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Abstract

This document specifies a set of QoS requirements for the SUIT system. On the one hand it addresses the air interface of the different networks, defining mainly low layer QoS parameters that can be monitored and measured with existing and/ or modified equipment and tools. On the other hand it also specifies IP related quality parameters, and provides background information on QoS issues in the different systems.

Keyword list: Quality-of-Service, DVB, WiMAX, WiFi

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QoS requirements

SUIT_164

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1 Introduction

The QoS requirements given in this Deliverable complement the requirements for the user terminal and for all-IP support. Together they are part of the system requirements which are compared with the system features as evaluated during the implementation and demonstration phase.

They are also the basis for the definition of the system architecture (*Figure 1-1*) and the reference scenarios, in other words, they describe the target to be reached by the Suit project.

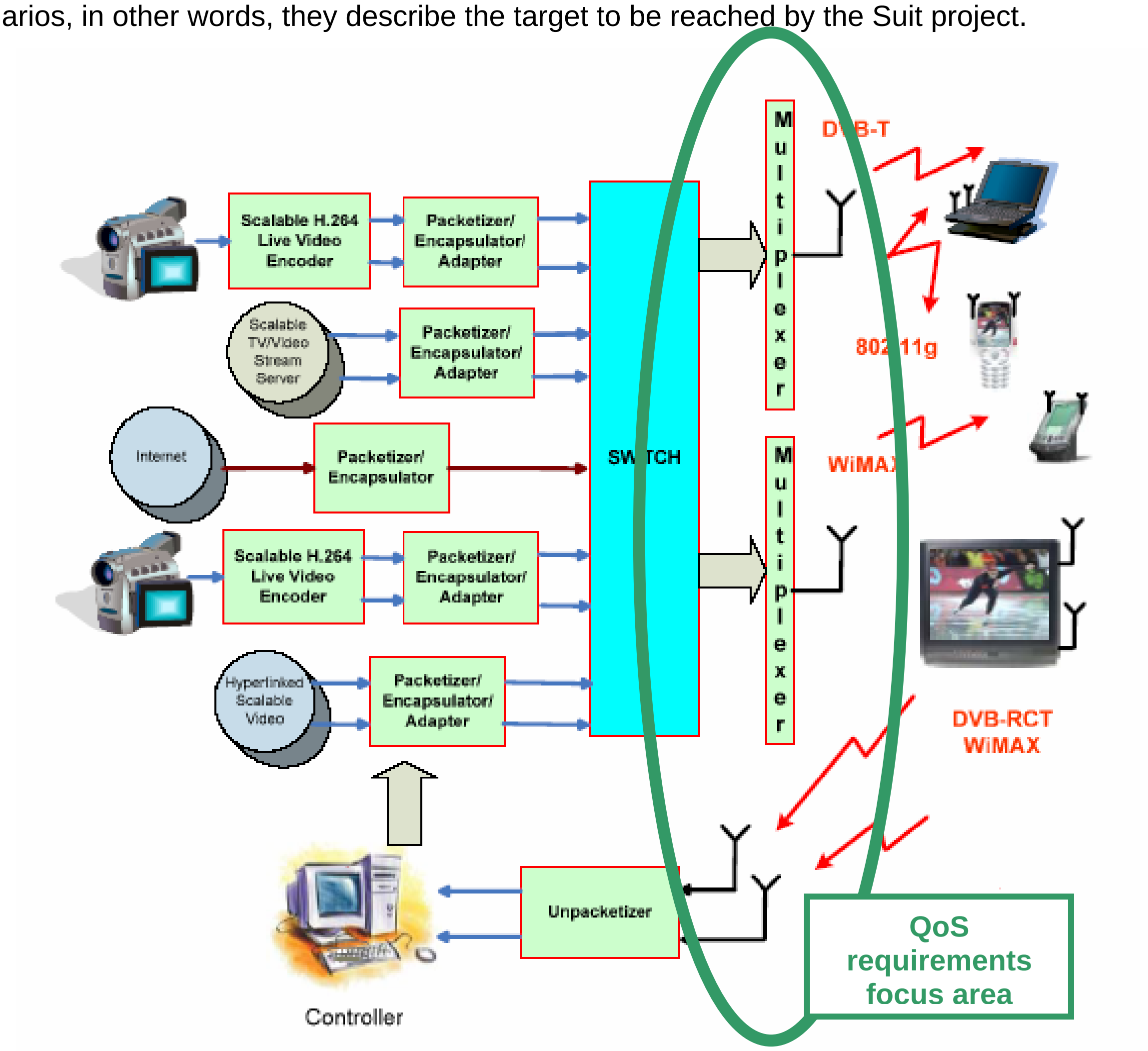

Figure 1-1: Overall Suit system architecture

For a complex system as the one being set up in the Suit project, it is advisable to differentiate between mandatory features and desirable features which are reflected in mandatory and desirable requirements. To transmit services with sufficient quality so that they can be received by the end-user, only a limited set of QoS requirements seems to be necessary. The fulfilling of these mandatory requirements facilitates the SUIT system to work properly in a basic configuration.

The desirable features are the ones that are useful or even highly recommended to make the handling of the system much easier and the performance of the system much better, but are not necessary for the basic functionalities.

Requirements should be defined as values of technical parameters, as far as this is possible. For example, the requirement that a certain broadcast signal can be received in a pre-defined area with a certain location probability and a certain time probability can be described as a minimum value of the electro-magnetic field strength in this area at a certain average height above ground.

The objective of this Deliverable is to define a minimal set of QoS parameters to be checked permanently or intermittently, and to define values for these parameters that have to be met to ensure a proper function of the system and a good quality of service for the end user, i.e. the values should always remain above the threshold of acceptability by a certain margin.

In the Suit project, various hybrid scenarios are considered which are combinations of DVB-T/H networks, WiMAX networks, DVB-RCT networks and WiFi networks. DVB-T/H and WiMAX networks can be utilised for broadcasting multicasting and unicasting services. Particularly, unicasted video could be transmitted over DVB-RCT (UP-Link) networks.

The following cases are considered:

- Broadcast over DVB-T/H
- Broadcast over WiMAX
- Multicast over WiMAX
- Multicast over DVB-T/H
- Multicast over WiFi
- Unicast over WiMAX
- Unicast over DVB-T/H
- Unicast over DVB-RCT
- Unicast over WiFi

The general block diagram of SUIT platform is shown in Figure 1-2. Terminals will be fed via two different wireless broadband last mile networks, WiMAX and DVB-T/H. The playout selects the right network to deliver any contents. The figure shows two homes and therefore two home gateways. In each home two terminals are connected to each gateway.

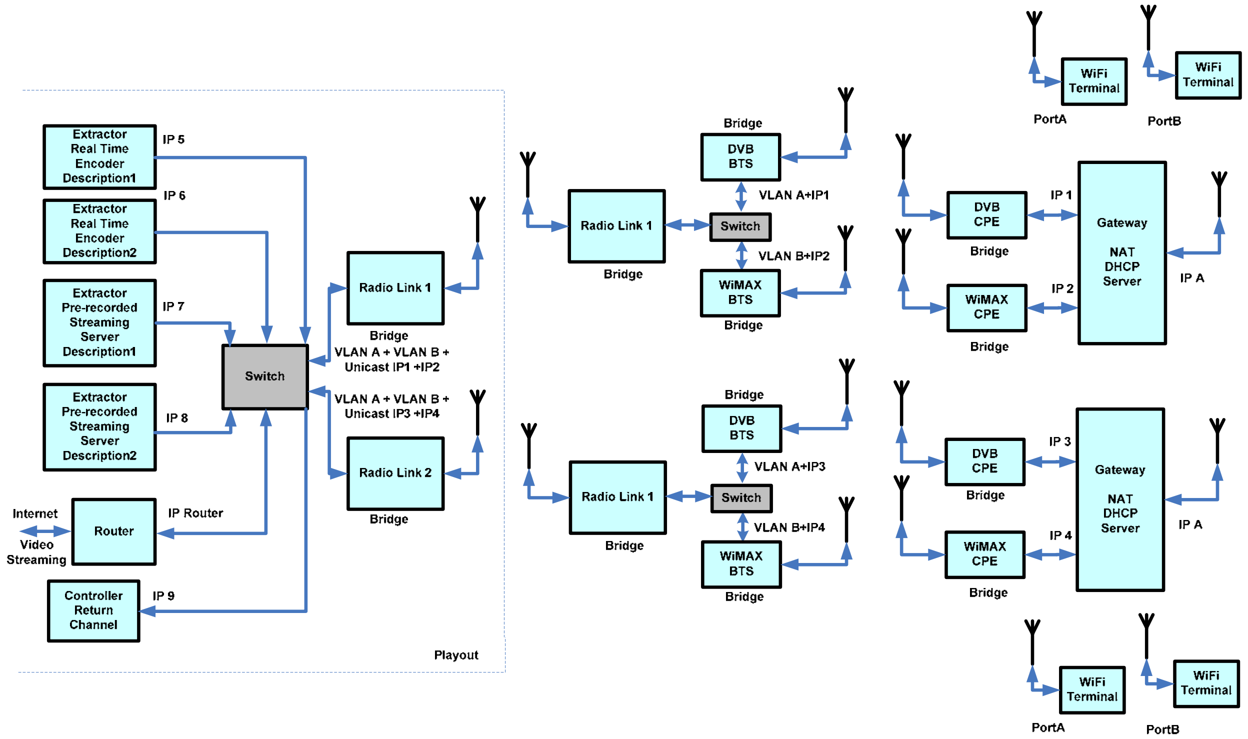

Figure 1-2: Suit platform

More details about service and network scenarios can be achieved from deliverables D1.1, D1.2, D1.4 and D3.1.

2 QoS requirements

2.1 DVB-T/H related QoS requirements

2.1.1 Reference architecture

The diagram in Figure 2-1 depicts a simplified DVB-T/H chain from the input of the IP encapsulator to the IP output of the receiver. The interfaces labelled as IP1, TS1, RF1 and RF2, TS2, IP2 are those normally accessible for test probes. The reason for this is that the functional blocks

- IP encapsulator and Mux,
- Modulator

are normally different entities interconnect by cables or similar means. Typically they are also equipped with a monitoring interface at input and/ or output.

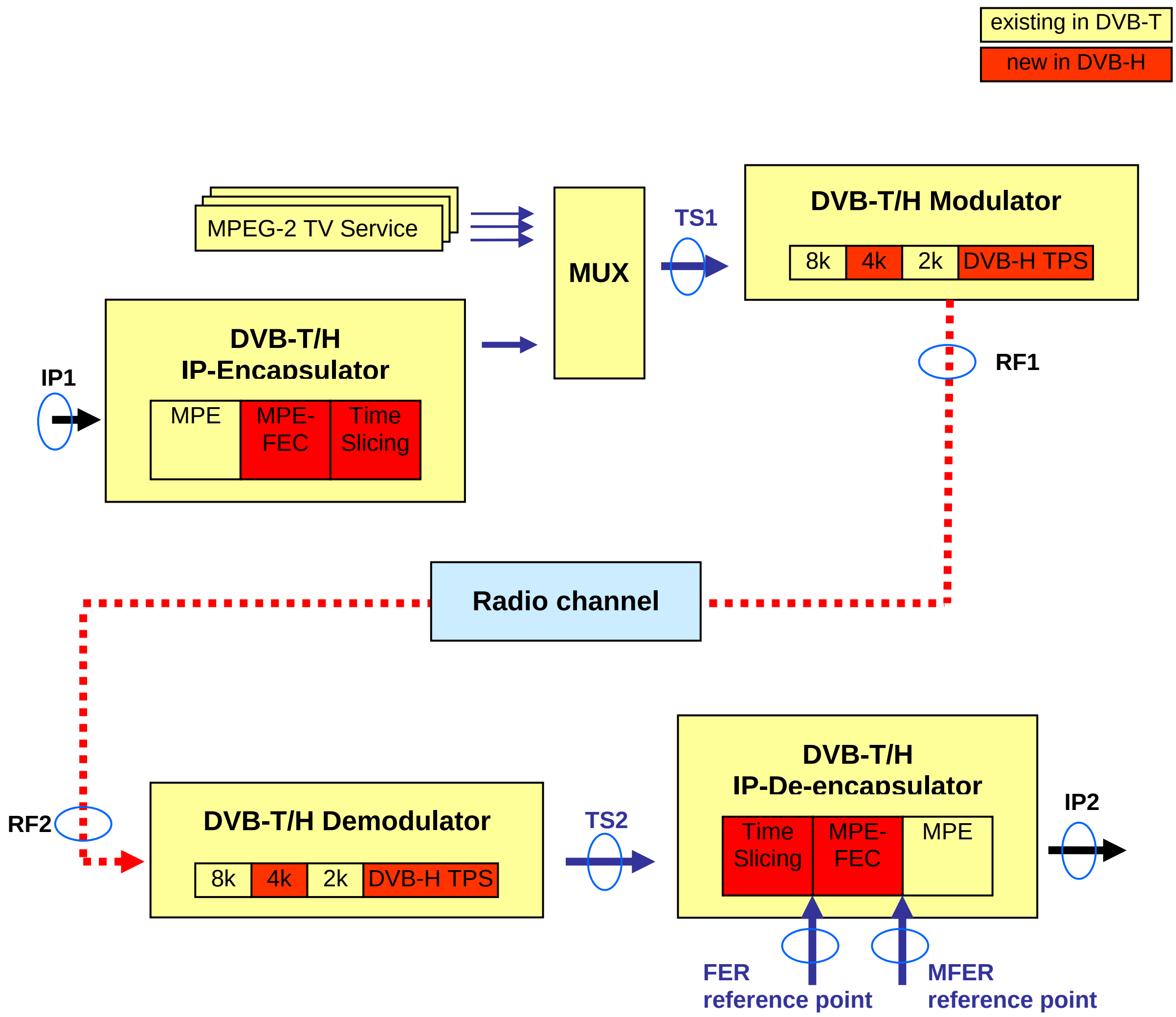

Figure 2-1: DVB-T/H chain

Therefore, the QoS requirements listed in the following sections refer to QoS parameters that can be measured at these interfaces [10] .

The three interfaces at the transmission and at the reception side of the chain correspond to a certain extent with the different layers of the completed signal:

- IP > network layer
- TS > data link layer
- RF > physical layer.

The additional reference points for FER (Frame Error Rate) and MFER (MPE Frame Error Rate) are internal measurement points which are normally only accessible in a test receiver.

2.1.2 QoS requirements at interfaces IP1 and IP2

The most important QoS parameters at IP level in a DVB-T/H system are supposed to be

- Service Bit Rate
- IP Packet Error Rate
- IP Packet Jitter.

2.1.2.1 Service bit rate

QoS Parameter	Service bit rate
Interface	IP1, IP2
Description	Evaluation of the bit rate of a service (e.g. video service, audio service) including the IP, UDP, RTP header
QoS Requirement	Service bit rate remains within a pre-defined band around a pre-defined nominal value
Example	256 kbps +/- 5 % or 4.0 Mbps +/- 100 kbps

2.1.2.2 IP packet error rate

QoS Parameter	IP packet error rate
Interface	IP1, IP2
Description	Number of corrupted IP packets vs. number of all IP packets
QoS Requirement	at IP1, i.e. the input of the encapsulator, the IP packet error rate should be very low considering the high number of potential users of a broadcast service at IP2 a certain degradation has to be taken into account due to the demanding radio channel in a mobile environment In both cases, the service bit rate and the acceptable probability of visible distortions determine the threshold for the IP packet error rate.
Example	at IP1: 10^{-6} (for a service of 4 Mbps this corresponds roughly to 1 visible artefact per hour) at IP2: $3 * 10^{-5}$ (at the end-user's terminal a higher IP packet error rate is acceptable, e.g. 1 visible artefact every 2 minutes)

2.1.2.3 IP Packet Jitter

QoS Parameter	IP packet jitter
Interface	IP1, IP2
Description	Variation of delay of IP packets expressed as peak-to-peak value
QoS Requirement	at IP1, i.e. the input of the encapsulator, the IP packet jitter is normally not so critical since the IP Encapsulator can buffer the jitter of incoming IP packets up to a certain extent at IP2 the IP packet jitter should be rather low, otherwise it could point to potential problems in the receiver buffer management
Example	at IP1: < 120 ms p-p at IP2: < 40 ms p-p

2.1.3 QoS requirements at interfaces TS1 and TS2

2.1.3.1 TS Synchronisation Loss

QoS Parameter	TS_sync_loss (as defined in TR 101 290 [1])
Interface	TS1, TS2
Description	Loss of synchronization of the MPEG2 Transport Stream
QoS Requirement	at TS1, i.e. the input of the DVB-T/H modulator, the TS has to be long term stable; any synchronisation loss points to a severe problem and should be attended to immediately; at TS2 the synchronisation loss is indicative for a lack of coverage for the receiver in a mobile environment; this could be caused by insufficient C/N as a result of insufficient field strength or too high speed in a mobile environment;
Example	at TS1: once per month at TS2: no synchronisation loss as long as the receiver is moved in the nominal coverage area with a Doppler shift of not more than 80 Hz

2.1.3.2 Sync Byte Error

QoS Parameter	Sync_byte_error (as defined in TR 101 290 [1])
Interface	TS1, TS2
Description	Sync_byte not equal 0x47
QoS Requirement	at TS1, the correct spacing between sync byte needs to be check to avoid synchronisation problems in the receiver population at TS2, frequent sync byte errors are an early indication of rising synchronisation problems, in many cases cause by to small a margin
Example	at TS1: once per month at TS2: no synchronisation loss as long as the receiver is moved in the nominal coverage area with a Doppler shift of not more than 80 Hz

2.1.3.3 BER (Bit Error Rate) before Reed-Solomon decoding

QoS Parameter	BER (see also ' BER before RS decoder' in TR 101 290 [1])
Interface	TS2
Description	BER before RS can be measured quickly and provides a good overview of the BER after RS value that can be expected. The relation between BER before RS and BER after RS dependents on the channel characteristics, i.e. only in certain circumstances a BER before RS of 10^{-4} corresponds to QEF after RS decoding.
QoS Requirement	low BER before RS to obtain an acceptable level of visible distortions;
Example	at TS2: $2*10^{-4}$

2.1.4 QoS requirements at interfaces RF1 and RF2

2.1.4.1 RF level

QoS Parameter	RF level (see also 'RF/IF signal power' in TR 101 290 [1])
Interface	RF1, RF2
Description	The RF level is measured as the signal power of the DVB-T/H signal within the nominal channel bandwidth.
QoS Requirement	at RF1, the RF level needs to be within a narrow tolerance of the specified transmitter output level at RF2, the RF level should be above the minimum input level of the receiver frontend by a certain margin; the minimum input level required by the receiver to produce a data stream with a BER below a pre-defined threshold, depends on the selected transmission mode and the channel conditions including the speed with which the receiver is moved;
Example	at RF1: 100 W +/- 0.5 dB at RF2: -100 dBm + theoretically required C/N (mode and channel dependent) + Margin (e.g. 10 dB)

2.1.4.2 C/(N+I) (Carrier to noise and interference ratio)

QoS Parameter	C/(N+I) (see also ' RF/IF signal power' and ' Noise power' in TR 101 290 [1])
Interface	RF2
Description	The carrier to noise and interference ratio is measured at the input of the receiver in order to estimate the margin available for a certain mode and channel. Both noise and interference signals influence the ability of the receiver to properly demodulate and decode the input signal.
QoS Requirement	at RF2, the C/(N+I) measurement can help identifying potential problems with the coverage in certain areas;
Example	at RF2: C/(N+I) > theoretically required C/N (mode and channel dependent) + Margin (e.g. 3 dB)

2.1.4.3 MER (Modulation Error Ratio)

QoS Parameter	MER (see also ' Modulation Error Ratio (MER)' in TR 101 290 [1])
Interface	RF1, RF2
Description	MER provides a single "figure of merit" analysis of the modulated signal. It includes various signal degradations such as noise, amplitude imbalance, quadrature error and residual errors, e.g. from insufficient carrier suppression or coherent interferers.
QoS Requirement	at RF1, MER gives a good overview of the quality of the transmitter output signal; at RF2, the MER measurement can help identifying potential problems in the receiver;
Example	at RF1: MER > 30 dB at RF2: MER > theoretically required C/N (mode and channel dependent) + Margin (e.g. 3 dB)

2.1.5 QoS requirements for handover (HO)

2.1.5.1 IP HO packet loss rate

QoS Parameter	IP HO packet loss rate
Interface	IP2
Description	Number of lost IP packets/ frames or GOPs in the receiver due to a handover (horizontal or vertical)
QoS Requirement Example	The number of lost packets should be so low that not more than a pre-defined number of video frames or a pre-defined number of GOPs is affected. at IP2: < 10 lost IP packets per handover or at IP2: max. 2 GOPs lost

2.2 WiMAX related QoS requirements

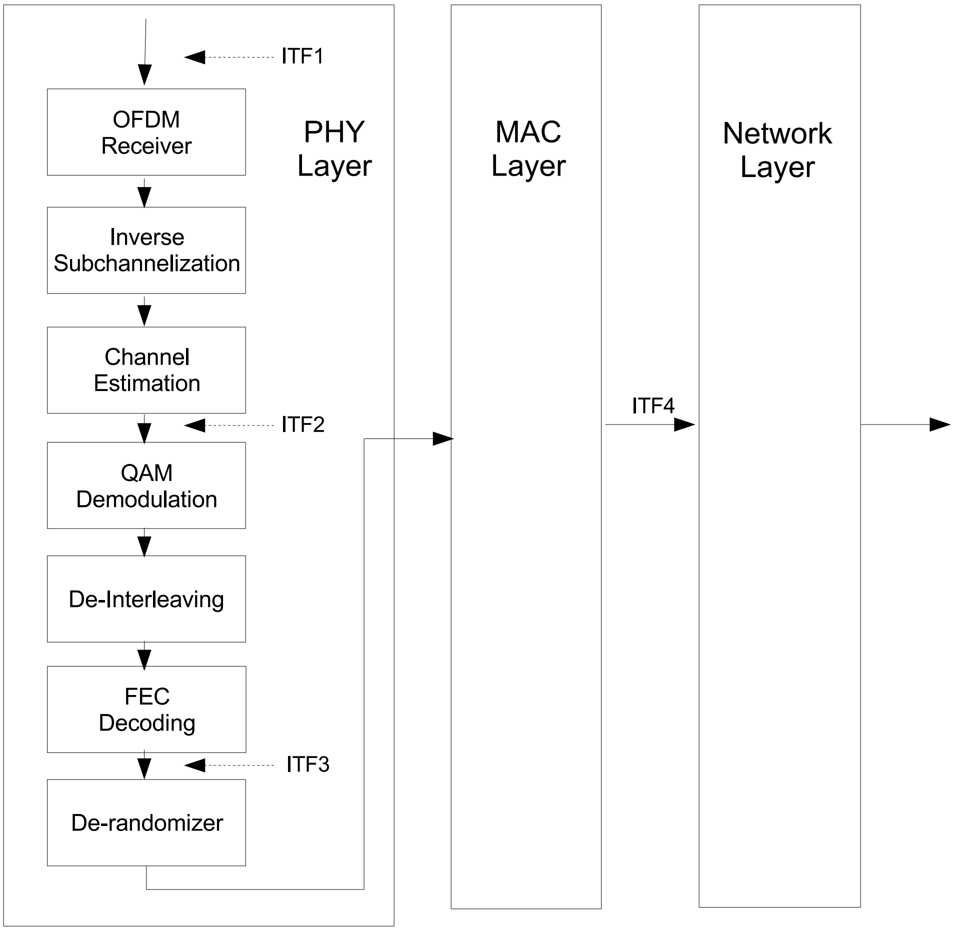

Figure 2-2 WIMAX receiver chain

The diagram in Figure 2-2 above illustrates a simplified WIMAX receiver chain within the OSI layer. The interfaces, ITF1, ITF2, ITF3 and ITF4 are the reference points for the specification of QoS requirements.

2.2.1 Service Bit Rate

QoS Parameter	Service bit rate
Interface	N/A
Description	Total service bit rate for the WIMAX network.
QoS Requirement	17.75 Mbps (as specified in Table 8, Section 7.7 of [2]). 6.25 Mbps for real time broadcasting, 6.25 Mbps for on demand video, 0.5 Mbps for hyperlinked video and 4.25 Mbps for streaming. In total 6.25 + 6.25 + 0.5 + 4.25 = 17.75Mbps.

2.2.2 RF Level

QoS Parameter RF level

Interface ITF1

Description The RF level is the signal power strength within the channel bandwidth. The signal power strength should be several times higher than the receiver noise and interference depending on the transmission mode and error performance requirement i.e. modulation and coding and BER.

QoS Requirement The required RF level, RFL, can be derived from simplified equation below:

$$RFL = SNR + IN + imploss$$

where

- SNR is the signal to noise ratio required for the selected transmission mode. The SNR value for given transmission mode can be obtained from Table 1
- IN is the sum of receiver thermal noise and interference within the measurement bandwidth
- $imploss$ is the implementation loss which includes non-ideal receiver effect like channel estimation, tracking errors, quantization errors and phase noise. This parameter is receiver dependent.

The receiver thermal noise and interference level can be calculated, for example, using equation 144 in [8].

2.2.3 Carrier to Noise and Interference Ratio (CINR) and Bit Error Rate (BER)

QoS Parameter CINR, BER

Interface ITF2, ITF3

Description CINR and BER are somewhat related QoS parameter since increasing CINR reduces the BER. The CINR is measured at the input of QAM demodulator (after the OFDM demodulator). CINR specifies the minimum QoS requirement for a target modulation and coding scheme for achieving a certain BER after the FEC decoding. CINR at this interface also includes the effect of receiver noise, imperfect channel estimation and implementation loss.

QoS Requirement

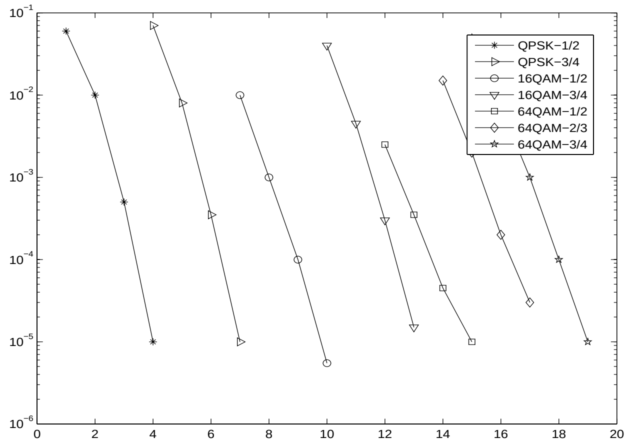

Figure 2-3 BER performance of WIMAX using CC in AWGN channel

Table 1 Required CINR for BER of 1e-6

Modulation	Coding Rate	Required CINR
QPSK	1/2	5
	3/4	8
16QAM	2/3	10.5
	3/4	14
64QAM	1/2	16
	2/3	18
	3/4	20

Assuming BER of 1×10^{-6} at interface ITF2, the required CINR value [7] for the mandatory tail-bitting Convolutional Code (CC) is specified in above table. Required CINR for other channel coding will be available shortly when WP2 simulation is complete. Note the required CINR values for each modulation scheme is obtained using AWGN simulation. The complete AWGN BER simulation results [7] are shown in graph above. Real channel measurement can be translated into above equivalent AWGN CINR using, for example, Exponential Effective SIR Mapping (EESM) [6] method. The mapping method is not standardized and can be defined by the manufacturer.

2.2.4 IP Packet Error Rate

QoS Parameter IP Packet Error Rate

Interface ITF4

Description Number of corrupted IP packets vs. the total number of transmitted packets for Scalable Video Coding (SVC) and Multiple Description Coding (MDC). For a given channel BER, the WIMAX MAC packetization scheme should fragment or packed packets to maximize throughput while ensuring packet error rate is not larger than a threshold. This threshold also depends on the modulation and channel coding and error requirement of SVC and MDC. Optimal packet size can be calculated from instantaneous BER and target packet error rate of SVC or MDC.

In WIMAX, three transmission modes are allowed: 1) Unicast 2) Multicast 3) Broadcast. In general, QoS for multicast and broadcast case can be grouped together as they do not support retransmission or Automatic Retransmission request (ARQ). For unicast case, higher channel packet error rate is allowed as reliability is achieved by ARQ at the cost of delay. Since fast response time is required for SUIT terminal, ARQ should be turned off. Thus its packet error rate requirement is the same to multicast and broadcast case.

QoS Requirement Note that for MDC case, packet error rate is measured over both channels, since video stream is split into two descriptions which are then transmitted over DVB-T and WIMAX respectively. As for SVC, the packet error rate is measured over a single wireless interface assuming the interface carries all the video packets.

MDC: Packet error rate on both DVB-T and WIMAX channel < 0.2, for a single WiMAX or DVB-T channel it should be < 0.1 [3]

SVC: Packet error rate on for either DVB-H or WIMAX channel < 0.05 [4]

2.2.5 IP Packet Jitter

QoS Parameter IP Packet Jitter

Interface ITF4

Description Variation of delay of IP packets expressed as peak-to-peak value. Packet jitter is less critical in WIMAX network since there is usually dejittering buffer at the end application. Use of dejittering buffer in WiMAX network is not recommended as it might introduce extra delay [5]. However at ITF3 the jitter should not but too large to avoid buffering problem.

QoS Requirement < 40 ms p-p after WiMAX receiver (following value from 2.1.2.3.)

2.2.6 Vertical Handover from WIMAX to DVB-T/H Network

2.2.6.1 Handover Delay

QoS Parameter Handover Delay

Interface N/A

Description This is the delay experienced at receiver due to a vertical handover. Assuming hard handover (break before make), the delay will have direct impact on number of packet/ frame loss. As SUIT terminal have both wireless interfaces i.e. DVB-T/H and WIMAX, soft handover (make before break) utilizing two tuners is possible thus minimizing packet loss/ frame loss. The parameter below serves as an upper bound.

QoS Requirement	For frame loss < 2 GOP, the handover delay must be less than 1 GOP time since receiver may start losing packet from current GOP continuously up to next GOP (if the packet loss hits an I or P frame). Assuming typical 25 fps and 12 frame GOP structure, this roughly corresponds to maximum handover delay of 500 ms (i.e. 1 GOP = 12 frame = ~ 500 ms).

2.2.6.2 Parameters to Trigger Handover

Parameter	IP packet loss rate
Interface	ITF2
Description	This is the parameter used to trigger vertical handover from WIMAX network to DVB-T/H. The user terminal will maintain a list of active transmitter and continuously monitor the channel CINR on WIMAX and DVB-T/H network for handover purposes. The current channel CINR can be measured and mapped to an estimated packet loss rate. Based on this measurement and comparison to DVB-T/H estimated packet loss rate, decision whether to perform handover is then taken. This decision should depend on a hysteresis function and a delay window to avoid “ping-pong” effect. The decision can also depend on the defined network policy, i.e. should a mobile connect to different network (e.g. DVB-T/H) with lowest packet loss rate performance although it can still find transmitter on the same network (e.g. WIMAX) but with higher packet loss rate although both networks satisfy the QoS requirement of video application? In any case, IP packet loss rate is probably the best trigger parameter for vertical handover as it is independent from the physical layer of the wireless interface and a comparable parameter between different wireless networks.
QoS Requirement	N/A

2.3 DVB-RCT related QoS requirements

The QoS in RCT is based on the CPE to BST connections management. It supports all configuration according to 802.1p, 802.1q, Diffserv, RSVP, CIR, MIR, Average delay. After connections are created it defines traffic nature and QoS (average delay, max jitter, CIR, MIR etc). In order to understand how the QoS is managed the following subjects should be clarified:
(All definitions are with regard to RCT BS3 mode that is suitable to Ethernet/ IP transmission.)

- Upstream is divided into Logical Channels – Sub-Channels
- Channel can be dedicated to connection or shared by several connections
- Each connection can receive n sub-channels for upstream transmission (n=1..54 in BS3)
- Several sub-channels are used to send MC-CDMA codes (anonymous)
- Channel is divided into:
  - –Ranging slots (shared by all CPEs)
  - –Contention slots (shared by all CPEs)
  - –Reserved slots (dedicated to a single CPE)
- Upstream traffic is divided into:
  - –Ranging codes
  - –MAC messages on contention slots
  - –MAC messages on reserved slots
  - –User data packets

The Figure 2-4 describes RCT sub-layer division:

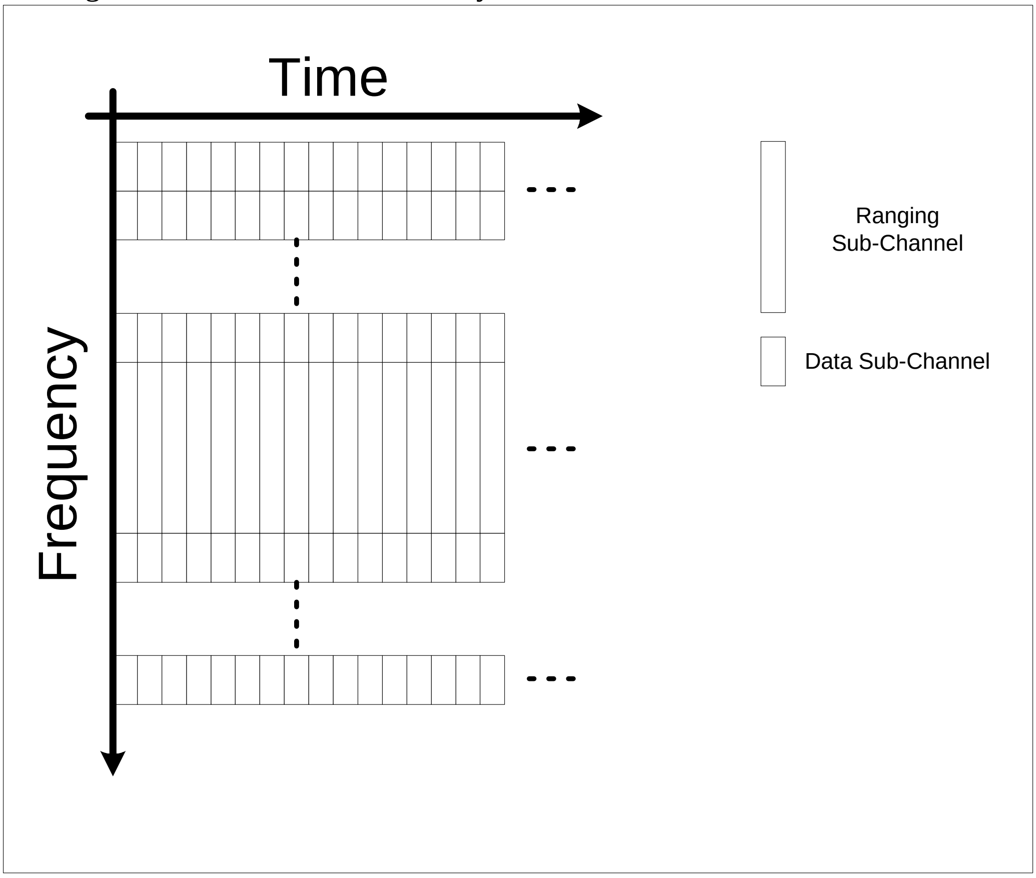

Figure 2-4: RCT sub-layer division

- Data packets encapsulated in ATM AAL5 format
- AAL5 packet is made of user data and ALL5 trailer
- ALL5 packets are sent from CPE to BST
- AAL5 is fragmented into ATM cells
- ATM cell format is:
  - – 5 bytes - ATM header (VPI, VCI, HEC etc)
  - – 48 bytes - user data
- Sub-channels are allocated in multiples of ATM cells

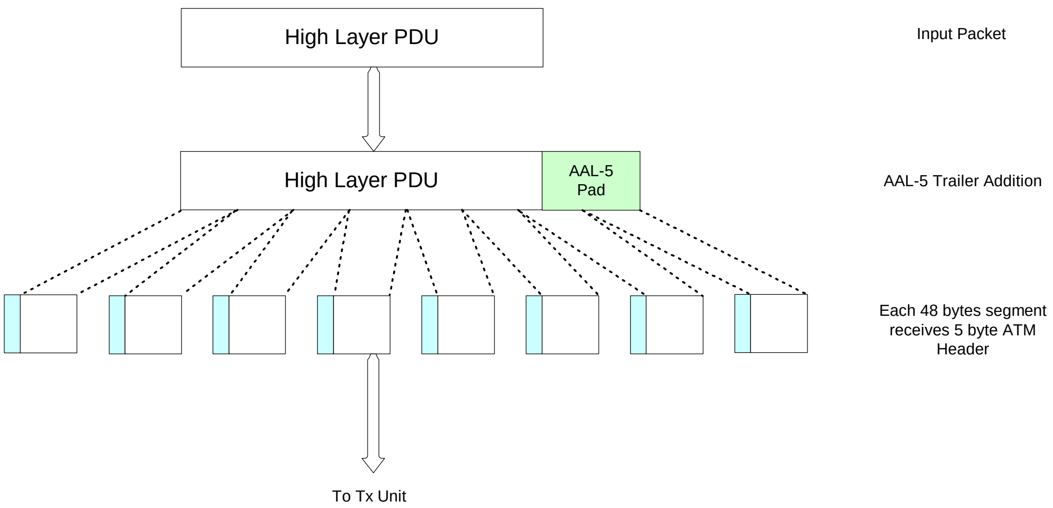

Figure 2-5: Sub-channel allocation to ATM cells

- CPE is using one of the shared sub-channels to request bandwidth allocation:
  - –Ranging slots – send CDMA request
  - –Contention slot – send MAC message
  - –Piggy-Back – on Data messages
- CPE use dedicated sub-channels to send:
  - –MAC message (bandwidth allocation, etc)
  - –User data packets
  - –Fillers/ null information

- Allocate bandwidth grants:
  - –According to “regular” BW request from the user (e.g. piggy-back, Reservation Request message)
  - –According to CDMA BW request.

Figure 2-6 describes the data flow from the CPE to the BST.

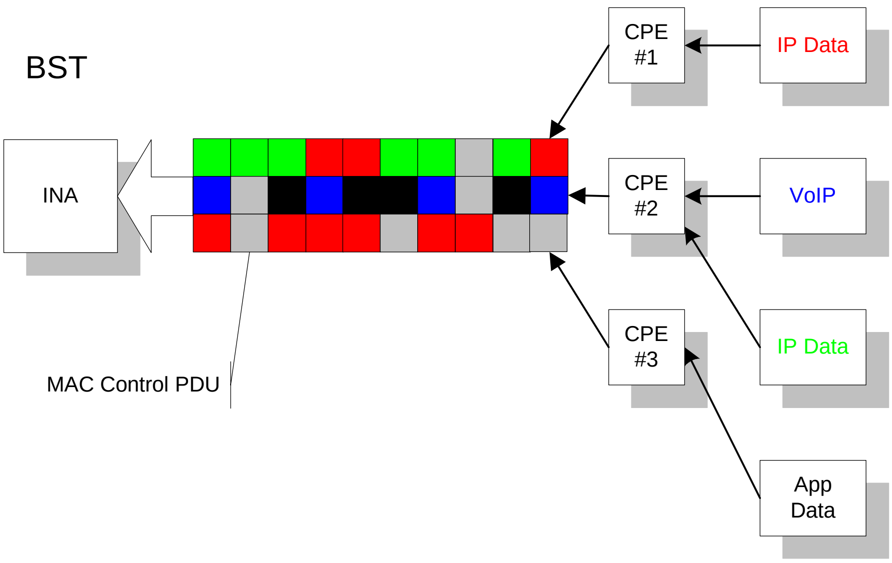

Figure 2-6: Data flow from the CPE to the BST

The RCT QoS parameters are as follows:

- Bandwidth is divided to connection to satisfy the following rules (according to priority):
  –Committed Information rate (CIR) – satisfy jitter
  –CIR – satisfy Average delay
  –CIR – according to priority
  –All other packets – less then connection MIR (Maximum Information Rate)
- Keep-alive packets (hot line)
- If allocated sub-channels
- High priority MAC messages
- Fix allocation data connections
- Low priority MAC messages
- Other data connections
- Idle/ null
- When no sub-channels allocated
- Send CDMA code
- Bandwidth allocation per connection

Figure 2-7 below describes DVB-RCT MAC Queues Overview.

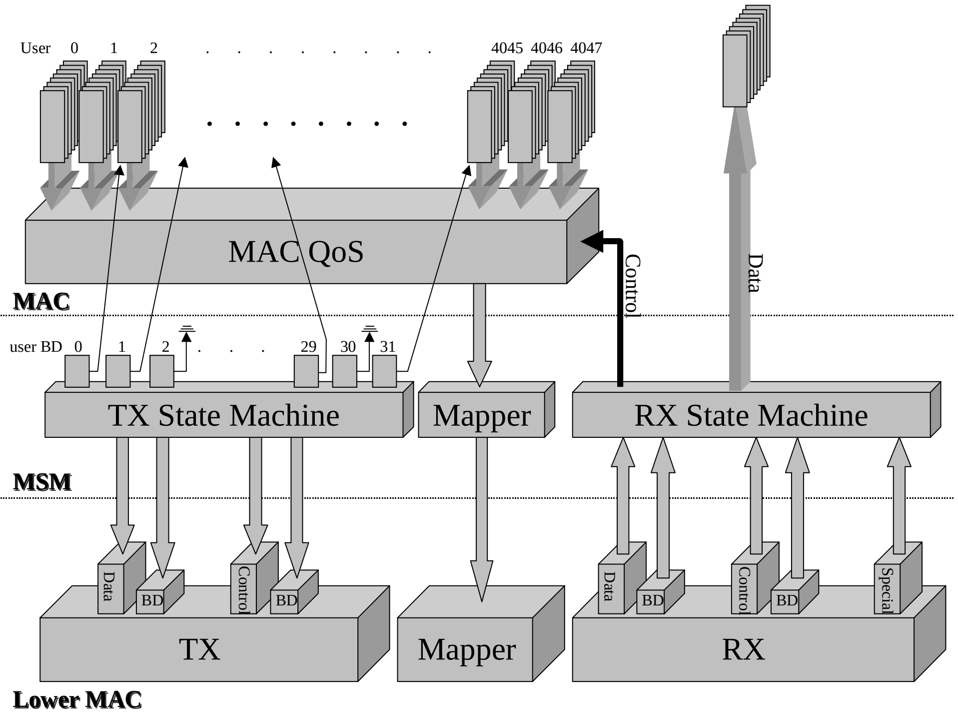

Figure 2-7: DVB-RCT MAC Queues Overview

- 8 level QoS (support 802.1p, 802.1q, diffserv etc)
- BW request per connection (Vs CPE)
- Multi connection per CPE (virtual connection)
- Adaptive Power/Synchronization control
- High level of security
  - –Authentication
  - –Encryption
- Support Broadcast, Multicast and Unicast traffic
- Small delays (10msec)
- Minimal jitter (3msec)

The RCT standard [9] defines the following MAC primitives for the QoS

8.4.9.3 <MAC> Reprovision Message (Singlecast Downstream)

The <**MAC**> **Reprovision Message** is sent by the INA to the NIU to reassign upstream resources (maintaining the originally requested QoS parameters at the establishment of the connection). This message is intended for fixed rate based channel maintenance by the INA to redistribute or reassign resources allocated to a NIU.

Table 66: Reprovision Message structure

Reprovision_Message (){	Bits	Bytes	Bit Number/Description
Reprovision_Control_Field Reserved Delete_Reservation_Ids Reserved New_Upstream_Channel_Included New_Downstream_Channel_Included New_Cyclical_Assignment_Included New_Slot_List_Included	 1 1 2 1 1 1 1	1	 7 6: {no,yes} 5..4: 0 3: {no,yes} 2: {no,yes} 1: {no,yes} 0: {no,yes}
if (Reprovision_Control_Field and= New_Upstream_Channel_Included) {			
New_Upstream_Channel	(8)	(1)	
}			
if (Reprovision_Control_Field and= New_Downstream_Channel_Included) {			
New_Downstream_Channel	(32)	(4)	
}			
if (Reprovision_Control_Field and= New_Slot_List_Included || New_Cyclical_Assignment_Included || Delete_Reservation_IDs){			
Number_of_Connections	(8)	(1)	
for(I=0;I<Number_of_Connections;i++){			
Connection_ID	(32)	(4)	
if(Reprovision_Control_Field and= New_Slot_List_Included){			Fixed Rate Access
Number_Slots_Defined	(8)	(1)	
for(i=0;i<Number_Slots_Assigned;i++){			
Slot_Pattern	(64)	(8)	
}			
}			
if (Reprovision_Control_Field and= New_Cyclic_Assignment_Included){			Fixed Rate Access
Fixedrate_Start_Pattern	(64)	(8)	
Fixedrate_Distance	(16)	(2)	
Fixed_Rate_End_Slot_Number	(16)	(2)	
}			
}			
}			
}			

Reprovision Control Field:

Reprovision_Control_Field specifies what modifications to upstream resources are included. It consists of the following sub fields:

- **Delete_Reservation_IDs** is a Boolean that indicates that the NIU/RCTT deletes all Reservation_IDs that have been assigned to the Connection_IDs contained in this message.
- **New_Upstream_Channel_Included** is a Boolean that indicates that a new upstream channel is specified in the message.
- **New_Downstream_Channel_Included** is a Boolean that indicates that a new downstream channel is specified in the message.
- **New_Cyclical_Assignment_Included** is a Boolean that indicates that a new cyclical assignment is specified in the message. If the connection has already cyclic fixed rate slots or a slot list assigned, these slots are lost. Having Cyclic Assignments and Slot List Assignments for the same Connect_ID at the same time is not allowed.
- **New_Slot_List_Included** is a Boolean that indicates that a new slot list is specified in the message. If the connection has already cyclic fixed rate slots or a slot list assigned, these slots are lost. Having Cyclic Assignments and Slot List Assignments for the same Connect_ID at the same time is not allowed.

New Upstream Channel:

This is an 8-bit unsigned integer representing upstream channel identifier.

New Downstream Channel:

This is a 32-bit unsigned integer representing the reassigned upstream carrier centre channel. The unit of measure is Hertz (Hz).

Number_of_Connections:

This is an 8-bit unsigned integer that represents the number of connections to which the reprovisioning of slots applies.

Connection_ID:

This is a 32-bit unsigned integer identifying the connections that are affected by the reprovisioned slots.

Number of Slots Defined:

This is an 8-bit unsigned integer that represents the number of slot assignments contained in the message. The unit of measure is slots.

Slot Pattern:

This is a structure that represents the Fixed Rate based pattern assigned to the NIU.

Fixed Rate Start Pattern:

This is the slot pattern structure described above, representing the first pattern that is granted to the NIU/RCTT within the fixed rate access region.

Fixed Rate Distance

This 16-bit unsigned number represents the distance in time duration slots between additional slots assigned to the NIU. The NIU is assigned all slots that are a multiple of Fixedrate_Distance from the Fixedrate_Start_slot that do not exceed Fixedrate_End_slot.

2.4 WiFi related QoS requirements

2.4.1 Introduction

Between the SUIT gateway and the WIFI terminals a wireless connection compliant to IEEE802.11g will be established.
Figure 2-8 depicts the functional blocks of the relevant SUIT devices to be operated by a WIFI-connection. In blue are depicted relevant measurement points of QoS.
The SUIT gateway interfacing with broadband networks DVB-T/DVB-H/WIMAX also interfaces to WLAN (IP1). The gateway bridges either original or trans-coded scalable contents at UDP/IP level via an 802.11g connection to one or more WIFI end-user terminals.
The access to the wireless medium of both gateway and WIFI terminals can be monitored at measurement points labelled RF1, RF2 and RF2'.

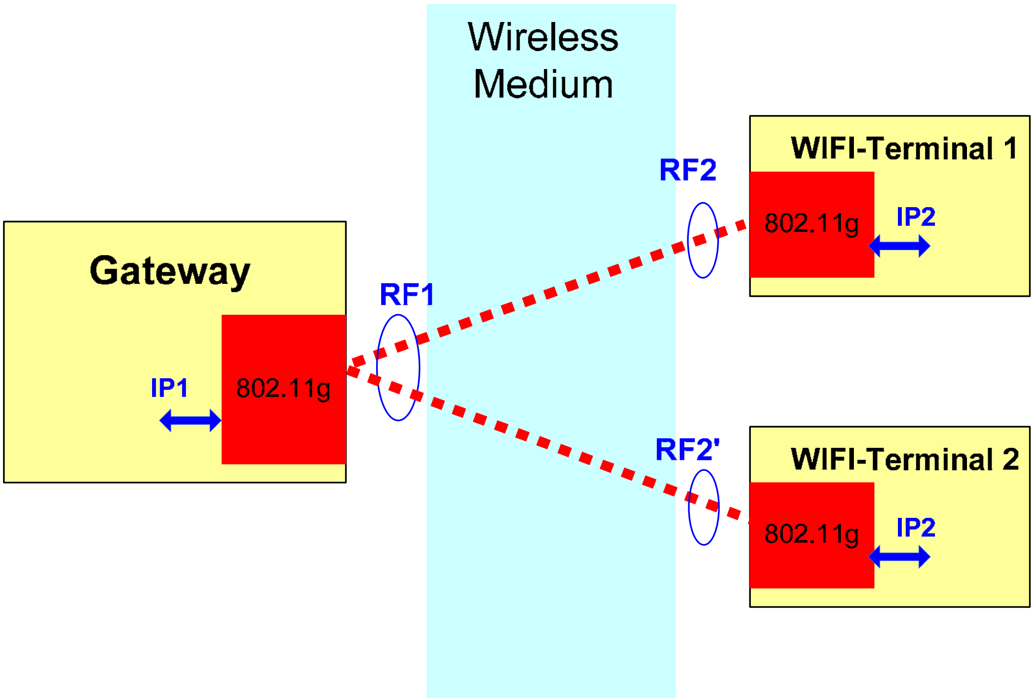

Figure 2-8: Relevant monitoring points for the WIFI connection

2.4.2 WLAN overview

WLAN 802.11g operates at the licence-free ISM-band at 2.4 GHz band deploying OFDM-technique.

WLAN Access Medium methods

In 802.11g networks, stations transmit and receive over the same shared channel (half duplex communication). If more than one station transmits a frame at the same time, a collision occurs and the packet will not be received.
The existing 802.11 standard has defined two wireless medium (WM) access methods which may co-exist:

the distributed coordination function (DCF)

and

the point coordination function (PCF).

DCF:

When a station needs to access the WM, it monitors the channel to determine whether someone else is transmitting. If the medium it is not idle, the station with the traffic to send waits a random backoff period before listening again and trying to transmit.
DCF is a medium access method with contention among the partitioning stations. With the DCF mechanism, over a long period of time all devices on a given channel are provided equal access ("fairness") to the WM. This works well for traditional data applications like ftp transfers, web-browsing, and other non-multimedia based applications.
DCF provides no priority mechanism. Quality of Service cannot be guaranteed for any data-traffic.

***PCF*:**

As an optional Medium Access Mode there is defined the “Point Coordinated Function PCF”.
An Access Point (AP) using PCF controls channel use by polling stations for traffic. So, the AP decides which client is allowed to communicate and how long exclusively. Stations with traffic to send are assigned "contention free" periods during which they can transmit.
In respect to the wireless access, PCF provides more QoS than DCF, however, although PCF enables AP control over channel utilization, it provides no mechanism for the AP to know what kind (priority) of traffic each station has to send.
PCF is optional and not implemented in most products on the market.

Reliability

802.11 specified an Acknowledgement mechanism at MAC-layer level applied after each transmitted packet. In case a data frame is not received correctly, e.g. due to a occurred collision or interferences, the absence of an ACK initiates the re- transmission of the previously sent data frame. With this handshake mechanism, high reliability can be achieved on 802.11 networks, but with the price of loose of data throughput.

Data throughput

The link rate of 802.11g is up to 54 Mbit/s, however, the real data throughput is much lower.
The management of the wireless network requires quite a lot of data exchange between the clients and the base stations. Due to this overhead, real network throughput is much lower than the link rate and does not exceed more than 25Mbit/s. In practice, net data throughput is often even lower, depending significantly on environment conditions (signal reception, interferences), and on the number of stations accessing the wireless medium.

Co-Channel Interference

In Europe, WLAN channels run from channel 1 to 13 in the frequency range between 2400 and 2483,5 MHz (802.11b/g). This frequency range is divided in channels by a channel separation of 5MHz. As already one channel requires a bandwidth of 22 MHz, only 3 channels (e.g. channel-number 1, 7 and 13) do not overlap other channels. Co-channel interference slows down data rate.

Interference between channels has to be avoided in particular for multicast. A good signal quality without interference is important where reliability on link layer level is not ensured anymore by disabled ACK mechanism.

Compatibility to Legacy 802.11b

As the 802.11g standard was designed to be interoperable with IEEE802.11b, switching between OFDM and High Rate DSSS (802.11b) methods is supported.
However, when operating in interoperability mode to 802.11b, the maximum usable data rate is even reduced significantly to max.15 Mbit/s.

Adaptive Modulation

Dependent on the selection of the modulation scheme, more or less bits per symbol can be transmitted. 64-QAM allows achieving higher throughputs or better spectral efficiencies than e.g. 16-QAM..
However, it must also be noted that when using a modulation technique such as 64-QAM, better signal-to-noise ratios (SNRs) are needed to overcome any interference and maintain a certain bit error ratio (BER).
In WLAN 802.11, the decision, which parameters to be chosen, is triggered from measured incoming S/N-ration and performed dynamically and swiftly. Adaptation of the transmission mode in accordance with the condition of the channel is called “adaptive modulation”.

In Table 2 below, dependent on typical signal reception strength values, link rates to be expected are listed. For the maximum link rate of 54 Mbit/s, sufficient signal strength is required. Within a range of about 20dB link rate changes from 54 Mbit/s down to 6 Mbit/s

Signal reception strength	Link rate	Modulation	Convolutional Coderate	Bits / OFDM-Symbol	1 OFDM Symbol
> -71dBm	54Mbit/s	64-QAM	¾	288	48 x 64QAM-sub-carriers
-73dBm	48Mbit/s	64-QAM	$^{2}/_{3}$	288	48 x 64QAM-sub-carriers
-78dBm	36Mbit/s	16-QAM	¾	192	48 x 16QAM-sub-carriers
- 82dBm	24Mbit/s	16-QAM	½	192	48 x 16QAM- sub-carriers
- 84dBm	18Mbit/s	QPSK	¾	96	48 x QPSK- sub-carriers
- 86dBm	12Mbit/s	QPSK	½	96	48 x QPSK- sub-carriers
- 90dBm	9Mbit/s	BPSK	¾	48	48 x BPSK- sub-carriers
- 92dBm	6Mbit/s	BPSK	½	48	48 x BPSK- sub-carriers

Table 2: Adaptive modulation – interdependence of link rate and signal reception strength

2.4.3 Considerations on QoS on IEEE 802.11g

In fact, WLAN 802.11g works on a “best effort” basis not providing real QoS. Due to the non-deterministic nature of DCF (PCF is not mandatory and does not introduce real QoS either), access to the wireless medium is statistically fair for all stations sharing the medium, but not fair for time critical applications. In addition to that, interference on the wireless channel occurs which causes packet retransmission and an impact on QoS.
Therefore, it is not possible—except in a very lightly loaded network—to expect some more reliable values for bandwidth, jitter, and latency.

However, despite the fact that WLAN802.11g does not offer guaranteed QoS, best service quality might be accomplished when following a set of arrangements:

- Reception signal strength: in order to have highest link rate available (54Mbit/s) it should be ensured that due to Adaptive Modulation stations have a good reception signal. Signal level should be higher than -65 dBm.
- Legacy: only 802.11g devices should share the same channel to avoid operation in interoperability mode to 802.11b with significantly reduced bandwidth.
- Channel planning: to avoid interferences with adjacent channels (“co-channel interference”), it should be ensured that the chosen channel does not overlap with other channels in the same geographical coverage area.
- Reduced contention: Minimum values for jitter and latency can be achieved by avoiding -to the extent possible - re-transmissions. One step into this direction is to minimise or even to avoid contention between stations in order to prevent collisions. The optimum case is to permit only one station connecting to the Access point, in other words to establish a point to point connection. In that special case, available bandwidth needs not to be shared amongst other accessing stations.
- Network load: heavy load imposed by many station typically leads to increased values of latency and jitter. In the worst case, packets may be lost when a maximum number of re-transmission is exceeded.

In case of Multicast and Broadcast, Acknowledge mechanism providing reliability due to re-transmission of corrupted packets is not applicable anymore.
Hence, for multicast/broadcast all requirements stated above become crucial, in order to keep packet loss as low as possible.

2.4.4 Improvements with the QoS-Standard IEEE 802.11e

As previously defined standards do not specify dedicated mechanism for Quality of Service, the 802.11e standard emerged. 802.11e extends IEEE 802.11 a, h and g for an improved support for multimedia applications and Voice over IP (VoIP).
For e.g. an HDTV application, there needs to be an assurance to the application that the high bandwidth it requires is available and will remain available and will not be pre-empted by applications that don't require high bandwidth such as traditional data applications.

In 802.11e, a hybrid coordination function (HCF) is added which provides QoS capabilities not available from the DCF. The HCF provides two WM access methods: enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA).

Priority scheme EDCA
EDCA (Enhanced Distributed Channel Access) is a superset of the 802.11 DCF. In the DCF, all stations compete for the WM with the same priority. In EDCA, on the other hand, this mechanism is extended to four levels of priorities or access categories (AC) that are used to prioritize traffic to provide enhanced multimedia support:

- Background Priority	low priority traffic (file downloads, print jobs)
- Best Effort Priority	traffic less sensitive to latency, but affected by long delay (internet surfing)
- Video Priority	video traffic
- Voice Priority	highest priority, low latency Voice-over IP calls.

The four access categories of EDCA are mapped from the eight priorities defined in 802.1d.

This "prioritized QoS" EDCA mechanism makes use of shorted maximum backoff time for higher priority traffic. A higher priority AC wins access to the WM more frequently than the lower priority AC. Therefore, statistically, the packets with the highest AC are given access to the medium more frequently for longer durations, than lower access category traffic
However, due to the non-deterministic nature of EDCA, it is not possible—except in a very lightly loaded network—to guarantee parameters such as bandwidth, jitter, and latency. In addition, due to the backoff mechanism, the medium usage is much less efficient than HCCA

HCCA (optional)
Unlike the non-deterministic behaviour of EDCA with its statistical uncertainty of random backoffs, HCCA (controlled channel access) provides deterministic behaviour and a high level of control and fidelity to multimedia applications that require parameterized QoS such as bandwidth, jitter, and latency.

HCCA has similarity to PCF and is generally considered the most advanced (and complex) coordination function. With the HCCA, priority for traffic can be configured with great precision. QoS-enabled stations have the ability to request specific transmission parameters (data rate, jitter, etc.) which should allow advanced applications like VoIP and video streaming to work more effectively on a Wi-Fi network.

HCCA support is not mandatory for 802.11e APs. In fact, few (if any) APs currently available are enabled for HCCA..

2.4.5 Improvements on WLAN 802.11n

802.11n is in the state of final draft now (October 2006). So-called pre-n products based on the joint draft already proved for what the 802.11n is supposed to improve - the enhancement of bandwidth efficiency and consequently, this will also impacts on resulting Quality of Service for Wireless Lan. However, 802.11n is not expected to be standardised before summer 2007.

Enhancement of throughput:
New operation modes in the physical layer (PHY) enable higher link rates up to 600 Mbit/s
This gain at the link rate will lead to higher quality for the transmission for an application due to reduced contention between different clients.

MIMO
802.11n will enable the support of MIMO-technique (Multiple Input Multiple Output) promising significant boost in performance for OFDM-WLAN systems.
A MIMO system takes advantage of the spatial diversity that is obtained by spatially separated antennas in a dense scattering environment. MIMO results in an increase of diversity gain to combat impairments like interference and noise which produces signal fading and/or to enhance the system capacity on time-variant and frequency-selective channels.

2.4.6 Specification of QoS Requirements on WLAN802.11

2.4.6.1 WIFI receiver chain

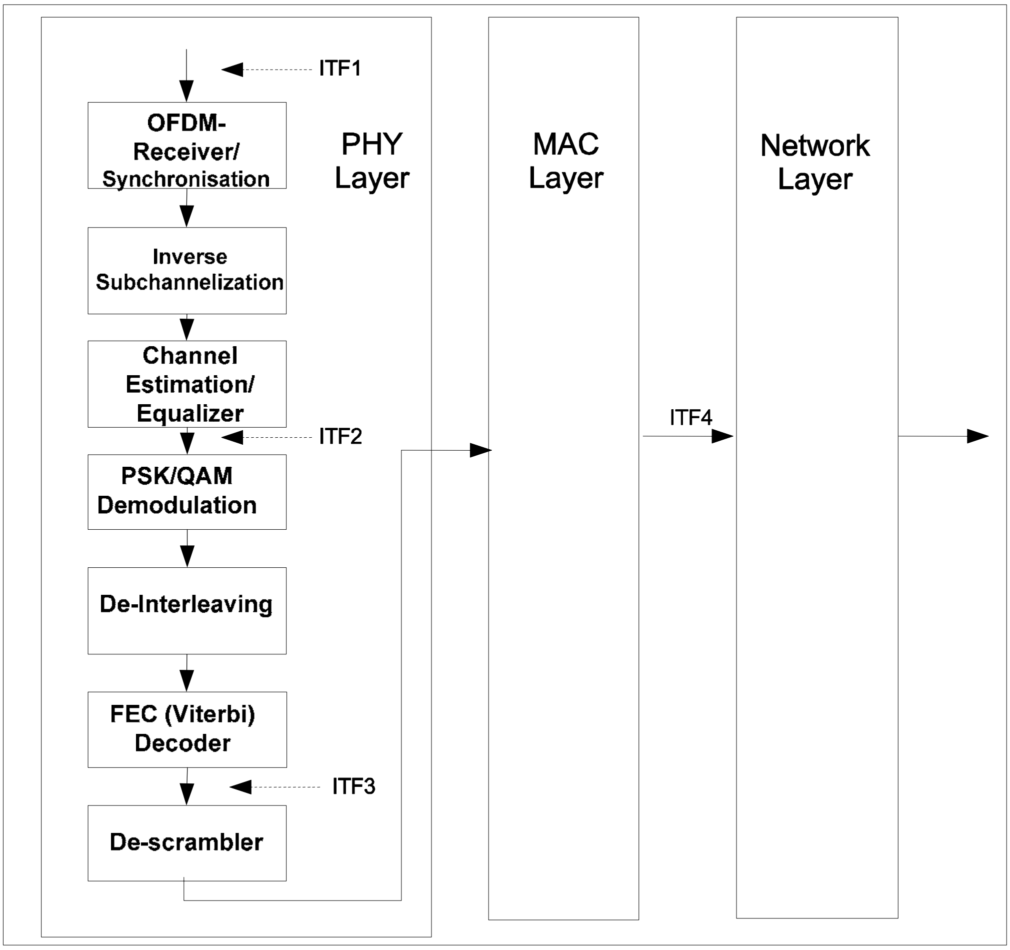

Figure 2-9: WIFI receiver chain

The diagram in Figure 2-9 above illustrates a simplified WIFI receiver chain within the OSI layer. The interfaces, ITF1, ITF2, ITF3 and ITF4 are the reference points for the specification of QoS requirements.

2.4.6.2 Service Bit Rate

QoS Parameter	Service bit rate
Interface	IP2 / IP2' (Figure 2-9)
Description	Total service bit rate for the WIFI network. As 802.11g does not support prioritized traffic, a minimum bandwidth needs to be assigned
QoS Requirement	Service bit rate > 15 Mbit/s

2.4.7 RF Level

QoS Parameter	RF level
Interface	ITF1 (Figure 2-9) / RF1, RF2, RF2' (Figure 2-8)
Description	The RF level is the signal power strength within the channel bandwidth. The signal power strength should be several times higher than the receiver noise and interference depending on the transmission mode and error performance requirement i.e. modulation and coding and BER. The RF level should be high enough for a link rate of 54Mbit/s (Adaptive Modulation)
QoS Requirement	The required RF level shall be > -65dBm

2.4.8 Carrier to Noise and Interference Ratio (CINR) and Bit Error Rate (BER)

QoS Parameter	CINR, BER
Interface	ITF2, ITF3
Description	*CINR* and *BER* are somewhat related QoS parameter since increasing *CINR* reduces the *BER*. The *CINR* is measured at the input of QAM demodulator (after the OFDM demodulator). *CINR* specifies the minimum QoS requirement for a target modulation and coding scheme for achieving a certain *BER* after the FEC decoding. *CINR* at this interface also includes the effect of receiver noise, imperfect channel estimation and implementation loss.
QoS Requirement	to be defined following results from WP2

2.4.9 IP Packet Error Rate

QoS Parameter	IP Packet Error Rate
Interface	ITF4 (Figure 2-9), IP2, IP2' (Figure 2-8)
Description	Number of corrupted IP packets v.s. the total number of transmitted packets for Scalable Video Coding (SVC). In WIFI, three transmission modes possible: 1) Unicast 2) Multicast 3) Broadcast. In general, - provided the wireless network is not heavy traffic loaded and the reception condition is sufficient- for DCF in unicast mode, high reliability of the channel is achieved, thanks to the MAC Acknowledgement

	mechanism, however at the cost of bandwidth. In multicast and broadcast, it is required retransmission or Acknowledgement (ACK) is not supported.
QoS Requirement	As for SVC, the packet error rate is measured over a single wireless interface assuming the interface carries all the video packets. SVC: max. permitted Packet error rate (to be defined in WP3)

2.4.10 IP Packet Jitter

QoS Parameter	IP Packet Jitter (RFC1889)
Interface	ITF4 (Figure 2-9) / IP2, IP2' (Figure 2-8)
Description	Variation of delay of IP packets expressed as peak-to-peak value.
QoS Requirement	< 10 ms p-p after WIFI receiver

2.4.11 IP Response Time

QoS Parameter	IP Response Time
Interface	ITF4 (Figure 2-9) / IP2, IP2' (Figure 2-8)
Description	Maximum Response time of the system. This value is dependent on traffic load and is influenced by latency.
QoS Requirement	Max. Response time < 25 ms

2.5 QoS criteria evaluated to trigger DVB-T/H → DVB-T/H handover

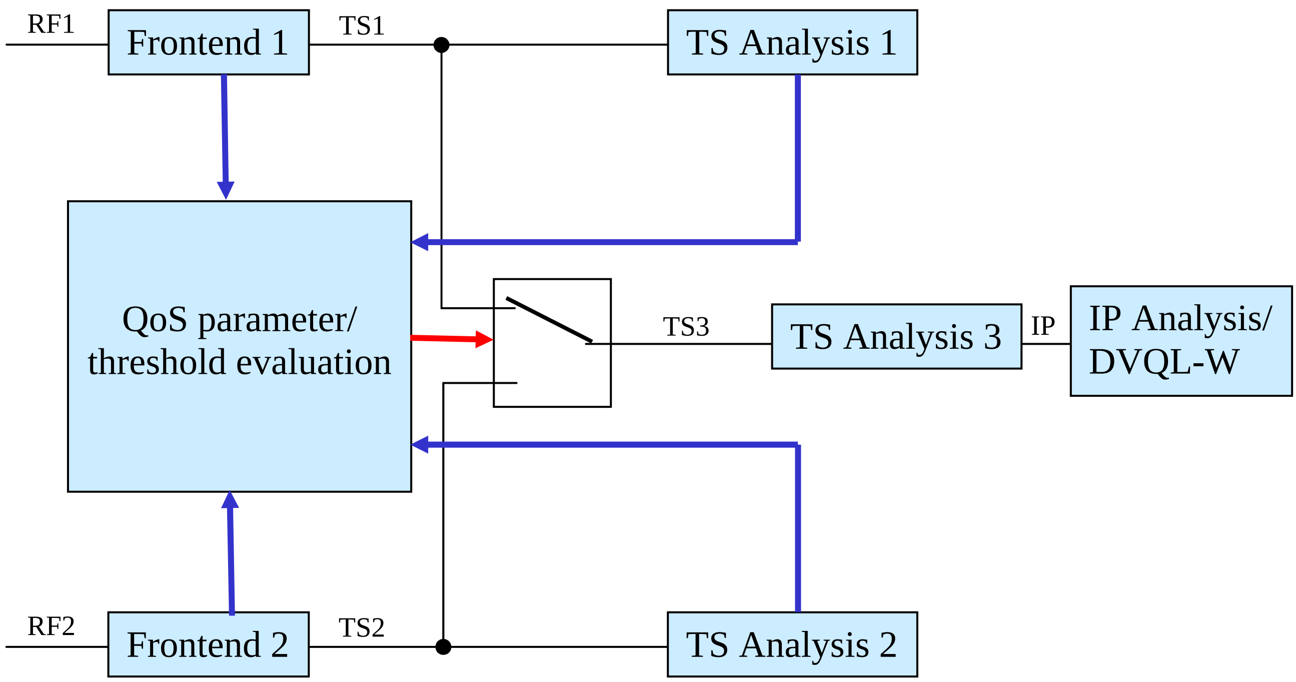

Figure 2-10: Test receiver set-up for investigation of horizontal handover (DVB-T/H → DVB-T/H)

QoS Parameter	Transport_error (see Tr 101 290, clause 5.2.2 [1])
Interface	TS1, TS2 see Figure 2-10
Description	A horizontal handover from one DVB-T/H cell to another is triggered if - the PSI/SI data indicate that the same service is available in another DVB-T/H cell which the terminal can receive - the Transport_error_indicator is set for 5 % of the Transport Stream Packets of the respective service over a period of 10 sec - the Transport_error_indicator for the alternative service shows a significantly lower level of distortions (e.g. 2 % over a period of 10 sec)
QoS Requirement Example	significant difference between reception quality on both channels, measured as averaged rate of Transport_error_indicator Transport_error_indicator > 5 % over 10 sec and Transport_error_indicator of alternative service < 2 % over 10 sec

2.6 QoS criteria evaluated to trigger DVB-T/H → WiMAX handover

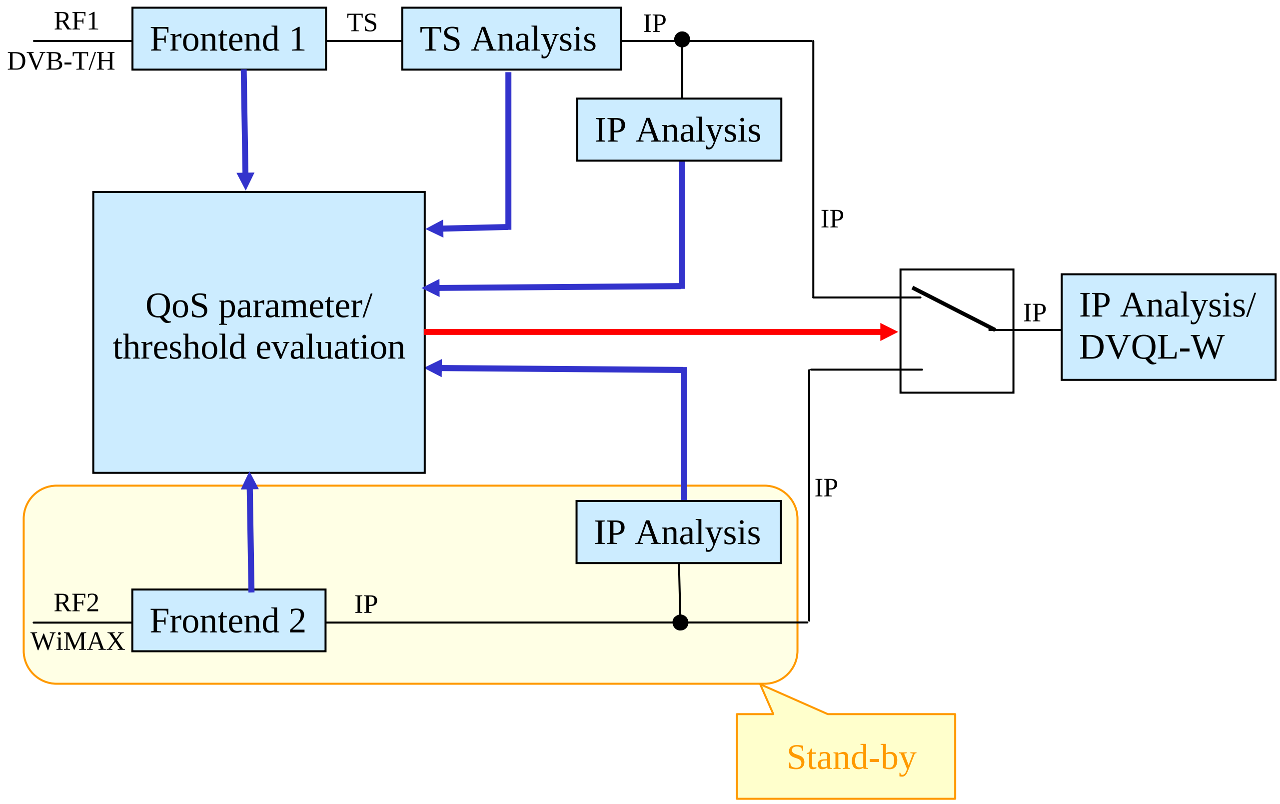

Figure 2-11: Test receiver set-up for investigation of vertical handover (DVB-T/H → WiMAX)

QoS Parameter	IP packet error rate
Interface	IP in DVB-T/H branch of test receiver IP in WiMAX branch of test receiver see Figure 2-11
Description	A vertical handover from a DVB-T/H cell to a WiMAX cell is triggered if - the same service is available in the WiMAX cell which the terminal can receive - the IP packet loss rate is more than 5 % for the respective service over a period of 10 sec - the IP packet loss rate for the alternative service over WiMAX shows a significantly lower level of distortions (e.g. 2 % over a period of 10 sec)
QoS Requirement Example	significant difference between reception quality on both channels, measured as averaged IP packet loss rate IP packet loss rate for DVB-T/H service > 5 % over 10 sec and IP packet loss rate for the same service over WiMAX < 2 % over 10 sec

2.7 Microwave links related QoS requirements

The base stations shall be linked to the playout via an IP connection, which will be implemented with microwave links on the trial platform.

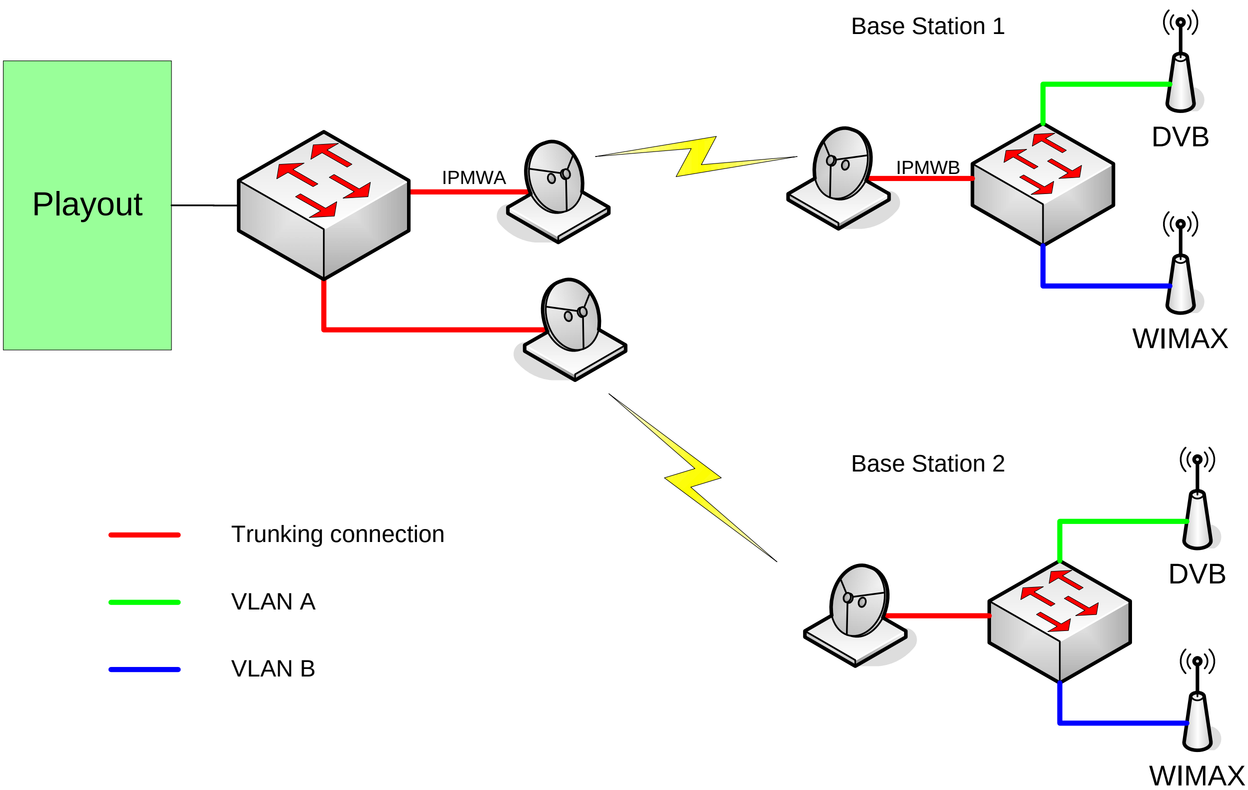

Figure 2-12: Transmission solution for the Base Stations via microwave links

The reason to deploy the demonstrator with microwave links as backhaul is that it makes it possible to have a demonstrator self supported on transmission. Moreover, the use of unlicensed band is appropriate, as long as performance requirements are fulfilled, because unlicensed band solutions are portable to the several EU countries (as long as ETSI certified band is used).

For general use in EU, the most suited unlicensed band is the one in the range 5470MHz-5725MHz, since this band permits the highest EIRP (Equivalent Isotropic Radiated Power), it is allocated for high performance outdoor networks (Hiperlan) and it provides eleven 20MHz wide non-interfering channels, making it very flexible to mitigate interference.

2.7.1 Throughput

The throughput requirements for each base station have been specified as 49Mbps.
A microwave link throughput depends on the radio equipment technology , frequency band, distance, output power, antennae, cables and propagation scenario. Depending on the link budget corresponding to the installation scenario conditions, the radios operate in the modulation scheme that provides a specified throughput, with a specified packet loss, guaranteeing some specified availability.

QoS Parameter	Throughput
Interface	IPMWA, IPMWB (Figure 2-12)
Description	Net IP half-duplex bit rate (from IPMWA to IPMWB)
QoS Requirement	Throughput > 49 Mbit/s

2.7.2 Latency

QoS Parameter	IP packets latency
Interface	IPMWA, IPMWB (Figure 2-12)
Description	Maximum delay between consecutive packets To measure the latency of a high throughput IP connection, one can use a “ping” command, specifying a small number of bytes, e.g. 64 bytes.
QoS Requirement	Latency < 6msec

2.7.3 IP packet loss

QoS Parameter	IP packet loss
Interface	IPMWA, IPMWB
Description	Number of corrupted IP packets / number of all IP packets The IP Packet loss is a consequence of errors in the bit stream leading to IP packet errors resulting in packet discarding. On a correctly deployed microwave link errors occur due to propagation phenomena (fading, cintilation,etc) and in a much smaller extent due to resilient electronics impairments.
QoS Requirement	The packet loss at interface resulting from the link operation should no worst than $1*10^{-6}$ at the link throughput

2.7.4 Availability

QoS Parameter	Link availability
Interface	IPMWA, IPMWB
Description	The availability of a microwave link is specified for a fixed throughput and fixed BER. It states the percentage of time in which a link guarantees those

	specified throughput and BER. Voice telecommunications operators typically design their networks to guarantee one availability level, e.g. 99.999% corresponds to a 5min yearly outage time. A link can be designed for up to 100% availability.
QoS Requirement	99.995% availability for a minimum of 49Mbps Throughput, and 0,001% packet loss

3 Acronyms

BER	Bit Error Rate
C/(N+I)	Carrier to noise and interference ratio
C/N	Carrier to noise ratio
CIR	Committed Information rate
DCF	distributed coordination function
DVB	Digital Video Broadcasting
DVB-RCT	DVB Return Channel Terrestrial
DVB-T/H	DVB Terrestrial/ Handheld
EDCA	enhanced distributed channel access
HCCA	HCF controlled channel access
HCF	hybrid coordination function
HO	Handover
IP	Internet Protocol
kpbs	kilobit per second
Mbps	Megabit per second
MER	Modulation Error Ratio
MIR	Maximum Information Rate
PCF	point coordination function
QoS	Quality of Service
RF	Radio Frequency
RS	Reed-Solomon
TS	MPEG-2 Transport Stream
WM	wireless medium
DVQL-W	Digital Video Quality Level - Weighted
FER	Frame Error Rate
IPMWA	Internet Protocol MicroWave interface A

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