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TECHNICAL SPECIFICATION

Speech and multimedia Transmission Quality (STQ); Digital reference point for speech communication in packet based networks

Reference

RTS/STQ-315

Keywords

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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Introduction

The present document focuses on establishing a 0 dBm0 IP reference point to standardize the evaluation of VoIP devices in modern IP-based networks. With the transition from analog and ISDN networks to all-IP systems, it highlights the need for clear, consistent testing methods to ensure high speech quality. The reference point will enable precise, independent assessments of key performance indicators such as frequency response, loudness, distortion, and latency, without interference from external network components. The present document aims to provide reliable and comparable results for VoIP terminal evaluations.

1 Scope

The present document introduces the concept of the 0 dBm0 IP reference point, which enables precise and comparable measurements of VoIP speech processing and device performance, including VoIP terminals. In modern telecommunications, ensuring high-quality speech communication over IP-based networks is a critical requirement. With the transition from analog and ISDN networks to all-IP systems, it is essential to establish new reference points and standards for evaluating the quality of Voice over IP (VoIP) devices.

The present document focuses on developing a framework for evaluating VoIP device characteristics in a controlled environment. Such a framework minimizes the influence of external network factors, ensuring consistent analysis of parameters such as frequency response, loudness, distortion, and latency.

However, the present document does not mandate the implementation of a physical 0 dBm0 IP point; instead, it provides guidelines and requirements for its definition and application.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found in the ETSI docbox.

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The following referenced documents are necessary for the application of the present document.

[1] Recommendation ITU-T P.10/G.100: "Vocabulary for performance, quality of service and quality of experience".

[2] Recommendation ITU-T G.100.1: "The use of the decibel and of relative levels in speechband telecommunications".

[3] Recommendation ITU-T G.111: "Loudness Ratings (LRs) in an international connection".

[4] ETSI ES 202 718: "Speech and multimedia Transmission Quality (STQ); Transmission Requirements for IP-based Narrowband and Wideband Home and Network Media Gateways from a QoS Perspective as Perceived by the User".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

Not applicable.

3 Definition of terms, symbols and abbreviations

3.1 Terms

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

0 dBm0 IP reference point: standardized reference level used for testing and evaluating VoIP devices in IP networks

codec: device or program that compresses or decompresses digital media, especially audio or video signals

decibels relative to overload (dBov): unit of measurement for signal levels in digital systems, relative to the maximum possible signal level

distortion: unwanted alterations to audio signals during transmission

frequency response: measure of how effectively a VoIP device reproduces audio frequencies

Integrated Services Digital Network (ISDN): digital network technology for transmitting voice and data over traditional phone lines

IP networks: networks based on Internet Protocol for transmitting data across interconnected devices

jitter buffer: device or software component that compensates for packet delay variations in VoIP transmissions

latency: delay in transmitting data, often measured in milliseconds, affecting communication quality

loudness: perceived intensity of sound, essential for ensuring clear VoIP communication

Public Switched Telephone Network (PSTN): traditional analog telephony network used for voice communication

Voice over IP (VoIP): technology allowing voice communication over Internet Protocol networks

3.2 Symbols

Void.

3.3 Abbreviations

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

dB	deciBel
dBm	dB [relative to] 1 mW
dBm0	[absolute level] dBm [relative to] 0 [dBr]
dBov	dB relative to overload [point]
dBr	dB relative [level]
GW	GateWay
IP	Internet Protocol
ISDN	Integrated Services Digital Network
MGCP	Media Gateway Control Protocol
MGW	Media GateWay
mW	milliWatt
PSTN	Public Switched Telephone Network
SIP	Session Initiation Protocol
VoIP	Voice over IP

4 Units and Measurement

4.1 Introduction

The requirements stipulated in Recommendations ITU-T P.10/G.100 [1], G.100.1 [2] and G.111 [3] apply.

4.2 The Unit "dBr"

The dBr unit describes relative signal levels between two points in a path, using one point as a reference (0 dBr). It facilitates network design by enabling consistent characterization of equipment power handling and signal levels.

4.3 The Unit "dBm0"

The dBm0 unit expresses absolute signal levels at the 0 dBr reference point. It standardizes transmission measurements, allowing comparison of different signals relative to this fixed reference.

4.4 The Unit "dBov"

The dBov unit is used in the digital domain to measure levels relative to the overload point of a system. Unlike dBm0, dBov focuses on the codec's maximum power limits.

4.5 Relationship between Units

The relationship between dBm, dBm0, and dBr is defined as:

dBm = dBm0 + dBr

dBm0 = dBm - dBr

For a digital system which has an overload point x_{over}, the maximum signal power will be P_0 = 1,0. In this case, the power level for a digital signal in decibels relative to the overload point (dBov, where the characters "ov" arbitrarily mean digital overload signal level) is defined by:

$$L_{ov} = 10\ log_{10}\left(\frac{P}{P_0}\right)\ \text{dBov}$$

Relationship between overload (dBov) and maximum levels (dBm0):

The relationship between dBov and dBm0 (decibel milliwatts at zero level) allows for the conversion of signal levels across different contexts. This is particularly useful for speech codecs, where input levels are specified relative to dBov, while transmission levels in telecommunications networks are expressed in dBm0.

The conversion between both representations can be generically expressed as:

y(dBm0) = z(dBov) + C

with factors y and z and an additive constant of C.

The **overload point** is defined to establish a consistent and standardized reference for signal levels across different codecs and network implementations. This ensures compatibility and enables accurate measurement and evaluation of codec performance. Without a defined overload point, comparing signal levels and ensuring uniformity in testing VoIP and telecommunications devices would be challenging.

5 Context

5.1 Adjusting the Correct Reference Level

In all-IP networks, identifying the correct reference point for testing is critical. Historically, analog PSTN or ISDN networks used simple 0 dBm0 reference points in the digital path. However, with the disappearance of these technologies, new approaches are required.

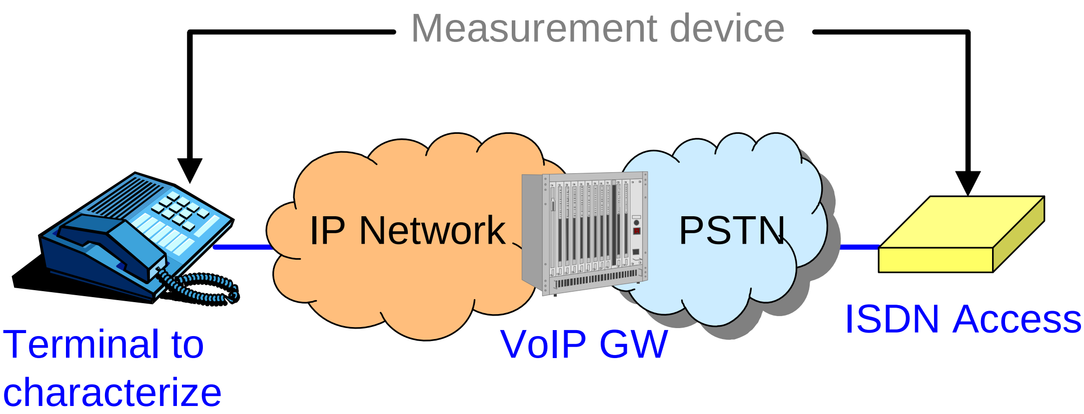

Figure 1: Historical standard configuration for performance evaluations of IP terminal

5.2 Historical Use of the Overload Point

The overload point is a key concept in telecommunication systems, historically used to determine the maximum allowable signal power level before distortion occurs in the system:

- Analog Systems: In traditional analog systems, the overload point was defined in terms of voltage or power (e.g. 0 dBm). It provided a clear physical reference for signal measurement and calibration.
- Digital Systems: In digital telecommunication, the overload point is defined relative to the maximum digital power handling capacity, measured in dBov. It marks the threshold beyond which signal clipping and distortion occur, compromising audio quality.
- Legacy Networks: PSTN and ISDN networks used the overload point as a reference for defining signal processing levels and assessing system performance under high signal conditions.

5.3 Current Challenges with the Overload Point in IP Networks

In IP-based systems, the relevance of the overload point has diminished due to several factors:

- Codec-Dependent Performance: Different codecs have varying overload characteristics, making a single overload point less universally applicable.
- Jitter and Packet Loss: The nature of IP transmission introduces variables like jitter and packet loss, which impact signal quality independently of the overload point.
- Dynamic Range Variability: Modern audio processing systems often feature expanded dynamic ranges, reducing the emphasis on overload thresholds.

5.4 Move to Reference Gateway Approach

To address these challenges, the reference gateway shifts the evaluation point entirely into the IP domain, ensuring independent and controlled performance testing of VoIP devices.

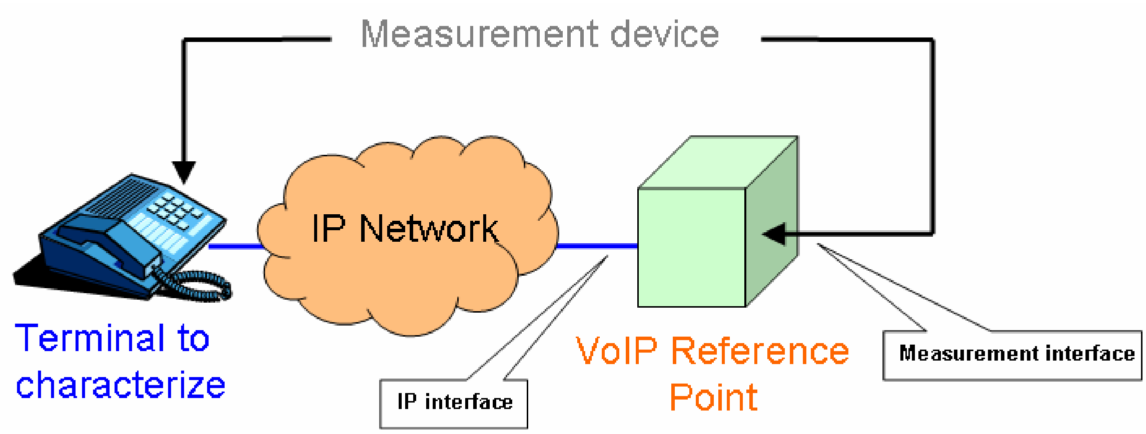

Figure 2: Configuration for performance evaluations of IP end points utilizing a reference point on the IP network

6 Concept of the 0 dBm0 IP Reference Point

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

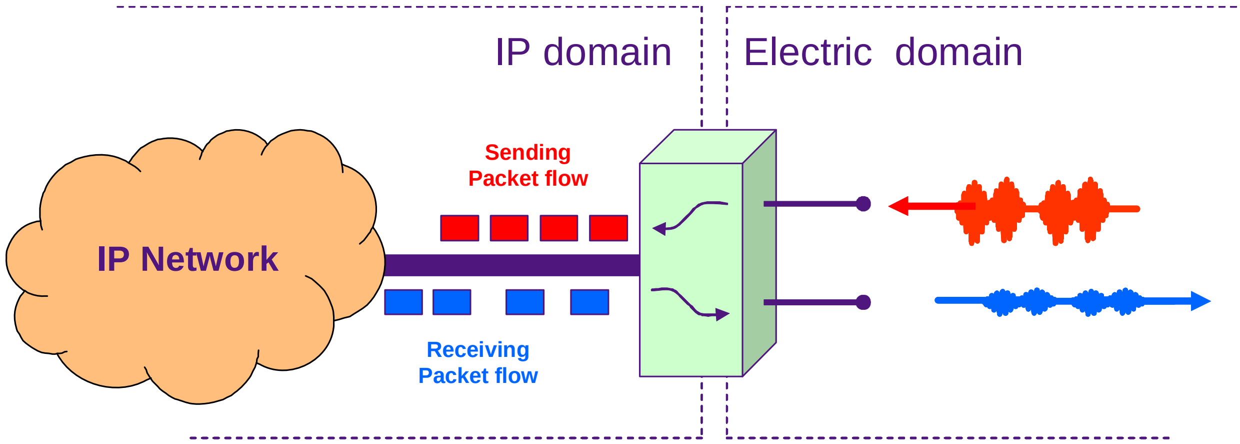

Figure 3: Presentation of the concept of the VoIP reference point

The VoIP 0 dBm0 Reference Point serves as an interface between the IP packet domain and the analog domain for speech signals. Its key characteristics include:

- Codec Compliance: Ensures no distortion due to codec implementation.
- Amplitude Calibration: Maintains correct signal amplitudes for testing.
- Delay Measurement: Tracks processing delays for accurate latency calculations.
- Logging and Traceability: Records events affecting device performance.

7 Characteristics of the Reference Point

7.1 Key Features

- Signal Level Calibration: Precise adjustment to ensure signals meet 0 dBm0 requirements.
- Protocol Compatibility: Supports SIP, H.323, and MGCP for interoperability.
- Jitter Buffer Control: Manages packet delay variations effectively.
- Measurement Interfaces: Includes analog and digital interfaces.

7.2 Performance Metrics

The reference point shall exhibit controlled characteristics for evaluating:

- Frequency Response
- Loudness
- Signal-to-Noise Ratio
- Distortion

7.3 Conclusion

The 0 dBm0 reference point provides a robust, standardized framework for evaluating VoIP devices, overcoming limitations of traditional overload point measurements. It ensures consistent device performance testing, supports interoperability, and simplifies quality assurance across diverse network implementations.

8 Test setup for Test Signal Levels Tests

The Test Signal Levels and the Measurement Methods for Circuit Loudness Rating and Linearity Range for CLR are described in ETSI ES 202 718 [4]. The application scenarios of the 0 dB IP reference point are depicted in figures 4 through 7.

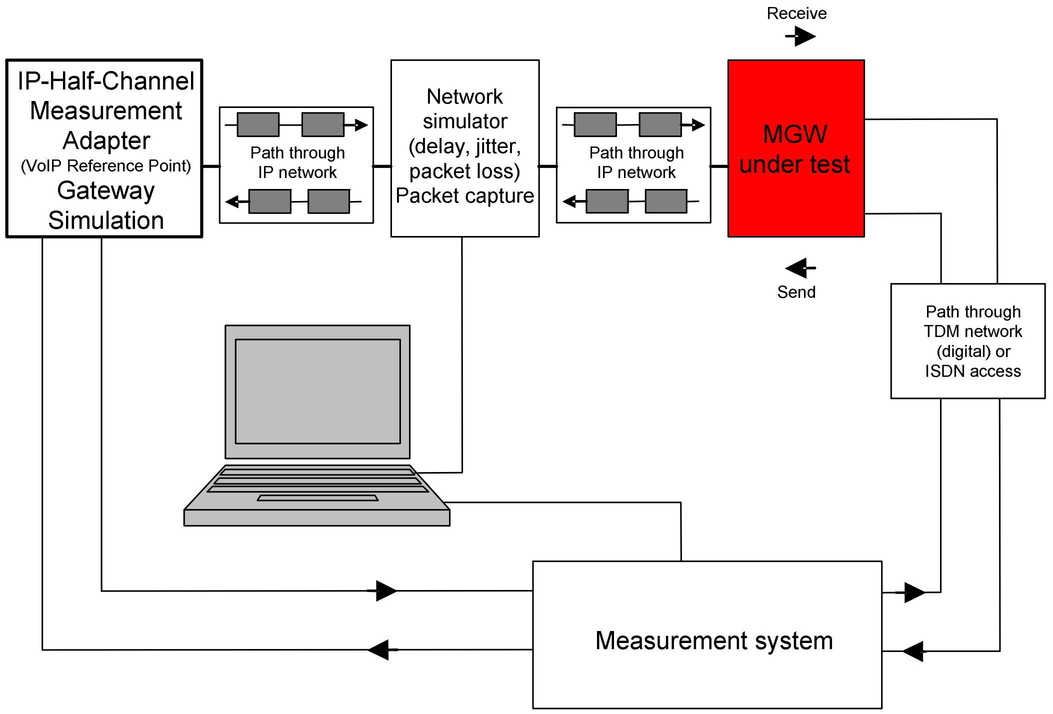

Figure 4: Half channel measurement for MGW with 4-wire interface

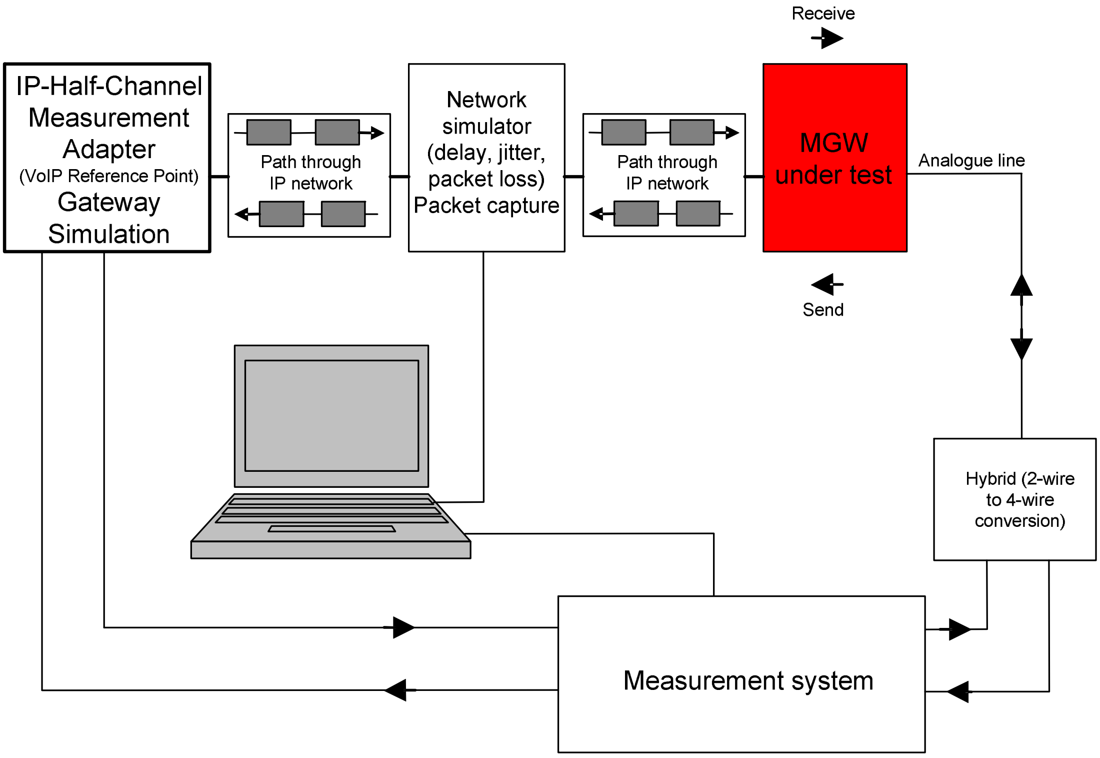

Figure 5: Half channel measurement for MGW with 2-wire interface

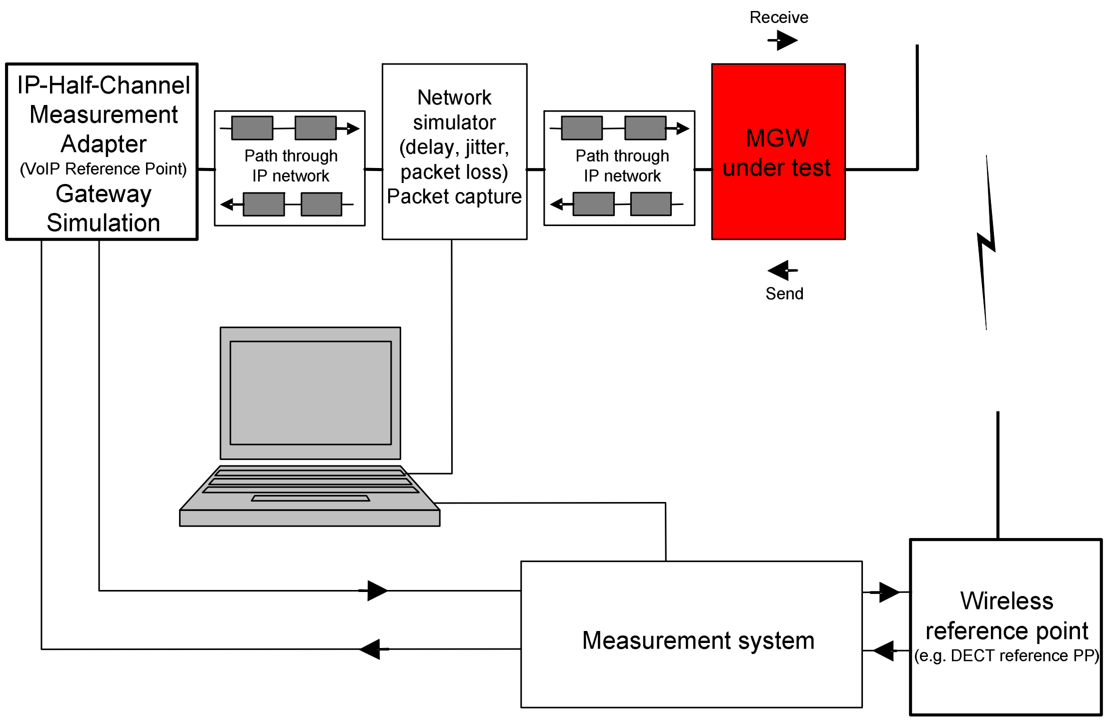

Figure 6: Half channel measurement for MGW with wireless access

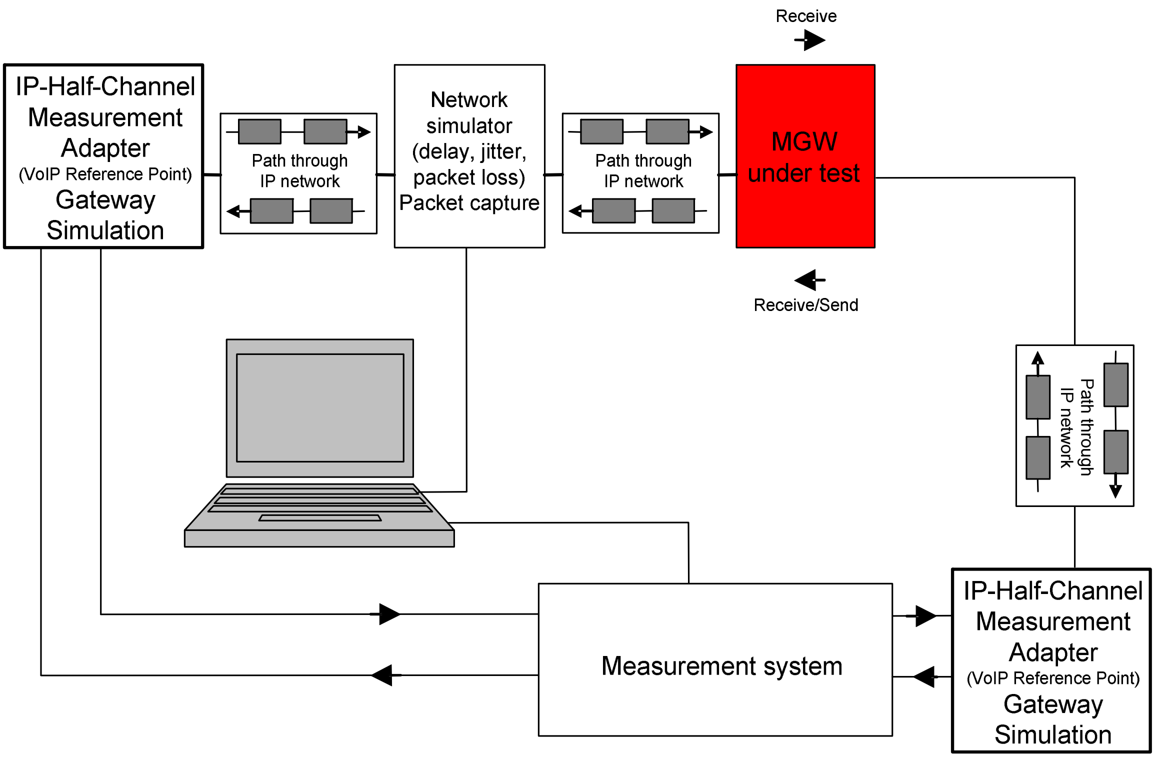

Figure 7: Half channel measurement for IP-to-IP MGW

History

Document history		
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