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Speech Processing, Transmission and Quality Aspects (STQ);
Definition and implementation of VoIP reference point

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Contents

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Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ).

1 Scope

The speech quality perceived by the users is the result of combined impairments generated by the network but also by the terminals. This is why the characterization of the terminal performances is a major point in the deployment of telephony services. It is generally agreed that an ISDN access at 0 dBr represents a reference point for the PSTN, on which reference testing suites like TBR 21 [1] or TBR 38 [2] for instance can be implemented. For IP networks, it is really different, actually there is no reference access. For this reason, the characterization of VoIP end points is usually performed on a configuration using an ISDN access, which requires the use of an IP-PSTN gateway. Thus, the characterizations of VoIP end points performance are impacted by the performances of other equipments, generally the gateway.

The present document specifies the characteristics of an "IP reference device" for voice over IP services.

The present document provides information about the architecture of the system. The present document describes the different elements constituting the device and specifies the operation of each.

Information about a practical implementation of this device is annexed to the present document.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
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[1] ETSI TBR 21: "Terminal Equipment (TE); Attachment requirements for pan-European approval for connection to the analogue Public Switched Telephone Networks (PSTNs) of TE (excluding TE supporting the voice telephony service) in which network addressing, if provided, is by means of Dual Tone Multi Frequency (DTMF) signalling".

[2] ETSI TBR 38: "Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe".

[3] ITU-T Recommendation G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[4] ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation channels".

3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACG	Amplitude Control Gain
DHCP	Dynamic Host Control Protocol
EFR	Enhanced Full Rate
HATS	Head And Torso Simulator
IP	Internet Protocol

ISDN	Integrated Services Digital Network
LAN	Local Area Network
MGCP	Media Gateway Control Protocol
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
SIP	Session Initiation Protocol
TDM	Time Division Multiplexing
VLAN	Virtual LAN
VoIP	Voice over Internet Protocol
WAN	Wide Area Network

4 Overview of the context

During performance evaluations of IP terminals or end points, one of problems encountered is the lack of a reference point or reference access on the IP network as the one existing on the PSTN with the ISDN. This lack of reference points on the IP network penalizes and makes it difficult to characterize the performance of IP equipments. Generally, the characterization of an IP terminal is performed through an ISDN access connected to the IP network by a VoIP gateway (figure 1).

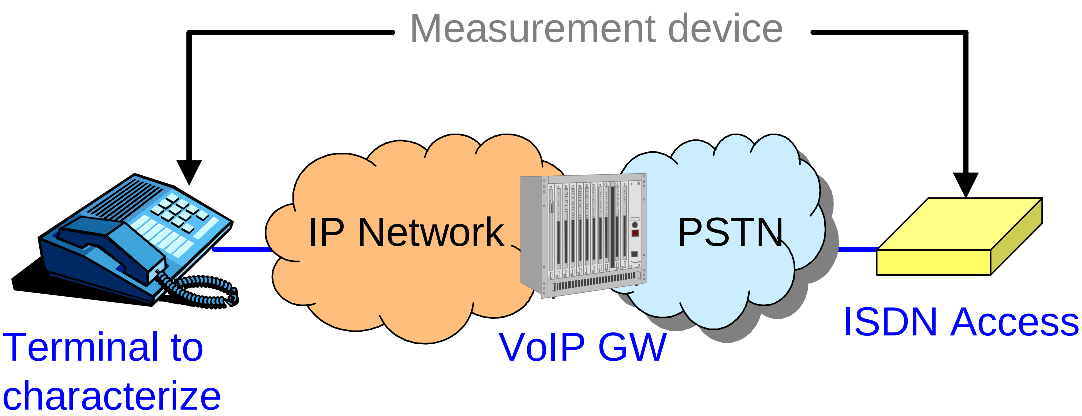

Figure 1: Standard configuration for performance evaluations of IP terminals

The implementation of an ISDN access for the characterization of a terminal involves contaminations of results by the performances of the VoIP gateway used to connect the IP network to the PSTN. Indeed the sending characteristics of a terminal to be evaluated are seen as a whole with the characteristics of the VoIP gateway in the IP to PSTN direction. In the same way, the receiving characteristics of the terminal are seen as a whole with the characteristics of the VoIP gateway in the PSTN to IP direction. It is obvious that this situation creates some problems when the performances of the gateway are badly known or out of control. The idea to improve the situation is to bring back the reference point at the level of the IP side the voice over IP gateway (figure 2). So with the implementation of a "reference gateway", the reference point moves from the switched network to the IP network.

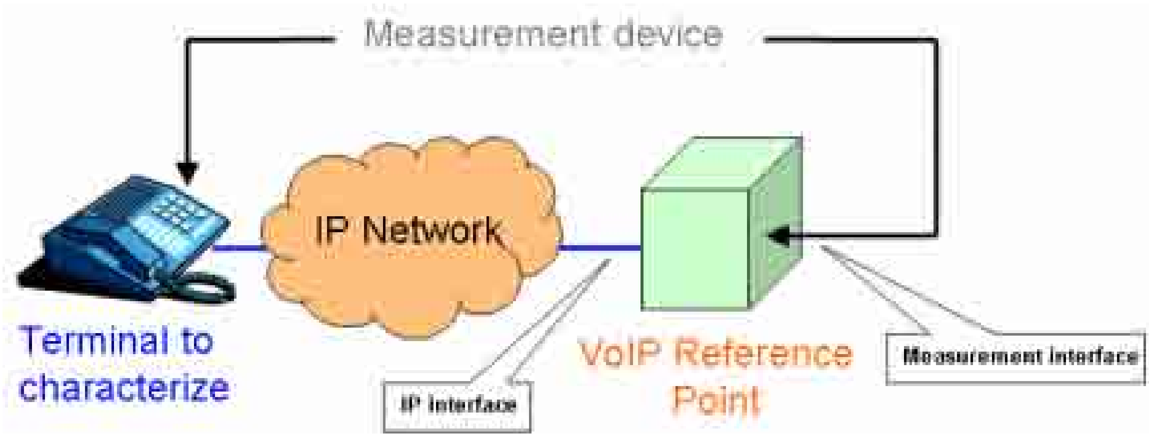

Figure 2: Proposed configuration for performance evaluations of IP end points: use a reference point on the IP network

If the characterization of VoIP services or telephony architectures does not generate major difficulties, for the characterization of VoIP equipments, there are several reasons to develop a reference point on IP networks:

- The increase of development and marketing of IP terminals like IPphones, videophones or domestic gateways introduced an important need of characterization for these new equipments. For this reason it becomes urgent to be able to evaluate the performances of these terminals compared to a same measurement point, whose performance is perfectly controlled.
- For VoIP services, a dissymmetry between the behaviour and characteristics of the two transmission ways is frequently noticed. This is particularly true when different kinds of terminals (e.g. an IPphone and an IP to PSTN network gateway) are implemented on the same link. An IP reference point must give the possibility to determine, absolutely and independently, the sending and receiving performances of terminals. So with the reference device it should be possible to identify the differences of behaviour or characteristics.
- Similarly, on IP communications problems on the speech signal level are frequently noticed. These problems often come from a bad implementation of the telephony applicative part and in particular a bad implementation of the codecs. The implementation of a reference element on the IP network makes it possible to check the compliance of sending and receiving loudness ratings.
- Frequency responses, loudness ratings, sidetone, distortions, noise levels are essential characteristics for telephony terminals. Therefore, a reference point on an IP network is a device which allows this type of characterization independently of any other equipment performance (Network gateway for example).
- For VoIP termination equipment, the evaluation of intrinsic speech quality (distortion of the voice signal, reduction and delay of acoustic echo, etc.) is currently very difficult or impossible. The evaluation of the performance of this equipment is impacted by the characteristics of the voice over IP system present at the other end of the IP segment of the transmission path. The development of a reference point makes it possible to get rid of this difficulty and to determine specific performances to the tested terminal.

5 Concept of VoIP reference point

When you move the reference point from the PSTN to the IP network, the reference point became a sort of "reference gateway".

The concept of the VoIP reference point is presented on figure 3.

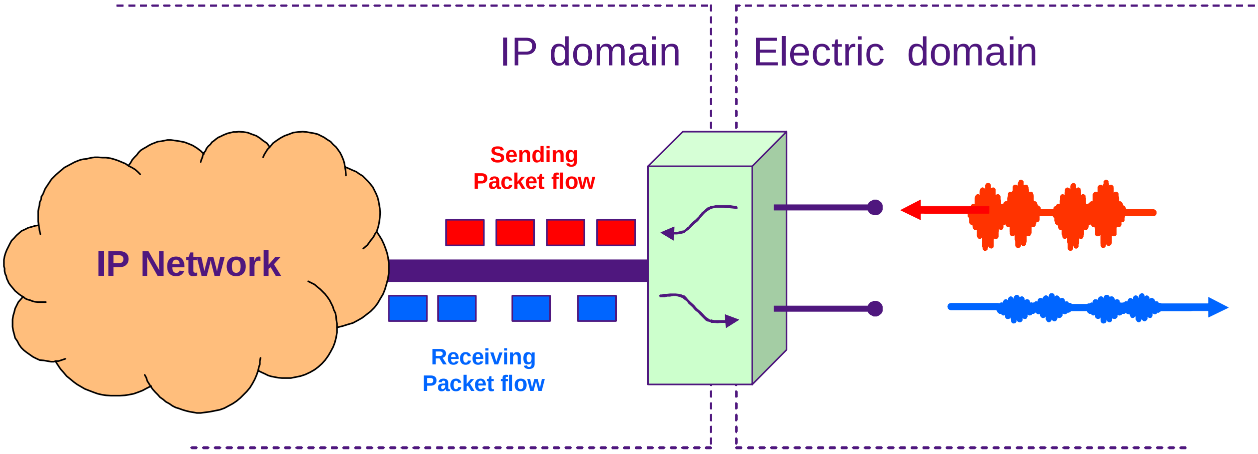

Figure 3: Presentation of the concept of the VoIP reference point

It is an interface between 2 domains, interface with controlled and well known characteristics.

The reference point is a device making the connection between an IP packet domain and an electric domain for speech signals.

This reference device is characterized by four important characteristics:

- Compliance codec implementations.
- Conform amplitudes for speech signals.
- Control of the processing delays.
- Tracing any characteristic modification.

These four characteristics ensure:

- no specific impairments due to a bad implementation of codecs;
- a correct amplitude for the speech signals send on the IP network and no amplitude modification for the speech signals receive from the IP network;
- the knowledge of the respective processing delay of the sending and receiving parts to make possible the determination of the processing delays of terminals to be tested;
- a recording of all events linked with characteristic modifications to make possible a correct interpretation of terminal evaluation results.

6 Architecture of the reference point

6.1 General characteristics of the VoIP reference point

6.1.1 Network interface

As IP telephony terminal, the reference point must have an interface to allow connection to IP networks. It must be able to reach with a wire connection to different types of network: dedicated LAN, company LAN, WAN, etc.

The privileged interface to comply with these requirements is the Ethernet interface. The specificity of this device as a reference element in IP network implies having only one type of interface and excluded wireless interfaces like Wifi.

6.1.2 Call establishment

This device must be able to establish a communication with any VoIP terminal. In particular, it must be able to be in communication with IP terminals such as IPphones, softphones and the domestic gateways.

The reference point must also communicate with terminals of the PSTN like POTS and ISDN equipment. It must also be able to be in communication with 2G and 3G mobile. At the opposite of IP to IP calls, these communications are established via VoIP network gateways.

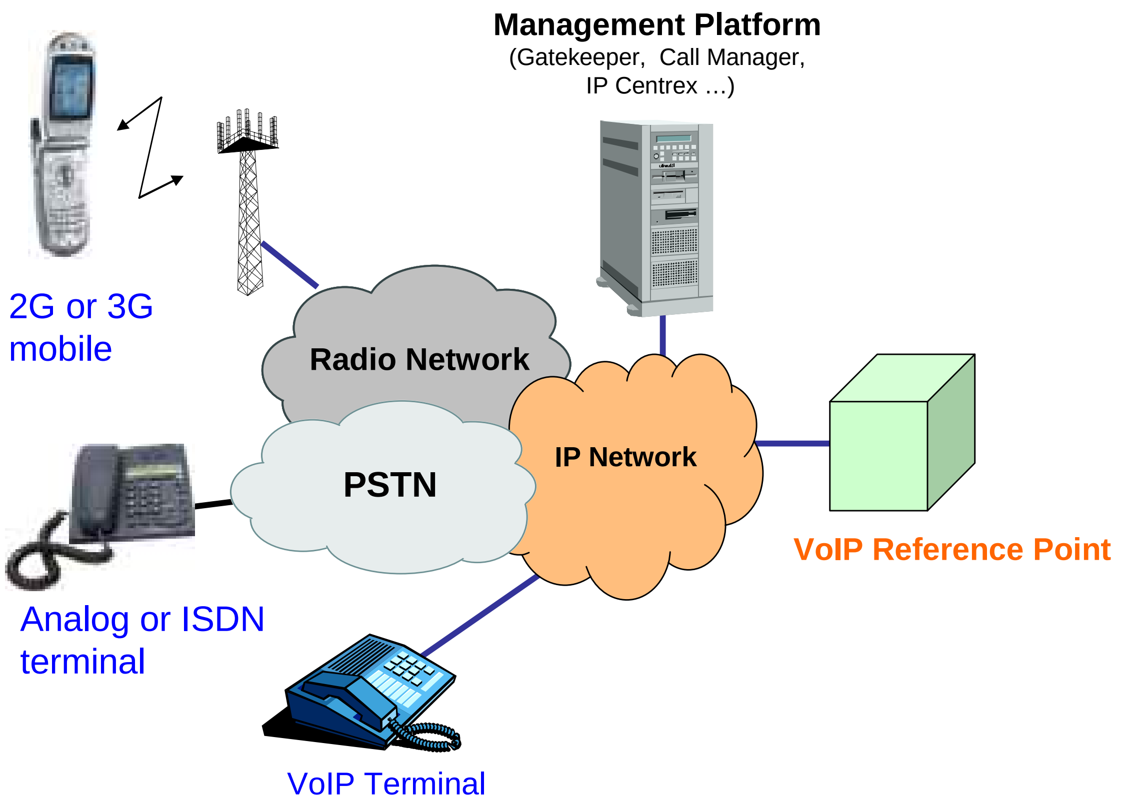

Figure 4: Inter connection of the VoIP reference point with other terminals

6.1.3 Registration on VoIP services platforms

This device must be able to be registered on any VoIP platform. To be used under the greatest possible number of conditions, it is essential that the VoIP reference point has the functionalities of registration on network or service equipment like the gatekeeper or the call manager.

6.1.4 Performances

Taking into account the reference aspect of the device, it is necessary that the characteristics of the VoIP reference point are perfectly known and controlled. The characteristics must be highest as possible.

They must be known and if there is modification of the characteristics during the use of the device, the modifications must be notified and traced by writing for example in a log file with time-stamping of the events.

6.2 Operating modes of the device

The reference point is dedicated to running on two specific configurations.

6.2.1 Connection with a terminal

Figure 5: Inter connection of the VoIP reference point with VoIP terminals

The reference point connected to a VoIP terminal is the principal configuration to use the reference device (figure 5). This makes an improvement compared to a characterization via an ISDN access because it is possible to characterize the equipment under test without contaminations of results by the performance of the VoIP gateway used to connect the IP network to the PSTN.

6.2.2 Connection with a Gateway

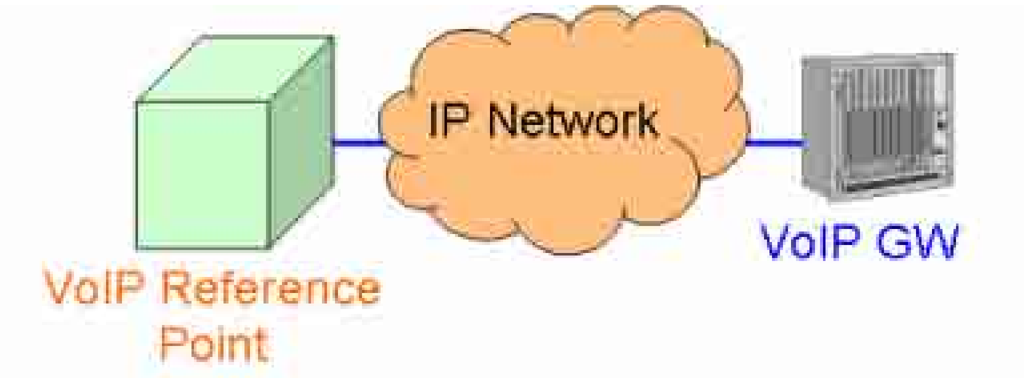

Figure 6: Inter connection of the VoIP reference point with Gateways

The reference point connected to a Gateway is a case close to the previous configuration (figure 6). In this configuration, it is possible to characterize the non-IP part of an architecture or telephony service.

6.3 Specific characteristics of the VoIP reference point

6.3.1 IP Interface

Figure 2 points out the IP interface on the synoptic of measurement configuration.

6.3.1.1 Wire interface

The reference point must have an interface to perform the connection with the IP network. The usual interface for the device is the Ethernet because it accommodates most VoIP terminals (IPphone, PC with softphone, domestic gateway adapter, etc.).

6.3.1.2 Wireless interface

A wireless interface could also be implemented on the reference device, but caution has to be exercised for avoiding impairments due to radio interface (for example limitation of bandwidth) and characteristic variations.

6.3.2 Measurement interfaces

Figure 2 points out the measurement interface on the synoptic of measurement configuration.

6.3.2.1 Analog interface

It is an electrical interface with input and output access on the reference point. The electrical signal on the interface is analogue. The main use of the interface is to connect measurement devices for terminal characterizations.

6.3.2.2 Digital interface

This second interface on the reference point has also an input and an output for signals. The electrical signal on the interface is digital. The interface can be used to play or to record audio signal from an external digital device like a hard disk.

6.3.3 Call establishments

To establish calls, the reference point must have all the functionalities of a end VoIP terminal.

6.3.3.1 Configuration

The point of reference is able to run with a fixed IP address, with a dynamic IP address (use of DHCP protocol for IP address assignment) and with fixed IP address with DHCP server. It must be possible to assign to the reference device a phone number (an E.164 number or a short one).

It is also possible to assign an alias.

6.3.3.2 Registration

The reference point shall support Internet Protocol version 4 (IPv4). However, it could also support Internet Protocol version 6 (IPv6).

For inter operating with any telephony service, the reference point must have the capability to be registered on any service platform. Thus it should be possible to define:

- IP address of the gatekeeper.
- IP address of the default gateway.
- A port number.
- A name of a VLAN.

- An address of a proxy.
- And so on.

6.3.3.3 Dialing

The reference point can establish a communication as a caller and a called end point.

Taking into account, that it is possible to assign an IP address, a phone number and an alias, the communication with the remote end terminal can be established either by IP address, phone number or alias. IP addresses, phone numbers and alias can also be used by the remote end terminal to establish a communication with the point of reference.

6.3.4 Signalization protocols

To be able to run on the greatest number of platforms of service with the greatest number of terminals, in the greatest number of configurations, the reference point shall implement the greatest possible number of signalization protocols such as:

- H.323.
- SIP.
- MGCP.
- And so on.

Non standardized signalization protocols (like Skinny) could also be implemented on the reference point.

6.3.5 Codecs

Any codec implemented by the reference point must respect the algorithms defined by the standards or by the experts. This condition is absolutely necessary to give the character of reference to the device.

The reference point implements narrow band and wide band codecs. These codecs could be either standardized or not.

For the same reasons as for the signalization protocols, the reference point implements maximum of codecs with different bite rate:

- G.711 Alaw.
- G.711 µlaw.
- G.726 (40 kbps, 32 kbps, 24 kbps and 16 kbps).
- G.729 main.
- G.729 A.
- G.723.1 (5,3 kbps and 6,3 kbps).
- G.722.
- EFR.
- AMR (23,85 kbps, 23,05 kbps, 19,85 kbps, 18,85 kbps, 18,25 kbps, 14,25 kbps, 12,65 kbps, 8,85 kbps and 6,6 kpbs).
- And so on.

6.3.6 Clock accurracy

The clock accurracy of the VoIP reference point should be as good as or better than 10 ppm.

For measuring clock skew the reference point should be at least 10 times better than the clock accurracy of the device under test.

Alternatively the option of adjusting the clock of the reference device to the clock of the system under test may be provided in order to conduct measurements with low delay drift during the measurements.

6.3.7 Jitter buffer

The purpose of the jitter buffer is to ensure that during the measurement all frames can be played out regularly without any loss or duplication. The size of the jitter buffer has to be large enough to fulfil this requirement.

One solution is to set the fill-in time to have the middle of the memory full at the beginning of the measurement. In addition it is recommended to reach the half filled state in advance of each new measurement.

6.3.8 Delay characteristics

The processing delay of the sending of the VoIP reference point due to the elements between the measurement interface and the network interface (like coder and IP stack) must be perfectly controlled. The processing delay at the reception due to the elements between the network interface to the user interface (like IP stack, jitter buffer and decoder) must be also perfectly controlled. The processing delays due to the sending and receiving part of the reference device must be well known. The control of these different processing delays allows to determine the sending and receiving processing delays of the system under test connected to the reference point. The sending and receiving processing delays of the system under test will be determined by the measurement of the end to end delay corrected by the delays inserted by the reference point.

6.3.9 Full duplex performances

The reference point is a full duplex access point without any additional signal processing like echo canceller, ACG and so on, except speech coding.

6.3.10 Configuration and control of the device

The reference point should have settings and control interfaces.

Via these interfaces it is possible to configure the reference device. For example it is possible to define the signalization protocol, the IP address, the codec implemented and so on.

It is also possible to configure if necessary the port number, the default gateway, the VLAN identification, etc.

A specific control should be implemented to select codecs to be used. If it is necessary to perform the reference point into a specific configuration with a specific codec (for example the codec chosen by the operation of this VoIP service) it could be possible to select one specific codec into the list of codecs available on the reference device.

A control interface should allowed the control of call establishment or setup and release the communication.

Annex A (informative): Example of an implementation of the VoIP reference point

A.1 Introduction

France Telecom R&D has developed a version of VoIP reference point. This device is used to characterize VoIP terminals in the same manner as the ISDN access is used to characterize analog or digital PSTN terminals.

The VoIP reference point developed by France Telecom R&D is based on software embedded on a computer platform (figure A.1).

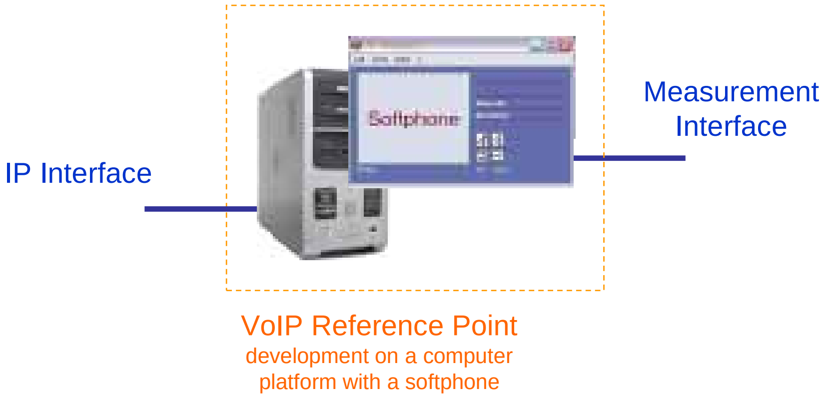

Figure A.1: Principle of the implementation

This solution has been developed because computers natively have an IP interface and an analog electrical interface (i.e. the sound card).

A.2 Advantages of the implementation

The advantages of a reference point based on a softphone embedded on a computer are mainly:

- Quick development of the device, because all elements for call establishment like dialler and codecs are available in a softphone. In addition, the softphone or the operating system of the computer has the IP stack necessary for terminal registration on service platform and for signalization control during communication establishment.
- Very cheaper solution because there is no specific hardware development. This solution requires only a few software modifications on the telephony application (softphone).

A.3 Technical characteristics

A.3.1 General architecture

The elements of the VoIP reference point implemented by France Telecom R&D are (figure A.2):

- a computer with the Operating System Windows XP;
- a softphone;
- an external sound card;

- an adapter unit.

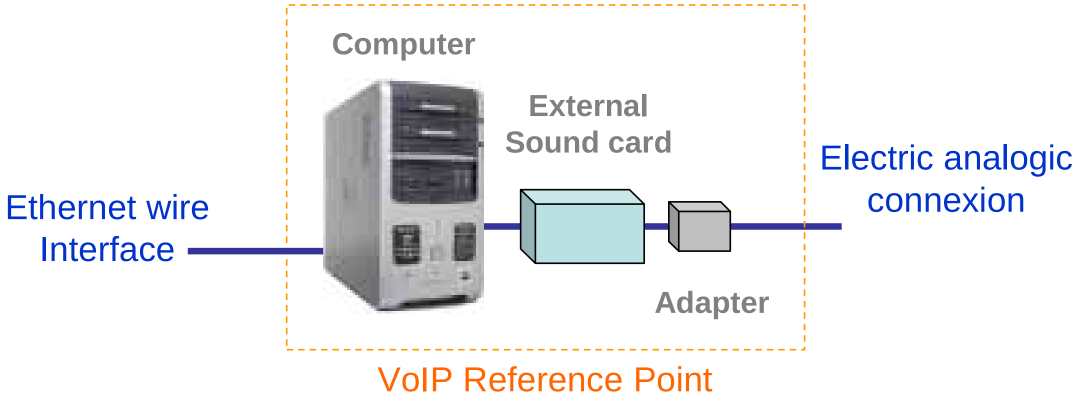

Figure A.2: General architecture of the VoIP reference point based on a computer

In order to enable assessments in the field, the target platform chosen is a laptop.

The softphone implemented is a **telephony** and **visiophony** application developed by France Telecom.

As a "standard" sound card is installed in the Laptop is not adapted to an implementation of VoIP reference point as it offers too poor quality performances, an external high quality sound card is used for the measurement interface. This card is connected to the PC via an USB port.

An adapter unit is connected to the analog access of the external sound card. The objective of this module is to adjust the level of the speech signal at the input and output of the sound card.

A.3.2 Interfaces

The device includes an Ethernet wire interface for the connection to the IP network. This Ethernet interface is performed by the network card of the computer. For the measurement interface, the VoIP reference point uses an analogue electrical connection. The measurement interface is performed by the external sound card associated to an adapter unit.

In addition to the analog accesses, the external sound card has also digital accesses, but these accesses are not used in this version of VoIP reference point.

A.3.3 Registration and configuration

The possible registration of the VoIP reference point on service platforms like gatekeeper or callserver is carried out via the setting interface of the softphone.

As the registration, the selection of codec (to use during the call) is performed via the setting interface. The setting menu proposes a list of codecs supported by the softphone. The choice of codecs is carried out via selection of check boxes.

A.3.4 Call establishments

Call establishments are performed via the dialer of the softphone. This dialer allows the use of E.164 phone number, IP address and alias to establish communications.

A.3.5 Signalization protocols and codecs

The reference point operates with the signalization protocols H.323 and SIP. Theses two protocols are embedded in the softphone.

The codecs available in the implementation of VoIP reference point are:

- G.711 Alaw and µlaw.
- G.729 main and G.729 A.
- G.723.1 (5,3 kbps and 6.3 kbps).
- G.722.
- G.729.1.
- AMR WB.

Note that proprietary codecs may also be implemented, as needed.

A.3.6 Jitter buffer

No specific modification has been done on the jitter buffer of the softphone. Nevertheless all modifications associated to the management of this buffer (like size modification, erased or duplicated audio frames management) are plotted out in a log file. In the same manner, all degradations on speech signal (like erased audio frames) are notified in real time on the computer screen via a specific display.

This notification of events associated to the jitter buffer management is an important feature for VoIP reference point use. This information displayed in real time enables to identify critical moments when the device generates degradations (erased or duplicated audio frames). In particular, it is the case when the softphone adjusts the time drift.

A.4 Calibration

A.4.1 Softphone calibration

Reference audio files, with different defined levels, stored on a disk, are used to calibrate the softphone and to check the level stability of the telephony application.

On the ingress direction, the reference audio file is packetized and sent on the IP link via the softphone and the Ethernet interface. A sniffer captures the IP packets; the audio signal is rebuilt and saved in a received file on the sniffer. Then a level comparison can be done between the reference and the received audio files (see figure A.3).

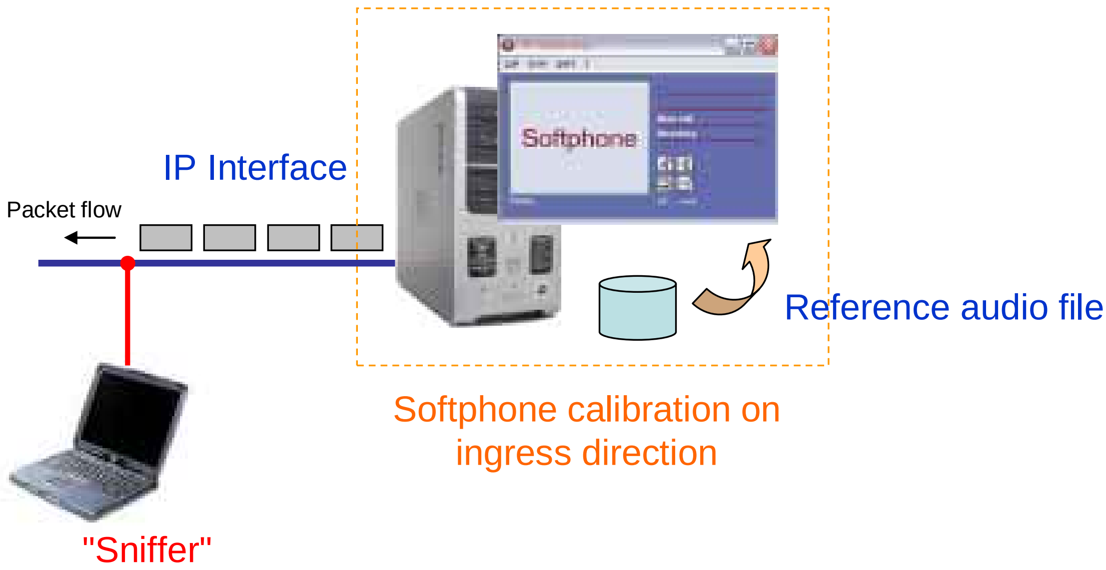

Figure A.3: Synoptic of the softphone calibration on ingress direction

On the egress direction, the packet flow (input audio signal) sent to the reference point is captured by a sniffer; the audio signal is rebuilt and saved in a file on the sniffer. At the same time, this audio signal is decoded by the softphone and saved on the disk of the reference point. Then a similar level comparison can be done between the input and the received audio files (see figure A.4).

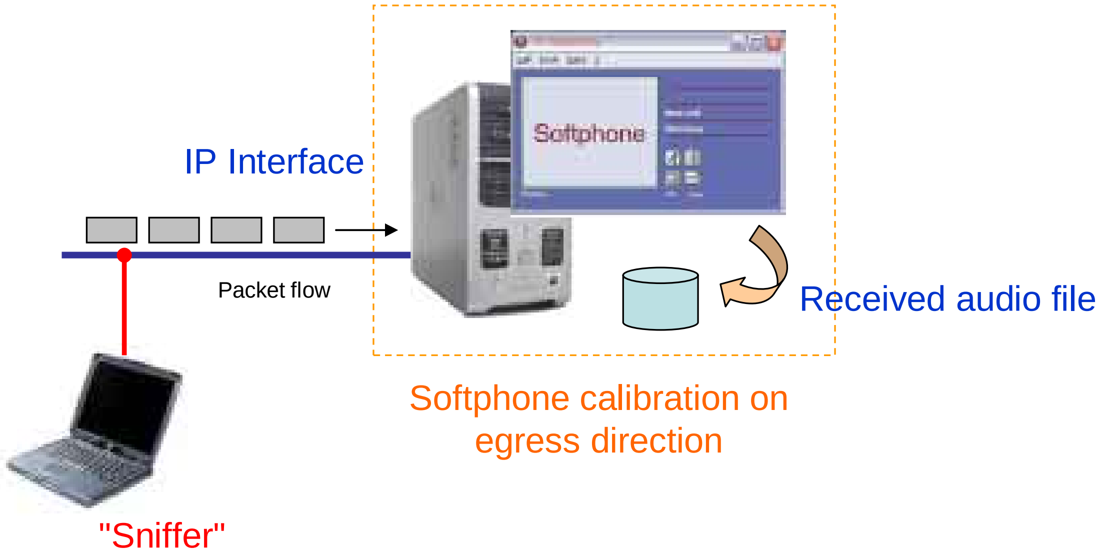

Figure A.4: Synoptic of the softphone calibration on egress direction

This methodology allows to check the level of transparency of the softphone.

A.4.2 Reference point calibration

The measurement interface is calibrated in the same way as the one used for the softphone calibration. A communication is established between the reference point and an ISDN access located on an other Gateway (see figure A.5). The gateway performance should be "the best possible", the ISDN access should conform to ITU-T Recommendation G.711 [3] and ITU-T Recommendation G.712 [4]. An audio-ISDN adapter is connected to the gateway to allow call establishment and to produce an analog audio access.

So, an analog electrical speech signal is sent from the ISDN adapter to the reference point. The level comparison between the sent and received signals (on the analog access of the reference point audio adapter) is possible. A broadband signal (e.g. pink noise) can be used to check the frequency response characteristics of the VoIP reference point. The comparison allows to adjust the level and the frequency response characteristics on the TDM output access of the VoIP reference point. In case of deviations from the desired 0 dB reference frequency response a correction of the frequency response can be applied. A noise measurement should be performed in order to ensure that the system noise is below the noise requirements of the system under test.

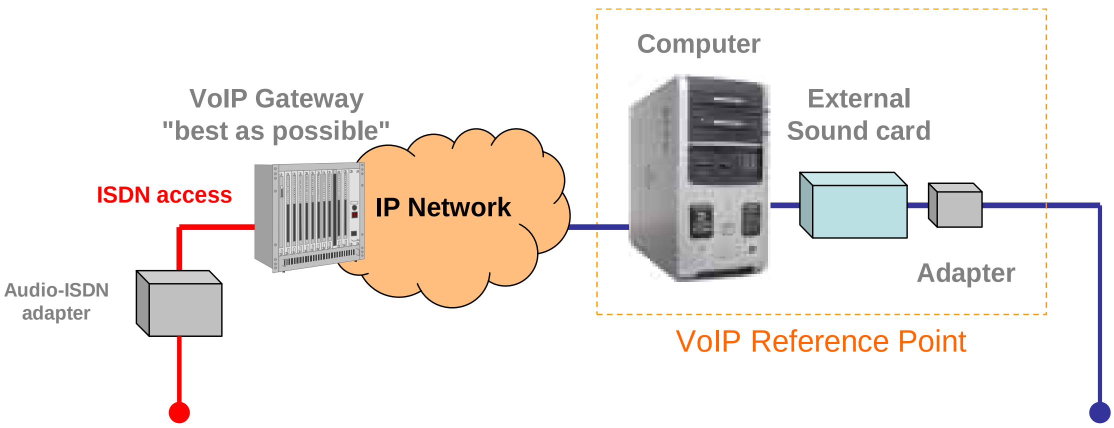

Figure A.5: Synoptic of the VoIP reference point calibration

Similarly, an analog electrical speech signal is sent from the input VoIP reference point to the audio-ISDN adapter. A level comparison between the sent and received signals is also possible. This comparison allows to adjust the level signal on the TDM input access of the VoIP reference point. Again, the frequency response characteristics and the noise levels are verified.

This methodology allows to check the level transparency of the reference point.

A.5 Use for terminal characterization

The main use of the VoIP reference point is the characterization of VoIP terminal. As shown on figure A.6, it is possible to connect on the measurement interface an external analyser. This analyser is also connected to an artificial head/HATS.

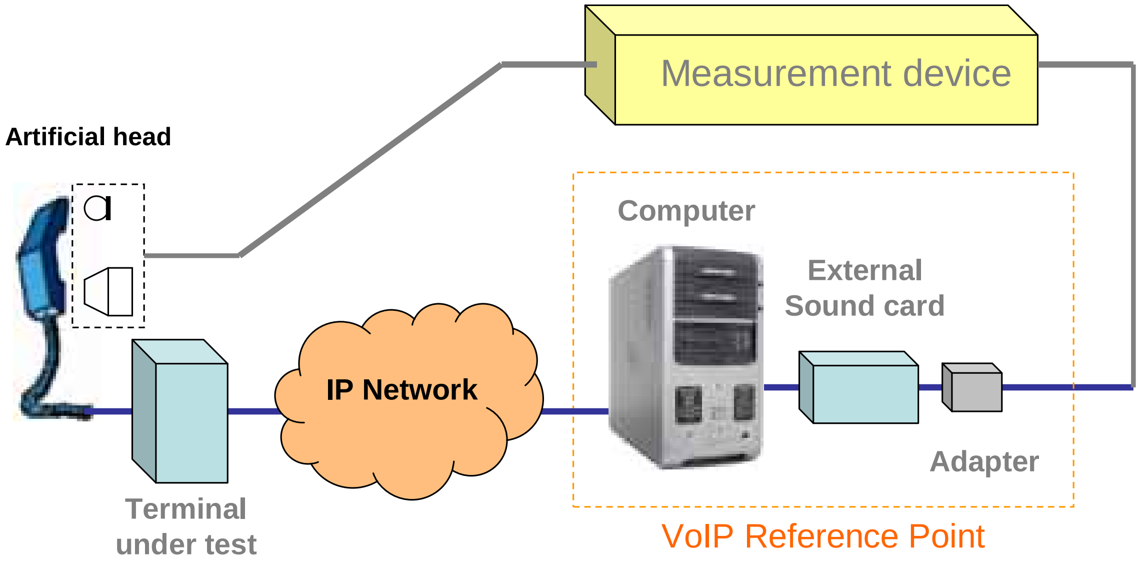

Figure A.6: Implementation of the VoIP reference point for terminal characterization

Under this condition, the following metrics have been implemented in the VoIP reference point:

- SLR (Send Loudness Rating).
- RLR (Receive Loundness Rating).
- TCLw (Terminal Coupling Loss weighted).
- Send frequency response.
- Receive frequency response.
- Linearity range for SLR.
- Linearity range for RLR.
- Send and receive idle noise.

Other metrics could be implemented in the VoIP reference point, such as:

- Terminal behaviour in double talk situation.
- Switching quality.

A.6 Limits of the implementation

The main disadvantage of this type of implementation (computer + softphone) is the lack of synchronization between the softphone and the operating system running on the computer. The lack of synchronization introduces an inaccuracy in the processing delay associated with the reference point (this inaccuracy is about 40 ms, very variable, and depending on PC platform and the sound card chosen). The processing delay to cross the device from IP to measurement interface is indeed very variable between different calls. We noticed the same behaviour in the other way round.

Under these conditions, delay performances of the reference point can not be controlled, with the appropriate accuracy. As a consequence, parameters needing time accuracy, such as sending delay and receiving delay, cannot be assessed accurately.

History

Document history		
V1.1.1	February 2007	Publication