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SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Objective measuring apparatus

One-view visualization of speech quality measurement results

ITU-T Recommendation P.505

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ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

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ITU-T Recommendation P.505

One-view visualization of speech quality measurement results

Summary

Nowadays, the numerous complex parameters that determine the speech quality of telecommunication equipment as well as the end-to-end speech quality can be interpreted by technical experts only. This Recommendation provides a novel quality representation methodology which is easy to use and also easy to understand for non-experts and which can serve as a basis for commercial decisions on a management or marketing level.

Source

ITU-T Recommendation P.505 was approved on 29 November 2005 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation P.505

One-view visualization of speech quality measurement results

1 Scope

This Recommendation provides a novel quality-representation methodology of parameters that determine the speech quality of telecommunication equipment as well as the end-to-end speech quality. This methodology is easy to use and also easy to understand for non-experts and it can serve as a basis for commercial decisions on a management or marketing level.

This Recommendation does not provide methods for the acquisition of speech quality measurement results; it is assumed that the user of this Recommendation has readily at hand those test results which are needed as an input for the representation methodology recommended here; furthermore, this Recommendation does not state any requirements with respect to the parameters mentioned herein.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.

[2] ITU-T Recommendation P.501 (2000), *Test signals for use in telephonometry*.

[3] ITU-T Recommendation P.502 (2000), *Objective test methods for speech communication systems using complex test signals*.

[4] ITU-T Recommendation P.800.1 (2003), *Mean Opinion Score (MOS) terminology*.

[5] ITU-T Recommendation P.862 (2001), *Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs*.

[6] ITU-T Recommendation G.168 (2004), *Digital network echo cancellers*.

[7] ITU-T Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[8] ITU-T Recommendation G.723.1 (1996), *Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

[9] ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*.

[10] VDA HFT V 1.5: (2004), *Test specification for hands-free terminals*.

3 Introduction

Due to the increasingly implemented signal conditioning, the quality of modern telecommunication equipment (cellphones, VoIP terminals and gateways) can only be described by using the most advanced measurement and analysis methods. The reason is obvious: signal processing mechanisms, which until recently were found in hands-free terminals only (echo compensation, noise reduction, voice controlled attenuation and amplification adjustments), are now employed in virtually all modern telecommunication devices.

The implementation of these demanding signal-processing mechanisms is necessary because, on the one hand, cellphones and other terminals are used in very noisy environments and, on the other hand, their geometry continues to be minimized. Therefore, the built-in loudspeaker and microphone cannot be acoustically de-coupled in a sufficient manner. Echo-reducing measures such as those that are typically used in hands-free terminals are required.

In addition, the different algorithms implemented influence each other. This principle is independent of manufacturers; however, the implementations themselves are not. This leads to significant differences in quality.

The quality of modern terminals (and network equipment) is characterized by numerous speech quality parameters. In order to allow a reliable quality statement, and to detect possible quality problems in advance by laboratory measurements, the telecommunication devices brought to market typically undergo intensive speech quality tests. During these measurements all parameters related to the implemented signal processing and relevant to speech quality are measured and the limit value violation is tested. These measurements guarantee the best possible assessment of speech quality problems that may occur during real use of the corresponding telecommunication device. However, since the numerous complex parameters determining the speech quality of modern telecommunication equipment, as well as the end-to-end speech quality, can be interpreted by technical experts only, a quality representation is required which is easy to use and also easy to understand for non-experts and which can serve as a basis for commercial decisions on a management or marketing level.

Another desirable aspect of these measurements is to show the most important parameters in a visual representation, thus giving a quick overview of all speech quality parameters. This representation should reveal at one glance the strengths and weaknesses as well as the limit value violations.

4 Derivation of the one-view visualization methodology

The requirements for the one-view visualization (OVV) methodology can be summarized as follows:

- Quick and easy recognition of expected speech quality problems for selected parameters ***(limit value violation)***;
- Assessment of strengths and weaknesses of signal processing implemented in a terminal or other telecommunication equipment, including end-to-end considerations ***(quality statement)***;
- Easy comparison of different equipment or connections based on the corresponding representations;
- Easy extension of the representation by new parameters relevant to quality in the future.

A representation based on circle segments ("pie diagram", "star plot") is recommended (see Figure 1).

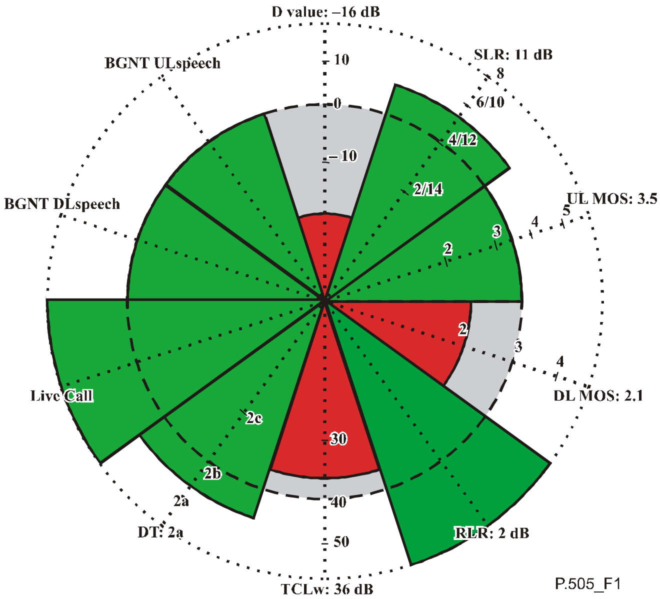


Figure 1/P.505 – Representation "Pie diagram"
(Example with fictitious values of a cellphone)

The number of the represented parameters determines the size of the individual circle segments. Similar to a "cobweb" representation the axes are shown with a common origin. The individual circle segments have the same size (spanned angle 360°/number of selected quality parameters). It is recommended that the number of different parameters visualized in one diagram should not exceed twelve. Moreover, the representation of individual segment sizes is not interdependent, thus guaranteeing the independence of the different quality parameters from each other. Therefore, this pie diagram offers the following advantages:

- Independent representation of individual quality parameters.
- Segment sizes are determined by the number of selected parameters and are identical. In a possible extension step the segment size could be adjusted according to the contribution of individual speech quality parameters to total quality. However, unless such a measure or weighting rule is available, an identical segment size is recommended.
- Segment size (radius) is a measure for the quality of a phone regarding this parameter.
- By means of a suitable axis scaling, a concentric circle around the origin can be defined which represents a minimum quality measure. Falling below this segment size (radius) indicates a non-compliance with this limit value.
- By means of a suitable colour selection results lying within the tolerance or transgressing the limit values can be easily visualized.

Figure 1 gives a representation example for a selection of 10 parameters. Note that this representation does not correspond to a real phone, but only serves as an example. It easily reveals the strengths and weaknesses as well as the limit value violation of a single device – and by representing the results of different devices on one sheet, an easy comparison of the different implementations is possible.

5 Selection of parameters

It is the responsibility of the user of this Recommendation to select a set of parameters as an input for the OVV methodology. Based on recent experience with speech quality tests, this clause provides a list of parameters that may be considered; this list is by no means exhaustive and does not exclude the use of additional or other parameters.

In sending direction (uplink)

- Send loudness ratings (in dB);
- MOS-LQO value [4][1].

NOTE 1 – The listening speech quality determined by MOS-LQO or by the objective MOS-value can be used to show the system performance under different packet-loss conditions, in addition to the listening speech quality with no packet loss.

In receiving direction (downlink)

- Receive loudness rating (in dB);
- MOS-LQO value [4][1].

(See Note 1.)

For End-to-end configurations

- Overall loudness ratings (in dB);
- MOS-LQO value [4].

NOTE 2 – The listening speech quality determined by MOS-LQO can be used to show the system performance under different packet-loss conditions, in addition to the listening speech quality with no packet loss.

For networks

- Junction loudness ratings (in dB);
- MOS-LQO value [4].

(See Note 2.)

Echo attenuation

- TCL_w value.

Double talk performance

- Characterization according to ITU-T Rec. P.340 [1].

"Live Call"

- Phone behaviour during a test call via a real network.

Quality of background noise transmission

- With simultaneous speech in receiving direction.

  The modulation of the background noise (level variation), caused by a receive signal and the echo cancellation signal processing thus activated in the sending direction, is used as a quality measure.

[1] For electrical connections based on ITU-T Rec. P.862 [5], objective MOS-value for acoustical connections currently under study in ITU, alternative methods may be used.

- With simultaneous speech in sending direction.

  The modulation of the background noise (level variation), caused by a send signal, is used as a quality measure.
- D value calculation.

This selection comprises the single-talk performance in the sending and receiving directions, the double talk performance, the quality of background noise transmission, and the echo attenuation as well as the behaviour during a real test phone call.

6 Scaling of axes

For the axis-scaling of the different parameters the following is recommended. Where applicable, the numbers given in this clause, are taken from the relevant Recommendations; in the absence of such requirements, the numbers are examples and are provided for guidance.

Send loudness rating (SLR)

This parameter should be within a range of 8 ± 3 dB according to the acoustical quality tests of phones. A double scaling of this axis was therefore selected. It rises from the origin of the diagram radially to the outside up to 8 dB and in addition radially to the inside up to 16 dB (again in the origin). The outer range to be kept thus lies between 5 and 11 dB.

Receive loudness rating (RLR)

The receive loudness rating measured in dB is set to a nominal value of 2 dB via the volume control of the phone at the beginning of the measurements. Lower values correspond to a louder transmission. Again, a double scaling of this axis was selected. It rises from the origin of the diagram (–6 dB, loud transmission) radially to the outside up to 2 dB (nominal value) and also radially to the inside up to 10 dB (again in the origin). The outer range that should be kept therefore lies between –1 and 5 dB. Usually an RLR value of 2 dB can be realized for one loudness setting of the phones.

Overall loudness rating (OLR)

The overall loudness rating measured in dB is set to a nominal value of 10 dB via the volume control of the phone at the beginning of the measurements. Lower values correspond to a louder transmission. Again, a double scaling of this axis is recommended.

Junction loudness rating (JLR)

The junction loudness rating measured in dB is expected to be at a nominal value of 0 dB. Again, a double scaling of this axis is recommended.

MOS-LQO value in sending direction

This value describes the sound quality of the speech transmitted in uplink mode. These values are used for the scaling of this axis.

For electrical connections the limit value is determined by the codec used.

For cellphones the limit value, which should be kept, lies at 3.2.

For terminals in other networks, e.g., in VoIP networks, the limit value depends on the codec used. As a general rule the MOS-LQO value should not drop more than 0.2 points compared to the value measured for the codec without any other signal processing.

MOS-LQO value in receiving direction

This value describes the sound quality of the speech transmitted in downlink mode. These values are used for the scaling of this axis.

For electrical connections the limit value is determined by the codec used.

For cellphones the limit value, which should be kept, lies at 2.5.

For terminals in other networks, e.g., in VoIP networks, the limit value depends on the codec used. As a general rule the MOS-LQO value should not drop more than 0.7 points compared to the value measured for the codec without any other signal processing involved.

MOS-LQO value in end-to-end configurations

This value describes the sound quality of the speech transmitted end-to-end. These values are used for the scaling of this axis. The limit value, which should be kept, lies at 2.5.

Echo attenuation as TCL_w value

This axis is scaled between 20 dB (origin) and 60 dB. The limit value to be kept lies at 46 dB.

Echo attenuation during double talk as TCL_{wdt} value

This axis is scaled between 0 dB (origin) and 40 dB. The limit value to be kept lies at 27 dB.

Double talk performance (characterization of phones or echo cancellers)

Based on ITU-T Rec. P.340 [1] as well as the VDA specification for mobile hands-free terminals [10], the phones are characterized based on their double talk performance. Two uncorrelated composite source signals according to ITU-T Rec. P.501 [2] and the analysis method according to ITU-T Rec. P.502 [3] are used for the measurement. A characterization between 3 (incapable of double-talk, origin), 2c, 2b, 2a and 1 (unlimited double talk capability) scales this axis. A limit value according to "2b" should be kept.

"Live call"

Within the framework of objective quality tests of phones, an additional short test phone call may be made by an expert ("live call"). This serves to verify whether, in the real network, additional impairments exist which influence the quality beyond the laboratory measurements. The axis only has 2 values. If impairments are detected during this phone call, the segment size is reduced to a red area within the circle which characterizes the minimum requirements. If no obvious impairments are found, the segment covers the maximum possible area.

Quality of background noise transmission with speech-like test signal in receiving direction

This axis is scaled between –20 dB (origin) and 0 dB. While feeding a speech-like test signal and simultaneously transmitting a background noise in the sending direction, the level modulation of the transmitted impairment noise caused by attenuation insertion is measured. The limit value is 10 dB (see also [1]).

Quality of background noise transmission with speech-like test signal in sending direction

This axis is scaled between –20 dB (origin) and 0 dB. While feeding a speech-like test signal in the sending direction and simultaneously transmitting an impairment noise (also in the sending direction), the resulting level modulation is measured. The limit value is 10 dB.

Performance of the implemented VAD respectively automatic gain control

This axis is scaled between –20 dB (origin) and 0 dB. While feeding a speech-like test signal in the receiving direction and simultaneously transmitting a background noise in the sending direction, the level matching of a comfort noise injection is measured. The limit value is 10 dB.

D value

This value scales the axis between –15 dB and 10 dB. The recommended limit value is 0 dB.

Annex A

Examples of the application of the OVV methodology

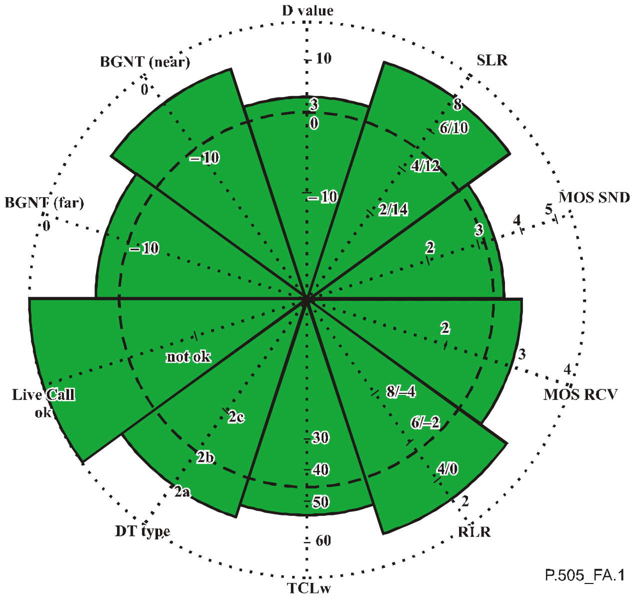
Subsequently, some example applications are illustrated for the suggested selection of parameters and their arrangement in a "pie diagram". Here, typical effects separated by individual conversational aspects are shown (quality parameters in the sending and receiving directions, echo and double talk performance, as well as the quality of background noise transmission).

A.1 Application of the OVV methodology to cellphones

These examples do not represent real cellphones, but are fictitious in order to illustrate the principles and the possibilities of interpretation.

Example A.1 – High speech quality in all conversational aspects

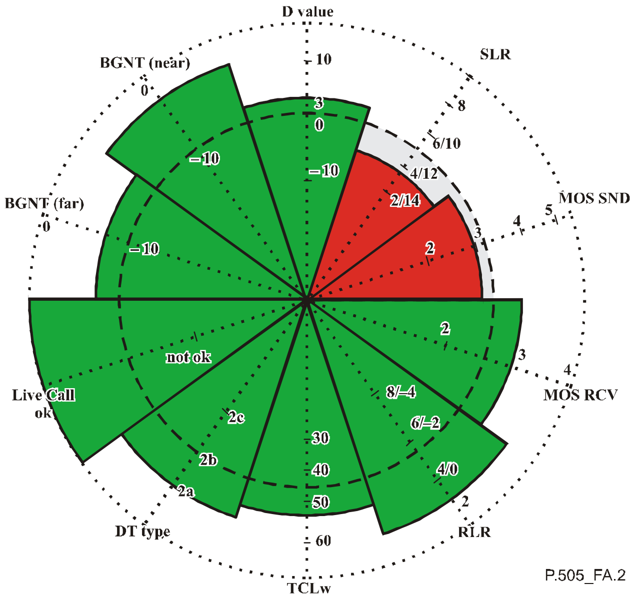
All measured values lie above the minimum requirements. The radius of all circle segments exceeds the medium circle describing the minimum quality (dashed line). In addition, this is indicated by the (light) green colouring.



Example A.2 – Quality impairments in sending direction

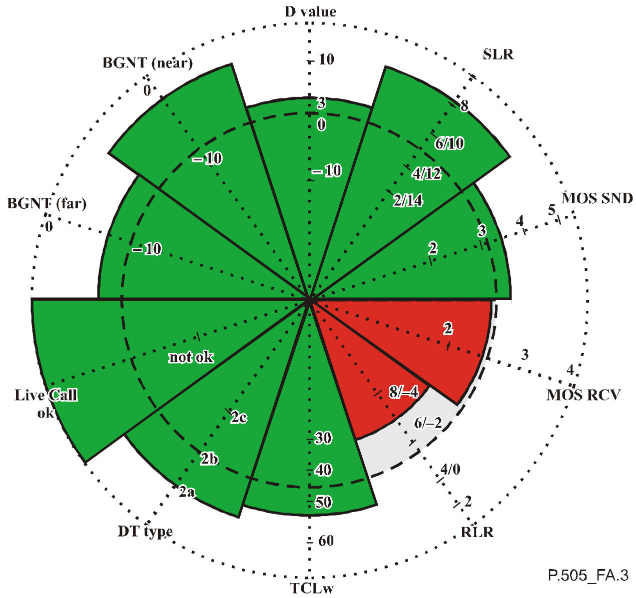
In this example, the SLR value measured as 13 dB would exceed the tolerance of 8 ± 3 dB. Moreover, the quality value for the sound of transmitted speech (MOS-LQO) is 3.0 and thus lies below the recommended limit value of 3.2.

Both circle segments are coloured in (dark) red to allow a better visualization.



Example A.3 – Quality impairments in receiving direction

The MOS-LQO value in the receiving direction is 2.4 and lies below the recommended limit value of 2.5. The circle segment (axis name "MOS RCV") is correspondingly coloured in red. The circle segment for receive loudness rating (RLR) also coloured in (dark) red indicates that an RLR value of 2 ± 3 dB within the tolerance required could not be achieved by any of the loudness settings.



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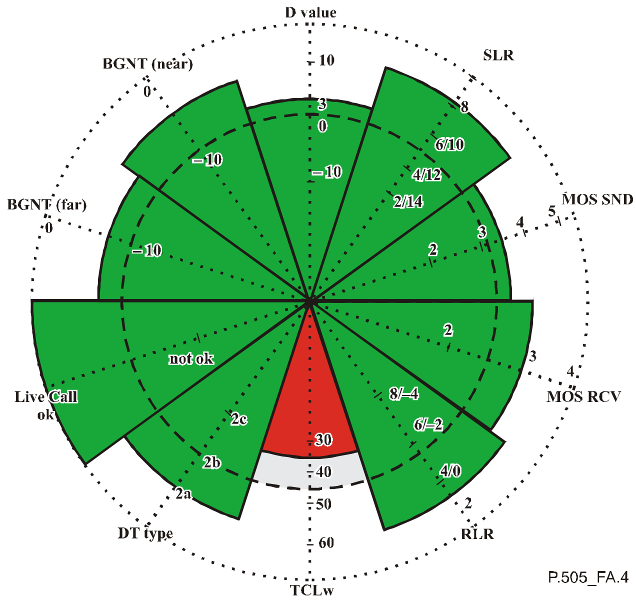
Example A.4 – Insufficient echo attenuation

The echo attenuation of 36 dB TCL_w (fictitiously measured) lies below the required limit value of 46 dB.

The circle segment indicated by the axis name "TCL_w" is coloured in (dark) red.



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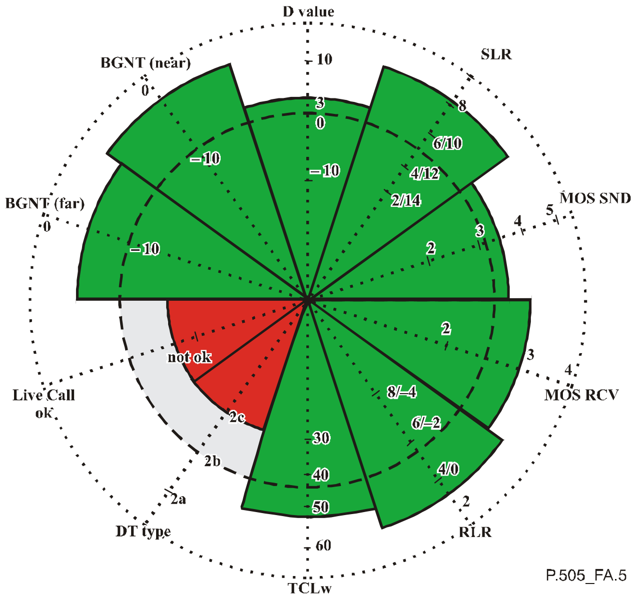
Example A.5 – Quality impairments during double talk and during "test call"

This cellphone would be characterized as "Type 2c" referring to double talk performance. It therefore only has a limited "double talk capability" and does not achieve the recommended characterization "2b" (dashed inner circle), "2a" or "1".

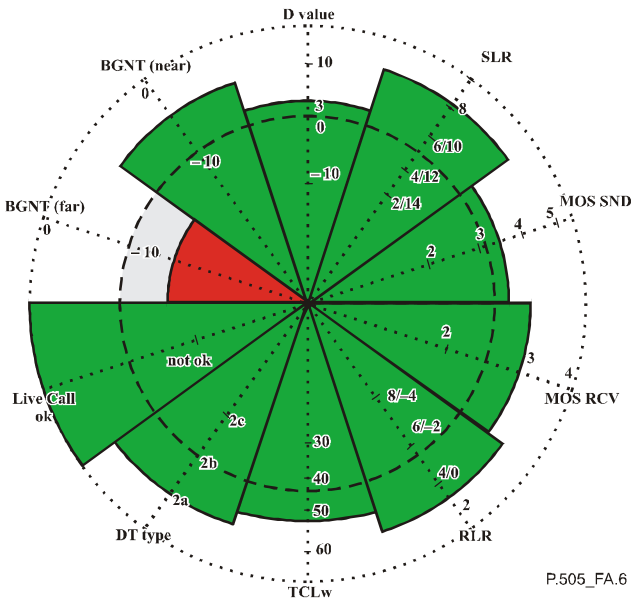
The (dark) red colouring of the circle segment with the axis name "Live Call" indicates that during an informative test phone call via a real GSM network obvious quality impairments were perceived.



Example A.6 – Quality impairments in the transmission quality of background noise during simultaneous feeding of a receive signal (downlink signal, far-end signal)

If the cellphone is used in a noisy environment, the signal transmitted in the sending direction (uplink) during simultaneous feeding of a receive signal (downlink) is level modulated. The level variations in this example were determined as 13 dB and thus exceed the limit value of max. 10 dB.

The circle segment with the axis name "BGNT (far)" is correspondingly coloured in (dark) red.

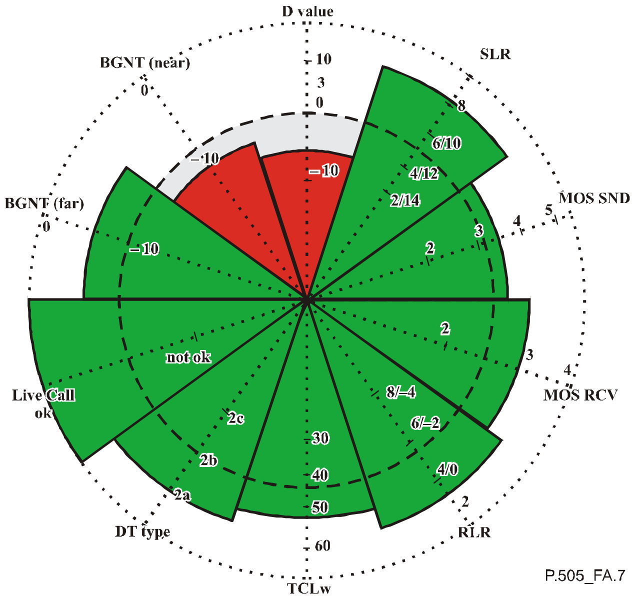


Example A.7 – Quality impairments in the transmission quality of background noise during simultaneous feeding of a send signal (uplink signal, near-end signal)

If the cellphone is used in a noisy environment, the signal transmitted in the sending direction (uplink) during simultaneous feeding of a speech-like test signal (uplink) is level modulated. The transmitted background noise is "pumping", the level varies with the transmitted speech of the cellphone user.

The level variations in this example were determined as 12 dB and thus exceed the limit value of max. 10 dB.

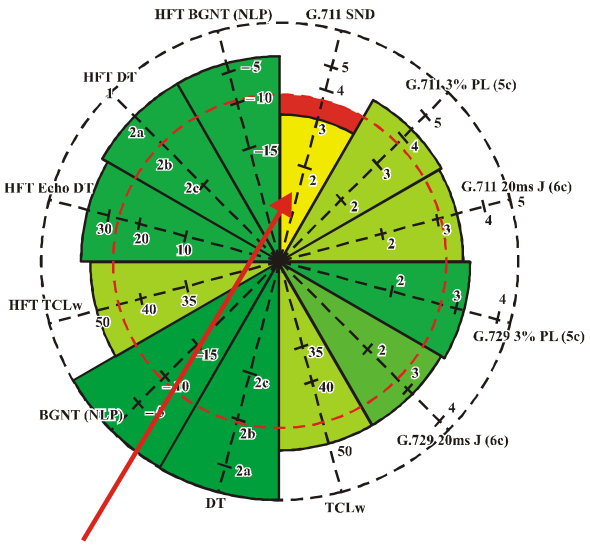
Moreover, the (dark) red colouring of the circle segment named "D value" shows a violation of the limit value for this parameter. The fictitious 4 dB presented here is below the recommended value of 0 dB.



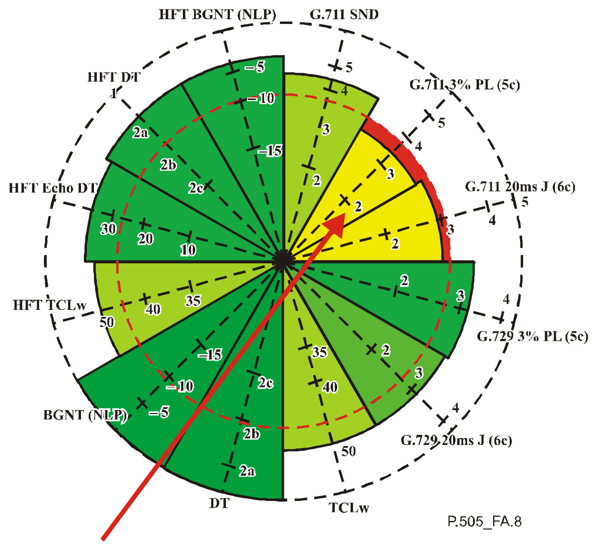
A.2 Application of the OVV methodology to VoIP terminals

The following examples explain each transmission quality parameter ("pie slice") with its scaling and requirement in detail. These examples are not derived from real existing IP terminals.

Example A.8 – Quality impairments, listening speech quality



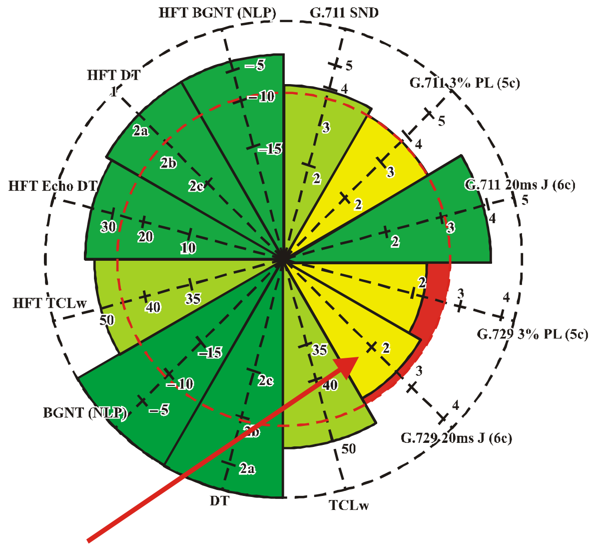
"Listening speech quality in sending direction using G.711 [7] (handset) below average"



"G.711 [7] listening speech quality in receiving direction (handset) below average results"

The listening speech quality result measured in the sending direction is represented by the first slice. In the receiving direction each speech coder is represented by two slices, one for the packet loss condition 3%, one for a jitter condition (20 ms jitter, 1% packet loss). The values are taken from MOS-LQO results. Each axis is scaled between 1 and 5 representing the MOS scale.

The limit (radius of the (dark) red circle) is given by the average MOS-LQO result under this test condition. It should be considered that these limits are different for each test condition and each speech coder.

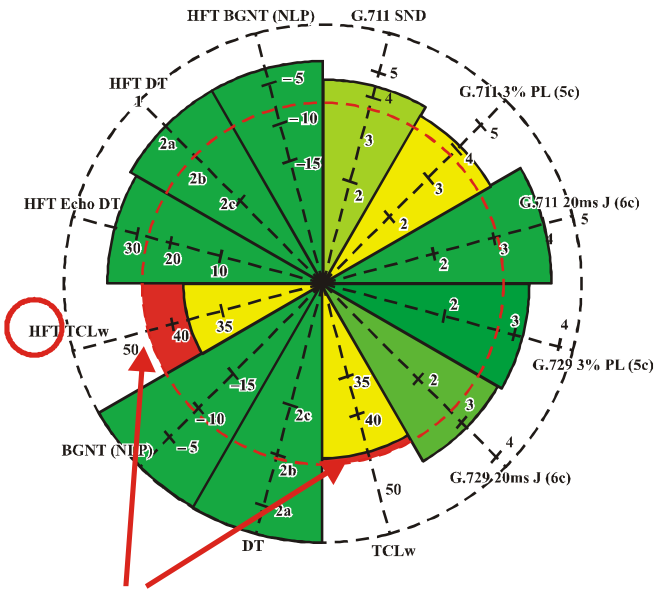


"G.729 [9] listening speech quality in receiving direction (handset) below average results"

Example A.9 – Quality impairments, echo during single talk and double talk

The terminal coupling loss (TCLw) is measured for the terminals in the handset and the hands-free mode.

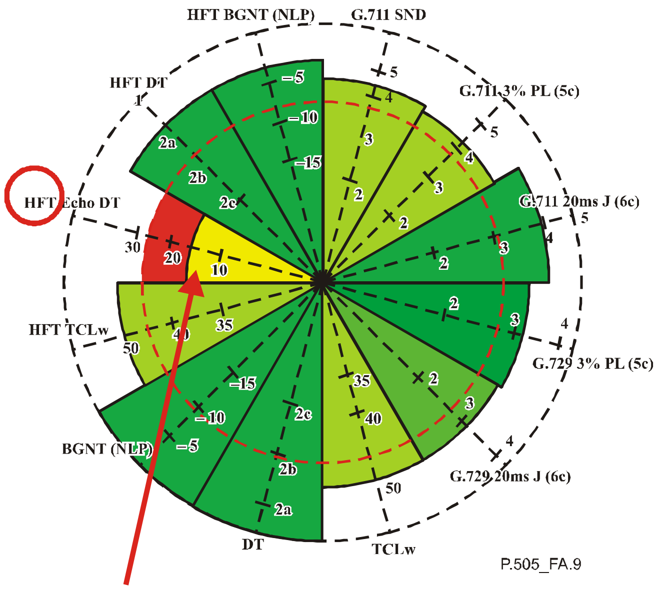
The requirement represented by the inner (dark) red circle is 46 dB.



"Echo attenuation according to G.122 under single talk condition below 46 dB"

The echo attenuation of the hands-free implementation during double talk is measured as described in ITU-T Rec. 502 [3].

The minimum attenuation (indicated by the inner (dark) red circle) is 27 dB. This value, derived from subjective tests can be found in ITU-T Rec. P.340 [1]. 27 dB echo attenuation during double talk would lead to a full duplex characterization assuming a 100 ms one-way delay in the network. This value can be regarded as a minimum requirement.

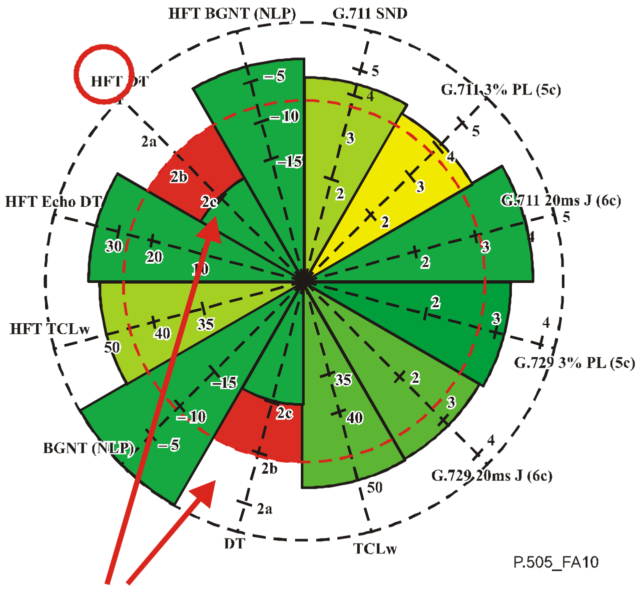


"Echo attenuation under double talk conditions lower than recommended"

Example A.10 – Quality impairments, attenuation in the sending direction during double talk, characterization

The double talk performance is influenced by the attenuation inserted during a double talk period. The tests are conducted according to ITU-T Rec. P.502 [3].

The level of the transmitted signal is referred to the near-end signal level (double talk signal) and analysed vs time. In this example, the attenuation in the sending direction leads to a type 2c characterization in handset and in HFT modes.



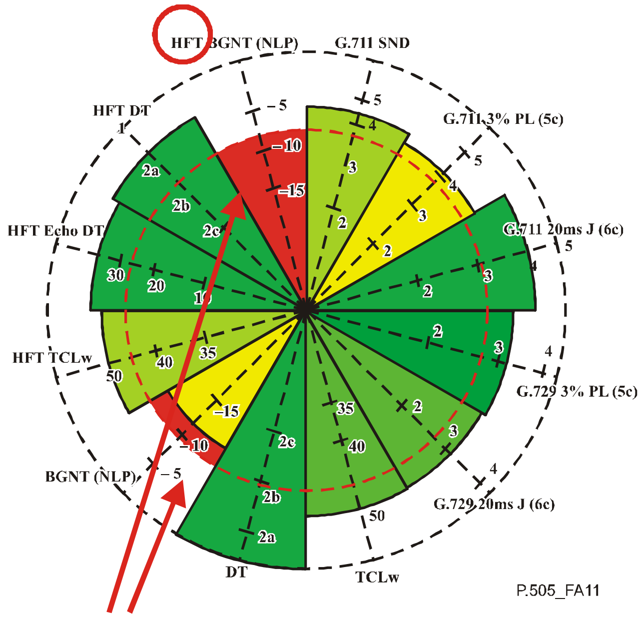
"Double talk performance influenced by level variation leads to 'type 2c' characterization"

Example A.11 – Quality impairments, quality of background noise transmission with far-end signal

During the application of far-end signals the echo suppression unit may introduce audible and disturbing noise modulation (level variation).

The level difference between the transmitted signal with and without the application of far-end signals is measured.

This difference should not exceed 10 dB, either for the pub noise or for the café noise.

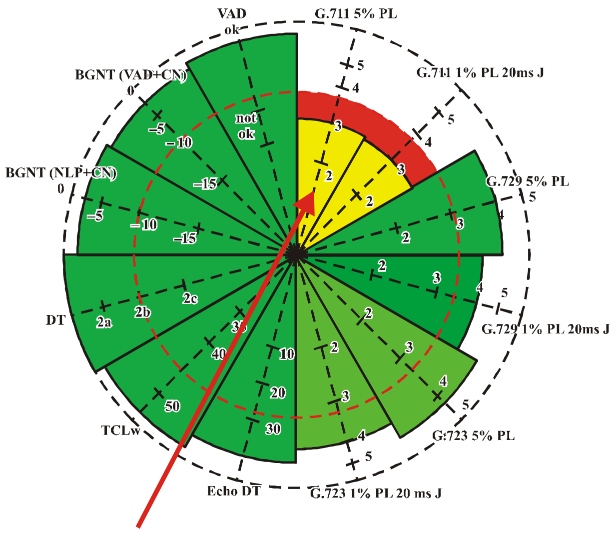


"Background noise modulation introduced by echo suppression and/or comfort noise generation too high"

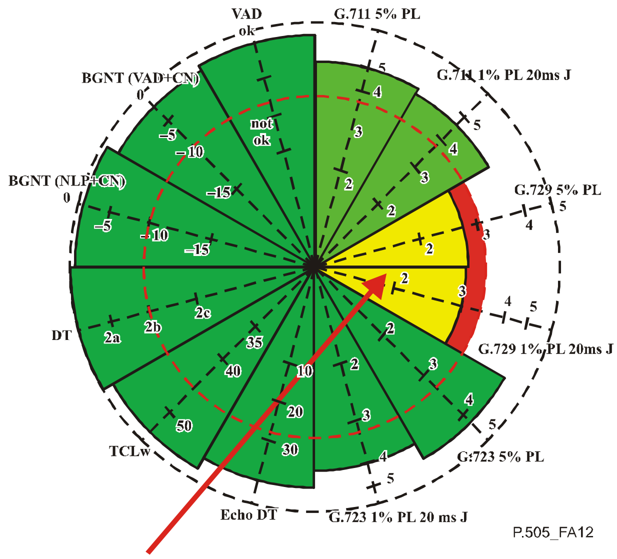
A.3 Application of the OVV methodology to VoIP gateways

The following examples explain each transmission quality parameter ("pie slice") with its scaling and requirement in detail. These examples are not derived from real existing gateways.

Example A.12 – Quality impairments, listening speech quality



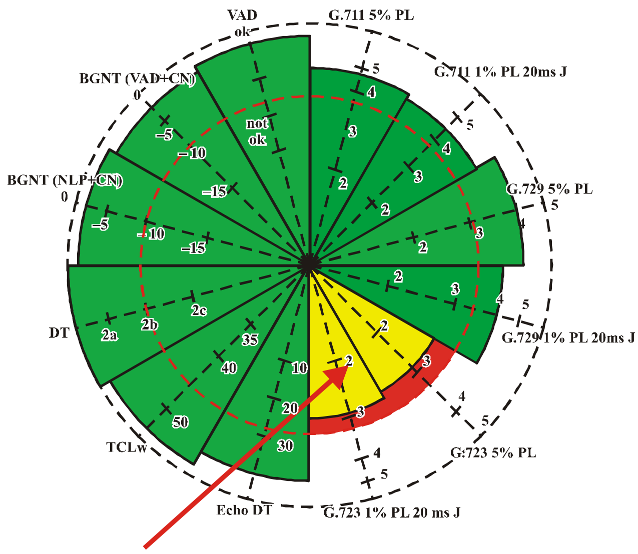
"G.711 [7] listening speech quality below average results"



"G.729 [9] listening speech quality below average"

The listening speech quality result for each speech coder is represented by two slices, one for the packet loss condition 5%, one for a jitter condition (20 ms jitter, 1% packet loss). The values are taken from the MOS-LQO results for the ITU-T Recs G.711 [7], G.729 [9] and G.723.1 [8] speech coders. Each axis is scaled between 1 and 5 representing the MOS scale.

The limit (radius of the (dark) red circle) is given by the average MOS-LQO result over all gateway implementations used in a comparison test. It should be considered that this limit is codec dependent, thus the limits are different for the three speech coders.

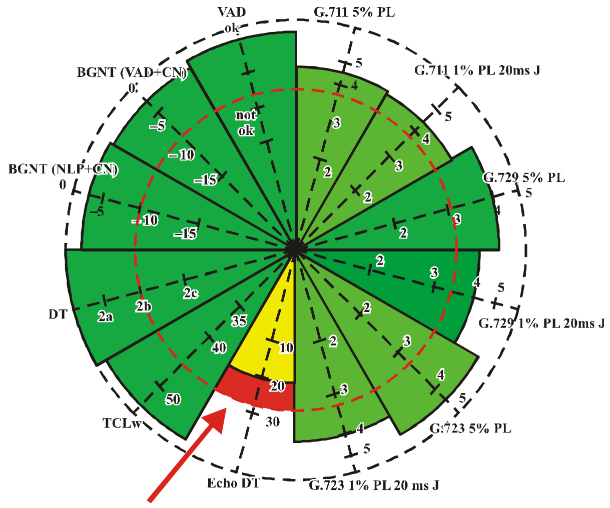


"G.723.1 [8] listening speech quality below average results"

Example A.13 – Quality impairments, attenuation in sending direction during double talk, characterization

The echo attenuation during double talk is measured according to the methods described in ITU-T Rec. P.502 [3] and defining specific echo paths for the tests (e.g., 40 dB ERL and 6 dB ERL).

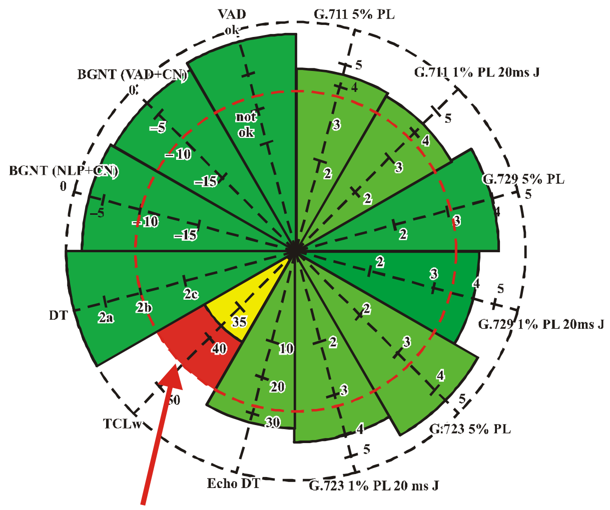
The minimum attenuation (indicated by the inner (dark) red circle) is 27 dB. This value, derived from subjective tests, can be found in ITU-T Rec. P.340.



"Echo attenuation under double talk conditions lower than recommended"

The echo is measured according to ITU-T Rec. G.168 [6]. The relevant results for this representation can be taken from the different echo paths used, e.g., 6 dB and the 40 dB ERL measurement.

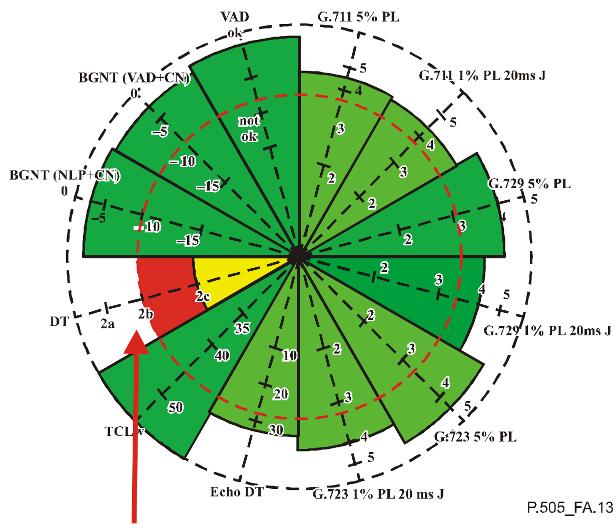
The lower value from both measurements is used for the pie. The requirement represented by the inner (dark) red circle is 46 dB.



"Echo attenuation according to G.122 under single talk condition below 46 dB"

The double talk performance is influenced by the attenuation inserted during a double talk period.

The tests can be conducted as described in ITU-T Rec. P.502 [3]. The level of the transmitted signal is referred to the near-end signal level (double talk signal) and analysed vs time. The average level difference is used to classify the double talk performance.



P.505_FA.13

"Double talk performance influenced by level variation"

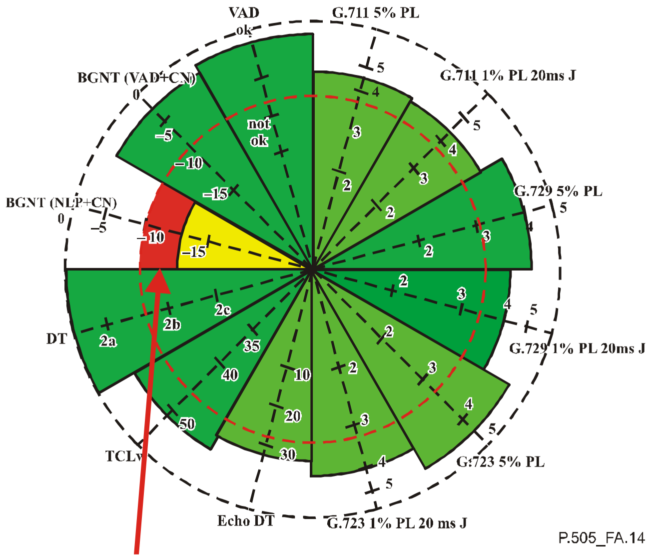
Example A.14 – Quality impairments, quality of background noise transmission with far-end signal

During the application of far-end signals the echo suppression unit may introduce audible and disturbing noise modulation (level variation).

For the tests, realistic background noises should be used.

The level difference between the transmitted signal with and without the application of far-end signals is measured.

This difference should not exceed 10 dB for all background noises used in the test.



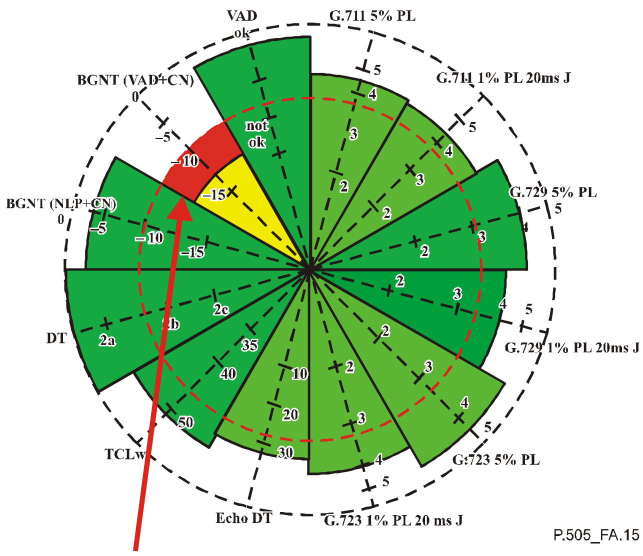
"Background noise modulation introduced by echo suppression and/or comfort noise generation too high"

Example A.15 – Quality impairments, quality of realistic background noise transmission with far-end signal

Realistic background noise scenarios like the pub noise or the café noise used should be transmitted without significant level variation.

The level difference between the transmitted signal with and without VAD is measured.

This difference should not exceed 10 dB, either for the pub noise or for the café noise.

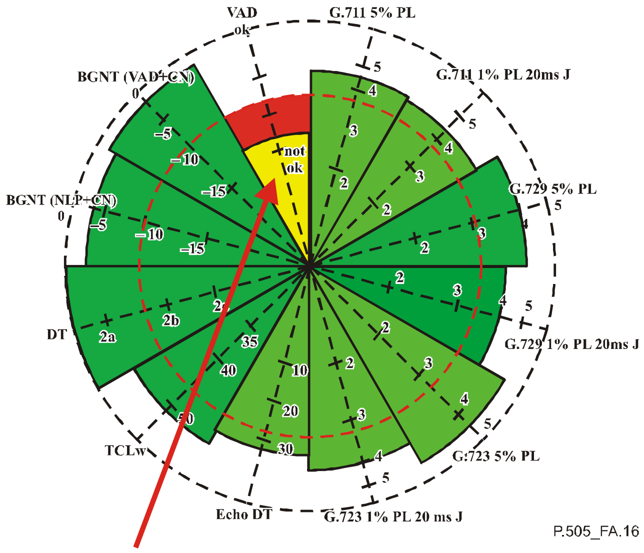


"Background noise modulation introduced by VAD or comfort noise generation too high"

Example A.16 – Quality impairments, VAD and AGC test

The level of a transmitted test signal should follow the original test signal level if VAD is enabled. Comfort noise – if implemented – should be level adaptive.

The level difference of the transmitted signal should not exceed 10 dB.



"Level of transmitted signal violates the tolerance scheme"

A.4 Further considerations for OVV application to end-to-end configurations

While for the application of the one-view visualization methodology to end-to-end configurations, in general, the same principles apply, which have been outlined in clauses A.1 through A.3, there are some additional aspects that need to be considered in such an application of OVV.

As described above, the application of OVV to telecommunication ***components*** provides the comparison of a number of similar devices, e.g., cellphones. However, in case of ***end-to-end configurations*** OVV can be applied with the two different strategies:

- "General Approach":

  Here, different kinds of end-to-end configurations are compared by means of OVV in order to evaluate the cross-technology satisfaction of the user.

  A typical example of such an OVV campaign would be an in-house comparison of one vendor's product portfolio, involving for example ISDN-to-ISDN, IP-to-IP and hybrid IP-to-ISDN connections.

- "Application Approach":

  There is one selected kind of end-to-end configuration, based on which the user's satisfaction for products from different vendors is compared.

  A typical example of such an OVV campaign would be a public test event, involving for example IP phones from different vendors in IP-to-ISDN connections.

In any case, care should be exercised to clearly indicate, together with the OVV diagrams, which of the aforementioned approaches has been used, which are the configurations, which are the components contained therein, etc.

Appendix I

Analysis examples

I.1 Analysis examples of different cellphones

Subsequently the results of some cellphones measured recently are analysed in the suggested form of representation. These cellphones were selected randomly from the models of different manufacturers.

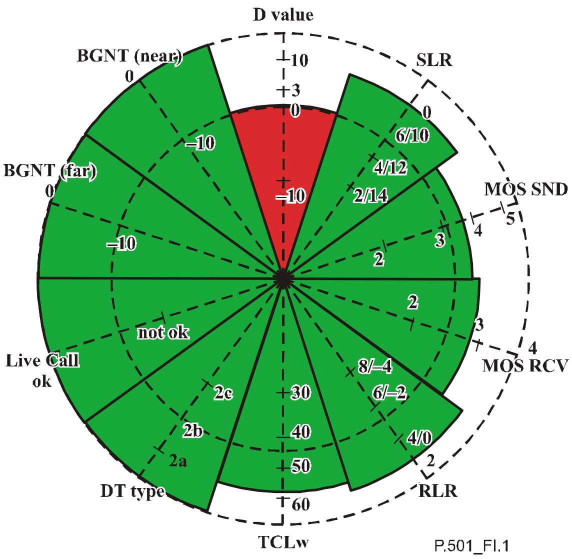


Figure I.1/P.505 – Cellphone 1

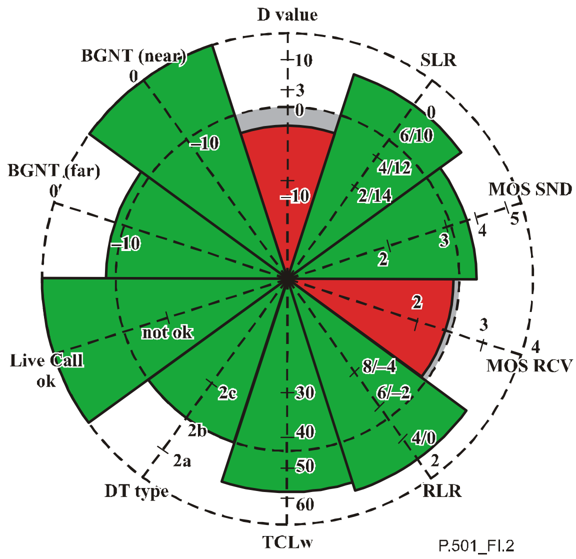


Figure I.2/P.505 – Cellphone 2

For both cellphones the parameter "D value" exceeds the tolerance of > 0 dB (–0.4 dB cellphone 1, Figure I.1 and –3.3 dB cellphone 2, Figure I.2). In addition, cellphone 2 reveals a lower sound quality in the receiving direction, the calculated value lies below the limit value. Cellphone 1 shows advantages for the parameters double talk performance ("DT type") and transmission quality of background noise during simultaneous feeding of a receive signal (downlink signal, far-end signal, axis "BGNT (far)").

The comparison of these two cellphones of different manufacturers shows some interesting differences in this form of representation, revealing clear advantages for the implementation in cellphone 1 compared to the cellphone 2.

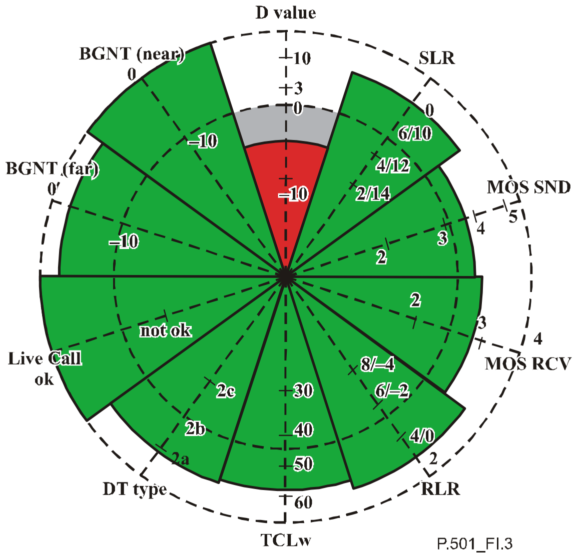


Figure I.3/P.505 – Cellphone 3

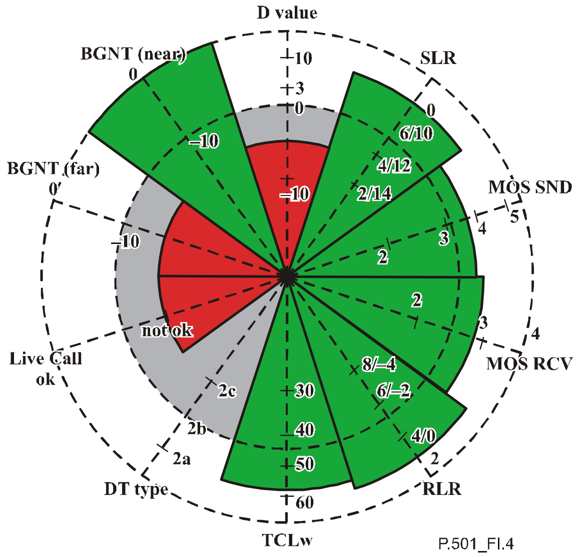


Figure I.4/P.505 – Cellphone 4

Figures I.3 and I.4 show the performance of two devices from one manufacturer. The direct comparison shows big differences. Besides the D value, the values measured for the parameters double talk performance ("DT type") and transmission quality of background noise during simultaneous feeding of a receive signal (downlink signal, far-end signal, axis name "BGNT (far)") clearly exceed the tolerance for cellphone 4. The other parameters for both devices are similar (TCL_w, SLR, MOS SND, MOS RCV).

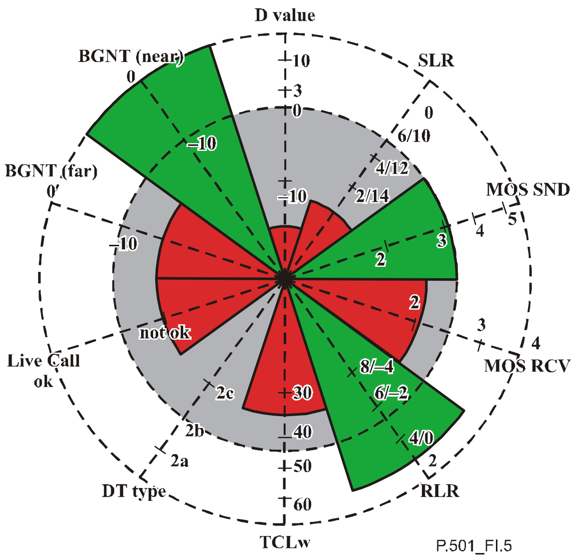


Figure I.5/P.505 – Cellphone 5

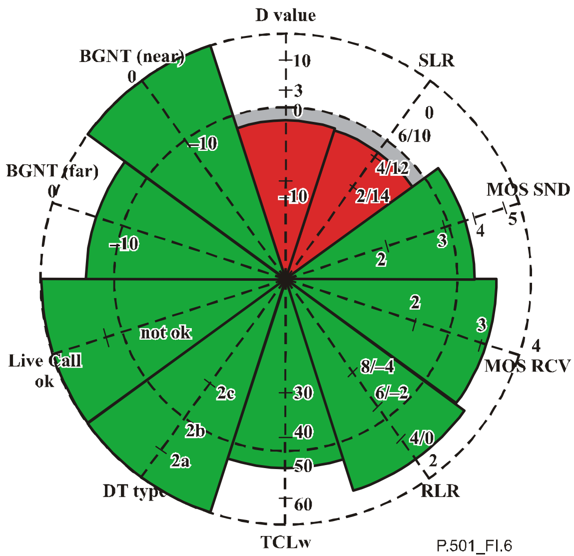


Figure I.6/P.505 – Cellphone 6

In Figures I.5 and I.6 clear quality impairments can be derived from this form in the representation of cellphone 5 (Figure I.5): With 19 dB the SLR is clearly too high and exceeds the maximum limit value by 8 dB. The echo attenuation is too low ("TCL_w") and the phone was characterized as "Type 3" (double talk incapability) based on the double talk performance measurements ("DT type"). In real use the conversation partner can expect clear quality impairments in noisy environments. The level variations in the sending direction during simultaneous feeding of a receive signal (downlink signal, far-end signal, axis name "BGNT (far)") also clearly exceed the tolerance.

At one glance, the example given in Figure I.6 for cellphone 6 reveals a quite balanced implementation with the exception of the D value and SLR parameters which are both slightly too low.

I.2 Analysis examples of different VoIP terminals

Subsequently the results of some VoIP terminals measured recently in the ETSI VoIP speech quality test events were analysed in the suggested form of representation.

In Figure I.7, the listening speech quality in the sending direction is comparable to the average score.

Under the influence of jitter, the listening speech quality is lower than the average performance during the event for both speech coders. Both PLC implementations (ITU-T Recs G.711 [7] and G.729 [9]) lead to listening speech quality scores comparable to the average scores.

The echo attenuation under single conditions is below the recommended value, but this result is mainly due to the high noise level. Only slight level variations occur in the transmitted background noise.

In hands-free mode the echo attenuation is higher than the recommended value, but double talk performance is characterized as type 3. The near-end signal is not transmitted. The activation of echo suppression also leads to disturbing noise modulation.

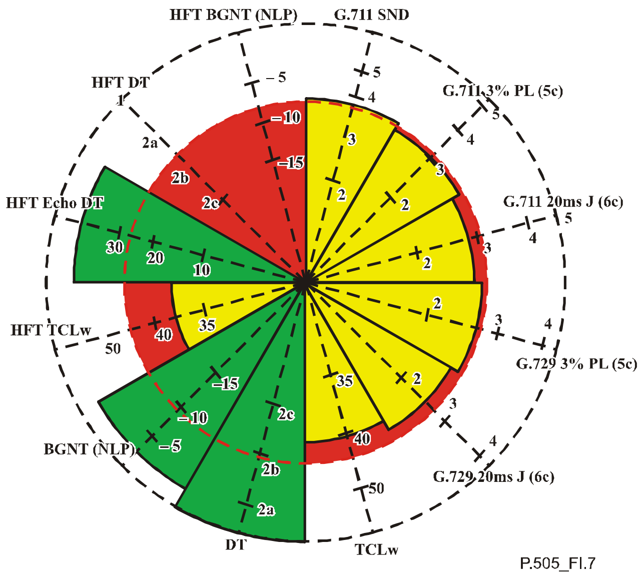


Figure I.7/P.505 – VoIP terminal 1

In Figure I.8, the listening speech quality in the sending direction corresponds to the average score.

Under the influence of jitter and packet loss the listening speech quality is comparable to or higher than the average performance during the event for both speech coders.

The echo attenuation under single talk conditions is higher than the recommended value. Only slight level variations occur in the transmitted background noise.

The double talk performance in hands-free mode is characterized as type 3 due to level variation in the sending direction. The activation of echo suppression also leads to disturbing noise modulation.

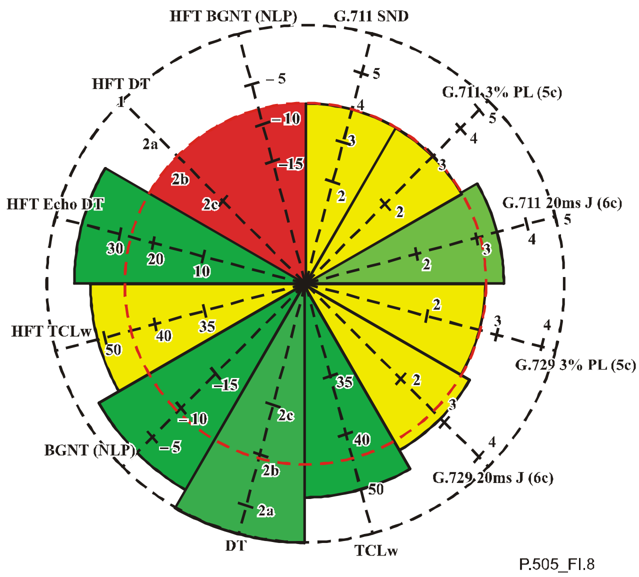


Figure I.8/P.505 – VoIP terminal 2

In Figure I.9, the listening speech quality in the sending direction corresponds to the average score.

Under the influence of jitter and packet loss the listening speech quality is comparable to or higher than the average performance during the event for both speech coders.

The echo attenuation under single talk conditions is higher than the recommended value. Level variations occur in the transmitted background noise.

Although the hands-free implementation is relatively "smooth", allowing some residual echo during double talk, the double talk performance is characterized as type 3 due to level variation in the sending direction. The activation of echo suppression also leads to disturbing noise modulation.

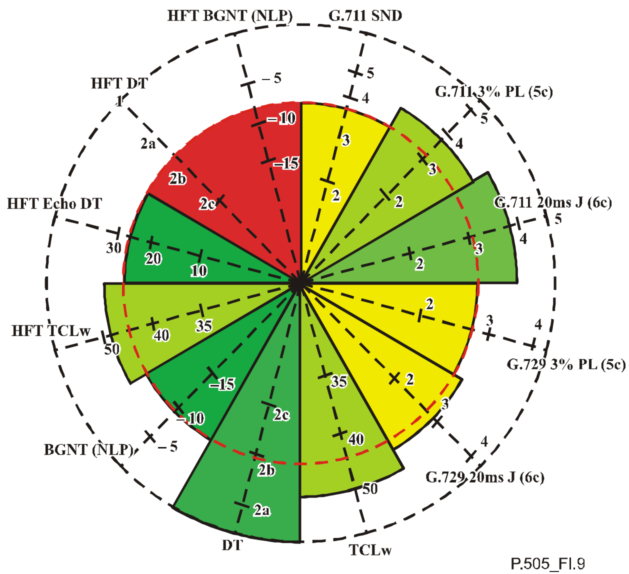


Figure I.9/P.505 – VoIP terminal 3

In Figure I.10, the listening speech quality in the sending direction is slightly lower than the average score.

Under the influence of jitter and packet loss the listening speech quality is lower than the average performance during the event.

The G.729 [9] speech coder was not tested during the event.

The echo attenuation under single conditions fulfils the recommended value. The transmission of background noise and double talk signals is not impaired by level variations.

The hands-free implementation was not tested during the event.

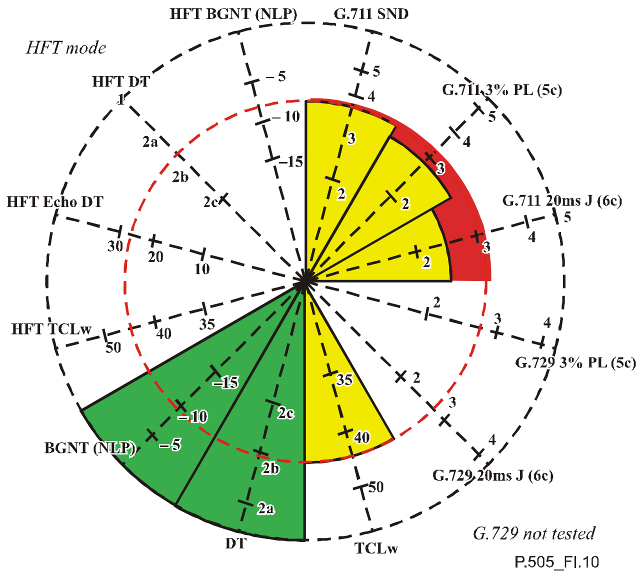


Figure I.10/P.505 – VoIP terminal 4

I.3 Analysis examples of different VoIP gateways

Subsequently, the results of some VoIP gateways measured recently in the ETSI VoIP speech quality test events were analysed in the suggested form of representation.

Except for the performance of the G.711 [7] PLC implementation, the listening speech quality scores are comparable to the average scores.

The echo attenuation under single and double talk conditions exceeds the recommended values. Double talk performance is characterized as "full duplex" for comparable near-end and far-end signal levels.

The activation of echo suppression leads to audible and disturbing noise modulation (tested with infinite ERL).

VAD and comfort noise generation do not significantly modulate the transmitted pub and café noises.

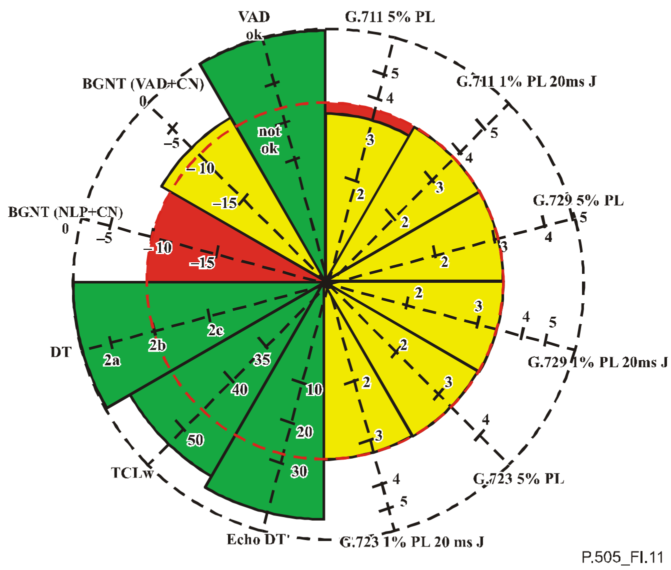


Figure I.11/P.505 – VoIP gateway 1

In Figure I.12, the listening speech quality scores are comparable or slightly higher (ITU-T Rec. G.711 [7], jitter) than the average scores.

The echo attenuation under single and double talk conditions exceeds the recommended value. Double talk performance is characterized as "full duplex" for comparable near-end and far-end signal levels.

The activation of echo suppression does not lead to disturbing noise modulation (tested with infinite ERL).

VAD and comfort noise generation do not significantly modulate the transmitted pub and café noises.

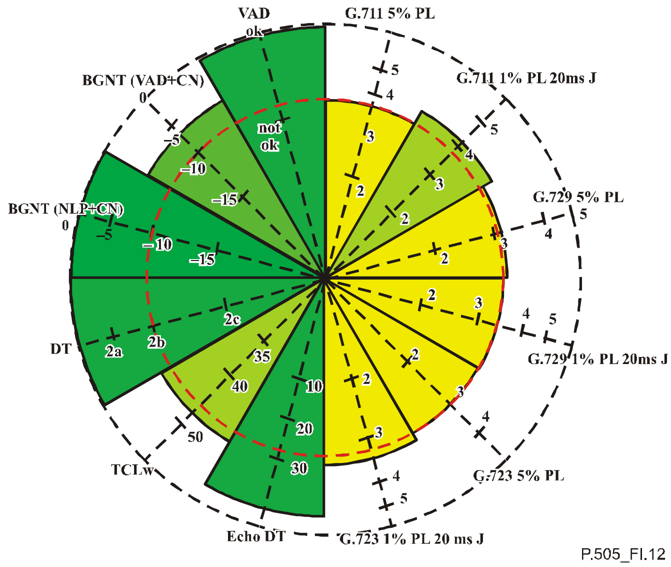


Figure I.12/P.505 – VoIP gateway 2

In Figure I.13, the listening speech quality scores are comparable to the average scores, but the performance with jitter for the G.711 [7] coder is disappointing.

The echo attenuation under single and double talk conditions exceeds the recommended values. Double talk performance is characterized as "full duplex" for comparable near-end and far-end signal levels.

The activation of echo suppression may lead to noise modulation (tested with infinite ERL).

VAD and comfort noise generation do not significantly modulate the transmitted pub and café noises.

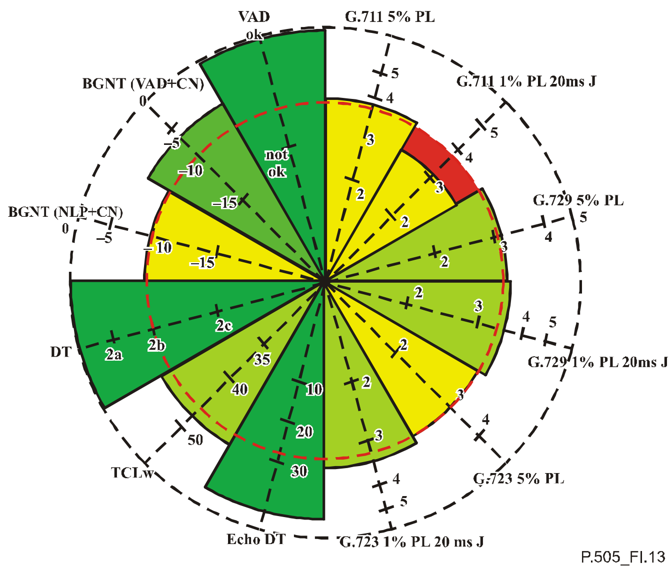


Figure I.13/P.505 – VoIP gateway 3

In Figure I.14, the listening speech quality scores are comparable or slightly higher (ITU-T Rec. G.711 [7], jitter) than the average scores.

The echo attenuation under single and double talk conditions exceeds the recommended values. Double talk performance is characterized as "full duplex" for comparable near-end and far-end signal levels.

The activation of echo suppression leads to audible and disturbing noise modulation (tested with infinite ERL).

Background noise is modulated in one-way transmission scenarios.

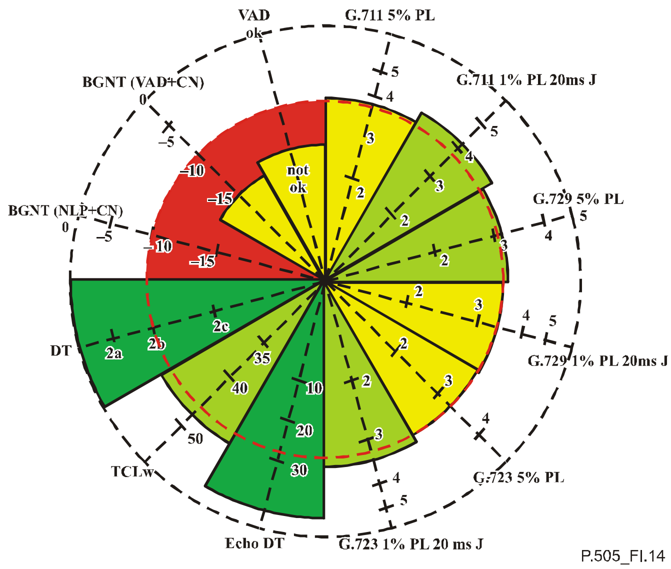


Figure I.14/P.505 – VoIP gateway 4

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