Summary .......................................................................................................................... 63

4 Performance Metrics for SIP-based VoIP Applications over MANET ......................... 65
  4.1 Voice Codecs .................................................................................................................. 65
  4.2 SIP Signaling Performance Metrics ............................................................................... 67
    4.2.1 SIP Signaling Performance Metrics in the Literature ............................................... 67
    4.2.2 RFC 6076: SIP end-to-end Performance Metrics ...................................................... 68
    4.2.3 RFC 6076 Related Research ..................................................................................... 72
    4.2.4 User Agent Registration ............................................................................................ 73
    4.2.5 Call Setup Delay ........................................................................................................ 74
    4.2.6 Call Termination Delay .............................................................................................. 75
  4.3 Voice Quality .................................................................................................................... 76
4.3.1 Objective Methods

4.3.1.1 Throughput

4.3.1.2 Delays

4.3.1.3 Jitter

4.3.1.4 Packet Loss

4.3.1.5 Other Related Performance Metrics for Objective Methods

4.3.2 Subjective Methods

4.3.2.1 Mean Opinion Score (MOS)

4.4 Routing Protocol Performance Metrics

4.4.1 Performance Metrics for MANET Routing Protocols

4.4.1.1 Qualitative Metrics

4.4.1.2 Quantitative Metrics

4.4.1.3 Internal Efficiency Metrics for the Routing Protocol

4.4.1.4 Network Performance Parameters

4.4.2 Definition of Mobility Terms

4.4.3 Performance of AODV Routing Parameters

4.4.4 Performance of OLSR Routing Parameters

4.5 Summary

5 Research Framework and Methodology

5.1 Research Scenarios and Implementations

5.2 Research Methodology

5.3 Simulation Parameters and Assumptions

5.4 Data Representation Method for the Simulation Results

5.5 The Implementation of SIP end-to-end Performance Metrics

5.6 The Evaluation Methods for the SIP signaling performance and QoS for VoIP Applications over MANET

5.7 The Design and Implementations for the Proposed Cross-Layer Approaches
5.7.1 Cross-Layer Algorithms for AODV-based MANET ........................................ 105
  5.7.1.1 CLAODV for the Call Registration Process ........................................ 106
  5.7.1.2 CLAODV for the Call Setup Process .................................................. 109
  5.7.1.3 CLAODV for the Call Termination Process ......................................... 113
5.7.2 Cross-Layer Algorithms for OLSR-based MANET ................................... 116
  5.7.2.1 CLOLSR for the Call Registration Process ....................................... 116
  5.7.2.2 CLOLSR for the Call Setup Process .................................................. 120
  5.7.2.3 CLOLSR for the Call Termination Process ....................................... 125
5.8 Analysis Methods for the Research Study ................................................. 128
5.9 Summary ......................................................................................................... 128

6 QoS and Performance Evaluation for SIP-based VoIP over MANET ............. 130
  6.1 Relevant Research Efforts ............................................................................. 131
  6.2 VoIP Calls Evaluation .................................................................................. 133
  6.3 SIP Signaling Evaluation for VoIP Applications ......................................... 135
    6.3.1 SIP end-to-end Performance Metrics .................................................... 136
      6.3.1.1 RRD Values ...................................................................................... 136
      6.3.1.2 SRD Values ...................................................................................... 137
      6.3.1.3 SDD Values ...................................................................................... 138
    6.3.2 SIP Registration Intervals ...................................................................... 139
    6.3.3 SIP Call Setup Intervals ......................................................................... 141
    6.3.4 SIP Call Termination Intervals ............................................................... 142
    6.3.5 SIP Server Efficiency ............................................................................. 143
  6.4 Voice QoS ...................................................................................................... 144
    6.4.1 Throughput ............................................................................................. 144
    6.4.2 RTP One-way Delay ............................................................................... 146
    6.4.3 Jitter ......................................................................................................... 147
    6.4.4 Packet Loss ............................................................................................. 148
List of Figures

Figure 1-1: The research scope of the thesis................................................................. 4
Figure 2-1: The signaling flow for a SIP-based VoIP application using
B2BUA-based SIP server.......................................................................................... 12
Figure 2-2: Signaling flow of messages over Peer-to-Peer SIP................................. 13
Figure 2-3: Signaling flow of SIP messages over Multiple SIP Servers.................. 15
Figure 2-4: Signaling flow of SIP messages over Multiple SIP Servers.................. 19
Figure 2-5: SIP over TCP for P2P .............................................................................. 20
Figure 2-6: SIP over UDP for P2P .............................................................................. 20
Figure 2-7: State diagram of the M/M/1/K Model...................................................... 25
Figure 2-8: ROHC compressor states ........................................................................ 26
Figure 2-9: ROHC decompressor states .................................................................... 27
Figure 2-10: ROHC compression mode of operation ............................................... 27
Figure 2-11: AODV Mechanism adopted from [50].................................................. 31
Figure 2-12: An example of network topology for MPR set selection [53]................. 35
Figure 2-13: MPR flooding in the network [57].......................................................... 36
Figure 2-14: Mobility pattern in the Random Waypoint mobility model .................. 43
Figure 3-1: Literature survey of types of SIP signaling system implementations over MANET.. 48
Figure 4-1: IPv4/IPv6 Voice Packet [134]................................................................. 66
Figure 4-2: SIP Performance Metrics over SIP call stages [1] .................................. 71
Figure 4-3: Main Factors Affects the Mouth-to-Ear Voice Quality. Adopted from [181].... 77
Figure 5-1: SIP-based VoIP implementation over closed MANET ............................ 93
Figure 5-2: MANET Model’s implementations in OPNET® Modeler
at simulation time 0 Second..................................................................................... 97
Figure 5-3: MANET RWP Mobility Model at simulation time 20 Seconds .................. 98
Figure 5-4: The identified timers for the registration process of the SIP signaling system .... 100
Figure 5-5: Flowchart representation for Algorithm 5.1........................................... 100
Figure 5-6: The identified timers for the call setup process of the SIP signaling system .... 101
Figure 5-7: Flowchart representation for Algorithm 5.2........................................... 102
Figure 5-8: The identified timers for the call termination process of the SIP signaling system .. 102
Figure 5-9: Flowchart representation for Algorithm 5.3........................................... 103
Figure 5-10: The identified parameters of the proposed Cross-Layer Algorithms
for the SIP signaling implementations over AODV and OLSR based MANET .... 105
Figure 5-11: CLAODV representation over MANET OSI for the proposed performance enhancement approaches of the SIP processes ................................................. 106
List of Figures

Figure 5-12: Flowchart representation for Algorithm 5.4 .................................................. 108
Figure 5-13: Flowchart representation for Algorithm 5.5 .................................................. 111
Figure 5-14: Flowchart representation for Algorithm 5.6 .................................................. 112
Figure 5-15: Flowchart representation for Algorithm 5.7 .................................................. 115
Figure 5-16: CLOLSR representation over MANET OSI for the proposed performance
enhancement approaches of the SIP processes ................................................................. 116
Figure 5-17: Flowchart representation for Algorithm 5.8 .................................................. 119
Figure 5-18: Flowchart representation for Algorithm 5.9 .................................................. 123
Figure 5-19: Flowchart representation for Algorithm 5.10 ............................................... 124
Figure 5-20: Flowchart representation for Algorithm 5.11 ................................................. 127
Figure 6-1: Average call durations in seconds .................................................................... 135
Figure 6-2: Average SIP call setup time in seconds ............................................................ 142
Figure 6-3: Average number of active SIP/TCP connection per 5 seconds in the SIP server .... 144
Figure 6-4: Average voice traffic sent from the caller (Node 1) to the callee (Node 24) ......... 145
Figure 6-5: Average voice traffic received by the callee (Node 24) from the caller (Node 1) ... 146
Figure 6-6: Average voice data end-to-end delay for traffic from (Node 1) to (Node 2) ........ 147
Figure 6-7: Average voice MOS value for VoIP applications ............................................. 150
Figure 6-8: Average consumed bandwidth for routing data in (Bits/Sec) for AODV-based and OLSR-based MANET ................................................................. 151
Figure 6-9: Average routing traffic sent by the caller node (Bits/Sec) ................................. 152
Figure 6-10: Average routing traffic received by the caller node (Bits/Sec) ................. 153
Figure 6-11: Average number of hops between caller (Node 1) and callee (Node 24) ......... 154
Figure 6-12: Average route discovery time for the caller (Node 1) ..................................... 155
Figure 6-13: Average number of route requests sent by the caller (Node 1) ................. 156
Figure 6-14: Average packet queue size on the caller Side (Node 1) ................................. 157
Figure 6-15: Average HELLO traffic sent in MANET ...................................................... 158
Figure 6-16: Average number of MPR in MANET .............................................................. 159
Figure 6-17: Average number of OLSR Topology Control (TC) messages
forwarded in MANET ........................................................................................................ 160
Figure 7-1: Average call durations in seconds for AODV-based MANET ...................... 171
Figure 7-2: Average call durations in seconds for OLSR-based MANET ............................. 172

Figure 7-3: Average SIP call setup time in seconds for AODV-based MANET ............... 180
Figure 7-4: Average SIP call setup time in seconds for OLSR-based MANET .................. 181
Figure 7-5: Average number of active SIP/TCP connections per 5 seconds in the SIP server
for AODV-based MANET .................................................................................................. 182
Figure 7-6: Average number of active SIP/TCP connections per 5 seconds in the SIP server
for OLSR-based MANET .................................................................................................. 183
# List of Figures

- Figure 7-7: Average route discovery time for the caller (Node 1) ......................................... 184
- Figure 7-8: Average number of route requests sent by the caller (Node 1) .......................... 186
- Figure 7-9: Average packet queue size on the caller Side (Node 1) .................................... 187
- Figure 7-10: Average HELLO traffic sent in MANET with ROHC implementations .......... 189
- Figure 7-11: Average number of MPR in MANET with ROHC implementations ........... 190
- Figure 7-12: Average number of OLSR TC messages forwarded in MANET with ROHC implementations ................................................................. 191
- Figure 8-1: Average call durations in seconds for AODV-based MANET using CLAODV approach ................................................................. 200
- Figure 8-2: Average SIP call setup time in seconds for AODV-based MANET using CLAODV approach ............................................................... 203
- Figure 8-3: Average number of active SIP/TCP connections per 5 seconds in the SIP server for AODV-based MANET.................................................. 205
- Figure 8-4: Average route discovery time for the caller (Node 1) for AODV-based MANET using CLAODV approach ........................................... 207
- Figure 8-5: Average route requests sent by the caller (Node 1) for AODV-based MANET using CLAODV approach ................................................. 209
- Figure 8-6: Average routing traffic sent by the caller node in (Bits/Sec) for AODV-based MANET using CLAODV approach ..................................... 210
- Figure 8-7: Average consumed bandwidth for routing data in (Bits/Sec) for AODV-based MANET using CLAODV approach ...................................... 212
- Figure 8-8: Average CPU utilization for the B2BUA SIP server with the CLAODV implementations ................................................................. 213
- Figure 8-9: Average call durations in seconds for OLSR-based MANET using CLOLSR approach ................................................................. 220
- Figure 8-10: Average SIP call setup time in seconds for OLSR-based MANET using CLOLSR approach ................................................................. 222
- Figure 8-11: Average number of active SIP/TCP connections per 5 seconds in the SIP server for OLSR-based MANET .................................................. 225
- Figure 8-12: Average HELLO traffic sent in MANET for OLSR-based MANET using CLOLSR approach ................................................................. 227
- Figure 8-13: Average number of OLSR TC messages forwarded in MANET for OLSR-based MANET using CLOLSR approach ................................ 228
- Figure 8-14: Average routing traffic sent by the caller node in (Bits/Sec) for OLSR-based MANET using CLOLSR approach ........................................ 229
- Figure 8-15: Average consumed bandwidth for routing data in (Bits/Sec) for OLSR-based MANET using CLOLSR approach ...................................... 230
- Figure 8-16: Average CPU utilization for the B2BUA SIP server with the CLOLSR implementations ................................................................. 232
- Figure 9-1: Summary of the investigations performed and contributions in the thesis .......... 241
- Figure B-1: The Node model of the MANET nodes ............................................................. 266
- Figure B-2: SIP functions representation in the header block of the process model in the application layer ................................................................. 267
Figure B-3: The representation of the SIP protocol in the function block.......................... 267
List of Tables

Table 2-1: The request methods of SIP signaling system .......................................................... 11
Table 2-2: The classes used by SIP request methods ................................................................... 11
Table 4-1: Voice codecs and bandwidth consumption; adopted from [182] and [183] .................. 78
Table 4-2: One-way delay constraints for voice data [185] ......................................................... 79
Table 4-3: ITU-T Rec. G.107 E-model, and MOS voice quality scores and descriptions [190] ... 81
Table 5-1: System Configuration in OPNET® Modeler [124] ......................................................... 96
Table 6-1: Number of successful VoIP calls .............................................................................. 134
Table 6-2: Number of rejected VoIP calls ................................................................................... 134
Table 6-3: RRD for SIP signaling over AODV and OLSR MANET in milliseconds (ms) ......... 137
Table 6-4: SRD for SIP signaling over AODV and OLSR MANET in milliseconds (ms) ............ 138
Table 6-5: SDD for SIP signaling over AODV and OLSR MANET in milliseconds (ms) ............ 139
Table 6-6: SIP registration time for SIP clients over AODV and OLSR MANET
in milliseconds (ms) ................................................................................................................ 140
Table 6-7: Number of SIP/TCP retransmission attempts ............................................................ 141
Table 6-8: Maximum jitter variation for voice packets flow in milliseconds .............................. 148
Table 6-9: RTP/UDP packet loss percentage for connected VoIP calls .................................... 149
Table 6-10: Maximum route discovery time for AODV in seconds ........................................... 155
Table 7-1: Number of successful VoIP calls for AODV-based MANET ................................. 169
Table 7-2: Number of successful VoIP calls for OLSR-based MANET .................................... 169
Table 7-3: Number of rejected VoIP calls for AODV-based MANET ....................................... 170
Table 7-4: Number of rejected VoIP calls for OLSR-based MANET ....................................... 170
Table 7-5: Average RRD values for SIP signaling over AODV and OLSR MANET
in milliseconds (ms) with ROHC implementations ................................................................. 174
Table 7-6: Average SRD values for SIP signaling over AODV and OLSR MANET
in milliseconds (ms) with ROHC implementations ................................................................. 175
Table 7-7: Average SDD values for SIP signaling over AODV and OLSR MANET
in milliseconds (ms) with ROHC implementations ................................................................. 176
Table 7-8: Average SIP registration time for SIP clients over MANET
in milliseconds (ms) with ROHC implementations .................................................................. 178
Table 7-9: Number of SIP/TCP retransmission attempts for AODV-based MANET ............... 179
Table 7-10: Number of SIP/TCP retransmission attempts for OLSR-based MANET .............. 179
Table 7-11: Maximum route discovery time for AODV in seconds ........................................ 185
Table 8-1: Set of Benchmarked values for SIP Performance metrics over MANET in milliseconds (ms) ................................................................. 198
Table 8-2: Number of successful VoIP calls for IPv4 AODV-based MANET ........................................ 199
Table 8-3: Number of successful VoIP calls for IPv6 AODV-based MANET ........................................ 199
Table 8-4: Number of rejected VoIP calls for IPv4 AODV-based MANET ........................................ 199
Table 8-5: Number of rejected VoIP calls for IPv6 AODV-based MANET ........................................ 199
Table 8-6: Average SIP registration time for SIP clients over IPv4 AODV-based MANET in milliseconds (ms) for CLAODV implementations ................................................................. 202
Table 8 7: Average SIP registration time for SIP clients over IPv6 AODV-based MANET in milliseconds (ms) for CLAODV implementations ................................................................. 202
Table 8-8: Average SIP termination time for SIP callers over IPv4 AODV-based MANET in milliseconds (ms) for CLAODV implementations ................................................................. 204
Table 8-9: Average SIP termination time for SIP callers over IPv6 AODV-based MANET in milliseconds (ms) for CLAODV implementations ................................................................. 205
Table 8-10: Maximum route discovery time for SIP callers over IPv4 AODV-based MANET in seconds using CLAODV approach ................................................................. 207
Table 8-11: Maximum route discovery time for SIP callers over IPv6 AODV-based MANET in seconds using CLAODV approach ................................................................. 208
Table 8-12: The performance enhancement percentage for the SIP processes with the implementations of the CLAODV approaches with different set values for the SIP Performance metrics of Table 8-1 ................................................................. 215
Table 8-13: Number of successful VoIP calls for IPv4 OLSR-based MANET ........................................ 218
Table 8-14: Number of successful VoIP calls for IPv6 OLSR-based MANET ........................................ 218
Table 8-15: Number of rejected VoIP calls for IPv4 OLSR-based MANET ........................................ 218
Table 8-16: Number of rejected VoIP calls for IPv6 OLSR-based MANET ........................................ 218
Table 8-17: Average SIP registration time for SIP clients over IPv4 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations ................................................................. 221
Table 8-18: Average SIP registration time for SIP clients over IPv6 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations ................................................................. 221
Table 8-19: Average SIP termination time for SIP callers over IPv4 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations ................................................................. 224
Table 8-20: Average SIP termination time for SIP callers over IPv6 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations ................................................................. 224
Table 8-21: The performance enhancement percentage for the SIP processes with the implementations of the CLOLSR approaches with different set values for the SIP Performance metrics of Table 8-1 .............................................................. 234
## Glossary of Terms

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Full Form</th>
</tr>
</thead>
<tbody>
<tr>
<td>AODV</td>
<td>Ad hoc On-Demand Distance Vector</td>
</tr>
<tr>
<td>ARP</td>
<td>Address Resolution Protocol</td>
</tr>
<tr>
<td>B2BUA</td>
<td>Back-to-Back User Agent</td>
</tr>
<tr>
<td>BER</td>
<td>Bit Error Rate</td>
</tr>
<tr>
<td>Bps</td>
<td>Bytes per second</td>
</tr>
<tr>
<td>CBR</td>
<td>Constant Bit Rate</td>
</tr>
<tr>
<td>CI</td>
<td>Connectivity Index</td>
</tr>
<tr>
<td>CRC</td>
<td>Cyclic Redundancy Check</td>
</tr>
<tr>
<td>CSeq</td>
<td>Command Sequence for SIP Messages</td>
</tr>
<tr>
<td>CTS</td>
<td>Clear to Send</td>
</tr>
<tr>
<td>DES</td>
<td>Discrete Event Simulation</td>
</tr>
<tr>
<td>DHCP</td>
<td>Dynamic Host Configuration Protocol</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Server</td>
</tr>
<tr>
<td>dSIP</td>
<td>Distributed SIP</td>
</tr>
<tr>
<td>DSR</td>
<td>Dynamic Source Routing</td>
</tr>
<tr>
<td>ETSI</td>
<td>European Telecommunications Standards Institute</td>
</tr>
<tr>
<td>GPRS</td>
<td>General Packet Radio Service</td>
</tr>
<tr>
<td>GSM</td>
<td>Global System for Mobile Communications</td>
</tr>
<tr>
<td>HA</td>
<td>Home Agent</td>
</tr>
<tr>
<td>HSPA</td>
<td>High Speed Packet Access</td>
</tr>
<tr>
<td>I-CSCF</td>
<td>Interrogating - Call State Control Function</td>
</tr>
<tr>
<td>IETF</td>
<td>Internet Engineering Task Force</td>
</tr>
<tr>
<td>IMS</td>
<td>IP Multimedia System</td>
</tr>
<tr>
<td>IOS</td>
<td>International Organization for Standardization</td>
</tr>
<tr>
<td>IP PBX</td>
<td>IP Private Branch Exchange</td>
</tr>
<tr>
<td>ISAs</td>
<td>Ineffective Session Attempts</td>
</tr>
<tr>
<td>ITU</td>
<td>International Telecommunication Union</td>
</tr>
<tr>
<td>ITU-T</td>
<td>ITU – Telecommunication Standardization Sector</td>
</tr>
<tr>
<td>Kbps</td>
<td>Kilo bit per second</td>
</tr>
<tr>
<td>KPIs</td>
<td>Key Performance Indicators</td>
</tr>
<tr>
<td>LAN</td>
<td>Local Area Network</td>
</tr>
<tr>
<td>MAC</td>
<td>Media Access Control</td>
</tr>
<tr>
<td><strong>Term</strong></td>
<td><strong>Definition</strong></td>
</tr>
<tr>
<td>-------------</td>
<td>---------------------------------------------------</td>
</tr>
<tr>
<td>MANET</td>
<td>Mobile Ad-hoc Network system</td>
</tr>
<tr>
<td>MIP</td>
<td>Mobile IP</td>
</tr>
<tr>
<td>MN</td>
<td>Mobile Nodes</td>
</tr>
<tr>
<td>MOS</td>
<td>Mean Opinion Score</td>
</tr>
<tr>
<td>MPR</td>
<td>Multipoint Relay</td>
</tr>
<tr>
<td>NGN</td>
<td>Next Generation Networks</td>
</tr>
<tr>
<td>NWK</td>
<td>Network Layer (OSI)</td>
</tr>
<tr>
<td>OLSR</td>
<td>Optimized Link State Routing</td>
</tr>
<tr>
<td>OSI</td>
<td>Open Systems Interconnection</td>
</tr>
<tr>
<td>P2P</td>
<td>Peer-to-Peer</td>
</tr>
<tr>
<td>P2PSIP</td>
<td>Peer-to-Peer Session Initiation Protocol</td>
</tr>
<tr>
<td>PAN</td>
<td>Personal Area Network</td>
</tr>
<tr>
<td>P-CSCF</td>
<td>Proxy-Call State Control Function</td>
</tr>
<tr>
<td>PHY</td>
<td>Physical Layer (OSI)</td>
</tr>
<tr>
<td>PRO</td>
<td>Proactive Route Optimization</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>QoE</td>
<td>Quality of Experience</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>RDT</td>
<td>Route Discovery Time</td>
</tr>
<tr>
<td>RERR</td>
<td>Route Error</td>
</tr>
<tr>
<td>RIP</td>
<td>Routing Information Protocol</td>
</tr>
<tr>
<td>RNG</td>
<td>Random Number Generator</td>
</tr>
<tr>
<td>ROHC</td>
<td>Robust Header Compression</td>
</tr>
<tr>
<td>RRD</td>
<td>Registration Request Delay</td>
</tr>
<tr>
<td>RREP</td>
<td>Route Reply</td>
</tr>
<tr>
<td>RREQ</td>
<td>Route Requests</td>
</tr>
<tr>
<td>RSVP</td>
<td>Resource Reservation Protocol</td>
</tr>
<tr>
<td>RTCP</td>
<td>RTP Control Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real-Time Transport Protocol</td>
</tr>
<tr>
<td>RTS</td>
<td>Request to Send</td>
</tr>
<tr>
<td>RTSP</td>
<td>Real-Time Streaming Protocol</td>
</tr>
<tr>
<td>RWP</td>
<td>Random Waypoint</td>
</tr>
<tr>
<td>RWP-All</td>
<td>Random Way Point All</td>
</tr>
<tr>
<td>SCR</td>
<td>Session Completion Ratio</td>
</tr>
<tr>
<td>S-CSCF</td>
<td>Serving-Call State Control Function</td>
</tr>
<tr>
<td>SDD</td>
<td>Session Disconnect Delay</td>
</tr>
<tr>
<td>SDP</td>
<td>Session Description Protocol</td>
</tr>
<tr>
<td>Abbreviation</td>
<td>Full Form</td>
</tr>
<tr>
<td>--------------</td>
<td>-----------</td>
</tr>
<tr>
<td>SDT</td>
<td>Session Duration Time</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SLA</td>
<td>Service Level Agreement</td>
</tr>
<tr>
<td>SLP</td>
<td>Service Location Protocol</td>
</tr>
<tr>
<td>SNR</td>
<td>Signal-to-Noise Ratio</td>
</tr>
<tr>
<td>SRD</td>
<td>Session Request Delay</td>
</tr>
<tr>
<td>SSD</td>
<td>Session Setup Delay</td>
</tr>
<tr>
<td>TC</td>
<td>Topology Control</td>
</tr>
<tr>
<td>TCP</td>
<td>Transport Control Protocol</td>
</tr>
<tr>
<td>TLS</td>
<td>Transport Layer Security</td>
</tr>
<tr>
<td>TORA</td>
<td>Temporally Ordered Routing Algorithm</td>
</tr>
<tr>
<td>TTL</td>
<td>Time-to-Live</td>
</tr>
<tr>
<td>UAC</td>
<td>User Agent Client</td>
</tr>
<tr>
<td>UAS</td>
<td>User Agent Server</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>UE</td>
<td>User Equipment</td>
</tr>
<tr>
<td>UMTS</td>
<td>Universal Mobile Telecommunications System</td>
</tr>
<tr>
<td>URI</td>
<td>User Resource Identifier</td>
</tr>
<tr>
<td>URL</td>
<td>Uniform Resource Locator</td>
</tr>
<tr>
<td>VoIP</td>
<td>Voice over IP</td>
</tr>
<tr>
<td>VoWLAN</td>
<td>Voice over WLAN</td>
</tr>
<tr>
<td>WLAN</td>
<td>Wireless Local Area Network</td>
</tr>
<tr>
<td>WPAN</td>
<td>Wireless Personal Area Network</td>
</tr>
<tr>
<td>WSN</td>
<td>Wireless Sensor Networks</td>
</tr>
<tr>
<td>ZWPAN</td>
<td>ZigBee Wireless Personal Area Network</td>
</tr>
</tbody>
</table>
Chapter 1

1 Introduction

Voice over Internet Protocol (VoIP) and multimedia are nowadays some of the most popular Internet applications because of their wide range of implementations over various platforms. One of the most common wireless network systems is the Mobile Ad-hoc Network system (MANET) that provides simple wireless connectivity for mobile nodes over different mobility systems with flexible, low cost, and providence infrastructure. However, the unstable conditions of the wireless channels, connectivity, nodes’ mobility, complexity of the dynamic routing, and resource limitations represent the main performance challenges that could face the implementations of real-time applications over MANET. MANET has different routing protocols, and each protocol has its own characteristics over different applications and mobility models. The reactive and proactive routing protocols are commonly used with MANET implementations.

The Session Initiation Protocol (SIP) signaling system is used to control and manage the implementations of the VoIP/multimedia applications. Implementing the SIP signaling over MANET has many challenges over all SIP call stages. Different factors affect the Quality of Service (QoS) of SIP-based VoIP over MANET’s routing protocols, such as the mobility model, voice codec, physical distance between calling parties, number of hops, node capacity, Wireless Local Area Network (WLAN) technology, the behaviour of the transportation protocol, and call duration. When route signaling is lost, expired, or delayed, the SIP-based applications will be affected and the VoIP performance will be lowered. The implementation of SIP-based VoIP depends on three main processes: caller’s registration, call initiation, and call termination. These processes depend on the SIP server to relay the connectivity between different callers. The delays in SIP signaling of all processes affect the performance of the VoIP/multimedia calls. In addition, the SIP signaling delays are affected by the connectivity status between the call parties, which are the caller, the callee, and the SIP server. For the implementations of a SIP server with a Back-to-Back User Agent (B2BUA), the main challenging issue is the fact that SIP generally relies on a centralised architecture between the callers and the SIP server. Therefore, the SIP signaling performance has an important role in the overall QoS for the different network systems.

A number of research efforts have been made to enhance the performance of real-time applications over MANET protocols based on the routing parameters and mobility factors. However, few of these
researchers considered the SIP signaling performance over MANET without identifying or using any effective performance metrics for the SIP signaling system. In addition, many standards for evaluating the performance of telephony signaling protocols have been proposed; however, none of those metrics were used to address the SIP signaling performance until the IETF had proposed the RFC 6076, basic telephony SIP end-to-end performance metrics [1]. However, no numerical values or benchmark objectives for the RFC 6076 SIP performance metrics have been proposed yet.

1.1 Problem Statement

To provide the implementations of SIP-based VoIP applications over MANET, the research area needs further investigations and performance enhancement efforts. The importance of these implementations relates to its role as a substitute communication scheme in disasters, emergency recovery system, military operations, and collaborative applications. However, the mobility of MANET nodes and related routing issues increase the required time for the registration, call setup and call termination processes that reflect on the general performance of the SIP-based VoIP/multimedia applications. Moreover, the centralised nature of the SIP signaling system is not suitable for the MANET nature because of the nodes’ mobility, routing mechanism, and signaling issues such as noise, interference, and fading. Therefore, when the route signal is lost or delayed, the SIP signaling system will be affected and the performance of VoIP applications will be lowered. To enhance the SIP signaling performance in MANET, the routing parameters need to be modified dynamically during the SIP processes based on a determined level for the performance enhancement metrics to support the SIP signaling system. This dynamic adoption for the routing parameter during the SIP processes reflects positively on the SIP signaling efficiency, thus on the performance of the VoIP applications. Therefore, the SIP protocol can be used as a key component for MANET, to enhance its usability and capabilities regarding its implementations.

The performance enhancement studies for the SIP signaling system over MANET mostly related to specific network systems or multimedia applications that cannot employed for the SIP signaling implementations over MANET. Furthermore, the SIP implementations over MANET is still a challenging issue, as many routing factors affect the SIP performance such as nodes’ mobility and dynamic hop number between nodes. Few studies proposed performance evaluation methods for the SIP signaling performance over different platforms. These methods considered the adjustments of the signaling timers of SIP, routing enhancement, traffic labelling, or service re-establishment for the SIP based applications. However, there is no efficient performance enhancement methods with a standardized system that proposed and implemented for the SIP signaling system over MANET.
1.2 Research Objectives

The aim of this research study is to investigate the performance issues of SIP-based VoIP/multimedia applications over MANET. The research objectives are to enhance the performance of the SIP signaling performance for VoIP/multimedia applications with an effective, simple and reliable approach without the need to change the basic mechanism for the SIP signaling system. The SIP call processes and QoS parameters of VoIP will be evaluated for IPv4 and IPv6 MANETs. The implementation values for IPv4-based MANET will be used as reference values to validate and verify the analysis for the proposed performance enhancement approaches. The main related performance metrics of the RFC 6076 will be measured for the analysis and to improve the performance. Novel approaches will be suggested to improve the performance of SIP signaling for VoIP calls over MANET based on the evaluation studies in this research. The registration, call setup, and termination processes of the SIP calls over MANET will be optimised to achieve this goal. These proposed approaches will reduce the delays in the SIP processes for VoIP calls in both IPv4 and IPv6 MANETs. The modification level on the functionality of the SIP protocol will be at its minimal level to ensure the actual implementations of the network system. In addition, the proposed approaches aim to be flexible for future implementations with different MANET scenarios or even over different platforms. The proposed approaches considered the MANET routing protocol, bandwidth consumption, SIP signaling performance, and important VoIP metrics. Although the work is focused on VoIP, but the results are equally applicable to multimedia applications (including video and other data types). The objectives of this thesis include:

i. To study SIP-based VoIP and the QoS parameters to support real-time applications over AODV-based and OLSR-based MANET. These parameters will be investigated in IPv4-based and IPv6-based MANETs over different mobility models. A comparison will be conducted to analyse the results and improve the SIP signaling performance.

ii. To benchmark the end-to-end SIP performance metrics in RFC 6076 for the SIP-based VoIP applications. The benchmarked values will support any related assumptions or evaluation studies for SIP-based VoIP applications over any network systems.

iii. To enhance the performance of the SIP-based VoIP applications by employing the related end-to-end SIP performance metrics of the RFC 6076 in the proposed novel approaches. These novel approaches will consider the evaluation results for the current state of the art to enhance the performance level for the SIP processes and reduce the total delays of the SIP signaling system. In addition, the study of the proposed approaches will consider four sets of the benchmarked values of the employed end-to-end SIP performance metrics of the RFC 6076.
1.3 Scope of the Study

This research study considered simple, closed, AODV-based and OLSR-based MANET scenarios with a moderate capacity of mobile nodes and mobility models that use SIP-based VoIP to communicate together. In addition, this study assumed a single B2BUA SIP server in the MANET network system that provides the SIP registration, initiation, and termination mechanism for SIP calls. Furthermore, the network and information security issues are out of the scope of this research study. Figure 1-1 shows the research scope of this study where the titles (in bullets) were considered in the study.

![Figure 1-1: The research scope of the thesis](image)

1.4 Research Methodology

The research study is conducted on OPNET © Modeler for academic and research purposes that is licensed for the Centre for Communication Systems Research (CCSR), University of Surrey. This
Chapter 1. Introduction

simulations tool is a discrete event Modeler that provides a wide range of approved and authentic simulation models including SIP-based VoIP, VoIP QoS, MANET, IPv4, and IPv6.

The research approach that was taken in this thesis was first by studying and reviewing the current state of the art for the SIP-based VoIP over MANET. The literature review had shown that the current research efforts have not covered the performance evaluation for the SIP signaling system with efficient benchmarked performance metrics for SIP-based VoIP applications over MANET. In addition, the performance enhancement methods for the SIP-based VoIP applications were not providing the appropriate level of performance enhancement that meets with the dynamic nature of MANET over both IPv4 and IPv6. Therefore, simulation efforts have been conducted on OPNET© to evaluate the SIP signaling performance and benchmarking the related end-to-end SIP performance metrics of RFC 6076 for SIP-based VoIP applications over MANET. The outcomes from this evaluation study set the key optimisation parameters for the SIP signaling performance over MANET. Then, a novel performance enhancement approach was proposed to optimise the SIP signaling performance over MANET for the registration, initiation and termination processes of the SIP calls based on the conducted evaluation studies. In addition, this proposed approach had been implemented and evaluated with different sets of values over different mobility models for both IPv4 and IPv6 for validation. The performance enhancement level had been identified and compared with the evaluation results for the normal implementations. The analysis results of the proposed approach proved its efficiency optimisation for the SIP signaling performance over other existing approaches. Although the proposed approach shows its scalability through the simulation study, it was not possible for the researcher to confirm this by experimental implementations. This is due to the size limitation of the considered MANET systems and lack of support for SIP applications in mobile nodes. Furthermore, the future work and research directions had been identified by the end of this research study.

1.5 Main Contributions

The main contributions of this thesis are summarised as following:

i. Evaluated the performance of the SIP-based VoIP implementations over AODV-based and OLSR-based MANET. The evaluation considered the SIP signaling performance and VoIP QoS.

ii. Benchmarking the main SIP end-to-end performance metrics of the RFC 6076 that are related to the registration, initiation, and termination processes of the SIP calls.

iii. Evaluated the implementations of the Robust Header Compression (ROHC) system for SIP-based VoIP applications over IPv6 MANET.

iv. Proposed novel approaches to enhance the performance of the SIP processes over AODV-based and OLSR-based MANET. These approaches are designed to reduce the routing overhead of...
MANET during the implementations of the SIP-based VoIP applications. The proposed approaches are applied over the caller’s side or/and the SIP server’s side. In addition, these approaches provide an efficient usage for MANET bandwidth. Furthermore, the research study evaluated the proposed approaches over MANET for both IPv4 and IPv6. The evaluation results illustrated that the proposed approaches performed well and provided an enhanced level of performance for the SIP-based calls.

1.6 Thesis Structure

This thesis has the following structure:

Chapter 2 introduces the SIP signaling system and MANET. It covers the SIP signaling processes, supportive media transport protocols, ROHC, MANET routing protocols, mobility models, and the implementations of IPv4 and IPv6 for MANET.

Chapter 3 covers the SIP over MANET in the literature review. It also reviews the supportive simulation tools for SIP-based applications over MANET.

Chapter 4 discusses the performance metrics for SIP-based VoIP applications over MANET. This chapter considers the related the performance metrics for SIP signaling, voice quality, and MANET routing performance metrics.

Chapter 5 explains the research framework and methodology. It also presents the simulation parameters and assumptions, data representation methods, and the analysis methods. In addition, it describes the implementations of the conducted SIP end-to-end performance metrics. It also represents the design and implementation for the proposed novel algorithms of the performance enhancement approaches.

Chapter 6 evaluates the performance of the SIP-based VoIP implementations over AODV-based and OLSR-based MANET. In addition, it benchmarks the SIP end-to-end performance metrics of the RFC 6076 that are related to the registration, initiation, and termination processes of the SIP calls.

Chapter 7 evaluates the performance of the SIP-based VoIP applications over IPv6 MANET with the implementations of the Robust Header Compression (ROHC) system.

Chapter 8 implements and evaluates the proposed performance enhancement algorithms over AODV-based and OLSR-based MANET. The evaluation considers different sets of values within the proposed algorithms to compare the performance enhancement level over SIP processes. In addition, it compares the new approaches with the basic one for both IPv4 and IPv6.

Finally, Chapter 9 discusses the conclusions and future directions for further research.
In addition, the system configuration and models’ representation of the implemented scenarios in OPNET © Modeler have been reported in the appendices. The code representations of the proposed algorithms and its implementations in MANET, have also been attached.

1.7 List of Publications

The related publications for this research study are as follows:

- M. Alshamrani, “SIP over Next Generation Mobile Ad Hoc Networks (MANETs),” the 5th IEEE International Symposium on a World of Wireless, Mobile and Multimedia Networks (WoWMoM), PhD forum, IEEE Computer Society, Madrid, Spain, 4 -7 Jun 2013.
Chapter 2

2 Introduction to SIP Signaling System and MANET

The SIP signaling is widely used to manage and control voice calls over IP based network systems. The main functions of the SIP signaling are inviting other parties to initiate a call, adding media streams during the calls, changing the encoding system during the call, transferring or holding voice calls. The capabilities of the SIP signaling system depend on the implementation systems of the SIP signaling that is used, and the level of support that the network system provided for the application layer services.

In this chapter, a brief background about the SIP signaling system will be provided to help with studying and understanding the implementations of the SIP signaling system over MANET during the research and the evaluation efforts. The chapter begins with an introduction about SIP protocol then it discusses the SIP system components, messages, and related implementations. In general, there are four types of implementations for the SIP signaling system. These implementations are the Peer-to-Peer SIP signaling system, the single server based SIP signaling system, the multiple servers based SIP signaling system, and the IMS-based SIP signaling system. Each type will be represented and explained in this chapter. In addition, this chapter will represent the implementations of the SIP signaling protocol over the transport layer protocols to help with understanding the nature of the messaging system for SIP-based applications. Furthermore, this chapter will introduce two other supportive protocols for the SIP signaling system: the Session Description Protocol (SDP) and the Real-Time Transport Protocol (RTP). Both SDP and RTP are used to support the SIP signaling system to fulfil the required level of VoIP applications. In addition, this chapter will introduce the Robust Header Compression (ROHC) system as one of the most efficient methods for IPv6-based communication systems to enhance the applications’ performance by using a smaller size of packet headers. In addition, this chapter will end with a description of the SIP signaling support for the mobility-based implementations.

On the Other hand, a Mobile Ad Hoc Network (MANET) is a self-organizing, infrastructure-less, and multi-hop network that consists of unlike groups of nodes with limited capabilities and energy constraints. MANET features, such as nodes mobility, the network dynamic nature and multipath communication scheme tends this network system with variable topology with frequent change. The communicating nodes in a MANET usually seek the help of other intermediate nodes to establish
communication channels, as each node in a MANET works as host as well as router. This adds another challenge: along with the dynamic nature of the network is the unpredictable connectivity change [2]. Therefore, developing efficient dynamic routing protocols is the key challenge in Mobile ad hoc networks to support the proper functioning of MANET. The implementations in MANET are considering both IPv4 and IPv6 systems, as they are part of the existing future network systems [3], [4], and [5]. In this section, the classification of the MANET routing protocol will be discussed in the main with a brief representation of the most common routing protocols. The research study in this thesis had considered AODV and OLSR routing protocols as they are the most widely accepted routing protocols for MANET applications. Thus, this chapter covers the functionality and performance for both AODV and OLSR. In addition, the chapter briefly discusses the mobility models and their related performance issues for the MANET environment.

2.1 SIP Signaling System

The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard for signaling protocol released by RFC 3261 [6]. SIP is commonly used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP). SIP is used for initiation, managing, and termination of multimedia sessions such as voice calls over IP based networks. This session can be either a two-way call, which is either unicast or collective multimedia calls, which is multicast. These features have made SIP a better choice for providing VoIP services in the last few years when compared with other signaling systems for VoIP applications such as H.323 [3, 4, and 5]. SIP is an application layer protocol, which serves five main functions for the multimedia calls [6]. These functions are: User Location, User Availability, User Capability, Session Setup, and Session Management. The User Location is used to determine the location of the end user, while the User Availability examines the end user willingness to participate in the call session. The User Capability supports the applications compatibility between different communication systems and users to determine the required methods and standards for the requested multimedia applications. The Session Setup provides the resources to setup and establish the communication. Finally, the Session Management function supports the call management services in different ways such as adding, transferring and modifying the session parameters. SIP is rather a component which works in a framework with other IETF protocols to build a complete multimedia architecture. The most common protocols which are used in this architecture are: Real-Time Transport Protocol (RTP) for real time data transportation, Real-Time Streaming Protocol (RTSP) for controlled delivery of streaming media, and Session Description Protocol (SDP) for multimedia session description.
2.2 SIP Components

SIP works collectively as a conjunction with different protocols and technologies. SIP consists of two basic components known as User Agents and SIP servers. The User Agents are the end points of the call, while the SIP servers facilitate by sending the responses back to the requested client. User Agents are self-sufficient for initiating a session with other nodes in the network. Each node consists of two fundamental components known as User Agent Server (UAS) and User Agent Client (UAC). UAC is responsible for initiating a new session, whereas UAS handles all the connection requests of the clients [6, and 7]. The SIP server is responsible for handling the user name and the IP addresses of the user agent which connects to the SIP server. There are four different SIP servers that are used to handle the calls’ interconnection processes to different user agents in the network [7]. These SIP servers are: proxy server, location server, registration server, and redirect server. The proxy server is responsible for forwarding the requests on behalf of user agents. The location server is used to find the information about possible locations for the callee. Mostly the location server is incorporated within the proxy server features. The address registered to the register server is stored in the location server. The registration server is used for registering a user agent when it is logged into the network. Hence, the registration servers are responsible for registering the location of the user agents. The registration server is used to find out the IP address of the user agents and then map the IP address to the related user name. Finally, the redirect server is responsible for redirecting the clients to the user agents with whom they want to initiate the call session. The redirect server sends back the IP address of the user agent with whom other clients want to communicate. The main difference between the proxy server and the redirect server is that the proxy server forwards on behalf of the UAC, whereas the redirect server provides the IP address so that the UAC can contact directly to other UAC’s.

2.3 SIP Messages

SIP is a text-based protocol similar to HTTP, which is used for forwarding the information between UAC and UAS, by using several requests and responses [8]. The request methods that used in SIP signaling system represented in Table 2-1 with its related functions. The request methods are replied with one of the response codes used by SIP. The request methods used by SIP consist of six classes as represented in Table 2-2. When a user agent wants to initiate a session with another user agent, the queries of the client are processed by specific servers. Figure 2-1 shows the message flow for a simple scenario of an invitation and termination transactions between two users through the B2BUA SIP server. There is a difference between the B2BUA-based SIP server and the proxy-based SIP server regarding the SIP signaling flow. The B2BUA maintains the whole call state and participates with all call requests. It is involved in the call initiation, management, and termination processes. Therefore, the B2BUA system of the SIP server is providing a secure, reliable communication system for different
User Agents (UAs) where all SIP signaling messages and voice data need to go through the SIP server. The secure connectivity is provided as the IP addresses, port numbers, and locations of the users are hidden from each client where only the B2BUA SIP server knows this information. The B2BUA SIP signaling system is very common for privacy approved VoIP implementations, such as military applications and secured call services. The main disadvantages of a B2BUA-based SIP server are the single point of failure and congestions overhead. On the other hand, the proxy-based SIP server is relaying the SIP signaling system only for the registration stages of the SIP call processes by maintaining the transaction state of the SIP calls. The proxy-based SIP server has a low level of security as the IP addresses and locations of the connected clients could be exposed by the callers.

### Table 2-1: The request methods of SIP signaling system

<table>
<thead>
<tr>
<th>Request</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>REGISTER</td>
<td>register when a user agent initially logged on to the network</td>
</tr>
<tr>
<td>INVITE</td>
<td>Invite other UAC’s for establishing communication and then to start a new SIP session between them</td>
</tr>
<tr>
<td>OPTIONS</td>
<td>query the server for finding the capabilities of other User Agents</td>
</tr>
<tr>
<td>ACK</td>
<td>acknowledge a session before exchanging the related messages</td>
</tr>
<tr>
<td>CANCEL</td>
<td>cancel the pending request</td>
</tr>
<tr>
<td>BYE</td>
<td>terminate the session</td>
</tr>
</tbody>
</table>

### Table 2-2: The classes used by SIP request methods

<table>
<thead>
<tr>
<th>Class</th>
<th>Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td>1xx</td>
<td>to inform that the request is received and processed by having its provisional response, such as 180 ringing</td>
</tr>
<tr>
<td>2xx</td>
<td>For success or (acknowledging), such as 200 OK</td>
</tr>
<tr>
<td>3xx</td>
<td>redirection requests which tells that the request cannot be completed and needs redirection of the user agent, such as 302 moved</td>
</tr>
<tr>
<td>4xx</td>
<td>belongs to the client error to inform that the server cannot process, such as 407, which means that the SIP server authentication is required even for the Back-to-Back User Agent (B2BUA) where the SIP server is acting as a UAS</td>
</tr>
<tr>
<td>5xx</td>
<td>belongs to the server errors to inform that the server cannot process the request, such as 503, that means that the service is unavailable</td>
</tr>
<tr>
<td>6xx</td>
<td>belongs to the server response code which is known as the global error which informs that the server cannot process globally, such as 603, which means decline</td>
</tr>
</tbody>
</table>

The interactions in Figure 2-1 show the employing of the SIP methods INVITE, Ringing, and BYE through the SIP Sever. The SIP server here is recording all the interactions and is used as the coordinator of the Internet working system between the two ends except the media transmissions. The Media Data is mostly depending on the Real-Time Protocol (RTP) and Real-Time Control Protocol (RTCP). The call setup time is mostly consuming more time compared with the termination time. The termination messages could be generated from both ends depending on the type of the application and the connection system. In general, the proxy, redirect, registrar, and location servers are implemented
within the B2BUA SIP Server as represented in Figure 2-1. The interactions between the entities of the SIP server are integrating together to provide the SIP services depending on the connectivity methods.

![Figure 2-1: The signaling flow for a SIP-based VoIP application using B2BUA-based SIP server](image)

### 2.4 SIP Implementations

Many VoIP phone companies allow clients to use their own SIP devices, as SIP-capable telephone sets, or soft phones. The market for consumer SIP devices continues to expand and there are many devices such as SIP Terminal Adapters, SIP Gateways, etc. The free software community started to provide more and more of the SIP technology required to build both end points as well as proxy and registrar servers leading to a commoditisation of the technology, which accelerate global adoption. As an example, the open source community at SIP foundry actively develops a variety of SIP stacks, client applications, in addition to entire IP Private Branch Exchange (IP PBX) solutions that compete in the market against mostly proprietary IP PBX implementations from established vendors [9]. SIP-enabled video surveillance cameras can make calls to alert the owner or operator that an event has occurred, for
example to notify that motion has been detected out-of-hours in a protected area. Other feasible application examples include video conferencing, streaming multimedia distribution, instant messaging, presence information, file transfer and online games. In general, there are four types of implementations for the SIP signaling system. These implementation types are: Peer-to-Peer SIP system, multiple servers based SIP system, single server based SIP system, and IMS based SIP system.

2.4.1 Peer-to-Peer SIP System

Most of the SIP signaling or traditional SIP signaling is based on Client/Server architecture. In Peer-to-Peer (P2P) architecture clients have capabilities of both client and server, and are capable of starting a new session with each other and requesting services [6]. Each node is capable of providing services and resources, and in case any node is unable to provide the services then the next node can be contacted. Nodes in P2P architecture have the features of both UAC and UAS. Therefore, P2P SIP provides instant messaging or VoIP services with the help of P2P architecture, where session initiation and communication between users is facilitated by the SIP protocol. As Client/Server architecture needs a SIP server for handling requests and responses, there is no need of SIP servers in P2P based SIP architecture. To-tag, From-tag and Call-ID are collectively used for handling the dialogue between UAC and UAS in P2P-based SIP [10]. A message exchange between two devices using P2P SIP is shown in Figure 2-2.

![Figure 2-2: Signaling flow of messages over Peer-to-Peer SIP](image)

The SIP protocol stack is handled by the various protocols based on the media protocol stack, such as at the transport layer the TCP/UDP protocol is used. In P2P SIP, two users are initiating the communication process as shown in Figure 2-2 and no SIP server is used. In this case users need not to register at any SIP servers. As TCP protocol is used at the transport layer, hence initially a TCP SYN packet is sent for opening the connection. SYN consist of an initial sequence number to be used in
communication further. The callee responds with the SYN message consisting of the initial sequence number and the ACK message, which confirms that the callee has received the SYN frame from the UAC. Then the UAC sends the TCP ACK message consisting of the UAC acknowledgement number and completes the 3-way handshake [10]. Hence, the connection is becoming open. The UAC caller exchanges the message by sending a SIP INVITE message to the UAS callee. The INVITE message consists of various details, such as session type, which means it can be either a multimedia session or a voice session. There are various other fields in the INVITE message. The first header field in the INVITE message is Via, which is usually a host name and further maps to the IP address using DNS query. In addition, the header field consists of the SIP version, transport layer protocol used, host name and port number. The next header fields are To and From, which dictates the sender and receiver details of the SIP request. Call-ID header field is the next header field which is used for keeping track of a particular SIP session [11]. To-tag, From-tag, and Call-ID are collectively used as identifying parameters and are known as tags. The initial INVITE message consists only of From-tag and the UAC caller generates an INVITE message and it consists of both From-tag and Call-ID. In response to the INVITE message, the user agents who respond to this message will generate the To-tag. The SIP parameters From-tag, To-tag, and Call-ID are used to identify an initiated session. Furthermore, the Content-type and Content-length header fields are used to represent the message body as the SDP protocol. The SDP Content-type describes the media information using various SDP fields, such as media format port number, IP address, media transport protocol, media encoding, and sampling rate [11].

After receiving the INVITE message, the UAS callee responds back by a responded 1xx or 180 ringing. The UAS callee creates a 180 ringing message by copying several header fields from the INVITE message [11], such as From, To, and Call-ID. The 180 ringing message consists of a header field known as the CONTACT header field, which informs an address at which the UAC callee can be contacted. Once the UAC callee is ready to initiate the session, a 200 OK response is sent back to the UAC caller. Finally, acknowledgement ACK is sent by the UAC caller to start the media session. Hence, a media session is established between the UAC and UAS using another protocol for media data transferring.

The major advantage of P2P-based SIP is scalability [10]. As in P2P SIP, a user agent need not register with a central server; instead the user agent needs to register with an overlay network formed by UAC in the system [10]. Client/Server based SIP needs more maintenance and configuration, whereas P2P-based SIP is more scalable and reliable as there cannot be a single point of failure [12]. In addition, P2P SIP does not need maintenance and configuration including NAT and Firewall. However, all these benefits come at the cost of the increased number of security threats and look-up delays [12]. As in client server based SIP, look-up cost is very low, whereas in P2P SIP look-up cost is comparatively very high. Security features such as authentication, and reputation is another major drawback of P2P SIP.
2.4.2 Multiple Servers Based SIP System

The multiple server SIP is also based on client/server architecture in which all the servers, such as proxy server, location server, and registration server, respond to the request sent by the UAC separately. Multiple servers use the Redirect server for initiating a session between a UAC caller and a UAS callee. The Redirect server does not forward the request on behalf of the UAC; it only returns the location address of the UAS callee. Hence, the UAC caller directly contacts the UAS callee after getting the location of the UAS. The message flow between the UAC caller and the UAS callee through the redirect server, registration server and location server is shown in Figure 2-3.

![Figure 2-3: Signaling flow of SIP messages over Multiple SIP Servers](image)

The UAC caller registers itself with the Registration server by sending a REGISTER message. After receiving the REGISTER message by the registration server, it extracts the user name, IP address, and port number then stores them into the location server [11]. A contact header field of the REGISTER message tells the lifespan of the registration. Similarly, the UAS callee also registers itself at the registration server. Now the location details of both the UAC caller and the UAS callee are stored in the location server. A further INVITE message is sent by the UAC callee to the redirect server. The
Chapter 2. Introduction to SIP Signaling System and MANET

INVITE message consists of the header fields, such as INVITE, Via, Max-Forwards, To, From, Call-ID, CSeq, Subject, Contact-type, and Content-length [7]. The Redirect server performs a look-up within the database of the location server for the intended recipient. Then the location information of the user is sent back to the UAC in a redirection class response. The response Moved Temporarily (302) contains the message format having header fields SIP moved temporarily, Via, To, From, Call-ID, CSeq, Contact-type and Contact-length. After getting the response, the UAC callee acknowledges by an ACK response. Hence, the redirection process and the exchange process is completed. Further, a new INVITE message is sent directly to the UAS callee as the location is obtained from the control header field of Moved Temporarily in response to the redirect server. The new INVITE message contains a new Call-ID.

In response to the INVITE message, a direct 200 OK response is sent instead of the 180 ringing response. The UAC caller responds to the UAS callee by acknowledging it by an ACK response. Thus, a session is initiated between the UAC caller and the UAS callee using a redirect server. After initiating the session, the media session is started between the UAC caller and the UAS callee using the RTP protocol. Once the media session is completed the session is terminated by sending a BYE request. Once it is acknowledged by the UAC caller, the complete session is terminated. In multiple server based SIP, the redirect server does not forward session initiation requests for the UAC caller as is done by the proxy server. As the redirect server does not initiate the request, hence a lower state overhead is needed while comparing with a proxy server. Multiple server based SIP uses the redirect server and it has very few messages to process, therefore it has high processing capacity [11].

2.4.3 Single Server Based SIP System

The Single SIP server is based on Client/Server architecture in which the client send requests to the server, and the server replies to the corresponding request of the client for establishing communication. A UAC requests the services and SIP servers, such as the SIP server, redirect server, or register server, respond to those requests. The single server based SIP signaling system is Back-to-Back User Agent (B2BUA) implementations, as shown in Figure 2-1. Initially, the caller sends a REGISTER request to the SIP server. After receiving the REGISTER message, the information in the request message of the caller is updated in the database used by proxies. The REGISTER message sent by a caller consists of the address of the SIP server [11]. The REGISTER request contains To and From header fields. The To header field consists of the User Resource Identifier (URI), to be registered on the server. The next Contact header field containing the SIP URI is stored by the registrar [8]. Then the SIP server acknowledges the caller by sending a 200 OK response message. Similarly, the callee also registers himself on the SIP server. In this case the SIP server is playing the role of both registration and location service [11]. After completing the registration process, the caller is not aware of the callee’s current location. The caller also needs to check whether the callee is available for the session initiation process.
or not. Hence, the SIP server is used for inviting the callee, as the SIP server forwards the request on behalf of the user agent. Initially the DNS look-up is performed by the caller SIP URI. It returns the IP address of the SIP server to handle the callee domain. Then the INVITE message is sent to that mapped IP address of the SIP server. Furthermore, the SIP server looks up in its own database to locate the callee’s current location. The process consists of two major steps: the DNS look-up step which is performed by the user agent to find the IP address of the SIP server, then the database look-up which is performed to locate the IP address of the SIP server. Then an INVITE message is forwarded by the SIP server to the callee’s IP address using a Via header field, having the address of the SIP server [8].

As the INVITE message consists of two Via header fields, therefore the callee comes to know that an INVITE message has been routed through the SIP server. After receiving the INVITE message, the callee sends back a 180 ringing response code to the caller. The 180 response code is created by copying the header fields, such as To, From, Call-ID, and Cseq from the INVITE request. A response code is sent to the callee through the SIP server. The first Via header field contains the received parameters whereas the second Via header field contains the IP address in the URI. After receiving the 180 ringing response by the SIP server, the SIP server checks the contents of the first Via header field. Furthermore, when the SIP server finds the first Via header field consists of its own address, it removes the first Via header field and forwards the response to the address within the second Via header field.

As the callee is ready to start the session with the caller, it sends back a 200 OK message through the same set of proxies. The SIP server follows a similar process by removing the first Via header field and forwards a 200 OK message back to the caller. The contact header field of the callee in the 200 OK message allows the caller to send an ACK message directly to the callee by bypassing the SIP server. However, it needs to be noticed that the request is sent to the callee’s contact URI not in the address of the contact header field. After getting the ACK message from the callee, the session is started between the caller and the callee. Hence, the transmission session is established between the caller and callee using the RTP protocol. In this scenario, the SIP server is used for contacting and locating both end points. The SIP server can drop the path if there is no exchange of media. In the SIP protocol, the path of the signaling message is different from the path of media packets. Then, after the successful transfer of voice data, the connection is terminated using a BYE message. Once the BYE message is received by the callee, it responds by sending back a 200 OK message. After receiving the 200 OK message, the media session terminates the transmission process.

In this case, the SIP signaling is performed using a single SIP server, which forwards the request on behalf of the user agent. A SIP SIP only forwards the message at the application layer level and it is allowed to modify both request and response, as defined in RFC 3261 [11]. Hence, the SIP server establishes end-to-end communication and preserves end-to-end transparency. As the SIP server can be either a stateful or stateless proxy, all the requests and responses that have been received in the past are
tracked by a stateful proxy and can be beneficial for future processing of requests; one such example is
the transactional stateful proxy [11]. Reliability is ensured while the TCP protocol is used in a stateful
proxy. However, a stateless proxy does not keep track of the request and response messages. A stateless
SIP server has higher processing capacity. Major benefits of the SIP server include reliability using
replication, flexibility and the use of stateful or stateless proxies. If the number of proxies handling the
message are exceeded in limits calculated by the Max-forwards header field, then the SIP server
discards the messages. Hence, if the SIP server is not scaled properly then it can have a potential
overload.

2.4.4 IMS-based SIP System

The IP Multimedia System (IMS) is a concept for providing multimedia services regardless to the media
type using a common architectural framework for most of the media applications. The signalling within
the IMS bases on the SIP signaling system. For supporting multimedia services functionalities, the IMS
consists of multiple SIP proxies known as Call Session Control of Function (CSCF) which are used for
SIP signaling with other variants, which are P-CSCF (Proxy-CSCF) which is the first contact point for
an IMS terminal and Internet with Gateway GPRS Support Node (GGSN) for resource allocation. The
P-CSCF is assigned to an IMS terminal before registration, I-CSCF (Interrogating CSCF) which works
similarly to the work performed by the registration server and is responsible for routing to the S-CSCF
(Serving S-CSCF), and the S-CSCF which facilitates control and service triggers [9]. The IMS provides
more efficient services and provisioning of capabilities than circuit and packet switched networks [9].
Initially, when any user registers to the IMS, a Subscriber Service Profile (SSP) is downloaded by S-
CSCF from a Home Subscriber Server (HSS) [9]. The IMS-based SIP system is shown in Figure 2-4.
Initially both of the User Equipment (UE’s) devices register themselves in the network. Session
establishment between both UE1 and UE2 can have some scenario, such as both originating and
terminating; UE has ready resources before sending INVITE and response messages [14]. The SIP-IMS
message flow for the initiating session between both UE’s begins from the caller UE-1 to the callee UE-
2. At the start of the communication processes, UE-1 sends an INVITE message to the P-CSCF. The
INVITE message contains various header fields, such as From, To, Call-ID, Cseq, Via, Max-forwards,
Route, P-preferred identity, Privacy, Proxy-require, Security-verify, Contact, Allow, Content-type, and
Content-length. After adding itself to record the route header, it forwards an INVITE message to S-
CSCF then I-CSCF. The I-CSCF requests the DNS look-up of the location of user UE-2 and sends a
Location Information Request (LIR) to the HSS. The HSS replies with a Location Information Answer
(LIA) by providing the address of the S-CSCF of the terminating subscriber. Then an INVITE message
is forwarded to the S-CSCF of the terminating visited network. The S-CSCF forwards the INVITE
message to UE-2 via the P-CSCF. Then a message of 183 is sent back to UE-1 which says the session
is in progress. After getting the 183 response code, UE-1 sends a Provisional Acknowledgement
(PRACK) to UE-2. In corresponding to the PRACK, a 200 OK message is sent back to UE-1 for Policy Decision Point (PDP) activation, and resource reservation [15].

![Figure 2-4: Signaling flow of SIP messages over IMS-based SIP System](image)

The IMS system implementations are using the UPDATE messages from UE-1 to UE-2 and a response code 200 OK is sent back to UE-1 for enabling QoS utilization. Since the resources are ready with UE-2, hence it sends a 180 ringing response to UE-1 via the S-CSCF, I-CSCF and the originating I-CSCF, S-CSCF and P-CSCF. It consists of the header fields, such as From, To, Call-ID, Cseq, Via, Record route, Contact, Privacy, P-Asserted identity, Privacy, and Content-type [9, and 15]. UE-1 acknowledges the UE-2 180 ringing message with a PRACK response. The PRACK consists of header fields, such as From, To, Call-ID, P-Access Network, Cseq, Via, Max-forward, Route, Ack, and Content-length [15]. A 200 OK response is generated and sent back to the UE-1 acknowledging the PRACK request. After acknowledging the PRACK request by ACK, a session is initiated between UE-1 and UE-2 using the RTP protocol. The IMS SIP has enabled the provision for service such as multimedia services over IP, VoIP, and IMS. It has a very modular design with open interfaces. Hence, it provides flexibility for providing multimedia services over IP networks.

### 2.5 SIP over the Transport Layer Protocols

SIP depends on the supported Internet protocols, and it includes many elements of the text-based protocols. Moreover, the SIP protocol is an Application Layer protocol designed to be independent of the existing transport layer. It can run on top of the Transmission Control Protocol (TCP) Figure 2-5, or the User Datagram Protocol (UDP) Figure 2-6 [16, 17]. This important characteristic allows the SIP protocol to provide various features and services for the network systems, such as call control services, mobility, interoperability with existing telephony systems, and more.
2.5.1 SIP over TCP

TCP was not designed for signaling, so it cannot fully apply to act with SIP. It was developed to transfer non real-time data in bulk. It requires a connection setup using a three-way handshake before data transfer. SIP exchanges small messages, approximately 512 bytes in size, in a client/server model. These small messages are interdependent. TCP’s flow control and congestion control is the best for bulk data, but not for such small messages. Thus, TCP does not perform up to the mark for SIP signaling. Above all, connection set-up adds significant delays. SIP messages cannot be exchanged before the three-way handshake. In a relatively prolonged connection, this time can be ignored. However, a SIP session is of short duration, hence connection setup time is not negligible. The three-way handshake can be adversely affected by packet loss and it further delays SIP transactions, which is very unacceptable. TCP maintains the sequence of packets, as it a connection-oriented protocol. A whole SIP message, due to its small size, can be encapsulated in a TCP packet. The keep in order delivery of packets is useless for signaling. The check and balance on the order of packets is a shortcoming in the case of packet loss. All the packets delivered after the lost packet will be queued at the destination, until the successful arrival of that packet. Therefore, the packets carrying SIP messages, required to complete a SIP transaction, are pointlessly delayed to maintain the sequence of packets. It results in Head of the Line (HOL) blocking, a phenomenon, also encountered by packets waiting in router queues.

2.5.2 SIP over UDP

SIP signaling is not affected by the connection establishment time as the protocol is connectionless. In applications that use UDP at the transport layer, the application layer is responsible for detecting and recovering from packet loss. The SIP specification defines a retransmission policy to guarantee the delivery of SIP messages [6]. SIP transactions involve several two-way handshakes: INVITE-100
Trying, 200 OK-ACK and BYE-ACK. There are two types of retransmission – one is for INVITE transactions and the other is for non-INVITE transactions.

In the INVITE transaction, the client sends the request INVITE, and on receiving it, the server returns the response “100 Trying”. The client manages the retransmission of the INVITE with a timer. This timer starts at $T_1$ seconds, doubling after every retransmission. The client transaction stops retransmission when it receives the provisional response “100 Trying” or when $64 \times T_1$ seconds have passed since the initial INVITE was sent. The default value for $T_1$ is 500 ms, thus, the INVITE is retransmitted at intervals of 0.5, 1, 2, 4, 8 and 16 seconds. If there is no response after 32 seconds, the client ceases retransmission. The retransmission for the non-INVITE transaction is basically the same. A new timer $T_2$ is introduced. For UDP, the “200 OK” or BYE is retransmitted at an interval starting at $T_1$. The interval is doubled after every retransmission, capping off at $T_2$. The default value of $T_2$ is 4 seconds. Therefore, the “200 OK” and BYE are retransmitted at intervals of 0.5, 1, 2, 4, 4, 4, … seconds. After 32 ($64 \times T_1$) seconds in total, the retransmission is ceased.

### 2.6 Supportive Protocols for SIP-based VoIP

For SIP-based VoIP applications, the SIP signaling system is managing and controlling the calling sessions between different parties of the VoIP applications. Several supporting protocols had proposed to enhance the VoIP implementations. This section presents an overview about the related protocols for the SIP-based VoIP applications.

#### 2.6.1 Session Description Protocol (SDP)

The Session Description Protocol (SDP) is a format for describing streaming media initialization parameters [13]. SDP was first published by the IETF in 1998 as RFC 2327, then the revised specification Proposed Standard as RFC 4566 [13, 18]. SDP is entirely textual while the grammar for SDP is very structured and strict when it says what a session description should look like. The SDP is proposed to describe the multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation. SDP is not delivering the media itself, but it is designed to establish negotiations between the end points about the media types, formats, and all associated properties. The set of properties and parameters are often called a session profile [19]. SDP is designed to be extensible to support new media types and formats. SDP started as a component of the Session Announcement Protocol (SAP), but found other uses in conjunction with the Real-time Transport Protocol (RTP), Real-time Streaming Protocol (RTSP), Session Initiation Protocol (SIP) and even as a standalone format for describing multicast sessions.
2.6.2 Real-Time Transport Protocol (RTP)

The Real-time Transport Protocol (RTP) defines a standardized packet format for delivering audio and video over IP networks [20]. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push-to-talk features. RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP is used to monitor transmission statistics and Quality of Service (QoS) and aids synchronization of multiple streams. When both protocols are used in conjunction, RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number. It is one of the technical foundations of Voice over IP and in this context is often used in conjunction with a signaling protocol which assists in setting up connections across the network. RTP was developed by the Audio-Video Transport Working Group of the Internet Engineering Task Force (IETF) and was first published in 1996 as RFC 1889, superseded by RFC 3550 in 2003.

2.7 SIP Implementations with Mobility Scenarios

Mobility support for IP-based communication is one of the important features, as wireless and mobile communication is increasing the use of portable devices. Therefore, it is important to realise the support capabilities for mobility in the SIP system and mobility impacts on the performance of real-time services. As SIP is an application layer protocol for establishing a multimedia session between users, the SIP protocol uses centralized servers, such as proxy server, registration server, redirect server, and location server, for providing IP-based multimedia services. In a SIP signaling system, while using the proxy servers for initiating the session between the caller and the callee, the SIP server itself forwards the request on behalf of the user agent or the callee. However, while using the redirect server in SIP, the redirect server does not forward the request; it only returns the address of the intended destination to the caller. After getting the address of the callee, the caller itself initiates a session with the callee. Therefore, the redirect server’s properties have similar resemblance to the Home Agent (HA) in Mobile IP [21]. SIP provides support for the personal mobility in which a user can be independent of its location. This means that the node can have access to its communication services on any terminal at any location [21, 22]. Similarly SIP also has support for service mobility which provides the features of keeping similar services, terminal mobility which allows devices to move between subnets and is still reachable from incoming requests and session mobility.

Support for personal mobility in SIP is provided by the REGISTER method. It allows the mobile device to change its IP address as well as the point of connection to the Internet, but is still capable of receiving the calls. Initially, when a node is registered with a registration server, then SIP temporarily binds the contact URI of the device with the User URI. As the IP address of a node changes then this information
is automatically updated within the network [11]. As if the caller is itself a mobile host, then the caller’s regular address is contained by the From header field used for identification of the caller. As the caller moves to a new location its new address is contained by the Contact header field, which is part of the SDP message. Finally, the caller updates its registration to the home server. Hence, the call can be redirected without any error. In this way, SIP provides support for personal mobility. The major benefit of personal mobility in SIP is that there will be no distinction between personal and device mobility [21]. Hence, in IP-based mobility applications, the user can either move to a new location or the user can change their device, such as from a phone to a computer [22]. SIP also supports terminal mobility. The terminal mobility impacts SIP at three different stages that is the pre-call, mid-call, and recover from partition. In pre-call mobility, the caller which is a mobile host gets a new address prior to receiving a call. However, in mid-call mobility, the mobile host resends another INVITE message to the callee without going to proxies, which is an updating message about its new location [22]. The SIP-based mobility is known as terminal mobility, which is less favourable for TCP-based applications because the nodes have to maintain TCP connections across the subnet [21].

Session mobility facilitates the user to maintain the media session even after having the terminal mobility and personal mobility. For example, the caller presently carries a session using a mobile device. After a certain time interval, the user wants to carry the session using the desktop in place of a mobile device. Both IPv4 and IPv6 do not provide any support for session mobility while SIP does provide session mobility [22]. The session mobility makes the SIP-based application more efficient to use. The service mobility facilitates the user having access to service even while the caller, which is a mobile host, is moving, changing the device, or changing the network system [20]. SIP provides a mechanism for synchronizing the services across the servers. The SIP application registers with a registration server and updates whenever the network address is changed. The registration server broadcasts this message, consisting of the user’s current network address, device properties, and configuration elements [21].

SIP collectively, with the support of SDP, has the ability to change the transport addresses. Thus, SIP can be used to support the mobility without changing the IP stack of the caller or the mobile host. The SIP-based mobility supports all forms of mobility such as terminal mobility, session mobility, service mobility, and personal mobility. These SIP mobility capabilities are highly suitable for wireless networks such as 802.11 in home, office, outdoors, or in any public area. However, some other commercial applications could be built, like wireless telephony systems, using a SIP signaling system. Wireless SIP clients can also take the advantage of voice codecs to make the network highly tolerant against packet loss. In this way, SIP mobility can be highly beneficial for wireless-based applications.
2.8 Related Queuing Theory for SIP Server Systems

In the communication systems, a queue is a mathematical model in which a number of jobs are waiting to be serviced by one or more servers. The Markovian distributions mean that inter-arrival and service times are exponential. The M/M/n model is one of the models used for SIP server implementations, where n presents the number of servers that are serving a specified queue [23]. In the queuing theories, the distribution of service time for definite period, are constrained to being Markovian [24] distributions (M for short). The Markovian distributions mean that inter-arrival and service times are exponential. The M/M/n model is one of the models used for SIP server implementations, where n presents the number of servers that are serving a specified queue. A system of a single SIP server with one queue can be written as M/M/1, where the first M denotes the arrival process, the second M presents the departure process and 1 is for the number of servers. “M” stands for ‘Markovian’, and that means exponential inter-arrival and exponential service times.

The system with the M/M/1 queue [25] has times between arrivals (or inter-arrival times) as well as service times which are distributed exponentially with defined parameters. This system is with a single server and usually utilizes FIFO service scheduling. The waiting line has infinite size and it is simple to find the suitable Markov chain. A Markov chain [25] is a system that goes through transitions from one state to another state on a state space, and it is a random process typically described as memoryless. Memoryless means that the next state is depending only on the present state and not on the series of events that occurred before it. The number of the clients in this kind of SIP server system is used as the system state. For this kind of system, at any point of time, at least one event can happen. The event can be the arrival of a novel client or the completion of the service of the client. It can be concluded that for this system the service and arrival rates are not state-dependent. On the other hands, M/M/m stands for a multiple servers with multiple queues SIP system. In fact, it represents the generality of the queue M/M/1 that includes only one single server. The M/M/m queue compared with M/M/1 has the same service time and inter-arrival time distributions, but there are M servers in the system and the waiting time in this case is shorter. M/M/m is also a pure birth-death system like M/M/1. A birth-death system or process means a skip-free Markov chain, and it denotes that the process could move with a single step to the neighbouring state [26].

Systems with one server, exponential inter-arrival and service times and queue size only for K clients (including the one in service) can be written as M/M/1/K [27]. This system has exponential service time and inter-arrival time distributions, each one with the particular parameters $\mu$ and $\lambda$. The clients in this case are served with FIFO distribution. There is a single server and the system can hold up to K clients. If a novel client arrives and the system is already involved with K clients, the novel client is regarded as lost, and it drops from the system, which known as ‘blocking’. This is essential for this system,
because otherwise the arrival process will not be Markovian (if the client waits outside pending for a free place). M/M/1/K is also a pure birth-death system like M/M/1 and it is a better approximation of the ‘real systems’ (as routers) because the buffer space is always finite. Figure 2-7 presents the functioning of state diagram of the M/M/1/K system. M/M/1/K is modelled in Figure 2-7 as a birth and death process with states from 0 to K.

![State diagram of the M/M/1/K Model](image)

For the M/M/1/K system [28], the utilization factor (traffic intensity) $\rho$ can be expressed as $\rho=\lambda/\mu$, where $\mu$ is the SIP system’s average service rate, $\lambda$ is arrival rate of the requested messages. The average delay that messages have is equal to the waiting time that messages have in the queue plus the average service time. One of the advantages of the M/M/1/K queue with limited size $K$ is that it is much more realistic comparing it with the infinite queue. Because this model has a finite number of states, it can be concluded that it is a stable process not depending on the values of $\mu$ and $\lambda$. On the other hand, the main disadvantage of the M/M/1/K queue is that it is computationally, to some extent, harder to deal with.

### 2.9 Robust Header Compression (ROHC)

The header size of the IP packet should be smaller in size to reduce the end-to-end delay and jitter. Therefore, in the case of low bandwidth and a mobility environment, IPv6 can create challenges for better performance. Hence, there is a need for an IP header compression technique to decrease packet overhead. IP header compression can be defined as ‘to compress the header of the protocol before sending it on to the link and decompress it at the receiving end’. IETF has developed two header compression algorithms as Internet Protocol Header Compression (IPHC) and Robust Header Compression (ROHC).

The IPHC header compression technique is defined in RFC 2507 and designed for low bit error links. It facilitates compression of dynamic header fields, such as TCP, and sequence numbers. Incremental header fields use Delta-based differential encoding schemes for encoding. In case of any sequential packet loss, the compression and decompression context will be desynchronized. The major drawback of the IPHC compression scheme is it has very poor performance over high bit error links [29]. To provide better performance for high bit error links, the ROHC header compression technique was
developed. ROHC is a standardized compression scheme defined in RFC 3095 and 3096, to support both wired and wireless links with high error rates and long RTT [30]. It is used to provide an adopted suitable form of compressed data for IP-based communication systems, such as 3GPP, which act as the core technology for 3G cellular networks [31]. ROHC is used to compress the IP, UDP, RTP, and TCP packet headers to provide the best case with lower size of packet headers for IP-based communication. ROHC was first devised to improve voice and data transmission quality in one hop cellular networks; it could have different characteristics when it is adopted into multi-hop wireless mesh backhaul networks. ROHC was designed to overcome the overhead for streaming applications such as VoIP, where IPv4 normally requires 40 Bytes compared with 60 Bytes for IPv6. The mechanism of ROHC depends on placing the compression system before the link port of the source side with limited capacity and the decompression system after the link port of the destination side. The compression and the decompression systems of ROHC are used to convert large amounts of overhead traffic headers to fewer amounts of traffic headers’ overheads for IPv6 network systems. ROHC performs better than other types of compression schemes for IP-based links with high rates of packet loss, such as wireless network systems.

2.9.1 Overview of Robust Header Compression

Header compression of protocols is possible with the availability of redundancy in header fields of the same packet, the same with consecutive packets [32]. Each header compressor technique has two operations: compressor state, which is before sending the packet into the links, and decompressor state, which is before receiving the packet at the destination.

2.9.1.1 ROHC Compressor State

The ROHC Compressor state consists of three stages, which are Initialization and Refresh (IR), First Order (FO), and Second Order (SO) [33]. The states represent the correctness of the level at the decompression side, so that if there occurs an error indicated by the decompressor side, the compressor moves to the lower state to send packets, as shown in Figure 2-8.

![Figure 2-8: ROHC compressor states](image)
2.9.1.2 ROHC Decompressor State

The ROHC Decompressor state also consists of three stages: No Context (NC), State Context (SC), and Full Context (FC), as represented in Figure 2-9 [32]. The starting state of decompressor is with NC, and as with successful decompression of an IR packet, it moves to the Full Context state after getting a new packet; if any error occurs it will again move to the lower state.

![ROHC decompressor states diagram](image)

Figure 2-9: ROHC decompressor states

2.9.1.3 Mode of Operation

The ROHC algorithm consists of three different modes of operation: Unidirectional mode (U-mode), Bidirectional Optimistic mode (O-mode), and Bidirectional Reliable mode (R-mode) [32]. Selection of any mode depends on variation in the header size, error percentage, and feedback from the decompressor, as represented in Figure 2-10. In U-mode, data packets are sent in unidirectional mode from compressor to decompressor, whereas in O-mode the decompressor sends feedback like error message, successful context, and update, and in R-mode the feedback channel is used for packet loss avoidance. ROHC header compression is used in very few headers like IPv6 (UDP/RTP), IP, etc.

![ROHC compression mode of operation diagram](image)

Figure 2-10: ROHC compression mode of operation
2.9.2 ROHC Header Compression Performance

ROHC header compression consists of Initialize and Refresh (IR), which starts the compression by sending a start packet to the decompressor side. The packet size of IR is larger than the original packet which puts the compressor and decompressor in a high state for a longer period of time. Since IPv6 ROHC varies with the specific application, prior studies suggest that operation modes also have different efficiencies for IPv6 [33, 34]. The R-mode uses the confidence system (L) which is not used by the U-mode, and with small headers frequently rather than larger headers, hence efficiency is better than O-mode and U-mode. However, in the case of an error propagation from the decompressor, the performance of O-mode is better than R-mode as O-mode uses the downlink rarely in comparison to R-mode. However, in the case of a noisy link, the downlink is used more than the uplink therefore R-mode gives better performance rather than O-mode. In the case of a wireless link, the ROHC header compression has different errors. First, either there is an error in the header compression and this error is detected by the decompressor CRC and it is dropped, or the error is with the payload; in this case, the decompressor CRC is unable to detect the error in payload and forward it to the network layer [34]. The second kind of loss is unavoidable for real-time applications. With the help of prior studies, it can be concluded that if the bit error rate is increased then it affects the application rather than the ROHC loss. Hence, ROHC compression with increasing bit error rate affects payload rather than the header. The loss of payload and application with increasing bit rate leads to degradation in the performance. The performance of efficient ROHC compression cannot be explained with ROHC specifications because it depends on the scenario, the application, and its implementation [35].

2.10 Classification of MANET Routing Protocols

Routing in MANET is a challenging task as it has a dearth of research efforts. This has led to the development of various routing protocol strategies for MANET, and each new proposed routing algorithm is supposed to be an improved version over some of the previous algorithms, taking consideration of the previous literature reviews by the authors. Since each protocol has its pros and cons when comparing them to other protocols, on the basis of certain attributes and different network scenarios, therefore to analyse and compare Mobile ad hoc network protocols, an appropriate categorization method is important, so that it will be helpful to understand the nature and distinct properties of available routing protocols. There are various ways to classify the routing protocols in Mobile ad hoc networks, but most of these classifications are done on the basis of certain attributes such as routing strategy and network structure [36, 37]. Routing strategy is either table driven or source-initiated, so protocols can be categorized as either table-driven protocols or source-initiated protocols, while on the network structure protocols are classified as flat routing, hierarchical routing and
geographical position as proposed by the authors in [38]. In general, there are three types of routing protocols in MANET [39, 40]:

A. Reactive Routing Protocols

The Reactive routing protocols are on-demand protocols that discover the routes between the source and the destination when needed using the route discovery process. These routes are considered as source-initiated route discovery protocols. The most widely accepted and used reactive routing protocols are the Dynamic Source Routing (DSR) [41], Ad hoc On-Demand Distance Vector (AODV) [42], Temporally Ordered Routing Algorithm (TORA) [43], and Associativity Based Routing (ABR) [40].

B. Proactive Routing Protocols

The proactive routing protocols are traditional distributed protocols that use the shortest paths based on periodic updates. The proactive routing protocols are table driven where all possible routes to all destinations are determined at the start and uses periodic route updates, however they have a high routing overhead. The most widely accepted and used proactive routing protocols are the Optimized Link State Routing (OLSR) [44], Destination Sequenced Distance Vector (DSDV) [38], Fisheye State Routing (FSR) [45], and Topology Broadcast Reverse Path Forwarding Protocol Fisheye State (TBRPF) [46].

C. Hybrid Routing Protocols

The Hybrid routing protocols have combined functionality from both reactive and proactive routing protocols, with hybrid routing capabilities. The most widely accepted and used proactive routing protocol is Zone Routing Protocol (ZRP) [47].

2.11 Ad hoc On-Demand Distance Vector (AODV)

AODV was determined by the Internet Engineering Task Force (IETF) MANET working group as the RFC 3561 [48]. AODV is an efficient reactive routing protocol that computes routes in an on-demand base when a node wants to send data to another node using a flat routing strategy. In this scheme, the AODV packet only consists of the destination address and maintains the routing table at each node. In route discovery, the Route Request (RREQ) rebroadcasted from the next hop then it maintains a bidirectional link between nodes. When the Route Reply (RREP) comes from the destination it consists of only the destination IP and a sequence number. Thus, the route discovery is initiated in AODV from source to destination. The major advantages of this algorithm are the route discovery is on demand; the destination sequence number can be used for finding the latest routes, and it is beneficial for highly dynamic networks. Having an old sequence number with the source and a higher sequence number than
the intermediate node can lead to a false route reply to the source by an intermediate node and, in case of link failure, extra bandwidth utilization; these are some disadvantages of this routing algorithm.

2.11.1 AODV Characteristics

AODV maintains only one active route per destination inside the routing table between nodes that need to communicate. AODV provides a fast adaptation for dynamic links, low network utilization, and low processing and memory overhead [49]. AODV depends on a per-node sequence number to avoid loops and to select the most updated routing path. AODV is a hop-by-hop routing system which depends on simple route request and reply messages to find out the routes of the requested connectivity. Each route has its own lifetime that is renewed once it is used, otherwise the route will be discarded in case it is not used during this lifetime interval. The routing table maintains information about the destination node IP address and sequence number, the next hop, the route lifetime, and the required routing flags. AODV provides multicast and unicast connectivity, and supports the Quality of Service (QoS) of MANET connections.

2.11.2 AODV Route Discovery (RREQ and RREP)

Once the source node (S) needs to send a data packet to a destination node (D), it searches its routing table for a route. If the required route is found, a route discovery is raised and a Route Request packet (RREQ) is flooded to the next nodes in the network as shown in Figure 2-11a. The RREQ forwarded packet indicates the IP address of the destination node (D), the IP address of the source node (S), the last known sequence number for the destination node (D) and the current sequence number. In addition, the RREQ packet includes other details such as the hop count, which was initiated with zero value and the RREQ ID, which is continuously increased the counter for each individual node. The RREQ ID counter is incremented each time a RREQ message has been invoked. A unique RREQ is identified from the source IP address of source node (S) and the RREQ ID value, which is used to avoid duplicated RREPs [50]. Figure 2-11 displays the flooding of the AODV Route messages.
Once a node receives a RREQ packet from a node, a reverse route to that node will be created. The node that the RREQ message was received from is the next hop in the route to the source node (S) that initiated the RREQ. Then the hop count in the RREQ packet will be incremented to calculate the hop distance from the source node. The routing table of the node receiving the RREQ checked for a route that has not exceeded its determined lifetime with an active and unexpired route for the destination node (D). If such a route does not exist, the node re-forwards the RREQ packet to its neighbours and increments the hop count value by one. The route search to the destination node (D) is repeated by re-forwarding the RREQ in case no route was available in the routing table when not receiving a RREP. When the active route entry inside the routing table has a sequence number greater or equal to the destination sequence number found inside the RREQ packet, a route reply will be generated.

The node's route table entry is at least as fresh as the source node's last known route to the destination node and can create a Route Reply (RREP) message, as shown in Figure 2-11b. The most recent route to the destination is selected and concurrently ensures loop freedom. The RREP packet contains the IP address for both the source node and the destination node. In addition, it contains the destination's sequence number as found inside the node's route table for the destination node. The hop count field inside the RREP packet is fixed to be equal to the node's distance number, which is the destination count number. The hop count is assigned a value of zero in case the RREP packet has been created by the destination node. The RREP is then directly transmitted towards the source node (S) using the reverse route that was created when the RREQ packet was received. This reverse route will be used to deliver
the RREP packet to the source node (S). The next hop nodes that receive the RREP packet will update their routing table with the routing details of the forwarded RREP packet as shown in Algorithm 2.1.

Algorithm 2.1: AODV route calculations algorithm [48]

<table>
<thead>
<tr>
<th>Step 1:</th>
<th>Node A receive control packet.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2:</td>
<td>Find the New_dest_seqno</td>
</tr>
</tbody>
</table>
| Step 3:     | If (entry to new source exists = No)
              | Add the new entry in routing table
              | else if (New_dest_seqno > Old_dest_seqno) Update Old entry
              | else if (New_dest_seqno = Old_seqno) && [(New_hop_count+1)] < Old_hop_count
              | Update old entry
              | else
              | no update in routing table      |
| Step 4:     | Routing table updated           |

The hop count assigned to this route is the hop count inside the RREP packet, incremented by one. These details of the forwarded packet will be used in case the source node selects this path to transmit the data packets to the destination node as the RREP depends on the hop-by-hop forwarding scheme. As soon as the source receives the RREP packet, it will use the routing path for data transfer. If the source node receives more than one RREP packet, it will select the route with the greatest sequence number and the smallest number of hop counts [50].

2.11.3 Route Maintenance

Because of the nodes' mobility, the topology changes inside the network, this breaks the links in the routing paths. The routes by then need to be repaired to avoid the packet drops. The broken active route to the nearest consequent nodes to the source node (S) marks the routes to the unreachable destinations inside its routing table and a Route Error (RERR) message is created as shown in Figure 2-11c. The RERR message contains the list of the destination nodes that are no longer reachable as a result of the broken link. Once the RERR message is created, it sends to its upstream neighbours that were using this link. The upstream nodes mark the broken routes as invalid, generating a RERR message, and send it to their upstream neighbour nodes that were using the broken link. The RERR message will be sent back to the source node by using the reverse path. Once the RERR message reaches the source node, the source node uses it to repair the route. The other nodes search on their routing table as soon as they
receive the RERR message to find a route for the unreachable destinations that are listed in the RERR. In the case that no route was found for the requested destination, it is marked as invalid and the node broadcasts a new RERR message to its neighbouring nodes. This stage will be repeated until the source node receives a RERR message. After marking the listed routes as invalid, a new RREQ will be generated if the route is still needed [50, 51].

### 2.11.4 AODV Performance

The behaviour of AODV affects its performance with different data transmission processes. AODV does not retransmit lost data packets and does not guarantee the packets delivery, however the delivery percentage is almost to 100% for a small few number of nodes [51]. The packet delivery ratio drops with the increased mobility ratio of nodes. On the other hand, the overhead in AODV packets needs to be on its minimum possible level as it is related to the RREQ, RREP and RERR messages’ behaviour. The number of overhead packets increases with the increase of the nodes’ mobility ratio as this increases the frequent link breaks and the route discovery time. For route optimisation in AODV, RREQ messages are initially sent with a small Time-to-Live (TTL) field to limit their propagation, as shown at the route discovery mechanism in Algorithm 2.2. AODV gives the nodes the ability to repair the broken active links locally instead of notifying the source node (S) with less overhead, delay and packet loss; however, longer delays and greater packet loss may happen in case unsuccessful repairs occur [52].

**Algorithm 2.2**: Route discovery algorithm [48]

<table>
<thead>
<tr>
<th>Goal:</th>
<th>Route discovery</th>
</tr>
</thead>
<tbody>
<tr>
<td>Declaration:</td>
<td>Route Request (RREQ), Source (S), Destination (D), TTL</td>
</tr>
<tr>
<td>Step 1:</td>
<td>RREQ generate</td>
</tr>
<tr>
<td>Step 2:</td>
<td>if (RREQ reached to node = yes)</td>
</tr>
<tr>
<td></td>
<td>2.1 else if (check new node= D)</td>
</tr>
<tr>
<td></td>
<td>2.1.1 Generate route reply</td>
</tr>
<tr>
<td></td>
<td>2.2 else if (check TTL&lt;=1)</td>
</tr>
<tr>
<td></td>
<td>2.2.1 Exit and send RERR</td>
</tr>
<tr>
<td></td>
<td>else</td>
</tr>
<tr>
<td></td>
<td>Rebroadcast RERR</td>
</tr>
<tr>
<td>Step 3:</td>
<td>Route Found</td>
</tr>
</tbody>
</table>

### 2.12 Optimized Link State Routing (OLSR)

OLSR was determined by the IETF MANET working group as the RFC 3626 [53]. OLSR is a routing protocol that uses the basic functionality of link state routing and distance vector. Being a proactive routing protocol, OLSR provides the immediate routes available to the destination. In this protocol,
each node maintains topology information and updates it periodically using link state messages [54]. OLSR provides certain optimization mechanisms to improve its functionality, which are Multipoint Relay Selectors (MRS), Multipoint Relay (MPR) and Control Messages. The Multipoint Relay is used to minimize the control overhead and the number of rebroadcasting nodes. Each node’s one-hop and two-hop neighbours are found using periodic broadcasting of HELLO messages. Then each node selects its one-hop neighbour to be an MPR, so that all two-hop neighbours are reachable from at least one member of an MPR set. Only nodes that are MPRs of at least one node rebroadcast the packet for the selection of an MPR in the entire network, whereas the nodes that are not part of an MPR set receive and process each control message but does not rebroadcast it. On the basis of one-hop and two-hop information, each node calculates its route to destination and saves it into its own routing table. Thus the OLSR protocol is designed for a complete distributive nature platform, and does not require any kind of reliable transmission for its control messages, like HELLO messages and similar. The protocol uses a sequence number in its control message in order to deliver the messages. The protocol is designed to perform a hop-by-hop routing, which guarantees the packet delivery to the destination node frequently [55, 56].

2.12.1 Multipoint Relays

Multipoint Relay (MPR) is a key concept used by the OLSR protocol to minimize the flooding of rebroadcasting messages in the network. Based upon its one hop neighbourhood and two-hop neighbourhood, each node selects an MPR such that its two-hop neighbour can be in a reachable state. It must be at least a one member set of MPR. This set consisting of neighbour nodes is known as Multipoint and each node selects its own MPR set, as represented in Algorithm 2.3. The selection of an MPR set of ‘N’ nodes is denoted by MPR (N) and is selected such that each node which is in the two-hop neighbourhood of N must have a symmetric link towards MPR (N) which should be a reachable state to MPR (N), and the MPR set should be minimized so that one can minimize the traffic control overhead [57, 58]. For the Multipoint characteristic, each node in the network maintains information about the neighbourhood node which is in the MPR set. This set is called a Multipoint Selector set (MS). This information can be obtained using periodic HELLO messages received by the neighbour. Figure 2-12 shows an example of a network topology for MPR set selection.
**Algorithm 2.3:** MPR selection algorithm

**Goal:** node x has to select its MPR set of a set of nodes in 1-hop neighbourhood $N_1(x)$, which provide a 2-hop neighbourhood $N_2(x)$ for node x.

Step 1: Select nodes of $N_1(x)$ such that it covers the isolated nodes of $N_2(x)$.

Step 2: Select those nodes in $N_1(x)$ which are not selected in step 1, to provides a larger coverage to $N_2(x)$.

Step 3: Continue step 2 until all nodes of $N_2(x)$ are covered.

In Figure 2-12, the 1-hop and 2-hop neighbour sets of F are:

1-hop neighbourhood set = \{C, G, H, E, \ldots\}

2-hop neighbourhood set = \{A, B, D, P, Q, \ldots\}

The minimum set needs to be determined where node E has the minimum number of neighbours, therefore:

MPR set \{F\} = \{C, G, H, E\}, where C=5, G=4, H=4, and E=3

![Diagram of network topology for MPR set selection](image)

Figure 2-12: An example of network topology for MPR set selection [53]

In this way, the MPR set and the MS set will be selected within any network topology. In Figure 2-13, another simple MPR flooding structure is represented, where a broadcasting message to all nodes of a 1-hop neighbourhood is taking place. Then a rebroadcasting message will be generated by only those nodes which are in the MPR set as a 2-hop neighbour to the 1-hop neighbour nodes.
2.12.2 OLSR Control Messages

The OLSR protocol consists of three kinds of control messages to support its functionality. These are the HELLO message, the Topology Control (TC) message, and the Multiple Interface Declaration (MID) message. The HELLO messages are periodically broadcasted by each node to find out its one-hop and two-hop neighbours. These messages are generated and distributed for each link which is participating in the network. The HELLO messages consist of a link code, a link message size, and a neighbour interface address in its HELLO packet format, as discussed in [56]. The HELLO messages are utilized to perform the link sensing, neighbour detection, and MPR selection tasks. When a node is selected as an MPR, it advertises itself and its MPR selector set to its neighbour nodes by propagating a Topology Control (TC) message. This message is used for updating by the nodes that are not in the MPR set. Each TC message consists of a sequence number and the neighbours’ main address in its packet format. Furthermore, the MID messages are used by the node when it has more than one interface; then the node itself propagates these additional interfaces periodically to other nodes with the help of the MID messages.

Figure 2-13: MPR flooding in the network [57]
2.12.3 Routing Table Calculation

Each node has its routing table for routing the data to the destination. The routing table contains the link set information, network topology, and it gets updated as each node gets a control message. The format of the route entries in the routing table consists of the routing destination address (R_dest), the next hop address (R_next_addr), the topology destination address (T_dest), and the interface address (R_inface_addr) [54]. Each destination route entry is recorded in the routing table. The routing table needs to be updated when a change occurs in the link set, 2-hop neighbour set, neighbour set, topology set, and multipoint interface associative information base. It is suggested that each time when the neighbourhood node of a node is lost or adds a new one, a 2-hop neighbour node is lost or a new one is added, then the routing table needs to be recalculated. The routing table is calculated [54] as in Algorithm 2.4.

**Algorithm 2.4:** Routing calculation algorithm

**Goal:** Update routing table

**Step 1:** Remove old entries from table

**Step 2:** Add one-hop neighbour (h=1) entries as destination in table

2.1 If (one-hop neighbour is bidirectional)
   Then add new routing table entries as destination such that
   \[ R_{\text{dest}} = R_{\text{next}}; \text{ and } \text{Dist}=1; \text{ else} \]
   2.2 Break;

**Step 3:** Increase value of h by 1, and record destination node for 2-hop neighbour

3.1 Continue, till no new node found

3.2 If (destination addr in topology table ≠ corresponding dest addr of routing entry)
   Then
   3.2.1 Add \( R_{\text{dest}} = T_{\text{dest}} \) and \( R_{\text{dest}} = h+1; \)

**Step 4:** Routing table updated

2.12.4 OLSR Performance

The OLSR protocol is a proactive routing protocol. The major advantage of this protocol is to minimize the size of each control message and the number of rebroadcasting nodes by using a multipoint relay strategy. The OLSR consists of three tables, which are the routing table, the neighbour table, and the Topology table. For \( N \) number of nodes in the network, the complexity of memory overhead of the OLSR protocol is \( O(N^2) \) and control overhead complexity is also \( O(N^2) \) [55]. The route maintenance task in the OLSR protocol is performed after packet delivery or data information is timed-out, in case of link failure. Due to this MPR set updates, the routing table blocks the router until those routes are
terminated. In the case of large networks, a larger time computation is required to update the routing table in the MPR with larger values of Topology Control (TC). Whenever the values of the TC parameter increased, the update intervals of the routing values become more effective [55]. OLSR maintains routes periodically to all the destinations, as this is beneficial for such networks where a set of nodes are communicating with another set of nodes. Since routes are provided quickly to each destination, OLSR benefits the applications where the transmission delay is not negotiable. As the performance of any networking protocol is based on different specifications and conditions (like mobility, scalability, transmission, and others), OLSR has the best-suited performance for networks with high density where communication takes place between the largest sets of nodes.

2.13 IPv4 and IPv6 MANET

Recent trends in ubiquitous computing have provided a new direction to MANET implementations. Characteristics and application domains of Mobile ad hoc networks have already been discussed in previous sections. Routing is a challenging task in Mobile ad hoc networks and each node in the network needs to be identified uniquely. Hence, the Internet Protocol (IP) must be an integral part of Mobile ad hoc networks. Both IPv4 and IPv6 are considered as the main platforms for different network systems [36]. Therefore, in this research study, the implementations are considering the IPv4 and IPv6 for the investigated research area.

2.13.1 Challenges and Issues in IPv4 and IPv6 Mobile Ad hoc Networks

With the successful deployment of various applications in Mobile ad hoc networks, it opens many challenges in IP-based Mobile ad hoc networks. Some of them are summarized in this review. Each node in Mobile ad hoc networks needs to be identified by a unique global address, and addresses need to be configured globally so that it can communicate to other nodes as well as to any node in the Internet. Since support for IP address configuration can be provided by either DHCP server or by manual configuration in a wired environment, this is a challenging task in case of Mobile ad hoc networks. Therefore, for assigning IP addresses to each node there is a need for auto-configuration in Mobile ad hoc networks and this process is easier in IPv6 since it has larger address space compared to IPv4 and it is easy to generate global addresses in IPv6.

In Gont’s work [60], a proposed approach for an IPv6 stateless auto-configuration scheme to provide a link’s local address in case of unavailability of a DHCP server. Thus this process can be modified for facilitating the support to Mobile ad hoc networks, and many researchers have worked on this method, such as [61, 62, and 63]. The use of mobile IP is a new key concept and has become very popular for mobility support in IPv4 and in IPv6 for ad hoc networks [64 and 65]. This approach uses the concept of two addresses: home address and care of address. A home address is attached for each node and when
this node moves to any new location it is provided with a care of address by a remote agent, and the message is delivered to the node via the care of address. To access the Internet from a Mobile ad hoc network is another aspect in new generation IP-based Mobile ad hoc networks. The gateway node has access to the Internet, and the mobile nodes who want to connect to the Internet have to discover the gateway node first. The challenges in Internet access in IPv4 are more complex due to less address space, use of the NAT process for private addresses in networks. As MANET is a self-configuring network, so nodes may leave or join the network as they wish. When a node leaves the network, it needs to leave its allocated addresses, as other nodes should be aware about the nodes that have left the network, so that its IP address can be recovered to provide the dynamic connectivity and prevent any security issues in MANET.

2.13.2 QoS in IP Mobile Ad hoc Network

Special attention needs to be given for achieving a reliable QoS model for Mobile ad hoc networks. Different QoS classes are needed, like QoS provisioning for routing, MAC layer level, scheduling, and resource reservation [66] to provide the support to the Mobile ad hoc network for its variety of applications and features. In the dynamic environment of Mobile ad hoc networks in IPv6, there are many challenges like hidden terminals, link fluctuations (due to node mobility), and end-to-end QoS support needs to solve these for reliable service in Mobile ad hoc networks. In [67, 68, and 69], reviews were provided on various QoS provisioning approaches including various classes like routing, MAC, and others. They had concluded that many QoS features like bandwidth, throughput, and end-to-end packet delivery can be implemented through some available methods. For obtaining the QoS support in Mobile ad hoc networks using the routing strategy only is not sufficient. It needs to include many other attributes like resource availability, bandwidth, and power consumption, and others. The IETF has proposed two mechanisms to support the QoS: IntServ and DiffServ. Each packet receives a particular level of QoS as the type of service in IPv4 is octet, and the traffic class in IPv6 is octet. The IPv6 header has two separate fields for QoS as a 20-bit flow level that uses the IETF Integrated service, and an 8-bit traffic class indicator that uses the IETF Differentiate service.

2.14 Mobility models in Mobile Ad hoc Networks

Nodes in Ad hoc networks are connected via wireless link which proves to be a less reliable media rather than the wired environment, hence routing in such networks proves to be more complex. Frequent changes in topology makes routing a challenging task in ad hoc networks, so these mobility issues need to be addressed properly. Prior studies suggest that there exist a number of mobility models that are designed to analyse the pattern of mobility in mobile nodes, like the location, change of velocity over time, power consumption level, and others. These models are important for the behavioural study of
nodes in specific environments. Thus, while evaluating the Mobile Ad hoc network routing protocol, it is necessary to select a specific mobility model to analyse its performance regarding the routing performance for different applications.

### 2.14.1 Classification of Mobility Models

Prior studies suggest that there has been significant literature research available on mobility models. In [70], two models are proposed and broadly used for mobility analysis, which are the Trace model and the Synthesis model. The Trace model mobility patterns are analysed in real-time events; therefore, these models provide accurate information about node mobility behaviour. However, within a new network the environment cannot be modelled if traces are not available, so Synthesis models are used in such a scenario. The Synthesis model provides a realistic behaviour for the mobile nodes, without having the traces. Further specific categorization of these mobility models was proposed in [71]. The most widely accepted and used mobility models for MANET implementation are as follows:

1. **Random Waypoint (RWP)** [54] is the most widely accepted model in Mobile ad hoc networks. In this mobility model, mobile nodes wait in their respective location for a random time period, and wait for changes like direction, velocity, and others. When the waiting time is over, the mobile node chooses its direction (i.e. Destination) and speed (from the time interval \([V_{\text{max}}, V_{\text{min}}]\)) randomly. Now the mobile node moves to its newly selected destination and again waits for a specific time period to continue the process. The specific time slot is also arbitrary for each movement, and the node’s movement is also arbitrary. There are many other variants of RWP models available.

2. **Random Walk Mobility Model** [72] is also known as Brownian mobility, basically used in Cellular networks. This model is memory-less, and mobile nodes’ previous moves and directions are unknown. In this model, each mobile node moves to a new location with random direction and random speed. The new speed of the mobile node lies between \([\text{minsp}, \text{maxsp}]\), with its direction in the interval \([0, 2\pi]\). Each movement in this model has either a constant time interval or travel distance, therefore at the end of each movement, new speed and direction is calculated. Since the nature of the mobile node is unpredictable, therefore it is utilized for erratic movement of mobile nodes. This mobility model can also be utilized to test the mobility of a node around its start point without having any prior concern of its reachable point.

3. **Random Direction Mobility Model** [2] is equivalent to the Random Walk mobility model. In this model, the mobile nodes move randomly within the network but as soon as nodes reach the boundary, nodes will stop there for a specific time period (pause time). After the completion of the pause time, it again starts in a random direction. By this implementation, it overcomes the
clustering of the nodes within the specific area (which happens in the case of the Random Waypoint mobility model).

4. **Referential Point Group Mobility Model** [73] is a group mobility model which represents random motion of the group as well as the mobile nodes. It is designed for a group of similar nodes (Cluster). Within each group or cluster there will be a cluster head, which dictates the behaviour of the whole group. In this model, a logical centre is used to calculate the group motion using a group motion vector. Each group has a group centre and the motion of this group centre decides the movement of its group (including velocity, direction, and others). All the nodes of a cluster depend upon the cluster head to travel.

5. **Pursue Mobility Model** [74] is a mobility model which emulates a specific case, where mobile nodes are following a particular target. In this model, tracker nodes move towards the target node with random speed but are limited so that they can target the node successfully. It uses a single equation for each mobile node’s new position as:

\[
\text{new\_position} = \text{old\_position} + \text{acceleration (target old\_position)} + \text{random\_vector}
\]

where random\_vector = random offset of each mobile node

6. **Column Mobility Model** [75] is a model in which mobile nodes have a specific motion around a given straight line and the motion is in a forward direction. An initial grid is defined in this model and then the mobile node is placed in this initial grid such that it is at its referential point. Then the mobile node moves randomly around its referential point, and the new reference point is calculated as:

\[
\text{new\_ref\_point} = \text{old\_ref\_point} + \text{adv\_vector}
\]

where, adv\_vector= predefined offset.

7. **Gauss Markov Mobility Model** [76] is a model specified for a fixed timeslot (known as tuning parameter). At the start, each mobile node is assigned a particular speed and direction, which will change after a specific time interval (say t). Now the new speed and new direction at the \((t^{th})\) instance is calculated as:

\[
V_t = \beta \cdot V_{t-1} + (1 - \beta) \hat{V} + \sqrt{1 - \beta^2} \cdot V_{x_{n-1}}
\]

Where, \(\beta=\) tuning parameter, \(\hat{V}=\) means value of speed and Direction

\[
d_t = \beta \cdot d_{t-1} + (1 - \beta) \hat{d} + \sqrt{1 - \beta^2} \cdot d_{x_{n-1}}
\]

Where \(V_{x_{n-1}}\) and \(d_{x_{n-1}}\) are Gaussian variables
This is a summary of most acceptable and available mobility models found in research work done by researchers in Mobile ad hoc networks. These Mobile ad hoc network mobility models are very useful in evaluating the performance of routing protocols.

2.14.2 Random Waypoint Mobility Model

The Random Waypoint (RWP) mobility model is a benchmark and the most commonly used mobility model in ad hoc networks [67, 68, 71, and 77]. This algorithm uses pause time before changing the direction. The detailed RWP algorithm is represented in Algorithm 2.5. The mobility pattern of mobile nodes in the RWP model is shown in Figure 2-14, the randomly selected mobile node start point is at location (x0, y0). This node waited for the pause time, and afterwards it selected the new destination/direction with constant speed with coordinates (x1, y1) and this process goes on. Thus was found a mobility pattern in the RWP mobility model.

<table>
<thead>
<tr>
<th>Algorithm 2.5: Random Waypoint (RWP)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Goal:</strong> To find the mobility of mobile nodes in a Mobile ad hoc network</td>
</tr>
<tr>
<td>Declaration: Mobile nodes, Mobile node direction, Speed (V), Pause Time (t)</td>
</tr>
</tbody>
</table>

Step 1: Mobile nodes stay at a location (e.g. [x0, y0]) for predefined pause time (t).

Step 2: if (t expires)

then

2.1 choose a random destination, and move directly to it
2.2 select speed from uniform interval [Vmin, Vmax]
2.3 on arrival at new destination
2.4 go to step 2.1 if the pause time (t) still not activated

Step 3: Mobility calculation
Figure 2-14: Mobility pattern in the Random Waypoint mobility model

RWP is the most applicable mobility model because of its simplicity and resemblance to real life different implementations. Initially all the mobile nodes are distributed randomly. Each mobile node determines the speed from its time interval:

\[
\text{Speed Node } (x) = [V_{\text{min}}, V_{\text{max}}] \\
\text{Average Speed } = [(V_{\text{max}}+V_{\text{min}})/2] \\
\text{if } V_{\text{avg}} \to 0, \text{ then } V_{\text{min}}=0, \quad \text{where, } V_{\text{avg}} \text{ is average speed, and } V_{\text{min}} \text{ is the minimum speed.}
\]

The actual average speed is found to be much smaller in the RWP model. As each mobile node starts with a random position and starts the same procedure to move to its next destination, so it is expected that each point is visited with similar probability. However, the location probability distribution is found to be highly skewed. The system stabilization process in the RWP model is very slow, and if the minimum speed becomes zero, then the system moves to an unstable state.

2.15 Mobility of Nodes in Real-time Application

The dynamic topology nature of Mobile ad hoc networks makes this an unpredictable and challenging task for the deployment of real-time application. There are various applications that are successfully deployed in Mobile ad hoc networks, but varying mobility features of nodes are expected to have significant impact on the performance of real-time application. With the proliferation of Internet technologies there is a paradigm shift from traditional computing to future Internet technologies like Cloud computing, WSN, VANET, and others, all on the same platform accessing through wireless technologies. Thus, there will be an enormous increase in the number of applications. There are various real-time applications like video conferencing, multimedia applications, weather report prediction, remote area monitoring, VoIP, and others, which need to be deployed over the ad hoc wireless
environment. Mobile ad hoc network is widely accepted and successfully deployed as a wireless ad hoc network with dynamic distribution of its mobile nodes, which move within different mobility systems [70, 74, and 78].

Infrastructure-less, multi-hop communication, and dynamic topology features of Mobile ad hoc networks differentiate it from conventional wireless networks. The Mobile ad hoc network has already proven itself of great importance as it is noticing the deployment of various application areas. Self-configuring natures, infrastructure-less features are best suited for emergency applications, military operations, collaborative applications, and multimedia applications, and others. In emergency applications where previously available infrastructure is destroyed due to sudden catastrophes (e.g. Tsunami, earthquake, and others), Mobile ad hoc networks can be used to establish communications with the emergency units, and similarly in collaborative applications like video conferencing, multimedia chat, within classrooms, business meetings, and others. These applications can be one-to-many and many-to-many implementations; therefore Mobile ad hoc networks need to be deployed in various environments like broadcast and multicast nature. There has been significant research going on into real-time communication deployment to facilitate support for multicasting and unicasting. Real-time applications in Mobile ad hoc networks include Wireless Sensor Networks, VANET, video conferencing, and video transmission from the battlefield, and others, without wireless infrastructure. The major requirements of these applications are very low delays and packet sensitivity [69], so retransmission mechanisms cannot be applied to real-time applications.

In [54], the author proposed that there is a group mobility in the nodes in many real-time applications whereas other nodes move independently; for example, in a battlefield application a particular group of nodes will move in a group as they all are working on the same mission. In [80], a virtual military-based mobility model to provide applicability to military applications is proposed. In this model various approaches of node mobility are defined, when group mobility occurs. In [81], a proposed Reference Point group mobility model provides an insight on the relationship between mobile nodes and mobility. The study tried to find out the impact of mobility on the performance of routing protocols. In [82] the effects of human mobility on the links and available routes in Mobile ad hoc networks had been discussed. The study concluded that human mobility and collisions have a significant impact on the links and routes for MANET based applications. In both [57] and [79], the nodes’ mobility for MANET environments had been classified into three different scenarios: the source mobility, the destination mobility, and the network group mobility. Specific protocols had been used for analysing the scenarios and concluded that AODV and DSDV protocols provide an appropriate level of performance for source mobility and network group mobility. In addition, the reactive routing protocols had shown good performance over different mobility models compared with Proactive and Hybrid routing protocols. Most of the existing research for real-time applications is based on the study of the impact of nodes’ mobility over different routing protocols and their performance issues.
2.15.1 Impact of Node Mobility on Real-time Applications

The implementation of real-time services over wireless communication systems provides a wide range of applications for MANET platforms such as remote location monitoring, VoIP, video conferencing, and others. Real-time applications are an appropriate communication mechanism to interconnect various components in a future network environment. Furthermore, the real-time applications in MANET must guarantee reliability with timeliness, and minimize various constraints like latency, control overhead, connectivity, and others. To fulfill these requirements, there is a need to minimize the energy consumption, support quality of service for real-time data, and provide bandwidth efficient broadcasting and enabling the multicasting features of the routing protocols [30]. With the current available infrastructure of Mobile ad hoc networks, that includes the routing protocols, mobility models, security mechanisms and others, the impact of the node mobility on real-time application can be summarized as follows:

1. **Nodes’ Mobility**: The Mobile ad hoc network is infrastructure-less and self-configuring where the nodes build their own communication infrastructure and communicate with each other in such a way that each node works either as a host or as a router itself depending upon the situation. As any node moves from its position in the network, the network topology gets changed. Therefore, the link failure in MANET due to the nodes’ mobility is a primary hurdle in routing [82]. Link failure rate is directly proportional to the nodes’ mobility. When the node mobility increases, the link failure rate will increase as the link status becomes unpredictable. Therefore, the control overhead will increase because of the increase in the update messages, the network congestion increases, as will the elapsed time for the updating processes for the routing tables and the applications status. In such cases, high mobility of the nodes can prove a serious performance problem for real-time applications.

2. **Routing Protocol**: Routing is an important phenomenon in ad hoc networks that ensures packet delivery from source to destination. Previous studies have proved that mobility affects the performance of routing protocols. In addition, mobility affects various characteristics including node connectivity, number of nodes, packet delivery ratio, and control overhead. Therefore, node mobility makes real-time routing information, in such real-time applications, more peculiar. Routing for real-time applications must ensure the availability of resource while maintaining latency.

3. **End-to-end Delay**: The average of end-to-end delay needs to be minimized to ensure the timeliness requirement for real-time applications. Higher numbers of nodes with high mobility increase the probability of collisions and link failure in the network that leads to unpredictable latency for media applications. This is an unavoidable condition for real-time communication, where each node is bounded to a real-time event. With high probability of collision in the network, the timeliness requirement for real-time applications cannot be achievable.
4. **Bandwidth:** The bandwidth constraint in MANET is a very important issue as it is sometimes related to the power consumption levels. Due to the higher level of nodes’ mobility, both control overhead and memory overhead increase due to the routing conditions, link failure, broadcasting/flooding of messages, and others. Therefore, there is a trade-off where ensuring real-time application constraints can be limited by extra traffic overheads.

In conclusion, there is a significant effect of node mobility on real-time applications to achieve reliable communication for real-time events. Other related characteristics need to be addressed and analysed to enhance the general performance for real-time applications over MANET. A number of studies tried to focus on the impact of nodes’ mobility to enhance the QoS for specific real-time applications over MANET, such as VoIP and video conferencing [67, and 70].

### 2.16 Summary

In present IP-based multimedia environments, VoIP is emerging as an alternative for traditional telephony. SIP signaling empowers the user for initiating and managing multimedia sessions over IP networks. With the enhancement in IP communication, SIP provides various features, such as flexibility to the user for connecting with any new user in the network on the fly. This Chapter gave a comprehensive background about the SIP signaling system and its entities, interaction messages, supportive protocols, services, and implementations. SIP can work over the TCP protocol mostly for text-based applications and messages while SIP also can work over the UDP protocol for real-time applications such as audio and video.

The SIP signaling system has various implementations such as P2P SIP, proxy server based SIP, Redirect server based SIP, and IMS-based SIP. Single and multiple servers based SIP depends on the client/server architecture to provide more security and reliability services with better end-to-end communication systems for large numbers of nodes. However, P2P SIP is faster for small simple network systems. In addition, the P2P SIP has no centralised architecture which means that its usage is free from the performance threats of the single point failure, while the IMS-based SIP is considered as the major type for the next generation networks that can be used over different platforms and communication systems. However, IMS-based SIP has more complicated architecture that causes lots of synchronisation faults and longer delays for the SIP signaling system. In addition, this chapter considered the implementations of SIP signaling over transport layer TCP and UDP protocols. It shows that SIP/TCP is more common in use than the SIP/UDP for future IP network systems. In addition, the Queuing Theory for the SIP Server Systems with B2BUA services and capabilities was introduced. The main drawbacks of the of B2BUA SIP server implementations are the single point of failure and bottleneck that could affect the performance of the SIP signaling during the registration, initiation, and termination stages. The chapter showed that the SIP server efficiency is dependent on the queuing
system that is used for server interactions with the received packets of different SIP processes. Furthermore, the SIP signaling system showed an acceptable level of support for mobility implementations over simple network systems. However, this support is affected by different factors, such as connectivity with the SIP server, the synchronization mechanism, and the availability of the system entities.

The Mobile ad hoc network represents a complex wireless system that comprised of mobile nodes that move freely in the network, and it is self-configuring to establish a network infrastructure allowing communication between humans and mobile devices in remote areas with no pre-established infrastructure. In this chapter, it first explained the need of new routing algorithms for Mobile ad hoc networks and their classification, then provided a comprehensive overview of various routing algorithms used in Mobile ad hoc networks. This chapter focused on two of the most widely accepted protocols, AODV and OLSR, as both have been explained in detail. The routing algorithm issues are important and an integral part of Mobile ad hoc network performance, however, there are various constraints that affect the routing performance, such as nodes’ mobility. A review of various mobility models and their main features have been discussed. The Random Waypoint (RWP) is the simplest mobility model and is widely used, and was further explained in detail. In the future Internet, ubiquitous computing is a new paradigm, so there are various real-time applications that need to integrate with Mobile ad hoc networks. Hence, the impact of node mobility on real-time applications in MANET has been investigated as well.
Chapter 3

3 SIP over MANET – A Literature Review

An overview of the existing literature with research focused on the SIP signaling performance over MANET is presented in this chapter. In addition, an extensive survey is provided on the related work. This review is focused on the studies about SIP signaling over MANET and the performance enhancement approaches for SIP-based VoIP applications. Generally, SIP is implemented over MANETs with four different types of SIP signaling systems as represented in Figure 3-1. The first of them is peer-to-peer SIP over MANET. The main purpose in this case is not to use SIP servers. This chapter explains in detail much more about this kind of system in the following review of the existing research in this direction. The second type of SIP signaling system is SIP with multiple servers over MANET that includes registration, redirect and proxy. The third type of SIP signaling system that exists and is researched is SIP with a single SIP server that acts as a registration, redirect and proxy server over MANET. The research work in the presented thesis is focused on this kind of signaling system. The fourth type of SIP signaling system over MANET is SIP with an IMS system. This chapter will provide in the following exposition the current state of the research, results, gaps, advantages and disadvantages regarding the above-mentioned SIP signaling systems over MANET. Furthermore, the available performance enhancement methods for SIP signaling over MANET will be discussed in the current chapter.

Figure 3-1: Literature survey of types of SIP signaling system implementations over MANET
The four implementation types for SIP signaling systems had been introduced in Chapter 2. In this chapter, a survey about the SIP signaling system implementations over MANET will be provided to have a comprehensive review of the current state of the art. There are a number of researchers who focused on adapting the SIP to MANETs. Generally, they can be classified in two classes, according to which of the nodes take action as SIP servers in the network. The first class is characterized with the implementation of the SIP servers to all nodes and each node is registering locally or is broadcasting the location information in the entire network. The other class is distinguishing some nodes to take action as SIP servers. This literature review represents the current state of the proposed research area in terms of the investigation, the evaluation and the service enhancement efforts. Moreover, the supportive simulation tools and test-beds for SIP signaling implementations for MANET will be discussed in this chapter.

### 3.1 Peer-to-Peer SIP Signaling Implementations over MANET

The authors of [83] propose two solutions for enabling SIP in MANETs: dSIP and sSIP. According to the dSIP, a REGISTER request is broadcasted from each node in order to notify all nodes in the network with the information of its location. Discovery of the members in the network is accomplished by local probing in the cache. To enable Session Initiation Protocol in MANETs, the Service Location Protocol (SLP) [84] is used by sSIP [83]. An SLP request is broadcasted from the node that is connecting with the ad hoc networks in order to ask for bindings of the users that are available. After this, every node that receives an SLP request responds using an SLP reply that includes its binding. However, as mentioned earlier, using this kind of solution can cause flooding. That is the reason for problems arising when they are used in larger ad hoc networks.

The authors of [85] employ peer-to-peer cover that is structured and related to Chord [86]. With the purpose of mapping the users with the relevant connection information, a Distributed Hash Table (DHT) protocol is used by the nodes. Hence, when some of the nodes connect to the Chord cover, they will be in charge for keeping the information related with the part of the cover that is mapping to its estimated Node-Identification (Node-ID). High control overhead is caused by maintaining the hash tables. Registration in [87] is realized utilizing the multicast mechanism with IPv6. A REGISTER request is multicasted by the node to announce its appearance in the whole network. User-List-Cache is updated by each node when REGISTER updates are received. After that, each node is replying by providing the information to the correspondent employing unicast. However, this kind of alternative is also not effective because, for large ad hoc networks, preserving the User-List-Cache gives poor results.

Initial research on the subject of SIP over MANET was started in 2003 by Khlifi et al [88]. In this work, a framework for conference signaling using SIP is presented allowing a MANET user to discover, initiate conferences, and join existing conferences with other users. After this work, the next research
on SIP over MANET was in 2004 in [89], where SIP is set up over OLSR utilizing a cross layer, integrated application and routing layer, in order to assist proxy-less and proxy-based systems. A proxy-less system is without a proxy server and proxy-based SIP MANET contains at least one SIP proxy server. Research in the field of proxy-less SIP MANET, i.e. SIP peer-to-peer over MANET without SIP servers is presented in [90]. The authors in this work proposed a signaling system that is unique and is used for sessions in P2P ad hoc networks. The framework when SIP is used in ad-hoc networks provided in [88] is enhanced in [90] by establishing hierarchical clustering architecture. This concept is confirmed by performing testbed running on eight computers. The benefit of the system proposed in [90] is in generating a lower number of overhead messages compared to [88].

The majority of the P2P SIP over MANET approaches employs resource discovery mechanisms in order to offer SIP user location discovery. Hence, P2P SIP over MANET approaches could be also classified into P2P SIP without overlay network and on the overlay network. Although the majority of the SIP over MANET approaches employs SIP register and user discovery operations in MANET, they do not deal with the working of their protocols in heterogeneous networks in order to supply interoperability between MANET and Internet SIP users. Research with solutions for Internet connected MANET environments are presented in [83, 91, 92, and 93]. Yet, this research relies on a centralized SIP registrar/proxy that can be positioned at the Internet or at the MANET gateway. However, this kind of centralized registrar/proxy is the reason for creating a traffic bottleneck when SIP requests are sent to the gateway and for creating a single point of failure. With centralized architecture, users’ SIP binding information is kept on MANET by one or a few MANET gateways, named SIP gateways. The role of the SIP gateways is also to forward the received SIP register requests from MANET users to an external SIP registrar on Internet.

In [87] the authors design and implement the pseudo Session Initiation Protocol (p-SIP) server embedded in each mobile node in order to provide the ad-hoc VoIP services. The benefit from this paper is that the implemented p-SIP server is compatible with common VoIP user agents, and it integrates the standard SIP protocol with SIP presence in order to handle SIP signaling and discovery mechanism in the ad-hoc VoIP networks. The advance of this work is also that the implementation is based on real equipment. The implementation of p-SIP is done on IBM ThickPAD x32 laptops, equipped with IEEE 802.11g wireless communication, using the Ubuntu Linux 6.10 and applying Kphone 4.2 as UA on top of the embedded p-SIP server. These authors, with the implementation of the testbed and the performance measurements from the experimental setup, have shown valuable analysis of the ad-hoc VoIP network. The results of this work demonstrate the possibility to achieve ad-hoc VoIP services using the implemented p-SIP servers. However, in this work more complete presentations of different UDP packet sizes, injection rates and contention scenarios are not provided. What is provided is the information on the influence of TCP/UDP traffic that contend VoIP streams in ad-hoc networks. Further possible research in the direction of the work in [87] is the influence of the ad-hoc nodes’ density on
the performance and the limitation of forwarding hop counts to realize acceptable VoIP QoS in the ad-hoc network.

The authors of [94] suggest a framework for service provisioning in stand-alone MANETs. Contributions that this work provides are in the new model of business that is harmonized with the features of MANETs and that allows invocation and execution of the services, and in the allocation system of the SIP servlets and overlay networks as a service execution environment. Because the proposed model does not have a central unit, and it is adaptable when dedicating functions in order to grant the functionalities, any user can take part in possessing the required features. The suggested functional distribution by the authors of this work deals with the number of independent units and the loose coupling. In [94] the authors also propose a covering network for execution of the services in MANETs that are stand-alone, based on the framework of the SIP servlets. The benefit of the paper is also in the prototypes built as verification of ideas for the model of business and the allocated system. Provided solutions in this work are proving that the model and the scheme are reasonable with a satisfactory time of response. The covering network protocol in the results is also formally validated. However, yet more detailed validation would be needed here in this presented work.

In [95] these authors propose the architecture of a MANET emulator suitable for SIP services on a MANET. The proposed architecture in this work supports real-time audio/video communication, node mobility, and peer-to-peer-type communication. The SIP_MANET emulator has been developed in [95] based on the proposed architecture, and it is confirmed that solid communication quality can be maintained with SIP applications. Communication quality evaluation is also conducted to confirm the effectiveness of the simulator. In order to make achievable usage of the MANET emulator for verifying a SIP application, it is suggested to add the capabilities to translate the IP address and port numbers, to give priority to AODV packets, and to process transmission/reception of packets in multiple threads. The percentage of successful audio and video communication in a SIP application is approximately 95% when the nodes stay still, hence the communication quality in this case is satisfactory. From the given results in this work when the nodes are moving, this percentage is decreased to approximately 77%. Still, in the presented work in [95] multi-path protocols are not taken into consideration and are not included in the testing simulations. Hence, to enhance the communication quality when nodes are in movement, additional research of multi-path protocols is needed here.

The work in [96] represents an innovative Peer-to-Peer (P2P) framework for SIP on MANET. The focus here is on distributed P2P resource lookup mechanisms for SIP that tolerate failures resulting from the node mobility. The authors of [96] proposed a novel P2P lookup architecture based on a Structured Mesh Overlay Network (SMON) that enables P2P applications to perform fast resource lookups in the MANET environment. Their approach extends the traditional SIP user location discovery utilizing DHT in SMON in order to distribute SIP object identifiers over SMON. Simulation and experiment results
are conducted from which it is concluded that SIPMON provides the lowest call setup delay comparing it with the existing broadcast-based approaches. In [96] the authors also propose a new OLSR Overlay Network (OON), which is a single overlay network that contains MANET nodes and nodes on the Internet. Testbed experiments are conducted proving that extended SIPMON (SIPMON+) gives better performance expressed in terms of call setup delay and handoff delay compared with MANET for Network Mobility. These authors also contribute in [96] with a proof-of-concept and prototype of P2P multimedia communication based on SIPMON+ for post-disaster recovery missions. This concept is evaluated with experimentation in real disaster situations – Vehicle to Infrastructure scenarios - and it is concluded that the proposed prototype outperforms MANEMO-based approaches expressed in packet loss, call setup delay, and deployment time. The proposed framework in [96] can be easily implemented to day-to-day growth of the Internet connectivity. It will be interesting to see this work to be continued in addressing how TCP-based applications can be provided on SIPMON+. In this direction, other mobility issues like session mobility are motivating for research.

### 3.2 Implementations of Multiple SIP Servers over MANET

The authors of [97] propose AdSIP, that is a protocol that is distributed and that allows SIP in MANETs. Realization of this protocol is on the network simulator ns-2 where comparison is made with the Tightly Coupled Approach (TCA) in terms of pertinent metrics, average session establishment time, failure rate and consumed bandwidth. Comparison shows that the proposed protocol in [97] performs improved adaptability and scalability to the mobility of the nodes. The solution in this work called AdSIP, chooses a group of nodes that are mobile to operate as SIP servers, and they establish a virtual infrastructure as overlay on top of the physical network. A new distributed algorithm is built to construct the topology and to assign dynamically previously explained functionality to a group of nodes in the network. The simulation results obtained using the ns-2 simulator clearly state that the proposed AdSIP protocol is well-adapted to mobile ad hoc networks, giving lower session establishment time, low control overhead and high service availability. However, this work lacks confirmation of the proposals with real results that could be obtained with a real scenario, instead of the ns-2 simulation tool.

Proactive route optimization in SIP mobility is introduced in [98] in order to achieve a reduction in the session setup latency. According to the proposed Session Initiation Protocol – Proactive Route Optimization (SIP-PRO), during the location registration step the mobility binding information is pre-fetched and used for session establishment. Reduced session setup latency in this work is accomplished by eliminating traverse over multiple SIP servers using the proactive route optimization. Hence, in this work these authors propose a novel idea called proactive route optimization. It achieves, when a session is initiated, direct establishment of the session with the callee if the caller has valid mobility binding information. A mobility-aware pre-fetching scheme is developed where only the mobility binding
information with lower mobility is selected, as it is more probable that such information could be used for session establishment. In [99] the authors also propose a new session setup procedure where mobility information with a sufficient residual time is used. Still, this work fails to perform extensive simulations with the developed analytical models in order to verify the proposed optimization and procedures.

### 3.3 Implementations of Single SIP Server over MANET

The implementations of Single SIP Server over MANET will be used in this research study as will be introduced in Chapter 6. The overview for the current state of the art will help with understanding the SIP implementation issues and available efforts to reference the efforts of this research study with the related researches in the literature. In [100], An intelligent VoIP system with embedded pseudo SIP server in an ad-hoc network is implemented. The embedded pseudo SIP server presented in this work is compatible with common VoIP user agents using SIP and it acts like middleware between the application and the transport layer. The VoIP quality level of the service is performed with the transmission delay for signaling and voice packets and it shows acceptable results with the conducted testbed. The proposed pseudo SIP server in [100] utilizes SIP presence to discover the mobile device and exchange the signaling over an ad-hoc network. However, this work lacks other performance metrics (not just transmission delay) in the experimental results to confirm the quality of the proposed SIP server.

Converting IP addresses and port number and rewriting SIP messages is required in order to enable a MANET emulator to provide SIP services. However, the disruptions may arise between SIP clients, and the real-time performance can fall. The authors of [101] propose an architecture of a MANET emulator and local multipath routing appropriate for SIP services. A SIP_MANET emulator is developed and correct operation of the SIP_VoIP call is confirmed. The proposed routing method from the authors of this work provides high probability of retaining the required path. Their developed system is well described and the evaluation results are presented in detail. The proposed routing method is compared with AODV and the disjoint multipath routing, using the MANET emulator and the described evaluation model. Call holding time is measured, defined as the time from the start to the disconnection of the call. Path retaining probability is also calculated and it verifies the effectiveness of the proposed local multipath routing. The proposed routing method uses a spare path when some node in the used path fails and that is the reason why its path retaining probabilities are higher than that of AODV. It would be very useful if the proposed local multipath routing in [101] is compared with AODV on more varied network models to have more detailed results in this domain.

A SIP-based mobile network architecture for Network Mobility (NEMO) in vehicular applications is developed in [100]. The focus in this work is on developing a MANET that contains hosts that are mobile and that are in a vehicle or in a group of vehicles. In [100] the MANET is linked with a SIP-
based Mobile Network Gateway (SIP-MNG) to the outside, equipped with external wireless interfaces and internal 802.11 interfaces. In this direction, SIP-MNG is supporting call admission control and resource management for the MHs. The authors are proposing a boost mechanism with message service that is short for the purpose of waking up in an on-demand manner the wireless interfaces. Signaling with details is presented regarding this mechanism. Additionally, this system is completely well-matched with the SIP standards that are accessible. Prototyping practice and performance measurements outcomes are presented. The proposed system in this work saves fees for internet access that is beneficial for operators of public transport or users, allowing the sharing of one interface from multiple sessions. Furthermore, with this kind of design, vehicles could offer access to the Internet to travellers supporting the group mobility. A proposed push mechanism allows SIP-MNG to stay off-line when calling activity is not present and to be activated when there is a need. Maintaining global accessibility of the users, the proposed push approach also saves call charges and energy. From the presented experimental results, it is demonstrated that for PHS, WCDMA, and 802.11 networks, it is possible for multiple stations to share one interface. Because for the proposed push mechanism, the call setup time is around 20s, the push server is also designed to select the session temporarily and to use the REFER scheme in order to transmit the session to the client within SIP-MNG. However, still the reconnection time of the wireless interface takes too long and further research is needed in this direction to reduce the reconnection time.

The authors of [102] depict a middleware infrastructure for session establishment and management in MANETs, called SIPHoc. SIPHoc is designed in order to be independent of the underlying network topology, so it supports both mobile and static MANETs. Hence, SIPHoc avoids the problem of having to elect nodes for specialized tasks and replacing them when conditions change. SIPHoc differs from the SIP standard in the fully decentralized implementation which does not require any centralized components, while they both provide the same interfaces. It is also presented that SIPHoc is message efficient through routing message piggybacking and is independent of the routing protocols. It is also shown that SIPHoc does not impose any topology allowing seamless interaction with the Internet. In this work the architecture and the implementation of SIPHoc is represented and its performances are evaluated. From the results presented in [102], SIPHoc has a message efficient system and provides a low dial-to-ring delay. In addition, SIPHoc allows the usages of the SIP-based applications in MANETs without modifications. This is proved by presenting how SIPHoc supports VoIP conversations within MANET and between the end-points and the MANET on the Internet. VoIP application is used in this work in order to study the SIPHoc performances and to prove that the resulting overhead is near the optimal and comparable with the results of the standard operations on MANETs.

Two approaches enabling SIP-based session setup in ad hoc networks are proposed in [103]. One of them is the loosely coupled method, where endpoint discovery of SIP is decoupled from the procedure of routing. The other approach is the tightly coupled method, which incorporates the endpoint discovery
with the cluster supported routing protocol that is fully distributed and that constructs a virtual topology for effective routing. This work also presents simulation experiments that prove that the tightly coupled method achieves improved results for the latency of the session setup of SIP over multihop wireless networks that are quite static, compared with the loosely coupled method. On the other hand, results show that the loosely coupled method generally has improved performance in networks that are characterized with random node mobility. In [103] the authors set up the problem related to basic deployment over ad hoc networks. Furthermore, integration of ad hoc routing protocols with SIP is done by the authors. Anyway, subjects that are associated with SIP supported application use for ad hoc networks are not addressed in this work. Essential SIP supported session setup for the applications is provided in [103] without considering special applications like, for instance, the SIP supported conferencing application. In the direction of the research in this work, it will be interesting to research the load balancing methods, deployment and design of specialized SIP supported applications.

3.4 IMS-based SIP Signaling Implementations over MANET

The IMS is a developing technology with a lot of potential for its usage in MANETs in order to offer a multimedia Internet experience for different kinds of users operating with a lot of various applications in a mobile environment. The employment of the IMS over MANETs and modern wireless and mobile networks is featuring a lot of needs and challenges. IMS is using a number of protocols, but its driving force is founded on the SIP. The IMS [104] is 3GPP/3GPP2 architecture standardized of the Next Generation Networks (NGN). The goal of this system is to fill the gap that exists between the cellular and the Internet worlds. Hence, the IMS permits operators to use the benefit of the interoperability and quality of telecoms and the modern progress of the Internet [105]. According to the work presented in [106], IMS is proposing a SIP servlets-based application server. Still, exploiting this pattern in MANETs for service provisioning requires a signaling layer. SIP servlets as an option are the best alternative according to the proposed SIP-based architecture for signaling in MANETs in [107].

Three main entities are related to the service provisioning in IMS: HSS, CSCF and the SIP AS. The most important data that are stored in the HSS are made up of user identities, registration information and security information. The main part represents the user profile. It resolves the services that are offered to each of the users and states the rules for triggering the services. The job of the S-CSCF is to download the user profile or its part from the HSS as soon as the user registers with that S-CSCF for the first time. The S-CSCF also is evaluating the initial filter criteria and communicates with the proper application server. Connections among the HSS, the S-CSCF and the AS are achieved with standardized IMS interfaces.
3.5 Performance Enhancement Approaches for SIP-based Applications over MANET

Current performance enhancement methods for SIP-based applications over MANET vary in terms of system features, requirements, implementation possibility, integration with existing systems, and costs. In general, the main performance enhancement methods are related to the dynamic adjustments for SIP timers, dynamic adjustments for the routing protocol parameters, implementations for supportive signaling systems, infrastructural based solutions, or service distribution features for the system users.

The dynamic adjustments for SIP timers are providing flexible implementation for SIP-based applications over different platforms. However, these thoughts are related to the theoretical studies where, in reality, the SIP adjustments need to consider the nature of the network systems that SIP signaling is working on. The wireless and mobility characteristics of MANET are affecting the SIP signaling performance [100, 106]. Therefore, applying the dynamic adjustments for SIP signaling systems is not a proper solution that can be taken over MANET systems unless the nature of MANET systems had considered this method. On the other hand, the dynamic adjustments for the parameters of MANET routing protocols have shown an efficient enhancement for different implemented applications. This method depends on accommodating the routing parameters to provide the best level of service for the implemented applications [99]. For SIP-based applications, using this method show an enhanced level in the performance for the SIP signaling and voice data transfer in general [99, 101]. This method is considered as one of the most effective performance enhancement methods. However, no efficient level of implementations had been shown for this method, especially for SIP-based VoIP over MANET for emergency and backup scenarios.

The implementation for supportive signaling systems for SIP is considered as one of the effective solutions. Therefore, the SDP signaling system improves the SIP signaling performance over MANET as it supports the management features of the SIP signaling system. However, the lossy nature of MANET is also affecting the SDP performance itself which increases the performance problems of SIP signaling [101, and 108]. Furthermore, the majority of the studies in the literature review implemented SIP without SDP in their research efforts. The synchronisation issues between SIP and SDP protocols had shown some concerns about the performance of SIP signaling especially in network systems that have variable natures or mobility related implementations [94, 96, 100, and 103].

The infrastructural based solutions are using different ways to enhance the SIP implementations over MANET. One of the suggested methods is used to implement multiple SIP servers with high performance to support larger numbers of MANET nodes [97]. However, this method is difficult to be implemented for emergency or communication backup scenarios because of the required
synchronisation functionality between multiple SIP servers for the mobile callers [99, and 106]. This method could be supported by using the IMS infrastructure as the synchronisation functions are secured by its infrastructure. The P2P SIP implementations are considered as the most direct and easiest infrastructural performance enhancement solutions, as described in Section 4.1 [96]. However, regardless of the QoS issues, large numbers of MANET-based callers are difficult to be communicated without a central SIP server [83]. Other infrastructural methods which are suggested include controlling the nodes’ speed, limiting the hop numbers, and reducing the background traffic of other simultaneous applications [87, 97, and 99]. Other research efforts suggested employing the service distribution features over the system users by scheduling the calls’ setup processes. These solutions control the user’s ability to initiate voice calls in certain conditions related to number of users and bandwidth portions. The main purpose of these methods is to reduce the concurrent calls by applying the time distribution features over the service users to increase the QoS level for the provided services [87, 109].

The performances of the reviewed performance enhancement methods vary in terms of the enhancement level and implementations requirements. However, both dynamic adjustment methods for SIP parameters and MANET parameters had shown a good level of performance enhancement. Thus, the most efficient method for enhancing the SIP signaling performance over MANET is to qualify the SIP signaling behaviour to meet with the mechanical nature of MANET systems. Combining both dynamic methods for SIP and MANET has a promised level of performance enhancement with lower costs and simple implementation. However, this enhancement method needs to be based over the evaluation studies for the current state of the art for SIP signaling over different MANET scenarios. In addition, the implementations of these enhancement methods has not fully investigated over clear identified mobility models for MANET nodes. The simulation or test-bed tools that used even not reflect reliable results that can considered as a reference results for the investigated methods. In addition, non of the proposed solutions in this section have considered any performance metrics for both SIP signalling systems and MANET routing parameters for the SIP-based applications over MANET.

### 3.6 SIP implementations with Cross-Layer Approaches

A limited number of Cross-Layering approaches employed SIP signalling for performance enhancements of a network system. Most of the implementations focused on the IMS systems to support the QoS and the performance efficiency for SIP-based applications. In [110], a proposed Cross-Layer approach for mobility management is based on the integration of Mobile IP (MIP) and SIP to support seamless mobility and scalability in IMS-based network systems for multimedia services. The approach module considered between home agent and S-CSCF that supported the information exchange between network layer and application layer. The approach suggested binding the update messages received by the home agent and transferring it to the S-CSCF to activate the registration procedure in IMS to control
and reduce the signaling between mobile nodes and the home network. The study proposed three mobility management solutions that were analysed theoretically and in simulations. The simulation results showed lower delays and lower signalling overhead over the IMS network system compared with the plain IMS system. In [111], a Cross-Layer fast handoff scheme for the IPv6 network proposed to utilize the link-layer triggers and IP address pre-allocation to improve the handoff speed for multimedia applications over IMS. The study integrated both MIP and SIP to support the mobility over different types of transport protocols. The experimental results showed that the handoff latency for SIP mobility in the proposed scheme is 77% shorter than the original scheme for mobile IPv6. In addition, the average handoff frequency had improved 3% with random movement scenarios and 27% with straight movement scenarios for IPv6 Mobile Nodes.

In [112], a Cross-Layer handover approach is proposed for SIP applications based on the Media-Independent Pre-Authentication and the Redirect Tunneling (MPA-RT). The approach is designed to reduce the packet loss ratio in Media-independent Pre-Authentication (MPA) systems for Mobile Nodes by integrating SIP signalling, MPA-RT, and Media Independent Handover (MIH), to provide a seamless handover for SIP applications. The simulation effort in NS-2 showed an enhanced level of performance for the proposed MPA-RT system in terms of packet transmission delay and the temporary buffer utilization over the network layer. In [113], a novel Cross-Layer Localized Authentication Mechanism (CLAM) proposed to secure the node’s mobility in Next Generation Networks depending on the network-based mobility protocols. Both SIP and Proxy MIPv6 were employed to secure real-time and non-real-time communications. CLAM was designed to reduce the handover latency, the signaling cost, the communication overhead, and packet loss. The research implemented CLAM in mobile devices using cryptographic and one-way hash operations. The study analysed and compared CLAM performance with other existing schemes where the conducted analysis results showed that CLAM has better performance compared with other schemes.

The 3GPP and IETF organizations have accepted Session Initiation Protocol (SIP) as the multimedia signaling standard in omnipresent wireless networks. However, challenges are still present at the handovers of real-time applications in the SIP-based wireless LAN. The most sensitive topic during these handovers is delay, which cannot be too long for real-time applications. The authors of [114] researched in this direction, so they developed a Cross-Layered Handover Management architecture for the SIP signaling system (CHM-SIP). Their solution gives lower service disruption time in scenarios where mobile nodes are making frequent handovers in the WLAN. Furthermore, the authors of this work designed in detail the Cross-Layer Handover Management for Voice over WLAN (VoWLAN) that is SIP-based. It uses for this purpose information about the speed of the mobile nodes and the handover signaling delay in order to improve the performance of the handovers. Hence, the authors in
When mobility management is handled using both the MIPv6 protocol and SIP as the usual protocol for session control, the replication of functions can lead to ineffectiveness. This challenge is researched by the authors of [115] who suggest an integrated scheme that reduces the delay of the session setup. The benefits of this work are supported by theoretical analysis of the delay and by simulation results. The authors suggest and analyse the optimizations in the session establishment. The results obtained from the simulations done with the ns-2 simulator prove that the optimized session initiation sequence is faster compared with the standard situation, especially in the case of greater link delays caused, for example, with long distance calls, or larger packet losses in the wireless connections. Still, this work lacks more thorough simulation results changing more parameters in order to get more reliable results.

The relationship between Peer-to-Peer Session Initiation Protocol (P2PSIP) and MANETs is not clearly defined in the existing literature. Hence, the authors of [116] are suggesting a new Cross-Layered scheme for managing the P2PSIP overlay. The benefit of this work is in offering helpful information that can deal with the main research problems of the P2PSIP contained by the MANETs.

### 3.7 Supportive Simulation Tools for SIP-based Applications over MANET

The implementation of SIP signaling as a RFC 3261 [2] over MANET protocols is available in few number of simulation and test-beds tools. Research that has been carried out for SIP signaling and MANET are utilizing simulation tools [117], although the consistency of the simulation results has been taken into consideration [118]. As a result of this, comparative researches have been published in order to confirm the achieved results [119, 120]. As is mentioned in [117], there are a lot of discrete-event network simulators that are accessible for the community of MANET [121]. In [117], the survey investigations shown that the most utilized simulator in MANET research is the Network Simulator-2 (ns-2) with 43.8% of the analysed papers. There are also a lot of self-developed simulators, 27.3% according to this study. The GloMoSim simulator is included with 10% of the MANET simulations, QualNet with 6.3%, OPNET® with 6.3%, the CSIM simulator with 2.5% and MATLAB with 3.8% from the 80 analysed papers in [117]. The OMNeT++ simulator is also used for simulations of the MANET technology. Its simulation programs possess a modular structure. The OMNET++ simulator includes delay as a function of the distance of the nodes, and in ns-2 the delay is a constant that is defined in the configuration file. Because of this, the same kind of parameters will give diverse results although the simulation scenario for MANET could be exactly the same in both simulators. MANET can be also simulated with the ns-3 simulator, which is an improved version of the ns-2 simulator. MANET routing protocols that can be simulated in ns-3 [122] are Ad-Hoc on Demand Distance Vector (AODV), Destination-Sequence Distance Vector (DSDV), Dynamic Source Routing (DSR), and
Optimized Link State Routing (OLSR). From all the simulators used for MANET and SIP signaling the most eminent and widely-used network simulators are ns-2 [123] and OPNET® [124]. There are significant deviations at various levels between these two simulators. Consequently, a number of modifications are required when it is wanted to repeat the results obtained using the ns-2 simulator with the OPNET® simulator.

The majority of the simulation parameters that are used in ns-2 and OPNET® simulators are equal, but there are parameters as the wireless buffer size or the transmission range which are different and they could influence the simulation results considerably. For example, if we use the 802.11 technology with 54 Mbps data rate, in ns-2 the default transmission range equals to 250 meters, but in the OPNET® simulator that default value of this parameter is 371 meters. If we analyse the parameter buffer size, in the ns-2 simulator the default size equals to 50 packets (the packet size is 512 bytes that is 204,800 bits), but in the OPNET® simulator the default value of this parameter is 256,000 bits.

Significant dissimilarity that is noticed between these simulators is the significance of a few key metrics, like throughput and load. When these parameters are processed with the ns-2 simulator, they are computed from the Application level aspect. More precisely, the presented load is assessed by putting the transmitted data from the application layer on the source node. On the other side, throughput is calculated by putting up the received data from the application level at a target node. The OPNET® simulator considers the parameters load and throughput on the MAC level, and that is a reason for two straight outcomes. The first outcome is that overhead is included that comes from the headers of the MAC frame, MAC control packets, and network protocol. The other consequence is that both statistics are assessed bearing in mind all the nodes that are in the network, and not just the source and destination nodes. That means that if any of the nodes is retransmitting some packets, the entire load is also increased, although the specific nodes could be intermediate nodes. In the same way, when some intermediate node is receiving some packets, the matching cumulative throughput is also incrementing. Because of these facts, there are differences in the concluding results. This problem can be solved if a statistic, like end-to-end, is assessed, which is on the Application level, similar to the ns-2 simulator.

Another important matter when comparing ns-2 with OPNET® is that error bars are contained in the outcomes from the OPNET® simulator corresponding to the average of 90% confidence interval. In the ns-2 simulator, the graphs do not illustrate error bars for the reason that they are not observable, although a confidence interval of 99% can be reached [119]. This inconsistency is shown because of a few constraints that are discovered in the Random Number Generator (RNG) of the OPNET® simulator [125] or weak points of the RNG of the ns-2 simulator [126]. Hence, from the above comparison analysis conducted between ns-2 and OPNET®, it can be concluded that the OPNET® simulator has better performances and it is closer to reality, because it utilizes approved and supported simulation models and it gives more reliable results.
Chapter 3. SIP over MANET – A Literature Review

3.8 SIP-based Applications over MANET Using Asterisk

Asterisk is an open source solution for deploying VoIP over different platforms [129]. Asterisk can help in designing a wide range of applications such as Internet Protocol Private Branch Exchange (IP PBX), VoIP gateway, and Interactive Voice Response (IVR). VoIP can be implemented using any standard signaling protocol such as SIP and H.323. Asterisk utilizes the functionalities of the SIP protocol for signaling, and it also supports protocols such as Inter Asterisk Exchange (IAX) and Skinny Call Control Protocol (SCCP). The IAX collectively puts both the features which are signaling and multimedia transport in the same protocol stack. One fundamental concept of Asterisk is the use of channels, in which various terminals can either connect to different VoIP protocol or telephony interfaces. Although Asterisk supports SIP, the inner architecture does not reflect the concept of the SIP protocol. The Asterisk architecture supports its use of the provided system at those places where RTP traffic needs to be routed through the server [95]. This is also helpful in resolving the issues like NAT and firewall. However, Asterisk can also be used as an intermediate system between incompatible SIP terminals [128]. Asterisk can also be helpful in translating one codec to another while used as media gateway. It means that while Asterisk is used as a media gateway, it is also used for connecting the SIP client to traditional telephony networks. Asterisk can be considered as a PBX, which runs under most of the operating systems such as Linux ©, Windows ©, Berkeley Software Distribution (BSD) or UNIX, and Macintosh ©. It is capable for connecting with most of the Plain Old Telephone Service (POTS) standards.

SIP signaling provides instant messaging and multimedia sessions over the Internet and network systems. The SIP protocol architecture is basically client/server architecture which uses centralized proxies such as a redirect server, proxy server, and registration server. The centralized servers are used in SIP for locating the end points but they may not exist in the case of MANET. Therefore, the SIP protocol cannot be deployed as an isolated MANET [10, 127]. In an Internet-connected MANET, the mobile nodes’ location can be found, hence SIP services can be deployed in the Internet-connected MANET environment. However, the centralized nature of MANET raises many problems when SIP services are connected to MANET. In an Internet-connected MANET, the mobile nodes or the end nodes located within the ad-hoc network can reach to other nodes in the Internet using gateway nodes for the communication. However, in case both the caller and callee mobile nodes lie in the same ad-hoc network and need to communicate using SIP, then both nodes need to communicate via a gateway. Hence it can put a limitation on its performance. Various researches have been done to find out the alternative approaches for SIP implementation. Examples of these approaches are the distributed SIP (dSIP), Proxy-based SIP, and P2PSIP. These approaches are used to provide the integration with an Internet-based MANET [10, 83, and 127]. The proxy-based SIP uses a SIP proxy/registration module within mobile nodes, and the P2PSIP uses the DHT module within mobile nodes to support SIP services.
in an Internet-based MANET [128]. Other research suggested that to enable the SIP communications in the mobile nodes, a proxy-based SIP can be used. The proxy-based SIP uses the integration of a gateway with the proxy features which proves to be more efficient [83, and 127].

3.8.1 MANET implementations in Asterisk

VoIP is a low cost solution over the traditional MANET. Thus, the system integration of VoIP with support to mobility can be a new paradigm for providing various voice communication solutions using an Ad-hoc network. The traditional way of implementing VoIP applications is infrastructure based. However, in MANET the nodes are mobile; the IP address of each node is auto-configured and can be variably changed, too. Hence, VoIP applications cannot be implemented over MANET using the traditional way. Several researchers studied solutions to implement VoIP over MANET. Asterisk can provide a working solution to MANET for VoIP support. Some of the previous researches on implementation of VoIP over wireless network using Asterisk are like emergency services, implementation of VoIP service over Wi-Fi based systems, emergency services using voice communication, and VoIP over Ad-hoc networks [130, and 131].

For VoIP applications over remote area wireless mesh networks, a suggested implementation used to provide VoIP services using Asterisk installed on various peers of the networks. All VoIP components are communicating with each other using the standard SIP protocol, whereas Asterisk is used to communicate with other peers or nodes using Inter Asterisk Exchange (IAX) protocol. However, the wireless network in these applications does not operate completely in ad-hoc mode. To implement the VoIP using Asterisk, it is suggested that each mesh network node uses its wireless links in both the modes such as in ad-hoc mode as well as in infrastructure mode [130]. Ad-hoc mode can be used by nodes to communicate with each other, whereas infrastructure mode can be useful for obtaining Internet access or to allow mobile nodes to have the network address. Ad-hoc mode also helps the devices for auto-configuration with Asterisk [130]. Further, Asterisk runs over wireless mesh nodes with some routing protocol too, installed on those nodes. Whenever a call is made, nodes perform the internal routing whereas SIP servers are used for routing for the VoIP calls [130].

To enable VoIP applications over Wi-Fi-based networks, an analyzed performance of an Asterisk-based embedded system, which is used for providing IP-based multimedia communication, had been represented by [131]. This embedded system board uses protocols such as SIP and IAX which are integrated with GSM codecs. The nodes run this Asterisk-based server over a Linux-based operating system [131]. However, in this architecture, Asterisk is used to manage the calls within the same subnet. Results concluded that for a low bit rate Wi-Fi network, the IAX protocol has good performance, whereas for a high bit rate Wi-Fi network, the SIP protocol has better performance. In applications such
as emergency warning systems, a study has provided an insight for implementing IP-based multimedia communication such as VoIP services [132]. In this scheme, the research has proposed architecture in which PSTN is connected to an Asterisk server, to interconnect the IP network with PSTN. The performance of the emergency warning system proves to be good; however, the major drawback of this architecture is that no mobile devices are included [132].

MANET can be implemented in a 4G network to provide multimedia services like VoIP. Signaling protocols such SIP and H.323 can be used for supporting IP-based multimedia communication. SIP is basically a centralized-based architecture, whereas MANET is a decentralized architecture. Hence SIP cannot deploy directly in MANET. Various methods have been proposed previously to deploy VoIP services over MANET. However, these methods have their own drawbacks such as the effect of SIP endpoint mobility is not addressed. To enable VoIP services over MANET, architecture has been proposed to implement SIP in which end point users inside a MANET are connected to an external SIP end point [133]. The external SIP endpoint is connected through the Internet. To interconnect different MANETs, a gateway has been used. The Asterisk is used to enable one of the mobile nodes to work as a voice communication server. The Asterisk server is used for routing the VoIP calls too. The results conclude that the performance of voice calls is improved with this architecture and the overall delay is dependent on the node movement velocity [133]. Thus, Asterisk can be a better solution for providing VoIP services in the MANET. In the case of complete ad-hoc mode without Internet connectivity, it can be difficult to provide a complete solution with better quality of service. However, Asterisk can be used within the mobile nodes to establish communication within the mobile nodes. Using an Asterisk server and the SIP signaling protocol, voice calls can be managed within an ad-hoc network. Therefore, it can be concluded that Asterisk can support MANET for VoIP applications.

3.9 Summary

The literature survey has shown that enhancing the SIP signaling performance is the most efficient method that can be considered when compared with other solutions. In addition, combining the dynamic adjustment methods for SIP signaling and MANET routing parameters can improve the performance level with efficient and simple implementations. In addition, this chapter considered the possible performance methods for SIP signaling over MANET. The presented research works in the field of SIP signaling over MANET are lacking in the inclusion of the terminal mobility using SIP. That is why in [96] terminal mobility as well as low call setup delay and fast network operations are considered when proposing the Easy Disaster Communication (EasyDC) framework. However, other mobility issues had raised in some of the proposed approach, for instance the session mobility which was not address in the approaches design and implementations. Furthermore, some research works represented every node of MANET is having the role of register/proxy of SIP which will overload the nodes with the routing
traffic of the SIP processes [83, 88, and 93]. Hence, a SIP user agent within MANET is flooding the SIP REGISTER messages into the entire network in order to register its presence. However, the SIP systems that using multiple SIP servers over MANET has better performance that avoids the single point of failure when compared with the centralised architecture of SIP over MANET. Other approaches considered that the callers' nodes in MANET have the SIP functionalities and act exactly as a proxy or registrar server. In addition, the user location in the SIP system could be determined dynamically within the MANET system. Different from this kind of broadcast-supported register/proxy of SIP, is the group-supported register/proxy of SIP where the role of registers/proxies takes only the cluster heads as in [90]. However, these kinds of mechanisms are utilizing the flooding requests of SIP between the nodes. The results shown a very high overhead for the network that causing challenges with the adaptability level in the network systems that make these approaches not appropriate for MANET based implementations.

In general, the essential problems that the SIP signaling over MANET are facing the lookup time of the SIP agents, the mobility assistance of the terminal, and the interoperability among the Internet and the users of the SIP over MANET. Using a reliable performance metrics to enhance the SIP signaling performance for SIP-based applications over different platforms is still an open research issues. Although the performance metrics need to consider the best and worst cases during the dynamic implementations of the SIP signaling system over MANET. Furthermore, the security issues for SIP-based applications are still open and need to investigate the implementations hazards over MANET based applications such as Denial of Service (DoS), Man in the Middle, and Sniffing attacks.
Chapter 4

4 Performance Metrics for SIP-based VoIP Applications over MANET

The evaluation studies need to investigate a determined performance metrics to understand and evaluate the examined scenarios. SIP-based Voice over IP (VoIP) applications over MANET have two main performance categories related to the Quality of Service (QoS). The main performance metrics that are considered for the evaluation processes in this research are the SIP end-to-end Performance metrics as defined by the RFC 6076 [1]. The main performance metrics are related to the registration, the call setup, and the call termination processes. In this research study, the SIP performance metrics are based on a single SIP proxy server. For voice data, the QoS evaluation is based on two methods: the Objective method and the Subjective method. The Objective method considers the traffic throughput, end-to-end delays, packet loss, and jitter, while the subjective method considers the Mean Opinion Score (MOS), which is mostly related to the end users’ experience during voice calls. The voice codec represents the compression system for voice data that is used during the calling session and affects the construction of the voice traffic volume. The other related metrics for SIP-based VoIP over MANET are the routing performance metrics. The routing mechanism of the considered MANET routing protocol is responsible for the average bandwidth consumed for routing data, the average routing traffic sent and the average routing received. In this chapter, the main performance metrics for SIP signaling and voice data are presented to identify the evaluation methods for SIP-based VoIP applications over MANET platforms. These metrics will be considered during the evaluation stages of the research efforts in chapter 6, 7, and 8.

4.1 Voice Codecs

The voice applications used to compress the analog voice signals into digital signals uses different types of voice codecs. Voice codecs are audio data compression algorithms for use for different types of voice based applications [134]. This basic stage happens on the caller’s side to make the voice data transferable over the PSTN or the Internet for far distances. For wireless based VoIP applications, the voice compression is critical, as the voice signal needs to be compressed as much as possible to fit with the loose nature of wireless communications. This compression effectively reduces the bandwidth.
consumption and transmission power over wireless network systems. In addition, the voice compression systems create smaller packets, which reduce the packet loss ratio, and end-to-end delays that support the voice quality as the number of received voice packets relatively increase [135]. The present researcher studied the SIP-based VoIP applications over MANET using four common voice codecs:

- G.723.1 [136] is one of the most common voice codecs for VoIP applications that operates at 5.3 Kbit/s or 6.3 Kbit/s and is officially known as Dual Rate Speech Codec for Multimedia Communications Transmitting at 5.3 and 6.3 Kbit/s.
- G.729 [137] is another common voice codec for VoIP applications because of its low bandwidth requirements. It operates at 8 Kbit/s and formally known as Coding of Speech at 8 Kbit/s Using Conjugate-Structure Algebraic Code-Excited Linear Prediction Speech Coding (CS-ACELP).
- The GSM voice codec is developed by the European Telecommunication Standards Institute (ETSI). It is widely used in mobile telecommunications as it operates at 13 Kbit/s and has good performance over CPU demands that support the nodes’ mobility nature [138].
- G.728 [139] is a speech coding algorithm which operates at 16 Kbit/s and is described by the International Telecommunication Union – Telecommunication Standardization (ITU-T) as the Coding of Speech at 16 Kbit/s Using Low-Delay Code-Excited Linear Prediction (LD-CELP).

In this research, the GSM voice codec is considered as the main voice codec in the simulation results investigation for research study scenarios. The main outcomes of the evaluation study in this research are meeting with the implementation results for other voice codecs such as G.723.1, G.729, and G.728. The encapsulated voice data of VoIP applications are transferring over a special multimedia transport protocol known as the Real Time Protocol (RTP), which runs over the User Datagram Protocol (UDP) [135]. As RTP is depending on an unreliable transferring process, the voice packets have a losing ratio which depends on the network health and connectivity status between source and destination. In general, voice data has the priority over different types of network traffic to support the QoS of voice application in different network systems. For SIP-based VoIP applications, the call initiation stage activates the media data transfer process for RTP directly between the caller and callee using one of the supported voice codecs. The payload of voice packets and the headers are shown in Figure 4-1, where IPv4 packets have an overhead of 20 Bytes and IPv6 packets have an overhead of 40 Bytes. This difference between IPv4 and IPv6 packet size influences the traffic transferring and the QoS for voice applications.

<table>
<thead>
<tr>
<th>IPv4/IPv6</th>
<th>UDP</th>
<th>RTP</th>
<th>Voice Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>20/40</td>
<td>8</td>
<td>12</td>
<td>Bytes</td>
</tr>
</tbody>
</table>

*Figure 4-1: IPv4/IPv6 Voice Packet [134]*
4.2 SIP Signaling Performance Metrics

The signaling performance of SIP plays an important role in affecting the overall Quality of Service (QoS) and Quality of Experience (QoE) in next generation networks. SIP signaling delays are relating to the connectivity status between the SIP call parties, which are the Caller, Callee, and SIP proxies. These delays mainly happen during the Caller’s registration process, call initiation, call termination, and/or call management [140]. In addition, SIP signaling is affected by the behaviour of the Transportation Protocol (TCP or UDP) that SIP relies on during the different connectivity processes of SIP calls. Many standards have been proposed for the performance evaluation of telephony signaling protocols, however none of these metrics were used to address the SIP signaling performance until the IETF proposed the RFC 6076, the SIP end-to-end Performance Metrics [1]. However, there are no numerical values or benchmark objectives for the RFC 6076 SIP performance metrics.

4.2.1 SIP Signaling Performance Metrics in the Literature

As many approaches were introduced to evaluate and standardize the telephony signaling protocols, none of these approaches studied the SIP signaling performance until the appearance of RFC 6076 in 2011 [1]. The concerns about SIP signaling performance began with assumptions and investigations for a set of related performance metrics for SIP signaling over different platforms. The ITU-Recommendations proposed two SIP performance metrics for circuit-switched network systems, the Post Selection Delay (PSD) to measure the call setup times for SIP calls, and the Network Effectiveness Ratio (NER) to detect the ability of the User Agent or the proxy server during the session establishment of SIP calls [141].

Beside the standardization efforts, a number of studies tried to identify different performance metrics for SIP signaling using different related network elements for SIP sessions. One of the early studies [142] proposed to examine the main processing entities for SIP network elements and identify a number of SIP performance metrics in 3GPP IMS systems. The study provided some experimental results related to the processing efforts for SIP sessions regarding CPU usage and memory consumption with an analysis of SIP servers’ performance during SIP sessions. In [143], an alarm message was proposed for the sent INVITE requests for SIP calls known as Session Setup Delay (SSD). In the same way, the Call Setup Delay (CSD) performance metric for IMS supports end-to-end QoS over fixed multimedia applications and converged mobile services [144]. The CSD depends on three main metrics: the Post Dialing Delay (PDD), Answer Signal Delay (ASD) and Call Release Delay (CRD). The PDD is the time interval between the call initiation by the caller and the time the caller hears the callee’s terminal ringing while the time interval between when the callee picks up the line and the caller receives the callee’s response is known as ASD. The CRD is the time interval between call disconnect and initiating/receiving a new call by the same calling party. The values of these metrics are affected by the
number of hops and the distance between the call parties. In [145], a performance evaluation concept was proposed for a single SIP transaction known as Quality of Signaling (QoSg) that consists of the following metrics: User to User Delay (UUD), Processing Delay (PD) and Response Delay (RpD). The UUD is the time interval between sending an INVITE request until it is received by its destination. The PD is the time needed by the User Agent Server (UAS) to process a request and send its response message, while the RpD is the time interval between sending a request by a User Agent Client (UAC) and receiving its response message.

In [146], two performance metrics were proposed, the Session Negotiation Time (SNT) and the Session Re-Negotiation Time (SRNT). The SNT is used in the negotiation process of a calling session to measure the required time interval between a sent SIP INVITE request until an applicable 200 OK response is received. The SRNT is used to describe the negotiation process initiated by sending an INVITE request or by a SIP update request that carries updated information regarding SIP sessions. A few studies tried to benchmark SIP performance metrics for IMS core components by identifying different metrics such as Registration Time, Initial Ringing Time, Initial Response Time, Disconnect Request Time, and other system estimation metrics like CPU and power consumption [147, 148]. All the reviewed studies considered non-standardized performance metrics for SIP-based applications. They used different metrics with limited levels of usability that make it difficult to be applied over different platforms and network systems. In addition, the limitations of applying these performance metrics provided non-comparable and incompatible results for the performance metrics.

### 4.2.2 RFC 6076: SIP end-to-end Performance Metrics

The Internet Engineering Task Force (IETF) adopted standardized end-to-end Performance Metrics for a basic SIP-based signaling system as defined in RFC 6076 [1]. These metrics provide Key Performance Indicators (KPIs) and Service Level Agreement (SLA) indicators to support the SIP-based telephony systems and enhance the network utilization. The RFC 6076 defined the following SIP end-to-end Performance Metrics:

1. **Registration Request Delay (RRD)**

   The RRD is used to determine the response delay time for the User Agent REGISTER request. The RRD helps to measure and analyze the successful Registration requests and at the originating User Agent as represented in Figure 4-2. The output values for RRD should be in milliseconds (ms). This metric is calculated using equation (4.1):

\[
RRD = \text{Time of Final Response} - \text{Time of REGISTER Request}
\]  

(4.1)
The RRD is calculated only for successful registrations. In addition, when the load of SIP calls increases in the network systems, the value of RRD also increases. When there is low load in the network system the value of RRD will be in the range of the lowest values.

2. Ineffective Registration Attempts (IRAs)

The IRA is a metric that detects the failures or the impairments that cause the inability of the registrar to receive the User Agent REGISTER request. The IRA is a percentage parameter of the total number of unsuccessful registrations of the REGISTER requests and is calculated using the following equation (4.2):

\[
IRA \ [%] = \frac{Number \ of \ IRAs}{Total \ number \ of \ REGISTER \ Requests} \times 100
\]  

(4.2)

3. Session Request Delay (SRD)

The SRD is a metric designed to detect the faults or defects that cause delays in responding to INVITE requests. SRD considers both successful and unsuccessful session setup requests where the duration for success and failure responses are varied. A simple representation for SRD related to the SIP flows is shown in Figure 4-2. The SRD is calculated using the following equation (4.3):

\[
SRD = Time \ of \ Status \ Indicative \ Response \ - \ Time \ of \ INVITE
\]  

(4.3)

4. Session Disconnect Delay (SDD)

The SDD is designed to calculate the time interval between the time that the session completion message (BYE) is sent and the last subsequent acknowledgement of the session completion response received (2xx). The SDD is used to detect the failures or impairments that cause the delays for a session to end. The SDD measures both successful and failed session disconnects where the output values are in milliseconds (ms). The SDD is calculated using the following equation (4.4):

\[
SDD = Time \ of \ (2xx) \ or \ Timeout \ - \ Time \ of \ Completion \ Message \ (BYE)
\]  

(4.4)

5. Session Duration Time (SDT)

The SDT is designed to define the time interval between the receipt of the (200 OK) response to the INVITE request and the receipt of the last associated (BYE) request. The SDT is measured at the originating User Agent and the terminating User Agent. Therefore, the SDT
value measured at the originating User Agent is different from the value that is measured at the terminating User Agent. The SDT is calculated using the following equation (4.5):

\[ SDT = Time \ of \ BYE \ or \ Timeout - Time \ of \ (200 \ OK) \ response \ to \ INVITE \]  

(4.5)

### 6. Session Establishment Ratio (SER)

The SER is considered to define the ability of terminating the User Agent or proxy server during the session establishment process. This metric is used to calculate the ratio of the INVITE requests that result in the (200 OK) responses and the difference of the total number of the INVITE requests and INVITE requests results from the 3xx responses. The equation of this metric is as following (4.6):

\[
A = \frac{\text{Total Number of INVITE Requests}}{\text{Associated (200 OK)}} \\
B = \text{Total Number of INVITE Requests} - \frac{\text{Number of INVITE Requests}}{\text{3XX Responses}} \\
SER [\%] = \frac{A}{B} \times 100
\]

(4.6)

### 7. Session Establishment Effectiveness Ratio (SEER)

The SEER metric is complementary to the SER, however it is used to exclude the potential effects of an individual user of the target User Agent from the metric. The SEER is defined as the ratio of INVITE requests resulting in a (200 OK) response and the INVITE requests resulting in a 480, 486, 600 or 603 response, to the total number of the initiated INVITE requests less the INVITE requests resulting in a 3xx response. The SEER is calculated using the following equation (4.7):

\[
C = \frac{\text{Total Number of INVITE Requests}}{\text{Associated (200,480,486,600,or 603)}} \\
SEER [\%] = \frac{C}{B} \times 100
\]

(4.7)

### 8. Ineffective Session Attempts (ISAs)

The ISA is a metric that is used when SIP entities are damaged or overloaded. The ISA is calculated as a percentage of the total session setup requests using the following equation (4.8):

\[
ISA [\%] = \frac{\text{Number of ISAs}}{\text{Total number of Session Requests}} \times 100
\]

(4.8)
9. Session Completion Ratio (SCR)

The SCR is used to represent the percentage of the successfully completed sessions over the total number of sessions. This metric is similar to the Call Completion Ratio (CCR) in the telephony applications of SIP. The SCR is calculated using following equation where the output indicates the percentage of successfully completed sessions (4.9):

\[
SCR\ [%] = \frac{\text{Number of Successfully completed Sessions}}{\text{Total number of Session Requests}} \times 100 \quad (4.9)
\]

In this research study, three RFC 6076 performance metrics will be considered during the evaluation study in Chapters 6 and 7, and with the proposed approaches in Chapters 8, 9 and 10. These metrics are the Registration Request Delay (RRD), the Session Request Delay (SRD), and the Session Disconnect Delay (SDD).

![Figure 4-2: SIP Performance Metrics over SIP call stages [1]](image-url)
Chapter 4. Performance Metrics for SIP-based VoIP Applications over MANET

4.2.3 RFC 6076 Related Research

A limited number of research efforts studied the RFC 6076 SIP performance metrics for VoIP applications. The early efforts to identify a specific set of performance metrics to evaluate the SIP signaling performance was proposed in [149]. The research measured the overall performance of a SIP server by defining the parameters and methodology for benchmarking the SIP-based VoIP infrastructure. This method was designed to investigate the performance of the SIP server which is acting as a Back-to-Back User Agent (B2BUA) for VoIP applications. The study proposed and examined two parameters, Registration Request Delay (RRD) and Session Request Delay (SRD). The study was updated in [150] and proposed new methods of stress testing for the SIP management and control approaches that generated high SIP-based traffic. The efforts had extended later to study the B2BUA-based SIP server and proposed a stress test of SIP signaling and benchmarking of the performance metrics based on the behavioural analysis of the SIP environment [151]. The SIP registration burst load for B2BUA-based SIP servers was examined in [152]. The research studied the RRD performance metric for a B2BUA-based SIP server where the effectiveness of handling the burst loads for SIP Registration requests was investigated in an Asterisk PBX system. In [153], the SIP performance metrics were used in the methodology of the performance tests for calls and registrations for the Interactive Video and Audio System (IVAS) which had been developed for the INDECT project which aims to develop the tools for enhancing the security of citizens and protecting the confidentiality of recorded and stored information. The study analysed the central elements of the IVAS system in VoIP PBX Asterisk by investigating the call registrations and initiations at the end users’ devices.

In [154], the RRD and SRD SIP performance metrics of the RFC 6076 were employed to measure and evaluate the SIP signaling performance on B2BUA-based SIP signalling system using an Asterisk based implementations. The research used open-source applications such as jQuery, Python, JSON and the cornerstone SIP generator SIPp. The results show that the SIP performance metrics could provide accurate values in busy network systems. The values collected for the SIP performance metrics in this research are limited to the RRD and SRD values, and could applied with simple SIP-based applications. However, no other performance metrics had investigated in this research and the results are limited to the LAN systems only. In [155], a simulation-based optimization algorithm of SIP signaling procedures in IMS is presented to improve the SIP signaling performance by assigning a high priority value for SIP messages to reduce the network congestion and improve the overall QoS. The simulation efforts were conducted with ns-2 and the results were analysed in terms of the RRD, SRD, and SDD. The results have not compared with other related published results because the measurements used are implemented under different conditions and environments. In [156], the SIP performance metrics were used to evaluate the implementation of the Rich Communication Suite (RCS) services in IMS platforms. The study employed the RRD, SRD, SDD, and SDT as timing parameters to optimise the SIP signaling services for the proposed system during its implementation. In [157], the SIP performance metrics were
used to measure and compare the overhead of using Transport Layer Security (TLS) compared with TCP and UDP for secure SIP implementations. The experimental results show a noticeable decrease in the performance of VoIP services based on the RRD and SRD when using SIP over a TLS based signaling system. The RFC 6076 performance metrics had been used in [158] as performance metrics for context based charging in a 3GPP network environment. The metrics were used to measure the response times of the messages related to the resource reservation and online charging procedures to support the QoS. The RRD was used in [159] as a performance metric to evaluate the SIP Retransmission timers in HSPA 3G Networks for both the SIP server and the User Agents. The comparison with other work is still a challenging issue as no unified approach is provided to support the achieved studies.

4.2.4 User Agent Registration

The first stage for SIP-based VoIP calls is the Client’s registration with the SIP server for non Peer-to-Peer SIP applications as shown in Figure 4-2. The SIP server is applying the Back-to-Back User Agent (B2BUA) for the exchanged SIP signaling system. The registration process is initiated whenever the User Agent joins the network system or updates its status. The SIP calls cannot setup if the other call parties are not registered with the SIP server. Whenever a Caller wants to call an unregistered UA, the UAS will fail the call setup process. A few number of research studies investigated the performance of SIP registration processes while other research focused on the related security issues. An optimization approach had been proposed for the SIP registration process over the mobile environment where the network access and IP addresses are changing frequently [160]. The SIP clients need to proactively register with the SIP server and update its current parameters using identified dynamic registration intervals relating to the probability of links’ disconnection to reduce the wasted resources and overloaded links. This approach is based on using an extended version of the Kalman filter to indicate the dynamic registration version. The simulation results show an enhanced performance for SIP registration processes of using the proposed registration method compared with the constant update intervals method of the registration processes. In [161] and [162], the registration process is measured using the RRD performance metric of RFC 6076 for a simple scenario using a SIP testing platform designed in the VSB – Technical University of Ostrava. The research is in the context of benchmarking SIP performance metrics to improve the performance evaluation methodology of a SIP signaling system by using an Asterisk testing platform. The study investigated the performance of the SIP server during the registration process by collecting the RRD performance metrics for the clients. The examination used a separate registration stress over an Asterisk PBX server during the registration process for a number of simultaneous registrations. The test results show that the SIP server has the main effectiveness of handling the burst loads for the registration requests.
The delays in the call registrations of the SIP clients consistently affect the call setup processes based on the nature of the SIP signaling system for the SIP processes and the network nature. As a simple scenario, if the callers delayed in the registration processes with the SIP server, any scheduled or intended call initiations will be delayed as well. This will also affect the general performance of related SIP services that provided by the SIP server [161, and 162]. The registration process investigated within different proposed approaches to enhance the SIP signaling performance over MANET platforms. In [87], a design of Ad hoc VoIP using an embedded p-SIP server provided an enhanced algorithm for the registration process over the SIP server. This approach reduced the registration delays by handling the route discovery mechanism over clients and SIP server for the registration and call setup processes. The study succeeded with reduction of the registration delays by improving a p-SIP Register algorithm to reduce the discovery time and the hops number between the call entities. In [163], the design of the Proactive Route Optimization (PRO) in SIP mobility considered the registration processes to maintain the client’s location information to reduce the call setup time. In [97] and [164], developed designs for the Registration process in MANET platform employed the client location and the relaying priority to enhance the SIP server performance.

4.2.5 Call Setup Delay

The Call Setup Delay is defined as the elapsed time between sending the initial INVITE request and receiving the Acknowledgment (ACK) response message by the Callee node [165]. The call setup time for each call is the difference between the absolute times of the INVITE and corresponding ACK message that belong to the same call. The call identifier headers are used in addition to the To and From headers, to categorise each SIP session as the SIP Method header can identify the type of the message, e.g., INVITE, and ACK. The Call Setup Delay is also known as the Post Dial Delay (PDD) and is considered as one of the required parameters for QoE evaluation. A number of studies have evaluated the call setup time for SIP-based network systems. The SIP call setup time was analysed for a reliability model over a SIP server in [166].

In [167], the SIP session setup delay was investigated for correlated fading channels for 3G wireless networks. The SIP call setup time and the Real Time Protocol (RTP) were evaluated for one-way delay using IPv4 and IPv6 for basic network systems [168]. In [169], the SIP-based IMS establishment sessions for WiMax-3G networks had considered. The call setup latency for SIP-based VoIP over wireless local area networks was analysed using an ns-2 simulator in [170]. In [171], a call setup model is presented for SIP-based stateless calls for next generation networks. The model was designed based on the queuing models and the call setup delays for single domain and multiple domain scenarios. In [172], an evaluation was made of SIP-based call setup time and other QoS parameters of VoIP over IPv4 and IPv6 satellite environment based on the unreliable User Datagram Protocol (UDP) at the transport protocol layer. The call setup time is increased in the satellite networks environment due to
the long propagation delay. The research recommended that an improvement for SIP signaling should be applied to reduce the number of SIP messages and call setup time over satellites. In [173] the call setup delay had been surveyed for a SIP-based VoIP LANs Cisco environment. The study concluded that the average SIP call setup delay for various network loads is in the range of 200 to 300 milliseconds. On the other hand, the simulation and analytical results in [166] proved that the call setup latency is sensitive to the number of mobile nodes in a WLAN.

The previous approaches investigated and enhanced the SIP setup time for VoIP applications. The results could be applied or examined over different network systems and scenarios, however, these approaches could not considered as benchmark results for SIP-based call setup processes because it is not dependent on standard performance metrics for SIP signaling systems. The ITU-T recommendations defined considerable constraints for SIP Calls Setup Delay where the mean value is equal to 800 ms and the maximum value is equal to 1500 ms [174]. In [165], the SIP call setup delay had been investigated based on the values in [174] by using different QoS and QoE factors for the SIP signaling evaluation over IPv4 and IPv6 IMS networks. In addition, it provided some theoretical analysis of SIP setup delays and provided results that show that the values of Call Setup Delays when using IPv6 are greater than those using IPv4 for radio access links with a bit rate less than 128 kbps. In [175], the Call Setup Delay is represented by using the RFC 6076 performance metrics. The research tried to consider both the RRD and SRD values to evaluate the Call Setup Delays for SIP-based VoIP Services in the context of the high loss, high latency, and bandwidth constrained Airborne Network environment using the OPNET® simulation tool. In [176], a model and algorithm for end-to-end call setup time calculations had been proposed for SIP-based VoIP applications. This model was designed to check all possible situations during the call setup processes that affect the general performance of the SIP call when dealing with highly loaded and congested networks. The study presented numerical calculations and simulation comparisons for the presented approaches using a cumulative distribution function for a trapezoid model. Furthermore, the study had shown that the probability of successful call setup in networks with a large number of nodes is very time-consuming. However, with low traffic networks, the SIP retransmission values and the call setup time depend on transmission times.

4.2.6 Call Termination Delay

The call termination represents the closing signal for SIP call sessions between the caller and the callee. The call termination process is initiated by one of the callers to end the call. In addition, it happens when the voice QoS begins to drop or when the connectivity is lost between both ends. The long delay for the call termination process could hold the caller’s status as busy for a longer time which affects its availability in the network system as other callers may be trying to contact them through the SIP server. A limited number of research efforts studied the performance of termination processes within the
general investigation of the SIP system. The impact of end users’ response delays on the SIP server was examined in [177]. The study provided an analytical performance evaluation for an open source SIP Proxy server by using the proposed SIP Performer testing tool with a central test host and multiple distributed test agents to simulate different traffic models including the users’ response delays. The SIP Performer considered the call termination delays in its testing platform. The study proposed an enhanced termination mechanism for SIP calls within the overall proposed SIP signaling system. However, this approach has not been evaluated within different platforms where the proposed design is tight to a simple scenario that consists of two nodes and a SIP server. A SIP-based QoS management system was proposed to provide a consistent QoS control system for multimedia applications over WLAN by considering the users’ priorities and the providers’ objectives [178]. This approach employed both data and control planes to detect the congestion events of SIP processes and apply suitable actions to overcome the system deficiency within bandwidth and end-to-end delays for SIP-based multimedia applications. Furthermore, the approach architecture considered the session termination process to enhance the SIP applications QoS by sending BYE messages whenever delays or congestion was detected. The simulation efforts of the study show that the overall delays of the session termination process should not exceed one second (1s) to provide a good level of QoS.

4.3 Voice Quality

Voice Performance evaluation for VoIP applications has two main measurement methods: Objective and Subjective methods [179]. The Objective methods are mathematical-based methods that are used to measure different physical quantities of voice traffic such as Packet Loss, Delays, Jitter, amount of Sent and Received Traffic [180]. In general, measuring VoIP applications is more objective and based on the performance calculations of the traffic transferred on the IP networks. On the other hand, the Subjective methods are human-based methods that depend on the average user’s perception of the voice quality. It investigates the callers about the quality of the voice through simple questionnaires with limited classified choices.

4.3.1 Objective Methods

The objective methods depend on two mathematical testing techniques for voice quality: the intrusive or the non-intrusive techniques. The intrusive technique is used to inject a testing voice signal into the network system without the existence of the application users. This technique is mainly used during the system development stages when the users are offline or not beginning to use the provided services. On the other hand, non-intrusive techniques are used to examine live real-time traffic without the need to determine a reference signal. It depends on the network impairment parameters such as delays, packet loss ratio, and jitter. In general, the non-intrusive techniques provide larger numbers of live and real-
time tests, however it has less accuracy compared with the intrusive techniques. In this research effort, the non-intrusive technique is used with the objective methods to measure the end-to-end or Mouth-To-Ear voice quality. The main performance metrics in this investigation are summarized in Figure 4-3 [181], where each metric has a direct influence on the voice transmission for VoIP applications [140].

4.3.1.1 Throughput

The throughput is identified as the maximum number of bytes that are received in the receiver side out of the total sent voice traffic during an interval of time. For wireless networks, IEEE 802.11 standards have different data rates for different wireless applications. In this research work, the IEEE 802.11n standard has been used with a bit rate of 13 Mbps. In addition, the required bandwidth for VoIP calls is mainly determined by the voice codec system used. Limited bandwidth could cause traffic congestions and packet delays that affect the general performance of the voice data. The complexity of the voice codec system affects the coding speed and the required bandwidth. As long as the coding complexity increased, the coding speed reduced and the required bandwidth increased. The amount of the consumed IP bandwidth for a voice call using one of the identified voice codecs in the previous section are computed by the following equations [182]:

\[ \text{Total packet size} = \text{IP/UDP/RTP header} + \text{Voice payload size} \] (4.10)

Number of Packets needed per second to deliver the codec rate:

\[ \text{PPS} = \frac{\text{Codec bit rate}}{\text{Voice payload size}} \] (4.11)

\[ \text{Bandwidth} = \text{Total packet size} \times \text{PPS} \] (4.12)

Where PPS is the number of packets required per second to deliver the amount of data for the voice codec.
Table 4-1 represents the voice codecs that were used in this research study, the payload size, and the Mean Opinion Score (MOS) for each codec [183]. The MOS parameter will be explained in detail in Section 4.3.2.1. The bandwidth calculations for a voice call are computed using equations (4.10), (4.11), and (4.12). The header size for IPv4 and IPv6 voice packets are 20 Bytes and 40 Bytes, respectively, as represented in Figure 4-1. The table also shows the percentage of difference between IPv4 and IPv6 for the maximum required bandwidth for a single voice call. The Full Rate voice call is a call that uses the full voice rate of the voice codec to improve the performance of the voice codec and it is mostly used to examine the maximum possible bandwidth of the voice channel [183]. The Full Rate call is used to show the difference in the bandwidth consumptions. IPv6 consumes more bandwidth compared with IPv4 over all voice codecs with different ratios, as calculated in Table 4-1.

Table 4-1: Voice codecs and bandwidth consumption; adopted from [182] and [183]

<table>
<thead>
<tr>
<th>Codec Information</th>
<th>Bandwidth Calculations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Codecs</td>
<td>Mean Opinion Score (MOS)</td>
</tr>
<tr>
<td>G.723.1</td>
<td>5.3 Kbps 678.4 Bps 3.80</td>
</tr>
<tr>
<td>G.729</td>
<td>8 Kbps 1024 Bps 3.92</td>
</tr>
<tr>
<td>GSM</td>
<td>13 Kbps 1664 Bps 3.50</td>
</tr>
<tr>
<td>G.728</td>
<td>16 Kbps 2048 Bps 3.61</td>
</tr>
</tbody>
</table>

In reality, the transmission throughput is much below the maximum defined bit rate over different wireless standards because of the overhead caused by the network protocols. For example, the maximum throughput achieved in WLANs is between 50% and 70% of the maximum transmission rate. This is low compared with the maximum throughput achieved in Ethernet, which is in between 80% and 90% of the transmission rate [184]. The throughput investigation is used to estimate the VoIP performance; when low throughput of voice packets is received that does not meet the expected amount of voice traffic, a concern about the voice performance must be raised [141]. Low throughput indicates
that the traffic has problems in the network connection such as a high percentage of traffic congestion. Another metric is the number of the reported dropped packets on the receiver side, which indicates the buffer overflow or problems with the network signaling system.

### 4.3.1.2 Delays

The voice delay is defined as the time taken for a person communicating with another person to speak a word and for it to be heard at the other end. As represented in Figure 4-3, Voice data delays in VoIP applications are accumulative delays from several parameters, algorithms, and transferring processes [185]. The delays can be classified into: delays at source, network delays, and delays at receiver. The voice algorithmic delays result from voice compressing and decompressing at the source and destination from analog voice signals into digital signals and vice versa. The delays at the source and destination depend on the complexity and speed of the codec system used [183]. In addition, the length of the voice frames and its headers count in the total delays for VoIP applications during the packetization and de-packetization processes. Furthermore, in wireless network systems, there are extra delay factors for VoIP applications compared with wired network systems because of its connectivity nature. In MANET, the delays vary as VoIP data transportation is affected by the features of Mobile Ad hoc nodes such as the hops numbers between source and destination, Ad hoc routing protocol, node movement, mobility model, and traffic congestion. In summary, the total delays for voice data transport from source to destination are considered as a One-way delay. The One-way delay is the time difference for the same packet at source and destination [186]. VoIP is a real-time application where RTP transmits the Voice packets. The delays of voice data packets should not exceed the acceptable level of One-way delays as identified by the ITU-T recommendations and shown in Table 4-2 [185, 186].

<table>
<thead>
<tr>
<th>ITU-T Recommendations G.114 for one-way voice data delays</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Under 150 ms</td>
<td>Acceptable</td>
</tr>
<tr>
<td>From 150 ms to 400 ms</td>
<td>Acceptable with limitations</td>
</tr>
<tr>
<td>Over 400 ms</td>
<td>Unacceptable</td>
</tr>
</tbody>
</table>

### 4.3.1.3 Jitter

The successful arrival of sequential voice packets to the destination with different time delays is known as jitter. As voice packets transmit by RTP, it identifies the voice stream by using a unique Synchronization Source identifier (SSRC), port numbers, sequence numbers and timestamps. The variations in delay happen because of the different delays in the sender side or in the network [135]. The buffer in the receiver side used to overcome part of the jitter problem by reordering the received packets depending on the RTP timestamps and then it plays out the reordered received packets. When the buffer overflows because of the jitter, any more received packets will be dropped. Because of that,
jitter could seriously affect the quality of voice as it is involved in the overall delays and packet drops of voice traffic. The jitter can be estimated using the equation below (4.13), where \( R \) represents the arrival time of a packet, \( S \) represents the RTP timestamp, and \( D(x, y) \) represents the arrival difference between two packets, packet \( x \) and packet \( y \) [187]. The acceptable jitter for VoIP applications is 50 ms [188].

\[
D(x, y) = (R_y - R_x) - (S_y - S_x) = (R_y - S_y) - (R_x - S_x)
\]  

\[ (4.13) \]

**4.3.1.4 Packet Loss**

The Packet loss for voice data is identified as when one or more voice packets fail to reach its destination which degrades the voice quality at the receiver side. Voice packets are carried on UDP, which is an unreliable transport layer protocol where packet loss could happen because of the traffic congestion, signal interference, or signal noise in the network systems. It also happens at the receiver side because of the buffer delay or the buffer overflow. In addition, delayed voice packets that arrive to the receiver side behind the scheduled voice playtime will be discarded [189]. The probability of a packet being lost depends on the network and receiver status. The Packet loss percentage for mouth-to-ear voice traffic can be calculated by equation (4.14) below. As the percentage of the packet loss increases, the voice quality on the receiver side decreases. In VoIP calls, up to 10% of packet loss is acceptable [185]. There are different ways to reduce the percentage of packet loss for voice traffic such as sending redundant phases of voice packets and using interleaving packets [140].

\[
Packet \ Loss = \left( \frac{(Packet \ Sent - Packets \ Received)}{(Packets \ Sent)} \right) \times 100
\]  

\[ (4.14) \]

**4.3.1.5 Other Related Performance Metrics for Objective Methods**

The packet latency that happens when it travels using a longer route until reaching its destination can cause the packet to be dropped because of its late arrival. The packet latency can be used as a performance metric for VoIP traffic. It could be avoided by using a shorter and faster route between the sender and the receiver. In addition, the out-of-order packet arrival is a performance metric which indicates that the received voice packets are using different routes. The out-of-order packet arrival causes the buffer to overflow, and packets drop. This could be resolved by using other supportive protocols for RTP that enhance the routing performance for voice packets [180]. Another QoS metric for VoIP applications is voice echo. Voice echo happens when the caller at the sender side hears his voice after he speaks where the callee does not notice this echo. Echo results from the reflection of the voice traffic that is sent back to the caller as an acoustic coupling between the phone’s microphone and speaker. In addition, echo results from the voice packets’ delays and becomes noticeable when its round trip delay becomes more than 50 ms [179]. Echo cancellers can be used when the round trip time for VoIP applications exceeds 100 ms to enhance the performance of the VoIP.
4.3.2 Subjective Methods

The Subjective method is a human-based approach to evaluate the voice performance. For SIP-based VoIP, the subjective method used to evaluate the VoIP applications performance depends on the average user’s perception about the voice quality.

4.3.2.1 Mean Opinion Score (MOS)

Most of the telephony network systems have been using MOS as a subjective performance measurement tool for a long time. It is used to evaluate the voice quality by obtaining the human listener’s opinion regarding the quality of the voice heard during a call session. The ITU-T Recommendation P.800 introduced an MOS subjective determination of voice quality based on a user’s perception using a ranged measurement scale from 1 to 5, where 1 is the lowest perceived voice quality, and 5 is the highest perceived voice quality [190]. ITU-T proposed different objective testing methods using MOS. In this research study, the E-model was used to investigate the MOS values during the simulation stages as becomes the most widely used tool for objective evaluation of speech quality. The E-model had been introduced by ITU-T Recommendation G.107 and improved by the European Telecommunications Standards Institute (ETSI), and the Telecommunications, and Industry Association (TIA). This model combines different impairment factors using an analytical equation to predict the voice QoS [191]. It estimates the R value which represents the voice quality in the range from 0 to 100 as shown in equation (4.15):

\[
R = R_0 - I_s - I_d - I_e + A
\]

Where: \( R_0 \) represents the signal to noise ratio, \( I_s \) represents the impairments such as too loud a speech level, \( I_d \) represents the mouth-to-ear delay, \( I_e \) represents impairments due to low bit rate voice codecs, and \( A \) represents the advantage of access that some systems have in comparison with PSTN which is determined by the G.107 standardization factors. \( R_0 \) and \( I_s \) values are related to the voice signal itself and not depending on the voice transmission process over IP networks. The R value of the E-model directly converted to its relative MOS value is shown in Table 4-3.

Table 4-3: ITU-T Rec. G.107 E-model, and MOS voice quality scores and descriptions [190]

<table>
<thead>
<tr>
<th>R: E-model</th>
<th>MOS</th>
<th>Voice Quality</th>
<th>Impairment Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>81 - 100</td>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>61 - 80</td>
<td>4</td>
<td>Good</td>
<td>Perceptible but Not Annoying</td>
</tr>
<tr>
<td>41 - 60</td>
<td>3</td>
<td>Fair</td>
<td>Slightly Annoying</td>
</tr>
<tr>
<td>21 - 40</td>
<td>2</td>
<td>Poor</td>
<td>Annoying</td>
</tr>
<tr>
<td>0 - 20</td>
<td>1</td>
<td>Bad</td>
<td>Very annoying</td>
</tr>
</tbody>
</table>
In this research effort, MOS is used as a subjective performance method for SIP-based VoIP applications. The default values of MOS measurements for OPNET® simulations are based on the G.107 and GSM recommendations as specified in the scenarios configuration in Chapter 5.

### 4.4 Routing Protocol Performance Metrics

Real-time applications like VoIP, video and audio streaming, and collaborative group work, are attaining remarkable attention of researchers and practitioners to be implemented over Mobile ad-hoc networks for establishing communication with remote locations, and emergency areas where there is no availability of predefined infrastructure. However, the implementation of these applications over such environments is a challenging task due to the nature of MANET, like dynamic topology, resource limitations, vulnerable wireless links, and without a central controlling authority. Routing is an important strategy in MANET and various routing protocols are used for it. Each routing protocol needs to manage these MANET features, when it transfers the packets from source to destination via a multi-hop network. Moreover, these protocols need to provide and satisfy the Quality of Service (QoS) requirements of real-time applications to provide reliable communication with higher performance. At the same time each real-time application has different characteristics, so different requirements in the network are needed to provide reliable and synchronized data communication.

The performance of any routing protocol can be analysed using various metrics like qualitative metrics and quantitative metrics. Qualitative metrics are beneficial for evaluating internal efficiency of the routing protocol like bandwidth utilization, memory utilization and power consumption. Quantitative metrics are used for a comparative analysis between routing protocols on the basis of various parameters like mobility model, latency, and routing overhead. Some performance metrics are [192]:

1. **Packet Delivery Ratio:** Packet delivery ratio is the ratio of total packet delivered to destination over total packet sent from source.

   \[
   \text{Packet delivery ratio} \% = \frac{\text{total packets reached at destination}}{\text{total packets sent from source}} \times 100 \tag{4.16}
   \]

   It is an important metric for real-time applications as applications are using the UDP protocol like multimedia communication in Mobile ad-hoc networks; a lower packet delivery ratio may reduce the quality of communication to the end user.

2. **Traffic packet counts:** The total number of packets in relation to CBR traffic is measured at different layers of the protocol stack. Bit rate is used to measure packet count at the MAC layer,
and the number of packets forwarded at the routing layer is also counted, to have a traffic packet count.

3. **Latency (average end-to-end delay):** Latency or end-to-end delay is the average time which is taken by any packet to reach its destination. This delay can be due to the delay in the route discovery process, processing delay, and delay from retransmission from multi-hops. This metric has wide applicability in real-time applications like VoIP and video conferencing, where time delay is not negotiable.

\[
\text{Latency (Average end – to – end delay)} = \frac{\sum (\text{CBR sent time} - \text{CBR receive time})}{\sum \text{CBR received packets}}
\]

where CBR = constant bit rate

4. **Routing traffic count:** Total number of packets, which are related to routing traffic of various protocols like AODV, DSR and OLSR, are counted at different layers of the IP stack.

5. **Data packet loss:** Data packet loss can be possible on both the MAC layer as well as on the Network layer. When the packet is received by any node then it can be lost if the buffer of the receiving node is full or the TTL time of packet expires. Packet loss performance can have a significant effect on real-time applications.

\[
\text{Packet loss} = (\text{total packet sent by source} - \text{packet receive by sink})
\]

6. **Throughput of received packets:** Throughput gives the fraction of the channel capacity used for useful transmission and the total number of packets received by the destination within a specified time frame.

\[
\text{Throughput} \% = \frac{\sum_n^1 \text{Total received packets (in bits)}}{\sum_n^1 \text{Simulation time interval (in seconds)}} \times 100
\]

7. **Routing overhead:** The total number of routing control packets sent by any node during data transmission is known as routing overhead. This metric is important to check the efficiency of any routing protocol.

\[
\text{Normalized routing load}\% = \frac{\sum_n^1 \text{Routing packets sent at MAC layer}}{\sum_n^1 \text{CBR traffic received at Agent layer (AGT)}} \times 100
\]

where CBR = constant bit rate
8. **Normalized routing load**: Normalized routing overload is the total number of packets which are transmitted per data packet delivered to the destination. For each hop-to-hop delivery, the transmission of routing is supposed to be one complete transmission. This is the total summation of all control packets sent by all nodes for route discovery. This routing load is also calculated in terms of bytes rather than packets.

\[
\text{Normalized routing load (bytes)\%} = \frac{\sum_1^1 \text{Routing bytes sent at routing layer} + \text{source header overhead}}{\sum_1^n \text{CBR traffic received at Agent layer}}
\]  
(4.21)

9. **Normalized MAC load**: It is measured as the total number of packets including the routing packet, Address Resolution Protocol (ARP) packet, and MAC control packets such as RTS, CTS, and ACK sent at the MAC layer for each data packet delivered to the destination. Normalized MAC load is also calculated using hop-to-hop transmission of routing and gives the measurement of both routing and MAC overhead associated with each data packet delivered. This is an indicator for media access utilization in a wireless medium. The lower the value of MAC load the better will be the protocol performance.

\[
\text{Normalized MAC load} = \frac{\sum_1^1 \text{MAC and routing packets sent at MAC layer}}{\sum_1^n \text{CBR message received at destination}}
\]
(4.22)

where CBR = constant bit rate

10. **Average hop count**: It is an estimation of path length from source to destination for packet delivery. The average hop count can be measured by dividing the total number of packets with constant bit rate transmitted over the MAC layer by the total number of packets with constant bit rate received at the destination node. It is calculated independently from routing protocol simulation.

\[
\text{Average hop count} = \frac{\sum_1^1 \text{CBR message sent at MAC layer}}{\sum_1^n \text{CBR message received at destination (Agent layer)}}
\]
(4.23)

These are some quantitative performance metrics, used to analyse the performance of any routing protocol within specific environments. In addition, this bandwidth, throughput, and reliability are the other metrics, which are equally important for performance measurement of routing protocols. In the case of real-time applications such as multimedia transmission, it demands high end-to-end delay, and high jitterness values, whereas applications like VoIP and video conferencing demands low jitterness.
with higher delay values, for providing the desired results. Hence, real-time application requirements differentiate based on the specific scenario of each application.

### 4.4.1 Performance Metrics for MANET Routing Protocols

Variable numbers of applications have emerged in the past decade to fulfill the various needs of the network and end users. Each application has its own specific characteristics, hence its demands too. With the specific scenario of MANET, various properties such as vulnerable wireless links due to node mobility, and Signals interference produces challenges in providing reliable communication. Hence, each routing protocol must guarantee reliable data transfer, with delay sensitivity while delivering data from source to destination in a real-time environment. The Internet Engineering Task Force (IETF) [192] suggested some recommendations in RFC 2501 for evaluating the performance of routing protocols. Certain base criteria and performance metrics are provided to evaluate the performance of routing protocols. Performance metrics are divided into two parts as qualitative and quantitative metrics [193].

#### 4.4.1.1 Qualitative Metrics

RFC 2501 suggests fundamental qualitative metrics for MANET routing protocols [194]:

1. **Loop Freedom:** MANET, having fixed bandwidth, interference with neighbouring nodes, and with high probability of packet collision, hence it is necessary to prevent packet looping before its delivery to a destination node. TTL values are indicating the packet’s dropping time if it reaches to its maximum hop count.

2. **Distributed Operation:** It suggests the way of interconnecting nodes under various distributed environments.

3. **Demand-based operation:** Bandwidth constraints in MANET or in wireless networks can be facilitated by the use of reactive-based or on-demand routing protocols to minimize the control packets in the network and preserves the power of mobile nodes, hence bandwidth utilization too.

4. **Proactive operation:** Proactive behaviour of the routing protocols provides availability of routes to any destination at any time. In case of any link failure no extra latency is required for establishing a new route to the destination.

5. **Sleep mode operation:** Power management is an important aspect in MANET as each node is operated with a limited battery power source. To make the protocol more efficient it must be capable of working even when some nodes are in sleep mode for short periods without affecting its performance.
6. **Unidirectional support:** Nodes in MANET must be capable of communicating with unidirectional links too. The routing protocol must ensure that they must support unidirectional and bidirectional links.

### 4.4.1.2 Quantitative Metrics

The quantitative metrics are defined in RFC2501 and can be used to compare and evaluate the performance of a routing protocol for different type of applications. The metrics are [193]:

1. **Route acquisition time:** It indicates the total time taken by a protocol to discover a route between two different ends. The higher the route discovery takes, the higher the latency in the network occurs. This metric helps with the evaluation processes for the actual status of the route performance.

2. **Out of order delivery:** The ratio of packets that delivered out of order, also affects the performance of higher layer protocols, such as when TCP based signalling systems for real-time applications.

3. **Efficiency:** The route efficiency from source to destination can be evaluated by calculating the total overhead used by a protocol to route the packets. The route performance could be enhanced using dynamic updated routing table with lower number of hops.

4. **End-to-end delay data throughput:** The total time needed to deliver amount of data per second from source to destination. This metric provides an evaluation tool for the routing performance for an application with variable amount of packets such as voice data.

### 4.4.1.3 Internal Efficiency Metrics for the Routing Protocol

Various ratios need to be analysed for tracking the internal efficiency of a routing protocol. Some of the ratios are [194]:

1. **Average number of data bits transmitted for each data bit delivered:** It is used for measuring the bit efficiency for delivering the data within a network. Hence, it also gives the average hop count to deliver the packet.

2. **Average number of control bits transmitted for each data bit delivered:** It is an indication of the bit efficiency of the protocol while expending the control overhead within a network to deliver data. This should not only include bits in the control packet but also the bits in the header of the data packet.

3. **Average number of both control bits and data packets transmitted for each data packet delivered:** It is the measure of bandwidth efficiency in contention based link layers.
These quantitative metrics are based on same network parameters. Such metrics can be useful to evaluate the bandwidth utilization, memory and power consumption. Evaluating the routing protocol efficiency offered to higher layer protocols is also discussed in RFC 2501, and are summarized in the next subsection.

### 4.4.1.4 Network Performance Parameters

Various network parameters also have a significant impact while studying the behaviour of various routing protocols. Some network parameters such as network size, mobility, and dynamic topology need to be taken into consideration. The main network parameters are [195]:

1. **Network size**: As MANET is a self-configuring network, therefore the nodes can join or leave the network system. The number of nodes in the network system can have a significant effect on routing protocol performance, hence the network size needs to be taken into consideration while analysing any routing protocol.

2. **Dynamic topology**: High node mobility in the network system leads MANET into a highly unstable state during the connections integration. Therefore, the node mobility has a significant impact on the general performance of the routing protocol.

3. **Network connectivity**: The network connectivity in MANET is the average of neighbours with whom a node is communicating within a single hop or more during the communication setup.

4. **Link capacity**: This parameter is considering the data rate between two nodes in MANET that remaining after packet loss due to the MAC access and data encoding processes. The data rate has a significant effect over the link capacity for wireless systems.

5. **Unidirectional link**: In a routing protocol with a full duplex communication, if unidirectional links were used, then it may affect the routing protocol efficiency. The usage of the unidirectional links depends on the type of application identified and applied over the network system.

6. **Mobility**: The use of different mobility models by a routing protocol for the node mobility can affect the performance of the routing protocol and the applications performance. The wireless connectivity nature of MANET also affected by the nodes’ mobility.

7. **Sleeping nodes frequency**: The number of sleeping nodes to live nodes during the MANET communication has a significant impact on the performance of the routing protocol.
4.4.2 Definition of Mobility Terms

Functioning of any routing protocol also depends on various factors like node mobility, connectivity of nodes, and node density. Such factors make the network dynamic in nature. Some of the mobility terms are [192]:

1. **Nodes Density**: It is defined as the total number of nodes within a specified region. Node density for a simulation region can be obtained as:

   \[
   \text{Node density} = \frac{\text{Number of nodes}}{\text{Simulation area}}
   \]  

   \(4.24\)

2. **Nodes Distance**: Node distance is the measurement of the total distance between two nodes. The distance between two nodes \(n_1\) and \(n_2\) within a network is given by a Euclidean distance.

   \[
   \text{Node distance}(n_1, n_2) = \sqrt{(x_{n_1} - x_{n_2})^2 + (y_{n_1} - y_{n_2})^2}
   \]  

   \(4.25\)

4.4.3 Performance of AODV Routing Parameters

AODV uses the mechanism of route discovery and route maintenance as represented in Section 2.11. The main advantage of the AODV protocol is its loop-free nature generating less traffic overhead in the network. As for the reactive nature of AODV, the node itself starts a route discovery process to a destination when needed. AODV is designed to minimize the flooding in the network, therefore it uses the shortest possible route to the destination node. Furthermore, AODV reuses an already known updated route to the destination by using the TTL increment features. Each node starts to generate a Route Request (RREQ) packet with an initial value of TTL in the IP header. If the source node does not receive a Route Reply (RREP) after a certain time interval, it resends the RREQ packet and increases the TTL value and waiting time, unless and until the source node gets a RREP or reaches its maximum number of retries. Therefore, this mechanism takes a sufficient amount of time or a considerable delay for the route discovery process to the farthest nodes. Hence, in such a scenario, this protocol can raise issues like time delays for the real-time application. In multimedia applications, the waiting time during the route discovery process runs smoothly at the start. However, in case of link failure, a rerouting process is generated using TTL values. Hence, in the case of the farthest nodes, this process takes a longer time. Therefore, the route discovery process that uses TTL has a significant effect on AODV performance and can degrade the quality of application for end users. The Route Discovery Time (RDT) is the total time taken by the AODV protocol to discover the route, and is very important for protocol performance. During the route discovery process, when any node receives a RREQ, it first checks in its
routing table whether a route to the destination exists or not. If it does not exist then it buffers the message and rebroadcasts the RREQ. Only after receiving a RREP message can the original message be sent. Hence as the number of nodes increases, the value of RDT will also be changed. Previous research also suggests that the value of RDT is not changing linearly [196]. This is because with each extra intermediate node in the route, the IP address has to be recorded at those intermediate nodes, hence the RREP will be larger in size. Hence, the size of RREP message will be larger for the farthest node in comparison to nearby nodes, and changes the RDT value too.

Further prior studies suggest that the use of HELLO messages by the AODV routing protocol has a significant effect on its performance while using with real-time applications [196, 197]. HELLO messages are used by the AODV routing protocol for knowing the link status of its neighbours. Thus, each node in AODV has a link table, which maintains nodes’ active communication by receiving constant HELLO messages from each neighbouring node. Each node analyses whether the links with its neighbouring nodes are stable or lost. Noise in the medium plays an important role in communicating within any application. If a HELLO message is received by any node after the waiting time period due to noise, impairment in the network or due to distance to farthest node, then the node will analyse that the link is lost and it will update the entries in its routing table. In case the deleted node entry belongs to any active route, the protocol needs to restart the route discovery process and produces higher packet loss and latency in the network. This can affect the continuity of any multimedia application streaming as there will be random delay in the AODV protocol, which is not a favourable condition for real-time applications. Prior research also suggests that the value of random delay in the AODV protocol is higher than in other routing protocols [196]. With further studies, authors have also analysed that the average routing overhead of the AODV routing protocol during communication has higher values in comparison to protocols like OLSR [198, 199]. Another issue is reaction time in AODV. While dealing with the nodes’ mobility during implementation of the AODV routing protocol for real-time applications and the nodes’ mobility is on the higher side, the reaction time is also on the higher side. This high reaction time in AODV is not acceptable as it leads to communication termination, and become an unavoidable condition for real-time applications. With the increased number of nodes in the AODV routing protocol, its performance is becoming more critical. This is because the increasing number of nodes that will increase the routing overhead in the network. Therefore, the routing performance will be decreased.

Prior studies suggest that the AODV routing protocol has the lowest average latency and lowest jitter while comparing it with other protocols like OLSR, DSR, DSDV, and others [200], whereas AODV shows the lowest packet loss rate. For delay sensitive applications like multimedia applications, a protocol with low value of average end-to-end delay and low packet delay will be best suited, like DSR [201]. Thus, it is concluded that AODV can represent good performance, while having higher routing overheads. The AODV routing protocol has a higher packet delivery ratio due to its reactive nature. In general, it is concluded that the DSR and OLSR protocols outperform AODV within real-time
applications [201]. Hence depending upon various situations, the AODV protocol varies in its QoS metric performance.

### 4.4.4 Performance of OLSR Routing Parameters

As OLSR is a proactive routing protocol, it maintains routing tables to provide instant routes and updates it periodically, as explained in Section 2.12. OLSR uses the Multipoint Relay (MPR) to minimize the rebroadcasting of HELLO messages in the network. Since each node in the OLSR routing protocol keeps a routing table, therefore a shortest path to destination is provided by this protocol. Because of the availability of various routes to the destination there will be no route discovery delay in case of any link failure. The OLSR protocol utilizes the topology control messages with information about link status after a fixed time interval and maintains a route to the destination.

OLSR uses various messages like the HELLO message and topology control messages in the network for the efficient working of the protocol. HELLO INTERVAL and TC_INTERVAL time slots are used by HELLO and Topology Control (TC) messages, while updating the table and deleting the entries if timeout occurs. It means HELLO_MESSAGE and TC_MESSAGE are used for maintaining each nodes’ state and are directly proportional to HELLO_INTERVAL and TC_INTERVAL simultaneously. Therefore, any change in HELLO_INTERVAL and TC_INTERVAL can change the performance of the OLSR protocol [203]. Within a network topology, there are various nodes, out of which some nodes are selected as MPR. In the case of any node, which is itself a MPR, and it moves away from the range of its neighbourhood nodes, then for a certain time period some of those neighbourhood nodes have stale information, unless these nodes get updated from some other new neighbouring nodes of that MPR node. It will take some multiple times of HELLO_INTERVAL. The performance of real-time applications will be affected more for higher time intervals in comparison to low time interval value.

Previous research also suggests that after any link failure, like an MPR left the network, new routes are available in the routing table but cannot be actually used until the topology control waiting time [204]. It can also degrade the performance of the applications. In the case where the value of the HELLO_INTERVAL is increased, it will improve the performance of the OLSR protocol, but it will increase the overhead in the network, which is not suitable for real-time applications. Prior research concluded that the OLSR protocol is better than AODV and DSR for real-time applications like the multimedia video streaming application, in the case of the route discovery process after the link failure [204].

Packet loss and erroneous packets are not acceptable in real-time applications such as VoIP, where multimedia streaming causes the quality degradation. In the OLSR protocol, neighbourhood hold time interval is used for link expiry. In case of any node leaving from the transmission range of the
neighbourhood node, the numbers of packet loss will increase hence the performance of the real-time applications will decrease. With further studies it can be concluded that HELLO_INTERVAL and TC_INTERVAL are used for route stability, but the DEFAULT values of both can be ineffective for route discovery as a two-hop link cannot be established with frequent node mobility and frequent route changes [205]. It means in this case the packet loss ratio will be higher and the delays for a new route discovery process will be higher, which needs to be addressed for reliability of real-time applications.

With the increase in the size of the network, the OLSR protocol outperforms other routing protocols in real-time scenarios, as OLSR has a lower ratio of packet drop because OLSR is a table-driven proactive multipath routing protocol. The average delay for the network is significantly small and almost constant with the variation of the network size. Since each node in OLSR has a routing table, therefore end-to-end delays in the OLSR protocol are almost constant. The packet delivery variation and jitter for the OLSR routing protocol is observed to be slightly higher. Therefore, high quality video and audio streaming in the OLSR protocol does not provide best results. The OLSR routing protocol has lower performance in terms of packet delivery ratio and jitter [200]. Even the proactive nature of the protocol does not guarantee lowest values of jitter. Therefore, it can be argued that OLSR cannot be a better choice for hard real-time applications broadly.

4.5 Summary

The SIP-based VoIP applications are affected by the SIP signaling performance, voice quality, and the routing performance. The SIP end-to-end performance metrics of RFC 6076 are considered as the most efficient approach to evaluate SIP signaling. However, there are no benchmarking values proposed for these performance metrics. In this research study the RRD, SRD, and SDD will be used for the evaluation efforts for SIP signaling for the registration, call setup, and termination processes. In addition, the study considered the throughput, packet loss rate, end-to-end delays, and jitter for voice quality evaluation. Furthermore, the research study considered routing performance metrics of VoIP application processes to evaluate the routing efficiency. The study of these performance parameters will help to improve the overall performance for SIP-based VoIP applications by controlling the values of these parameters to be within the acceptable range [206]. Further considerations regarding the performance metrics of MANET routing protocols in general have been discussed in this chapter. In addition, a review of related performance parameters for both AODV and OLSR have been covered as well.
Chapter 5

5 Research Framework and Methodology

A wide range of implementations are available for SIP-based applications over MANET. Disaster recovery and emergency backup systems are considered as very interesting implementations that MANET can effectively serve and support [58]. As discussed in Chapter 4, the most efficient method to enhance the performance of the SIP signaling system over MANET is to study and analyse the SIP signaling behaviour regarding the MANET nature. In addition, the literature review showed no benchmark values for the performance metrics of SIP signaling, SIP registration process, call setup process, and call termination process [1, 149].

In this chapter, an evaluation scheme will be presented to evaluate and benchmark the performance metrics for the SIP signaling system using a Back-to-Back User Agent (B2BUA) implementation for the SIP server over MANET. The system representation depends on the best effort scenarios. The research methodology to be followed during this research study will be presented. In addition, the scenarios and assumptions of this research study will be introduced, and the simulation setup parameters will be identified. The suitable tool for this study compared with other simulation tools is the OPNET® Modeler, regarding the simulation capabilities, accuracy, and support for SIP signaling and MANET implementations that meet with the results’ reliability and implementation requirements of this research study. The simulation efforts for this research study will be carried out using GSM voice codecs to evaluate SIP call processes and VoIP QoS parameters together over MANET. The simulation efforts will be used to evaluate and implement different scenarios for SIP-based VoIP applications over both AODV and OLSR, for both IPv4 and IPv6 MANETs. The SIP signaling and VoIP QoS parameters will be assessed on OPNET® Modeler to simulate the required scenarios for this research study. The best effort scenario will be represented with a Static model, where all the nodes are stable and not moving. Next the Uniform mobility model will be simulated, where all the nodes are moving (except the SIP server) in a uniform symmetric motion with the lowest possible changes in the routing tables and hops number between the calls’ entities [58, 71]. Furthermore, this chapter will introduce the design and implementation methods for a number of the SIP end-to-end performance metrics that will be used for the evaluation and performance enhancement efforts of this research study. At the beginning of this research study, the research considerations were about the SIP-based VoIP implementations over ZigBee-based Wireless Sensor Networks (WSN). However, the research area has been modified to
consider MANET rather than ZigBee WSNs because of the implementation difficulties for the SIP signaling system over ZigBee, as explained in Appendix A.

5.1 Research Scenarios and Implementations

The aim of this research is to study a simple, closed, AODV-based MANET scenario with a high density of mobile nodes using SIP-based VoIP to communicate together, as represented in Figure 5-1. The study assumed a system model with a single SIP server based on a B2BUA-based SIP signaling system for a MANET network system, which provides the SIP registration, initiation, and termination mechanisms for SIP calls. It also assumed that node A is the caller and node B is the callee. Node A and node B both need to register with the SIP server to identify their existence and IP addresses. When node A wants to call node B, the SIP initiation messages for the call setup process start flooding between both caller nodes through the B2BUA-based SIP server, as represented in Figure 4-2. When the call ends, the SIP termination message will be sent through the B2BUA-based SIP server to terminate the call.

![Figure 5-1: SIP-based VoIP implementation over closed MANET](image)

This scenario takes advantage of applying SIP-based VoIP applications as an alternative or backup communication system over mobile nodes that support the MANET network system. This system could be used for disaster and emergency recovery schemes when other communication systems are lost or break down. The infrastructure-less, multi-hop communication, dynamic topology features of a Mobile ad hoc network differentiate it from other conventional wireless networks. Self-configuring natures, infrastructure-less features are the best suitable form for emergency applications, military operations, and collaborative applications. In emergency applications where pre-available infrastructure is destroyed due to sudden catastrophes (e.g. Tsunami, earthquake, etc.), a Mobile ad hoc network can be used to establish communication with the emergency units, and similarly in collaborative applications like video conferencing, multimedia chat, etc. These implementations can be one-to-many and many-to-many; therefore a Mobile ad hoc network needs to be deployed in various environments like
broadcast and multicast natures. The major requirements of these applications are delays and packet sensitivity [79], where the retransmission mechanism cannot be applied for real-time applications. Thus, in this study, a MANET with a moderate node capacity and different types of mobility models will be considered based on the literature review and similar to real-life scenarios.

5.2 Research Methodology

In this research study, the methodology depends on the literature investigations of the current state of the art for SIP signaling over MANET as represented in Chapter 4. This research study is going to implement a simple closed MANET system with different mobility scenarios and apply the SIP-based VoIP application between two ends, as shown in Figure 5-1. Then, the study will follow a sequence of implementations to evaluate and enhance the SIP signaling system over MANET [71]. The SIP signaling and QoS parameters for VoIP over MANET will be assessed on the simulation tools to implement the research scenarios. In general, the evaluation of the SIP signaling performance for the simulated scenarios will depend on the end-to-end performance metrics of RFC 6076 and the call setup time that was discussed in Section 4.2. In addition, the evaluation of SIP signaling considered the B2BUA SIP server performance regarding SIP messages during the registration, initiation, and termination processes of SIP calls, as shown in Figure 4-2. This study considered the B2BUA-based SIP server because of its features among the proxy-based SIP server, as discussed in Section 2.3.

The first part of the study will evaluate and analyse the SIP signaling system and VoIP performance over MANET. This evaluation will include part of the determination efforts to benchmark the SIP end-to-end performance metrics of RFC 6076 for WLAN in general and MANET in particular for a B2BUA-based SIP signaling system [149]. In addition, this part will determine the main performance issues for the implemented systems that support the studies and implementations for new performance enhancement approaches. The second part of the study is going to employ the ROHC system over an IPv6 SIP-based VoIP application for the same MANET system that was used in the first part of the evaluation study. This implementation will be used to investigate the SIP signaling performance, in particular regarding the enhancement level that ROHC could provide for the investigated performance metrics [233]. The third part of this research study considers the development and enhancement methods to improve the performance of the SIP signaling system over MANET. This part will depend on the evaluation and analysis findings in the first and second parts. The dynamic adjustments for the parameters of MANET routing protocols will be applied for the performance enhancement approaches. These approaches will provide a flexible dynamic accommodation for the SIP signaling system for VoIP applications to meet with the variable connectivity nature of MANET. In addition, the third part is considered as the main contribution of this research study. The results of the performance
enhancement contribution will be analysed and studied against the current state of the art to show the enhancement level of the SIP signaling performance over MANET.

5.3 Simulation Parameters and Assumptions

In this research study, the IEEE 802.11n will be applied as the wireless network standard for MANET, due to its enhanced features, wide usage over WLAN devices, and good mobility support for MANET compared with 801.11a, b and g, according to [207] and [208]. As discussed and concluded in Section 3.6, OPNET® Modeler provides the best implementation capabilities over other simulation tools and test-beds for the SIP signaling system over both AODV and OLSR routing protocols. OPNET® Modeler is a Discrete Event Simulation (DES) tool that provides realistic and accurate implementations for SIP-based VoIP applications over MANET and for a large number of nodes with reliable implementations. Therefore, the simulation efforts will be carried out in OPNET® Modeler version 17.1.

The simulation works will be implemented over four types of mobility models: Static, Uniform, Random Waypoint (RWP), and RWP-All. These models represent the most common real-life mobility scenarios for MANET nodes [209]. The B2BUA-based SIP server is fully controlling and managing the SIP sessions over all the call stages. In addition, the SIP server is assumed to have a high performance for data processing over all received and sent SIP traffic during all the calling stages. In the Static model, the MANET’s nodes are stable and not moving. In the Uniform model, all nodes move in the same direction, with different speeds within the identified speed limitation range, except the SIP server which is in static state [71]. In the RWP model, the nodes move in different directions, but the SIP server is stable in the centre of the simulation area. In the RWP mobility model, every node in the simulation, except the SIP server, has its own mobility direction and speed, depending on the identified random functionality of the node parameters. In the RWP-All model, all nodes move in different directions, including the SIP server. The mobile nodes randomly move and at the same time act as routers that discover and maintain the route statistics for multi-hop communication [58]. The main characteristics that affect the implementations for these mobile nodes are the unpredictable topology changes, low bandwidth, high level of mobility, and variable connections. The reason for examining random mobility using two different models is to study and evaluate the effect of SIP server mobility for VoIP applications and the signaling QoS.

In general, the topology modelling system in the designed scenarios depends on the AODV and OLSR mechanisms for route selections in both AODV and OLSR which are well implemented in OPNET® Modeler. The topology selection depends on the algorithm of the routing protocol that considers the location, number of hops, and power issues. For both RWP and RWP-All scenarios, MANET has dynamic topologies between its nodes. Thus, the nodes are partially connected during the simulation time. The Queuing theory for the B2BUA-based SIP server system that is applied in the simulation
system is for a single server node with an M/M/1/K queue. The single server queue has a limited queue size (K) (i.e. buffer) as represented in Figure 2-9. In addition, the M/M/1/K queue system will be applied for the node’s routing system for each node in the MANET, as shown in the equations of (2.3), (2.4), and (2.5) [27, 28]. Furthermore, this study will consider both IPv4 and IPv6 MANETs in order to identify the difference in route overhead between the two IP systems. IPv6 will not apply the QoS features over MANET applications in this research study and simulation efforts, as it is not supported over MANET in OPNET® Modeler. The GSM voice codec will be used for VoIP applications because of its simplicity, wide usage, efficiency, and compatibility with MANET natures [138, 210]. In most of the simulation, voice data will travel through the B2BUA-based SIP server between the caller and the callee to provide a secure communication system [145]. Table 5-1 presents the simulation parameters that were identified depending on the features and capabilities of the MANET and SIP-based VoIP applications. This design and implementation will be used to investigate and evaluate the QoS for SIP-based VoIP over MANET using the previously identified mobility systems.

Table 5-1: System Configuration in OPNET® Modeler [124]

<table>
<thead>
<tr>
<th>A. MANET</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Simulations: 56</td>
</tr>
<tr>
<td>Variable Seed Number: 128</td>
</tr>
<tr>
<td>Simulation Time: 30 Minutes = 1800 Seconds</td>
</tr>
<tr>
<td>Mobility Models: Static, Uniform, RWP, and RWP-All</td>
</tr>
<tr>
<td>MANET Routing Protocols: AODV, OLSR</td>
</tr>
<tr>
<td>Background Traffic: 30% to 40%</td>
</tr>
<tr>
<td>Number of nodes: 25 nodes</td>
</tr>
<tr>
<td>Area Dimension: 1 km x 1 km</td>
</tr>
<tr>
<td>NODE Speed Range: uniform speed between 1.4 m/s (5 km/hr) and 9.8 m/s (35 km/hr)</td>
</tr>
<tr>
<td>WLAN Physical Characteristic: 802.11 n</td>
</tr>
<tr>
<td>Data Rate: 13 Mbps</td>
</tr>
<tr>
<td>Maximum Transmission Range between Nodes: 100 - 250 meters</td>
</tr>
<tr>
<td>Frequency Band: 2.4 GHz</td>
</tr>
<tr>
<td>Transmission Power: 0.001 W</td>
</tr>
<tr>
<td>Packet Size: 512 Bytes</td>
</tr>
<tr>
<td>Buffer Size: 32 Kbytes</td>
</tr>
<tr>
<td>Maximum Simultaneous Calls:</td>
</tr>
<tr>
<td>Total VoIP Calls in 1800 Seconds: 175 Calls</td>
</tr>
<tr>
<td>MAXIMUM Call Duration: 10 Seconds</td>
</tr>
<tr>
<td>caller: Node 1, callee: Node 24</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>B. AODV</th>
</tr>
</thead>
<tbody>
<tr>
<td>RREQ Retries: 3</td>
</tr>
<tr>
<td>Max Path Discovery Time (Seconds): 5</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>C. OLSR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hello Interval (Seconds): 2</td>
</tr>
<tr>
<td>TC Interval (Seconds): 3</td>
</tr>
<tr>
<td>Topology Hold Time (Seconds): 10</td>
</tr>
<tr>
<td>Neighbourhood Hold Time (Seconds): 5</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>D. Applications: SIP Based VoIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Server Connect Timeout: TCP Based</td>
</tr>
<tr>
<td>Voice Codec: GSM (13 Kbps)</td>
</tr>
<tr>
<td>VoIP Calls (Unlimited)</td>
</tr>
<tr>
<td>Call Duration: 10 Seconds</td>
</tr>
<tr>
<td>Caller: Node 1, callee: Node 24</td>
</tr>
<tr>
<td>Total VoIP Calls in 1800 Seconds: 175 Calls</td>
</tr>
<tr>
<td>Maximum Simultaneous Calls:</td>
</tr>
<tr>
<td>B2BUA-based SIP Server: Unlimited Calls</td>
</tr>
<tr>
<td>User Agent (Caller/Callee): 1 Call at time</td>
</tr>
<tr>
<td>Total VoIP Calls in 1800 Seconds in the SIP server: Unlimited Calls</td>
</tr>
</tbody>
</table>
All the assumptions and simulation setup parameters of this study are based on other similar studies from the literature, such as [71, 209, 211, 212, 213, and 214]. The MANET scenarios and implementation methods that are followed in this research study are driven from [210, 215, and 216]. Furthermore, the SIP server in this design will be used as a single SIP server. The SIP server will act as a B2BUA for SIP-based applications. Therefore, the SIP server is considering multiple SIP/TCP connections at the time of multiple instantaneous calls. Figure 5-2 shows the scenario implementations in OPNET® Modeler at the start stage of the scenario in the simulation works with regard to the identified parameters given in Table 5-1. This design depends on the features and capabilities of MANET for SIP-based VoIP implementation and will be followed over all mobility models of this study. The red box represents the mobility area of one square Kilometre (1 Km x 1 Km) for MANET. The simulation models will be used to generate an unlimited number of sequenced voice calls between node 1 and node 24 where each voice call has a duration of 10 minutes and only one active VoIP call at a time. The calls are in its best effort conditions where background traffic is generated at the same time. The total estimated number of VoIP calls initiated in the best effort conditions is 175 calls in 30 minutes between node 1 and node 24. The simulations will generate an unlimited number of sequenced voice calls between different nodes in MANET to provide the background traffic using the VoIP applications. This background traffic has a medium level of saturation for the provided wireless bandwidth in MANET, with 40% as represented in [151, 210, 212, 222 and 223]. Figure 5-3 shows the implementation of the RWP mobility model for MANET in OPNET® Modeler after 20 seconds from the beginning of the simulation. Each node, except the SIP server, has its own mobility direction and speed. The mobility direction for each node is represented by a red arrow.

![Figure 5-2: MANET Model's implementations in OPNET® Modeler at simulation time 0 Second](image)
In the simulation efforts, using static and simple SIP-based VoIP applications was assumed where the Session Description Protocol (SDP) signaling system is not applied in the simulation works of this research study [13]. Most of the research studies in the Literature Review considered SIP signaling without the SDP signaling system in their research efforts, such as in [151, 212, 213, 218, and 219]. This is because of the limitations and constraints on the analysis and implementation works when applying the SDP signaling system [5, 65]. Furthermore, the implementation of SDP signaling in the simulation tools that support SIP systems over OPNET® Modeler is not possible, especially for MANET. This is because these simulation tools do not provide SDP as the mobility issues have considerable problems over SIP/SDP signaling synchronization during the simulation. Therefore, this research study focused on the static simple mechanism of SIP signaling and its timers to evaluate and enhance the SIP applications.

5.4 Data Representation Method for the Simulation Results

The simulation results will be represented to show the minimum, average, and maximum records of the results. This is because the various representations of the simulation results are giving unclear and ambiguous readings for the efficiency of the overall system represented. In addition, the representation of the SIP-based VoIP applications will follow the structures of the simulation works in the main resources in the literature review, as in [217, 218, 219, and 220]. This dependency on the literature review implementations for the simulation setup parameters will support the credibility of the research findings of this research. Furthermore, these implementation methods are provided and recommended by OPNET® modulation tools [124]. For voice data packets, the RTP identifies the voice stream by its unique Synchronisation Source Identifier (SSRC). In addition, each packet is identified by the port
numbers, sequence numbers and timestamps [135]. The time difference of the same packet at source and destination results in its one-way delay, as suggested in [221].

### 5.5 The Implementation of SIP end-to-end Performance Metrics

The OPNET® Modeler supports both the SIP implementations for the B2BUA-based SIP server and MANET implementations for AODV and OLSR routing protocols. These implementation features give the OPNET® Modeler the advantages over other simulation and emulation tools, especially for this research study. However, the SIP end-to-end performance metrics are not implemented in OPNET® Modeler, which require implementation efforts to use it in the simulation works to meet with the purposes of this research study. Therefore, a number of performance metrics will be implemented to be used during the performance evaluation efforts and the performance enhancement contributions. Based on these performance metrics, the performance enhancement contributions will be proposed to understand the actual values of the performance metrics before and after applying the performance enhancement approaches. The performance metrics that will be designed and implemented in this research study are the performance metrics for RFC 6076, which are the Registration Request Delay (RRD), Session Request Delay (SRD), and Session Disconnect Delay (SDD), using the performance metric equations of (4.1), (4.3), and (4.4) that are represented in Figure 4-2. In addition, this research study will implement the call registration, call setup, and call termination values in OPNET® Modeler. The call setup time is considered as the main performance metric that is used during this research study. All the implementation works of this research study will employ the evaluation values of these performance metrics as a part of the benchmarking efforts, especially for the RFC 6076. The evaluation results for these performance metrics will be reported in Chapters 6, 7 and 8. In addition, based on the values of these performance metrics, the performance enhancement approach will be identified and examined in Chapter 9. The values of the end-to-end SIP performance metrics will be driven from the OPNET® simulation results and manipulated in Excel for analysis and representation purposes. For the registration process, this research identifies a number of timestamp values to help with the evaluation and the performance enhancement efforts in the next chapters, as shown in Figure 5-4. From equation (4.1), the RRD values for a registration process number \( n \) for both UAC (A) and UAC (B) can be calculated in milliseconds (ms), as follows:

\[
RRD1 \ (n) = R4A \ (n) - R1A \ (n) \tag{5.1}
\]
\[
RRD2 \ (n) = R4B \ (n) - R1B \ (n) \tag{5.2}
\]

where RRD1 is the registration request delay for UAC (A) and RRD2 is the registration request delay for UAC (B). Therefore, the total time for the registration process at simulation time \( t \) for both UAC (A) and UAC (B) can be found using Algorithm 5.1 that represented in the flowchart of Figure 5-5.
Figure 5-4: The identified timers for the registration process of the SIP signaling system

**Algorithm 5.1:** Total Time used for the Registration Process of SIP Call number \( n \)

\[ \text{REGISTRATION}(n) \]

**Inputs:** \( R1_A, R4_A, R4_B, R4_B, n \)

**Outputs:** \( \text{RegistrationInitiation}(n), \text{RegistrationFinish}(n) \)

1: if \( R1_A(n) \geq R1_B(n) \)
2: \( \text{RegistrationInitiation}(n) = R1_A(n) \)
3: else
4: \( \text{RegistrationInitiation}(n) = R1_B(n) \)
5: if \( R4_A(n) \geq R4_B(n) \)
6: \( \text{RegistrationFinish}(n) = R4_A(n) \)
7: else
8: \( \text{RegistrationFinish}(n) = R4_B(n) \)

Figure 5-5: Flowchart representation for Algorithm 5.1

The call setup performance depends on the SIP initiation messages sent and received by the call entities. For MANET systems with moderate to high node capacity, variable hop numbers, and node mobility, SIP performance metrics can be used to enhance the performance by adjusting the routing parameters.
to the required level. The SRD values are calculated for all calls initiated by the caller node UAC (A). As the call initiation signaling passes through the SIP server, all messages and parameters for INVITE messages are recognized by the SIP server and the caller to evaluate the SIP signaling performance, as represented in Figure 5-6.

The SRD in this implementation is the time difference between the INVITE message, sent at time TInt1, and Tx3, the time at which the callee’s response message was received by the UAC (A) for the call invitation, as represented in equation (4.3). Therefore, the SRD values for a SIP call number \( n \) can be calculated in milliseconds (ms) from the following equation:

\[
SRD \left(n\right) = T_{A1} \left(n\right) - T_{int1} \left(n\right)
\]  

(5.3)

where SRD is the delay for the SIP call request initiated by caller UAC (A) to the callee UAC (B). In the call setup process, the three-way handshake system is applied for the call setup, as it represents the basic SIP signaling flow between the call entities, and the SRD SIP performance metric can be applied. The call setup time is the time difference between TInt1 and TA3, where TA3 is the time of receiving the call acceptance acknowledgment (200 OK) by the callee UAC (B). Therefore, the total time used for the call setup process for the SIP call number \( n \) can be calculated using Algorithm 5.2 that represented in the flowchart of Figure 5-7.

---

**Figure 5-6: The identified timers for the call setup process of the SIP signaling system**
Algorithm 5.2: Total Time used for the Call Setup Process for SIP call number $n$

\[
\text{CALL\_SETUP}(n)
\]

Inputs: $T_{\text{INT}}1(n), T_A3(n)$

Output: CallSetupTime$(n)$

1: \hspace{1em} \text{if} \hspace{1em} T_{\text{INT}}1(n) > 0 \hspace{1em} \text{then}

2: \hspace{1em} \text{CallSetupTime}(n) = T_A3(n) - T_{\text{INT}}1(n)

3: \hspace{1em} \text{else return}

---

![Flowchart representation for Algorithm 5.2](image)

Figure 5-7: Flowchart representation for Algorithm 5.2

Finally, for the call termination process, this research identified timestamp values to determine the termination process for a SIP call $n$ as shown in Figure 5-8. From equation (4.4), the SDD values for a SIP call number $n$ can be calculated in milliseconds (ms) from the following equation:

\[
SDD(n) = TK3(n) - TD1(n)
\]  

(5.4)

where SDD is the disconnect process delay for a SIP call between UAC (A) and UAC (B). Therefore, the termination process has the same SDD values and can be calculated using Algorithm 5.3 that represented in the flowchart of Figure 5-9.

![Diagram showing identified timers for call termination process](image)

Figure 5-8: The identified timers for the call termination process of the SIP signaling system
Algorithm 5.3: Total Time needed for the Termination Process of SIP call number $n$

\[ CALL\_TERMINATION(n) \]

**Inputs:** $T_D1(n), T_K3(n)$

**Output:** CallTerminationTime($n$)

1: \[ \text{CallTerminationTime}(n) = T_K3(n) - T_D1(n) \]

---

5.6 The Evaluation Methods for the SIP signaling performance and QoS for VoIP Applications over MANET

The assumptions in this research study are designed to generate many VoIP calls within a short period to provide a comprehensive investigation for all sessions over different MANET models. OPNET® Modeler supports the implementations for SIP-based VoIP applications over AODV and OLSR routing protocols with the ability to analyse the simulation results. The B2BUA-based SIP server will be used to provide the SIP-based VoIP implementations in this research study. The B2BUA-based SIP server provides a secure and controlled communication system that provides a wide range of privacy with dynamic voice connectivity system over MANET. The B2BUA-based SIP server will be implemented using a single server node that uses the M/M/1/K queuing system with a single queue with limited buffer size (K). The SIP signaling and QoS parameters for VoIP are assessed in OPNET® Modeler. The simulations will consider simultaneous VoIP applications as background traffic that influences the performance of SIP applications. For Static and Uniform Scenarios, the SIP signaling performance is in its best effort state as the weakest link at the network performs well for routing and data transmission processes. In addition, it was assumed that the connectivity is available for all MANET nodes in the simulation area and no hardware problems are considered. On the other hand, the worst case scenario is represented with the mobility of the SIP server, as shown in the Random Way Point (RWP) scenario. By comparing the efforts over different mobility models, this study will highlight the differences between the SIP signaling performance metrics over these examined scenarios. During the results analysis, the average data representation will be used as it provides simple and comparable readings that help to understand the overall performance for SIP signaling.
To benchmark the SIP end-to-end performance metrics for a B2BUA-based SIP signaling system from the evaluation efforts, the study will depend on the results of the Static and Uniform mobility models. In addition, the results of the evaluation studies in (Chapter 6) will be used later as a reference to compare the performance for the ROHC implementations (Chapter 7) and the proposed Cross-Layer Approaches (Chapter 8) over MANET. These benchmarked values will be used to identify the SIP signaling issues that affect the general performance of the SIP-based applications based on the investigated performance metrics.

5.7 The Design and Implementations for the Proposed Cross-Layer Approaches

The RFC 6076 proposed end-to-end performance metrics for SIP signaling to provide a standardized method of evaluating SIP performance over different platforms. The benchmarking values for these metrics will be proposed and examined over MANET with different mobility models in Chapter 6. In this section, a design for a Cross-Layer approach will be proposed to enhance the SIP sessions’ performance of SIP-based VoIP over AODV-based and OLSR-based MANETs. This approach will employ the SIP performance metrics to maintain the SIP registration, call setup, and termination processes of the SIP calls using a dynamic adjustment system for the routing protocol parameters as shown in Figure 5-10. For the registration process, the Cross-Layer algorithm will be implemented on the caller agents: User Agent Client (A) and User Agent Client (B). On the other hand, for the call setup process, the Cross-Layer algorithm is applied over the UAC (A), and the UAS (SIP Server) where each of them has a different algorithm depending upon the related identified signaling system. Furthermore, the termination process will be implemented on the User Agent Client (A) as shown in Figure 5-10. Moreover, the proposed Cross-Layer approaches are applicable to be implemented over both IPv4 and IPv6 traffic systems.
5.7.1 Cross-Layer Algorithms for AODV-based MANET

The design of the Cross-Layer approaches for AODV (CLAODV) depends on the reactive nature and routing parameters for the AODV as represented in Chapter 2, Section 2.11. The routing parameters that are adjusted and modified by the CLAODV depend on the performance enhancement review for AODV at Chapter 2, Section 2.11.4. In addition, the design depends on the evaluation results and findings for SIP-based VoIP over AODV-based MANET in Chapter 6. Therefore, the CLAODV algorithms and implementations are not applicable to be implemented with other routing protocols unless they share the main routing concepts and parameters. Figure 5-11 represents the CLAODV representations over the Open Systems Interconnection (OSI) for all performance enhancement approaches of the investigated SIP processes over MANET. This section will represent the design for
the CLAODV algorithms over the SIP signaling processes that are modelled, implemented, and evaluated by the OPNET® Modeler.

CLAODV Examinations for SIP processes 1, 2, or 3:
1. Registration: RRD1, RRD2, REGISTER, 200 OK
2. Initiation: SRD, INVITE, 200 OK
3. Termination: SDD, BYE, 200 OK

Figure 5-11: CLAODV representation over MANET OSI for the proposed performance enhancement approaches of the SIP processes

5.7.1.1 CLAODV for the Call Registration Process

The performance of the SIP registration process depends on the SIP REGISTER messages that are sent by the UACs and received by the SIP Server. For MANET systems with moderate to high node capacity, variable hop numbers, and node mobility, the SIP performance metrics can be used to enhance the performance by adjusting the routing parameters to the required level. For the CLAODV approach, the RRD values of the SIP performance metrics can be used to enhance the performance of the registration process by adjusting the route discovery values to the required level, depending on the benchmarked RRD values that will be investigated in Section 6.3.1.1. The SIP registration intervals had will be evaluated as well for normal AODV-based MANET in Section 6.3.2. For the CLAODV approach of the registration process, the RRD values are independently calculated by the caller nodes using the identified timers for SIP call number \( n \) on the UAC (A) and UAC (B), as represented in Figure 5-10. The total registration time is the time difference between R1A or R1B for the initiated REGISTER message of UAC (A) or UAC (B), and the R4A or R4B for the 200 OK response message of the REGISTER message for the UAC (A) or UAC (B). The RRD values of a registration process for a SIP call number \( n \) is calculated using equations (5.1) and (5.2), and Algorithm 5.1. The proposed CLAODV approach for the SIP registration process is applied over the UAC (A) and UAC (B). Once the UAC (A) or UAC (B) recognized a delay, the CLAODV approach will be triggered to modify the AODV routing parameter using the Cross-Layer messages based on the investigated SIP performance metrics at the application layer, as represented in Figure 5-11.

The proposed CLAODV algorithm considered the RRD_BenchM parameter as the benchmark value for RRDs that is used to determine the registration process of the SIP call over MANET. The RRD_BenchM value will use a specific input value from the evaluated RRD values of the benchmark efforts of Section 6.3.1.1 that will be determined from the evaluation efforts over the Static and Uniform mobility models for best case scenarios. The RRD values that will be used for the RRD_BenchM are...
determined for both IPv4 and IPv6 traffic by the proposed benchmarked sets for the SIP performance metrics as in Section 8.2 and Table 8-1. During the registration session, the RRD values for both caller UAC (A) and UAC (B) will be compared with the determined RRD_BenchM. If the RRD values are greater than the RRD_BenchM, then the CLAODV will be triggered to update the parameters of the routing table by resending the RREQ messages with a longer TTL period, as represented in Algorithm 5.4 and the flowchart in Figure 5-12. The effectiveness of having longer TTL values over the application’s performance was discussed in Section 2.11.4. The CLAODV process will be triggered for a short time up to a RREP message received to provide the REGISTER message with an active updated route based on the AODV route discovery as represented in Algorithm 2.2. As soon as the RREP is received, a REGISTER message will be resent. In addition, the TTL value for the RREQ will continually degrade to its original value to save the CPU cycles and bandwidth. The sequence number for the registration process has the same sequence number as the SIP call number \( n \). The simulation time \( t \) is represented in seconds, \( T_{\text{Registration\_Wait}} \) is the maximum specified time in seconds before sending the re-REGISTER message, and \( T_{\text{End}} \) is the simulation end time. The total registration time for a registration process is the total of RRD1 and RRD2 or the time difference between the first registered UAC of the call parties with the SIP Server and the last received registration response from the call parties.

Algorithm 5.4: Registration process for Cross-Layer AODV. The Implementation of this algorithm is on the \( UAC(R(n)) \) for SIP call number \( n \), where \( R \) is the User Agent Client \( A \) or the User Agent Client \( B \).

\[
\text{CALL\_REGISTRATION}(n) \\
\text{Inputs: } RRD_{BM}, t_{rw}, t_{end} \\
1: \quad \text{for } t = 0, \text{ until } t \leq t_{end}, \text{ with step } t = t + 1 \\
2: \quad \text{if } \text{REGISTER}(n) \text{ is sent by the } UAC(n) \\
3: \quad \quad \text{if } OK_{200}(n) \text{ is received and REGISTER}(n) \text{ is received} \\
4: \quad \quad \quad \text{RRD}(n) = R4(n) - R1(n) \\
5: \quad \quad \text{else if } R3(n) > RRD_{BM} \text{ or } RRD(n) > RRD_{BM} \text{ or } RERR(n) == 1 \\
6: \quad \quad \quad \quad \quad \text{TTL}_{\text{Current}} = \text{TTL} \\
7: \quad \quad \quad \text{TTL} = \text{TTL} + 1 \\
8: \quad \quad \quad \text{resend RREQ}(n) \\
9: \quad \quad \text{if } \text{RREP}(n) \text{ is received} \\
10: \quad \quad \quad \text{resend CALL\_REGISTRATION}(n) \\
11: \quad \quad \quad \text{TTL} = \text{TTL}_{\text{Current}} \\
12: \quad \quad \quad \text{go to } \text{CALL\_REGISTRATION}(n + 1) \\
13: \quad \quad \text{else continue the next iteration in the for loop} \\
14: \quad \text{else return} \\
15: \quad \quad \text{else if } t > t_{rw} \\
16: \quad \quad \quad \text{resend CALL\_REGISTRATION}(n) 
\]
Figure 5-12: Flowchart representation for Algorithm 5.4
5.7.1.2 CLAODV for the Call Setup Process

The call setup performance depends on the SIP initiation messages that are sent and received by the call entities. In the CLAODV approach for MANET, the SRD is calculated for all calls initiated by the caller node UAC (A). The call initiation signaling passes through the SIP server, where all messages and parameters for INVITE messages are recognized by the SIP server and the caller to evaluate the SIP signaling performance, as represented in Figure 5-10. The call setup time is the time difference between TIn1 and TA3, where TA3 is the time of receiving the call acceptance acknowledgment 200 OK by the callee UAC (B). The SRD in this implementation is the time difference between the INVITE message, sent at time TIn1, and Tx3, the time at which the callee’s response message will be received by the UAC (A) for the call invitation, as represented in equation (4.3). The SIP call setup intervals will be evaluated as well for normal AODV-based MANET in Section 6.3.3. For the CLAODV approaches, the three-way handshake system is applied for the call setup, as it represents the basic SIP signaling flow between the call entities, and the SRD SIP performance metric can applied. The proposed approaches are applied over the UAC (A) and the UAS (SIP Server). When any of the SIP call parties recognize a delay, the CLAODV approach is triggered to enhance the routing performance using Cross-Layer messages with the network layer to modify the AODV routing parameters based on the analysed call setup performance metrics in the application layer, as represented in Figure 5-11.

The proposed algorithm in this section considered the SRD_BenchM parameter which is the benchmark value for SRD that determines and evaluates the call setup time performance over MANET. The SRD_BenchM value can even be used as a specific input value, or it can be found during the run time of the system. Because there are no approved benchmark values for the RFC 6076 performance metrics in general and especially for MANET, therefore this research study will evaluate and find the SRD values as part of the benchmark efforts in Section 6.3.1.2. The SRD values that will be employed for the SRD_BenchM are determined for both IPv4 and IPv6 traffic by the proposed benchmarked sets for the SIP performance metrics in Section 8.2 and Table 8-1. During the implementations, the SRD values of a call setup process for a SIP call number \( n \) is calculated using equation (5.3), and Algorithm 5.2. During the call setup session, the SRD values of caller UAC (A) will be compared with a determined SRD_BenchM. If the SRD is greater than SRD_BenchM, then CLAODV will be triggered to update the routing table by resending the RREQ messages with a longer TTL period, as represented in Algorithm 5.2. This process will be triggered for a short time until a RREP is received to provide the call setup message with the ability to reach its destination with an active updated route based on the AODV route discovery algorithm. When the RREP is received, a re-INVITE message will be resent, and the TTL value for the RREQ will degrade to save both the CPU cycles and the bandwidth. For the SIP call number \( n \), the sequence number for the call setup process has the same sequence number \( n \), where \( t \) is the simulation time in seconds, \( T_{Call\ Setup\ Wait} \) is the maximum specified time in seconds.
before sending the re-INVITE message, and T_END is the time that the simulation ends. The UAS uses
the signaling timers for INVITE messages and responses that are forwarded by the UAS between the
call ends during the call setup process. The UAS can observe the call setup performance for INVITE
messages between both UAC (A) and UAC (B) by investigating T_INT2, TR2, TX2, and TA2. The UAS
can determine the SRD values for the UAC (A) call setup process from the sequence numbers and time
stamps of the INVITE messages and its acknowledgments. When a delay is detected, the UAS initiates
the CLAODV process to adjust the routing parameters with the required level that allows the routing
table to be updated with active routes as represented in Algorithm 5.5 and the flowchart in Figure 5-13
for the UACs, and Algorithm 5.6 and the flowchart in Figure 5-13 for the UAS.

Algorithm 5.5: The Call setup process for the Cross-Layer AODV. The Implementation of this
algorithm is on the UAC(R(n)) for SIP call number n where R is the User Agent Client A.

CALL_SETUP_UAC(n)
Inputs: SRD_{BM}, t_{rw}, t_{end}
1: for t = 0, until t \leq t_{end}, with step t = t + 1
2: if INVITE(n) is sent by the UAC(n)
3: if RINGING_{180}(n) is received
4: if T_INT2(n) > SRD_{BM} or SRD(n) > SRD_{BM} or RERR(n) == 1
5: TTL_{Current} = TTL
6: TTL = TTL + 1
7: if RREP(n) is received
8: resend RREQ(n)
9: resend INVITE(n)
10: return
11: else continue the next iteration in the for loop
12: else return
13: else if t > t_{rw}
14: resend INVITE(n)

Algorithm 5.6: The Call setup process for the Cross-Layer AODV. The Implementation of this
algorithm is on the UAS(R(n)) for SIP call number n where R is the User Agent Client B.

CALL_SETUP_UAS(n)
Inputs: SRD_{BM}, t_{rw}, t_{end}
1: for t = 0, until t \leq t_{end}, with step t = t + 1
2: if RINGING_{180}(n) is received for INVITE(n)
3: if T_INT2(n) > SRD_{BM} or TR2(n) > SRD_{BM} or TX2(n) > SRD_{BM} ... or RERR(n) == 1
4: TTL_{Current} = TTL
5: TTL = TTL + 1
6: resend RREQ(n)
7: if RREP(n) is received
8: request new INVITE(n)
9: TTL = TTL_{Current}
10: return
11: else continue the next iteration in the for loop
12: else return
Figure 5-13: Flowchart representation for Algorithm 5.5
The CLAODV algorithm provides a reliable real-time detection system for the call setup delays and undeliverable SIP messages as it relays the SRD performance metric to adjust the level of the routing update values. In addition, it provides a self-adjustment mechanism for SIP signaling instead of feeding...
the system with the required actions to enhance the call setup performance over MANET. The SRD values can be determined by the caller node with continuous investigations to obtain self-evaluation records for SRD values during the application running time. This method can enhance the call setup performance with the dynamic reachability process to other nodes to reduce the connectivity delays, save the CPU cycles, and reduce the bandwidth. If the RREP message is not received during this approach, the call setup performance will not be enhanced, as additional delays may result because the algorithm needs to reuse the last known routing to send the re-INVITE message. The CPU cycles increase as the level of RREQ messages increases in order to update the routing tables, which also increases the nodes’ consumed power for routing purposes; bandwidth consumption also increases.

5.7.1.3 CLAODV for the Call Termination Process

The performance of the SIP termination process depends on the BYE message of the SIP signaling system that is sent by one of the calling parties during the SIP call by the UACs and through the B2BUA SIP Server. The CLAODV approach for MANET considered the SDD values from the SIP performance metrics that can be used to enhance the performance of the termination process through adjusting the route discovery values. The adjustment process is based on the benchmarked SDD values that represent the termination intervals of the SIP calls, as will evaluated in Section 6.3.1.3. For the CLAODV approach of the termination process, the SDD values are calculated by the caller nodes using the identified timers for SIP call number \( n \) on the UAC that requested the session termination, as represented in Figure 5-10. The termination time is the time difference between TD1 of the initiated BYE message by the UAC that requested the session termination and TK3 for the initiated 200 OK response message for the BYE message. The SDD values of a termination process for a SIP call number \( n \) is calculated using equations (5.4) and Algorithm 5.7. The proposed CLAODV approach for the SIP termination process is applied on the UAC that requested the session termination, which is in this case is UAC (A). Once the UAC (A) recognized a termination delay, the CLAODV approach will be triggered to modify the AODV routing parameter using the Cross-Layer messages based on the investigated SIP performance metrics at the application layer, as represented in Figure 5-11.

The CLAODV algorithm considered the SDD_BenchM parameter as the benchmark value for SDDs that will be used to determine the termination process of the SIP call over MANET. The SDD_BenchM value will be used as a definite input value from the investigated SDD values during the benchmark efforts in Section 6.3.1.3 that are determined by the Static and Uniform mobility models for the best case scenarios. The SDD values for the SDD_BenchM are determined for both IPv4 and IPv6 traffic by the proposed benchmarked sets for the SIP performance metrics, as in Section 8.2 and Table 8-1. During the termination session, the SDD values for both callers will be examined with the determined SDD_BenchM. The CLAODV will be triggered to update the parameters of the routing table by
resending the RREQ messages when the SDD values are greater than the SDD_BenchM. An update for the TTL values with a longer period will be applied as represented in Algorithm 5.7 and the flowchart in Figure 5-15. The CLAODV process will be triggered for a short time until a RREP message is received to provide the BYE message with an active updated route based on the AODV route discovery algorithm. When the RREP message is received, a BYE message will be resent. Furthermore, the TTL values of the RREQ will constantly degrade to the original values that help with CPU performance and bandwidth conservation. The sequence number for the termination process uses the same sequence number \( n \) of the SIP call. The simulation time \( t \) is represented in seconds, \( T_{\text{Termination}} \) is the maximum specified time in seconds before resending the BYE message, and \( T_{\text{End}} \) is the simulation end time.

Algorithm 5.7: The call termination process for the Cross-Layer AODV. The Implementation of this algorithm is on the \( UAC(R(n)) \) for SIP call number \( n \), where \( R \) is the User Agent Client \( A \) or the User Agent Client \( B \).

\[
\text{CALL\_TERMINATION}(n)
\]

\begin{align*}
\text{Inputs:} & \quad \text{SDD}_{BM}, t_{rw}, t_{end} \\
1: \quad & \text{for } t = 0, \text{ until } t \leq t_{end}, \text{ with step } t = t + 1 \\
2: \quad & \text{if } \text{BYE}(n) \text{ is sent by the } UAC(n) \\
3: \quad & \quad \text{if } \text{TK3}(n) \text{ is received for } \text{BYE}(n) \\
4: \quad & \quad \quad \text{SDD}(n) = \text{TK3}(n) - \text{TD1}(n) \\
5: \quad & \quad \text{else if } \text{TK3}(n) > \text{SDD}_{BM} \text{ or } \text{SDD}(n) > \text{SDD}_{BM} \text{ or } \text{RERR}(n) == 1 \\
6: \quad & \quad \quad \text{TTL\_Current} = \text{TTL} \\
7: \quad & \quad \quad \text{TTL} = \text{TTL} + 1 \\
8: \quad & \quad \quad \text{resend } \text{RREQ}(n) \\
9: \quad & \quad \text{if } \text{RREP}(n) \text{ is received} \\
10: \quad & \quad \quad \text{resend } \text{BYE}(n) \\
11: \quad & \quad \text{if } \text{RREP}(n) \text{ is received} \\
12: \quad & \quad \quad \text{TTL} = \text{TTL\_Current} \\
13: \quad & \quad \quad \text{go to } \text{CALL\_TERMINATION}(n + 1) \\
14: \quad & \quad \text{else return} \\
15: \quad & \quad \text{else if } t > t_{rw} \\
16: \quad & \quad \quad \text{resend } \text{BYE}(n)
\end{align*}
Figure 5-15: Flowchart representation for Algorithm 5.7

<table>
<thead>
<tr>
<th>Condition</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>$t &lt; t_{rw}$</td>
<td>Yes: resend BYE(n); No: $t = 0$</td>
</tr>
<tr>
<td>$t &lt; t_{end}$</td>
<td>Yes: $t = t + 1$; No: $t_{end}$</td>
</tr>
<tr>
<td>BYE(n) is received</td>
<td>Yes: terminator(n+1); No: resend REQ(n)</td>
</tr>
<tr>
<td>TK3(n) is received</td>
<td>Yes: $TTL_{current} = TTL(t)$; No: SDD(n) &gt; SDD_{host}</td>
</tr>
<tr>
<td>TK3(n) &gt; SDD_{host}</td>
<td>Yes: resend REQ(n); No: RERR(n) = 1</td>
</tr>
<tr>
<td>SDD(n) &gt; SDD_{host}</td>
<td>Yes: $TTL_{current} = TTL(t)$; No: SDD(n) = TK3(n) - TD1(n)</td>
</tr>
<tr>
<td>TTL_{current} = TTL(t)</td>
<td>Yes: terminator(n+1); No: resend BYE(n)</td>
</tr>
<tr>
<td>terminator(n+1)</td>
<td>Yes: terminator(n+1); No: resend BYE(n)</td>
</tr>
<tr>
<td>END</td>
<td>Yes: terminator(n+1); No: resend BYE(n)</td>
</tr>
</tbody>
</table>
5.7.2 Cross-Layer Algorithms for OLSR-based MANET

The proactive nature for the OLSR routing protocol is adding extra overhead routing traffic over the OLSR-based MANET, especially with real-time applications. This problem makes the design and implementations of the Cross-Layer approaches for OLSR-based MANET (CLOLSR) more challenging with low levels of performance enhancement when compared with the proposed CLAODV for AODV-based MANET. The basic design and operational mechanisms for OLSR had been discussed in Chapter 2, Section 2.12. The routing parameters that are considered for the adjustment and modifications within the proposed CLOLSR depend on the performance enhancement review for OLSR at Chapter 2, Section 2.12.4. In addition, the design of the CLOLSR considered the evaluation results for the SIP-based VoIP over OLSR-based MANET at Chapter 6. The CLOLSR considered the values of the HELLO messages and the Topology Control (TC) intervals to modify the values of the Multiple Point Relay (MPR) of the OLSR routing protocol. The MPR values increased by the increments of the HELLO messages and the TC intervals’ values with the considerations of the OLSR MPR selection mechanism in Algorithm 2.3. Figure 5-16 represents the CLOLSR over MANET OSI for all performance enhancement approaches of the investigated SIP processes. The proposed CLOLSR algorithms are not applicable to be implemented over MANET using other routing protocols unless they share the routing concepts and parameters. In this section, the design of the CLOLSR algorithms will be introduced to enhance the SIP signaling processes over OLSR-based MANET. Furthermore, the CLOLSR will be modelled, implemented, and evaluated by the OPNET® Modeler.

| Application Layer | SIP |
| Transport Layer   | TCP |
| Network Layer     | OLSR | Adjust: HELLO, TC Interval, MPR |

CLOLSR Examinations for SIP processes 1, 2, or 3:
1. Registration: RRD1, RRD2, REGISTER, 200 OK
2. Initiation: SRD, INVITE, 200 OK
3. Termination: SDD, BYE, 200 OK

Figure 5-16: CLOLSR representation over MANET OSI for the proposed performance enhancement approaches of the SIP processes

5.7.2.1 CLOLSR for the Call Registration Process

The REGISTER messages of the SIP signaling flow are considered for the performance enhancement approach of the SIP registration processes. The REGISTER messages are used to register the UACs with the SIP server as discussed in Chapter 4, Section 4.2.4. The SIP performance metrics are employed to enhance the performance of the registration processes by adjusting the routing parameters of the OLSR to the required level. The RRD values of the SIP performance metrics are employed within the
CLOLSR approach to enhance the performance of the registration process by adjusting the route discovery values to the required level depending on the benchmarked values of the RRD which will be investigated in Section 6.3.1.1. Furthermore, the registration intervals of each SIP call will be evaluated as well for OLSR-based MANET in Section 6.3.2. The RRD values in the CLOLSR are calculated for each SIP call using the identified timers of the SIP call on the UAC (A) and UAC (B) sides, as demonstrated in Figure 5-10. As identified in Section 5.5, the total registration time is the time interval between the initiated REGISTER messages of UAC (A) (R1A) or UAC (B) (R1B) and the 200 OK response message of the REGISTER message for the UAC (A) (R4A) or UAC (B) (R4B). Therefore, the CLOLSR considered the RRD values of the registration process for the related SIP call and calculated its values using equations (5.1) and (5.2), and Algorithm 5.1 as discussed earlier in this Chapter. The CLOLSR approach for the registration process is applied on the callers’ side: the UAC (A) and UAC (B). Once a delay is recognised by the UAC (A) or UAC (B), the CLOLSR approach will be triggered to adjust the OLSR routing parameters using the Cross-Layer messages. The adjustments process for the OLSR routing values are based on the investigated SIP performance metrics that passed from the application layer, as represented in Figure 5-12.

The CLOLSR algorithm considered the RRD_BenchM parameter as the RRD benchmarked values that will be used to determine the registration processes of the SIP calls over MANET. The values of the RRD_BenchM had been determined from the evaluated RRD values of the benchmark efforts in Section 6.3.1.1 by the evaluation efforts of the Static and Uniform mobility models of the best case scenarios. The RRD values that will be used for the RRD_BenchM will be determined for both IPv4 and IPv6 traffic by the proposed benchmark sets for the SIP performance metrics, as in Section 8.2 and Table 8-1. During the registration session, the sequence number for the registration process has the same sequence number as the SIP call number n where X refers to the current UAC node. The RRD values for both the caller UAC (A) and UAC (B) will be compared with the considered RRD_BenchM from the benchmarked set during the registration sessions. When the current RRD value becomes greater than the RRD_BenchM value, then the CLOLSR will update the routing table parameters by resending the HELLO messages using longer time periods for the Topology Control (TC) values, as represented in Algorithm 5.8 and the flowchart in Figure 5-17. The effectiveness of having longer TC intervals over the OLSR routing performance was discussed in Section 2.12.4. The CLOLSR process will be activated temporarily for a short period of time until the MPR values of the second hop neighbours (MPR_2) for the current node (X) become more than the values of the first hop neighbours (MPR_1) for the same node. This adoption mechanism is based on the OLSR MPR selection mechanism that is represented in Algorithm 2.3. Then when the MPR_2 received for node (X) and for the current registration process and its value is greater than the current MPR_1 value, the REGISTER request will be sent again and the TC value will be degraded to its original value to save the CPU cycles and reduce the bandwidth consumption. The proposed algorithm used the simulation time t which is represented in seconds,
T\_Registration\_Wait which is the maximum specified time in seconds before sending the re-
REGISTER message, and T\_End is the simulation end time. The total registration time for a registration
process is the total of RRD1 and RRD2, or it is the time difference between the first registered UAC of
the call parties with the SIP Server and the last received registration response from the call parties.

\textbf{Algorithm 5.8:} The call registration process for the Cross-Layer OLSR. The Implementation of this
algorithm is on the \textit{UAC}(\textit{R(n)}) for SIP call number \textit{n}, where \textit{R} is the User Agent Client \textit{A} or the User
Agent Client \textit{B}.

\texttt{REGISTER}(\textit{n})

\textbf{Inputs:} \texttt{RRD\_BM, t\_rw, t\_end}

\begin{algorithmic}[1]
\State \textbf{for} \textit{t} = 0, until \textit{t} \leq \textit{t\_end}, with step \textit{t} = \textit{t} + 1
\State \textbf{if} \texttt{REGISTER}(\textit{n}) is sent by the \textit{UAC}(\textit{n})
\State \quad \textbf{if} \textit{OK}_{200}(\textit{n}) is received \textbf{and} \texttt{REGISTER}(\textit{n}) is received
\State \quad \quad \texttt{RRD}(\textit{n}) = \texttt{R4}(\textit{n}) - \texttt{R1}(\textit{n})
\State \quad \textbf{else if} \textit{R3}(\textit{n}) > \texttt{RRD\_BM} \textbf{or} \texttt{RRD}(\textit{n}) > \texttt{RRD\_BM} \textbf{... or} \texttt{TC} \equiv 0 \textbf{or} \texttt{MPR1}(\textit{X}, \textit{n}) \geq \texttt{MPR2}(\textit{X}, \textit{n})
\State \quad \quad \texttt{TC\_Current} = \textit{TC}
\State \quad \quad \texttt{TC} = \textit{TC} + 1
\State \quad \textbf{resend HELLO}(\textit{n})
\State \quad \textbf{if} \texttt{MPR2}(\textit{X}, \textit{n}) is received \textbf{and} \texttt{MPR2}(\textit{X}, \textit{n}) > \texttt{MPR1}(\textit{X}, \textit{n}) \textbf{... and} \texttt{R}(\textit{n})! = \texttt{R}(\textit{n} + 1)
\State \quad \quad \textbf{resend REGISTER}(\textit{n})
\State \quad \texttt{TC} = \texttt{TC\_Current}
\State \quad \textbf{go to REGISTER}(\textit{n} + 1)
\State \quad \textbf{else return}
\State \textbf{else return}
\State \textbf{else if} \textit{t} > \textit{t\_rw}
\State \quad \textbf{resend REGISTER}(\textit{n})
\State \textbf{go to REGISTER}(\textit{n} + 1)
\State \textbf{else go to REGISTER}(\textit{n} + 1)
\end{algorithmic}
Figure 5-17: Flowchart representation for Algorithm 5.8
5.7.2.2 CLOLSR for the Call Setup Process

The performance of the SIP call initiation processes depend on the call setup messages that are interchanged between the call entities. For the CLOLSR approach over MANET, the SRD values are calculated for all calls that are initiated by the caller node UAC (A) as represented in Figure 5-10. The call initiation messages go through the SIP server and all the INVITE messages and parameters are recognised by the SIP server and the caller for the evaluations of the SIP signaling performance. The time difference between TInt1 and TA3 is considered as the call setup time where the SRD value is the time difference between the INVITE message, sent at time TInt1, and Tx3, as represented in equation (4.3) in Chapter 4. The SIP call setup intervals will be evaluated for OLSR-based MANET in Section 6.3.3. The CLOLSR approaches had considered the three-way handshake system for the call setup process between the SIP call entities. The SRD SIP performance metrics are also will be applied in the CLOLSR. This proposed approach is applied over the UAC (A) and the UAS (SIP Server). Whenever a delay is recognized by any of the SIP call parties, the CLOLSR approach will be activated to optimise the routing performance using the Cross-layer messages to modify the OLSR routing parameters based on the analysed performance metrics of the call setup process in the application layer, as represented in Figure 5-12.

The call initiation processes of the SIP-based VoIP implementation over OLSR have very long delays over RWP and RWP-All mobility models as discussed in Section 6.3.3. Therefore, the proposed CLOLSR algorithm is designed to determine and optimise the call setup performance over OLSR based on the evaluation values of SRD parameters in Section 6.3.1.2. The design of the CLOLSR algorithm considered the SRD_BenchM values to determine the proposed benchmarked sets of the SIP performance metrics, as in Section 8.2 and Table 8-1. In the proposed algorithm, the actual SRD value for a call setup process of the SIP call number $n$ is calculated using equations (5.3), and Algorithm 5.2. The SRD values of the caller agent UAC (A) is compared with the SRD_BenchM values within the SIP call setup process for the evaluation purposes. Once the SRD value of the current call is recognised by the CLOLSR that it is greater than the SRD_BenchM values, the routing update will be activated by sending the HELLO messages with longer intervals to support the Topology Control (TC) mechanisms of the MANET with the active routes, as represented in Algorithm 5.9 and the flowchart in Figure 5-18 for the UACs. The MPR values of the second hop neighbours (MPR_2) for the current node (X) will be updated with the received routing replies messages. Therefore, the MPR_2 values become more than the values of the first hop neighbours (MPR_1) for the same node. The performance of OLSR and the effectiveness of modifying the MPR selection mechanism have been discussed in Section 2.12.4 and represented in Algorithm 2.3. Once the updated MPR_2 is received for node (X) and for the current call setup process with values greater than the current MPR_1 value, the INVITE request will be resent again as a re-INVITE message and the TC values will be degraded to their previous values to save the
CPU cycles and reduce the bandwidth consumption. On the other hand, the UAS depends on the signaling timers for the INVITE messages and responses that are exchanged between both callers during the call setup processes. The UAS is also able to control the call setup performance for the INVITE messages between both UAC (A) and UAC (B) by examining the T|nt2, TR2, Tx2, and TA2 parameters. Therefore, the UAS can determine the SRD values for the UAC (A) call setup process from the sequence numbers and time stamps of the INVITE messages and their acknowledgments. Once a delay is detected, the UAS activates the performance enhancement approach to adjust the routing parameters with the required level to update the routing table with the required active routes as represented in Algorithm 5.10 and the flowchart in Figure 5-19. In these two proposed algorithms for the call setup processes, the X refers to the current UAC or UAC node. The SIP call number is n which also represents the sequence number of the call setup process. The simulation time is t in seconds, the T_Call_Setup_Wait is the maximum specified time in seconds before sending the re-INVITE message, and T_End is the simulation termination time.

**Algorithm 5.9**: The call setup for the Cross-Layer OLSR. The Implementation of this algorithm is on the UAC (R(n)) for SIP call number n where R is the User Agent Client A.

```plaintext
CALL_SETUP_UAC(n)
Inputs: SRD_BM, trw, t_end
1: for t = 0, until t <= t_end, with step t = t + 1
2: if INVITE(n) is sent by the UAC(n)
3: if RINGING180(n) is received
4: SRD(n) = TX3(n) - T INT1(n)
5: else if TR3(n) > SRD_BM or SRD(n) > 0 or MPR1(X, n) >= MPR2(X, n)
   ... or TC == 0 or MPR1(X, n) >= MPR2(X, n)
6: T_CCurrent = TC
7: TC = TC + 1
8: resend HELLO(n)
9: if MPR2(X, n) is received and MPR2(X, n) > MPR1(X, n) ... and R(n)! = R(n + 1)
   resend INVITE(n)
10: TC = T_CCurrent
11: go to CALL_SETUP_UAC(n + 1)
12: else return
13: else return
14: else if t > trw
15: resend INVITE(n)
16: go to CALL_SETUP_UAC(n + 1)
17: else go to CALL_SETUP_UAC(n + 1)
```
Algorithm 5.10: The call setup for the Cross-Layer OLSR. The Implementation of this algorithm is on the \( UAS(R(n)) \) for SIP call number \( n \) where \( R \) is the User Agent Client \( B \).

\begin{verbatim}
CALL_SETUP_UAS(n) Inputs: SRD_BM, t_rq, t_end
1: for t = 0, until t \leq t_end, with step t = t + 1
2: if RINGING\(_{180}(n)\) is received for INVITE(n)
3: if T\(_{INT}\)(n) > SRD\(_{BM}\) or TR2(n) > SRD\(_{BM}\) or TX2(n) > SRD\(_{BM}\) ...
   ... or TC == 0 or MPR1(X, n) \geq MPR2(X, n)
4: TC\(_{Current}\) = TC
5: TC = TC + 1
6: resend HELLO(n)
7: if MPR2(X, n) is received and MPR2(X, n) > MPR1(X, n) ...
   ... and R(n)! = R(n + 1)
8: request new INVITE(n)
9: TC = TC\(_{Current}\)
10: go to CALL_SETUP_UAS(n + 1)
11: else return
12: else return
13: else go to CALL_SETUP_UAS(n + 1)
\end{verbatim}

The proposed CLOLSR algorithm for the call setup process designed to provide a reliable detection system for any delays and undeliverable SIP messages that will take place during this process. The CLOLSR approach considers the SRD performance metric to adjust the level of the routing update values. This approach provides a self-adjustment mechanism for the SIP signaling instead to enhance the call setup performance over MANET. In addition, the SRD values will be employed to be evaluated by the caller nodes and the UAS with active monitoring during the call setup processes to enhance the performance. Whenever the MPR values not updated during this approach, the call setup performance will not be enhanced where additional delays occur, because the algorithm values need to be recovered with their previous values to complete the re-invitation process. In addition, the CPU cycles increase with the increase in the number of HELLO messages to update the routing table entries which also increases the bandwidth consumption with inactive updates for the routing tables.
Figure 5-18: Flowchart representation for Algorithm 5.9
Figure 5-19: Flowchart representation for Algorithm 5.10
5.7.2.3 CLOLSR for the Call Termination Process

The termination performance of a SIP-based VoIP call depends on the transportations of the BYE messages that sent from one of the callers during the call and go through the B2BUA SIP Server. The termination mechanism had been discussed and evaluated in Section 4.2.6 and will be considered in Section 8.3. The CLOLSR approach considered the SDD values during the performance enhancement for the call termination process to provide a dynamic adjustments for the route discovery values. The original existing SDD values had been evaluated in Section 6.3.1.3 and considered for the suggested the benchmark values in this section. The SDD values are determined by the caller nodes using the identified timers of the SIP call number \( n \) on the UAC that requested the session termination as illustrated in Figure 5-10. The SDD values for a SIP call number \( n \) is calculated using equations (5.4), and Algorithm 5.3 at Chapter 5. For the call termination process, the SDD values is the time difference between TD1 which is the time of sending the BYE message by the UAC, and TK3 which is the time of receiving the 200 OK response message of the BYE message. During the call session, the CLOLSR system observers the call termination processes between the caller agents. Once a delay being recognised by the UAC (A), the performance enhancement will be activated to adjust the routing parameters using the Cross-Layer messages based on the identified SIP performance metrics at the application layer as shown in Figure 5-12.

For the performance enhancement process of the call termination, the CLOLSR algorithm considered the SDD_BenchM parameter as the benchmarked values of the SDDs that used to determine any existing delays as represented in Algorithm 5.8. The SDD_BenchM values considered from the values of the investigated scenarios during the benchmarking efforts in Section 6.3.1.3. In addition, the SDD_BenchM values determined for both IPv4 and IPv6 implementations using the proposed sets of the benchmarked SIP performance metrics in Section 8.2 and Table 8-1. During the termination process, the sequence number for the registration process has the same sequence number the SIP call number \( n \). The CLOLSR will be activated to update the routing table parameters by regenerating the HELLO messages when the SDD values are greater than the SDD_BenchM. An update for the TC values using longer intervals will be applied as shown in Algorithm 5.11 and the flowchart in Figure 5-20. This activation for the CLOLSR process works temporarily until the MPR values for the second hop neighbours (MPR_2) of the current node (X) become more than the values of the first hop neighbours (MPR_1) of the same node (X). This modification mechanism of the MPR selection considers the basic MPR routing system in Algorithm 2.3. Once the MPR_2 values updated for node (X) with the current termination process with the required value, the BYE request will sent again and the current TC value for node (X) will be degraded to its original value to save the CPU cycles and reduce the bandwidth consumptions. The termination process sequence number is the same sequence number for the SIP call.
The simulation time \( t \) is represented in seconds, \( T_{\text{Termination\_Wait}} \) is the maximum specified time in seconds before resending the BYE messages, and \( T_{\text{End}} \) represents the simulation end time.

**Algorithm 5.11**: Call termination for Cross-Layer OLSR. The Implementation of this algorithm is on the \( UAC(R(n)) \) for SIP call number \( n \), where \( R \) is the User Agent Client \( A \) or the User Agent Client \( B \).

\[
\text{CALL\_TERMINATION}(n)
\]

\textbf{Inputs:} \( SDD_{BM}, t_{rw}, t_{end} \)

\begin{enumerate}
  \item for \( t = 0 \), until \( t \leq t_{end} \), with step \( t = t + 1 \)
  \item if \( BYE(n) \) is sent by the \( UAC(n) \)
  \item if \( TK3(n) \) is received for \( BYE(n) \)
  \item \( SDD(n) = TK3(n) - TD1(n) \)
  \item else if \( TK3(n) > SDD_{BM} \) or \( SDD(n) > SDD_{BM} \) …
    \quad \text{or } TC == 0 \text{ or } MPR1(X, n) \geq MPR2(X, n)
  \item \( TC_{\text{Current}} = TC \)
  \item \( TC = TC + 1 \)
  \item resend HELLO\((n)\)
  \item if \( MPR2(X, n) \) is received and \( MPR2(X, n) > MPR1(X, n) \) …
    \quad \text{and } R(n)! = R(n + 1)
  \item resend \( BYE(n) \)
  \item \( TC = TC_{\text{Current}} \)
  \item go to CALL\_TERMINATION\((n + 1)\)
  \item else return
  \item else if \( t > t_{rw} \)
  \item resend \( BYE(n) \)
  \item go to CALL\_TERMINATION\((n + 1)\)
  \item else go to CALL\_TERMINATION\((n + 1)\)
\end{enumerate}
Chapter 5. Research Framework and Methodology

Figure 5-20: Flowchart representation for Algorithm 5.11
5.8 Analysis Methods for the Research Study

The evaluation studies for SIP-based VoIP applications will be used to determine the SIP signaling performance for IPv4 and IPv6 for both AODV and OLSR over four different mobility models. Both Static and Uniform mobility models will be used to represent the best effort scenarios that will be used to compare the performance of the SIP signaling over MANET with the performance of other platforms, such as LAN, WLAN, Satellite, and WiMAX. This step will indicate the accuracy level of the evaluation works of this research study as the comparison efforts should meet with the expected level of similarity with other network systems, as provided in [151, 149, 210, and 212].

The RWP and RWP-All mobility models are expected to have higher levels of difference in their results when compared with the Static and Uniform mobility models. Thus, it will be hard to link these differences with other platforms of network systems as the communication natures are different and simulation parameters are not fully matched with the identified parameters in this research study. Therefore, this research study will focus on the evaluation works on OPNET® Modeler for SIP-based VoIP over AODV and OLSR to benchmark the end-to-end SIP performance metrics and the SIP signaling performance for the current state of the art. Then, these values will be used as a reference to compare them with the performance values of the proposed algorithms to determine the enhancement level that could be provided. The comparison will depend on different parameters for both SIP signaling and the MANET routing protocol. The evaluation works will analyse the results and show the differences over different mobility models and routing protocols based on the graphical representations and the benchmark values of the simulation results. In addition, the results analysis will consider the performance of both the B2BUA-based SIP server and the SIP callers to provide comprehensive investigations for the SIP signaling system.

5.9 Summary

The framework of this research study provides measurable implementations for SIP-based VoIP over both AODV and OLSR MANETs. This is due to the medium size of MANET system that is used in the framework, which is not limited to a few MANET nodes in the simulation scenarios. Furthermore, the network and information security issues are out of the scope of this research study. The methodology of this research is focusing on the evaluation studies for SIP signaling performance and benchmarking the end-to-end SIP performance metrics. Based on the evaluation study that will be implemented in Chapter 6, a performance enhancement approach will be introduced and evaluated. Therefore, the research study depends on the simulation works to determine the performance level for the SIP signaling over MANET that will determine the required enhancements for the SIP signaling system. In addition, this chapter introduced the design and implementation of the required SIP end-to-end performance
metrics for this research study. These performance metrics will be used during this research study for the evaluation, benchmarking, and the performance enhancement efforts that will be discussed in the next chapters. The proposed Cross-Layer approaches for the SIP signaling performance enhancements are represented and explained in this Chapter.
Chapter 6

6 QoS and Performance Evaluation for SIP-based VoIP over MANET

In this chapter, analyses for the performance metrics that define the quality of service (QoS) of SIP-based VoIP will be introduced. SIP-based VoIP applications over MANET systems have three main performance categories related to the QoS. These categories are the SIP signaling, voice data transmission, and MANET routing. The SIP signaling controls the VoIP calls initiation, termination, and modifications. The major QoS parameters of VoIP that are managed by SIP signaling are the registration intervals, call setup time, and call termination time. These QoS parameters are increased in MANET due to the nodes’ mobility that affects the routing calculations and the connectivity status. These necessitate mechanisms to reduce the delays in the MANET environment. The voice packets are transferred over the Real Time Protocol (RTP) which is encapsulated in the unreliable transport protocol using the User Datagram Protocol (UDP). In addition, the bandwidth consumption, delays, jitter and packet loss are QoS parameters that quantify VoIP performance. The bandwidth is dependent on the codec system used. The delays experienced by voice packets are one-way delays between the two calling ends and are affected by the routing and connectivity delays in the MANET environment. The jitter is related to the variations in the delay and the RTP tries to recover the loss for it, while the packet loss is related to the network congestion and erroneous links. A number of studies have been undertaken for these performance metrics that support the evaluation studies.

In this chapter, the simulation efforts have been carried out using GSM voice codecs to evaluate SIP call processes and QoS parameters together over MANET. The evaluation used to implement different scenarios for SIP-based VoIP applications over both AODV and OLSR, for both IPv4 and IPv6, was determined in Chapter 5. The SIP signaling and QoS parameters for VoIP have been assessed on the OPNET® Modeler simulation scenarios. The simulation efforts have not considered other simultaneous applications that could influence the performance of the SIP applications to provide the effort implementations. However, the assumptions considered a background traffic with 30% to 40% of the overall bandwidth.

In this evaluation study, the results for both Static and Uniform mobility models are representing the best effort of the implemented scenarios. The results for the Static and Uniform scenarios are meeting
with the evaluation results for similar scenarios over other network systems which support the reliability level of the findings of this research study. Hence, the investigated QoS parameters are considered as the benchmark values for SIP-based VoIP over AODV and OLSR MANET. These benchmarking efforts also considered the Registration Request Delays (RRD), Session Request Delays (SRD), and the Session Disconnect Delay (SDD) of the RFC 6076, as introduced in section 4.2.3. These main SIP end-to-end metrics had been implemented in this research study to provide an evaluation for the SIP signaling over VoIP application between the SIP calls’ entities as represented in Figure 4-2. The results show that these parameters are comparable for both IPv4 and IPv6 in AODV and OLSR MANET environments. Furthermore, the simulation efforts in this research study used to design and implement number of important parameters for SIP signaling performance evaluation as represented in Chapter 5, Section 5.5. These parameters are the RRD, SRD, and RDD from the RFC 6076 together with the parameters of the call registration delay, and the call setup delay.

6.1 Relevant Research Efforts

In general, limited numbers of researchers have studied and evaluated the performance of real-time applications over MANET. Most of the evaluation efforts considered Constant Bit Rate (CBR) or File Transfer Protocol (FTP) traffic with a different number of MANET nodes. For IPv4 MANET, a performance evaluation with OPNET® Modeler v14.5 for the reactive routing protocols, AODV and DSR, using GSM voice traffic, concluded that AODV has the lowest end-to-end delay and a lower network load compared with DSR [210]. Furthermore, AODV presents higher average throughput and received traffic while DSR does not scale well with large sized networks. Simulation results also showed that AODV reactive routing protocol is the best suited for MANET, while DSR recorded very poor QoS in MANET with high node capacity for GSM voice applications. However, this research did not consider VoIP applications with different mobility models [209]. An evaluation for AODV as a reactive routing protocol and OLSR as a proactive routing protocol in OPNET® Modeler with a variable number of MANET nodes concluded that the performance of routing protocols vary depending on the network type and the selection of accurate routing protocols that affect the application’s efficiency [209]. In [212], three RWP-based mobility models have been implemented for MANET reactive routing protocols over different performance parameters. The results show that AODV in the RWP Waypoint mobility model performs better than TORA and DSR in the RWP walk and RWP direction mobility models. The study also concluded that AODV can be used with intensive mobility models. In [216], a performance comparison of selected MANET routing protocols has been performed in varying network sizes with increasing area and numbers of nodes to investigate RWP mobility and scalability of the routing process with high CBR traffic flow. AODV performed very consistently and established quick connection between nodes without delays while TORA conceded high end-to-end delay due to the
formation of temporary loops within the network. AODV offers the best efficiency with high traffic applications compared to OLSR and TORA.

In [222], an evaluation study used to compare different performance parameters between the most popular routing protocols in MANET: OLSR (proactive), TORA (reactive), and Geographic Routing Protocol (GRP) (hybrid). The study exploited FTP traffic over simulation models with different node capacity. The results show that OLSR offers the best performance in terms of load and throughput. However, it suffers considerable delays and routing overhead compared with other routing protocols as a result of the MPR nodes’ selection. In [223], a comparison between OLSR and TORA in terms of delay, retransmissions and data drop has shown that OLSR interacts with different nodes’ update to reduce the delays and increase the throughput, knowing that the retransmission attempts are considered a real problem in OLSR. A method of proactive MANET routing protocol evaluation has been applied to the OLSR protocol, as proposed in [224]. The method analysed the performance of OLSR in medium sized MANET clusters using data from the MANIAC Challenge project specifically for OLSR evaluation. A performance evaluation study about AODV and OLSR routing protocols under realistic radio channel characteristics has already been implemented in ns-2 with Nakagami’s fading model in [225]. The study used CBR traffic with a uniform mobility model with a speed of 40kph. The results show that under realistic channel conditions in both routing protocols the system failed to deliver a good number of data packets to the destination nodes in highly fading environments.

In [226], a study has compared DSR, OLSR and ZRP using different mobile scenarios generated by the RWP Mobility model for MANET using CBR (UDP) traffic. The study used ns-2 and has shown that OLSR offers low average jitter and end-to-end delay with high throughput. The study in [227] has discussed the impact of mobility models and the density of nodes on the performances of OLSR using real-time VBR (MPEG-4) as well as Constant Bit Rate (CBR) traffic. The paper compared the performance in both cases in ns-2 over three mobility models: RWP Waypoint, RWP Direction, and Mobgen Steady State. The simulation results have shown that OLSR behaviour changes according to the traffic and the mobility model used, where RWP Waypoint has the optimal throughput. A proposed QoS extension model for OLSR MANET has been presented and evaluated in OPNET® Modeler for voice applications in [228]. The simulation result has shown an improvement of the packet’s delivery ratio by using the proposed QoS support model for voice communication over MANET compared with native OLSR. The study has focused on voice signaling using PCM voice codec. In [229], a study illustrated the performance of real-time streaming media over a mesh OLSR-based network. The study has examined the effect of mobility and background traffic on carried load and jitter for media applications using IEEE 802.11 MAC/PHY with the EMANE software emulator.

On the other hand, few research efforts studied SIP-based applications over IPv6 MANET where the main concern is mobility issues. In [230], a proposed solution for the optimization of SIP-based mobility
Chapter 6. QoS and Performance Evaluation for SIP-based VoIP over MANET

over IPv6 was considered in an 802.11b network with a theoretical evaluation of the actual performance and the proposed work. The study modified the cross layer triggers and Duplicate Address Detection (DAD) to reduce the handoff delays which improve SIP mobility. In [231], a design and implementation for a SIP-based MIP6-MANET system was proposed as an integration of Mobile IPv6 and SIP-based MANET. The system focused on providing an efficient handoff mechanism to reduce the handoff time and routing delay for Mobile Nodes (MN) roaming between different MANETs through IP clouds. In [232], a mobility handover scheme was designed and implemented for IPv6 based AODV MANET by reducing the mobility handover cost and shortening the mobility handover delay.

6.2 VoIP Calls Evaluation

The simulation works focused on evaluating the SIP signaling performance. Therefore, the study assumptions used to generate many VoIP calls with short period to provide a comprehensive investigation for all sessions of the SIP signaling during the simulations over different MANET models. The results shown in this research study used the average representation method as it provides simple and comparable data representations during the evaluation assessments [209, and 223].

The VoIP calls performance in general can be evaluated from the statistical results provided by OPNET® Modeler from the VoIP calls successful ratio, number of connected calls, and the call durations [221]. These parameters are related to the performance of the three SIP signaling sessions: registration, initiation, and termination. The total estimated number of the successful calls between the caller and the callee is 175 calls over all the scenarios where the duration of each call is 10 seconds. However, the simulation results shown that the percentage of the successful VoIP calls from the total estimated number of the successful VoIP calls in the best effort scenarios for AODV in the Static mobility model is 95.43% with IPv4 and 93.14% with IPv6, as shown in Table 6-1, while in the Uniform mobility model the percentage is 92.6% with IPv4 and 90.3% with IPv6. In the RWP mobility model, AODV has a percentage of 76% successful calls with IPv4 and 69.21% with IPv6. In the RWP-All mobility model, AODV has 56% with IPv4 and only 22.3% with IPv6. On the other hand, OLSR has a percentage of 99.43% with IPv4 and 96% with IPv6 in the Static mobility model, while in the Uniform mobility model the percentage is 94.3% with IPv4 and 89.1% with IPv6, as shown in Table 6-1. In the RWP mobility model, OLSR has only 28% calls with IPv4 and 22.9% with IPv6. In the RWP-All mobility model, OLSR has 19.4% with IPv4 and only 1.73% with IPv6. Both AODV and OLSR had shown good performance in the Static and Uniform models as it represents the best effort scenarios. OLSR had shown a better percentage of the successful calls compared to AODV. AODV showed an acceptable number of calls with the RWP mobility model while the RWP-All mobility model showed a low number of calls. OLSR has low numbers with the RWP mobility model and only three successful
calls with the RWP-All mobility model. IPv4 in general has shown better performance than IPv6 over both AODV and OLSR.

**Table 6-1: Number of successful VoIP calls**

<table>
<thead>
<tr>
<th></th>
<th>AODV IPv4</th>
<th>OLSR IPv4</th>
<th>AODV IPv6</th>
<th>OLSR IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>167</td>
<td>174</td>
<td>163</td>
<td>168</td>
</tr>
<tr>
<td>Uniform</td>
<td>162</td>
<td>164</td>
<td>158</td>
<td>155</td>
</tr>
<tr>
<td>RWP</td>
<td>133</td>
<td>49</td>
<td>121</td>
<td>40</td>
</tr>
<tr>
<td>RWP-All</td>
<td>98</td>
<td>34</td>
<td>39</td>
<td>3</td>
</tr>
</tbody>
</table>

The reduction in the VoIP calls’ initiation process can be remarked from the increased number of rejected SIP calls in Table 6-2 with the RWP mobility models. The calls’ rejections are caused by the SIP initiation packet drops between the two ends because of MANET connectivity conditions. In addition, the call rejections happen as a result of the callee being unreachable. The rejection percentage is very low with the Static and Uniform mobility models. The percentage is between 4.6% and 8% for AODV and between 2.3% and 9.14% for OLSR. The rejection percentage is increased with the RWP mobility model and is very high with the RWP-All mobility model where it is between 24% and 71.4% for AODV and between 76% and 98.8% for OLSR.

**Table 6-2: Number of rejected VoIP calls**

<table>
<thead>
<tr>
<th></th>
<th>AODV IPv4</th>
<th>OLSR IPv4</th>
<th>AODV IPv6</th>
<th>OLSR IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>8</td>
<td>4</td>
<td>12</td>
<td>7</td>
</tr>
<tr>
<td>Uniform</td>
<td>12</td>
<td>11</td>
<td>14</td>
<td>16</td>
</tr>
<tr>
<td>RWP</td>
<td>42</td>
<td>133</td>
<td>67</td>
<td>144</td>
</tr>
<tr>
<td>RWP-All</td>
<td>79</td>
<td>140</td>
<td>125</td>
<td>173</td>
</tr>
</tbody>
</table>

The average VoIP calls’ duration has been investigated for AODV and OLSR over different traffic models, as shown in Figure 6-1. The call duration for a single VoIP call is determined from the SIP call registration, initiation, and termination processes, as explained in Figure 4-2 in Chapter 4. The ideal call duration for successful VoIP calls in this simulation work is 10 seconds as implemented and configured in OPNET® Modeler scenarios. The shorter the possible call duration shows the better SIP signaling performance it has. OLSR shows shorter calls’ durations compared with AODV for both IPv4 and IPv6 over the Static and Uniform mobility models. The proactive behaviour of OLSR supported the registration and initiation processes for SIP signaling. In general, the optimised VoIP calls’ duration for AODV and OLSR are existing over the Static and RWP mobility models. With the RWP mobility model, AODV IPv4 had acceptable call durations, whereas with AODV IPv6 the call durations are between 10.64 and 21.80 seconds for the successful VoIP calls. In the RWP-All mobility model, AODV had call durations of 13.44 to 33.85 seconds for the successful VoIP calls. OLSR had call durations
between 15 and 21 seconds with IPv4 and 14 to 36 seconds with IPv6 for its successful VoIP calls over the RWP mobility model. For the RWP-All mobility model, the call durations for the successful VoIP calls were between 28 and 37 seconds with IPv4 and 35 to 55.36 seconds with IPv6. The long call durations for OLSR traffic is related to the long timers for SIP signaling for the initiation or the termination processes. In the simulation configuration steps for OLSR, the SIP termination timers for SIP calls were configured to be unlimited to allow VoIP calls to connect, otherwise no calls were connected over the OLSR RWP mobility models because of the default timers for SIP signaling.

1. Static

2. Uniform

3. RWP

4. RWP-All

Figure 6-1: Average call durations in seconds

6.3 SIP Signaling Evaluation for VoIP Applications

The evaluation of SIP signaling performance for the simulated scenarios depends on the RFC 6076 SIP end-to-end performance metrics and the call setup time that was discussed in Section 4.2. In addition, the evaluation of SIP signaling considered the B2BUA-based SIP server performance regarding the SIP messages over the SIP registration, initiation, and termination processes. As the main issue regarding the performance metrics is the benchmarking values, the simulation work shows results using the best effort scenarios over the Static and Uniform mobility models. Then, the collected results were compared
with the RWP scenarios to find the related performance differences for SIP signaling. In this research study, the best effort values for SIP signaling over the Static and Uniform mobility scenarios were considered as the reference values for SIP performance metrics to compare them with other scenarios using different mobility models. In addition, the best effort values are used as suggested benchmarking values for SIP end-to-end performance that could be used to compare the SIP signaling performance over MANET for different mobility models.

6.3.1 SIP end-to-end Performance Metrics

The SIP signaling performance had examined three RFC 6076 performance metrics: the RRD, SRD, and SDD. These metrics had been introduced and discussed in Section 4.2.2 and implemented in the simulation efforts of the research study.

6.3.1.1 RRD Values

The RRD values for SIP over MANET for both AODV and OLSR scenarios are shown in Table 6-3 by using equation (5.1). For AODV, RRD values are in its best effort with the Static mobility model over both IPv4 and IPv6, where the average RRD values are between 2 ms and 2.73 ms and the maximum RRD value was 10.45 ms. In the Uniform model, the average values increased to be in the range of 14.33 ms to 32.41 ms, where the maximum value was 205.89 ms with IPv6. The RRD values for IPv6 are 25.3% more than IPv4 in the Static AODV scenario, and 55.8% more than IPv4 in the Uniform AODV scenario. For the RWP mobility model, the performance was affected by the random mobility of MANET nodes as the average RRD values are in the range of 578.84 ms to 874.24 ms where the maximum value is 2.84 seconds. The mobility of the SIP server in the RWP-All scenario affected the RRD values to be in the range of 1.41 seconds to 2.11 seconds while the maximum value was 5.69 seconds. The RRD values for IPv6 are 33.8% more than IPv4 in the RWP AODV scenario, and 34.2% more than IPv4 in the RWP-All AODV scenario. The RRD values over AODV IPv4 have lower values compared with AODV IPv6 with different mobility scenarios.

For OLSR, the average RRD values in the Static model are in the range of 1.59 ms to 2.26 ms, and the maximum RRD value was 8.85 ms. In the Uniform mobility models, the average RRD values are in the range of 13.61 ms to 27.24 ms where the maximum RRD value was 191.51 ms. The RRD values for IPv6 are 30% longer than IPv4 in the Static OLSR scenario, and 50% longer than IPv4 in the Uniform OLSR scenario. For the RWP mobility model, the performance was influenced by the nodes’ random mobility where the average RRD values are in the range of 1.27 seconds to 3.18 seconds where the maximum recorded value is 13.18 seconds. Furthermore, the RRD values for the RWP-All scenario are in the range of 3.49 seconds to 8.36 seconds while the maximum value was 17.9 seconds. The RRD values for IPv6 are 60.2% more than IPv4 in the RWP OLSR scenario, and 58.2% more than IPv4 in
the RWP-All OLSR scenario. The RRD values with OLSR IPv4 showed smaller values compared with OLSR IPv6 over different mobility scenarios.

### Table 6-3: RRD for SIP signaling over AODV and OLSR MANET in milliseconds (ms)

<table>
<thead>
<tr>
<th></th>
<th>AODV</th>
<th>OLSR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>IPv4</td>
<td>IPv6</td>
</tr>
<tr>
<td></td>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td>Static</td>
<td>1.12</td>
<td>6.56</td>
</tr>
<tr>
<td>Uniform</td>
<td>4.47</td>
<td>182.34</td>
</tr>
<tr>
<td>RWP</td>
<td>108.43</td>
<td>2574.67</td>
</tr>
<tr>
<td>RWP-All</td>
<td>307.69</td>
<td>4758.98</td>
</tr>
</tbody>
</table>

#### 6.3.1.2 SRD Values

The SRD values for SIP over MANET for both AODV and OLSR scenarios are shown in Table 6-4 using equation (5.3). For AODV, the best values for SRD are with the Static mobility model where the values are between 16.32 ms and 24.76 ms and the maximum SRD value was 41.03 ms. In the Uniform model, the average value are in the range of 189.48 ms to 265.23 ms where the maximum value was 469.27 ms within IPv6 implementation. The SRD delays for IPv6 are 34.1% more than IPv4 in the Static AODV scenario, and 28.6% more than IPv4 in the Uniform AODV scenario. For the RWP mobility model, the random mobility of MANET nodes increased the SRD average value to be in the range of 1.03 seconds to 1.94 seconds where the maximum value is 3.32 seconds. On the other hand, the mobility of the SIP server in the RWP-All scenario increased the SRD values to be in the range from 2.24 seconds to 3.26 seconds while the maximum value was 12.28 seconds. The SRD values for IPv6 are 53.1% longer than IPv4 in the RWP AODV scenario, and 31.1% longer than IPv4 in the RWP-All AODV scenario. The AODV performance with IPv4 for SRD values is slightly better than IPv6 over different mobility scenarios.

For OLSR, the average SRD value in the Static model are in the range of 15.69 ms to 19.39 ms, and the maximum SRD value was 27.98 ms. In the Uniform mobility models, the average SRD values are in the range of 121.55 ms and 212.73 ms where the maximum SRD value was 581.27 ms. The average SRD values for IPv6 are 19% longer than IPv4 in the Static OLSR scenario, and 42.9% longer than IPv4 in the Uniform OLSR scenario. For the RWP mobility model, the performance of random mobility for MANET nodes is lower than Static and Uniform mobility models as the average SRD values are in
the range of 3.59 seconds to 6.78 seconds where the maximum recorded value is 18.88 seconds. Furthermore, the SRD values for the RWP-All scenario are in the range of 6.38 seconds to 11.16 seconds while the maximum value was 34.36 seconds. The SRD values for IPv6 are 47.3% more than IPv4 in the RWP OLSR scenario, and 42.83% more than IPv4 in the RWP-All OLSR scenario. The SRD values over OLSR IPv4 have smaller values compared with OLSR IPv6 over different scenarios.

<table>
<thead>
<tr>
<th>Scenario</th>
<th>IPv4</th>
<th>IPv6</th>
<th>AODV</th>
<th>IPv4</th>
<th>IPv6</th>
<th>OLSR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Minimum</td>
<td>Maximum</td>
<td>Average</td>
<td>Minimum</td>
<td>Maximum</td>
<td>Average</td>
</tr>
<tr>
<td>Static</td>
<td>3.08</td>
<td>28.56</td>
<td>16.32</td>
<td>5.11</td>
<td>41.03</td>
<td>24.76</td>
</tr>
<tr>
<td>Uniform</td>
<td>5.42</td>
<td>345.67</td>
<td>189.48</td>
<td>7.42</td>
<td>469.27</td>
<td>265.23</td>
</tr>
<tr>
<td>RWP</td>
<td>485.04</td>
<td>2141.25</td>
<td>1028.29</td>
<td>684.36</td>
<td>3322.79</td>
<td>1943.57</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1179.46</td>
<td>6147.81</td>
<td>2242.74</td>
<td>1408.15</td>
<td>12276.16</td>
<td>3255.35</td>
</tr>
</tbody>
</table>

**6.3.1.3 SDD Values**

The SDD values for SIP over MANET for both AODV and OLSR scenarios are shown in Table 6-5 using equation (5.4). For AODV, the SDD values are in its best case in the Static mobility model, where the average SDD values are between 1.48 ms and 2.02 ms and the maximum SDD value was 8.04 ms. For the Uniform model, the average values range increased to be in the range of 11.37 ms to 21.45 ms where the maximum value was 137.45 ms for IPv6. The SDD values for IPv6 are 26.7% more than IPv4 values in the Static AODV scenario, and 47% more than IPv4 in the Uniform AODV scenario. For the RWP mobility model, the average SDD values were affected by the random mobility of MANET nodes where the SDD values are in the range of 453.78 ms to 737.19 ms where the maximum value is 1.19 seconds. Furthermore, the mobility of the SIP server in the RWP-All scenario affected the SDD values to be in the range from 1.19 seconds to 1.79 seconds while the maximum value was 4.05 seconds. The SDD values for IPv6 is 38.4% more than IPv4 in the RWP AODV scenario, and 33.7% more than IPv4 in the RWP-All AODV scenario. The AODV performance for the SDD IPv4 has better performance compared with IPv6 over different mobility scenarios.

For OLSR, the average SDD values in the Static model are in the range of 1.03 ms to 1.95 ms, and the maximum SDD value was 6.79 ms. In the Uniform mobility models, the average SDD values are in the range of 11.08 ms and 19.39 ms where the maximum SDD value was 226.65 ms. The SDD values for
IPv6 are mostly 47.2% more than IPv4 in the Static OLSR scenario, and 42.9% more than IPv4 in the Uniform OLSR scenario. For the RWP mobility model, the performance was affected by the nodes’ random mobility where the average SDD values are in the range of 1.09 seconds to 1.9 seconds where the maximum recorded value is 6.71 seconds. Furthermore, the SDD values for the RWP-All scenario are in the range of 1.82 seconds to 5.41 seconds while the maximum value was 14.6 seconds. The SDD values for IPv6 are mostly 42.33% longer than IPv4 in the RWP OLSR scenario, and 66.41% longer than IPv4 in the RWP-All OLSR scenario. In general, the OLSR performance for SDD with IPv4 shows lower values and better performance compared with IPv6 over different mobility scenarios.

### Table 6-5: SDD for SIP signaling over AODV and OLSR MANET in milliseconds (ms)

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td><strong>AODV</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>1.16</td>
<td>4.13</td>
</tr>
<tr>
<td>Uniform</td>
<td>3.86</td>
<td>107.22</td>
</tr>
<tr>
<td>RWP</td>
<td>83.68</td>
<td>1558.64</td>
</tr>
<tr>
<td>RWP-All</td>
<td>252.36</td>
<td>3871.73</td>
</tr>
<tr>
<td><strong>OLSR</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>0.82</td>
<td>5.18</td>
</tr>
<tr>
<td>Uniform</td>
<td>3.71</td>
<td>146.14</td>
</tr>
<tr>
<td>RWP</td>
<td>489.13</td>
<td>3712.68</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1256.14</td>
<td>6452.16</td>
</tr>
</tbody>
</table>

The comparison of the results for RRD, SRD, and SDD have shown that AODV and OLSR are slightly similar with delays over the Static and Uniform scenarios for both IPv4 and IPv6. However, AODV has shorter delays for the RWP and RWP-All scenarios compared with OLSR that showed longer delays. In general, the SDD average values are lower than the RRD values as the registration process has a simple direct signaling system. Furthermore, the nature of session request processes affects the SRD values. The SRD values are representing the most considerable delays for SIP signaling over MANET for both AODV and OLSR.

#### 6.3.2 SIP Registration Interval

The registration process for SIP clients is the first stage of the SIP call initiation. The registration happens once in the beginning of the network setup and is repeated each time the calling nodes use it to change its status by re-joining the network or reconnecting with the SIP server. Table 6-6 shows the average registration time for both the caller and the callee to register with the SIP server during the provided registration period as discussed in Section 4.2.4. For AODV, the average registration time values for SIP sessions over the Static mobility model for both IPv4 and IPv6 are between 3.13 ms and
4.21 ms and the maximum registration time was 17.12 ms. In the Uniform model, the average registration time values increased to be in the range of 37.72 ms to 52.83 ms where the maximum value was 395.15 ms with IPv6. The registration time values for IPv6 are 25.7% more than IPv4 in the Static AODV scenario, and 28.6% more than IPv4 in the Uniform AODV scenario. For the RWP mobility model, the random mobility of MANET nodes increased the average registration time values to be in the range of 812.91 ms to 1.42 seconds where the maximum value is 3.83 seconds. Furthermore, the mobility of the SIP server in the RWP-All scenario added further delays to the registration time values to be in the range from 1.98 seconds to 3.79 seconds while the maximum value was 13.37 seconds. The registration time values for IPv6 are 42.56% more than IPv4 in the RWP AODV scenario, and 47.84% more than IPv4 in the RWP-All AODV scenario. The registration time values for AODV IPv4 have shorter registration times compared with AODV IPv6 over different mobility scenarios.

For OLSR, the average registration time values in the Static models are in the range of 3.32 ms to 5.13 ms, and the maximum value was 15.62 ms. In the Uniform mobility models, the average registration time values are in the range of 29.54 ms and 44.83 ms where the maximum registration time value was 434.45 ms. The average registration time values for IPv6 is 35.28% more than IPv4 in the Static OLSR scenario, and 34.12% more than IPv4 in the Uniform OLSR scenario. For the RWP mobility model, the nodes’ random mobility increased the average registration time values to be in the range of 2.49 seconds to 5.33 seconds where the maximum recorded value is 23.17 seconds. Furthermore, the average registration time values for the RWP-All scenario are in the range of 6.02 seconds to 13.82 seconds while the maximum value was 25.77 seconds. The average value of the registration time for IPv6 is 53.29% more than IPv4 in the RWP OLSR scenario, and 56.46% more than IPv4 in the RWP-All OLSR scenario. In general, the average registration time values with OLSR IPv4 have shorter registration times compared with OLSR IPv6 over different mobility scenarios.

Table 6-6: SIP registration time for SIP clients over AODV and OLSR MANET in milliseconds (ms)

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Minimum</td>
<td>Maximum</td>
</tr>
<tr>
<td><strong>AODV</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>2.05</td>
<td>9.72</td>
</tr>
<tr>
<td>Uniform</td>
<td>9.42</td>
<td>248.18</td>
</tr>
<tr>
<td>RWP</td>
<td>138.31</td>
<td>3112.1</td>
</tr>
<tr>
<td>RWP-All</td>
<td>698.26</td>
<td>9177.17</td>
</tr>
<tr>
<td><strong>OLSR</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>2.65</td>
<td>8.59</td>
</tr>
<tr>
<td>Uniform</td>
<td>8.31</td>
<td>207.17</td>
</tr>
<tr>
<td>RWP</td>
<td>981.36</td>
<td>9235.17</td>
</tr>
<tr>
<td>RWP-All</td>
<td>2853.24</td>
<td>17276.43</td>
</tr>
</tbody>
</table>
6.3.3 SIP Call Setup Interval

The call setup process for SIP-based VoIP has been discussed and investigated in Section 4.2.5. Table 6-7 shows the number of SIP/TCP retransmission attempts during the call setup process. The number of retransmission attempts increased when one or more of the call setup messages had been lost or delayed. With the Static model, AODV had 4 to 5 SIP/TCP retransmissions, while with the Uniform mobility model, it had 6 to 10 retransmissions. The number of retransmission attempts for IPv6 is 20% more than IPv4 in the Static AODV scenario, and 40% more than IPv4 in the Uniform AODV scenario. Because of the random mobility of MANET nodes, the number of SIP/TCP retransmission attempts increased to be 23 to 42 for the RWP mobility model, and 48 to 87 for the RWP-All mobility model. This also shows that the number of retransmission attempts for IPv6 has 45.24% more than IPv4 in the RWP AODV scenario, and 44.83% more than IPv4 in the RWP-All AODV scenario. On the other hand, OLSR has 14 to 19 SIP/TCP retransmissions, while with the Uniform mobility model, it has 20 to 36 retransmissions. Thus, the number of retransmission attempts for IPv6 is 26.32% more than IPv4 in the Static OLSR scenario, and 44.44% more than IPv4 in the Uniform OLSR scenario. Furthermore, the random mobility of MANET nodes increased the SIP/TCP retransmission attempts to be 49 to 83 for the RWP mobility model, and 80 to 141 for the RWP-All mobility model. Thus, the number of the retransmission attempts for IPv6 are 41% more than IPv4 in the RWP OLSR scenario, and 43.3% more than IPv4 in the RWP-All AODV scenario.

Table 6-7: Number of SIP/TCP retransmission attempts

<table>
<thead>
<tr>
<th></th>
<th>AODV IPv4</th>
<th>OLSR IPv4</th>
<th>AODV IPv6</th>
<th>OLSR IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>4</td>
<td>14</td>
<td>5</td>
<td>19</td>
</tr>
<tr>
<td>Uniform</td>
<td>6</td>
<td>20</td>
<td>10</td>
<td>36</td>
</tr>
<tr>
<td>RWP</td>
<td>23</td>
<td>49</td>
<td>42</td>
<td>83</td>
</tr>
<tr>
<td>RWP-All</td>
<td>48</td>
<td>80</td>
<td>87</td>
<td>141</td>
</tr>
</tbody>
</table>

In Figure 6-2, the average call setup time showed variable delays over the RWP mobility models compared with the static level of the call setup over the Static and Uniform scenarios. The average call setup time is always more than the values of the SRD performance metrics over all SIP calls. The average call setup time is similar on AODV and OLSR, as it is in the range of 17.15 ms to 278 ms. Furthermore, OLSR has shown slightly better performance compared with AODV for the Static and Uniform scenarios. In the RWP scenarios, the average call setup time for AODV is in the range of 0.93 seconds to 2.24 seconds, while in the RWP-All model the average call setup time is in the range of 1.56 seconds to 4.43 seconds. The average call setup time for IPv6 is 46.1% more than IPv4 for the RWP AODV scenario, and 39.3% more than IPv4 for the RWP-All AODV scenario. For OLSR, the average call setup time in the RWP scenarios is in the range of 2.93 seconds to 7.18 seconds, while in the RWP-
All model, the average call setup time is in the range of 5.14 seconds to 12.19 seconds. The average call setup time for IPv6 is 47.8% more than IPv4 for the RWP OLSR scenario, and 44.1% more than IPv4 for the RWP-All OLSR scenario. In general, IPv4 showed better performance over all scenarios compared with IPv6 for both AODV and OLSR. These results support the research findings in [165] and [175] which shows that IPv6-based SIP has longer delays, and size compared with IPv4-based SIP. In addition, the call setup process has the longest delays compared with other SIP call processes over both AODV and OLSR.

**Figure 6-2: Average SIP call setup time in seconds**

**6.3.4 SIP Call Termination Interval**

The call termination process between two callers in a MANET environment has a lower affect over the SIP signaling performance compared with the registration and the call setup processes. This is because of the call termination process usage for a simple direct form of SIP messages as represented in Figure 4-2. In addition, the SIP call termination time has the same values of the SDD performance metric, which makes it simpler to detect the delays and refer it to the termination performance for the benchmarking values as represented in Section 6.1.1.3. According to [178], the maximum
recommended delays for SIP call termination should not exceed 1 second. Thus, based on the SDD values for the investigated scenarios, the termination delays in both the Static and Uniform scenarios are at the accepted level of delays for both AODV and OLSR. However, for the RWP and RWP-All scenarios, the termination time had exceeded 1 Second which reflects on the termination performance over the overall QoS. Therefore, the average termination time for AODV over the RWP mobility model is in the range of 453.78 ms to 737.19 ms where the maximum time is 1.19 seconds as shown in Table 6-5. For the RWP-All, the average termination time for AODV RWP-All scenarios is in the range from 1.19 seconds to 1.79 seconds while the maximum time is 4.05 seconds. Thus, the average termination time for IPv6 is 38.4% more than IPv4 in the RWP AODV scenario, and 33.7% more than IPv4 in the RWP-All AODV scenario. For the OLSR RWP mobility model, the average termination time is in the range of 1.09 seconds to 1.9 seconds where the maximum time is 6.71 seconds. Furthermore, for the OLSR RWP-All scenario the average termination time is in the range of 1.82 seconds to 5.41 seconds while the maximum time was 14.6 seconds. Thus, the average termination time for IPv6 is mostly 42.33% more than IPv4 in the RWP OLSR scenario, and 66.41% more than IPv4 in the RWP-All OLSR scenario.

6.3.5 SIP Server Efficiency

The SIP server efficiency during the registration, call setup, and termination processes has an important role over the overall performance of the SIP call. Figure 6-3 shows the average number of active SIP/TCP connections each 5 seconds over the B2BUA-based SIP server during the simulation time. In the Static and Uniform scenarios, the average number of active SIP/TCP connections each 5 seconds on the SIP server is 6 to 8 per 5 seconds for both AODV and OLSR, whereas the SIP server performance for IPv4 scenarios is slightly better compared with IPv6. In the RWP scenarios, AODV scenarios have 5 to 7 average number of active SIP/TCP connections per 5 seconds in the SIP server, while the OLSR scenarios have an active number of SIP/TCP connections between 4 and 5 active SIP/TCP connections per 5 seconds in the SIP server. In the RWP-All scenarios, the IPv4 AODV has a performance of 4 to 5 average number of active SIP/TCP connections per 5 seconds in the SIP server, while in the IPv6 AODV and IPv4 OLSR scenarios, the number of active connections at the SIP server had dropped from the range of 4 to 5 per 5 seconds, to the range of 1 to 2 active connections per 5 seconds as most of the connections between MANET nodes had been lost after a while because of the nodes' movement start. For IPv6 OLSR over the RWP-All, the average of active SIP/TCP connections start with 2 to 3, and then it dropped down to 1 active connection per 5 seconds. In general, the number of active SIP/TCP connections reflects the status of the SIP signaling performance over different SIP processes. As the number of active SIP/TCP connections increased, the SIP signaling performs better.
6.4 Voice QoS

The investigated VoIP calls showed variable statistics over different voice performance metrics that were introduced in Section 4.3.1 in Chapter 4. When the caller receives the call initiation acknowledgment from the callee, the voice RTP/UDP traffic starts flooding between each end through different MANET nodes. The voice performance metrics had investigated the voice traffic from the caller (node 1) to the callee (node 24) over four mobility models. The voice performance metrics had investigated five measures, as detailed below.

6.4.1 Throughput

In OPNET® Modeler, voice streams can be analysed using RTP statistics to determine the total volume of sent and received voice traffic per second to find out the actual voice throughput transferred from the caller to the callee. The average number of sent voice traffic from the caller (node 1) for the investigated MANET protocols is shown in Figure 6-4. On the other hand, the received voice traffic by the callee (node 24) is shown in Figure 6-5. The average received voice traffic by the receiver needs to be as close
as possible to the average sent by the sender with a low percentage of packet drops. The best effort for the total sent and received voice traffic for both AODV and OLSR is with Static and Uniform mobility models.

For the RWP mobility models, in general, AODV has better throughput compared with OLSR. The average of voice traffic throughput for IPv4 AODV is 27.23% more than the IPv6 AODV average throughput. On the other hand, OLSR showed a very low throughput over RWP mobility models. Furthermore, the average of voice traffic throughput for IPv4 AODV is 48.52% more than the IPv6 AODV average throughput. In general, the results showed that the bandwidth consumption of IPv6 is higher than IPv4 because of its larger header size. However, for the RWP mobility models, IPv4 has higher consumptions than IPv6 because it has more successful VoIP calls. In addition, the VoIP calls in its best effort scenarios over Static and RWP mobility models consumed a convergent amount of bandwidth on the range of the calculated bandwidth in Table 4-1.

![Figure 6-4: Average voice traffic sent from the caller (Node 1) to the callee (Node 24)
6.4.2 RTP One-way Delay

The one-way RTP/UDP packet delays for voice streams had been examined between the caller and the callee for VoIP applications. It was used to find the time difference between the received RTP/UDP packets in the receiver side depending on the RTP packet parameters, which are Timestamps, Unique Synchronization Source identifiers (SSRC), port numbers, and sequence numbers. The average RTP one-way delays had been measured for all the generated VoIP calls over both IPv4 and IPv6 as represented in Figure 6-6. The one-way delays constraints in Table 4-2 were used to determine the delay ranges for the voice streams. The delay range for AODV and OLSR with Static and Uniform mobility are acceptable as it is from 143 ms to 170 ms as determined in Table 4-2. OLSR showed lower one-way delays compared to AODV for Static and Uniform mobility models because of the proactive routing nature for OLSR that provides more reachability choices for RTP/UDP packets with lower delays [225].
On the other hand, RWP mobility models showed longer delays for those successfully received RTP packets. For AODV, the delays are in the range of 156 ms to 166 ms for RWP mobility while it is in the range of 175 ms to 208 ms for the RWP-All mobility model. OLSR has lower packets delivered with end-to-end delays between 216 ms and 223 for RWP mobility, and between 196 ms and 235 ms for the RWP-All mobility model. The header processing for IPv4 and IPv6 RTP packets affects the one-way delays because of the header sizes. However, the overall average of end-to-end delays for RTP/UDP packets that were successfully received are within the acceptable delay range for VoIP applications. Therefore, there is no significant difference between IPv4 and IPv6 in the RTP one-way delays.

![Graphs showing end-to-end delay for traffic from Node 1 to Node 2 for different mobility models](image)

**Figure 6-6: Average voice data end-to-end delay for traffic from (Node 1) to (Node 2)**

### 6.4.3 Jitter

OPNET® Modeler implements the jitter determination equation (5.13) in Chapter 4 to calculate the maximum jitter for voice packets of the investigated VoIP applications over MANET as shown in Table 6-8. In the Static and Uniform mobility models, the maximum jitter for both AODV and OLSR varies between 2.86 ms and 5.23 ms which is an acceptable variation range for VoIP applications [232].
AODV in both RWP mobility models has an acceptable jitter range for both IPv4 and IPv6, except with RWP-All where AODV IPv6 exceeds the acceptable range of the maximum jitter. On the other hand, OLSR IPv4 showed an acceptable maximum jitter for both mobility models while OLSR IPv6 had an acceptable maximum jitter variation. The difference between IPv4 and IPv6 in the jitter variation is not related to the header size. It is related to the delay variation of the received consecutive voice packets.

Table 6-8: Maximum jitter variation for voice packets flow in milliseconds

<table>
<thead>
<tr>
<th></th>
<th>AODV IPv4</th>
<th>OLSR IPv4</th>
<th>AODV IPv6</th>
<th>OLSR IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>2.86</td>
<td>2.23</td>
<td>3.81</td>
<td>3.12</td>
</tr>
<tr>
<td>Uniform</td>
<td>4.37</td>
<td>4.78</td>
<td>5.58</td>
<td>5.23</td>
</tr>
<tr>
<td>RWP</td>
<td>23.52</td>
<td>37.33</td>
<td>45.89</td>
<td>53.41</td>
</tr>
<tr>
<td>RWP-All</td>
<td>38.64</td>
<td>42.75</td>
<td>54.18</td>
<td>62.42</td>
</tr>
</tbody>
</table>

6.4.4 Packet Loss

The packet loss percentage for RTP traffic was calculated using equation (5.14) in Chapter 5. The calculation depends on the difference between the average number of voice traffic sent from the caller (node 1) in Figure 6-4, and the average number of received voice traffic by the callee (node 24) in Figure 6-5. As voice data is encapsulated in RTP packets, the amount of received voice traffic should be convergent to the sent voice traffic and not exceed the maximum theoretically calculated bandwidth in Table 4-1. Static and Uniform mobility models had the lowest packet loss percentage of the total sent and received voice traffic for both AODV and OLSR. For AODV, the average sent traffic is between 1305 and 1610 Bytes per second, and the average received traffic is between 1280 and 1520 Bytes per second, with a traffic loss percentage between 0.62% and 1.11% for the overall successful voice traffic for IPv4 and IPv6 VoIP calls. The organized movement and the fixed number of hops between the source and destination nodes reduced the number of lost packets between the source and the destination nodes. However, OLSR showed a slightly better traffic compared with AODV where packet loss is lower, as it is between 0.56% and 1.02% of the overall VoIP calls.

For RWP mobility models, AODV has higher percentage of packet loss for the successful calls compared with OLSR. For AODV, the average sent traffic is between 1090 and 1490 Bytes per second for the first 900 seconds, and the average received traffic is between 770 and 1290 Bytes per second, with a packet loss percentage of 3.15% to 7.79%, where the percentage increased with the RWP-All mobility model that reached to 14.75% of the total successful VoIP calls. On the other hand, the average sent traffic is between 600 and 1150 Bytes per second for the first 735 seconds, and the average received traffic is between 100 and 540 Bytes per second, with a packet loss percentage of 5.18% to 14.75%. The percentage of the packet loss increased with the RWP-All mobility model. It reached to 17.13% of the total generated voice traffic for the very limited number of successful VoIP calls.
These increases in the packet loss percentage happen because of the variable increments in hop numbers between source and destination. In addition, the MANET nodes’ reachability during voice traffic transmission was reduced in the RWP mobility models which has another influence on the packet delivery ratio. Table 6-9 summarizes the RTP packet loss percentage for the evaluated voice traffic. The packet loss percentage of the voice calls in the Static mobility mode is less than 1%, which is an acceptable packet loss percentage in VoIP applications. With the Uniform mobility model, the packet loss is still acceptable as it is around 1%, which is the highest recommended percentage of packet loss. However, with the RWP mobility models the packet loss percentage increased more than the acceptable packet loss ratio, which reflects on the voice QoS for both AODV and OLSR. The mobility nature and limited destination reachability are responsible for this increased percentage of packet loss [230].

<table>
<thead>
<tr>
<th>Table 6-9: RTP/UDP packet loss percentage for connected VoIP calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
</tr>
<tr>
<td>Uniform</td>
</tr>
<tr>
<td>RWP</td>
</tr>
<tr>
<td>RWP-All</td>
</tr>
</tbody>
</table>

6.4.5 Mean Opinion Score

The subjective MOS performance metric for VoIP calls was evaluated using the E-model equation (5.15) in Chapter 4 over the simulation scenarios in OPNET®. The ITU-T determined that the average MOS values for GSM codec system is 3.5 out of 5 as shown in Table 4-1, which has Fair to Good voice quality as described in Table 4-3. In Figure 6-7, the best MOS values are with the Static model over all scenarios as the results swing between 3 and 3.5. For Uniform mobility, OLSR scenarios are slightly better when compared with AODV, and IPv4 performs better with both routing protocols. However, the MOS voice quality values for OLSR scenarios dropped to Fair with the RWP model and to Poor with the RWP-All mobility model while OLSR IPv6 had Poor results compared with OLSR IPv4. For AODV, the MOS values for IPv4 performs Fair with RWP and the RWP-All Mobility model while AODV IPv6 had a little poor MOS value over the RWP-All mobility model. This is because of the delays and the increased jitter over both RWP mobility models that affect the R factor of MOS parameters, as explained in Section 4.3.2.1. In general, the MOS values for OLSR are better with the Static and Uniform mobility models while AODV mostly have acceptable MOS values with both RWP mobility models.
Chapter 6. QoS and Performance Evaluation for SIP-based VoIP over MANET

6.5 General Routing Performance

The performance metrics for routing protocols had been discussed in Section 4.4. A relation between the applications' performance and MANET routing parameters had been shown. In this section, the considered performance metrics for the general routing performance are the consumed bandwidth, sent/received routing traffic, and hops number between the correspondent nodes. Figure 6-8 shows the average consumed bandwidth for the routing processes in the MANET during the simulation time of the investigated scenarios. In the Static and Uniform scenarios, the average consumed bandwidth for routing data in the MANET is in the range of 25 to 34.6 Kbits/s for AODV, and in the range of 31 to 40.5 Kbits/s for OLSR. The average of consumed bandwidth for IPv4 scenarios is slightly lower compared with the consumed bandwidth in IPv6. This is because of the packet overhead of IPv6 traffic that increased the amount of consumed traffic. OLSR has higher bandwidth consumptions compared with AODV because of the proactive nature of OLSR. On the other hand, the RWP scenarios showed that the average of total consumed bandwidth for routing messages IPv4 AODV is around 18 to 24
Kbits/s while IPv6 AODV has lower consumption with 13 to 16 Kbits/s. For IPv4 OLSR scenarios, the total consumed bandwidth for routing messages were in the range of 21 to 43 Kbits/s and for IPv6 OLSR in the range of 5 to 10 Kbits/s. For the RWP-All scenarios, the total consumed bandwidth for routing messages in IPv4 AODV is in the range between 20 and 24 Kbits/s while in IPv6 AODV it is in the range between 10 and 21 Kbits/s. While with IPv4 OLSR scenarios, the total consumed bandwidth for routing messages were in the range of 10.5 to 29.7 Kbits/s and for IPv6 OLSR in the range of 1.8 to 14.7 Kbits/s. This variance in the routing traffic bandwidth for RWP and RWP-All scenarios are related to the successful reachability and active calls over MANET nodes during VoIP applications. The routing performance in OLSR with RWP and RWP-All scenarios has considerable delays that affect the connectivity for VoIP applications and reduced the number of the established VoIP calls, thus the voice traffic had reduced as well.

**Figure 6-8: Average consumed bandwidth for routing data in (Bits/Sec) for AODV-based and OLSR-based MANET**
In Figure 6-9, the average routing traffic sent from the caller node regarding the established VoIP applications is represented. For the Static and Uniform scenarios, the average of sent routing traffic from the caller node is in the range of 450 to 800 bits/s for AODV, and in the range of 100 to 430 bits/s for OLSR. While in the RWP scenarios, the average of sent routing traffic is in the range of 480 to 600 bits/s for AODV, and in the range of 500 to 2000 bits/s for OLSR. For the RWP-All scenarios, the average of sent routing traffic is in the range of 220 to 680 bits/s for AODV, and in the range of 100 to 2700 bits/s for OLSR. The low sent routing traffic in both RWP and RWP-All scenarios is reflecting the fact that the voice traffic has not sent and that most of the sent routing requests are for the messages of the SIP call setup attempts.

On the other hand, Figure 6-10 shows the average routing traffic received by the caller node during the simulation time. For the Static and Uniform scenarios, the average received routing traffic by the caller node is between 1 and 2 Kbits/s for AODV, and in the range of 2 to 4 Kbits/s for OLSR. In the RWP scenarios, the average of received routing traffic is between 1 and 3 Kbits/s for AODV, and in the range of 2 to 13 Kbits/s for OLSR. For the RWP-All scenarios, the average is between 1 and 2.5 Kbits/s for AODV, and in the range of 3 to 16 Kbits/s for OLSR. The results showed that the received routing
traffic for both scenarios are mostly double or triple the sent routing traffic. The RWP and RWP-All scenarios showed a high number of received routing traffic compared with the sent routing traffic. This happens as a result of the high number of generated routing messages that try to find routes for the requested connections during the variable movements of MANET nodes.

1. Static
2. Uniform
3. RWP
4. RWP-All

Figure 6-10: Average routing traffic received by the caller node (Bits/Sec)

In Figure 6-11, the average number of hops between the caller node and the callee node during the generated VoIP calls is represented. For the Static and Uniform scenarios, the average number of hops is between 7 and 8 hops for both AODV and OLSR, while for the RWP scenarios, the average number of hops is between 3 and 7 for AODV, and 2 to 4 for OLSR. In the RWP-All scenarios, the average number of hops is between 3 and 5 for AODV, and 4 to 5 for IPv4 OLSR, while IPv6 OLSR has registered only two successful call setups at the beginning of the simulation. For the RWP and RWP-All scenarios, successful connections mostly happen when the hops number between the source and destination is lower than 4. Thus, both communicated nodes need to have a low number of hops between themselves during the nodes' mobility to be able to initiate the SIP-based VoIP calls over the RWP mobility models.
6.6 AODV Routing Performance

The effect of AODV routing parameters over applications' performance had been discussed in Section 4.4.3. In this section, a simple investigation is reported for related AODV routing parameters that affect the SIP-based VoIP performance over AODV MANET. The routing performance for related parameters will be evaluated throughout the examinations for the route discovery time, number of route requests sent by the caller, and the packet queue size on the caller side.

Figure 6-12 shows the average Route Discovery Time (RDT) for the caller node over AODV routing processes during the simulation time for the implemented scenarios, while Table 6-10 shows the maximum registered route discovery time for AODV throughout the simulations for both IPv4 and IPv6 traffic. For the Static and Uniform scenarios, the average route discovery time is up to 210 ms. The maximum registered value was 980 ms for IPv6 AODV. Hence, these values are meeting with the best effort conditions for AODV route discovery time that provide the best performance for real-time applications. In the RWP scenarios, the average route discovery time increased to be in the range of
0.93 to 2.4 seconds where the maximum registered value was 24.25 seconds for IPv6 AODV, while in RWP-All, the average was between 0.96 and 6.92 seconds and the maximum value was 47.24 seconds for IPv6 AODV.

Table 6-10: Maximum route discovery time for AODV in seconds

<table>
<thead>
<tr>
<th>Scenario</th>
<th>AODV IPv4</th>
<th>AODV IPv6</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>0.48</td>
<td>0.66</td>
</tr>
<tr>
<td>Uniform</td>
<td>0.59</td>
<td>0.98</td>
</tr>
<tr>
<td>RWP</td>
<td>17.35</td>
<td>24.25</td>
</tr>
<tr>
<td>RWP-All</td>
<td>19.45</td>
<td>47.24</td>
</tr>
</tbody>
</table>

Figure 6-13 shows the average number of Route Requests (RREQ) per 5 seconds that is sent by the caller node over AODV for the implemented scenarios. For the Static and Uniform scenarios, the average number of route requests sent by the caller is between 4 and 9 requests per 5 seconds, while for the RWP scenarios, the average number has reduced to 3 to 7 requests per 5 seconds. In the RWP-All
scenarios, the average number of route requests sent by the caller is between 2 and 6 per 5 seconds. The IPv6 AODV showed a slightly higher number of sent route requests compared with IPv4.

Figure 6-13: Average number of route requests sent by the caller (Node 1)

Figure 6-14 shows the average number of queued packets by the middle nodes between the caller and callee for data sent from the caller side during SIP-based VoIP implementations over AODV MANET. The average number of queued packets by middle nodes shows its related affects over the RDT and the RREQ performance for AODV routing that affect the SIP-based VoIP applications over MANET. As much as the queue size increases, the RDT time increases and the number of RREQ messages increase which reduce the performance of the VoIP calls. For the Static scenarios, the average number of queued packets is between 9 and 21 packets per second, while in the Uniform scenarios, the average number is between 7 and 42 packets per second. In the RWP scenarios, the average number is between 12 and 83 packets per second. With the RWP-All scenarios, the average number of queued packets is between 14 and 273 packets per second. In general, IPv6 AODV representations have larger numbers of queued packets compared with IPv4.
6.7 OLSR Routing Performance

The OLSR routing parameters that affect the applications' performance had been discussed in Section 4.4.4. In this section, a simple investigation is reported for related OLSR routing parameters that affect the SIP-based VoIP performance over OLSR MANET. The related OLSR routing parameters had been evaluated throughout the simulation results by examining the traffic of HELLO messages sent, the number of Multipoint Relay (MPR) messages sent, and the number of OLSR Topology Control (TC)
messages that were forwarded in MANET. Figure 6-15 shows the average number of HELLO traffic sent over OLSR routing processes for MANET during the simulation time for the implemented scenarios. The traffic amount of HELLO messages represents the actual status of route discovery for the required connectivity between different nodes. As the amount of HELLO traffic increases, so the routing overhead increases in the MANET. For the Static and Uniform scenarios, the average HELLO traffic sent in MANET is between 4.3 Kbits/s and 5.1 Kbits/s where these values are meeting with the best effort conditions for OLSR HELLO traffic that provide the best performance for real-time applications. In the RWP scenarios, the average HELLO traffic sent increased to be in the range of 4.4 Kbits/s to 5.9 Kbits/s for IPv4 OLSR, and in the range of 9.2 Kbits/s to 13.6 Kbits/s for IPv6 OLSR, while in the RWP-All, the average HELLO traffic sent is between 6.5 Kbits/s and 7.8 Kbits/s for IPv4 OLSR, and in the range of 9.6 Kbits/s to 14.4 Kbits/s for IPv6 OLSR.

![Figure 6-15: Average HELLO traffic sent in MANET](image)

Figure 6-16 shows the average number of MPR messages sent over the OLSR routing process during the simulation time for the implemented scenarios. The functionality of MPR is to minimise the rebroadcasting of HELLO messages in the network. Hence, the decreased number of MPR values
indicate that high numbers of HELLO messages are generated from MANET nodes and the MPR mechanism is not able to control the high number of generated HELLO messages. For the Static and Uniform scenarios, the average number of MPR messages sent is between 12 and 17 message per second, while for the RWP scenarios, the average number has reduced to be between 10 and 12 messages per second. In the RWP-All scenarios, the average number of MPR messages sent in MANET is between 6 and 10 messages per second.

![Graphs](image.png)

**Figure 6-16: Average number of MPR in MANET**

Figure 6-17 shows the average number of OLSR Topology Control (TC) messages that were forwarded in OLSR MANET during the SIP-based VoIP implementations. The number of TC messages sent indicates the route status of MANET. The increased number of TC messages reflects the status of the correct routing data that provide the best route between two nodes in the MANET. The TC values decrease with the increase of the nodes' mobility factor. For the Static and Uniform scenarios, the average number of TC messages is between 1240 and 1330 messages every 10 seconds, while in the RWP scenarios, the average number of TC messages is between 750 and 1350 messages every 10 seconds. With the RWP-All scenarios, the average number of TC messages is between 520 and 1230 messages every 10 seconds.
messages every 10 seconds. Furthermore, IPv6 OLSR representations showed larger numbers of TC messages compared with IPv4.

The performance of OLSR routing parameters showed a considerable level of effect over the performance of SIP-based VoIP implementations. The increased number of HELLO messages increased the routing overhead for route discovery in the network. In general, the number of HELLO messages increased with the RWP and RWP-All scenarios because of the increased impact of the nodes’ mobility and route updates of OLSR routing tables. Furthermore, the RWP scenarios had shown lower numbers of TC messages and weak performance for the MPR mechanism compared with the Static and Random scenarios. Controlling the HELLO_INTERVAL and TC_INTERVAL time slots that are used by HELLO and TC messages can improve the MPR mechanism performance with frequent updates for the routing table and reducing the routing overhead.
6.8 Results Discussion for the Benchmarking and Evaluation Efforts of the SIP Processes

The delays in the SIP signaling processes and voice end-to-end delays for collected statistics in this evaluation chapter for both Static and Uniform scenarios for MANET VoIP implementations are in the same range as the results of the implementations of SIP-based applications over other network systems, such as in [4-27, 4-33, 4-40, and 4-43]. The similarity between the findings of SIP-based VoIP performance in this chapter with other research efforts supports the accuracy level of the simulation efforts and the reliability of the research implementations. The evaluation results showed that IPv6 SIP has longer delays, call setup time and throughput compared with IPv4 SIP for both RWP and RWP-All scenarios, which support the results’ findings in [4-36, 5-4, 5-9, 5-10, and 5-11].

For the SIP signaling performance, the time intervals of the registrations, calls setup, and calls termination processes are quite long within the RWP and RWP-All MANET. These long time intervals are related to the nodes’ mobility which affects the general routing performance. For example, the call setup time can be reduced with the dynamic modifications for SIP timers using the optimisation algorithms for SIP messages’ retransmission processes depending on the routing status. Further improvements in terms of call setup delays can be achieved based on these modifications. In general, a set of QoS parameters from the upper three layers of the Open Systems Interconnection (OSI) could be considered to enhance the SIP signaling performance over MANET. This is because the routing processes during the nodes’ mobility had been shown to have a direct effect on the SIP signaling performance over MANET. Therefore, the SIP performance is only as strong as the weakest link at the MANET and performs well where there are no hardware problems. The performance evaluation of the examined systems revealed the following results regarding the SIP processes:

- As part of the benchmarking efforts, the values of the investigated performance metrics of the RFC 6076 have been determined from the evaluation efforts of the Static and Uniform mobility models for the best effort scenarios. The RRD values are up to 35 ms, the SRD values are up to 270 ms, and the SDD values are up to 25 ms for both IPv4 and IPv6 representations.
- The average registration time values for AODV IPv4 have shorter registration, initiation and termination times when compared with AODV IPv6 over different mobility scenarios as shown in Section 6.3.2. The registration time value for IPv6 is about 43% longer than IPv4 in the RWP AODV-based MANET. In addition, the registration time for IPv6 is about 48% more than for IPv4 over the RWP-All AODV-based. In the OLSR-based MANET, the average value for the registration intervals for IPv6 is 54% more than IPv4 in the RWP OLSR scenario and 57% more than IPv4 in the RWP-All scenario. In general, the average values of the registration time for
Chapter 6. QoS and Performance Evaluation for SIP-based VoIP over MANET

OLSR-based IPv4 have shorter registration times compared with OLSR-based IPv6 over different mobility scenarios.

- The call setup process has the longest delays compared with other SIP call processes over both AODV and OLSR. The average call initiation time for an IPv6-based SIP-based call is about 46.1% longer than IPv4 for the RWP AODV scenario, and 39.3% longer than IPv4 for the RWP-All AODV scenario. For OLSR, the average call setup time for IPv6 is 48% more than IPv4 for the RWP OLSR scenario and 44% more than IPv4 for the RWP-All OLSR-based scenario.

- The average termination time for IPv6 is about 38% more than IPv4 in the RWP scenario and about 34% more than IPv4 in the RWP-All scenario for AODV-based MANET. For OLSR, the average termination time for IPv6 is about 42% more than IPv4 in the RWP scenario and 66% more than IPv4 in the RWP-All for OLSR-based scenario.

- In general, IPv4 implementations have better performance over all scenarios when compared with IPv6 implementations for both AODV and OLSR. The findings of the evaluation efforts support the research findings in [4-33] and [4-43] that showed that IPv6-based SIP has longer delays and traffic size when compared with IPv4-based SIP. Both IPv4 and IPv6 implementations had been considered in the investigated scenarios without the mobility support, which clearly influenced the performance of the IPv6 traffic when compared with IPv4 traffic.

The differences for the VoIP QoS for SIP-based VoIP applications over IPv4 and IPv6 are in different ranges depending on the nature of the network system, mobility, bandwidth, and the connectivity statues. The VoIP metrics such as end-to-end delays, jitter, throughput, and packet loss are quite comparable for both IPv4 and IPv6 over RWP and RWP-All MANET scenarios. The IP networks still cannot meet the required level of QoS for VoIP applications; however, the VoIP QoS can be improved by controlling the values of the VoIP performance parameters to be within the acceptable range as declared in [5-4] and [5-10]. As this research considered the GSM as the voice codec for the implemented VoIP applications, the voice end-to-end performance metrics could differ with the usage of other voice codecs. From the evaluation efforts, the majority of successful VoIP calls within the random mobility scenarios happened in the first half of the simulations because of the initial positions of the nodes that provided the best possible connectivity and reachability level until the nodes’ distribution increased the percentage of the changed routing values. In general, MANET mobility characteristics are considered unfriendly with the VoIP nature, as shown in the RWP and RWP-All mobility models. The comparison results of different QoS parameters for SIP-based VoIP in the MANET for IPv4 and IPv6 are as follows:
The average bandwidth consumption of IPv6 is higher than IPv4 because of its larger header size in IPv6. In addition, IPv6 in the implemented systems of this research effort does not support the mobility features. However, for the RWP mobility models, IPv4 has higher consumption than IPv6 because it has more successful VoIP calls. In addition, the VoIP calls are in their best effort conditions over the Static and Uniform scenarios, where the RWP mobility models consumed a convergent amount of bandwidth related to the identified and calculated bandwidth in Table 4-1. In general, OLSR implementations have higher bandwidth consumption compared with AODV implementations because of the proactive nature of OLSR.

The average one-way RTP/UDP packet delays for voice streams between the caller and the callee for VoIP applications have longer delays for those successfully received voice data. The header processing for IPv4 and IPv6 RTP packets affects the one-way delays; however, the overall average of the end-to-end delays for RTP/UDP packets that were successfully received are within the acceptable delay range for VoIP applications. Therefore, there is no significant difference between IPv4 and IPv6 for the RTP one-way delays.

The maximum jitter variations for both AODV-based and OLSR-based MANETs for the Static and Uniform scenarios are in the acceptable variation range for VoIP applications [6-11]. For the RWP mobility models, the jitter range for both IPv4 and IPv6 are mostly in the acceptable range. The difference between IPv4 and IPv6 in the jitter variations is related to the delay variation of the received consecutive voice packets not to the header size.

The average packet loss ratio for Static and Uniform scenarios is between 0.5% and 1% of the overall voice data for VoIP calls over both AODV and OLSR implementations. For RWP mobility models, the implementations of AODV-based MANET has a higher percentage of packet loss for the successful calls compared with OLSR-based MANET. For AODV, the average packet loss ratio is between 3% to 15% of the overall voice data for the VoIP calls, and between 5% to 17% for OLSR implementations. The mobility nature and limited destination reachability are responsible for the increased percentage of the packet loss [6-9].

The average Mean Opinion Score (MOS) value is Good (3.5 out of 5) for VoIP calls over both Static and Uniform mobility models. The MOS value dropped to Fair with the RWP model and to Poor with the RWP-All mobility model. IPv6 implementations have Poor MOS results compared with IPv4. In general, the MOS values for OLSR implementations are better with the Static and Uniform mobility models while AODV implementations mostly have acceptable MOS values with both RWP mobility models.

This evaluation in this chapter is focused on the SIP process delays and VoIP QoS. The related RFC 6076 performance metrics in the evaluation of the SIP processes are employed. The performance metrics values precisely indicate the performance level of the SIP signaling of the SIP processes. In [149, 150, and 154], the RRD values are in the range of (10ms to 30ms) for the registration processes.
with the B2BUA SIP server. Furthermore, the SRD values are in the range of (25ms to 100ms) for the call setup processes with the same resources. The results of the RRD and SRD representations concur with the evaluation results in this chapter. The evaluation methods for the SIP signaling in [149], [150], [154], and [162] do not consider a large number of hops between the nodes for the generated traffic as the main concern in this research was the performance of the SIP server. In addition, the research efforts do not cover all of the SIP processes in the simulation work as the termination process was not considered. However, the evaluation and benchmarking efforts in this chapter do consider the performance of all of the SIP processes for the callers and also the SIP server. In addition, the benchmarked values are clearly identified and used in this research study. The SDD values in this chapter are in the range of (2ms to 21ms), which is in the acceptable range based on the related delay values for both the RRD and SRD performance metrics in this study and in [149, 154].

Compared with the other performance evaluation methods for the SIP signaling processes in [142, 145, and 146], the evaluation and benchmarking method considered in this thesis provide more reliable results for the registration, initiation and termination processes of the SIP-based VoIP. In addition, these benchmarked values can be used over different network platforms based on the identified equations for the RRD, SRD, and SDD in Section 4.2.2. Furthermore, the implementation of these performance metrics is considered a standardised method using a reliable simulation tool (OPNET). On the other hand, the values of the investigated SIP processes within the RWP and RWP-All mobility models cannot be used for the benchmarking results as the node connectivity is affected by the routing nature in MANET. This issue is considered the main performance problem for the SIP-based VoIP implementations over MANET [94, 207].

### 6.9 Summary

In this chapter an evaluation and comparison study for the SIP processes and VoIP performance metrics over AODV and OLSR for both IPv4 and IPv6 MANET have been represented. The results of this evaluation studies help with identifying the required efforts to enhance the performance of the SIP-based VoIP applications by employing the related end-to-end SIP performance metrics of the RFC 6076 to propose novel approaches for performance enhancements. These novel approaches will consider the evaluation results for the current state of the art to enhance the performance level for the SIP processes and reduce the total delays of the SIP signaling system. In addition, the study of the proposed approaches will consider sets of benchmarked values for the used end-to-end SIP performance metrics of the RFC 6076.

As IP networks still cannot meet the required QoS of VoIP, VoIP QoS is improved by controlling the related values of these parameters to be within the acceptable range as declared in [210, and 216]. The
differences in the call setup delays for SIP-based VoIP applications over IPv4 and IPv6 are in different ranges depending on the network system used, the bandwidth, and the connectivity status. The VoIP metrics such as end-to-end delays, jitter, throughput, and packet loss are quite comparable for both IPv4 and IPv6 over RWP and RWP-All MANET scenarios. The results show that most of the successful VoIP calls during MANET mobility scenarios occur in the first half of the simulation as the nodes’ initial positions provide better connectivity and reachability before they begin moving. Both Static and Uniform mobility models had shown the best effort for SIP end-to-end performance metrics and VoIP QoS over AODV and OLSR MANET. In general, MANET mobility characteristics have direct effects on the VoIP performance as shown in the RWP and RWP-All mobility models. Furthermore, the effects of such characteristics on different voice metrics have been studied in this chapter.

In terms of SIP signaling performance, the call setup time is quite long in RWP and RWP-All MANET which relates to the nodes’ mobility. The call setup time can be reduced with the dynamic modifications for SIP timers using optimisation algorithms for SIP messages’ retransmission processes, depending on the routing status. Further improvements in terms of call setup delays can be achieved based on these modifications. The results for both Static and Uniform mobility models are representing the best effort of the implemented scenarios that meets with the evaluation results for similar scenarios over other network systems. The evaluation results showed that these performance parameters are comparable for both IPv4 and IPv6 over MANET environments. With the comparisons of the evaluation results for RRD, SRD, and SDD, the implementations of the SIP-based VoIP over AODV-based and OLSR-based MANET are slightly similar in terms of the delays over the Static and Uniform scenarios for both IPv4 and IPv6. However, the implementations of the AODV-based MANET have shorter delays over the RWP and RWP-All scenarios compared with the OLSR-based that showed longer delays. In general, the average values of the SDD are lower than the average values of RRD because the SDD processes have a simple and direct signaling representation system. On the other hand, the nature of the session initiation requests affects the SRD values. Therefore, the session initiation processes represent the most considerable delays for the SIP signaling over MANET for both AODV-based and OLSR-based MANET representations. Furthermore, the values of the evaluated RFC 6076 metrics in this chapter could be used as reference values to evaluate the SIP signaling performance over the registration, call initiation, and termination processes for SIP-based VoIP applications over MANET.
Chapter 7

7 Employing ROHC for SIP Signaling over MANET

In this Chapter, the evaluation study considers the evaluation findings of the IPv4 and IPv6 examinations in Chapter 7. The study examines the employ of Robust Header Compression (ROHC) as a compression/decompression system for IPv6 headers over Static, Uniform, Random Way Point (RWP), and Random Way Point All (RWP-All) mobility models to study the SIP signaling performance and related routing performance. The simulation works in this part will run for both AODV-based and OLSR-based MANET over five types of IP Headers for SIP-based VoIP applications which are: IPv4, IPv6, IPv6 using ROHC-All for All TCP and UDP headers, IPv6 using ROHC-UDP for RTP UDP/IP headers only, and IPv6 using ROHC-TCP for TCP/IP headers only. The simulation results will be used to compare each type of the SIP-based VoIP applications. In this Chapter, the performance parameters that were designed and implemented for the performance evaluation studies for the SIP signaling system have been used in this chapter. These performance metrics had been reported in Section 4.5 and used in Chapter 6.

7.1 ROHC over MANET in the Literature

For ROHC-based MANET, very few testbed or field trial measurement efforts are reported in the reviewed literature. An evaluation of the transmission of GSM encoded voice using ROHC over wireless links is proposed in [233]. The research introduced an evaluation methodology that combines an elementary objective of voice quality metrics using a novel frame with a synchronization mechanism of voice transmission to provide an effective and accurate quality evaluation of voice packets. The research considered the impact of ROHC on the consumed bandwidth and the delay jitter in the voice signal regardless of the impact of ROHC on the voice quality. The research findings show that ROHC mostly reduced the bandwidth required for the transmission of GSM encoded voice to half for a wide range of error probabilities on wireless links. Furthermore, ROHC improved the voice quality compared to the voice transmissions without ROHC. The method of this research focused on general wireless networks without referring to the mobility issues that affect real-time applications’ performance.
A method system was designed for providing header compression guidance to mobile devices initiating certain applications and protocols [234]. It used a Selective Header Compression (SHC) to utilize the SIP messages to provide ROHC guidance for mobile devices to initiate SIP-based applications, protocols and service options. In addition, the ROHC guidance was designed to allow mobile devices to provide an appropriate selection for the applications source and/or destination ports for data flows requiring ROHC. The ROHC guidance recommended the mobile device request for ROHC-based VoIP flows but not for video flows as the guidance helped to specify the suitable range of RTP ports for the applicable ROHC-based flows. This method only provides a selective way for ROHC over RTP/UDP flows depending on the performance features of the SIP-based flows without any further testbed or simulation efforts with comparable results. In [235], is an implementation of the header compression algorithm and protocol for TCP/IPv6 transport over wireless links with slow/medium speed. This implementation was applied on the standardization work underway in the IETF ROHC working group. The simulation results showed that ROHC over TCP reduced the size of TCP/IPv6 headers with acceptable efficiency, while it is robust for the packet loss of wireless link. In addition, the research efforts indicate that the performance and the type of TCP affect the ROHC-based wireless link.

An evaluation of the behavioural effect of ROHC and packet aggregation over multi-hop wireless mesh networks is shown in [31]. The study mentioned that ROHC improved the IP-based data flow with around 20% of the total wireless link transmissions. However, even if the number of mesh routers increased, the total number of the achieved rate will gradually decrease. On the other hand, the ROHC cooperation with packet aggregation provides 4 to 10 times the achieved rate of improvements and results up to 6 times of reductions for the end-to-end delays. However, this cooperation has an effect on the processing time over the behaviour of the wireless mesh networks. The research study evaluated and improved the ROHC processing time using NS-2 simulation models. The simulation results show an improvement in the general performance for ROHC and packet aggregation where the hardware design is required to speed up the processing units for ROHC-based wireless mesh network systems. Accordingly, the study proposed and evaluated a hardware system model for ROHC and packet aggregation by using SystemC Hardware Description Language (HDL). Further simulation results show that the proposed built-in processor has improved the ROHC performance by enhancing the low speed of the processing power.

An analysis presentation of the primary functional blocks of ROHC with an extract for the architectural implications on the next generation network processor for wireless access is represented in [236]. The study focused on the memory space, bandwidth, and processing resource budgets for the wireless access hardware. The study examined the resource consumption and the gains of the achievable potential performance using the offloading computationally intensive ROHC functions for the applications with specific hardware assists. In addition, this paper discussed the design trade-offs for hardware which
assist in the form of a reconfigurable hardware for the ROHC design and functionality for the network processors and access infrastructure without investigating any real-time applications.

A study for VoIP traffic over fixed WiMAX was introduced in [237] to evaluate ROHC and the application layer aggregation of voice for VoIP performance in a fixed WiMAX testbed with one base station and two subscriber stations. The result shows that ROHC increases the number of simultaneous bidirectional emulated VoIP flows by 6% compared to plain VoIP, while the aggregation and ROHC allows 86% more flows than standard VoIP to be sustained. The evaluation steps of this research effort could be applied to ROHC over MANET to improve the general performance of SIP-based VoIP applications. In [238] is a performance evaluation for VoIP over Wireless MAN-OFDMA air interface of a state-of-the-art mobile WiMAX testbed operating at the 3.5 GHz frequency band and it quantifies the benefits of employing VoIP aggregation and ROHC. In addition, the study proposed VoIP aggregation with ROHC and evaluated it using simulation and modelling. The study concluded that the combined use of VoIP aggregation and ROHC over WiMAX can increase the number of effective data flows without loss by approximately three times the regular VoIP transmission. Furthermore, another study investigated ROHC over the Universal Mobile Telecommunications System (UMTS) to provide an optimal combination of Radio Access Bearers (RAB) for VoIP with IP Multimedia Subsystem in the Core Network (VoIMS) to provide an efficient use of ROHC to improve the QoS of the physical layer [239]. The main improvement of this RAB combination is with the adaption of the throughput for SIP signaling to improve the call setup delays for the transmission of the packet switch by using a Transport Format selection algorithm to combine flexible rate of data flows to match the physical layer. The researchers efforts in [237, 238, and 239] could inspire the research efforts of the evaluation and the improvements of VoIP over IPv6 MANET using ROHC.

7.2 Evaluation for SIP-based VoIP Calls with ROHC Implementations

The main concern in this evaluation investigation is the SIP signaling performance. VoIP calls over short periods were generated during the simulations over different MANET models. In addition, the results are represented using the average reading method because of its simplicity and the ability that it provides to compare the represented readings [209, and 223]. This section is following the evaluation efforts at Section 6.2. This section going to investigate the successful average ratio of SIP-based calls, number of connected calls, and calls’ duration, as these parameters are related to the registration, initiation, and termination stages of the SIP signaling. The evaluation will cover both AODV-based and OLSR-based MANET for IPv4, IPv6, and ROHC-based traffic. The total estimated number of the successful calls between the caller and the callee is 175 calls over all generated scenarios where the duration of each call is 10 seconds. The compression/decompression time for ROHC implementations over IPv6 traffic at the sender/receiver sides are set to the default value $1 \times 10^{-9}$ second/bit as represented.
in OPNET® [124]. Thus, for 1 KBps of IPv6 data traffic, the compression/decompression delay is 0.008 ms. For example, for GSM Full Rate voice codec, the maximum bandwidth for IPv6 traffic is 4.5 KBps, the compression/decompression delay ratio is 0.036 ms within the worst case implementations for GSM based voice traffic. For AODV-based MANET, the average percentage of successful VoIP calls from the total estimated number of VoIP calls is in its best effort for the Static mobility model is 95.43% for IPv4, 93.14% for IPv6, 19.43% for ROHC-All, 12.57% for ROHC-UDP, and 94.29% for ROHC-TCP implementations as shown in Table 7-1. In the Uniform mobility model the average percentage reduced to 92.6% for IPv4, 90.3% for IPv6, 8.57% for ROHC-All, 5.71% for ROHC-UDP, and 92% for ROHC-TCP implementations. In the RWP mobility model, the percentage dropped to 76% for IPv4, 69.14% for IPv6, 6.85% for ROHC-All, 5.14% for ROHC-UDP, and 72.57% for ROHC-TCP implementations. In the RWP-All mobility model, the average percentage of successful calls was limited to 56% with IPv4, 22.3% with IPv6, 4.57% for ROHC-All, 1.71% for ROHC-UDP, and 41.14% for ROHC-TCP implementations.

On the other hand, the implementation of OLSR-based MANET shows variable results with the implementations of ROHC systems over IPv6 traffic. In the Static mobility model, the average percentage of successful successful VoIP calls is 99.43% for IPv4, 96% for IPv6, 82.86% for ROHC-All, 60% for ROHC-UDP, and 98.29% with ROHC-TCP implementations, as shown in Table 7-2. For the Uniform mobility model, the average percentage had reduced to 94.37% for IPv4, 89.1% for IPv6, 53.14% for ROHC-All, 32.57% for ROHC-UDP, and 90.29% for ROHC-TCP implementations. In the RWP mobility model, the percentage dropped to 28% for IPv4, 22.86% for IPv6, 4.57% for ROHC-All, 2.86% for ROHC-UDP, and 2.51% for ROHC-TCP implementations. In the RWP-All mobility model, the average percentage of successful successful calls was limited to 19.43% for IPv4, 1.73% for IPv6, 1.14% for ROHC-All, 0.57% for ROHC-UDP, and 8% for ROHC-TCP implementations.

### Table 7-1: Number of successful VoIP calls for AODV-based MANET

<table>
<thead>
<tr>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>167</td>
<td>163</td>
<td>34</td>
<td>22</td>
</tr>
<tr>
<td>Uniform</td>
<td>162</td>
<td>158</td>
<td>15</td>
<td>10</td>
</tr>
<tr>
<td>RWP</td>
<td>133</td>
<td>121</td>
<td>12</td>
<td>9</td>
</tr>
<tr>
<td>RWP-All</td>
<td>98</td>
<td>39</td>
<td>8</td>
<td>3</td>
</tr>
</tbody>
</table>

### Table 7-2: Number of successful VoIP calls for OLSR-based MANET

<table>
<thead>
<tr>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>174</td>
<td>168</td>
<td>145</td>
<td>105</td>
</tr>
<tr>
<td>Uniform</td>
<td>164</td>
<td>155</td>
<td>93</td>
<td>57</td>
</tr>
<tr>
<td>RWP</td>
<td>49</td>
<td>40</td>
<td>8</td>
<td>5</td>
</tr>
<tr>
<td>RWP-All</td>
<td>34</td>
<td>3</td>
<td>2</td>
<td>1</td>
</tr>
</tbody>
</table>
Chapter 7. Employing ROHC for SIP Signaling over MANET

The number of rejected calls indicates the reduction level of the number of successful VoIP calls. This can be recognised from the high percentage of average rejected calls in RWP and RWP-All mobility models for both AODV-based and OLSR-based MANET as represented in Table 7-3 and Table 7-4. This rejection was caused as a result of the unreachability conditions for SIP messaging traffic and the dis-connectivity that occurs during different call sessions. For Static and Uniform mobility models, the average percentage of calls rejection is very low for IPv6. The ROHC-TCP reduced this percentage from 6.9% to 4% for AODV, and from 4% to 2.86% for OLSR in the Static mobility model. In the Uniform mobility model, a limited level of reduction in the rejected calls had been noticed with ROHC-TCP for both AODV and OLSR. In RWP, the percentage of the calls rejection for IPv6 had reduced by using ROHC-TCP from 38.29% to 32% for AODV, and from 82.29% to 80% for OLSR. In the RWP-All mobility model, the percentage of the rejected calls had reduced from 71.43% to 59.29% for AODV, and from 98.6% to 89.14% for OLSR implementations. The ROHC-TCP implementations over IPv6 traffic succeeded in enhancing the performance of SIP signaling in the call initiation processes, while, the implementations of ROHC-UDP and ROHC-All have increased the number of the rejected calls because of the high amount of ROHC compression processes.

AODV and OLSR implementations show a high percentage of successful SIP calls with a low number of rejected calls for IPv6 implementations in both Static and Uniform models. The ROHC system over TCP/IP had increased the number of the successful calls to an acceptable range of calls similar to IPv4-based VoIP implementations over all mobility models, while no considerable enhancements on the total number of successful calls had been shown with the ROHC-All and ROHC-UDP systems, especially with RWP and RWP-All mobility models.

Table 7-3: Number of rejected VoIP calls for AODV-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>8</td>
<td>12</td>
<td>153</td>
<td>161</td>
<td>7</td>
</tr>
<tr>
<td>Uniform</td>
<td>12</td>
<td>14</td>
<td>165</td>
<td>169</td>
<td>13</td>
</tr>
<tr>
<td>RWP</td>
<td>42</td>
<td>67</td>
<td>161</td>
<td>166</td>
<td>56</td>
</tr>
<tr>
<td>RWP-All</td>
<td>79</td>
<td>125</td>
<td>167</td>
<td>171</td>
<td>102</td>
</tr>
</tbody>
</table>

Table 7-4: Number of rejected VoIP calls for OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>4</td>
<td>7</td>
<td>29</td>
<td>69</td>
<td>5</td>
</tr>
<tr>
<td>Uniform</td>
<td>11</td>
<td>16</td>
<td>44</td>
<td>36</td>
<td>14</td>
</tr>
<tr>
<td>RWP</td>
<td>133</td>
<td>144</td>
<td>167</td>
<td>170</td>
<td>140</td>
</tr>
<tr>
<td>RWP-All</td>
<td>140</td>
<td>173</td>
<td>174</td>
<td>175</td>
<td>156</td>
</tr>
</tbody>
</table>

For each SIP-based call, the calls’ duration are determined by the call registration, initiation, and termination processes, as introduced in Figure 4-2 in Chapter 4. In Figure 7-1, the call durations for
SIP-based VoIP calls had been investigated for the AODV-based MANET. In the simulation configurations for both AODV and OLSR, the SIP termination timers for SIP calls were configured to be unlimited to allow VoIP calls to connect. This helps the calls to be connected over RWP mobility models depending on the flexibility of SIP signaling timers.

In the Static and Uniform mobility models, the average call durations for IPv6 are between 10.20 and 11.20 seconds, between 10.25 and 10.31 seconds for ROHC-TCP, and between 10.43 and 11.49 seconds for ROHC-All and ROHC-UDP implementations. In the RWP mobility model, the average call durations had increased to be between 10.64 and 21.80 seconds for IPv6, between 10.45 and 26.11 seconds for ROHC-TCP, and between 10.75 and 41.70 seconds for ROHC-All, and between 10.25 and 64.20 seconds for ROHC-UDP. In the RWP-All mobility model, due to the reachability and disconnectivity problems, the average call durations had high increases to be in the range between 13.44 and 33.55 seconds for IPv6, between 20.14 and 33.85 seconds for ROHC-TCP, and between 20.55 and 84.35 seconds for ROHC-All and ROHC-UDP implementations.

![Graphs showing average call durations in seconds for AODV-based MANET](image)

Figure 7-1: Average call durations in seconds for AODV-based MANET

On the other hand, the implementation of SIP-based VoIP calls over OLSR-based MANET have variable call durations as shown in Figure 7-2. For both Static and Uniform mobility models, the average
Chapter 7. Employing ROHC for SIP Signaling over MANET

call durations are at its optimal level. The average call durations for IPv6 and ROHC-TCP are between 10.01 and 10.28 seconds, between 10.25 and 10.31 seconds for ROHC-TCP, between 10.03 and 10.88 seconds for ROHC-All, and between 10.17 and 11.48 seconds for ROHC-UDP implementations. In the RWP mobility model, the average call durations had increased to be in the range between 10.64 and 37.25 seconds for IPv6, between 10.65 and 37.80 seconds for ROHC-TCP, and between 41.15 and 51.47 seconds for ROHC-All, and between 42.55 and 66.85 seconds for ROHC-UDP. In the RWP-All mobility model, the average call durations had high increases to be in the range between 35 and 55.36 seconds for IPv6, between 33.60 and 37.72 seconds for ROHC-TCP, between 73.15 and 96.05 seconds for ROHC-All, and between 171.35 and 193.32 seconds for ROHC-UDP implementations. The evaluation efforts are showing that OLSR implementations has long call durations which is related to the long timers for SIP signaling during the initiation or the termination processes.

![Graphs showing average call durations in seconds for OLSR-based MANET](image)

**Figure 7-2: Average call durations in seconds for OLSR-based MANET**

### 7.3 An Evaluation for a ROHC-based SIP Signaling System

The performance evaluation in this section is depending on the RFC 6076 SIP end-to-end performance metrics that were introduced in Section 4.2. The Registration Request Delay (RRD), Registration
Chapter 7. Employing ROHC for SIP Signaling over MANET

Request Delay (RRD), and Session Disconnect Delay (SDD) SIP performance metrics will be considered in this Section as well. In addition, the performance of the SIP server including the SIP registration, initiation, and termination processes will be investigated regarding the level of enhancement that will be compared with the results of the performance metrics in Section 6.3. The best effort scenarios are represented in the Static and Uniform mobility models where the concerns are mainly focusing on the RWP mobility models. The benchmarking values for the SIP end-to-end performance metrics had been evaluated in Chapter 7. In this section, the ROHC implementations will be studied regarding the enhancement level that it provides for IPv6 implementations for both AODV-based and OLSR-based MANET.

7.3.1 RRD Values for a ROHC-based SIP Signaling System

Table 7-5 shows the RRD values for SIP scenarios over both AODV-based and OLSR-based MANET with the implementations of the ROHC systems. For AODV, the RRD values are in its best effort with the Static mobility model over IPv6, where the average RRD values are around 2.73 ms for IPv6. The RRD values are around 2.95 ms for ROHC-All and 4.25 ms for ROHC-UDP implementations, whereas the best values were around 2.23 ms for ROHC-TCP implementations. For Uniform mobility models, the RRD values increased to be around 32.41 ms for IPv6, 34.84 ms for ROHC-All, 53.13 ms for ROHC-UDP, and enhanced with ROHC-TCP implementations that reduced it to 17.27 ms. The RRD values increased with the RWP model to reach to 874.24 ms for IPv6 where both ROHC-All and ROHC-UDP implementations failed to reduce the RRD values where the RRD values were around 887.75 ms to 1.1 seconds. However, ROHC-TCP showed a level of enhancement for the RRD values as the results were around 612.73 ms. For the RWP-All mobility model, a very high massive increase in the RRD values had shown for IPv6 traffic as it was around 2.11 seconds. Both ROHC-All and ROHC-UDP were not able to reduce the RRD values as it had values between 2.23 seconds and 2.7 seconds. The ROHC-TCP implementations succeeded in enhancing the RRD values for IPv6 to be around 1.5 seconds instead of 2.1 seconds.

For OLSR, the average RRD values in the Static model are around 2.26 ms for IPv6, 2.37 ms for ROHC-All, 3.17 ms for ROHC-UDP, and 1.84 ms for ROHC-TCP implementations. In the Uniform mobility models, the average RRD values are around 27.24 ms for IPv6, 30.48 ms for ROHC-All implementations, and 36.52 ms for ROHC-UDP implementations, while the RRD values had been enhanced to 14.94 ms with ROHC-TCP implementations. The RRD values increased to 3.18 seconds for IPv6 with the RWP model. The ROHC implementations had larger RRD values with 3.3 seconds for ROHC-All and 3.86 seconds for ROHC-UDP. The implementations of ROHC-TCP had efficiently reduced the RRD values for IPv6 traffic to 1.31 seconds. For the RWP-All mobility model, the average
RRD values were around 8.36 seconds for IPv6 and enhanced with the ROHC-TCP implementations to become 4.3 seconds.

In general, the ROHC-TCP succeeded in enhancing the RRD values for IPv6 implementations to be near to the average RRD values of IPv4 implementations over both AODV and OLSR. On the other hand, both ROHC-All and ROHC-UDP are not providing any enhancement level for RRD values of the IPv6 traffic.

### Table 7-5: Average RRD values for SIP signaling over AODV and OLSR MANET in milliseconds (ms) with ROHC implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>AODV</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>2.04</td>
<td>2.73</td>
<td>2.95</td>
<td>4.25</td>
<td>2.23</td>
</tr>
<tr>
<td>Uniform</td>
<td>14.33</td>
<td>32.41</td>
<td>34.84</td>
<td>53.13</td>
<td>17.27</td>
</tr>
<tr>
<td>RWP</td>
<td>578.84</td>
<td>874.24</td>
<td>887.75</td>
<td>1092.96</td>
<td>612.73</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1405.72</td>
<td>2105.41</td>
<td>2256.22</td>
<td>2703.68</td>
<td>1513.82</td>
</tr>
<tr>
<td><strong>OLSR</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>1.59</td>
<td>2.26</td>
<td>2.37</td>
<td>3.17</td>
<td>1.84</td>
</tr>
<tr>
<td>Uniform</td>
<td>13.61</td>
<td>27.24</td>
<td>30.48</td>
<td>36.52</td>
<td>14.94</td>
</tr>
<tr>
<td>RWP</td>
<td>1268.37</td>
<td>3184.35</td>
<td>3304.18</td>
<td>3871.23</td>
<td>1307.62</td>
</tr>
<tr>
<td>RWP-All</td>
<td>3495.62</td>
<td>8357.12</td>
<td>9682.17</td>
<td>14213.74</td>
<td>4254.71</td>
</tr>
</tbody>
</table>

### 7.3.2 SRD Values for a ROHC-based SIP Signaling System

The SRD values had been investigated using a ROHC system for both AODV and OLSR scenarios as shown in Table 7-6. For AODV, the Static mobility model has an average SRD value around 24.76 ms for IPv6 where ROHC-TCP implementation reduced the SRD value to 18.07 ms. Both ROHC-All and ROHC-UDP had not enhanced the SRD values as the results increased to be around 32.61 ms and 54.77 ms. For the Uniform mobility model, the SRD value of IPv6 had increased to be around 265.23 ms where the ROHC-TCP implementations reduced it to be around 217.83 ms. The ROHC-All has an average SRD value around 321.82 ms and the ROHC-UDP had an average SRD value around 397.12 ms. For the RWP mobility model, the average SRD value increased to be around 1.94 seconds for IPv6 and the ROHC-TCP implementation succeeded in reducing it to 2.46 seconds. The ROHC-All and ROHC-UDP failed to reduce the SRD values as the averages were between 2.2 seconds and 3.3 seconds. For the RWP-All mobility model, the SRD value is around 3.26 seconds for IPv6 traffic where limited enhancement levels had been shown with using ROHC-TCP as the SRD value turned to be around 2.46 seconds, while for both ROHC-All and ROHC-UDP implementations, very large values for SRD had been noticed where no enhanced level for the original SRD values of the IPv6 traffic had been recognised.
For OLSR, the average SRD value for the Static model is around 19.37 ms for IPv6. The ROHC-TCP implementations reduced this value to be around 16.49 ms, as shown in Table 7-6. The SRD values for both ROHC-All and ROHC-UDP have not enhanced the average SRD value for OLSR as it increased to be between 23.73 ms and 41.52 ms. For the Uniform mobility model, the SRD value of IPv6 increased to be around 212.73 ms, and the ROHC-TCP implementations reduced it to be around 157.23 ms. The implementations of ROHC-All has an SRD value around 284.81 ms and the ROHC-UDP has an SRD value around 455.47 ms. For the RWP mobility model, the SRD value increased to be around 6.78 seconds for IPv6 and the ROHC-TCP implementation succeeded in reducing it to 4.21 seconds. ROHC-All and ROHC-UDP also failed to reduce the SRD value as the values were between 10.43 and 21.87 seconds. For the RWP-All mobility model, the SRD value is high as it is around 11.26 seconds for IPv6 traffic. The ROHC-TCP implementations reduced the average SRD value to be around 8.93 seconds. On the other hand, both ROHC-All and ROHC-UDP implementations showed high increases in the SRD values for IPv6 traffic where the results are around 23.78 seconds and 33.85 seconds.

Table 7-6: Average SRD values for SIP signaling over AODV and OLSR MANET in milliseconds (ms) with ROHC implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>AODV</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>16.32</td>
<td>24.76</td>
<td>32.61</td>
<td>54.77</td>
<td>18.07</td>
</tr>
<tr>
<td>Uniform</td>
<td>189.48</td>
<td>265.23</td>
<td>321.82</td>
<td>397.12</td>
<td>217.83</td>
</tr>
<tr>
<td>RWP</td>
<td>1028.29</td>
<td>1943.57</td>
<td>2194.03</td>
<td>3267.53</td>
<td>1221.92</td>
</tr>
<tr>
<td>RWP-All</td>
<td>2242.74</td>
<td>3255.35</td>
<td>4513.34</td>
<td>5019.86</td>
<td>2457.18</td>
</tr>
<tr>
<td><strong>OLSR</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Static</td>
<td>15.69</td>
<td>19.37</td>
<td>23.73</td>
<td>41.52</td>
<td>16.49</td>
</tr>
<tr>
<td>Uniform</td>
<td>121.55</td>
<td>212.73</td>
<td>284.81</td>
<td>455.47</td>
<td>157.23</td>
</tr>
<tr>
<td>RWP</td>
<td>3578.45</td>
<td>6783.82</td>
<td>10427.09</td>
<td>21869.11</td>
<td>4210.93</td>
</tr>
<tr>
<td>RWP-All</td>
<td>6386.32</td>
<td>11164.74</td>
<td>23776.62</td>
<td>33851.42</td>
<td>8932.87</td>
</tr>
</tbody>
</table>

7.3.3 SDD Values for a ROHC-based SIP Signaling System

The SDD values examined using a ROHC system for both AODV and OLSR scenarios are shown in Table 7-7. The average SDD values are also representing the values of the termination intervals for the implemented SIP calls. In general, the call termination process for SIP calls showed lower effects over the SIP signaling performance when compared with the registration and call setup processes.

For AODV, the Static mobility model has an average SDD value around 2.02 ms for IPv6 where the ROHC-TCP implementations reduced the SDD value to 1.64 ms. With the ROHC-All and ROHC-UDP, the SDD values have not been enhanced as the results increased to be around 4.33 ms and 6.72 ms. For the Uniform mobility model, the SDD value of IPv6 had increased to be around 21.45 ms where the
ROHC-TCP implementations reduced the value to be around 14.52 ms. The ROHC-All has an average SDD value around 34.20 ms and the ROHC-UDP average SDD value is around 49.71 ms. For the RWP mobility model, the average SDD value increased to be around 0.74 seconds for IPv6 and the ROHC-TCP implementations succeeded in reducing it to 0.51 seconds. The ROHC-All and ROHC-UDP failed to reduce the SDD values as the averages were between 1.03 and 1.81 seconds. For the RWP-All mobility model, the SDD value is around 1.79 seconds for IPv6 traffic where limited enhancement levels had been shown with the usage of ROHC-TCP as the SDD value reduced to around 1.30 seconds, while for both ROHC-All and ROHC-UDP implementations, large values for SDD had been shown with no enhanced level for the original SDD values of the IPv6 traffic.

For OLSR, the average SDD value for the Static model is around 1.95 ms for IPv6. The ROHC-TCP implementations had reduced this value to be around 1.24 ms, as shown in Table 7-7. The SDD values for both ROHC-All and ROHC-UDP have not been enhanced for OLSR as it increased to be around 2.69 ms and 4.64 ms. For the Uniform mobility model, the SDD value of IPv6 increased to be around 19.39 ms, and the ROHC-TCP implementations reduced it to be around 15.29 ms. The implementations of ROHC-All has an SDD value around 25.25 ms and the ROHC-UDP has an SDD value around 32.18 ms. For the RWP mobility model, the SDD value increased to be around 1.89 seconds for IPv6 and the ROHC-TCP implementation succeeded in reducing it to 1.39 seconds. ROHC-All and ROHC-UDP also failed to reduce the SDD values as the values are between 2.45 and 3.58 seconds. For the RWP-All mobility model, the SDD value is high as is it around 5.41 seconds for IPv6 traffic. The ROHC-TCP implementations reduced the average SDD value to be around 2.50 seconds. On the other hand, both ROHC-All and ROHC-UDP implementations had shown a high increase in the SDD values for IPv6 traffic where the results are between 7.89 and 10.20 seconds.

Table 7-7: Average SDD values for SIP signaling over AODV and OLSR MANET in milliseconds (ms) with ROHC implementations

<table>
<thead>
<tr>
<th></th>
<th>AODV</th>
<th></th>
<th></th>
<th></th>
<th></th>
<th>OLSR</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>1.48</td>
<td>2.02</td>
<td>4.33</td>
<td>6.72</td>
<td>1.64</td>
<td>1.03</td>
<td>1.95</td>
<td>2.69</td>
<td>4.64</td>
</tr>
<tr>
<td>Uniform</td>
<td>11.37</td>
<td>21.45</td>
<td>34.20</td>
<td>49.71</td>
<td>14.52</td>
<td>11.08</td>
<td>19.39</td>
<td>25.25</td>
<td>32.18</td>
</tr>
<tr>
<td>RWP</td>
<td>453.78</td>
<td>737.19</td>
<td>1025.14</td>
<td>1813.32</td>
<td>507.92</td>
<td>1088.14</td>
<td>1886.68</td>
<td>2447.27</td>
<td>3576.04</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1187.81</td>
<td>1790.23</td>
<td>2809.29</td>
<td>3417.66</td>
<td>1300.74</td>
<td>1818.38</td>
<td>5412.85</td>
<td>7890.77</td>
<td>10201.43</td>
</tr>
</tbody>
</table>
7.3.4 SIP Registration Interval for a ROHC-based SIP Signaling System

The registration process for the SIP clients happens at the beginning of the SIP and could be repeated with each update for the caller’s status. The ROHC implementations have a variable level of enhancement for the registration intervals over the IPv6-based SIP signaling system, as shown in Table 7-8. For AODV, the Static mobility model has an average value for the registration intervals of 4.21 ms for IPv6 where the ROHC-TCP implementations reduced the values to 3.24 ms. With ROHC-All and ROHC-UDP the values of the registration intervals have not been enhanced as the results had increased.

For the Uniform mobility model, the registration interval value of IPv6 increased to 52.83 ms and the ROHC-TCP implementations reduced the average value of this interval to 43.84 ms. The ROHC-All has an average value for the registration intervals of 73.18 ms and the ROHC-UDP average value is around 98.24 ms. For the RWP mobility model, the average registration interval value increased to be around 1.42 seconds for IPv6 and the ROHC-TCP implementations succeeded in reducing its average value to 1 second. Both ROHC-All and ROHC-UDP failed to reduce the registration interval values as the average is between 2.55 and 3.11 seconds. For the RWP-All mobility model, the average value of the registration intervals is around 3.79 seconds for IPv6 traffic. Limited enhancement levels had been shown with ROHC-TCP as the value had reduced to around 2.91 seconds. In addition, both ROHC-All and ROHC-UDP implementations have very large values for the registration intervals as it had no enhancement level for the original values of the registration intervals for IPv6 traffic.

For OLSR, the Static mobility model showed an average value for the registration intervals of 5.13 ms for IPv6 where the ROHC-TCP implementations reduced the value to 3.48 ms. With ROHC-All and ROHC-UDP the values of the registration intervals also have not been enhanced as the values had increased for IPv6 traffic. For the Uniform mobility model, the registration interval value of IPv6 increased to 44.83 ms and the ROHC-TCP implementations reduced the average value of this interval to 35.17 ms. The ROHC-All has an average value for the registration intervals of 59.13 ms and the ROHC-UDP average value is around 72.78 ms. For the RWP mobility model, the average registration interval value had increased to be 5.33 seconds for IPv6 and the ROHC-TCP implementations succeeded in reducing its average value to 8.11 seconds. However, both ROHC-All and ROHC-UDP failed to reduce the registration interval values as the average is between 3.24 and 13.17 seconds. For the RWP-All mobility model, the average value of the registration intervals is around 13.82 seconds for IPv6 traffic with simple enhancement levels with ROHC-TCP as the average value had reduced to 8.25 seconds. The implementations of the ROHC-All and ROHC-UDP systems had shown very large values for the registration intervals as it had no enhancement level over the original values of the registration intervals for IPv6 traffic.
Table 7-8: Average SIP registration time for SIP clients over MANET in milliseconds (ms) with ROHC implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>3.13</td>
<td>4.21</td>
<td>6.77</td>
<td>8.19</td>
<td>3.24</td>
</tr>
<tr>
<td>Uniform</td>
<td>37.72</td>
<td>52.83</td>
<td>73.18</td>
<td>98.24</td>
<td>43.84</td>
</tr>
<tr>
<td>RWP</td>
<td>812.91</td>
<td>1415.16</td>
<td>2551.21</td>
<td>3108.94</td>
<td>1002.67</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1976.12</td>
<td>3789.19</td>
<td>5433.26</td>
<td>9824.11</td>
<td>2907.88</td>
</tr>
</tbody>
</table>

### 7.3.5 SIP Call Setup Intervals for a ROHC-based SIP Signaling System

During SIP call setup intervals, the number of SIP/TCP retransmission attempts increased as one or more messages of the call setup had been lost or delayed as explained in Section 4.2.5 and investigated in Section 6.3.3. Table 7-9 shows the number of SIP/TCP retransmission attempts during the call setup process with ROHC implementations for AODV-based MANET. For Static and Uniform mobility models, the retransmission attempts for IPv6 were 5 to 10 while the ROHC-TCP implementations had reduced the attempt numbers to be 4 to 6 attempts. For the RWP mobility model, the retransmission attempts were 42 attempts for IPv6 and 29 attempts for ROHC-TCP. For the RWP-All mobility model, the retransmission attempts were 87 for IPv6 traffic and this was reduced to 66 attempts with the implementations of the ROHC-TCP system.

For OLSR-based MANET, Table 7-10 shows the number of SIP/TCP retransmission attempts during the call setup process with ROHC implementations. For Static and Uniform mobility models, the retransmission attempts for IPv6 were 19 to 36 attempts. The ROHC-TCP implementations had reduced the retransmission attempt numbers to be 25 to 31 attempts. For the RWP mobility model, the retransmission attempts were 83 attempts for IPv6 and 63 attempts for ROHC-TCP. For the RWP-All mobility model, the retransmission attempts were 141 for IPv6 traffic which was reduced to 98 attempts with the implementations of the ROHC-TCP system. The retransmissions attempts for both ROHC-All and ROHC-UDP are very high and could not reduce the actual retransmission values of IPv6 traffic over all implementations of AODV-based and OLSR-based MANET.
Table 7-9: Number of SIP/TCP retransmission attempts for AODV-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>4</td>
<td>5</td>
<td>21</td>
<td>37</td>
<td>4</td>
</tr>
<tr>
<td>Uniform</td>
<td>6</td>
<td>10</td>
<td>25</td>
<td>74</td>
<td>6</td>
</tr>
<tr>
<td>RWP</td>
<td>23</td>
<td>42</td>
<td>63</td>
<td>97</td>
<td>29</td>
</tr>
<tr>
<td>RWP-All</td>
<td>48</td>
<td>87</td>
<td>107</td>
<td>133</td>
<td>66</td>
</tr>
</tbody>
</table>

Table 7-10: Number of SIP/TCP retransmission attempts for OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>14</td>
<td>19</td>
<td>86</td>
<td>101</td>
<td>25</td>
</tr>
<tr>
<td>Uniform</td>
<td>20</td>
<td>36</td>
<td>109</td>
<td>147</td>
<td>31</td>
</tr>
<tr>
<td>RWP</td>
<td>49</td>
<td>83</td>
<td>121</td>
<td>154</td>
<td>63</td>
</tr>
<tr>
<td>RWP-All</td>
<td>80</td>
<td>141</td>
<td>137</td>
<td>167</td>
<td>98</td>
</tr>
</tbody>
</table>

The average call setup time for ROHC implementations over IPv6 SIP-based VoIP calls are shown in Figures 7-3 and 7-4. For AODV, the average call setup time for the Static mobility model is represented in its optimal readings for IPv6 call systems where the average call setup time is less than 190 ms including the ROHC implementations which meet with the best efforts limit for the call setup time. For the Uniform mobility model, the ROHC-TCP implementations showed a level of enhancements for IPv6 traffic, however both ROHC-All and ROHC-UDP have high call setup delays that exceeded the highest boundary of the acceptable delays for the call setup time. For the RWP mobility model, the average call setup time for IPv6 is between 1.20 and 1.87 seconds while ROHC-TCP reduced the call setup time to be between 1 and 1.36 seconds. For the RWP-All, the average call setup time for IPv6 is between 3.17 and 3.87 seconds while the ROHC-TCP reduced the call setup time to be from 1.96 to 2.83 seconds. Both ROHC-All and ROHC-UDP implementations have a very high call setup time for SIP calls over RWP and RWP-All mobility models that even affect the representation of the call setup over the simulation time.

For OLSR, the average call setup times for the Static mobility model are represented in its optimal status for IPv6 call systems as shown in Figure 7-4. The average call setup time had begun with over 3 seconds for the earliest 200 seconds of the simulation time for IPv6 SIP calls and ROHC implementations. Then the call setup time had descended to be under 150 ms with the stability of the OLSR routing table, while in the Uniform mobility model, the average call setup time for IPv6 had begun from 3 seconds and descended to below 250 ms. The ROHC-TCP implementations showed very limited levels of enhancement over IPv6 traffic. Both ROHC-All and ROHC-UDP implementations have not enhanced the call setup time with the Uniform mobility model. For the RWP mobility model, the average call setup time for IPv6 is between 5.15 and 7.18 seconds while the ROHC-TCP enhanced
the call setup time to be from 1.70 to 8.16 seconds. For the RWP-All, the average call setup time for IPv6 is between 10.07 and 12.19 seconds while the ROHC-TCP had enhanced the call setup time to be from 7.32 to 9.73 seconds. While both ROHC-All and ROHC-UDP implementations showed high values for the average call setup time for SIP calls over RWP and RWP-All mobility models during most of the simulation time, the call setup time showed the longest delays for the SIP calls compared with other processes of the SIP call over both AODV and OLSR.

**Figure 7-3:** Average SIP call setup time in seconds for AODV-based MANET
7.3.6 SIP Server Efficiency for a ROHC-based SIP Signaling System

The number of active SIP/TCP connections reflects the status of the SIP signaling performance over different SIP processes. The SIP server efficiency during the SIP processes has an important effect on the overall performance of the SIP signaling system. The SIP signaling performs better when the number of active SIP/TCP connections has increased. Figure 7-5 shows the average number of active SIP/TCP connections each 5 seconds over the B2BUA-based SIP server including the ROHC implementations for AODV-based MANET. For the Static and Uniform scenarios, the average number of active SIP calls successful through the SIP server is 6 to 7.5 SIP/TCP connections per 5 seconds for IPv6. The ROHC-TCP implementations have slightly enhanced the IPv6 traffic SIP/TCP connectivity in the SIP server. The ROHC-All and ROHC-UDP implementations showed a lower number of active SIP/TCP connections on the SIP server with 4 to 6.5 SIP/TCP active connections per 5 seconds. In the RWP scenarios, the AODV scenarios have 5 to 6.5 average number of SIP/TCP calls per 5 seconds in the SIP server for both IPv6 and ROHC-TCP traffic, while ROHC-All and ROHC-UDP implementations have a lower number of active SIP/TCP connections, between 3 and 4.5 active
connections per 5 seconds. In the RWP-All scenarios, the IPv6 AODV has a performance of 2 to 5 average number of SIP/TCP active connections per 5 seconds in the SIP server. The ROHC-TCP implementations had enhanced the IPv6 traffic to be 2.5 to 5.5 active SIP/TCP connections, while the ROHC-All and ROHC-UDP implementations have lower numbers of active SIP/TCP connections between 1 to 3.5 active connections per 5 seconds.

Figure 7-5: Average number of active SIP/TCP connections per 5 seconds in the SIP server for AODV-based MANET

For the OLSR-based MANET, the average number of active SIP/TCP connections through the SIP server in the Static and Uniform scenarios is 6.5 to 7 SIP/TCP connections per 5 seconds for IPv6, as shown in Figure 7-6. The ROHC-TCP implementations showed an enhanced level for the IPv6 traffic of SIP/TCP connectivity in the SIP server, while the ROHC-All and ROHC-UDP implementations showed a lower number of active SIP/TCP connections on the SIP server which are 4.5 to 6 active connections per 5 seconds. In the RWP scenarios, the OLSR scenarios have 4 to 6 average number of SIP/TCP active connections per 5 seconds in the SIP server for both IPv6 and ROHC-TCP traffic, while ROHC-All and ROHC-UDP implementations showed a lower number of active SIP/TCP connections, between 2 to 3.5 active connections per 5 seconds. In the RWP-All scenarios, the IPv6 OLSR traffic
has a performance of 1 to 2 average numbers of SIP/TCP active connection per 5 seconds on the SIP server. The ROHC-TCP implementations had enhanced the IPv6 traffic to be 2.5 to 4 active SIP/TCP connections. Both ROHC-All and ROHC-UDP implementations have a very low number of active SIP/TCP connections, between 1 to 2 active connections.

Figure 7-6: Average number of active SIP/TCP connections per 5 seconds in the SIP server for OLSR-based MANET

7.4 AODV Routing Performance for SIP-based VoIP Using ROHC

The routing parameters are reflecting the health of the MANET during the implementations of SIP-based VoIP applications as discussed in Section 4.4.3 and investigated in Section 6.6. Figure 7-7 shows the average Route Discovery Time (RTD) for the caller node over AODV routing processes during the simulation time for the implemented scenarios, while Table 7-11 shows the maximum registered route discovery time for AODV throughout the simulations for IPv6 traffic and ROHC implementations. As much as the nodes’ locations are updated, the RDT values are increased.
The average RDT values for the Static and Uniform scenarios are 50 ms up to 210 ms for IPv6 traffic and all ROHC implementations. ROHC-TCP implementations enhanced the average RDT values to be similar to IPv4 traffic values. The maximum value for IPv6 was 980 ms and for ROHC-TCP implementations it was 650 ms where these values are meeting with the best effort conditions for the AODV route discovery time to provide the best performance for real-time applications. In the RWP scenarios, the average RDT values increased to be in the range of 0.93 to 2.4 seconds for IPv6 traffic and all ROHC implementations. The maximum registered value for IPv6 traffic was 24.25 seconds and for ROHC-TCP implementations it was 19.78 seconds. In the RWP-All mobility models, the average RDT values were between 0.96 and 6.92 seconds and the maximum value for IPv6 was 47.24 seconds and 35.72 seconds for ROHC-TCP implementations. For all mobility models, the implementations of ROHC-All and ROHC-UDP are not providing a sufficient level of performance enhancements for RDT values of the IPv6 traffic.

Figure 7-7: Average route discovery time for the caller (Node 1)
Chapter 7. Employing ROHC for SIP Signaling over MANET

Table 7-11: Maximum route discovery time for AODV in seconds

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>IPv6</th>
<th>IPv6 ROHC-All</th>
<th>IPv6 ROHC-UDP</th>
<th>IPv6 ROHC-TCP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Static</td>
<td>0.48</td>
<td>0.66</td>
<td>1.07</td>
<td>1.25</td>
<td>0.54</td>
</tr>
<tr>
<td>Uniform</td>
<td>0.59</td>
<td>0.98</td>
<td>1.36</td>
<td>1.54</td>
<td>0.65</td>
</tr>
<tr>
<td>RWP</td>
<td>17.35</td>
<td>24.25</td>
<td>28.66</td>
<td>32.41</td>
<td>19.78</td>
</tr>
<tr>
<td>RWP-All</td>
<td>19.45</td>
<td>47.24</td>
<td>55.72</td>
<td>61.18</td>
<td>35.72</td>
</tr>
</tbody>
</table>

The average number of Route Requests (RREQ) per 5 seconds that is sent by the caller node over the AODV-based MANET for the implemented scenarios is shown in Figure 7-8. For the Static mobility model, the average number of route requests sent by the caller is between 4 and 5 requests per 5 seconds for IPv6 traffic and between 5 and 6 for ROHC-TCP implementations. For the Uniform mobility model, the average number of route requests sent by the caller is between 5 and 7 requests per 5 seconds for IPv6 traffic and between 7 and 8 for ROHC-TCP implementations, while for the RWP scenarios, the average number has reduced to be between 3 and 5 requests per 5 seconds for IPv6 and between 3 and 6 for ROHC-TCP implementations. In the RWP-All scenarios, the average number of route requests sent by the caller is between 2 and 6 per 5 seconds for IPv6 and between 2 and 4 for ROHC-TCP implementations. The ROHC-TCP implementations reduced the RREQ values which enhanced the performance of the route reachability over AODV-based MANET. On the other hand, the implementations of ROHC-All and ROHC-UDP over all mobility models had shown large increases in the RREQ values that affect the ability to enhance the routing performance over IPv6 traffic.
Chapter 7. Employing ROHC for SIP Signaling over MANET

The average number of the queued packets by the middle nodes in the MANET between the caller and the callee for the traffic that is sent from the caller’s side during SIP-based VoIP implementations over AODV-based MANET is represented in Figure 7-9. The average values of the queued packets in the middle nodes shows its related effects over the RDT and the RREQ performance for AODV routing, which reflects on the SIP-based VoIP applications over MANET. The RDT values and the number of RREQ messages increase with the increase of the queue size which affects the general performance of the SIP-based VoIP calls.

Figure 7-8: Average number of route requests sent by the caller (Node 1)
Chapter 7. Employing ROHC for SIP Signaling over MANET

1. Static

2. Uniform

3. RWP

4. RWP-All

Figure 7-9: Average packet queue size on the caller Side (Node 1)

For Static mobility models, the average number of queued packets is between 9 and 17 packets per second for IPv6 traffic and ROHC-TCP implementations, while in the Uniform mobility models, the average number is between 14 and 42 packets per second for IPv6 traffic and between 18 and 28 packets per second for ROHC-TCP implementations. In the RWP scenarios, the average number is between 21 and 83 packets per second for IPv6 traffic and between 39 and 82 packets per second for ROHC-TCP implementations. For the RWP-All mobility models, the average number of queued packets is between 14 and 273 packets per second and between 84 and 127 packets per second for ROHC-TCP implementations. In general, the implementations of ROHC-TCP had reduced the number of queued packets by the middle nodes which enhanced the delivery ratio of data packets over AODV-based MANET. However, the implementations of ROHC-All and ROHC-UDP had massive increases in the queued packets that were not able to enhance the performance of the IPv6 traffic.

The AODV routing performance had shown a related effect over SIP-based VoIP performance over both IPv6 traffic and ROHC implementations. The ROHC-TCP implementations showed the best values that enhanced the IPv6 traffic. Throughout the implementations of the ROHC-TCP system, the
number of RREQ messages had increased which enhanced the routing table with the updated values. The enhancements on RDT values for IPv6 traffic was increased by the usages of the ROHC-TCP system, while both ROHC-All and ROHC-UDP have not shown any level of routing performance over all mobility models.

7.5 OLSR Routing Performance for SIP-based VoIP Using ROHC

The routing parameters that affect the performance of OLSR-based applications have been discussed in Section 4.4.4 and evaluated for SIP-based VoIP in Section 6.7. These OLSR routing parameters are the number of HELLO messages sent, the number of Multipoint Relay (MPR) messages sent, and the number of OLSR Topology Control (TC) messages that were forwarded in MANET. Figure 7-10 shows the average number of HELLO traffic for OLSR routing processes that were sent during the simulation time. The amount of HELLO traffic messages reflects the route discovery status between the nodes that request connectivity with other nodes in the MANET. The routing overhead increases with the increase of HELLO traffic in the MANET. The high increase of HELLO messages means that there are no available known routes between the communicating nodes in the MANET because of different reasons during the nodes’ mobility such as the reachability, availability, and disconnection.

For the Static and Uniform scenarios, the average HELLO traffic sent in the MANET is between 4.9 Kbits/s and 5.1 Kbits/s for IPv6 traffic, and between 4.3 Kbits/s and 4.8 Kbits/s for ROHC-TCP implementations. While for both ROHC-All and ROHC-UDP implementations, the average HELLO traffic sent in the MANET was between 8 Kbits/s and 11.3 Kbits/s. In the RWP mobility model, the average HELLO traffic sent increased to be in the range of 9.2 Kbits/s and 13.6 Kbits/s for IPv6 traffic, while the ROHC-TCP implementation reduced the average for IPv6 traffic to be in the range of 9.2 Kbits/s and 11.1 Kbits/s. In the RWP-All mobility model, the average HELLO traffic sent is between 9.6 Kbits/s and 14.4 Kbits/s for IPv6 traffic, and in the range of 7.6 Kbits/s to 13.6 Kbits/s with ROHC-TCP implementation. Both ROHC-All and ROHC-UDP implementations have not succeeded in increasing the average number of HELLO traffic as it showed a low average representation of HELLO messages in RWP and RWP-All mobility models.
Chapter 7. Employing ROHC for SIP Signaling over MANET

Figure 7-10: Average HELLO traffic sent in MANET with ROHC implementations

Figure 7-11 shows the average number of MPR messages sent over the OLSR routing process with ROHC implementations. The main purpose of MPR is to minimise the rebroadcasting of HELLO messages in the MANET. When MPR is not able to control the high number of HELLO messages that are generated in the MANET, the MPR values will be low and the number of HELLO messages will increase which affects the MANET routing performance. For the Static and Uniform scenarios, the average number of MPR messages sent is between 12 and 16 message per second for IPv6 traffic, and ROHC implementations, while for the RWP scenarios, the average number has reduced to be between 10 and 12 messages per second for IPv6 traffic and ROHC-TCP implementations, and between 7 and 9 messages per second for ROHC-All and ROHC-UDP implementations. In the RWP-All scenarios, the average number of MPR messages sent in MANET is between 6 and 10 messages per second for IPv6 traffic and ROHC-TCP implementations, and between 4 and 6 messages per second for ROHC-All and ROHC-UDP implementations. The results showed that the MPR values had reduced with the nodes’ mobility as the nodes’ numbers increase and that increased the number of generated HELLO messages.
Figure 7-11: Average number of MPR in MANET with ROHC implementations

Figure 7-12 shows the average number of OLSR Topology Control (TC) messages that were forwarded in OLSR MANET with ROHC implementations during the SIP-based VoIP implementations. The number of TC messages sent indicates the route status of the MANET. The TC values decrease with the increase of the nodes' mobility factor in the MANET. For the Static and Uniform scenarios, the average number of TC messages is between 1240 and 1330 messages for every 10 seconds for IPv6 and ROHC-TCP implementations, while it is between 750 and 1190 messages for every 10 seconds in the RWP scenarios. In the RWP-All scenarios, the average number of TC messages is between 520 and 1185 messages for every 10 seconds for both IPv6 traffic and ROHC-TCP implementations. The ROHC-All and ROHC-UDP have generated lower numbers of TC messages during their implementations. This shows that both ROHC-All and ROHC-UDP have limited levels of performance enhancement that could be provided over IPv6-based applications.
Figure 7-12: Average number of OLSR TC messages forwarded in MANET with ROHC implementations

For OLSR routing parameters, the implementations of ROHC-TCP showed an enhanced level of performance for SIP-based VoIP calls. For IPv6 traffic, the increased number of HELLO messages in the RWP and RWP-All scenarios increased the routing overhead of the route discovery processes and reduced the MPR values in the MANET, while the implementations of ROHC-TCP had reduced the overhead of HELLO messages as a result of the increased number of successful SIP connections between MANET nodes which increased both MPR and TC values.

7.6 Summary

The evaluation study in this chapter showed that IPv6 has long delays with poor performance for SIP-based VoIP over both AODV and OLSR MANET even when applying the ROHC system. The ROHC system has been proposed to reduce the large amount of Internet protocol header overhead when transmitting IPv6-based data. The ROHC implementations had been employed over IPv6 traffic for TCP/IP traffic only, RTP UDP/IP traffic only, or over all generated traffic in this evaluation study. The simulation results showed that ROHC for TCP/IP headers has slightly enhanced the signaling...
Chapter 7. Employing ROHC for SIP Signaling over MANET

performance and VoIP QoS while ROHC over RTP/UDP Headers and ROHC over All Headers for the investigated SIP-based traffic flows have weak performance that did not support the required level of the general VoIP quality.

With the implementations of the ROHC system for TCP/IP traffic, a considerable level of enhanced performance had been shown for SIP-based VoIP calls over IPv6 MANET. This level of enhancement occurred due to the traffic compression process which is limited on the TCP/IP traffic that showed an acceptable level of efficiency for the SIP signaling system. However, ROHC still has longer delays and poor performance compared with SIP-based VoIP over IPv4 MANET. The delays in SIP/TCP messages still affect the calls’ cancellations, terminations, and modifications of the SIP sessions. Therefore, the SIP signaling for IPv6 MANET reactive protocols require further enhancements for SIP/TCP registration and retransmission timers to be able to employ the capabilities of the IPv6 and ROHC system for SIP-based VoIP and real-time applications over MANET. In addition, ROHC implementation over both RWP and RWP-All mobility models had shown different effects on the performance that were related to the mobility, nodes’ capacity, SIP server mobility, and connectivity status of nodes during the SIP signaling transitions. The retransmission timers for SIP/TCP signaling over IPv6 MANET are still an open research issue. In conclusion, the implementations of the ROHC system are not able to improve the overall performance of SIP signaling for mobile-based systems, especially with the mobility of the SIP server.
Chapter 8

8 Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

The proposed Cross-Layer approaches provide dynamic performance enhancements for the SIP signaling system over MANET based on the end-to-end performance metrics as explained in Chapter 5. These approaches are designed to enhance the SIP signaling performance over AODV-based and OLSR-base MANET for VoIP applications. The design of the Cross-Layer performance enhancement models for both AODV and OLSR are considering simple and effective modifications on the related routing parameters to enhance the performance of the SIP signaling system with the lowest possible routing overheads. Both CLAODV and CLOLSR approaches employ the SIP performance metrics of RFC 6076 to provide dynamic adjustments for the routing parameters. The SIP performance metrics that are considered in the proposed performance enhancement approaches in this chapter are Registration Request Delay (RRD), Session Request Delay (SRD), and Session Disconnect Delay (SDD), to provide the proper enhancement level for the SIP signaling performance during the registration, call setup, and termination processes as discussed in Chapter 5. The proposed approaches are applied over the caller’s side or/and the SIP server side. When the caller node UA(A) or the SIP server UAS recognises a delay, the Cross-Layer approach is triggered to enhance the routing performance using cross-layer messages with the network layer to modify the routing parameters based on the analysed data in the application layer, as represented in Figure 5-7. The implementation and evaluation efforts for the proposed performance enhancement approaches of Section 5.7 are represented and investigated in this chapter. Furthermore, the evaluation works in this chapter are used to identify the exact level of performance enhancement and compare these results with the evaluation results for normal implementations in Chapter 6.

8.1 Cross-Layer Implementations over MANET in the Literature

In order to enhance the performance of MANET, a number of novel approaches have been suggested, depending on the Cross-Layer model. The Cross-Layer models have been applied for the purpose of improving the MANET performances for proactive and reactive routing protocols.
8.1.1 Cross-Layer Implementations over AODV-based MANET

A number of studies proposed different routing approaches of the CLAODV routing system to enhance the application performance over MANET. There is no optimal Cross-Layer solution for AODV-based MANET that can solve all the challenging performance issues, however these proposed mechanisms provide performance enhancement approaches for specific problems. Most of the proposed Cross-Layer approaches for AODV are dealing with the lower layers; between the Network layer and the Physical layer to provide an efficient enhancement level for AODV connectivity and data transmissions.

A design and implementation for CLAODV is proposed to improve the performance of AODV over Wireless Sensor Networks (WSN) in [49]. The proposed CLAODV tried to decrease the protocol overhead without compromising the performance of AODV by implementing a crossing routing sub-layer and MAC sub-layer to enhance the sending, receiving and processing mechanism of the timed HELLO messages. The implementations showed that the efficiency of data transferring over MANET had improved, the End-to-End delays were reduced, and the protocol overhead was reduced. The study has not discussed other related issues regarding the mobility model used and the CLAODV overhead regarding the power consumptions of the sensor nodes. In [240], a performance enhancement method for the AODV routing protocol using a Cross-Layer and Position-based forwarding technique known as AODV-PF is suggested to improve the route lifetime. The AODV-PF is a MAC layer-based approach to calculate, observe, and manage the consumed power of the received packets from other nodes. The Simulation efforts on OMNET++ show that AODV-PF improved the delivery ratio but has a scalable routing overhead for mobility, higher packet loss ratio, and higher traffic load. The study discussed the topology limitations, the routing table overhead, and the delays of the route establishment.

In [241], a Cross-Layer algorithm to calculate the channel availability at the link layer for an AODV-based Vehicular Network is proposed to improve the communication reliability and reduce the latency for vehicle safety applications. The study calculated the distance change rate between related nodes by investigating the AODV path discovery process regarding the forwarded RREQ packets to increase the route lifetime, decrease the route delays, and increase the transmission reliability. The study implemented this on a vehicular network using the NCTUs simulator with a realistic road-based mobility model. The implementation included a mobility prediction mechanism based on the knowledge from surrounded mobile nodes. In [242], a Cross-Layered multi-node cooperative routing protocol (AODV-CLC) is proposed, based on AODV multi-route and the ratio of the signal power consumption between MANET nodes. Each node compares the signal power between itself, the transmitter, and the receiver. The study was implemented on the IEEE 802.11 MAC protocol as the AODV-CLC interacts between the network layer and the MAC layer. The simulation works showed that AODV-CLC reduced the average communication delays and improved the packet delivery ratio.
In [243], a proposed design of the CLAODV optimization mechanism was based on multiple physical layer parameters, node load, and the ratio of packet delivery of the MAC layer and the minimum hop number of the network layer. This approach aimed to improve the optimization mechanism for congestions caused by the imbalance of node loads by avoiding the congested nodes and selecting the optimal route to improve the quality of the transmission link. The theoretical analysis together with ns-2 based simulation efforts showed that the routing performance had improved as lower frame loss rate, lower latency and lower throughput were reported. In [244], a modified AODV routing protocol based on Bit Error Rate (MAODV-BER) was proposed. The route discovery of AODV was modified to provide a stable route by obtaining Bit Error Rate (BER) information from the physical layer over the Cross-Layer approach. The design considered the QoS implementation for multimedia applications over MANET. The study minimised the BER values and the hop counts to improve the network performance. The results analysis of QoS metrics showed that MAODV performed well with variable level of node velocity, with higher bandwidth, and lower delays compared with the plain AODV. The improvement in the network performance is due to the selection of the MAODV, a stable route from source to destination by considering the BER and QoS metrics. The study concluded that the maintaining of the routing table has a considerable level of memory consumption and increasing processing overhead.

In [245], a modified version of the AODV routing protocol was proposed, based on the route discovery approach that depends on the Physical Layer information without relying on the hop count approach of the distance vector algorithm. The study designed a model that employed the received Signal-to-Noise Ratio (SNR) to find the required route. The study focused on the usage of the Polar Coding for the fast Fading Channel wireless system. The initial simulation efforts with OPNET® showed an enhanced level of performance for the SNR-based CLAODV regarding MANET traffic throughput, application response time, data dropped, and End-to-End delays.

The authors of [246] suggest a novel cross-layer routing protocol in MANET, named Type of Service, Power and Bandwidth Aware AODV (TSPBA-AODV). It simultaneously offers QoS assurance with the cross-layer approach. The authors obtained various simulation results in order to prove that their suggested protocol TSPBA-AODV has better results compared with the Cost Based Power Aware Cross Layer – AODV (CPACL-AODV). According to the presented results in this work, when the speed of the mobile nodes in the MANET is lower (maximum 40 Kph), the suggested protocol gives better results. Results for two kinds of traffic are expressed in terms of throughput, average end-to-end delay, control overhead and packet delivery ratio. However, the benefits of the suggested protocol of the authors are limited, because their solution gives degraded results when the speed of the mobile nodes gets higher than 40 Kph.
8.1.2 Cross-Layer Implementations over OLSR-based MANET

The proactive nature of the OLSR routing protocol gives it special concerns regarding the route overhead and mobility aspects during real-time applications such as SIP-based MANET. In [247], the authors suggest an heuristic methodology by employing the Bit Error Rate (BER) obtained from the physical layer, and by using the weighted Connectivity Index (CI). CI is a parameter from the network layer and it is a mixture between the capacity and the link connectivity. According to the suggested algorithm in [247], the aim is to optimize the route computation and the performances of the Multipoint Relays (MPR) selection algorithm using the lowest Bit Error Rate (BER) and the highest weighted CI. The proposed model is evaluated by the authors in [247] with simulations in which metrics are used, like the average end-to-end delay, packet delivery ratio, throughput, and control overhead. Furthermore, the results obtained from the simulations are compared with the OLSR protocol with a weighted CI and normal OLSR protocol. By analyzing the obtained simulation results, it is evident that the suggested model gives better results expressed in the mentioned evaluation metrics. Hence, the work in [247] describes a novel cross-layer model in order to improve the OLSR protocol performances using the weighted CI and BER as Quality of Service metrics of the network layer. The suggested algorithm in the scenario of a large number of nodes (over fifty) gives the greatest time complexity. Furthermore, the control overhead is increased with the suggested model as a result of the suggested MPR method. In this context, the suggested model raises the processing overhead during the computation of the weighted CI of every node and BER of every link.

It is well known that MANETs usually implement peer-to-peer architecture. If there is a disconnection between peer nodes (because of power drains, mobility or damage), the network capability is degraded and the maintenance of the route is difficult. The authors of [248] suggest a routing model that tries to proactively manage this kind of disconnection via leveraging and combining information obtained by a Cross-Layer model between multiple levels of the network protocol stack. The route maintenance is improved and the likelihood of correct service delivery is increased in this work via the Cross-Layer model. The crosswise protocol stack used to utilize the link state information like service type, link life prediction and mobility information. Hence, the authors in this work are improving the performances of MANETs by reducing the service interruptions. However, this work lacks detailed Cross-Layer interactions, because just a small part of the possible Cross-Layer interactions is taken into consideration.

In [249] the authors proposed a novel Quality Link Metric over the OLSR protocol. Comparison of the performances of three obtainable Quality Link Metrics (QLMs) is also presented. These three metrics are the Minimum Delay (MD), Expected Transmission Count (ETX), and Minimum Loss (ML). The benefit of the research in [249] is in improving the standard OLSR in terms of higher efficiency via
optimizing the routing latency and routing load. Firstly, the authors describe the proposed mathematical structure and after that they choose three performance parameters in order to confirm the proposed framework. Therefore, the obtained simulation results are expressed in end-to-end delay, normalizing the routing load and throughput, which prove that higher efficiency could be achieved by regulating the frequencies of exchanging the topological information. The crucial idea that is utilized in the OLSR protocol is the implementation of the MPRs or the chosen nodes that are forwarding the broadcast messages for the duration of the process of flooding. The authors of [250] found out that the message overhead is significantly reduced with this technique when compared with a pure mechanism of flooding. The algorithm for selecting the multipoint relays is proposed in this work, although improvements are needed in many aspects, for example, in the situation when multiple probable interface addresses exist for just one host.

In order to discover and disseminate the link state information all over the MANET, the OLSR routing protocol utilizes the Topology Control (TC) and hello messages. The influence of hello messages on the OLSR performance expressed in terms of throughput, load and delay, is presented in [251] employing the OPNET® simulator. Authors in this work conclude that the interval of hello messages has significant impact on different performance parameters, like delay, load, throughput and sent packets. It is clearly shown that the performance of the OLSR protocol is improved if the exchange of hello interval packets is greater. However, more tuning of the OLSR performances is needed in the future by utilizing the appropriate values of the hello interval. The routes that are used by the OLSR protocol are continuously discovered and updated, so they are accessible when they are needed. Hence, a very important factor in the process of the evaluation of the OLSR protocol is the route refresh interval time. The authors of [252] assessed the OLSR protocol using various route refresh intervals in a network with high mobility. Furthermore, delivery ratio and throughput are also examined in order to assess the effectiveness of the OLSR routing protocol. From the obtained simulation results in [252], it is obvious that if the hello message time is selected between 2 to 4 seconds, the throughput is not influenced significantly, but if the interval is increased to 8 seconds, the throughput is significantly influenced and the QoS is degraded.

8.2 Results and Discussion for the CLAODV-based Implementations

The performance evaluation for SIP-based VoIP calls with the implementations of the proposed CLAODV is represented in this section with the considerations of the CLAODV design in Section 5.7.1. The evaluation of the proposed CLAODV needs to study different values of the end-to-end SIP performance metrics for SIP signaling parameters. Therefore, the study considered four sets of the benchmarked values using the employed end-to-end SIP performance metrics of the RFC 6076. To investigate the CLAODV approach over SIP-based MANET, the proposed sets of the benchmarked
values are designed to support the SIP signaling system over AODV-based MANET. The Benchmark sets for the SIP performance metrics are driven from Tables 6-3, 6-4, and 6-5. These sets are representing the best efforts for the Cross-Layer implementations over the investigated scenarios of the SIP-based VoIP over MANET. The RRD, SRD, and SDD values are identified gradually from the longest to the shortest values of the benchmarked sets as represented in Table 8-1. Set A provides the longest possible values for the investigated RRD, SRD, and SDD parameters that can support the performance enhancement for the SIP signaling system over AODV-based MANET. These values decreased consequently in sets B, C, and D. Set D represents the best values with low parameters of the Benchmarked end-to-end SIP performance metrics.

Table 8-1: Set of Benchmarked values for SIP Performance metrics over MANET in milliseconds (ms)

<table>
<thead>
<tr>
<th></th>
<th>RRD</th>
<th>SRD</th>
<th>SDD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Set A</td>
<td>150</td>
<td>900</td>
<td>110</td>
</tr>
<tr>
<td>Set B</td>
<td>125</td>
<td>700</td>
<td>90</td>
</tr>
<tr>
<td>Set C</td>
<td>100</td>
<td>500</td>
<td>70</td>
</tr>
<tr>
<td>Set D</td>
<td>75</td>
<td>300</td>
<td>50</td>
</tr>
</tbody>
</table>

Using the study assumptions that were proposed in Section 5.3, a number of SIP-based VoIP calls were generated with short periods to fulfil the investigation efforts about all sessions of the SIP signaling system during the simulation time using different mobility models. The SIP signaling performance evaluated the registration, initiation, and termination sessions for the SIP-based VoIP calls. The statistical results in OPNET® Modeler used the successful ratio, number of connected calls, and the call durations for this evaluation [221]. The total number of the initiated calls between the caller and the callee is 175 calls over all the scenarios with a duration of 10 seconds for each call.

The benchmarked values of the SIP end-to-end performance metrics for both IPv4 and IPv6 AODV MANET in Chapter 6 were compared with the CLAODV implementations. The total number of initiated VoIP calls over IPv4 AODV-based MANET for both RWP and RWP-All are represented in Table 8-2. Set A showed a lower number of initiated calls compared with actual normal IPv4 traffic that was represented in Table 6-1. The number of the calls gradually increased with sets B and C, where set D represents the best implementations with highest possible efforts with 155 successful calls for RWP, and 142 successful calls for RWP-All. Set D enhanced the percentage of the successful calls to be 12.6% more than actual normal IPv4 traffic for RWP, and 25.14% more than actual normal IPv4 traffic for RWP-All. On the other hand, the total number of initiated VoIP calls over IPv6 AODV-based MANET for both RWP and RWP-All are represented in Table 8-3. Set A has a lower number of successful initiated VoIP calls compared with the actual normal IPv6 traffic for both RWP and RWP-All mobility models. The number of successful calls for RWP had increased to 148 calls with set D compared with 121 calls for the actual number of normal IPv6 traffic, with an increase in the percentage
to 15.46%. In addition, set D increased the number of successful calls for RWP-All to 65 compared
with 39 for actual number of normal IPv6 traffic, with an increase of the percentage to 14.84%.

<table>
<thead>
<tr>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>133</td>
<td>123</td>
<td>135</td>
<td>143</td>
</tr>
<tr>
<td>RWP-All</td>
<td>98</td>
<td>96</td>
<td>112</td>
<td>127</td>
</tr>
</tbody>
</table>

Table 8-3: Number of initiated VoIP calls for IPv6 AODV-based MANET

<table>
<thead>
<tr>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>121</td>
<td>119</td>
<td>128</td>
<td>139</td>
</tr>
<tr>
<td>RWP-All</td>
<td>39</td>
<td>35</td>
<td>41</td>
<td>53</td>
</tr>
</tbody>
</table>

On the opposite side, the total number of rejected VoIP calls over IPv4 and IPv6 AODV-based MANET
for both RWP and RWP-All mobility models are shown in Table 8-4 and Table 8-5. Set D has the
lowest number of initiated calls compared with actual normal IPv4 traffic that were represented in Table
6-1. The number of rejected calls decreased with sets B and C, where set D has the lowest number of
rejected calls for IPv4 traffic with 30 for RWP, and 46 for RWP-All. The percentage of the rejected
calls reduced to 20.7%, which is 3.7% less than the actual normal IPv4 traffic for RWP, and 26.3%,
which is 18.8% less than the actual normal IPv4 traffic for RWP-All. For the IPv6 traffic, the number
of rejected calls decreased with sets B, C and D. Set D has the lowest number of rejected calls for IPv6
traffic with 39 for RWP, and 101 for RWP-All. The percentage of the rejected calls has reduced to
22.3%, which is 6% less than the actual normal IPv6 traffic for RWP, and 57.7%, which is 14.43% less
than the actual normal IPv6 traffic for RWP-All.

Table 8-4: Number of rejected VoIP calls for IPv4 AODV-based MANET

<table>
<thead>
<tr>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>42</td>
<td>63</td>
<td>52</td>
<td>43</td>
</tr>
<tr>
<td>RWP-All</td>
<td>79</td>
<td>93</td>
<td>76</td>
<td>63</td>
</tr>
</tbody>
</table>

Table 8-5: Number of rejected VoIP calls for IPv6 AODV-based MANET

<table>
<thead>
<tr>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>67</td>
<td>71</td>
<td>59</td>
<td>41</td>
</tr>
<tr>
<td>RWP-All</td>
<td>125</td>
<td>135</td>
<td>121</td>
<td>113</td>
</tr>
</tbody>
</table>

The average duration of the SIP-based VoIP call durations had been investigated with the CLAODV
implementations over RWP and RWP-All mobility models, as represented in Figure 6-1. The call
duration for a single call is the total required time for the registration, initiation, voice data transmission,
and termination processes of a SIP call, as introduced in Figure 4-2 in Chapter 4. The total call duration for successful VoIP calls in the simulation efforts need to be in the acceptable range of call duration, which is 10 seconds for SIP-VoIP calls as implemented in the OPNET® Modeler scenarios.

In Figure 8-1, the call durations for SIP-based VoIP calls had been investigated for AODV-based MANET with CLAODV implementations. For IPv4 traffic with the RWP mobility model, the average call durations were between 10.52 to 11.95 seconds for IPv4, between 11.15 to 12.13 seconds for set A, between 10.83 to 11.77 seconds for set B, between 10.15 to 11.65 seconds for set C, and between 10.23 to 10.85 seconds for set D. In the RWP-All mobility model, the average call durations increased to be in the range between 13.73 to 18.37 seconds for IPv4 traffic, between 11.55 to 17.1 seconds for set A, between 11.47 to 14.88 seconds for set B, between 11.91 to 13.63 seconds for set C, and between 11.37 to 12.35 seconds for set D. From these results, the call durations for IPv4 traffic had been enhanced with the implementations of set D of the CLAODV. The call duration time had reduced for IPv4 traffic with an average of 1.10 seconds for RWP and 6.02 seconds for RWP-All.

![Figure 8-1: Average call durations in seconds for AODV-based MANET using CLAODV approach](image-url)
For IPv6 traffic with the RWP mobility model, the average call durations are between 10.64 to 21.80 seconds for IPv6, between 10.55 to 16.21 seconds for set A, between 10.52 to 13.82 seconds for set B, between 10.53 to 12.76 seconds for set C, and between 10.33 to 12.15 seconds for set D. In the RWP-All mobility model, the average call durations increased to be in the range between 13.44 to 33.85 seconds for IPv6, between 12.25 to 21.87 seconds for set A, between 12.07 to 17.44 seconds for set B, between 12.03 to 17.32 seconds for set C, and between 11.45 to 14.25 seconds for set D. The call durations for IPv6 traffic had been enhanced with the implementations of set D of the CLAODV. The call duration time had reduced for IPv6 traffic, with an average of 9.65 seconds for RWP and 19.55 seconds for RWP-All.

8.2.1 SIP Registration Interval with CLAODV

The registration process for the SIP clients is the first process that occurs at the beginning of the SIP-VoIP calls and is repeated with each update for the caller’s status, as introduced in Section 4.2.4. The implementations of the CLAODV have different levels of enhancements for the registration intervals of the SIP signaling system based on the proposed algorithm in Section 5.7.1.1. In addition, the CLAODV implementations for the registration process depend on the identified sets of SIP signaling performance optimisation methods in Table 8-1. In Table 8-6, the average registration time for both the caller and the callee to register with the SIP server for SIP sessions over the RWP mobility model for IPv4 traffic is 812.91 ms. The average registration time for IPv4 traffic is reduced by the implementations of the CLAODV to 789.56 ms with set A, 576.42 ms with set B, 334.03 ms with set C, and 275.31 ms with set D. The average time of the registration intervals had reduced for IPv4 traffic with an average of 537.6 ms for RWP. The enhancement percentage is 66.1% when compared with 812.91 ms which is the original average registration interval. For the RWP-All mobility model, the average registration time for IPv4 traffic is reduced by the implementations of the CLAODV from 1976.12 ms to 1115.23 ms with set A, 789.73 ms with set B, 487.65 ms with set C, and 410.74 ms with set D. The average time of the registration interval had reduced for the IPv4 traffic with an average of 1.6 seconds for the RWP-All. The enhancement percentage is 79.2% compared with 1.97 seconds which is the original average registration interval.

On the other hand, Table 8-7 shows the average registration time for both the caller and the callee to register with the SIP server for SIP sessions over the RWP mobility model for IPv6 traffic, which is 1.42 seconds. The average registration time for IPv6 traffic has been reduced by the implementations of the CLAODV to 1.1 seconds with set A, 688.81 ms with set B, 408.29 ms with set C, and 334.84 ms with set D. From these results, the average time of the registration intervals had reduced for IPv6 traffic with an average of 1.1 seconds for RWP, where the enhancement percentage is 76.3% compared with 1.42 seconds which represents the original average registered registration interval. For the RWP-All
mobility model, the average registration time for IPv6 traffic has also reduced by the implementation of the CLAODV from 3.79 seconds to 2.1 seconds with set A, 934.54 ms with set B, 591.65 ms with set C, and 496.33 ms with set D. The average time of the registration interval had reduced for the IPv6 traffic with an average of 3.3 seconds for RWP-All mobility model. The enhancement percentage is 86.9% compared with 3.79 seconds, which represents the original registered average registration time for SIP clients.

Table 8-6: Average SIP registration time for SIP clients over IPv4 AODV-based MANET in milliseconds (ms) for CLAODV implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>812.91</td>
<td>789.56</td>
<td>576.42</td>
<td>334.03</td>
<td>275.31</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1976.12</td>
<td>1115.23</td>
<td>789.73</td>
<td>487.65</td>
<td>410.74</td>
</tr>
</tbody>
</table>

Table 8-7: Average SIP registration time for SIP clients over IPv6 AODV-based MANET in milliseconds (ms) for CLAODV implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>1415.16</td>
<td>1065.72</td>
<td>688.81</td>
<td>408.29</td>
<td>334.84</td>
</tr>
<tr>
<td>RWP-All</td>
<td>3789.19</td>
<td>2103.56</td>
<td>934.54</td>
<td>591.65</td>
<td>496.33</td>
</tr>
</tbody>
</table>

8.2.2 SIP Call Setup Interval with CLAODV

The call setup process is the main process for the SIP-based VoIP calls which has been discussed in Section 4.2.5 and investigated in Section 6.3.3. Figure 8-2 represents the average call setup time with variable delays over RWP and RWP-All mobility models for AODV-based MANET with the implementations of the proposed CLAODV in Section 5.7.1.2. The average call setup time is always more than the values of the SRD performance metrics over all SIP calls.

For IPv4 traffic with the RWP mobility model, the average call setup time is between 0.93 to 2.24 seconds for IPv4 traffic, between 0.87 to 2.63 seconds for set A, between 0.77 to 1.67 seconds for set B, between 0.54 to 1.47 seconds for set C, and between 0.43 to 1.68 seconds for Set D. In the RWP-All mobility model, the average call setup time for IPv4 AODV is in the range of 1.56 to 2.77 seconds for IPv4, between 1.86 to 2.32 seconds for set A, between 1.62 to 2.22 seconds for set B, between 1.44 to 2.21 seconds for set C, and between 0.39 to 2.29 seconds for set D. The results of IPv4 implementations showed that set D of the CLAODV had enhanced the call setup process and reduced the average time by about 0.5 second for RWP and between 0.18 to 0.6 seconds for RWP-All. On the other hand, for IPv6 traffic with the RWP mobility model, the average call setup time is between 1.56 to 2.79 seconds for IPv6 traffic, between 1.41 to 2.18 seconds for set A, between 1.23 to 1.86 seconds for set B, between 0.89 to 2.12 seconds for set C, and between 0.74 to 1.91 seconds for set D. In the RWP-All mobility
model, the average call setup time for IPv6 AODV is in the range of 2.96 to 4.43 seconds for IPv6, between 2.71 to 3.63 seconds for set A, between 2.16 to 3.42 seconds for set B, between 1.73 to 3.32 seconds for set C, and between 1.42 to 2.57 seconds for set D. Set D of the CLAODV showed the best level of enhancement for the average call setup time for IPv6 traffic and reduced the average call setup time by 0.82 to 0.88 seconds for RWP and 1.54 to 1.86 seconds for RWP-All.

Figure 8-2: Average SIP call setup time in seconds for AODV-based MANET using CLAODV approach

8.2.3 SIP Call Termination Interval with CLAODV

The call termination process for the SIP-based VoIP calls is the last process that happens at the calls, as explained in Section 4.2.6 and examined for benchmarking purposes in Section 6.3.4. The actual call termination time is equal to the value of the actual SDD. The implementations of the proposed CLAODV in Section 5.7.1.3 have variable levels of enhancement for the call termination time over both IPv4 and IPv6 SIP-based VoIP. The enhancement approaches depend on the identified sets of the optimised approaches for the SIP signaling system in Table 8-1. As the call termination process has a lower effect over the performance of the SIP-based VoIP when compared with the registration and call
setup processes, however, the total performance of the SIP-based call needs to carefully consider this process.

In Table 8-8, the average call termination time for a SIP call between the caller and the callee over the RWP mobility model for IPv4 traffic is 453.78 ms. The average call termination time for IPv4 traffic is reduced by the implementations of the CLAODV to 428.52 ms with set A, 325.28 ms with set B, 256.22 ms with set C, and 137.29 ms with set D. The average time of the call termination intervals had reduced for IPv4 traffic with an average of 316.49 ms for RWP. The enhancement percentage is 69.7% for set A when compared with the original average call termination intervals of IPv4 traffic. For the RWP-All mobility model, the average call termination time for IPv4 traffic is reduced by the implementations of the CLAODV from 1187.81 ms to 753.21 ms with set A, 494.37 ms with set B, 313.91 ms with set C, and 276.64 ms with set D. The average call termination time had reduced for the IPv4 traffic with an average of 0.91 second for the RWP mobility models’ scenarios. The enhancement percentage is 76.7% compared with 1.19 seconds which is the original collected average call termination time for IPv4 traffic. On the other hand, Table 8-9 shows the average call termination time for a SIP call between the caller and the callee over the RWP mobility model for IPv6 traffic which is 737.19 ms. The average call termination time for IPv6 traffic is reduced by the implementations of the CLAODV from 719.82 ms with set A, 544.82 ms with set B, 311.61 ms with set C, and 221.82 ms with set D. The average call termination time had reduced for the IPv6 traffic with an average of 515.37 ms for RWP, where the enhancement percentage is 69.9%, compared with 737.19 ms which represents the original average call termination intervals. For the RWP-All mobility model, the average call termination time for IPv6 traffic is also reduced by the implementations of the CLAODV from 1.84 seconds to 1.27 seconds with set A, 808.58 ms with set B, 461.87 ms with set C, and 380.15 ms with set D. The average time of the termination intervals is reduced for the IPv6 traffic with an average of 1.46 seconds for the RWP-All mobility model. The enhancement percentage is 79.34% compared with 1.84 seconds, which represents the original registered average call termination time for IPv6 traffic. In general, the termination time had exceeded the RWP-All scenarios by 1 second for both IPv4 and IPv6 traffic, which reflects on the termination performance of the overall performance of the SIP-based VoIP. The average termination time for AODV-based MANET with the RWP mobility model is in the range of 453.78 ms to 737.19 ms, as shown in Table 8-8 and Table 8-9.

**Table 8-8:** Average SIP termination time for SIP callers over IPv4 AODV-based MANET in milliseconds (ms) for CLAODV implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>453.78</td>
<td>428.52</td>
<td>352.28</td>
<td>265.22</td>
<td>137.29</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1187.81</td>
<td>753.21</td>
<td>494.37</td>
<td>313.91</td>
<td>276.64</td>
</tr>
</tbody>
</table>
Table 8-9: Average SIP termination time for SIP callers over IPv6 AODV-based MANET in milliseconds (ms) for CLAODV implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>737.19</td>
<td>719.82</td>
<td>544.82</td>
<td>311.61</td>
<td>221.82</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1790.23</td>
<td>1274.53</td>
<td>808.58</td>
<td>461.87</td>
<td>380.15</td>
</tr>
</tbody>
</table>

8.2.4 SIP Server Efficiency with CLAODV

The SIP server performance during the processes of the SIP-based calls affects over the performance of the registration, call setup, and termination processes as discussed in Section 6.3.5. The number of active SIP/TCP connections on the SIP server reflects on the performance of the SIP signaling over the processes of the SIP-based VoIP. Therefore, the SIP signaling performs better when the number of active SIP/TCP connections is increased. Figure 8-3 shows the average number of active SIP/TCP connections over the B2BUA-based SIP server including the CLAODV implementations for AODV-based MANET.

Figure 8-3: Average number of active SIP/TCP connections per 5 seconds in the SIP server for AODV-based MANET
In the RWP scenarios, the IPv4 traffic has an average number of 4 to 6 active SIP/TCP connections per 5 seconds in the SIP server. The average number of active SIP/TCP connections in the SIP server has increased with the proposed CLAODV sets until it reached to 5 to 8 active connections. In the RWP-All mobility model, the average number of active SIP/TCP connections for IPv4 traffic on the SIP server is between 3 to 5 active connections per 5 seconds, while it increased to between 4 to 6 with CLAODV set D. For IPv6 traffic with the RWP mobility model, the average number of active SIP/TCP connections per 5 seconds in the SIP server is between 4 to 6 connections. This average number has increased with the proposed CLAODV sets until it came to be between 4 to 7.5 active connections. In the RWP-All mobility model, the average number of active SIP/TCP connections for IPv6 traffic on the SIP server starts with 2 to 4.75 active connections per 5 seconds, then it dropped down to 1 active connection per 5 seconds. However, the average number of active SIP/TCP connections increased with the CLAODV sets until it reached to a range between 2 to 6 with set D. The results for both IPv4 and IPv6 showed that the implementations of set D for the CLAODV had increased the load of the SIP server with about 1 to 2 extra SIP/TCP connections over both RWP and RWP-All mobility models. Therefore, the SIP signaling performance becomes better when the number of active SIP/TCP connections are increased.

8.2.5 Routing Performance with CLAODV

The routing parameters are employed in the CLAODV approaches to have dynamic adjustments during the processing of the SIP-based VoIP applications. This dynamic adjustment procedure is part of the performance enhancement for the SIP signaling processes in the proposed CLAODV approach, as discussed in Section 5.7. The effect of the routing parameters over the performance of the SIP-based applications has been discussed in Section 4.4.3. In this section, the routing performance for related parameters will be evaluated through the examinations for the route discovery time, and the number of route requests sent by the caller with the implementations of the CLAODV sets of Table 8-1. Figure 8-4 shows the average Route Discovery Time (RDT) for the caller node over the routing processes of the AODV-based MANET with the proposed approaches. Table 8-10 and Table 8-11 show the maximum registered route discovery time over AODV-based MANET throughout the simulations for both IPv4 and IPv6 traffic with the CLAODV implementations.

For IPv4 traffic, the average RDT values are from 1.15 to 2.41 seconds over the RWP mobility model. The average RDT values are reduced by the implementations of the CLAODV to be between 1.23 to 1.98 seconds with set A, 0.78 to 1.88 seconds with set B, 0.62 to 1.76 seconds with set C, and 0.47 to 0.97 seconds with set D. The average RDT values for IPv4 traffic over the RWP-All mobility model are from 1.35 to 6.98 seconds. The implementations of the CLAODV reduced the average RDT values to between 1.18 to 4.15 seconds with set A, 1.3 to 2.76 seconds with set B, 1.1 to 2.15 seconds with set
C, and 0.87 to 2.04 seconds with set D. Table 8-10 shows the maximum RDT value for AODV from
the caller side is 17.35 seconds for IPv4 traffic over the RWP mobility model which is reduced by the
implementations of the CLAODV to 10.89 seconds with set D. In addition, for the RWP-All mobility
model, the CLAODV has reduced the maximum RDT value from 19.45 seconds for IPv4 traffic to
12.91 seconds with set D.

Table 8-10: Maximum route discovery time for SIP callers over IPv4 AODV-based MANET in
seconds using CLAODV approach

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>17.35</td>
<td>18.87</td>
<td>16.67</td>
<td>13.44</td>
<td>10.89</td>
</tr>
<tr>
<td>RWP-All</td>
<td>19.45</td>
<td>20.91</td>
<td>18.72</td>
<td>14.26</td>
<td>12.91</td>
</tr>
</tbody>
</table>
Table 8-11: Maximum route discovery time for SIP callers over IPv6 AODV-based MANET in seconds using CLAODV approach

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>24.25</td>
<td>26.66</td>
<td>22.07</td>
<td>19.52</td>
<td>15.96</td>
</tr>
<tr>
<td>RWP-All</td>
<td>47.24</td>
<td>50.17</td>
<td>31.33</td>
<td>23.63</td>
<td>19.07</td>
</tr>
</tbody>
</table>

On the other hand, for IPv6 traffic, the average RDT values are from 0.72 to 0.97 seconds over the RWP mobility model. The average RDT values were activated by the implementations of the CLAODV for IPv6 traffic that activated the routing role to between 2.13 to 2.56 seconds with set A, 1.57 to 2.77 seconds with set B, 1.19 to 1.77 seconds with set C, and 0.87 to 1.43 seconds with set D. The average RDT values for IPv6 traffic over the RWP-All mobility model are from 0.92 to 1.21 seconds. The implementations of the CLAODV reduced the average RDT values to between 2.97 to 3.79 seconds with set A, 2.74 to 3.27 seconds with set B, 2.23 to 2.95 seconds with set C, and 1.92 to 2.39 seconds with set D. In Table 8-11, the maximum RDT value for AODV from the caller side is 24.25 seconds for IPv6 traffic over the RWP mobility model, which is reduced by the implementations of the CLAODV to 15.96 seconds with set D. Furthermore, for the RWP-All mobility model, the CLAODV has reduced the maximum RDT value from 47.24 seconds for IPv6 traffic to 19.07 seconds with set D. The RDT values for IPv6 traffic are not meeting with the best effort conditions for AODV-based route discovery time that provide the best performance for real-time applications. However, with the CLAODV implementations the RDT values became active to meet the required route values for SIP-based VoIP.

The average number of the Route Requests (RREQ) that are sent by the caller node over the AODV-based MANET for the implemented scenarios are represented in Figure 8-5. For IPv4 traffic, the average number of RREQs sent by the caller is between 3.5 to 6 requests per 5 seconds over the RWP mobility model. The average RREQ values are increased by the implementations of the CLAODV to between 4 to 7 requests per 5 seconds with set A, 4.25 to 6 requests per 5 seconds with set B, 5 to 7.5 requests per 5 seconds with set C, and 5.5 to 8.5 requests per 5 seconds with set D. For the RWP-All scenarios, the average number of RREQs sent by the caller has reduced to between 2.25 to 6.5 requests per 5 seconds for IPv4 traffic. The average RREQ values are increased by the implementations of the CLAODV to between 4 to 6.5 requests per 5 seconds with set A, 4.75 to 7 requests per 5 seconds with set B, 5.5 to 7.5 requests per 5 seconds with set C, and 6.5 to 8.5 requests per 5 seconds with set D.

On the other hand, for IPv6 traffic, the average number of RREQs sent by the caller is between 3 to 8 requests per 5 seconds over the RWP mobility model. The average RREQ values are increased by the implementations of the CLAODV to be between 4 to 6 requests per 5 seconds with set A, 4.75 to 6 requests per 5 seconds with set B, 5.25 to 6.5 requests per 5 seconds with set C, and 5.75 to 8 requests per 5 seconds with set D. For the RWP-All scenarios, the average number of RREQs sent by the caller...
is reduced to between 1.5 to 5.5 requests per 5 seconds for IPv6 traffic. However, the average RREQ values are increased by the implementations of the CLAODV to between 2.5 to 5.5 requests per 5 seconds with set A, 3.5 to 6.25 requests per 5 seconds with set B, 4.5 to 6.5 requests per 5 seconds with set C, and 5.25 to 7.25 requests per 5 seconds with set D.

Figure 8-5: Average route requests sent by the caller (Node 1) for AODV-based MANET using CLAODV approach

In Figure 8-6, the average routing traffic that is sent from the caller node for the SIP-based VoIP applications is represented for the investigated CLAODV. For IPv4 traffic, the average of sent routing traffic from the caller node is in the range of 480 to 600 bits/s over the RWP mobility model. The average of the sent routing traffic is increased by the CLAODV implementations to between 0.78 to 1.12 Kbits/s with set A, 1.04 to 1.22 Kbits/s with set B, 1.24 to 1.35 Kbits/s with set C, and 1.31 to 1.55 Kbits/s with set D. In the RWP-All scenarios, the average of sent routing traffic is in the range of 510 to 680 bits/s, which increased with the implementations of the CLAODV sets. The average of the sent routing traffic has increased to between 1.38 to 1.64 Kbits/s with set A, 1.51 to 1.86 Kbits/s with set B, 1.74 to 2.18 Kbits/s with set C, and 1.95 to 2.65 Kbits/s with set D. These increases resulted from the
increased number of the route adjustment processes for the AODV parameters that were caused by the implementations of the proposed CLAODV algorithms. On the other hand, for IPv6 traffic, the average of the sent routing traffic from the caller node is in the range of 410 to 485 bits/s for the RWP mobility model. The average of the sent routing traffic has increased by the CLAODV sets to between 0.58 to 0.84 Kbits/s with set A, 0.73 to 1.11 Kbits/s with set B, 0.96 to 1.71 Kbits/s with set C, and 1.28 to 1.51 Kbits/s with set D. For IPv6 traffic over the RWP-All mobility model, the average of the sent routing traffic from the caller node is in the range of 410 to 485 bits/s that were increased with CLAODV implementations. The average of the sent routing traffic increased to between 1.1 to 1.37 Kbits/s with set A, 1.28 to 1.46 Kbits/s with set B, 1.44 to 1.67 Kbits/s with set C, and 1.65 to 1.95 Kbits/s with set D. In general, IPv6 traffic showed a lower average of routing traffic that was sent by the caller when compared with IPv4 traffic. In addition, the RWP-All scenarios had a high number of generated routing traffic compared with the RWP scenarios because of the variable nature of the SIP server within the RWP-All scenarios. The routing messages increased to update the routing data for the requested connections during the variable movements of the SIP server and the MANET nodes.

**Figure 8-6:** Average routing traffic sent by the caller node in (Bits/Sec) for AODV-based MANET using CLAODV approach
8.2.6 Bandwidth Consumptions and CPU Utilisation for CLAODV Implementations

In the previous section, the routing performance of the proposed CLAODV had been discussed and had shown a relation between the enhancement level of the SIP performance and MANET routing parameters. This performance enhancement level also affected the performance metrics for the performance of the network system in general. The CLAODV implementations affected the total consumed bandwidth in the MANET, and the CPU utilisation level for the B2BUA SIP server for the SIP-based VoIP applications. Figure 8-7 shows the average consumed bandwidth for MANET routing processes during the simulation time of the investigated scenarios.

For IPv4 traffic, the average consumed bandwidth for routing data in the MANET is in the range of 18 to 24 Kbits/s over the RWP mobility model. The CLAODV implementations increased the average consumed bandwidth for routing data in the MANET to between 23.5 to 30 Kbits/s with set A, 25.3 to 34 Kbits/s with set B, 31 to 37.4 Kbits/s with set C, and 33 to 42 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for routing data is in the range of 20 to 24 Kbits/s which increased with the implementations of the CLAODV sets. The CLAODV implementations increased to between 21.8 to 29.7 Kbits/s with set A, 24.7 to 27.4 Kbits/s with set B, 26 to 31.2 Kbits/s with set C, and 27.6 to 32.6 Kbits/s with set D. For IPv6 traffic, the average consumed bandwidth for routing data in the MANET has lower consumption, with 13 to 16 Kbits/s over the RWP mobility model when compared with IPv4 traffic over RWP. The implementations of the CLAODV sets have increased the average consumed bandwidth for the routing data in the MANET to be between 16 to 25 Kbits/s with set A, 17 to 31 Kbits/s with set B, 23 to 31 Kbits/s with set C, and 24.5 to 31.8 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for routing data in the MANET is 10 to 21 Kbits/s over the RWP mobility model which lower than the average consumption with IPv4 traffic over the RWP-All. The CLAODV implementations increased the average consumed bandwidth for routing data in the MANET to between 16.5 to 19.6 Kbits/s with set A, 16.7 to 25.8 Kbits/s with set B, 18 to 28.4 Kbits/s with set C, and 19.5 to 27.7 Kbits/s with set D. Because of the packet overhead of the IPv6 traffic that increased the amount of consumed traffic, in reality the average of the consumed bandwidth for IPv4 scenarios is slightly lower compared with the consumed bandwidth in IPv6. However, the average of the successful SIP-based VoIP calls in IPv4 implementations is greater than the average of the IPv6 implementations. Therefore, the average of the consumed traffic for IPv4 scenarios is higher than IPv6 scenarios, as shown in the results of Figure 8-7.
Chapter 8. Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

Figure 8-7: Average consumed bandwidth for routing data in (Bits/Sec) for AODV-based MANET using CLAODV approach

The performance enhancement approaches for the SIP signaling system over AODV-based MANET have increased the routing traffic over MANET during the implementations of the CLAODV. In addition, the CPU utilisation level of the B2BUA SIP server has also increased. This increase happened as a result of the increased number of the processes that relate to the routing enhancement procedures of the CLAODV. Figure 8-8 shows the average level of the CPU utilisation for the B2BUA SIP server in the AODV-based MANET during the simulation time for the investigated scenarios. For IPv4 traffic, the average CPU utilisation level for the B2BUA SIP server is in the range of 5% to 24% per second for the total capacity of the CPU processes in the RWP mobility model. The CLAODV implementations increased the average CPU utilisation level in the SIP server to between 26% to 41% per second with set A, 37% to 51% per second with set B, 41% to 62% per second with set C, and 52% to 79% per second with set D. In the RWP-All scenarios, the average CPU utilisation level of the B2BUA SIP server is in the range of 18% to 51% per second for the total capacity of the CPU processes over the RWP mobility model which has increased with the implementations of the CLAODV sets. The
CLAODV implementations increased the average CPU utilisation level in the SIP server to between 42% to 63% per second with set A, 51% to 65% per second with set B, 69% to 82% per second with set C, and 72% to 96% per second with set D.

![IPv4 - RWP](image1)

![IPv4 - RWP-All](image2)

![IPv6 - RWP](image3)

![IPv6 - RWP-All](image4)

**Figure 8-8: Average CPU utilization for the B2BUA SIP server with the CLAODV implementations**

On the other hand, for IPv6 traffic, the average CPU utilisation level for the B2BUA SIP server is in the range of 5% to 27% per second for the total capacity of the CPU processes in the RWP mobility model. The CLAODV implementations increased the average CPU utilisation level in the SIP server to between 13% to 35% per second with set A, 24% to 48% per second with set B, 35% to 58% per second with set C, and 49% to 69% per second with set D. For the IPv6 traffic over the RWP-All scenarios, the average CPU utilisation level of the B2BUA SIP server is in the range of 11% to 39% per second for the total capacity of the CPU processes. The CLAODV implementations increased the average CPU utilisation level in the SIP server to between 29% to 58% per second with set A, 41% to 71% per second with set B, 54% to 80% per second with set C, and 66% to 91% per second with set D. In general, the average CPU utilisation level for the B2BUA SIP server in the RWP mobility model is lower than the level with the RWP-All mobility model because the change of the routing parameters with the RWP is lower than with the RWP-All. As the SIP server moves, the routing parameters update more frequently.
when compared with a MANET with a fixed SIP server. Therefore, the movement of the SIP server consumes more bandwidth and requires higher CPU processing.

8.2.7 Discussion about the CLAODV Implementations over SIP-based VoIP

The investigations for the proposed CLAODV approaches used a set of related performance metrics to evaluate the SIP-based VoIP calls. These investigations linked the results of the CLAODV with the results of the evaluations for the current state of the art for SIP signaling performance of SIP-based VoIP over MANET as represented in Chapter 6. All successful SIP-based calls within the implemented CLAODV systems have similar conditions and pass through the same procedures during the system implementations for the investigated SIP-based calls. However, most of the SIP calls at the beginning of the simulations have unstable readings as a result of the initial integrations of the AODV routing system which takes from 50 to 120 seconds by the beginning of the simulation time. The initial values of the AODV-based MANET system have larger routing values with longer related delays. After the system stability for the simulated events in OPNET, the SIP signaling delays were reduced to a comparable level over the different set of calls.

During the implementations of the CLAODV, the AODV routing parameters have applied variable changes during the registration, initiation, and termination processes. All the enhancements of the SIP signaling processes are related to an adequate level of the dynamic adjustments for the routing table, as proposed in Section 5.7.1. The active frequent modifications for the TTL values activate the Route Requests (RREQ) messages to update the routing table with the best valid routes for the required connectivity. The CLAODV algorithms have also considered the RREP messages during the SIP signaling processes of the investigated scenarios. In addition, the CLAODV algorithms considered saving the CPU cycles and reducing the consumed bandwidth by returning the TTL values for the RREQ to its original default values after reaching the required level of the performance enhancement for the SIP signaling. Because of these frequent adoptions in the routing parameters that depend on the application layer values for the SIP signaling system performance, the CLAODV implementations had consumed larger amounts of bandwidth with higher levels of CPU utilisation that had increased with the mobility of the SIP server, as shown in Section 8.2.6. The call setup process represents the larger part of the SIP signaling system that consumed larger amounts of traffic and requires higher CPU processing. The number of retransmission messages for the SIP/TCP signalling system in the call initiation processes has more delays compared with the registration and termination processes. Table 8-12 summarises the performance enhancement percentages for the investigated SIP processes with the implementations of the CLAODV approaches. The results considered the proposed set of values of Table 8-1 for the SIP performance metrics.
Table 8-12: The performance enhancement percentage for the SIP processes with the implementations of the CLAODV approaches with different set values for the SIP Performance metrics of Table 8-1

<table>
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<tr>
<th>CLAODV</th>
<th>performance enhancement values/percentage</th>
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<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>in (ms)</td>
<td>in %</td>
<td>in (ms)</td>
<td>in %</td>
</tr>
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<td><strong>Registration Process</strong></td>
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</tr>
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<td>576.42</td>
<td>+ 29</td>
</tr>
<tr>
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<td>IPv6</td>
<td>RWP</td>
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<td>+ 25</td>
<td>688.81</td>
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<td>+ 24</td>
<td>1223.17</td>
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<td>+ 12</td>
<td>1545.27</td>
<td>+ 25</td>
</tr>
<tr>
<td><strong>Call Termination Process</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv4</td>
<td>RWP</td>
<td>428.52</td>
<td>+ 6</td>
<td>352.28</td>
<td>+ 22</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>753.21</td>
<td>+ 37</td>
<td>494.37</td>
<td>+ 59</td>
</tr>
<tr>
<td>IPv6</td>
<td>RWP</td>
<td>719.82</td>
<td>+ 2</td>
<td>544.82</td>
<td>+ 26</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>1274.53</td>
<td>+ 29</td>
<td>808.58</td>
<td>+ 55</td>
</tr>
<tr>
<td><strong>Number of Initiated Calls</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv4</td>
<td>RWP</td>
<td>123</td>
<td>- 8</td>
<td>135</td>
<td>+ 2</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>96</td>
<td>- 2</td>
<td>112</td>
<td>+ 14</td>
</tr>
<tr>
<td>IPv6</td>
<td>RWP</td>
<td>119</td>
<td>+ 2</td>
<td>128</td>
<td>+ 6</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>35</td>
<td>- 10</td>
<td>41</td>
<td>+ 5</td>
</tr>
<tr>
<td><strong>Bandwidth Consumptions for Routing</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv4</td>
<td>RWP</td>
<td>26.75</td>
<td>+ 31</td>
<td>29.65</td>
<td>+ 59</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>25.14</td>
<td>+ 16</td>
<td>26.13</td>
<td>+ 20</td>
</tr>
<tr>
<td>IPv6</td>
<td>RWP</td>
<td>20.56</td>
<td>+ 32</td>
<td>23.85</td>
<td>+ 57</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>17.93</td>
<td>+ 15</td>
<td>21.25</td>
<td>+ 26</td>
</tr>
<tr>
<td><strong>CPU Cycles for the SIP server</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv4</td>
<td>RWP</td>
<td>26% to 41%</td>
<td>+ 3 to 19</td>
<td>37% to 51%</td>
<td>+ 13 to 26</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>42% to 63%</td>
<td>+ 1 to 13</td>
<td>51% to 65%</td>
<td>+ 2 to 16</td>
</tr>
<tr>
<td>IPv6</td>
<td>RWP</td>
<td>13% to 35%</td>
<td>+ 2 to 8</td>
<td>24% to 48%</td>
<td>+ 5 to 21</td>
</tr>
<tr>
<td></td>
<td>RWP-All</td>
<td>29% to 58%</td>
<td>+ 1 to 19</td>
<td>41% to 71%</td>
<td>+ 2 to 32</td>
</tr>
</tbody>
</table>

The performance of the AODV routing parameters in the performance evaluation efforts have related effects over the performance of the SIP signaling. The increased number of RREQ messages caused an increase in the number of the RREP messages. The usage of the CLAODV employed dynamic modifications for the TTL values that enhanced the routing performance. The AODV routing parameters show improved performance in the SIP signaling process.
performance had been enhanced because of the values of the routing table that have regular modifications. These regular modifications for the routing table affect the adjustment mechanism of the RREQ and RREP messages. Furthermore, the values of the RDT had increased with the RWP and RWP-All scenarios as a result of the increased impact of the nodes’ mobility and nodes’ lost connectivity that affected the routing efficiency. The performance evaluation of the proposed CLAODV approach had the following results:

- The percentage of the successful calls had increased with set D up to 13% more than the actual normal IPv4 traffic for the RWP, and 25% more than the actual normal IPv4 traffic for the RWP-All. For IPv6, the percentage of successful calls increased up to 16% for RWP and 15% for RWP-All.

- The registration algorithm of the CLAODV approach succeeded in enhancing the registration processes over a large number of calls when compared with the classical SIP registration process. The average time of the registration intervals had reduced for the IPv4 traffic up to 66% for the RWP mobility model and up to 79% for the RWP-All mobility model from the original average registration intervals. For IPv6, the enhancement percentage is about 76% for RWP and 87% for RWP-All from the original average registration intervals.

- The call initiation algorithm for the CLAODV approach reduced the total time intervals of the call setup processes with different percentages depending on the mobility models and the IP Protocol used. For IPv4 traffic, the average time of the call setup intervals had reduced by about 57% for the RWP mobility model and by about 48% for the RWP-All mobility model. For IPv6 traffic, the enhancement of the call setup interval reached to about 45% for RWP and 59% for RWP-All from the original average of the call setup intervals.

- The average call termination time had reduced for the IPv4 traffic with about 70% for the RWP and 77% for RWP-All mobility models from the original average of the call termination time. For IPv6 traffic, the average call termination time reduced to about 71% for the RWP and about 80% for RWP-All mobility models from the original average of the call termination time.

The values of the investigated sets for the proposed CLAODV had shown variable levels of the SIP signaling performance enhancements. Set D represents the best enhancement level for the SIP signaling system over both RWP and RWP-All scenarios, however the bandwidth consumptions and CPU cycles for the SIP server in set D are considered very high compared with the other sets with higher values such as set A or the actual representation of the SIP signaling system. Therefore, much of the set values of the CLAODV performance enhancements have reduced as the performance level of the SIP signaling system, bandwidth consumptions, and CPU utilisation of the SIP server have increased. Furthermore, the mobility of the SIP server in RWP-All scenarios had also increased the bandwidth consumption of the SIP-based VoIP calls and the CPU cycles of the SIP server as that affected the SIP server efficiency,
as discussed in Sections 8.2.4 and 8.2.6. However, the performance of the SIP signaling had not improved when compared with the RWP scenarios as longer delays had been registered with a lower number of successful calls. In addition, the average of consumed bandwidth for IPv4 scenarios for the CLAODV implementations is slightly lower compared with the consumed bandwidth in IPv6 because of the packet overhead of IPv6 traffic that slightly increased the amount of the consumed bandwidth. In addition, the IPv6 implementations in the research study are not supporting the mobility features for the mobile nodes.

### 8.3 Results and Discussion for the CLOLSR-based Implementations

The evaluation results for the proposed Cross-Layer OLSR approach of the SIP-based VoIP implementations will be discussed in this section with the considerations of the CLOLSR design in Section 5.7.2. The evaluations considered the four identified sets of the benchmarked values of the RFC 6076 for the end-to-end SIP signaling performance metrics as shown in Table 8-1. These sets proposed to provide the best effort values for the CLOLSR implementations over the investigated scenarios as had been considered in Section 8.2 for the CLAODV implementations. Set D represents the values of the best case and set A represents the values of the worst case for benchmarked performance metrics. The implementations of the CLOLSR depend on the proposed assumptions in Section 5.7.2. The SIP signaling performance for the CLOLSR implementations is affected by the identified benchmarked sets that reflect on the general performance for the registration, initiation, and termination sessions.

The total number of initiated VoIP calls over the IPv4 OLSR-based MANET for both RWP and RWP-All mobility models are represented in Table 8-13. Set D has a larger number of initiated calls when compared with the actual normal IPv4 traffic and set A has a lesser number of successful calls. The total number of the implemented calls is increased over sets B, C and D. Set D represents the best level of successful VoIP calls with 104 calls for RWP, and 61 calls for RWP-All. The enhancement percentage of successful calls for set D is 59.43% of the total implemented calls, which is 31.43% more than the IPv4 traffic implementations over RWP without CLOLSR. On the other hand, for the RWP-All mobility model, set D has 34.9% of successful calls which is more than the actual IPv4 traffic for RWP-All with 15.47%. For the IPv6 implementations for OLSR-based MANET, the implementations of the CLOLSR showed variable levels of enhancement upon the implemented VoIP calls with the proposed benchmarked sets, as shown in Table 8-14. Set A has a lower number of implemented VoIP calls when compared with the IPv6 based VoIP calls for both RWP and RWP-All mobility models, while the number of successful calls for RWP had increased to 88 calls with set D when compared with 40 calls for normal IPv6 traffic with a percentage of increase of 27.43%. Furthermore, set D had increased the number of successful calls for the RWP-All mobility model to 32 compared with only 3 for normal implementations of IPv6 traffic with a percentage increase that reached 16.58%. The increase by the
CLOLSR is still limited and does not provide a higher level of enhancements over the considered implementations. However, this level of enhancement over OLSR-based MANET with the considered mobility scenarios is still good when compared with the actual low level of performance for SIP calls.

Table 8-13: Number of initiated VoIP calls for IPv4 OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>49</td>
<td>68</td>
<td>79</td>
<td>91</td>
<td>104</td>
</tr>
<tr>
<td>RWP-All</td>
<td>34</td>
<td>42</td>
<td>49</td>
<td>56</td>
<td>61</td>
</tr>
</tbody>
</table>

Table 8-14: Number of initiated VoIP calls for IPv6 OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>40</td>
<td>61</td>
<td>69</td>
<td>76</td>
<td>88</td>
</tr>
<tr>
<td>RWP-All</td>
<td>3</td>
<td>17</td>
<td>21</td>
<td>24</td>
<td>32</td>
</tr>
</tbody>
</table>

The total number of rejected VoIP calls over IPv4 and IPv6 OLSR-based MANET for both RWP and RWP-All mobility models are considered in Table 8-15 and Table 8-16. The number of rejected calls is decreased with sets B and C, where set D has the lowest number of rejected calls for IPv4 traffic with 85 for RWP, and 128 for RWP-All. The total percentage of the rejected calls has reduced to 48.57%, which is 27.43% less than the actual normal IPv4 traffic for RWP, and 73.14% which is 6.86% less than the actual normal IPv4 traffic for RWP-All. For the IPv6 traffic, the number of rejected calls decreased with sets B, C and D. Set D has the lowest number of rejected calls for IPv6 traffic with 118 for RWP, and 140 for RWP-All. The percentage of the rejected calls has reduced to 67.43%, which is 14.86% less than the actual normal IPv6 traffic over RWP, and 80% which is 18.87% less than the actual normal IPv6 traffic for RWP-All.

Table 8-15: Number of rejected VoIP calls for IPv4 OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>133</td>
<td>116</td>
<td>107</td>
<td>98</td>
<td>85</td>
</tr>
<tr>
<td>RWP-All</td>
<td>140</td>
<td>138</td>
<td>135</td>
<td>132</td>
<td>128</td>
</tr>
</tbody>
</table>

Table 8-16: Number of rejected VoIP calls for IPv6 OLSR-based MANET

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>144</td>
<td>137</td>
<td>131</td>
<td>126</td>
<td>118</td>
</tr>
<tr>
<td>RWP-All</td>
<td>173</td>
<td>168</td>
<td>159</td>
<td>151</td>
<td>140</td>
</tr>
</tbody>
</table>

The total call duration is considering the SIP-based call processes and data transmission intervals, as illustrated in Figure 4-2 in Chapter 4. The call duration for a single call is the total required time for the registration, initiation, voice data transmission, and termination processes. The average durations of the
VoIP calls within the implementations of the CLOLSR approaches over OLSR-based MANET using the RWP and RWP-All mobility models are shown in Figure 8-9. The considered VoIP calls are for a duration of 10 seconds for SIP-VoIP calls as identified in Chapter 5 and implemented in OPNET® Modeler scenarios.

For IPv4 traffic with the RWP mobility model, the average call duration is between 15 and 21 seconds for IPv4 and is enhanced to between 13.45 and 17.82 seconds for set A, between 13.13 and 15.18 seconds for set B, between 12.72 and 17.33 seconds for set C, and between 11.86 and 13.77 seconds for set D. With the RWP-All mobility model, the IPv4 traffic has an average between 28 and 37 seconds. This average is enhanced with the CLOLSR implementations to between 22.78 and 29.65 seconds with set A, between 17.42 and 25.43 seconds with set B, between 14.93 and 21.87 seconds with set C, and between 12.76 and 17.13 seconds with set D. On the other hand, for the IPv6 traffic, the average call duration is between 14 and 36 seconds and is enhanced to between 13.75 and 24.37 seconds for set A, between 16.64 to 20.28 seconds for set B, between 16.14 to 17.73 seconds for set C, and between 14.26 to 16.92 seconds for set D. With the RWP-All mobility model, the IPv6 traffic has very long call durations with an average of 35 to 55 seconds. These average values had been enhanced with the implementations of the CLOLSR to between 31.46 and 38.85 seconds with set A, between 27.94 and 30.14 seconds with set B, between 25.1 and 30.12 seconds with set C, and between 21.86 and 25.74 seconds with set D. From the results shown in Figure 8-9, set D has the best enhancement level with implementations of CLOLSR over the total SIP-based call durations over both RWP and RWP-All mobility models and for both IPv4 and IPv6 traffic. The call duration intervals had reduced for IPv4 traffic with about 5 seconds for RWP and 14.65 seconds for RWP-All. For IPv6 traffic, the call duration intervals had reduced on average to 8.67 seconds for RWP and 21.82 seconds for RWP-All.
Chapter 8. Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

8.3.1 SIP Registration Interval with CLOLSR

The earliest stage of the SIP-based VoIP calls is the registration process. The implementations of the CLOLSR approach considered the registration processes for the user agents with the system SIP server, as proposed in Section 5.7.2.1. The implementations of the CLOLSR approach have different levels of performance enhancements over the intervals of the registration processes for both IPv4 and IPv6 SIP-based, with the considerations of the identified sets for the SIP signaling metrics in Table 8-1.

The average registration time for both the caller and the callee to register with the SIP server for the SIP sessions for IPv4 traffic over OLSR-based MANET with the RWP mobility model is represented in Table 8-17. The average registration time for IPv4 traffic is 2.49 seconds and is reduced by the implementations of the CLOLSR to 1.92 seconds with set A, 1.76 seconds with set B, 1.42 seconds with set C, and 973.83 ms with set D. The average time of the registration intervals had reduced for IPv4 traffic with an average of 1.2 seconds with the RWP mobility model. The enhancement percentage with set D is 39.12% when compared with 2.49 seconds of the average registration intervals for the
Chapter 8. Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

original IPv4 traffic. For the RWP-All mobility model, the average registration time for IPv4 traffic is reduced by the implementations of the CLOLSR from 6.02 seconds to 3.03 seconds with set A, 2.54 seconds with set B, 1.96 seconds with set C, and 1.44 seconds with set D. The average time of the registration intervals had reduced for IPv4 traffic with an average of 4.46 seconds with the RWP-All mobility model. The enhancement percentage with set D is 49.53% when compared with 6.02 seconds of the average registration intervals for the original IPv4 traffic.

On the other hand, the average registration for IPv6 traffic over OLSR-based MANET with the RWP mobility model is illustrated in Table 8-18. The average registration time for IPv6 traffic is 5.33 seconds and is reduced with the OLSR implementations to 3.78 seconds with set A, 2.88 seconds with set B, 1.97 seconds with set C, and 1.24 seconds with set D. Therefore, the average time of the registration intervals had reduced for IPv6 traffic with an average of 3.34 seconds for the RWP mobility model. The enhancement percentage with set D is 76.73% when compared with 5.33 seconds of the average registration intervals for the original IPv6 traffic.

For the RWP-All mobility model, the average registration time for IPv6 traffic reduced with the CLOLSR implementation from 13.82 seconds to 5.53 seconds with set A, 4.6 seconds with set B, 3.80 seconds with set C, and 2.80 seconds with set D. The average time of the registration intervals had reduced for IPv6 traffic with an average of 10.26 seconds over the RWP-All mobility model. The enhancement percentage with set D is 20.29% when compared with 13.82 seconds of the average registration intervals for the original IPv6 traffic.

| Table 8-17: Average SIP registration time for SIP clients over IPv4 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations |
|----------------|----------------|----------------|----------------|----------------|
|                | IPv4           | Set A          | Set B          | Set C          | Set D          |
| RWP            | 2489.19        | 1918.31        | 1757.82        | 1420.92        | 973.83         |
| RWP-All        | 6017.38        | 3029.72        | 2541.66        | 1963.59        | 1443.56        |

| Table 8-18: Average SIP registration time for SIP clients over IPv6 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations |
|----------------|----------------|----------------|----------------|----------------|
|                | IPv6           | Set A          | Set B          | Set C          | Set D          |
| RWP            | 5326.59        | 3779.22        | 2875.68        | 1972.04        | 1243.58        |
| RWP-All        | 13819.43       | 5530.92        | 4601.45        | 3799.73        | 2803.15        |

8.3.2 SIP Call Setup Interval with CLOLSR

The main process for the SIP-based VoIP calls is the call initiation process. The general performance of this process is related to the amount of signaling flow between the callers that go through the SIP server as represented in Section 4.2.5 and examined in Section 6.3.3. The average time spent for the call setup processes over the OLSR-based MANET with the implementations of the CLOLSR approach
Chapter 8. Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

is represented in Figure 8-10 and implemented in Section 5.7.2.1. The values of the average call setup time are always more than the values of the SRD performance metrics during all the initiated SIP calls.

The average call setup time for the IPv4 traffic with the RWP mobility model is between 2.93 to 5.18 seconds for IPv4 traffic, between 2.1 to 4.83 seconds for set A, between 1.84 to 3.95 seconds for set B, between 1.35 to 3.78 seconds for set C, and between 0.94 to 2.26 seconds for set D. In the RWP-All mobility model, the average call setup time for IPv4 traffic over OLSR-based MANET is in the range of 5.14 to 7.23 seconds for IPv4, between 4.13 to 6.32 seconds for set A, between 3.65 to 5.27 seconds for set B, between 2.18 to 4.98 seconds for set C, and between 1.87 to 3.96 seconds for set D. The results of the IPv4 implementations showed that the implementations of set D of the CLOLSR enhanced the call setup process and reduced the average call setup time to about 2.52 second for RWP and 3.1 to 3.93 seconds for RWP-All.

![IPv4 - RWP](image1.png) ![IPv4 - RWP All](image2.png)

![IPv6 - RWP](image3.png) ![IPv6 - RWP All](image4.png)

**Figure 8-10: Average SIP call setup time in seconds for OLSR-based MANET using CLOLSR approach**

On the other hand, the implementations of the IPv6 traffic with the RWP mobility model showed an average call setup time between 4.91 to 7.18 seconds. The implementations also showed a level of enhancement on the average call setup time which is between 4.27 to 5.67 seconds for set A, between
3.41 to 5.32 seconds for set B, between 2.21 to 4.43 seconds for set C, and between 1.83 to 3.97 seconds for set D. In the RWP-All mobility model, the average call setup time for IPv6 OLSR-based MANET is in the range of 9.88 to 12.19 seconds for IPv6 traffic. The average call setup time over IPv6 traffic is enhanced to be between 8.69 to 11.14 seconds for set A, between 7.38 to 9.43 seconds for set B, between 5.95 to 7.92 seconds for set C, and between 4.89 to 6.73 seconds for set D. The results conclude that set D of the CLOLSR has the best level of enhancement for the average call setup time for IPv6 traffic as it reduced the average call setup time with 2.93 to 3.26 seconds for RWP and 5.27 to 6.15 seconds for RWP-All.

8.3.3 SIP Call Termination Interval with CLOLSR

The call termination process is the last part of the SIP-based VoIP call where one of the call’s parties finishes the call by sending a call termination request through the B2BUA-based SIP server. The call termination process was discussed for OLSR-based MANET in Section 4.2.6 and investigated for benchmarking in Section 6.3.4. The effect of the call termination process over the performance of the SIP-based VoIP is lowest when compared with the registration and call setup processes. The call termination intervals are equal to the SDD values of the end-to-end SIP performance metrics. The CLOLSR implementations in Section 5.7.2.3 with identified sets for the call termination in Table 8-1 showed variable level of enhancements over both IPv4 and IPv6 traffic.

In Table 8-19, the average call termination time for the initiated SIP calls between the caller and the callee for IPv4 traffic over the RWP mobility model is 1.1 seconds. The implementations of the CLOLSR reduced the average call termination time for IPv4 traffic to 908.27 ms with set A, 712.21 ms with set B, 563.72 ms with set C, and 298.39 ms with set D. The total average time of the call termination intervals had been reduced for IPv4 traffic with an average of 524.37 ms for RWP. The enhancement percentage is 27.42% for set D when compared with the original average call termination intervals of IPv4 traffic. For the RWP-All mobility model, the average call termination time for IPv4 traffic is 1.82 seconds. The implementations of the CLOLSR reduced the average call termination time for IPv4 traffic to 1.42 seconds with set A, 1.13 seconds with set B, 917.59 ms with set C, and 723.87 ms with set D. Therefore, the average call termination time had reduced for IPv4 traffic with an average of 0.9 second for the scenarios of the RWP mobility models. The percentage of the enhancement level is 39.81%, compared with 1.82 seconds for the original average call termination time for IPv4 traffic.

On the other hand, Table 8-20 shows the average call termination time for the initiated SIP calls between the caller and the callee for IPv6 traffic over the RWP mobility model. The average call termination time for IPv6 traffic is 1.89 seconds and is reduced by the implementations of the CLOLSR to 1.18 seconds with set A, 912.76 ms with set B, 753.44 ms with set C, and 427.63 ms with set D. From the results, the average of the call termination time for IPv6 traffic reduced with an average of 743.33 ms
to 1313.74 ms for RWP, where the enhancement percentage is from 39.41% to 69.62% of the original average of the call termination intervals. For the RWP-All mobility model, the average call termination time for IPv6 traffic is reduced by the implementations of the CLAODV from 5.41 seconds to 3.91 seconds with set A, 2.95 seconds with set B, 2.01 seconds with set C, and 1.11 seconds with set D. The average time of the termination intervals reduced for IPv6 traffic with an average of 3.45 to 4.37 seconds for the RWP-All mobility model. The enhancement percentage is from 37.13% to 20.52%, compared with the original average of the call termination processes for IPv6 traffic.

Table 8-19: Average SIP termination time for SIP callers over IPv4 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv4</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>1088.14</td>
<td>902.71</td>
<td>712.21</td>
<td>563.72</td>
<td>298.39</td>
</tr>
<tr>
<td>RWP-All</td>
<td>1818.38</td>
<td>1420.66</td>
<td>1127.38</td>
<td>917.59</td>
<td>723.87</td>
</tr>
</tbody>
</table>

Table 8-20: Average SIP termination time for SIP callers over IPv6 OLSR-based MANET in milliseconds (ms) for CLOLSR implementations

<table>
<thead>
<tr>
<th></th>
<th>IPv6</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td>RWP</td>
<td>1886.68</td>
<td>1175.54</td>
<td>912.76</td>
<td>753.44</td>
<td>427.63</td>
</tr>
<tr>
<td>RWP-All</td>
<td>5412.85</td>
<td>3897.27</td>
<td>2952.36</td>
<td>2009.54</td>
<td>1109.35</td>
</tr>
</tbody>
</table>

8.3.4 SIP Server Efficiency with CLOLSR

For SIP-based VoIP implementations that are using the B2BUA SIP server, all the SIP processes are relayed through the SIP server, as discussed in Section 6.3.5. The SIP server manages and transfers the SIP/TCP messages between different parties. The number of active SIP/TCP connections in the SIP server reflects the performance of the SIP signaling system over the SIP processes. A high number of active SIP/TCP messages does not always mean that the performance is acceptable, however it shows a level of active communications are considered by the SIP server and the SIP processes are likely to perform better. In Figure 8-11, the average number of active SIP/TCP connections over the B2BUA-based SIP server is represented for each 5 seconds of the simulation time with the considerations of the CLOLSR implementations for the OLSR-based MANET.

For IPv4 traffic, the average number of active SIP/TCP connections per 5 seconds in the SIP server for the RWP mobility model is between 4 and 5. This average number increased with the CLOLSR implementations to between 4 to 6 with set A, between 4 to 6.25 with set B, between 4 to 6.75 with set C, and between 4 to 7 with set D. In the RWP-All mobility model, the average number of active SIP/TCP connections for IPv4 traffic on the SIP server is between 4 to 5 active connections per 5 seconds then it dropped to 1 to 2 active connection per 5 seconds. This drop is a result of the lack of connectivity of the OLSR routes that affect the dropped performance of the SIP/TCP connectivity. The number of active
connections per 5 seconds had increased to between 2.5 to 4.75 with set A, between 2.5 to 5 with set B, between 2.5 and 5.25 with set C, and between 2.5 and 5.75 with set D.

For IPv6 traffic with the RWP mobility model, the average number of active SIP/TCP connections per second in the SIP server is between 4 to 5 connections. The average number of active SIP/TCP connections increased with the implementations of the proposed CLOLSR sets to between 4 to 5.5 with set A, between 4 to 6 with set B, between 4 to 6.25 with set C, and between 4 to 6.5 with set D. In the RWP-All mobility model, the average number of active SIP/TCP connections for IPv6 traffic on the SIP server starts with 2 to 3 active connections per 5 seconds then it dropped down to 1 active connection per 5 seconds. The average number of active SIP/TCP connections increased with the implementations of the CLOLSR to between 1.75 to 3.25 with set A, between 2 to 3.25 with set B, between 2.75 to 3.25 with set C, and between 3 to 3.75 with set D. The implementations of set D within the CLOLSR had increased the number of active SIP/TCP connections over the SIP server for both IPv4 and IPv6 traffic with different mobility models. In general, the SIP signaling for the SIP processes performed better when the number of active SIP/TCP connections had increased.

Figure 8-11: Average number of active SIP/TCP connections per 5 seconds in the SIP server for OLSR-based MANET
8.3.5 Routing Performance with CLOLSR

The performance of the SIP signaling on MANET is affected by the routing performance. For OLSR-based MANET, the routing parameters have an important role over the performance enhancement for the implementations of the SIP-based applications. The main OLSR routing parameters have been discussed in Section 4.4.4 and evaluated for SIP-based VoIP in Section 6.7. In this section, the related OLSR routing parameters will be investigated regarding the enhancement level of the CLOLSR implementations over the SIP signaling performance. In Figure 8-12, the average number of HELLO messages for OLSR routing processes that were sent during the simulation time is represented. The route discovery status between the nodes within the dynamic mobility nature of MANET depends on the number of activated HELLO messages. As the number of HELLO messages increases, the connectivity between the nodes becomes better over the dynamic nature. However, the routing overhead increases with the increase of HELLO traffic in the MANET. The increased number of HELLO messages means that only a few of the known routes are known between the communicating nodes because of the nodes’ dynamic mobility that affects the reachability, and availability, and causes rapid disconnections. For IPv4 traffic over the RWP mobility model, the average HELLO traffic sent in the MANET is between 4.4 Kbits/s and 5.9 Kbits/s. The average number increased with the implementations of the proposed CLOLSR approaches to between 5.5 Kbits/s and 9.2 Kbits/s with set A, between 6.7 Kbits/s and 10.3 Kbits/s with set B, between 7.6 Kbits/s and 11.7 Kbits/s with set C, and between 7.3 Kbits/s and 12.2 Kbits/s with set D. For the RWP-All mobility model, the average number of HELLO traffic that were sent for IPv6 traffic implementations increased to be between 6.5 Kbits/s and 7.8 Kbits/s. With the implementations of the CLOLSR, the average number of HELLO traffic sent for IPv6 traffic increased to between 8.8 Kbits/s and 11.7 Kbits/s with set A, 9.1 Kbits/s and 12.5 Kbits/s with set B, 9.6 Kbits/s and 15.1 Kbits/s with set C, and 10.1 Kbits/s and 17.8 Kbits/s with set D.

On the other hand, for IPv6 traffic over the RWP mobility model, the average HELLO traffic sent in the MANET is between 10.1 Kbits/s and 12.9 Kbits/s. This average number increased with the CLOLSR implementations to between 10.3 Kbits/s and 14.4 Kbits/s with set A, between 10.9 Kbits/s and 15.4 Kbits/s with set B, between 11.7 Kbits/s and 16.8 Kbits/s with set C, and between 12.1 Kbits/s and 18.3 Kbits/s with set D. The average number of HELLO traffic sent for IPv6 traffic over the RWP-All mobility model increased to between 9.5 Kbits/s and 14.3 Kbits/s. With the implementations of the CLOLSR, the average number of HELLO traffic sent for IPv6 traffic increased to between 10.6 Kbits/s and 16.4 Kbits/s with set A, 11.2 Kbits/s and 18.2 Kbits/s with set B, 10.8 Kbits/s and 18.5 Kbits/s with set C, and 11.1 Kbits/s and 20.3 Kbits/s with set D. From the previous results, the implementations of the CLOLSR increased the number of HELLO messages over both RWP and RWP-All mobility models.
which indicated the active number of requested route updates during the nodes’ mobility for SIP applications.

Figure 8-12: Average HELLO traffic sent in MANET for OLSR-based MANET using CLOLSR approach

Figure 8-13 shows the average number of OLSR Topology Control (TC) messages that were forwarded in the OLSR-based MANET with the implementations of CLOLSR approaches during the SIP-based VoIP calls. The number of TC messages that were sent during the SIP processes indicates the status of the routes. The TC values decrease with the increase in the mobility factor of the nodes in the MANET. For IPv4 traffic over the RWP mobility model, the average number of the sent TC messages in the MANET is between 787 and 1343 messages for every 10 seconds. This average number decreased to the range between 846 to 1271 with set A, between 921 to 1315 with set B, between 1060 to 1349 with set C, and between 1036 to 1394 with set D. The average of the sent TC messages reduced in the RWP-All implementations for IPv4 traffic to become in the range of 583 to 1243 messages for every 10 seconds. The average number increased to the range between 688 to 1237 with set A, between 720 to 1285 with set B, between 779 to 1281 with set C, and between 907 to 1403 with set D. On the other
hand, for IPv6 traffic over the RWP mobility model, the average number of the sent TC messages in the MANET is between 763 and 1114 messages for every 10 seconds. The average number of TC decreased to between 774 to 1109 with set A, between 813 to 1178 with set B, between 911 to 1209 with set C, and between 923 to 1235 with set D. In the RWP-All scenarios, the average number of TC messages for IPv6 traffic is between 410 and 1169 messages for every 10 seconds. The average number of TC decreased to between 566 to 1092 with set A, between 617 to 1165 with set B, between 672 to 1154 with set C, and between 626 to 1128 with set D.

![Graph showing average number of OLSR TC messages forwarded in MANET for OLSR-based MANET using CLOLSR approach.](image)

Figure 8-13: Average number of OLSR TC messages forwarded in MANET for OLSR-based MANET using CLOLSR approach

Figure 8-14 shows the average routing traffic that is sent by the caller node for the implementations of the SIP-based VoIP applications with the investigations of the CLOLSR approaches. For IPv4 traffic, the average number of sent routing traffic from the caller node is in the range of 1.23 to 1.93 Kbits/s over the RWP mobility model. The average number of the sent routing traffic increased with the CLOLSR implementations to between 1.39 to 2.17 Kbits/s with set A, between 1.52 to 2.52 Kbits/s with set B, between 1.62 to 2.54 Kbits/s with set C, and between 1.91 to 2.92 Kbits/s with Set D. In the
RWP-All scenarios, the average of sent routing traffic is in the range of 1.48 to 1.57 Kbits/s, that increased with the implementations of the CLOLSR sets. The average number of the sent routing traffic increased to be between 1.75 to 2.67 Kbits/s with set A, between 2.26 to 2.97 Kbits/s with set B, between 2.41 to 3.99 Kbits/s with set C, and between 2.51 to 3.49 Kbits/s with set D.

![IPv4 - RWP](image1.png) ![IPv4 - RWP-All](image2.png)

![IPv6 - RWP](image3.png) ![IPv6 - RWP-All](image4.png)

**Figure 8-14:** Average routing traffic sent by the caller node in (Bits/Sec) for OLSR-based MANET using CLOLSR approach

On the other hand, for IPv6 traffic, the average number of sent routing traffic from the caller node is between 500 to 960 bits/s with the RWP mobility model. The average number of the sent routing traffic had increased with the CLOLSR implementations to between 0.98 bits/s to 1.51 Kbits/s with set A, between 1.05 to 1.67 Kbits/s with set B, between 1.11 to 2.28 Kbits/s with set C, and between 1.1 to 2.39 Kbits/s with set D. In the RWP-All scenarios, the average number of sent routing traffic from the caller node for IPv6 traffic is between 0.62 to 1.21 Kbits/s. The average number of the sent routing traffic increased to be between 1.41 to 2.46 Kbits/s with set A, between 1.72 to 2.41 Kbits/s with set B, between 1.82 to 2.56 Kbits/s with set C, and between 1.92 to 2.98 Kbits/s with set D. The increases in the total routing traffic sent resulted from the increased number of the HELLO and TC messages sent.
for the dynamic route adjustment processes of the OLSR parameters as a result of the implementations for the proposed CLOLSR approaches.

### 8.3.6 Bandwidth Consumptions and CPU Utilisation for CLOLSR Implementations

The implementations of the CLOLSR approaches for the three SIP processes have different levels of effect over the network performance in general and the SIP calls’ agents in particular. The CLOLSR implementations affect the total consumed bandwidth in the MANET and the CPU utilisation level of the B2BUA SIP server for the SIP-based VoIP calls. Figure 8-15 represents the average consumed bandwidth for the OLSR-based MANET routing processes during the implementations of the CLOLSR for the investigated scenarios.

*Figure 8-15: Average consumed bandwidth for routing data in (Bits/Sec) for OLSR-based MANET using CLOLSR approach*
For IPv4 traffic, the average consumed bandwidth for routing data in the OLSR-based MANET is between 21 to 43 Kbits/s over the RWP mobility model. The implementations of the CLOLSR increased the average consumed bandwidth for the routing data to between 37 to 38.1 Kbits/s with set A, 29 to 49.3 Kbits/s with set B, 31.2 to 51.5 Kbits/s with set C, and 31.5 to 58.5 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for the routing data of the IPv4 traffic is in the range of 10.5 to 29.7 Kbits/s, which increased with the CLOLSR implementations. The CLOLSR implementations increased the average of total consumed bandwidth to be between 12.5 to 32.8 Kbits/s with set A, 17.2 to 38.3 Kbits/s with set B, 20.1 to 41.5 Kbits/s with set C, and 22.7 to 48.6 Kbits/s with set D. The total consumed bandwidth for routing messages in the IPv6 OLSR scenarios over the RWP mobility model were in the range of 5 to 10 Kbits/s. The implementations of the CLOLSR had increased the total average of the consumed bandwidth to between 15.7 to 30.4 Kbits/s with set A, 15.7 to 46.4 Kbits/s with set B, 15.8 to 38.9 Kbits/s with set C, and 15.8 to 44.7 Kbits/s with set D. In the RWP-All scenarios, the average consumed bandwidth for the routing data of the IPv6 traffic is between 1.8 to 14.7 Kbits/s, which increased with the implementations of CLOLSR. The CLOLSR implementations increased the average of the total consumed bandwidth to between 17.5 to 21.7 Kbits/s with set A, 18.1 to 24.7 Kbits/s with set B, 18.6 to 28.7 Kbits/s with set C, and 18.9 to 30.3 Kbits/s with set D.

During the implementations of the CLOLSR for SIP-based VoIP calls, the CPU utilisation level for the B2BUA SIP server has variable increases depending on the identified values of the proposed sets. The increase in the CPU utilisation level results is normal as the number of the running processes of the routing enhancement procedures of the proposed algorithms had also increased. Figure 8-16 shows the average level of the CPU utilisation for the B2BUA SIP server in the OLSR-based MANET during the simulation time for the investigated scenarios. For IPv4 traffic, the average CPU utilisation level for the B2BUA SIP server is in the range of 12% to 39% per second for the total capacity of the CPU processes in the RWP mobility model. The CLOLSR implementations increased the average CPU utilisation level in the SIP server to between 29% to 54% per second with set A, 41% to 68% per second with set B, 58% to 78% per second with set C, and 64% to 89% per second with set D. For the IPv4 traffic over the RWP-All scenarios, the average CPU utilisation level of the B2BUA SIP server is in the range of 22% to 52% per second from the total capacity of the CPU processes. The CLOLSR implementations increased the average CPU utilisation level in the SIP server to between 41% to 66% per second with set A, 47% to 79% per second with set B, 61% to 86% per second with set C, and 71% to 99% per second with Set D.

On the other hand, for IPv6 traffic, the average CPU utilisation level for the B2BUA SIP server is in the range of 8% to 23% per second for the total capacity of the CPU processes in the RWP mobility model. With the implementations of the CLOLSR, the average CPU utilisation percentage in the SIP server increased to between 18% to 38% per second with set A, 23% to 46% per second with set B,
33% to 58% per second with set C, and 48% to 74% per second with set D. In the RWP-All scenarios, the average CPU utilisation level of the B2BUA SIP server for IPv6 traffic is in the range of 12% to 36% per second for the total capacity of the CPU processes which increased with the implementations of the CLOLSR sets. The CLOLSR implementations increased the average CPU utilisation level in the SIP server to between 25% to 52% per second with set A, 37% to 66% per second with set B, 49% to 76% per second with set C, and 60% to 86% per second with set D.

![Figure 8-16: Average CPU utilization for the B2BUA SIP server with the CLOLSR implementations](image)

**8.3.7 Discussion about the CLOLSR Implementations over SIP-based VoIP**

The proposed CLOLSR approaches for the implementations of SIP-based VoIP applications over OLSR-based MANET had been investigated in the previous subsections. The implementations of the CLOLSR considered the sets identified in Table 8-1. With the considerations of related performance metrics of RFC 6076 and the evaluation efforts of the current state at Chapter 6, the evaluation of the CLOLSR implementations had been studied and compared regarding the related OLSR routing performance factors. At the beginning of the simulations, all the MANET nodes were used to build its
own routing tables depending on the OLSR algorithms at Section 2.12. This process consumes a high amount of bandwidth and requires a longer time for MANET with a high density of nodes as in the implemented scenarios. During the simulations, the initial TC routing values of the OLSR-based MANET have longer delays that were reduced until the system reached its initial level of stability with the required initial entries of the routing table values. Mostly, the MANET reached its initial level of stability at 250 to 350 seconds from the beginning of the simulation. With continuous and dynamic mobility of MANET nodes, the update processes of the routing tables increased the number of sent HELLO and TC messages. However, this increase in the routing messages is not reflecting the actual health and efficiency of the MANET routes. In addition, the high number of route messages in OLSR does not mean that the reachability with the requested node/nodes does not exist. It means that the nodes used to update their routing tables based on the proactive nature of OLSR to fulfil with any requested routes in the MANET.

With the implementations of the CLOLSR algorithm over the three processes of the SIP-based VoIP calls, the OLSR routing parameters are relatively updated depending on the considered reachability delays or destination availability in the routing tables. The modifications on the routing parameters are related to the level of the dynamic adjustments in the routing table, as proposed in Section 5.7.2. The values of the HELLO and the TC intervals are used to modify the values of the Multiple Point Relay (MPR) of the OLSR routing protocol. Therefore, the MPR values increase with the increments of the HELLO and TC values with regard to the OLSR MPR selection mechanism in Algorithm 2.3. With the implementations of the CLOLSR, the SIP server succeeded in providing better performance by increasing the frequent updates for the routing table entries of the correspondent nodes. The performance levels for the registration and termination processes with the CLOLSR implementations provide good enhancement levels when compared with the actual current signaling system for both IPv4 and IPv6 traffic for the RWP and RWP-All mobility models. The call setup processes have also had an acceptable level of enhancement over the RWP mobility model. However, for RWP-All, the call setup processes showed variable levels of enhancement with CLOLSR implementations, especially with IPv6 traffic. The CLOLSR approaches also save the CPU cycles and reduce the consumed bandwidth by returning the MPR values for the correspondent nodes to their original default values. This happens just when the SIP calls’ processes reach the required level of the performance enhancement. However, the results showed that the CLOLSR implementations had consumed larger amounts of bandwidth with higher levels of CPU utilisations. This increase happened as a result of the frequent adoptions in the routing parameters because of the mobility of the correspondent nodes. The successful adoptions for the OLSR routing parameters depend on the application layer values to enhance the SIP signaling system performance that increased the level of CPU utilisation and the consumed bandwidth. The call initiation process consumed the largest amount of traffic and used higher CPU cycles when compared with the registration and termination processes. Table 8-21 summarises the performance enhancement.
percentages for the investigated SIP processes with the implementations of the CLOLSR approaches. The results considered the proposed set of values of Table 8-1 for the SIP performance metrics.

Table 8-21: The performance enhancement percentage for the SIP processes with the implementations of the CLOLSR approaches with different set values for the SIP performance metrics of Table 8-1

<table>
<thead>
<tr>
<th>CLOLSR</th>
<th>performance enhancement values/percentage</th>
<th>Set A</th>
<th>Set B</th>
<th>Set C</th>
<th>Set D</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>in (ms)</td>
<td>in %</td>
<td>in (ms)</td>
<td>in %</td>
<td>in (ms)</td>
</tr>
<tr>
<td><strong>Registration Process</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>1918.31</td>
<td>+ 23</td>
<td>1757.82</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>3029.72</td>
<td>+ 50</td>
<td>2541.66</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>3779.22</td>
<td>+ 29</td>
<td>2875.68</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>5530.92</td>
<td>+ 60</td>
<td>4601.45</td>
</tr>
<tr>
<td><strong>Call Setup Process</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>3455.13</td>
<td>+ 12</td>
<td>2867.72</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>5221.66</td>
<td>+ 21</td>
<td>4462.14</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>4967.91</td>
<td>+ 11</td>
<td>4365.85</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>9915.47</td>
<td>+ 18</td>
<td>8405.65</td>
</tr>
<tr>
<td><strong>Call Termination Process</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>902.71</td>
<td>+ 17</td>
<td>712.21</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>1420.66</td>
<td>+ 22</td>
<td>1127.38</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>1175.54</td>
<td>+ 38</td>
<td>912.76</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>3897.27</td>
<td>+ 28</td>
<td>2952.36</td>
</tr>
<tr>
<td><strong>Number of Initiated Calls</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>68</td>
<td>+ 39</td>
<td>79</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>42</td>
<td>+ 24</td>
<td>49</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>61</td>
<td>+ 53</td>
<td>69</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>17</td>
<td>+ 467</td>
<td>21</td>
</tr>
<tr>
<td><strong>Bandwidth Consumptions for Routing</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>35.56</td>
<td>+ 21</td>
<td>40.15</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>22.67</td>
<td>+ 40</td>
<td>35.79</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>23.68</td>
<td>+ 237</td>
<td>32.14</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>20.13</td>
<td>+ 217</td>
<td>23.97</td>
</tr>
<tr>
<td><strong>CPU Cycles for the SIP server</strong></td>
<td>IPv4</td>
<td>RWP</td>
<td>29% to 54%</td>
<td>+ 2 to 15</td>
<td>41% to 68%</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>41% to 66%</td>
<td>+ 1 to 14</td>
<td>47% to 79%</td>
</tr>
<tr>
<td></td>
<td>IPv6</td>
<td>RWP</td>
<td>18% to 38%</td>
<td>+ 2 to 15</td>
<td>23% to 46%</td>
</tr>
<tr>
<td></td>
<td></td>
<td>RWP-All</td>
<td>25% to 52%</td>
<td>+ 1 to 16</td>
<td>37% to 66%</td>
</tr>
</tbody>
</table>
Chapter 8. Cross-Layer Optimisation Approach for SIP Signaling Performance over MANET

The OLSR routing efficiency during the implementations of the SIP-based VoIP calls reflects on the general performance of the SIP signalling system as revealed from the Chapter 6 results. The CLOLSR used a frequent adoption for the number of the HELLO and Topology Control (TC) messages to reduce the route discovery time between the MANET nodes with a dynamic mobility nature. The dynamic routing enhancements of the CLOLSR approaches improve the efficiency of the Multipoint Relay (MPR) sets for the OLSR routing protocol. Therefore, with the implementations of the CLOLSR, the MPR selection mechanism will be enhanced with a dynamic modification for the values of the OLSR routing table. The CLOLSR provides a reliable detection system for any raised delays or undeliverable SIP messages over both RWP and RWP-All mobility models for the SIP processes. The proactive routing nature of the OLSR is still the main issue within the implementations of the CLOLSR approaches over MANET nodes. This can be enhanced by reducing the number of required second hops during the route discovery processes to provide routes with shorter update periods. The performance evaluation of the proposed CLOLSR approach has shown the following results:

- The percentage of successful calls had increased with the implementations of the CLOLSR with set D that reached to 31% more than the actual normal IPv4 traffic for the RWP, and 16% more than the actual normal IPv4 traffic over the RWP-All. For IPv6, the percentage of successful calls increased with 27% for RWP and 17% for RWP-All.
- The registration process had been enhanced with the implementations of the CLOLSR approach with the limited number of the successful initiated calls when compared with the classical OLSR MANET. The average time of the registration intervals had reduced for the IPv4 traffic to 39% for the RWP mobility model and up to 50% for the RWP-All mobility model of the original average registration intervals. For IPv6, the enhancement percentage is about 77% for RWP and 20% for RWP-All of the original average registration intervals.
- The implementation of the call initiation algorithm of the CLOLSR approach succeeded in reducing the total time intervals of the call setup processes with different percentages. For IPv4 traffic, the average time of the call setup intervals had reduced to about 61% for the RWP mobility model and to about 58% for the RWP-All mobility model. For IPv6 traffic, the enhancement of the call setup interval reached to about 56% for RWP and 63% for the RWP-All from the original average of the call setup intervals.
- The average call termination time reduced with the implementations of CLOLSR over the IPv4 traffic to about 27% for the RWP and 40% for RWP-All mobility models from the original average of the call termination time. For IPv6 traffic, the average call termination time reduced up to 70% for the RWP and up to 37% for the RWP-All mobility models from the original average of the call termination time.
The implementations of the CLOLSR approaches were used to determine a set of values to control the required enhancement level of the SIP signaling performance. Among all the proposed sets in Table 8-1, set D represents the best enhancement level for the SIP signaling system performance over different mobility models. This high level in the SIP signaling performance within set D also required a high level of CPU utilisation and bandwidth consumptions as shown in Section 8.3.6. Therefore, the OLSR routing parameters came to their best possible values for the CLOLSR implementations with the considered values of set D. However, the mobility of the SIP server still affects the overall performance, even with the implementations of set D, as long delays within the SIP processes were registered with a low number of successful calls. Therefore, the OLSR proactive routing nature of route discovery is still the main issue within the implementations of the CLOLSR approaches over MANET nodes over the different processes of SIP-based VoIP.

8.4 Discussion about the Proposed Cross-Layer Approaches

The MANET nodes’ density and the dynamic mobility represent the main challenges for the implementations of the proposed Cross-Layer approaches. In addition, the B2BUA SIP server location, mobility, and efficiency affect the performance enhancement level of the Cross-Layer approaches. This can be avoided by using a high performance B2BUA SIP server, using two or more B2BUA SIP servers, and a centralised location with high reachability in the MANET. The evaluation results for the SIP-based VoIP over the current classical MANET and the proposed Cross-Layer approaches identified a comparable difference between IPv4 and IPv6 implementations. Therefore, IPv4 can be nominated as the most appropriate protocol for the current implementations of the SIP-based applications over MANET. However, the performance enhancement efforts for IPv6 mobility issues can improve the capabilities of IPv6 and the implementations of the SIP signaling systems over a dynamic MANET. It can be concluded from the previous results that the difference between IPv4 and IPv6 protocols for SIP signaling implementations over MANET are not very significant. However, the IPv4 is still considered as the better choice for SIP-based VoIP over MANET even with the implementations of the Cross-Layer approaches, as the IPv6 has implementation issues regarding the nodes’ mobility support.

The size of the routing bandwidth for the proposed approach in [164] is acceptable when compared with the limited number of nodes (hops), call numbers, and the fixed SIP server that was used, which has no mobility features. However, the total consumed bandwidth for the routing data increased with an increase in the nodes and mobility ratio, which was in the range of 1 to 15 Kbits/s. In addition, the delays to the SIP processes are high in [164] compared to the evaluated delays in the proposed CLAODV/CLOLSR. This is due to the authorisation messages for the admission control system between the caller agents and the SIP server, which were identified as enhancing the performance of the SIP processes and the SIP server for the approach in [164]. The implementation of the Cross-Layer
approaches in this chapter shows that the bandwidth overhead is limited to the dynamic routing adoption to enhance the performance of the SIP processes. The investigation of the Cross-Layer approaches also shows a lower level of delays for the call initiation processes over the number of MANET nodes when compared with the AdSIP approach implementations in [97]. The session establishment delays were in the range of 0.88 to 1.60 seconds with a node capacity of between 20 and 30 nodes. This low range in the call setup time for the AdSIP representations is related to the decentralised nature of the AdSIP where each node acts like a SIP server to register and contact other caller nodes. The advantage of the Cross-Layer approach over the AdSIP is that it provides a centralised connection with the SIP server that shows a slightly better performance when compared with the AdSIP implementation results. In general, the reliable level of the CLAODV and CLOSR implementations is higher with OPNET when compared with the reliable level of implementation for ns-2 as in [97] and [164]. On the other hand, the Cross-Layer approaches show an acceptable level of performance enhancement regarding the SIP process delays when compared with research concerning the SIP messaging functionality enhancement. In [94] the self-organising approach for the SIP messages of the SIP processes over MANET (AODV and OLSR) has an average delay of between 5.75 seconds and 14.04 seconds for the proposed SIP Servlet Engine (SSE) over MANET scenarios. The Cross-Layer delays for the SIP processes are in the range of 300 ms to 5 seconds for the investigated scenarios.

8.5 Summary

The proposed CLAODV and CLOLSR approaches provide a compatible performance enhancement mechanism for the registration, call setup, and call termination processes of the SIP-based VoIP applications over MANET. The proposed approaches had been introduced and discussed in Chapter 6. These Cross-Layer approaches are designed to reduce the routing overhead of MANET during the implementations of the SIP-based VoIP applications. The proposed approaches are applied over the caller’s side and/or the SIP server’s side. In addition, these approaches provide an efficient usage for MANET bandwidth. The evaluation results illustrated that the proposed approaches performed well and provided an enhanced level of performance for the SIP-based calls.

This chapter began with a review for the Cross-Layer implementations for the SIP signaling system over both AODV-based and OLSR-based MANET. In addition, the simulation results of the implementations for the CLAODV and CLOLSR in the OPNET® Modeler were represented and discussed in this chapter. Furthermore, related SIP performance metrics, routing parameters, and system performance metrics had been investigated in this chapter regarding the proposed approaches and were compared with the performance of the actual existing systems. In general, the proactive nature of the OLSR routing protocol adds extra overhead for the routing traffic over the OLSR-based MANET, especially with the real-time applications. Therefore, the design and the implementations of the Cross-
Layer approaches for OLSR-MANET are more challenging with low levels of performance enhancement when compared with the proposed CLAODV for the AODV-based MANET. In addition, the overall performance enhancement level of the CLAODV implementations is higher than the overall performance enhancement level of the CLOLSR implementations, in terms of the total successful initiated calls, the performance enhancement of the SIP processes, the total MANET bandwidth consumption, and the utilisation level of the SIP server.

As the Cross-Layer approaches improved the SIP signaling performance, however, these approaches still have high levels of CPU utilisation, especially for the SIP server, that could increase the level of the energy consumption. Furthermore, the average of the CPU utilisation level for the B2BUA SIP server with the RWP mobility model is lower than the level with the RWP-All mobility model with the implementations of the Cross-Layer approaches. This difference in the CPU processing percentage happens because the changes in the routing parameters within the RWP mobility model are lower than the implementations with RWP-All. The routing parameters update more frequently with the movement of the SIP server when compared with a MANET system with a fixed SIP server. The movement of the SIP server increases the required number of updates for the routing data that consumes more bandwidth and increases the CPU's processing cycles of the SIP server. In general, the average of the consumed bandwidth with the implementations of the proposed Cross-Layer approaches had increased with enhancement of the SIP signaling performance. The main limitation of the Cross-Layer approaches is the implementations over different routing platforms with various routing clusters. As the CLAODV only supports the AODV-based MANET and the CLOLSR only supports the OLSR-based MANET, the Cross-Layer approaches need to be related to the routing protocol that is being worked over. In addition, the benchmarked values for the Cross-Layer approaches need to be identified as static values for the proposed approaches before the system is implemented.
Chapter 9

9 Conclusions and Future Work

This research has been conducted on the SIP-based VoIP applications over MANET. Both AODV and OLSR MANET routing protocols have been considered in this research study. The MANET nature is characterised by the nodes’ mobility, capacity, connectivity, and the routing protocol. Therefore, the implementations of the SIP-based applications in MANET are affected by both the SIP signaling representations and the MANET nature. In this thesis, a general overview about the three main performance factors for SIP-based VoIP over MANET has been given. These performance factors are the SIP signaling system, the VoIP QoS, and the MANET performance. A number of related research issues have been discussed in the literature review regarding the performance metrics of the SIP-based VoIP over MANET. As the SIP signaling performance over MANET is a main concern of this research study, the end-to-end SIP performance metrics of the RFC 6076 was used to identify the performance evaluation of the SIP-based VoIP applications as shown in Chapters 5, 6, and 7. In addition, the performance metrics have been employed for the Cross-Layer performance enhancement approaches of the SIP signaling systems over MANET. The proposed CLAODV and CLOLSR approaches have shown a good level of enhancements for both IPv4 and IPv6 MANET when compared with the normal classical implementations. The SIP signaling performance had been enhanced in the registration, initiation and termination processes. In addition, these Cross-Layer approaches have been proved to be bandwidth efficient with an acceptable level of resource utilisation. In the following sections, the conclusions are represented and the related future works are identified. The research efforts in this thesis support the implementations of SIP signaling over MANET for VoIP and multimedia applications.

9.1 Conclusions

The performance of SIP signaling system over MANET was evaluated in this thesis. In addition, novel Cross-Layer performance enhancement approaches had been proposed, implemented, and evaluated to enhance the performance of the SIP signaling system over MANET. It also provided a comparative analysis for the SIP signaling over IPv4 and IPv6 protocols. Figure 9-1 represents a summary of the investigations conducted and the contributions achieved in this thesis. In this research study, the benchmarking efforts for the Registration Request Delays (RRD), Session Request Delays (SRD), and the Session Disconnect Delays (SDD) of the RFC 6076 end-to-end SIP performance metrics have been
implemented, studied and investigated. The benchmarking values of the SIP-based VoIP applications for the Static and Uniform mobility models over AODV-based and OLSR-based MANET could be considered over other network systems as well. The benchmarked values of the SIP performance metrics of Chapter 6 have been used to compare the performance of ROHC implementations over IPv6 MANET for both AODV and OLSR routing protocols as represented in Chapter 7. Furthermore, the proposed Cross-Layer approaches have been designed, evaluated and compared based on the results of the benchmarked values as represented in Chapter 8.

It can be concluded from the evaluation results in chapter 6 that the difference between IPv4 and IPv6 is not very significant in Static and Uniform scenarios, where the impact of the mobility issues over the QoS are not present. The loss mostly depends on the network congestion and erroneous links. Therefore, both IPv4 and IPv6 can be used for Static and Uniform MANET systems, however, considerable delays with higher values of SIP end-to-end performance metrics had been registered over both RWP and RWP-All scenarios. In addition, the evaluation efforts showed that IPv6 has long delays with bad performance for SIP-based VoIP over both AODV and OLSR MANET even when applying ROHC. The ROHC system was employed to reduce the large amount of the Internet Protocol header overhead when transmitting the IPv6-based data. The best level of enhancements that ROHC can provide for IPv6 SIP-based VoIP implementation was on the TCP/IP traffic. In general, the implementations of the ROHC system for IPv6 MANET are not able to improve the overall performance of SIP signaling for mobile-based systems to the required level of enhancement.

The proposed CLAODV and CLOLSR algorithms in Chapter 8 provide better SIP signaling performance, and flexible adoptions for routing parameters depending on the SIP application status. In addition, the proposed approaches are better than the regular basic SIP signalling system and other related solutions from the literature in terms of the efficiency, the implementations' flexibility, and the QoS criteria. The usages of the determined sets in the proposed Cross-Layer approaches are considered from the benchmarking efforts at Chapter 6. These sets were proposed with regard to the required level of enhancements, where sets with too short values could not allow the SIP processes to be generated to provide good services. Both the CLAODV and CLOLSR provide a dynamic reachability nature for the correspondent nodes to reduce the connectivity delays, save the CPU cycles, and reduce the bandwidth. In addition, these approaches are applicable for both IPv4 and IPv6 implementations over MANET. Over all the investigated proposed sets of the benchmarked values in Table 8-1 for both CLAODV and CLOLSR implementations, set D has the best enhancement level regarding the SIP processes. Therefore, as much as the set values for the CLAODV or CLOLSR approaches reduced the performance level of the SIP signaling system, bandwidth consumptions and CPU utilisation of the SIP server increased. In addition, the average of consumed bandwidth for IPv4 scenarios for the CLAODV and CLOLSR implementations is slightly lower compared with the
consumed bandwidth in IPv6 because of the packet overhead of IPv6 traffic that slightly increased the amount of the consumed bandwidth. In addition, the IPv6 implementations in this research study do not support the mobility features for the mobile nodes.

### SIP Signaling over MANET:

1) **Benchmarking** the SIP end-to-end Performance Metrics of RFC 6076
   - From the **Static** and **Uniform** models:
     - **Benchmarked values** for different network systems. The benchmarked values of the SIP signaling system are: RRD, SRD, and SDD.
     - **Benchmarked values** for the registration, initiation, and termination processes.

2) **Performance evaluation** of the SIP-based VoIP over MANET
   - From **RWP** and **RWP** mobility models’ scenarios:
     - **Evaluation** of the current state of SIP-based VoIP implementations over AODV/OLSR MANET. (SIP signaling Performance/VoIP QoS)
       - **Main Findings:**
         - Very long delays for the SIP processes, low routing performance, a small number of successful calls, and low performance for the SIP server.

3) **Employing the ROHC system** for the SIP-based VoIP over MANET
   - **Investigate** the performance enhancement level for the SIP signaling processes with the implementations of ROHC for the **RWP** and **RWP** mobility models over AODV/OLSR MANET.
     - **Main Findings:**
       - Limited level of enhancements with the implementations of ROHC over SIP/TCP traffic only.
       - Other implementations of ROHC (RTP/UDP and All traffic) have worst performance.
       - ROHC is not an appropriate choice for the implementations of the SIP signaling system over MANET.

4) **Proposed Cross-Layer Approaches** to enhance the SIP Signaling performance over MANET
   - **Investigate** the performance enhancement level of the proposed Cross-Layer Approaches for the Registration/Call Initiation/Termination processes.
   - **Examine** the CLAODV/CLOLSR Implementations and **compare** them with the current classical implementations of the SIP-based VoIP over AODV-based/OLSR-based MANET.
   - **Examine** the AODV/OLSR routing performance with CLAODV/CLOLSR.
     - **Main Findings:**
       - Good level of performance enhancement for the SIP signaling processes with shorter delays, enhanced routing performance, increase in the number of successful calls, good performance for the SIP server.
       - Good level of bandwidth consumption and CPU utilization related to the enhanced performance.

Figure 9-1: Summary of the investigations performed and contributions in the thesis
9.2 Future Works

This thesis has investigated the SIP signaling system for VoIP applications over MANET using AODV and OLSR routing protocols. The study considered different mobility models for the Cross-Layer performance enhancement methods. However, still there are some future directions to continue with the line of this research:

i. Study and improve the SIP registration and retransmission timers over MANET to enhance the SIP signaling and QoS for SIP-based applications over MANET reactive/proactive routing protocols. This would include the topology modelling for the SIP-base VoIP applications over MANET.

ii. Study and enhance the SIP signaling system over TCP for both IPv4 and IPv6 as the evaluation results of the SIP/TCP performance showed bad performance over the implemented random mobility models. Furthermore, the retransmission timers for SIP/TCP signaling over MANET are still an open research issue. The retransmission of the SIP/TCP messages causes the duplication, latency, and delays for the INVITE/Re-INVITE messages.

iii. Employ the benchmarked values of the RFC 6076 for SIP-based VoIP applications in other platforms such as WLAN and LTE. The benchmarked values could be used as performance references for the SIP-based VoIP applications over different types of networks and system implementations.

iv. Employ the Session Description Protocol (SDP) with the functionality of the Cross-Layer performance enhancement methods for the SIP signalling over MANET. The SDP could add extra enhancements for the SIP signalling system over MANET by offering the correspondent nodes and the SIP server the choice to agree on the performance level for the SIP processes.

v. Evaluate the SIP-based VoIP applications over MANET systems using a realistic trace-based mobility model. This will include the investigations for the proposed Cross-Layer approaches to check the improvement level for the SIP signaling system in reality scenarios.

vi. Improve the efficiency of the proposed algorithms of the Cross-Layer approaches for the SIP processes over different mobility models (such as the Manhattan mobility model) with implementation of a realistic speed detection function. This enhancement will increase the reliability level and the system efficiency of the Cross-Layer implementations that reflect on the bandwidth consumption and CPU cycles as well.

vii. Develop a disaster recovery scenario for SIP-based VoIP applications over MANET using the proposed Cross-Layer optimisation approaches.

viii. Investigate the security issues that could be raised by the implementations of the Cross-Layer approaches for the performance enhancements of the SIP signaling system. The delays and retransmission processes for the SIP/TCP messages within the CLAODV/CLOLSR implementations represent a real level of security threats in the considered secured SIP sessions over the B2BUA SIP server.
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Appendix A

SIP-based Internetwork System between Future IP Networks and ZigBee-based Wireless Network Systems

The SIP-based internetwork system is an application layer communication system which is extensively used for Future IP Network systems. On the other hand, the wireless ZigBee networks are a growing technology which provides many useful sensing and actuating features. Therefore, it needs to effectively interact with other types of network systems through the IP network systems. The idea of connecting the wireless ZigBee networks with IP networks using the SIP protocol is a recently raised issue. Many approaches with different aspects had been suggested. In general, the SIP-based internetworking systems between Future IP Networks and wireless ZigBee networks depend on two main approaches: the Proxy-based approach, and the ZigBee Stack-based Approach. However, the investigations are still modest and the SIP-based approaches need to be improved to support its efficiency for the SIP/TCP and the SIP/UDP applications.

This research study evaluated the available approaches to improve the Quality of Service (QoS) of the SIP-based internetworking system between the ZigBee-based Wireless Personal Area Networks (WPANs) and the Future IP Networks as mentioned in Chapter 5. The initial research efforts showed that the Proxy-based approach, which is used to relay the SIP application through a Presence Server, has a considerable level of end-to-end delays for the SIP/TCP applications. These delays increase with the increase in number of the ZigBee WPAN nodes. In addition, the implementations of the VoIP applications over ZigBee-based WSNs are difficult with the Proxy-based approach because of the centralized nature of the SIP signaling system. The early efforts of this research study proposed the Combined approach, which is an improved approach that depends on the Proxy-based and the ZigBee Stack-based approaches. In addition, the research study proposed an initial design for the TinySIP translator to be implemented for the ZigBee Stack-based approach and the Combined approach. The three approaches need to be evaluated and improved to enhance the QoS for the SIP/TCP and the SIP/UDP (VoIP) applications. Furthermore, the investigations of the SIP-based VoIP over ZigBee-based WSNs have implementation problems over the simulation tools. The OPNET© Modeler provides
the ZigBee models as a trademarked technology that only allows the implementations and the simulations of the ZigBee-based applications in the modeler. However, the accessibility to the ZigBee code is constrained because of the copyright issues for the ZigBee Alliance. These constraints affected the ability to implement the proposed approaches for the improvements of the Quality of Service (QoS) of the SIP-based internetworking system between the ZigBee-based Wireless Personal Area Networks (WPANs) and the Future IP Networks. The Riverbed (the current owners of OPNET application tools) recently has released a limited level of accessibility for the ZigBee communication codes for the licensed versions of OPNET Modeler.
Appendix B

The SIP System Representation in OPNET Modeler

The implementations of the SIP-based VoIP application over MANET had been considered over the OPNET 17.1 Modeler. In this Appendix, the used node models are represented in simple snapshots. Figure B-1 shows the node model representation of MANET. It includes the OSI representation for the MANET nodes and the SIP server. The SIP implementations are located in the application layer of the node model. Figure B-2 shows the SIP functions representation in the header blocks of the process model. The complete representation of the basic SIP function is in the function block for that initiate and implement the SIP signaling system.

Figure B-1: The Node model of the MANET nodes
Appendix B: The SIP System Representation in OPNET Modeler

Figure B-2: SIP functions representation in the header block of the process model in the application layer

Figure B-3: The representation of the SIP protocol in the function block