INFSCO-ICT-216203 DA VINCI

D5.4.1 v1.0

Evaluation criteria for multimedia services, issue 1

Abstract:
This document evaluates the possible gains of DaVinci codes from the user perspective, namely Quality of Experience objective measurements following the ITU-T G.107 specification. The evaluation methodology was prepared and applied during cell level simulation measurements. According to the IEEE 802.16EMD cell level simulation should be treated as intermediary step between link level and system level tests performed in other deliverables (D2.1.2 and D2.3.2). In addition admission control mechanisms were studied and four of them implemented to evaluate DaVinci codes in operation in various SNR conditions and realistic user mobility models in comparison to other codes specified for use with WiMAX standards (e.g. RS-CC specified in the IEEE 802.16e). Moreover QoE ontologies were analyzed and specification of simple QoE ontology for DaVinci was presented. In addition simple tests for TCP over WiMAX with DaVinci were prepared.

Keyword list: QoE, QoS, MOS, HARQ, ACM, 802.16

Disclaimer:
Executive Summary

The main goal of this document is to evaluate potential gains of DaVinci codes applied in WiMAX system. In order to provide highest fidelity of results this document introduces thorough analysis of alternatives for preparing suitable simulation platform. The challenge here is that such platform should on one hand be capable of providing physical layer abstraction whose characteristics resemble DaVinci error curves on the other it should be implementing MAC and above layers of ISO/OSI stack in order to assess user experience as well as cell level performance. The most suitable platform chosen is ns2 simulator enhanced with standard-compliant extensions for managing service flows in WiMAX – the developed ns2 extension is called VIMACCS. The resulting solution provides comprehensive cell level capabilities for evaluating admission control together with objective QoE measurements. It is shown that it can be interesting to couple event based simulator with DSP processor that improves fidelity of physical layer abstraction at the cost of computational complexity (increased simulation time). However having gathered experience with DSP in the loop of ns2 authors suggest that DSP platform can be instead more convenient as a QoE “result browser”. It is interesting alternative for field results of potential subjective tests of HW solutions with DaVinci on a chip. Due to the fact of the lack of unavailability of the hardware solutions with DaVinci physical layer such tests are not feasible in reality. Eventually the simulation platform is composed of ns2 with physical layer implementing lookup table concept.

The key contribution of this document is the evaluation of DaVinci codes from the perspective of admission control algorithms performance and QoE objective assessment. The results show that for realistic mobility patterns of simulated users, depending on the SNR distribution achieved using two different coverage maps, DaVinci codes with specific ACM can improve system capacity of the similar system using different FEC schemes. On the other hand DaVinci codes can positively influence number of users that experience satisfactory VoIP call quality in conditions similar to WiMAX system with alternative coding mechanism. It should be stressed however that results presented are not fully addressing the potential of DaVinci as the simulations do not implement HARQ mechanisms described in WP5 deliverables. It is expected that HARQ developed in the project time frame increase robustness of the system though.

Finally the document proposes formalization of specific QoE capabilities of the DaVinci framework in a form of ontology. The ultimate goal of the ontology is to be used within running DaVinci system for automated assessment of QoE with real user feedback. Although the limitations stemming from the fact that DaVinci on a chip is currently on its way cause this section to be rather theoretic enhancement and proposal towards comprehensive evaluation suite of DaVinci solution once field test are feasible. Authors have also included proposal of the TCP test suite for DaVinci, however due to the limitations of simulation environment these were not executed.
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</table>
# Table of Contents

1. **Introduction** ................................................................. 9

2. **Simulation environment - introduction and motivation** .......... 11
   2.1 Network Simulator 2 platform.............................................. 12
   2.1.1 DaVinci WiMAX module with Admission and Congestion Control Support - ViMCCS module for Network Simulator 2 ........................................... 13
       2.1.1.1 Dynamic Service Management Messages Flow (DSX) ......................... 14
       2.1.1.2 Connection Admission Control and Process of creating new Service Flows .... 15
       2.1.1.3 Process of changing parameters of active Service Flows – DSC-x messages flow 16
       2.1.1.4 Process of deleting existing Service Flows – DSD-x messages flow ............. 16
       2.1.1.5 Service Flows - Multiple Service Flows per user, MIB .............................. 17
   2.2 Matlab/Simulink platform .................................................. 18
   2.3 Matlab/Simulink and Digital Signal Processor (DSP) ................ 19
   2.4 Platforms integration .......................................................... 20
       2.4.1 Platforms connectivity .................................................. 21
       2.4.2 ns2 nodes’ communication enhancement ..................................... 22
   2.5 Deployed link level model and DaVinci codes deployment .......... 22
       2.5.1 nb-LDPC Code deployment ............................................ 23
       2.5.2 Deployed code validation .............................................. 25
   2.6 LUT-based approach to model physical layer .......................... 28
   2.7 Conclusions ................................................................. 28

3. **Methodology for QoS/QoE evaluation** ................................. 29
   3.1 State of the art ................................................................. 29
   3.2 QoS/QoE models used ....................................................... 30
       3.2.1 QoS in WiMAX ......................................................... 30
       3.2.2 QoE evaluation methods .............................................. 32
       3.2.3 ITU-T Recommendation G.107 E-model ................................. 34
       3.2.4 The E-model concept - calculation of the transmission rating factor R ........... 34
           3.2.4.1 Basic signal-to-noise ratio ................................. 35
           3.2.4.2 Simultaneous impairment factor ............................... 35
           3.2.4.3 Delay impairment factor ....................................... 35
           3.2.4.4 Equipment impairment factor ................................ 35
           3.2.4.4 Advantage factor .............................................. 36
       3.2.5 Default values for E-model ........................................... 36
       3.2.6 Quality measure derived from the transmission rating factor R .................... 36
   3.3 Assumptions for QoE evaluation in DaVinci .......................... 38
       3.3.1 VoIP over WiMAX ..................................................... 39
       3.3.2 The impact of HARQ on VoIP transmission ......................... 40
       3.3.3 Adaptive Coding and Modulation .................................... 41
   3.4 QoE test methodology ....................................................... 41
       3.4.1 Test description template ............................................. 42
           3.4.1.1 Objective of the test ........................................... 42
           3.4.1.2 Test network scenarios ........................................ 42
           3.4.1.3 Measured parameters .......................................... 42
           3.4.1.4 Measurement models .......................................... 43
           3.4.1.5 Traffic conditions ............................................. 43
           3.4.1.6 Expected test results ........................................... 43
   3.5 QoE simulation methodology - packet transmission level .......... 43
       3.5.1 Scenario one ......................................................... 45
           3.5.1.1 Objectives ......................................................... 45
           3.5.1.2 Test network scenario ......................................... 46
           3.5.1.3 Measured parameters .......................................... 47
           3.5.1.4 Measurement models .......................................... 48
           3.5.1.5 Traffic conditions ............................................. 48
           3.5.1.6 Expected results .............................................. 48
3.5.2 Scenario two ........................................................................................................... 48
  3.5.2.1 Objectives ........................................................................................................... 48
  3.5.2.2 Test network scenario ...................................................................................... 49
  3.5.2.3 Measured parameters ....................................................................................... 50
  3.5.2.4 Measurement models ....................................................................................... 51
  3.5.2.5 Traffic conditions ............................................................................................ 51
  3.5.2.6 Expected results .............................................................................................. 51
3.5.3 Scenario Three ......................................................................................................... 51
  3.5.3.1 Objectives .......................................................................................................... 51
  3.5.3.2 Test network scenario ...................................................................................... 51
  3.5.3.3 Measured parameters ....................................................................................... 52
  3.5.3.4 Measurement models ....................................................................................... 53
  3.5.3.5 Traffic conditions ............................................................................................ 53
  3.5.3.6 Expected results .............................................................................................. 53
3.5.4 Scenario Four .......................................................................................................... 53
  3.5.4.1 Objectives .......................................................................................................... 53
  3.5.4.2 Test network scenario ...................................................................................... 53
  3.5.4.3 Measured parameters ....................................................................................... 54
  3.5.4.4 Measurement models ....................................................................................... 55
  3.5.4.5 Traffic conditions ............................................................................................ 55
  3.5.4.6 Expected results .............................................................................................. 55
3.6 QoE trials and experiments .......................................................................................... 55
  3.6.1 Scenario one .......................................................................................................... 55
    3.6.1.1 Simulation result for Case 1A ......................................................................... 55
    3.6.1.2 Simulation results for Case 1B ........................................................................ 57
  3.6.2 Scenario Two .......................................................................................................... 59
    3.6.2.1 Simulation results for case 2A ....................................................................... 59
  3.6.3 Scenario three ........................................................................................................ 61
    3.6.3.1 Simulation results for case 3A ....................................................................... 61
  3.6.4 Scenario four ......................................................................................................... 63
    3.6.4.1 Simulation results for case 4A ....................................................................... 63
3.7 Conclusions for DaVinci ............................................................................................. 65

4. Evaluation of admission control algorithms for WiMAX ........................................ 66
  4.1 State of the art .......................................................................................................... 66
    4.1.1 Complete Sharing .............................................................................................. 66
    4.1.2 Guard Channel (GC) based CAC ..................................................................... 66
    4.1.3 Fair Connection Admission Control .................................................................. 66
    4.1.4 Cost–based Connection Admission Control ..................................................... 66
    4.1.5 Power Reservation Connection Admission Control ........................................... 66
  4.2 Admission control algorithms for DaVinci .............................................................. 67
    4.2.1 Complete Sharing Connection Admission Control Algorithm ......................... 67
    4.2.2 Dynamic Hierarchical (DHCAC) and Modified Dynamic Hierarchical (mDHCAC) 
        Connection Admission Control Algorithm .............................................................. 67
      4.2.2.1 Original DHCAC algorithm ....................................................................... 68
      4.2.2.2 Modification introduced to DHCAC algorithm ........................................ 68
    4.2.3 Fair Connection Admission Control Algorithm ............................................... 71
  4.3 Admission Control versus system capacity estimation - ACM and Symbol Reservation 
      Schemes ...................................................................................................................... 73
  4.4 Test scenarios ............................................................................................................ 74
    4.4.1 Scenario I – verification of proper operation of ViMACCS ................................. 75
    4.4.2 Scenario II – evaluation and comparison of CSCAC, original DHCAC, modified DHCAC 
        and Fair CAC algorithms ......................................................................................... 76
    4.4.3 Scenario III – CAC and Lookup Tables (ns2+LUT) .......................................... 78
    4.4.4 Scenario IV – CAC and ACM - Symbols Reservation Schemes (SRS) .............. 79
    4.4.5 Scenario V – CAC, ACM and Symbols Reservation Schemes for two FEC schemes .. 80
  4.5 Evaluation of results .................................................................................................. 81
    4.5.1 Scenario I – verification of proper operation of ViMACCS ................................. 81
    4.5.2 Scenario II – evaluation and comparison of CSCAC, original DHCAC, modified DHCAC 
        and Fair CAC algorithms ......................................................................................... 82
    4.5.3 Scenario III – CAC and Lookup Tables ............................................................. 85
5. TCP over WiMAX ................................................................. 93
   5.1 TCP over WiMAX state of art ............................................ 93
   5.2 Tools and techniques for TCP performance assessment ....... 93
   5.3 TCP evaluation requirements ........................................... 94
      5.3.1 Slow Start ............................................................... 94
      5.3.2 Congestion avoidance .............................................. 95
      5.3.3 Fast Retransmit ....................................................... 95
      5.3.4 Fast Recovery ......................................................... 95
   5.4 Test methodology ........................................................... 96
      5.4.1 Scenario One .......................................................... 97
         5.4.1.1 Objectives ......................................................... 97
         5.4.1.2 Test network scenario ....................................... 98
         5.4.1.3 Measured parameters ......................................... 98
         5.4.1.4 Measurement models ......................................... 98
         5.4.1.5 Traffic conditions ............................................... 98
         5.4.1.6 Expected results ................................................ 98
      5.4.2 Scenario Two ............................................................ 98
         5.4.2.1 Objectives ......................................................... 99
         5.4.2.2 Test network scenario ....................................... 99
         5.4.2.3 Measured parameters ......................................... 99
         5.4.2.4 Measurement models ......................................... 99
         5.4.2.5 Traffic conditions ............................................... 99
         5.4.2.6 Expected results ................................................ 99
      5.4.3 Scenario Three .......................................................... 99
         5.4.3.1 Objectives ......................................................... 100
         5.4.3.2 Test network scenario ....................................... 100
         5.4.3.3 Measured parameters ......................................... 100
         5.4.3.4 Measurement models ......................................... 100
         5.4.3.5 Traffic conditions ............................................... 100
         5.4.3.6 Expected results ................................................ 100
   5.5 Test results assessment .................................................... 101

6. QoE ontology ................................................................. 102
   6.1 Available ontologies desktop research .............................. 102
   6.2 Machine readable formalisms .......................................... 108
      6.2.1 OWL ................................................................. 109
      6.2.2 RDF ................................................................. 109
      6.2.3 SWRL .............................................................. 109
      6.2.4 Protégé .............................................................. 110
   6.3 QoE ontology for DaVinci .............................................. 110
   6.4 Applied ontology and E-model ........................................ 113
   6.5 Conclusions for DaVinci .............................................. 115

7. Conclusions and Prospects .............................................. 116

8. References ................................................................. 117
# List of Acronyms and Abbreviations

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Abbreviation</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACK</td>
<td>Acknowledgement</td>
</tr>
<tr>
<td>AMC</td>
<td>Adaptive Modulation and Coding</td>
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<tr>
<td>AQ</td>
<td>Assessed QoS</td>
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<td>AQA</td>
<td>Auditory Quality Assessment</td>
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<td>AQOS</td>
<td>Application Quality of Service</td>
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<td>ARQ</td>
<td>Automatic Repeat Request</td>
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<tr>
<td>BE</td>
<td>Best Effort</td>
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<tr>
<td>BP</td>
<td>Blocking Probability</td>
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<td>BS</td>
<td>Base Station</td>
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<tr>
<td>BTC</td>
<td>Block Turbo Codes</td>
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<tr>
<td>BU</td>
<td>Bandwidth Utilization</td>
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<tr>
<td>BWA</td>
<td>Broadband Wireless Access</td>
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<tr>
<td>CAC</td>
<td>Connection Admission Control</td>
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<tr>
<td>CBR</td>
<td>Constant Bit Rate</td>
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<td>CC</td>
<td>Chase Combining</td>
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<td>CCSRS</td>
<td>Congestion Control SRS</td>
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<td>CID</td>
<td>Connection ID</td>
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<tr>
<td>CRC</td>
<td>Cyclic Redundancy Check</td>
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<tr>
<td>CS</td>
<td>Complete Sharing</td>
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<tr>
<td>DHCAC</td>
<td>Dynamic Hierarchical Call Admission Control</td>
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<td>DP</td>
<td>Dropping Probability</td>
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<td>DSA</td>
<td>Dynamic Service Add</td>
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<td>DSC</td>
<td>Dynamic Service Change</td>
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<tr>
<td>DSD</td>
<td>Dynamic Service Delete</td>
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<tr>
<td>DSP</td>
<td>Digital Signal Processor</td>
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<tr>
<td>ertPS</td>
<td>Extended real time Polling Service</td>
</tr>
<tr>
<td>FCAC</td>
<td>Fair Connection Admission Control</td>
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<tr>
<td>FDD</td>
<td>Frequency Division Duplexing</td>
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<tr>
<td>FTP</td>
<td>File Transfer Protocol</td>
</tr>
<tr>
<td>GC</td>
<td>Guard Channel</td>
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<tr>
<td>H-ARQ</td>
<td>Hybrid Automatic Repeat Request</td>
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<tr>
<td>IQ</td>
<td>Intrinsic QoS</td>
</tr>
<tr>
<td>IQA</td>
<td>Instrumental Quality Assessment</td>
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<tr>
<td>IR</td>
<td>Incremental Redundancy</td>
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<tr>
<td>LDPC</td>
<td>Low Density Parity-check Codes</td>
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<td>LTE</td>
<td>Long Term Evolution</td>
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<tr>
<td>LUT</td>
<td>Look Up Tables</td>
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<td>MCS</td>
<td>Modulation and Coding Scheme</td>
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<td>mDHCAC</td>
<td>Modified DHCAC</td>
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<tr>
<td>MIB</td>
<td>Management Information Base</td>
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<tr>
<td>MNB</td>
<td>Measuring Normalized Blocks</td>
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<tr>
<td>MOS</td>
<td>Mean Opinion Score</td>
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<tr>
<td>nb-LDPC</td>
<td>Non-binary Low Density Parity-check Codes</td>
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<tr>
<td>NQOS</td>
<td>Network Quality of Service</td>
</tr>
<tr>
<td>nrtPS</td>
<td>Non-real time Polling Service</td>
</tr>
<tr>
<td>OFDM</td>
<td>Orthogonal Frequency Division Multiplexing</td>
</tr>
<tr>
<td>OSI</td>
<td>Open System Interconnections</td>
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<tr>
<td>PAMS</td>
<td>Perceptual Analysis Measurement System</td>
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<tr>
<td>Acronym</td>
<td>Full Form</td>
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<tr>
<td>PESQ</td>
<td>Perceptual Evaluation of Speech Quality</td>
</tr>
<tr>
<td>PQ</td>
<td>Perceived QoS</td>
</tr>
<tr>
<td>PSQA</td>
<td>Pseudo-Subjective Quality Assessment</td>
</tr>
<tr>
<td>PSQM</td>
<td>Perceptual Speech Quality Measure</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>QoE</td>
<td>Quality of Experience</td>
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<tr>
<td>QoS</td>
<td>Quality of Service</td>
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<tr>
<td>RFSRS</td>
<td>Reservation Factor SRS</td>
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<tr>
<td>rtPS</td>
<td>Real time Polling Service</td>
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<tr>
<td>SCN</td>
<td>Switched Circuit Network</td>
</tr>
<tr>
<td>SFID</td>
<td>Service Flow ID</td>
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<tr>
<td>SNR</td>
<td>Signal-to-Noise-ratio</td>
</tr>
<tr>
<td>SQ</td>
<td>Speech Quality</td>
</tr>
<tr>
<td>SRS</td>
<td>Symbol Reservation Scheme</td>
</tr>
<tr>
<td>SS</td>
<td>Subscriber Station</td>
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<tr>
<td>TCL</td>
<td>Tool Command Language</td>
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<tr>
<td>TCP</td>
<td>Transport Control Protocol</td>
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<tr>
<td>TDD</td>
<td>Time Division Duplexing</td>
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<tr>
<td>TLC</td>
<td>Target Language Compiler</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>UGS</td>
<td>Unsolicited Grant Service</td>
</tr>
<tr>
<td>VBR</td>
<td>Variable Bit Rate</td>
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<tr>
<td>VoIP</td>
<td>Voice over IP</td>
</tr>
<tr>
<td>WCSRS</td>
<td>“Worst-case-scenario” SRS</td>
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1. Introduction

The goal of IEEE 802.16 standardization efforts is to provide a comprehensive and accurate new generation broadband wireless standard specification. New generation of BWA networks are expected to be capable of handling hundreds of user connections over a wide area assuring high QoS for different user traffic. The requirement for IMT-Advanced systems are concerned by IEEE 802.16m specification, since by 2010 it is assumed that such system will provide throughputs (three times) higher than these of current IMT-2000 systems (30Mb/s) and depending on user mobility level, offer throughputs from 100Mb/s for mobile users moving at high speed up to 1Gb/s for stationary solutions. Ongoing efforts to improve current candidate-systems performance focus on areas like channel coding effectiveness, optimizing H-ARQ techniques, turbo channel estimation, resource allocation with special focus on Connection Admission Control techniques (CAC) and QoS provisioning. Most of IEEE 802.16 specifications do not either provide detailed description on QoS provisioning or deploy call admission control decision engines, leaving it up to the vendor’s decision. Accurate QoS and admission control management solutions could assure WiMAX competitiveness on the market, as LTE is considered a very strong candidate for systems working in IMT-Advanced bandwidth. Little effort has been made to investigate the current CAC algorithms solutions. It is worth pointing out that in order to assure an appropriate level of service, BS has to allocate an adequate amount of resources to a connection. Although the IEEE 802.16 standards do not state how to implement the scheduling or CAC algorithms they specify the set of mechanisms used in the physical and MAC layer to support QoS management. Whereas in physical layer of WiMAX new coding schemes including binary and non-binary turbo encoders are being investigated and evaluated as a potential contribution to IEEE 802.16m. The nb-LDPC codes have recently shown a potential in comparison to other codes but the work is being carried in the direction of comprehensive evaluation of such codes being subject to cross-layer, multi-mechanism synergy towards improved resource management. The efforts to propose cross-layer (opportunistic) solutions that in order to address the natural non-determinism of radio resources benefit from cooperation of MAC and physical layer are becoming increasingly visible.

The remaining part of the document is structured as follows. In Chapter2 the analysis of the simulation requirements of the T5.4 is made with respect to the capabilities of the available simulators. Optimal choices are presented (both integrated ns2/Matlab platform, DSP processor kit and standalone ns2) and eventually VIMACCS simulation module for DaVinci is described. Chapter 3 introduces the concept and the formalisms behind objective methods for assessing QoE of the potential DaVinci users. This chapter also presents methodology designed for measuring and evaluation of the QoE/QoS parameters. Results of measurement are described afterwards.

Admission control algorithms applied in the DaVinci context (using simulation platform proposed in chapter 2) are introduced and evaluated in Chapter4. Dedicated test scenarios were introduced herein in order to assess effectiveness of the algorithms (in terms of blocking/dropping probability, bandwidth utilization and system throughput). Eventually the results are presented and DaVinci gains evaluated in the final sections of this chapter.

Scenarios for evaluating DaVinci solutions with the TCP applications are presented in Chapter5. Mechanisms typical for TCP are first introduced however the results of the proposed scenarios are still pending due to required simulator modifications.

Finally Chapter 6 addresses aspects of describing the concepts of QoE/QoS within DaVinci enabled WiMAX networks. The literature study of available QoE/QoS ontologies is provided and some of the most suitable solutions are described in more details. The main idea behind this chapter is to provide methodology for systematic assessment of the real (measured with end users) QoE when using DaVinci. As the full fledged, deployable system with DaVinci codes is not available at the moment of writing this deliverable we only show how to benefit from having results of Chapter3 in order to presents subjective score of future DaVinci users. The latter can be interesting “visualization” tool for the project review meeting.

Eventually results gathered in this deliverable are summarized in the last section of the document.
2. Simulation environment - introduction and motivation

The performance enhancing techniques and mechanisms located in physical layer include among all OFDM technology, logical and physical channels, adaptive modulation and coding, intelligent antennas, MIMO technology, error detection and correction and power control.

The IEEE 802.16 MAC layer is far more complex than its counterpart in WLAN (IEEE 802.11). The key part in the MAC processing is realized by the Common Part Sub-layer (CPS), one of the three MAC sub-layers defined in the standard. The CPS enables provision of QoS parameters that in WiMAX depend on the variety of factors like residual bandwidth, fluctuating channel capacity due to changes in radio channel or terminal transmission capabilities that vary as a function of terminal location with respect to the serving base station (BS). In the IEEE specification the key resource management functions and admission control are located in the management plane (Fig. 1). The concept of guaranteeing QoS in 802.16 networks is based on a few mechanisms:

- service flow management
- dynamic service creation
- two step resource activation.

The IEEE 802.16 defines five classes of service in the CPS sub-layer. Classes are differentiated by means of QoS parameters. In order to deliver requested QoS to connections BS has to allocate enough resources, however it is out of scope of the standard to decide how such task is to be accomplished. The IEEE 802.16 specification does not define either admission control or scheduling mechanisms for WiMAX. That is why the research on the aforementioned algorithms by simulations is so important. The benefits from adding mechanisms for service scheduling and user admission can further improve QoS and have a direct impact on the quality as perceived by the end user – defined as QoE.

Moreover, in order to assess performance improvements found in 4G candidate-systems, particularly using solutions introduced by DaVinci, a comprehensive link level and cell level simulations are required. The need to evaluate multiple performance enhancing techniques like MIMO, novel channel coding schemes like non-binary LDPC codes, together with standards like IEEE 802.21 that aim at enabling handover and interoperability between heterogeneous network types make simulations even more important. The real challenge is to introduce the representative characteristics of nb-LDPC and related mechanisms. Simulation environment equipped with both MAC and physical layer modeled at the
satisfactory level of fidelity and standard compliance is needed. Integration proposals covering most representative approaches have been presented in Fig. 2 and, among all, include:

- Network Simulator 2 (ns2)
- Matlab
- Matlab + Digital Signal Processor (DSP)
- Lookup Tables (LUTs)

Therefore in this chapter we present our work that contributes to the creation of an integrated, open source, WiMAX simulation environment. Throughout study of proposed solutions was presented in number of papers submitted by the authors of this document to international conferences [bcom] [kkr][etq].

![Fig. 2 ISO/OSI layers' coverage by integrated platform](image)

### 2.1 Network Simulator 2 platform

Network Simulator (in this deliverable referred to as “ns2”) is a widely used open-source simulation tool that implements a variety of entities, such as wireless node (IEEE 802.11), wireless channel and protocols. Although by default ns2 does not implement model of the IEEE 802.16 networks, a few WiMAX extension modules for ns2 are available. Among all, the ones with the biggest potential are:

- NIST (802.16-2004) module [nis]
- NIST+ (802.16-2004) [nip]
- PMP (802.16-2004) module [pmp]
- LRC (802.16-2004) module [lrc]

Another WiMAX module for ns2 is called WINSE [winse], but is not open – source, and therefore not available on the web.

The first of the modules presented above, (NIST) has been developed by National Institute of Standards and Technology. It supports OFDM PHY with configurable modulation and delivers most comprehensive set of PHY layer blocks from described open – source modules. This module lacks error correction mechanisms. Fragmentation is implemented, but packing in not supported. Although data structures for service flows have been defined, NIST does not support service flow classes. Therefore there is no possibility to adjust parameters for QoS provisioning and all connection are treated by schedulers as BE connections.

WiMAX module presented in [nip] (in this paper referred to as “NIST+”) is a package expanding default functionality of aforementioned NIST module. It adds QoS support, management of the QoS requirements and introduces three service flow classes (UGS, rtPS, BE). It also implements scheduling mechanisms for UGS compliant with the IEEE 802.16e standard and implements three scheduling algorithms for rtPS and BE connections. Moreover it supports unicast and contention request mechanisms. NIST+ shares most of PHY parameters with NIST.
The PMP module supports only fixed WiMAX (IEEE 802.16d) and it does not provide means for QoS provisioning [pmp]. However it is worth mentioning, that authors of this module already took into consideration possibility of implementing admission control algorithms in future, as there are already void CAC functions definitions in code. Nevertheless this module has been discontinued and its future is unsure.

LRC module supports MAC QoS features [lrc]. Also as the only one of open-source ns2 WiMAX modules LRC implements all five service flow classes compliant with the IEEE 802.16e standard. Still, this WiMAX module lacks some MAC features e.g. it is impossible to change coding and modulation schemes.

WINSE described in [winse] is commercial WiMAX extension for ns2. It supports both OFDM and OFDMA, implements ARQ and HARQ mechanisms and supports fragmentation and packing. Moreover it supports all five service flow classes defined in 802.16e. It is possible to configure modulation type and coding schemes. Although this module is the most advanced ns2 WiMAX extension, it is being developed for Nokia and is not available on the web.

Table 1 compares described modules together with OPNET and ns2 extension developed by ITTI. Although other WiMAX modules exist, they either focus on simulating aspects of WiMAX which are out of scope of our research (such as [mesh] which focuses on MESH mode of WiMAX) or are not available for download (such as [chmsa]).

ViMACCS (DaVinci WiMAX module with Admission and Congestion Control Support) – has been developed by ITTI and is based on NIST+ as it implements three most important service classes with unicast and contention request modes. NIST and NIST+ originally lack DSD-x (dynamic service delete) and DSC-x (dynamic service change) messages, which are very important approaching evaluation of admission control algorithms. Both NIST and NIST+ implement a dynamic service addition (DSA-x) message, which is a good base for implementing other dynamic service messages, such as DSD-x and DSC-x. Additional advantage of this patch for ns2 is good documentation of the underlying NIST core part.

### Table 1 Comparison of ns2 WiMAX modules and OPNET

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>PPM</th>
<th>NIST</th>
<th>NIST+</th>
<th>LRC</th>
<th>WINSE</th>
<th>OPNET</th>
<th>ViMACCS</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSA-x</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>---</td>
<td>x (?)</td>
<td>x (?)</td>
<td>x</td>
</tr>
<tr>
<td>DSC-x</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>x (?)</td>
<td>x (?)</td>
<td>x</td>
</tr>
<tr>
<td>DSD-x</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>no data</td>
<td>no data</td>
<td>x</td>
</tr>
<tr>
<td>MCS</td>
<td>---</td>
<td>---</td>
<td>x</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>OFDM</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>OFDMA</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>---</td>
</tr>
<tr>
<td>fragmentation</td>
<td>?</td>
<td>x</td>
<td>x</td>
<td>?</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>packing</td>
<td>?</td>
<td>---</td>
<td>---</td>
<td>?</td>
<td>x</td>
<td>x</td>
<td>---</td>
</tr>
<tr>
<td>CAC</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>no data</td>
<td>---</td>
<td>x</td>
</tr>
<tr>
<td>SF classes</td>
<td>---</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>BW req mechanisms</td>
<td>---</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>BW grant mechanisms</td>
<td>---</td>
<td>---</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
<td>x</td>
</tr>
<tr>
<td>mode</td>
<td>TDD</td>
<td>TDD</td>
<td>TDD</td>
<td>TDD</td>
<td>TDD/FDD</td>
<td>TDD/FDD</td>
<td>TDD</td>
</tr>
<tr>
<td>topology</td>
<td>PMP</td>
<td>PMP</td>
<td>PMP</td>
<td>PMP</td>
<td>PMP</td>
<td>PMP</td>
<td>PMP</td>
</tr>
<tr>
<td>code</td>
<td>3.5</td>
<td>5</td>
<td>4.5</td>
<td>3</td>
<td>no data</td>
<td>no data</td>
<td>----</td>
</tr>
</tbody>
</table>

#### 2.1.1 DaVinci WiMAX module with Admission and Congestion Control Support - ViMACCS module for Network Simulator 2

One of D5.4.1 tasks is to – despite limited capabilities of WiMAX simulators- evaluate selected Admission Control algorithms. The most common performance indicators used to evaluate Admission Control algorithms are:

- Blocking Probabilities (BP’s)
- Dropping Probabilities (DP’s)
Bandwidth Utilization (BU’s).

for different classes of services. Therefore a new ns2 extension module called ViMACCS (DaVinci WiMAX module with Admission and Congestion Control Support) has been developed. The new module is based on WiMAX module presented in [nip] (referred to as “NIST+”) and has been designed for version 2.29 of ns2. The underlying NIST+ implements QoS mechanisms and various schedulers for different classes of services, but by default it does not implement any CAC algorithms.

Although NIST supports DSA-x message flows and NIST+ adds Service Flow parameters to DSA-x messages, there is no support for managing time intervals between arrivals of new Service Flows requests or to terminate existing connections. The later is basic requirement for admission control evaluation to be successful. Therefore both NIST and NIST+ are insufficient in order to evaluate CAC algorithms performance, as it is impossible to generate figures of Blocking or Dropping Probabilities. Therefore ViMACCS expands functionality of NIST and NIST+ in order to allow generation of these characteristics. Most important features of ViMACCS have been presented below:

- four Connection Admission Control algorithms implemented
- implemented DSC-x messages flow
- implemented DSD-x messages flow
- modified Initial Ranging procedures
- added support for multiple service flows per User
- implemented Poisson generator

The most important of aforementioned functionalities have been described in more detail in the following subsections of this chapter.

2.1.1.1 Dynamic Service Management Messages Flow (DSX)

Dynamic Service messages (DSX) allow creating, modifying and deleting Service Flows [d-std]. In order to support more complex management of Service Flows, authors implemented DSC-x and DSD-x

![ViMACCS DSX-messages flow](image-url)
messages flows and modified DSA-x message flow. The figure (Fig. 3) presents state diagram of DSX messages implemented in our module. This section describes some of the major features of ViMACCS, such as modified DSA-x, DSD-x and DSC-x messages flow and support for multiple service flows per user.

2.1.1.2 Connection Admission Control and Process of creating new Service Flows

Formally (according to IEEE specifications) admission control mechanism could be invoked at the level of ranging procedures (accept SS or not) or at level of creating new service flow. Used solution implements CAC algorithm triggered by the service flow addition message.

The figure (Fig. 4) shows process of handling a new connection request. When Subscriber Station (SS) wants to create new Service Flow it sends DSA-req (Dynamic Service Addition request). When BS receives this message Connection Admission Control algorithm is performed in order to determine, if BS has enough resources to meet new Service Flow QoS parameters. If CAC decides BS has enough resources, BS creates new connection and sends a DSA-rsp containing unique CID and SFID for this connection. If respective SS receives DSA-rsp with positive response, it adds new service flow's SFID and CID to its list of connections. If CAC algorithm decides BS does not have resources for new connection, it rejects incoming request and sends a DSA-rsp with negative response. Conversation ends with SS sending a DSA-ack message.

All implemented algorithms are triggered accordingly to the above process, even if they use different decision making criteria. As main CAC handling routine is always called at the same moment (receiving a DSA-req), changing CAC algorithm is just a matter of calling different CAC algorithms.
2.1.1.3 Process of changing parameters of active Service Flows – DSC-x messages flow

Fig. 5 shows process of handling Dynamic Service Change (DSC-x) messages. When SS decides to change QoS parameters of active service flow, it sends DSC-req message containing service flow’s SFID, CID and requested new QoS parameters. When BS receives this message it checks if it has sufficient resources to meet new QoS requirements. The algorithm is the same as admission control algorithm. If BS has enough resource, it admits new QoS parameter set, updates management information base (MIB) accordingly and creates DSC-rsp with positive response. If there are not enough resources, BS creates DSC-rsp message with negative response. Then BS sends DSC-rsp containing Service Flow’s admitted QoS parameters set. Upon receiving DSC-rsp, SS updates his admitted QoS parameter set. Conversation ends with SS sending a DSC-ack message.

2.1.1.4 Process of deleting existing Service Flows – DSD-x messages flow

Fig. 6 shows process of handling Dynamic Service Delete (DSD-x) messages. When SS decides to delete active service flow, it sends DSD-req message containing service flow’s SFID and CID. When BS receives this message it deletes given service flow and updates management information base (MIB) accordingly. From this moment resources previously reserved by this service flow can be reused by other...
connections. Then BS sends a DSD-rsp. Upon receiving DSD-rsp, subscriber station deletes given service flow from list of active service flow and associated connection.

### Fig. 6 DSD messages flow – deleting Service Flow

#### 2.1.1.5 Service Flows - Multiple Service Flows per user, MIB

A major feature of one of the implemented Admission Control algorithms – namely FCAC, which is described in chapter 4.2.3 - is the fairness offered. Fairness is achieved by calculating the amount of bandwidth already utilized by user’s ongoing Service Flows during admission process (see chapter 4.2.3). Therefore for this algorithm it is assumed that user can have multiple active connections (Service Flows) at the same time. NIST+ allows only one Service Flow per simulation node. In order to evaluate CAC algorithms, that take into consideration user’s ongoing service flows, NIST+CAC has been modified to support multiple Service Flows per user. This has been achieved by assigning multiple physical nodes to a single user and relating traffic to one user by appropriate identification of flows. Also an object which
contains information about node-user associations has been created. Each user is assigned unique ID which is used during simulation to identify which nodes belong to him. Assigning nodes and IDs to a user is done at the level of TCL (Tool Command Language) script (Fig. 7). Moreover BS schedulers have been modified in order to take into account users.

Different CAC algorithms take into consideration different operational variables. Therefore for each implemented type of CAC algorithm dedicated Management Information Base (MIB) has been created as part of each BS information structure. For all algorithms MIB stores information about remaining resources as well as bandwidth consumed by each class of service on aggregate and per-connection level.

2.2 Matlab/Simulink platform

One of the major drawbacks of ns2 WiMAX physical layer components is insufficient support for different channel models (AWGN, Rayleigh), limited availability of channel coding schemes (e.g. LDPC) and MIMO techniques. The Matlab/Simulink (Mathworks) by default offers much more solutions in this area. Default elements coming with the Simulink library meet most IEEE 802.16 PHY requirements. Simulink community provides a variety of custom blocks, so it is not unusual to find implementation of (binary/non-binary) LDPC codes and Turbo-Codes. Moreover Matlab offers more advanced and sophisticated tools for radio characteristics analysis (modulation constellation analysis, automatic BER assessment, built-in radio calibration models). Although Simulink is well suited for PHY simulation, it is not intended for simulating higher ISO/OSI layers. As the authors aimed the evaluation of the cell-level performance of WiMAX, a lot of programming effort would have to be put in order to model MAC and higher layers in Matlab/Simulink. On the contrary the availability of higher-layer protocols libraries, agent-nodes and the ability to define trajectory of mobile terminals places ns2 as a solution naturally complementary to Matlab (with Communication Toolbox). Developed Simulink model is presented in the figure (Fig. 8). The model was first calibrated with DaVinci error curves as presented by other Partners.

![Fig. 8 IEEE 802.16e PHY Layer Simulink model](image-url)

The main features of the developed Simulink model are gathered in Table 2. The model support DaVinci coding scheme as well as traditional RS-CC coding. Transmission can be configured for SISO, MISO or MIMO channel. Estimation is performed on preambles added in the Uplink and Downlink direction. The model supports OFDM with FFT equal to 256 and Cyclix Prefix size of 1/4, 1/8, 1/16 and...
1/32 as specified in [d-std],[e-std]. Pilots are added before OFDM symbol assembling in the Uplink and Downlink direction. The signal is transmitted over an AWGN channel.

Table 2 Specification of IEEE 802.16 PHY Layer Simulink model

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bandwidth</td>
<td>2.5 MHz</td>
</tr>
<tr>
<td>Coding</td>
<td>RS-CC, DaVinci codes</td>
</tr>
<tr>
<td>Rate</td>
<td>½, 2/3, ¾</td>
</tr>
<tr>
<td>Modulation</td>
<td>BPSK, QPSK, 16-QAM, 64-QAM</td>
</tr>
<tr>
<td>Channel</td>
<td>AWGN</td>
</tr>
<tr>
<td>$N_{used}$</td>
<td>192</td>
</tr>
<tr>
<td>$N_{pilot}$</td>
<td>8</td>
</tr>
<tr>
<td>$N_{FFT}$</td>
<td>256</td>
</tr>
<tr>
<td>Antenna configuration</td>
<td>MIMO (2x2), MISO (2x1), SISO (1x1)</td>
</tr>
<tr>
<td>Channel estimation</td>
<td>LS, MMSE</td>
</tr>
<tr>
<td>Cyclic prefix</td>
<td>1/1/1/1/1/4/8/16/32</td>
</tr>
</tbody>
</table>

A simplified version of the model depicted in Fig. 8 was used for calibration case conducted in WP2.

2.3 Matlab/Simulink and Digital Signal Processor (DSP)

With respect to physical/link layer the built-in Matlab functionality is essential when trying to assess/distinguish cell level performance according to requirements from [requ] by means of simulation. The Real-Time workshop (RTW) the Matlab extension package, enables automated code generation, compilation and consolidation for a target DSP board (e.g. TI DSK 6000) based on Simulink model. Thus basically DSP seems to be interesting denominator combining potential of the two virtues, namely ns2 and physical layer abstraction in Matlab.

Providing fixed-point hardware-oriented implementation of required WiMAX physical layer radio communication system blocks with DaVinci encoder/decoder this option enables interesting DSP deployment capabilities. Due to advanced profiling features it is possible to assess each block's computational complexity. When looking at the pros and cons of ns2, it is important to stress the fact that this environment is typically useful for cell-level evaluation, but mainly with respect to wired networks. The ns2 simulation capabilities of L1-L2 ISO/OSI layers with respect to DaVinci requirements have been presented in Table 3.

One of the biggest advantages of using Matlab is that it allows the moving forward from standard code writing to high level model-based project designing. Model-based design makes it easy to express a design concept, simulate the model to verify the algorithms, automatically generate the code to deploy it on a hardware target, and verify exactly the same operation on silicon. Moreover it is possible to implement the design on a DSP, and verify its on-target performance in real time, simply by adding dedicated block (Target Preferences Block) from target processor block set library. The advantages of the integrated platform proposed herein stem from the following facts:

- various high-level PHY models can be easily deployed on DSP and integrated with ns2 without a line of coding
- ease of modifying/deploying model updates (e.g. LDPC coder) brings the platform closer to the concept of rapid prototyping
- it is possible to use the DSP hardware-in-the-loop (HWIL) extension in real-time (emulation) or virtual time (simulation) mode
- in addition to the built-in Ethernet card, DSP board supports JTAG communication with host computer introducing real-time communication with the board
- single DSP can be used to model several nodes (multi-user) with slight (but generic) modifications to ns2
- measuring real DSP processing delays, BER can be introduced and analyzed if needed

With the model code optimized for target DSP (e.g. LDPC decoder) the integrated ns2/DSP platform can be used in real-time (emulation) mode; the DSP equipped with additional HW modules enables real RF communication.

**Table 3 Summary of simulators’ capabilities with respect to the DAVINCI requirements**

<table>
<thead>
<tr>
<th>Block</th>
<th>Matlab/Simulink</th>
<th>Network simulator</th>
<th>DaVinci requirements</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>R2008b</td>
<td>Patch 1 (NIST)</td>
<td>Patch 2 (PMP)</td>
</tr>
<tr>
<td>Modulation</td>
<td>*BPSK, *QPSK, *4-QAM</td>
<td>*BPSK</td>
<td>*QPSK</td>
</tr>
<tr>
<td></td>
<td>*16-QAM, *64-QAM</td>
<td></td>
<td>*16-QAM, *64-QAM</td>
</tr>
<tr>
<td>OFDM/OFDMA</td>
<td>yes=yes</td>
<td>yes/no</td>
<td>yes/no</td>
</tr>
<tr>
<td>Coding</td>
<td>* Convolutional Coding</td>
<td>no data</td>
<td>no data</td>
</tr>
<tr>
<td></td>
<td>* Turbo Coding</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>* Convolutional</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Turbo Coding</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>* LDPC</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Channel</td>
<td>* AWGN</td>
<td>* free space (FS)</td>
<td>* free space (FS)</td>
</tr>
<tr>
<td>(Propagation model)</td>
<td>* Rayleigh (no fading, flat-fading, selective fading)</td>
<td>model</td>
<td>model</td>
</tr>
<tr>
<td></td>
<td>* Doppler spread</td>
<td>* shadowing model</td>
<td>* shadowing model</td>
</tr>
<tr>
<td>MIMO</td>
<td>2x1, 2x2</td>
<td>No</td>
<td>no</td>
</tr>
<tr>
<td>Coding rates</td>
<td>1/2, 2/3, 3/4, 5/6</td>
<td>1/2, 3/4</td>
<td>1/2, 3/4</td>
</tr>
</tbody>
</table>

The key disadvantage of the solution using DSP in the loop of ns2 simulator is that by default (e.g. without applying optimizations to the Simulink model) the DSP processing time of the deployed model is not suitable for real-time emulation model (only virtual-time simulation mode applies).

### 2.4 Platforms integration

Functional complementarities of aforementioned simulation environments (ns2, Matlab/DSP) in the area of IEEE 802.16 standard compliance create an opportunity to merge their virtues by the means of an integrated platform. The idea of proposed integration solution has been presented in figures (Fig. 2, Fig. 9). The key enabler of this solution is a physical Ethernet link between the MAC layer implemented in ns2 and WiMAX physical layer blocks designed in Matlab and deployed on a target DSP board.
2.4.1 Platforms connectivity

With the use of Matlab extension package namely the Instrument Control Toolbox (ICT) it is possible to connect Matlab model with an external (e.g. ns2) server (see Host1 in the figure Fig. 10) in a simple way by means of TCP/IP communication.

The implementation using two optional network communication interfaces IA1 and IA2 (IEEE 802.3), that connect ns2 and PHY model, has been presented in the figure (Fig. 10). Duality of the interfaces indicates that it is possible to deploy/run model on both – Matlab/Simulink environment and target DSP board (Texas Instruments DSK C6455) [texdsk] either using relay PC (Host2) or directly communicating via Ethernet connection (IA2). The DSK C6455 kit is by default equipped with two communication interfaces: a) a 10/100Mb/s Ethernet card and b) JTAG (IEEE 1149.1) USB emulator for real-time DSP CCS communication. During first phase of development, platform incorporated an older version of TI DSP board, namely DSK c6416. DSK c6416 did not have a built-in Ethernet interface, thus Host2 had to act as a proxy during communication between ns2 and DSP target board. However both floating-point computations typical for running Matlab models and non-Real-Time operating system (WindowsXP) that introduces non-deterministic allocation of Host2 processor's resources, resulted in poor performance and low stability of the physical layer model. Moreover sending data packets to/from WiMAX MAC layer (ns2) from/to DSP module utilized RTDX (Real Time Data eXchange), an extension that enables real – time tracing and debugging of a running program. It allows not only reading but also sending data to a program running on DSP processor. This approach benefits from small DSP load footprint during host – target board communication. An interesting feature of the TI C6000 DSP Toolbox is that it provides a set of blocks dedicated (optimized) to perform computationally expensive calculations using processor-specific instructions set. The RF DaVinci model functions that would benefit from this capability are among all FFT/IFFT, various matrix operations etc.
2.4.2 ns2 nodes' communication enhancement

The ns2 introduces capabilities to support communication with external nodes. To the best authors knowledge two methods exist to support external communication:

- running the ns2 in emulation mode
- modifying current MAC – PHY API to handle external communication with HW module

When in emulation mode, the simulated nodes can – by means of special objects called “tap-agents network objects” – exchange packets with host computer (Host1) that is connected to a physical network [nsman]. The other approach requires some additional programming effort, as it is necessary to modify simulator's source code. As a result it gives a better control over ns2 task scheduler and additional degrees of freedom when choosing the right ns2 WiMAX module (patch). Changes made to generic ns2 wireless node were summarized in table shown in Table 4.

<table>
<thead>
<tr>
<th>Network node component (ns2)</th>
<th>Changes in code</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wireless Phy</td>
<td>Some functionality for packet processing was turned off. Uplink and Downlink transmission was only allowed.</td>
<td>Power management turned off (packets would be bounced back if the terminal is turned off).</td>
</tr>
<tr>
<td>Channel</td>
<td>The module was replaced by a Matlab/Simulink model communicating through a TCP/IP block</td>
<td>The channel class is responsible for estimating the propagation time. That is why it was excluded from the test and replaced with a Matlab/Simulink model. The model doesn't add any additional delay.</td>
</tr>
<tr>
<td>Radio Propagation Model</td>
<td>This function was turned off</td>
<td>No need for propagation model. Propagation time estimation and noise adding provided by Matlab block</td>
</tr>
</tbody>
</table>

Changes presented in Table 4 include - among all - replacing ns2 channel by Matlab/Simulink model communicating through a TCP/IP block. Also default ns2 propagation model was switched off as propagation time estimation and noise addition is done by Matlab.

2.5 Deployed link level model and DaVinci codes deployment

The link layer model used for preliminary tests of the combined ns2/DSP platform includes AWGN channel with simple transmitter and receiver implementing the nb-LDPC codes. The sequence of steps leading to fully operational simulation platform included the following actions:

- take encoder/decoder source C file
- adapt it to Simulink (wrapping)
- verify and validate the model (against “pure C” results of the original, C-language nb-LDPC implementation)
- attach Ethernet Receiver/Sender to the Matlab/Simulink model
- deploy the resulting model on DSP
- verify and validate the effect of deployment
- make basic DSP measurements of the physical
- layer abstraction (BER, profiling, delay)
- connect DSP to ns2 via Ethernet cable
• perform ns2 cell level, end-to-end, measurements (virtual time simulation).

It essential to underline that the above process of deploying the target physical layer blocks on the DSP is done with the following assumptions:

• the physical layer blocks use single threaded implementation
• no performance optimization of encoder/decoder code is performed

To ensure proper behavior of the integrated simulator a test suite was prepared and successfully performed.

2.5.1 nb-LDPC Code deployment

In order to proceed with DaVinci cell level tests and QoS/QoE tests, according to the concept of combined emulation-simulation case, it was essential to build a PHY model that is easily deployable (partially or whole as an application) on different platforms (namely the DSP and PC). Such an approach allows maximum benefit to be drawn from particular platform advantages (e.g. DSP is appropriate for computing FFT, while PC will be faster for sequential code). Therefore to meet this goal the RTW MATLAB’s toolkit was adopted (see appendix A for further details about RTW). The Real-Time Workshop (RTW) toolbox is an extension of capabilities of the Simulink and MATLAB products that allows automatic source code and packages generation to create real-time application on variety of systems. However adopting different source codes (C/C++, m-files, mex-files) to work with Simulink (and DSP) requires different approaches. DaVinci sources of the developed nb-LDPC codes were available as:

• C-ANSI program simulating the non-binary LDPC
  The Monte Carlo simulation of nb-LPDC codes. The encoder, decoder, and channel code is executed several times in loops to gather the BER vs. Eb/No characteristics.

• Matlab mex-functions wrappers for C-ANSI program
  The LDPC code ported to mex-functions is logically divided into three blocks: encoder, decoder, and channel (the code is modular), which allows adaptation of this code for different Simulink simulation (Fig. 11)

![Fig. 11 nb-LDPC adapted to Simulink](image)

Matlab mex and m functions are ideal to work with Simulink just by calling those functions in simulation’s block (this is explained in
Table 5). Table below presents the concept of mapping available nb-LDPC implementation towards instantiation closer to DSP deployment.
Table 5. M-function and its corresponding block in Simulink

<table>
<thead>
<tr>
<th>m-function/mex-functions</th>
<th>Simulink block</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>function [m, std] = stats(vals) ;</code></td>
<td><img src="image" alt="Simulink block" /></td>
</tr>
</tbody>
</table>

- The m-functions are set of Matlab commands executed sequentially to provide particular output.
- The mex-files are compiled as ANSI-C functions and can be called both from Simulink (inside a block) and from Matlab m-function.

However the m-function and mex-functions are not directly deployable to DSP or to PC platforms (as a binary file), thus it required special deployment strategy to be applied. Detailed description of deploying nb-LDPC as both mex-functions and the C-based standalone version has been presented in Appendix B.

\[2.5.2\] Deployed code validation

This section presents description of functional tests performed on physical layer model designed for DAVINCI project. The tests verify that the module behaves as specified and provided interfaces perform as expected. To ensure a successful implementation (and deployment) of physical layer abstraction model for DaVinci project the following tests were designed and performed:

- **Model deployment validation on PC/DSP** - This test validates TLC scripts (TLC is macro-based language used for mapping model’s C code onto target DSP binary) by checking content of files created (by TLC) as an output of DSP deployment process. For different initial conditions, generated output project is checked against compilation errors and common variables/constants proper initialization.

- **Iterative simulation on PC/DSP** - This test validates the short term model stability. For different signal inputs test verifies if model returns output signal during 10 subsequent repetitions of a simulation.

- **Model output validation for different input signals (DSP/PC)** - This test validates model response for different signal inputs. The output is checked with output collected from the legacy Simulink simulation.

Physical layer model validation tests were prepared to assure that fully operational model can be successfully deployed both on PC and DSP giving the same results. Moreover separate test suite was introduced to validate outputs of each radio model blocks separately. This group of tests aims at verifying that each of the modules:

- nb-LDPC encoder output validation
- AWGN output validation.

can be deployed separately without compilation errors. Additionally the BER, CWER, and FER vs. EbNo characteristics (for model shown in) gathered using ANSI-C application (Fig. 12) were compared with the same characteristics obtained for DSP - ns2 platform (Fig. 13). Obtained results show that BER / CWER /
FER characteristics for and DSP – ns2 platform are compliant with those gathered for ANSI-C application.

Fig. 12 Results for nb-LDPC codes for original code

Fig. 13 Results for nb-LDPC codes deployed on DSP board

The Table 6 shows average computation times of encoder and channel blocks. For PC platform these times are constant and equal to 0.34ms and 11ms respectively. The average computation times of encoder and channel block for DSP platform are also constant and equal to 41ms and 5ms respectively.

Table 6 . Average computation times of encoder and channel

<table>
<thead>
<tr>
<th></th>
<th>PC [ms]</th>
<th>DSP [ms]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel</td>
<td>11</td>
<td>41</td>
</tr>
<tr>
<td>Encoder</td>
<td>0.34</td>
<td>5</td>
</tr>
</tbody>
</table>
During validation process the computation time of nb-LDPC decoder deployed on PC (dual-core 2GHz, 2 GB RAM - Fig. 14) was also compared with the same code deployed on DSP (Fig. 15). During these tests Eb/No was changed from 1 to 5.

![Fig. 14 Computation time of code deployed on PC vs. Eb/No](image1)

![Fig. 15 Computation time of code deployed on DSP vs. Eb/No](image2)

Performance test results presented in Fig. 14 and Fig. 15 show that due to lack of code optimization of code deployed on DSP platform and due to the strong sequential nature of nb-LDPC decoder it was impossible to beat the PC dual-core (2GHz clock) using the DSP single-core (1GHz clock).
2.6 LUT-based approach to model physical layer

Although approach integrating ns2 and 802.16 PHY model deployed as Simulink model is interesting from the perspective of functional capabilities of the resulting model, it can also be computationally demanding for simulations with large number of users, as Simulink channel is shared by all SS and BS nodes. That is why physical layer abstraction based on the idea of Look Up Tables (LUT) for simulations for various FEC codes (nb-LDPC, LDPC, BTC) was also used. The idea of LUT is presented in Fig. 16. The process of using LUT is divided into two phases. In phase I channel is simulated in MATLAB and its characteristics (EbNo vs. BER/WER/CWER) are saved into a file. In phase II ns2 uses such a file as a lookup table (LUT) to mimic MATLAB channel. The figure (Fig. 16) presents also an approach to QoE tests, which have been presented more elaborately in section 3.

![Fig. 16 LUT-based approach to model the physical layer](image)

2.7 Conclusions

In this chapter the integration of ns2 and Simulink platform was presented and described in details. Also a new ns2 WiMAX extension called ViMACCS, that implements functionality required to evaluate Connection Admission Control algorithms, was presented. Particularly the different platforms connectivity and code deployment problems were stressed to provide complete methodology, which allows creating, run and evaluate CAC and QoE simulation scenario for WiMAX. Furthermore it was shown that the concept of ns2 and Simulink integration platform in terms of cell level WiMAX simulation allows to analyze whole simulation in details, thus the resulting scenarios does not have to pay a cost of a trade-off between fidelity of MAC and physical layer. However (according to the performed tests) such approach can be computational expensive, therefore for CAC and QoE simulations we use generalizations of the physical layer by means of the lookup tables. Nevertheless DSP can be also used for presentation purposes and platform for QoE assessment (see section 3). Finally, topics analyzed in this chapter have also been studied in papers submitted by the authors of this chapter to international conferences [bcom][kkr][etq].
3. Methodology for QoS/QoE evaluation

The main goal of this chapter is to provide the relevant background information on QoS/QoE evaluation methodology and show potential benefit from using DaVinci codes at the user (thus subjective) level. In particular QoS and QoE aspects of VoIP applications are considered. Further on the authors propose several test cases which could show potential QoE gain of DaVinci over several traditional coding schemes (Reed Solomon Convolutional Codes, Low Density Parity-check Codes, and Block Turbo Codes). The performance of ACM and CAC with DaVinci codes is evaluated.

3.1 State of the art

The concept of QoS has evolved during the past decade. More and more network resource consuming applications emerge and by the time IPv6 protocol has been fully deployed QoS capable systems will play an important role in future IP-based wireless broadband networks. QoS is the quality a service provider has to maintain to provide services with specific demands to the end user as specified in the agreement. A general QoS model can be further divided into three notions [evol]:

- Intrinsic QoS (IQ) – technical notion, evaluates measured and expected characteristics expressed by network parameters such as delay and loss,
- Perceived QoS (PQ) - reflects user satisfaction while using a particular service. It is a subjective measure and the only method to capture it is to involve human subjects in the measurement process,
- Assessed QoS (AQ) - extends the notion of PQ and refers to secondary aspects of user satisfaction such as service price, availability, usability and mobility.

A general QoS model is depicted in Figure 1 presents PQ as a function of IQ and an element of AQ [evol]. However the notions are usually considered in isolation and neither IETF nor ITU-T describes AQ to a full extent.

![Fig. 17 PQ,IQ and AQ relation](evol)

According to ITU-T we can express QoS as the ‘degree of objective service performance’ and QoE the ‘overall acceptability of an application or service, as perceived subjectively by the end user’ [evol]. According to these definitions some general assumptions can be made. According to [evol] QoS is described by IQ and QoE is described by AQ. Nevertheless, QoE is generally associated with Speech Quality (SQ) which is an element of AQ.

While QoS evaluation is only a matter of measuring physical parameters on the other hand QoE measurement are much more complicated as they involve human subjects in the measurement process. Despite this, QoE is considered the ultimate measure and in order to include it in systems the interdependence between QoS and QoE has to be addressed. One should observe that QoE is a function of QoS (Fig. 17). QoS is described through physical parameters and a relation has to be established to map these parameters to user experience with a service. QoE measurement process has been captured in ITU-T Recommendation [p.800]. The leading QoE evaluating method for voice is the Mean Opinion Score.
Current researches focus on the evaluation of quality of service as experienced by end users. Different approaches are proposed and a variety of solutions are investigated on how to evaluate VoIP quality over a wireless link – especially WiMAX networks. Some articles focus on the subjective measurement approach as a method for evaluating quality of experience [rosa],[dery] and some try to correlate the subjective measurements with objective approach [webster]. Objective approach measurements usually use PESQ or PSQA as proposed in [pesq01], [psqa01], and [psqa02]. Due to the constrains present in both methods other objective measurement approaches are proposed. The e-model approach was described in several publications [cole],[carv],[walk] as a method for evaluating QoE over a wireless link using VoIP applications. Variations of the e-model implementation [medd] as well as new approaches [kim][raja] are investigated to evaluate QoE under QoS-aware mobility mechanisms [bern]. Authors focus on QoE solutions designed for wireless environments especially WiMAX systems [evol][sham].

3.2 QoS/QoE models used

QoS capable systems play vital role in future broadband wireless networks. Although the IEEE 802.16 family of standards [d-std][e-std] do not address the methods for provisioning enhanced QoS for scheduling and admission control algorithms it defines the specific service classes and performance parameters that have to be considered when evaluating service quality efficiency. Section 6.3.5 of the IEEE 802.16-2004 standard describes the available scheduling services and the service flows related to the specific MAC scheduler [d-std]. Furthermore section 11.13.4 of IEEE 802.16-2004 standard gives a brief description of available QoS parameter set types [d-std],[e-std].

IEEE 802.16m EMD document states how to evaluate QoS performance over a wireless WiMAX link. Based on [80216emd] and IR2.1.2 [ir212] authors have investigated the problem of Quality of Experience measurement and the QoS metrics which have impact on voice transmission were selected. Latter on a simplified model for QoE assessment is described and evaluated. The next section of this document describes the different QoS WiMAX specific classes of service for broadband wireless IP networks. Furthermore a description of the QoE measurement methods is given.

3.2.1 QoS in WiMAX

The IEEE 802.16-2004 defines four service classes extended with a fifth as defined in IEEE 802.16e-2005. These are: UGS (Unsolicited Grant Service), rtPS (Real-time Polling Service), nrtPS (Non-real-time Polling Service), BE (Best Effort) and ertPS (Extended Real-Time Polling Service). Descriptions and related QoS service flow parameters for the five service classes are given in Table 7.

<table>
<thead>
<tr>
<th>Service class</th>
<th>Description</th>
<th>QoS service flow parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td>UGS</td>
<td>Designed to support real-time data streams consisting of fixed-size data packets issued at periodic intervals, such as T1/E1 and Voice over IP without silence suppression</td>
<td>Maximum Sustained Traffic Rate, Maximum Latency, Tolerated Jitter, Request/Transmission Policy</td>
</tr>
<tr>
<td>rtPS</td>
<td>Designed to support real-time data streams consisting of variable-sized data packets that are issued at periodic intervals, such as moving picture experts group (MPEG) video</td>
<td>Minimum Reserved Traffic Rate, Maximum Sustained Traffic Rate, Maximum Latency, Request/Transmission Policy</td>
</tr>
<tr>
<td>nrtPS</td>
<td>Designed to support delay-tolerant data streams consisting of variable-sized data packets for which a minimum data rate is required, such as FTP</td>
<td>Minimum Reserved Traffic Rate, Maximum Sustained Traffic Rate, Traffic Priority, Request/Transmission Policy</td>
</tr>
<tr>
<td>BE</td>
<td>Designed to support data streams for which no minimum service level is required and therefore may be handled on a space-available basis</td>
<td>Maximum Sustained Traffic Rate, Traffic Priority, Request/Transmission Policy</td>
</tr>
<tr>
<td>ertPS</td>
<td>Designed to support real-time service flows that generate variable size data packets on a periodic basis, such as Voice over IP services with silence suppression</td>
<td>Maximum Sustained Traffic Rate, Minimum Reserved Traffic Rate, Maximum Latency, Request/Transmission Policy</td>
</tr>
</tbody>
</table>
The related QoS parameters for every service class are described in Section 11.13.4 of the IEEE 802.16-2004 standard [d-std]. As stated in this section a service class is defined by specific QoS parameters like: Minimum Reserved Traffic Rate, Maximum Sustained Traffic Rate, Maximum Latency, Tolerated Jitter, Traffic Priority, and Request/transmission Policy. In accordance to IEEE 802.16-2004 the description of each parameter is given further:

a) **Traffic priority** Specifies the priority assigned to a particular service flow. A higher priority number means that a service flow must be handled in a different way as service flows with the same QoS but lower priority. This includes a lower delay and higher buffering preferences for service flows with greater priority. For otherwise non-identical service flows, the priority parameter should not take precedence over any conflicting service flow QoS parameter. The specific algorithm for enforcing this parameter is not mandatory here.

b) **Maximum Sustained Traffic Rate** Expressed in bits per second and defined as the peak information rate of the service. The algorithm for policing this parameter if left to vendor differentiation and is outside the scope of the standard.

c) **Minimum Reserved Traffic Rate** This parameter specifies the minimum rate reserved for this service flow. The rate is expressed in bits per second and specifies the minimum amount of data to be transported on behalf of the service flow when averaged over time. If this parameter is omitted, then it defaults to a value of 0 bits per second (i.e. no bandwidth is reserved for the flow).

d) **Request/transmission policy** Used to specify certain attributes for associated service flow, including PDU formation and, for uplink service flows, restrictions on the types of bandwidth request option that may be used.

e) **Tolerated Jitter** Defines the maximum delay variation (jitter) for the connection.

f) **Maximum Latency** Specifies the latency between the reception of the packet by the BS or SS on its network interface and the forwarding of the packet to its RF Interface.

The above mentioned parameters are relevant when dealing with WiMAX schedulers and differentiating between concurrent traffic flows. Other parameters like Maximum Traffic Burst and Minimum Tolerable Traffic Rate are also defined in [d-std] but the impact on QoS performance and scheduling schemes are usually evaluated through parameters described from a) to f).

However to capture the systems specific QoS parameters, which determine the user experience during a VoIP session, the measurable transmission parameters - delay, jitter and packet loss, have to be taken into account. The QoS requirements and related physical parameters that are considered through this chapter are limited only to IP networks. Transmission in IP networks is characterized by several parameters such as delay, jitter, packet loss, error rate, etc. The ITU-T organization defined the range of values for each parameter. Two recommendation issued by ITU-T - Y.1541 and G.1010, describe the requirements for transmission in IP networks. QoS service flow parameter values are defined in [y.1541],[g.1010],[fund],[kim]. The values specified in Table 8 and Table 9 summarizes the requirements for IP services and the allowed range for each parameter.

### Table 8 Traffic parameters for broadband wireless access [fund]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Interactive Gaming</th>
<th>Voice</th>
<th>Streaming Data</th>
<th>Data</th>
<th>Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Data Rate</td>
<td>50 Kbps-85 Kbps</td>
<td>4 Kbps-64 Kbps</td>
<td>5 Kbps-384 Kbps</td>
<td>0.01 Mbps – 1 Mbps</td>
<td>&gt; 1 Mbps</td>
</tr>
<tr>
<td>Example application</td>
<td>Interactive gaming</td>
<td>VoIP</td>
<td>Music, speech, video clips</td>
<td>Web browsing, e-mail, instant messaging (IM), telnet, file downloads</td>
<td>IPTV, movie download, peer-to-peer video sharing</td>
</tr>
<tr>
<td>Traffic flow</td>
<td>Real time</td>
<td>Real-time continuous</td>
<td>Continuous, bursty</td>
<td>Non-real time, bursty</td>
<td>Continuous</td>
</tr>
<tr>
<td>Packet loss</td>
<td>Zero</td>
<td>&lt; 1%</td>
<td>&lt; 1% for audio</td>
<td>Zero</td>
<td>&lt; 1e-8</td>
</tr>
<tr>
<td>Delay variation</td>
<td>Not applicable</td>
<td>&lt; 20 ms</td>
<td>&lt; 2 sec</td>
<td>Not applicable</td>
<td>&lt; 2 sec</td>
</tr>
<tr>
<td>Delay</td>
<td>&lt; 50ms -150ms</td>
<td>&lt; 100ms</td>
<td>&lt; 250ms</td>
<td>Flexible</td>
<td>&lt; 100ms</td>
</tr>
</tbody>
</table>
Table 9 Allowed QoS parameters and range in IP networks [kim]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>QoS parameter and range</th>
<th>Note</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Delay (D)</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>70 ms ~ 100ms</td>
<td>Y.1541</td>
</tr>
<tr>
<td></td>
<td>100 ms ~ 150ms</td>
<td>G.1010</td>
</tr>
<tr>
<td></td>
<td>150 ms ~ 200ms</td>
<td></td>
</tr>
<tr>
<td><strong>Jitter (J)</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>30 ms ~ 50ms</td>
<td>Y.1541</td>
</tr>
<tr>
<td></td>
<td>50 ms ~ 60ms</td>
<td></td>
</tr>
<tr>
<td></td>
<td>60 ms ~ 70ms</td>
<td></td>
</tr>
<tr>
<td><strong>Packet loss rate (L)</strong></td>
<td>10^{-2}  ~ 10^{-3}</td>
<td>Y.1541</td>
</tr>
<tr>
<td></td>
<td>10^{-3}  ~ 10^{-4}</td>
<td>DSL - Forum</td>
</tr>
<tr>
<td></td>
<td>10^{-4}  ~ 10^{-5}</td>
<td></td>
</tr>
<tr>
<td><strong>Packet error rate (E)</strong></td>
<td>~ 10^{-6}</td>
<td>Y.1541</td>
</tr>
<tr>
<td></td>
<td>10^{-6}  ~ 10^{-5}</td>
<td>DSL – Forum</td>
</tr>
<tr>
<td></td>
<td>10^{-5}  ~ 10^{-4}</td>
<td></td>
</tr>
<tr>
<td><strong>Bandwidth (B)</strong></td>
<td>Voice 256 Kbps ~ 128 Kbps</td>
<td>G.1010</td>
</tr>
<tr>
<td></td>
<td>128 Kbps ~ 80 Kbps</td>
<td>DSL-Forum</td>
</tr>
<tr>
<td></td>
<td>80 Kbps ~ 64 Kbps</td>
<td>WiMAX</td>
</tr>
<tr>
<td></td>
<td>Video 3 ~ 2 Mbps</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2 ~ 1 Mbps</td>
<td></td>
</tr>
<tr>
<td></td>
<td>1 Mbps ~ 512 Kbps</td>
<td></td>
</tr>
</tbody>
</table>

3.2.2 QoE evaluation methods

Call quality assessment, or more specific – VoIP user QoE estimation, can be done on a subjective (Auditory Quality Assessment) or objective (Instrumental Quality Assessment) basis. Subjective measurements assume a subject listening to a sample audio file and rating the perceived quality. The leading subjective measurement methodology for voice quality is the mean opinion score (MOS) defined by ITU-T organization in Recommendation P.800 [p.800]. This simple rating method defines a five point scale where 1 means very low quality and 5 – very high quality as perceived by the test subjects (Table 10). To evaluate MOS, a large number of human subjects have to listen to audio (the specific samples are defined by ITU-T [p.800]) and rate the call quality. However performing the above mentioned measurement can be difficult and expensive [mexp]. The psychological factors also have to be taken into account. The voice quality is not perceived in the same way by different people thus it is mandatory to perform the test on large amount of test subjects to average the results. Fortunately ITU’s Recommendation P.800 [p.800] gives advices how perform such test using human subjects.

Table 10 MOS values and their corresponding level of satisfaction

<table>
<thead>
<tr>
<th>Quality</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Best</td>
<td>5</td>
</tr>
<tr>
<td>High</td>
<td>4</td>
</tr>
<tr>
<td>Medium</td>
<td>3</td>
</tr>
<tr>
<td>Low</td>
<td>2</td>
</tr>
<tr>
<td>Poor</td>
<td>1</td>
</tr>
</tbody>
</table>

In commercial networks it is however necessary for Service Providers to estimate user satisfaction in real-time (e.g., during a VoIP call) and relate those measurements to specific values of delay, jitter and packet loss. Such results can be obtained using objective (Instrumental Quality Assessment) measurements. The objective approach can be divided into two groups of measurement methods:

- Signal-based methods (also referred to as black-box),
- Parameter-based methods (also referred to as glass-box).
The principle of objective measurements is to correlate physical and measurable magnitudes with quality as perceived by a human subject. Such relations exist in theory, but so far, it is not easy to establish them in practice, even for very simple applications. The two above mentioned methods for assessing voice perception quality give a considerable precision and consistency. Usually the results obtained by those methods are mapped to a rating scale. Various standards which are based on the concept of objective measurements have been developed during the past few years for voice quality estimation:

- PSQM (ITU P.861) / PSQM+ Perceptual Speech Quality Measure [p.861],
- MNB (ITU P.861) Measuring Normalized Blocks [p.861],
- PESQ (ITU P.862) Perceptual Evaluation of Speech Quality [p.862],
- PAMS (British Telecom) Perceptual Analysis Measurement System,
- The E-model (ITU G.107) [g.107].

The PSQM, PSQM+, MNB, and PESQ are signal-based measurement methods and part of a succession of algorithm modifications starting in ITU standards P.861. The PAMS (developed by British Telecom) and ITU-T's PSQM algorithms are very similar. The signal-based measurement algorithms require a reference signal to be feed into the assessed network and then compare the reference signal with the signal received at the other end of the connection. Several traditional voice measurement tools available at the market have implemented PESQ and PAMS measurement [ptool], [pmea]. The signal-based method approach has limitations which are relevant in evaluating QoE in terms of QoS. These methods give only estimates of voice quality without looking into the networks impairments which have impact on the signal. These measurements are a good solution in test labs to analyze the quality of individual devices. For example PSQM can be used to evaluate the quality of a telephone handset. However, the black-box methodology is not suited to assess call quality in data networks, where multiple users are talking at the same time, since signal-based methods do not “know” anything about data networking.

To mitigate the imperfections of the signal-based method and to relate the measurement methods with QoS parameters the parameter-based approach was proposed. One example of objective parameter-based measurement method is the E-model. The E-model, described in Recommendation G.107, was proposed by ITU-T to relate QoS parameters to quality of experience when assessing voice quality. The output of the E-model calculation is a single factor, called an ‘R-value’, derived from delays, impairment factors and voice transmission imperfections present in the network during transmission. The model described in G.107 is established to assess the QoE in telephone wire networks, thus some modifications are needed to be done to prepare the model for voice transmission assessment over wireless IP networks (e.g. WLAN, WiMAX, LTE) [cole],[medd],[walk]. The model gives reliable results for PSTN networks and is used for transmission planning and evaluation of system-level performance. Researchers have focused on the use of E-model for VoIP call quality assessment, and have shown that the results produced by the model are consistent and reliable, especially in outdoor WiMAX testbed [cole], [carv], and [walk]. Additionally the obtained R-factor can be mapped onto an estimated MOS as described in ITU-T Recommendation [g.107]. E-model is simple and easy to implement and was selected for assessing QoE in wireless environment with DaVinci codes.

![Fig. 18 Methods for assessing Quality of Experience for voice application](image)

The figure above (Fig. 18) summarizes the outcomes of this section. The next section provides theoretical background for E-model and provides guidance on QoE evaluation planning in IEEE 802.16 standard compliant networks.
3.2.3 ITU-T Recommendation G.107 E-model

A computational model for use in transmission planning, known as E-model, was defined by the standardization group ITU-T. The E-model has been approved as a transmission planning tool, for assessing the combined effects of variations in several transmission parameters that affect conversational quality. The definitions necessary to understand the ITU-T standards are gathered in ITU-T Recommendation G.100 [g.100]. Furthermore the ITU-T Recommendation G.107 [g.107] describes the requirements for QoE evaluation tests of voice transmission. ITU-T Recommendation G.108 [g.108] gives directives on how to prepare the e-model for different transmission environments. Authors have investigated the use of the E-model in a wireless environment based on concluded and ongoing researches and implemented the solution using Matlab. Furthermore the Matlab E-model was used to assess call quality using simulation platform defined in Chapter 2. The E-model implementation in BASIC is described in ITU-T Rec. G.107.

3.2.4 The E-model concept - calculation of the transmission rating factor R

The QoE evaluation is described in the E-model in form of a transmission rating factor R. Figure (Fig. 19) presents the reference model with different impairments which are present during a call over PSTN network. The E-model algorithm is evaluated according to those impairments.

![Fig. 19 Reference connection of the E-model [g.107]](image)

During voice transmission several imperfections of the transmission line may affect the transmitted signal as well as the caller surrounding (e.g., echo, background noise). The figure (Fig. 19) depicts the transmission impairments in a 4-wire loop telephone network and additional background noise sources present on both sides of the call. The definitions for each of the described impairment are given in ITU-T Recommendation G.100 [g.100]. Based on those transmission impairments a voice quality rating value called the R-factor is calculated. The R rating factor which describes the performance for voice transmission is composed of [g.107]:

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

Where:
- \( R_0 \) represents the basic signal-to-noise ratio, including noise source such as circuit noise and room noise,
• $I_s$ - is the combination of all impairments which occur more or less simultaneously with the voice signal,
• $I_d$ - represents the impairments caused by delay,
• $I_e$ - represents impairments caused by low bit rate codes,
• A – represents the Advantage Factor and allows for compensation of impairment factors when there are other advantages of access to the user.

The mentioned factors can be further subdivided into specific impairment values. We start with the description of each impairment factor and then show how to implement an E-model for VoIP quality assessment. More information on calculating the impairment factors can be found in ITU-T Rec. G.107 [g.107].

### 3.2.4.1 Basic signal-to-noise ratio.

The basic signal-to-noise ratio is defined by the following equation [g.107]:

$$R_0 = 15 - 1.5(SLR + N_q)$$

Where the equations for $SLR$ and $N_q$ are defined in [g.107].

### 3.2.4.2 Simultaneous impairment factor

The $I_s$ parameter is the sum of all impairments which may occur more or less simultaneously with the voice transmission. Three further parameters are specified to describe the $I_s$ impairment as follows [g.107]:

$$I_s = I_{olr} + I_{st} + I_q$$

Where the different impairments factors are:
• $I_{olr}$ - represents the decrease in quality caused by too-low values of OLR,
• $I_{st}$ - represents the impairment caused by non-optimum side-tone,
• $I_q$ - represents impairment caused by quantizing distortion.

### 3.2.4.3 Delay impairment factor

The delay impairment factor represents all impairments due to delay of voice signals are further subdivided into the three factors $I_{dte}$, $I_{dle}$, $I_{dd}$, $I_{dle}$, $I_{dd}$ [g.107]:

$$I_d = I_{dte} + I_{dle} + I_{dd}$$

Where:
• The factor $I_{dte}$ gives an estimate for the impairments due to Talker Echo,
• The factor $I_{dle}$ represents impairments due to Listener Echo,
• The factor $I_{dd}$ represents the impairment caused by too-long absolute delay $I_a$, which occurs when with perfect echo canceling.

### 3.2.4.4 Equipment impairment factor
The Equipment Impairment Factor \( I_e \) describes impairments introduced by low bit rate codecs \([g.107]\). They are depending on the subjective mean opinion score test results as well as on network experience. Appendix I/G.113 contains the recommended values of \( I_e \).

### 3.2.4.4 Advantage factor

The Advantage factor has a special meaning thus there is no relation to all other transmission parameters. The use of the A factor depends on the transmission environment and application and its use is up to the planner’s decision. The Advantage Factor describes the user tolerance of a given service. For example a user may tolerate longer delays in exchange for mobility or lower traffic pricing. ITU-T defines some of the provisional values for parameter A (Table 11), but it is not advisable to use the A factor during network planning as no reasonable studies were conducted towards the use of this parameter in VoIP networks [cole].

**Table 11 Provisional examples for the advantage factor A**

<table>
<thead>
<tr>
<th>Communication system example</th>
<th>Maximum value of A</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conventional (wirebound)</td>
<td>0</td>
</tr>
<tr>
<td>Mobility by cellular networks in building</td>
<td>5</td>
</tr>
<tr>
<td>Mobility in a geographical area or moving in a vehicle</td>
<td>10</td>
</tr>
<tr>
<td>Access to hard-to-reach locations, e.g. via multi-hop satellite connections</td>
<td>20</td>
</tr>
</tbody>
</table>

The A parameter values stated above are taken from ITU-T Recommendation G.113 and are only provisional (but when planning should be considered as upper limits).

### 3.2.5 Default values for E-model

To verify the proper set up of the model the Rating factor should be equal to 93.2 when default impairment values are used (please refer to Appendix I to find more about the impairment factors and their default values). The Matlab QoE calculation implementation was verified according to this assumption and the results are depicted in the next section.

### 3.2.6 Quality measure derived from the transmission rating factor R

The final output of the E-model is a scalar value R. The Rating factor R ranges from 0 to 100 and describes the measured voice quality. Values of the R factor below 60% are considered unacceptable. However the E-model provides objective measurements ratings which can be mapped onto subjective measurements ratings by calculating the Mean Opinion Score. The MOS scale can be obtained from the R factor and is represented in the scale from 1 to 5. Figure (Fig. 20) presents the algorithm implemented in Matlab according to ITU-T Recommendation G.107 \([g.107]\).
As a result of the depicted algorithm (Fig. 20) a relation between R factor and MOS value can be established. Figure below (Fig. 21) presents the MOS curve as a function of R factor.

![Fig. 20 MOS calculation algorithm](image)

![Fig. 21 MOS in function of the R factor](image)

Based on the picture above the MOS values are categorized as in Table 12.
Table 12 Quality of voice rating in terms of R-factor and MOS [kim]

<table>
<thead>
<tr>
<th>R-factor</th>
<th>Quality of voice rating</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>90&lt;R&lt;100</td>
<td>Best</td>
<td>4.34 - 4.5</td>
</tr>
<tr>
<td>80&lt;R&lt;90</td>
<td>High</td>
<td>4.03 - 4.34</td>
</tr>
<tr>
<td>70&lt;R&lt;80</td>
<td>Medium</td>
<td>3.60 - 4.03</td>
</tr>
<tr>
<td>60&lt;R&lt;70</td>
<td>Low</td>
<td>3.10 - 3.60</td>
</tr>
<tr>
<td>50&lt;R&lt;60</td>
<td>Poor</td>
<td>2.58 - 3.10</td>
</tr>
</tbody>
</table>

MOS is considered as the ultimate quality rating scale but is limited to subjective measurement. To map subjective and objective measurements R-factor calculation derived from the E-model should be applied. Given a value of R-factor one can estimate how the user perception (his MOS) of the service would be.

### 3.3 Assumptions for QoE evaluation in DaVinci

The E-model was designed to support PSTN planners and telephone service providers to evaluate the system requirements for telephone line. This can be considered a disadvantage when dealing with VoIP but several publications have shown a consistent and reliable approach towards the adoption of the E-model in an IP wireless environment to assess VoIP call quality. ITU-T defined in Recommendation G.108 [g.108] the end-to-end user requirements for voice transmission in IP networks that have to be adjusted to prepare the E-model for IP packet-switched networks. However we are using a more simplified model by adjusting some modifications to equations proposed in [g.107].

The R-factor in telephone networks is expressed as the sum of four terms:

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

Where \( R_0 \) is the signal-to-noise ratio and can be calculated using the default values, \( I_s \) is the signal-to-noise impairments associated with typical SCN paths, \( I_s \) is calculated from the default values, \( I_d \) is the impairment associated with the mouth-to-ear delay of the path and \( I_e \) is an equipment impairment factor associated with the losses within the audio codecs and \( A \) is the Advantage Factor, also called the Expectation Factor. Applying the default values to the R-value equation we get:

\[
R = 93.35 - I_d - I_e + A
\]

The model can be now described as a function of delay and packet loss contributions isolated in \( I_d \) and \( I_e \), respectively. On the other hand the Advantage Factor covers those quantities that are difficult (or impossible) to quantify. This term describes the customer tolerance of lower quality in exchange for, e.g., the convenience afforded by mobility, or in exchange for a lower price for the service provided by the ISP. It is difficult to estimate the Advantage Factor, but the G.107 states that it is appropriate to set the value to 10 for cellular networks. However, no researches have been conducted towards the use of A factor in VoIP thus, in our case, the A factor can be dropped [cole].

The delay components in the impairment factor \( I_d \) include: the average, absolute one-way mouth-to-ear delay \( T_a \), the average, one-way delay from the receive side to the point in the end-to-end path where a signal coupling occurs as a source of echo \( T_e \), and the average one trip delay in the four-wire loop \( T_r \) as described in Table 6. ITU-T G.107 gives a fully analytical expression for the impairment factor \( I_d \) in terms of \( T_a \), \( T_e \), and \( T_r \). Since WiMAX is an IP packet-switched network some simple generalizations may be made [cole]. We can take:

\[
d = T_a = T = \frac{T_r}{2}
\]

Because it is assumed that in IP networks \( T_a \) is half the time of \( T_r \) and no signal coupling occurs we can make some simplification and adopt the results in the E-model for wireless VoIP transmission. Parameters other than \( T_a \), \( T_e \), and \( T_r \) in the equation for \( I_d \) are set to their default values. The relation between the delay impairment factor and mouth-to-ear delay time \( d \) as well as the relation between \( R \) and mouth-to-ear delay time \( d \) is presented in the figures below (Fig. 22).
Values for the $I_e$ factor are calculated using the results for packet loss obtained during simulation. The equation for the $I_e$ is given below:

$$I_e = PL + (95 - PL) \cdot \left( \frac{pPL}{pPl \cdot BurstR} + bPL \right)$$

Where $PL$ is the packet loss, $pPL$ is the packet loss probability, $bPL$ is the packet loss robustness factor and $BurstR$ is the Burst Ratio.

The final output of the E-model is a scalar value $R$ [g.107]. Using the algorithm proposed in [g.107], and described in Section 3.2.6, the $R$ factor can be mapped to MOS scale.

### 3.3.1 VoIP over WiMAX

Voice over Internet Protocol (VoIP) provides an alternative to the telephone service offered by traditional Public Switched Network (PSTN) by using an IP network to carry digitized voice. However, packet switched networks differ from circuit switched networks and are more sensitive to delay and packet loss [evol]. To satisfy the requirements for VoIP application over a wireless link, researches have performed numerous studies to investigate the performance of IP telephony with different performance enhancing algorithms. Some techniques implemented to increase the performance gain over wireless links during VoIP conversation are presented in Table 13. Technologies like H-ARQ and their impact on delay and packet loss are also in the scope of this document.
VoIP applications use compression/decompression (CODEC) techniques for VoIP. The input audio stream is transformed into a digital bit stream. The speech samples are then compressed to produce a bit stream of 8-12 kbps that can be carried over the IP network. VoIP over wireless networks is affected by the choice of codec, packet loss, delay and jitter. The fluctuations in the channel typically cause packet loss and increase latency. In order to keep the mouth-to-ear round trip latencies at a reasonable level of 250-300ms, the delay budget for transmission over the air interface is 50-80 ms. H-ARQ is used to keep the delays within acceptable limits [eval][ahson].

### 3.3.2 The impact of HARQ on VoIP transmission

As stated in Section 3.3.1 H-ARQ can increase the reliability of the service using retransmission and timeouts to ensure service delivery within acceptable time limits. Hybrid Automatic Repeat Query is the successor of ARQ protocol used in cellular networks. Some of the basic ARQ implementations include:

- Stop-and-wait ARQ [sawARQ],
- Go-Back-N ARQ [gbNARQ].

### Table 14 VoIP traffic parameters [abu]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value (1 connection per SS)</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRTR</td>
<td>25 kbps</td>
</tr>
<tr>
<td>Average Traffic Rate</td>
<td>44 kbps</td>
</tr>
<tr>
<td>MSTR</td>
<td>64 kbps</td>
</tr>
<tr>
<td>Maximum latency</td>
<td>100 ms</td>
</tr>
<tr>
<td>Tolerated Packet Loss</td>
<td>10 %</td>
</tr>
<tr>
<td>Talk Spurt Length</td>
<td>Exponential random, ( \mu=147 ) ms</td>
</tr>
<tr>
<td>Silent Period Length</td>
<td>Exponential random, ( \mu=167 ) ms</td>
</tr>
<tr>
<td>Packet Size</td>
<td>23 bytes</td>
</tr>
</tbody>
</table>
• Selective Repeat ARQ [srARQ].

These protocols reside in the Data Link or Transport Layers of the OSI model. To speed up the retransmission of frames received with error, ARQ was transferred from the MAC layer to the PHY layer [d-std], [e-std]. The main difference between H-ARQ and ARQ is that in ARQ, error detection bits (CRC) are considered to decide for retransmission [d-std]. In HARQ, in addition to error detection bits, error correction bits are also added. This increases the robustness when the signal experiences bad channel. The two fundamental forms of Hybrid ARQ are chase combining (Type I - CC) and incremental redundancy (Type II - IR).

Both Type I and Type II HARQ protocols are investigated by DaVinci to use with new coding scheme [pfle]. It has to be taken into account that the time for H-ARQ retransmission has to be kept within acceptable limits [evol]. Especially for time sensitive applications like VoIP. Despite this H-ARQ is a very suitable solution to use with connectionless protocols like UDP. The retransmissions at the PHY Layer can significantly increase the performance of the network [sham],[caleb]. Although H-ARQ works fine with UDP some research has been conducted to evaluate network performance with H-ARQ and TCP [cela]. Some results considering the use of H-ARQ with VoIP are presented in [tero]. The H-ARQ mechanism was not implemented in the VIMACCS patch. H-ARQ mechanism for DaVinci was described in detail in [d531].

3.3.3 Adaptive Coding and Modulation

Adaptive Modulation and Coding is a technique introduced to wireless networks to increase the system capacity by constantly adjusting the coding rate and modulation to changing channel conditions. It is critical for time-varying channels because the Signal-to-noise ratio can change rapidly depending on the user location. ACM find they way to improve QoS of the transmission and improve performance [gong]. The results from evaluating capacity of some conventional modulation schemes in AWGN were presented in [tuan],[gong],[alas]. Researches in [tuan] also present an overview of the performance of different modulation schemes as well as results for the efficiency of transmission for each modulation. The performance evaluation in Rayleigh fading channels was presented in [tuan] and [gold] where the coding gain under fading was simulated. Moreover it is shown that ACM techniques give reasonable results and achieve better spectral efficiency than systems without ACM [alas] and moreover give more reasonable results when used with ARQ techniques [wang3]. Additionally researches toward the test of ACM on LDPC performance in DVB-S2 were carried out and the results can be found in [smol]. Although the technique is very promising some of the basic issues for using it with broadband wireless communication system are underlined in [sampei]. ACM mechanism for DaVinci was described in detail in [d511].

3.4 QoE test methodology

QoE evaluation is based on QoS measurements which are obtained during simulations using the platform described in Chapter 2. The system-level simulation approach was also proposed in IR 2.1.2 [ir212] and D.2.1.4 [d124]. However some modifications had to be made to make the approach more suitable for ns-2 VIMACCS module. The authors adjusted the simulation for a cell-based simulation environment as it is important to learn the characteristics of a single-cell performance simulation before proceeding to system level simulation. IEEE 802.16m EMD [80216emd] evaluation model states that the extension of the link-level analysis methods to a system-level analysis may start with adding multiple users in a single-cell setting. Furthermore the simulation components are addressed within IEEE 802.16m EMD [80216emd] and depicted in figure (Fig. 23). Authors put special focus on the Simulation component, where the following operations had to be made:

- Gathering results from a Link-level Matlab WiMAX PHY Layer Simulator,
- Generating lookup table for use with ns-2 (Physical Layer Abstraction).
According to the figure above a simulation environment should be build upon different components to improve the handling of data sets/simulation. The Link-level performance curves are captured in the Modeling Requirements block. In particular this block holds the information about the propagation environment, codes which are being used and, if applied, the traffic model. The performance curves together with the performance metrics gathered in the Evaluation Criteria component build the input for the Simulation component. The Simulation component is the core element for our simulations and ns-2 VIMACCS model is located at that layer.

3.4.1 Test description template

Based on [80216emd] authors have proposed a template for cell-level simulation case. The use of a template in the simulation methodology is advisable for assuring a defined and structured approach for simulation test cases which then can be easily used by third parties to produce simulation results. The recommended template for the simulation methodology is grouped in the following 6 points:

1. Objective of the test
2. Test network scenarios
3. Measured parameters
4. Measurement tools / models
5. Traffic conditions
6. Expected results

Each of the mentioned points will be given a brief description below.

3.4.1.1 Objective of the test

The objective of test should be clearly specified and well motivated.

3.4.1.2 Test network scenarios

Thus section refers to the details of test environment containing information about location of terminals, base station configuration and applications used. In particular, for the network we should specify topology and configuration of network devices (parameters corresponding to scheduler type, Base Station configuration, channel bandwidth, FEC coding scheme, etc.)

3.4.1.3 Measured parameters

This point should include list of connected characteristics. In some cases, if a given parameter is not well known – its definition should be recalled. The following QoS metric are measured:

- **Performance metrics for Delay Sensitive Applications:**
  - Packet Delay: Packets that are dropped or erased may or may not be included in the analysis of packet delays depending on the traffic model specifications. For
example, in modelling traffic from delay sensitive applications, packets may be dropped if packet transmissions are not completed within a specified delay bound. The impact of such dropped packets can be captured in the packet loss rate,

- **User Average Packet Delay (minimum and maximum packet delay per user),**
- **The averaged packet delay per cell** is defined as the ratio of the accumulated delay for all packets for all UEs received by the sector and the total number of packets. The delay for an individual packet is defined as the time between when the packet enters the queue at transmitter and the time when the packet is received successively by the UE. If a packet is not successfully delivered by the end of a run, its ending time is the end of the run.

**Packet losses:** Can be represented by the One-way Packet Loss Metric [RF2C680]. This metric is measured per packet and is set to 0 when the packet sent by the source node reaches the destination node within a reasonable period of time. Otherwise, is set to 1. Exact and complete methodology can be found in [RF2C680].

**Throughput Performance Metrics:**
- User and cell (average and cell edge) throughputs,
- Average cell throughput [kbps/cell] is used to study the network throughput performance, and is measured as:

\[ R = \frac{b}{kT} \]

where \( b \) is the total number of correctly received data bits in all data UEs in the simulated system over the whole simulated time, \( k \) is the number of cells in the simulation and \( T \) is the simulated time. In the case of only evaluating the center cell site, \( k \) is the number of sectors.

**Other metrics**
- Number of active users per cell,
- CAC Blocking probability: blocking probability due either to signal quality or resource (applicable ONLY if CAC is enabled).

### 3.4.1.4 Measurement models

This point should specify the list of models required for performing QoE tests.

### 3.4.1.5 Traffic conditions

This section specifies traffic conditions assumed for the test case. In particular, we should specify the generated traffic by type of traffic profile, rate, source-destination relation, etc. In general, we evaluate the system capacity under one and two different traffic types.

### 3.4.1.6 Expected test results

This point presents the expected test results including short justification.

The test methodology described above will be applied for test scenarios in Cell Level Simulation.

### 3.5 QoE simulation methodology - packet transmission level

The objective of the packet transmission level simulation is to evaluate quality of experience performance using service classes defined in IEEE 802.16 family of standards. In particular the UGS traffic with CBR (Constant Bit Rate) shall be investigated as the service class used to carry VoIP traffic across a wireless, IEEE 802.16 compatible link. The simulations are performed using the ns-2 VIMACCS patch. The PHY Layer abstraction was received from previous simulations (using MATLAB and SIMULINK) and integrated with ns-2 Lookup Tables as specified in Chapter 2. The test scenario objectives are evaluated according to the six point template structure described in section 3.4. We assume four test network configuration scenarios to evaluate QoE with ACM, CAC and DaVinci codes. Due to ns-2 limitations a bandwidth of 3.5 MHz and 256 point FFT are assumed. Taking into account the above expectations, we propose to perform the following groups of trials:
• Voice users only (only UGS traffic admitted to the network),
• Average voice users in cell (60% of users using VoIP application within a cell),

Each trial has to be performed under following conditions:
• Tests should be performed for one user as well as for a maximum of $N$ users ($N$ depends on MCS),
• Simulations under maximum bandwidth utilization have to be conducted to evaluate the maximum cell capacity,
• Each simulation should be performed for various encoders (BTC, WiMAX, LDPC, RS-CC, DaVinci codes) to see the changes in performance,
• Simulations have to be conducted for standalone CAC and ACM as well as for the full system profile (ACM + CAC),
• Simulations are performed in the Uplink direction,
• The UL scheduler performance should be evaluated,
• A receiver node, belonging to a group of sink stations, is presented in Fig. 24.

![Fig. 24 Uplink transmission with sink nodes](image)

In most cases only VoIP CBR traffic will be considered as the scope of this document is to evaluate performance of voice application. A VoIP traffic generator is used to feed the simulator according to the traffic requirements described in IR 2.1.2. A detailed description of application traffic modeling is presented in IR.2.1.2 and Table 15 summarized the traffic requirements for each application including VoIP traffic.

<table>
<thead>
<tr>
<th>Service category</th>
<th>QoS classes mapping</th>
<th>Loss tolerance</th>
<th>Real-time</th>
<th>Bandwidth requirements</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>UGS</td>
<td>Yes</td>
<td>Yes</td>
<td>Low-Medium: Skype: 3-16 kbit/s; 8 kbit/s (G.729) – 64 kbit/s (G.711)</td>
</tr>
<tr>
<td>Video conference</td>
<td>UGS</td>
<td>Yes</td>
<td>Yes</td>
<td>Medium-High: 160 kbit/s</td>
</tr>
<tr>
<td>Streaming</td>
<td>UGS</td>
<td>Yes</td>
<td>Yes</td>
<td>Low (radio 8 kbit/s) – High (TV 2 Mbit/s)</td>
</tr>
<tr>
<td>File transfer</td>
<td>BE</td>
<td>No</td>
<td>No</td>
<td>Low-High</td>
</tr>
<tr>
<td>Web Browsing</td>
<td>nrtPS</td>
<td>No</td>
<td>No</td>
<td>Low-Medium</td>
</tr>
<tr>
<td>E-mail</td>
<td>BE</td>
<td>No</td>
<td>No</td>
<td>Low</td>
</tr>
<tr>
<td>E-commerce</td>
<td>rtPS/nrtPS</td>
<td>No</td>
<td>Yes/No</td>
<td>Low-Medium</td>
</tr>
<tr>
<td>On-line gaming</td>
<td>rtPS/nrtPS</td>
<td>No</td>
<td>Yes/No</td>
<td>Medium</td>
</tr>
</tbody>
</table>

Table 15 Mapping between applications and CoSs with traffic requirements [ir212]
Apart from modeling VoIP traffic another source of traffic generator should be present to test the cell performance in the presence of concurrent traffic types. In particular the second feed generator will be based on the rtPS traffic class model.

To evaluate QoE several scenarios with different cases have to be applied. The scenarios proposed by authors are gathered in Table 16 and each scenario will be described further on in this chapter. Additionally we defined in Table 16 the priorities for each simulation test case according to the current status of work.

**Table 16 Scenarios description**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Test case</th>
<th>Description</th>
<th>Priority</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario One</td>
<td>1A</td>
<td>System capacity – basic</td>
<td>High</td>
<td>Basic system performance evaluation</td>
</tr>
<tr>
<td></td>
<td>1B</td>
<td>System capacity with different SNR</td>
<td>High</td>
<td>Basic system performance evaluation</td>
</tr>
<tr>
<td>Scenario Two</td>
<td>2A</td>
<td>System capacity with ACM</td>
<td>Medium</td>
<td>System capacity performance evaluation with ACM enabled. Two coding schemes RS-CC and DaVinci</td>
</tr>
<tr>
<td>Scenario Three</td>
<td>3A</td>
<td>System capacity with different traffic types and ACM</td>
<td>Low</td>
<td>Two traffic types - UGS and rtPS with ACM enabled. Two coding schemes RS-CC and DaVinci</td>
</tr>
<tr>
<td>Scenario Four</td>
<td>4A</td>
<td>System capacity with ACM and CAC</td>
<td>Low</td>
<td>CAC algorithms deployed in scenario for evaluating system capacity with ACM for two coding schemes – RS-CC and DaVinci</td>
</tr>
</tbody>
</table>

Simulation test cases 3A and 4A have a low priority assigned. H-ARQ algorithms based on Incremental Redundancy are considered impossible implement in ns-2 due to DV_HARQ unavailability at the moment of writing this paper. Test cases with H-ARQ retransmission are therefore not taken into account. For ACM simulation it is critical to define the Lookup Tables from Matlab simulation. The ACM threshold values are presented in Table 17.

**Table 17 ACM threshold values for DaVinci codes**

<table>
<thead>
<tr>
<th>Modulation</th>
<th>BPSK</th>
<th>QPSK</th>
<th>16-QAM</th>
<th>64-QAM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Coding</td>
<td>1/2</td>
<td>1/2</td>
<td>3/4</td>
<td>3/4</td>
</tr>
<tr>
<td>DaVinci</td>
<td>-0.88</td>
<td>1.79</td>
<td>4.77</td>
<td>14.18</td>
</tr>
</tbody>
</table>

The following sections contain description of the Scenarios mentioned in Table 16.

### 3.5.1 Scenario one

Scenarios are described according to the template proposed in Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate VoIP QoE when used with DaVinci codes.

#### 3.5.1.1 Objectives

The objectives of this simulation are to test the maximum system capacity and performance at the cell level. We assume a system with one WiMAX base station and several users spread throughout the covered area. Several test cases were applied to test the system capacity. In Case 1A we assume that each
user is experiencing similar channel conditions. In Case 1B the users are first located near the Base Station and moved away to the Cell Edge of the system. Afterwards for the three steps (Minimum Distance, Maximum Distance, and Cell Edge) the maximum capacity of the system is evaluated.

### 3.5.1.2 Test network scenario

The basic network configuration is presented in Fig. 25. Two test scenarios for this configuration are assumed:

- Case 1A – measures the basic system capacity,
- Case 1B – the system capacity under varying SNR values is measured.

![Fig. 25 Basic configuration for Cell Level Simulation](image)

Tables Table 18 and Table 19 describe the system parameters used for this configuration. The following cases were evaluated:

- **CASE 1A** (Basic system capacity)

Each user has the same channel condition (constant value of SNR). All users are using VoIP application. Transmission parameters are measured in the Uplink direction. Each simulation is performed with DaVinci codes as the coding scheme.

**Table 18 Parameters used in Case 1A**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>Variable from 1 to N (“N” limit depends on modulation type – max. 64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>UGS (VoIP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Constant (BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci codes</td>
</tr>
<tr>
<td>Rate</td>
<td>$\frac{1}{2}$</td>
</tr>
<tr>
<td>$K_{bin}$</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>VoIP over UDP, packet size ~120 B</td>
</tr>
<tr>
<td>Scheduler</td>
<td>UGS Scheduler</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Constant</td>
</tr>
<tr>
<td>ACM</td>
<td>Off</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>
• CASE 1B (System capacity for different SNR)

Users are using VoIP application and experience different channel conditions (variable SNR). Transmission is in the Uplink. DaVinci coding scheme is used (Fig. 26).

**Table 19 Parameters used in Case 1B**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>Variable from 1 to N (“N” limit depends on modulation type – max. 64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>UGS (VoIP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci codes</td>
</tr>
<tr>
<td>Rate</td>
<td>½</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>VoIP over UDP, packet size ~120 B</td>
</tr>
<tr>
<td>Scheduler</td>
<td>UGS Scheduler</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>Off</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

**3.5.1.3 Measured parameters**

According to IR 2.1.2. and authors studies the following QoS metrics are used to evaluated QoE in ns+2:

- Performance metrics for Delay Sensitive Applications:
  - Packet Delay,
  - User Average Packet Delay (minimum and maximum packet delay per user),
  - The averaged packet delay per cell.
- Packet losses.
- Throughput Performance Metrics
  - User and cell (average and cell edge) throughputs,
  - Average cell throughput [kBps/cell].
- Other metrics
  - Number of active users per cell,
  - CAC Blocking probability.
The list above presents the QoS specific parameters which should be measured when performing Cell Level Simulation. Not all metrics described in IR2.1.2 are listed as it was decided not to implement them in ns-2.

### 3.5.1.4 Measurement models

The simulation platform described in Chapter 2 and the Matlab E-model implementation described in Chapter 3 was used to evaluate QoE performance.

### 3.5.1.5 Traffic conditions

VoIP traffic generator with CBR (Constant Bit Rate) is used. UDP protocol is used for network data transmission from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

### 3.5.1.6 Expected results

We expect to evaluate DaVinci codes in a basic system configuration and show how a coding scheme can have influence on user experience when using VoIP.

### 3.5.2 Scenario two

Scenarios are described according to the template proposed in Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate VoIP QoE when used with DaVinci codes.

#### 3.5.2.1 Objectives

The objective of this test is to estimate the performance and system capacity at the cell-level with ACM enabled. SNR values should not be constant and change in time. In order to evaluate behavior of WiMAX system in various propagation conditions we have decided to prepare two distinct geographical maps (with respect to the SNR distribution). We generated two SNR matrices for rural and hilly terrains limited to 16 square km area. Mobility models follow Levy-walk distribution and are generated using Matlab source files provided by [leavy]. Both user mobility patterns and SNR maps are combined in order to generate modulation transition trace files functioning as lookup tables for NS2 simulation (Fig. 27). We generated both maps for base station with omni directional antenna, transmit power 42 dBm and of BS antenna’s height of 35m.
In order to generate map of SNR values the Radio Mobile freeware [radio] was used. Radio Mobile is freeware software, which main functionality includes generating radio coverage areas for a given geographical region and elevation data. In simulations we are using SRTM project [shuttle] elevation data as it provides the best elevation resolution for the Europe region.

In order to generate realistic mobility schemes of simulated users the human mobility trace data measured in New York and Disneyworld were adopted [leavy]. User movements are facilitated within simulation in the form of lookup tables (imported to ns2).

In this scenario several users are moving within the covered area. We assume that each user’s channel conditions are changing in time and ACM is enabled. The tests are performed for two coding schemes – RS-CC and DaVinci.

3.5.2.2 Test network scenario

The generated map with the human movement pattern is presented in the figure (Fig. 28). One test scenarios for this configuration are assumed:

- Case 2A – the system capacity under varying SNR values with ACM enabled is measured for different coding schemes.
Table 20 describes the system parameters used for this configuration. The following case was evaluated: Users are transmitting VoIP traffic and moving within the cell (SNR is changing). Adaptive Coding and Modulation is enabled. Transmission is in the Uplink. Different coding schemes are applied for each simulation.

**Table 20 Parameters used in Case 2A**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>Variable from 1 to N (“N” limit depends on modulation type – max. 64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>UGS (VoIP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci codes, RS-CC</td>
</tr>
<tr>
<td>Rate</td>
<td>1/2</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>VoIP over UDP, packet size ~120 B</td>
</tr>
<tr>
<td>Scheduler</td>
<td>UGS Scheduler</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>On</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

**3.5.2.3 Measured parameters**

The parameters measured where described in Section 3.3.5.1.3. The following metrics are taken under consideration:
• Performance metrics for Delay Sensitive Applications:
  o Packet Delay,
  o User Average Packet Delay (minimum and maximum packet delay per user),
  o The averaged packet delay per cell.
• Packet losses.
• Throughput Performance Metrics
  o User and cell (average and cell edge) throughputs,
  o Average cell throughput [kBps/cell].
• Other metrics
  o Number of active users per cell,
  o CAC Blocking probability.

3.5.2.4 Measurement models

The simulation platform described in Chapter 2 and the Matlab E-model implementation described in Chapter 3 was used to evaluate QoE performance.

3.5.2.5 Traffic conditions

VoIP traffic generator with CBR (Constant Bit Rate) is used. UDP protocol is used for network data transmission from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

3.5.2.6 Expected results

We expect to evaluate the gain of DaVinci codes over RS-CC and show how coding schemes have influence on user experience when using VoIP. Enabling ACM should further enhance the system capacity and performance gain should be achieved.

3.5.3 Scenario Three

Scenarios are described according to the template proposed in Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate VoIP QoE when used with DaVinci codes.

3.5.3.1 Objectives

We assume a system with one WiMAX base station and several users moving within the covered area (half of the users are transmitting UGS traffic, the other rtPS). A test case was applied to test the system capacity with ACM under concurrent traffic types (UGS, rtPS). In Case 3A we assume that each user’s channel conditions are changing in time and ACM is enabled. We enhance the system with CAC in case 3B.

3.5.3.2 Test network scenario

The generated movement pattern map is presented in the figure (Fig. 29). The following test case was assumed for this configuration:
  • Case 3A - the system capacity under varying SNR values with ACM enabled was measured.
Table 21 describes the system parameters used for this configuration. The following cases are assumed: Users are transmitting VoIP and MPEG traffic (60% and 40% of users respectively) and moving within the cell (SNR is changing). Adaptive Coding and Modulation is enabled. Transmission is in the Uplink. Different coding schemes are applied for each simulation.

### Table 21 Parameters used in Case 4A

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>From 1 to N (“N” limit depends on modulation type – max. 64-QAM, min. BPSK) 40% of users transmitting rtPS</td>
</tr>
<tr>
<td>Traffic class</td>
<td>UGS (VoIP), rtPS (MPEG)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci codes, RS-CC</td>
</tr>
<tr>
<td>Rate</td>
<td>1/2</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>VoIP, MPEG</td>
</tr>
<tr>
<td>Scheduler</td>
<td>UGS Scheduler</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>On</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

#### 3.5.3.3 Measured parameters

The parameters measured where described in Section 3.3.5.1.3. The following metrics are taken under consideration:

- Performance metrics for Delay Sensitive Applications:
  - Packet Delay,
  - User Average Packet Delay (minimum and maximum packet delay per user),
  - The averaged packet delay per cell.
• Packet losses.
• Throughput Performance Metrics
  o User and cell (average and cell edge) throughputs,
  o Average cell throughput [kBps/cell].
• Other metrics
  o Number of active users per cell,
  o CAC Blocking probability.

3.5.3.4 Measurement models

The simulation platform described in Chapter 2 and the Matlab E-model implementation described in Chapter 3 was used to evaluate QoE performance.

3.5.3.5 Traffic conditions

VoIP traffic generator with CBR (Constant Bit Rate) and MPEG generator are used. UDP protocol is used for network data transmission from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

3.5.3.6 Expected results

We expect to evaluate the gain of DaVinci codes over RS-CC and show how coding schemes have influence on user experience when using VoIP in an environment where some users are streaming MPEG. Enabling ACM algorithm should further enhance the system capacity and performance gain should be achieved.

3.5.4 Scenario Four

Scenarios are described according to the template proposed in Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate VoIP QoE when used with DaVinci codes.

3.5.4.1 Objectives

We assume a system with one WiMAX base station and several users moving within the covered area (half of the users are transmitting UGS traffic, the other rtPS). A test case was applied to test the system capacity with ACM and CAC mechanism. It is assumed that each user’s channel conditions are changing in time and ACM is enabled. We enhance the system with CAC in case 4A.

3.5.4.2 Test network scenario

The generated movement pattern map is presented in Fig. 30. The following test case was assumed for this configuration:
• Case 4A - the system capacity under varying SNR values with ACM and CAC enabled was measured,
Table 22 describes the system parameters used for this configuration. The following cases are assumed: Users are transmitting VoIP and MPEG traffic (60% and 40% of users respectively) and moving within the cell (SNR is changing). Adaptive Coding and Modulation is enabled. Transmission is in the Uplink. The system is enhanced with CAC mechanism. Two coding schemes were applied (RS-CC and DaVinci).

### Table 22 Parameters used in Case 4A

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>From 1 to N (&quot;N&quot; limit depends on modulation type – max. 64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>UGS (VoIP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci codes, RS-CC</td>
</tr>
<tr>
<td>Rate</td>
<td>½</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>VoIP</td>
</tr>
<tr>
<td>Scheduler</td>
<td>UGS Scheduler</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>On</td>
</tr>
<tr>
<td>CAC</td>
<td>On</td>
</tr>
</tbody>
</table>

### 3.5.4.3 Measured parameters

The parameters measured where described in Section 3.3.5.1.3. The following metrics are taken under consideration:

- **Performance metrics for Delay Sensitive Applications:**
  - Packet Delay,
  - User Average Packet Delay (minimum and maximum packet delay per user),
• The averaged packet delay per cell.
• Packet losses.
• Throughput Performance Metrics
  o User and cell (average and cell edge) throughputs,
  o Average cell throughput [kBps/cell].
• Other metrics
  o Number of active users per cell,
  o CAC Blocking probability.

3.5.4.4 Measurement models

The simulation platform described in Chapter 2 and the Matlab E-model implementation described in Chapter 3 was used to evaluate QoE performance.

3.5.4.5 Traffic conditions

VoIP traffic generator with CBR (Constant Bit Rate) and MPEG generator are used. UDP protocol is used for network data transmission from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

3.5.4.6 Expected results

We expect to evaluate the gain of DaVinci codes over RS-CC and show how coding schemes have influence on user experience when using VoIP in cases where some users are streaming MPEG. Enabling ACM algorithm should further enhance the system capacity and performance gain should be achieved. Additional when using CAC the system should guarantee resources for users and show stable performance for all admitted users.

3.6 QoE trials and experiments

The following section present the results gathered from our VIMACCS patch ns-2 simulator described in Chapter 2. The scenarios are described in Section 3.4 of the current Chapter.

3.6.1 Scenario one

The following Section contains simulation results for DaVinci codes scenarios described in Section 3.5.1.

3.6.1.1 Simulation result for Case 1A

Configuration: DaVinci, BTC, WiMAX and LDPC codes, BPSK with rate=1/2

Simulations were conducted to estimate the performance gain of DaVinci codes over BTC and LDPC. We assume an AWGN (EbNo is equal 2 dB) and BPSK modulation. The throughput curves are depicted in Figure 10. In can be seen that for DaVinci codes and BTC with Kbin = 144 there is no gain in performance. For Kbin = 288 DaVinci codes outperform WiMAX LDPC codes as seen on Fig. 31 and Fig. 32.
Fig. 31 Throughput degradation due to number of users for Scenario One Case 1A

We assume that the delay for BTC was infinite because of the packet loss results (100%). It can be seen that for an AWGN channel with EbNo = 2dB conversation is only possible for DaVinci codes. The rest of the codes fall below 60% of the R-value.

Fig. 32 Simulation results for Simulation One Case 1A

The next section shows the results obtained in case 1B.
3.6.1.2 Simulation results for Case 1B

Configuration: DaVinci codes, BPSK, QPSK, 16-QAM, 64-QAM, with rate = 1/2

Simulations were conducted to evaluate QoE for different modulation schemes using DaVinci codes curves for AWGN channel. The average throughput per user was measured and the results are depicted in Fig. 33. The highest possible number of users connected to the system was achieved when using high order modulation – namely 64-QAM 2/3. When using more robust modulation like BPSK ½ a small number of users could be served within the cell. This is because the number of OFDM symbols available for a user decreases when low order modulations are used.

![Graph showing throughput degradation due to number of users for Simulation One Case 1B](image)

Fig. 33 Throughput degradation due to number of users for Simulation One Case 1B

The increasing number of users has impact on latency and packet losses as presented in Fig. 34. The R-factor was calculated according to the achieved results of QoS measurements including both the delay and loss of the packets.
Fig. 34 Simulation results for Scenario One Case 1B

It can be seen that for 64-QAM 2/3 for up to 50 users transmission is still possible and satisfactory to most of the simulated users. Although the one-way delay is within acceptable limits for user count greater than 50 it is impossible to maintain a reliable VoIP transmission link as the packet losses are too high due
to congestion the buffer size has also influence on the system performance [jing]. The figure (Fig. 35) depicts the MOS ratings calculated from the R-value.

### 3.6.2 Scenario Two

The following Section contains simulation results for DaVinci codes scenarios described in Section 3.5.2.

#### 3.6.2.1 Simulation results for case 2A

Configuration: DaVinci, R-S CC, rate=1/2, two maps with different SNR distribution, ACM enabled

The performance comparison of DaVinci and RS-CC codes is presented in the figures (Fig. 36 and Fig. 37). Authors have shown below that DaVinci codes achieve slightly better throughput than RS-CC (Fig. 36) although for small number of users the throughputs for both codes are nearly equal. Still the DaVinci codes are less prone to packet loss and delay than RS-CC. The R-factor degradation shows that when using RS-CC the QoE drops slightly faster than for DaVinci.

![Fig. 36 Throughput for Scenario Two Case 2A](image)

When evaluating DaVinci codes performance in a system with ACM a small performance gain was observed over R-S CC codes. When using DaVinci codes slightly more users (up to 5 more users) are experiencing good QoE. In case where RS-CC codes fall below 60% of the R-value communication is still possible with DaVinci codes (R-value is close to 70%).
Fig. 37 Results for Scenario Two Case 2A

Fig. 38 Mean Opinion Score results for Scenario Two Case 2A

The figure above (Fig. 38) depicts the MOS ratings calculated from the R-value.
The next section describes the results for scenario three.

### 3.6.3 Scenario three

The following Section contains simulation results for DaVinci codes scenarios described in Section 3.5.3.

#### 3.6.3.1 Simulation results for case 3A

Configuration: DaVinci, R-S CC, rate = $\frac{1}{2}$, two maps with different SNR distribution, ACM enabled, mixed traffic types (UGS, rtPS).

Performance comparison of DaVinci and RS-CC codes under mixed traffic type (UGS and rtPS traffic) is presented in Fig. 39 and Fig. 40. Authors have shown that DaVinci codes achieve better results when comparing to RS-CC codes. The throughput gain of DaVinci is higher than for RS-CC.

![Fig. 39 Throughput for Scenario Three Case 3A](image)

Figure below (Fig. 40) presents the result for packet loss and delay. When comparing to previous figure (Fig. 37) it can be seen that under mixed traffic types greater performance gain was achieved by DaVinci codes over RS-CC.
Transmission is still possible for DaVinci codes (R value close ~70%) whereas RS-CC codes fall below 60%. When using DaVinci codes more users are experiencing good QoE (up to 7 users when compared to RS-CC). The figure (Fig. 41) depicts the MOS ratings calculated from the R-value.
3.6.4 Scenario four

The following Section contains simulation results for DaVinci codes scenarios described in Section 3.5.4.

3.6.4.1 Simulation results for case 4A

Configuration: DaVinci, R-S CC, rate=1/2, two maps with different SNR distribution, ACM enabled, CAC enabled, UGS traffic types

Connection Admission Control algorithms were evaluated in Chapter 5. For the purpose of evaluating VoIP QoE the CAC mechanism was employed and used with different coding schemes (DaVinci, RS-CC). To properly interpret the results for case 4A it is necessary to point out that each user’s call is held for a limited number of time, and users not able to start a transmission are not admitted in the network (for more information about CAC please refer to Chapter 5). Therefore the results are nearly constant when increasing the number of nodes. Users have guaranteed resources for their transmission. Once the transmission ends the resources are assigned to the next call.

![Fig. 42 Throughput Scenario Four Case 4A](image)

It can be seen that a reliable transmission is maintained when using Connection Admission Control algorithms (Fig. 42 and Fig. 43). CAC is responsible for assigning resources for incoming calls and in case a call cannot be served by a BS, the user is not admitted in the network. More references and descriptions of CAC algorithms were given in Section 5.
Results collected in this section have shown a performance gain of DaVinci codes over traditional (RS-CC, BTC) and new (WiMAX, LDPC) coding schemes. Reliable transmission can be maintained when CAC mechanisms for resource allocation are used. The figure (Fig. 44) depicts the MOS ratings calculated from the R-value.
3.7 Conclusions for DaVinci

The above sections presented results of evaluation of the QoE of VoIP sessions in IEEE 802.16 network with DaVinci FEC codes implemented. DaVinci codes have proven to be a reliable coding scheme which leads to better QoE results when compared to BTC, WiMAX or LDPC codes. DaVinci codes could further benefit from networks mechanisms supporting reliable transmission and keep QoS within acceptable limits. In this chapter DaVinci codes were evaluated together with ACM and CAC mechanisms. A slight QoE performance gain of DaVinci over RS-CC codes in IEEE 802.16 network with ACM was observed. Still when considering scenario with mixed traffic types QoE performance gain of DaVinci codes further increases. To ensure that users holding a VoIP call don’t experience a degradation of performance the CAC algorithm described in Chapter 5 was introduced to the simulator. The results have shown that in the presence of CAC mechanism users admitted in the network are experiencing high quality of service which stems from the guaranteed resource allocation controlled by CAC. The evaluation of the system with H-ARQ was not possible due to limitations on reliable H-ARQ implementation in ns2. Furthermore the approach of integrating LUT tables, gathered from Matlab, into ns-2 PHY Layer have proven to be consistent and the future integration of MAC ns2 and PHY Matlab/Simulink will include SINR calculation as described in [80216emd].
4. Evaluation of admission control algorithms for WiMAX

In order to provide an adequate QoS level to different service flows and to prevent base station (BS) from running out of resources an adequate Connection Admission Control (CAC) algorithm has to be implemented. Since the introduction of IEEE 802.16e standard, a lot of studies have focused on development of CAC. This section presents evaluation of selected Connection Admission Control algorithms using WiMAX ns2 extension module introduced in section 2.1.1 and called ViMACCS (DaVinci WiMAX module with Admission and Congestion Control Support).

4.1 State of the art

Since the introduction of IEEE 802.16e standard, a lot of studies have focused on development of CAC, as it is a vital part in the QoS provisioning process. An adequate connection admission control (CAC) should prevent base station (BS) from running out of resources, and therefore provide an adequate QoS level to different service flows. This subsection reviews some of CAC algorithms proposed in literature.

4.1.1 Complete Sharing

Complete Sharing (CS) is the least complicated CAC algorithm. This technique assumes that BS accepts all connections until it runs out of resources. CS is easy to implement, but it is sufficient only if the system has to cope with only one class of service. As mentioned earlier IEEE 802.16e defines five service classes, which makes classic CS insufficient for WiMAX.

4.1.2 Guard Channel (GC) based CAC

Classic approach to CAC in mobile wireless networks assumes allocation of dedicated resources e.g. bandwidth or channel reservation, for handoff connections (so called Guard Channel) [misra]. This kind of solution has been originally proposed in [hong] for cellular networks. In this technique a dedicated share of resources always remains reserved for higher priority connections (so called Fixed Guard Channel CAC). In [wang1] [kuo][guo] it is presented how to adapt this technique to WiMAX in order to prioritize handoff connections over arriving connection requests and thus ensure required QoS for handoff connections. In fixed guard channel, if there are multiple service classes present, an optimal value of guard channel is calculated usually using multidimensional Markov chains. However this process is relatively computational-intensive and would be difficult to conduct in real-time for changing radio environment. That is why in [guha] authors introduce a simple dynamic GC scheme, which calculates GC value based on recent incoming connection request thus enabling system to set the GC value dynamically. In case of medium network load this result in smaller new connection blocking probability and handoff dropping probability.

4.1.3 Fair Connection Admission Control

Another method for resource reservation in the process of admitting new connections considers maximum bandwidth allocated by ISP to user [fair]. This technique – FCAC – makes the assumption that it is pointless to allocate more bandwidth than a user can actually use. This is due to the limitations introduced by user’s ISP. When BS receives connection request, the algorithm first checks if user who sent the connection request has enough of remaining bandwidth, then it checks remaining resources of BS. Moreover while making decision it takes into consideration all active user’s ongoing connections. In this manner the probability that a new connection from a user that is utilizing smaller part of his admitted bandwidth will be blocked decreases. This makes decisions more “fair” to users and at the same time the algorithm does not affect overall resources utilization.

4.1.4 Cost–based Connection Admission Control

Next connection admission control algorithm utilizes COL (Competitive On-Line) [chang] [liang] in order to make a decision based on the remaining BS bandwidth [chou]. In this technique for each connection COL function computes appropriate cost. Numerical results presented in [chou] show that this kind of solution can considerably increase performance, in situations when BS polls each user at each frame.

4.1.5 Power Reservation Connection Admission Control

Most CAC algorithms designed for wireless networks do not take into consideration power as a scarce resource. This can lead to unnecessary energy loss, due to ineffective power allocation. Power
Reservation CAC introduced in [qin] reserves both types of resources – power and bandwidth. Also two separate thresholds are set in order to find an optimal tradeoff between handoff dropping probability and new call blocking probability. This technique achieves high energy efficiency by controlling the downlink transmit power. This technique can be seen as an alternative to guard channel reservation techniques that could lead to high power consumption at BS side.

4.2 Admission control algorithms for DaVinci

Currently ViMACCS module, used to evaluate Admission Control algorithms (see section 2.1.1) supports four CAC algorithms basing on concept of reserving bandwidth in the admission process (in this chapter referred to as “bandwidth – based CAC algorithms”). These algorithms are namely Complete Sharing CAC, Dynamic Hierarchical CAC (DHCAC) [dhcac], and Fair CAC (FCAC) [fair] and introduced by us in [icumt] modified version of DHCAC - modified Dynamic Hierarchical CAC. Bandwidth – based Connection Admission Control algorithms have been chosen, as bandwidth, particularly in presence of ACM and changing SNR environment, can be considered a scarce resource (see subsection 4.3). All evaluated algorithms have been described in the following subsections. Moreover throughout study of proposed solutions was presented in number of paper submitted by authors of this document to international conferences [icumt] [bwncp][aina].

4.2.1 Complete Sharing Connection Admission Control Algorithm

Complete Sharing (CS) is the least complicated CAC algorithm. Let us assume, that total bandwidth that can be assigned by Base Station is equal to $B_{\text{total}}$, bandwidth already assigned to ongoing connections is given by $B_{\text{used}}$ and bandwidth required by requested new connection is given by $B_{\text{req}}$. In this case, CSCAC algorithm can be described as:

$$D = \begin{cases} 
1, & \text{if} \quad (B_{\text{used}} + B_{\text{req}} \leq B) \\
0, & \text{other}
\end{cases}$$

Where $D$ is the decision, 0 meaning rejecting and 1 accepting new connection. This technique assumes that BS accepts all connections until it runs out of resources. No other factors e.g. service flow class are taken into consideration. CS is easy to implement, but it is sufficient only if the system has to cope with only one class of service. IEEE 802.16e defines five service classes, which makes classic CS insufficient for WiMAX. Nevertheless this algorithm has been implemented in order to assess if our method of estimating remaining Base Station’s resources is correct. Example of CSCAC algorithm’s functioning has been presented in Fig. 45.

![Fig. 45 Conceptual example of CSCAC algorithm’s functioning](image)

4.2.2 Dynamical Hierarchical (DHCAC) and Modified Dynamical Hierarchical (mDHCAC) Connection Admission Control Algorithm

In this subsection first the original Dynamical Hierarchical Connection Admission Control Algorithm has been presented and discussed. Next Modified Dynamical Hierarchical (mDHCAC) Connection Admission Control Algorithm has been presented as introduced by us in [icumt].
4.2.2.1 Original DHCAC algorithm

Other algorithm that has been implemented, was introduced in [dhcac]. DHCAC is a CAC algorithm basing on the concept of guard channel. Algorithm itself required initial modifications before it could be implemented in ViMACCS module(2.1.1). Decision logic behind the proposed CAC scheme is as follows:

\[ \text{if } (B_{used} + B_{rtPS} \leq B - U) \text{ (1)} \]
accept new rtPS connection;

\[ \text{else if } (B_{rtPS} - (B - U - B_{used}) \leq \sum_{i \in BE} (B_{BE}^i - B_{BE}^{min})) \text{ (2)} \]
accept new rtPS connection;

where:
- \(B_{used}\) - bandwidth used by all connections.
- \(B_{req}\) - BW requested by new rtPS connection
- \(U\) - threshold for UGS connections
- \(B_{BE}^i\) - bandwidth reserved by i-th BE connection
- \(B_{BE}^{min}\) - minimal BW reserved for BE connection (authors assume, that BE connections also have minimal QoS requirements).

Pseudo – code of original DHCAC algorithm has been presented in figure (Fig. 47).

As described in [dhcac] when deciding, whether a new rtPS connection should be admitted, algorithm first checks if total bandwidth decreased by bandwidth reserved by all ongoing connections plus \(B_{rtPS}\) is less than or equal to value \(B - U\). If this condition is met, the connection should be allowed. Example of DHCAC algorithm’s functioning has been presented in Fig. 46.

![Fig. 46 Conceptual example of DHCAC algorithm’s functioning](image)

4.2.2.2 Modification introduced to DHCAC algorithm

Condition (1) in the original algorithm (4.2.2.1) may pose a problem, as original DHCAC algorithm does not take into consideration bandwidth already used by ongoing UGS connections. In order to explain, let us assume an exemplary situation in which:
a) connection type == UGS:
if $B_{\text{USED}} + B_{\text{UGS}} \leq B$
accept connection;
else
reject connection

b) connection type == rtPS:
if $B_{\text{USED}} + B_{\text{rtPS}} \leq B - U$
accept connection;
else if $(B_{\text{rtPS}} - (B - U - B_{\text{USED}})) \leq \sum_{i \in \text{rtPS}} (B_{BE}^i - B_{BE}^{\text{min}})$
accept connection;
else
$$B_{BE}^i = \sum_{i \in \text{rtPS}} (B_{BE}^i - B_{BE}^{\text{min}})$$
if $(B_{\text{rtPS}} - (B - U - B_{\text{USED}} - B_{BE}^i)) \leq \sum_{i \in \text{rtPS}} (B_{\text{rtPS}}^{\text{min}} - B_{\text{rtPS}}^{\text{min}})$
accept connection;
else
$$B_{\text{rtPS}} = \sum_{i \in \text{rtPS}} (B_{\text{rtPS}}^{\text{min}} - B_{\text{rtPS}}^{\text{min}})$$
if $(B_{\text{rtPS}} - (B - U - B_{\text{USED}} - B_{BE}^i - B_{\text{rtPS}})) \leq \sum_{i \in \text{rtPS}} (B_{\text{rtPS}}^{\text{min}} - B_{\text{rtPS}}^{\text{min}})$
accept connection;
else
reject connection;

c) connection type == nrtPS:
if $B_{\text{USED}} + B_{\text{nrtPS}} \leq B - U$
accept connection;
else if $(B_{\text{nrtPS}} - (B - U - B_{\text{USED}})) \leq \sum_{i \in \text{nrtPS}} (B_{BE}^i - B_{BE}^{\text{min}})$
accept connection;
else
$$B_{BE}^i = \sum_{i \in \text{nrtPS}} (B_{BE}^i - B_{BE}^{\text{min}})$$
if $(B_{\text{nrtPS}} - (B - U - B_{\text{USED}} - B_{BE}^i)) \leq \sum_{i \in \text{nrtPS}} (B_{\text{nrtPS}}^{\text{min}} - B_{\text{nrtPS}}^{\text{min}})$
accept connection;
else
reject connection;

d) connection type == BE:
if $B_{\text{BE,USED}} + B_{\text{BE}} \leq B_{\text{BE}}^{\text{min}}$
accept connection;
else if $(B_{\text{USED}} + B_{\text{BE}} \leq B - U)$
accept connection;
else
reject connection;

Fig. 47 original DHCAC algorithm
According to equation (2) we get:

\[ 0.6 + 0.1 \leq 1 - 0.5 \]

\[ 0.7 > 0.5 \]

So, even though UGS is using more bandwidth than its Guard Channel, any new non-UGS connection will not be allowed. In worst-case scenario, this could lead to situation in which a part of bandwidth (equal to the size of UGS Guard Channel) would never be used, even if ongoing UGS connections are using more than \( U \) bandwidth, and there are new non-UGS service flows sending connection requests. That is why the ViMACCS module includes also modified version of original DHCAC. The original algorithm has been modified in order to take into consideration bandwidth already used by UGS connections:

\[
\text{if} \ (B_{\text{used}} + B_{\text{rtPS}} \leq B - (U - B_{\text{UES}})) \quad (3) \\
\text{accept new rtPS connection;}
\]

Admission Control using this modified formula in the following sections is referred to as the mDHCAC (modified DHCAC).

If the first condition (1) is not met, CAC algorithm tries to degrade connections with lower, or equal priority. In simulator used (2.1.1)QoS schedulers are responsible for guaranteeing required minimal bandwidth. Decision whether particular connection should be granted more bandwidth than indicated by its \( B_{min} \) value is made on a per-frame basis and performed by QoS schedulers. In turn, there is no need to perform bandwidth degradation at CAC level. This means, CAC needs only to evaluate \( B_{min} \) of arriving connections, which means that additional conditions as well as connection degradation do not have to be performed. As part of the contribution ITTI has modified DHCAC algorithm also by introducing thresholds for rtPS connections, not only UGS connections. The modified algorithm is presented in the figure (Fig. 48). Example of mDHCAC algorithm’s functioning has been presented in Fig. 49.

Fig. 48 mDHCAC algorithm
4.2.3 Fair Connection Admission Control Algorithm

Next implemented CAC algorithm is Fair Connection Admission Control (FCAC) [fair] algorithm. This method of resource reservation, in the process of admitting new connections, considers maximum bandwidth allocated by ISP to user (so called “Legal Bandwidth”) [fair]. FCAC makes assumption that it is pointless to allocate more bandwidth than user can actually use. This is due to limitations introduced by user’s ISP. When BS receives connection request, algorithm first checks if user has enough remaining Legal Bandwidth, then it checks remaining resources of BS. Moreover while making decision it takes into consideration all active user’s ongoing connections. In this manner the probability that a new connection from a user that is utilizing smaller part of his admitted bandwidth (e.g. user with small number of active, not bandwidth-demanding connections) will be blocked should decrease. This makes decisions more “fair” to users and according to [fair] FCAC algorithm should not affect overall resources utilization. In brief the original decision process of presented CAC algorithm can be described as.

\[
D = \begin{cases} 
1, & RB_i - B_{req,i} \geq 0 \quad (4) \\
0, & other 
\end{cases} 
\]

if \((RB_i - B_{req,i} \geq 0)\) (4)

and \(B_a - B_{req,i} \geq TH_i\) (5)

accept new connection;

else reject new connection;

where:

\(B_{req,i}\) - is the reserved bandwidth requested by the new connection

\(B_a\) - is the available bandwidth of the BS

\(RB_i\) - (Remaining Bandwidth) is the amount of bandwidth which can be used to admit new connections of the user

\(TH_i\) - is the value of threshold for a connection i
Also here, authors had to apply some modifications in algorithm itself before implementing it in ns2. In [fair] authors define $RB_i$ as:

$$RB_i = \sum_{all \, connections \, of \, Ui} MRTR + \beta \cdot (MSTR - MRTR) \quad (6)$$

where:
- MRTR – Minimum Reserved Traffic Rate
- MSTR - Maximum Sustained Traffic Rate
- $\beta$ – reserved bandwidth ratio factor in the range of [0, 1]

And $B_{req,i}$ as:

$$B_{req,i} = MRTR_{req,i} + \beta \cdot (MSTR_{req,i} - MRTR_{req,i}) \quad (7)$$

In [fair] authors also define so called “bandwidth acquirement ratio” as:

$$\eta_i = 1 - \frac{RB_i}{LB_i} \quad (5)$$

Where $LB_i$ is Legal Bandwidth and $RB_i$ is remaining bandwidth of i-th user. New value of threshold $TH$ is generated for user requesting new connection according to formula:

$$TH_i = 2B_{sf} \ln(1 + \sum_{\mathcal{N}} LB_{i} / \sum_{i=1}^{N} LB_{i})$$

Where:
- $N$ – number of users
- $B_{sf}$ - average bandwidth required by a connection
- $S_j = \{ j \mid \eta_j < \eta_i , 0 < j \leq \mathcal{N} \}$ - group of users with $\eta_j$ lower than $\eta_i$ of user requesting new connection
- $\sum_{k \in S_j} LB_k$ - sum of Legal Bandwidths of all users with $\eta_j$ lower than $\eta_i$ of user requesting new connection
- $\sum_{i=1}^{N} LB_i$ - sum of Legal Bandwidths of all users

Fairness is achieved by comparing requesting user’s bandwidth acquirement ratio, to $\eta_i$ of other users during the admission process. Next, based on result of that comparison, new value of threshold $TH$ is generated for user requesting new connection.

Condition (6) in the original algorithm may pose a problem. Let us assume that a new user enters empty cell and wants to create a new connection. It is his first connection, so value of $RB_i$ according to (6) will be equal to zero. Equation (7) is always greater than 0. This means that result of subtraction in (4) will always be a negative number. What follows, the algorithm will never admit any connection. That is why equation (6) of FCAC implemented in ViMACCS has been modified as follows:

$$RB_i = B_{ISP} - \sum_{all \, connections \, of \, Ui} MRTR + \beta \cdot (MSTR - MRTR) \quad (8)$$

Where $B_{ISP}$ is the amount of bandwidth granted to a user by respective Internet Service Provider.

Four bandwidth based CAC algorithms (CSCAC, DHCAC, mDHCAC and FCAC) including modified version of DHCAC - modified Dynamic Hierarchical CAC - have been presented. Bandwidth - based Connection Admission Control algorithms have been chosen, as bandwidth, particularly in presence of ACM and changing SNR environment, can be considered a scarce resource (see subsection 4.3).
4.3 Admission Control versus system capacity estimation - ACM and Symbol Reservation Schemes

In OFDM–based systems the overall capacity can be expressed by the number of available OFDM symbols. In WiMAX however this does not provide reliable capacity information in terms of bits per symbol, as the capacity depends on modulation and coding scheme (MCS) used. Moreover for systems with adaptive coding and modulation (ACM) an MCS for a given connection can change over time, which means that number of symbols required meeting particular connection’s QoS demand will vary and thus - influence capacity. This fact can be expressed after [cur] in the following equation:

\[ C_i(t) = N_r \times R_i(t) \]  

where \( R_i(t) \) is the number of packets that can be carried by one time slot and is determined by the channel quality of connection \( i \) via AMC as in table (Table 30). Either \( R_i(t) \) or \( C_i(t) \) indicates the channel quality or capacity.

The process of estimating and reserving an adequate number of symbols at CAC level in this chapter is referred to as Symbol Reservation Scheme (SRS). Although estimating and reserving number of symbols required to serve connection is a crucial part of the QoS provisioning stage in WiMAX systems, it has been given very little attention when researching Admission Control algorithms. Methods that could be used in order to cope with this problem have been presented below. Moreover, ViMACCS, which has been used to evaluate Admission Control algorithms is – to our best knowledge – the only open source WiMAX ns2 module that provides support for symbol reservation schemes.

“Worst-case-scenario” SRS (WCSRS): One method to cope with this problem could be to assume “worst-case-scenario” - estimate number of required symbols according to most robust MCS, therefore reserving maximum number of symbols for a connection. If a connection is using less symbols than reserved maximum, free slots could be used by low priority connections e.g. BE connection. Although this approach ensures no connections will be dropped due to lack of resources, bandwidth utilization for this method would not be optimal in case most terminals use less robust MCS and there is few BE connections. This kind of solution has been used in [icuntKoucheravy].

SRS with reservation factor (RFSRS): Other method could be to estimate number of required symbols according to formula:
\[ S_{\text{rsvd}} = S_{\text{min}} + \lambda \cdot (S_{\text{max}} - S_{\text{min}}) \] (2)

Where - \( S_{\text{rsvd}} \) – is the number of reserved symbols
\( S_{\text{min}} \) - denotes minimum number of symbols required to serve connection
\( S_{\text{max}} \) - denotes maximum number of symbols required to serve connection
\( \alpha \) - symbol reservation factor in range from 0 to 1.

Although this method could result in higher bandwidth utilization and lower connection blocking probability, it is necessary to find an optimal value of \( \alpha \). The latter analysis will not be investigated.

**SRS with Congestion Control (CCSRS):** Another method dealing with system capacity is reserving number of symbols required for the particular MCS as requested at time of creating new connection. In turn an admission control algorithm would be triggered each time the MCS changes [oll]. If there is not enough resources to serve connection using a new MCS, connection is dropped. In this approach admission control algorithm works as combined Admission Control and congestion control (CC) algorithm. As proposed in [oll] switching to higher order modulation (less robust) would always be accepted, as it requires fewer resources (higher spectral efficiency). This method should result in higher bandwidth utilization and lower connection blocking probability of new connections as we are always reserving only a number of symbols that is required to serve all connections at a given moment. The disadvantage of this method is the fact there is a possibility that already accepted connections could be dropped.

![Fig. 51 changing SNR environment and CCSRS](image)

To present the importance of the symbol reservation method let us consider scenario in which user (\( U \)) using VoIP application moves from location in POSITION I to location in POSITION III (fig. 51). When a mobile terminal is changing its location from the area B to area A the CAC working also as congestion control algorithm is triggered. As user \( U \) is switching to higher order modulation (less robust) his connection is maintained, and a part of symbols previously reserved for user \( U \) is now available for newly arriving connection requests. Later when user \( U \) is crossing from area A to area B, his connection gets dropped, because when he was in area A (using higher order modulation), BS had reused unused symbols by accepting new connections from other users.

Therefore user’s connection is eventually dropped. This could be avoided by performing congestion control algorithm that would use different criteria e.g. drop lower priority connections.

### 4.4 Test scenarios

This subsection describes simulation scenarios used to verify, evaluate and compare implemented Connection Admission Control algorithms. All test scenarios assume single-cell environment. In order to
minimize simulation time, the bandwidth for test scenarios has been set to 2MHz unless indicated otherwise. Connection arrival rate (for scenarios where applicable) varies between simulations from 0.1 up to 3 connections per minute. These values have been chosen accordingly to bandwidth (and therefore available resources) and average duration of existing connections. For scenarios with more resources (e.g. bandwidth 3.5 MHz) higher values of arrival rate could be chosen. Results obtained for described scenarios are presented in 4.5. The list of test scenarios together with their aims has been presented in table 23.

**Table 23 List of scenarios**

<table>
<thead>
<tr>
<th>Test</th>
<th>Objective</th>
<th>Channel</th>
<th>Chapter</th>
<th>Results</th>
</tr>
</thead>
<tbody>
<tr>
<td>SCENARIO I</td>
<td>verification of ViMACCS proper operation</td>
<td>ns2 flat channel</td>
<td>4.4.1</td>
<td>4.5.1</td>
</tr>
<tr>
<td>SCENARIO II</td>
<td>evaluation and comparison of CSCAC, original DHCAC, modified DHCAC and Fair CAC algorithms</td>
<td>ns2 flat channel</td>
<td>4.4.2</td>
<td>4.5.2</td>
</tr>
<tr>
<td>SCENARIO III</td>
<td>performance comparison of CAC and for different FEC schemes</td>
<td>DV, BTC and LDPC codes as LUT</td>
<td>4.4.3</td>
<td>4.5.3</td>
</tr>
<tr>
<td>SCENARIO IV</td>
<td>performance comparison of CAC with ACM and two Symbols Reservation Schemes (SRS)</td>
<td>ns2 flat channel</td>
<td>4.4.4</td>
<td>4.5.4</td>
</tr>
<tr>
<td>SCENARIO V</td>
<td>performance comparison of CAC, ACM with Congestion Control for two FEC schemes</td>
<td>CWER &lt; 0.01</td>
<td>4.4.5</td>
<td>4.5.5</td>
</tr>
</tbody>
</table>

4.4.1 Scenario I – verification of proper operation of ViMACCS

In this scenario situation in which one Base Station has to cope with increasing number of arriving connection requests is considered. For this scenario we conduct simulations without CAC, and compare them to results obtained during simulation with CAC enabled. Therefore two test scenarios for this configuration are assumed:

- Case 1A – Connection Admission Control disabled
- Case 1B – Connection Admission Control enabled, CSCAC

Table 24 describes system parameters used for this configuration. The following cases are assumed:

- CASE 1A

In this scenario the number of arriving connection request will be increased and CAC will be disabled.

- CASE 1B

In this scenario the number of arriving connection request will be increased and CAC will be enabled. The Connection Admission Control algorithm used will be CSCAC.

Parameters shared by all scenarios:

All users experience the same channel conditions (constant value of SNR). Users send either UGS or rtPS connection requests. Transmission parameters are measured in the Uplink direction. We are using TRS-RR scheduler for rtPS [nip]. For this test bandwidth has been set to 3.5MHz.
Table 24 Parameters used in scenario I

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of UGS connections</td>
<td>depending on case variable from 1 up to 36</td>
</tr>
<tr>
<td>Number of rtPS connections</td>
<td>4</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Data transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Constant - QPSK</td>
</tr>
<tr>
<td>Coding Rate</td>
<td>$\frac{1}{2}$</td>
</tr>
<tr>
<td>SF classes</td>
<td>UGS, rtPS</td>
</tr>
<tr>
<td>Traffic type</td>
<td>UDP, packet size 200B (UGS)</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Constant</td>
</tr>
<tr>
<td>AMC</td>
<td>Off</td>
</tr>
<tr>
<td>CAC</td>
<td>Off (case A)/CSCAC (case B)</td>
</tr>
<tr>
<td>cyclic prefix</td>
<td>1/8</td>
</tr>
<tr>
<td>Frame length</td>
<td>20ms</td>
</tr>
</tbody>
</table>

Results together with analysis for this scenario have been presented in subsection 4.5.1

4.4.2 Scenario II – evaluation and comparison of CSCAC, original DHCAC, modified DHCAC and Fair CAC algorithms

In this scenario we measure blocking probabilities and bandwidth utilization for connection admission algorithms supported by ViMACCS. We also compare results obtained for original DHCAC algorithm presented in [dhcac] and DHCAC algorithm modified by us as proposed in [icumt] (referred to as “mDHCAC”). Therefore three test scenarios for this configuration are assumed:

- Case 2A – CSCAC as Admission Control Algorithm
- Case 2B – DHCAC as Admission Control Algorithm
- Case 2C – mDHCAC as Admission Control Algorithm
- Case 2D – FCAC as Admission Control Algorithm

Table 25 - Table 27 describes system parameters used for this configuration. All users experience the same channel conditions (constant value of SNR). All connection requests are generated according to Poisson process. Times of all existing connections are exponentially distributed with mean value 5 min for UGS and 8 min for rtPS connections. Simulations for this test scenario are conducted using ns2 flat channel, meaning that there are no errors introduced in channel (i.e. BER). For this scenario as well as for following test scenarios bandwidth has been set to 2MHz in order to minimize simulation time.

- Case 2A – CSCAC

In this case we measure blocking probabilities and bandwidth utilization for CSCAC

Table 25 Parameters used in Case 2 A

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>VALUE</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAC algorithm</td>
<td>CSCAC</td>
</tr>
<tr>
<td>total bandwidth</td>
<td>2MHz</td>
</tr>
<tr>
<td>Data transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Arrival Rate</td>
<td>0.1 - 3 con/min</td>
</tr>
<tr>
<td>avg. UGS</td>
<td>rtPS con. time</td>
</tr>
<tr>
<td>rtPS scheduler</td>
<td>Round-robin</td>
</tr>
<tr>
<td>cyclic prefix</td>
<td>0.125</td>
</tr>
</tbody>
</table>
Case 2B / C – (m)DHCAC

In this case we measure Blocking Probabilities, Bandwidth Utilization and compare results for original DHCAC and (m)DHCAC algorithms.

Table 26 Parameters used in Case 2B / C

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>VALUE</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAC algorithm</td>
<td>mDHCAC</td>
</tr>
<tr>
<td>UGS GC</td>
<td>0%</td>
</tr>
<tr>
<td>total bandwidth</td>
<td>2MHz</td>
</tr>
<tr>
<td>Data transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Arrival Rate</td>
<td>0.1 - 3 con/min</td>
</tr>
<tr>
<td>avg. UGS</td>
<td>rtPS con. time</td>
</tr>
<tr>
<td>rtPS scheduler</td>
<td>Round-robin</td>
</tr>
<tr>
<td>cyclic prefix</td>
<td>0.125</td>
</tr>
<tr>
<td>Modulation</td>
<td>QPSK ½</td>
</tr>
<tr>
<td>frame duration</td>
<td>20ms</td>
</tr>
<tr>
<td>rtPS traffic</td>
<td>VBR - 500B - 800B</td>
</tr>
<tr>
<td>UGS traffic</td>
<td>CBR - 1000B</td>
</tr>
</tbody>
</table>

Case 2D – (m)DHCAC

In this case we measure Blocking Probabilities, Bandwidth Utilization and compare results for original DHCAC and (m)DHCAC algorithms.

Table 27 Parameters used in Case 2D

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>VALUE</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAC algorithm</td>
<td>FCAC</td>
</tr>
<tr>
<td>total bandwidth</td>
<td>2MHz</td>
</tr>
<tr>
<td>Data transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Arrival Rate</td>
<td>0.1 - 3 con/min</td>
</tr>
<tr>
<td>avg. UGS</td>
<td>rtPS con. time</td>
</tr>
<tr>
<td>rtPS scheduler</td>
<td>Round-robin</td>
</tr>
<tr>
<td>cyclic prefix</td>
<td>0.125</td>
</tr>
<tr>
<td>modulation</td>
<td>QPSK ½</td>
</tr>
<tr>
<td>OFDM frame duration</td>
<td>20ms</td>
</tr>
<tr>
<td>rtPS traffic</td>
<td>VBR - 500B - 800B</td>
</tr>
<tr>
<td>UGS traffic</td>
<td>CBR - 1000B</td>
</tr>
<tr>
<td>Number of users</td>
<td>5</td>
</tr>
<tr>
<td>max con. per user</td>
<td>10 rtPS</td>
</tr>
<tr>
<td>Legal BW</td>
<td>1Mb/s</td>
</tr>
</tbody>
</table>

Results together with analysis for this scenario have been presented in subsection 4.5.2.
4.4.3 Scenario III – CAC and Lookup Tables (ns2+LUT)

In this scenario we use physical layer abstraction based on the idea of Look Up Tables (LUT) for simulations for various FEC codes (nb-LDPC, LDPC, BTC) (see chapter 5.2). ns2 uses characteristics (EbNo vs. BER/WER/CWER) that were previously saved into a file as a lookup table (LUT) to mimic channel behavior. In order to evaluate the potential gain of the nb-LDPC the cell level scenarios with the use of BTC (Block Turbo Codes), WiMAX LDPC and DaVinci codes together with CSCAC and mDHCAC algorithm were applied. Therefore three test scenarios for this configuration are assumed:

- Case 3A – nb-LDPC
- Case 3B – WiMAX LDPC
- Case 3C – Block Turbo Codes

Table 28 describes system parameters used for this configuration. All users experience the same channel conditions (constant value of SNR). All connection requests are generated according to Poisson process. Times of all existing connections are exponentially distributed with mean value 5 min for UGS and 8min. for rtPS connections.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>CAC algorithm</td>
<td>CSCAC</td>
</tr>
<tr>
<td>BS bandwidth</td>
<td>2MHz</td>
</tr>
<tr>
<td>Cyclic prefix</td>
<td>0,125</td>
</tr>
<tr>
<td>modulation</td>
<td>BPSK 1/2</td>
</tr>
<tr>
<td>frame duration</td>
<td>20ms</td>
</tr>
<tr>
<td>rtPS traffic</td>
<td>15 - 20B at 1ms</td>
</tr>
<tr>
<td>Arrival Rate</td>
<td>0,1 - 3 con/min</td>
</tr>
<tr>
<td>rtPS con. time</td>
<td>Avg. 8 min</td>
</tr>
<tr>
<td>rtPS scheduler</td>
<td>Round-robin</td>
</tr>
<tr>
<td>UGS GC</td>
<td>30 %</td>
</tr>
<tr>
<td>Eb/No</td>
<td>1,5 – 2,0</td>
</tr>
</tbody>
</table>

Error sensitivity characteristics of the error correction codes deployed and the performance gain areas of nb-LDPC were presented in the Fig. 52.

| BTC, WiMAX LDPC and DAVINCI codes [dvdel] |

Results together with analysis for this scenario have been presented in subsection 4.5.3
4.4.4 Scenario IV – CAC and ACM - Symbols Reservation Schemes (SRS)

This scenario aims at measuring efficiency of CAC with ACM and two symbol reservation schemes. Measurements are conducted assuming varying SNR environment and CSCAC is used as admission control algorithm. Therefore two test scenarios for this configuration are assumed:

- Case 4A – CAC and “worst-case-scenario” SRS (WCSRS)
- Case 4B – CAC and SRS with Congestion Control algorithm (CCSRS)

Bandwidth utilization (BW), blocking probabilities (BP) and dropping probabilities (DP) have been measured for two symbol reservation schemes - “worst-case” SRS (WCSRS) and SRS with Congestion Control (CCSRS). For WCSRS we used CSCAC. In case of CCSRS we are using CSCAC as both admission control and congestion control algorithm. CSCAC does not prioritize connections due to their service flow class and we are interested mainly in number of symbols required by connection due to MCS changes. Therefore we consider only CBR traffic.

Table 29 Parameters used in Case 4 A / B

<table>
<thead>
<tr>
<th>PARAMETER</th>
<th>VALUE</th>
</tr>
</thead>
<tbody>
<tr>
<td>total bandwidth (BW)</td>
<td>2MHz</td>
</tr>
<tr>
<td>Arrival Rate (AR)</td>
<td>0.1 - 3 conn./min</td>
</tr>
<tr>
<td>avg. connection time</td>
<td>8 minutes</td>
</tr>
<tr>
<td>rtPS scheduler</td>
<td>Round-robin</td>
</tr>
<tr>
<td>cyclic prefix (CP)</td>
<td>0.125</td>
</tr>
<tr>
<td>Modulation (MCS)</td>
<td>BPSK ½ - 16QAM ¾</td>
</tr>
<tr>
<td>frame duration</td>
<td>20ms</td>
</tr>
<tr>
<td>rtPS traffic</td>
<td>CBR - 200B</td>
</tr>
</tbody>
</table>

Table 29 describes system parameters used for this configuration. All connection requests are generated according to Poisson process. Holding times of connections are exponentially distributed with mean value of eight minutes duration. Modulation and coding schemes for all users adapt during simulation in response their movement and the underlying SNR map. We assume user cannot have higher modulation than 16QAM ¾ for sake of simulation execution time. For simplicity we assume SNR values are the same for both Uplink and Downlink. ACM thresholds for Reed Solomon Convolutional Coding (RS-CC) codes were used (Table 30) which comes from the IEEE 802.16e specification.

Fig. 53 SNR map/user mobility models for scenario IV (Map1)
The map has been generated for a village near Warsaw (Poland). Process of map generation and idea of modulation transition files functioning as lookup tables for ns2 has been described in section 3.4.5.2.1. SNR thresholds are set for Reed-Solomon convolutional coding (RS-CC) as defined in IEEE 802.16e standard. The map showing SNR distribution and the history of movements of three exemplary users has been presented in the Fig. 53. Results together with analysis for this scenario have been presented in subsection 4.5.4

### 4.4.5 Scenario V – CAC, ACM and Symbols Reservation Schemes for two FEC schemes

In this scenario we consider simulation cases for two distinct maps with two FEC schemes – DaVinci nb-LDPC and Reed-Solomon convolutional coding (RS-CC). Therefore four test scenarios for this configuration are assumed:

- Case 5A – Reed Solomon Convolutional Coding (RS-CC) with Map 1
- Case 5B – DaVinci nb-LDPC with Map 1
- Case 5C – Reed Solomon Convolutional Coding (RS-CC) with Map 2
- Case 5D – DaVinci nb-LDPC with Map 2

In order to be able to compare the gains of two different FEC codes the two sets of ACM thresholds were taken (Table 30). The thresholds are optimized for two particular FEC codes – Reed Solomon Convolutional Coding (RS-CC) and DaVinci nb-LDPC (DV). The first one comes from the IEEE 802.16e specification.

<table>
<thead>
<tr>
<th>modulation</th>
<th>coding rate</th>
<th>BPSK</th>
<th>QPSK</th>
<th>16QAM</th>
<th>64QAM</th>
</tr>
</thead>
<tbody>
<tr>
<td>DaVinci</td>
<td>1/2</td>
<td>-0.88</td>
<td>1.79</td>
<td>4.77</td>
<td>6.75</td>
</tr>
<tr>
<td>RS-CC</td>
<td>1/2</td>
<td>3</td>
<td>6</td>
<td>8.5</td>
<td>11.5</td>
</tr>
<tr>
<td>DV Gain</td>
<td>2/3</td>
<td>3.88</td>
<td>4.21</td>
<td>3.73</td>
<td>4.75</td>
</tr>
</tbody>
</table>

The above ACMs were used against two different SNR scenarios to present results of system capacity assessment with ACM enabled and various FEC codes. Moreover CAC algorithm with SRS behavior is studied for varying SNR environment. In order to provide more realistic simulation parameters authors have proposed to combine the output of the radio coverage planning (SNR matrices) with the realistic mobility patterns of mobile users (taken from field measurements facilitating DTN networks).

![Fig. 54 SNR map/user mobility models for scenario V Case C / D](image)
For DV nb-LDPC we assume code word error rate is always less than 1% (CWER < 0.01). First map utilized here comes from Scenario IV. Fig. 54 shows second SNR map used. It presents less optimistic SNR distribution than compared with the map in Scenario1. The map has been generated for a village near Katowice (Poland). Results together with analysis for this scenario have been presented in subsection 4.5.5 Results together with analysis for all of the aforementioned scenarios have been presented in next subsection of this chapter.

4.5 Evaluation of results

This subsection presents results obtained for scenarios introduced in section 4.4. Connection Admission Control algorithms and resource reservation techniques introduced in sections 4.2 and 4.3 have been evaluated in terms of Blocking Probabilities per class of service, Bandwidth Utilization (BW utilization as perceived by Base Station’s schedulers), Effective Bandwidth Utilization (BW utilization as perceived by users) and Dropping Probabilities (where applicable). Moreover Connection Admission Control algorithm has been evaluated together with two reservation techniques in varying SNR environment.

4.5.1 Scenario I – verification of proper operation of ViMACCS

This subsection presents results together with analysis for scenario described in 4.4.1. As it can be seen in the figure (Fig. 55) when there is no CAC algorithm the average throughput of a connection falls dramatically. This is because at some point it will be impossible to provide satisfactory level of QoS both for rtPS and UGS connections.

![Average Throughput per active connection](image)

**Fig. 55 Average connection throughput vs. number of connection request**

For low load, delays for both scenarios are equal. When there is no CAC algorithm working at BS side, at some point there is starvation of resources, which leads to high delays (Fig. 56). If we apply the CAC, BS does not let new connections in, if it doesn’t have enough resources to meet connection QoS level.
These results show that CAC method of estimating remaining Base Station’s resources for this scenario is correct and, therefore, all of the admitted connections meet their required QoS levels.

4.5.2 Scenario II – evaluation and comparison of CSCAC, original DHCAC, modified DHCAC and Fair CAC algorithms

This subsection presents results together with analysis for scenario described in 4.4.2. Figures (Fig. 57 and Fig. 58) compare results obtained for original and modified DHCAC algorithm. It can be observed, that modified DHCAC algorithm achieves higher (by about 10%) bandwidth utilization than original DHCAC.
At the same time modified DHCAC offers lower blocking probabilities for rtPS connection. This proves that assumptions made in chapter 4.2.2.2 and [icumt] were right, e.g.: by modifying the algorithm we were able to achieve smaller BPs and higher BW utilization at the same time.

**rtPS Blocking Probability**

for original and modified DHCAC algorithm

Fig. 58 rtPS BPs for original and mDHCAC

**UGS Blocking Probability**

for different UGS GC

Fig. 59 UGS BPs for mDHCAC and CSCAC
Figure (Fig. 59) presents blocking probabilities for UGS connections for different values of GC and for CSCAC. When we increase UGS GC value, the blocking probability is dropping, but at the same time BW utilization is dropping (Fig. 61). It is worth noticing, that for GC equal zero obtained results for CSCAC and DHCAC are the same. It is because CSCAC is in fact a variant of DHCAC with GC = 0%.

**UGS Blocking Probability**
for FCAC, CSCAC and DHCAC

![UGS Blocking Probability Graph](image)

**Bandwidth Utilization**
for FCAC, DHCAC and CSCAC

![Bandwidth Utilization Graph](image)

Fig. 60 UGS BPs for FCAC, CSCAC, mDHCAC

Fig. 61 BW utilization for FCAC, CSCAC, mDHCAC
Although fairness is supposed to be major feature of FCAC in [fair] authors do not propose metrics that could reliably show fairness of this approach. According to our simulation results FCAC slightly degrades BW utilization. From the figure (Fig. 61) one can observe that FCAC is characterized by lowest bandwidth utilization of all four connection admission control algorithms supported by ViMACCS module and at the same time offers highest UGS blocking probabilities (Fig. 60).

4.5.3 Scenario III – CAC and Lookup Tables

This subsection presents results together with analysis for scenario described in 4.4.3. Figure (Fig. 62) presents results obtained for DaVinci codes and CSCAC for different settings of code length (Kbin) and Eb/No values. For all simulations DaVinci codes with Kbin=288 outperform codes with Kbin=144, which results in higher effective (perceived by user) BW utilization for simulations with DaVinci Kbin=288. This is because for lower BER fewer packets are lost due to errors introduced in channel.

Fig. 62 Effective BW utilization for DaVinci codes, CSCAC and different Eb/No settings

 Figures below (Fig. 63 and Fig. 64) present bandwidth utilization curves obtained for physical layer abstraction with BTC, WiMAX LDPC and DaVinci codes for DHCAC and CSCAC respectively. CAC algorithms, DaVinci Codes with code length (Kbin) equal to 288 outperform WiMAX LDPC codes. It is worth noticing, that bandwidth utilization at given Eb/No is lowest for Block Turbo Codes for all cases and bandwidth utilization for those codes is almost equal to zero due to highest BER.
Fig. 63 Effective BW utilization for BTC, WiMAX LDPC and DaVinci codes

Fig. 64 Effective BW utilization for BTC, WiMAX LDPC and DaVinci codes and CSCAC
4.5.4 Scenario IV – CAC and ACM - Symbols Reservation Schemes (SRS)

This subsection presents results together with analysis for scenario described in 4.5.4. Figure (Fig. 65) presents effective (perceived by users) BW utilization for two SRS scenarios - “worst-case” SRS, which reserves maximum number of symbols and SRS combined with CSCAC functioning also as Congestion Control algorithm. Also as a reference in the fig. 65 we presented values of BW utilization for CSCAC and constant SNR (resulting in QPSK ½ MCS) where symbol reservation is not an issue.

As expected the SRS with congestion control achieves lower BP’s and significantly higher BW utilization than “worst-case” SRS. The higher arrival rate the higher dropping probability for SRS with CC (fig. 67). This is because for higher AR the load increases and in turn there is higher probability, that at a given moment more BS’s resources will be consumed and therefore probability of refusing change to lower order modulation (that requires more symbols) rises (fig. 67).

![Bandwidth Utilization for CSCAC and different SRS scenarios](image)

Fig. 65 Effective BW utilization for scenario with two SRS

The figure (Fig. 66) presents blocking probabilities for two SRS scenarios. “worse-case-scenario” SRS has the highest BPs from simulated scenarios. It can be noticed that BPs for CSCAC with congestion control are significantly lower (by almost 40%) than for CSCAC with constant SNR. This is because it is possible for a connection to change to higher order modulation – therefore capacity of system increases and BP for given AR decreases.
**Blocking Probability**
for CSCAC and different SRS scenarios

Fig. 66 BPs for CSCAC and two SRS

**Dropping Probability**
for CSCAC with Congestion Control

Fig. 67 DPs and refuse prob. for CSCAC with CC
4.5.5 Scenario V – CAC, ACM and Symbols Reservation Schemes for two FEC schemes

This subsection presents results together with analysis for scenario described in 4.4.5. The figures from Fig. 68 to fig. 72 show average performance values obtained for each of 4 maps averaged from 10 successive simulation sets, each set consisting of 8 simulations (4x10x8 = 320 separate simulations in total). Each simulation includes approximately 1.5 hour trace of users’ movements. After each simulation collected data was processed as storage of all the trace data would be impossible in terms of required disk capacity. In this scenario symbol reservation scheme with congestion control is used whereas the rest of the parameters are the same as for Scenario IV. Dropping probabilities for scenario VI are shown in the Fig. 70. For Map1 dropping probability is lower for simulations with DaVinci ACM thresholds. This is because for both the DaVinci ACM thresholds and satisfactory (on average) SNR conditions there is less MCS transitions which lead to lower dropping probabilities. This in turn results in higher connection blocking probabilities (fig. 69), as less resources are being freed by congestion control algorithm.

![Bandwidth Utilization](image)

**Fig. 68 Average BW utilization for scenario VI**

For Map2 dropping probabilities as well as blocking probabilities are similar for both ACM thresholds. Bandwidth utilization from the perspective of Base station’s schedulers (percent of scheduled symbols) can be assessed in the Fig. 68. For all simulated cases bandwidth utilization is similar, but in presence of ACM and variable SNR this does not account for equality in throughput values between the scenarios. The average throughput is thus plotted in the Fig. 71. For Map1 both, RS-CC and DaVinci codes achieve similar throughputs (which is an expected behavior, as the probability of high SNR is significant). For map 2 higher throughputs are achieved for simulations with ACM thresholds for DaVinci codes. This is due to different efficiency of MCS (fig. 72). For the map 2 the usage of more robust modulations is higher for scenarios with RS-CC ACM thresholds therefore resulting in lower throughputs (by about 15-20%) for simulations with RS-CC.
Fig. 69 Blocking Probabilities for scenario VI

Fig. 70 Dropping Probabilities for scenario VI
Fig. 71 Average throughput for scenario VI

Fig. 72 MCS usage for Arrival Rate 1.5 calls/min.
4.6 Conclusions for DaVinci

Four bandwidth-based algorithms have been evaluated using ViMACCS ns2 extension developed by ITTI. Simulation results show, that – as expected – CSCAC is characterized by the highest Bandwidth Utilization for all values of Arrival Rates, but at the same time does not provide any means to control Blocking Probabilities for higher – priority connections. DHCAC and mDHCAC algorithms do provide means to lower BP’s of higher – priority connections at the expense of Bandwidth Utilization, which is inversely proportional to value of Guard Channel. Although FCAC is supposed to ensure fairness among users, it is in fact – for most cases - characterized by the lowest Bandwidth Utilization and highest Blocking Probabilities of all evaluated admission algorithms. **According to simulations, impact of using different error correction codes is most significant in case of varying SNR environment. Using more robust FEC schemes (e.g. DAVINCI codes as in chapter 4.5.5) results in higher average cell throughputs while keeping similar Dropping Probabilities - when coping with bad SNR conditions. When coping with good SNR conditions – it results in lower Dropping Probabilities while keeping similar average cell throughputs. Finally, WCSRS can be considered only in systems that experience very poor SNR conditions.** Symbol Reservation Schemes directly impact Bandwidth Utilization and Blocking Probabilities, therefore more attention should be brought to the topic of SRS combined with adaptive scheduling and admission control.
5. TCP over WiMAX

The main goal of this chapter is to provide the relevant background information on TCP over WiMAX performance evaluation. In particular the available TCP protocols for WiMAX and their mechanisms are described later on in this chapter. Further on the authors propose several test cases which evaluate TCP performance over WiMAX with DaVinci codes implemented as FEC coding scheme.

5.1 TCP over WiMAX state of art

To assure successful delivery of packets between two nodes of an IEEE 802.16 standard based network, a transport protocol which handles the retransmission of packets should be deployed. Since IEEE 802.16 compliant networks are based on IP protocol the TCP protocol, residing at the Transport Layer, is used during transmission. TCP is the most important protocol in the world wide network and numerous studies have been accomplished to analyze the performance of the Transport Control Protocol in wireless environment [mort],[yan],[cant],and [bakshi]. A survey on TCP performance evaluation and modeling was conducted in [pagan]. However the performance requirements for wired and wireless networks differ. Several publications consider the performance evaluation of TCP over wireless links. Although the behavior of TCP over WiMAX networks has not been fully researched, some reference papers like [bill] describe the basic ideas behind TCP over WiMAX. In [hale] a measurement study has been conducted to evaluate the system performance of WiMAX networks with four TCP variants, including protocols like: TCP New Reno, TCP Cubic, TCP Vegas, TCP Veno. The work presented in [bill] and [hale] created the fundamentals for other TCP research papers. It was clear that ARQ techniques have influence on TCP performance since the retransmission of packets could be limited at the transport layer by resending lost ARQ blocks at the transmission link level [yong]. However this approach has its limitations which are described in [ivan] and should be taken into account when planning to use TCP with ARQ on a WiMAX link. Other publications [abde] consider the analysis of latency on Wireless Links with FEC/ARQ for error recovery. When deployed in an appropriate way the cooperation could be beneficial for the overall system performance. The work on efficient TCP and ARQ utilization has lead to studies on the cross-layer design for IEEE 802.16 compliant networks. A research was conducted in [hwang]. The TCP performance over 802.16e compatible systems was presented in [jing] describing the ns-2 WiBRO implementation. The results for TCP measurements in a real WiMAX network were conducted in [gron],[perez].

5.2 Tools and techniques for TCP performance assessment

TCP is the most common protocol in the network. It resides in the Fourth Layer of the Seven Layer ISO/OSI model (Fig. 73) and is responsible for establishing and maintaining a connection between two end points.

---

Fig. 73 TCP and the 8 Layer OSI model
TCP protocol establishes a connection during the three-way handshake process. This ensures that received packets are acknowledged after receiving – the receiving node issues an ACK message to the sending host to confirm that the packet was received error free. If the sender does not receive an ACK message during the timeout period it resends the packet thus extending the delay of the transmission and utilizing more bandwidth. The three-way handshake is presented in Fig. 74.

![TCP Three-way Handshake process](image)

Fig. 74 TCP Three-way Handshake process

Test performed in Chapter 3 were carried out using UDP protocol. UDP is considered a connectionless protocol whereas TCP is connection oriented protocol. In comparison to UDP the Transmission Control Protocol is slower which results from the need for acknowledging every received packet. A research on the variations of ACK schemes was conducted in [jing]. Although TCP is slower, and its use with time sensitive applications like VoIP is limited it’s still an excellent solution for application which requires a reliable connection for data sending (example FTP). Moreover the performance of TCP is higher than UDP when dealing with packet loss or throughput metrics. This is achieved because TCP uses a congestion control mechanism to avoid oversubscription. A network can be congested when network resources are limited and users are constantly requesting new services. The impact on congestion window size on TCP performance in WiMAX is presented in [hale]. The tests were conducted in a real WiMAX testbed using Motorola modem RSU-2510F for connection. Another test on TCP behavior in a real testbed was conducted in [perez].

According to [jing],[hale],[pagano] TCP has influence on:

- Throughput – retransmission limits the available throughput per user as some resources have to be allocated for ACK messages,
- Delay – the average delay per user increases as packets has to be acknowledged,
- Packet loss – the number of dropped packets decreases due to the nature of TCP protocol (retransmission, congestion control).

5.3 TCP evaluation requirements

When evaluating TCP performance all the characteristic of the protocol have to be taken into account as it will justify some of the results obtained during simulation. RFC 2001[rfc2001] contains information about the algorithm used with TCP, in particular the slow start, congestion avoidance, fast retransmit and fast recovery algorithm.

5.3.1 Slow Start

Is the algorithm used to avoid overloading the queue of routers or other network devices. Slow start operates by observing the rate of the acknowledgement arriving at the sender side and adjusting the rate at which new packets are send according to this observation. Additional slow start adds another window to the sender’s TCP: the congestion window “cwnd”. When a new connection is established between two
sides from different networks, the congestion window is initialized to one segment (i.e., it can be set to the segment size announced by the other end, or to the default value, typically 536 or 512). The congestion window increases its size by one segment each time an ACK is received. The sending side starts by transmitting one segment and waiting for the corresponding ACK. When ACK arrives at the sender side, slow start increments the window size from one segment to two segments, thus more data can send. For each ACK the congestion window size is increased by one, thus when receiving two ACKs the window size is increased to four. At some point the capacity of the link is reached and an intermediate router will start discarding packets. The sender then knows that its congestion window size is too large and it gets decreased to a size of one segment again and slow start starts over.

5.3.2 Congestion avoidance

Congestion can occur when data is put fast LAN segment out to a slower WAN connection. Congestion can also occur when multiple input streams arrive at a router whose output capacity is less than the sum of the input streams. To mitigate the impact of congestion on lost packets the congestion avoidance algorithm is introduced. In the network there are two situations that signal a packet loss – either the timeout has expired or duplicate ACKs arrive at the sender side.

In practice congestion avoidance is implemented together with slow start because when congestion occurs TCP must slow down its transmission rate of packets into the network and then call slow start to start transmitting at a smaller window size. Both, congestion avoidance and slow start, require two variables to be defined – on the cwnd window and the slow start threshold size sstresh. The combination of the two algorithms works as follows:

- Initialize – set cwnd to one segment and ssthresh to 65535 bytes,
- When congestion occurs – set ssthresh to the size of half the current window size (must be at least the size of two segments) cwnd. If the congestion is indicated by a timeout – set cwnd to one segment,
- When data is acknowledged by the other end, increase cwnd – if cwnd is less than or equal to ssthresh, TCP is in slow start mode and cwnd is incremented by one segment each time an ACK is received. Otherwise cwnd is increased by one segment size each round-trip time.

5.3.3 Fast Retransmit

The sender side may receive a duplicate ACK when an out-of-order segment is received. The purpose of the duplicated ACK is to inform the transmitter what sequence number is expected. Since TCP does not know what the cause was for the duplicate ACK, it waits for small number of duplicate ACKs to be received. This means that the packet with the sequence in the duplicate ACKs has been lost. It is assumed that the reordering process at the receiver side takes time thus duplicate ACK may be received by the sender. However, if three duplicate ACKs are received it is indicated that a segment has been lost. TCP then performs a retransmission of what appears to be the missing segment, without waiting for a retransmission timer to expire.

5.3.4 Fast Recovery

After fast retransmission of the missing segment the congestion avoidance is called for a fast recovery. Fast Recovery was introduced to allow high throughput under moderate congestion, especially in the case when large windows are used. The reason why congestion avoidance is used instead of slow start is that when TCP receives a duplicate ACK it is information for the sender side that there is still data flowing between the two ends and flow reduction is not needed. The fast recovery algorithm is implemented together with the fast retransmission algorithm. The combination of the two algorithms works as follows:

- After receiving the third duplicate ACK in a row, sstresh is set to one half of the congestion window size (not less than two segments). After that the missing segment is retransmitted. Cwnd is set to ssthresh plus 3 times the segment size. This inflates the congestion window by the number of segments that have left the network and which the other has cached,
- Cwnd is incremented by the segment size each time a duplicate ACK arrives. This inflates the congestion window for the additional segment that has left the network. If allowed, the new packet is transmitted with the new value of cwnd.
• When the retransmitted data is acknowledged the value of cwnd is set to sssthresh. This ACK additionally acknowledges all the intermediate segments sent between the lost packet and the receipt of the first duplicate ACK (the packets are still in the receiver buffer).

5.4 Test methodology

To assess TCP performance over WiMAX it is required to feed the system with TCP traffic (for example FTP traffic). Once the traffic is set traffic traces are collected at the sink node and tstat can be further used to post-process the data. Another application is Netperf and can be used to measure various aspects of networking performance. The tools are described in [net].

When assessing TCP performance over WiMAX the following should be taken into account:

• The congestion widow has influence on performance thus test for different window sizes should be conducted,
• The buffer size of the Base Station has influence on performance as the buffer size has influence on jitter (jitter is absorbed when large buffer sizes are used),
• TCP retransmission significantly decreases packet loss but the average delay increases. Retransmissions have also influence on throughput. One way to improve performance is to reduce the number of ACK as they consume network resources. In particular the normal ACK, delayed ACK, filtered ACK schemes have to be evaluated [jing]. ACKs are carried using the BE service class.

According to the figure above (Fig. 75) only TCP Uplink performance is taken under consideration during the simulation. We already justified this approach in Section 3. Some preliminary results on TCP Downlink and Uplink performance were obtained in [jing].

TCP over WiMAX assessment has to be carried out for different variations of the TCP protocol to select the most suitable one to be used with DaVinci codes. Different variations of the protocol present different performance gains according to the environment where the protocol is being deployed [hale]. The commonly known variations of TCP used in wireless networks are:

• TCP New Reno [reno] – this TCP variant is characterized by slow start algorithm and congestion avoidance phase. When using the slow start algorithm a sender is allowed to grow his window to capture bandwidth. Congestion avoidance phase grows the window linearly promoting fairness and stability. Loss is detected when a third duplicate ACK arrives at the sender (in this case the congestion widow size is set to half the current window) or the RTO (Retransmission Timeout) is reached,
• TCP Cubic [cubic] – is characterized by an aggressive window growth function that is independent of RTT. In case loss is detected the TCP Cubic variation reduces its window size by 0.8 each time. Cubic is used in Linux kernels as the default TCP variant,
• TCP Vegas [vegas] – this variation can linearly change the congestion window size. It is a New Reno variation with early loss detection. In comparison to the New Reno implementation it is possible to retransmit loss packet before the third ACK arrives at the
sender side. This implementation due to window stability and low retransmission rates is
the most suitable one for wireless networks,
• TCP Veno [veno] – this implementation is targeted at wireless networks. It combines Reno
and Vegas mechanisms to deduce whether a network is congested or if random loss is more
likely.

For test scenarios we made the same assumptions as in Chapter 3. The differences are listed as
follows:
• TCP protocol is used instead of UDP,
• FTP traffic generator is used instead of VoIP traffic generator,
• FTP is considered a rtPS service class, which is similar to nrtPS (nrtPS is not supported by
VIMACCS ns-2 patch).

Among all the possible measurements the following are selected to assess TCP performance over
WiMAX:
• Transfer time – defined by the time interval between the first SYN segment and the last
SYN segment carrying data,
• Throughput – computed by dividing the packet size D by the transfer time,
• Round Trip Time – especially the maximum, minimum and average RTT per user in a cell
shall be investigated,
• Number of retransmission segments – this parameter has influence on the delay and packet
loss metric thus the number of retransmitted segment should be taken into account,
• Congestion windows size – the average cwnd size is computed as the difference between
the highest data sequence number send and the highest acknowledgement number received.
The size of cwnd has influence on the throughput performance.

The test methodology follows the 6 point template described in section 3.4.

### Table 31 Scenarios description

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Description</th>
<th>Priority</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scenario One</td>
<td>TCP performance with DaVinci code over WiMAX</td>
<td>High</td>
<td>Basic system performance evaluation with FTP application</td>
</tr>
<tr>
<td>Scenario Two</td>
<td>TCP performance with different coding schemes (DaVinci, RS-CC codes) over WiMAX with AMC</td>
<td>Medium</td>
<td>System capacity performance evaluation Two coding schemes RS-CC and DaVinci. AMC is enabled.</td>
</tr>
<tr>
<td>Scenario Three</td>
<td>TCP performance with different coding schemes (DaVinci, RS-CC codes) and mixed traffic types over WiMAX with AMC</td>
<td>Low</td>
<td>Two traffic types - UGS and rtPS with ACM enabled. Two coding schemes RS-CC and DaVinci.</td>
</tr>
</tbody>
</table>

The following sections describe the scenarios employed to evaluate TCP performance over WiMAX.
A brief description on each scenario is given in Table 31.

### 5.4.1 Scenario One

Scenarios are described according to the template proposed in Chapter 3 Section 3.4. First the
objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic
conditions are further determined. The measurement models for this scenario are described later on. This
section concludes with the expected results for this scenario. The aim of this scenario is to evaluate TCP
performance with DaVinci codes over WiMAX.

#### 5.4.1.1 Objectives

The objectives of this simulation are to evaluate TCP performance in a single-cell scenario. We
assume a system with one WiMAX base station and several users spread throughout the covered area.
One test case was applied to test the system capacity. The users are sending files to FTP server (flow is in the Uplink direction). The service class for FTP is rtPS. DaVinci codes are used as the FEC coding scheme. Simulations are conducted for BPSK ½, QPSK (1/2 and 2/3) 16-QAM (1/2 and 2/3) and 64-QAM 2/3.

5.4.1.2 Test network scenario

This parameters for test network scenario are gathered in Table 32. It is assumed that each user is experiencing similar channel conditions at the same time.

Table 32 Parameters used in Scenario One

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>From 1 to N (“N” limit depends on modulation type – max.64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>rtPS (FTP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci</td>
</tr>
<tr>
<td>Rate</td>
<td>1/2, 2/3, 3/4</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>FTP</td>
</tr>
<tr>
<td>Scheduler</td>
<td>Round Robin ??</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>Off</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

5.4.1.3 Measured parameters

According to IR 2.1.2 and our research the following metrics are used to evaluate TCP performance in ns-2:

- Throughput,
- Packet loss,
- One-way delay,
- Number of retransmitted segments,
- Congestion window size.

5.4.1.4 Measurement models

The simulation platform described in Section 2 was used to evaluate TCP over WiMAX performance.

5.4.1.5 Traffic conditions

FTP traffic generator with VBR (Variable Bit Rate) is used. TCP protocol is used for data transport from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

5.4.1.6 Expected results

We expect to evaluate performance of TCP with DaVinci codes over WiMAX. In particular authors want to estimate how TCP retransmissions and congestion window size have influence on WiMAX performance when using DaVinci codes.

5.4.2 Scenario Two

Scenarios are described according to the template proposed in Chapter 3 Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate TCP performance with DaVinci codes over WiMAX.
5.4.2.1 Objectives
The objectives of this simulation are to evaluate TCP performance in a single-cell scenario. We assume a system with one WiMAX base station and several users spread throughout the covered area. One test case was applied to test the system capacity with AMC. The users are sending files to FTP server (flow is in the Uplink direction). The service class for FTP is rtPS. DaVinci and RS-CC codes are used as the FEC coding scheme. Simulations are conducted for BPSK ½, QPSK (1/2 and 2/3) 16-QAM (1/2 and 2/3) and 64-QAM 2/3.

5.4.2.2 Test network scenario
This parameters for test network scenario are gathered in Table 33. It is assumed that each user is experiencing similar channel conditions at the same time.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>From 1 to N (“N” limit depends on modulation type – max.64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>rtPS (FTP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci, RS-CC</td>
</tr>
<tr>
<td>Rate</td>
<td>1/2</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>FTP</td>
</tr>
<tr>
<td>Scheduler</td>
<td>Round Robin</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>On</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

5.4.2.3 Measured parameters
According to IR 2.1.2 and our research the following metrics are used to evaluate TCP performance in ns2:

- Throughput,
- Packet loss,
- One-way delay,
- Number of retransmitted segments,
- Congestion window size.

5.4.2.4 Measurement models
The simulation platform described in Chapter 2 was used to evaluate TCP over WiMAX performance.

5.4.2.5 Traffic conditions
FTP traffic generator with VBR (Variable Bit Rate) is used. TCP protocol is used for data transport from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

5.4.2.6 Expected results
The authors expect to evaluate the potential gain of DaVinci codes over RS-CC when using TCP. In particular the authors want to estimate how the congestion window size and number of retransmissions change when using different coding schemes.

5.4.3 Scenario Three
Scenarios are described according to the template proposed in Chapter 3 Section 3.4. First the objectives of Scenario One are defined. The test network scenarios, measured parameters and traffic
conditions are further determined. The measurement models for this scenario are described later on. This section concludes with the expected results for this scenario. The aim of this scenario is to evaluate TCP performance with DaVinci codes over WiMAX.

5.4.3.1 Objectives
The objectives of this simulation are to evaluate TCP performance in a single-cell scenario. The authors assume a system with one WiMAX base station and several users spread throughout the covered area. One test case was applied to test the system capacity with ACM. Some users are sending files to FTP server (flow is in the Uplink direction) and several users are holding a VoIP call. The service class for FTP is rtPS and for VoIP - UGS. DaVinci and RS-CC codes are used as the FEC coding scheme. Simulations are conducted for BPSK ½, QPSK (1/2 and 2/3) 16-QAM (1/2 and 2/3) and 64-QAM 2/3.

5.4.3.2 Test network scenario
This parameters for test network scenario are gathered in Table 34. It is assumed that each user is experiencing similar channel conditions at the same time.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of users</td>
<td>From 1 to N (“N” limit depends on modulation type – max.64-QAM, min. BPSK)</td>
</tr>
<tr>
<td>Traffic class</td>
<td>rtPS (FTP), UGS (VoIP)</td>
</tr>
<tr>
<td>Bandwidth [MHz]</td>
<td>3.5</td>
</tr>
<tr>
<td>Transmission direction</td>
<td>Uplink</td>
</tr>
<tr>
<td>Modulation</td>
<td>Variable (64-QAM, 16-QAM, QPSK, BPSK)</td>
</tr>
<tr>
<td>Coding</td>
<td>DaVinci, RS-CC</td>
</tr>
<tr>
<td>Rate</td>
<td>1/2</td>
</tr>
<tr>
<td>Kbin</td>
<td>144</td>
</tr>
<tr>
<td>Traffic type</td>
<td>FTP, VoIP</td>
</tr>
<tr>
<td>Scheduler</td>
<td>Round Robin ??</td>
</tr>
<tr>
<td>SNR [dB]</td>
<td>Variable</td>
</tr>
<tr>
<td>ACM</td>
<td>On</td>
</tr>
<tr>
<td>CAC</td>
<td>Off</td>
</tr>
</tbody>
</table>

5.4.3.3 Measured parameters
According to IR 2.1.2 and our research the following metrics are used to evaluate TCP performance in ns2:

- Throughput,
- Packet loss,
- One-way delay,
- Number of retransmitted segments,
- Congestion window size.

5.4.3.4 Measurement models
The simulation platform described in Section 2 was used to evaluate TCP over WiMAX performance.

5.4.3.5 Traffic conditions
FTP traffic generator with VBR (Variable Bit Rate) and a VoIP generator with CBR (constant bit rate) are used. TCP protocol is used for data transport from the user terminal (send node) to the receiver (sink node). Traffic direction is in the Uplink.

5.4.3.6 Expected results
The authors expect to evaluate the potential gain of DaVinci codes over RS-CC when using TCP in a mixed traffic class environment. In particular the authors want to estimate how the congestion window size and number of retransmissions change when using different coding schemes.
5.5 Test results assessment

Tbd.
6. QoE ontology

Ontology as a model of knowledge about some domain, is considered as a useful element in many technological solutions or applications. It plays a role of knowledge database and supports reasoning about this data. In DaVinci project, ontology has been considered as a part of QoS/QoE measuring system. Intention of the ontology was support of measurement architecture for combined subjective and objective QoS/QoE evaluation. That is why QoE ontology for DaVinci project has been designed and developed. Description of this ontology, with their concepts and relationships is shown in this section. Moreover overview of existing QoE/QoS ontology is presented, as well as machine readable formalisms used to ontology creation.

6.1 Available ontologies desktop research

Many different domains are described by ontologies since they are promising and increasingly developing method for modeling some particular domain of concepts. There are several interesting approaches for presenting QoS and QoE issues as an ontology model.

QoE metrics are assumed as a subjective user perception of network QoS objective parameters. That is the reason that QoS and QoE concepts in many cases are related to each other. Researchers try to map QoS parameters to QoE concepts, to provide understanding subjective quality of experience parameters to objective QoS parameters. QoS/QoE measurement techniques and mechanisms were reviewed and presented in section 3.

Ontologies more often describe QoS domain than only QoE that is why we present here some QoS/QoE ontology approaches. In [sill] agent based platform for measure QoE parameters is presented. It is performed by agent, which control network QoS parameters and trying to match the QoE. The authors propose managing of QoS by arbitrator using a series of rules, references tables and end-user feedback. QoE agent based platform summarization process is presented in Fig. 76:

![Fig. 76 QoE agent based platform summarization process [sill]](image-url)

---

[sill]: Author's note or reference to the original source.
Framework presented in Fig. 76, considers both subjective feedback and objective performance metrics to manage the QoS. The platform has been designed to meet the QoE requirements during the transmission. As a result it generates a mapping of the QoS metrics to QoE. The authors assumed that network and application QoS (AQoS and NQoS) can be controlled separately using weightings. Normalized QoE can be defined as a function of AQoS and NQoS:

\[ NQoS + AQoS = QoE. \]

There are two ways of weights representation – the objective weights by Ametric e.g. Aloss, and subjective weights by Wmetric e.g. Wdelay. The result of platform activity (Fig. 76) is QoS to QoE mapping in a way of equation:

\[ NQoS = \frac{Wdelay \times Adelay + Wjitter \times Ajitter + Wloss \times Aloss}{2} \]

To deal with defined task QoE ontology has been developed. The intention of the QoE ontology was supporting of QoE framework. This QoE ontology includes agent actions and concepts. There are 6 basic actions, which are connected to configuration of the packetiser/depacketiser, network and transmission/reception requests. Concepts defined in ontology support these actions. Each concept is derived from a series of slots or attributes. Concepts, which are included by QoE ontology are presented in Fig. 77, and QoE ontology actions in Fig. 78.
Although agent-based platform and steps performed using it, and experiments conducted using it are described in [sill], ontology components are shortly shown. This approach seems to be interesting, using ontology to QoS/QoE mapping, but not many details are presented. Definitions of the ontology concepts are not included. Ontology actions are not defined as well. That is why we didn’t use this approach to QoE ontology for DaVinci project. Agent based platform is described in some details, but it is not applicable for DaVinci project goals, because of user feedback needs. We can only notice some elements in ontology classes, which are also included in QoE DaVinci ontology, like QoS Parameters for example.

An extension of QoE ontology presented above can be found in [gall]. The authors conducted QoS ontologies research and decided to improve ontology described in [sill]. Purpose of these improvements was to support QoE by full QoS management capabilities. The authors refer to requirements from [dobs], which say that QoS ontology should:

- decide which QoS mechanisms is better to fits the user needs;
- perform QoS monitoring and detection of SLA violations;
- carry out QoS adaptation.

New QoE ontology actions are presented in Fig. 79. The new QoE ontology is used for agent-based platform, and improvements of ontology should bring enhancements in this platform. The QoS management process is conducted as follows. At the beginning there is assumed that quality of the transmission is optimal, and the arbitrator takes the default parameters from the QoS Reference Table. This table called QoS RT provides the objective network performance metrics. In next stage the arbitrator waits for any user request and checks that the QoS RT performance metrics are met. The arbitrator is responsible for reaction on users request or when the network conditions are compromised. In these situations arbitrator have to put necessary actions to meet the desired QoS level. It could be achieved by changing the QoS mechanism or its parameters.

Similarly to approach in [sill], the QoE framework end user experience is measured considering two metrics: AQoS (Application Quality of Service – multimedia coding, packetisation) and NQoS (Network Quality of Service – the routers, switches, strategies for admission control, policy, scheduling, reservation etc.). Activity of this platform is based on user feedback. Real time quality improve/reduce requests are made by user during the media transmission. These requests are translated into a series weighting factors which are applied to the metrics $AQoS$ and $NQoS$.

This approach requires end user feedback. Within Da Vinci environment/conditions this couldn’t be performed. There is not any testing environment, which can perform tests with high amount of users to ensure objective measures. There is no possibility of interaction between DaVinci solutions and high amount of users.

\begin{table}[h]
\centering
\caption{QoE Ontology Actions}
\begin{tabular}{|l|l|}
\hline
Agent Action Name & Description \\
\hline
TxReq & Transmission request \\
RxReq & Reception request \\
NetConfReq & Network configuration request \\
NetDelConfReq & Delete network configuration request \\
NetStatusQuery & Network status query \\
PktConfReq & Packetiser configuration request \\
DepktConfReq & Depacketiser configuration request \\
PollPerformanceMetrics & Polling performance metrics \\
MonitorEndUserReq & Monitoring end-user requests \\
CodecConfigReq & Codec Configuration request \\
\hline
\end{tabular}
\end{table}

Fig. 79 QoE ontology actions [gall]
In QoSOnt is presented. That is ontology, designed to provide generic model for QoS concepts, applicable in multiple domains. It is intended to future reuse and extension. It means that this ontology can be base for other ontologies. New users can create their own ontologies for many different and special tasks or applications, using elements of QoSOnt. It can be also used for ontology-based application exploited by multiple participants to cooperate with each other. Thus particular ontology parts can be replaced by other, adjusted to required conditions by different participants. QoSOnt is in a form of modular ontology, where every module is an ontology itself. This ontology is constructed of 3 layers, which are presented in Fig. 80.

The Base QoS layer includes generic, main concepts relevant to QoS. Unit ontologies are also at this level. Example of unit ontologies is Time, which represents units of time and how to convert between them. In the next layer there are ontologies defining particular QoS attributes and their metrics. On top of this is the domain-specific layer, which links the lower layers to specific types of computer system. Lower layers are used to build ontologies of higher layers, which are specialization of them. Concepts with properties of the Base QoS ontology are presented in the Fig. 81.
In [sanch] knowledge-based framework for mappings between technical QoS and QoE for automatic calculation of the level of QoE is proposed. This framework is based on ontologies and rules. The ontologies which are used are quality of service ontology models defined in [lock] with their instances. Framework for QoE calculations is presented in Fig. 82.
Fig. 82 Framework for QoE calculations [sanch]

Framework for QoE calculations starts its activity by retrieving special data to get set of measures about network and applications. These measures are collected and are used as instances of the quality of service ontology. To define relationships, formulas or functions between the different quality domains rule language is used. To processes defined rules, reasoning is needed, and in the result of rule processing, information about QoE level by new instances related to the QoE domain is provided. In this way QoS to QoE mapping is performed.

For representation QoS-QoE relationships, ontology in OWL and rules in SWRL are proposed, but SRWL does not support mathematical calculations. To proceed mathematical calculations usage of OpenMath tool is proposed.

Process of QoS-QoE relationships representation has the following steps to take. First specific SWRL built-in type (mathext), which is needed to apply a mathematical formula defined in OpenMath to a set of instances should be created. Then the generation of a class (QoSFunction) and a property (hasOM) to hold the OpenMath function should be done. The value of the property is an OpenMath XML expression. At the end existing reasoner and interfaces to communicate with a Mathematical tool should be use and the result should be generated.

This approach seems to be very interesting, especially because of using SWRL rules. We considered rules as a good way of mapping QoS parameters to QoE. Unfortunately any practical examples are not presented in [sanch]. Any examples of the rules and defined instances of ontologies. Any mathematical expression for calculation QoE metrics from QoS is not given as well. That is why this approach could be only inspiration to develop our own solutions, to design QoS/QoE ontology and rules, with potential usage of OpenMath for mathematical calculations. In [vuong] QoS Ontology for Web Services WS-QoSOnto is presented and core QoS properties of the QoS Ontology is shown in Fig. 83.

Fig. 83 Core QoS properties [vuong]
In [vuong] requirements for QoS Ontology are also given. These requirements correspond with questions how to model QoS properties and how to support the usage of QoS properties. Summary comparison of meeting these requirements by several existing QoS ontologies is presented. Aware of QoS ontology requirements can be helpful in creating, designing and implementing QoS and QoE ontologies.

In [papa] QoS Ontology language for web-services is proposed. This QoS ontology language provides a standard generic model for QoS attributes, by defined the nature of relationships between QoS attributes and the way they are measured. In this QoS ontology describes every QoS attribute by the same set of classes.

The classes defined QoS attributes:

- QoSParameter
- Metric
- QoSImpact
- Type
- Nature
- Aggregated
- Node
- Relationship.

In [papa] descriptions of these classes are presented, which could be useful in defining QoS parameters in ontology.

Table 35 Comparison of QoS/QoE ontology approaches

<table>
<thead>
<tr>
<th>No.</th>
<th>Approach</th>
<th>Described domain</th>
<th>Usage for DaVinci ontology</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>Agent Based Platform to Map Quality of Service to Experience</td>
<td>Describes platform for measuring QoS and QoE parameters.</td>
<td>However this approach assumes user feedback that is why it couldn’t be directly apply to DaVinci project.</td>
</tr>
<tr>
<td>2.</td>
<td>An ontology for the Quality of Experience Framework</td>
<td>Concepts and actions in QoE ontology are presented.</td>
<td>QoE ontology concepts and actions are not described in details, so we can derive from it only examples of relevant concepts in QoS/QoE domain.</td>
</tr>
<tr>
<td>3.</td>
<td>QoSOnt: a QoS Ontology for Service-Centric System</td>
<td>Modular ontology model, for describe QoS domain, for future reuse by different users.</td>
<td>We developed our own QoS/QoE ontology, we considered it more useful for DaVinci scope that existing QoSont ontology.</td>
</tr>
<tr>
<td>4.</td>
<td>A Framework for the Automatic Calculation of Quality of Experience in Telematic Services</td>
<td>Framework for measuring QoE, using SWRL rules and OpenMath calculations.</td>
<td>Approach is not described in details, and any example of rule is not given, thus it could be only inspiration for creation our own solutions.</td>
</tr>
<tr>
<td>5.</td>
<td>QoS Ontology for Web Services WS-QoSOnto</td>
<td>QoS Ontology with QoS properties is presented.</td>
<td>This ontology does not include all properties and parameters significant from DaVinci point of view.</td>
</tr>
<tr>
<td>6.</td>
<td>QoS Ontology language for web-services</td>
<td>QoS Ontology language with classes, which should describe QoS attributes.</td>
<td>This approach does not reflect directly DaVinci aims and we developed our own ontology with QoS attributes.</td>
</tr>
</tbody>
</table>

Presented ontology approach is not directly applicable for DaVinci scope. Additionally small amount of details cause, that authors developed own ontology for describing QoS and QoE parameters. However surveyed ontologies gave us overview on QoS and QoE domains. Comparison of QoS/QoE ontology approaches described above is shown in Table 35.

6.2 Machine readable formalisms

Machine readable formalisms used to develop QoE ontology for DaVinci project are presented in this section. This section includes ontology language, rule language and platform for ontology creation.
6.2.1 OWL

Comparison of different ontology languages can be found in [gome]. To create QoE DaVinci ontology OWL language has been used. The OWL (Web Ontology Language) is a language intended to define semantic documents in world wide web system. It is based on languages like RDF (Resource Description Framework), DAML (DARPA Agent Markup Language) or DAML+OIL (DAML+Ontology Inference Layer). It was standardized by W3C consortium (World Wide Web Consortium) and both with RDF are recommended to developing semantic web. OWL is a language of knowledge representation, enable ontology creation, and OWL ontologies were usually serialized by using RDF/XML syntax.

OWL can be divided into three sub-languages, with reference to their complexity:
- OWL Lite – with the simplest syntax, enables taxonomy of concepts creation, defining these concepts by simple restrictions and class hierarchy.
- OWL DL – extension of OWL Lite, ensures developing complex structures of concepts with several restrictions, by compatibility with reasoning systems, rule engines supports reasoning about ontology concepts instances.
- OWL Full – complete language, without reasoning capability, and without restrictions given for ontology relations.

OWL Lite and DL are based on Description Logics and OWL Full has been developed for sustain some features and compatibility with RDF schema. Creating ontology, we should decide which type of OWL language we prefer. Choosing OWL DL or Lite can be determined with respect to if simple construction of OWL Lite will be enough. Comparing OWL DL and Full, we should define if the reasoning support is needed. QoE DaVinci ontology has been created in OWL DL language to provide reasoning features.

OWL is a language of formal ontology description. It contains classes (domain concepts), objects (instances, individuals of classes), and relations (properties of concepts, or relationships between classes and objects). Through ontology, developed with all those components, OWL semantic specifies logical consequences, facts not literally present in the ontology, but caused by the semantics [owl].

6.2.2 RDF

RDF is a common representation of information contained in web resources, using XML syntax. RDF has been developed by W3C consortium, and both with OWL are recommended for semantic web creation. RDF ensures information representation about web resources and systems, which used them. RDF is intended to enable interoperability between applications or automated processing of Web information and software agents. RDF provides simple data models, formal semantics and reasoning. RDF model is a statement about resources in a form of RDF triples. It has three types of objects:
- subject – resource, anything described by statement
- predicate - attribute, feature or relation, which describe a resource;
- object – (literal) – the property value, which can be resource or literal.

RDF/XML syntax is used to serialized OWL ontologies. RDF schema is RDF’s vocabulary description language, which provides mechanisms for describing groups of related resources and relationships between them. RDFS defines properties and classes of RDF resources, and a semantics for hierarchy of them. OWL uses RDF Schema. Examples of RDF Schema features used in OWL are: Class, Property, subClassOf, Property domain and range or Individual [rdf].

6.2.3 SWRL

SWRL is a proposition for semantic web for integration rules with OWL. SWRL enables cooperation with several different rule engines and cooperation with defined by user methods libraries, which can be used to create new rules. SWRL has built-in libraries, which include sets of strings, mathematical operators and temporary operators. These operators can be used to effective change SWRL rules in Query language. Query language provides simple, but effective ways for gain information from OWL ontology. SWRL rules are in a form of IF …THEN statement (IF condition THEN instruction). It is an implication between an antecedent (body) and consequent (head). Rule constructed in this way can be written as: every time when conditions defined in the antecedent are met, then the conditions defined in the consequent must also be met [swrl].

Head and body of the rule, are parts which analyze elements in knowledge base, using classes and properties defined in OWL ontologies. Elements of body can be considered as queries to knowledge base and head as a consequential new modification of knowledge base.
SWRL standard provides only semantic language. To proceed rules usage of rule engine is required. User, designer should choose rule engine, implement SWRL rules and built-in functions of SWRL and execute semantic in accordance with these rules [wang2].

Popular rule engine is Jess. To integrate rule engine Jess with Protege-OWL SWRL Bridge is used. Bridge ensures mechanism to import SWRL rules and classes, instances, properties and description from OWL ontology, load this knowledge to rule engine or semantic reasoner, then reasoning is proceed, new knowledge is returned to the bridge and introduced to OWL ontology [wang2].

Example of rule in SWRL language:
Vulnerabilities(?x) ^ Network_Interconnection(?y) ^ hasResources(?y, ?z) ^ related(?x, ?y) ^ hasVulnerabilities(?y, ?x) → Vulnerability_expose_Resources (?x, ?z)

6.2.4 Protégé

Protégé is a free, open-source platform, which provides a suite of tools to construct domain models and knowledge-based applications with ontologies. Protégé implements a rich set of knowledge-modeling structures and actions that support the creation, visualization, and manipulation of ontologies in various representation formats [prot]. These formats are among all [knub]:
- Relational Database (ODBC - Open DataBase Connectivity),
- CLIPS,
- UML / XMI,
- XML / XML Schema,
- RDF,
- Topic Maps,
- DAML+OIL,
- OWL.

Protégé can be extended by plug-ins and a Java-based Application Programming Interface (API) for building knowledge-based tools and applications. Protégé is a set of tool for developing and usage of ontology through:
- Intuitive and easy in use Graphical User Interface
- Scalability – Protégé can be used with high amount of data
- Expendability of plug-ins architecture – Protégé can be extended by these plug-ins provides new functionalities, like for e.g. visualization.

Protégé supports two main methods of ontology modeling:
- Protégé –Frames editor – enables building and populating ontologies based of frames, in accordance with OKBC protocol (Open Knowledge Base Connectivity).
- Protégé-OWL editor – enable building ontology for semantic web, in standardized language OWL (Web Ontology Language).

In this model ontology consists of:
- Set of classes, organized in hierarchy of concepts to represent main, relevant domain concepts;
- Set of properties related to classes, to describe their attributes, relationships;
- Set of instances of classes – individuals, typical examples of concepts, which have values for their properties.
- Protégé-OWL editor – enable building ontology for semantic web, in standardized language OWL (Web Ontology Language).

In this model OWL ontology also consists of classes, properties and individuals. In addition the OWL formal semantics defines how to derive its logical consequences, facts not literally present in ontology, but entailed by the semantic.

6.3 QoE ontology for DaVinci

Described in 6.1 QoS/QoE ontologies were an inspiration during creation and development QoE DaVinci ontology. The most interesting approach seems to be presented in [sanch]. SWRL rules could be good solution for improving, make more automatic data processing. Usage of OpenMath should be considered if needed to mapping QoS to QoE parameters. Unfortunately many of presented ontologies, frameworks, approaches are too general. The authors do not described them in details, which causes that
they do not give finished/ready solutions to reuse. Some of them are also not applicable within Da Vinci environment and are beyond the Da Vinci scope.
That is why authors designed dedicated QoS/QoE Da Vinci ontology with put special emphasis on WiMax network and its parameters and configuration modes.

Ontology is a kind of knowledge database, a model of some domain. Ontology includes:
- Classes – concepts of described knowledge domain;
- Properties – attributes of classes, and relationships between them;
- Individuals – instances, basic objects of classes, concepts.
Although ontologies reflect designers point of view, there are defined steps, which are very helpful in ontology creation process [noy]. During ontology development the following steps should be considered:
1. What is domain and scope of ontology
2. Ontologies are designed to share knowledge and reuse of it that is why desktop research of QoS/QoE ontology has been done.
3. Listed the most important concepts and condition
4. Define classes and classes hierarchy
5. Define class properties
6. Define types of properties
7. Define instances

These 7 rules make ontology more useful, unique and more clear. QoE ontology is a part of overall QoS ontology. Some of the concepts are QoE metrics. QoE metrics are similar to QoS metrics in some way. Thus in the ontology we classify them into the same general concept: Metric. The classification of this class is:

Metric:
- Simple Metric
  - Which includes QoS Parameters:
    - QoS Parameter_Delay
    - QoS Parameter_Jitter
    - QoS Parameter_Loss
    - QoS Parameter_Reliability
    - QoS Parameter_Througput
- Standard Metric
  - Which includes classes of service CoS of different network:
    - CoS_WiMax
      - WiMax_BE (Best Effort)
      - WiMax_ertPS (Extended Real-Time Polling Service)
      - WiMax_nrtPS (Non-real-time Polling Service)
      - WiMax_rtPS (Real-time Polling Service)
      - WiMax_UGS (Unsolicited Grant Service)
    - CoS_e2e
    - CoS_Ethernet
    - CoS_UMTS
    - CoS_WLAN
- Subjective Metric
  - Which includes QoE metrics:
    - E-model with R factor
    - MoS scale:
      - MoS_Best
      - MoS_High
      - MoS_Low
      - MoS_Medium
      - MoS_Poor

We focus here only on description of classes related to QoE and QoS. These classes have its properties, which describe some particular features of them. The hierarchy of Metric class is presented in Fig. 84.
One of possible measured QoE metric is R factor, reflected MoS scale – Mean Opinion Score. It could be measured by e-model, but before we get value of R factor, particular instances of WiMax Network are needed. Instances of WiMax Network are basic objects of overall WiMax Network class in ontology. This means that individual examples of WiMax Network configuration, with a set of parameters will be given. These instances should have parameters characteristic only for WiMax networks to show differences between particular network settings and QoE metrics. Different instances should have different values of parameters. For this reason we defined properties of WiMax network class as follows:

- Network has CoS support
- Network has ARQ mechanism
  - ARQ
  - HARQ I
  - HARQ II
- Network has channel
  - AWGN
  - Non linear channel
  - Mobile channel
  - v=30 km/h, v=90 km/h, v=120 km/h
- Network has coding
  - Convolution coding
  - Turbo codes
    - Block turbo codes
  - LDPC codes
    - Da Vinci codes
      - Coding has rate (1/2, 2/3, ¾)
      - Coding has codeword (N=96, N=192, N=384)
- Network has modulation
  - QPSK
  - 16QAM
  - 64QAM
- Network has QoE MoS
- Network has scheduler
- Network has Admission control strategy
Classes of these network attributes are shown in Fig. 85 and Fig. 86.

![Diagram](image1)

**Fig. 85 Logical Component class of QoE DaVinci ontology**

![Diagram](image2)

**Fig. 86 Coding class of QoE DaVinci ontology**

Having defined particular parameters values we can measure R factor by e-model. This R factor should reflect MOS scale. To relate instances of network with MOS scale value for them, SWRL rule was created:

\[
p2:\text{Network}(?x) \land \text{Network\_has\_QoE\_R\_factor}(?x, ?y) \land \text{R\_factor\_to\_MOS\_scale}(?y, ?z) \rightarrow \text{Network\_has\_QoE\_MOS}(?x, ?z)
\]

The rule could be interpreted as follows:

*when for instance of Network class - x, with individual settings, R factor is y, and factor R is mapped on MoS scale as a value z, then Network x has QoE MoS value z.*

Rules can be used for automatic reclassification of some parameters and their values. This means that by define only a few values, we can get a set of different values of parameters, which are related to each other. Through the rule definition we can improve data processing and reasoning.

### 6.4 Applied ontology and E-model

Using ITU-T E-model, the QoE level could be measured with various settings of WiMAX network instances. Such QoE level could be determined/specified by measuring R factor and MoS scale. Result of this could be base for creating SWRL rules, which will define how QoE assumed level is changing according to particular values of WiMax network instances parameters. Real time subjective
tests (Fig. 87) could be performed to gain user feedback about subjective QoE parameters. This could in turn be used for creation of rules in ontology. These rules could show to what extent the measured QoE level (measured by e-model) reflect subjective QoE as experienced by real user.

For example it could be necessary to perform subjective test to verify the results from Section 3.6. However no subjective test results were conducted towards performance evaluation of VoIP in WiMAX with DaVinci codes due to unavailability of its hardware realization. It is why authors suggest utilizing DSP board as a straightforward hardware testbed for expressing voice quality degradation (audible) experience as a function of the R-factor. In particular authors enable to show how the changes of the R-factor influence the transmitted voice signal. The authors are using a DSK6455 platform (Fig. 88) and predefined libraries in Simulink which include the necessary blocks for targeting DSP processors. A simple Simulink model developed by authors was deployed on the DSP processor. The idea of the model is to add noise to the voice signal received from the microphone. The noise can be added in accordance to the characteristics of the plots presented in the section 3.6.

In addition the noise is added in such a way that it corresponds to the equivalent MOS scale. We take our results obtained in Section 3.6 and apply rules according to Fig. 89. The idea of the model is to answer the following question: What was the user experience with the signal if he rated it 3 (in the MOS scale)?
In particular, the model implemented on DSK6455 allows us to hear the voice degradation respectively to the measured MOS value. The MOS values can be changed by changing the position of switches depicted in Fig. 88.

### 6.5 Conclusions for DaVinci

Existing approaches to ontology-based description of QoS and QoE domains were presented in this section. Lack of publicly accessible QoE ontologies makes it impossible to be used in QoE ontology for DaVinci project. Moreover most of them are different from the original scope of the DaVinci project. The proposed QoE ontology for DaVinci project was presented in this section includes objective QoS metrics as well as subjective user QoE with reference to MOS scale. Individual WiMax network instances with parameters characteristic only for WiMax networks are presented. These instances with different settings would be used to assess QoE level for customized WiMAX network. An interesting alternative of evaluating DaVinci (QoE) results using DSP was presented and described in paper [ict2010].
7. Conclusions and Prospects

In this deliverable we have evaluated the cell level performance of DaVinci codes using the two key dimensions: admission control performance indicators and QoE assessment. We have first studied and eventually implemented WiMAX simulation environment suitable to execute various admission control algorithms (namely VIMACCS ns2 patch). The simulation platform was selected as a tradeoff between DaVinci requirements and computational complexity of the simulator. The results show that depending on the admission control solution implemented DaVinci codes outperform other codes (e.g. RS-CC) with respect to system capacity (effective throughput is up to 20% higher) while providing lower dropping and blocking probabilities. Bandwidth utilization is usually similar for a given admission control algorithm although is typically higher with longer codewords. The gain is higher in worst radio conditions, as the codes are more robust (assuming DaVinci ACM thresholds and realistic user mobility patterns). Careful selection of proper admission control guarantees yet better results. The cell level capacity gains are less visible in cells with high SNR. In order to get more insights into WiMAX system with DaVinci codes next step would be to evaluate results gathered after applying HARQ and TCP within the simulation platform.

From the point of QoE assessment the gain is less visible although DaVinci codes provides more users with satisfactory QoE compared to other codes applied in the WiMAX system configured with the same parameter values. The QoE results can be visualized using the developed DSP solution that enables “replaying” the QoE results for given simulation conditions.
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Appendix A. Simulink Real Time Workshop (RTW)

In order to deploy Simulink model on DSP platform the Code Composer Studio need to be adapted. Code Composer Studio IDE offers robust, mature core functions with easy-to-use configuration and graphical visualization tools for faster system design. In case of MATLAB CCS is used as a middle layer between host(PC) and the device (DSP), allowing project deployment on embedded platform. As it is shown on Fig. 90 the default deployment path ends up with regular Win32 *.exe file. When CCS is configured to be used with MATLAB Simulink the process becomes slightly different since CCS is started up to compile the created simulation and fulfill the device’s requirements.

![Diagram of model deployment process](image)

**Fig. 90 Difference between regular Win32 and DSP application deployment.**

The one of the biggest advantages of using Matlab as layer between CSS and developer is that it allows the programmer the high level model-based project designing, which makes easy to express a design concept, automatically generate the code to deploy it on a hardware target, and verify exactly the same operation on silicon. Moreover it is possible to implement the design on a Texas Instruments DSP, and verify its on-target performance in real time, simply by adding Target Preferences Block from c6000 blockset library.

![Diagram of Target Preferences Block](image)

**Fig. 91 Adding Target Preferences Block from c6000 blockset library allows model deployment on DSP**

The model-based design programmer the high level model-based project designing, which makes and deployment. However the deployment process is not only dedicated for DSP, since there is also support for xPC, which may provide developer with additional information on system’s performance. After running application on the target machine, developer can quickly conduct multiple iterations to optimize the design in order to meet assumed performance requirements. Particularly Simulink provides profiling functionalities in Link for Code Composer Studio to identify the most computation-intensive
segments of the algorithm. For example shown in Fig. 92 the most time consuming was DECODER block, which used about 84% of simulation time.

### Simulink Profile Report: Summary

- Total recorded time: 5.19 s
- Number of Block Methods: 31
- Number of Internal Methods: 5
- Number of Nonvirtual Subsystem Methods: 4
- Clock precision: 0.00000005 s
- Clock Speed: 200 MHz

### Function List

<table>
<thead>
<tr>
<th>Name</th>
<th>Time</th>
<th>Call Time</th>
<th>Self Time</th>
<th>Location</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sim</td>
<td>5.187s</td>
<td>5.187s</td>
<td>0.00000005 s</td>
<td>denv64.mat</td>
</tr>
<tr>
<td>Model Execute</td>
<td>4.64075s</td>
<td>4.64075s</td>
<td>0.00000005 s</td>
<td>denv64.mati</td>
</tr>
<tr>
<td>qm64 matl (Output)</td>
<td>4.59867s</td>
<td>4.59867s</td>
<td>0.00000005 s</td>
<td>denv64.matl</td>
</tr>
<tr>
<td>Major Outputs</td>
<td>4.59867s</td>
<td>4.59867s</td>
<td>0.00000005 s</td>
<td>denv64.mati</td>
</tr>
<tr>
<td>qm64 matl/Subsystem2/nb-LDPC decoder (Output)</td>
<td>4.00025s</td>
<td>4.00025s</td>
<td>0.00000005 s</td>
<td>denv64.matl/Subsystem2/nb-LDPC decoder</td>
</tr>
</tbody>
</table>

### Fig. 92 Profiling report

According to analysis, developer may change the model parameters, apply more efficient algorithm or replace the block used in the model with target-optimized block.

In Table 36 there are proposed TI C6000™ DSP dedicated block that may be used to improve performance of the model. However the optimization of nb-LDPC codes was not the key issue and was out of project’s scope.

### Table 36 Optimization suggestions

<table>
<thead>
<tr>
<th>Block name</th>
<th>Suggested block increasing performance</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Encoder</td>
<td>Although the encoder block is not computational expensive, however the “matrix multiply” can be applied to speed up some of the for-loops in the encoder</td>
<td></td>
</tr>
<tr>
<td>Channel</td>
<td>The FFT and IFFT DSP blocks are extremely rapid, thus those block should be used to implement different channels (AWGN, Rayleigh)</td>
<td></td>
</tr>
</tbody>
</table>
Appendix B. Deployment of mex-function version of nb-LDPC

B.1 Making “nb-LDPC mex-functions” deployable

According to assumption that new versions of nb-LDPC code will be maintained as Matlab mex-functions, it was necessary to allow:

- easy model re-configuration
  To meet this goal global configuration script was created, which makes access to simulation parameters (such as LDPC-decoder properties, I/O vectors length, etc.) fast and comfortable.

- system modularity
  This functionality allows divide project in three parts (encoder, decoder and AWGN channel), which allows to spread model execution over several nodes (PC,DSP etc.). Furthermore modularity enables easily channel replacement.

- easy LDPC codes replacement
  It is important to provide solution that would allow easy replacement in order to avoid redundant work.

![Diagram of Simulink model conversion](image)

Fig. 93 During deployment Simulink model is converted to C/C++ code and then compiled to platform dependent binary file

The encoder and decoder mex-files are exactly implemented in ANSI-C code, therefore the aim of wrapper shown in Fig. 93 is to put appropriately the mex-code in the final output file, that will be finally compiled. The necessary information describing how to create the wrapper function is provided in appendix B.

B.2 The approach to adopt the C-based, standalone version of NB-LDPC.

The C-ANSI program simulating the non-binary LDPC is a standalone application which requires set of input arguments defining particular settings of encoder, decoder and AWGN channel properties. It runs in loop and returns the BER value for provided Eb/N0. Therefore to meet all requirements introduced to
mex version of nb-LDPC codes, it was necessary to modify the contributed application in order to adopt it to Simulink.

<table>
<thead>
<tr>
<th>Before adaptation</th>
<th>After adaptation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loop</td>
<td>Break loop and provide sender and receiver</td>
</tr>
<tr>
<td>Random number generator</td>
<td>Receiver (socket)</td>
</tr>
<tr>
<td>Encoder</td>
<td>Encoder</td>
</tr>
<tr>
<td>Channel</td>
<td>Channel</td>
</tr>
<tr>
<td>Decoder</td>
<td>Decoder</td>
</tr>
<tr>
<td>Sender (socket)</td>
<td></td>
</tr>
</tbody>
</table>

Fig. 94 Adaptation details

To adopt the ANSI-C application modules and make it able to work with Simulink it was required to break the main loop in the application and separate the code inside this loop in three independent modules (Encoder, Channel, and Decoder - Fig. 94). Then it was necessary to replace the random number generator with receiver (UDP socket) and to connect the output of decoder to sender socket (to enable communication with other elements of the integrated platform).

### B.3 S-Function Builder

The separated modules (encoder, channel, and decoder) were put in ANSI-C functions, which are named as follows:

1. `sim_nb_ldpc_encoder(input, encoded_info)`
2. `sim_nb_ldpc_channel(encoded_info, encoded_info_plus_noise)`
3. `sim_nb_ldpc_decoder(encoded_info_plus_noise, decoded_info)`

Thanks to the Simulink S-Function Builder it was possible to link the modified ANSI-C application (and newly created encoder, channel, and decoder functions) and create independent blocks as it is shown in Fig. 95

<table>
<thead>
<tr>
<th><code>sim_nb_ldpc_encoder()</code></th>
<th><code>sim_nb_ldpc_channel()</code></th>
<th><code>sim_nb_ldpc_decoder()</code></th>
</tr>
</thead>
</table>

Fig. 95 Separated channel, encoder, and decoder
B.4 Deployment of mex-function version of nb-LDP

Deployment process is shown in Figure B1 and consists of three important elements:

- Simulink model (the *.mdl file)
- Wrapper (the *.tcl file)
- Compiled binaries

Depending on TLC script and building configuration user can achieve totally different compiled binaries dedicated for particular platforms. Generated output for DSP platform looks like typical “Code Composer Studio” project with compiled and linked *.out file (application being uploaded on DSP platform). In case of standard (*.exe) generated binary whole compiled and linked project contains additionally all sources files (*.cpp, *.h, *.c) and make files, what makes it possible to recompile project without MATLAB environment.

“Target Language Compiler” allows developer to chose the target language the code will be generated. TLC compiler uses TLC wrappers (*.tlc files) to manage this process. GNU Make application uses the standard make-file to build sources generated by TCL compiler.

Depending on chosen platform different target binaries are built.

Fig. 96 Model deployment process

As is it mentioned in Figure B1 the TLC-compiler uses the *.tlc file to generate the source code. This is strictly required by MATLAB building process and plays the following functions:

- TLC works with the Simulink software to generate target specific code
- Thanks to those scripts the constants and global variables can be ported from m-file to ANSI-C format, providing single-point of application initialization
- Allows to include additional source code (C-file, header files)
- Allows to control building and linking process (linking shared libraries, object files)

The example explaining how to the TLC compiler and *.tcl files were used for DSP deployable code generation is shown in Fig. B2. The enter point is the Simulink model. It uses the globals_initialization.m
script to load all encoder, decoder, and channel settings. This file allows the whole simulation to be initialized and set up in one place.

The encoder, decoder, and channel blocks (that use mex-files) can not be simply linked with the final binary file, because those require input arguments (input vector, decoder/encoder settings, input/output dimensionalities) that are passed via TLC wrapper from `globals_initializatin.m` file, providing final C-functions, that can be simply compiled with C language compiler.

Fig. 97 DSP code deployment using mex version of nb-LDPC codes
Appendix C. E-model impairment values

The ITU-T Recommendation G.107 [g.107] provides the default parameters and their permitted range for the E-model. It is recommended for calibration cases to test the system with the parameters set to their default values.

Table 37 contains all the E-model related parameters and their corresponding values.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Abbr.</th>
<th>Unit</th>
<th>Default value</th>
<th>Permitted range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Send Loudness Rating</td>
<td>SLR</td>
<td>dB</td>
<td>+8</td>
<td>0 … +18</td>
</tr>
<tr>
<td>Receive Loudness rating</td>
<td>RLR</td>
<td>dB</td>
<td>+2</td>
<td>-5 … +14</td>
</tr>
<tr>
<td>Sidetone Masking Rating</td>
<td>STMR</td>
<td>dB</td>
<td>15</td>
<td>10 … 20</td>
</tr>
<tr>
<td>Listener Sidetone Rating</td>
<td>LSTR</td>
<td>dB</td>
<td>18</td>
<td>13 … 23</td>
</tr>
<tr>
<td>D-Value of Telephone, Send Side</td>
<td>Ds</td>
<td>-</td>
<td>3</td>
<td>-3 … +3</td>
</tr>
<tr>
<td>D-Value of Telephone, Receive Send</td>
<td>Dr</td>
<td>-</td>
<td>3</td>
<td>-3 … +3</td>
</tr>
<tr>
<td>Talker Echo Loudness Rating</td>
<td>TELR</td>
<td>dB</td>
<td>65</td>
<td>5 … 65</td>
</tr>
<tr>
<td>Weighted Echo Path Loss</td>
<td>WEPL</td>
<td>dB</td>
<td>110</td>
<td>5 … 110</td>
</tr>
<tr>
<td>Mean one-way Delay of the Echo Path</td>
<td>T</td>
<td>msec</td>
<td>0</td>
<td>0 … 500</td>
</tr>
<tr>
<td>Round Trip Delay in a 4-wire Loop</td>
<td>Tr</td>
<td>msec</td>
<td>0</td>
<td>0 … 1000</td>
</tr>
<tr>
<td>Absolute Delay in echo-free Connections</td>
<td>Ta</td>
<td>msec</td>
<td>0</td>
<td>0 … 500</td>
</tr>
<tr>
<td>Number of Quantization Distortion Units</td>
<td>qdu</td>
<td>-</td>
<td>1</td>
<td>1 … 14</td>
</tr>
<tr>
<td>Equipment Impairment Factor</td>
<td>Ie</td>
<td>-</td>
<td>0</td>
<td>0 … 40</td>
</tr>
<tr>
<td>Circuit Noise referred to 0dBref-point</td>
<td>Nc</td>
<td>dBm0p</td>
<td>-70</td>
<td>-80 … -40</td>
</tr>
<tr>
<td>Noise Floor at Receive Side</td>
<td>Nfor</td>
<td>dBm0p</td>
<td>-64</td>
<td>-</td>
</tr>
<tr>
<td>Room Noise at Send Side</td>
<td>Ps</td>
<td>dB(A)</td>
<td>35</td>
<td>35 … 85</td>
</tr>
<tr>
<td>Room Noise at the Receive Side</td>
<td>Pr</td>
<td>dB(A)</td>
<td>35</td>
<td>35 … 85</td>
</tr>
<tr>
<td>Advantage Factor</td>
<td>A</td>
<td>-</td>
<td>0</td>
<td>0 … 20</td>
</tr>
</tbody>
</table>