PUBLIC

IST R&D. FP6-Priority 2.
SPECIFIC TARGETED RESEARCH PROJECT
Project Deliverable

Doc Number	SUIT_427
Project Number	IST-4-028042
Project Title	SUIT
Deliverable Nature	Report
Deliverable Number	D1.5
Contractual Delivery Date	30.09.2007
Actual Delivery Date	15.11.2007
Title of Deliverable	Architecture and Reference scenarios
Contributing Workpackage	WP1
Dissemination Level	PU
Author(s)	Alois Zistler (IRT), Christoph Dosch (IRT), Moti Goldshstein (Runcom), Francesc Enrich (URL), Antonio Navarro (IT), Olivier Guye (Vitec), Juergen Lauterjung (R&S)

Abstract

This document describes the SUIT architecture as well as the proposed references scenarios for research and demonstration activities. It is an updated version of Deliverable D1.4.

Keyword list: Architecture, reference scenarios

PUBLIC

Architecture and reference scenarios

SUIT_427

15-11-2007

Table of Contents

1 Introduction

1.1 Scope

This document is part of WP1 - Architecture Requirements and Specification - and relates to Activity 1.4 ”Architecture and Reference Scenarios”.

It has strong relationships to other Activities within WP1 as well as to other Work-packages, in particular to WP4 “Playout, Terminal and Networks Resource Management and Optimisation”.

Deliverable D1.5 is the second document resulting from Activity A1.4. It corresponds to an improved version of the previous edited document D1.4, especially in respect of planned user scenarios.

1.2 Objective

One main objective of this document is to give an overview of the SUIT architecture supporting broadcast and non-broadcast services over converged broadcast/broadband networks encompassing DVB-T/DVB-RCT, DVB-H and WiMAX.

An important part of the overall architecture is the playout. Hence, special emphasis is given to the main units of the playout – like multiplexers, servers, encoders, packet inserters and network interfaces – in this document.

Besides the requirements of the end-to-end system, scenarios are described according to two main axes:

- Spatial and channel organization of video transmission (network reference scenarios)
- Some kind of usage (service scenarios)

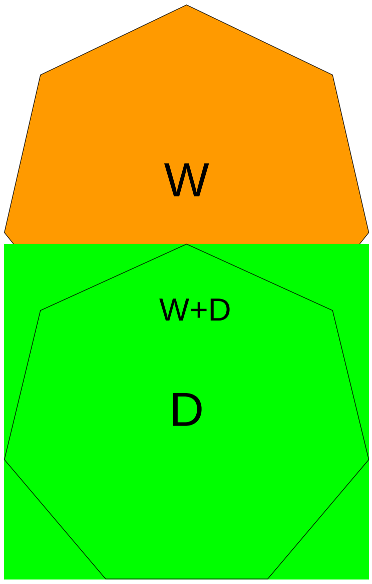
These two dimensions can help us to highlight two different kinds of results:

- Technical innovations for the first one
- Support for service innovations for the second one

By distinguishing them, we expect to get results that are more valuable than these obtained by translating technical innovations into value-added usages. This should be of advantage in respect of exploitation and dissemination activities (WP7).

2 Network Reference Scenarios

Possible SUIT network scenarios resulting from the availability of converging networks are outlined in the following sub-sections. In extension of the "Description of Work", SUIT is focusing on tools for both DVB-T and DVB-H.



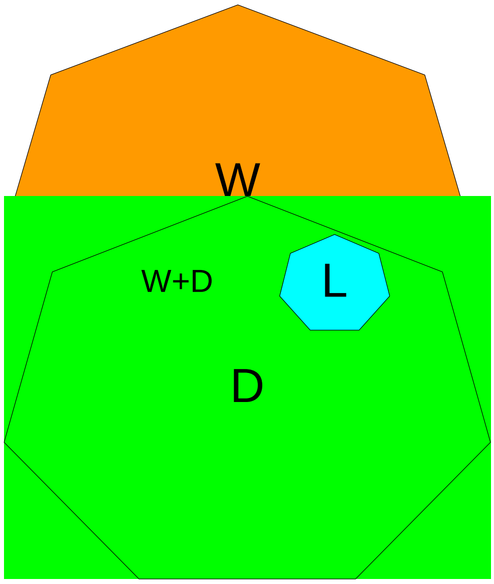
2.1 Network scenario no. 1 (DVB-T / WiMAX / DVB-T+WiMAX)

A WiMAX cell (W) and a DVB-T cell (D) share an overlapping coverage area where both networks are accessible (W+D). Table 1 shows the demonstration classes which are relevant for this network scenario. SUIT proposes MDC (Multiple Description Coding) as a solution to address vertical handover issues for broadcast services.

Type of Demonstration	relevant	Remarks
Mobility, Speed	x	
Handover	x	"hard" vertical
Interactivity		
Low Delay	x	
Scalability		
Intelligent Multiplexing	x	
JSCC+UPA (Extended Coverage)	x	
Multiple Description	x	seamless vertical handover

Table 1: Demonstration classes for network scenario 1

2.2 Network scenario no. 2 (DVB-T / WiMAX / DVB-T+WiMAX+WiFi)



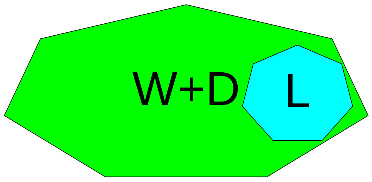
Similar to network scenario 1, a WiMAX cell and a DVB-T cell share an overlapping coverage area (W+D) where both networks are accessible. Additionally, in network scenario 2, a local WiFi spot (L) is available (WLAN 802.11g) within the overlapping coverage area.

Type of Demonstration	relevant	Remarks
Mobility, Speed		
Handover		
Interactivity	x	
Low Delay		
Scalability	x	
Intelligent Multiplexing		
JSCC+UPA (Extended Coverage)		
Multiple Description	x	

Table 2: Types of demonstration for network-scenario 2

Network scenario 2 is a dedicated scenario for the use of the gateway interfacing with the WiFi terminal.

2.3 Network scenario no. 3 (DVB-T+WiMAX / WIFI)

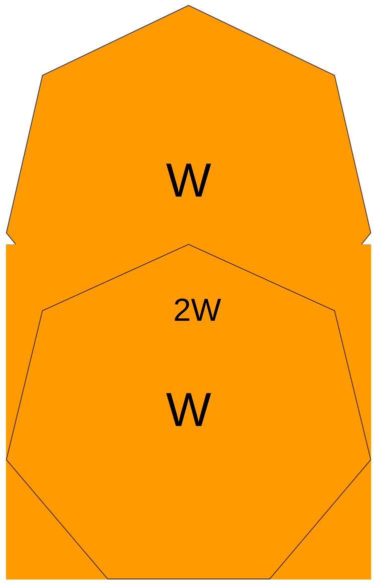


In network-scenario 3, there is only one cell in which both WiMAX and DVB-T are accessible. A local WiFi spot (L) is located within this coverage area.

Type of Demonstration	**relevant**	**Remarks**
Mobility, Speed	**x**	
Handover		
Interactivity		
Low Delay	**x**	
Scalability		
Intelligent Multiplexing		
JSCC+UPA (Extended Coverage)	**x**	
Multiple Description	**x**	

Table 3: Types of demonstration for network-scenario 3

2.4 Network scenario no. 4 (WiMAX / WiMAX)

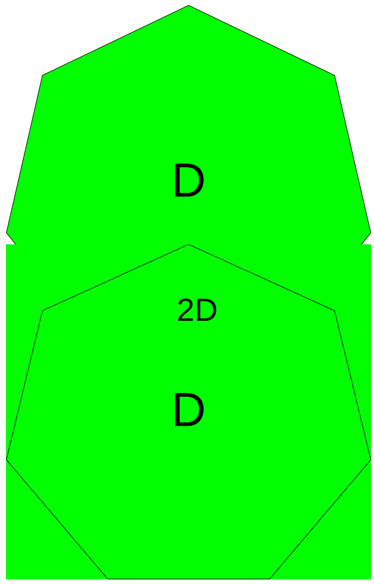


Two separate WiMAX cells share an area (2W) in which connectivity to both cells is available.

Type of Demonstration	**relevant**	**Remarks**
Mobility, Speed		
Handover	**x**	horizontal
Interactivity		
Low Delay		
Scalability		
Intelligent Multiplexing	**x**	
JSCC+UPA (Extended Coverage)		
Multiple Description		

Table 4: Types of demonstration for network-scenario 4

2.5 Network scenario no. 5 (DVB-T / DVB-T)



Two separate DVB-T cells share an area (2D) in which connectivity to both cells is available.

Type of Demonstration	relevant	Remarks
Mobility, Speed		
Handover	x	horizontal
Interactivity		
Low Delay		
Scalability		
Intelligent Multiplexing	x	
JSCC+UPA (Extended Coverage)		
Multiple Description		

Table 5: Types of demonstration for network scenario no.5

Note: In case of DVB-RCT (uplink channel), 2 pairs of frequencies are required as DVB-RCT is a FDD system. Both, downlink channel (DVB-T) and uplink channel (DVB-RCT), operate in separate frequencies. To demonstrate horizontal handover based on RUNCOM's solution using DVB-RCT, each cell requires a set of two frequencies (UL+DL). If each user throughput in UL is low, it can be considered to use 1 frequency (instead of 2) and to share the BW allocations between all users.

2.6 Summing up demonstration activities

Main technical innovations in SUIT are:

1. To design scalable multiple-description video coding approaches adaptable to the dynamic network characteristics and the multitude of user terminals types.
2. To design adaptive joint source and channel coding techniques (JSCC+UPA) for optimal network resource allocation while taking advantage of source scalability and channel conditions.
3. To optimize the overall rate-distortion performance for a given set of user preferences (in terms of resolution and frame-rate) and fixed network conditions.
4. To implement handover and Intelligent Multiplexing.

Table 6 summarizes the technical innovations to be demonstrated in the various network scenarios.:

Scenario / **Concept**	**1**	**2**	**3**	**4**	**5**
Mobility, Speed	**x**		**x**		
Handover	**x**			**x**	**x**
Interoperability		**x**			
Low Delay	**x**		**x**		
Scalability		**x**			
Intelligent Multiplexing	**x**			**x**	**x**
JSCC+UPA (Extended Coverage)	**x**		**x**		
Multiple Description	**x**	**x**	**x**		

Table 6: Summary of network scenarios and technical features demonstrated

2.7 BSTs location

In order to fulfill all network scenarios mentioned in the previous sections, we selected the following locations (marked in yellow) for placing the Basestations as shown in Figure 1. The distance between each site is 1.5 km.

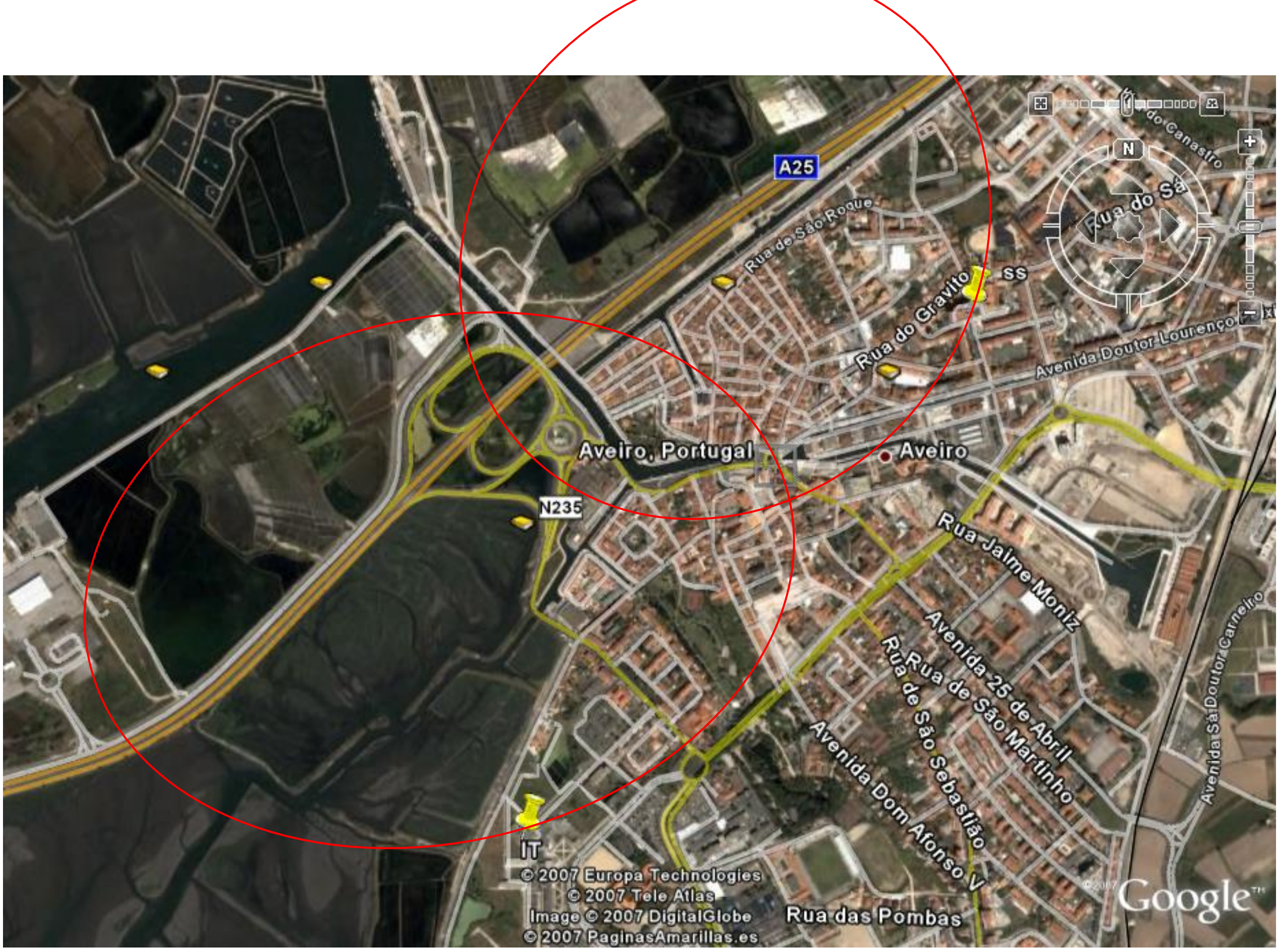


Figure 1: BSTs Location

The UHF frequencies requested to the Portuguese frequency regulation body are:

BST SS – DL -> canal 43, UL -> canal 53

BST IT – DL -> canal 57, UL -> canal 67

We are still considering using only one channel for both BSTs UL, channel 67. The two WiMAX frequencies are 10 MHz bandwidth centred somewhere from 2.5 to 2.6 GHz.

3 Service scenarios

3.1 Quad-play services bundle presentation

Presently main attractive service offers refer to triple-play services. They consist in bundles proposed on broadband networks and are commonly composed of the following basic services:

- local and international phone calls;
- broadband Internet access;
- a given bunch of digital TV and radio channels.

Compared to previous passive ways of audiovisual content consumption, innovation in triple-play services stands in proposing interactive services like:

- presence services (push-to-talk and instant messaging);
- unified messaging (text, voice and video);
- service on demand (gaming and video on demand).

They are gathered in the term of IPTV in which interactivity is provided by a simultaneous use of Internet and TV.

Mobility appears now as a new major source of services and can be added to previous bundles of services to provide quadruple-play services. Dual-band handsets have been recently launched to provide a continuity of service inside home, but they are still limited by the use of WiFi technology that does not allow to play roaming services outside home.

In this scope, WiMAX technology should soon afford a true broadband access that should be widely deployed and that could be efficiently used in conjunction with a DVB-T/H receiver for providing a full bundle of interactive multimedia services, continuously accessible on the move. That is the aim of the SUIT project to demonstrate that a truly usable quadruple-play offer can be now set up.

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 playout to the terminal, where the terminal can feed an HDTV screen or a small pocket-sized display.

Since the introduction of a broadcast operator in SUIT consortium, the initial service scenarios described in the previous deliverable have been modified in the following way:

- now each cell is containing a DVB-T/H base station as well as a WiMAX one;
- only 20 Mbps multiplexes, either for DVB-T/H or WiMAX, are taken in account;
- a full multimedia bundle of services will be demonstrated, that includes Internet, voice and video;
- more video channels are allocated on each multiplex.

Concerning interactivity and mobility:

- Internet and VoIP are mainly implemented over WiMAX. However, in some cases, the user can also have those unicast services over DVB-T/H and DVB-RCT;
- QoS is still ensured by transmitted multiple descriptions over the two networks DVB-T/H and WiMAX;
- Horizontal handover and vertical handover will provide the continuity of services when moving.

According to the kind of implemented MDC scheme, the bandwidth allocation for providing services may vary:

- schemes using unbalanced descriptions will need only a narrow bandwidth to transport redundant information;
- at the opposite, balanced schemes share equally non-redundant and redundant information over the two networks.

So, SUIT is proposing an unique model of services but implemented in three different ways (MDC-1, MDC-2, MDC-3) depending on the amount of redundancy of broadcasting services over WiMAX.

The first one, MDC-1, repeats the baselayer. The second, MDC-2, generates two descriptions for all layers at the expense of 50% redundancy. The third, MDC-3, is similar to MDC-1, but both descriptions at the baselayer have 50% redundancy instead of 100% as occurring in MDC-1. As described in some previous deliverables (for instance D4.4), herein three table services associated with service bitrates. Concerning the abbreviations used in them, their meanings are the following:

- 1D and 2D, first and second descriptions;
- CIF is corresponding to a common video format 320x176@25fps (delivered at 0.5 Mbps);
- SD is corresponding to a standard video format 640x352@25fps (delivered at 1.5 Mbps);
- HD is corresponding to a high definition format 1280x704@25fps (delivered at 4.0 Mbps);
- SVC SD is including CIF and SD video resolutions;
- SVC HD is including HD video resolution moreover.

SUIT is also proposing an Electronic News Gathering system using SD resolution.

3.2 MDC-1: Repeating information

DVB-T		WiMAX	
Service	Bit Rate (Mbps)	Service	Bit Rate (Mbps)
1 D SVC HD Real Time Broadcasting	4 – 6	2 D SVC HD Real Time Broadcasting	0.5
1 D SVC HD Recorded Broadcasting	4 – 6	2 D SVC HD Broadcasting (on QoS demand)	0.5
1 D SVC SD Real Time Broadcasting	1.5 – 2	2 D SVC SD Real Time Broadcasting	0.5
1 D SVC SD Recorded Broadcasting	1.5 – 2	2 D SVC SD Recorded Broadcasting	0.5
1 D CIF Hyperlinked Video	0.5	2 D CIF Hyperlinked Video	0.5
Internet	1 p.u	Internet	1 p.u
VoIP	0.080 p.u	Streaming (VoD)	0.5-4 p.u
		VoIP	0.080 p.u
Total	11-17.5	Total	2-20

Table 7- MDC-1, Network/services scenarios

3.3 MDC-2: Dual quantization

DVB-T		WiMAX	
Service	Bit Rate (Mbps)	Service	Bit Rate (Mbps)
1 D SVC HD Real Time Broadcasting	3.125-4.625	2 D SVC HD Real Time Broadcasting	3.125-4.625
1 D SVC HD Recorded Broadcasting	3.125-4.625	2 D SVC HD Recorded Broadcasting (on QoS demand)	3.125-4.625
1 D SVC SD Real Time Broadcasting	1.25-2	2 D SVC SD Real Time Broadcasting	1.25-2
1 D SVC SD Recorded Broadcasting	1.25-2	2 D SVC SD Recorded Broadcasting	1.25-2
1 D CIF Hyperlinked Video	0.5	2 D CIF Hyperlinked Video	0.5
Internet	1 p.u	Internet	1 p.u
VoIP	0.080 p.u	Streaming (VoD)	0.5–4 p.u
		VoIP	0.080 p.u
Total	8.75-17.5	Total	8.75-20

Table 8- MDC-2, Network/services scenarios.

In MDC-2 mode with a 50% overhead and assuming SVC HD resolution, the lowest bit rate is given by [(4-0.5)*1.5]/2+0.5=3.125 Mbps. Both descriptions are balanced.

3.4 MDC-3: Splitting information

DVB-T		WiMAX	
Service	Bit Rate (Mbps)	Service	Bit Rate (Mbps)
1 D SVC HD Real Time Broadcasting	3.875 - 5.875	2 D SVC Real Time Broadcasting	0.375
1 D SVC HD Recorded Broadcasting	3.875 - 5.875	2 D SVC Broadcasting (on QoS demand)	0.375
1 D SVC SD Real Time Broadcasting	1.375 - 1.875	2 D SVC SD Real Time Broadcasting	0.375
1 D SVC SD Recorded Broadcasting	1.375 - 1.875	2 D SVC SD Recorded Broadcasting	0.375
1 D CIF Hyperlinked Video	0.375	2 D CIF Hyperlinked Video	0.375
Internet	1 p.u	Internet	1 p.u
VoIP	0.080 p.u	Streaming (VoD)	0.5-4 p.u
		VoIP	0.080 p.u
Total	9.6-17.5	Total	1.5-20

Table 9- MDC-3, Network/services scenarios.

In the MDC-3 mode, we considered that two descriptions are generated for the baselayer at the CIF resolutions and consequently required baselayer bit rate is lower than in MDC-1 and MDC-2 modes.

3.5 Services description

Each table describes the service types, the transmission network and the correspondent bit rate range.

The service described in the first row is a real time broadcasting composed by two descriptions, each delivered to a particular network. In the case of transmission over error prone channels a terminal receiving both descriptions will be able to display a better quality video.

In the second row, a 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 requesting 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 on a corner on the top of the main video.

Third and forth rows describe the real time and pre-recorded broadcasting services in the standard resolution (SD) version. As described before, these services use both networks in order to reach higher performance and video quality, and to cover DVB dead zones.

In the fifth row is described the hyperlinked video service, as described above, which requires a low delay communication and is unicasted (downloaded) to a particular terminal. Therefore, the intelligent playout will upload the hyperlinked video descriptions through both networks, selecting them intelligently in order to ensure low latency. The terminal/gateway may compose one video stream with 1D slices from each description. In other words, it is not required that the terminal/gateway has to compose a stream from both full descriptions. Besides, If needed to ensure low latency, the playout should reduce the bit rate allocated to each broadcasted service described in previous rows in the tables, up to the minimum limit, i.e. for MDC-3, it means 3.875 (1.375) Mbps. This is also a strategy for Internet as described below.

In sixth row, the SUIT playout will serve (unicast), again intelligently, a terminal with internet contents. Unused bandwidth by the first four services (Real Time Broadcasting, Recorded Broadcasting, SD Real Time Broadcasting and SD Record Broadcasting) will be used to serve internet service requests. If this available bandwidth is fully used, then playout will select the most appropriate network depending on available empty slots (packets) in each network or by reducing the bit rate associated to the broadcasted material with negligible quality loss. However, under a combined network solution (WiMAX+DVB-T) and the same channel conditions, it is expected that most of Internet traffic will be delivered over WiMAX.

Finally, last row defines VoIP service in DVB network and Streaming in WiMAX network. VoIP service needs a low network bandwidth since each phone call requires just about 80 kbps. Over WiMAX network, any terminal may request a video streaming service from the playout server or even from outside the playout. Again, the playout may need to reduce the bit rate associated to the broadcasted material delivered over WiMAX, especially in MDC-2.

4 Architecture

4.1 Reference Architecture

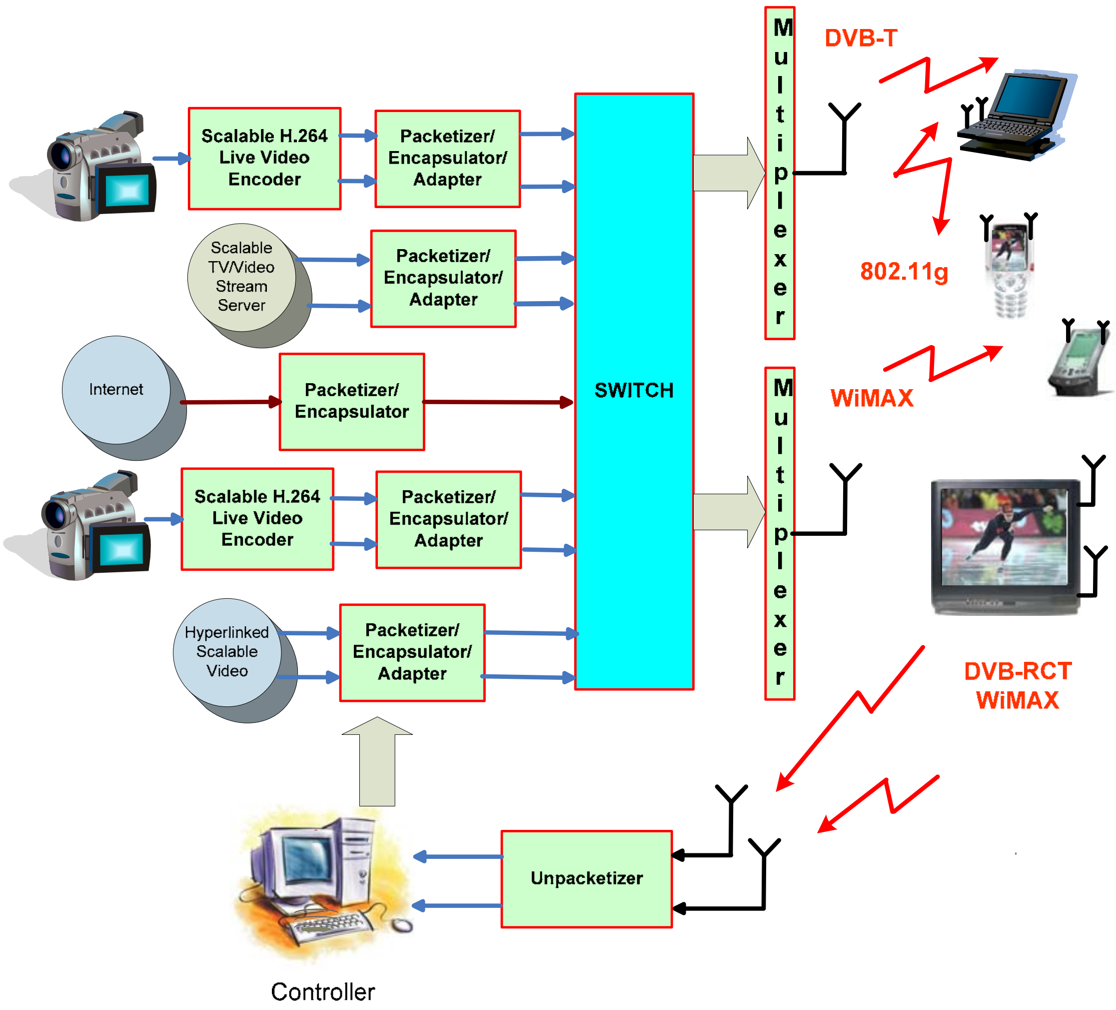


Figure 2: Suit overall architecture

Figure 2 above outlines the overall SUIT reference architecture. The radio interfaces are fed by live scalable contents, pre-recorded scalable contents and internet data. The playout will dynamically and optimally manage all those resources and adapt them according to the network conditions. Once the SUIT terminal interfaces to the WLAN, it will deal locally with instantaneous variations of QoS and thus minimizing the effects by reacting as swift as possible. The connection between the playout and the transmitters will be by the core network, possibly radio links that allow us to implement different field trials easily.

In SUIT, there will be two types of end-user terminals to demonstrate the various network and service scenarios:

- WiMAX/DVB-T/H/-RCT terminal – to demonstrate -amongst others- mobility by reception of multiple description scalable video content (including handover)

- WiFi terminal – to receive rate-adapted content via a IEEE 802.11g (WiFi) connection from a deployed WiMAX/DVB-T/H gateway

Consequently, different use conditions are envisaged as described in the following three sections.[1]

4.2 Network scenario – Platform 1 – Home

The WiFi 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 playout about the network and terminal characteristics. The playout 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 3** shows two homes and therefore two home gateways. In each home two terminals are connected to each gateway.

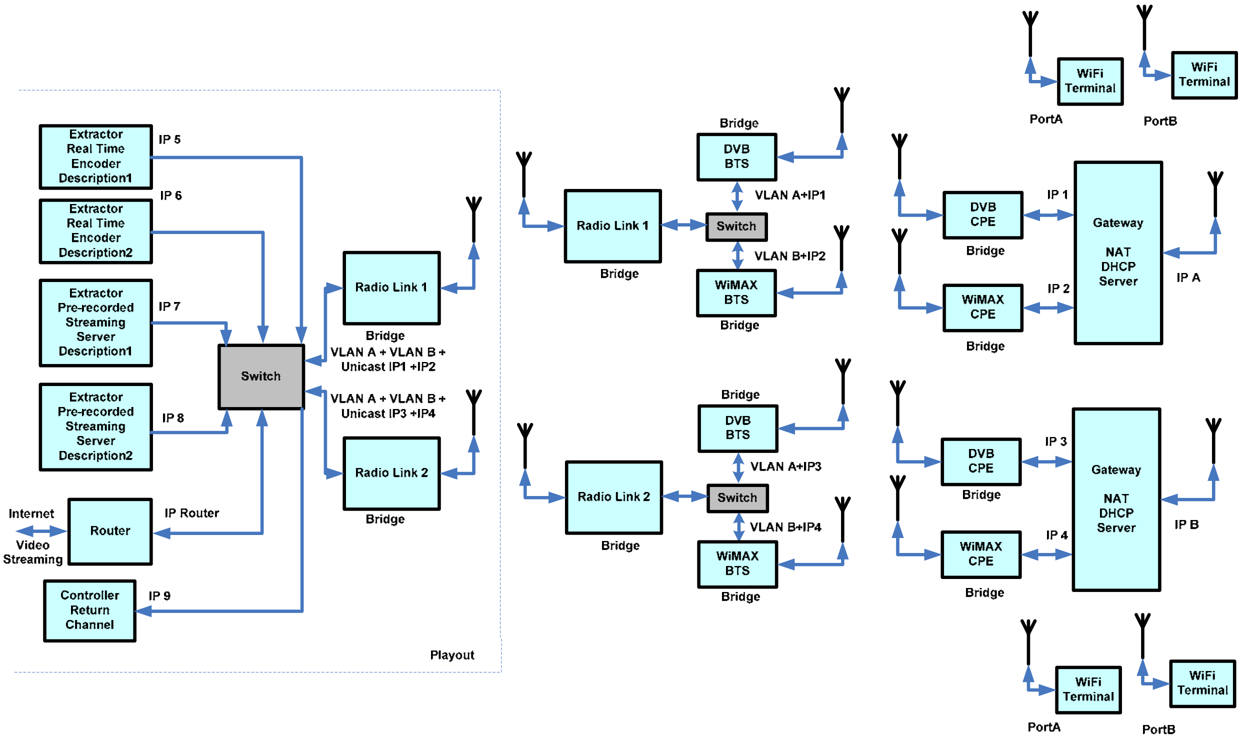


Figure 3: Block diagram of the home network scenario

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. The big constraint is that in most countries UHF frequencies are widely used by analogue and/or digital broadcast transmitters. We are, however, confident to obtain the necessary temporary test licenses.

[1] Refer to Deliverable D1.3 for more in-depth information on the IP requirements of these platforms.

4.3 Network scenario – Platform 2 - Mobile

For the sake of simplicity, we assume one transceiver in each cell. In this scenario, the WiMAX/DVB-T/H/RCT terminal (as shown in **Figure 4**) replaces the gateway of the home network scenario. In other words, some of the gateway functionalities, like the combiner, have moved to the terminal. In order to cope with handover issues, the controller within the playout receives information about the network characteristics and the terminal capabilities to redirect the information from the playout by acting on the switch.

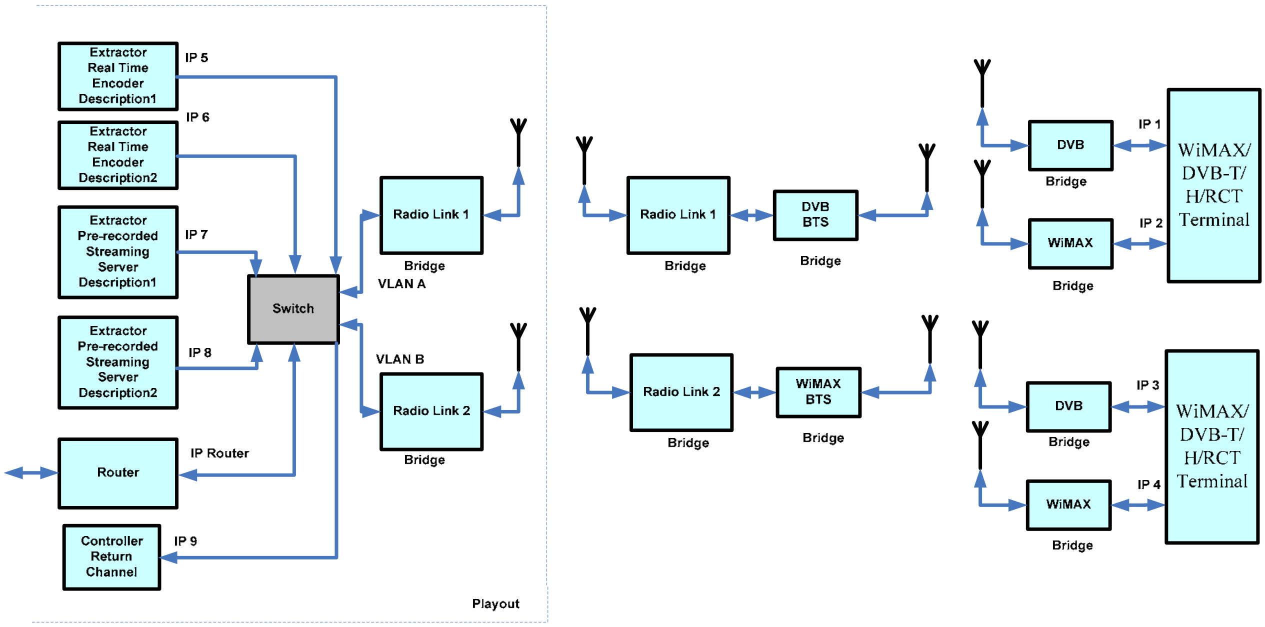


Figure 4: Block diagram of the mobile network scenario

4.4 Playout architecture

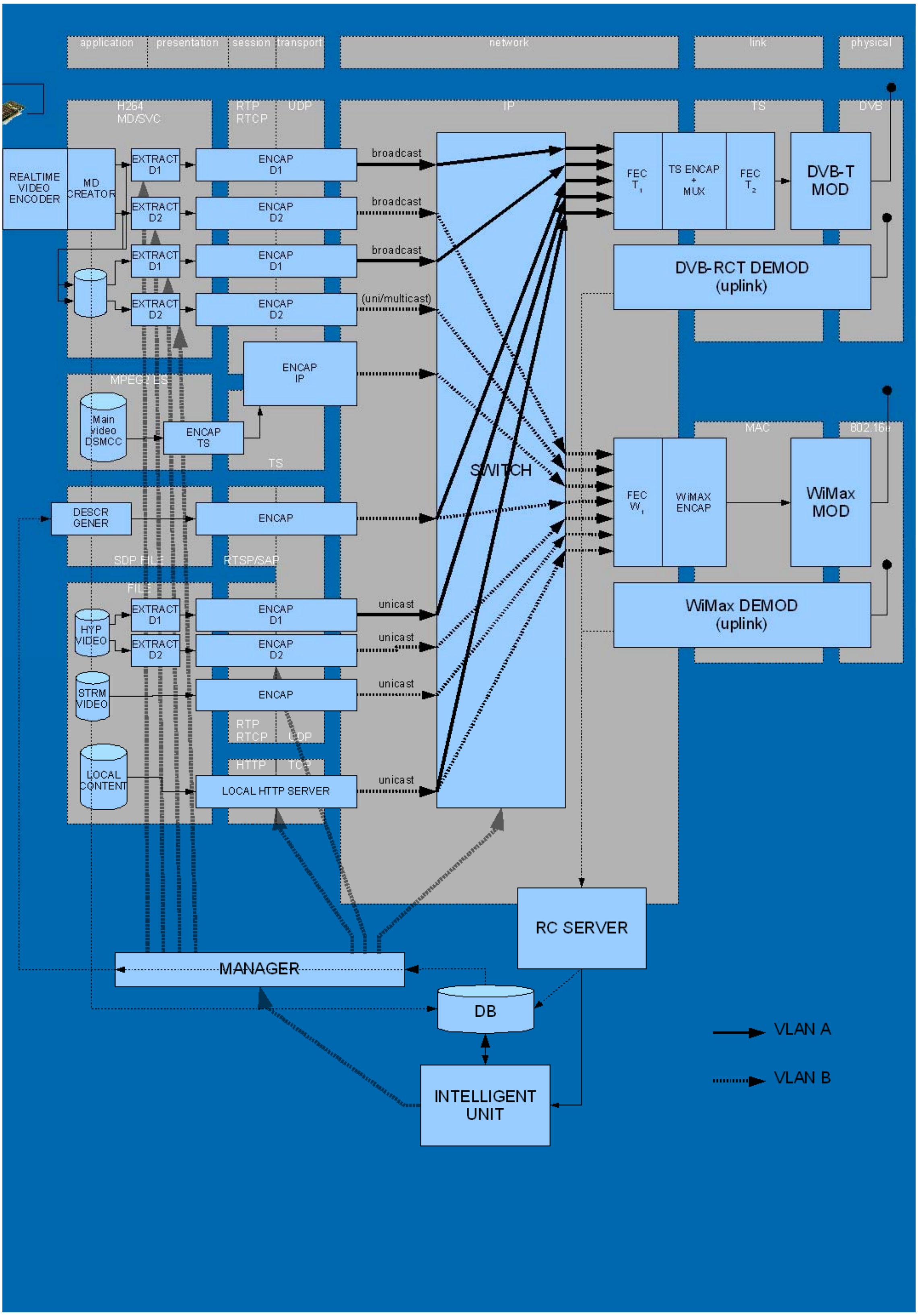


Figure 5: SUIT Playout Architecture

Figure 5 illustrates in detail the components required for the SUIT playout. The different protocols used in SUIT are depicted step-by-step from content suppliers to front-end equipment. The illustrated structure is solely conceptual; which means that in a final implementation separate modules can be combined for optimization.

Next, the module functionalities are detailed:

4.4.1 H.264 MD/SVC content

- **Realtime MDC encoder + extractor + encapsulator (D1- D2)**

  This module is developed within WP3 and WP5, and supplies two MDC descriptions in H.264/SVC format over RTP/UDP/IP. It also generates RTCP packets.

  Depending on the network bandwidth available, the *intelligent unit* or the *manager* informs the extractor to select the correct bitrate using MGS or CGS techniques.

  The extractor and the encapsulator for the two descriptions will be implemented in a stand-alone application.

  This module deals with the real-time video scenario.

- **Extractor + encapsulator (D1-D2)**

  The extractor module delivers the NAL (network abstraction layer) units of two descriptions from a pre-recorded H.264/SVC video. The encapsulator performs the RTP/UDP/IP encapsulation from this output. It also generates RTCP packets.

  The extractor and the encapsulator for the two descriptions would be implemented in a stand-alone application.

  This module deals with the pre-recorded video scenario.

4.4.2 - File content:

- **Hyperlinked video**

  The extractor module delivers the NAL units of two descriptions from a pre-recorded H.264/SVC video when a web page link is selected. The encapsulator performs the RTP/UDP/IP encapsulation from this output. It also generates RTCP packets. This scenario is conducted as a unicast session.

- **Video streaming**

  The encapsulator performs the RTP/UDP/IP encapsulation from a pre-recorded H.264/AVC file. The session is initiated and controlled by the RTSP protocol It generates also RTCP packets. This scenario is conducted as a unicast session.

- **Local content**

  The Internet content (HTTP) should be redirected in a transparent way through the switch to the client.

4.4.3 - Description content:

- **Session & description protocols (SDP, SD&S)**

  This module is intended to generate all the session descriptions and the service discovery information delivered to the SUIT terminals.

  The SD&S information will be delivered using the DVBSTP protocol.

  In order to announce multicast sessions, SDP over HTTP will be implemented.

  In order to announce to control and announce streaming applications SDP over RTSP will be implemented.

4.4.4 - Network:

- **Switch**

  This module will be implemented with a MAC level switch hardware equipment. This switch establishes two different VLANs (Virtual Local Area Network) according to each description and it distributes the content to the different available platforms (DVB-T, WIMAX).

  Usually, D1 will be sent to DVB-T and D2 to WiMAX according to the proposed scenario.

- **TS encapsulator + multiplexer + FECs + DVB-T/RCT modulator**

  This module is implemented internally in the RUNCOM DVB-T/RCT equipment. It encapsulates each stream in transport stream (TS) packets using multi-protocol encapsulation (MPE). After this, it multiplexes all TS packets into a final TS which it is sent to the DVB-T modulator. The system is all-IP based; therefore it works in a transparent way at IP level.

- **System manager**

  This module will manage the entire system. It allows the configuration of the playout services, signaling and the two platform managing.

  The system manager will be connected to a general database, which should contain information about the system configuration, user accounting, available bandwidth, etc...

  The system manager acts over all the other system components over SOAP messages through a web service called Intelligent Unit. This web service also contains the optimization algorithms that maximizes the service bit rate in the system.

- **Return channel server (RCIC)**

  This module will deal with all information of the return channel. It will be connected to a general database with information on network conditions, terminals and available services.

  It is connected to the system manager and the DVB-T, RCT and WiMAX modules through SOAP and UDP messages.

- **FEC + WiMAX encapsulator + Wimax modulator**

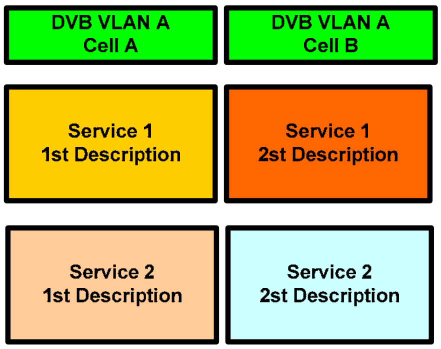
  This module is implemented internally in the RUNCOM WiMAX equipment. The system is all-IP based and therefore works in a transparent way at IP level.

5 Handover in DVB-T/H system

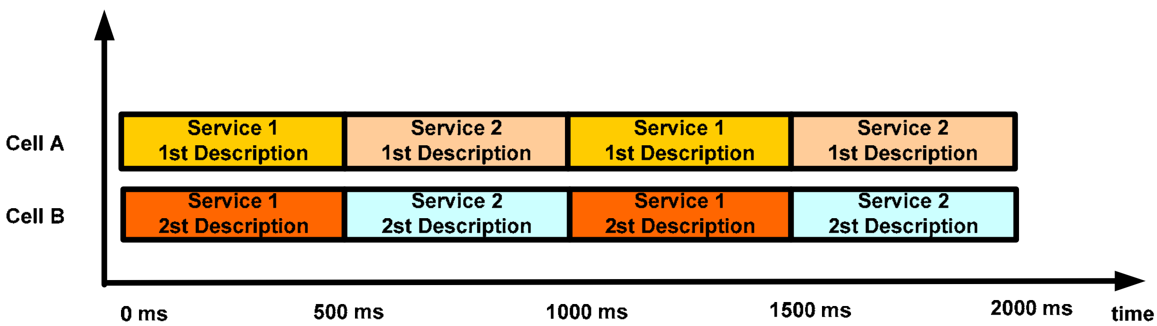
SUIT main objective is to evaluate MDC techniques for broadcasting services. Redundant technique like MDC may provide a more robust communication system and lower delay than the conventional stack of channel encoders. MDC will also provide a convergent solution for broadcasting over IEEE 16e and DVB-T/H as presented in previous sections. Besides, from several previous meetings discussions and as SUIT platforms should support vehicles moving at high speed crossing different cells, we would like to propose some solutions for horizontal and vertical handover (HO). Some of proposals in the following sections are just ideas which need to be validated. More confident solutions will be discussed in the next related deliverable, D1.5. Thus, Section 5 presents an HO solution for DVB in the down-link (DL). Then, Section 6 deals with the same subject but in the DVB up-link (UL). HO in 16e (WiMAX) is discussed in Section 7. Finally, Section 8 discusses vertical HO.

5.1 Handover considerations in DVB-T/H system

In this section, let us firstly focus on horizontal handover between two DVB-T/H cells in the DL (see Section 2.5). In DVB-H, or even in DVB-T or WiMAX (a solution proposed in Section 7), we can use time-slicing and therefore the terminal may not loose any data. For instance, let us assume a 20 Mbps multiplexer (similar to SUIT multiplexers) carrying two broadcasted services. Each service is transported in a 10000 bits packet and thus a packet has duration of 500 ms. **Figure 6** depicts this concept.



(a)



(b)

Figure 6: Two broadcasting services, each with two descriptions. (a) Services per cell. (b) Representation in time.

As two descriptions (between cells) representing the same service could be synchronized, 500 ms is the required time to Switch-over horizontally, the mobile terminal will be able to switch to the next cell by using a similar principle standardized in DVB-H as a mechanism called time-slicing. The HO process in the same network DVB-T to DVB-T is below 100 ms (the target is 50 ms). The problem remains on how to synchronize the both descriptions in different base stations.

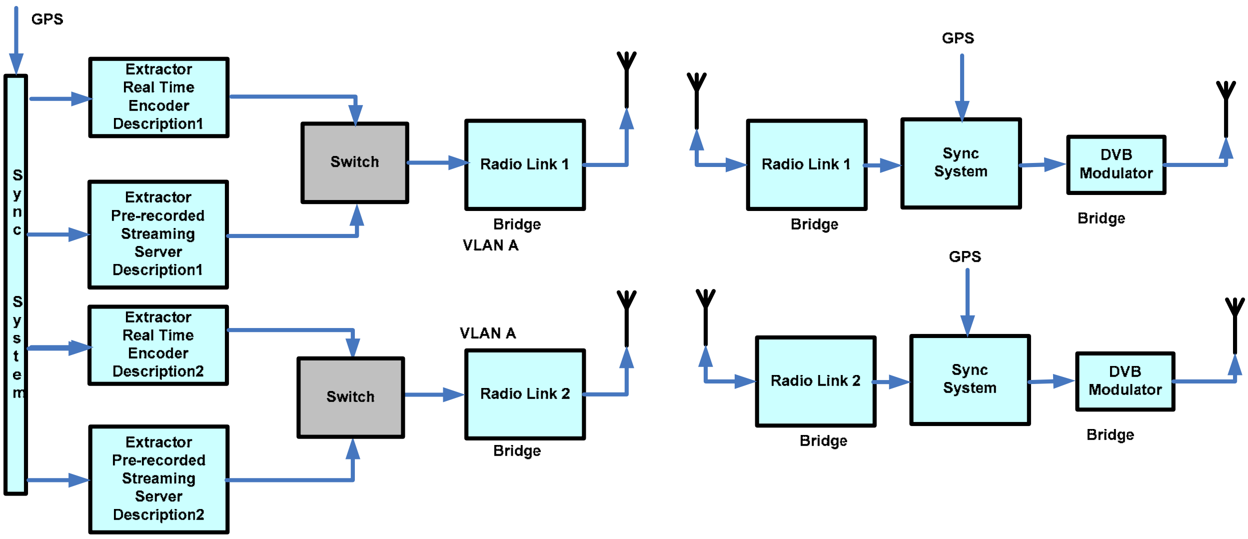


Figure 7: Distribution network synchronized with GPS

Despite SUIT is proposing an MFN solution, it is possible to adopt a similar solution as proposed by DVB for SFN which is described in ETSI TR 101 190. Assuming that the DVB modulators in the BSTs provide a fixed delay from the input to the air interface and the switch/multiplexer (see Figure 3) in the playout is split into two in order to ensure two parallel streams to both cells (A and B), the desynchronization is mainly caused by the distribution network. Obviously, despite in SUIT demonstrator, the distribution network is composed of radio links (assuming equal radio equipments) with a fixed delay, both streams carrying the same information are synchronized but this is not the general case. Therefore, SUIT proposes to synchronize both streams descriptions by using GPS as shown in **Figure 7**.

SUIT has to define a structure like MIP (Mega-frame Initialization Packet) which points to the start of next IP packet. So, MIP_M points to the starting point of packet M+1. Besides, MIP_M carries the Synchronization Time Stamp (STS) which is the information related to the difference between the last GPS pulse (1 pps) and the start of M+1 packet. The maximum delay time can also be inserted in MIP. In this way, all transmitters buffer the data in order to synchronize them with the maximum delay path. However, it is expected that the maximum delay path is less than 1 second. Otherwise a GPS clock divider should be designed. There is no need to synchronize with the DVB scrambler as required in SFN. Therefore, SUIT neither need to define a structure like mega-frame, nor use a fixed packet size like TS.

We have presented an HO solution for broadcasting services including the hyperlinked video since in the later case, the playout will deliver two descriptions. The horizontal HO for unicasting services, like Internet, over DVB-T/H, works in a similar way by making use time slicing. However, the traffic must be redirected to the new cell. As the internet streams are using TCP, the terminal requests a new connection to the playout.

6 Handover in DVB-RCT system

6.1 Handover considerations in DVB-RCT system

First, it is important to understand that the DVB-RCT system standard has been designed for fixed environments. For mobile environments, special considerations have to be taken into account in order to become able to perform handover (HO) within the SUIT project mobile demonstration. As a result, Runcom has implemented HO capabilities based on hardware and software infrastructures that already exist in the system.

The HO design had to take into account that there is no simple way to synchronize between two DVB-RCT BSTs due to the fact that there is no way to synchronize IP sources such that the same IP packets will run in two different BSTs at the same time. If these two BSTs will operate in the same frequencies there will be interference between them and no good communication will be available. Hence, two sets of frequencies are required to prevent any interference between the BSTs. Since the RCT system operates is FDD mode, four (or three) frequencies of 8 MHz bandwidth are needed in the UHF spectrum.

In Sections 6.2-6.4, a detailed description is given of Runcom's approach to design the HO in the simplest way without any need to change the basic CPE and BST concept or structure.

In addition, this contribution describes in detail the differences between *hard HO* and *soft HO* and gives the explanation on how Runcom has implemented the soft HO based on two transceivers.

6.2 Hard handover – L1/L2 considerations

With hard handover, the link to a prior (serving) base station is terminated just when the user is transferred to the new cell's (target) base station (**see Figure 8**). The mobile user is linked to no more than one base station at any given time. Initiation of the handover may begin when the signal strength at the mobile received from base station 2 is greater than that of base station 1. The signal strength is measured as real signal levels averaged over a chosen period of time. This averaging is necessary because of the Rayleigh fading nature of the environment in which the cellular network resides.

A major problem for hard HO is that the received signals of both base stations often fluctuate. When the mobile user is between the base stations, this effect is to cause the mobile to wildly switch links between either station. The base stations bounce the link with the mobile back and forth. Hence, the phenomenon is called *ping-ponging.* Besides ping-ponging, this simple approach generally allows for too many handovers. It has been shown in previous studies that for much of the time a selected link was well adequate and that handover occurred unnecessarily. A better method is to use the averaged signal levels relative to a threshold and to a hysteresis margin for handover decision. Furthermore, the condition should be imposed that the target base station's signal level should be significantly greater (over a specified period of time) than that of the current base station.

Figure 8 depicts the situation for hard handover. The connection is interrupted before completing the transition: F1/F2 BST is disconnected and only after a certain period F3/F4 is connected (*break-before-make mechanism*).

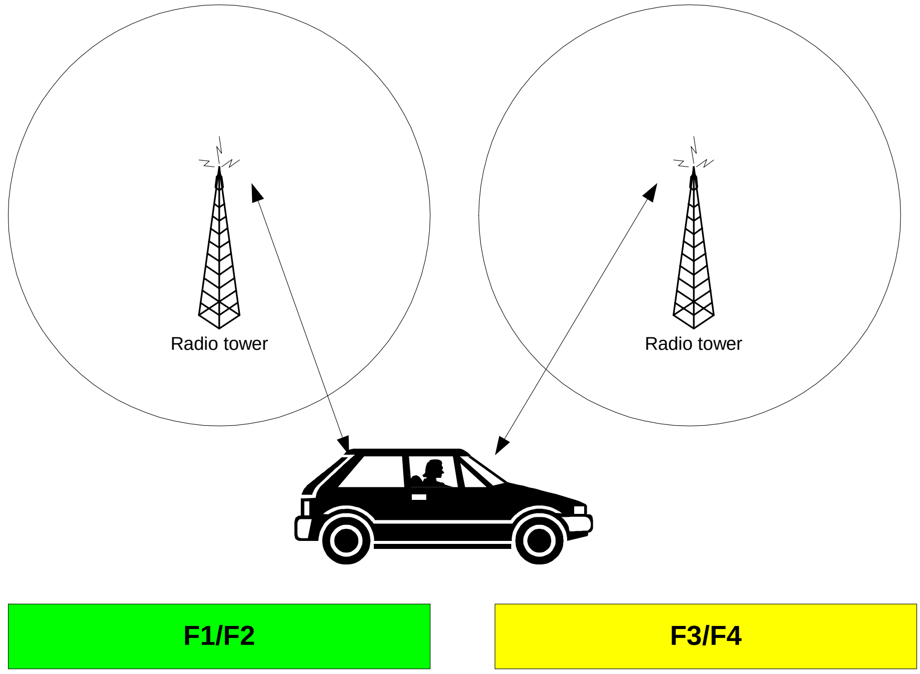


Figure 8: Hard handover

6.3 Soft handover – L1/L2 considerations

In soft handover the CPE can be connected to several BSTs or sectors simultaneously.

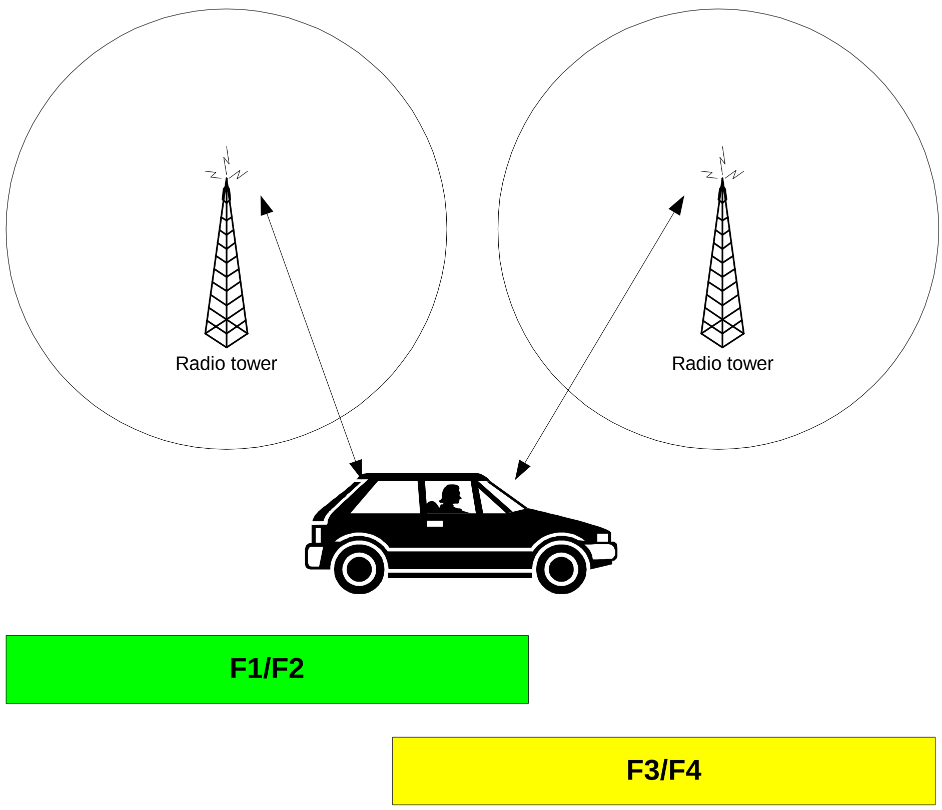


Figure 9: Soft handover

In **Figure 9**, the concept of soft handover is illustrated. Here, the CPE is connected to F3/F4 BST before it is disconnected from F1/F2 BST (*make-before-break mechanism*).

The DVB-RCT system Runcom has implemented soft handover by using two transceivers in one CPE box. Transceiver 1 can be synchronized only to F1/F2 BST and the transceiver 2 can be synchronized only to F3/F4. The coverage design should take into account that there will be an overlapping area in which both transceivers are synchronized to their corresponding sector (BST).

Figure 10 illustrates the functional phases of the soft handover process:

Phase 1: Received signal strength measurements (RSSI), (or C/N or PER measurements)

Phase 2: Decision, bases upon the established threshold

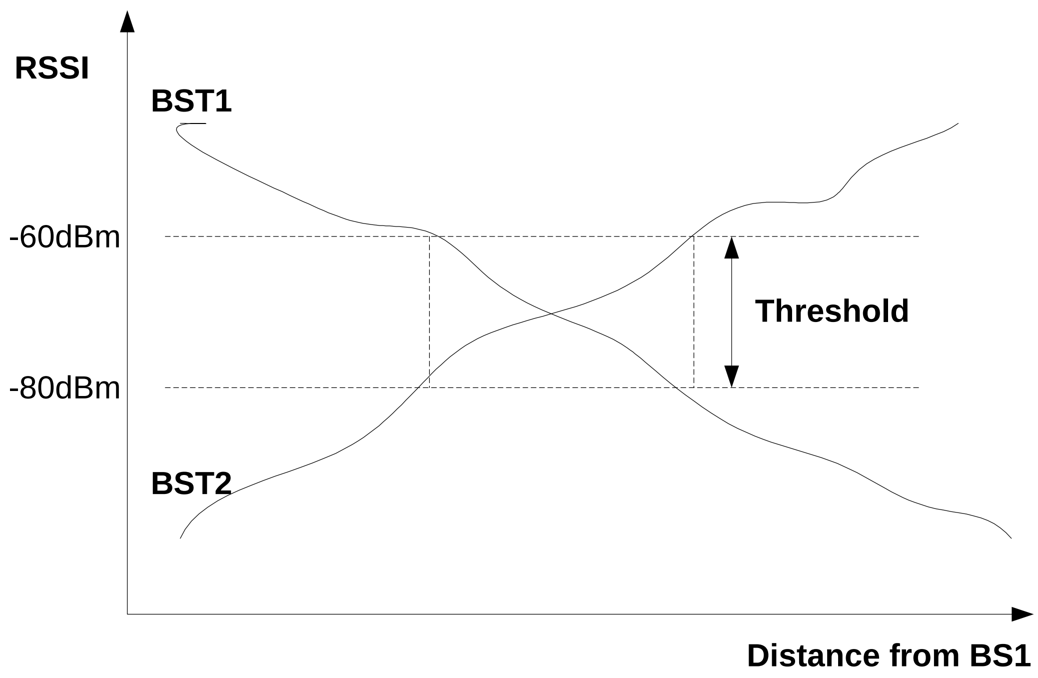


Figure 10: Phases in soft handover

6.4 Runcom's HO implementation – General view

Figure 11 shows the CPE that encompasses the two transceivers and the hub connected through the Ethernet interface. As an application, an encoder for "MPEG-4 over IP" (MP4oIP) is connected to the hub, transferring the MP4oIP video to the BSTs via the CPE in the uplink. Despite the MP4oIP application has been developed for the up-link, it can also be connected to the BST which may transfer the MP4oIP video to the CPE via the BST in the downlink.

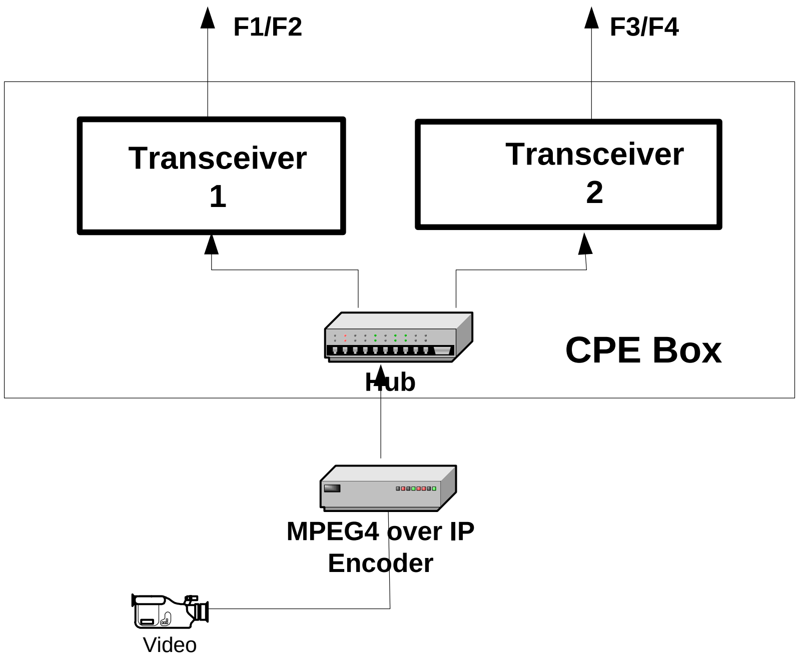


Figure 11: Connection for soft handover implementation in the end-user terminal

In the soft handover implementation, there are master/slave transceivers activated according to the master transceiver decision. The master transceiver can be the one that was initiated with the first synchronization to the BST. Only one transceiver will transmit at a time making the video distribution in the BST side easier.

Figure 12 presents the CPE box block diagram in more detail. It includes two transceivers, two power amplifiers (for each transceiver a separate amplifier is recommended to achieve good reception) and duplexers in order to use the same antenna for the received and transmitted signals. (Note: For single-antenna operations a power splitter/combiner is needed in addition.) One Ethernet port is used for both, the communication between the transceivers and the provision of the MP4oIP application.

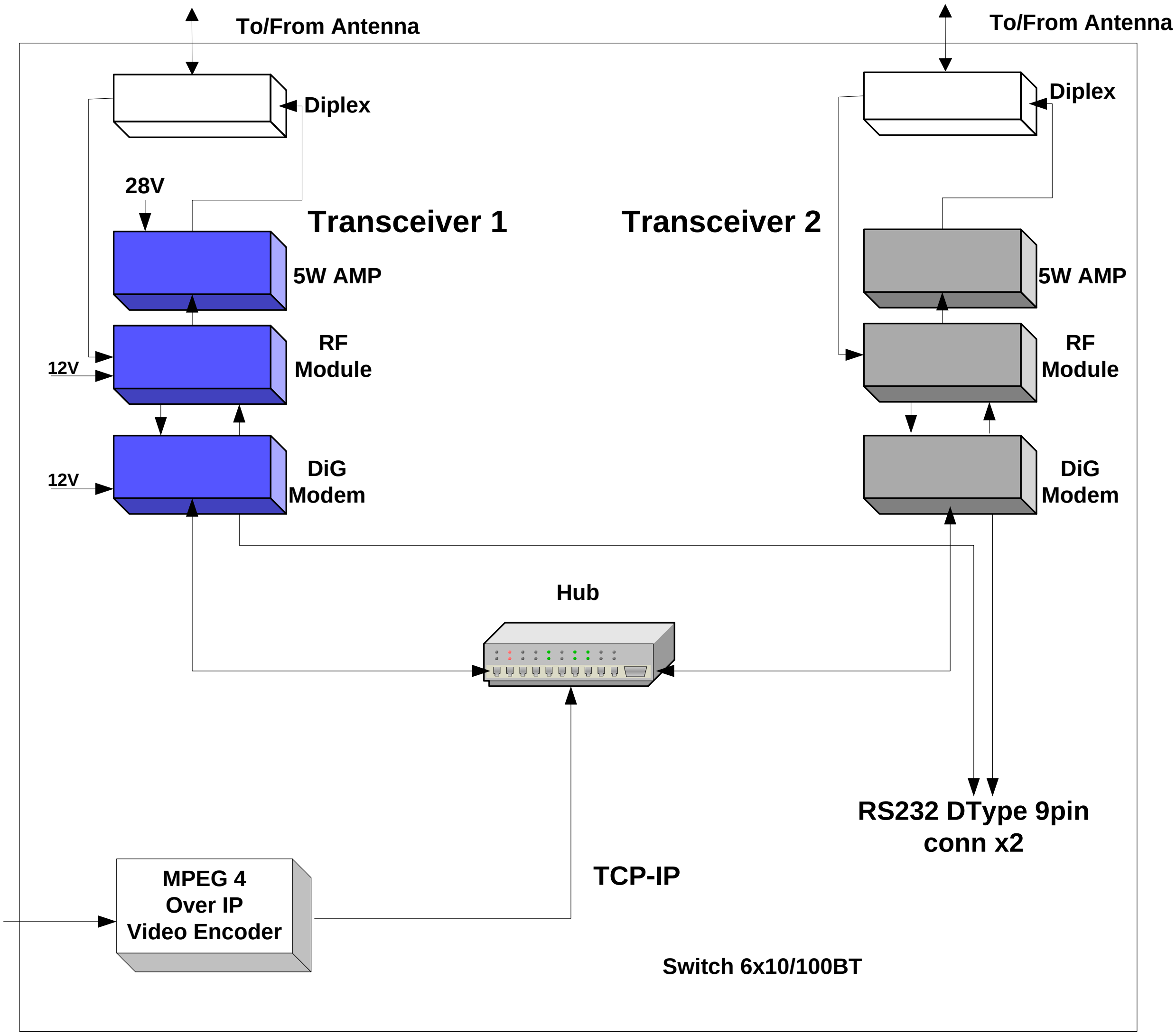


Figure 12: Block diagram of CPE box containing a solution for soft handover

7 Handover in mobile WiMAX 802.16e

This chapter explains the handover process for a mobile system based in a mobile WiMAX 802.16e environment, The approach concentrates on MAC messages in order to describe the HO process demonstration which will be presented by SUIT. It starts with the design and architecture implementation of the HO layer 2 for the communication between the serving BST and the user terminal (layer 2), and continues with the description of the layer 3 communication between the serving BST and the target BST (using ACR - Access Control Router).

7.1 System

The purpose of this chapter is to define the system components from the single-cell level up to the multiple cell level, and to define their operation scenario. In a single cell HO is possible between sectors. In the case of multiple cells HO can occur between BSTs.

7.1.1 Single-cell block diagram

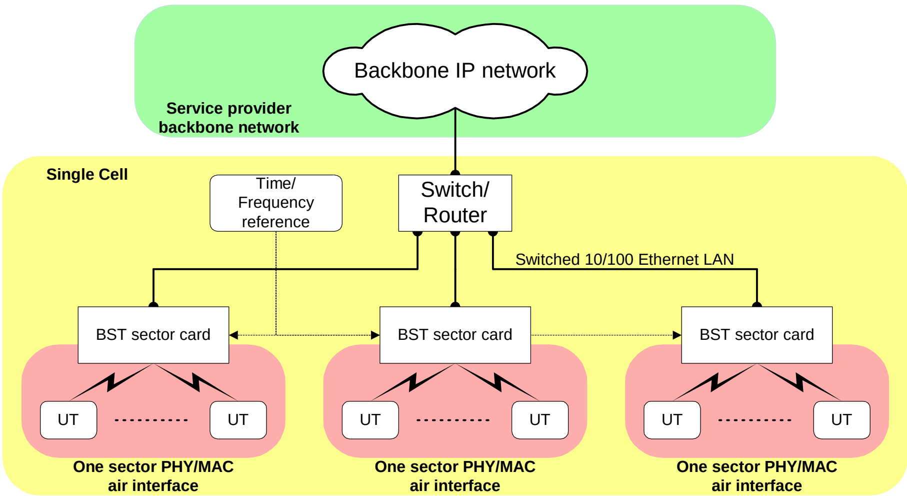


Figure 13: Single-sector block diagram

7.1.2 Multiple-cell block diagram

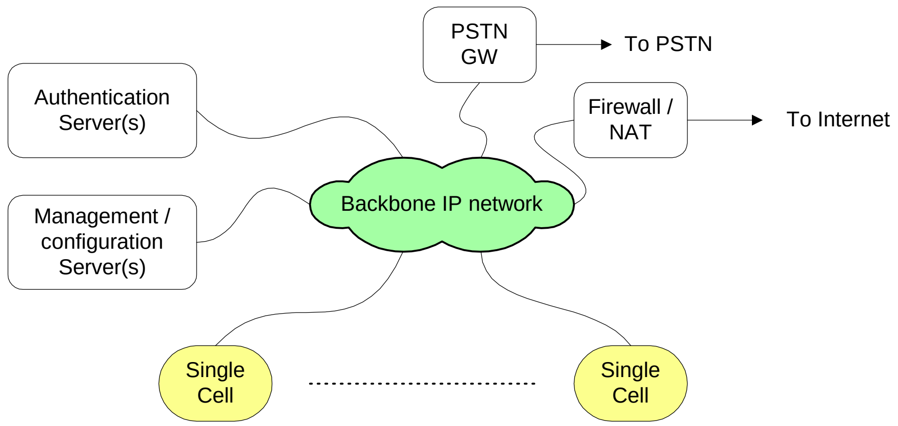


Figure 14: Multiple-sector block diagram

7.1.3 Description of operation state (before HO)

7.1.3.1 <u>BST (base station)</u>

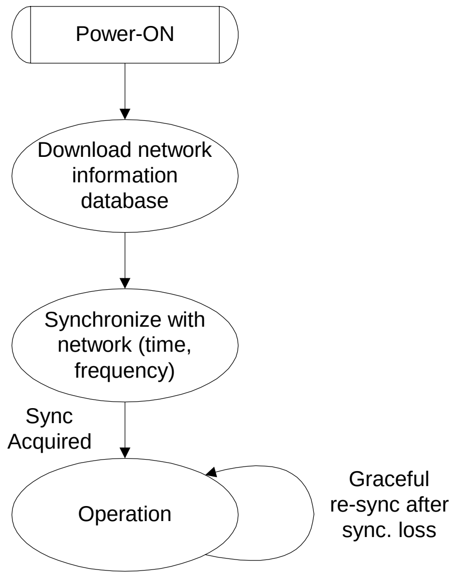


Figure 15: BST sector operation states

Upon connection of a serving BST to the network, the BST becomes manageable through the backbone IP network. It can access the management and configuration servers in order to obtain data relating to the network topology (i.e. frequency and RF parameters, neighbors, network-related parameters, server addresses, etc.). At the next stage, the BST has to synchronize its timing and frequency to the timing and frequency settings used in the system. The time/frequency source would normally be a GPS receiver located at the BST site, but other methods are also possible to provide a time/frequency reference. After the BST is synchronized to the network, it starts its air interface and starts supporting the user terminal (UT). The BST uses the backbone (or

ACR) to access to the provisioning and authentication servers whenever information about a UT is required.

7.1.3.2 User terminal (before HO)

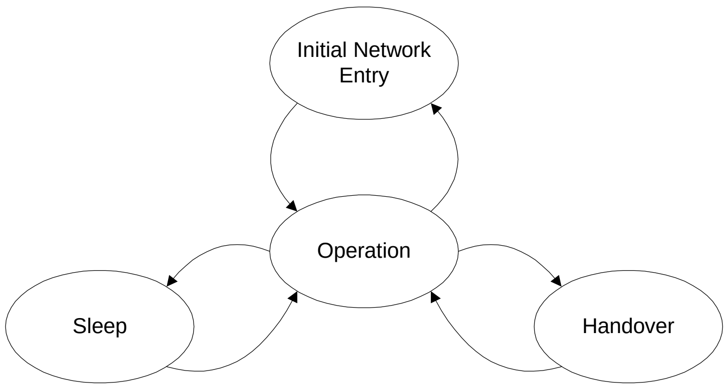


Figure 16: UT operation states

The user terminal (UT) features three major operational states. Upon power-on, the UT attempts *initial network entry*. During this process the UT scans for a suitable downlink channel and synchronizes with a specific BST sector. After having achieved physical level synchronization, the UT registers with the network through the BST. The network-entry process terminates once the UT is assigned an IP address and is capable of opening transport connections for data. It should be emphasized that the network entry process does not include any setup operations required in order to transmit data in layers above layer 2.

After initial network entry, the UT is in *operation mode*, where it is capable of full functionality and bi-directional transfer of data in compliance with the PHY/MAC specifications of the air-interface.

The UT may leave the operational mode to go into *sleep mode*. During sleep the UT wakes up occasionally in a manner synchronized with the BS, and goes back to sleep if no data is available for it. In case that there is data for a sleeping UT, the UT returns to the operation mode.

The last major of the UT is handover. Handover is the process where a UT registered with a given BST sector seeks transition to another BST sector which may be in the same cell or in a different cell (see Section 7.1). The process involves tearing down the existing data connection at the “old” BST and re-establishing it with the “new” BST after synchronization and registration with this “new” BST.

7.1.3.3 Synchronization with a GPS system

The BST platform has two inputs to a GPS reference. One input is a 10 MHz frequency reference and the other input is a 1 pps timing reference. The BST can query the GPS at the time of each timing pulse through a RS-232 or RS-422 interface.

The BST system clock is synthesized directly from the 10 MHz reference by a PLL as shown in the drawing of **Figure 17** below.

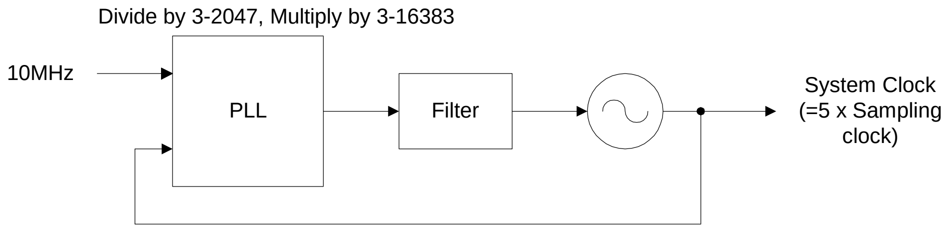


Figure 17: BST system clock generation at BST

The TDD frame is synthesized such that we define some date in the past (e.g. 1/1/1990 00:00 UTC) as the starting point. Given this starting point and the known length of the TDD frame as well as the 1 pps pulse location and the knowledge to what second in what year/day/hour this pulse refers to, we know where the TDD frames should be located relative to the 1 pps pulse.

The scheme to lock the TDD frame to the GPS is to measure the actual TDD frame phase at each 1 pps pulse, and compare it to the calculated phase. The difference is used for an according shift of the phase of the circuitry that generates the TDD frame clock from the system clock (that is locked to the 10 MHz frequency reference).

In order to perform HO between BSTs these BSTs need to be synchronized. GPS is the prefered way to achieve synchronization between BSTs. A similar GPS solution has been proposed for DVB-T in Section 5.1.

7.1.4 HO architecture

Figure 18 reflects the overall system.

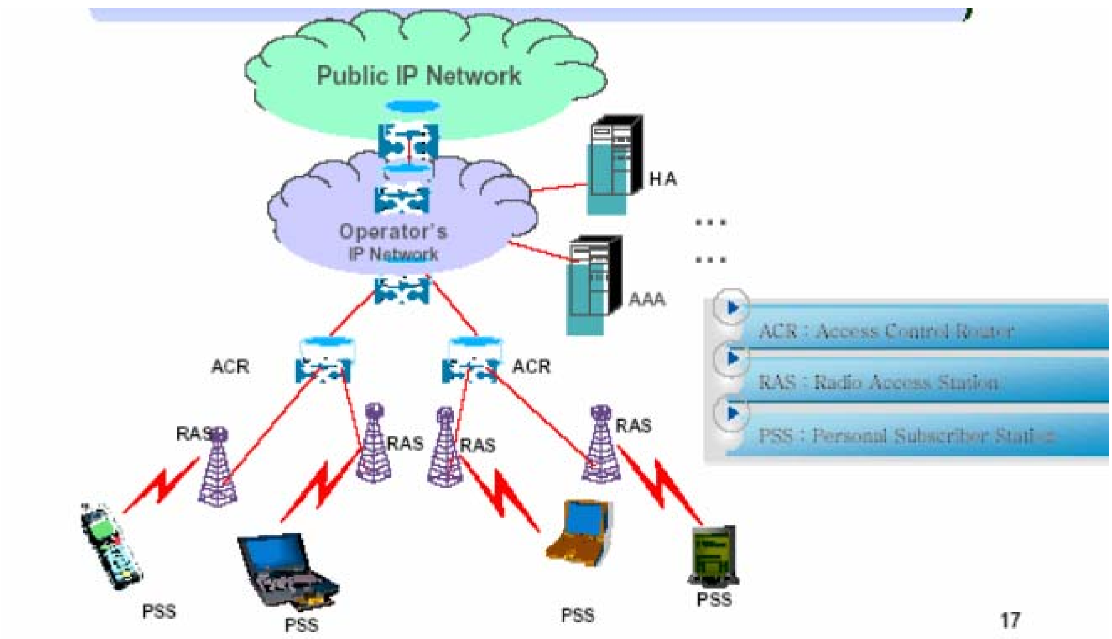


Figure 18: Overall system concept for HO operation

The ACR (Access Control Router) is the interface between the BSTs, and it is the main player in the HO process. In **Figure 18**, the BSTs are represented by the Radio Access Stations (RASs).

The WiMAX standard knows two kinds of handover processes concerning the Frequency Assignment (FA):

- Inter-FA HO – Handover between BSTs with different frequencies
- Intra-FA HO – Handover between BSTs of the same frequency

Note that Intra-FA does not require the interleaved scan.

7.1.4.1 Layer-2 HO messages

Below, the HO process in layer 2 is presented step by step :

- After sync, the user terminal is constantly measuring the CINR of the serving BST
- Periodically, the user terminal is receiving MOB_NBR-ADV (all neighborhood BSTs list – serving BST “advertises” its neighboring BSTs to the UT)
- When the CINR measurement of the serving BST is below a certain value (x dB) the UT requests the serving BSTs to scan for adjacent (neighboring) BSTs (when serving BST’s CINR < Scn_threshold: send MOB_SCN-REQ).
- After MOB_SCN-RSP: Interleaved scan (scan/data/scan/data…)
- Serving BST responses to scan request
- UT reports its scan results

- For neighbor BSs with CINR > Req_threshold (X-YdB): send MOB_MSHO-REQ (The Mobile Station (MS) is sending the threshold CINR received to the serving BST and ask a permission to make HO to target BST.
- Receive MOB_BSHO-RSP
- Serving BST response (to handover request)
- When serving BST’s CINR < Ind_threshold (Cross+ZdB): send MOB_HO-IND
- UT notifies HO (confirmation/cancellation/rejection)
- When serving BST’s CINR < Tran_threshold (Cross): Handover (transition)
- Serving BST reports (aggregate) neighbour BST’s RNG-RSP
- Serving BST requests UT to Handover to a target BST

X, Y and Z are parameters and could be changed according to the link requirements. As an example X=8dB Y=5dB and Z=2.5dB. For the illustration of the HO refer to **Figure 10**.

7.1.4.2 Layer-3 HO

The protocol stack between the BST and the UT (including the ACR) is described in **Figure 19** shown below:

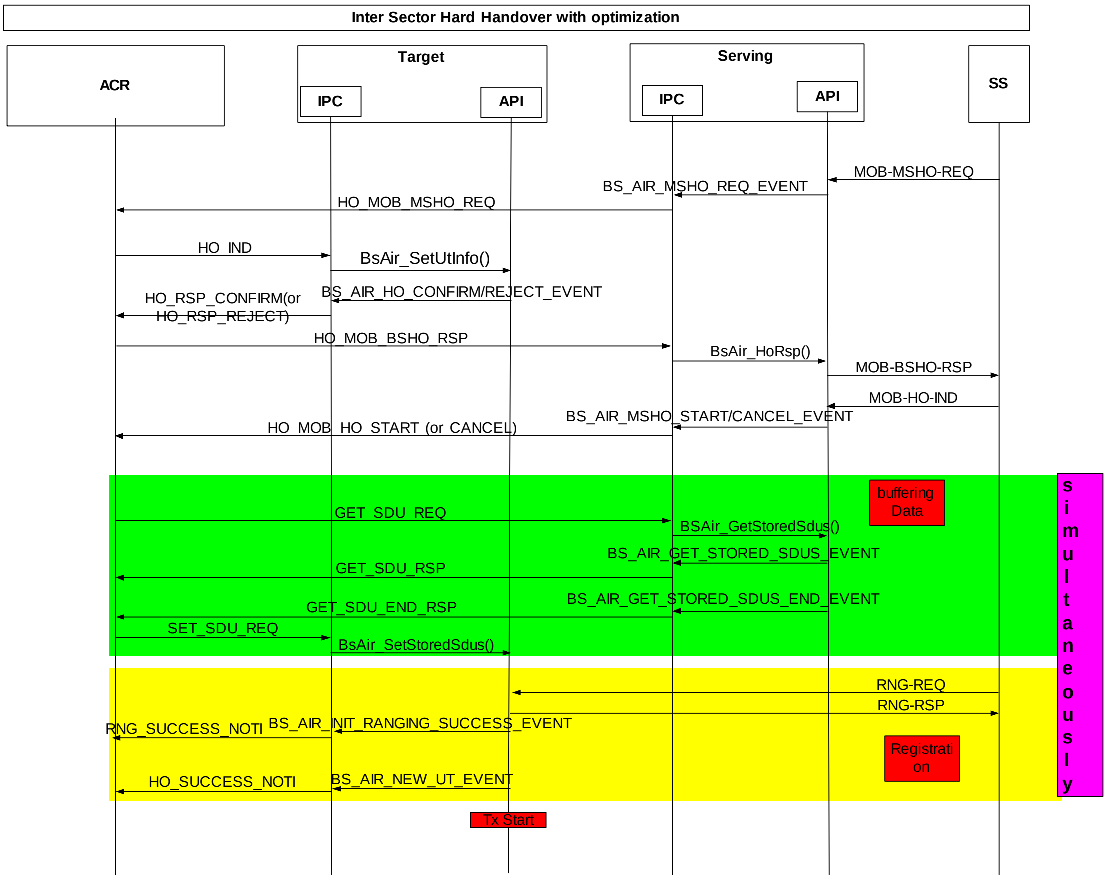
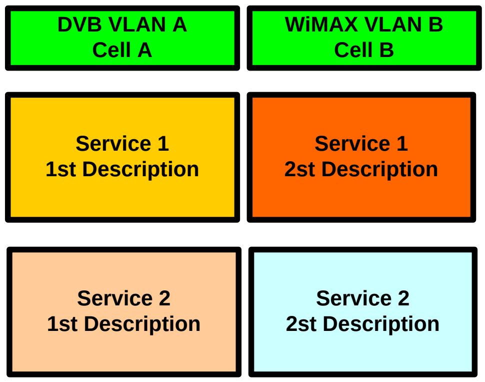


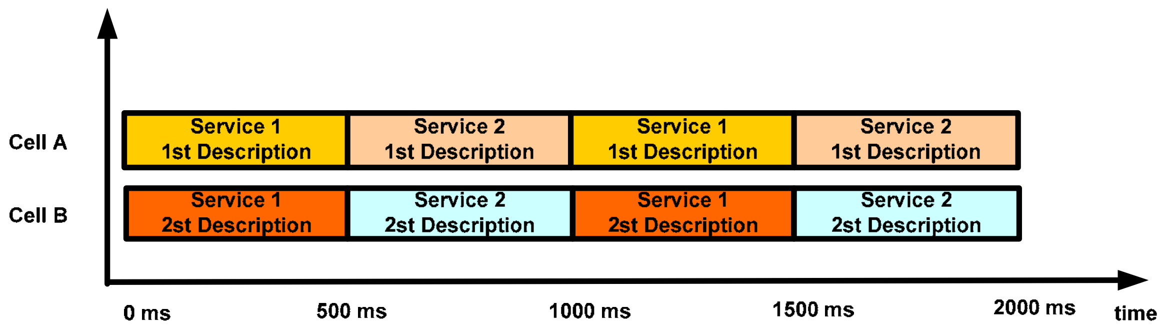
Figure 19: Protocol stack for layer-3 handover (example of inter-sector hard handover)

8 Vertical handover between DVB-T/H and WiMAX systems

In previous sections, we focused on horizontal handover between two DVB-T/H cells in the DL, two WiMAX cells in the DL and two WiMAX cells in the up-link. In this section, we discuss vertical HO. As proposed in Section 5 and because SUIT proposed MDC, we can also use time-slicing and therefore the terminal may not loose any data during the vertical HO. Each cell will broadcast the same related information (two descriptions, 1D and 2D) synchronized in time as shown in **Figure 20**.



(a)



(b)

Figure 20: Two broadcasting services, each with two descriptions, each one delivered over different network . (a) Services per cell. (b) Representation in time.

The HO process is quite similar to the one propose for horizontal HO in DVB-T/H cells. **Figure 21** shows SUIT architecture to support vertical HO. Again seamless HO will not be achieved for non-real time services like Internet.

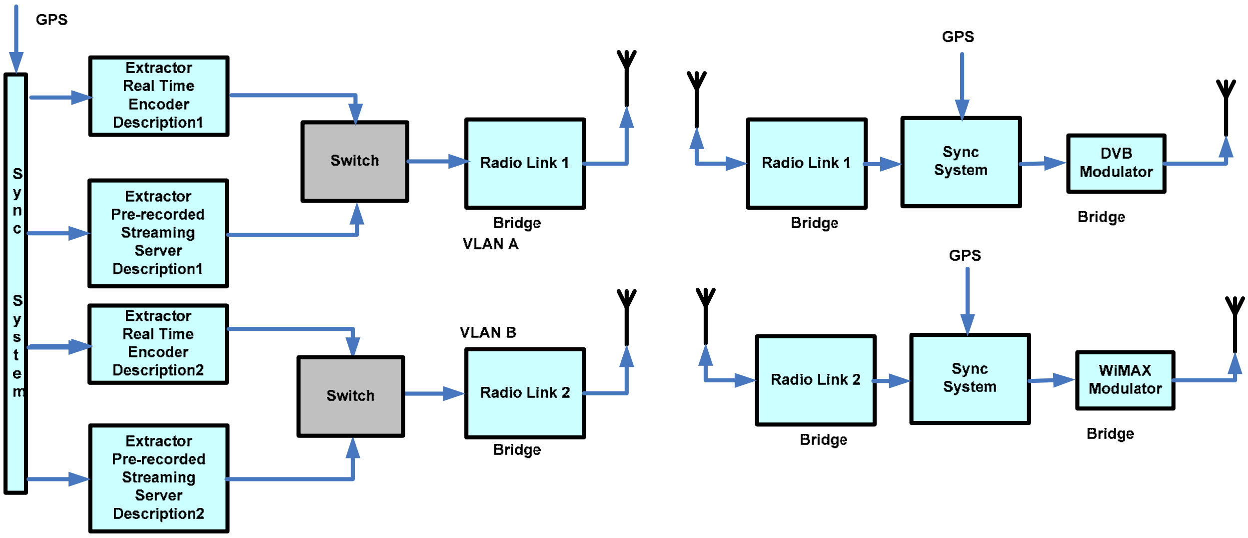


Figure 21: Distribution network to DVB and WiMAX synchronized with GPS

It is expected that the modulators in the BSTs, DVB and WiMAX will not provide the same delay from the input to the air interface despite they are fixed. Again, the switch/multiplexer (see Figure 3) in the playout is split into two in order to ensure two parallel streams to both VLANs (A and B), the desynchronization is mainly caused by the distribution network. Therefore, SUIT proposes to synchronize both descriptions by using GPS as shown in **Figure 21**.

SUIT also proposes to use a structure like MIP (Mega-frame Initialization Packet) in order to support vertical HO which points to the start of next IP packet. For further information, the reader is invited to read Section 5.1.

Balanced MDC will facilitate HO because both streams will be composed by the same packet duration.

9 Conclusions

This document gives an overview of different architecture and reference scenarios within the SUIT project. It is the updated version of the initial document D1.4 that has been previously published in the time line of Activity A1.4. The main improvements are standing in the reference scenarios that will apply in WP6.

10 Acronyms

ACR	Access Control Router
BW	Bandwidth
BST	Base station
CIF	Common Intermediate Format
C/N	Carrier-to-Noise Ratio
CINR	Carrier-to-Interference-and-Noise Ratio
CPE	Consumer Products Equipment
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
DVBSTP	Digital Video Broadcasting SD&S Transport Protocol
DVB-T	Digital Video Broadcasting –Terrestrial
ENG	Electronic News Gathering
HDTV	High Definition Television
HO	Hand Off / Hand Over
FDD	Frequency Division Duplex
FGS	Fine Grain Scalability
GPS	Global Positioning System
IETF	Internet Engineering Task Force
IP	Internet Protocol
IRT	Institut für Rundfunktechnik
IT	Instituto de Telecomunicações
JMF	Java Media Framework
JSCC	Joint Source Channel Coding
UPA	Unequal Power Allocation
MAC	Media Access Control
MDC	Multiple Description Coding
MFN	Multi-Frequency Network
MHP	Multimedia Home Platform
NAL	Network Abstraction Layer
PDA	Personal Digital Assistant
PER	Packet Error Rate
QoS	Quality of Service

QCIF	Quarter Common Intermediate Format
RSSI	Received Signal Strength Measurements
RTCP	Realtime Transport Control Protocol
SAP	Session Announcement Protocol
SDC	Single Description Coding
SDP	Session Description Protocol
SDTV	Standard Definition Television
SD&S	Service Detection & Selection
SFN	Single Frequency Network
S/N	Signal-to-Noise Ratio
STS	Synchronization Time Stamp
SVC	Scalable Video Coding (in accordance with MPEG-4 Part 10)
TDD	Time Division Duplex
UDP	User Datagram Protocol
UT	User terminal
UTC	Universal Time Clock
UHF	Ultra High Frequency
VLAN	Virtual Local Area Network
WIFI	Wireless Fidelity
WIMAX	Wireless Local Area Network (in accordance with IEEE 802.16e)
WLAN	Wireless Local Area Network (in accordance with IEEE 802.11g)