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Abstract

This document describes the QoS management strategies for the SUIT system. It addresses the QoS requirements as laid down in D1.2 and details how they are translated into procedures for the resource management of the play-out, especially for the router/ switch and the encapsulator. It also describes the implementation of QoS parameter evaluation for handover initiation.

Keyword list: QoS management, resource management, handover.

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QoS Management and Handover

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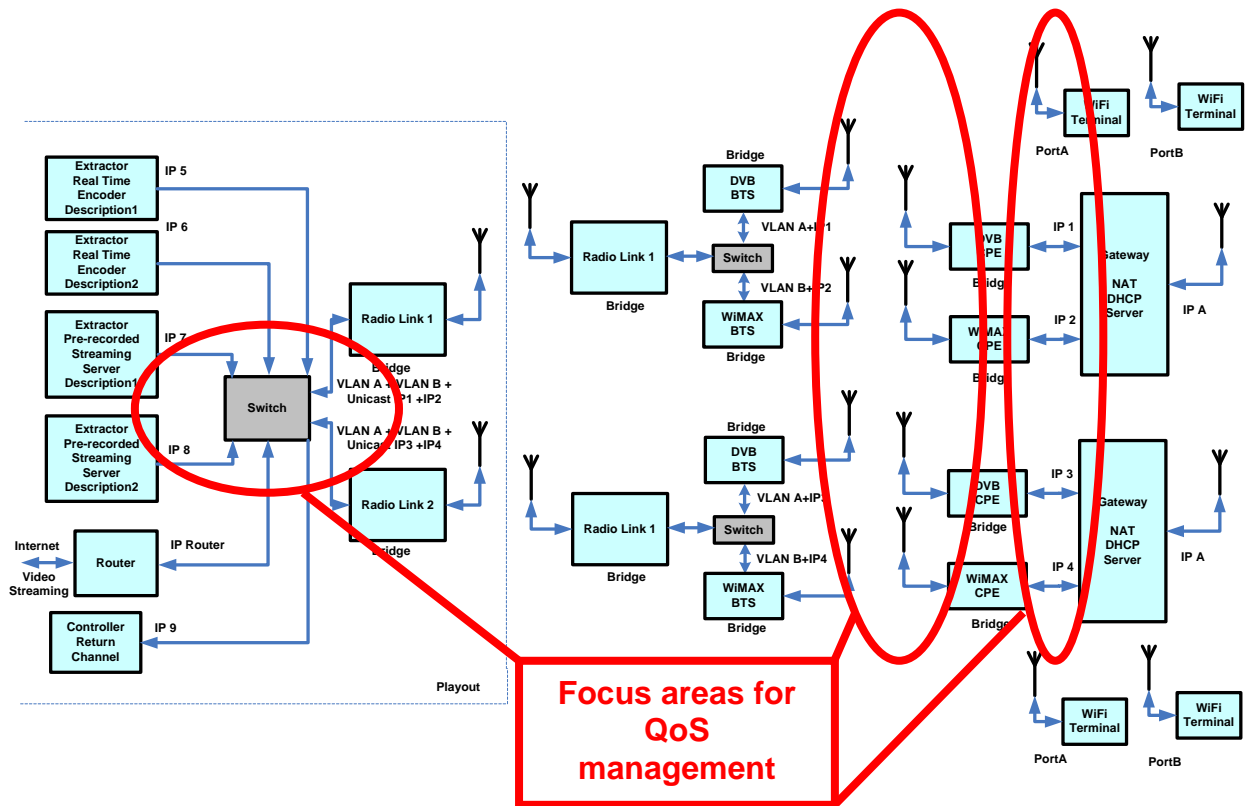# Table of Contents

1 Introduction

Figure 1-1: Suit platform

The architecture of the SUIT platform is shown in Figure 1-1.

This deliverable D4.3 "QoS management and handover" provides an overview of the QoS management issues related to the play-out, the QoS dependent handover strategies in DVB-T/H systems and the WiMax mobile handover mechanisms as they are currently worked on in SUIT.

2 QoS management

2.1 QoS management strategies for the play-out

The SUIT project aims at providing QoS, in terms of bandwidth and delay, for the end-to-end services. According to the SUIT architecture the play-out has to preserve the QoS in two focus areas:

- at the IP router/switch (from the video server to the DVB-T/H & WIMAX encapsulators input) and

- at the DVB-T/H & WIMAX interfaces (from the encapsulators input to the de-encapsulators output).

In the following sections these two focus areas are detailed.

2.1.1 Switch/ router related issues

Several techniques can be used to satisfy QoS requirements at different levels: at layer-2, taking advantage of QoS provided by the wireless channels (if any); at IP level if the version of the IP supports it; and at higher levels using specific protocols or mechanisms such as 'Diffserv'.

Some applications (e.g. real-time ones) need a guaranteed quality of service, in terms of percentage of packet loss, jitter, delay, etc. Depending on the network characteristics, several approaches to solve this problem are suitable. If enough bandwidth is available, the most practical solution is over-dimensioning resources reserved for such applications. Otherwise, different techniques at IP level have been proposed in Deliverable 1.3 [1]:

- **Multiprotocol Label Switching (MPLS)**: as no routers are present in the play-out because the RTP encapsulator will deliver the packets directly through the VLANs (defined by the switch module), this protocol is not used.

- **Integrated Services (IntServ)**: there are no routers in the current SUIT architecture where IntServ could be implemented.

- **Differentiated Services (DiffServ):** DiffServ demands packet priorities to be set up at network edge routers. As this causes an inevitable increase in end-to-end delay, this method will not be used in SUIT.

According to Deliverable 1.2 [2] the play-out management should take into account several parameters at IP level:

- *Service bit rate*: the play-out will deal with this parameter, according to the available bandwidth, network characteristics and terminal capabilities.
- *IP packet error rate*: the RTP encapsulator will deal with these parameters in the streaming scenario.
- *IP Packet Jitter*: the RTP encapsulator will deal with these parameters in the streaming scenario.

The play-out is intended to deliver all the services configured in the play-out manager and watch over the whole SUIT architecture to ensure the correct content delivery to the consumers. In order to do this, the play-out system takes advantage of the information retrieved from the terminals and the different network equipment in order to apply QoS algorithms acting over the network layer at the interface IP1 (defined in Deliverable 1.2 [2]) and video coding, to make much better use of the network and providing a better experience to the end-user.

The management of different types of traffic: huge amounts of traffic from different types (non-real time, real time…) must be managed efficiently, providing also several grades of QoS using different

QoS management strategies defined by SUIT, with regard to the location on the architecture and the scenario where it has to be applied.

The play-out includes two resident Web Services (WS) modules. Both WS acquire information about network conditions, analyse them and take decisions for controlling the different play-out modules using QoS algorithms. These modules are called Adaptation Decision Taking Engine web service (ADTE WS) and Intelligent Unit web service (IU WS).
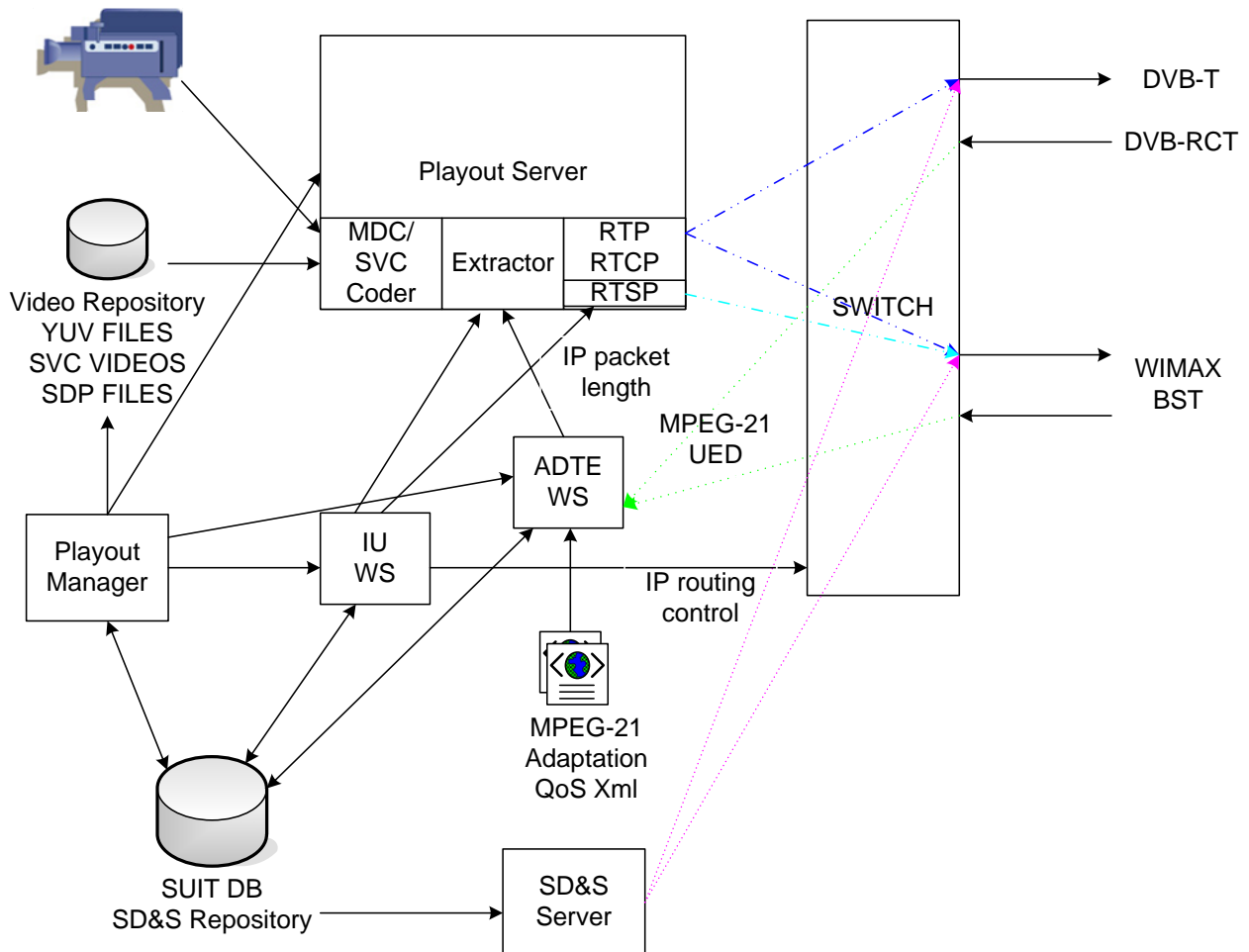

Figure 2-1: Play-out architecture

The play-out (Figure 2-1) has the knowledge of the total amount of bit rates to be delivered to each platform. The finality of this strategy consists of an optimally distribution of the sessions towards each platform, taking into account network characteristics such as available bandwidth, jitter, content priority, etc. The IU web service will retrieve these parameters from the SUIT database, triggered by an alarm message generated by ADTE WS, and will decide the optimal content distribution of the unicast content.

According to the SUIT architecture, the two descriptions delivered over the network already have a predefined path through the two VLANs defined in the switch. Each description is always sent over the same VLAN, so the IP router only redistributes the additional contents, to be delivered to the most capable network.

In the next sections the two play-out web services are briefly described, though these modules will be explained in more detail in Deliverables 3.3, 4.4 and 4.5.

2.1.1.1 Adaptation Decision Taking Engine web service: (ADTE WS)

- **Unicast Scenarios**: the return channel capability offered by the last-mile networks is exploited in order to create an intelligent play-out system. Feedback information about the terminal and the network can be used to adapt the unicast video stream, and make it more suitable for the terminal in question, or to avoid network congestion.

In the return channel between gateway and play-out, tools from the MPEG-21 Digital Item Adaptation (DIA) suite will be used. The return channel work flow can be broken down into these parts:

1. Feedback information, such as device information and network statistics, are collected by the terminal.
2. This feedback information is gathered in a standardized format, so that it can be understood by the play-out. In SUIT, this format will conform to the MPEG-21 DIA-UED (Usage Environment Description) specification.
3. The document tailored in steps 1 and 2 is periodically sent to the play-out via the return channel. In the SUIT demonstrator, the play-out will offer a Web Service to which clients can connect and upload their DIA-UED documents.
4. In the play-out, the received document is parsed and the feedback information is extracted. For this task, an MPEG-21 DIA parser is employed (Figure 2-2).
5. Using MPEG-21 AdaptationQoS, the feedback information is converted to the parameters needed to drive the bit stream extractor.

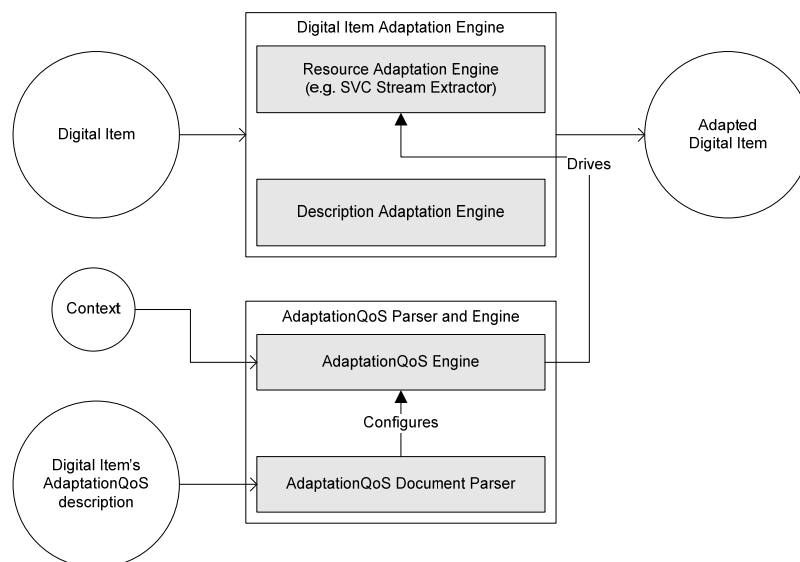*Figure 2-2: Digital Item Adaptation parser and engine*

These protocols are well defined in Deliverable 4.2 [3] chapter 4.

In the SUIT project, the following AdaptationQoS implementation is proposed. The bit stream extractor is driven by three parameters: temporalLevels and spatialLayers, indicating the number of entire layers to be removed; and qualityReduction, indicating the amount of FGS information to

truncate. In unicast scenarios, the input parameters to the AdaptationQoS engine are extracted from the Usage Environment parameters that are supplied by the terminal; these are:

- The average available bandwidth;
- The maximum horizontal display resolution;
- The maximum vertical display resolution.

- **Broadcast Scenarios:** a single parameter will be used: the target bit rate. In this case, this parameter is generated by the play-out, which estimates the bit rate to be allocated to a stream based on the amount of streams to be broadcast and the amount of other information (internet, hyperlinked video,…) that has to be sent alongside the regular broadcasts.

A single *UtilityFunction* module will be sufficient to map these input parameters to the three adaptation operators that drive the bit stream extractor. This UtilityFunction module will be set up so that:

1. (Unicast only:) For defining *spatialLayers*, it gives precedence to the maximum display resolution. This way, the adapted bit stream's resolution is never greater than the maximum resolution the terminal can display;
2. Then temporal and/or more spatial layers are removed based on the available bandwidth. Which layers are to be removed first is expressed in the AdaptationQos file.
3. Finally, *qualityReduction* is calculated in order to fill the remainder of the available bandwidth.

For the SUIT demonstrator, the MPEG-21 parser and AdaptationQoS engine will be integrated in a Java web service, using a SOAP RPC front end. In the unicast scenarios, terminals can periodically submit their UED document by invoking a SOAP remote procedure call on the web service. In the broadcast scenario, an RPC method will be provided through which other parts of the gateway can set the target bit rate. Inside the play-out, an inter-process communication (IPC) channel is required between the web service and the bit stream extractors it has under its control. For this purpose, a simple UDP or TCP based message system will be employed.

2.1.1.2 Intelligent Unit web service: (IU WS)

This module works as a resident web service waiting for commands from the Play-out Manager and update triggers from the ADTE WS. The IU web service basically applies some of the following methods, to provide the expected QoS:

- Control of the RTP Encapsulator
    - Modification of IP packet length
    - Aggregation and fragmentation packets
    - Repetition of SPS and PPS NALs.
    - Different protection of NAL units depending of their importance.
    - Dynamic frame-rate

- Switch IP routing control
    - Decide which is the best network where deliver additional content, such as Internet.

The different techniques will be applied according to the scenario:

-Unicast Scenario

Feedback information about the terminal and the network can be used to adapt the unicast video stream, and make it more suitable for the terminal, or to avoid network congestion.
MPEG-21-DIA information will be stored by the ADTE WS in the SUIT database, then the IU WS will access the database in order to analyse the feedback information.
The required information extracted from MPEG-21 will be:

- *Network capabilities*. It defines a network's static attributes, such as the ***maximum capacity*** of a network and the ***minimum guaranteed bandwidth***.
- *Network conditions*. It describes network parameters that tend to be more dynamic and time varying, such as the available bandwidth, error and ***delay*** characteristics. The error is specified in terms of ***packet loss and bit error rates***. Delay types include one-way and two way packet delay, as well as delay variation.

RTCP packets could be used instead of MPEG-21 because they contain the necessary information for quality of service monitoring.

According to these parameters the IU WS will perform the defined actions:

- *Modification of IP packet length*

This method consists of selecting the optimal packetization size for the RTP encapsulation according to the networks characteristics and the transport technology used in the last-mile. It will allow optimization and error resilience. Some algorithms will be applied to calculate the optimal MTU for each session, with regard to the network that will consume its content. The IP packet length will be set according the network channel models simulations described in WP2.

- *Aggregation and fragmentation packets*

This method consists of selecting the optimal packetization mode for the RTP encapsulation according to the networks characteristics.
Single NAL unit packets will be used when the NAL information plus RTP, UDP, IP header length doesn't exceed the optimal packet length calculated for each platform.
Fragmentation packets will be used when NAL information plus RTP, UDP, IP headers length exceeds the optimal packet length instead of leaving the network to fragment the IP packets. This method provides more control over the packet delivery and reduces the network delays.

Aggregation packets will be used in order to reduce network overhead, decreasing the total number of packets delivered for one frame and optimizing delays and jitter.

- *Repetition of SPS and PPS NALs*

In case the packet loss and bit error rates increase, then the RTP encapsulator should repeat the SPS NAL and PPS NAL, because these are very important for the receiver to decode the SVC stream. Another possibility is to control the encoder to send these types of NALs more frequently.

- *Different protection of NAL units according of their importance*

Another issue is to change the protection of the packets according the NRI values of the NAL units. NAL units with NRI greater than the others have more importance in the decoding process. To provide different levels of protection at IP level, the play-out should apply extra Forward Error Corrections, which implies another delay to be added at the end-to-end system delay, so FEC algorithms will be applied only at the BSTs.

- *Dynamic frame-rate*

A higher packet loss due to bad network conditions or a receiver buffer overflow (congestion), could be solved by changing the velocity of the packets sent by the play-out, so a dynamic frame rate could be applied to reduce the frames per second. This method depends basically of the SVC encoder implementation.

- *IP routing control*:

The play-out has the knowledge of the total amount of bit rates to be delivered to each platform. The finality of this strategy consists in distributing optimally the sessions through each platform, with regard to the network characteristics such as available bandwidth, jitter, content priority, etc… In this case, the IU web service will acquire networks characteristics parameters from the SUIT database, triggered by new commands or by some alarm message and will decide the optimal content distribution of the unicast content.

-Multicast/ Broadcast Scenario

In the case of a point to multipoint connection a large number of receivers, possibly spread all over the world, are receiving multicast real-time data and are sending back receiver reports as MPEG-21 descriptions. The question is, should we decrease the bandwidth of a video session only because of one link on the other end of the world suffers from high packet loss?
The response is that the play-out won't act in this case, and it will be only focused in the unicast scenario.
For the broadcast scenario, the ADTE WS is responsible to adapt the bit rate according the play-out target bit rate.

2.1.2 Encapsulator related issues

The QoS related to the DVB-T/H & WIMAX encapsulators will be managed according to the parameters defined in Deliverable 1.2 [2], so both base stations will try to guarantee the QoS applying their own algorithms.

DVB-T/H and WIMAX base stations will deal with some parameters to assure the QoS at different levels:

- **DVB-T/H:**
    - **QoS at TS**
    - **QoS at RF**
    - **QoS Handover (HO)**
- **WIMAX:**
    - **QoS at RF**
    - **Carrier to Noise and Interference Ratio (CNIR) and Bit Error Rate (BER)**

2.1.3 Gateway related issues

The QoS related to the WIFI interface will be detailed in Deliverable 3.3 "Optimization of the entire system", where the models and error pattern bit streams from WP2 are incorporated in order to form the entire system. In that deliverable the gateway rate-control methods to provide the desirable QoS for each terminal will be described.

3 QoS dependent handover strategies in DVB-T/H systems

This section describes the handover scenarios that are investigated on a testbed which is set up as part of WP4 in SUIT.

The major objective of this activity is the identification of those QoS parameters that are best suited to trigger handovers. Therefore, tests with different parameter sets are being prepared that focus on either physical layer parameters, Transport Stream related parameters or IP level related parameters. Of special interest is the hysteresis effect that is expected to result from the inherent delay in the analysis of higher layer parameters, and if this could substitute the calculation of average values of physical layer parameters to avoid ping-pong effects.

3.1 Description of the Handover Scenario Simulator

3.1.1 Goals and Features

The goal of Handover Scenario Simulator (Figure 3-1) is to simulate a real broadcast environment with one mobile receiver switching between two DVB-T/H transmitters providing different signal quality at two different frequencies.

The Handover Scenario Simulator is capable of controlling two Broadcast Test Systems (R&S type: SFU) over the LAN interfaces and of simulating signals with variable quality. To achieve a signal quality degradation/ improvement, one or more of SFU's transmit parameters are varied over time.
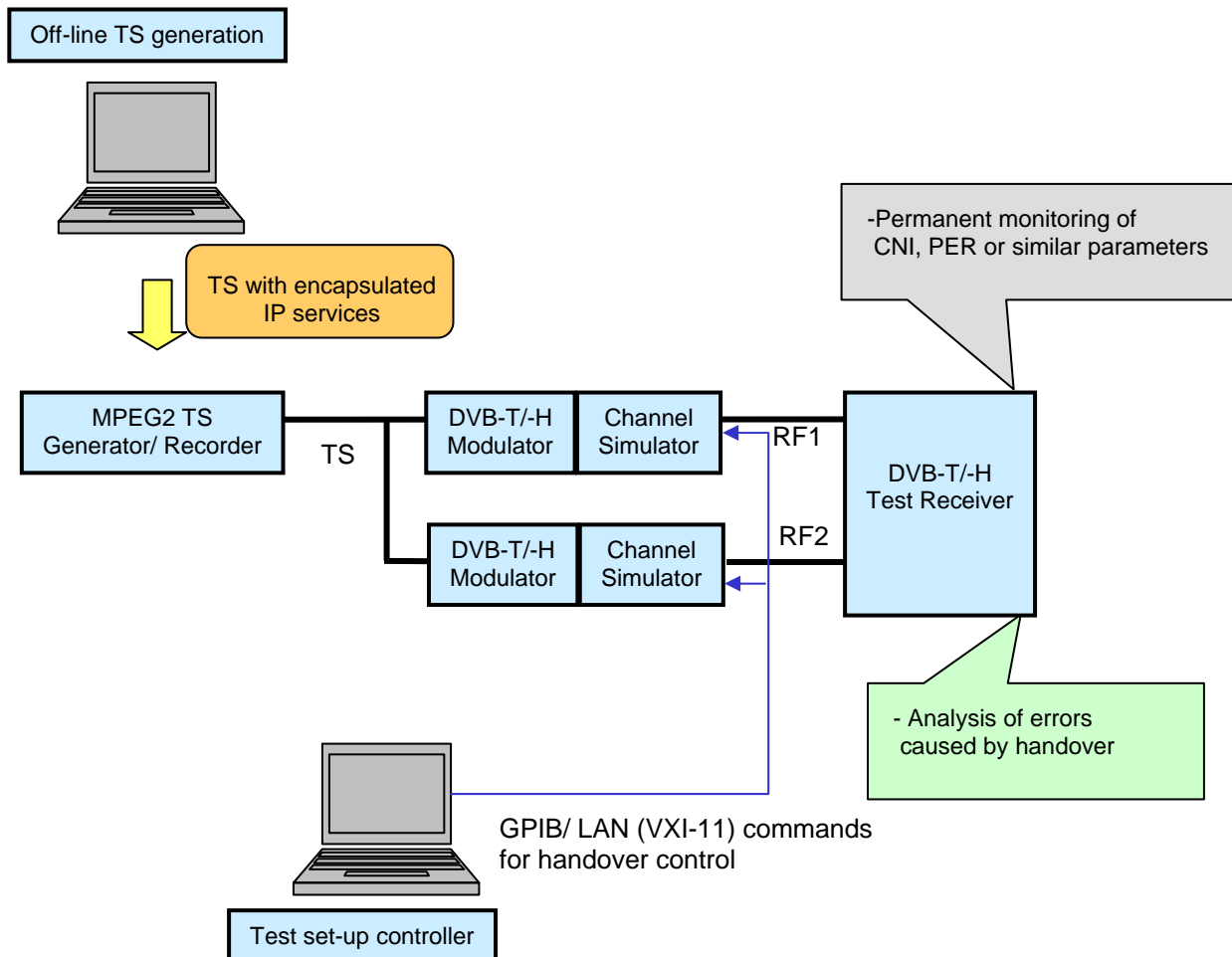

Figure 3-1: Test set-up for horizontal handover testing in DVB-T/H systems

The following two sweep scenarios are implemented:
1. RF-Output level is varied, signal-to-noise ratio remains constant
2. RF-Output level remains constant, signal-to-noise ratio is varied.

Other transmit parameters such as channel characteristics/ fading profiles, can also be configured via the user interface.

3.1.2 User Interface
The user interface was created using CVI/ LabWindows programming environment. It consists of several program panels which control panel-specific functions:
1. Main Panel
2. Connection Panel
3. Configuration Panel
4. Scenario 1 Panel (RF Level Sweep)
5. Scenario 2 Panel (Noise Level Sweep)
6. Test Functions (SCPI Commands) Panel

3.1.2.1 Main Panel

The Main panel (Figure 3-2) is the first interactive panel that is visible after the application start. It can also be accessed from the menu bar on the top of every panel.
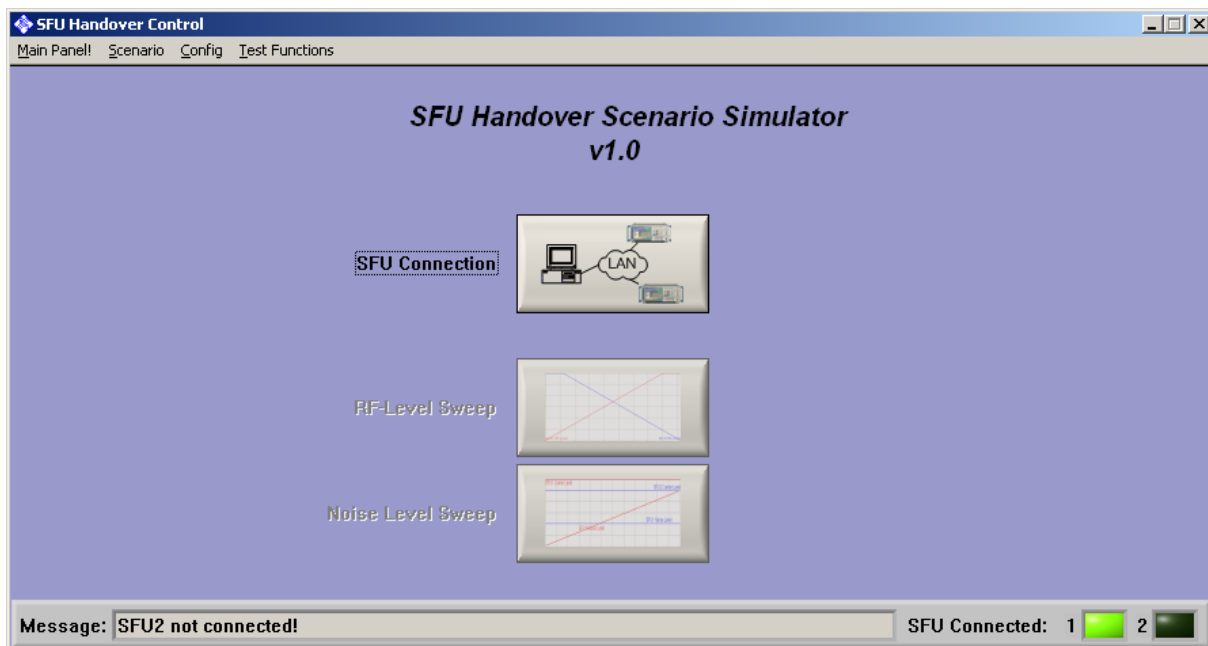

Figure 3-2: Main panel of the SFU handover scenario simulator

The main panel provides access to the most important program functions:
- LAN Connection Panel
- RF-Level Sweep Panel
- Noise Level Sweep Panel

At the bottom of the main panel there is a status bar which signals the connection to the SFU Broadcast Test System, and a text box for error messages. This status panel and the menu bar are visible on every program panel.

The scenario panels and menu items are only accessible when both SFUs are connected.

3.1.2.2 SFU Connection Panel

The connection panel (Figure 3-3) can be accessed either from the main panel by clicking the appropriate control item or from every other panel by choosing an according menu item ('Config → Ethernet connection').
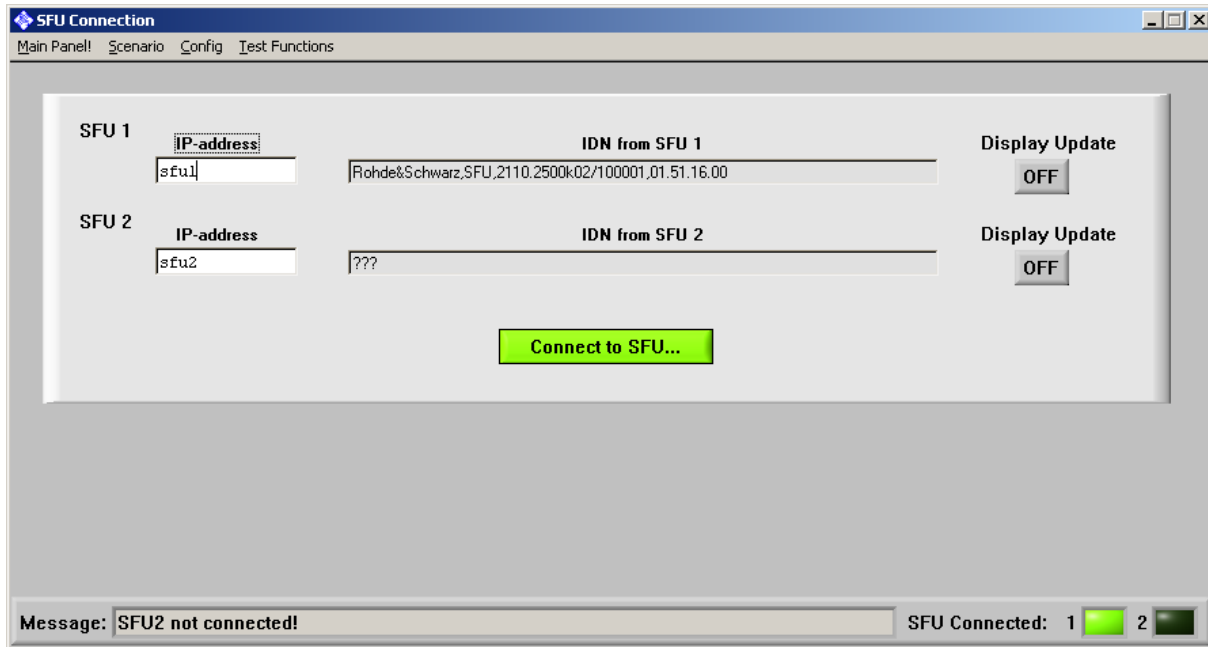

Figure 3-3: SFU connection panel

The application connects to the SFU via a LAN interface. The SFU Connection Panel provides an interface for connecting an instrument at a given IP-Address or host name (as defined in HOSTS file). A device specific ID Number is displayed. The user can decide whether the instrument display should be updated during the connection.

3.1.2.3 Configuration Panel

The configuration panel (Figure 3-4) can be activated from the menu bar ('Config → SFU configuration').
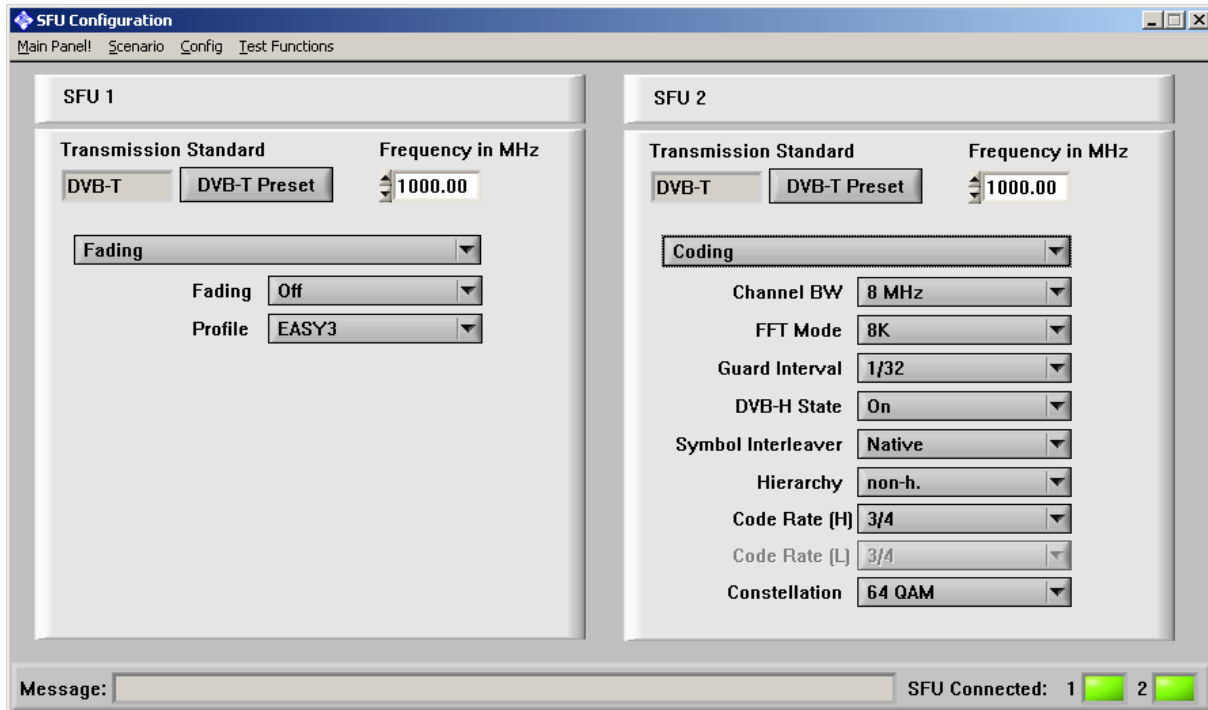

Figure 3-4: configuration panel

It provides access to basic application-related SFU parameters for both instruments individually:

- RF-Frequency
- Input Signal
    - o Signal Source
    - o Signal Input
    - o Stuffing
- DVB-T Coding Parameters
    - o Channel Bandwidth
    - o FFT Mode
    - o Guard Interval
    - o DVB-H Status
    - o Symbol Interleaver
    - o Hierarchy
    - o Code Rate
    - o Constellation
- Fading
    - o Status (on/off)
    - o Profile

The 'DVB-T Preset' control sets the instrument to a predefined state equivalent to the PRESET hardkey, except for modulation standard (here: DVB-T with DVB-H Status on)

3.1.3 Signal level triggered handover/ RF Level Sweep Panel

The RF level sweep panel (Figure 3-5) can be accessed only if both SFUs are connected, either from the main panel by clicking the appropriate control item or from every other panel by choosing the menu item 'Scenario → RF-Level Sweep'.
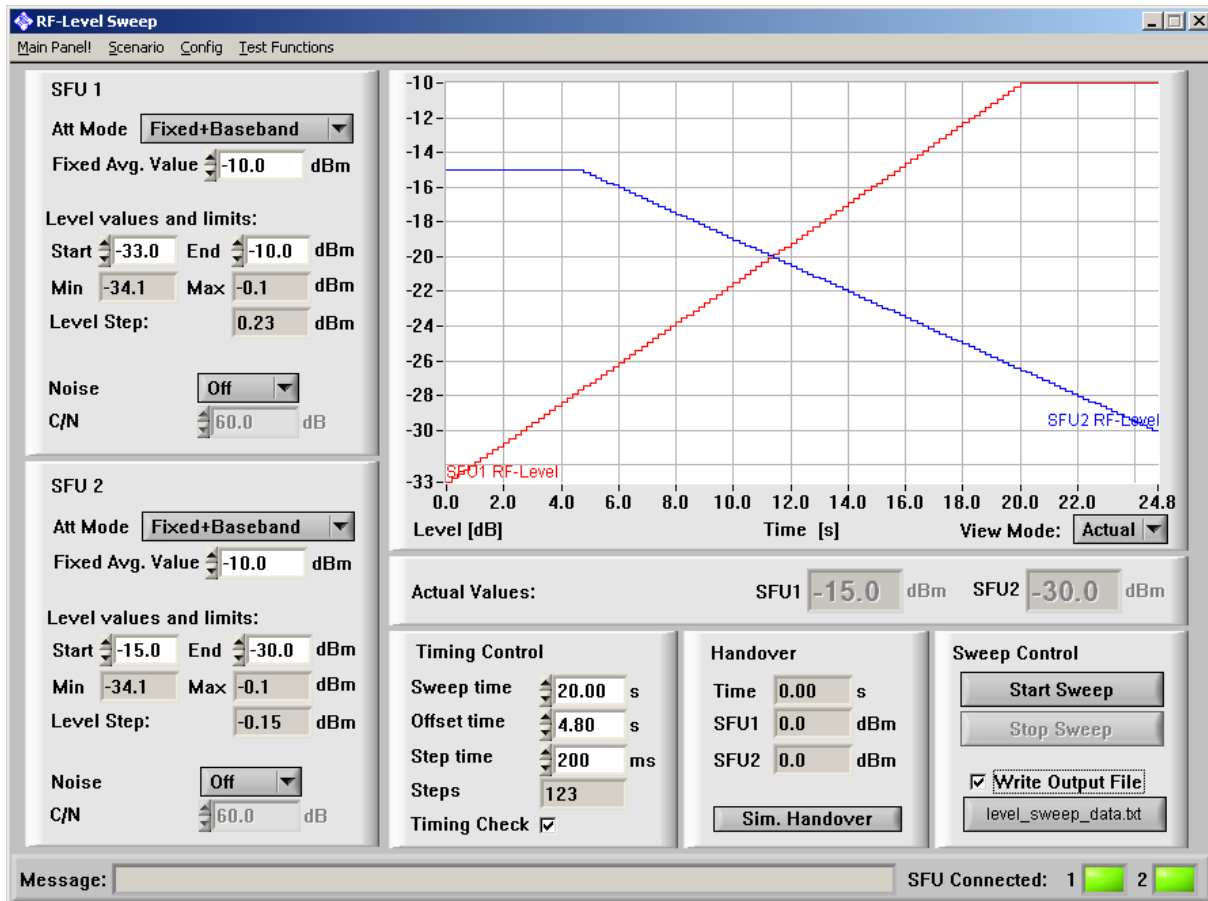

Figure 3-5: RF level sweep panel

The RF Level Sweep Panel provides a control interface for configuring and triggering an RF-level sweep.

The sweep configuration includes:

1. Configuration of attenuator mode (independent for both connected instruments). The RF-output level of an SFU instrument can only be varied continuously if the electronic attenuator is in FIXED mode and the automatic level control (ALC) is 'ON'. The output level range in this mode is limited by the ATT. FIXED RANGE LOWER and ATT. FIXED RANGE UPPER values. The user can choose between the three attenuator modes:
    a. **Auto**: the level is not limited by the attenuator fixed range but the output level is not continuous.
    b. **Fixed**: the level range is limited.
    c. **Fixed + Basisband**: the level range is limited. By additional baseband attenuation the dynamic range of output level is 20 dB greater than in FIXED mode, however signal quality decreases due to quantisation noise.

When the mode a. or b. is chosen, the user must define the average output level for the level sweep **(Fixed Avg. Value)**. The application determines the related level range.

2. Configuration of start and end value for the sweep: **Start**, **End** (independently for both connected instruments).

3. Configuration of AWGN noise state (on/off) and C/N value: **Noise, C/N** (independently for both connected instruments).

4. Configuration of timing parameters.

    a. **Sweep Time:** duration of the sweep (offset time not included).

    b. **Offset Time:** positive offset time means that the level sweep on SFU2 is delayed compared to the sweep on SFU1; when Offset Time is negative the sweep of SFU1 is delayed. Total time equals sweep time + offset time

    c. **Step Time:** defines the sweep resolution; must be at least 100 ms.

The total sweep time is a sum of the sweep time and the offset time. Both total time and offset time must be multiples of the time step.

    d. **Timing check**. When activated, a timing report is displayed at the end of the sweep. This includes:

        1. Switching time between step values. Min, max and average value are displayed, as well as the goal step time.

        2. Total sweep time vs. total time goal.

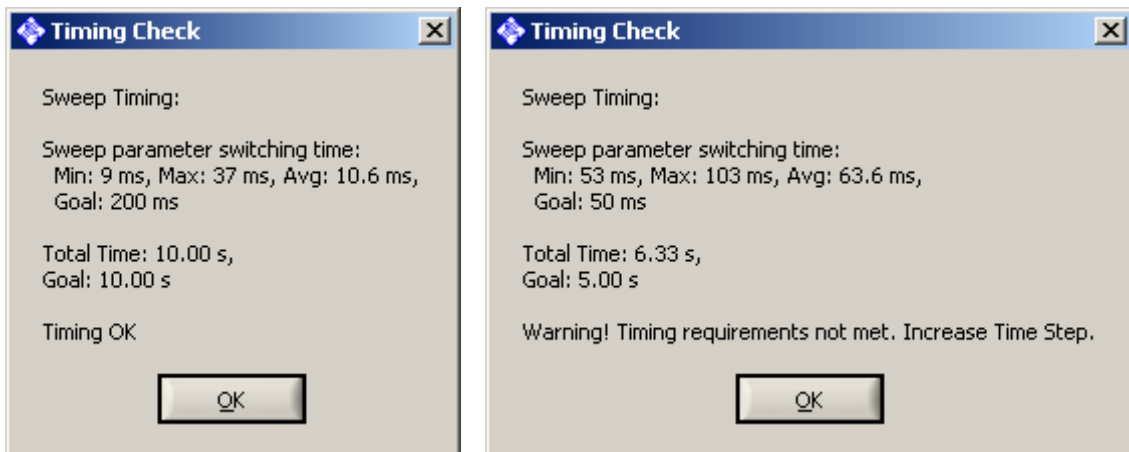If the timing requirements cannot be met, a warning occurs (see Figure 3-6).

Figure 3-6: Timing check information

The actual level sweep configuration is visualised on the time plot (see Figure 3-5). The user can choose between the view of idealised sweep function and the view of actual sweep function (steps): control **View Mode**.

The control **Start Sweep** triggers a new RF level sweep with the current level and timing parameters. The actual level values can be observed on the plot or as numeric values displayed on the panel.

Additionally, an output data file with sweep values can be defined. An example file is shown in Table 3-1 below.

```
** SFU Handover Scenario output file
** RF-Level Sweep
** SFU1: RF_Freq:1000.000MHz Noise:C/N=40.00dB Fading:EASY3
** SFU2: RF_Freq:1000.000MHz Noise:C/N=30.00dB Fading:off
index        relative_time[ms]     SFU1_Level[dBm]     SFU2_Level[dBm]
0            0                     -33.00              -15.00
1            100                   -32.77              -15.15
2            200                   -32.54              -15.30
3            300                   -32.31              -15.45
4            400                   -32.08              -15.60
5            500                   -31.85              -15.75
6            600                   -31.62              -15.90
7            700                   -31.39              -16.05
8            800                   -31.16              -16.20
9            900                   -30.93              -16.35
...
```

Table 3-1: Example for RF level sweep output file

An incoming handover event that is signalled by the test receiver is indicated on the current plot during the live update of the level sweep application window. The time of the handover event and the RF levels at that time are in this case also displayed on the panel. This is presented on the screenshot in Figure 3-7 below.
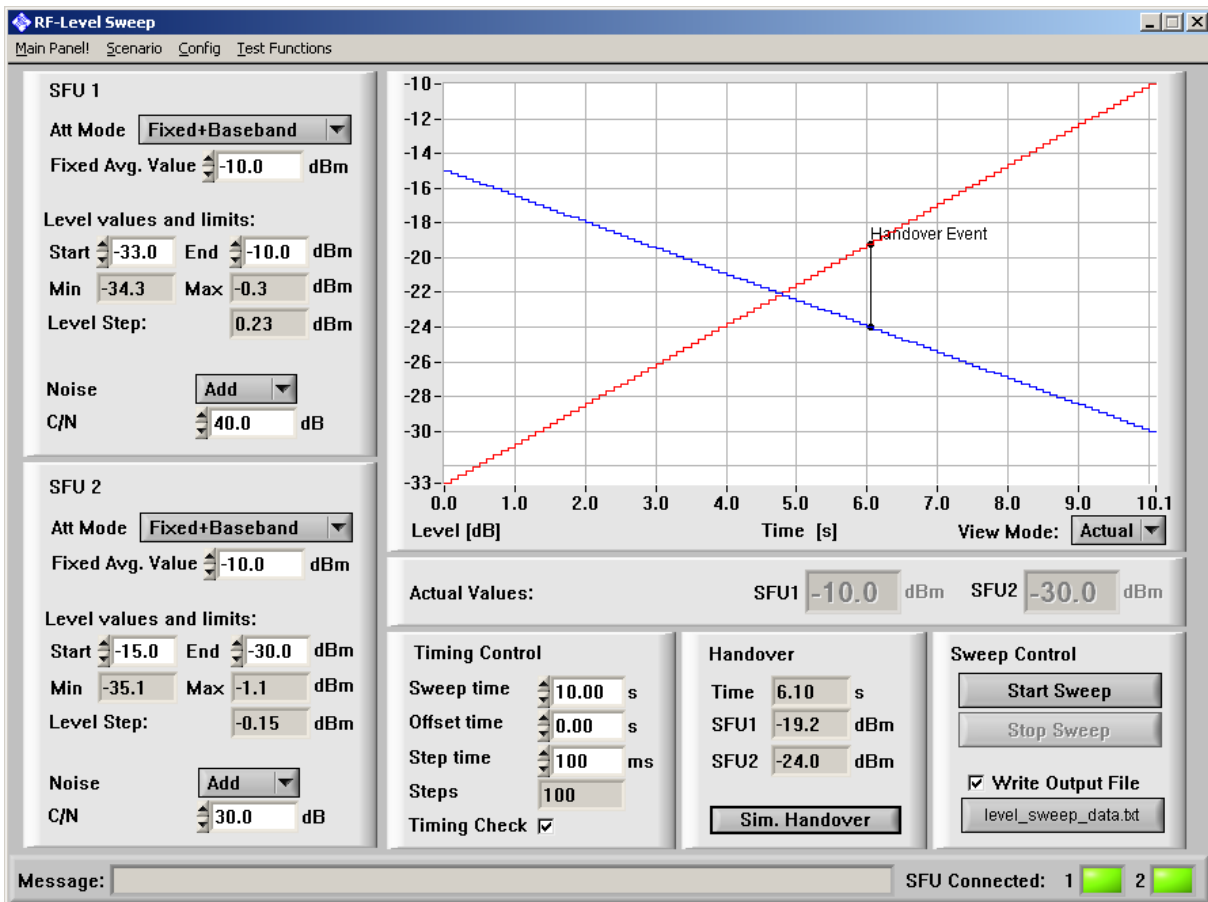

Figure 3-7: Indication of a HO event in the level sweep application window

3.1.3.1 Validation of the test set-up (generator part)

For checking of the accurate settings, the actual RF level can be observed during the simulated level sweep on a spectrum analyser with Frequency Span = 0. In this case a sine wave is used as the modulation source and the RF signal is fed directly to the input of the analyser. Due to the low signal bandwidth of sine wave modulated signal, the power level observed on the analyser equals the output power of the SFU. Figure 3-8 displays measurements for different parameter sets (each one signal source only):

-   start level: -30 dBm
-   end level: -5 dBm
-   sweep time 10 s
-   step time 100 ms
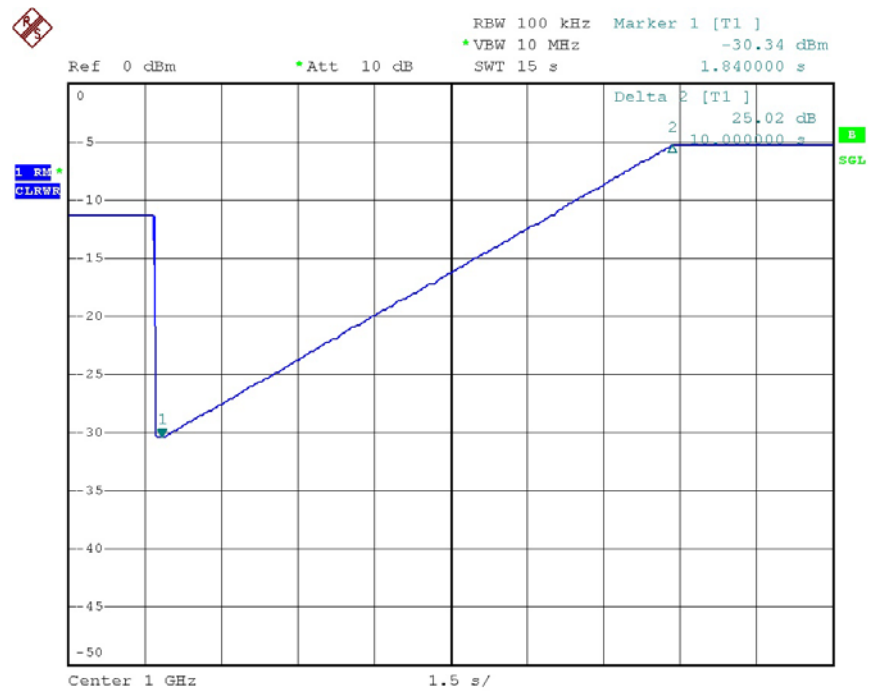
-   start level: -15 dBm
-   end level: -45 dBm
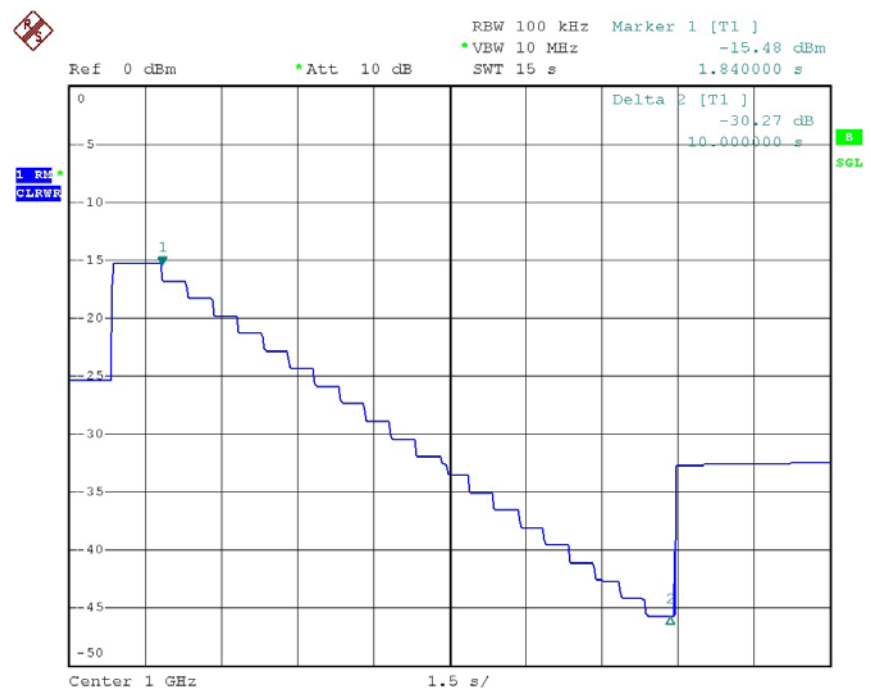-   sweep time 10 s
-   step time 500 ms

Figure 3-8: RF level measurement with a spectrum analyser

3.1.4 Interference level triggered handover (Noise Sweep Panel)

The Noise Sweep Panel can be accessed only if both SFUs are connected, either from the main panel by clicking the appropriate control button or from every other panel by choosing the menu item Scenario → Noise Level Sweep (see Figure 3-9).
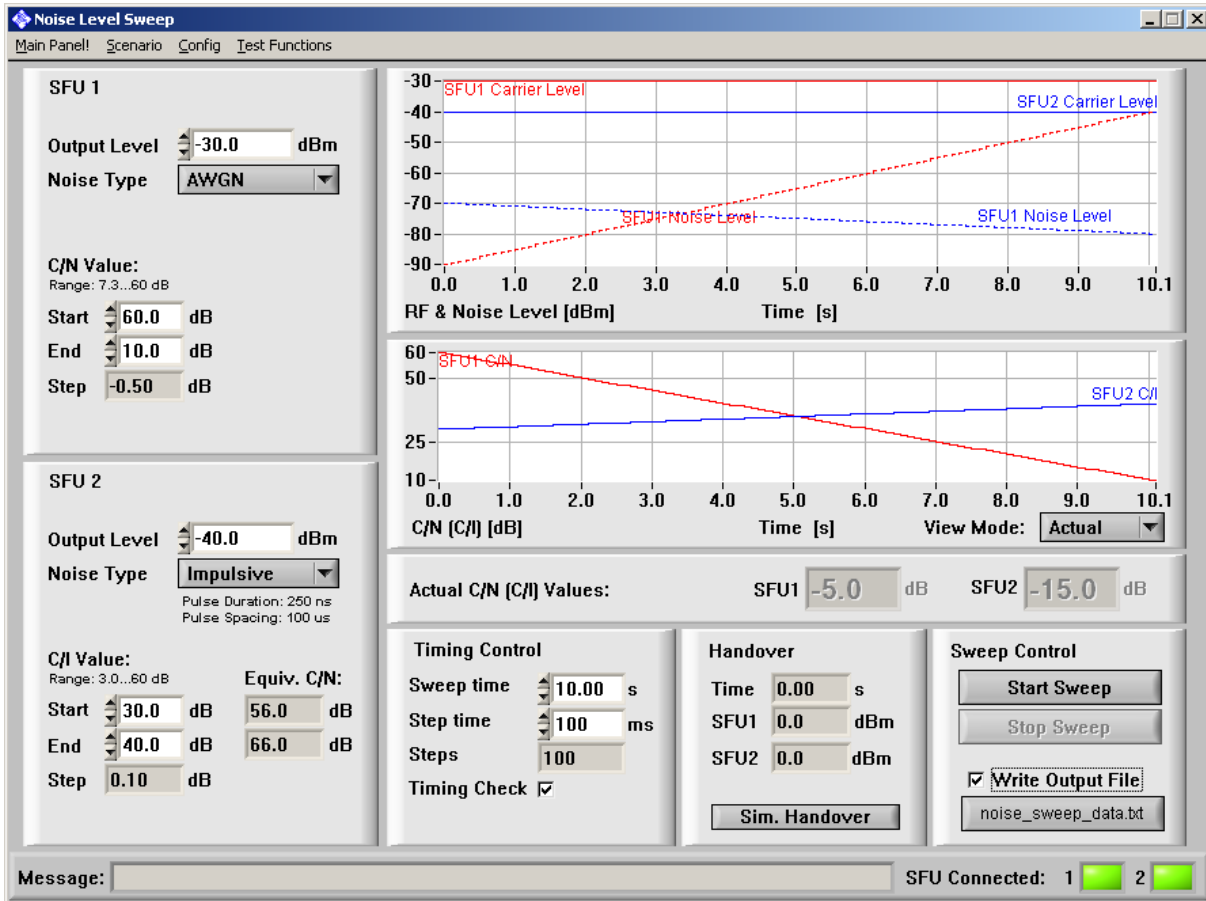

Figure 3-9: Noise Level Sweep panel

In this scenario the user can choose between two noise types for each instrument individually. The carrier-to-noise ratio (for AWGN) or carrier-to-impulse ratio (for impulsive noise) is swept between start and end value over the defined time period, while the defined output levels remain constant. The instruments must be operated in 'Attenuator fixed' mode to prevent level discontinuity during the sweep.

The configuration of the sweep includes:

1. Setting up **output level**. For each output level the current range for noise parameters (limited due to attenuator fixed mode) is obtained and displayed on the panel.
2. Configuration of **noise type** for each instrument:
   a. **Off**
   b. **AWGN**
   c. **Impulsive** noise: consists of equally spaced impulses of 250 ns duration. During an impulse, an AWGN noise is added to the carrier with the defined carrier-to-noise ratio (here: carrier-to-impulse, C/I). The impulse spacing equals 100 μs. The effective carrier-to-noise ratio (with regard to pulse duration and spacing) is 26 dB higher than the C/I ratio.

3. Configuration of the **start** and **end** values for C/N (C/I) noise parameters. These values are limited in order to obtain continuous output signal. The actual range is displayed.
4. Timing configuration
   a. **Step time** (minimal value 100 ms)
   b. **Sweep time** (must be a multiple of step time)
5. **Timing check**. When activated, a timing report is displayed at the end of the sweep. (see also 3.1.3)

The actual noise sweep configuration is visualised on two time plots. The first plot shows the RF output and noise level scaled in dBm. The second one shows the variation of C/N (C/I) parameter over the sweep time. The user can choose between the view of idealised sweep function and the view of actual sweep function (steps): control **View Mode**.

The control **Start Sweep** triggers a new noise sweep with the current noise and timing parameters. The actual values can be observed on the plot or as numeric display controls on the panel.

There is a possibility to log sweep values into a defined file. An example of log file is shown in Table 3-2:

```
** SFU Handover Scenario output file
** Noise sweep
** SFU1: RF_Freq: 1000.000MHz RF_Level:-30.00dBm; noise_type:AWGN;
Fading:EASY3
** SFU2: RF_Freq: 1000.000MHz RF_Level:-40.00dBm;
noise_type:impulsive; Fading:off
index      relative_time[ms] SFU1_C/N[dB]    SFU2_C/N[dB]
0          0                 60.00           30.00
1          100               59.50           30.30
2          200               59.00           30.60
3          300               58.50           30.90
4          400               58.00           31.20
5          500               57.50           31.50
6          600               57.00           31.80
7          700               56.50           32.10
8          800               56.00           32.40
9          900               55.50           32.70
10         1000              55.00           33.00
...
```

Table 3-2: Handover scenario output file

Another feature of the noise sweep application is to mark an incoming handover event from the test receiver on the current plot during the live update. The time of the handover event and the RF level at that time would also be displayed on the panel. This is shown in the screenshot in Figure 3-10.
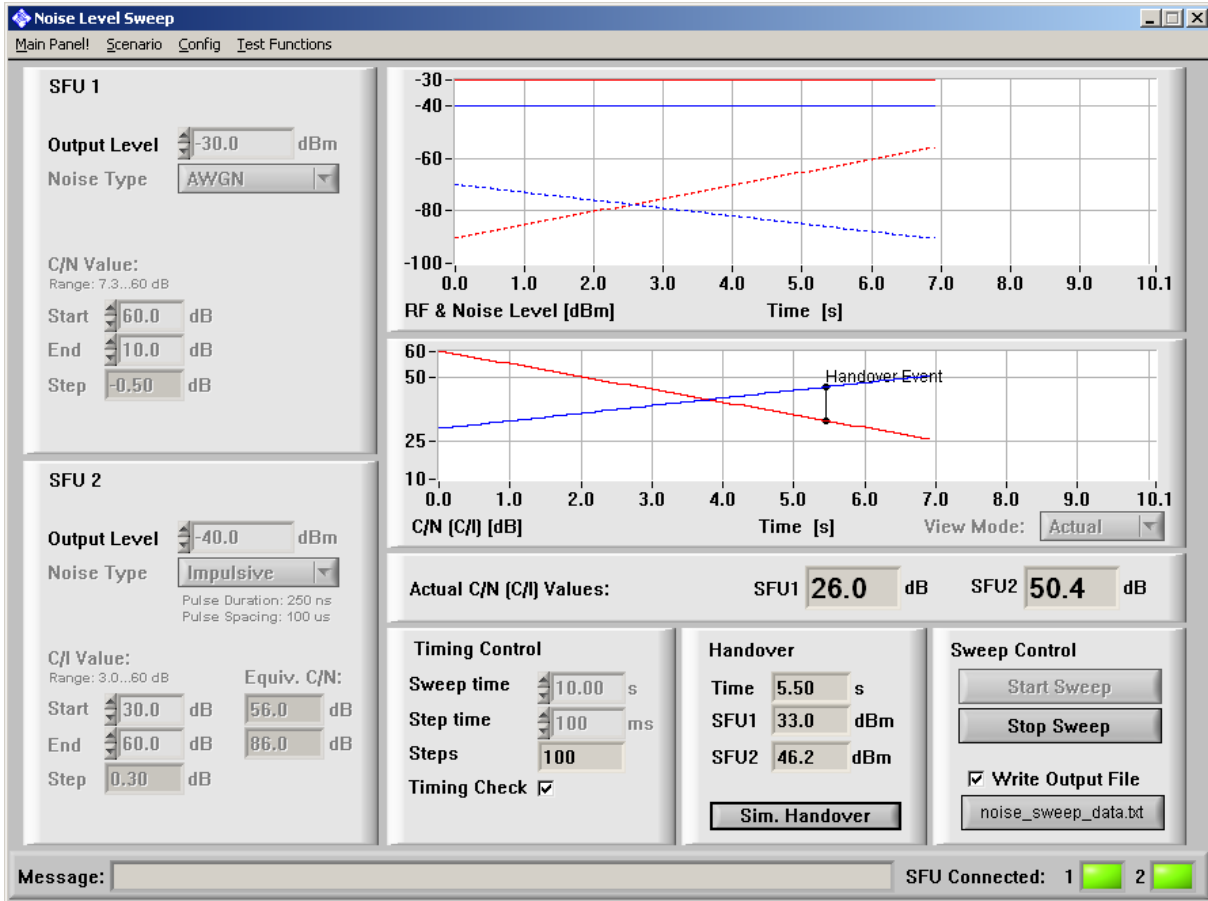
Figure 3-10: Screenshot with handover event displayed

3.1.4.1 Validation of the test set-up (generator part)

The validation of the test signal generator set-up was performed with a spectrum analyser.
The noise level can be measured indirectly on a spectrum analyser. The measured 1 GHz signal is DVB-T modulated (Bandwidth 8MHz) and fed to the analyser. The analyser is set to measure the signal power at the frequency of 1.045 GHz width a resolution bandwidth of 100 kHz. This is visualised below (Figure 3-11):
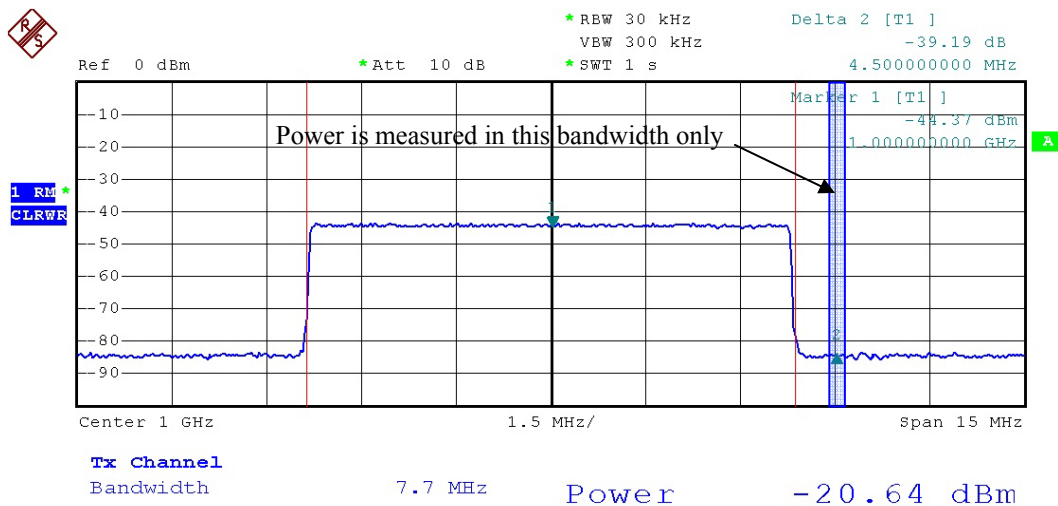

Figure 3-11: Spectrum of a DVB-T/H signal

The following screenshot (Figure 3-12) presents the relative variation of signal noise over time for the following parameters:

- Output level: -20 dBm
- AWGN noise
- C/N initial value: 40 dB
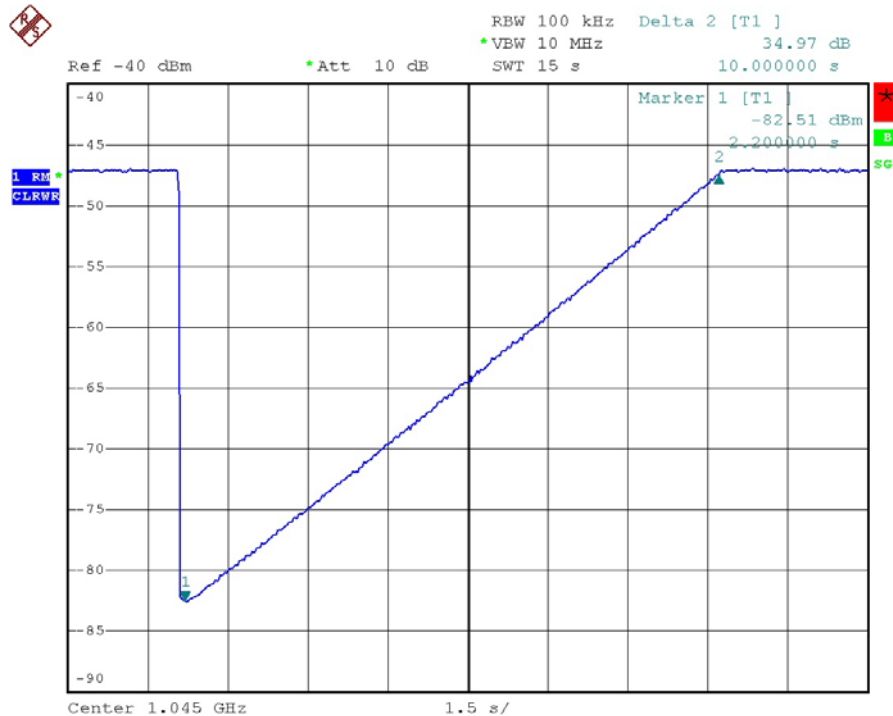- C/N end value: 5 dB
- Sweep time: 10 s
- Time step: 100 ms

Figure 3-12: C/N variation over time (displayed on a spectrum analyser)

3.2 Criteria for horizontal DVB-T/H handover initialisation

As described in deliverable D1.2, section 2.5, the original plan was to use the Transport_error indicator to trigger the horizontal DVB-T/H handover. A more detailed analysis has since led to the conclusion that the analysis of physical layer parameters may be preferential since Transport Stream parameters may be too slow to allow for a prompt reaction in an ever changing mobile environment. This is because the analysis of Transport Stream parameters requires a demodulated Transport Stream. The demodulation process depends to a certain extent on the size of the TS buffer which needs to be approximately half filled before the TS can be forwarded to the next module. This inherent delay may be comparable to a rather long averaging of TS parameters.
On the other hand, the analysis of Transport Stream parameters could create a hysteresis effect that would then replace the calculation of average values of physical layer parameters.
The conclusion is to investigate physical layer parameters first (as a working assumption) for a faster reaction to changing reception conditions. These physical layer parameters are extracted from the DVB-T/H frontends directly via a SNMP MIB. Three parameters are used as candidates and their reaction in different mobile environments is to be investigated:

- RF level
- C/(N+I) (Carrier to noise and interference ratio)

- MER (Modulation Error Ratio)

The Transport Stream parameters will also be analysed and their suitability for triggering the handover will be investigated. They are also accessible via the SNMP MIB.
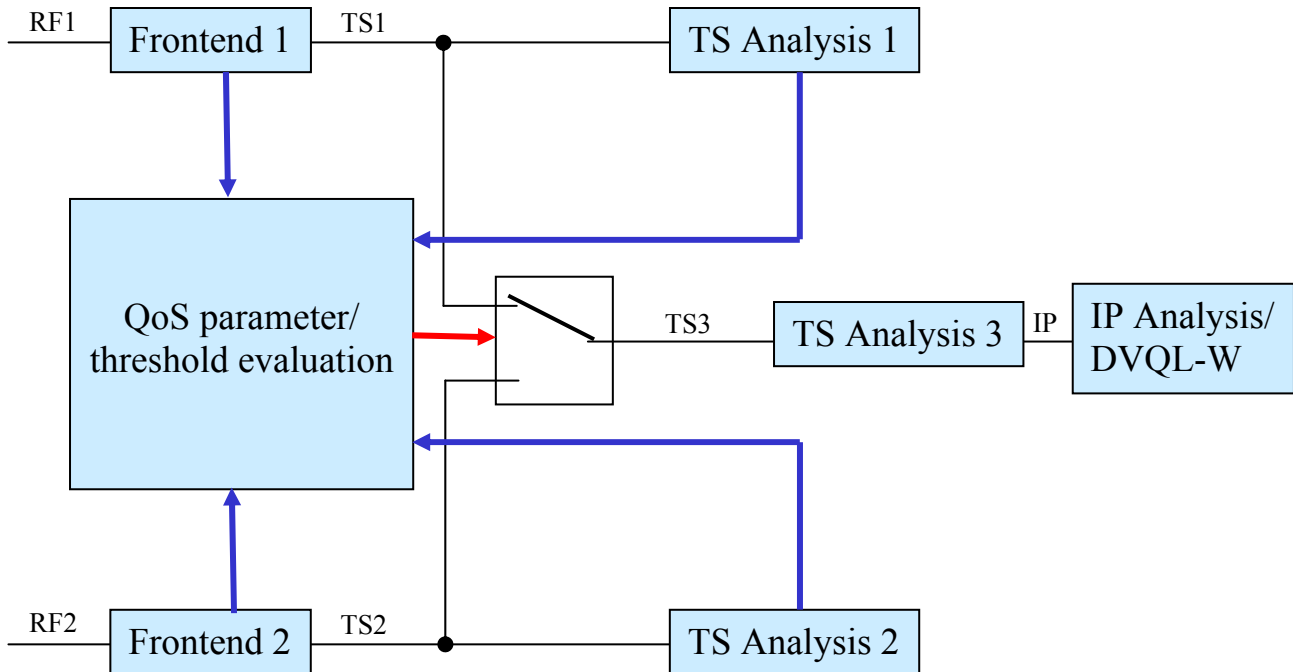
3.3 QoS parameter monitoring

Figure 3-13: Test receiver set-up for investigation of horizontal handover (DVB-T/H → DVB-T/H), from [2]

The DVM400 Test receiver (Figure 3-13) that is part of the handover testbed for horizontal handover in DVB-T/-H systems, allows the evaluation of QoS parameters at the physical layer and at the Transport Stream level. Tests are planned to analyse the impact of delays and reaction time if QoS parameters from different layers are selected to initiate the handover.

The next step is than the integration of a WiMax frontend into the test receiver's PC platform (Figure 3-14), that enables the equipment to perform a vertical handover between the DVB-T/H system and the WiMax system. Here a three-layer approach is possible: QoS parameters can be analysed at physical layer, in the case of DVB-T/H at the Transport Stream level, and at IP level. Again the tests are planned in such a way that the suitability of different QoS parameters or parameter sets can be judged.
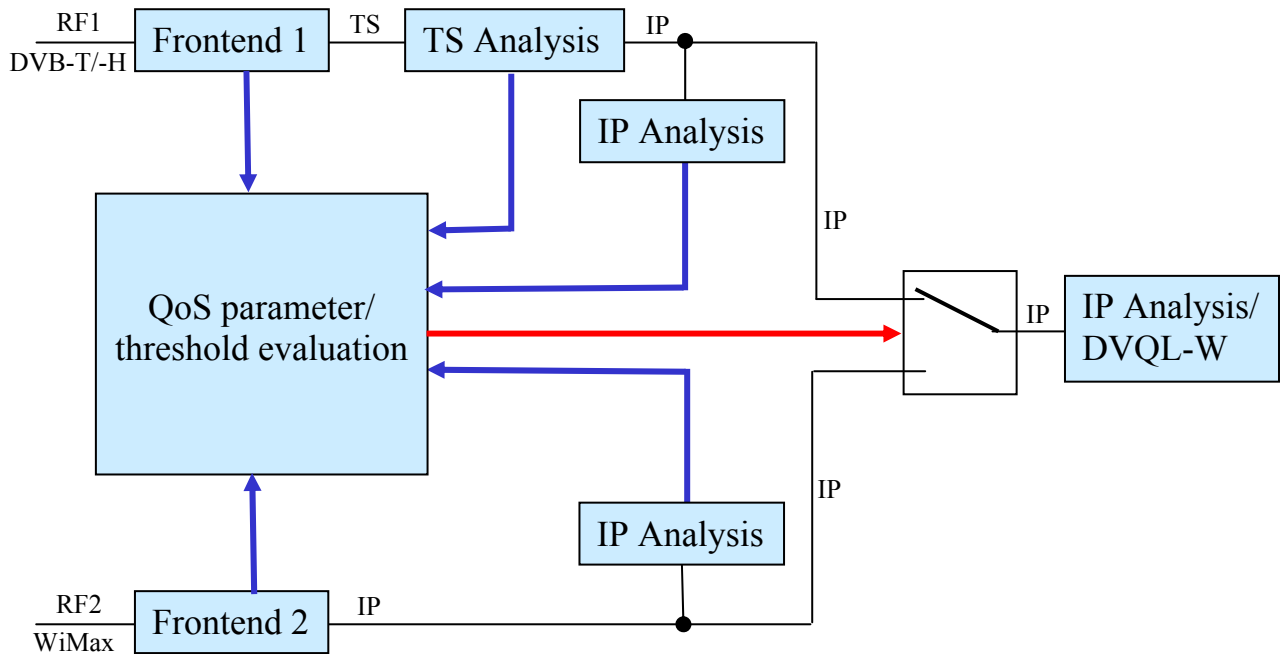
Figure 3-14: Test receiver set-up for investigation of vertical handover (DVB-T/H → WiMAX), from [2]

To evaluate the impact of handovers on the quality of the application, several parameters are monitored during the handover process. For the first tests, four parameters were selected:

- TS Synchronisation Loss
- Sync Byte Error
- BER (Bit Error Rate) before Reed-Solomon decoding
- IP HO packet loss rate.

The inclusion of physical layer parameters depends on the hardware and software of the WiMax RF frontend. Different product are being tested for their suitability.

4 WiMax Horizontal Handover (HO)

One of the important features of the Mobile WiMax system (802.16e) is the fast HO capability. For the Suit application, it is very important that the video stream, the control messages and all other traffic pass (when HO is in process) from the current serving BST to the target BST.
The clock synchronization, illustrated in Figure 4-1 below, is based on GPS and connected to both BSTs.
The CPE is connected to 2 Base stations as depicted in Figure 4 below. The HO process starts when the CNIR (or S/N) in the CPE receiver is below a pre-defined threshold level which is assumed as +10dB.
Adjusting the attenuators causes changes in the receiver CNIR. When it becomes low than +10dB the CPE starts an automatic scan of the other frequencies. The detailed HO process is described in section 4.1.1 below.
The maximum HO switch time measured in the tests was 120ms. It should be emphasized that a special SW is implemented in the ACR module, in the BST cards, so no addition SW is needed to make all traffic and control transformations in the HO process. The result is that there are no video packets lost or at least no video disturbances visible on the video screen.
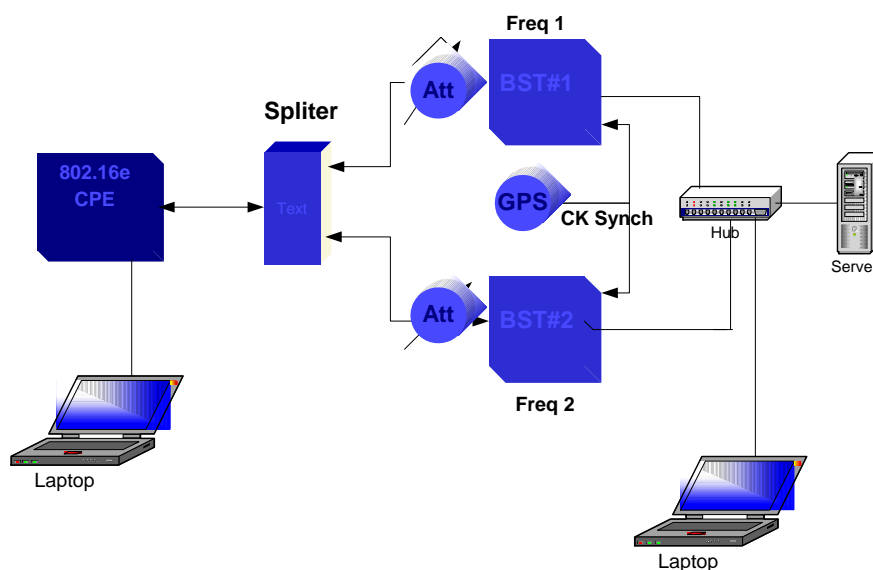

Figure 4-1: Handover laboratory set-up description

4.1.1 WiMax handover step by step

- After synchronization of the CPE to the serving BST, constantly measure Serving BST's CNIR
- Periodically the CPE receives MOB_NBR-ADV (Mobile Neighbours Advertising, i.e. the CPE receives from the serving BST the list of all the neighbouring BSTs)

- If the serving BST's CNIR < Scn_threshold (10dB): send MOB_SCN-REQ (Mobile Scan Request)
- After MOB_SCN-RSP: Interleaved scan (scan/data/scan/data…)
- For neighbouring BSTs with a CNIR > Req_threshold (6dB): send MOB_MSHO-REQ (Mobile_Mobile Station(CPE) HandOver–Request)
- [Receive MOB_BSHO-RSP] (Mobile_BaseStationHandOver)
- If Serving BST's CNIR < Ind_threshold (Cross+3dB): send MOB_HO-IND
- If Serving BST's CNIR < Tran_threshold (Cross): Handover (transition)

Inter and Intra HO:

- Inter FA(Frequency Assignment) HO – Handover between BSTs with different frequencies.
- Intra FA(Frequency Assignment) HO – Handover between BSTs with the same frequency Intra FA(Frequency Assignment) HO doesn't require the interleaved scan.

Messages between serving BST and CPE:

MOB_NBR-ADV – Serving BST Advertising its Neighbours (BSTs) to CPE

Neighbours MOB_SCN-REQ – CPE requests scanning intervals to scan Neighbours BST's

MOB_SCN-RSP – Serving BS response (to scan request)

MOB_SCN-REP – CPE reports its scan results to the serving BST

MOB_MSHO-REQ – CPE requests permission to Handover to a Target BST.

MOB_BSHO-RSP – Serving BST response (to Handover request).

MOB_HO-IND – CPE notifies HO (confirmation / cancellation / rejection).

MOB_BSHO-REQ – Serving BS requests CPE to Handover to a Target BS.

4.1.2 QoS and delay parameters

Two QoS connections were defined and tested:
Best effort (BE) and Unsolicited Grand Service (UGS).
Delay figures in both connections are presenting in the table below:

	UGS (ms)	BE (ms)	Comments
Downlink	30	40	
Uplink	40	70	
Round Trip Delay	70	110	
HO		120	Should be less than 100ms

5 Acronyms

ACR	ACR - Access Control Unit
ADTE WS	Adaptation Decision Taking Engine Web Service
BE	Best Effort
BER	Bit Error Rate
BST	Base Station
C/(N+I), CNIR	Carrier to Noise and Interference ratio
C/N	Carrier to Noise ratio
CIR	Committed Information Rate
CPE	Consumer Premises Equipment
DCF	Distributed Coordination Function
DIA	Digital Item Adaptation
DVB	Digital Video Broadcasting
DVB-RCT	DVB Return Channel Terrestrial
DVB-T/H	DVB Terrestrial/ Handheld
DVQL-W	Digital Video Quality Level - Weighted
EDCA	Enhanced Distributed Channel Access
FER	Frame Error Rate
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Function
HO	Handover
IU WS	Intelligent Unit Web Service
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
SW	Software
TS	MPEG-2 Transport Stream
UED	Usage Environment Description
UGS	Unsolicited Grand Service
WM	Wireless Medium
WS	Web Services

6 References

[1]   SUIT Deliverable D1.3 "All-IP Support Requirements", October 2006

[2]   SUIT Deliverable D1.2 "QoS requirements", November 2006

[3]   SUIT Deliverable D4.2 Session and Description Protocols", February 2007