2 METHODOLOGY

The methodology is based on the performance of end-to-end automatic tests, thus making it possible to identify the quality of service on the field, giving as much a realistic perspective of the networks' performance as possible, from the user's standpoint.

Measurement collection is made using *drive-tests*. Besides providing an assessment from the user's standpoint, this approach makes it possible to carry out the tests independently from the correct functioning of the networks themselves, i.e., for example, also analysing the areas with poor or no coverage at all.

On the other hand, the use of a sole testing system to assess the services provided by the three mobile networks makes the results highly comparable, regarding time and space.

2.1 FUNDAMENTALS

This study's methodology is based on three basic characteristics:

a) **End-to-end measurements** - measurements reflect all aspects that impact the quality of a service;

b) **Impartiality**: Measurements are carried out under equal terms for the three operators (OPTIMUS, VODAFONE and TMN);

c) **Objectivity**: Tests are carried out in a totally automatic way, eliminating the subjectivity inherent to human intervention or decision.

2.2 MAIN QOS INDICATORS

From the user's standpoint, the use of mobile services presents the following stages (different features of the Quality of Service):

a. **Network Availability** – Shows that the mobile network is present;

b. **Network Access** – Shows that it is possible to use the services (usually it corresponds to the indication of the network's name on the screen of the terminal equipment and the indication of the availability of GPRS and/or 3G);

c. **Service Access** – When the user intends to use a service, the mobile operator provides the access to that service (*e.g.* to set up a voice call);

d. **Service Integrity** – Corresponds to the Quality of Service (QoS) during its use (*e.g.* Audio Quality during a voice call; Video Quality during a video-telephony call);

e. **Service Consistency** – Corresponds to the way the use of the service is ended (according or not to the user's will).

The main Quality of Service Indicators were analysed for each of the QoS features.

2.2.1 Regardless of the Service

2.2.1.1 Radio Network Availability (Coverage)

Network availability is the probability of the mobile services being available to a user (radio network coverage).

$$\text{Radio Network Availability}\left[\%\right] = \frac{\text{No. of Measurements with Availability Mobile Services}}{\text{Total No. of Measurements}} \times 100\%$$

Mobile services are considered to be available when the radio signal shows values above the minimum levels that make its use possible. These levels may be adjusted by mobile operators and normally present different values for GSM and WCDMA[1].

The used testing and measurement system, through an RF Scanner, makes it possible to continuously measure each network's signal levels. These measurements are geographically referenced, thus rendering possible their representation on maps and making it easy to visualize the coverage levels of mobile networks on the routes under study.

[1] Wideband Code Division Multiple Access – Technology used on the radio networks of UMTS communications systems.

Table 1 – Coverage Levels

Coverage	GSM	WCDMA
Good	RxLev ≥ –85 dBm	CPICH RSCP ≥ –95 dBm
Acceptable	–95 dBm ≤ RxLev < –85 dBm	–105 dBm ≤ CPICH RSCP < –95 dBm
Bad	–110 dBm ≤ RxLev < –95 dBm	–115 dBm ≤ CPICH RSCP < –105 dBm
Non-existent	RxLev < –110 dBm	CPICH RSCP < –115 dBm

2.2.2 TELEPHONY SERVICES

2.2.2.1 SERVICE ACCESSIBILITY (VOICE OR VIDEO-TELEPHONY)

Service accessibility is the probability that the user has of having access to the service (voice or video-telephony), i.e., success probability when establishing a (voice or video-telephony) call.

It is considered that a call was "Set Up with Success" if it reaches the called terminal (one hears the "calling signal" on the calling terminal).

$$\text{Service Accessibility}\left[\%\right] = \frac{\text{No. of Calls Set Up with Success}}{\text{Total No. of Attemps to Set Up Calls}} \times 100\%$$

2.2.2.2 CALL SET UP TIME (VOICE OR VIDEO-TELEPHONY)

Call set up time is the period of time elapsing from the sending of a complete destination address (target telephone number) to the setting up of a call.

$$\text{Call Set Up Time}\left[s\right] = t_{\text{address\_sending}} - t_{\text{calling\_signal}}$$

$t_{address\_sending}$ – moment when the user presses the send button

$t_{calling\_signal}$ – moment when the call is successfully set up

(One hears the "calling signal" on the caller terminal).

2.2.2.3 CALL TERMINATION RATE (VOICE OR VIDEO-TELEPHONY)

Call termination rate is the probability of a call being maintained, after its set-up, during a given period of time, ending normally, i.e., according to the user's will.

$$\text{Call Termination Rate}\left[\%\right] = \frac{\text{No. of Normally Ended Calls}}{\text{No of Succefully Set Up Calls}} \times 100\%$$

2.2.2.4 CALL AUDIO QUALITY (VOICE OR VIDEO-TELEPHONY)

This indicator quantifies how well the conversation is perceived during a (voice or video-telephony) call. Both-ways communication is assessed and only calls with normal termination are considered.

The assessment of this QoS indicator is made by comparing the sent original audio sample, *X(t)* with the corresponding received degraded sample, *Y(t)*, on the other end of the call, by applying the *PESQ*[2] algorithm.

The objective audio quality index obtained by applying this algorithm is close to what would be obtained if sample *Y(t)* were submitted to the subjective appreciation of a panel of service users.

$$\text{Call Audio Quality}_{\text{side A}}\left[\text{MOS\_LQO}\right] = f\left\{X_B(t); Y_A(t)\right\}$$
$$\text{Call Audio Quality}_{\text{side B}}\left[\text{MOS\_LQO}\right] = f\left\{X_A(t); Y_B(t)\right\}$$

side A; side B – name of both ends of a voice call.

MOS_LQO – perceived audio quality quantification scale (Mean Opinion Score - Listening Quality Objective).

f – function corresponding to the application of the reckoning algorithm and conversion function of the results in MOS_LQO values.

$X_A(t)$; $X_B(t)$ – original audio sample sent from side A (B).

$Y_A(t)$; $Y_B(t)$ – degraded audio sample sent from side A (B), resulting from the transmission of the original sample $X_B(t)$ ($X_A(t)$).

The results of the algorithm application are shown on a MOS (Mean Opinion Score) type scale from 1 to 5 named MOS_LQO (Mean Opinion Score – Listening-only Quality Objective), such as shown on Table 2. The MOS scale quantifies the effort that it takes to understand a conversation. Its limits are 0 (zero) when there is no communication and 5 (five) when the communication is perfect. Value “zero” never shows on the results since they only consider situations where the connection was set up and kept during a given period. “Five” also never shows on the results because the *CoDec*[3] used by mobile networks does not render possible such high voice or video quality values (the voice or video quality reached with the usually used *CoDec* presents MOS values lower than 4.5).

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[2] PESQ – Perceptual Evaluation of Speech Quality. Recommended by the ITU-International Telecommunications Union (ITU-T Recommendation P.862 (02/2001); ITU-T Recommendation P.862.1 (11/2003)).

[3] CoDec – Codifier/De-codifier.

Table 2 - MOS_LQO / MOS_VQO Scale

MOS	Quality
5	Excellent
4	Good
3	Acceptable
2	Poor
1	Bad

In situations where each direction of the same call sends and receives several audio samples {$X_1(t), \ldots, X_n(t); Y_1(t), \ldots, Y_n(t)$}, the *Call Audio Quality* indicator is reckoned through the arithmetic average of the values obtained by applying the formula presented above to each pair of audio samples, i.e.:

$$\text{Call Video Quality}_{\text{side A}}\left[\text{MOS\_LQO}\right] = \frac{f\left\{X1_B(t); Y1_A(t)\right\} + \ldots + f\left\{Xn_B(t); Yn_A(t)\right\}}{n}$$

$$\text{Call Video Quality}_{\text{side B}}\left[\text{MOS\_LQO}\right] = \frac{f\left\{X1_A(t); Y1_B(t)\right\} + \ldots + f\left\{Xn_A(t); Yn_B(t)\right\}}{n}$$

2.2.2.5 VIDEO-TELEPHONY CALL VIDEO QUALITY

This indicator quantifies the communication's visual quality during a video-telephony call. Both directions of the communications are evaluated and only calls that ended normally are considered.

The evaluation process of this indicator is similar to the one used for *Call Audio Quality*.

$$\text{Call Video Quality}_{\text{side A}}\left[\text{MOS\_VQO}\right] = f\left\{W_B(t); Z_A(t)\right\}$$

$$\text{Call Video Quality}_{\text{side B}}\left[\text{MOS\_VQO}\right] = f\left\{W_A(t); Z_B(t)\right\}$$

side A; side B – name of both ends of a voice call.

MOS_VQO – perceived audio quality quantification scale (Mean Opinion Score – Visual Quality Objective).

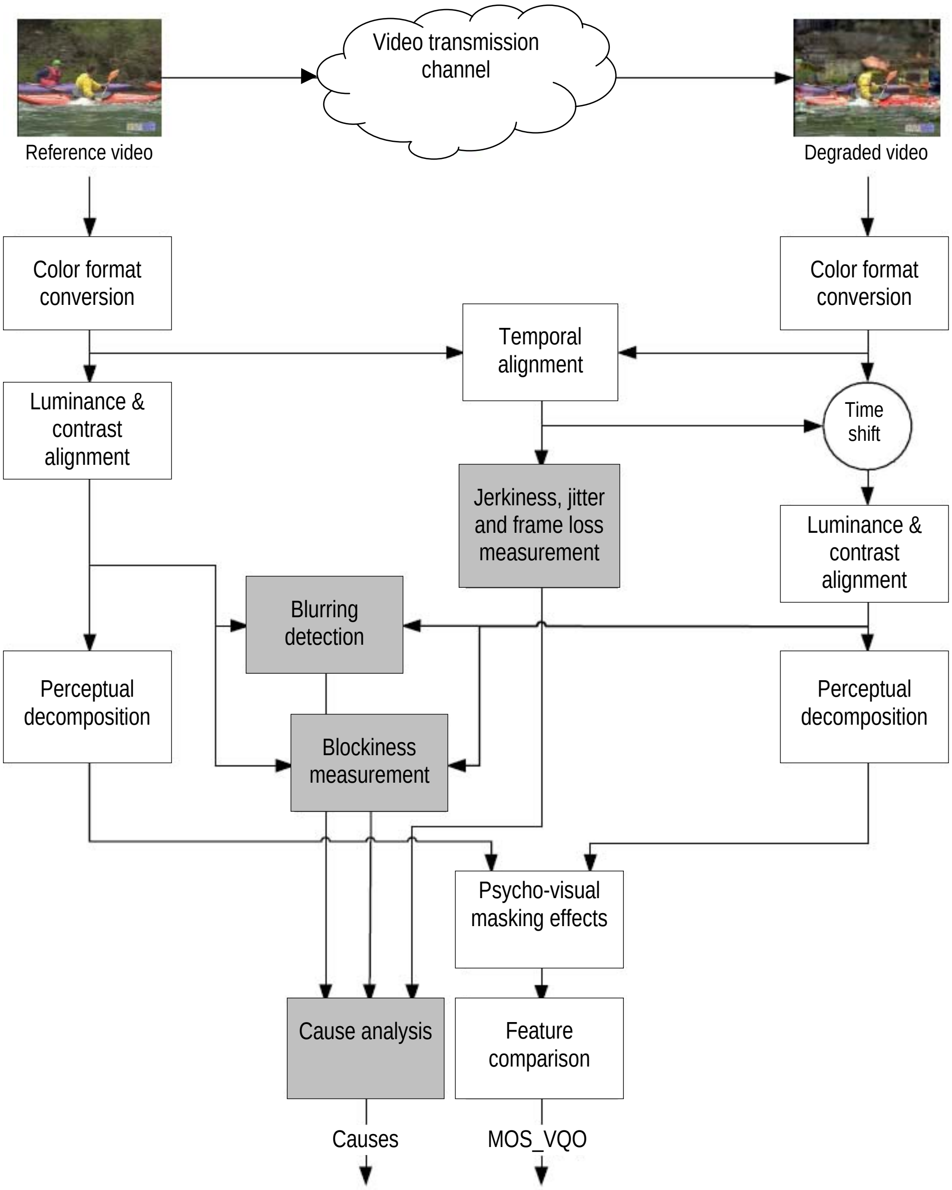
f – function corresponding to the application of the reckoning algorithm and conversion function of the results in MOS_VQO values.

$W_A(t)$; $W_B(t)$ – original video sample sent from side A (B).

$Z_A(t)$; $Z_B(t)$ –degraded video sample sent from side A (B), resulting from the transmission of the original sample $W_B(t)$ ($W_A(t)$).

There is currently no algorithm being recommended by international standard organizations to evaluate video quality. However, some measurement system manufacturers have developed their own algorithms taking into account the guidelines established by ETSI (ETSI TR 102 493 V1.1.1 (2005-08)) and by *VQEG* – Video Quality Experts Group ("Multimedia Group Test Plan", Draft Version 1.16, February 7, 2007). That is the case with SwissQual, AG, supplier of the testing and measurement system used in this study, which uses in its products its own algorithm, named VQuad - Objective Model for Video Quality Assessment.

Figure 25 shows the functional diagram of the VQuad algorithm. This algorithm is based on a full-reference type model, i.e., on a perceptual comparison of the degraded video sample with its reference. A reference video sequence (sample) is carried through the mobile network being tested. At the destination, the video sequence is captured and objectively validated through the perceptual comparison with the reference video sequence. It results in a global visual quality index (MOS_VQO) and other specific quality parameters (block distortion, blurring, jerkiness, level, PSNR, frame jitter, frame loss, lip-sync, etc.).



The global visual quality index, resulting from the application of the VQuad algorithm, is presented in a MOS (Mean Opinion Score) type scale from 1 to 5 named MOS_VQO (Mean Opinion Score – Visual Quality Objective), such as shown on Table 2.

In situations where each direction of the same call sends and receives several video samples $\{W_1(t), \ldots, W_n(t); Z_1(t), \ldots, Z_n(t)\}$, the *Call Video Quality* indicator is reckoned through the arithmetic average of the values obtained by applying the formula presented above to each pair of video samples, i.e.:

$$\text{Call Video Quality}_{\text{side A}}\left[\text{MOS\_VQO}\right]=\frac{f\left\{W1_B(t);Z1_A(t)\right\}+\ldots+f\left\{Wn_B(t);Zn_A(t)\right\}}{n}$$

$$\text{Call Video Quality}_{\text{side B}}\left[\text{MOS\_VQO}\right]=\frac{f\left\{W1_A(t);Z1_B(t)\right\}+\ldots+f\left\{Wn_A(t);Zn_B(t)\right\}}{n}$$

2.3 MEASUREMENT PROFILES[4]

Measurement profiles define a set of conditions that must the verified in order to correctly assess the services' quality and to guarantee the reliability of the tests. They also include process standardization and the definition of testing and measurement parameters, thus making it possible to perform analyses and compare results.

2.3.1 GENERAL FEATURES

Tests are performed automatically and using the Seven Five system (there is no human intervention or decision during the caring out of a test).

Voice tests are made by manually selecting the 2G (GSM) infrastructure, while video-telephony tests are performed with automatic selection of the 2G or 3G (GSM/UMTS) infrastructures.

Measurements are carried out in moving vehicles and with outdoor antennas (without gain). All collected parameters are geographically referenced and can be later shown by digital cartography.

2.3.2 COVERAGE

Network coverage assessment is made by measuring the downlink signal levels, RxLev (Received

[4] The measurement profiles presented here are supported on the technical specifications ETSI TS 102 250, namely part 5 (ETSI TS 102 250-5 V1.3.1 (2005-11)), and ETSI EG 202 057, namely parts 3 and 4 (ETSI EG 202 057-3 V1.1.1 (2005-04) and ETSI EG 202 057-4 V1.1.1 (2005-10)).

signal Level) for GSM and CPICH RSCP (Common Pilot Channel Received Signal Code Power) for WCDMA, along each analysed route.

Measurements are made through a RF Scanner device adapted and exclusively dedicated to this task, so that the measured signal levels correspond to the effective levels. The measurement equipment has the ability, at each point, to collect signal samples from all GSM and WCDMA radio channels used by the operators under analysis, at 1 and 2.5 second intervals, respectively. These samples are later analysed and only the best signal level results obtained for each point, technology and operator are considered.

Each measurement point is geographically referenced so that signal levels can be later represented on digital cartography, thus making it easier to visualise coverage levels of the mobile networks along the routes under study and to identify the locations with a poor or non-existent coverage.

2.3.3 TELEPHONY SERVICES

These services are evaluated end-to-end, using a “call” as the basic test unit.

Test calls are made between two mobile terminal devices, where at least one of them is of a mobile type (MS – Mobile Station or UE – User Equipment). This MS or UE moves along the studied route/location, and the calls originated from this terminal equipment are named MOC (Mobile Originated Call).

In order to minimize the uncertainty that is always accompanies measurement procedures, the second end of the test calls must present good performance levels and great stability. This end’s impact on the services’ performance indicators is intended to be the minimum. The solutions includes using fixed network terminals (ISDN) to assess the performance of voice services, and to use mobile terminals (UE) to assess the performance of the video-telephony service. UE are kept motionless in locations with proper (good) radio coverage, minimum interference and with a (virtually) 100% probability of accessing the video-telephony service. Calls originated on this end, at the ISDN or UE terminal, and ended at the mobile terminal that is under test, are named MTC (Mobile Terminated Call).

After a test call is established, the communication’s integrity is analysed (audio quality, for the voice service, and audio/video quality for the video-telephony service), alternately in both directions, regardless of the end that started the call.

With the purpose of comparing the performance of the several operators (benchmark), a fixed time frame is used for making each call during the test sessions. When a call malfunction occurs, either when establishing a call or in the conversation phase, the next call is only started when the next time frame arrives.

2.3.3.1 VOICE SERVICE

The analysis of the voice service, at a given location, includes the abilities to establish and to end calls.

Since the aim is to study the normal use of the voice service, the duration of test calls is close to the average duration of calls routed on the networks. Besides the call's own duration, the time frame considers time periods that make possible the setting up and ending of a call, and also a 20 second pause between consecutive calls, to prevent possible network constraints regarding signalling or mobility management.

The test parameters used for the analysis of the voice service present the following values:

- Relationship between MOC/MTC: 1/1;
- Duration of the test calls: 120 seconds;
- Time frame for making a test call: 180 seconds;
- Maximum call set up time: 20 seconds.

2.3.3.2 VIDEO-TELEPHONY SERVICE

The test parameters for the analysis of the video-telephony service are similar to those used for the voice service. The difference relies on the time frame, which is larger, since the time needed for establishing calls and negotiating audio/video communications between terminals is longer.

The test parameters used for the analysis of the video-telephony service present the following values:

- Relationship between MOC/MTC: 1/1;
- Duration of the test calls: 120 seconds;
- Time frame for making a test call: 210 seconds;
- Maximum call set up time: 20 seconds;
- Maximum audio and video communication set up time: 30 seconds

2.4 Test/Measurement and Post-Processing System

The Seven.Five/NetQual system, conceived and developed by SwissQual, A.G. (Switzerland), was used for measurements on the field and for their post-processing. This is a tool specifically designed for the analysis and benchmarking of mobile communications systems.

The system is made up of the following modules:

a. *7.5 Multi* – Mobile Unit, with an RF scanner and commercial mobile terminal devices (NOKIA 6680 terminals were used in the study carried out);

b. *Land Unit* – Fixed Unit, with ISDN interface cards, used for voice tests;

c. *Video Call Server* – Fixed Unit, with commercial mobile terminal devices (also NOKIA 6680), used for video-telephony tests;

d. *Media Server* – Fixed Unit, server used for data and video streaming tests;

e. *NQDI* – Post-processing System, for analysis and reporting of the completed measurements.

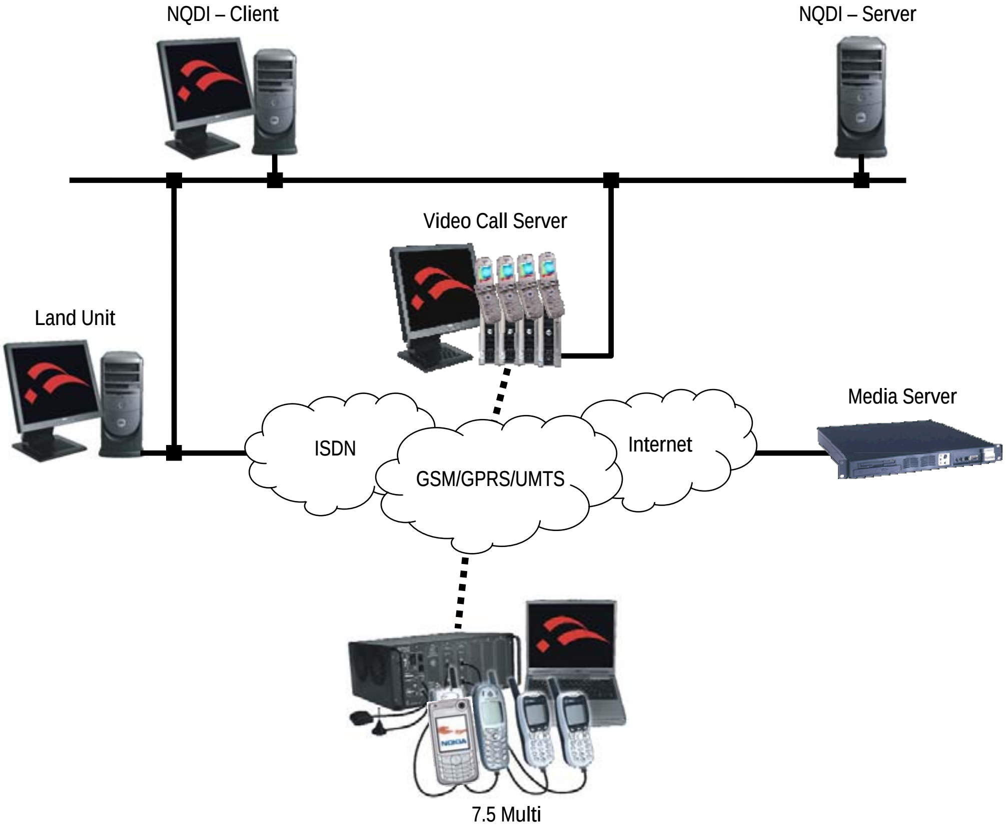


Figure 26 – Seven.Five/NetQual system architecture