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Abstract

This document addresses the specifications for the design of the SUIT gateway and terminal, identifying the main functionalities and proposing a general architecture. As a gateway, it acts as the interface between the broadband networks (DVB-T/H and WiMAX) and the wireless local area network (Wi-Fi), connecting in this way the play-out with the user terminals. As the gateway functionalities can move to the terminal (mobile terminal), two different types of terminals are considered: the Wi-Fi terminal fed by the gateway, and the SUIT terminal that is able to receive MD-SVC video streams directly from DVB-T/H and WiMAX outdoor networks.

Keyword list: Gateway, scalable video coding, multiple description coding, rate control.

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Gateway Specifications

SUIT_210

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1 Introduction

1.1 Scope

This document is part of the WP5 – “Components for the Test-bed/Field Trials”, which is the responsible for designing some required components not available in the market but required by the play-out infrastructure in order to adequately serve scalable contents through two last mile wireless networks to user terminals. Specifically, the document deals with the SUIT Gateway and terminals.

1.2 Objective

The aim of this report is to address the specifications of the SUIT component acting like a gateway with two main tasks:

- on the one hand, interfacing to the last mile wireless networks (DVB-T/H/RCT and WiMAX) on one side and the wireless local network (Wi-Fi 802.11g) on the other, this way allowing local devices to access to the play-out audio-visual scalable resources,
- on the other hand, processing Scalable Multiple Description coded contents (combining and performing a rate control taking advantage of the use of scalability) coming from the play-out, and delivering them with the appropriate bit-rates to the wireless local network.

This report also tackles the functional specifications of the end user terminal.

2 Gateway functionalities

2.1 Introduction

Figure 1 illustrates the basic gateway architecture within the SUIT platform. SUIT gateway will interface, on one hand, two local loop broadband networks (DVB-T and WiMAX) which connect the gateway with the play-out. On the other hand it will also interface a WLAN (802.11g) through which the video contents will be delivered to end users.

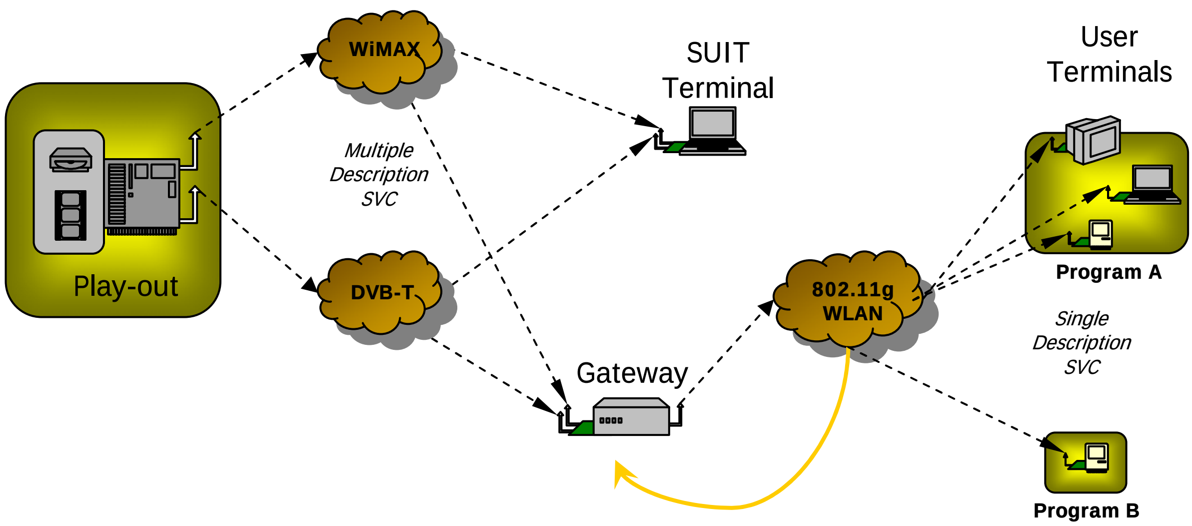

Figure 1 - SUIT platform general view.

In a home network scenario, the gateway has to provide a set of functionalities which can be divided into four categories:

- Stream management. Video contents will be delivered by the play-out using both last mile networks. Contents will be encoded using a Scalable Multiple Description coding approach resulting in two different scalable descriptions. One of them will be delivered by the WiMAX network, while the other will be transmitted via the DVB connection. These two descriptions have to be combined in a single scalable video stream for the Wi-Fi terminals. More details on Scalable Multiple Description video coding can be obtained from D3.1 – "Design of scalable multiple-description video coding".
- WLAN transmission. SUIT has adopted 802.11g for the wireless local network due to its bandwidth capacity to deliver video based contents. However, the local network characteristics may vary dynamically (local devices move far away from each other, interferences, etc.). It will be a task of the gateway to deal with this random behaviour by adapting transmitted contents to the WLAN dynamic characteristics.
- Communication with the play-out. In some situations, feedback from the gateway to the play-out is convenient to report about the statistics of the transmission, or even to select the characteristics of the coded streams (single/multiple description, bit rate, etc.)
- Internet access. One of the services offered to the end users is access to the Internet. Therefore, the gateway has to support routing capabilities within the scenario depicted.

2.2 *Stream management*

As described before, the gateway receives a two-description coded sequence via its interfaces with the last mile networks. These two descriptions has to be combined in order to generate a single scalable bit-stream which is delivered to the WLAN terminals according to the network and/or terminal characteristics, as is depicted in Figure 2.

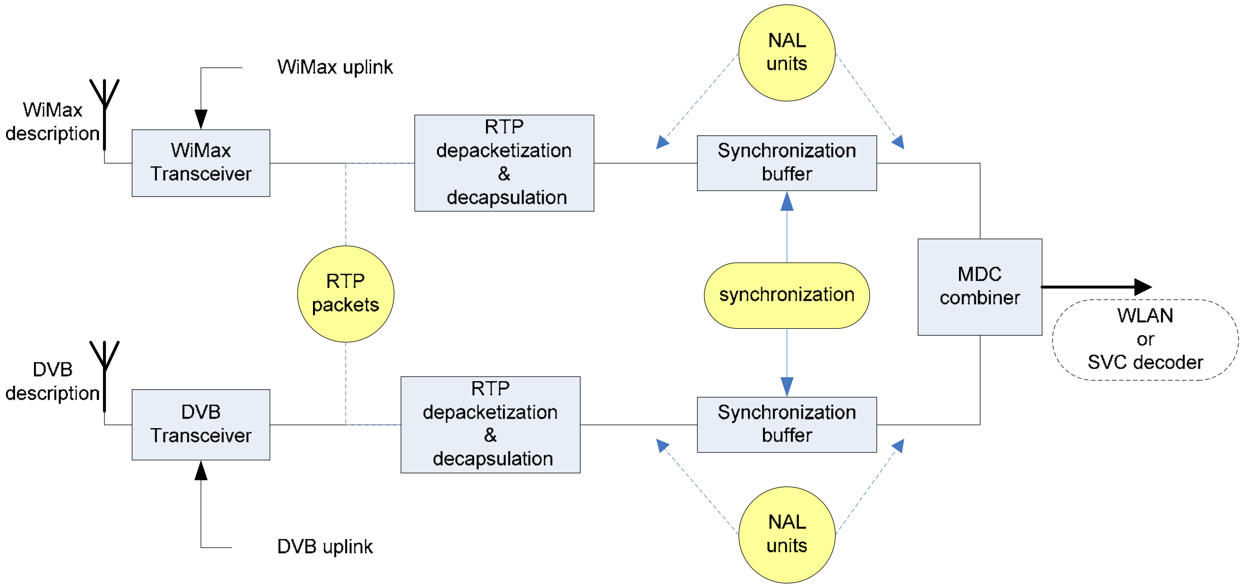

Figure 2 - Gateway video bit-streams paths.

Correctly received link-layer packets of both descriptions are reassembled and their UDP and IP headers are removed as they traverse the UDP/IP protocol stack. For each of both descriptions, the resulting stream of RTP packets is delivered to a RTP depacketization and decapsulation module which recovers the NAL units (NALUs) contained inside the RTP packets.

The resulting streams of NALUs may not be synchronized with each other due to unequal delays in the WiMAX and DVB transmission chains. Furthermore, due to packet loss in one of both networks NALUs that are recovered from one description may have been erased in the other description, or vice versa. Therefore, synchronization between both NALU streams is required before they are delivered to the MDC Combiner.

In the following, the operations required to produce the single scalable bit-stream are described more in depth.

2.2.1 Synchronization of video descriptions

2.2.1.1 <u>RTP level</u>

SUIT project has adopted Real Time Protocol (RTP) [1] over UDP for the transmission of the scalable video sequences. Therefore, the transmission of the video contents is not reliable nor packets are guaranteed to be received orderly. These characteristics pose difficulty to the combination of the two video descriptions, since it requires the synchronization of both descriptions at NAL level.

The first step to achieve this synchronization is performed at a RTP level. Figure 3 shows the format of an RTP packet. For a given video content, RTP packets are to be ordered firstly based on the information of the standard RTP fields, such as the timestamp field and the sequence number.

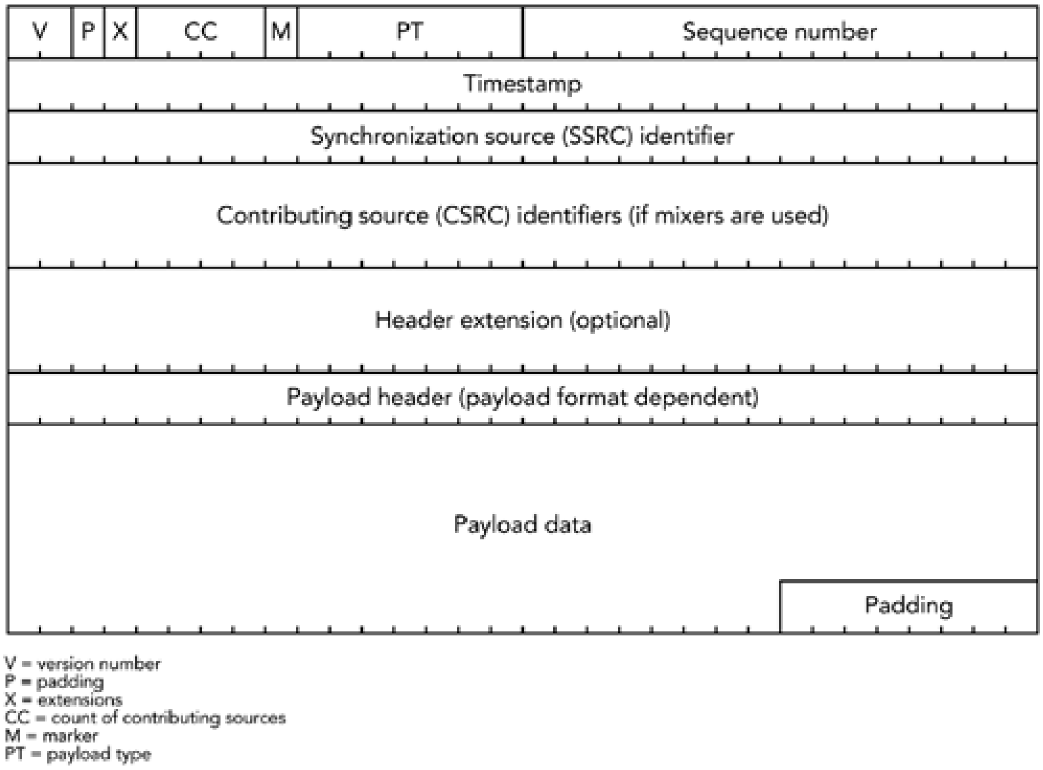

Figure 3 - RTP packet structure.

RTP packets will host in their payload NALUs of the coded video sequences. Since there is not a unique correspondence between a NALU and an RTP packet (either NALU aggregation or fragmentation can be used) RTP header information is not sufficient to order the NALUs. To overcome this limitation, specific data will be transmitted in the RTP payload along with the NALUs data, such as the frame number, or the slice number the NALU belongs to as described in the following section.

So, the tasks performed at the RTP level are:

- Reordering of the received RTP packets based on their RTP sequence number,
- Depacketization of RTP Aggregate Packets (if used); i.e. recovering the NAL units one by one,
- Ensambling of RTP Fragmented Packets (if used); i.e. ensambling NAL units when required,
- Delivery of NALUs for their synchronization.

2.2.1.2 NAL level

Looking at the transmission of an SVC bit-stream, synchronization must be ensured at the Access Unit (AU) level. Inside a bit-stream, it must be known that the different spatial and quality information pieces, that contribute to describe a picture, have the same timestamp. Timestamps are evolving along the temporal axis within the different temporal layers of a Group of Pictures (GOP). As the encoder samples pictures a coding and then a decoding order, it is then generally used to label AU with timestamps. It may be done by dividing the picture number, according the decoding order, by the frame rate. In a decimal numbering, it provides timestamps described in seconds and fractions of seconds for filling the corresponding fields in an RTP header.

Besides this feature due to scalable coding, some descriptive NALUs, specifying parameters used to encode a sequence or a picture (SPS, PPS, SEI), can appear previously to video coded information (VCL) NALUs. They generally inherit the same AU timestamp.

For improving encoding efficiency or error resilience, pictures can be decomposed into slices that may be coded separately into different NALUs, but sharing also the same timestamp.

An SVC decoder will start processing when it will get a complete AU, even it some spatial or quality layers are missing or truncated: it will do its best with the remaining information. Bit-stream truncation could then apply at the play-out side as well at the gateway level in order to adapt video content to the available bandwidth or terminal capabilities. In this aim, spatial or quality layers can be discarding only using their level numbering or by relying on some complementary quality information stored in a given SEI message. To avoid the gateway to decode such messages, the following information could be sent as extra fields in the extension header of the RTP packets (in order to maintain the compatibility with others terminals):

- frame number (3 bytes);
- slice number (2 bytes);
- NALU size (3 bytes);
- temporal layer (1 byte);
- spatial layer (1 byte);
- quality layer (1 byte);
- quality measure (1 byte).

All this information is available in the SVC file format at the play-out (see D5.2 – “Real-Time SVC Encoder Specifications”).

In a first approach, the additional fields of the extension header will be the frame_number (3 bytes) and slice_number (2 bytes), because the other information can be obtained from SEI messages or computed from NALU and RTP headers without requiring a huge computational cost. Nevertheless, in the final solution, sent information could vary due to other constrains.

2.2.2 Combiner

In D3.1 – “Design of scalable multiple-description video coding” three different MD coding schemes for the SUIT project are proposed. In the following, their respective MD combiners will be described.

2.2.2.1 Unbalanced MDC

Synchronized NALUs representing the output of the two synchronizations buffers arrive at the MD combiner. The role of this one is reduced to dropping the redundant NALUs and signalizing the missing NALUs (in the case of errors in both descriptions).

2.2.2.2 MDC based on redundant slices

The combination of both descriptions into a single compliant bit-stream can be performed in the network abstraction layer (NAL). At the play-out side, the MDC generator will tag each NALU with a sequence number, a frame number and/or a timestamp. This will aid in synchronizing and combining both descriptions with each other.

The requirements for the MDC combiner depend upon the considered approach:

1. When using the H.264/AVC support for signalling redundant slices, the combiner has to parse the slice header of all coded slices in order to find out whether or not the slice is redundant, and must then decide whether it can be discarded. Unavoidably, this increases the computational requirements and hence the delay of the combiner. As an alternative, when a decoder with support for H.264/AVC redundant slices is available, the task of removing the

redundant slices can be offloaded to this decoder. However, this will increase the bandwidth requirement for the last-mile WLAN network, since both redundant and non-redundant slices need to be transmitted through this network.

2. When not using the H.264 syntax to mark redundant slices, the MDC combiner will instead investigate the numbers with which each NAL unit was tagged at the play-out side. This allows the combiner to detect which slices were lost during transmission, and hence to correctly replace lost slices with their redundant copy.

2.2.2.3 MDC based on EMDSQ

The MD combiner module requires access to the quantized coefficients of both descriptions; therefore entropy decoding shall first be performed on both incoming video streams. Then, the actual reconstruction of both side-descriptions into a single description is performed, according to the number of descriptions available. Entropy coding will have to be re-applied in order to obtain an SVC compliant bit-stream to be sent over the local WLAN. The EMDSQ combiner is in fact an inverse IA matrix which is mapping pair of coefficients into the reconstructed central descriptions as described in D3.1.

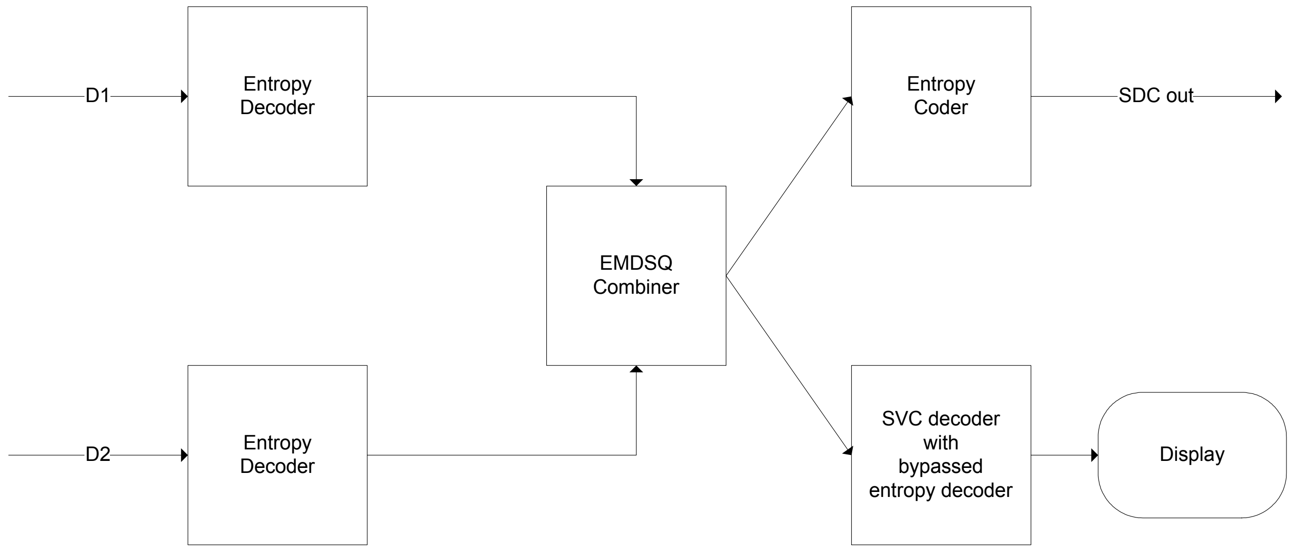

Figure 4 - EMDSQ Combiner.

2.3 WLAN Transmission (SVC transrating & RTP packetization)

The output of the combiner is a scalable video bit-stream which will be transmitted to the end-users' terminals through a Wi-Fi network. At this point, the gateway will perform rate control in order to dynamically adapt the output bitrate to the varying conditions of the WLAN channel. Appropriate and optimized rate control strategies will be developed taking into account:

- the scalability of the coded bit-stream,
- information about the Rate-Distortion (R-D) characteristics of the coded sequence,
- the transmission scheme used (multicast or unicast),
- information about the behaviour of the WLAN channel.

More specifically, rate control strategies will take advantage of the different types of scalable layers (spatial, temporal and SNR) and their dependencies to provide an efficient use of the available bandwidth. Specific enhancement layers may be dynamically discarded to accommodate the SVC bitrate to the channel bitrate. Moreover, in case Fine Granularity Scalability (FGS) is provided, the bit rate of the SNR enhancement layers can be customized according to the fluctuations of the bandwidth of the WLAN channel. Rate control strategies will also use R-D information of each layer when available. In this sense, the NALU size and its quality measure (as described in D5.2) will be the R-D data considered in the rate-distortion optimization.

In addition, rate control strategies will distinguish between multicast and unicast sessions since in the former packet losses are more likely to occur than in the second one. In multicast, the communication of the Access Point (AP) with the terminals is not protected at link level via retransmissions. Therefore, rate control strategies may consider selective retransmissions in a multicast scenario as a mechanism to minimize the degradation of the decoded sequence due to the occurrence of packet losses.

The characteristics of the WLAN channel will be inferred from the terminals feedback. This feedback will be obtained by means of the protocols adopted (such as RTCP) or by solutions specifically developed for the SUIT project, if required. Measurements such as the Packet Error Rate (PER), or the length of the bursts will be estimated at the gateway, and incorporated to the rate control strategies via the development channel models described in D2.4 – “802.11g WLAN network modelling”.

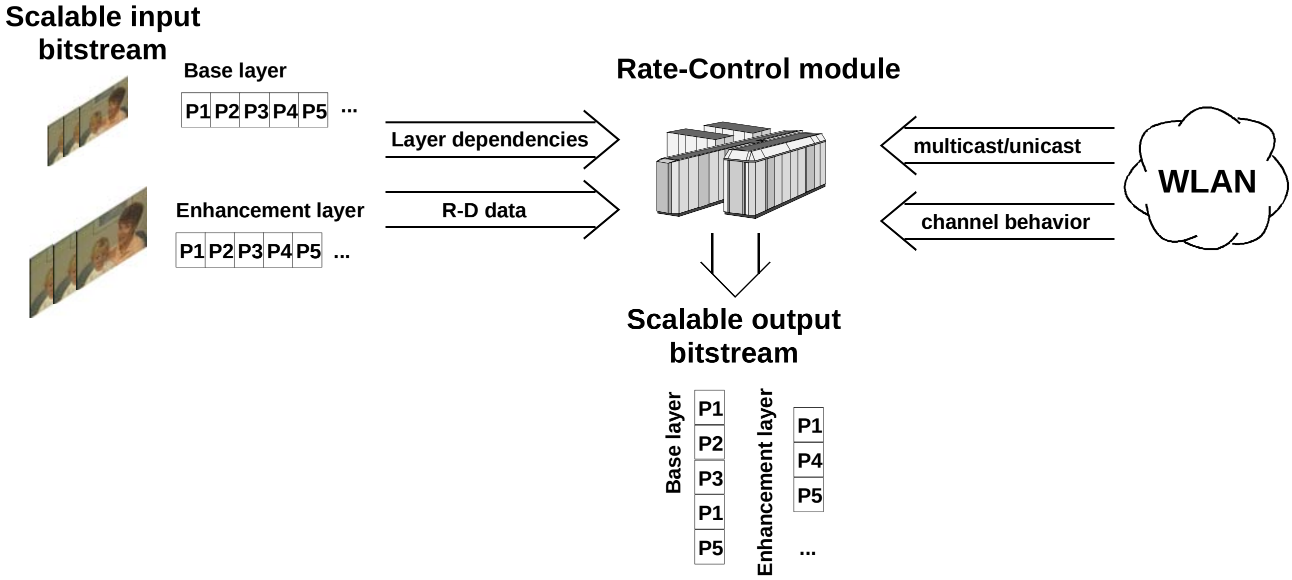

Figure 5 - Diagram of the rate control approach.

Figure 5 summarizes the elements considered in the rate control scheme. The rate control algorithms will determine which parts of the SVC bit-stream are transmitted at each transmission time, with the objective of:

- fulfilling the channel constraints,
- minimizing the degradation of the decoded sequence due to the effect of dropping parts of the bit-stream and packet losses.

Moreover, they may also act on the input to the combiner in order to save computational resources. If specific layers of the scalable stream are not going to be transmitted due to the channel limitations, they will be signalized to the synchronization module so as to the corresponding NALUs are prevented from being combined.

In a first stage, heuristic and basic rate control strategies will be implemented based on a subset of the elements described before. Once all the elements are completely identified and defined, the rate control problem will be formulated within an optimization framework aiming at computing optimized algorithms.

The output SVC bit-stream will be encapsulated following the same approach used for the transmission of the video contents at the play-out described in D4.1”Synchronization and encapsulation”, taking into account the characteristics of the specific scenario addressed by the gateway. In this sense, timestamps given at the play-out will be maintained, and the packetization algorithm will be adapted according to the WLAN characteristics. The customization of parameters such as the use of aggregation or fragmentation of the different types of NALUs will be also tackled and decisions will be made based on the results of both simulated and real tests.

2.4 Communication with the play-out

Deliverable D1.4 – “Architecture and Reference Scenarios” identified three network scenarios, Home, Mobile and MHP-IPTV. The first one, Home, requires a gateway as depicted, below, in Figure 6. The Wi-Fi end-user terminals will be fed via the gateway interfacing to two different wireless broadband last mile networks, WiMAX and DVB-T/H. The gateway informs the play-out about the network and terminal characteristics. The play-out selects the right network to deliver the content and may reduce the bit-rate for e.g. video-on-demand (unicast) according to the network conditions. Figure 6 shows two homes and therefore two home gateways. In each home two terminals are connected to each gateway.

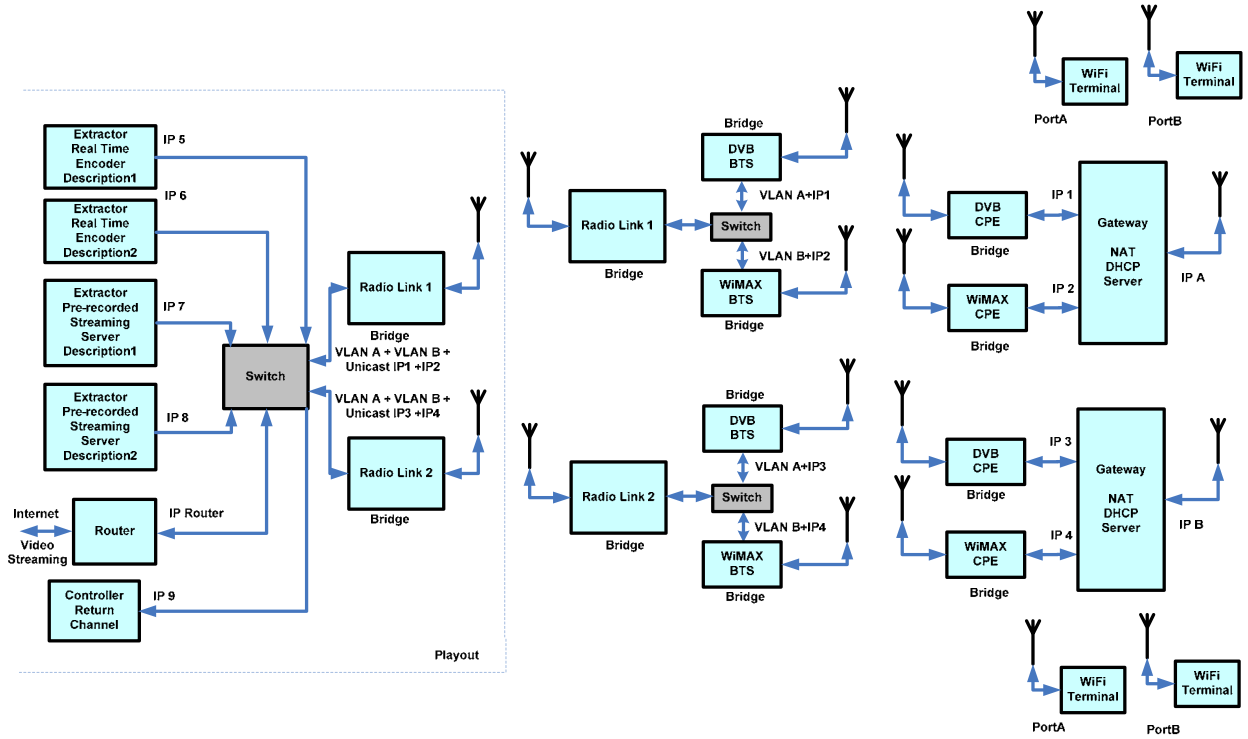

Figure 6 - Block diagram of the home network scenario.

SUIT will set up four base stations in two cells, where they will be co-sited in pairs. So, each cell will have one DVB-T/H base station and one WiMAX base station. This network scenario allows us to test different types of services and functionalities. However, it requires four experimental frequencies and the associated licenses in order to perform the field trials.

Deliverable D1.4 also mentioned service scenarios. With the application of new techniques in the SUIT project, various innovating service scenarios will be enabled when converging the two broadband mobile networks IEEE 802.16e (WiMAX), ETSI/EN 300 744 (DVB-T) and ETSI/EN 302 304 (DVB-H). SUIT will deliver a layered description of each of those last mile networks. By using two different video descriptions, SUIT will push video scalability into broadcasting and telecom networks in a fruitful way. As a final objective, SUIT intends to demonstrate an end-to-end communication system, from the play-out to the terminal, where the terminal can feed an HDTV screen or a small pocket-sized display.

Deliverable D1.1 – “User Terminal Requirements” proposed a series of service scenarios (see Table 1 and Table 2 below) associated to relatively high and relatively low data rates and for two different DVB-T/H multiplexes. In the configurations described in the two tables, only one operative base station is considered in each cell. Therefore, one cell contains one DVB-T/H base station and the other cell contains one WiMAX base station.

DVB-T cell		**WiMAX cell**	
Service	**Bit rate (Mbps)**	**Service**	**Bit rate (Mbps)**
1st D SVC real-time broadcasting	6.25	2nd D real-time broadcasting	6.25
1st D SVC broadcasting	6.25	2nd D broadcasting (on QoS demand)	6.25
1st D hyperlinked video	0.5	2nd D hyperlinked video	0.5
Internet	0-?	Internet	0-?
		Streaming	0.5-4.25
Total	**13**	**Total**	**13.5-17.25**[1]

Table 1 - Service scenarios for the 13 Mbps multiplexer of IRT.

Note: 1st D = 1st description, 2nd D = 2nd description;
SVC - HD: 1280x704x25 (4.25 Mbps);
SD: 640x352x25 (1.5 Mbps);
CIF: 320x176x25 (0.5 Mbps)
(All video formats are progressively scanned)

DVB-T cell		**WiMAX cell**	
Service	**Bit rate (Mbps)**	**Service**	**Bit rate (Mbps)**
1st D SVC real-time broadcasting	10	2nd D real-time broadcasting	10
1st D SVC broadcasting	10	2nd D broadcasting (on QoS demand)	10
1st D hyperlinked video	0.5	2nd D hyperlinked video	0.5
Internet	0-?	Internet	0-?
		Streaming	0.5-8
Total	**20.5**	**Total**	**21-28.5**[1]

Table 2 - Service scenarios for the 20.5 Mbps multiplexer of IT.

Note: 1st D = 1st description, 2nd D = 2nd description;
SVC - HD: 1280x704px25 (8 Mbps);
SD: 640x352x25 (1.5 Mbps);
CIF: 320x176x25 (0.5 Mbps)
(All video formats are progressively scanned)

The service described in the first row in the tables is a real-time broadcasting composed by two descriptions, each delivered to a particular network. In the case of transmission over error-prone

[1] In contrast to DVB-T, the WiMAX system uses adaptive modulation techniques, allowing adjusting the signal modulation scheme depending on the actual signal-to-noise ratio (SNR) of the radio link. When the radio link is high in quality, the modulation scheme of highest bit-efficiency is used, resulting in maximum transmission capacity. During a signal fade, the WiMAX system can shift to a more robust but less efficient modulation scheme in order to maintain the connection quality and link stability. This feature allows the system to overcome time-selective fading.

channels, a terminal receiving both descriptions will be able to display a better quality video. In the second row, pre-recorded material will be broadcasted over DVB and its second description can be unicasted over WiMAX to a particular terminal requesting better video quality. In other words, this service is multicasted to all terminals which request a better video quality. This situation can occur mainly in the cities where WiMAX can cover DVB dead zones or when the mobile is moving at high speed. This main pre-recorded material has a hyperlink to a short video. For instance, the viewer is watching a football match and wants to watch a short spot (<10 min) of the best goal scored by one player. The hyperlinked video will then be displayed as an overlay in a corner of the main video. The hyperlinked video requires a low-delay communication and is unicasted (downloaded) to a particular terminal. Therefore, the intelligent play-out will upload the hyperlinked video descriptions through both networks, selecting them intelligently in order to ensure low latency. To ensure low latency, the play-out should reduce the bit-rate allocated to each broadcasted service described in the two top rows in the tables.

In the forth row, the SUIT play-out intelligently serves a terminal with internet content via a unicast connection. To do so, the play-out selects the most appropriate transmission path depending on available empty slots (packets) in each network or by reducing the bit-rate associated to the broadcasted material with negligible quality loss.

Finally, any terminal may request a video streaming service from the play-out server or even from outside the play-out. This service will be unicasted only via WiMAX. Again, the play-out may need to reduce the bit rate associated to the broadcasted material delivered over WiMAX.

In conclusion, a system like an IP inserter looks for empty slots to insert IP (e.g. unicast) traffic. Normally, the broadcasting services are using most of the available bandwidth. Other type of traffic over DVB will happen only in situations where the play-out can manage it without corrupting any of the broadcasted services and simultaneously ensuring a low-delay communications for interactive services (hyperlinked video).

In the future, after the discontinuation of analog TV transmission, it may be technically possible to transmit non-real time unicast services along with broadcasting services. SUIT is focusing on the concept of all-IP but still considers TS (MPEG transport stream) in DVB-T/H.

The service scenarios in Table 1 and Table 2 describe three spatial layers from quasi-CIF to quasi-HD. In the first year, however, SUIT will restrict itself to the service scenarios described in the second line of Table 1 and Table 2**.** We want to demonstrate the advantages of layered MDC as soon as possible. We will do this for two layers, at CIF format, the base layer plus one or two FGS layers. This testbed will be upgraded progressively by adding spatial scalability in order to serve different sizes of terminal screens.

The network and service scenarios presented, above, in this section allow us to define all communication requirements from the terminals and gateway to the play-out. Thus,

a. If the WLAN conditions are getting worse, as for instance, a Wi-Fi terminal is moving away from the Access Point, the gateway is able reduce the MPEG-4 SVC bit rate by cutting the SVC bit-stream. Under this situation and for broadcasting material (first and second rows in Table 1 and Table 2), that particular gateway is not required to inform the play-out because the broadcasting material are also delivered to other homes. However, for unicasted material like streaming (fifth row in Table 1 and Table 2), the gateway should inform the play-out about its decisions. Thus, the play-out can also reduce the streamed bit rate saving WiMAX bandwidth. In order to inform the play-out, the gateway will create an MPEG-21 Usage Environment Description (UED) document. This XML document contains feedback information about the network conditions, such as the bandwidth that is currently available. This document is then sent to the play-out, e.g. using web services, where it is parsed and used to steer the bit-stream adaptation process. It should be noted that the streaming material is also SVC encoded in order to allow an efficient handling by the gateway and play-out.

b. The gateway may request a 2nd Description (second row in Table 1 and Table 2) via WiMAX. If broadcasters wish to charge consumers for the second description, RTSP must be used by the terminal to access this description; else the simpler IGMP protocol can be used by the terminal to subscribe to the 2nd description. We should point out that in Mobile network scenario, the gateway functionalities are moved into each terminal. The 2nd Description request is based on SNR or BER obtained from DVB-T/H CPE (see in 6).

c. The hyperlinked video (third row in Table 1 and Table 2) is the most quality demanding service in SUIT. It must be delivered to the terminal at a bit rate greater or equal to 500 kbps in each last mile network. The play-out will decide to deliver the hyperlinked video to the terminal by using either both last mile networks or one of them dynamically in order to minimize the delay. Therefore, during this transference, the gateway must provide the play-out with the last mile networks conditions to be obtained from DVB and WiMAX CPEs (SNR or BER). As in (a), this information is supplied to the play-out by means of an MPEG-21 UED document.

2.5 Internet access

The gateway must have LB (Load Balance) NAT (Network Address Translator) and DHCP (Dynamic Host Configuration Protocol) configured in order to provide Internet access to the Wi-Fi user terminals. The appropriate scenario should be the one depicted in Figure 7.

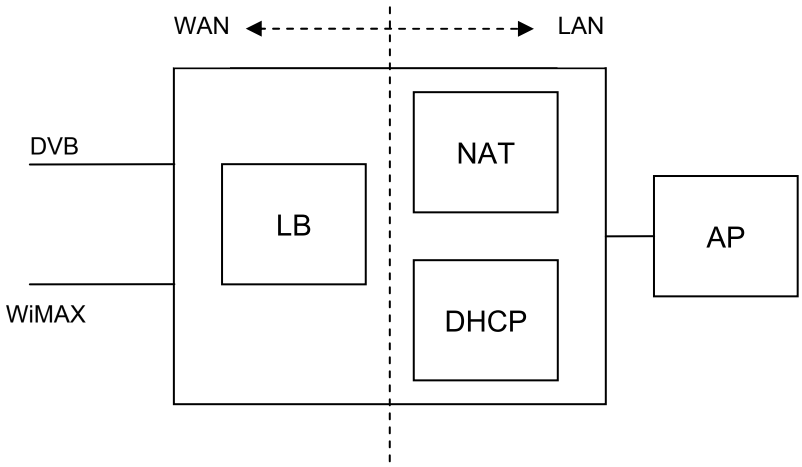

Figure 7 - Internet access in the gateway.

2.5.1.1 LB

In computing, load balancing is a technique (usually performed by load balancers) to spread work between many computers, processes, hard disks or other resources in order to get optimal resource utilization and decrease computing time.

A load balancer consists of a virtual server (also referred to as vserver or VIP) which, in turn, consists of an IP address and port. This virtual server is bound to a number of physical services running on the physical servers in a server farm. These physical services contain the physical server's IP address and port. A client sends a request to the virtual server, which in turn selects a physical server in the server farm and directs this request to the selected physical server.

In the following example we can see the importance of LB. Assuming that we have both WiMAX and DVB networks present and suddenly we just have DVB (for example, if we get out the WiMAX cell coverage) we need a mechanism capable of routing the traffic only through DVB. The LB is responsible for this action.

2.5.1.2 NAT

In computer networking, the process of network address translation (NAT, also known as network masquerading, native address translation or IP-masquerading) involves re-writing the source and/or destination addresses of IP packets as they pass through a router or firewall. Most systems using NAT do so in order to enable multiple hosts on a private network to access the Internet using a single public IP address (see gateway).

In a typical configuration, a local network uses one of the designated "private" IP address subnets (the RFC 1918 Private Network Addresses are 192.168.x.x, 172.16.x.x through 172.31.x.x, and 10.x.x.x), and a router on that network has a private address (such as 192.168.0.1) in that address space. The router is also connected to the Internet with a single "public" address (known as "overloaded" NAT) or multiple "public" addresses assigned by an ISP.

As traffic passes from the local network to the Internet, the source address in each packet is translated on the fly from the private addresses to the public address(es). The router tracks basic data about each active connection (particularly the destination address and port). When a reply returns to the router, it uses the connection tracking data it stored during the outbound phase to determine where on the internal network to forward the reply; the TCP or UDP client port numbers are used to demultiplex the packets in the case of overloaded NAT, or IP address and port number when multiple public addresses are available, on packet return. To a system on the Internet, the router itself appears to be the source/destination for this traffic.

2.5.1.3 DHCP

DHCP is a protocol used by networked computers (clients) to obtain IP addresses and other parameters such as the default gateway, subnet mask, and IP addresses of DNS servers from a DHCP server. It facilitates access to a network because these settings would otherwise have to be made manually for the client to participate in the network.

The DHCP server ensures that all IP addresses are unique, that is, no IP address is assigned to a second client while the first client's assignment is valid (its lease has not expired). Thus IP address pool management is done by the server and not by a human network administrator.

The Dynamic Host Configuration Protocol (DHCP) automates the assignment of IP addresses, subnet masks, default gateway, and other IP parameters. The assignment occurs when the DHCP-configured machine boots up or regains connectivity to the network. The DHCP client sends out a query requesting a response from a DHCP server on the locally attached network. The query is typically initiated immediately after booting up and before the client initiates any IP based communication with other hosts. The DHCP server then replies to the client with its assigned IP address, subnet mask, DNS server and default gateway information.

The assignment of the IP address generally expires after a predetermined period of time, at which point the DHCP client and server renegotiate a new IP address from the server's predefined pool of addresses. Typical intervals range from one hour to several months, and can, if desired, be set to infinite (never expire). The length of time the address is available to the device it was assigned to is called a lease, and is determined by the server.

DHCP is a broadcast-based protocol. As with other types of broadcast traffic, it does not cross a router unless specifically configured to do so. Users who desire this capability must configure their routers to pass DHCP traffic across UDP ports 67 and 68.

3 Gateway architecture

Figure 8 shows a more detailed architecture of the gateway in a general view of the SUIT system. It depicts the gateway functional components. This section will describe each one of these components and their relationships. For the sake of a better clarity, only the play-out to user terminal direction will be described.

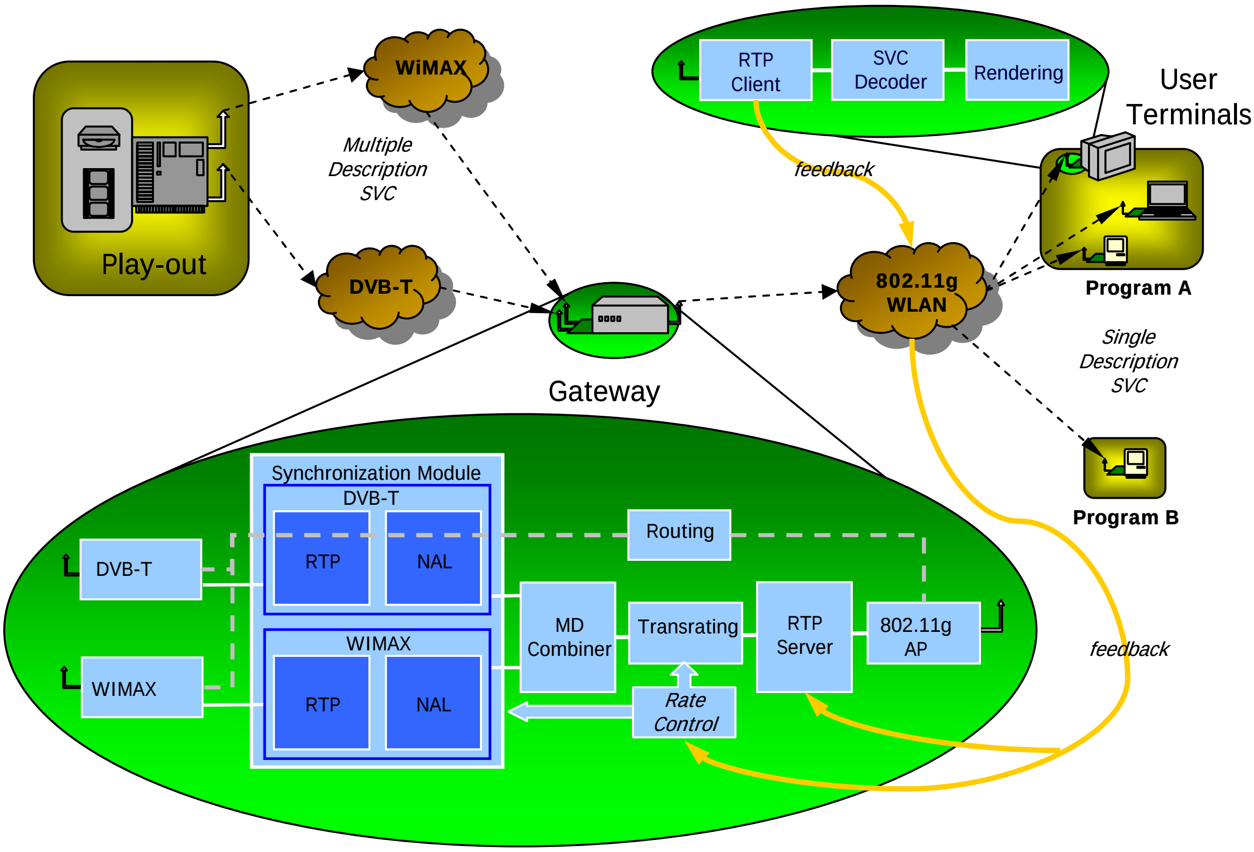

Figure 8 - General view of the SUIT system.

The gateway is divided into three big parts: last mile networks interfaces, gateway core and Wi-Fi network interface.

3.1 Last mile networks interfaces

This is the specific HW to be developed in the SUIT project for interfacing both wireless broadband last mile networks (see D5.3 – “SoC+FPGA Specifications”).

<u>DVB-T/RCT transceiver</u>:

Description: The DVB-T/RCT interface.

Input/Output: DVB-T/DVB-RCT Standards compliant radio signal (2 antennas)

Output/Input: Ethernet layer 2 packets (RJ45 connector).

<u>WiMAX transceiver</u>:

Description: The WiMAX interface.

Input/Output: WiMAX Standard compliant radio signal (1 antenna using Time Division Multiplexing).

Output/Input: Ethernet layer 2 packets (RJ45 connector).

3.2 Gateway core

This part carries out the main functionalities of the gateway. As a sole module, its inputs and outputs are the Ethernet layer 2 packets coming for the three radio interfaces. Specific SW will be developed for the implementation of the majority of its functionalities, based on existing RPT stack libraries.

The required HW will be a PC, running a Microsoft Windows operating system, equipped with 2 Ethernet cards interfacing the two broadband last mile networks interfaces, and another one interfacing the Wi-Fi network interface.

<u>Synchronization module</u>:

Description: This module is in charge of the synchronization of the two video descriptions coming from the two bit-streams paths (WiMAX and DVB). Due to the different delays presented in equivalent packets in both descriptions, to the non reliable transmission and to the possible non ordered reception of the packets, a temporal buffer is necessary for waiting the arrival of them. After the timeout has expired, the packets are synchronized at RTP level (see 2.2.1) and NAL level.

Input: Ethernet layer 2 packets.

Output: Two descriptions NALUs.

<u>MD combiner module</u>:

Description: This module receives two synchronized video descriptions of SVC, combines them (see 2.2.2), and takes out a unique description of the scalable video description.

Input: Two descriptions SVC NALUs.

Output: One description SVC NALUs.

<u>Rate control and transrating modules</u>:

Description: These modules perform rate control to adapt the output bitrate to the varying conditions of the Wi-Fi network channel. The Transrating module will use the Rate Control directives to perform the bit-stream cutting (see 2.3).

Input: One description SVC NALUs, information from the user terminals.

Output: One description SVC NALUs.

<u>RTP server module</u>:

Description: This module is in charge of the RTP packetization of the NALUs coming from the Transrating module. In this sense, SVC bit-stream will be encapsulated following the same approach used at the play-out (see deliverable D4.1 – “Synchronisation and Encapsulation”). In addition, this module is in charge of the establishment of the RTP session with the Wi-Fi terminals for the RPT packets transmission. Moreover, some error protection tasks dealing with the user terminals feedback (information about packet losses, channel quality…) could be implemented. For example, this module, in some circumstances, could retransmit some lost packets.

Input: One description SVC NALUs, information from the user terminals.

Output: Ethernet layer 2 packets encapsulating RTP.

Routing module:

Description: It provides Internet access to the user terminals located in the Wi-Fi LAN. The routing capabilities of the hosting operating system will be used for this purpose.

Input/Output: Ethernet layer 2 packets.

Output/Input: Ethernet layer 2 packets.

3.3 Wi-Fi network interface

This is the HW interfacing the Wi-Fi wireless LAN, where the user terminals are located. No specific HW will be developed in the SUIT project but a commercial Access Point will be use for this purpose.

Access Point:

Description: The Wi-Fi interface. This component will also act as a DHCP server for the user terminals.

Input/Output: Ethernet layer 2 packets (RJ45 connector).

Output/Input: Wi-Fi Standard compliant radio signal (1 antenna).

4 Gateway components

The gateway will use specific HW to be developed in the SUIT project, like the last mile network transceivers from Runcom, and generic HW available in the market, like the router acting as an access point, and the PC acting as the core component (running the specific SW to be developed in the SUIT project).

4.1 DVB-T/DVB-RCT/WiMAX transceivers

4.1.1 DVB-RCT transceiver

DVB-RCT system is a wireless P2MP (Point To Multi Point) system and as all wireless systems the main modules are consist of Base Station (BST) and User Terminal (UT). DVB-RCT system can act as a 2nd layer bridge with Ethernet connections. Paragraph 4.1.1.1 presents the DVB-RCT UT transceiver will be used in the gateway, while, for completeness, Annex 10.1 presents the DVB-RCT BST transceiver.

4.1.1.1 DVB-RCT User Terminal

RN-2821 User Terminal is a complete standard-based (ETSI EN 300 744, ETSI EN 301 958) DVB-RCT UT based around the RN-2821MOD modem (Runcom's chip), in order to achieve the following main characteristics:

- Provide full fast IP applications (Internet, Video over IP, Voice over IP).
- Provide a broadband return channel for interactive applications.
- Enable indoor-to-outdoor communications.
- Provide an OFDMA uplink and a COFDM downlink.
- Provide 20 Mbps uplink and 31.6 Mbps downlink data rates at 8 MHz channel.
- Immunity to multi-path and interference.

The RN-2821UT is a complete modem; it implements Physical (PHY) and MAC layer functions and includes a UHF transceiver. Runcom RN-2821 implements the PHY and low-MAC layer, while the higher MAC layer is implemented in SW in a MIPS32 based CPU.

Runcom will provide, upon request, a comprehensive package of RN-2821MOD documentation and schematics for CPE or STB developers and manufacturers.

The UT consists of the followings main modules and interfaces:

- RN-2821MOD modem
- UHF RF transceiver
- Interfaces
- 220 V AC to 12 V DC PS

The RN-2821MOD implements the PHY and MAC layers of the DVB-T/DVB-RCT standards (EN 300 744 / EN 301 958). DVB-T is used for the downstream path, and DVB-RCT is used for the upstream path/return channel. A block diagram of the module is shown below in Figure 9

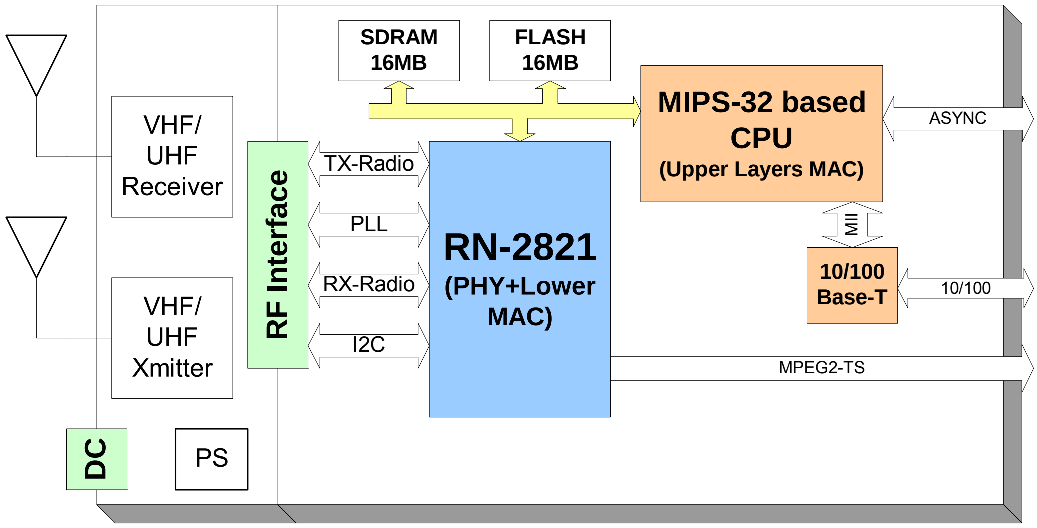

Figure 9 - RN-2821MOD block diagram.

The RN-2821MOD includes a 16MB SDRAM, 16MB FLASH memory, MAC CPU, Ethernet PHY and the RN-2821 chip.

The CPU implementing the MAC functionality with embedded RN-2821MAC software is a MIPS-32 based derivative (AU-1000 from AMD).

Communications with the host CPU is intended, primarily, to take place over the common interface. When the module is not connected via the common interface, the USB, ASYNC or 10/100BaseT interfaces may be used for communicating with the application processor. Communications between the application processor and the MAC CPU are done via a logical interface, called the MAC-API, with a specific driver to support the selected physical interface.

The 10/100BaseT interface is intended to be available directly to the CPE user. These interfaces may connect directly to a PC or to a home-gateway. By virtue of their high data-rate capability, these interfaces provide added value services, such as Fast Internet access and VoIP.

The UHF RF bi-directional transceiver consists of a DVB-T receiver (tuner) and a DVB-RCT transmitter packed closely together. The board is designed to use two antennas, or it can be used with one antenna connected to duplexer as an option.

The following described the applications available in the UT:

All “three play” applications:

- Video (MPEG 2 TS).
- Voice (Voice over IP).
- Data (Turbo internet, FTP, Video over IP and all IP applications).

Figure 10 shows a block diagram of the RN-2821 chip.

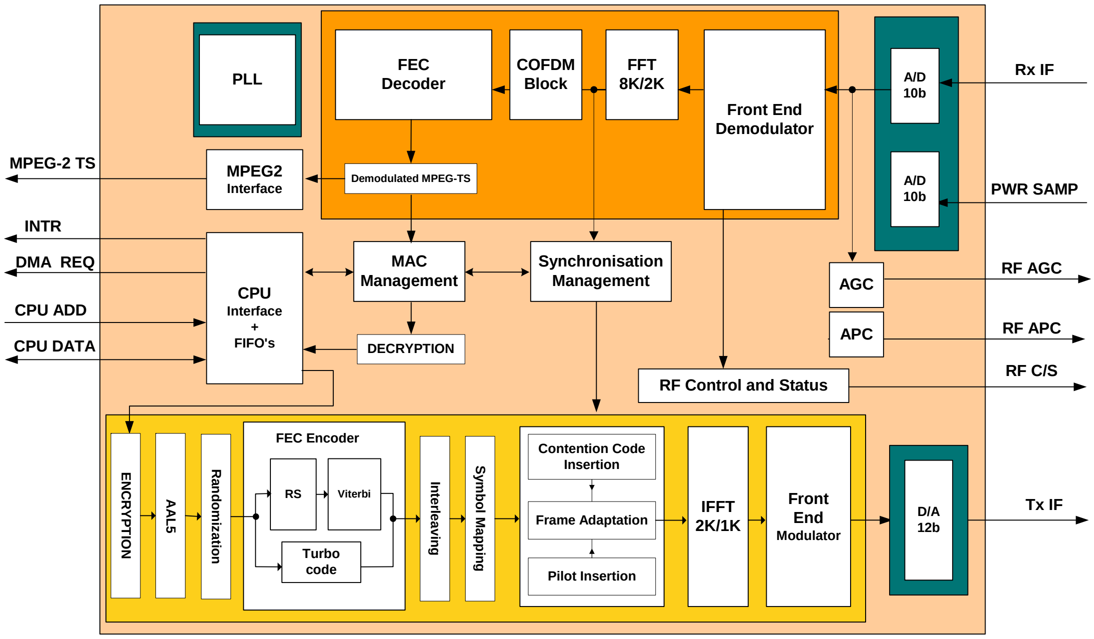

Figure 10 - RN-2821 block diagram.

The RN-2821 is a full duplex, COFDM/OFDMA standard modem designed as a System-on-a-Chip (SoC) for CPE or STB development and manufacture. The RN-2821 has operation modes for various point- to-multi-point broadband Interactive Terrestrial TV implementations and data applications. The RN-2821 implements a DVB-T COFDM downlink receiver (ETSI EN 300 744 standard compliant), based on a 2K or 8K FFT, and can drive a 31.6 Mbps @ 8 MHz bandwidth. The RN-2821 implements a DVB-RCT OFDMA (ETSI EN 301 958 standard compliant) upstream transmitter based on a 2K or 1K FFT. It can drive up to a 25 Mbps @ 8 MHz bandwidth for the upstream channel.

The chip implements a PHY layer and lower MAC layer. It integrates internal analog modules for 10-bit ADC, 12-bit I/Q DAC, IQ modulator/demodulator, 2K FFT.

The FEC block includes Turbo Code, Concatenated RS and Tail Biting convolutional coding 1/2, 2/3, 3/4, 5/6, 7/8. The chip supports QPSK, 16-QAM and 64-QAM modulations.

The chip interfaces to a 32 bit data CPU that implements higher MAC functions. The chip interfaces with an MPEG2-TS decoder chip or decoder core, integrated into the application CPU. It interfaces with low IF TX /RX signals to RF modules.

The chip includes block aggregation, which eases the MAC block structuring. The MAC data is received as MPEG-2 TS packets. This data is transferred only after parsing the MAC data through a CPU/DMA bi-directional interface. The chip also integrates a 240 MHz PLL and six channels of 1-bit Sigma-Delta.

The RN-2821 is designed in advanced 0.18 µm CMOS technology. The chip package is 300 pins LBGA.

The following is the connectors list:

- RF Out: 50 ohm SMA connector.
- RF In: 75 ohm BNC connector.
- DATA: for monitoring RS-232 UART connector

  for IP payload Ethernet 10/100BT RJ45 connector.
- Video Out.
- DC Input: 12 V, two pin.

For mobile application the following list is recommended:

- DC to DC converter (12 V to 28 V).
- 5 W power amplifier – to achieve better coverage.
- Duplexer – in order to operate with one antenna.

4.1.2 WiMAX transceivers

WiMAX (for mobile environment –IEEE802.16e) system is a wireless P2MP (Point To Multi Point) system and as all wireless systems the main modules are consist of Base Station (BST) and User Terminal (UT). WiMAX system can act as a 2nd layer bridge with Ethernet connections. Paragraph 4.1.2.1 presents the UT WiMAX transceiver will be used in the gateway, while, for completeness, Annex 10.2 presents BST WiMAX transceiver.

4.1.2.1 WiMAX User Terminal

WiMAX UT has the following main characteristics:

Modem

- Compatibility with IEEE 802.16e and 802.16-2004 standards OFDMA mode.
- TDD operation.
- Full Air Link Capacity.
- Adaptive modulation, 64QAM, 16QAM, QPSK, with up to 5 b/s/Hz spectral efficiency.
- Supports 1K FFT and channel BW up to 20 MHz (8.75 MHz for SUIT project).
- Supports advanced Convolutional Turbo Code (CTC) FEC scheme.
- Open and closed loop power control with CQI reporting.

System

- Full MAC & PHY Layers.
- Advanced Quality-of-Service.
- Supports ARQ.
- Low Power, supports SLEEP and IDLE mode operation.
- Security implementation based on AES-CCM.
- Supports HO in both reuse=1 and reuse<1 operation scenarios.

Radio

- High-performance dual ADC and DAC support base-band I/Q radios.
- On-Chip, wide dynamic range, AFC, AGC and APC control loops.
- Complete reference design includes and integrated radio solution.

Integration

- High performance ARM-11 processor implements MAC functionality.
- Integrated interfaces for SDRAM and Flash memories.
- Multiple Integrated interfaces include PC-Card, Cardbus, USB 2.0, SDIO, USIM, Ethernet 10/100.

Figure 11 shows the WiMAX UT block diagram.

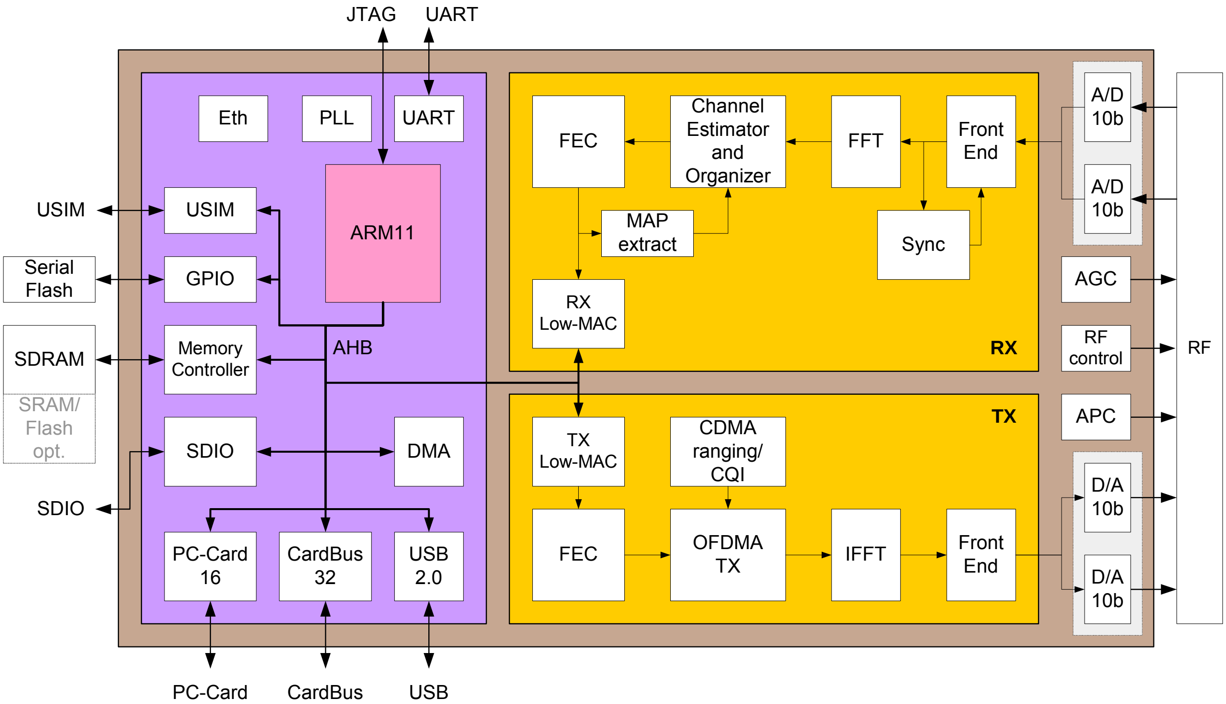

Figure 11 - WiMAX TU block diagram.

The ASIC implements the PHY and MAC layers of the IEEE802.16e specification. The upper MAC layers are implemented in SW on an ARM-11CPU, which requires external SDRAM and serial or parallel FLASH memories. The low-MAC and the PHY up to the analog interface to the RF unit are implemented in HW. The ASIC can interface a host processor through USB 2.0, Compact Flash (CF), Card Bus, Host-port, Ethernet and SDIO. USIM and UART interface are also provided.

4.2 Gateway core

The hardware for the gateway core will be a general purpose computer with 2 Ethernet cards and a connection to the Access Point (Ethernet/USB). The computer will be a PC with at least a Pentium 4 processor, and running a Microsoft Windows operating system.

4.3 Access point (Wi-Fi networks)

The considered Access Point for the Wi-Fi IEEE 802.11g LAN will be an 802.11b/g router for WLAN networking with multicast capabilities. This router must be sufficiently flexible to allow the modification of the main Wi-Fi protocol configuration parameters. A router from the Cisco 800 series for small offices has been selected for this purpose.

5 Terminal functionalities

5.1 Introduction

The SUIT project is dealing with two kinds of terminals:

- a SUIT terminal that is able to receive MD-SVC video streams over DVB-T and WiMAX combined outdoor networks;
- a user terminal (Wi-Fi) that can deal with single SVC video streams sent over an indoor WLAN network.

The SUIT terminal has to provide some of the gateway functionalities in addition to the Wi-Fi user terminal. More specifically:

- Stream management, since it is needed to generate a single SVC video stream from the MD-SVC video streams to feed the SUIT decoder.
- Communication with the play-out. In some situations, feedback from the SUIT terminal to the play-out as well as to the gateway is convenient to report about the statistics of the transmission.

A detailed description of these functionalities can be found in Section 2. Therefore, in this section, attention will be focused on user terminals fed with video streams from a home gateway.

5.2 Video Formats

The use of scalable video streams enables to send a same stream to different capabilities terminals. The SUIT project is mainly addressing a range of terminals equipped with progressive screens with a 16:9 aspect ration. It corresponds to the following kinds of terminals:

- handheld terminals that are used to display CIF pictures;
- personal computers on which SD streams can be shown;
- and TV sets that can performed HD video streams.

All these different formats can be merged into a single scalable video description in which a terminal can find the right resolution to display.

Corresponding bit-stream characteristics are listed in Table 3 (see other deliverables for further information, e.g. D3.1 – “Design of scalable multiple-description video coding”).

Format	Resolution	Frame rate	Bit rate
CIF	320 x 176	25 Hz	0.5 Mbps
SD	640 x 352	25 Hz	1.5 Mbps
HD	1280 X 704	25 Hz	8 Mbps

Table 3 - Video Format List.

5.3 Video Decoder Characteristics

Knowing its screen resolution and its computing power, a user terminal must be able to select in the incoming bit-stream the right information to display on its screen and a convenient frame rate for refreshing pictures. In an SVC bit-stream, three different types of scalability are multiplexed together:

- temporal scalability that allows to select the right frame up to a maximum value;

- spatial scalability that allows to find the right video format;
- and SNR scalability that allows to choose a given level of visual quality.

Spatial and SNR quality scalabilities are built starting from a base layer information to which has been added enhancement or refinement layers. Select the right information is then equal to keep the only necessary layers that enable to display a stream at a given resolution and with a given quality.

In order to satisfy bandwidth availability, the gateway may adapt the bit-stream by discarding some refinement layers. In this case, the decoder must be able to perform its task by decoding the remaining information present in the bit-stream, even if it is not fulfilling the capabilities of the terminal.

At the opposite, a handheld terminal including a low resolution screen and low computation power may only need to decode and render base layer information. As SVC encoder is defined as the extension of the AVC standard, only an AVC decoder will be necessary for processing base layer information from an SVC bit-stream.

5.4 Video Rendering

In order to keep low computational expenses for rendering, it should be preferred to use the graphics processor for rendering the decoded video sequence. It should happen when such a processor may support rendering libraries like DirectX, OpenGL or SDL.

5.5 Encapsulation and Transport Protocols

Video encoded streams should be encapsulated in RTP packets to insure their real-time transports. The terminal must be able to process them and to retrieve synchronisation information provided by RTP timestamps. A transport session must be controlled by the gateway according to the information sent by the terminal about the Quality of Service observed during the session, using RTCP protocol, or by solutions specifically developed for the SUIT project, if required for rate control purposes.

5.6 Service Discovery

The terminal should be able to select a service among several proposed in the SUIT network. Interpretation of service announcement and session protocols must apply (see D4.2 – “Session protocols” for more details).

6 Terminal architecture

Figure 12 shows the user terminal functional components. This section will describe each one of these components and their relationships.

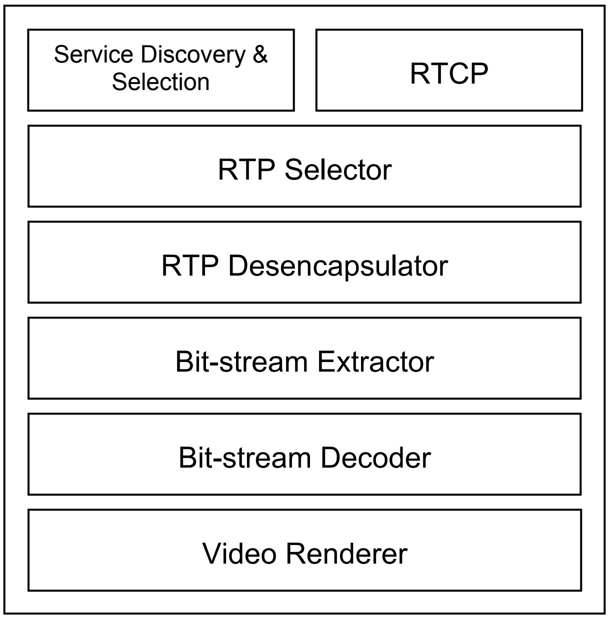

Figure 12 - User Terminal Architecture.

<u>Service Discovery & Selection module:</u>

Description: This module receives information from the play-out, via the gateway, in order to know the available service. This is not a primary module in the context of SUIT objectives.

Input: Information from the SUIT network.

Output: Available service.

<u>RTCP module:</u>

Description: RTCP or other solutions specifically developed for the SUIT project, which allow reporting about the statistics of the transmission in either the WLAN or in the last mile networks (no gateway case).

Input: RTP packets.

Output: Statistics on lost packets during the transmission.

<u>RTP Selector:</u>

Description: Extracts the RTP packets corresponding to video stream to display. It is expected in SUIT not to multiplex different layers bits in one RTP packet. Thus, a terminal interested in decoding only the lower layers does not need to decapsulate the RTP packets related to upper layers.

Input: RTP packets and address and port information.

Output: Selected RTP packets.

RTP Decapsulator:

Description: It enables to receive video streams over the Wi-Fi network and depacketizes the encapsulated bit-stream.

Input: RTP packets containing the video stream.

Output: SVC or AVC video bit-stream.

Bit-Stream Extractor:

Description: For SNR (FGS) scalability, the terminal may not be able to decode the full SNR bit stream due to lack of computational capabilities.

Input: SVC or AVC video bit-stream.

Output: SVC or AVC video bit-stream.

Video Bit-stream Decoder:

Description: It decodes the incoming bit-stream and rebuilds a video sequence of pictures. It is made from an AVC decoder if it is a low performance terminal (handheld) or an SVC decoder if it has higher computational performances (personal computer or HDTV set).

Input: SVC or AVC video bit-stream.

Output: YUV sequence of pictures.

Video Renderer:

Description: It displays the decoded video. It can be embedded into HTML code (webpage) in order to have a hyperlink and test interactivity.

Input: YUV sequence of pictures.

7 Conclusions

This document presents the specifications for the design of the gateway within the SUIT system. Its main functionalities have been identified and a general architecture has been proposed.

In a home network scenario, the gateway acts as the interface between the broadband networks (DVB-T and WiMAX) and the wireless local area network (Wi-Fi), connecting in this way the play-out with the user terminals. It has to provide a set of functionalities which can be divided into four categories:

- Stream management, to combine MD-SVC streams into a single SVC stream.
- WLAN transmission, which involves a rate control module (that adapts the single SVC stream to the WLAN dynamic characteristics) together with an RTP packetization module.
- Communication with the play-out, to report the play-out about the statistics of the transmission.
- Internet access.

The gateway will be implemented using general components like a PC (for the core functionalities) with Ethernet cards, and an access point (for interfacing with the Wi-Fi network), and specific components developed within the SUIT project like the broadband networks transceivers (for interfacing with the DVB-T and WiMAX networks).

Secondly, the user terminal has been addressed. Two different types of terminals will be used in the SUIT project:

- a SUIT terminal that is able to receive MD-SVC video streams over DVB-T and WiMAX combined outdoor networks;
- a user terminal (Wi-Fi) that can deal with single SVC video streams sent over an indoor WLAN network.

The main functionalities of the user terminal are related to the tasks of RTP management, SVC decoding, rendering and session management. In addition, the SUIT terminal has to provide some of the functionalities provided by the gateway (stream management and communication with the play-out).

8 Acronyms

ASIC	Application-Specific Integrated Circuit
AU	Access Unit
AVC	Advanced Video Coder
BER	Bit Error Rate
BST	Base Station
CIF	Common Intermediate Format
COFDM	Coded Orthogonal Frequency-Division Multiplexing
CPE	Customer Premises Equipment
CPU	Central Processing Unit
CTC	Convolutional Turbo Code
DHCP	Dynamic Host Configuration Protocol
DMA	Direct Memory Access
DNS	Domain Name System
DVB	Digital Video Broadcasting
DVB-IPI	Digital Video Broadcasting- Internet Protocol Infrastructure
DVB-H	Digital Video Broadcasting-Handhelds
DVB-RCT	Digital Video Broadcasting- Return Channel Terrestrial
DVB-T	Digital Video Broadcasting-Terrestrial
EMDSQ	Embedded Multiple Descriptions Scalar Quantizers
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FGS	Fine Grain Scalability
PHY	Physical Layer
FPGA	Field Programmable Gate Array
IGMP	Internet Group Management Protocol
GOP	Group Of Pictures
HD	High Definition
HDTV	High Definition Television
HTML	HyperText Markup Language
IP	Internet Protocol
ISP	Internet Service Provider
LB	Load Balance
MAC	Medium Access Control
Mbps	10^6 bit/s
MDC	Multiple Description Coding
MDSQ	Multiple Descriptions Scalar Quantizers

MD-SVC	Multiple Description Scalable Video Coder
MHP-IPTV	Multimedia Home Platform -
MMDS	Multichannel Multipoint Distribution Service
MPEG	Moving Picture Expert Group
NAL	Network Abstraction Layer
NALU	Network Abstraction Layer Unit
NAT	Network Address Translation
OFDMA	Orthogonal Frequency Division Multiple Access
PER	Packet Error Rate
PLL	Phase-Locked Loop
PPS	Picture Parameter Set
PS	Power Supply
P2MP	Point To Multi Point
QoS	Quality of Service
R-D	Rate-Distortion
RF	Radio Frequency
RTP	Real-time Transport Protocol
RTCP	RTP Control Protocol
RTSP	Real Time Streaming Protocol
SD	Standard Definition
SDC	Single Description Coding
SDL	Simple DirectMedia Layer
SD&S	Service Discovery and Selection
SEI	Supplemental Enhancement Information
SNR	Signal-to-Noise Ratio
SoC	System-on-a-Chip
SPS	Sequence Parameter Set
STB	Set-Top-Box
SVC	Scalable Video Coding
RX	Reception
TCP	Transmission Control Protocol
TS	Transport Stream
TX	Transmission
UART	universal asynchronous receiver/transmitter
UDP	User Datagram Protocol
UED	Usage Environment Description
UT	User Terminal
VCL	Video Coding Layer

VIP	Virtual IP address
VoIP	Voice over IP
UHF	Ultra High Frequency
WAN	Wide Area Network
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network
XML	eXtensible Markup Language

9 References

[1] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, “RTP: A Transport Protocol for Real-Time Application", RFC 1889, (Jan. 1996).

10 Annexes

10.1 DVB-RCT Base Station Transceiver

Runcom's RN-BS28 Base Station is a cost effective solution for Broad-Band wireless data IP applications and Interactive Digital Terrestrial TV. RN-BS28 is DVB-T/DVB-RCT-standard compliant and uses COFDM/OFDMA technology to leverage broadband wireless communications in both downstream and upstream transmissions.

The RN-BS28 technology satisfies the increasing demand for Fast internet, all IP data and high quality digital television broadcasting merged with data services in all existing bands, from VHF/UHF and up to the MMDS frequencies.

The RN-BS28 enables the development of a Base Station site with the following main characteristics:

- Provides all fast IP and data applications based on IP (Video over IP, Voice over IP etc.)
- Provides a broadband return channel for interactive applications.
- Enables indoor-to-outdoor communication.
- Provides an OFDMA up link and a COFDM downlink.
- Standard based (ETSI EN 300 744, ETSI EN 301 958).
- Provides data rates of 20 Mbps uplink and 31.6 Mbps downlink @ 8 MHz channel.
- Provides immunity to multi-path and interference.

The Base Station site that can be developed with the RN-BS28 is extendable, from one sector supported by two antennas (TX and RX), up to many sectors supported by an equivalent number of directional antenna. The RN-BS28 can be configured locally through its front panel RS-232 port using any terminal emulator application, or remotely through its Ethernet port.

The following are the BST features:

- Highly efficient operation in non-line-of-sight, enabling both indoor-to-outdoor and outdoor-to-indoor operation.
- Supports OFDMA upstream and COFDM downstream.
- Supports the DVB-T Standard (EN 300 744) for downstream transmission.
- Supports the DVB-RCT standard (EN 301 958) for upstream transmission.
- Uses 2K FFT size with BS3 mode for Broadband upstream.
- Uses 2K FFT size for Broadband downstream.
- Supports sub-channelization for upstream transmissions from 1 and up to 59 sub-channels.
- Supports automatic power adjustment to meet variable transmission conditions.
- Supports up to 8 MHz channel bandwidth. (Higher BW's is upon customer request)
- Supports dynamic bandwidth allocation.
- Supports adaptive modulation schemes upstream and downstream: QPSK, 16QAM, 64QAM.
- Uses either advanced turbo coding or concatenated RS and convolutional coding for upstream FEC.
- Maximum number of CPEs/STBs per BST modem: 8192.

- High efficiency: up to 4 bit/sec/Hz for downstream, and up to 3.5 bit/sec/Hz for upstream.
- Supports RX and TX diversity.
- Supports DES-56 Encryption & Decryption for downstream and upstream transmissions.
- Increases coverage and immunity in adverse multi-path environment.
- Outstanding performance in terms of Phase Noise and Backoff levels compared to OFDM systems.
- Available as an Add-On-Init to implement return channel for existing DVB-T base stations.
- Supports outdoor RF module with dual IF cables.
- Scalable solution: from single sector to multiple sectors Base Station, and from supper-cell to micro-cell networks.

General system view is presented in Figure 13:

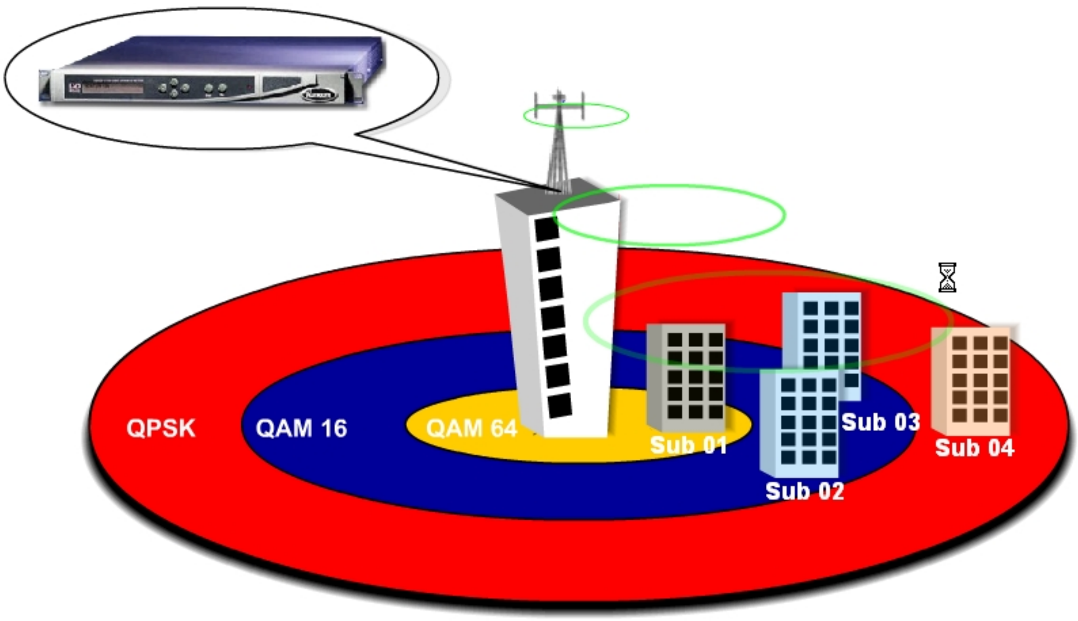

Figure 13 - General system view.

A complete Base Station is packaged in an enclosure with the following main components:

- Back-plane.
- Radio transceiver (RF module) and adapter.
- Power supply unit.
- RN-BS28PM card.

An LCD, keypad and LED panel on the front panel of the unit provide operator controls and indicators.

The RN-BS28 block diagram is shown in Figure 14.

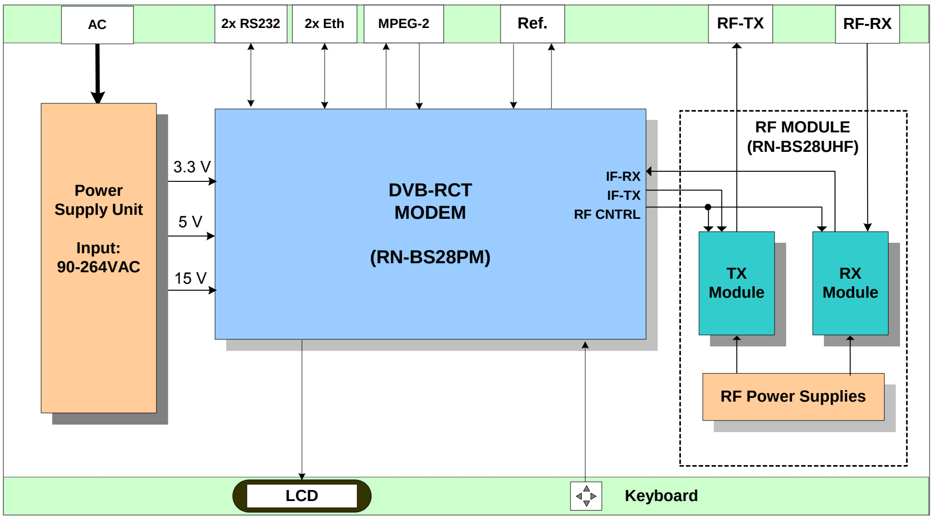

Figure 14 - RN-BS28 Block Diagram.

Table 4 lists the interface connectors of the BST.

INTERFACE	TYPE	DESCRIPTION
General		
ETHERNET	RJ45	Ethernet traffic interface
ETHERNET	RJ45	TBD
RF-TX	N-Type-50Ω	Transmit signal to antenna or IF interface to ODU.
RF-RX	N-Type-50Ω	Receive signal to antenna or IF interface to ODU.
DIVERSITY	50-pin	TX/RX Diversity port
POWER	AC 3-pin	90 – 264 V AC power input (2 A)
Time and Frequency Reference		
GPS	DB-9	GPS receiver control port (RS-232/RS-485)
REF-IN	BNC-50Ω	10 MHz clock reference input from external GPS receiver
	BNC-50Ω	1PPS time reference input from external GPS receiver
REF-OUT	BNC-50Ω	10 MHz clock reference output
	BNC-50Ω	1PPS time reference output
External DVB-T Modulator Support		
Tx-MOD-CTRL	DB-9	External DVB-T modulator control
MPEG2-IN	BNC-75Ω	ASI port out
MPEG2-OUT	BNC-75Ω	ASI port in

Table 4 - Interface Description.

The BST consists of a complete modem, including MAC and PHY layers. The BST is a full duplex COFDM/OFDMA modem implementing the DVB-RCT standard for Base Stations. The RN-BS28 has various operation modes for P-MP fixed Broadband Wireless Access and Broadband Interactive TV implementations and Data applications.

The RN-BS28PM implements a DVB-T COFDM Downstream modulation transmitter supporting 2K or 8K IFFT modes, that can drive a 31.7 Mbps @ 8 MHz Bandwidth. This Bandwidth represents more than 12 movie channels on a single analog PAL channel. For MMDS implementations, it is capable of driving a 45.6 Mbps @ 12 MHz Bandwidth.

The RN-BS28PM implements the DVB-RCT OFDMA de-modulator receiver supporting 2K or 1K IFFT modes, capable of receiving up to 27 Mbps @ 8 MHz Bandwidth for the Upstream channel. RN-BS28PM is also designed for MMDS implementations, being capable of driving a 40.5 Mbps @ 12 MHz Bandwidth.

The RN-BS28PM modem integrates high speed and high-density FPGA chips and implements COFDM/ FDMA PHY layer functions and lower MAC and FEC encoding/decoding.

Figure 15 shows the functionality block diagram of the RN-BS28.

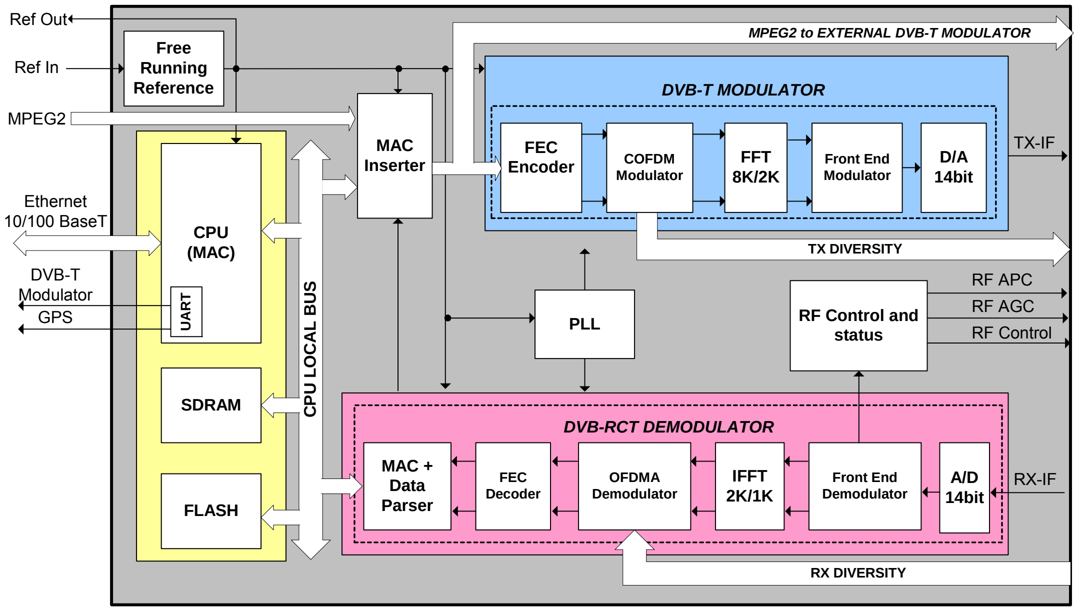

Figure 15 - RN-BS28PM Block Diagram.

Table 5 shows the BST specification.

ITEM	DESCRIPTION
Standard Compliance	Downstream ETSI EN 300 744 DVB-T Upstream ETSI EN 301 958 DVB-RCT Implementing Burst Structure 3 (BS3)
Transmission Technology	Downstream: COFDM. Upstream: OFDMA
FFT Size	Downstream: 2K. Upstream: 2K
Bandwidth Channelization	8 MHz
Number of Sub Channels for Return Channel (BS3)	59 Sub Channels for 2K size mode 29 Sub Channels for the 1K size mode
Modulation Schemes	QPSK, 16-QAM, 64-QAM downstream and upstream
Guard Intervals	1/4, 1/8, 1/16, 1/32 for both directions (for upstream, rectangular shaping only)

Interleaving	Downstream: Ramsey type 3 approach with I = 12 Upstream: based on PRBS
Coding	Downstream: Concatenated RS and Convolution Coding rates: 1/2, 2/3, 3/4, 5/6, 7/8. Upstream: Turbo code or Concatenated RS and Convolution Coding rates: 1/2, 3/4
Duplexing Techniques	FDD
Efficiency	Downstream: Up to 4 bits per Second per Hertz Upstream: Up to 3.5 bits per Second per Hertz
Throughput	Downstream: Up to 31.67 Mbps (for 8 MHz channel) Upstream: Up to 20 Mbps (for 8 MHz channel)
RF control signals	AGC range: 70 dB APC range: 64 dB
IF Frequency of Operation	30 – 44 MHz (downstream and upstream)
CPU	PPC750CXE-500
Operating Temperature	0 °C – 45 °C

Table 5 - BST specification.

10.2 WiMAX Base Station transceiver

Figure 16 below shows the mechanic of RNU2000 in a BST - Single Sector Unit.

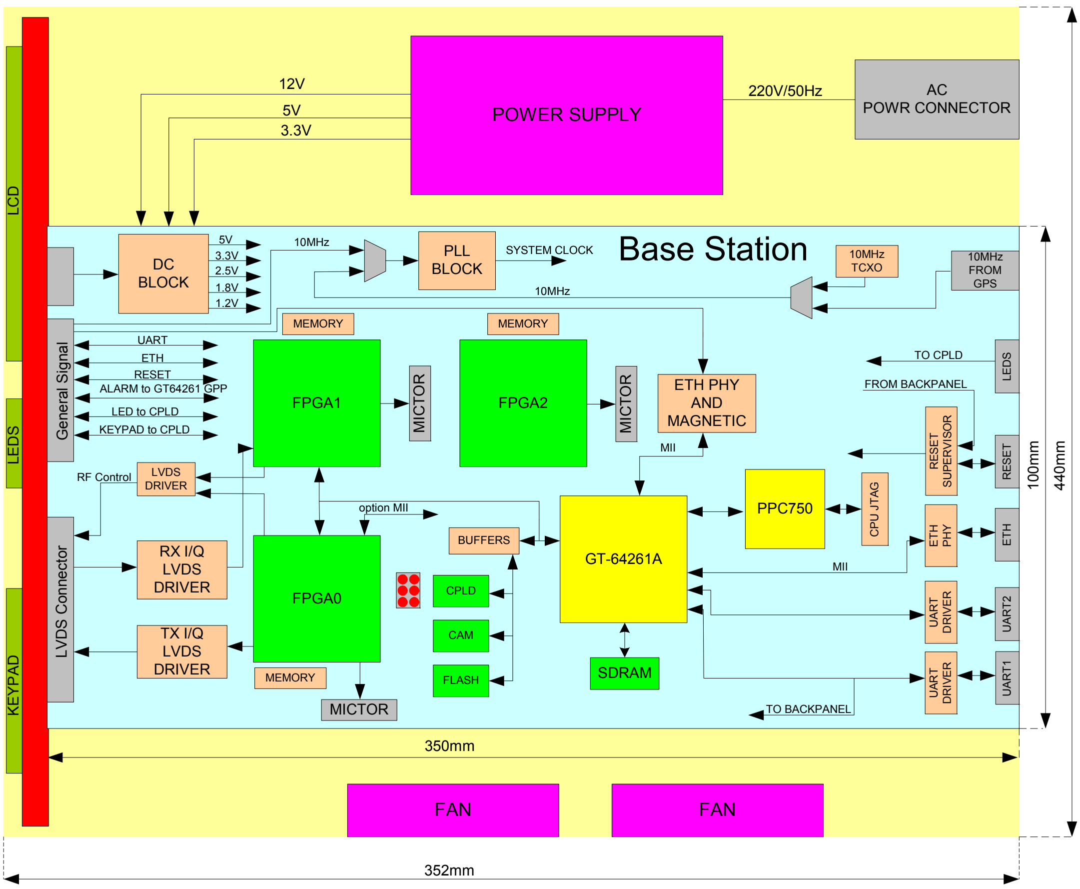

Figure 16 - RNU2000N Block Diagram

Figure 17 and Figure 18 show the front and rear side of the stand alone BST. It is compatible for rack mountable, provided in 19" length, 2U high and 30 cm deep.

Figure 17 - RNU2000 Front Panel.

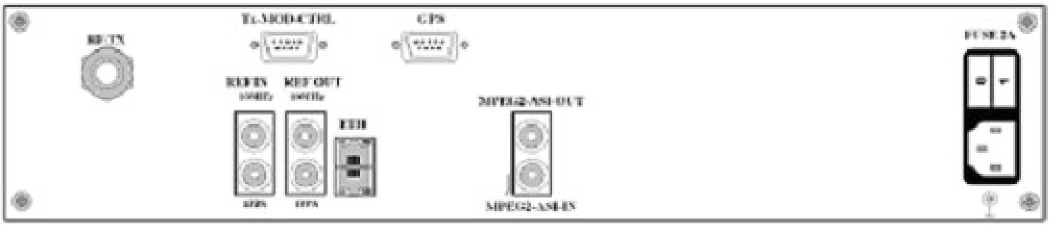

Figure 18 - RNU2000 Rear Panel.

Table 6 describes the external interfaces.

INTERFACE	TYPE	DESCRIPTION
General		
2 x Ethernet 10/100BT	RJ45	One interface is used for both Ethernet traffic interface and control messages.
RF-TX&RX	N-Type-50Ω	Transmit signal to antenna or IF interface to ODU.
POWER	AC 3-pin	90-264 V AC power input (2 A)
Time and Frequency Reference		
REF-IN	BNC-50Ω	10 MHz clock reference input from external GPS receiver
	BNC-50Ω	1PPS time reference input from external GPS receiver
REF-OUT	BNC-50Ω	10 MHz clock reference output
	BNC-50Ω	1PPS time reference output
Control		
Tx-MOD-CTRL	DB-9	RS-232 local terminal

Table 6 - Interface Description.

The block diagram in Figure 19 shows the PHY (consists of Tx and Rx chains) and MAC (CPU and peripherals).

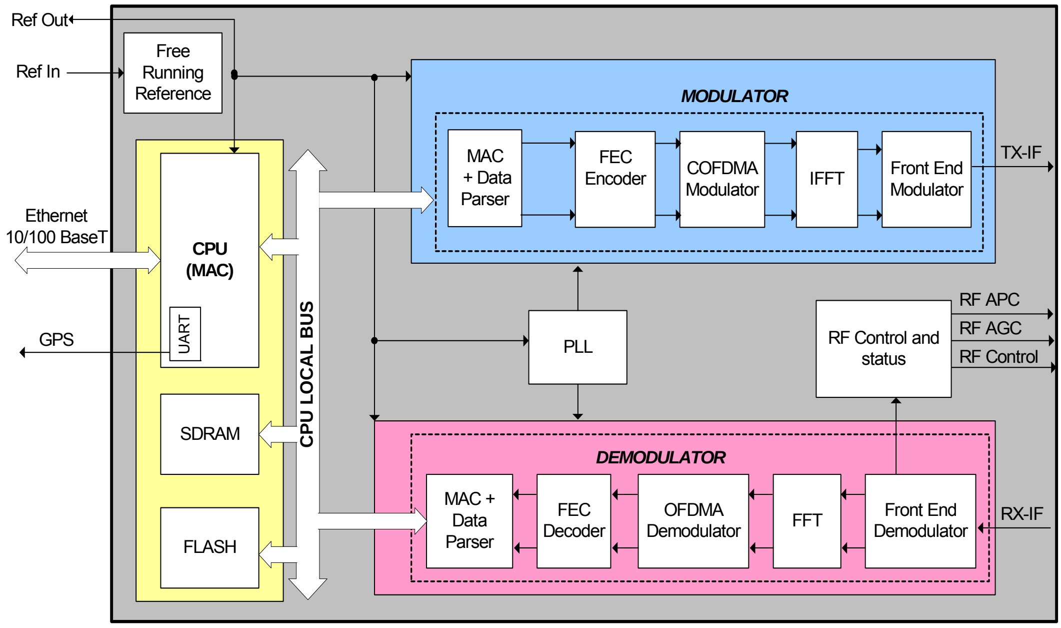

Figure 19 - PHY and MAC modem block diagram