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Abstract

This document compiles the definitions of the performance metrics which are used to describe the performance (i.e. the operational quality) of the SUIT delivery system. It addresses the performance metrics related to the play-out, especially for the router/switch and the encapsulator, as well as for the gateway, terminal and the handover. It proposes some equipment and software packages to measure those metrics. This deliverable also gives examples from the first series of tests, despite most of tests will be described in D6.3-Test bed Report and D6.4-Field Trials Report.

Keyword list: Performance metrics, play-out, gateway, handover, terminal

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Performance metrics definition

SUIT_437

25 January 2008

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1 Introduction

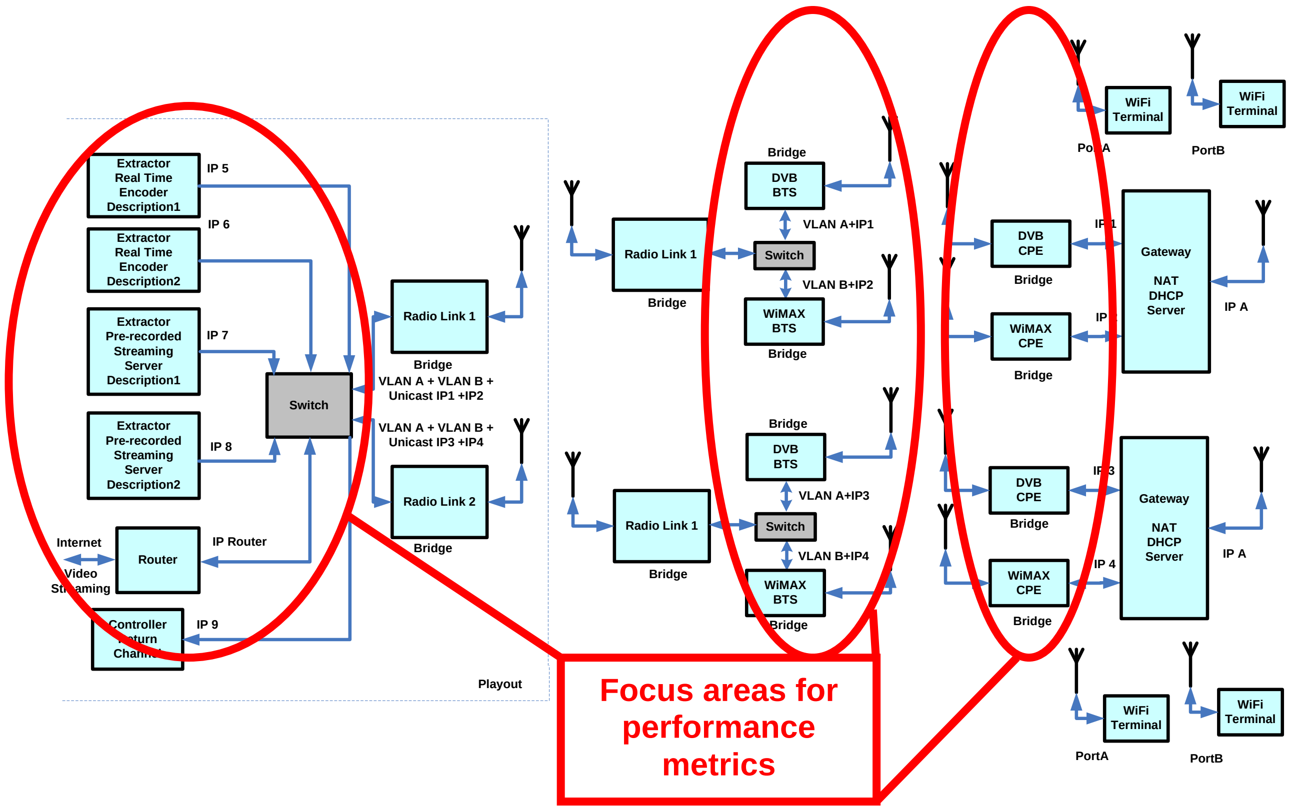

Figure 2.1-1 Suit platform

In the main section of this Deliverable 6.1, i.e. in section 2, the performance metrics for the three areas

- play-out,
- network (in particular during handover),
- gateway and
- terminal

are described. This includes a description of the modules necessary to provide quality related information as well as the metrics themselves.

2 Performance metrics

2.1 Play-out related performance metrics

In Figure 1 below, the SUIT playout functional blocks through which content passes and within which content changes, are detailed. These functional blocks can introduce undesirable changes to the content or its transportation might modify the overall system behaviour. In this section, some reference points to be measured, are identified.

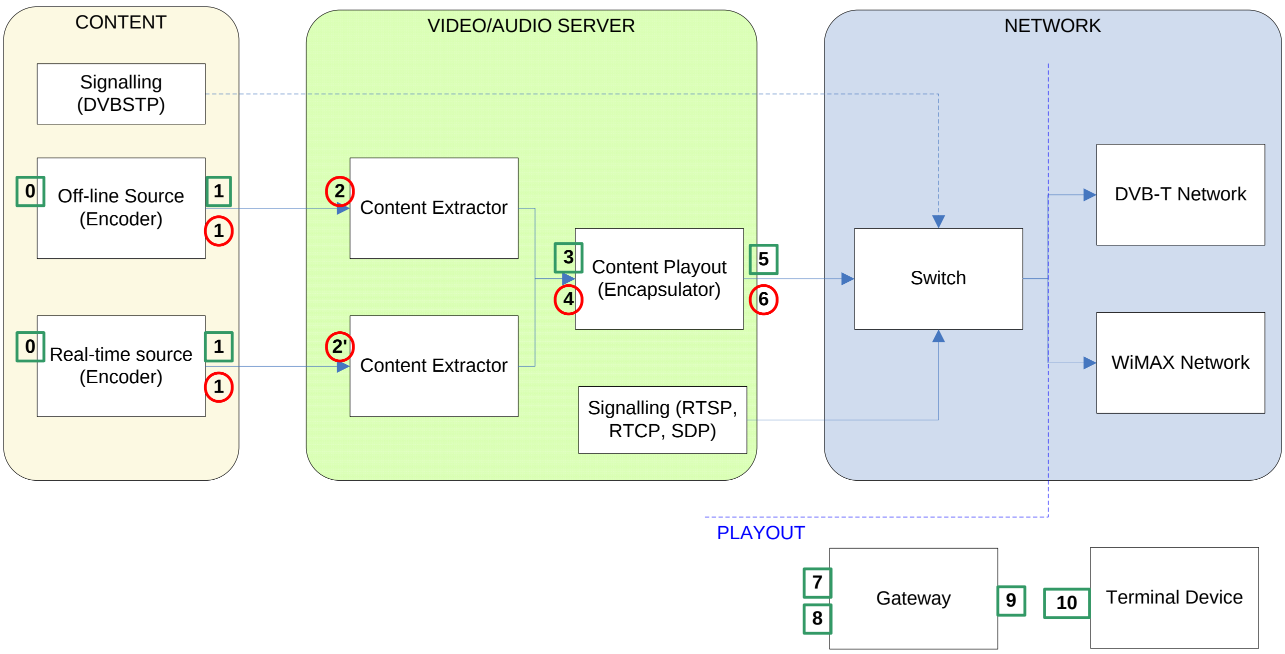

Figure 2.1-1 Functional blocks of the SUIT play-out

2.1.1 Content related performance metrics

The SUIT playout receives two types of video content:

- **Off-Line Source (Encoder):** A YUV video file is encoded in a H.264/SVC file and stored in the playout repository. The input **0** is the quality of the YUV video content and the output **1** is also the source quality for the video content encoded.
- **Real-Time Source (Encoder):** The acquisition from a real-time video is encoded in H.264/SVC video and delivered through the network interface. The input **0** is the quality of the incoming video and the output **1** is also the source quality for the video content encoded.

The quality of these sources will be the input to the Video Server Playout, Points **2**.

To measure the communication between point 1 and 2, we advise to use Wireshark or Ethereal running in the encoder and in the Extractor. This procedure allows you to measure any packet losses. Besides, given a stored file in the Off-line Encoder, send it to the Server at the expected bitrate and compare both files using for instance Hex Edit.

In addition, the SUIT playout will receive two types of audio content:

- **Off-Line Source (Encoder):** A raw audio file (e.g PCM) is encoded to an AAC file and stored in the playout repository. As for the video, the input **0** is the quality of the raw audio content and the output **1** is also the source quality for the encoded audio content.
- **Real-Source (Encoder):** The audio captured in real-time is encoded to AAC, and is delivered through the network interface. The input **0** is the quality of the incoming audio, and the output **1** is the source quality for the encoded audio content.

To measure the communication between points 1 and 2, we advise to use Wireshark or Ethereal running in the encoder and in the Extractor. This procedure allows you to measure any packet losses. Besides, given a stored file in the Off-line Encoder, send it to the Server at the expected bitrate and compare both files using for instance Hex Edit.

2.1.2 Video Server related performance metrics

The SUIT video server is LIVE-555 based and includes several modules, each one working in a specific way. We can distinguish these modules:

- **Extractor off-line:** This module extracts two descriptions from a video SVC file, and two descriptions from an AAC audio file. These descriptions are delivered to the Encapsulator module. The input **2** is the quality of the ingested content and the output **3** is also the source quality and transport rate for content encapsulating.
- **Extractor real-time:** This module extracts two descriptions from a SVC video and AAC audio, received through the network interface. These descriptions are delivered to the Encapsulator module. The input **2** is the quality of the received content and the output **3** is also the source quality and transport rate for content encapsulating.
- **Content Playout (Encapsulator):** This module receives the content from the extractor and encapsulates it into single RTP sessions. The output is the transport rate for the content delivery through network.
- **Signalling (RTSP, SDP):** This represents the authentication and signalling module. The measurement point represents the server end for the timing of client-server transactions.

To measure the communication between points 1 and 2, we advise to use Wireshark or Ethereal running in the encoder and in the Extractor. This procedure allows you to measure any packet losses. Besides, given a stored file in the Off-line Encoder, send it to the Server at the expected bitrate and compare both files using for instance Hex Edit.

2.2 Network

This domain covers all the network components required to transport the source contents to its destination, in our case to the base stations. It also includes the BSTs.

- **Switch/Router:** It provides connectivity between the Video Server and the base stations. This receives the content transport streams from one or more Video Servers and the signalling streams from the SD&S module.

To measure the communication between points either 5 (description 1) or 6 (description 2) and the Switch outputs, we advise to test whether both Switch VLAN inputs are working properly. To ensure that, follow usual ping procedures from the Server, point 5, to DVB-T BST and from the Server, point 6, to WiMAX BST. To be sure both VLAN are separated, exchange cables connectors between points 5 and 6 and repeat the ping procedures. In the alter case, it is expected malfunctioned. To test in multicast, use VLC server in the Server and a PC connected right after the switch.

- **BSTs:** To test the connections between the CPE to the BST, it is recommended to check the connection parameters in the Hyper Terminal. If the parameters of radio connection are correctly we can ping the BTS to the CPE using the Hyper Terminal tools.

2.2.1 Bandwidth

The bit rate in the radio channel in both directions, i.e. from the BTS to the CPE (DL) and vice-verse can be measured with the software IxChariot. The procedure is to connect the PC with the IxChariot to the BST and connect another PC with the application scripts to the CPE.

RF bandwidth can be measured by connecting a Spectrum Analyser to the BST output.

2.2.2 Radio Frequency

RF parameters like Carrier-to-Interference and Noise Ratio and Received Power can be measured in the CPE. In the DVB-T case, the Audemat receiver can be used.

The modulation and FEC can be monitorized through the specific application (MONGUI) of the RUNCOM BSTs

2.2.3 Packet jitter and Timestamp jitter

The interarrival jitter is an estimate of the statistical variance in network transit time for the data packets sent by the reporter synchronization source.

To calculate the variance in network transit time, it is necessary to measure the transit time. Because sender and receiver typically do not have synchronized clocks, however, it is not possible to measure the absolute transit time. Instead the relative transit time is calculated as the difference between a packet's RTP timestamp and the receiver's RTP clock at the time of arrival, measured in the same units. This calculation requires the receiver to maintain a clock for each source, running at the same nominal rate as the media clock for that source, from which to derive these relative timestamps. (This clock may be the receiver's local playout clock, if that runs at the same rate as the source clocks.) Because of the lack of synchronization between the clocks of sender and receiver, the relative transit time includes an unknown constant offset. This is not a problem, because we are interested only in the variation in transit time: the difference in spacing between two packets at the receiver versus the spacing when they left the sender. In the following computation the constant offset due to unsynchronized clocks is accounted for by the subtraction.

If S_i is the RTP timestamp from packet i, and R_i is the time of arrival in RTP timestamp units for packet i, then the relative transit time is $(R_i - S_i)$, and for two packets, i and j, the difference in relative transit time may be expressed as

$$D(i,j) = (R_j - S_j) - (R_i - S_i)$$

The interarrival jitter is calculated as each data packet is received, using the difference in relative transit times D(i,j) for that packet and the previous packet received (which is not necessarily the previous packet in sequence number order). The jitter is maintained as a moving average, according to the following formula:

$$J_i = J_{i-1} + \frac{(|D(i-1,i)| - J_{i-1})}{16}$$

To measure the Timestamp jitter, we can use the Wireshark or Ethereal running in the Gateway point 7 and 8. The procedure is calculate the time difference the capture two sequential packets and the jitter is a difference between this value and the timestamp difference.

To measure the Packet jitter we can use the Wireshark or Ethereal running in the Gateway point 7 and 8. It is the difference of time between receipt the same package of the two networks.

2.2.4 Packet loss (or drop)

It can be defined as the difference in the cumulative number of packets lost during an interval. The difference in the extended last sequence numbers gives the number of packets expected during the interval. The ratio of these values is the fraction of packets lost.

To measure the Packet loss, we advise to use Wireshark or Ethereal running in the Gateway points 7 and 8. In this software, we can see the Packet loss (or drop). Another method is use the IxChariot, this software can provide the packet loss in relation to the bit rate required in the complete LAN.

2.2.5 Packet format as in RFC3984

Another stream quality indicator is to ensure that the playout sends packets complain to SVC RFC (it is still a draft). The best way to verify the encapsulation process is to locally encapsulate a video stream, deencapsulate it and compare with the original. Packets size, in other words, NALUs size should be less than MTU, 1500 bytes. It means fragmentation/defragmentation is required.

2.2.6 Signalling Measurements (RTSP, SDP)

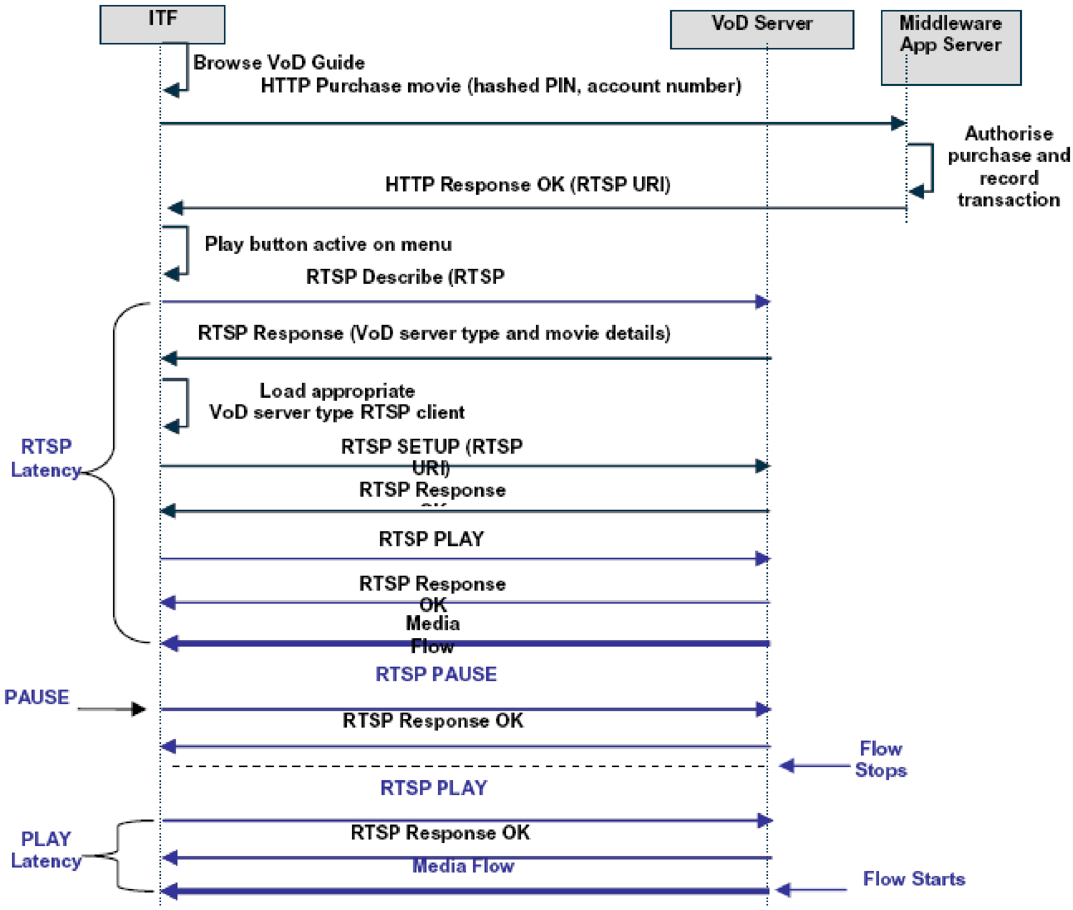

Figure 2.2-1 Signalling chart for the play-out

Given this trick features message flow, the following are examples of metrics that could support the Trick features Use Case.
Trick Latency: QoE metrics for VoD transaction quality are expressed by the following indicators:

♦ Video selection process delay: “Timing period from the time when the subject is selected to the time when content is displayed.”

♦ Play Delay: “Timing period from the time when the Play entry was selected to the time the content is displayed.”

♦ Stop Delay: “Timing period from the time when the Stop play video entry was selected to the time the content is stopped playing as indicated by video content display.”

♦ Rewind Delay: “Timing period from the time when the Rewind video entry was selected to the time the rewind action is executed as indicated on display device.”

♦ Pause Delay: “Timing period from the time when the Pause video entry was selected to the time the pause action is executed as indicated on display device.”

♦ FFW Delay: “Timing period from the time when the Fast Forward video entry was selected to the time the FFW action is executed as indicated on display device.”

To measure the signalling, we advise to use Wireshark or Ethereal running in the Playout. We can see in payload of the RTSP and SDP the content of the message and measure the latency for each RTSP message.

2.3 Gateway related performance metrics

2.3.1 RTP encapsulator/deencapsulator module

The performance metrics detailed above in the section for the Media Stream Quality Measurement can be also applied to the RTP encapsulator module build inside the Gateway.

To measure the communication between points 7-8 and 9, you should follow the procedure described in 2.2.5.

To measure the delay in the Gateway, we can use the Wireshark or Ethereal running twice (input and output) in the Gateway.

2.3.2 Combiner module

To evaluate the combiner, we can simply disconnect one input Ethernet cable alternatively, WiMAX and then DVB-T.

2.4 Terminal

Due to the way the content payload is encapsulated, and the nature of the changes that can be introduced by the function components, it is not practical to measure, payload, transport or network quality at every point within the model. The recommended points are included in the diagram as listed below.

2.4.1 Content Quality Measurement

For video:

Ref. Point	Description	Format	Potential Degradation	Measurements
0	Source Content Quality	YUV	Original Quality	Picture Quality
1	Output of Encoding	H.264/SVC	Coding artifacts	Content Quality
3	Output of Extracting	GOP,NAL units	Lost payload, Layers	Frame Loss, Layers

For audio:

Ref. Point	Description	Format	Potential Degradation	Measurements
0	Source Content Quality	PCM/WAV	Original Quality	Audio Quality, Sampling Frequency, Bits per Sample
1	Output of Encoding	AAC	Coding artifacts	Content Quality
3	Output of Extracting	LATM	Lost payload, Layers	Audio Frame Loss

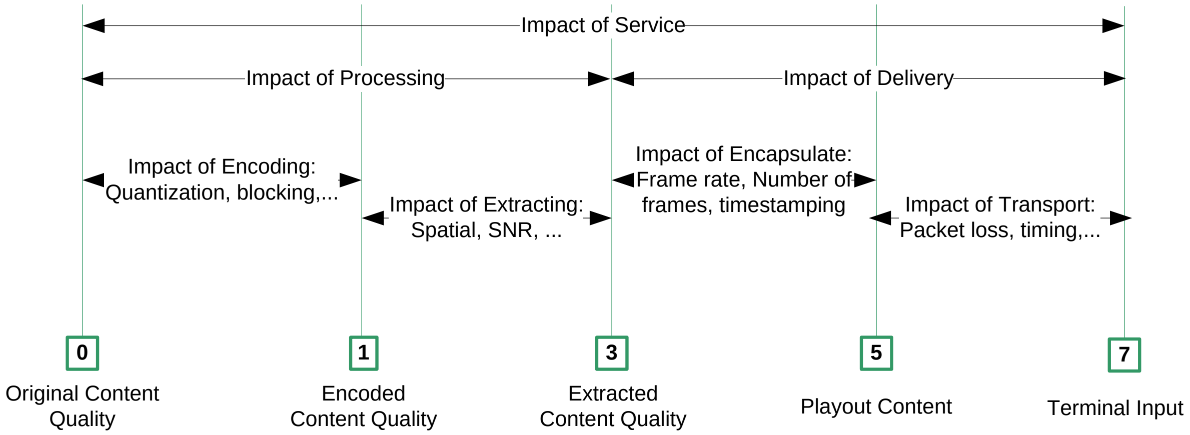

Figure 2.4-1 Impact of various stages in the delivery chain on quality from a content point of view

2.4.1.1 Video Quality

In literature, dozens of different algorithms for objectively measuring the quality of a video are described. They vary in computational complexity, correlation with subjective video quality measurement, and accessibility.

The Video Quality Experts Group (VQEG), containing experts from ITU-T study groups 9 and 12, is currently working on a standardized model for objective video quality measurement. They plan on finishing their work regarding a high definition model by the end of 2008. Until then, the common available and accepted metrics will have to be used for measurements. For the work in the SUIT project, all measurements are done using either the structural similarity or the peak signal-to-noise ratio method.

For the SUIT project, all measuring is done offline using a full reference model. This means that both the original video, and the processed video, that is, the one which was encoded, transmitted and decoded again, are compared. Three different scenarios are considered in the SUIT project: both descriptions are received correctly, only one description is received correctly, or no descriptions are received.

For real time video quality measures measurements, we propose to use Tektronix PQA300 or R&S DVQ

2.4.1.2 Structural Similarity

The Structural SIMularity (SSIM) is an objective video quality metric based on the idea that the human vision system is highly specialized in extracting structural information from the viewing field and not in extracting errors.

The SSIM index is calculated as follows:

$$SSIM = \frac{\left(2\overline{x}\overline{y} + C_1\right)\left(2\sigma_{xy} + C_2\right)}{\left(\overline{x}^2 + \overline{y}^2 + C_1\right)\left(\sigma_x^2 + \sigma_y^2 + C_2\right)}$$

with $\overline{x}$ being the mean of x, $\overline{y}$ being the mean of y, σ_x being the variance of x, σ_y being the variance of y, and σ_{xy} being the covariance of x and y. The function returns a decimal value between 0 and 1. 0 would mean zero correlation with the original image, and 1 means the exact same image. 0.95 SSIM, for example, would imply half as much variation from the original image as 0.90 SSIM.

The best way to assess the subjective video quality is to watch the video on screen with enough resolution.

2.4.1.3 Peak Signal-to-Noise Ratio

The Peak Signal-to-Noise Ratio, PSNR for short, is currently the most used objective measurement method for analyzing a video stream. It is the ratio between the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation. It is defined via the mean square error (MSE). For two $m \times n$ monochrome images K and L where one of the images is considered a noisy approximation of the other, the MSE is defined as:

$$MSE = \frac{1}{mn} \sum_{i=0}^{m-1} \sum_{j=0}^{n-1} \| K(i,j) - L(i,j) \|^2$$

The PSNR is then defined as:

$$PSNR = 10.\log_{10}\left(\frac{MAX_K^2}{MSE} \right) = 20.\log_{10}\left(\frac{MAX_K}{\sqrt{MSE}} \right)$$

with MAX being the maximum pixel value. For an image with 8 bit color channels (the most commonly used ones), this is 255.

The use of PSNR over SSIM is often preferred due to the high computational complexity of the latter.

For the SUIT project, IBBT delivered a command line tool which accepts both the original and processed video sequence and returns both the SSIM index and the PSNR of the luma and chroma components.

Measurements obtained by the tool can be found in several other deliverables.

2.4.1.4 Audio Quality

There are several issues that must be considered when measuring the audio quality provided by the SUIT system:

1) **Test material**
   Audio encloses a substantial range of different types of signal. The impacts of the employed coding algorithms and the transmission channel errors can differ according to the type of the audio signal. Therefore, it is important to consider a wide and commonly heard range of test sequences during audio quality assessment tests. EBU test sequences [3] that include a wide range of material, from concert music to speech have been chosen as the test material for the SUIT field trials.

2) **Quality Assessment Methods:**
   a. Subjective methods: MUSHRA (MUltiple Stimuli with Hidden Reference and Anchor), MOS (Mean Opinion Score) etc.
   b. Objective methods: SNR (Signal to Noise Ratio), PEAQ (Perceptual Evaluation of Audio Quality), PESQ (Perceptual Evaluation of Speech Quality), etc.

   Subjective tests generally necessitate much man power and effort. Since there is limited time, objective testing will be employed. Although it is mathematically formulated, PEAQ

[4] utilises psychoacoustical properties of the audio signals when comparing it the reference and resulting signal. PESQ [5] is similar to PEAQ and specifically designed for measuring speech quality. They are widely used in the literature and will be employed in this project.

3) **Joint vs separate evaluation with video**
Joint evaluation of audio and video implies subjective testing. As subjective testing is not being performed, only separate testing will be used for audio and video.

2.4.2 Media Stream Quality Measurement

Ref. Point	Description	Format	Potential Degradation	Measurements
1	Source Content Quality	H.264/SVC	Original Source	Frame check, Timestamping
2	Input of Extractor	File transfer	Read/Write process	Frame check, Timestamp
2'	Input of Extractor	UDP packets	Transport process	Frame check, Timestamp
4	Output of Extractor	GOP, NAL units	Transcoding process	Timestamp, jitter
6	Output of Delivery	Video streaming	Encapsulating process	Timestamp, jitter

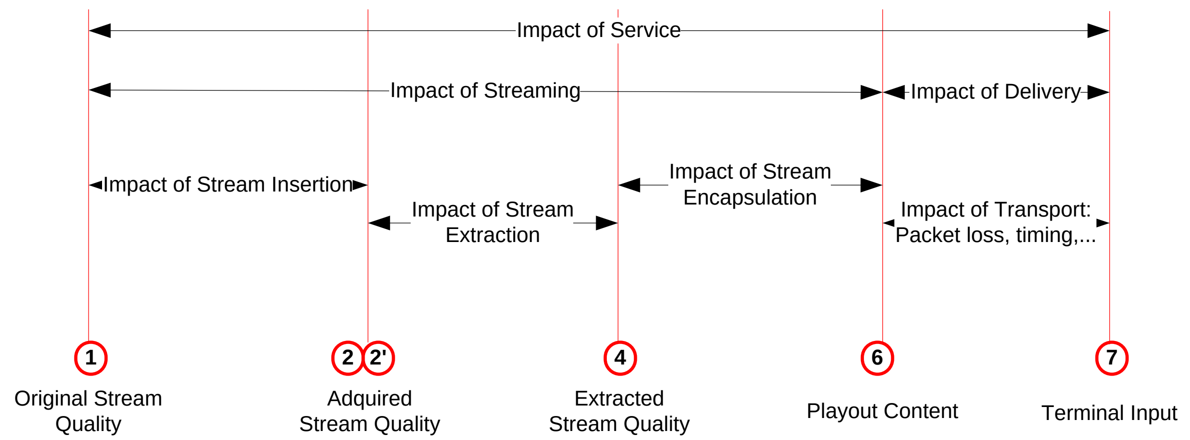

Figure 2.4-2 Impact of various stages in the delivery chain on quality from a streaming point of view

Specific test stream formats and content are unlimited, but some specific metrics which might be manipulated include:

2.5 Performance metrics for handover scenarios

2.5.1 Handover test set-up

The principle test set-up for the handover tests is described in Deliverable D4.3 [2]. Some modifications were made to arrive at a more general structure of the testbed where some components can be easily exchanged. That is the reason why now two DVB-T/H receivers are used. This allows the independent retrieval of physical layer and link layer parameter values such as signal level, MER and Transport Stream errors via the polling of the respective SNMP MIB.

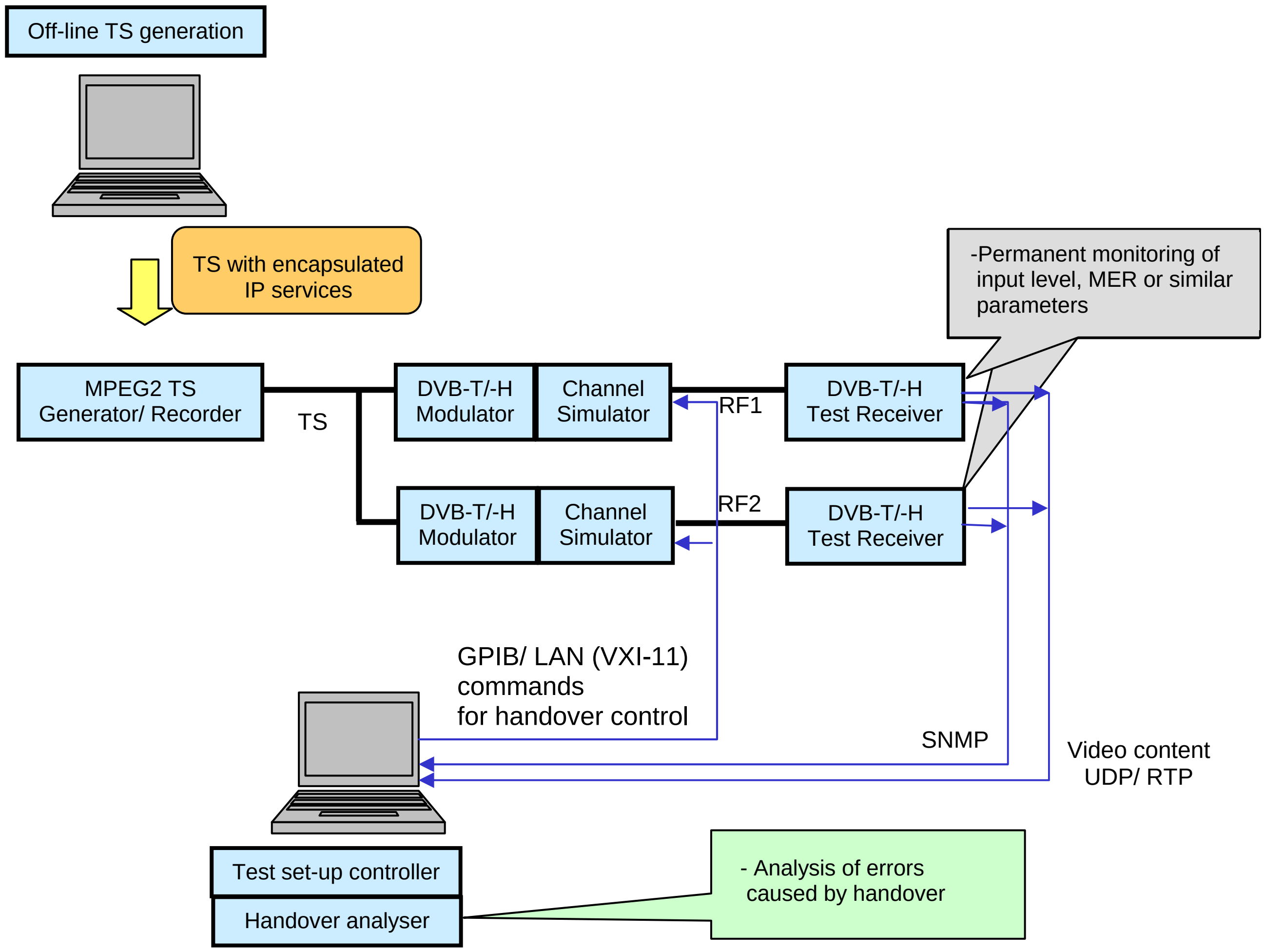

Figure 2.5-1 Testbed for handover tests

2.5.2 Handover analyser

For the purpose of the tests in SUIT, a new set of tools was assembled to analyse in detail the various influences on the perceived video quality. Some of these tools were newly developed, some were modified and some others already existed in a usable form.

The core component that was developed in the framework of the project, is the handover analyser which is usable for all types of handover because it processes the incoming IP data streams from both receivers may they both be DVB-T/H receivers or of a different type.
This section provides a short walk-through of the handover analyser and describes briefly its building blocks.
The handover analyser comprises the following modules:

- UDP RX module
- Input switch module
- Modification module
- Output switch module
- UDP TX module
- SNMP RX module
- Handover control module
- Log and monitoring module

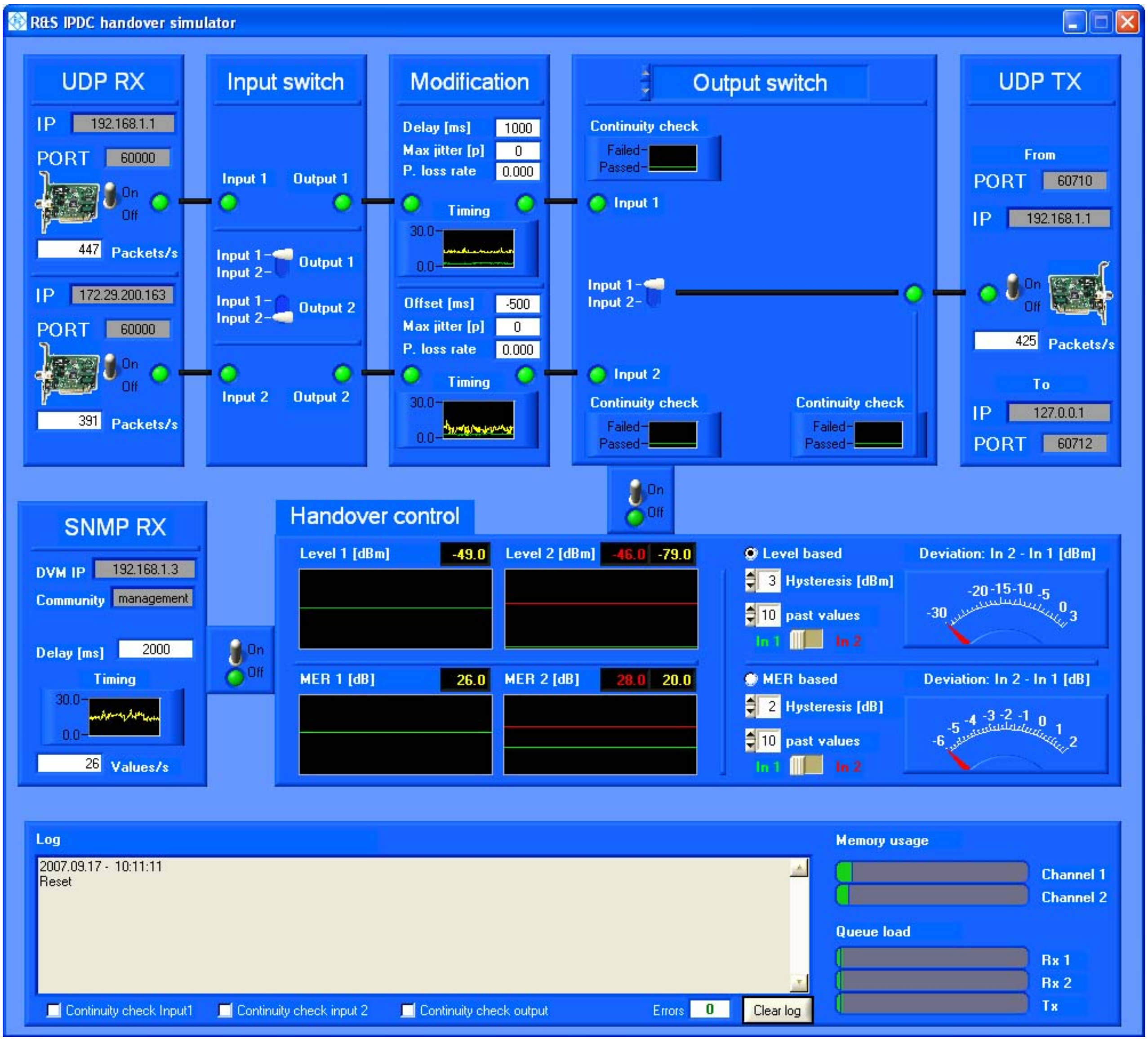

Figure 2.5-2 Handover analyser user interface

The HO analyser tool allows in principle for the selection of different QoS parameters to initialise handover. First validation tests have shown that low layer parameters such a signal level and MER

(Modulation Error Ratio) are best suited to trigger handovers. This is in line with the "IP datacast over DVB-H: Implementation Guidelines for Mobility" [11] where it is stated that e.g. BER is not suitable in mobile environment due to the rapid changes of the reception conditions and the long intervals for which BER has to be measured to arrive at a reasonably accurate value.
The handover requires a permanent reception/ probing of the two input signals during the period that is relevant for the handover. The serive identification via the respective tables carrying txxxhe Service Information (SI) and Program Specific Information (PSI) is not part of these tests. Here the testbed is set up in such a way that both transport streams carry the same service. This is achieved by feeding both modulators with the same Transport Stream.

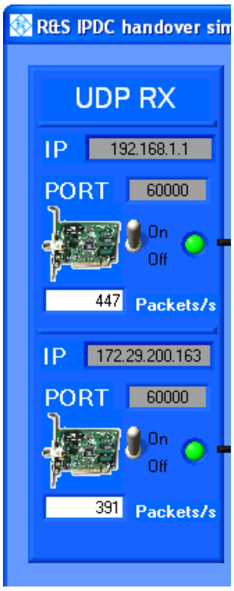

Figure 2.5-3 UDP RX part of the HO analyser user interface

The two test receivers are used to receive both multiplexes on different RF frequencies. They forward the extracted IP packets which were encapsulated in the TS according to the MPE (multi-protocol encapsulation) standard [12] directly to the HO analyser tool.
The HO analyser tool is resident on a separate PC which receives the two independent UDP streams on its UDP RX module (Figure 2.5-3) which provides two internal channels (1 and 2). Each channel is associated with a certain user-defined IP addresse and port number. At both inputs the received IP packets are counted and the received packet rates are displayed.

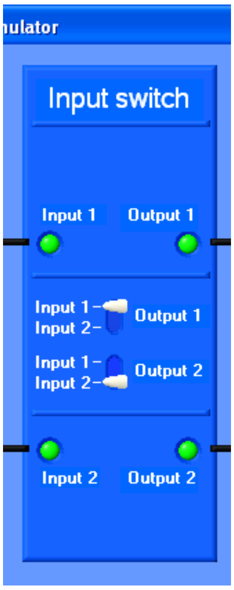

Figure 2.5-4 Input switch of the HO analyser user interface

The input switch provides a matrix where the two inputs can be connected to the two ouputs. It is also possible to connect the same input, e.g. Input 1 to both outputs.

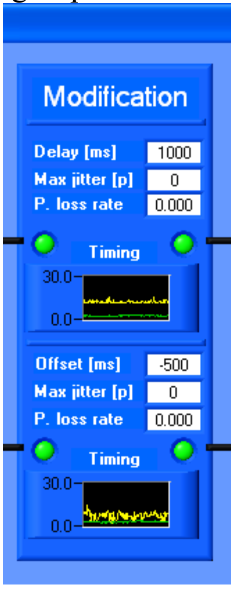

Figure 2.5-5 Modification part of the HO analyser user interface

The Modification tool is an option that is very useful for laboratory tests. It allows the simulation of relevant network impairments by defining an additional delay/ time shift, a scalable packet jitter and packet loss rates.

Delay means the absolute delay of Ch 1, Offset defines the delay of Ch 2 relative to Ch 2 and the Packet loss rate can be set independently for both channels. The Timing windows enable a continuous visual monitoring of the packet delay during the tests.

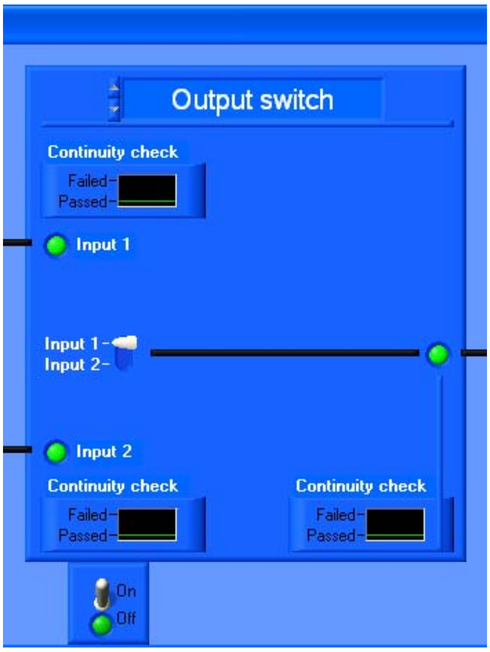

Figure 2.5-6 Output switch of the HO analyser user interface

The module Output Switch provides continuity checks for the two input signals and for the output signal. It also indicates which input (1 or 2) is connected to the output. The continuity checks identify discontinuities in both input streams and in the output stream based on the analysis of the RTP sequence number.
The position of the switch indicates the usage of signal from Input 1.

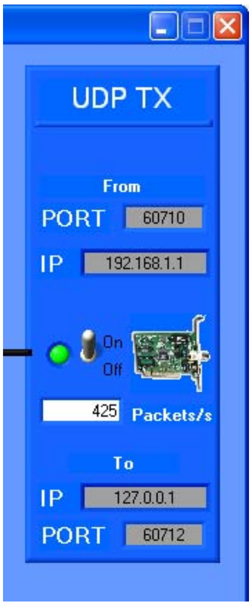

Figure 2.5-7 UDP Tx part of the HO analyser user interface

The UDP Tx part forwards the IP data stream to its destination IP address and port number. In the test set-up this is a separate PC on which the video quality monitor tool is resident. The UDP Tx part also displays a count of the output packets.

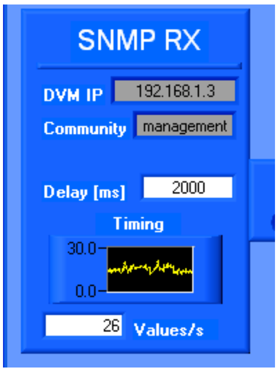

Figure 2.5-8 SNMP Rx part of the HO analyser user interface

The SNMP Rx module of the handover analyser receives from the DVB-T/H test receivers periodically the values of input level and MER for both channels. These values are forwarded to the Handover Control module where the decision for handover is taken and executed.

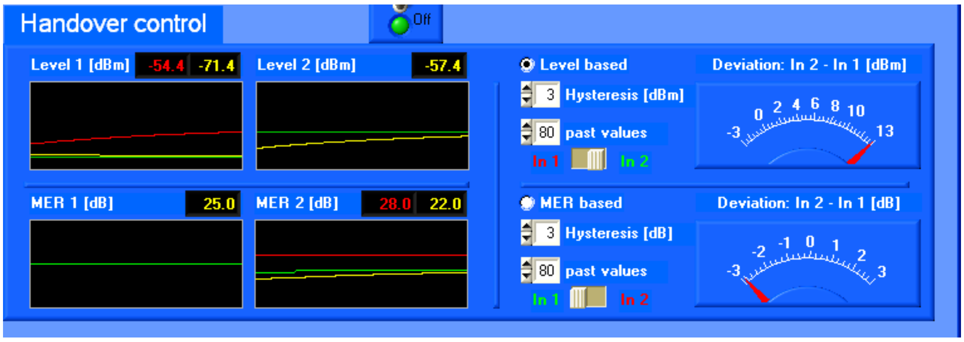

Figure 2.5-9 Handover Control part of the HO analyser user interface

The Handover Control module can be configured to use either the signal level or the MER value of the respective input channel as a relevant parameter for the initiation of a handover (MER is equivalent to SNR as long as no impairments from the original modulation are present in the signal).

The HO algorithms are designed to avoid a ping-pong effect during the handover phase and contain therefore an averaging function and a hysteresis.

The Handover Control panel provides an overview of the variation of the parameters signal level and MER during the handover phase. The green curve indicates the current value of the parameter, the yellow curve gives the averaged value and the red curve displays the threshold for set for the handover.

The instrument panel gives some information on the difference between the two channels and the handover decision related parameters. The positioning of the switch in the field for the parameter that triggers the handover, gives the information on which channel has been selected and is forwarded to the output. In the example in Figure 2.5-9 the result of the handover process (based on the evaluation of the signal level) is the switch-over to Input 2 during handover.

Figure 2.5-10 Log file of the HO analyser user interface

The Log file contains the most relevant information of each handover: date and time of the test, the parameter that is used to initiate the handover, and the information from which to which input the switch-over took place.
The recording of the results of the continuity checks (on both inputs and the output) can be activated, and the errors during handover are also registered.

2.5.3 Test results of handovers using the HO analyser

The most important issue for the end user is that the quality of the video and audio is not noticably impacted by the handover. To avoid subjective assessment of such quality variations, an objective measurement tool for the video quality is used. The focus is on video because it occupies a relatively large bandwidth and is more susceptible to impairments based on packet loss.
This objective measurement tool analyses the impairments in each video frame of H.264 encoded video material during the decoding process. It translates the identified impairments onto a quality scale from 0 to 100 where 0 is the worst and 100 the best quality value.
In its current version the quality monitor also analyses the spatial activity (SA) and the temporal activity (TA) in the video. In Figure 2.5-11 below the peak in the curve of spatial activity indicates a typical scene cut.

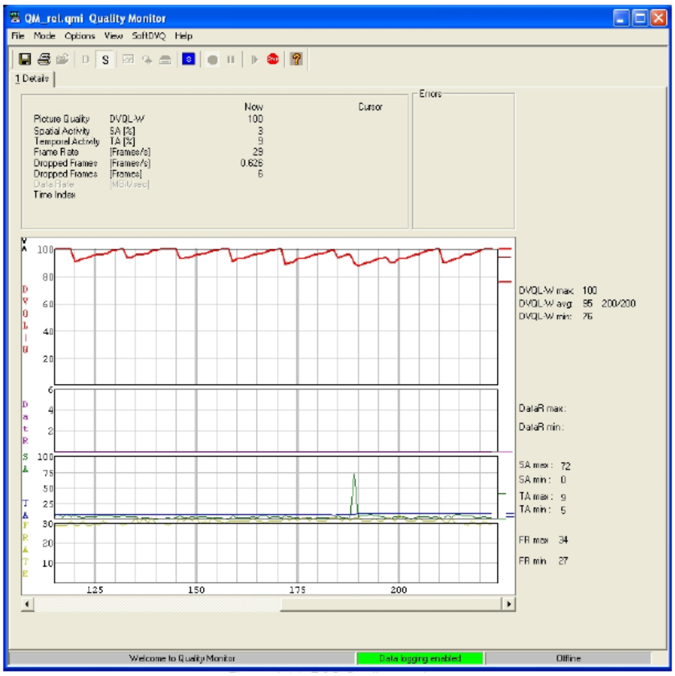

Figure 2.5-11 User interface of the Quality Monitor

The Quality Monitor provides a measurement value of the perceived video quality for each frame or an averaged value for a given number of frames. The numerical value for the video quality ranges from 0 to 100, where values between 0 and 20 indicate poor picture quality and values between 80 and 100 point to excellent picture quality.

The saw tooth structure in the example in Figure 2.5-11 stems from the GOP structure of the video. The I-frames at the beginning of each GOP require a higher momentary bitrate than is available. As a consequence, the measured picture quality of the I-frame drops slightly and recovers to its original value during the consecutive frames.

The test sequences used for the handover tests show an original video quality in the range of 70 to 100. The loss of packets (Transport Stream packets and/ or IP packets) during the handover procedure results in a measurable decrease of the video quality. Minor distortions (down to a measured value of 60) or hardly noticable.
The quality metric for the handover impact on video quality is therefore based on the units the measured video quality drops below the threshold of 60.

The figure below (Figure 2.5-12) shows the measured results of a typical handover.
The light blue and dark blue curves describe the decreasing and increasing level of the signals from the two transmitters (as it is the case if a mobile receiver e.g. in a car is moved from one coverage cell to the next). The orange and red curves give the corresponding TS packet loss rates. The yellow curve indicates the impact on video quality. The maximum value of the difference between the actually measured video quality and the threshold of 60 (as mentioned above) is normalised to 100.
The intelligent buffer management alone reduces the impact on the perceived video quality by about 90 % (dark green curve). If the buffer management functionalities are extended to include the combination of undistorted packets from both signals for the period of the handover (comparable to diversity reception for this period), the impact of the handover on video quality is below the threshold of detectability.

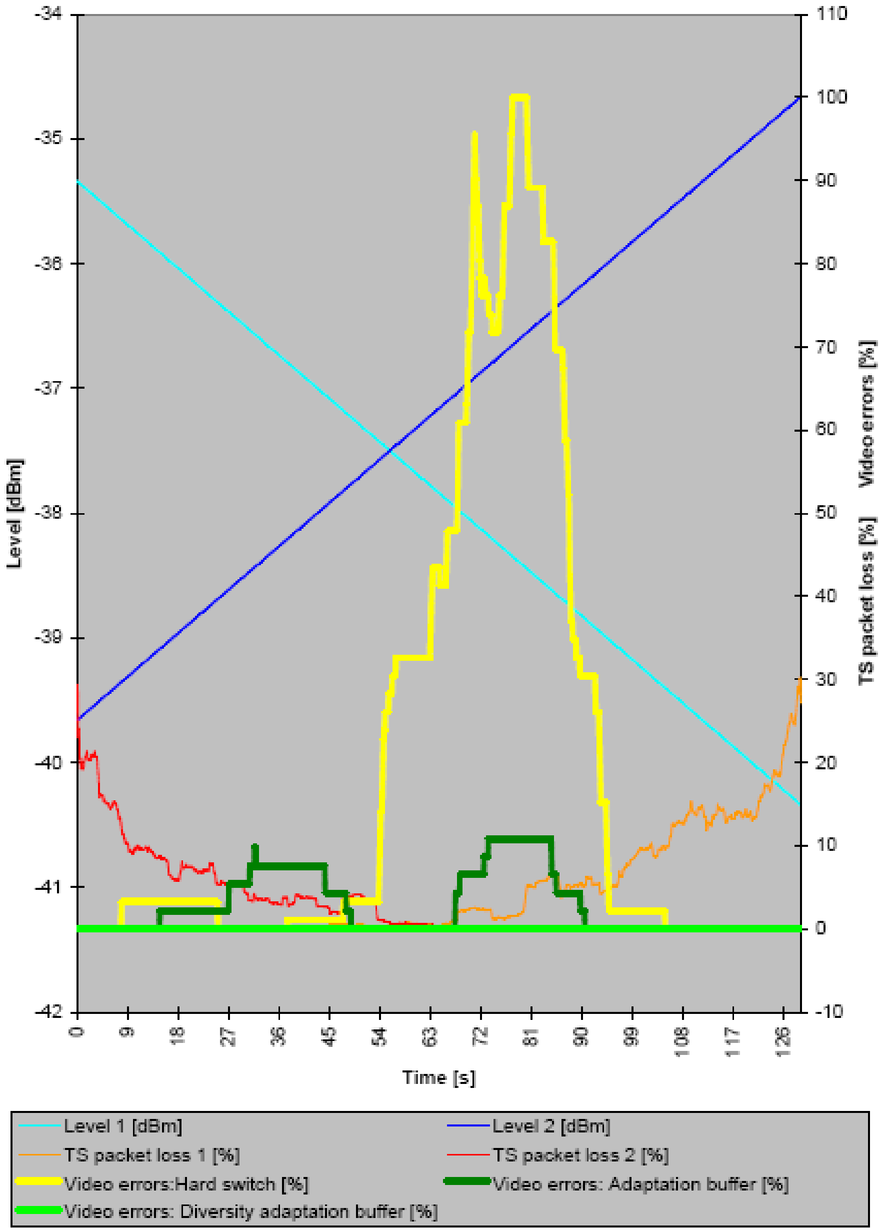

Figure 2.5-12 Example of handover test results

2.5.4 Conclusion

This document show all points of interest for measuring and thus evaluation the quality of the communication system proposed in SUIT.
Concerning handover, its initiation is based on input signal level or SNR (MER) because these low layer parameters are available from the receiver frontends without demodulation of the signal and therefore without considerable delay.
As an objective metric of the impact of handovers on the signal quality at the level of the IP data stream, a count of packet loss is used. For the testing of the impact that handovers have on the perceived quality, a software tool is deployed that measures objectively the equivalent of the perceived video quality.
The strategy for handovers can be extended by intelligent buffer management in such a way that for standard scenarios, a visual distortion is hardly detectable.

3 Acronyms

BER	Bit Error Rate
DVB	Digital Video Broadcasting
DVB-T/H	DVB Terrestrial/ Handheld
HO	Handover
IP	Internet Protocol
kpbs	kilobit per second
Mbps	Megabit per second
MER	Modulation Error Ratio
QoS	Quality of Service
RF	Radio Frequency
TS	MPEG-2 Transport Stream

4 References

[1] ETSI TR 101 290 V1.2.1 (2001-05) Digital Video Broadcasting (DVB); Measurement guidelines for DVB systems

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[3] http://sound.media.mit.edu/mpeg4/audio/sqam/

[4] http://www.peaq.org/

[5] http://www.pesq.org/

[6] A One-way Delay Metric for IPPM (RFC 2679)

[7] A One-way Packet Loss Metric for IPPM (RFC 2680)

[8] A Round-trip Delay Metric for IPPM (RFC 2681)

[9] Packet Delay Variation Metric for IPPM (RFC 3393)

[10] IP Performance Metrics (IPPM) metrics registry (RFC 4148)

[11] ETSI: TS 102 611 IP Datacast over DVB-H: Implementation Guidelines for Mobility

[12] ETSI EN 301 192 DVB specification for data broadcasting