SEVENTH FRAMEWORK PROGRAMME

Remote Collaborative Real-Time Multimedia Experience over the Future Internet

ROMEO

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D6.5

Second report on server, peer, user terminal, security and content registration modules development

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1 INTRODUCTION

1.1 Purpose of the document

This document aims to provide the development status of the ROMEO project for both server and peer components. While this document is used to check whether the development of components is aligned with the timeline presented in D6.3, it also provides a good summary on the development efforts for the whole ROMEO system.

1.2 Scope of the work

The scope of this document is to describe the development made so far for each module. The detailed description of the developed components, APIs to be used by the developed components as well as the common APIs to be shared by all components are provided in this document.

1.3 Structure of the document

The document has three main sections:

- Server Components
- Peer Components
- Network Infrastructure Components

Each section includes detailed information on the module development status.

2 SERVER COMPONENTS

2.1 Content Generation

2.1.1 Content Capturing

Three sets of raw audio and video contents corresponding to three capturing sessions have been delivered between the 8th and the 16th month of the project.

At the end of the shooting sessions, two kinds of content were available:

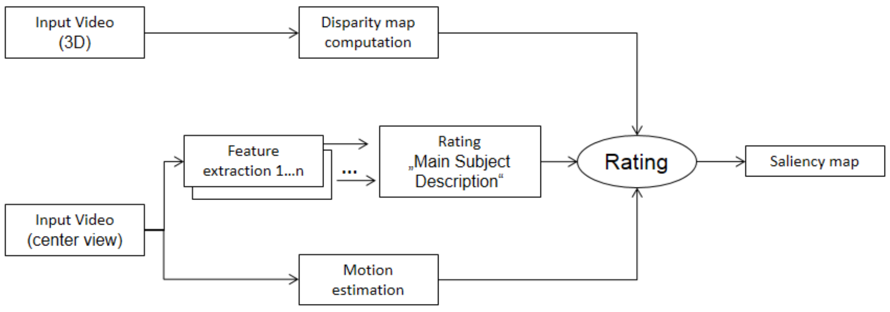
- Raw content generated from the professional equipments for the next processing stages
- Raw content generated from the user terminals (UGC - User Generated Content)

Before delivering this content, the raw data has been post-processed and tested/validated thanks to the first versions of the rendering modules. The acquisition and post-processing modules are specified in D3.1. After post-processing, the audio and video contents are delivered in raw data format (uncompressed).

2.1.2 Visual Attention Modelling

The Visual Attention Modelling module is an offline video processing tool that calculates saliency maps for 2D and 3D video sequences. An overview of this module is shown in Figure 1.

The development includes the module itself as well as an evaluation that compares the module's outcome with user test results generated by an eye tracker test. Implementation is done in C++ and based on the open source library openCV (openCV.org). The module is configured via a script file. In there filter parameters and the filter chain can be set. The implementation of the module is completed and a detailed description was reported in Deliverable D3.5.



Input Video (3D)
Disparity map computation
Feature extraction 1...n
...
Rating „Main Subject Description"
Rating
Saliency map
Input Video (center view)
Motion estimation

Figure 1 - Overview of the Visual Attention Modelling module

2.1.3 Media Encoding and TS Generation

2.1.3.1 Video Encoding

Compression of raw 3D video as a result of the content generation block is an offline process within ROMEO server architecture, which is not an integral of the live demonstrator. The output of this block is Annex G (SVC) of MPEG-4/H.264 AVC complying bit-streams. The modified reference MPEG scalable video codec (Joint Scalable Video Model – JSVM) v9.19 is used for encoding the descriptions independently. Independent encoding ensures that the decoding of descriptions can also be carried out independently. JSVM is based on a set of libraries written in C++, such as H264AVCEncoderLibStatic (implementation of motion estimation and entropy coding), H264AVCCommonLibStatic (implementation of data structures used in the encoder), H264AVCVideoIoLibStatic (implementation of input/output operations within the encoding process).). Interested readers can download the open software project JSVM using a CVS client following the instructions given in [19]. Video encoding is preceded by a multiple description generation step that produces the descriptions to be subsequently encoded. Some of the different description generation schemes had been compared against each other in Deliverable 4.2 (Report on streaming/broadcast techniques for 3D multi-view video and spatial audio) in terms of R-D performance as well as error resilience capability, in the context of stereoscopic viewing. Side-by-Side and Top-Down description generation mechanisms have shown very close performance. Figure 2 shows the process of description generation and scalable encoding.

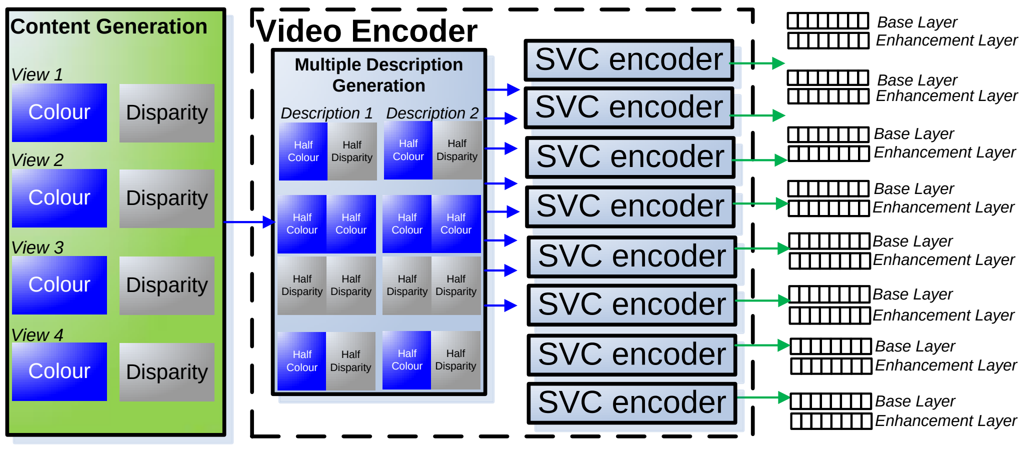
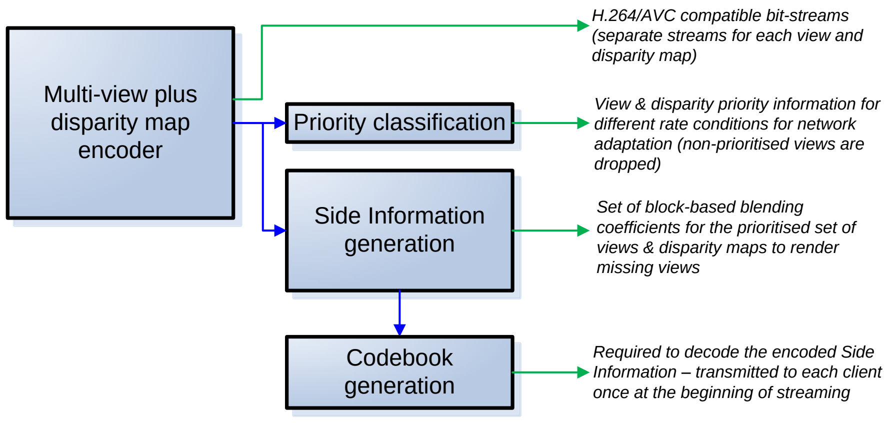


Figure 2 - Video Encoder architecture deployed in ROMEO server, to be used in the demonstrator

Sixteen descriptions in total are generated, one of which (base layer of the first description of the stereoscopic pair – views 2 & 3) is sent through DVB after encapsulating into MPEG-2 TS (described in Figure 2).

The reference SVC encoder has been modified to read an additional set of configuration parameters as well as a gray-scale map, which represents the macroblock-based saliency information (based on the low-level features within the scope of the visual attention research carried out in WP3) for regionally adaptive bit-rate assignment. The additional configuration parameters include the Quantisation Parameter (QP) offset to be applied on each defined saliency region. More details are provided in Deliverable 4.3 (Report on the developed audio and video codecs).

The Side-Information (SI) assisted multi-view video coding and streaming is not considered as an integral part of the real-time demonstrator, the performance of which is assessed separately outside of the multi-tree P2P multicast distribution system. More details about the architecture of the SI assisted multi-view encoder are provided in Deliverable 4.3. Figure 3 briefly outlines the breakdown of the SI-based encoder’s output.



Multi-view plus disparity map encoder
Priority classification
Side Information generation
Codebook generation
H.264/AVC compatible bit-streams (separate streams for each view and disparity map)
View & disparity priority information for different rate conditions for network adaptation (non-prioritised views are dropped)
Set of block-based blending coefficients for the prioritised set of views & disparity maps to render missing views
Required to decode the encoded Side Information – transmitted to each client once at the beginning of streaming

Figure 3 - Brief overview of the SI-based encoder structureAudio Encoding

2.1.3.2 Audio Encoding

The audio encoding module is in working order, as reported previously in D6.2. In channel-based approach, the offline encoding module accepts the multichannel audio of the captured contents as the input and produces the AAC (Advanced Audio Coding)-encoded multichannel audio in ADTS (Audio Data Transport Stream) format. For the object-based approach, it accepts each audio object as a mono input, and produces the encoded output which is to be interleaved with other objects as the Elementary Stream (ES) for transmission with the video, scene description and other relevant information. A software application has been developed which can generate the ES either as the encoded object(s) or packetised as PES (Packetised Elementary Stream). In the case of multiple objects, single frames from each object are extracted and interleaved to form the PES payload. The PES header information is attached in front of the payload. The PTS (Presentatin Time Stamp) value to be used for the synchronisation can be set as the input and is inserted accordingly in the header data, increasing with the frame count. This PES can be used for the TS encapsulation.

On the other hand, a GUI-based real-time audio capturing/encoding application has been developed. Instead of an audio file, multichannel audio stream through a microphone array is used as the input. The frames are encoded in real-time and can be streamed via TCP or stored as a 5.1-channel AAC file. Further development has also been made in the Analysis-by-Synthesis (AbS) coding technique, separately from the implementation of the practical encoding module towards the integration and demonstration. The details of its operation are described in D4.3. It is a new SAC (Spatial Audio Coding) technique based on the principle of closed-loop system. An algorithm for selecting sub-optimal signals and parameters through iterations has also been integrated, which offers complexity scalability, providing a trade-off between the complexity of the encoder and the quality of the reconstructed audio signals. The experimental results, for encoding five-channel audio signals at bit-rates ranging from 40 to 96 kb/s per audio channel using two reasonable complexity levels of the encoder, demonstrate that significant improvement of performances is achieved compared with the conventional open-loop techniques. At the lowest complexity level, the encoder is capable of working in a real-time implementation, although at a higher complexity level for enhanced quality, the computation time increases approximately 72 times.

2.1.3.3 TS Generation

2.1.3.3.1 Structure of the MPEG2 Transport Stream in ROMEO

The generation of MPEG2 Transport Streams (TS) is based on the scheme of camera views, video descriptions and layers as described in Deliverable D4.1 [7]. For the purpose of the ROMEO project, all views, descriptions and layers plus all audio signals of a certain test sequence are combined into one comprehensive TS.

One part of this TS is broadcast over the DVB transmitter, the other part can be accessed by each peer via the servers which form part of the ROMEO network structure. The comprehensive TS contains all necessary tables of program specific information and service information (PSI/SI) that enable a DVB receiver to identify and then decode the required video and audio streams.

The TS generation takes the Elementary Streams (ES) as input. These ES were encoded by ROMEO partner University of Surrey and each contains two layers of one of several view – description combinations.

The PSI/SI data are organised in those DVB-standard-compliant tables which are mandatory for a TS that is to be distributed over a DVB network [8]. These tables are

- PAT Program Association Table
- NIT Network Information Table
- SDT Service Description Table
- EIT Event Information Table
- TDT Time and Date Table
- PMT Program Map Table
- RST Running Status Table

For all test sequences that are used in ROMEO, the various parts of the TS are ordered into the same scheme of Packet IDentifiers (PID) as depicted in *Figure 4*.

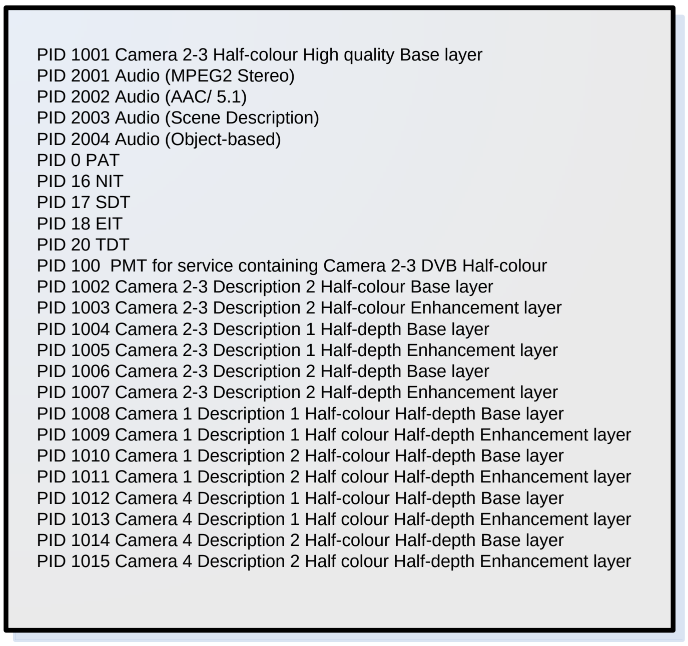

PID 1001 Camera 2-3 Half-colour High quality Base layer
PID 2001 Audio (MPEG2 Stereo)
PID 2002 Audio (AAC/ 5.1)
PID 2003 Audio (Scene Description)
PID 2004 Audio (Object-based)
PID 0 PAT
PID 16 NIT
PID 17 SDT
PID 18 EIT
PID 20 TDT
PID 100 PMT for service containing Camera 2-3 DVB Half-colour
PID 1002 Camera 2-3 Description 2 Half-colour Base layer
PID 1003 Camera 2-3 Description 2 Half-colour Enhancement layer
PID 1004 Camera 2-3 Description 1 Half-depth Base layer
PID 1005 Camera 2-3 Description 1 Half-depth Enhancement layer
PID 1006 Camera 2-3 Description 2 Half-depth Base layer
PID 1007 Camera 2-3 Description 2 Half-depth Enhancement layer
PID 1008 Camera 1 Description 1 Half-colour Half-depth Base layer
PID 1009 Camera 1 Description 1 Half colour Half-depth Enhancement layer
PID 1010 Camera 1 Description 2 Half-colour Half-depth Base layer
PID 1011 Camera 1 Description 2 Half colour Half-depth Enhancement layer
PID 1012 Camera 4 Description 1 Half-colour Half-depth Base layer
PID 1013 Camera 4 Description 1 Half colour Half-depth Enhancement layer
PID 1014 Camera 4 Description 2 Half-colour Half-depth Base layer
PID 1015 Camera 4 Description 2 Half colour Half-depth Enhancement layer

Figure 4 - General structure of the Transport Streams used in ROMEO

Depending on the production material, not all PIDs may be integrated into the comprehensive TS but a minimum set of PIDs should include the video, audio and tables for the DVB part of the TS, i.e. PID 1001, PID 2001, PID 0, PID 16, PID 17, PID 18, PID 20 and PID 100.

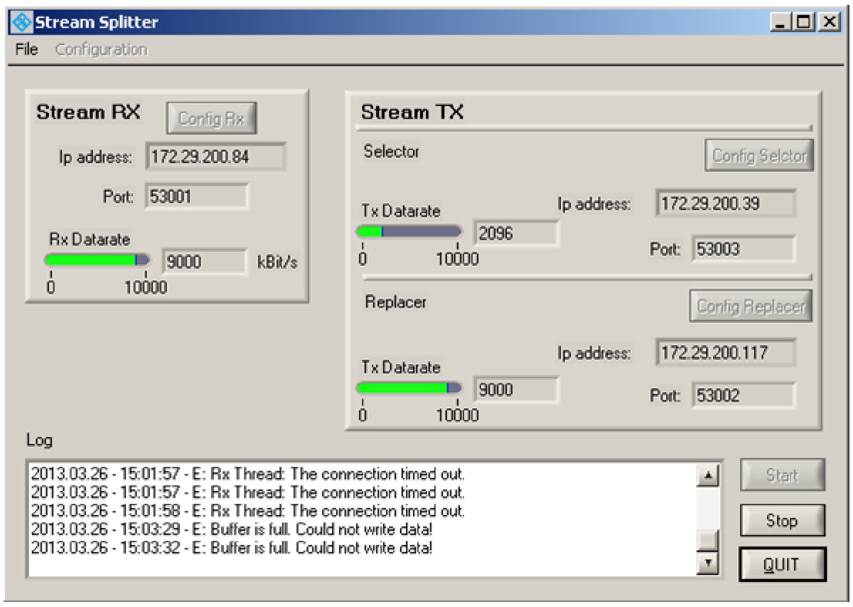
2.1.3.3.2 Transport Stream generation tool

The TS generation is worked on by ROMEO partner R&S and is based on a tool for which the necessary modifications were developed as part of WP6 in the ROMEO project.

Working with the Elementary Streams as input, the first steps concern the modifications required for compliance with DVB standards.

- The ES that contains the video part that is to be distributed over the DVB system is modified to maintain compliance with the H.264/ AVC standard [13]. Therefore the SVC header information [14] SSPS (Subset Sequence Parameter Set) and PPS (Picture Parameter Set) is removed.
- To generate a seemingly endless TS, a short sequence is processed in such a way that it plays out repeatedly without causing a loss of synchronisation or similar in the receiver. As a suitable length for a video / audio sequence for test purposes a sequence length of approx. one minute was identified. This one-minute-sequence can then be looped into 10 or 15 minute test session material.
- Since video frames and audio frames have different standard length (Video frame length: 40 msec, Audio frame length: 21.333 msec), a suitable duration that accommodates a certain number of complete video and audio frames is 59.84 sec.
- Subsequently, the Elementary Streams are cut for seamless looping containing 1496 video frames (25Hz) and 2805 AAC audio frames which gives an equal length of 59.840 seconds.
- The synchronisation between video and audio in the processed sequence is checked and if necessary, corrected to obtain lip-sync.

For test purposes, the comprehensive TS can be split into two parts. The first part is the input signal for the DVB-T/T2 modulator, the second part contains up to 14 video-related PIDs of the various views, descriptions and layers for the P2P (Peer-to-Peer) server. The tool developed for this task is named ‘Stream Splitter’ and can be operated by a graphical user interface as shown in Figure 5.



Stream RX

Ip address: 172.29.200.84

Port: 53001

Rx Datarate 9000 kBit/s (0 – 10000)

Stream TX

Selector

Tx Datarate 2096 (0 – 10000)

Ip address: 172.29.200.39

Port: 53003

Replacer

Tx Datarate 9000 (0 – 10000)

Ip address: 172.29.200.117

Port: 53002

Log

```
2013.03.26 - 15:01:57 - E: Rx Thread: The connection timed out.
2013.03.26 - 15:01:57 - E: Rx Thread: The connection timed out.
2013.03.26 - 15:01:58 - E: Rx Thread: The connection timed out.
2013.03.26 - 15:03:29 - E: Buffer is full. Could not write data!
2013.03.26 - 15:03:32 - E: Buffer is full. Could not write data!
```

Figure 5 - Stream Splitter GUI

2.1.3.3.3 New descriptor “Additional Content URL Descriptor”

The Transport Streams used for tests in ROMEO need to contain the necessary information so that the user terminal can find and identify available additional video views, descriptions, layers, as well as additional audio information. This information and the additional streams are available from the ROMEO project server where the user terminal can find the details under a specific URL. This URL is included in a new descriptor in the PSI/SI of the Transport Stream that is broadcasted over the DVB network. Since the assumption is that the DVB signal can be received by each user terminal, the URL is therefore known by each terminal.

The descriptor notifies the terminal that additional content (views) is available over the IP (P2P) network and further information can be retrieved under the URL from the respective internet page. The following parameters have to be included:

- Original Network ID
- Transport Stream ID
- Service ID
- possibly Event ID

The descriptor is located inside the DVB-related PSI/SI data:

- permanently in the SDT (Service Description Table)

or

- temporarily as an event description in the EIT (Event Information Table).

The syntax of the “Additional Content URL Descriptor” is defined following other descriptor definitions in DVB. Its details are given in Table 1.

Syntax	Bits	Identifier
additional_content_URL_descriptor(){		
descriptor_tag	8	uimsbf
descriptor_length	8	uimsbf
descriptor_tag_extension	8	uimsbf
reserved_future_use	8	uimsbf
for (i=0;i<N;i++){		
additional_content_id	16	uimsbf
URL_base_length	8	uimsbf
for (j=0;j<URL_base_length;j++){		
URL_base_byte	8	uimsbf
}		
URL_extension_count	8	uimsbf
for (k=0;k< URL_extension_count;k++){		
extension_id	16	uimsbf
URL_extension_length	8	uimsbf
for (l=0;l< URL_extension_length;l++){		
URL_extension_byte	8	uimsbf
}		
}		
}		

Table 1 - Syntax of the "Additional Content URL Descriptor"

Definitions:

descriptor_tag: The descriptor tag is an 8-bit field which identifies each descriptor. Those values with MPEG-2 normative meaning are described in the MPEG2 standard ISO/IEC 13818-1 [9]. The value of the descriptor_tag is **0x7F** for the **extension_descriptor()**.

descriptor_length: The descriptor length is an 8-bit field specifying the total number of bytes of the data portion of the descriptor following the byte defining the value of this field.

descriptor_tag_extension: The descriptor tag extension is an 8-bit field which identifies each extended descriptor. The value of the descriptor_tag_extension has to be defined by the DVB Working Group tm-gbs. (Proposal: 0x12).

reserved_future_use: When used in the clause defining the coded bit stream, it indicates that the value may be used in the future for ETSI defined extensions.

NOTE: Unless otherwise specified all "reserved_future_use" bits are set to "1".

additional_content_id: This 16-bit field uniquely identifies the additional available content to the service or event (Table 2).

additional_content_id	Description
0x0000	Reserved
0x0001	ROMEO content
0x0002 to 0x0FFF	Reserved for future use
0x1000 to 0xFFFF	User defined

Table 2 - Values for additonal_content_id

URL_base_length: This 8-bit field specifies the length in bytes of the following base part of the URL.

URL_base_byte: This is an 8-bit field. These bytes form the first part of a HTTP URL conforming to HTTP 1.0 (see RFC 1945 [10]), or the first part of an HTTPS URL conforming to RFC 2818 [11] or the first part of another URL conforming to RFC 3986 [12].

All bytes interpreted as text shall be encoded as UTF8, but shall not include the null character, e.g. "www.ROMEO.Service-Guide.tv**".**

URL_extension_count: This 8-bit field indicates the number of URL extensions conveyed by this descriptor.

URL_extension_id: This 8-bit field identifies a purpose (e.g. **ROMEO service number**) of the URL_extension.

URL_extension_length: This 8-bit field indicates the number of bytes in the extension part of the URL.

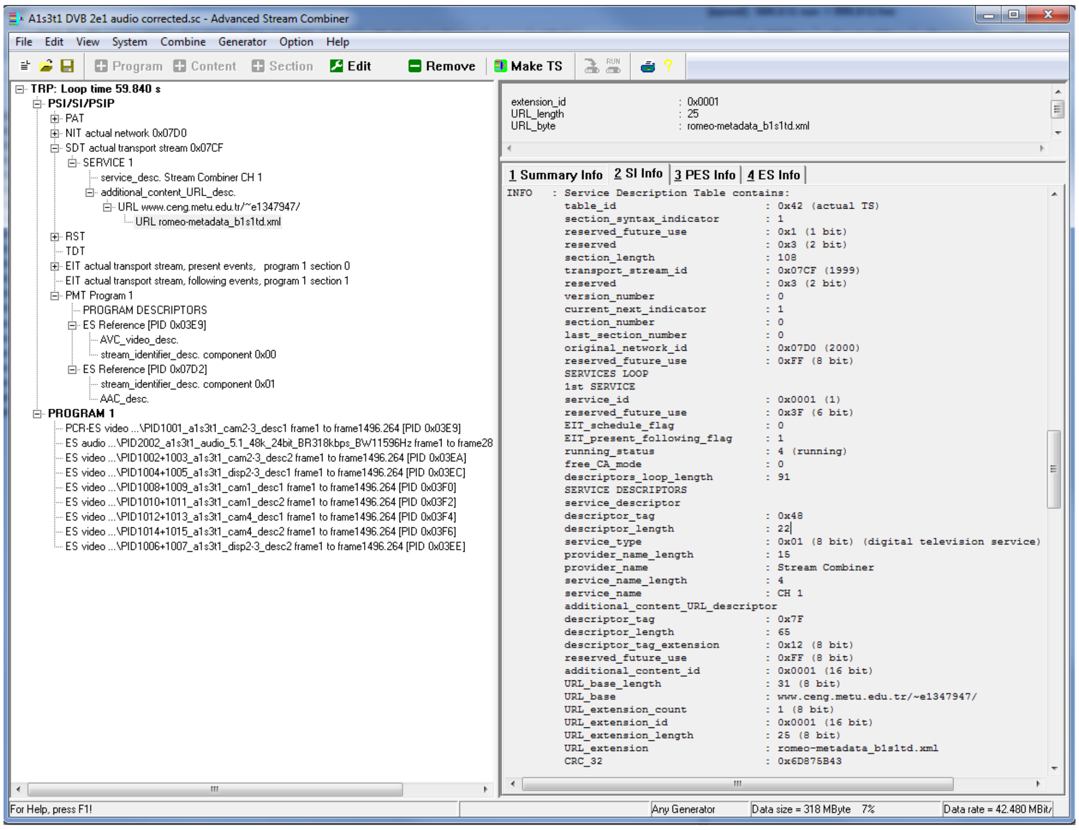
URL_extension_byte: These bytes form the extension part of an HTTP URL conforming to HTTP 1.0 (see RFC 1945 [10]), or the extension part of an HTTPS URL conforming to RFC 2818 [11] or otherwise a URL whose scheme is supported by a registered interaction channel transport service provider implementation.

All bytes interpreted as text shall be encoded as UTF8, but shall not include the null character.

URLs are formed by concatenating the URL extension with the preceding URL base. The URL so formed either identifies a file system directory or a specific XML file, e.g. "/romeo_esg.xml".

The Classes (modules) that generate the DVB additional_content_URL_descriptor for the ROMEO project were integrated with the existing version of the DVB service information generation module in the TS generation tool.

The user interface of the TS generation tool is depicted in Figure 6. On the left it shows the structure of the PSI/SI tables and the Elementary Streams to be processed. On the right it shows all details of the content of the respective component, in this case the location of the xml file that contains the ROMEO metadata.


A1s3t1 DVB 2e1 audio corrected.sc - Advanced Stream Combiner
File Edit View System Combine Generator Option Help
Program Content Section Edit Remove Make TS RUN
1 Summary Info
2 SI Info
3 PES Info
4 ES Info
For Help, press F1!
Any Generator
Data size = 318 MByte 7%
Data rate = 42.480 MBit/

```
TRP: Loop time 59.840 s
  PSI/SI/PSIP
    PAT
    NIT actual network 0x07D0
    SDT actual transport stream 0x07CF
      SERVICE 1
        service_desc. Stream Combiner CH 1
        additional_content_URL_desc.
          URL www.ceng.metu.edu.tr/~e1347947/
            URL romeo-metadata_b1s1td.xml
    RST
    TDT
    EIT actual transport stream, present events,   program 1 section 0
    EIT actual transport stream, following events, program 1 section 1
    PMT Program 1
      PROGRAM DESCRIPTORS
      ES Reference [PID 0x03E9]
        AVC_video_desc.
        stream_identifier_desc. component 0x00
      ES Reference [PID 0x07D2]
        stream_identifier_desc. component 0x01
        AAC_desc.
  PROGRAM 1
    PCR-ES video ...\PID1001_a1s3t1_cam2-3_desc1 frame1 to frame1496.264 [PID 0x03E9]
    ES audio ...\PID2002_a1s3t1_audio_5.1_48k_24bit_BR318kbps_BW11596Hz frame1 to frame28
    ES video ...\PID1002+1003_a1s3t1_cam2-3_desc2 frame1 to frame1496.264 [PID 0x03EA]
    ES video ...\PID1004+1005_a1s3t1_disp2-3_desc1 frame1 to frame1496.264 [PID 0x03EC]
    ES video ...\PID1008+1009_a1s3t1_cam1_desc1 frame1 to frame1496.264 [PID 0x03F0]
    ES video ...\PID1010+1011_a1s3t1_cam1_desc2 frame1 to frame1496.264 [PID 0x03F2]
    ES video ...\PID1012+1013_a1s3t1_cam4_desc1 frame1 to frame1496.264 [PID 0x03F4]
    ES video ...\PID1014+1015_a1s3t1_cam4_desc2 frame1 to frame1496.264 [PID 0x03F6]
    ES video ...\PID1006+1007_a1s3t1_disp2-3_desc2 frame1 to frame1496.264 [PID 0x03EE]
```

```
extension_id         : 0x0001
URL_length           : 25
URL_byte             : romeo-metadata_b1s1td.xml
```

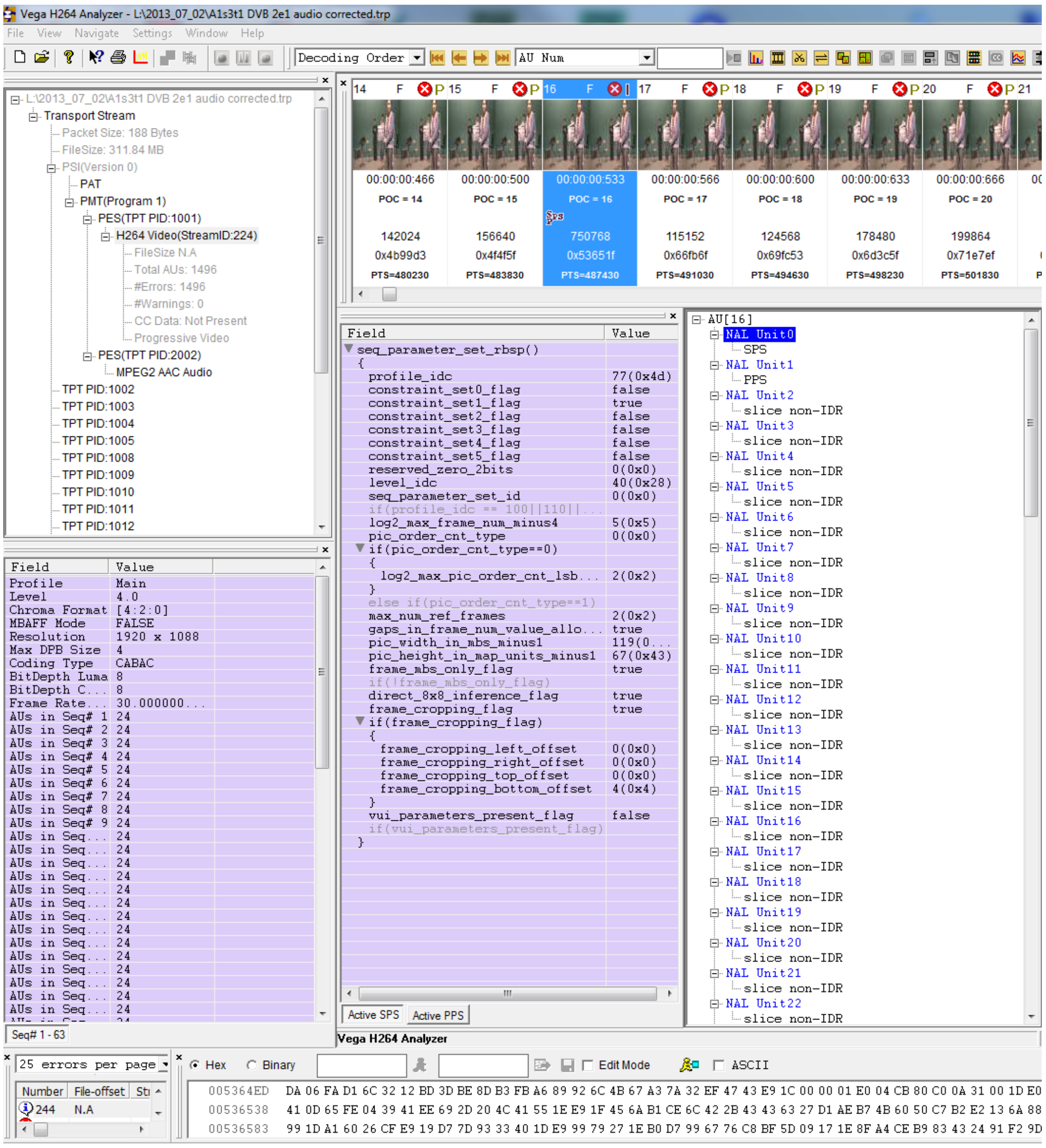
```
INFO   : Service Description Table contains:
         table_id                       : 0x42 (actual TS)
         section_syntax_indicator       : 1
         reserved_future_use            : 0x1 (1 bit)
         reserved                       : 0x3 (2 bit)
         section_length                 : 108
         transport_stream_id            : 0x07CF (1999)
         reserved                       : 0x3 (2 bit)
         version_number                 : 0
         current_next_indicator         : 1
         section_number                 : 0
         last_section_number            : 0
         original_network_id            : 0x07D0 (2000)
         reserved_future_use            : 0xFF (8 bit)
         SERVICES LOOP
         1st SERVICE
         service_id                     : 0x0001 (1)
         reserved_future_use            : 0x3F (6 bit)
         EIT_schedule_flag              : 0
         EIT_present_following_flag     : 1
         running_status                 : 4 (running)
         free_CA_mode                   : 0
         descriptors_loop_length        : 91
         SERVICE DESCRIPTORS
         service_descriptor
         descriptor_tag                 : 0x48
         descriptor_length              : 22
         service_type                   : 0x01 (8 bit) (digital television service)
         provider_name_length           : 15
         provider_name                  : Stream Combiner
         service_name_length            : 4
         service_name                   : CH 1
         additional_content_URL_descriptor
         descriptor_tag                 : 0x7F
         descriptor_length              : 65
         descriptor_tag_extension       : 0x12 (8 bit)
         reserved_future_use            : 0xFF (8 bit)
         additional_content_id          : 0x0001 (16 bit)
         URL_base_length                : 31 (8 bit)
         URL_base                       : www.ceng.metu.edu.tr/~e1347947/
         URL_extension_count            : 1 (8 bit)
         URL_extension_id               : 0x0001 (16 bit)
         URL_extension_length           : 25 (8 bit)
         URL_extension                  : romeo-metadata_b1s1td.xml
         CRC_32                         : 0x6D875B43
```

Figure 6 - User interface of the Transport Stream generation tool

2.1.3.3.4 Special considerations for Transport Streams in ROMEO

For the distribution of TSs containing video and audio over IP networks, it is essential that the decoding of the Transport Stream can continue as quickly as possible when one or more IP packets are lost. Therefore, the headers of the PID streams containing video Elementary Streams are modified so that the SPS, SSPS, PPS are repeated at every Intra frame (I-frame).

The result of the TS generation process is monitored with a standard H.264 analyser tool as shown in Figure 7.



```
L:\2013_07_02\A1s3t1 DVB 2e1 audio corrected.trp
  Transport Stream
    Packet Size: 188 Bytes
    FileSize: 311.84 MB
    PSI(Version 0)
      PAT
      PMT(Program 1)
        PES(TPT PID:1001)
          H264 Video(StreamID:224)
            FileSize N.A
            Total AUs: 1496
            #Errors: 1496
            #Warnings: 0
            CC Data: Not Present
            Progressive Video
        PES(TPT PID:2002)
          MPEG2 AAC Audio
    TPT PID:1002
    TPT PID:1003
    TPT PID:1004
    TPT PID:1005
    TPT PID:1008
    TPT PID:1009
    TPT PID:1010
    TPT PID:1011
    TPT PID:1012
```

```
14    F  P 15    F  P 16    F  I 17    F  P 18    F  P 19    F  P 20    F  P 21
00:00:00:466  00:00:00:500  00:00:00:533  00:00:00:566  00:00:00:600  00:00:00:633  00:00:00:666  00
POC = 14      POC = 15      POC = 16      POC = 17      POC = 18      POC = 19      POC = 20
                            SPS
142024        156640        750768        115152        124568        178480        199864
0x4b99d3      0x4f4f5f      0x53651f      0x66fb6f      0x69fc53      0x6d3c5f      0x71e7ef
PTS=480230    PTS=483830    PTS=487430    PTS=491030    PTS=494630    PTS=498230    PTS=501830    P
```

Field	Value
Profile	Main
Level	4.0
Chroma Format	[4:2:0]
MBAFF Mode	FALSE
Resolution	1920 x 1088
Max DPB Size	4
Coding Type	CABAC
BitDepth Luma	8
BitDepth C...	8
Frame Rate...	30.000000...
AUs in Seq# 1	24
AUs in Seq# 2	24
AUs in Seq# 3	24
AUs in Seq# 4	24
AUs in Seq# 5	24
AUs in Seq# 6	24
AUs in Seq# 7	24
AUs in Seq# 8	24
AUs in Seq# 9	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24
AUs in Seq...	24

```
Field                                   Value
seq_parameter_set_rbsp()
{
  profile_idc                           77(0x4d)
  constraint_set0_flag                  false
  constraint_set1_flag                  true
  constraint_set2_flag                  false
  constraint_set3_flag                  false
  constraint_set4_flag                  false
  constraint_set5_flag                  false
  reserved_zero_2bits                   0(0x0)
  level_idc                             40(0x28)
  seq_parameter_set_id                  0(0x0)
  if(profile_idc == 100||110||...
  log2_max_frame_num_minus4             5(0x5)
  pic_order_cnt_type                    0(0x0)
  if(pic_order_cnt_type==0)
  {
    log2_max_pic_order_cnt_lsb...       2(0x2)
  }
  else if(pic_order_cnt_type==1)
  max_num_ref_frames                    2(0x2)
  gaps_in_frame_num_value_allo...       true
  pic_width_in_mbs_minus1               119(0...
  pic_height_in_map_units_minus1        67(0x43)
  frame_mbs_only_flag                   true
  if(!frame_mbs_only_flag)
  direct_8x8_inference_flag             true
  frame_cropping_flag                   true
  if(frame_cropping_flag)
  {
    frame_cropping_left_offset          0(0x0)
    frame_cropping_right_offset         0(0x0)
    frame_cropping_top_offset           0(0x0)
    frame_cropping_bottom_offset        4(0x4)
  }
  vui_parameters_present_flag           false
  if(vui_parameters_present_flag)
}
```

```
AU[16]
  NAL Unit0
    SPS
  NAL Unit1
    PPS
  NAL Unit2
    slice non-IDR
  NAL Unit3
    slice non-IDR
  NAL Unit4
    slice non-IDR
  NAL Unit5
    slice non-IDR
  NAL Unit6
    slice non-IDR
  NAL Unit7
    slice non-IDR
  NAL Unit8
    slice non-IDR
  NAL Unit9
    slice non-IDR
  NAL Unit10
    slice non-IDR
  NAL Unit11
    slice non-IDR
  NAL Unit12
    slice non-IDR
  NAL Unit13
    slice non-IDR
  NAL Unit14
    slice non-IDR
  NAL Unit15
    slice non-IDR
  NAL Unit16
    slice non-IDR
  NAL Unit17
    slice non-IDR
  NAL Unit18
    slice non-IDR
  NAL Unit19
    slice non-IDR
  NAL Unit20
    slice non-IDR
  NAL Unit21
    slice non-IDR
  NAL Unit22
    slice non-IDR
```

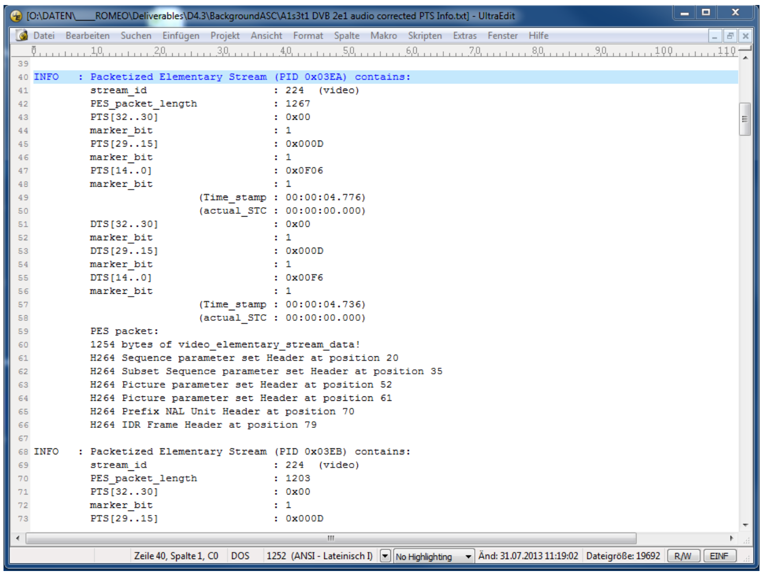
Number	File-offset	Str
244	N.A	

```
005364ED   DA 06 FA D1 6C 32 12 BD 3D BE 8D B3 FB A6 89 92 6C 4B 67 A3 7A 32 EF 47 43 E9 1C 00 00 01 E0 04 CB 80 C0 0A 31 00 1D E0
00536538   41 0D 65 FE 04 39 41 EE 69 2D 20 4C 41 55 1E E9 1F 45 6A B1 CE 6C 42 2B 43 43 63 27 D1 AE B7 4B 60 50 C7 B2 E2 13 6A 88
00536583   99 1D A1 60 26 CF E9 19 D7 7D 93 33 40 1D E9 99 79 27 1E B0 D7 99 67 76 C8 BF 5D 09 17 1E 8F A4 CE B9 83 43 24 91 F2 9D
```

Figure 7 - User interface of the H.264 analyser tool

The next step was the modification of the PES (Packetised Elementary Stream) Packetisation module of the TS generation tool. The specification of the Transport Streams in ROMEO requires that every NAL unit is packed in to a separate PES packet.

An additional module for the TS generation tool was developed for the synchronisation of the PTS values of the video PESs (for the different descriptions and layers) to guarantee that all corresponding video PES packets have identical PTS/DTS (Presentation Time Stamp/ Decoding Time Stamp) values throughout the entire sequence, not only at the start (see example in Figure 8).


[O:\DATEN\____ROMEO\Deliverables\D4.3\BackgroundASC\A1s3t1 DVB 2e1 audio corrected PTS Info.txt] - UltraEdit
Datei Bearbeiten Suchen Einfügen Projekt Ansicht Format Spalte Makro Skripten Extras Fenster Hilfe
Zeile 40, Spalte 1, C0
DOS
1252 (ANSI - Lateinisch I)
No Highlighting
Änd: 31.07.2013 11:19:02
Dateigröße: 19692
R/W
EINF

```

INFO   : Packetized Elementary Stream (PID 0x03EA) contains:
         stream_id                    : 224  (video)
         PES_packet_length            : 1267
         PTS[32..30]                  : 0x00
         marker_bit                   : 1
         PTS[29..15]                  : 0x000D
         marker_bit                   : 1
         PTS[14..0]                   : 0x0F06
         marker_bit                   : 1
                           (Time_stamp : 00:00:04.776)
                           (actual_STC : 00:00:00.000)
         DTS[32..30]                  : 0x00
         marker_bit                   : 1
         DTS[29..15]                  : 0x000D
         marker_bit                   : 1
         DTS[14..0]                   : 0x00F6
         marker_bit                   : 1
                           (Time_stamp : 00:00:04.736)
                           (actual_STC : 00:00:00.000)
         PES packet:
         1254 bytes of video_elementary_stream_data!
         H264 Sequence parameter set Header at position 20
         H264 Subset Sequence parameter set Header at position 35
         H264 Picture parameter set Header at position 52
         H264 Picture parameter set Header at position 61
         H264 Prefix NAL Unit Header at position 70
         H264 IDR Frame Header at position 79

INFO   : Packetized Elementary Stream (PID 0x03EB) contains:
         stream_id                    : 224  (video)
         PES_packet_length            : 1203
         PTS[32..30]                  : 0x00
         marker_bit                   : 1
         PTS[29..15]                  : 0x000D
```

Figure 8 - Example of syntax of streams with corrected PTS

To comply with the requirements of the ROMEO project, the SVC stream (which normally contains all views in one PID) had to be separated into two separate PIDs for base layer and extended layer for each view which are then put into different PIDs.

2.1.3.3.5 Summary of TS generation tool modifications

The modifications of the Transport Stream generation tool were developed to implement all ROMEO-specific features into Transport Streams that can then be used for tests and demonstrations throughout the project.
After the definition of the numbering of the PIDs for all contained streams, the required PSI/SI tables are generated. The Elementary Streams are split in such a way that compatibility with the H.264 AVC standard [13] is maintained. The sequences are cut to a suitable length and the synchronisation between all video views and the audio streams is enforced.
The Additional_content_URL_descriptor is defined and implemented in the PSI/SI tables, and the correct PTS values are inserted into each NAL unit.
With these measures, the Transport Streams fulfil the requirements defined in ROMEO in previous deliverables and can be used for integration test purpose and for the user evaluation tests.

2.2 P2P

P2P components in the server are responsible for streaming the content to top layer peers of the ROMEO system. In this respect, P2P Packetisation component responsible for providing P2P chunks including the main content to the P2PTransmitter component for the delivery. Topology Builder is the component who informs P2PTransmitter about the list of peers which are connected to the server directly so that P2P Transmitter stream the content them.

2.2.1 Topology Builder/Multicast Tree Manager

The Topology Builder (TB) is a software module that performs the following functions at the main server/super-peer:

- listens for new peer connection requests;

- acts as an authentication proxy (authenticator) for user authentication with the main server;
- creates multiple P2P application-level multicast trees for content distribution;
- computes peer insertion on the P2P trees;

When a peer is redirected to a super-peer, it is the responsibility of the TB to compute the peer position in the P2P multicast tree at access network level. The steps in the computation are: (i) to group peers according to the requested content; (ii) group peers according to their common edge router - geographical aggregation and; (iii) sort peers by evaluation, a metric explained in ROMEO deliverable D5.1 [1].

The Multicast Tree Manager (MTM) is intrinsically related with the TB operations and has the following functions:

- It collects/aggregates network monitoring data (percentage of packet loss, average delay, jitter and available bandwidth), from all connected peers, providing the TB with updated peer's network conditions;
- It allows peers to perform bandwidth tests with the server/super-peer.

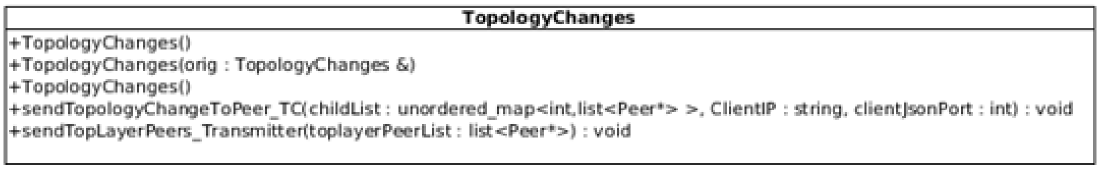
Next section describes the most relevant classes used by the TB and MTM to perform their operations.

2.2.1.1 Provided Classes

The TB and MTM software modules are constituted by the following classes:

`TopologyChanges class`

The *TopologyChanges* class is responsible to dispatch tree updates to all connected peers. This class is also responsible to send the top-layer peers to the *P2P Transmitter* module. These class functions are listed in Figure 9.

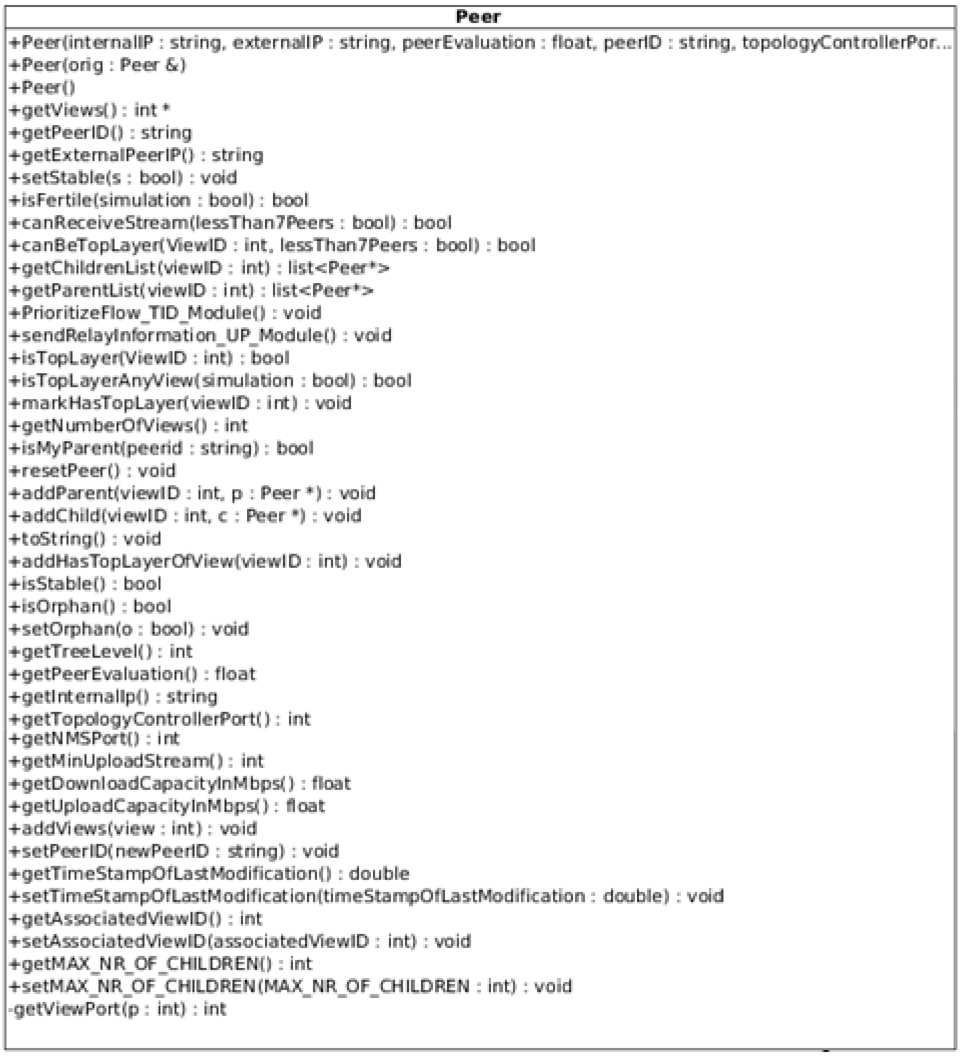


```
TopologyChanges
+TopologyChanges()
+TopologyChanges(orig : TopologyChanges &)
+TopologyChanges()
+sendTopologyChangeToPeer_TC(childList : unordered_map<int,list<Peer*> >, ClientIP : string, clientJsonPort : int) : void
+sendTopLayerPeers_Transmitter(toplayerPeerList : list<Peer*>) : void
```

Figure 9 - The TopologyChanges class and its functions

`Peer class`

The *Peer* class is responsible to identify a peer. This class collects peer relevant information (such as its evaluation, parent list, children list, status - stable, fertile, orphan), dispatchs the prioritization flow to the *Virrtualization* component, for QoS optimization at the access network, and is also responsible to send the children list to the *P2P Chunk Selection* module. These class functions are listed in Figure 10.

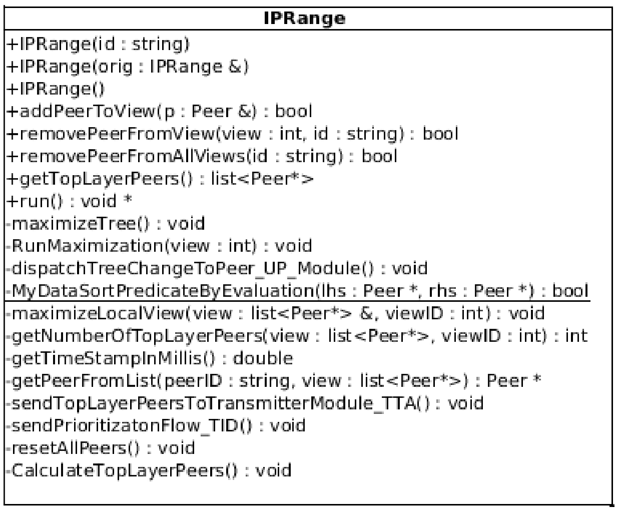


```
Peer
+Peer(internalIP : string, externalIP : string, peerEvaluation : float, peerID : string, topologyControllerPor...
+Peer(orig : Peer &)
+Peer()
+getViews() : int *
+getPeerID() : string
+getExternalPeerIP() : string
+setStable(s : bool) : void
+isFertile(simulation : bool) : bool
+canReceiveStream(lessThan7Peers : bool) : bool
+canBeTopLayer(ViewID : int, lessThan7Peers : bool) : bool
+getChildrenList(viewID : int) : list<Peer*>
+getParentList(viewID : int) : list<Peer*>
+PrioritizeFlow_TID_Module() : void
+sendRelayInformation_UP_Module() : void
+isTopLayer(ViewID : int) : bool
+isTopLayerAnyView(simulation : bool) : bool
+markHasTopLayer(viewID : int) : void
+getNumberOfViews() : int
+isMyParent(peerid : string) : bool
+resetPeer() : void
+addParent(viewID : int, p : Peer *) : void
+addChild(viewID : int, c : Peer *) : void
+toString() : void
+addHasTopLayerOfView(viewID : int) : void
+isStable() : bool
+isOrphan() : bool
+setOrphan(o : bool) : void
+getTreeLevel() : int
+getPeerEvaluation() : float
+getInternalIp() : string
+getTopologyControllerPort() : int
+getNMSPort() : int
+getMinUploadStream() : int
+getDownloadCapacityInMbps() : float
+getUploadCapacityInMbps() : float
+addViews(view : int) : void
+setPeerID(newPeerID : string) : void
+getTimeStampOfLastModification() : double
+setTimeStampOfLastModification(timeStampOfLastModification : double) : void
+getAssociatedViewID() : int
+setAssociatedViewID(associatedViewID : int) : void
+getMAX_NR_OF_CHILDREN() : int
+setMAX_NR_OF_CHILDREN(MAX_NR_OF_CHILDREN : int) : void
-getViewPort(p : int) : int
```

Figure 10 - The Peer class and its functions

IPRange class

The *IPRange* class is responsible to: (i) aggregate peers by geographical location, based on the peers IP addresses; (ii) maximize and trim the multicast tree for each content; (iii) add and remove peers from specific views, as a result of the user preferences. These class functions are listed in Figure 11.

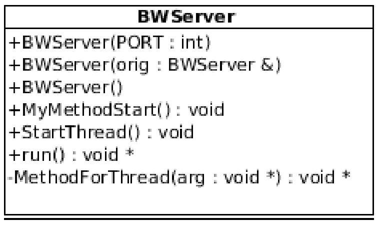


```
IPRange
+IPRange(id : string)
+IPRange(orig : IPRange &)
+IPRange()
+addPeerToView(p : Peer &) : bool
+removePeerFromView(view : int, id : string) : bool
+removePeerFromAllViews(id : string) : bool
+getTopLayerPeers() : list<Peer*>
+run() : void *
-maximizeTree() : void
-RunMaximization(view : int) : void
-dispatchTreeChangeToPeer_UP_Module() : void
-MyDataSortPredicateByEvaluation(lhs : Peer *, rhs : Peer *) : bool
-maximizeLocalView(view : list<Peer*> &, viewID : int) : void
-getNumberOfTopLayerPeers(view : list<Peer*>, viewID : int) : int
-getTimeStampInMillis() : double
-getPeerFromList(peerID : string, view : list<Peer*>) : Peer *
-sendTopLayerPeersToTransmitterModule_TTA() : void
-sendPrioritizatonFlow_TID() : void
-resetAllPeers() : void
-CalculateTopLayerPeers() : void
```

Figure 11 - The IPRange class and its functions

`BWServer class`

The *BWServer* class is responsible to instantiate individual handlers for each peer that has requested a link test. These class functions are listed in Figure 12.

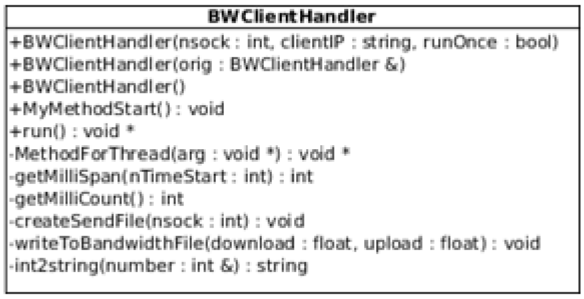


```
BWServer
+BWServer(PORT : int)
+BWServer(orig : BWServer &)
+BWServer()
+MyMethodStart() : void
+StartThread() : void
+run() : void *
-MethodForThread(arg : void *) : void *
```

Figure 12 - The BWServer class and its functions

`BWClientHandler class`

The *BWClientHandler* class is responsible, at the server/super-peer, to perform the link test towards one peer. These class functions are listed in Figure 13.

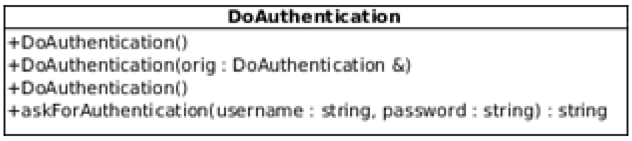


```
BWClientHandler
+BWClientHandler(nsock : int, clientIP : string, runOnce : bool)
+BWClientHandler(orig : BWClientHandler &)
+BWClientHandler()
+MyMethodStart() : void
+run() : void *
-MethodForThread(arg : void *) : void *
-getMilliSpan(nTimeStart : int) : int
-getMilliCount() : int
-createSendFile(nsock : int) : void
-writeToBandwidthFile(download : float, upload : float) : void
-int2string(number : int &) : string
```

Figure 13 - The BWClientHandler class and its functions

`DOAuthentication class`

The *DOAuthentication* class is responsible to relay the peer authentication request to the *Authentication, Registration and Security* ROMEO component. These class functions are listed in Figure 14.

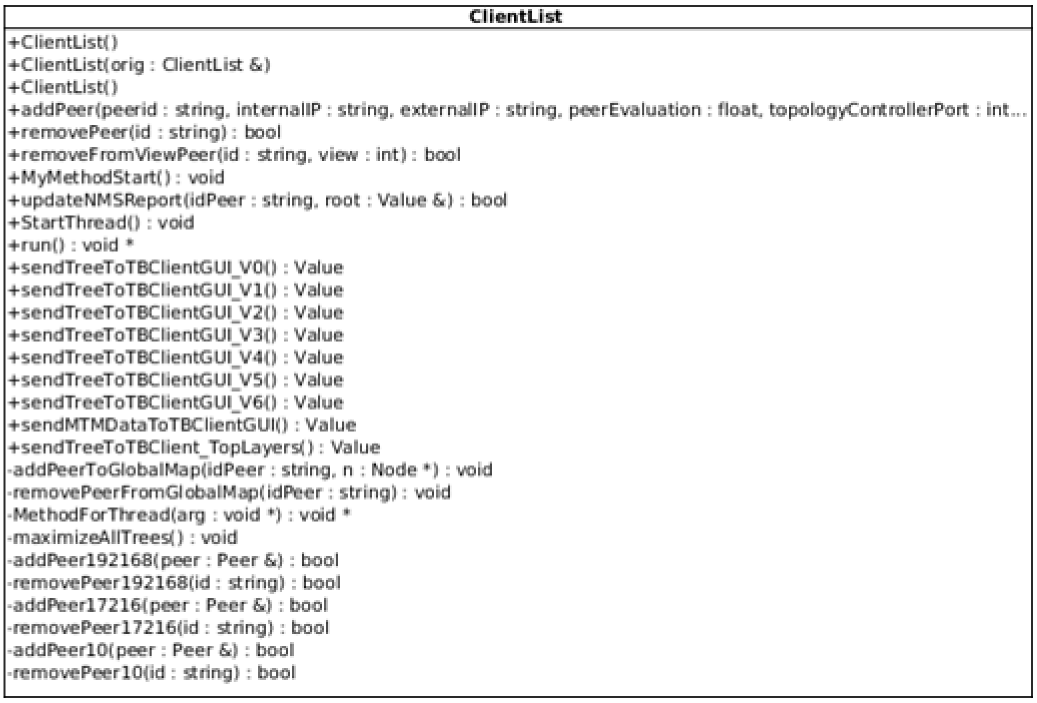


```
DoAuthentication
+DoAuthentication()
+DoAuthentication(orig : DoAuthentication &)
+DoAuthentication()
+askForAuthentication(username : string, password : string) : string
```

Figure 14 - The BWClientHandler class and its functions

`ClientList class`

The *ClientList* class is responsible to: (i) insert peers in the *IPRange* class; (ii) provide the TB graphical user interface -- an external java application – with the details (nodes, their relations and hierarchy) of each tree. These class functions are listed in Figure 15:

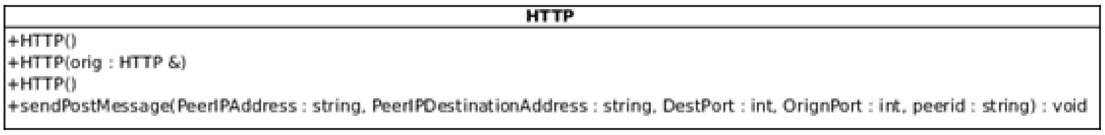


```
ClientList
+ClientList()
+ClientList(orig : ClientList &)
+ClientList()
+addPeer(peerid : string, internalIP : string, externalIP : string, peerEvaluation : float, topologyControllerPort : int...
+removePeer(id : string) : bool
+removeFromViewPeer(id : string, view : int) : bool
+MyMethodStart() : void
+updateNMSReport(idPeer : string, root : Value &) : bool
+StartThread() : void
+run() : void *
+sendTreeToTBClientGUI_V0() : Value
+sendTreeToTBClientGUI_V1() : Value
+sendTreeToTBClientGUI_V2() : Value
+sendTreeToTBClientGUI_V3() : Value
+sendTreeToTBClientGUI_V4() : Value
+sendTreeToTBClientGUI_V5() : Value
+sendTreeToTBClientGUI_V6() : Value
+sendMTMDataToTBClientGUI() : Value
+sendTreeToTBClient_TopLayers() : Value
-addPeerToGlobalMap(idPeer : string, n : Node *) : void
-removePeerFromGlobalMap(idPeer : string) : void
-MethodForThread(arg : void *) : void *
-maximizeAllTrees() : void
-addPeer192168(peer : Peer &) : bool
-removePeer192168(id : string) : bool
-addPeer17216(peer : Peer &) : bool
-removePeer17216(id : string) : bool
-addPeer10(peer : Peer &) : bool
-removePeer10(id : string) : bool
```

Figure 15 - The ClientList class and its functions

`HTTP class`

The *HTTP* class is responsible to create and send a HTTP POST message, containing the traffic details (IP and port information) between each parent and its children, to the Virtualization component with the goal of providing QoS assurances at the peer access network. These class functions are listed in Figure 16:

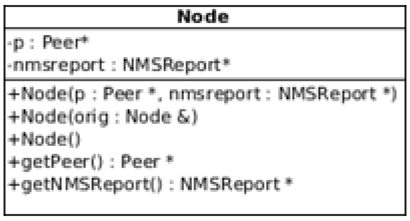


```
HTTP
+HTTP()
+HTTP(orig : HTTP &)
+HTTP()
+sendPostMessage(PeerIPAddress : string, PeerIPDestinationAddress : string, DestPort : int, OrignPort : int, peerid : string) : void
```

Figure 16 - The HTTP class and its functions

`Node class`

The *Node* class aggregates information on each peer. The collected information is defined by the *NMSReport* class, see Figure 18. These class functions are listed in Figure 17:

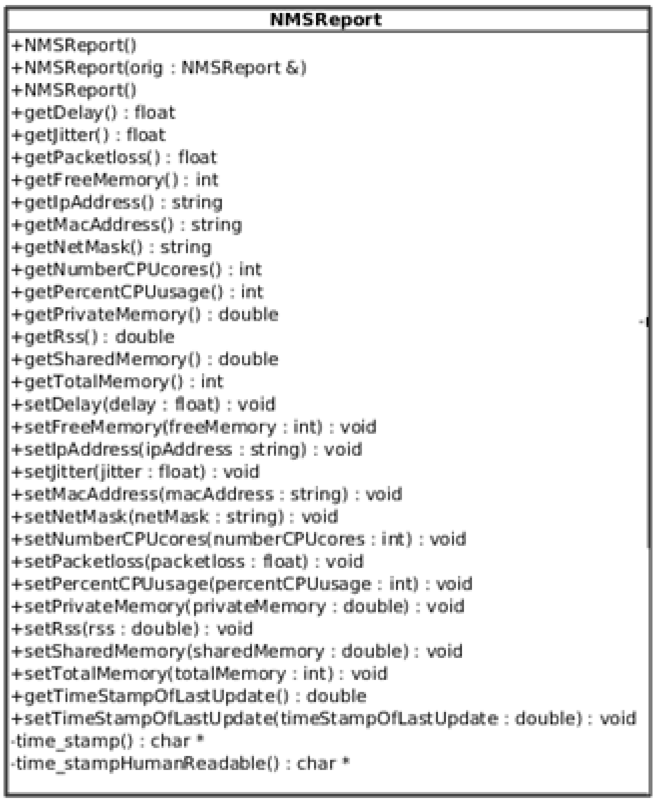


```
Node
-p : Peer*
-nmsreport : NMSReport*
+Node(p : Peer *, nmsreport : NMSReport *)
+Node(orig : Node &)
+Node()
+getPeer() : Peer *
+getNMSReport() : NMSReport *
```

Figure 17 - The Node class and its functions

`NMSReport class`

The *NMSReport* class defines: (i) the Network Monitoring Subsystem (NMS) report structure; (ii) the functions to parse a received JSON NMS report and; (iii) the functions to update the node report for a specific peer. These class functions are listed in Figure 18:



```
NMSReport
+NMSReport()
+NMSReport(orig : NMSReport &)
+NMSReport()
+getDelay() : float
+getJitter() : float
+getPacketloss() : float
+getFreeMemory() : int
+getIpAddress() : string
+getMacAddress() : string
+getNetMask() : string
+getNumberCPUcores() : int
+getPercentCPUusage() : int
+getPrivateMemory() : double
+getRss() : double
+getSharedMemory() : double
+getTotalMemory() : int
+setDelay(delay : float) : void
+setFreeMemory(freeMemory : int) : void
+setIpAddress(ipAddress : string) : void
+setJitter(jitter : float) : void
+setMacAddress(macAddress : string) : void
+setNetMask(netMask : string) : void
+setNumberCPUcores(numberCPUcores : int) : void
+setPacketloss(packetloss : float) : void
+setPercentCPUusage(percentCPUusage : int) : void
+setPrivateMemory(privateMemory : double) : void
+setRss(rss : double) : void
+setSharedMemory(sharedMemory : double) : void
+setTotalMemory(totalMemory : int) : void
+getTimeStampOfLastUpdate() : double
+setTimeStampOfLastUpdate(timeStampOfLastUpdate : double) : void
-time_stamp() : char *
-time_stampHumanReadable() : char *
```

Figure 18 - The NMSReport class and its functions

`JsonServer class`

The *JsonServer* class is responsible for sending and receiving JSON messages to/from the server application submodules. These class functions are listed in Figure 19:

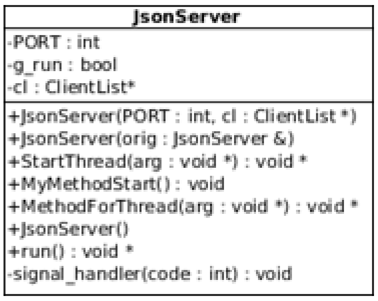

JsonServer
-PORT : int
-g_run : bool
-cl : ClientList*
+JsonServer(PORT : int, cl : ClientList *)
+JsonServer(orig : JsonServer &)
+StartThread(arg : void *) : void *
+MyMethodStart() : void
+MethodForThread(arg : void *) : void *
+JsonServer()
+run() : void *
-signal_handler(code : int) : void

Figure 19 - The JsonServer class and its functions

2.2.1.2 Messages

The TB/MTM modules currently send the following JavaScript Object Notation (JSON) messages:

From	To	Message	Description
TB	Topology Controller (TC)	307-TOPOLOGY_JOIN_RESPONSE	Informs a peer on the list of parents
TB/MTM	TC	320-TREE_STATE_READY	Informs a peer on the list of backup parents
TB	Transmitter	328-TOPLAYER_PEERS_INFO*	Informs the P2P Transmitter on the top-layer peers
TB	TC	330-USER_RESPONSE*	Informs the peer on it's: (i) PeerID; (ii) authentication status; (ii) port and IP address to use for the link tests.
TB	Virtualization	331-PRIORITY_FLOW_INFO*	Informs the Virtualization component on the parent-child traffic characteristics (IP and port information) for QoS flow prioritization at the access network

*Messages not defined in ROMEO Deliverable 2.2 [2]

2.2.2 P2P Transmitter

2.2.2.1 Description

P2P Transmitter is a multithreaded UDP sender application which supports multiple ROMEO view streams and runtime configurable destination IP lists. It counts the number of streams that has different TS PID values which are declared to be sent in the metadata. Each stream has separate threads and each thread parses the content in server-transmitter.xml.

2.2.2.2 Provided Functions

`StreamSenderThread::StreamSenderThread(StreamConfig *cfg) :`

This function is used to configure the thread with the readed configurations from romeo-metadata.xml and server-transmitter.xml.

`void StreamSenderThread::parseBaseLayerPESandSendToNw(void):`

This function is used to send encapsulated UDP packets to IPs which are in the destination IP list with a stable framerate. It reads the given TRP file and parses the MPEG 2 TS packets. It pushes the parsed chunks to the Packetisations Module. When the data is ready, it checks the previous send time. And calculates the next send time and sleep times to achieve 40 ms average time difference for each frames.

The destinanion IP lists is a thread safe list and used to communicate with Topology Builder component for active destination IP's.

`bool TBMessageHandlerThread::initConnection():`

This function is used to initiate TCP/IP Sockets of Topology Builder Message handler thread. It checks the connection state and inits/re-inits the TCP Socket for new communications.

`void TBMessageHandlerThread::run():`

This function is a handler for JSON messages from Topology Builder component. If the message is correct then the destination IP list is updated by this function.

`bool TBMessageHandlerThread::isMessageValid(Json::Value &root):`

This function checks the received JSON message structure and generates console output for erroneous messages.

`void GetAppConfig::readConfigFiles(string& metaDataFile, string& appConfigFile):`

This function tests the access and availability of the XML configuration file. It configures the xerces-c DOM parser,reads and extracts the pertinent information from the XML config file. It reads all configurations from romeo-metadata.xml and server-transmitter.xml.

2.2.2.3 FutureWork

This module is successfully integrated into the ROMEO system at Second ROMEO Workshop 2013. Necessary updates will be performed based on the integration test results.

2.2.3 P2P Packetisation

2.2.3.1 Description

This module is responsible for encapsulation of the media as P2P chunks to be carried over the P2P network.

2.2.3.2 Provided Functions

```
void Packetisation::prepare_header(P2PHeader& header,short int size,int pid,
uint64_t pcr,std::string &audio_video_indicator,    unsigned short int
viewID,unsigned short int descriptorID,std::string &layerType,unsigned int
crc):
```

This function is used to prepare P2P Header for encapsulating chunks.

```
void Packetisation::TS_header_decode(unsigned char *TS_raw_header,
TS_header& tsHeader):
```

This function is an utiliy function to decode TS headers.

```
int Packetisation::PES_header_decode (unsigned char* data,uint32_t& pts):
```

This function is an utility function to decode PES header and calculates PTS value.

`int Packetisation::SetPTSValue(unsigned char* data, uint32_t pts):`

This function is used to set the PTS values for continuously streaming of the same stream.

2.2.3.3 FutureWork

This module is successfully integrated into the ROMEO system at Second ROMEO Workshop 2013. Necessary updates will be performed based on the integration test results.

2.3 Authentication, Registry and Security

Authentication, Registry and Security component maintains a list of valid ROMEO users and uses this list during user authentication. User creation, deletion and update are performed upon the requests send from UI component and as a result of these requests, valid user list is modified. When a user wants to join the ROMEO system, authentication request is sent to Authentication, Registry and Security component by Topology Builder component and Authentication, Registry and Security component validates the user by using its valid user list.

2.3.1 Description

Authentication, Registry and Security component has the capabilities of creating a user, deleting a user, updating a user and authenticating a user. The module is implemented on a REST server, which use Zend PHP Framework and Apache Server.

The provided API for Authentication, Registry and Security module is as follows.

`Creating User:`

To register into the ROMEO system, ROMEO user interface must send an HTTP_POST request to the REST server. This call will be used to create a new ROMEO user and a random hash value for the user to sign the API calls.

postAction() function

Description:

This function lets a user to be added to the database.

Input parameters:

- name: Name of the user
- password: Password of the user

Return Values:

The return value is included in the HTTP response’s body as a JSON message.

- status: NOT_OK → if failed and status : OK → if successful

If status: OK

- hash: A random hash value

If status: NOT_OK

- error: USER_NOT_CREATED → If user cannot be created
- error: WRONG_API_FORMAT → Check if all required fields are filled

Example API Call:

http://localhost/api/user?name=yuri&password=gagarin

Example Return Value:

{"status":"OK","hash":"WbZVD7k7dlm871Y9FZcYL4PfX0M676aZOVXhFgKi4bglfFli71ggU3bD2X8LfO8kAYO8ilNFIKS767Vfba6dlKfUJRBGMeMmc5GbNclHJOdMG0WFi3c2RUHUkX06CBKU2JIaNO3NK5VB0BG6XPlRNKbANe1H6hkSHK3NQ4CF224KW8Dh39iKceQ0SBVbcYQXAQ8hCC99SD9kaNW0KQTGI4BbMT22VHd7Wi7BUB01W8IN02kemOIBmF0f6eS2hOPS"}

Updating User:

To change the user password in the ROMEO system, ROMEO user interface must send an HTTP_PUT request to the REST server. This call will be used to update ROMEO user password. If the user password is updated by using this call, a new random hash value will be generated to sign the API calls.

putAction() function

Description:

This function will update the user password and generate a new hash value for the user to sign the API calls.

Input Parameters:

- name: Name of the user
- password: Password of the user
- new password: New password of the user
- signature: Signature created by the hash function

Return Values:

The return value is included in the http response’s body as a JSON message.

- status : NOT_OK → if failed and status : OK → if successful

If status : OK

- hash : A random hash value

If STATUS : NOT_OK

- error : USER_NOT_EXIST → No user with the given name exists
- error : WRONG_PASSWORD → Given password does not match the one in database
- error : WRONG_API_FORMAT → Check if all required fields are filled.
- error : AUTHENTICATION_FAILED → The signature from user does not match the signed content in server

Example API Call:

http://localhost/api/user?name=neil&password=armstrong&new_password=armstrong123&signature=98e1a0eeb13eb6302b50380f5c071578518488e9cf42cf8d6b3f0bd716759899

Deleting User:

To delete an existing user in the ROMEO system, ROMEO user interface must send an HTTP_DELETE request to the REST server. This call will be used to delete the ROMEO user from the ROMEO system.

deleteAction() function

Descrition:

This function will delete the existing user from the database.

Input parameters:

- name: Name of the user
- password: Password of the user
- signature: Signature created by the hash function

Return Values:

The return value is included in the http response's body as a JSON message.

- status : NOT_OK → if failed and status : OK → if successful

If STATUS : NOT_OK

- error : USER_NOT_EXIST → No user with the given name exists
- error : WRONG_PASSWORD → Given password does not match the one in database
- error : WRONG_API_FORMAT → Check if all required fields are filled.
- error : AUTHENTICATION_FAILED → The signature from user does not match the signed content in server

Example API Call:

http://localhost/api/user?name=neil&password=armstrong&signature=278980210a4e6edb122f69c46405b54dede9de94097603313abf332ab3a8bc24

Authenticating User:

To be authenticated in the ROMEO system, ROMEO user interface must send an HTTP_GET request to the REST server.

getAction() function

Description:

This function will return ROMEO user's authentication result.

Input parameters:

- name: Name of the user
- password: Password of the user
- signature: Signature created by the hash function

Return Values:

The return value is included in the http response's body as a JSON message.

- status : NOT_OK → if failed and OK → if successful

If status : NOT_OK

- error : USER_NOT_EXIST → No user with the given name exists
- error : WRONG_PASSWORD → Given password does not match the one in database
- error : WRONG_API_FORMAT → Check if all required fields are filled.
- error : AUTHENTICATION_FAILED → The signature from user does not match the signed content in server

Example API Call:

http://localhost/api/user/id?name=yuri&password=gagarin&signature=2a1ebd5a332ddf6f2609dc527e86a7527e691a4611cb1a498743b43c59491aff

Example Return Value:

{"status":"NOT_OK","error":"WRONG_API_FORMAT”}

2.4 A/V Communication Overlay

ROMEO A/V Communication Overlay mainly consists of two totally independent components: Server and Client components as shown in Figure 20. In that diagram, server, peer and IRACS (Internet Resource & Admission Control Subsystem, which is responsible for QoS adjustments for A/V) are connected to each other with control (red) and stream (green) data. Whereas the server component is used as a hub for collaborating users who communicate with each other, the client component is resided at the user devices in order to let users join in the video conference.

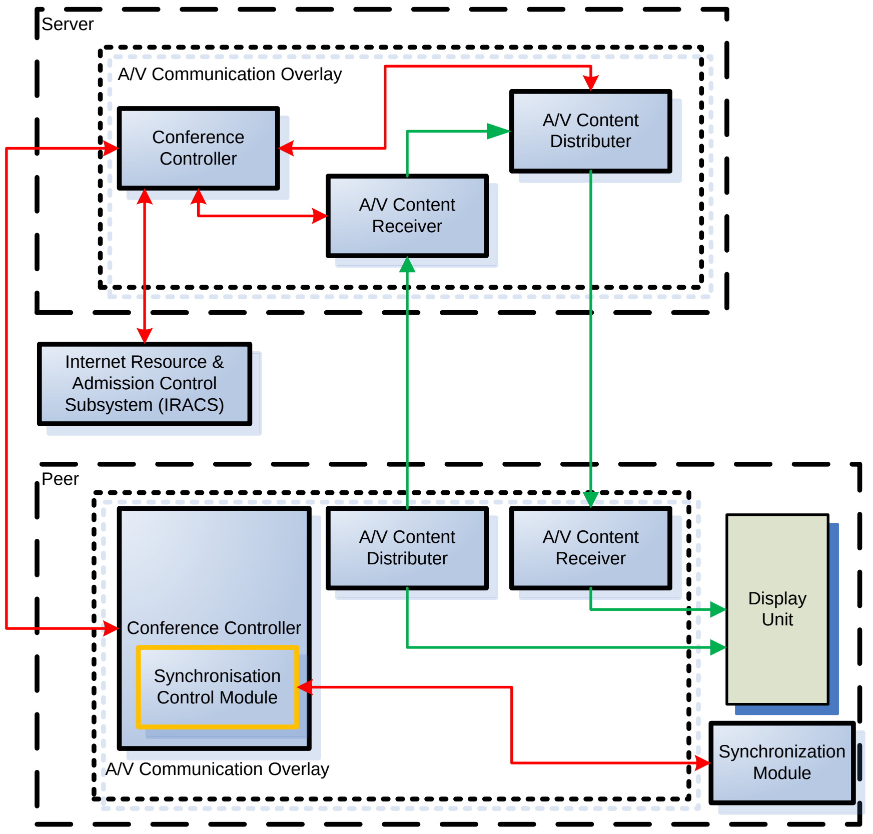


Figure 20 - A/V Communication Overlay overall system description (Green - stream, Red - control data)

Server module needs to broadcast the user video collected from users. Moreover, that module needs to hold the user information to control the authentication and flow of stream. In order to satisfy those needs, a stream server including authentication and user /room/session management is needed. Therefore, the Apache open source project OpenMeetings, satisfying the project requirements is preferred and used in A/V Communication Overlay part. OpenMeetings project included in Red5 server is illustrated in Figure 21. In the same figure, modification strategy of OpenMeetings is given.

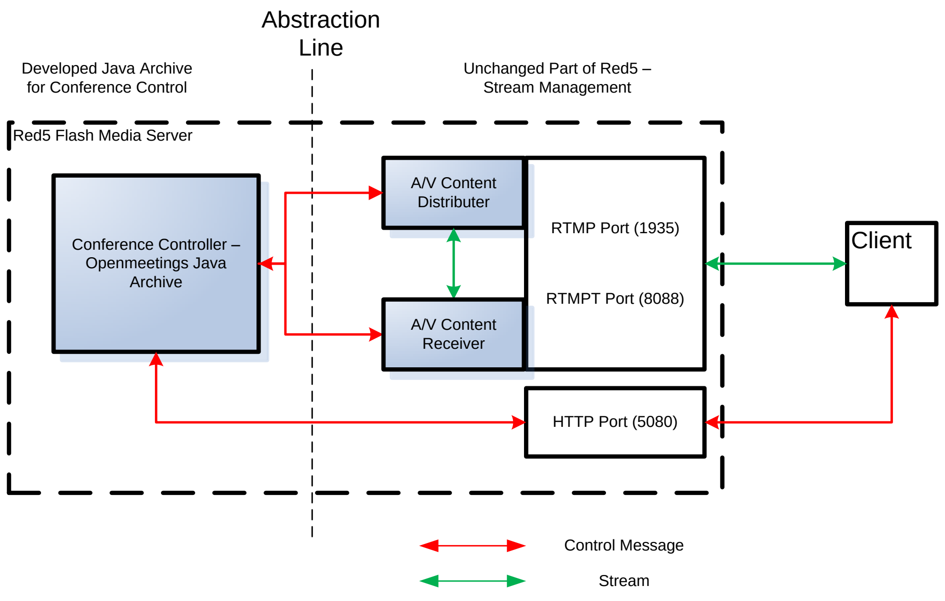


Figure 21 - Media Distribution and Distribution Control Mechanism of Red5

The server module is a modified version of the server used in OpenMeetings open source video conference solution. It is a Red5 flash server which includes a Java Enterprise Edition (EE) container that is responsible for giving collaborating users a web service in order to let them access flash server functionalities.

For ROMEO project, the OpenMeetings server module has been modified by adding a new extra-delay module. The aim of this module is to calculate a system-wide delay, which is necessary to make the members of same collaborating group consume the same live content almost at the same time (with an unnoticeable delay). That module obtains and stores remote users' collaboration delays, and calculates the optimum collaboration delay value. Optimum delay is the needed minimum delay that lets the users communicate with full synchronization. Delay calculations are performed in the server, since those delays included are highly varying factors depending on the network conditions. This modification is given in Figure 22.

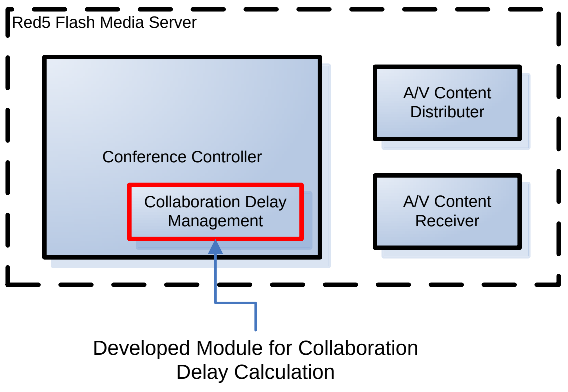


Figure 22 - Modification under Conference Controller

2.4.1 Conference Controller

The Java Web service, contained in Red5 stream server in OpenMeetings project, corresponds to the conference controller in the ROMEO project. In the conference controller development phase, the Java Web Server is modified to have a ROMEO-specific conference controller delay manager. It is updated by adding extra tables to the server and writing getDelay and setDelay functions accessible by users (Figure 23). Sequence diagram of the newly developed message flow is illustrated in Figure 24.

With those modifications, the delay management module is accessible via Web service functions on the server. Therefore, the controller modules on the client in user devices have the ability to synchronize 3D video content via those server functions.

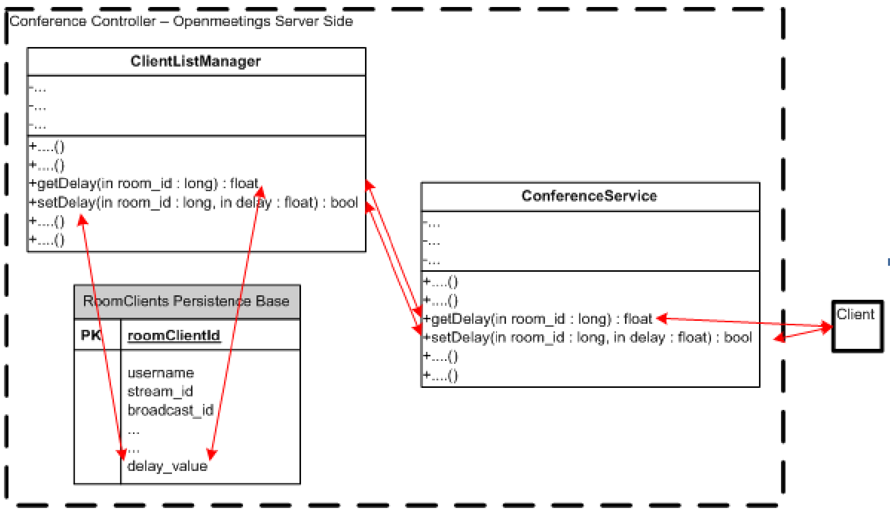

Conference Controller – Openmeetings Server Side
ClientListManager
-...
-...
-...
+....()
+....()
+getDelay(in room_id : long) : float
+setDelay(in room_id : long, in delay : float) : bool
+....()
+....()
RoomClients Persistence Base
PK
roomClientId
username
stream_id
broadcast_id
...
...
delay_value
ConferenceService
-...
-...
-...
+....()
+....()
+getDelay(in room_id : long) : float
+setDelay(in room_id : long, in delay : float) : bool
+....()
+....()
Client

Figure 23 - Added functions & attributes & connections on Server for delay

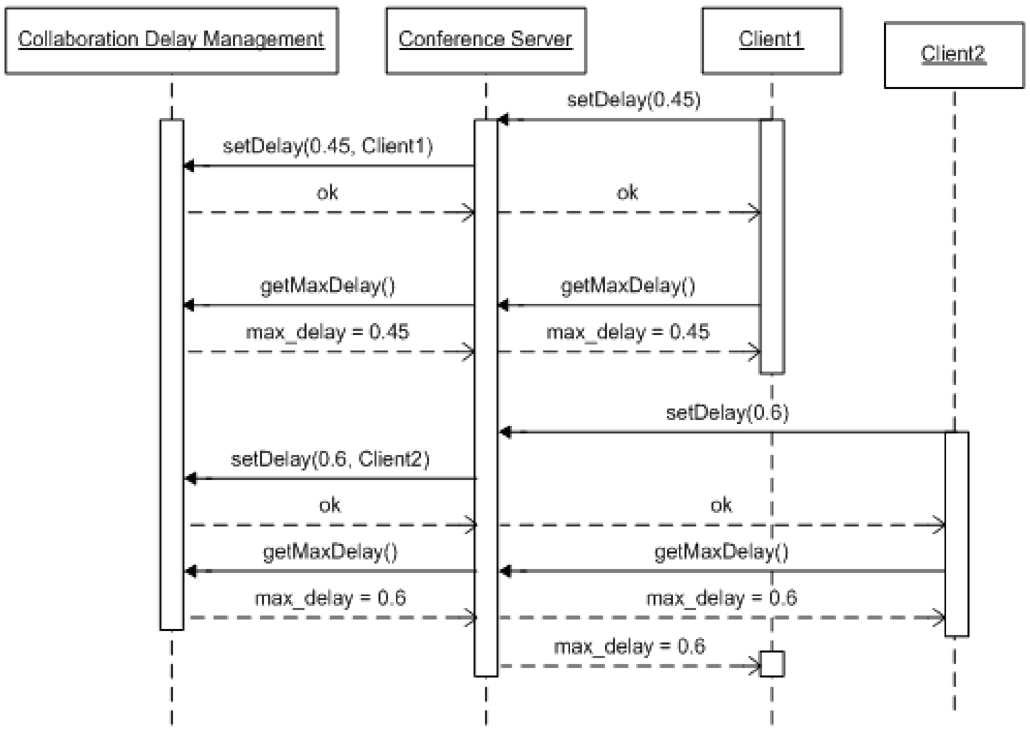


Figure 24 - Design for setDelay and getDelay functions

2.4.2 A/V Content Distributor

The role of A/V Content Distributer module is distributing all media collected from remote users. A/V Content Distributor with streaming abilities is contained in Red5 server core libraries, and the streaming mechanisms of that module are not changed for OpenMeetings. Red5 flash server modification strategy has already been shown in Figure 21.

2.4.3 A/V Content Receiver

The role of A/V Content Receiver module is receiving all media generated by remote users. A/V Content Receiver with RTMP port listening abilities is contained in Red5 server core libraries, and the content receiving mechanisms of that module are not changed for OpenMeetings. Red5 flash server modification strategy has already been shown in Figure 22.

3 PEER COMPONENTS

3.1 P2P

3.1.1 Topology Controller

The Topology Controller (TC) is a software module that runs at the peer with the following purposes:

- Initial contact with the main server for user authentication and redirection to the nearest super-peer;
- Compute the peer evaluation using peer's hardware characteristics and network statistics, as provided by the Network Monitoring Subsystem;
- Inform the TB of its intention to consume specific 3D content;
- Perform P2P tree operations as commanded by the TB (parent, parent/child or child);
- Establish connections with parents for content request and accept connections from children peers for content forwarding.

Next section describes the most relevant classes used by the TC to perform its operations.

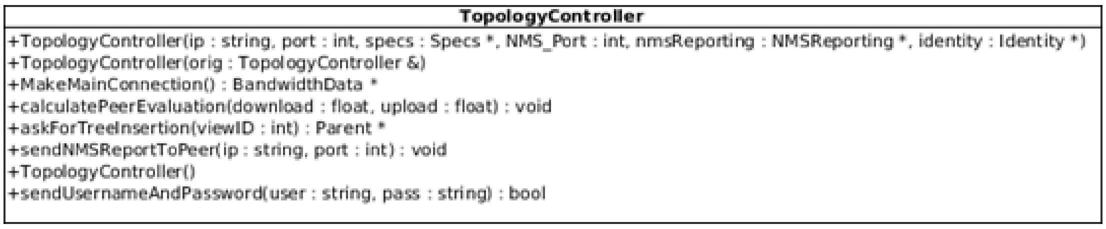
3.1.1.1 Provided Classes

The TC software module is constituted by the following classes:

`TopologyController class`

The *TopologyController* class is responsible for: (i) triggering the peer authentication; (ii) requesting NMS link test; (iii) compute the peer evaluation; (iv) request for tree insertion; (v) triggering the NMS reporting service.

These class functions are listed in Figure 25.

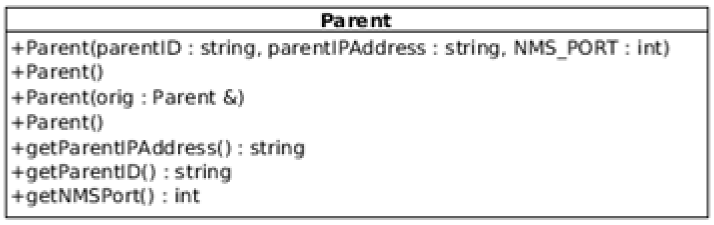


```
TopologyController
+TopologyController(ip : string, port : int, specs : Specs *, NMS_Port : int, nmsReporting : NMSReporting *, identity : Identity *)
+TopologyController(orig : TopologyController &)
+MakeMainConnection() : BandwidthData *
+calculatePeerEvaluation(download : float, upload : float) : void
+askForTreeInsertion(viewID : int) : Parent *
+sendNMSReportToPeer(ip : string, port : int) : void
+TopologyController()
+sendUsernameAndPassword(user : string, pass : string) : bool
```

Figure 25 - The TopologyController class and its functions

`Parent class`

The *Parent* class provides the list of parents (active and backup) for each peer and content type. These class functions are listed in Figure 26.

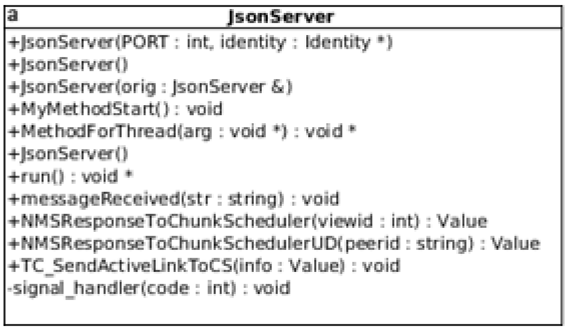


```
Parent
+Parent(parentID : string, parentIPAddress : string, NMS_PORT : int)
+Parent()
+Parent(orig : Parent &)
+Parent()
+getParentIPAddress() : string
+getParentID() : string
+getNMSPort() : int
```

Figure 26 - The Parent class and its functions

`JsonServer class`

The *JsonServer* class is responsible for sending and receiving JSON messages to/from the peer application submodules. These class functions are listed in Figure 27.



a **JsonServer**

```
+JsonServer(PORT : int, identity : Identity *)
+JsonServer()
+JsonServer(orig : JsonServer &)
+MyMethodStart() : void
+MethodForThread(arg : void *) : void *
+JsonServer()
+run() : void *
+messageReceived(str : string) : void
+NMSResponseToChunkScheduler(viewid : int) : Value
+NMSResponseToChunkSchedulerUD(peerid : string) : Value
+TC_SendActiveLinkToCS(info : Value) : void
-signal_handler(code : int) : void
```

Figure 27 - The JsonServer class and its functions

3.1.1.2 Messages

The TC module currently sends the JSON messages in Table 3:

From	To	Message	Description
TC	TB	306-TOPOLOGY_JOIN	Requests to join the multicast distribution trees
TC	TB	329-USER_AUTHENTICATION_REQ*	Sends user credentials for authentication purposes
TC	CS	333-LINK_STATUS_RESPONSE*	Provides network related information (packet loss, delay, jitter, upload and download capacities) for a specific link.

*Messages not defined in ROMEO Deliverable 2.2 [2] and Deliverable 2.3 [17]

Table 3 - TC module messages

3.1.2 Chunk Selection

3.1.2.1 Module description

The Chunk Selection module performs the following operations:

1. It requests from NMS information regarding network performance like upload capacity of peers, delay and packet loss.
2. It receives information from TC regarding the topology, i.e. the new set of children peers in each tree after a topology change.
3. It dictates peer P2PTransmitter component which children peers to forward chunks to in each tree. Under limited upload capacity of a peer, the children peers that belong to higher-priority trees are favoured. Priority of a stream that traverses a tree is defined by: (i) view, (ii) description and (iii) quality layer.
4. If a peer gets disconnected from a tree, ChunkScheduler class requests missing chunks from its remaining parents that are still connected at the tree from which the peer got disconnected, if any.

3.1.2.2 Provided classes

`class ChunkScheduler`

It updates children tables according to information received by TC and NMS, it also updates the children status, computes the incoming bitrate from P2PReceiver and updates the DestinationIPList of PeerP2PTransmitter. The ChunkScheduler class includes the following methods:

```
ChunkScheduler(int, GetAppConfig*);
void ResetChildrenStatus();
void InsertChild(int, int, std::string);
void RemoveChildren();
void ActivateChild(int, int, std::string);
void DeactivateChild(int, int, std::string);
void SetUploadCapacity(double);
void SetPktLoss(double);
void ComputeBitrates(int);
void ApplySelectionPolicy();
~ChunkScheduler();
```

`class InterfaceToNMSforCapacity`

It connects to NMS, requests report on the upload capacity of a peer and updates the peer status. The `InterfaceToNMSforCapacity` class includes the following methods:

```
InterfaceToNMS forCapacity (ChunkScheduler*);
void QueryNMS forCapacity (std::string, std::string, uint16_t);
void *run();
```

`class InterfaceToNMSforPktLoss`

It connects to NMS, periodically requests reports on network performance metrics (packet loss, jitter and delay) and updates the children status. The `InterfaceToNMSforPktLoss` class includes the following methods:

```
InterfaceToNMSforPktLoss (ChunkScheduler*);
void QueryNMSforPktLoss (std::string, std::string, uint16_t);
void *run();
```

`class InterfaceToTC`

It connects to TC, receives updates about the current children and updates the children list accordingly. The `InterfaceToTC` class includes the following methods:

```
InterfaceToTC(ChunkScheduler*);
void ReceiveEventsFromTC();
void UpdateChildren(int, int, std::string);
```

```
        void *run();
```

`class InterfaceToCS`

It connects to CS of parent peers in all trees and requests missing chunks when necessary. At the parent side, it responds with its availability to provide missing chunks. At the child side, it asks the parent peer to initiate transfer of missing chunks. The `InterfaceToCS` class includes the following methods:

```
        InterfaceToCS(ChunkScheduler*);
        void QueryCS (std::string, std::string, uint16_t);
        void ReceiveEventsFromCS();
        void InitiateTransfer(int);
        void CheckAvailability(int);
        void UpdateChildren(int, int, std::string);
        void *run();
```

3.1.2.3 Messages

Chunk selection sends the JSON messages to other modules in Table 4.

From	To	Message	Description
CS	CS	342-MISSING_CHUNKS_REQ	It requests from parent CS whether it has mssing chunks.
CS	CS	343-MISSING_CHUNKS_RESP	It responds to child CS with the possibility to provide missing chunks.
CS	CS	344-INITIATE_TRANSFER	It requests from parent CS to initate transfer of missing chunks.
CS	NMS	331-REQUEST_METRICS	It request network performance metrics from NMS.
CS	NMS	110-REQUEST_CAPACITY	It requests upload capacity from NMS.

Table 4 - Messages from CS to other modules

3.1.3 P2P Transmitter/Receiver

3.1.3.1 Receiver

3.1.3.1.1 Description

P2P Receiver is a multithreaded UDP receiver application which supports multiple ROMEO view streams. It counts the number of streams that has different TS PID values which are declared to be received in the metadata file. Each stream has a separate reception thread.

3.1.3.1.2 Provided Functions

`bool UDPReceiver::recvOverUDP(char* rcvedData, int size)`

This function is used to read *size* bytes of data to given memory over UDP. If there is no data to read it returns false otherwise it returns true.

`UDPReceiver::UDPReceiver(int port)`

This function is used to configurethe receiver to listen in a given port.

`void GetAppConfig::readConfigFiles(std::string& metaDataFile, std::string& appConfigFile)`

This function is used to read configuration files to configure receiver.

3.1.3.2 PeerP2PTransmitter

3.1.3.2.1 Description

The PeerP2PTransmitter is a multithreaded UDP transmitter application which supports multiple ROMEO view streams. It is used to re-transmit the received data to the peers selected in ROMEO network. The main difference between P2PTransmitter and PeerP2PTransmitter is P2PTransmitter send all streams to its destination IP list but PeerP2PTransmitter sends selected streams to selected peers.

3.1.3.2.2 Provided Functions

`void SocketListenerThread::SocketListenerThread(int tsPID, int port)`

This function is used to receive given TS PID identified stream from given UDP port and stores the chunks to a separate stream queue.

`void StreamSenderThread::transmitChunks(void)`

This function is used to transmit its chunks to its pre-selected IP list.

3.1.4 P2P Depacketisation

3.1.4.1 Description

This module is responsible for decapsulation of the media from P2P chunks received over the P2P network.

3.1.4.2 Provided Functions

`void StreamReceiverThread::GetChunksOverUDP()`

This function is used to decapsulate the P2P packet, decrypt the data and send it to Chunk Selection and Synchronisation modules. It is also a part of receiver module.

`bool UDPTransmitter::sendDataOverUDP(char* dataToSend, int size, char* dest_addr, int port)`

This function is used to send received and depacketised data to given IP and port. It is required for communicating with Chunk selection and Synchronisation modules.

3.2 DVB Reception

3.2.1 Fixed/Portable Terminals

DVB reception on peer terminals is provided by PCTV NanoStick, which is capable of receiving DVB-T/T2 signals with terrestrial signal receiver antennas (or direct cable) with MCX connectors. The NanoStick hardware is connected to the platform via an USB 2.0 interface. PCTV provides software with the hardware (tuner, remote control and antenna). The PCTV package has player software for DVB transport stream. This software can be installed on a platform running MS Windows. PCTV does not give any support for Linux OS. Linux OS needs some library installations over standard Linux distributions. Libraries, from Linux DVB API, that are listed below have been included in the DVB reception module.

- FRONTEND

- DEMUX
- DVR

By the help of these libraries, actions listed below have been achieved and software to perform these actions has been developed:

- getting tuner capabilities
- setting DVB tuning parameters
    - Frequency
    - Bandwidth
    - FEC rate
    - Inversion
    - Constellation
    - Transmission Mode
    - Guard Interval
    - Hierarchy Information
- controlling DEMUX filters
- receiving the Transport Stream via DVR

User Interface module provides an interface for setting the DVB parameters (Section 3.8.1). The DVB reception module takes the required parameters from the user interface. All other operations such as tuning, DEMUX filtering, and starting DVR are performed automatically by the DVB reception module.

After the Transport Stream is successfully received via DVB, it is handed down the synchronization module, which includes a stream parser for MPEG Transport Stream, extracts the elementary streams and service information (DVB-SI) tables from the stream. The extracted data is then used for synchronizing the streams and feeding the decoders.

The working principle of DVB reception module on portable and fixed terminals and its connection to the synchronization module can be summarized as follows:

The received stream is parsed via the stream parser. The parser extracts required information and processes them as needed. It sends PCR to both video and audio renderers whenever it receives. Additionally, it creates elementary stream frames (video, audio) and stores them in a buffer. The frames are stored in the buffer with their Presentation Time Stamp (PTS). The time stamp information is then used by synchronization module to synchronize the DVB reception buffer with the P2P receiver's buffers. Then, synchronization module's clock starts, which determines the time stamps (PTS) of the frame to be sent to the decoders. As the DVB reception module receives timestamp from the synchronization module, it sends the related frame to the decoders (Figure 28).

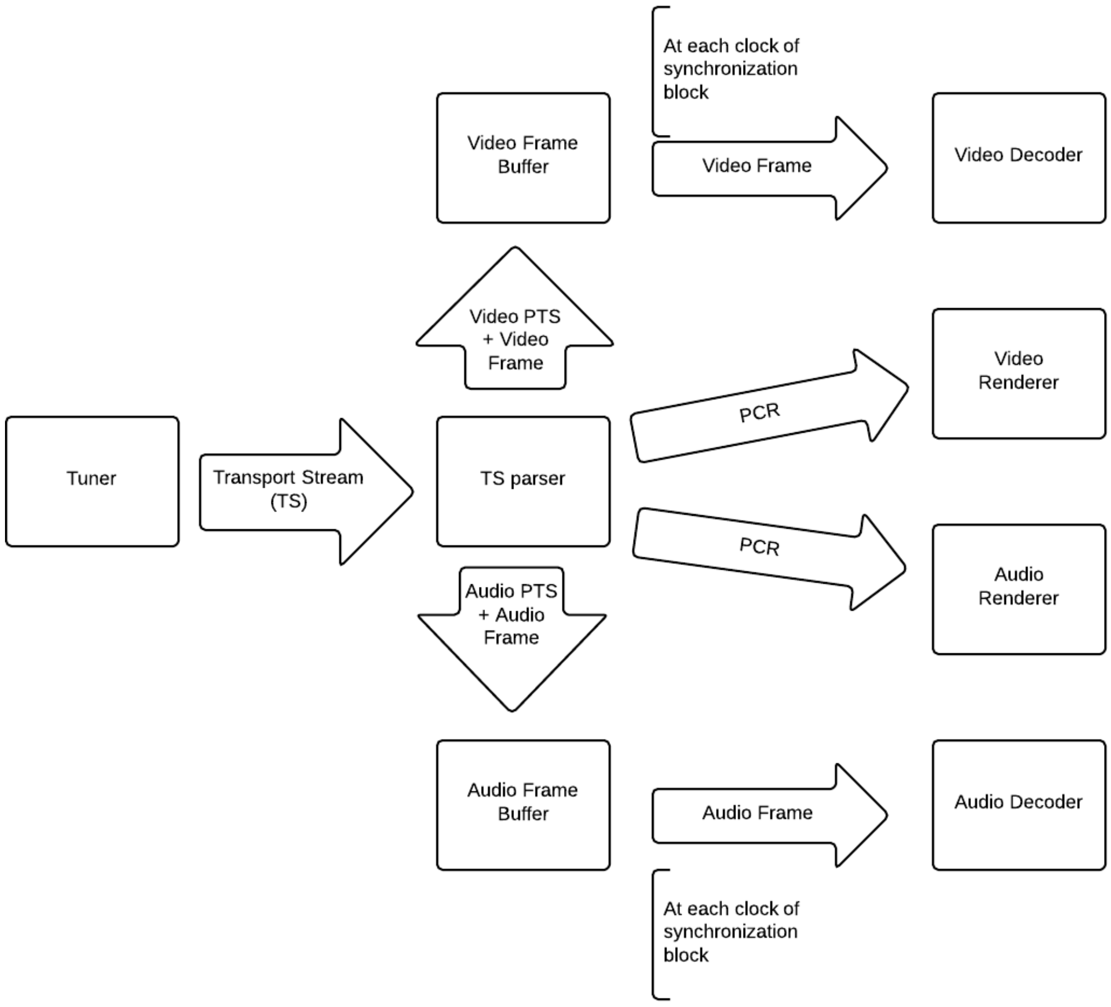


Figure 28 - DVB Reception Block Diagram

3.2.2 Mobile Terminals

The reception of DVB signals on Mobile Terminal Device is possible, when using USB DVB tuner. In ROMEO project the tuner PCTV nanostick 290e is used. With this tuner it is possible to receive DVB-T and DVB-T2 signals. This device uses special Linux driver. This driver is present in Kernel 3.0 or higher versions.

Reading from the PCTV nanostick is realized with the multimedia framework Gstreamer. For this project the standard version of this framework is not used, but also some additions to support Texas Instruments OMAP platforms. Special plugin for this framework can read from DVB devices. It is comfortable to use this plugin, because it will be easier to process the data later. Another importand element in receiving the DVB is the demuxer. It demuxes transport streams to audio and video elementary streams. Once these streams are extracted, video data is passed to the video decoder through suitable parser, depending on the video format. It is better to use the hardware acceleration of the OMAP platforms. This will improve performace of the host device and also rendering the video to the display. In order to use these accelerators a special plugin for gstreamer is needed, called gst-ducati. This plugin uses two importand libraries: libdce and libdrm. After this decoding the output data is in format NV12 and it is suitable for direct rendering. For rendering the video the OMAP SGX graphical processor is used. The audio data is passed to standard audio decoder, depending of the format of the audio stream. Pulseserver is used for audio sink element. The process of DVB reception, decoding and rendering is shown in the diagram in Figure 29.

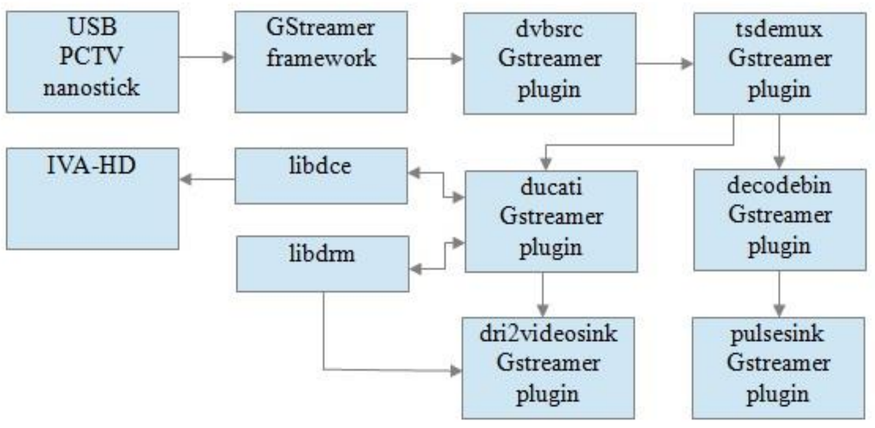


Figure 29 - Overview process of DVB reception, decoding and rendering

With this implementation of DVB reception, decoding of SD TV channels (PAL resolution: 720x576) at 25 frames per second is achieved.. Decoding of FULL HD channels (1920x1080) is still not developed. The playback is almost smooth, but there are some dropped frames. The reason for this problem is still not known. Audio and video streams have to be synchronized, but there is some mismatch between them (about a second). This may be because of dropped frames. Chart with some host device performance parameters is shown in Table 5.

Decoding	CPU Usage [%]	RAM Usage [k]
Video	25	21956
Audio + Video	36	24652

Table 5 - Performance for DVB-Reception

As the chart shows, using this technology and hardware accelerators improves performace of the host device. Just for reference, using software decoder only for video decoding increase CPU usage to about 85%.

3.3 Synchronisation

In ROMEO platform synchronisation is required at several places. Simultaneous reception of content by users requires the following synchronisations to be achieved:

- Play-out synchronisation
- Collaborative synchronisation

3.3.1 Play-out Synchronisation

3D content will be delivered through DVB and P2P networks. Multi-view 3D experience will be achieved by collecting all data from these networks and by rendering it synchronously on terminals. Play-out synchronisation module will be responsible for this task. Firstly, it will receive transport stream data from DVB reception and P2P reception modules. Secondly, it will parse the data as audio and video content and check their timings with respect to DVB timing information. As a third phase it will buffer audio and video content separately to send the content to the related decoders without interruption.

Synchronisation is based on DVB signals. ROMEO platform presumes that DVB will be always available. As a result, PCR information in DVB will be used as base timing information for all views. Each view will be delivered with the corresponding timing information and the synchronisation component will buffer all required data to synchronise all views.

Transport stream data contains video, audio and metadata information. Synchronisation block is expected to demultiplex these data and store them accordingly. Video data from all views will be collected under video buffer whereas audio data will be collected under audio buffer. All buffered data will be transmitted to audio and video decoders. Each data packet will be stamped again with proper timing information. As a result video and audio decoders will also know which packet is for which timing interval.

Play-out Synchronisation module has several functional blocks to execute particular tasks and also uses UDP ports for streaming and TCP ports for control communication. Figure 30 summarises the concept.

- *Communication Interface* block is responsible for all communications related to either configuring the blocks or sending control messages to other components of ROMEO platform.
- *Stream Input Interface* blocks are mainly the front end of Play-out Synchronisation module. They receive IP packets from DVB reception component and Decryption component. DVB front end path has an extra block to extract PCR info from received transport stream. This PCR information is used as base timing information through play-out synchronisation module.
- *Demux Block* is mainly a demultiplexer to parse audio and video data from all front ends. Demux block feeds video and audio buffers with active content.
- *Buffers* are going to be used as synchronisation elements to keep all users to have same content at the same time. There are possible risks of buffer underrun and overrun scenarios. These blocks communicate with A/V adaptation component of ROMEO to have reliable and continuous streaming.
- *PCR Sync Block* will be the final control point before transmitting streams to media decoding components. *PCR sync* uses Collaboration synch delay value and buffer status. It combines with reference timing information and delivers Video and Audio elementary streams to stream output Interface block.
- *Stream Output Interface* is basically a UDP port for video and audio elementary streams.

All streaming data is IP packets with Transport Stream (TS) packets as payload at input side. On the other hand streaming data is elementary streams on output side. Besides streaming there are Javascript Object Notation (JSON) messages. Coordination with other ROMEO components will be handled through these messages.

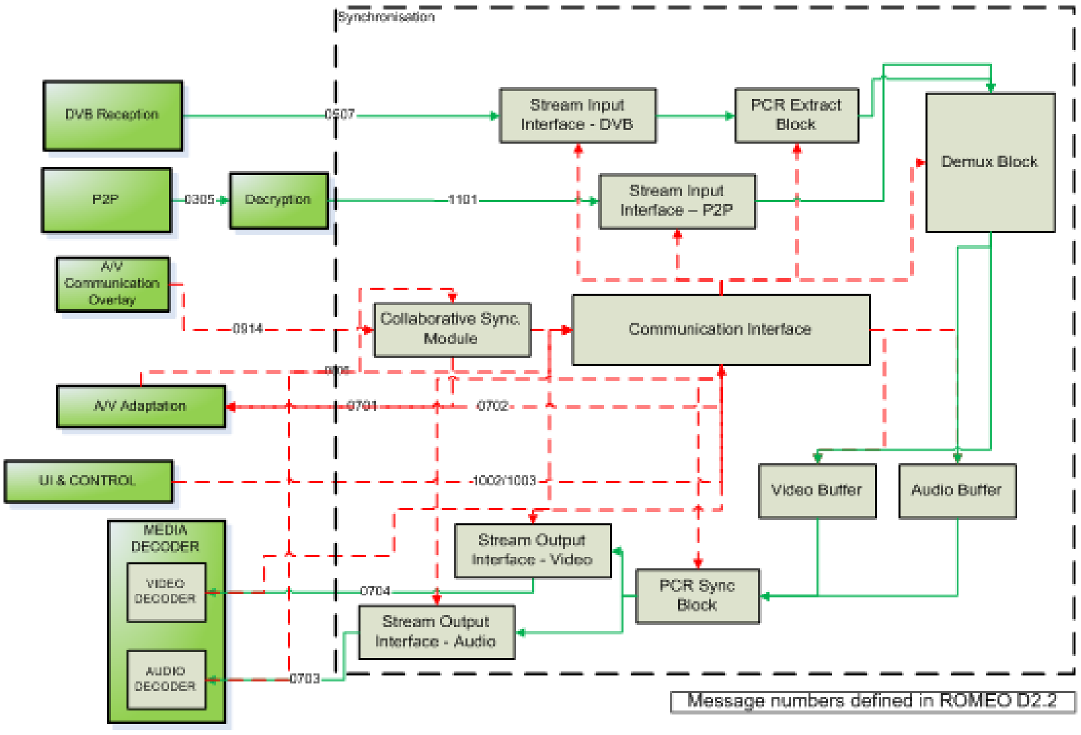


Figure 30 - Play-Out Synchronisation Blocks

3.3.2 Collaborative Synchronisation

This module is implemented as a configurable buffer unit between the network components, P2P transmitter/receiver and DVB receiver, and media decoder units. To achieve synchronisation among collaborating users, this module inserts delay calculated by *Collaborative Synchronisation Controller* unit. This total delay is calculated per consumed content and is not changed during the consumption period of the content.

3.4 A/V Communication Overlay

A/V Overlay software allows user to connect to OpenMeetings server and perform basic operations on server. Client application is written in Java, because the used protocol for communication with the server is RTMP (AMF3) and in Java there is a good implementation for this protocol.

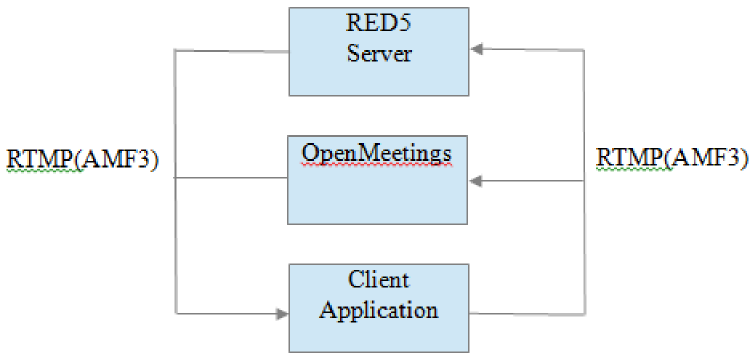


Figure 31 - Message communication between OpenMeetings server and Client

However, because of performance reasons, Java is used only for server-client message communication.(Diagram is shown in Figure 31). For audio and video transfer, GStreamer is used. This is multimedia framework, which enables encoding and decoding of multimedia streams. Usage of this framework improves performance of the audio and video communication significantly. Another advantage of this technology is that it is open source, it is multi platform and there is a lot of useful documentation. Process for communication is described below and is shown in Figure 32. Client makes RMI Java methods calls to OpenMeetings server over RTMP. These calls are used for user connecting to server, user parameters, user authentication, users in rooms and etc. Once, the connection between server and user is established, the Java layer calls a Python application, which manages GStreamer processes to handle audio and video data. GStreamer uses special plug-in, which read user camera data through video for Linux framework. Then another plug-in encodes data in flash video format. Finally encoded data is published directly to the server over RTMP protocol. For decoding, audio and video streams are received over RTMP protocol. Then one GStreamer plug-in decodes the video stream (Flash Video) and another plug-in decodes the audio stream (Nelly Moser). Finally the video streams are showed directly on the screen and audio stream is played through Pulse server, which enables audio playing from many users at the same time. The process is shown in Figure 32.

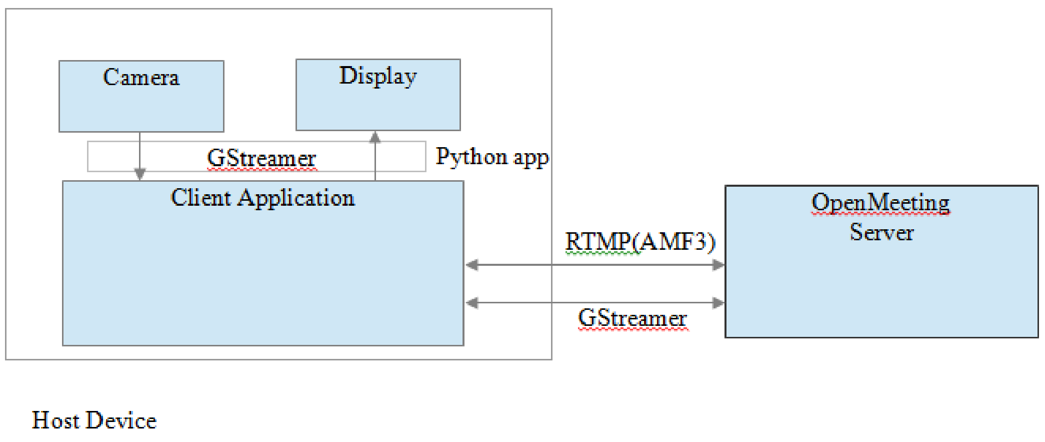


Figure 32 - Gstreamer usage for audio and video transfer

Some performance tests were made on this system. In Table 6 performance parameters are shown on the host device for four video decodings and one encoding.

	CPU [%]	RAM [k]	FPS Cam A	FPS Cam B	FPS Cam C	FPS Cam D	FPS CamPanda
1 Decoding	57	12956	9	-	-	-	19
2 Decodings	70	21512	6	15	-	-	19
3 Decodings	90	30804	6	14	19	-	15
4 Decodings	121	38992	6	14	19	12	15

Table 6 - A/V Overlay Performance Test Results

Notes:

- Host device is with dual core CPU, so CPU usage parameter is allocated between the two cores.

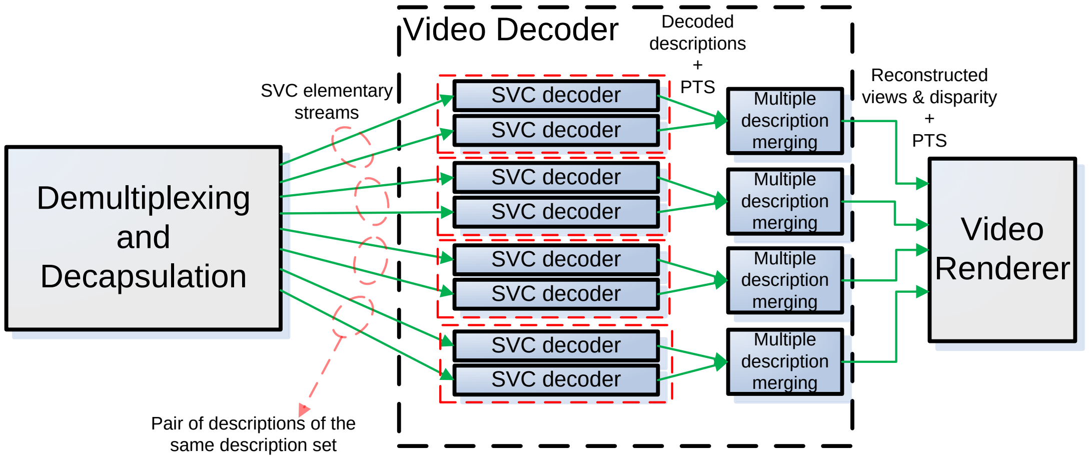
- The results depend on the server usage, network connection, used cameras, light conditions and others.

Final result is good communication between users in OpenMeetings room with synchronized audio and video. Depending on the different environment of the users, video with up to 30fps may be achieved.

3.5 Video Decoding

3.5.1 Video decoding for fixed and portable terminals

The video decoder platforms used in the fixed and the portable client terminals for the real-time demonstrator are identical. The decoder unit is composed of two sub-units: core decoders (multiple SVC decoders running in parallel for each delivered description) and multiple-descriptions merging sub-units (each sub-unit in charge of combining the two descriptions of the same description set). These two sub-units operate in parallel, the core decoders feeding the input of the multiple-descriptions merging sub-units with decoded description frames appended with the associated Presentation Time Stamp (PTS). Figure 33 depicts the overview of the decoder platform architecture.



Video Decoder
Decoded descriptions + PTS
SVC elementary streams
Demultiplexing and Decapsulation
SVC decoder
Multiple description merging
Reconstructed views & disparity + PTS
Video Renderer
Pair of descriptions of the same description set

Figure 33 - Video decoder for fixed and portable terminals

The number of concurrently running SVC decoder instances can be scaled depending on the context (adaptation mode, i.e., whether all views are received for multi-view display or a subset of views are retrieved for stereoscopic rendering) and the terminal type. For fixed terminals, the maximum number of concurrently running SVC decoder instances is 8 (4 views + 4 disparity maps), whereas for portable terminals, according to the ROMEO requirement settings the maximum number of concurrently running SVC decoder instances is 4. Fixed terminal is envisaged to consist of 2 to 4 average 2-core work stations (> 2.8 GHz), depending on the eventual decoding and merging speed of all descriptions.

The core SVC decoders are based on the OpenSVC project (found freely in http://sourceforge.net/projects/opensvcdecoder). Multiple-descriptions merging sub-unit is written in C++, that runs in parallel with a pair of SVC decoder instances. The means of synchronisation between the decoders and the MDC merging sub-units are the PTS information appended to each decoder description packet. The communication means between the SVC decoders and the Demuliplexing & Decapsulation block, and between the SVC decoders and the MDC merging blocks is TCP. The communication means between the MDC merging sub-units and the video renderer is TCP over infiniband network adapter. More information of the video decoder architecture for fixed and portable terminals is given in Deliverable 4.3 (Report on the developed audio and video codecs).

The decoder platform for SI-based multi-view coding and distribution is different from that of the fixed and portable terminals' decoder architecture. As mentioned earlier in Section 2.1.3.1, the SI-based codec is not an integral part of the real-time ROMEO system demonstrator. The SI-based decoder is integrated with an in-built video renderer, which is based on the MPEG View Synthesis Reference Software (VSRS) tool that works non real-time. Information on that decoder platform is also provided in Deliverable 4.3 (Report on the developed audio and video codecs).

3.5.2 Video decoding for the mobile terminal

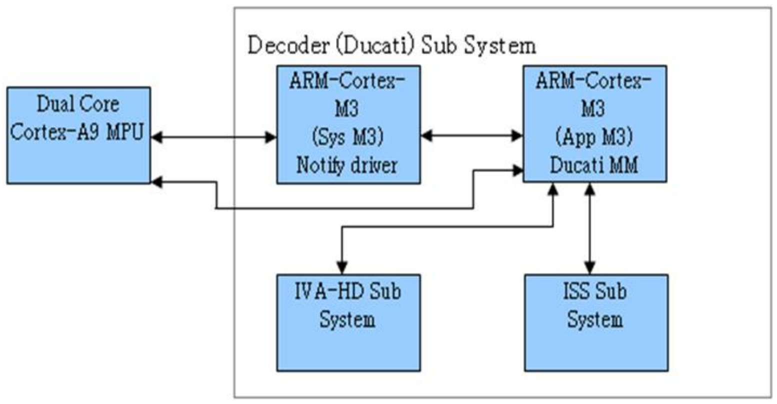

Decoder (Ducati) Sub System
Dual Core Cortex-A9 MPU
ARM-Cortex-M3 (Sys M3) Notify driver
ARM-Cortex-M3 (App M3) Ducati MM
IVA-HD Sub System
ISS Sub System

Figure 34 - Overview diagram of Ducati subsystem used to perform video decoding

Ducati is a hardware accelerator subsystem, which performs most of the video and image coding and processing operations on the Open Multimedia Application Platform (OMAP) processor. An overview diagram is depicted in Figure 34. Ducati subystem comprises two ARM® Cortex-M3 processors (CPUs), Image Video Accelerator – High Definition (IVA-HD) subsystem and Imaging SubSystem (ISS). The two Cortex-M3 processors are known as "Sys M3" and "App M3" and they constitute the MPU of Ducati Subsystem. Sys M3 runs the notify driver, which accepts commands from the High-Level Operating System (HLOS) software and then provides these to the other Cortex-M3 processor (App M3). The entire Ducati multimedia software is executed on App M3. OpenMax^{TM} (OMX) components on the App M3 processor invoke the necessary APIs for video or image processing on the IVA-HD or the ISS subsystems.

IVA-HD is composed of hardware accelerators, which enable video encoding and decoding up to 1080 lines (progressive/interlaced) at 30 frames per second (fps). ISS deals with the processing of pixel data coming from an external image sensor, or from the memory. ISS is one of the most important camera subsystem components which is included instill image capturing, camera view finder and video record use-cases.

The video decoder on the mobile terminal device supports up to 1080p at 30fps video decoding. It supports hardware acceleration for the folowing formats:

- H.264/AVC: Constrained Baseline Profile, Main Profile, High Profile (For 1080p30 decoding, bit-rates up to 20 Mbps are supported, which is enough concerning ROMEO requirements)
- MPEG-4 Visual: Simple Profile (SP) and Advanced Simple Profile (ASP)
- H.263: Profiles 0 and 3
- MPEG-2: Upto Simple Profile Main Level, Main Profile High Level
- VC-1: Simple Profile Low Level, Simple Profile Medium Level, Main Profile Low Level, Main Profile Medium Level, Main Profile High Level, Advanced Profile Level 0, Level 1, Level 2 and Level 3

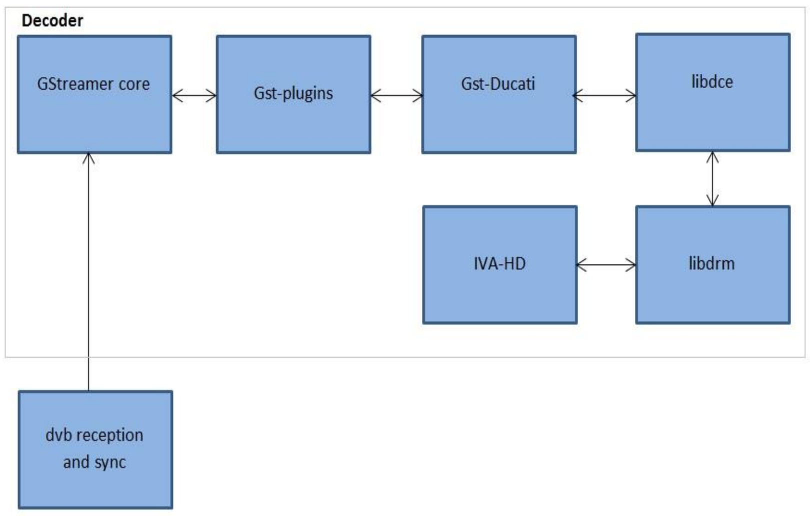


Figure 35 - Functional blocks for video decoding process

GStreamer is open source multimedia framework for creating media applications. The GStreamer core function is to provide a framework for plugins, data flow and media type handling. Gstreamer is based on plugins, known as Gst-plugins, that will provide the various functionalities. Gst-Ducati is GStreamer plugin, which enables using of Ducati video decoder hardware acceleration. It uses two importand libraries:

- libdce (Distributed Codec Engine) - it provides remote access to hardware acceleration for decoding

- libdrm (Direct Rendering Manager) - it provides routines for the X Window System, which is a computer software system and network protocol that provides a basis for Graphical User Interfaces and rich input device capability for networked computers, to directly interface with video hardware.

- IVA-HD (Image Video Accelerator - High Definition) are hardware accelerators in OMAP processors, which enables HD video decoding and stereoscopic 3D.

The functional block diagram is given in Figure 35.

3.5.3 Audio Decode and Rendering

3.5.3.1 Interface from Audio Decoder to Audio Renderer

The transport of decoded audio data to the corresponding renderer engine is implemented via IP transmission. This allows a physical separation of the hosts for rendering and decoding. For renderers running on the same host as the decoder data packets are sent to the local loopback. Each decoded audio stream goes out to an individual port using the UDP transport protocol. The packets have got a header containing information about the samples included. The time reference for synchronization – the Presentation Time Stamp (PTS) – is put in here, too. Renderers on mobile and portable terminals receive the program clock reference (PCR) on a TCP network socket. The PCR is packetized in the JSON format. The audio samples are getting buffered and rendered when the PCR matches the PTS value.

3.5.4 Audio Decoder

The audio decoder software application has been developed and distributed. Named aacDecoderApp.exe, this decodes incoming AAC frames and sends the decoded frames to another terminal over UDP/TCP connection. It accepts the following as the input parameters:

- **protocol** (UDP or TCP)
- **SrcPort** (port number to receive the frames from)
- **DstIPAddr** (destination IP address)
- **DestPort** (destination port number)
- **WriteToFilename.pcm** (the first channel of the decoded frames is written as a local file in raw PCM format for later playback and debugging)

The format of each incoming aac packet is supposed as a complete AAC encoded frame prefixed with 8-byte PTS information as in Figure 36:

PTS (8 bytes)	Encoded AAC frame (variable length)

Figure 36 - Incoming audio packet structure

The PTS is increased by 1920 for each following frame. The number of 1920 comes from this assumption: 48 KHz sampling rate, 1024 samples was encoded in each frame; the PTS is 90 KHz, so that the frame time in terms of PTS ticks is 1024/48000 * 90000 = 1920, which is equivalent to 21.333 ms.

The output data for each decoded audio frame has the format as in Figure 37 and Figure 38.

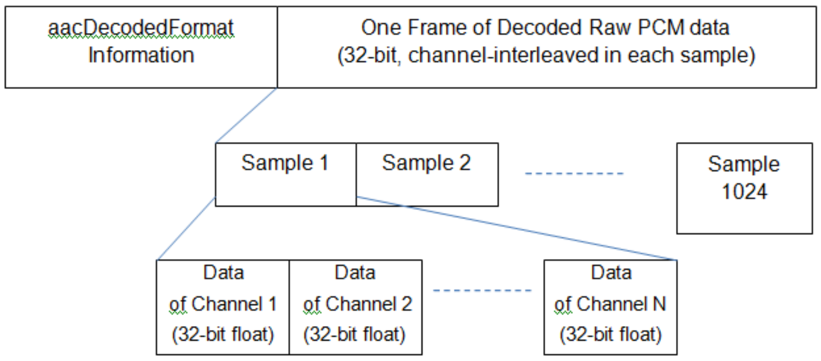
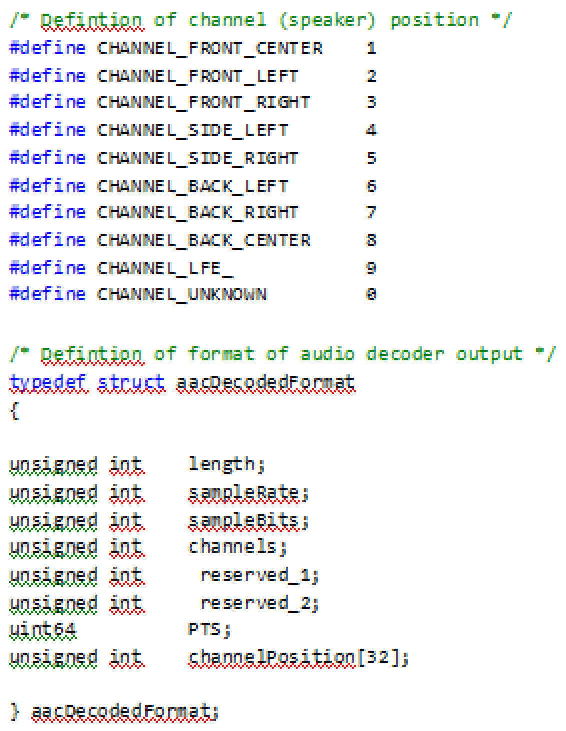

aacDecodedFormat Information
One Frame of Decoded Raw PCM data (32-bit, channel-interleaved in each sample)
Sample 1
Sample 2
Sample 1024
Data of Channel 1 (32-bit float)
Data of Channel 2 (32-bit float)
Data of Channel N (32-bit float)

Figure 37 - Decoded output audio data structure

The aacDecodedFormat Information is a group of structured data which includes the following information:

- **length**: the total number of bytes in this decoded audio frame including the the structured data shown in Figure 37
- **sampleRate**: sampling rate of the audio data, default 48,000 Hz
- **sampleBits**: number of bits in one sample, 16 or 32 bits. 24-bit is packed in 32-bit integer
- **channels**: number of channels in the decoded audio frame
- **PTS**: the PTS of this decoded frame. 8 bytes in length

- **channelPosition**[]: an array of unsigned integers to indicate the positions of each channel in the decoded audio frame. For example, if channelPosition[0] == 3, it indicates that the first 4-byte float data in each sample is for the Front-Right channel (seeFigure 37).



```
/* Defintion of channel (speaker) position */
#define CHANNEL_FRONT_CENTER    1
#define CHANNEL_FRONT_LEFT      2
#define CHANNEL_FRONT_RIGHT     3
#define CHANNEL_SIDE_LEFT       4
#define CHANNEL_SIDE_RIGHT      5
#define CHANNEL_BACK_LEFT       6
#define CHANNEL_BACK_RIGHT      7
#define CHANNEL_BACK_CENTER     8
#define CHANNEL_LFE_            9
#define CHANNEL_UNKNOWN         0

/* Defintion of format of audio decoder output */
typedef struct aacDecodedFormat
{

unsigned int    length;
unsigned int    sampleRate;
unsigned int    sampleBits;
unsigned int    channels;
unsigned int     reserved_1;
unsigned int     reserved_2;
uint64          PTS;
unsigned int    channelPosition[32];

} aacDecodedFormat;
```

Figure 38 - Description of the structure of aacDecodedFormat Information within the decoded output data

3.5.5 Renderer for Fixed Terminal

As previously introduced in D6.2, the default audio rendering format for the fixed terminal is the 5.1-channel configuration. The 5.1-channel audio renderer has been developed, integrated with the decoding module, and is in working order, utilising ASIO (Audio Stream Input/Output) protocol with USB or Firewire connection between the audio rendering hardware and the controlling PC. This conventional operation 5.1-channel rendering is described in D3.6 in detail. The synchronisation feature has been implemented, which enables the decoded frames to be passed to the renderer with the corresponding JSON PCR messages, such that the video and audio frames with the same timing information can be rendered at the same time.

An advanced immersive spatial audio renderer based on WFS has been implemented in a studio. The feasibility of the 3D audio technique has been numerically and experimentally validated with broadband audio sources in the listening room. Various scenarios with audio object localisation have been demonstrated, e.g., multiple virtual sources, a focused source, and a 360° moving source. The uncertainty and sensitivity of the system have been evaluated in the measurement to guarantee the accuracy of output results. In addition, subjective evaluations with the audio objects also have been conducted in the studio.

3.5.6 Renderer Interface for Portable and Mobile Terminal

The renderer interface module allows other modules to interconnect to the renderer on mobile as well as portable terminals. Besides the audio samples the renderer interface receives also the scene description updates (including the corresponding timestamp as PTS) and the incoming PCR messages. The scene description is formatted in SpatDIF. Synchronization between the renderer and other modules is achieved by matching the PCR to the SpatDIF- and audio packets of similar presentation time stamps.

The renderer is now capable of parsing the scene description language and translating the information into rendering commands, which means calculating proper filters, impulse resonses and other parameters needed to generate the audio output. SpatDIF commands are accepted live over the UDP socket receiving the packets from the demux module or alternatively from a SpatDIF formatted file for offline usage.

Headtracking is now possible on portable devices as well as mobile ones. The first prototype of the mobile head tracker is currently in the testing phase. Instead of tracking the gyration angle of the head from an absolute position, the mobile head tracker uses three different measurements to calculate the relative gyration angle.

3.5.7 Renderer for Portable Terminal

This renderer will be used on the portable terminal for binaural rendering of mainly object-based audio.

In the reporting period the binaural rendering of object-based scenes on the portable terminal have been optimized and refined. This allows rendering more complex scenes and higher number of objects. Furthermore, the rendering algorithms have been extended and optimized to allow moving sound objects and dynamic adaption of ADL scene properties. All core module draft implementations – as described in deliverable D6.2 – have been extended to form an integrated rendering engine suitable for the Portable Terminal. The synchronization module described in 3.3 and the interface to the audio decoder module have been integrated. Therefore the rendering of time-varying object based audio scenes is now possible, both for file-based and stream-based ADL scenes.

Future work will be mainly focussed on integrating the audio rendering modules with other non-audio modules to finalize the Portable Terminal.

3.5.8 Renderer for Mobile Terminal

This renderer will be used on the mobile terminal for binaural rendering of mainly 5.1 surround sounds.

As the basic principles of binaural rendering of multi-channel surround sound are known, the main challenge of this task is to transform and port existing high end and computationally expensive solutions to mobile platforms. As development platform for the mobile terminal an OMAP4 ARM architecture was chosen, namely a Pandaboard by Texas Instruments. This board allows low latency audio processing on a dual core ARM processor as it is often used in mobile devices. The CPU power is sufficient for running a rendering job with continuous head tracking, which could have been verified in first test runs. Several optimizations on software level reduced the CPU load by round 30%. Further optimizations are possible by integrating the newest versions of the libraries for audio processing and convolving. This requires an upgrade of the OS to a version supporting GCC 4.7 as a minimum.

The platforms supported in the moment are Linux derivates. After initial tests of the mobile head tracker, this device as well as the software components must be integrated with the mobile terminal platform provided by MMS.

3.6 Video Rendering

3.6.1 Video rendering for fixed terminal

In the list of displays targeted as fixed terminal within the ROMEO project, the most demanding ones in terms of video processing are multi-view displays (e.g. Alioscopy or

Dimenco). Since more than four views are required, there is a need for view interpolation using incoming video streams associated with incoming disparity maps. A MVD4 (MultiView 4 views + 4 Depth) multi-view rendering scheme is illustrated in Figure 39. It uses the full range of video and disparity information sent through the ROMEO chain. A MVD2 version of it is only using two videos and the two associated disparity maps.

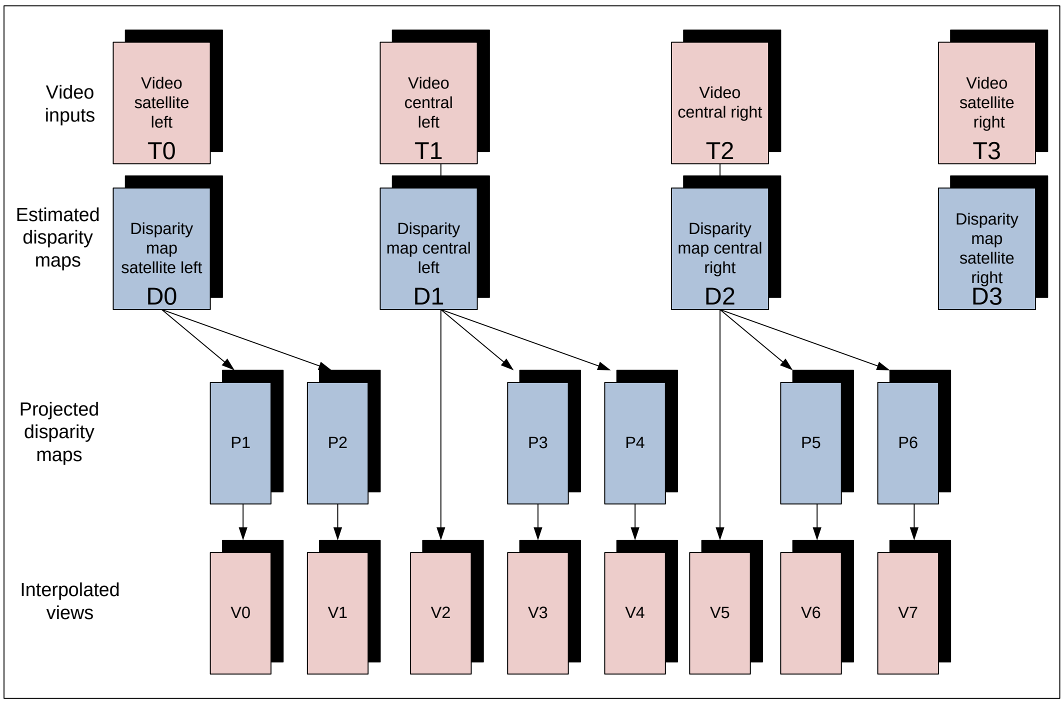


Figure 39 - MVD4 multi-view rendering scheme

The architecture of the video renderer has been defined to achieve in real-time view interpolation processing. A General-Purpose Graphic Processing Unit (GPU) is used to speed-up the processing.Figure 40 illustrates the global architecture of the renderer inside the GPU.

A first part of the GPU is dedicated to synchronization and decision processing to ensure the right mode is rendered and correctly synchronized with the rest of the system (including audio). A PCR information is decoded to know when a given time stamp has to be displayed. Following this extraction, the PTS information is decoded from incoming video/disparity streams. The different buffers receive the 8 streams in a non synchronized way. The PTS decoding is used to re-synchronize the used streams before sending them to the renderer.

Once the streams are read from the different buffers in a synchronized way, the decision block select the ones which are presented and used for the rendering. A full range MVD4 rendering requires the whole set of video and disparity whilst a reduced MVD2 mode requires only two video and one disparity map. “Video selection” and “Disparity selection” blocks apply this decision and transmit to the next block the selected streams. The final block (“View Interpolation + display sub pixel shuffling”) receives the video and the projected disparity map to perform the view interpolation.

An effort has been made to adapt the video rendering algorithm to realtime constraints: the algorithm has been implemented on a GPU based processor with some adaptation to fit with hardware constraints. The work done to adapt the basic algorithm is presented in the D3.3 and D3.6 deliverables. The real time aspect has been achieved for MVD2 and MVD4 processing with the range of disparity of ROMEO sequences.

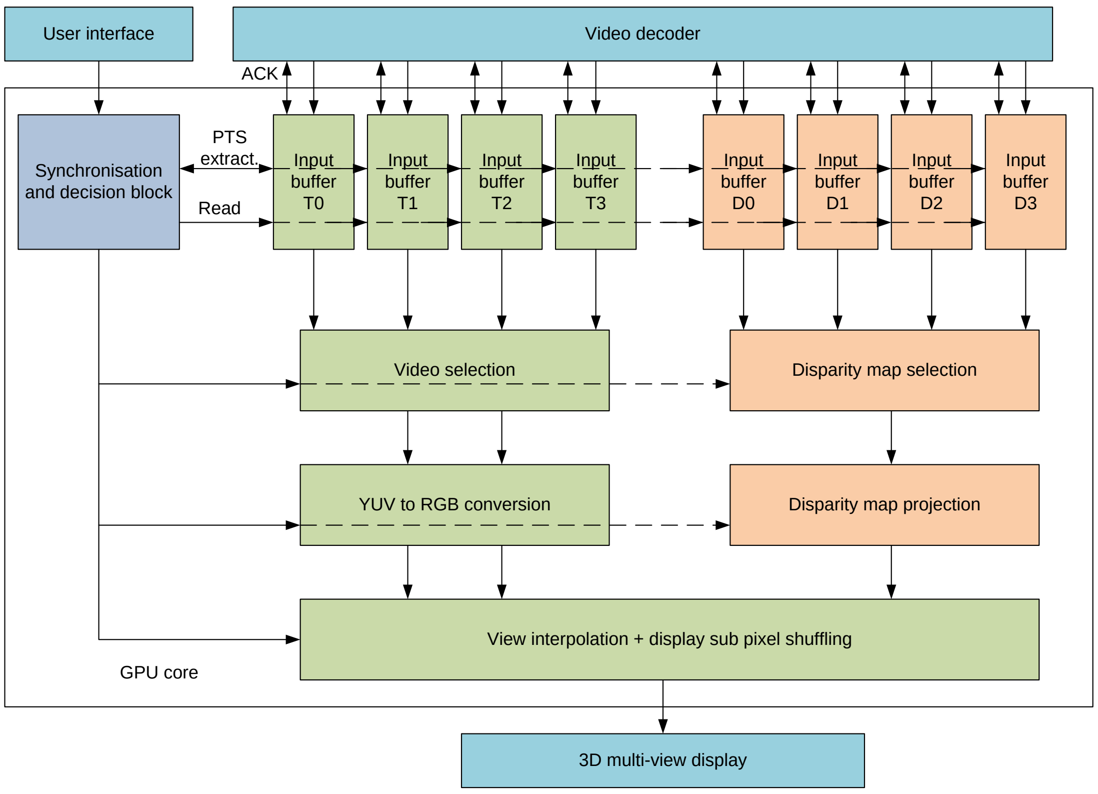


Figure 40 - GPU core architecture

3.6.2 Video rendering for mobile terminal

The output of the mobile terminal video decoder is rendered on the display by the multimedia framework GStreamer. Special video sink element (plug-in) is used in order to enable using of OMAP graphical processing unit (GPU) – PowerVR SGX540. The video sink plug-in uses library called libdrm. This library provides core routines for the X Window System to directly interface with video hardware using the Linux kernel's Direct Rendering Manager (DRM), which is kernel's portion of the Direct Rendering Infrastructure (DRI). The DRI is used on Linux to provide hardware accelerated OpenGL drivers. Without using the DRI and hardware acceleration, the decoder has to use CPU for rendering, which degrades overall performance of the system. The new rendering infrastructure (DRI2) improves several shortcomings of the old design, including removing internal locks and adding proper support for off-screen rendering, so that compositing and XVideo/OpenGL applications are properly managed. On this stage of development only mono video rendering is realized.

Structure of the renderer is showed on the diagram in Figure 41.

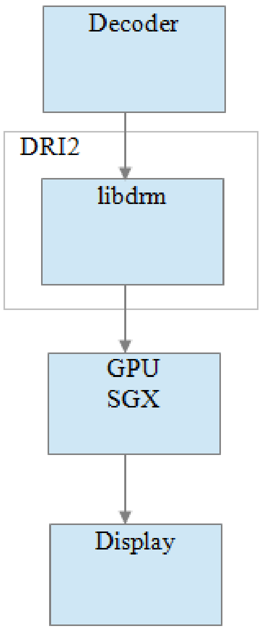


Figure 41 – Mono video rendering structure

3.7 A/V Adaptation and Network Monitor

3.7.1 Audio/video adaptation decision block

Several adaptation scenarios have been defined to ensure the end user is able to request a specific video mode. The network can also be responsible for not being able to deliver all streams. DEPTH_LEVEL and VIEW_POINT are the two parameters the user can modify to adapt the content to his expectation. Based on these parameters, six different video modes are possible to render multi-view contents depending on the number of views (and associated disparity used).

The six different modes are:

- MVD4 (4 views + 3 disparity maps)
- MVD3_1: (3 views + 2 disparity maps using satellite left information)
- MVD3_2: (3 views + 2 disparity maps using satellite right information)
- MVD2_1: (2 views + 1 disparity maps using satellite left + central left information)
- MVD2_2: (2 views + 1 disparity maps using central left + central right)
- MVD2_3: (2 views + 1 disparity maps using central right + satellite right information)

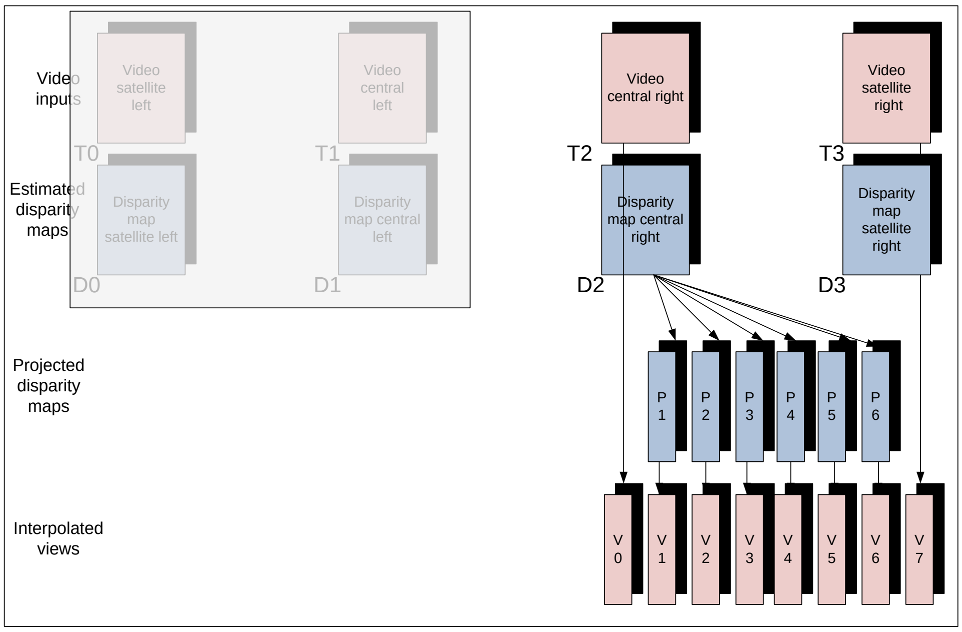


Figure 41 - MVD2_3 scheme

A more detailed explanation of all available mode is described in deliverable D3.4 as well as in the future D3.7 (due in M27). The video rendering block is able to adapt the output to the user requirement choosing the right mode from both DEPTH_LEVEL and VIEW_POINT parameters. “View selection” block of the GPU core receives from the “decision” block the key to select the right input to be sent to the interpolation block. The phase of the interpolation is also sent to the “disparity map projection” block and to the “view interpolation” block.

Another adaptation capability of the system is to address different multi-view displays. The Alioscopy 42” and the Dimenco 55” are the two multi-view displays selected. The system is also able to address stereoscopic display in a side-by-side mode (corresponding to a 2-view display) for the portable application.

3.7.2 Network Monitoring Subsystem

The Network Monitoring Subsystem (NMS) is a software module running at the peer with the following functions:

- Collects peer hardware and software characteristics;
- Collects network traffic statistics (packet loss, average delay, jitter, available bandwidth) for each conection;
- Periodically reports the collected data to its parent (it chooses a different parent in each iteration using a round-robin approach) or to the MTM (in case this is a top level peer).
- Computes the peer stability, a metric defined in ROMEO deliverable D5.1 [1] that reflects the stability potential of this peer based on previous sessions;
- Informs the MTM about detected changes in the network, such as parent disconnection;

The NMS report contains both the information defined in the *NMSReport* class (at the server) and the *Specs* class (at the peer), see Figure 18 and Figure 42. To save resources and simplify socket management at the receivers, reports are sent to one of the parents, who then collects all the received reports during a time-window and sends all collected reports to its own parent (one by one in a sticky TCP connection). This process goes on until the data gets to the highest peers in the P2P tree hierarchy, who then send the bundle of all collected reports to the MTM (NMSCollector submodule).

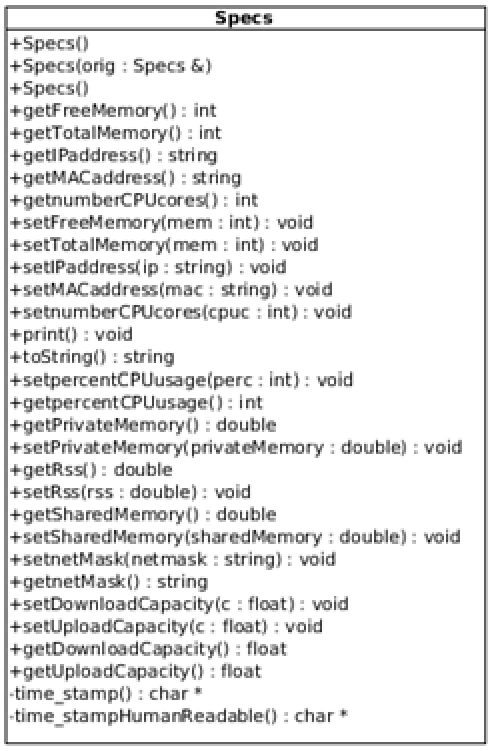
Next section describes the most relevant classes used by the NMS to perform its operations.

3.7.2.1 Provided Classes

The NMS software module is constituted by the following classes:

Specs class

The *Specs* class provides the functions to retrieve the peer's hardware information. These class functions are listed in Figure 42.

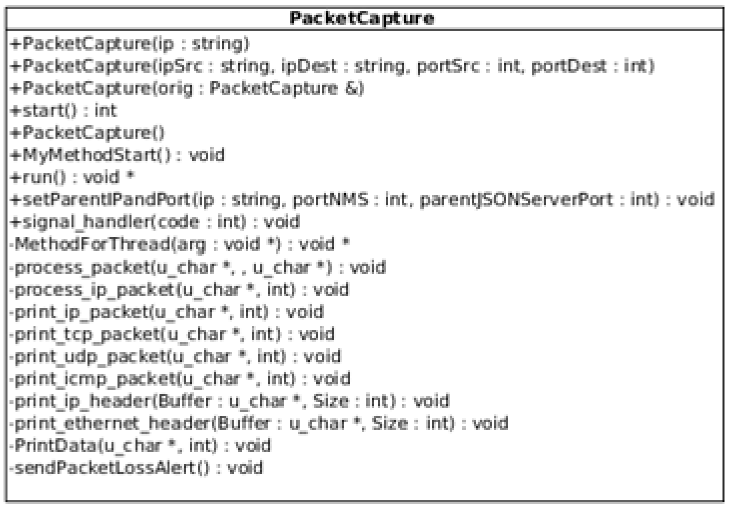


```
Specs
+Specs()
+Specs(orig : Specs &)
+Specs()
+getFreeMemory() : int
+getTotalMemory() : int
+getIPaddress() : string
+getMACaddress() : string
+getnumberCPUcores() : int
+setFreeMemory(mem : int) : void
+setTotalMemory(mem : int) : void
+setIPaddress(ip : string) : void
+setMACaddress(mac : string) : void
+setnumberCPUcores(cpuc : int) : void
+print() : void
+toString() : string
+setpercentCPUusage(perc : int) : void
+getpercentCPUusage() : int
+getPrivateMemory() : double
+setPrivateMemory(privateMemory : double) : void
+getRss() : double
+setRss(rss : double) : void
+getSharedMemory() : double
+setSharedMemory(sharedMemory : double) : void
+setnetMask(netmask : string) : void
+getnetMask() : string
+setDownloadCapacity(c : float) : void
+setUploadCapacity(c : float) : void
+getDownloadCapacity() : float
+getUploadCapacity() : float
-time_stamp() : char *
-time_stampHumanReadable() : char *
```

Figure 42 - The Specs class and its functions

PacketCapture class

The *PacketCapture* class is responsible to collect network data such as, connection history (start time, duration) or traffic statistics (packet loss, delay, jitter), by exploiting the libpcap [3] library. These class functions are listed in Figure 43.

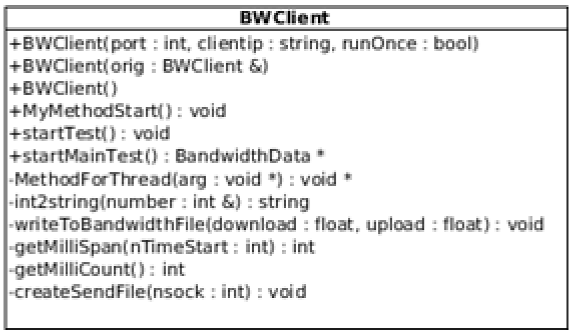


```
PacketCapture
+PacketCapture(ip : string)
+PacketCapture(ipSrc : string, ipDest : string, portSrc : int, portDest : int)
+PacketCapture(orig : PacketCapture &)
+start() : int
+PacketCapture()
+MyMethodStart() : void
+run() : void *
+setParentIPandPort(ip : string, portNMS : int, parentJSONServerPort : int) : void
+signal_handler(code : int) : void
-MethodForThread(arg : void *) : void *
-process_packet(u_char *, , u_char *) : void
-process_ip_packet(u_char *, int) : void
-print_ip_packet(u_char *, int) : void
-print_tcp_packet(u_char *, int) : void
-print_udp_packet(u_char *, int) : void
-print_icmp_packet(u_char *, int) : void
-print_ip_header(Buffer : u_char *, Size : int) : void
-print_ethernet_header(Buffer : u_char *, Size : int) : void
-PrintData(u_char *, int) : void
-sendPacketLossAlert() : void
```

Figure 43 - The PacketCapture class and its functions

BWClient class

The *BWClient* class is responsible, at the peer, to perform the link test towards the MTM running at the server/super-peer. These class functions are listed in Figure 44.

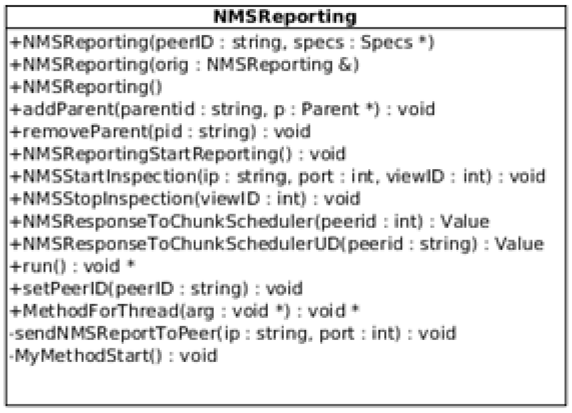


```
BWClient
+BWClient(port : int, clientip : string, runOnce : bool)
+BWClient(orig : BWClient &)
+BWClient()
+MyMethodStart() : void
+startTest() : void
+startMainTest() : BandwidthData *
-MethodForThread(arg : void *) : void *
-int2string(number : int &) : string
-writeToBandwidthFile(download : float, upload : float) : void
-getMilliSpan(nTimeStart : int) : int
-getMilliCount() : int
-createSendFile(nsock : int) : void
```

Figure 44 - The BWClient class and its functions

NMSReporting class

The *NMSReporting* class is responsible for: (i) knowing the list of parents of the peer. (ii) initiating the packet inspection mechanism; (iii) creating the NMS periodic reports; (iv) sending the NMS reports towards a parent (via *JsonServer* class at the TC); (v) collect NMS reports from children peers. These class functions are listed in Figure 45.



```
NMSReporting
+NMSReporting(peerID : string, specs : Specs *)
+NMSReporting(orig : NMSReporting &)
+NMSReporting()
+addParent(parentid : string, p : Parent *) : void
+removeParent(pid : string) : void
+NMSReportingStartReporting() : void
+NMSStartInspection(ip : string, port : int, viewID : int) : void
+NMSStopInspection(viewID : int) : void
+NMSResponseToChunkScheduler(peerid : int) : Value
+NMSResponseToChunkSchedulerUD(peerid : string) : Value
+run() : void *
+setPeerID(peerID : string) : void
+MethodForThread(arg : void *) : void *
-sendNMSReportToPeer(ip : string, port : int) : void
-MyMethodStart() : void
```

Figure 45 - The NMSReporting class and its functions

3.7.2.2 Messages

The NMS module currently sends the following JSON messages as seen in Table 7:

From	To	Message	Description
NMS	NMS or MTM	314-TREE_PROBE	Provides the NMS report to a parent peer (or MTM in case the reporting peer is a top-layer peer)

Table 7 - Messages from NMS to other modules

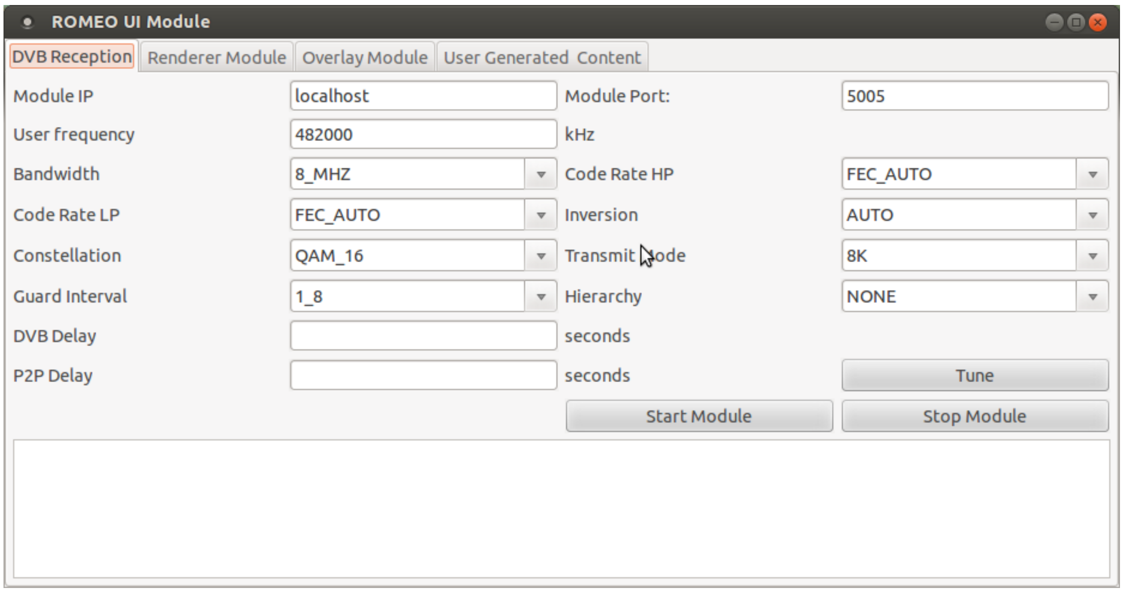
3.8 User Interface and Control

The graphical user interface (GUI) for the fixed and mobile terminals consists of 4 coexisting interfaces (control of the DVB reception (sync) module, renderer and overlay, and user generated content).

3.8.1 DVB Reception UI

The DVB Reception UI allows the control and configuration of the DVB receiver. Frequency, bandwidth and all the relevant DVB configurations are done from this screen (Figure 46).

Also the sync block delay can be configured from this screen (Section 3.2.1).



ROMEO UI Module

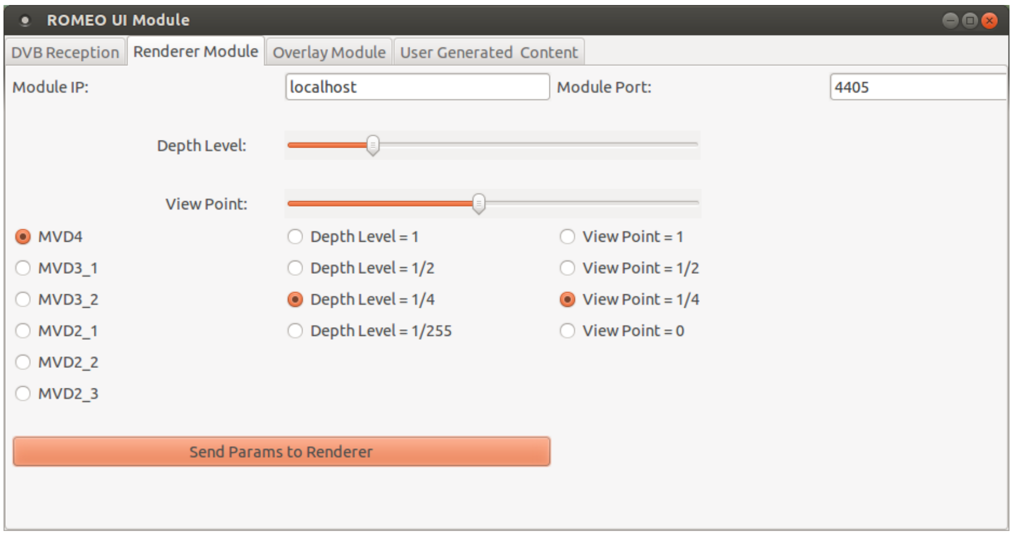
DVB Reception | Renderer Module | Overlay Module | User Generated Content

Module IP	localhost	Module Port:	5005
User frequency	482000	kHz	
Bandwidth	8_MHZ	Code Rate HP	FEC_AUTO
Code Rate LP	FEC_AUTO	Inversion	AUTO
Constellation	QAM_16	Transmit Mode	8K
Guard Interval	1_8	Hierarchy	NONE
DVB Delay		seconds	
P2P Delay		seconds	Tune

Figure 46 - DVB Reception UI

3.8.2 Renderer Module UI

This user interface allows the user to experience the multiview 3D aspect of the ROMEO project. Depth level and view point can be configured from this interface. Also the streams to be used can be forced/configured from this screen (Figure 47).



ROMEO UI Module

DVB Reception | Renderer Module | Overlay Module | User Generated Content

Module IP: localhost Module Port: 4405

Depth Level:

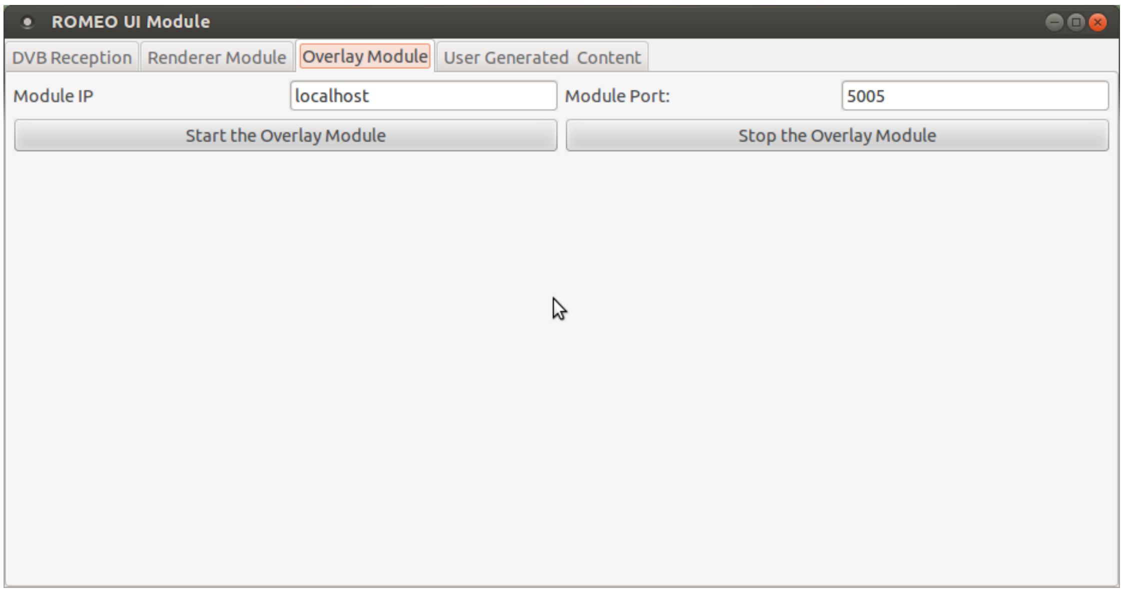
View Point:

[✓ MVD4]	Depth Level = 1	View Point = 1
MVD3_1	Depth Level = 1/2	View Point = 1/2
MVD3_2	[✓ Depth Level = 1/4]	[✓ View Point = 1/4]
MVD2_1	Depth Level = 1/255	View Point = 0
MVD2_2		
MVD2_3		

Figure 47 - Renderer Module UI

3.8.3 Overlay Module UI

This interface is used to start the overlay module to allow the users to collaborate, sync and chat via video and text (Figure 48).



Module IP: localhost
Module Port: 5005

Figure 48 - Overlay Module UI

3.8.4 User Generated Content UI

The user generated content GUI uses login mechanism for user authentication purposes, which is user name and password and will be inserted in front of the payload for each transaction with the server.

For the file upload procedure the GUI lists the users participating in the overlay communication, see ROMEO deliverable 5.3 [16]. A subset of the participants can be selected, together with a local file, to be uploaded to the UGC server. After uploading the file, all selected participants are able to download it.

For file download, the user interface constantly polls the server (as described in [15]) for the files that are available to the user. The user can select which transcoding format is requested, and then download the selected file from the server.

4 NETWORK RELATED COMPONENTS

4.1 Mobility

The Mobility Component comprises three basic modules and they are described in the following sections:

4.1.1 Media Independent Handover (MIH)

4.1.1.1 Module description

The MIH module performs the following tasks:

1. It periodically monitors all available access networks as well as all active mobile nodes and collects parameters such as network load, mobile node throughput and signal strength.
2. It uses XML format to encode and decode the MIH reports between the monitored entities and the MIH server.
3. It creates the UDP sockets that are used for the communication with the MIH server.
4. It stores the values of the monitored parameters in a database at the server side.
5. It performs handover management by sending MIH commands from the MIH server to the mobile node.

4.1.1.2 Provided source files

The MIH module functionalities are distributed across the following network entities.

1. At the server side

`mih_udp_alert.{h,c}`

At the MIH server side the mih_udp_alert.{h,c} source files contain the code that creates a UDP socket and listens for incoming MIH reports from the various access networks and registered mobile nodes. Based on this information the handover management is enforced by sending MIH commands to the mobile nodes.

```
int mih_udp_comm_init();

int mih_udp_create_socket(int, str, unsigned int *);

int mih_udp_comm_start();

void mih_udp_comm_loop();

void mih_udp_receive(int);

void mih_udp_comm_destroy();

void_mih_udp_send_command();
```

`mih_xml_codec.{h,c}`

After receiving the MIH reports which are in XML format the server decodes the messages and retrieves the values of the monitored parameters. To do so, it exploits the libxml2 library which is an XML C parser and toolkit developed for Linux.

```
void mih_xml_decoding(str, mih_xml_info_t*);

int mih_xml_decode_nn (str );
int mih_informer_xml_encode (xml_writer*, mih_report_t*,
MIH_subscription_id_t*);
```

db_mih.{h,c}

Finally, the server updates the database by performing update operations according to the monitored entity reports. For that purpose the libdrizzle library was used which implements the Drizzle and MySQL protocols to support the functionalities of a relational database management system.

```
int Server_db_mih_check_subs_client(MIH_subscription_id_t*, str);

int Server_db_mih_ue_insert_info(str, float*, float*, str*,
MIH_subscription_id_t*);

int Server_db_mih_ue_update_info(float*, float*, str*,
MIH_subscription_id_t, str);

int Server_db_mih_nn_update_info (str*, str*, nn_info_t*);

int Server_db_mih_check_network_node(str*);

int Server_db_mih_nn_insert_info(str*, nn_info_t*, str*);
```

2. At the Access Network side

mih_monitor.{h,c}

At access network side the relative entities (i.e. EPDG and eNodeB) periodically monitor the network load.

```
int mih_network_report();
```

mih_xml_codec.{h,c}

Then the network load value is passed to the xml encoder to form the MIH report message. As previously described, the libxml2 library was used for XML encoding

```
int mih_informer_xml_encode (xml_writer*, mih_report_t*);
```

mih_udp_comm.{h,c}

Finally, the report is sent to the MIH server. To this end, a UDP socket was created to communicate with the MIH server.

```
int mih_timer_udp_alert(void *ptr);

int mih_start_udp ();

static int open_mih_udp_socket(uint16_t ai_family);

static void close_mih_udp_socket(int *socket);

str get_mih_xml (mih_report_t *report);
```

```
int mih_udp_send_msg(void);
int mih_stop_udp (void);
```

3. At the Mobile Node side:

`mih_monitor.{h,c}`

Similarly with the access network entities the mobile node reports its related parameters such as the signal strength and throughput.

```
typedef mih_report_t (*mm_wlan_mih_report_f)();
mih_report_t mm_wlan_mih_report();
```

`mih_xml_codec.{h,c}`

The parameters value are imported into an xml message and sent to the server.

```
int mih_xml_encode_parameter(xml_writer*xml, mih_report_t*, str );
int mih_informer_xml_encode (xml_writer*, mih_report_t*,
MIH_subscription_id_t*);
```

`mih_udp_comm.{h,c}`

Finally, the report is sent to the MIH server over UDP transport protocol.

```
int udp_comm_init();
int udp_comm_start();
void udp_comm_destroy();
```

4.1.2 Media Aware Proxy

4.1.2.1 Module description

The Media Aware Proxy (MAP) is a transparent user-space module used for low delay filtering of scalable multiple description video streams. Its functionality extends from the Network Layer to the Application Layer providing video stream adaptation by taking into account the clients' wireless network conditions. Therefore based on the load of each access point, MAP is able to either drop or forward packets that carry specific layers of a stream to the receiving users. The need for such a middlebox stems from the fact that in a real environment each client must be able to have a guaranteed QoE. Moreover, the need for transparency derives from the fact that each new platform should provide backwards compatibility in order to be able to work with existing networks. So in the current implemented module there is no need to modify neither the client's feedback channel nor the streamer's signaling channel.

4.1.2.2 Provided source files

`media_aware_proxy.{h,c}`

The "media_aware_proxy" module provides the main method which is responsible for the initialization and invoking of all MAP's modules.

```
int main(int argc, char **argv)
void ctrlC(int sig);
void close_map(const char error_message[100]);
void showUsage();
```

map_thread.{h,c}

The “map_thread” is the most important module since it is responsible for the traffic management.

- It binds to the “Netlink” socket and receives IP/UDP/P2P packets by exploiting the libraries of “ip_queue” module and “Netfilter” mechanism.
- It identifies the packet address by parsing the UDP/IP headers.
- It identifies the video information by parsing the P2P header through “p2pParser” module.
- It identifies the client through a comparison of the packet’s destination address and the “key_id” field of the mih_clients_mobility table included in MIH database
- It either forwards or drops the packets based on the “check_p2p_packet” method, which uses the decision of the “adte_thread” module.

```
void *media_aware_proxy(void*);
int check_p2p_packet(client_info*);
int initialize_ip_queue();
int create_map_thread();
void destroy_map_thread();
void queue_error(struct ipq_handle*);
```

adte_thread.{h,c}

The “adte_thread” module is responsible for the correct decision of which views/layers/descriptions can safely be received by each client.

- It uses the “client_info” provided by “global_variables” module in order to identify the data rate for each client’s receiving view/description/layer.
- It uses the “networks_info” structures provided by “global_variables” module in order to identify the load of each access network (Wi-Fi, LTE).
- It invokes the “update_console” method of the “console” module in order to update the console with the information (client’s traffic, access point statistics) of the last second.
- It compares the load of each access point with the maximum acceptable load and it calculates the available bandwidth for each client.
- It allows the forwarding, only of those layers, that fit to the available bandwidth. The prioritization of each description is predefined with the descriptions of the stereo pair being more important than the side cameras as well as the base layers of the side cameras being more important than the corresponding enhancement layers.

```
void *adte_thread(void*);
void calculate_clients_available_bw();
void update_clients_available_bw(const char*,float);
void clients_adte_decision();
void calculate_client_data_rate_of_layers(client_info*);
void execute_adte(client_info*);
void reset_client_statistics(client_info*);
int create_adte_thread();
void destroy_adte_thread();
```

`database_thread.{h,c}`

The “database_thread” module is responsible for the connection to the MIH database provided by the MIH server.

- It updates the information of each connected client (id, wireless type, signal, throughput, IP address, subscription id) from the “mih_clients_mobility” table
- It retrieves the statistics for all monitored access networks (network id, type, IP address, network load) stored at the “mih_network” table.

```
void *update_from_db_thread(void*);
int connect_to_db();
int create_db_thread();
void destroy_db_thread();
```

`console.{h,c}`

The “console.c” module exploits the “ncurses” library in order to provide a legible overview of the MAP statistics collected by the “map_thread”, “adte_thread” as well as “database_thread” module.

```
void *update_console();
void init_wins(WINDOW**, int);
void write_to_wins(WINDOW*, WINDOW*, WINDOW*, WINDOW*);
void win_show(WINDOW*, const char*, int);
int initialize_console();
void destroy_console();
```

`ip_udp_header.{h,c}`

The “ip_udp_headers” module implements the IP and UDP structures as well as the aforementioned method which checks if the transport protocol is supported by the application. This auxiliary module is invoked by the “p2pParser” module in order to parse the IP/UDP headers of each packet.

```
const char* get_transport_protocol(__u8 number);
```

p2pParser.{h,c}

The “p2pParser” implements all the methods of “p2pParser” provided by P2PTransmitter in order to parse the P2P header of each packet. Moreover, it implements the “fast_p2p_parse” which is responsible for the IP/UDP/P2P header parsing of a packet.

```
int initialize_p2p_parser();
void destroy_p2p_parser();
int fast_p2p_parse(struct ipq_handle*, ipq_packet_msg_t*, client_info*);
uint64_t getPTS(P2PHeader*);
short getChunkId(P2PHeader*);
short GetViewId(P2PHeader*);
short GetDescriptiorId(P2PHeader*);
short getPID(P2PHeader*);
short getContentID(P2PHeader*);
short getLayerID(P2PHeader*);
short getPriority(P2PHeader*);
short getAudioIndicator(P2PHeader*);
short getMetadataFlag(P2PHeader*);
int SHORT_little_endian_TO_big_endian(int);
int INT_little_endian_TO_big_endian(int);
char* INT_little_endian_TO_big_endian_string_ip(int);
```

xml_parser.{h,c}

The “xml_parser” module exploits the libxml2 library in order to parse the “map_config.xml” file which contains information about the initialization of MAP.

```
int initialize_xml_parser(const char* file_name);
void destroy_xml_parser();
xmlNodePtr xml_find_element(const char *node_name,xmlNodePtr node);
int xml_decoding();
parameters_info* get_parameters();
```

log_files.{h,c}

The “log_file” module provides logging methods to each module.

```
void initialize_p2p_log_file(const char*);
void print_to_p2p_log_file();
void print_ok();
void print_dropped();
void close_file();
void initialize_delay_log_file(const char*);
void write_time_to_delay_file(long, long int);
void close_delay_file();
void initialize_db_log_file(const char*);
void write_client_info_to_db_file(client_info*);
void close_db_file();
void initialize_adte_log_file(const char*);
void write_to_adte_file(const char*);
void close_adte_file();
```

global_variables.{h,c}
The “global_variables” module holds all the definition, structures, variables, linked lists shared among the different modules. There are a variety of methods which enable the creation, update, search and deletion of a client’s node in the linked list “client_info”.

```
int initialize_global_variables();
void destroy_global_variables();
void set_error_occurred();
void initialize_new_mutex(pthread_mutex_t);
void destroy_old_mutex(pthread_mutex_t);
client_info* new_client(int);
client_info* search_client_by_ip(char);
client_info* search_client_by_id(int);
int update_client(client_info*, MYSQL_ROW);
void clear_client(client_info*);
void delete_client(void);
void destroy_linked_list();
void initialize_global_log_file(const char*);
void write_global_log_file(const char*);
void close_global_log_file();
```

4.1.3 Proxy Mobile IP

4.1.3.1 *Module description*

Proxy Mobile IP is responsible for allowing mobile nodes to move from one access network to another while maintaining their IP connectivity and without requiring their involvement in the overall process.

1. It detects the movement of mobile node in order to be aware of each node’s current position by monitoring link layer operations.
2. It implements the exchange of signalling messages between the Mobile Access Gateway (MAG) and the Local Mobility Anchor (LMA) that enables the creation and maintenance of a binding between the mobile node's home address(es) and its serving access network.
3. It creates bidirectional tunnels between the LMA and the serving MAG in order to deliver packets that are destined to the mobile node.

4.1.3.2 *Provided source files*

`lma.{h,c}`

LMA is the home agent for the mobile node in a Proxy Mobile IP domain. It is the topological anchor point for the mobile node's home network address(es) and is also the entity that manages the mobile node's binding state. The main module of the LMA component contains the logic for handing the requests (Proxy Binding Update message - PBU) coming from the MAGs of the various Access Network Gateways. It parses the PBUs and if it is an attachment or detachment it updates the routing tables and establishes or closes respectively the one end of the bi-directional tunnel to the requesting MAG. If it is an update it refreshes the life time of the respective Binding Cache entry. Finally it responds with a Proxy Binding Acknowledgement (PBA) message to the MAG.

```
lma_procedure_result_e lma_attach(gw_binding_t*, gw_binding_apn_t*);

lma_procedure_result_e lma_detach(gw_binding_t*, gw_binding_apn_t*);

lma_procedure_result_e lma_handover(gw_binding_t*, gw_binding_apn_t*);

lma_procedure_result_e lma_refresh(gw_binding_t*, gw_binding_apn_t*,
ip_address);

lma_procedure_result_e lma_detach_during_handover(gw_binding_t*,
gw_binding_apn_t*);

lma_procedure_result_e lma_attach_during_handover(gw_binding_t*,
gw_binding_apn_t*);

int lma_del_routes(gw_binding_t*);

int lma_del_routes_apn(gw_binding_t*, gw_binding_apn_t*);

int lma_add_routes(gw_binding_t*);

int lma_add_routes_apn(gw_binding_t*, gw_binding_apn_t*);

int lma_init_routes();

void lma_first_packet_callback(gw_binding_t*, gw_binding_apn_t *);

int lma_del_leftover_teid_route(void*);
```

`mag.{h,c}`

The main module of the MAG component contains the logic for detecting the mobile node’s attachments and detachments and then informing the LMA for these events by sending a Proxy Binding Update (PBU) message. Additionally it sets up its endpoint of the bi-directional tunnel to LMA upon receiving the PBA message.

```
int mag_init_routing_table();
int mag_del_routes(gw_binding_t*);
int mag_del_routes_apn(gw_binding_t*,gw_binding_apn_t*);
int mag_add_routes(gw_binding_t*);
int mag_add_routes_apn(gw_binding_t*,gw_binding_apn_t*);
int mag_initial(gw_binding_t*, gw_binding_apn_t*, uint8_t, ip_address,
ip_address);
int mag_refreshing(gw_binding_t*,gw_binding_apn_t*, uint8_t, ip_address,
ip_address);
int mag_confirmed();
int mag_waiting_for_dhcp();
int mag_deleting(gw_binding_t*, gw_binding_apn_t*, uint8_t);
int mag_handover(gw_binding_t*, gw_binding_apn_t*);
int mag_reply_to_source(mag_callback_operation_e, gw_binding_t*,
gw_binding_apn_t*);
```

4.2 Virtualisation

4.2.1 Description and current status

The development carried out in the virtualisation platform involves shifting the layer 3 functionalities, currently tackled by the Residential Gateway (RGW), to the operator premises aiming a layer 2 access network until the residential environment. The configuration of the BRAS, ONT, OLT and MAN and the development and integration of the DHCP server was reported in [18] .

The status of the virtualisation testbed up to year two, regarding the developments and configurations performed within the network elements that compose the access network is described in following sections:

4.2.1.1 Development of the Configuration Portal

The configuration portal is the web-based tool that allows end users to configure their virtualized router functionalities like the DHCP options, the IP addressing scheme, the port-forwarding configuration, etc. In addition, the Configuration Portal has the ability to manually enforce the flow prioritization in order to demonstrate the QoS enforcement at access network level.

The configuration portal has the following capabilities:

- To know the public IP address assigned to the virtual router
- To know the local network information (the IP subnet, the network mask, the gateway, the broadcast address and the primary and secondary DNSs)
- To know the IP addresses that are reserved to be assigned with the DHCP server
- To know the IP address and MAC address associations (same devices will obtain the same IP address from the DHCP server) and the possibility of delete the associations
- To know the current assigned IP addresses and the MAC address of the device
- To manually configure the IP address that will be assigned to an specific device (identified by its MAC address)

- To define and consult port-forwarding rules, in order to forward an incoming TCP or UDP flow (identified by the used port) to an internal device
- To manually define an incoming flow prioritization

4.2.1.2 Development of the Network API

The Network API is an application-programming interface that simplifies the communication of third-party services with the PCRF owned by the network operators. The point of the Network API is to offer a simple HTTP-based interface to the upper services (instead of complicated diameter-based protocol as Rx, defined by the 3GPP) and to hide the network complexity.

The Network API is running over Apache and is based in Python. The Super Peer will send the information to the Network API using an HTTP POST message using the endpoint "http://<network_api_address>/api/set-perfil". The parameters of that function are:

- Profile: The name of the physical profile applied to the end-user physical line
- User: Identification of the end-user on the PCRF.
- DestIP: IP address of the SuperPeer
- DestPort: Source port of the flow to be prioritised
- OrigPort: Destination port of the flow to be prioritosed

4.2.1.3 Configuration of the Policy and Charging Rules Function (PCRF)

The PCRF is a 3GPP standardized function responsible for defining the QoS rules that will be applied to a particular user. It will receive from the Super Peer though the Network API the physical parameters that identify a certain user and communicates with the Policy and Charging Enforcement Function (PCEF, that is hosted in the NATIE) in order to enforce the rules in the access network.

The PCRF is integrated with the Network API by using the Rx interface and also with the NATIE (BRAS) by using the Gx interface. There is a user identifier configured on the PCRF with a real fiber access line, placed on TID premises at Madrid.

4.2.1.4 Configuration of the Policy and Charging Enforcement Function (PCEF)

The PCEF function is the one in charge to enforce the QoS policies applied from the PCRF and it is embedded within the BRAS (the NATIE).

There is a predefined profile on the NATIE that priotizes a flow over background traffic. The PCRF will send to the PCEF the flow definition (IP addresses and transport ports) in order to get the required QoS on the ROMEO flow.

4.2.2 Future work

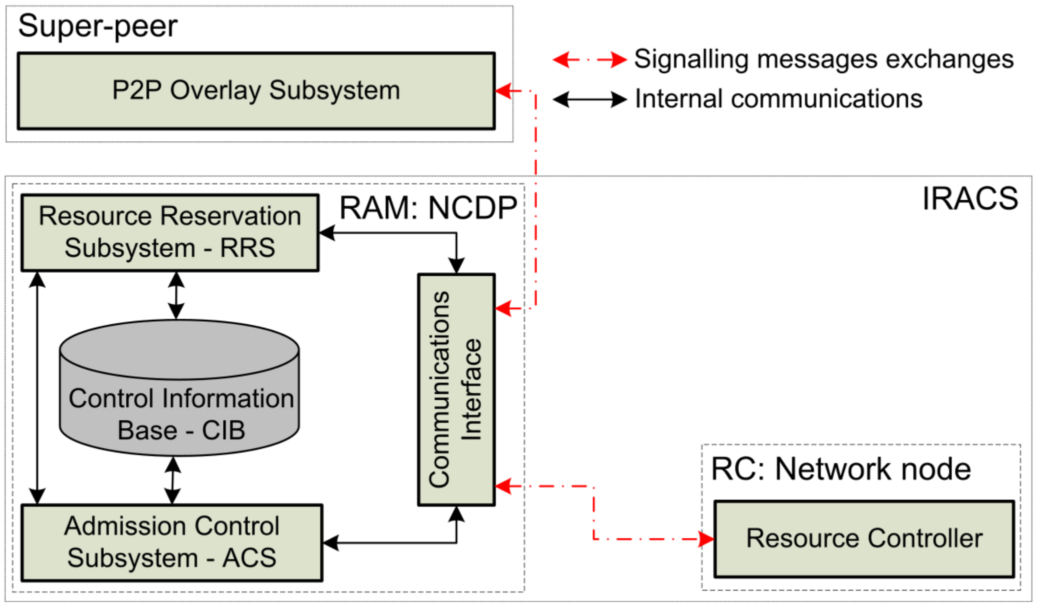
The next steps that need to be taken in the virtualisation platform are the development of the Virtual Software Execution Environment (VSEE), which has just started, that will enable the hosting of the execution capabilities that currently run on the RGW (e.g. UPnP control point) or other ROMEO capabilities that will be identified as potential candidates to be virtualized on the network too.

4.3 Internet Resource and Admission Control System

As it is depicted in Figure 49, the Internet Resource and Admission Control Subsystem (IRACS) is a set of software modules composed of a Resource and Admission Manager (RAM) component and a Resource Controller (RC) component with the following functions:

The RAM is responsible for defining proper control policies and managing access to the underlay network resources. It includes several functional software modules such as a Resource Reservation Subsystem (RRS), a Control Information Base (CIB), an Admission and Control Subsystem (ACS), and one interface for communications with external nodes (e.g., P2P Overlay Subsystems as reported in [1] [2]). It is worth noting that the overlay subsystem development is described in earlier sections.

Besides, each network node (e.g., router) implements a Resource Controller (RC) which enables them to enforce control decisions based on the instructions dictated by the RAM.



Super-peer
P2P Overlay Subsystem
Signalling messages exchanges
Internal communications
RAM: NCDP
IRACS
Resource Reservation Subsystem - RRS
Control Information Base - CIB
Communications Interface
Admission Control Subsystem - ACS
RC: Network node
Resource Controller

Figure 49. Cross-layer QoS control architecture.

The implementation of the IRACS is carried out in the well known Network Simulator version 2 (ns-2). Therefore, all the codes are developed in C++ [4] and in TCL [4] for the scripts. In particular, the ns-2 is a discrete event network simulator, popular in academia, for network protocols simulations whether wired or wireless. It provides a split objects model, meaning that the ns was built in C++ and provides a simulation interface through OTcl which is an object-oriented language of Tcl where each object created in OTcl has a corresponding object in C++. This model allows users to describe network topologies with specific parameters using OTcl scripts which will be simulated by the main ns program according to the specified topologies and configuration parameters.

Further details on the implementation status of the RAM and RC modules will be provided in subsections 4.3.1 and 4.3.2, respectively.

4.3.1 Resource and Admission Manager (RAM)

The RAM is a software module running at the Operator's network control server side with the following functions:

- Receive connection requests to use the underlay network resources;
- Control the allocation of the network resources (bandwidth) by granting or denying access to the latter in such a way as to improve the network utilization while guaranteeing differentiated QoS for all admitted sessions;

- Deploy efficient resource over-reservation techniques to improve QoS control scalability, that is, without overwhelming networks with resource reservation signalling, states and processing overhead;
- Assure service convergence by prevent some Internet applications (e.g., bandwidth demanding P2P communications) from starving other applications of resources using appropriate control policies;
- Perform traffic load balancing to avoid unnecessary congestion occurrence. This provides support for packet delay and jitter control to improve synchronization operations of multiple views that end-users may receive from diverse technologies or multiple communication paths;
- Maintain good knowledge of network topology and related links resource status;

In purely centralized networks, a single RAM takes overall control of the network. However, in hierarchical scenarios spanning multiple domains (see Deliverable 5.1), a RAM is deployed per core network for scalability reasons, and inter-domain connections are performed according to pre-defined Service Level Agreements (SLAs) between the domains operators. In this way, the RAM functions aim to optimize the network utilization across the core domains by enforcing scalable QoS measures.

The subsequent subsection describes the most relevant mechanisms and functions used by the RAM to perform its operations.

4.3.1.1 Implemented RAM Mechanisms and Functions

The RAM software module is implemented through a *RAMAgent* class to provide the functions required by the Network Control Decition Point (NCDP) described in [1]. It is therefore an agent to be attached to the NCDP server in TCL script. It includes the following functions:

At system initialisation when the nodes boot up, the TCL script file calls the commands *“ConfigInterfaces”* and *“NetInitialisation”* for the NCDP server running *RAMAgent agent.* The *“ConfigInterfaces”* gets links states information such as Link Bandwidth, and delay, among other variables, dynamically from the simulator, and the *“NetInitialization”* exploits the information to configure the network initial parameters (e.g., compute initial ingress-egresses paths as defined by the ACS module [1]). Further, the *storeInitialisationInformation()* function collects the information and stores them in the CIB database. Then, the system enters normal operation state where the ACS is idle and waits for requests to come from TCL commands, as summarised in Figure 50.

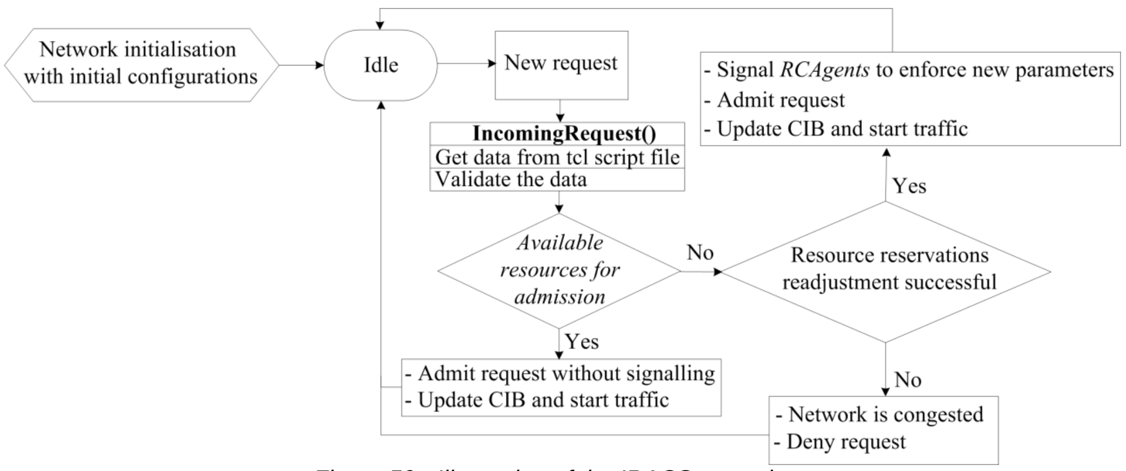

Network initialisation with initial configurations
Idle
New request
IncomingRequest()
Get data from tcl script file
Validate the data
Available resources for admission
Yes
- Admit request without signalling
- Update CIB and start traffic
No
Resource reservations readjustment successful
Yes
- Signal RCAgents to enforce new parameters
- Admit request
- Update CIB and start traffic
No
- Network is congested
- Deny request

Figure 50 - Illustration of the IRACS operations

When a new service request comes, the function IncomingRequest() is called to parse and to validate the variables passed through the TCL command (e.g., avoid parsing negative CoS or bandwidth values, check uniqueness of flows and session IDs). After that, the ACS takes into account the requested amount of bandwidth and the available bandwidth in the network for the admission decisions. The ACS always selects the best paths to accommodate an incoming flow, this procedure may involve several steps depending on the network resource conditions and the service demand as in the following:

i) If there is sufficient available bandwidth on existing paths for the requested service, the ACS selects the path which has the most available resources and accommodates the request. After that, the *updateCIB()* function is called to update the CIB information accordingly (e.g., requested Class of Service (CoS), bandwidth, session IDs, flows IDs, protocol IDs, ports IDs, and paths information). Then, the traffic is generated in the simulation by calling the function *CreateTraffic()* and the ACS system returns to idle state to be able to pasre another request upon need.

ii) In case the requested bandwidth is above the available bandwidth in the network, ACS triggers the RRS module and the latter calls *ReservationComputation(path, requested_bandwidth,CoS)* function to define new resource reservation parameters for the existing network paths. The resource reservation readjustment algorithm is described in [6]. If the reservation parameters are readjusted successfully on a potential path, that is, the requested CoS is increased with sufficient available resources from other existing CoSs on a path, the nodes on the concerned path are signaled where the RC module described later in subsection 4.3.2 is responsible for enforcing the new parameters defined for them along the path. Then, ACS admits the pending request, the CIB database is updated accordingly and the traffic is generated in the simulation environment as we described earlier. In this way, the reservation parameters are defined and readjusted dynamically in a way to prevent CoS starvation or unnecessary waste of resources while the QoS control signalling frequency is reduced for scalability.

iii) In the situation where the network is congested and requested bandwidth is above the total unused resource on all paths, the ACS denies the incoming to prevent control performance degradation.

iv) When a running session terminates, the ACS stops the related traffic and the CIB database is updated accordingly by using the *updateCIB()* function.

In order to enable operations over multiple domains, an interDomainOperations() interface is defined in RAMAgent to allow interactions between the RAM and external nodes such as other RAMs, super-peer, a signle peer or the ROMEO server. Also, the ACS system implements recv() and send() functions in order to effectively receive and to send commands and messages while being able to issue commands and messages with control instructions forwards the RCAgents inside the network. The mechanisms and functions implemented by the RC module to allow a proper coordination with the RAM to enforce control decisions and policies specified by the latter are described in subsequent subsection.

4.3.2 Resource Controller (RC)

The RC is a software module running mainly in the routers at the ISP network with the following functions:

- Deploy elementary transport functions to enable UDP port recognition (routers are permanently listening on a specific UDP port) or IP Router Alert Option (RAO) [1] on nodes to properly intercept, interpret and process control messages.

- Receive resource reservation and multicast tree creation instructions from the RAM and properly enforce them on the nodes;
- Interact with Resource Management Functions (RMF) [1] to properly configure schedulers on nodes [1], thus ensuring that each CoS receives the amount of bandwidth allocated to it to provide QoS-aware data transport across the network;
- Assure basic network functions such as packets forwarding and routing;
- Exploit legacy control databases such as, but are not limited to, the *Management Information Base* (MIB), *Routing Information Base* (RIB) or *multicast Routing Information Base* (MRIB) and *Forwarding Information Base* (FIB), according to the control instructions received from the RAM;
- Learn inter-domain routing information from the traditional Border Gateway Protocol (BGP) [1] for proper communications between various network domains and operators.

4.3.2.1 Implemented RC Mechanisms and Functions

The RC software module is implemented by means of an *RCAgent* class to provide the functions required by the network nodes (e.g., routers) described in [1]. It is an agent that must be attached to each node in TCL script and includes the following functions:

In the TCL script, the command *“ConfigInterfaces”* must be called at system initialisation for all *RCAgents* in the network so every node collects the links states information and configures its local interfaces CoSs and related initial reservations parameters. The initial reservation parameters are computed by using the function *CoS_initial_Reservation(interface)* that takes one interface as input parameter according to the policies defined in the algorithm described in [6]. Then, the computed values are enforced in the simulator environment upon successful computation by means of an *addQueueWeights* command. This command receives each CoS with the related weight *p*, which is a percentage of the link bandwidth to be assigned for the CoS.
Besides, the RCAgent also implement the recv() and send() functions to be able to intercept commands and messages conveyed to it or to forward or send messages, respectively. Upon receiving a message (e.g., from RAM), the RCAgent is able to interpret the information and instructions conveyed by calling a parseInformation() function. In addition, a pathReservationEnforcement(PATH, Links_Parameter_s) function is implemented by the RCAgent in order to properly enforce resource reservation parameters on nodes along communications paths to ensure that each CoS receives the bandwidth destined to it.

5 CONCLUSION

This deliverable provides a detailed report on the development status of ROMEO system components at the end of the second year of the project. The document divides the ROMEO components into two main groups: Server components and Peer components and focuses on each part separately. Since most of the development tasks of the project are near the end, this report includes almost complete descriptions for most of the modules.

Besides being used as a development reference for the whole system, this deliverable will also provide an opportunity to check the components' timelines for integrated solution which is provided in detail in the deliverable D6.3 - Second report on system components development plan.

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APPENDIX A: GLOSSARY OF ABBREVIATIONS

A	
AAA	Authentication, Authorization and Accounting
ACS	Admission and Control Subsystem
ADL	Audio Description Language
AAC	Advanced Audio Coding
ADTE	Adaptation Decision Taking Engine
AGC	Automatic Gain Control
AMF	Action Message Protocol
AP	Access Point
ARC	Arcelik A.S.
ARM	Advanced RISC Machines
ASIO	Audio Stream Input/Output
A/V	Audio / Video
AVC	Advanced Video Coding
B	
BCC	Binaural Cue Coding
BGRP	Border Gateway Reservation Protocol
BRAS	Broadband Remote Access Server
BRIR	Binaural Room Impulse Responses
BRS	Binaural Room Synthesis
BRTF	Binaural Room Transfer Functions
C	
CDN	Content Delivery Networks
CN	Correspondent Node
CoA	Care of Address
CoS	Class of Services
CPE	Customer Premises Equipment
CPU	Central Processing Unit
CS	Chunk Selection
CSRC	Contributing Source
D	
D/A	Digital to Analog
DHCP	Dynamic Host Configuration Protocol
DON	Decoding Order Number
DoW	Description of Work
DVB	Digital Video Broadcast
DVR	Digital Video Recorder
F	
FA	Foreign Agent
FEC	Forward Error Correction
FFT	Fast Fourier Transform

FM	Frequency modulation
FMIP	Fast Mobile Internet Protocol
Fps	Frames Per Second
FTP	File Transfer Protocol
FU	Fragmentation Unit
G	
GBCE	Global Brightness Contrast Enhancement
GM	General Meeting
GPON	Gigabit Passive Optical Network
GPS	Global Positioning System
GPU	Graphics Processing Unit
GUI	Graphical User Interface
H	
HA	Home agent
HD	High Definition
HDMI	High-Definition Multimedia Interface
HLOS	High Level Operating System
HMIP	Hierarchical Mobile IP
HRTF	Head Related Transfer Function
HoA	Home Address
I	
ICC	Inter-Channel Coherence
ICLD	Inter-Channel Level Difference
ICTD	Inter-Channel Time Difference
IETF	Internal Engineering Task Force
IFFT	Inverse Fast Fourier Transform
IGMP	Internet Group Management Protocol
ILD	Inter-aural Level Differences
IP	Internet Protocol
IPR	Intellectual Property Rights
IPTV	Internet Protocol Television
IRT	Institut für Rundfunktechnik GmbH
IT	Instituto de Telecomunicações
ITD	Inter-aural Time Differences
J	
JSON-RPC	JavaScript Object Notation-Remote Procedure Call
L	
LCD	Liquid Crystal Display
LFE	Low Frequency Enhancement
LMA	Local Mobility Anchor
LTE	Long Term Evolution
LVDS	Low Voltage Differential Signalling
M	
MAC	Media Access Control
MAG	Mobility Access Gateway

MANE	Media Aware Network Element
MANET	Mobile Ad-Hoc Network
MAP	Mobility Anchor Point
MARA	Multi-user Aggregated Resource Allocation
MCX	Micro Coaxial
MDC	Multiple Description Coding
MDCT	Modified Discrete Cosine Transform
MD-SMVD	Multiple Description Scalable Multi-view Video plus Depth
MICS	Media Independent Command Service
MIES	Media Independent Event Service
MIH	Media Independent Handover
MIIS	Media Independent Information Service
MIP	Mobile Internet Protocol
MISO	Multiple Input Single Output
MM	Mobility Management
MMS	MM Solutions AD
MN	Mobile Network
MPEG	Moving Pictures Experts Group
MPLS	Multiprotocol Label Switching
MST	Multi-Session Transmission
MT	Mobile Terminal
MTAP	Multi-Time Aggregation Packet
MTD	Mobile Terminal Display
MULSYS	MulSys Limited
MVC	Multi View Coding
N	
NAL	Network Abstraction Layer
NAT	Network Address Translation
NEM	Networked Electronic Media
NGN	Next Generation Networks
NMS	Network Monitoring Subsystem
NS-2	Network Simulator 2
O	
OFDM	Orthogonal Frequency Division Multiplexing
OLT	Optical Line Terminal
OMAP	Open Multimedia Application Platform
ONT	Optical Network Terminal
P	
P2P	Peer-to-Peer
PAPR	Peak to Average Power Ratio
PC	Personal Computer
PHY	Physical
PLP	Physical Layer Pipe
PMIP	Proxy Mobile Internet Protocol
PoA	Point of Attachment

PoP	Package on Package
POTS	Plain Old Telephone Service
PPP	Point-to-Point Protocol
PPSP	Peer-to-Peer Streaming Protocol
PSW	Priority Sliding Window
R	
R-OTT	Reverse One-To-Two
R&S	Rohde & Schwarz GmbH & Co. KG
RAT	Remote Access Technology
RF	Radio Frequency
RGB	Red Green Blue
RISC	Reduced Instruction Set Computer
RMI	Remote Method Invocation
RO	Route Optimization
RRP	Return Routability Protocol
RSVP	Resource ReServation Protocol
RTCP-XR	Real Time Control Protocol Extended Report
RTMP	Real Time Message Protocol
RTP	Real-time Transport Protocol
RTSP	Real Time Streaming Protocol
Q	
QAM	Quadrature Amplitude Modulation
QAT	Quality Assurance Team
QoE	Quality of Experience
QoS	Quality of Service
S	
SAC	Spatial Audio Coding
SD	Standard Definition
SDP	Session Description Protocol
segSNR	segmental Signal-to-Noise Ratio
SEI	Supplemental Enhancement Information
SIDSP	Simple Inter-Domain QoS Signalling Protocol
SMVD	Scalable Multi-view Video plus Depth
SoC	System on Chip
SSRC	Synchronisation Source
SST	Single Session Transmission
STAP	Single-Time Aggregation Packet
SVC	Scalable Video Coding
T	
TB	Topology Builder
TC	Topology Controller
TEC	Technicolor R&D France
TFT	Thin Film Transistor
TID	Telefónica I+D
TTA	Türk Telekomünikasyon A.Ş.

TS	Transport Stream
U	
UDP	User Datagram Protocol
UHF	Ultra High Frequency
UP	University of Patras
US	University of Surrey
V	
VAS	Value Added Service
VITEC	VITEC Multimedia
VoD	Video On Demand
W	
Wi-Fi	Wireless Fidelity
WFS	Wave Field Synthesis
WG	Working Group
WP	Work Package
WVGA	Wide Video Graphics Array
WXGA	Wide eXtended Graphics Array
X	
XGA	eXtended Graphics Array
Numbers	
4G	4th Generation