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Speech and multimedia Transmission Quality (STQ); Guidelines, objectives and results of speech quality analysis in the context of interworking Plugtests for multiplay services; Part 2: Results

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Foreword

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

The present document is part 2 of a multi-part deliverable covering Guidelines, objectives and results of speech quality analysis in the context of interworking Plugtests for multiplay services, as identified below:

Part 1: "Guidelines and objectives";

Part 2: "Results".

1 Scope

The present document shows the results obtained on technological watch platforms on Triple Play offerings. The determinate indicators and the used measurement methods are presented in part 1 of this multi-part deliverable [i.6]. The results shown come from a survey of various service performance, and show the applicability of the method provided in the part 1 of this multi-part deliverable [i.6] intended for Plugtests.

The main part of the present document defines the generic technological platform, the test conditions and provides general information about the test campaigns.

The different annexes detail each measurement campaign and the results obtained during these campaigns.

2 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 reference document (including any amendments) applies.

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2.1 Normative references

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

Not applicable.

2.2 Informative references

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.

[i.1] ETSI EG 202 765-2: "Speech Processing, Transmission and Quality Aspects (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics".

[i.2] ITU-T Recommendation P.862: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs".

[i.3] ITU-T Recommendation P.862.1: "Mapping function for transforming P.862 raw result scores to MOS-LQO".

[i.4] ITU-T Recommendation P.56: "Objective measurement of active speech level".

[i.5] ITU-T Recommendation P.505: "One-view visualization of speech quality measurement results".

[i.6] ETSI TR 102 716-1: "Speech and multimediaTransmission Quality (STQ); Guidelines, objectives and results of speech quality analysis in the context of interworking Plugtests for multiplay services Part 1: Guidelines and objectives".

3 Definitions and abbreviations

3.1 Definitions

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

platform: premise installed in residential environment where the accesses to different Multi Play offers proposed by ISP on the same country are made available

NOTE: This platform is generally installed in the centre of a city.

3.2 Abbreviations

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

ADSL Asymmetric Digital Subscriber Line
CPE Customer Premise Equipment
DTMF Dual Tone Multi-Frequency
HGW Home GateWay

NOTE: Referenced also as Residential Gateway.

IP Internet Protocol
IPTV IP TeleVision

NOTE: System where a digital television service is delivered using Internet Protocol.

ISP Internet Service Provider
ITU-T International Telecommunication Union - Telecommunication standardization sector
MGCP Media Gateway Control Protocol
MOS Mean Opinion Score
MOS-LQON Mean Opinion Store - Listening Quality Objective Narrowband
PDD Post Dialling Delay
PESQ Perceptual Evaluation of Speech Quality
PSTN Public Switched Telephone Network
SIP Session Initiation Protocol
TDM Time Division Multiplexing
VoIP Voice over Internet Protocol

4 Context

To have an overview of the performances of Triple Play offers deployed in France (and used by customers), several platforms dedicated to technological watch on Triple Play offerings were organized. These platforms consist in installing in the same place all the offers proposed by different ISP to residential customers. The offer subscriptions are made from the point of view of the user. Particular care is taken to make sure that the ISP cannot be aware of the real use of these offers. This is an important point because in such a way we can objectively determine the quality offered to the users. In fact, if the ISP is aware that an offer is made as part of a platform, it is then possible that the operator will adjust (or optimize) the functioning of this offer.

The results presented in the present document, concern the performances of VoIP service associated to Triple Play offers implemented on these technological platforms installed in France.

The results are not obtained during Plugtests but the implemented methodology and the indicators are perfectly in accordance with elements presented in TR 102 716-1 [i.6]. The interest of these results is to give an overview of the performance of deployed telephony services and not an overview of the performance of the prototype on a test platform.

Annex A presents results obtained in October 2009 on two technological platforms installed in two geographic locations for IP to PSTN call configuration.

Annex B presents results obtained during one year, from November 2008 till October 2009. These results refer to two platforms intalled in two cities in France. As for annex A, performance results are for IP to PSTN call configuration.

Annex C presents call performances obtained in IP to IP calls between different operators in comparison with IP to PSTN configuration. These measurements were performed in the same platform in January 2010.

5 Platform presentation

The platform is a premise installed in residential environment and in which there are the accesses to different Multi Play offers proposed by ISP on the same country. This platform is in most cases installed in the centre of a city.

The platform is characterized by:

- Ability to implement of different offers concerning different ISPs.
- Each offer proposes 2 or 3 services: Internet access and VoIP, and when possible IPTV.
- Access to services is obtained by an Home GateWay (HGW).
- Access to the network can be ADSL or cable type depending on the ISP.
- A PSTN access line is available for the speech quality analyses (IP to PSTN).
- As for network access, signalization protocol for voice service depend of ISP. So signalization protocol is not identical on all the offers and H.323, SIP or MGCP are implemented.

6 Presentation of test conditions

The indicators and the implemented method are identical for the platforms.

Concerning the tests, there is no difference between the offers and between the platforms. That allows the comparison of performance between the offers of the same platform and globally comparison of performance between the two platforms.

6.1 Indicator description

The indicators are the following ones.

6.1.1 Post Dialling Delay

Definition	Post Dialling Delay is the time interval between the end of dialling by the caller and the reception back by him of the appropriate ringing tone or recorded announcement. This indicator characterizes only the caller part of the call configuration.
Assessment method	Several measurements are performed sequentially and the mean value of measurement results represents the determined value of the indicator.
Unit	Millisecond.

6.1.2 Listening speech quality

Definition	Represents the intrinsic quality of speech signal after transmission. This indicator takes into account the degradations generated on the signal by the transmission links.
Assessment method	Voice quality is evaluated by using the ITU-T Recommendation P.862 [i.2] with the mapping functions according to ITU-T Recommendation P.862.1 [i.3]. Several MOS scores are determined in series during the same call. So listening speech quality performance during the call is defined by the mean value of MOS-LQON measurements (in the same transmission way). The voice quality indicator is determined in the two transmission directions by alternating the transmission way at each MOS score determination. For each transmission direction, 10 analyses are performed. As the duration of the voice sample for speech analysis is about 20 seconds and a MOS score is determined every 30 seconds, the duration of a test call is about 10 minutes.
Unit	Rating between 1 (= very bad) and 5 (= excellent) determines on MOS-LQON scale.

6.1.3 Listening speech quality stability

Definition	This metric represents the stability of the voice quality during a communication of several minutes long. This indicator takes into account the signal degradation due to the transmission links.
Assessment method	The MOS scores determined for speech quality evaluation are used to calculate the indicator characterizing speech quality stability. The methodology to perform this metric is described in EG 202 765-2 [i.1]. The major steps of stability indicator calculation are: - Determination of difference between successive MOS scores. - Evaluation of an instability level. - Transfer on a stability scale by using a linear function. This indicator is determined in the two directions of transmission.
Unit	Statistics on MOS score variation are plotted on a 0 to 100 scale.

6.1.4 End to end delay

Definition	Represent the global delay from one access to the other one. This indicator takes into account the transmission delay on networks but also processing delay in sending and receiving terminals.
Assessment method	Measuring the end to end delay is necessary to ensure a synchronization of both transmission ends of the measurement device. Because all communication terminations are co-located in the same area, the synchronization is done directly by the analyser. Several delay measurements are performed in series during the same call. The end to end delay during the call is defined by the mean value of delay measurements (in the same transmission way). The end to end delay is determined in the two directions of transmission by alternating the transmission direction at each delay measurement. For each transmission direction, 10 analyses are performed. End to end delay and MOS score are determined in the same test communication which has a duration of 10 minutes.
Unit	Millisecond.

6.1.5 End to end delay variation

Definition	This metric defines the stability of end to end delay during a communication of several minutes.
Assessment method	The values determined for end to end delay evaluation are used to calculate the indicator characterizing the delay stability. The methodology to perform this metric is described in EG 202 765-2 [i.1]. The major steps of stability indicator calculation are: - Determination of difference between successive end to end delay values. - Evaluation of an instability level. - Transfer on a stability scale by using a linear function. This indicator is determined in the two directions of transmission.
Unit	Statistics on delay variation are plotted on a 0 to 100 scale.

6.1.6 Level of active speech signal at reception

Definition	This indicator is the amplitude of speech signal received after transmission.
Assessment method	The received decoded signal used to determine MOS score (by using ITU-T Recommendation P.862 [i.2]) can also be used to assess this parameter. A typical method for the measurement of this parameter, based on a sample by sample approach and a moving threshold between noise and speech, is given in ITU-T Recommendation P.56 [i.4]. Several determinations of level are performed in series during the same call. So level of active speech signal at reception is defined by the mean value of level measurements (in the same transmission way). The level of active speech signal is determined in the two directions of transmission.
Unit	dBm

6.1.7 Noise level at reception

Definition	The metric is the level of noise determined at reception in non-speech segment of speech sample.
Assessment method	The received decoded signal used to determine MOS score (by using ITU-T Recommendation P.862 [i.2]) can also be used to assess this parameter. The measurement of these parameters is performed as for speech signal level but on the samples identified as non-speech. Several determinations of noise level are performed in series during the same call. So noise level at reception is defined by the mean value of noise level determinations (in the same transmission way). Noise level is determined in the two directions of transmission.
Unit	dBm0p

6.1.8 DTMF integrity

Definition	The metric characterizes the capability of telephony service to transmit correctly DTMF codes.
Assessment method	A specific test call is established. After call establishment, from caller part, all DTMF codes (0 1 2 3 4 5 6 7 8 9 A B C D * #) are sent in series. On called part, the received DTMF sequence is saved and analysed (for each DTMF code, frequencies and durations characteristics are checked). The call is released after reception of DTMF sequence. 10 tests are performed and for each test a specific call is established. So the 10 analyses are performed during different communications. The test is considered as "passed" if all DTMF codes of the 10 analyses are correctly transmitted and identified after reception. The test is considered as "failed" if one or more codes are not idenfitied after transmission.
Unit	Boolean (Passed or failed)

6.2 Use of platforms

The platforms may be used for two approaches:

- Measurement campaigns to analyze a specific topic or parameter and to have an overview of the performance,
- Regularly repeated analyses to follow performance changes over time.

6.3 Pie diagram presentation

An interesting presentation of the results is used within the framework of this activity is the Pie diagram (derived from ITU-T Recommendation P.505 [i.5]). This type of presentation offers on a single figure an overview of the performances. It is possible to present several metrics on the same graph by maintaining each indicator on its own scale. This type of presentation allows to easily display the strengths and weaknesses of each offer. The Pie diagram also allows to easily compare the offer performances.

Within the framework of these platforms, those indicators are presented on a Pie Diagram (derived from ITU-T Recommendation P.505 [i.5]).

An example of this type of graph is presented in figure 1. The color also allows discriminating between Mandatory and Optional indicators. In the case presented in figure 2, only the noise in the reception is Optional according to EG 202 765-2 [i.1].

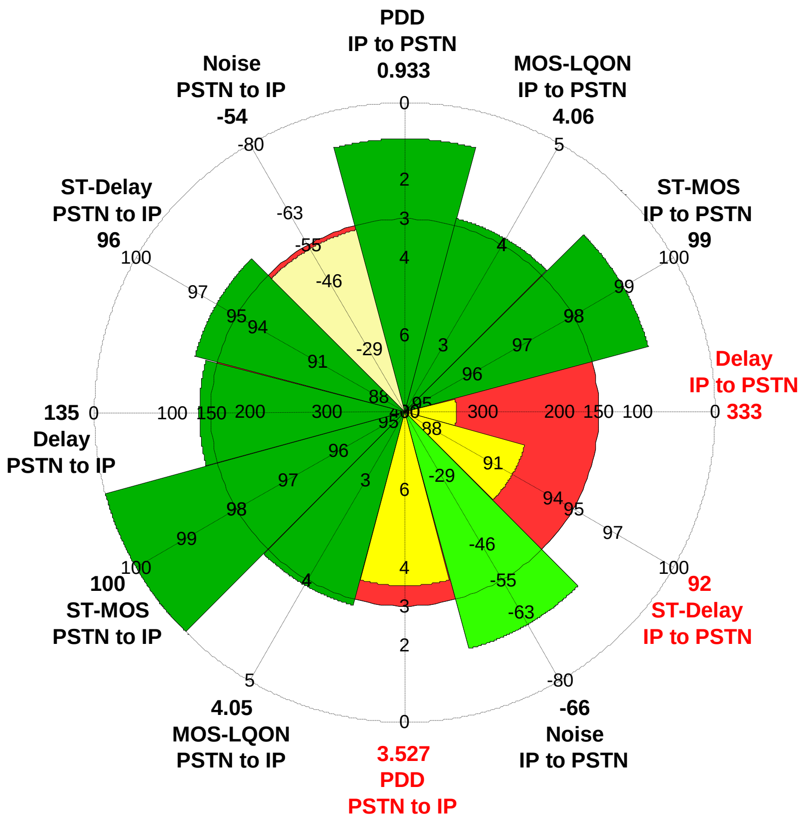


Figure 1: Example of Pie diagram with indicators determined within the framework of the two platforms of the VoIP offers

These 12 indicators correspond to 6 metrics by transmission way or call attempt direction: Post Dialling Delay, Listening speech quality, Stability of listening speech quality, End to end delay, Stability of end to end delay and Noise level at reception.

These indicators are presented in reference to acceptability thresholds.

The acceptability thresholds are represented by a red circle. The indicator value is green above and yellow below the threshold.

Annex A: Performance of VoIP for IP to PSTN connection condition. Overview of results obtained in October 2009

Annex A presents results obtained on two technological platforms installed in two geographic areas. These results concern IP to PSTN call configuration.

A.1 Platforms presentation

The technological platform N°1 is installed in a city of less than 250 000 residents whereas the platform N°2 is installed in a city of less than 25 000 people.

In addition to information given in clause 5, the platform N°1 is characterized by:

- Implementation of 7 offers concerning 7 different ISPs.
- Each offer proposes 3 services: Internet access, VoIP and IPTV.
- Distance between HGW and first digital equipment is about 350 meters (length of the ADSL line).
- For VoIP services, codec G.711 is implemented on each offer.

And the platform N°2 is characterized by:

- Implementation of 6 offers concerning 6 different ISPs.
- Each offer proposes 2 services: Internet access and VoIP.
- Distance between HGW and first digital equipment is about 2 000 meters (length of the ADSL line).
- For VoIP services, 2 codecs are implemented: G.711 and G.726 32 kbps.

Platform N°1	**Platform N°2**
7 offers (ISP1, IPS2, ISP3, ISP4, ISP6, ISP7 and ISP8) Access technology to network: ADSL and cable Distance to first digital equipment: 350 m Codec deployed: G.711	6 offers (ISP1, ISP3, ISP4, ISP5, ISP7 and ISP8) Access technology to network: ADSL Distance to first digital equipment: 2 000 m Codec deployed: G.711 and G.726 32 kbps

Note that between the 2 platforms, there is 5 common ISP (ISP1, ISP3, ISP4, ISP7 and ISP8).

A.2 Description of the methodology

A monthly analysis is made on both platforms. Every month, on each platform, an analysis of vocal quality is made on each offer. The methodology allows to have every month an overview of the quality of VoIP service proposed to the users and to see how this quality progresses on a rather long duration (one year for example).

A campaign of measurements is performed every month on each platform. The analysis of speech quality is made on the IP to PSTN configuration. The determined indicators are presented in clause 6.1.

Figure A.1 presents the overview diagram of the implemented chain of measurement.

The analysis is made between two electric accesses of the test communication. One of the two accesses of the analyzer is connected to the PSTN, the other one is connected to a switch interface which allows a sequential connection to the analog access of every HGW. This switch interface allows to analyze sequentially the different VoIP offers on the IP to PSTN configuration.

For each offer, the test protocol is identical:

- Calibration of the measurement chain.
- Measurement of the different indicators inside the same call.
- Measurement of PDD concerning IP to PSTN call establishment.
- Measurement of PDD concerning PSTN to IP call establishment.

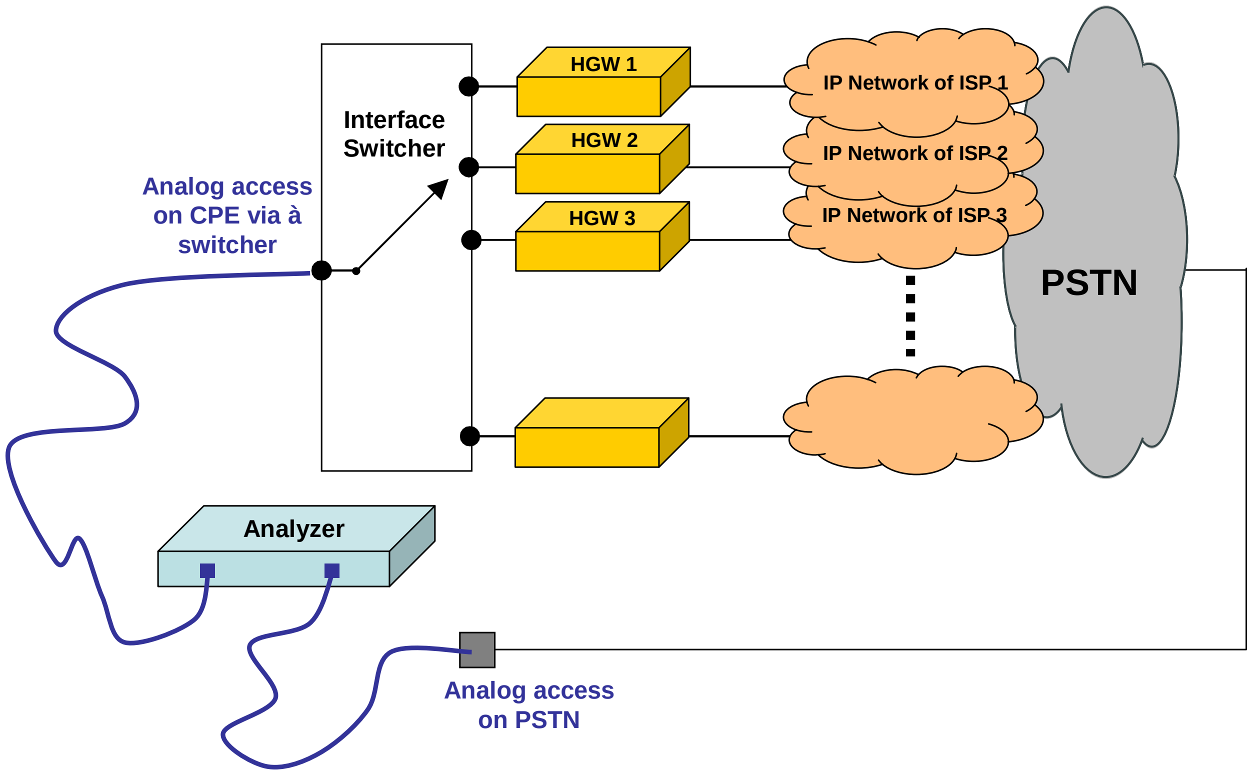


Figure A.1: Overview diagram of the measurement chain deployed on both platforms of follow-up

Analysis sequencing is always the same: analysis of offer 1, analysis of offer 2, analysis of offer 3 and so on. Globally, every month the offers are analyzed in similar time slots.

Notice that the performances of the VoIP services are determined in absence of load (without other streams associated of other applications like Internet or IPTV).

NOTE: The delay introduced between DSLAM and Home Gateway depends on the specific brand of the Home Gateway, manufacturer and the DSLAM manufacturer and their combination, as well as other factors such as bandwidth, interleaving, etc. As usual practice, this has not been taken into account for the survey of various service performances as presented in the present document. However it should be taken into account for future Plugtests.

A.3 Results presentation

Pie diagrams obtained for October 2009 on Platform N°1 are presented in figure A.2.

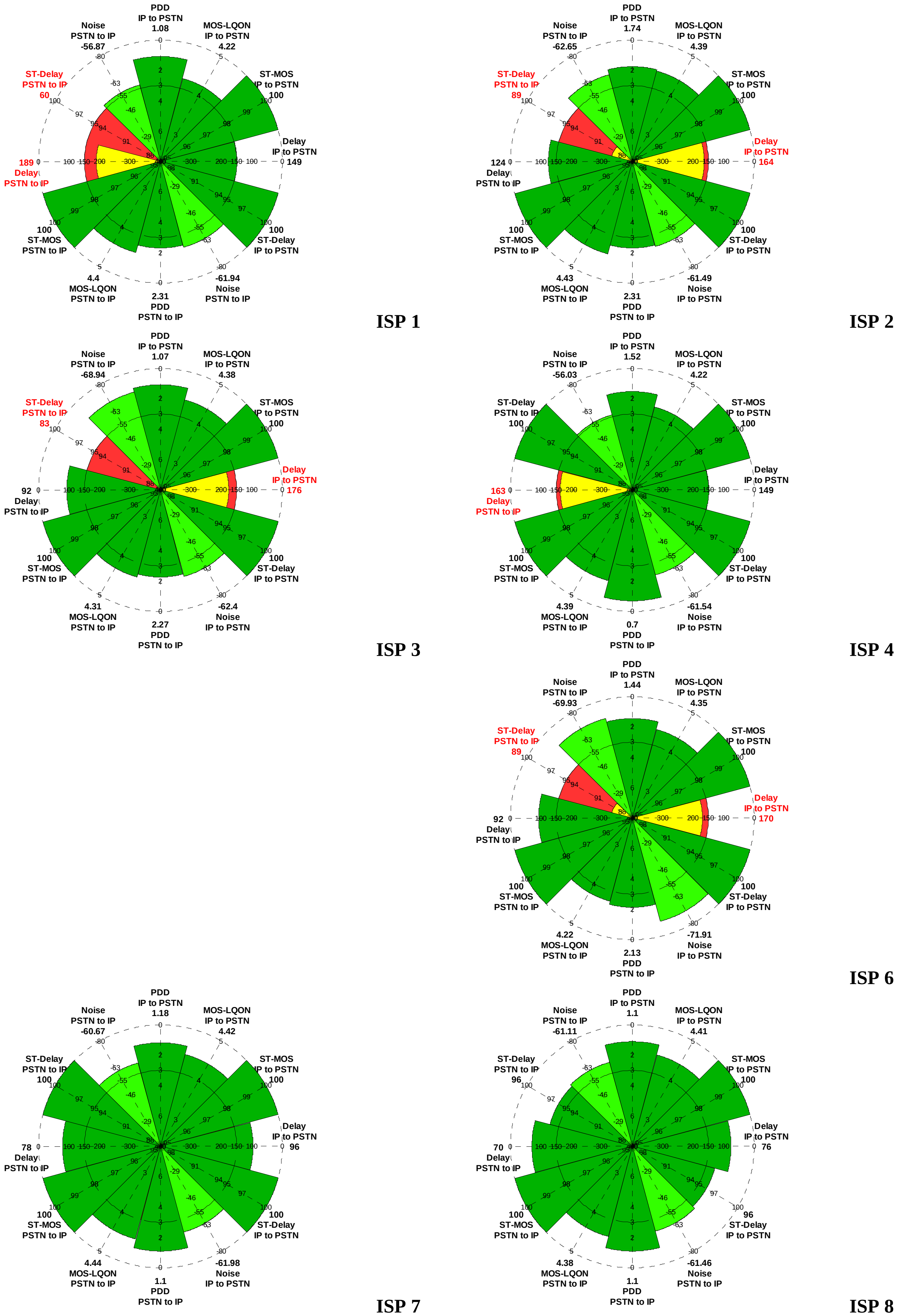
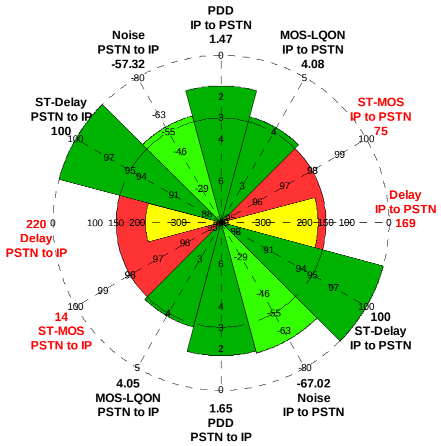
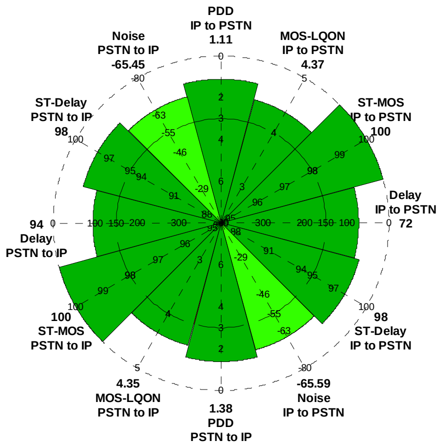


Figure A.2: Results obtained in October 2009 on Platform N°1

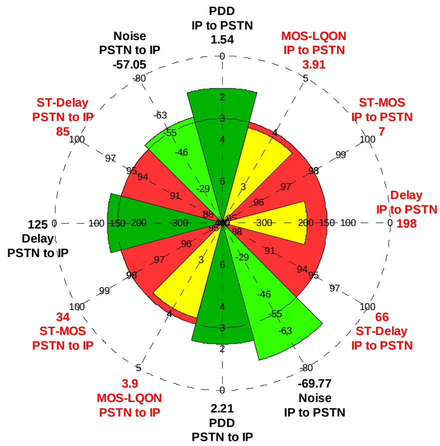
Pie diagrams obtained for October 2009 on Platform N°2 are presented in figure A.3.



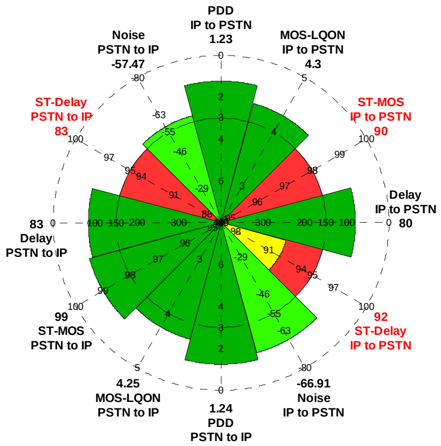
ISP 1



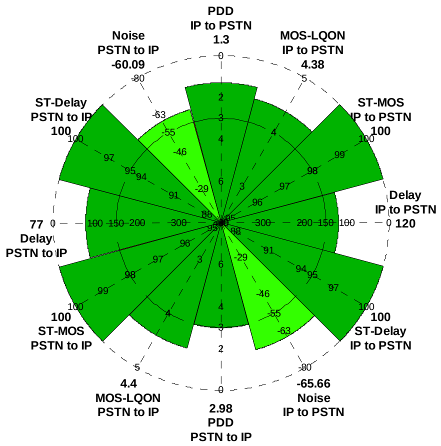
ISP 3



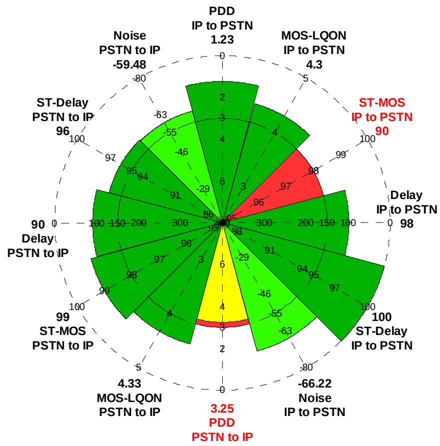
ISP 4



ISP 5



ISP 7



ISP 8

Figure A.3: Results obtained in October 2009 on Platform N°2

The results obtained on platform N°1 show that the call establishment performances are correct. PDD values are lower than 3 seconds for all the offers in the 2 directions of call.

Speech quality is characterized by MOS scores higher than 4,2. The performances are thus in accordance with the quality level expected with codec G.711. The results show also that the transmission does not degrade significantly the quality of speech signal.

Besides, the ST_MOS metric is equal to 100 % for all test configurations. Speech quality is thus perfectly stable during a communication of 5 minutes (test communication duration).

Concerning the end to end delay, the measurements indicate that this characteristic is lower than 200 ms on all configurations (for each offers and each transmission direction). We notice that for two offers (ISP7 and ISP8) the delay is lower than 150 ms in the two transmission directions.

For all the offers, in the transmission direction IP to PSTN, the delay variation is low. But in the transmission direction PSTN to IP, the variation is more significant for four of the offers.

Concerning the noise at the reception, even if in certain cases the level is high (close to -56 dBm) the related performance fullfills the standardized value (-65 dBm) in most cases.

Concerning all the indicators, the offer ISP7 and ISP8 show a performance slightly superior to the other offers.

On the platform N°2, the call establishment performances are lower than 3 seconds except for the offer IPS8 in the call direction PSTN to IP where PDD is slightly higher than 3 seconds.

Speech quality is characterized by MOS scores higher than 4,0 except for the offer IPS4 where average MOS scores are equal to 3,9. This lower performance for the offer ISP4 results from the codec used. On this offer the negotiated codec is G.726 32 kbps while on the other offers the negotiated codec is G.711.

For the offers ISP3 and ISP7, the MOS stability is optimal (ST_MOS=100 %). On the other hand the stability associated to speech quality on the other offers is lower. A weak MOS stability is shown for the offer ISP4 in both transmission directions.

End to end delay is lower than 200 ms for all the offers except for IPS1 in the direction PSTN to IP where the average delay is 220 ms.

As on the platform N°1, the noise at the reception, even if in certain cases the level is high (close to 57 dBm) this performance is globally correct.

Concerning all the indicators, offers ISP3 and ISP7 show performances slightly higher than the other offers.

If we compare the results obtained on the two platforms, we notice that the performances are slightly superior on the platform N°1. We also note that the offer ISP7 presents very correct and very similar performances on both platforms. On the other hand we also note that the offer ISP4 presents different performances on both platforms. This observation can be partially explained by the difference between negotiated codecs.

A.4 Conclusion

The implementation of technological platforms on Triple Play offerings allows to obtain an overview of performance of VoIP offers proposed to residential customers.

The measurement results show that the performance for the same ISP can be globally slightly different on different geographical areas (tests were performed on two significantly distant areas). We notice also that the performance can be significantly different between ISPs, which is the case for one particular offer. We also notice that for one particular offer the characteristics are almost identical for the two geographical areas.

Annex B: Performance of of VoIP for IP to PSTN connection condition. Overview of results obtained over one year

This annex presents a summary of the results obtained over one year, from November 2008 till October, 2009. The results refer to the platforms and performances are presented indicator by indicator and detailed in annex A.

B.1 Platforms and methodology presentation

Both technological platform and methodology are presented in annex A (clauses A.1 and A.2)

For each metric, the summary is presented by 8 graphs where both platforms and both transmission directions (or call attempt direction for PDD) are represented.

B.2 Post Dialling Delay

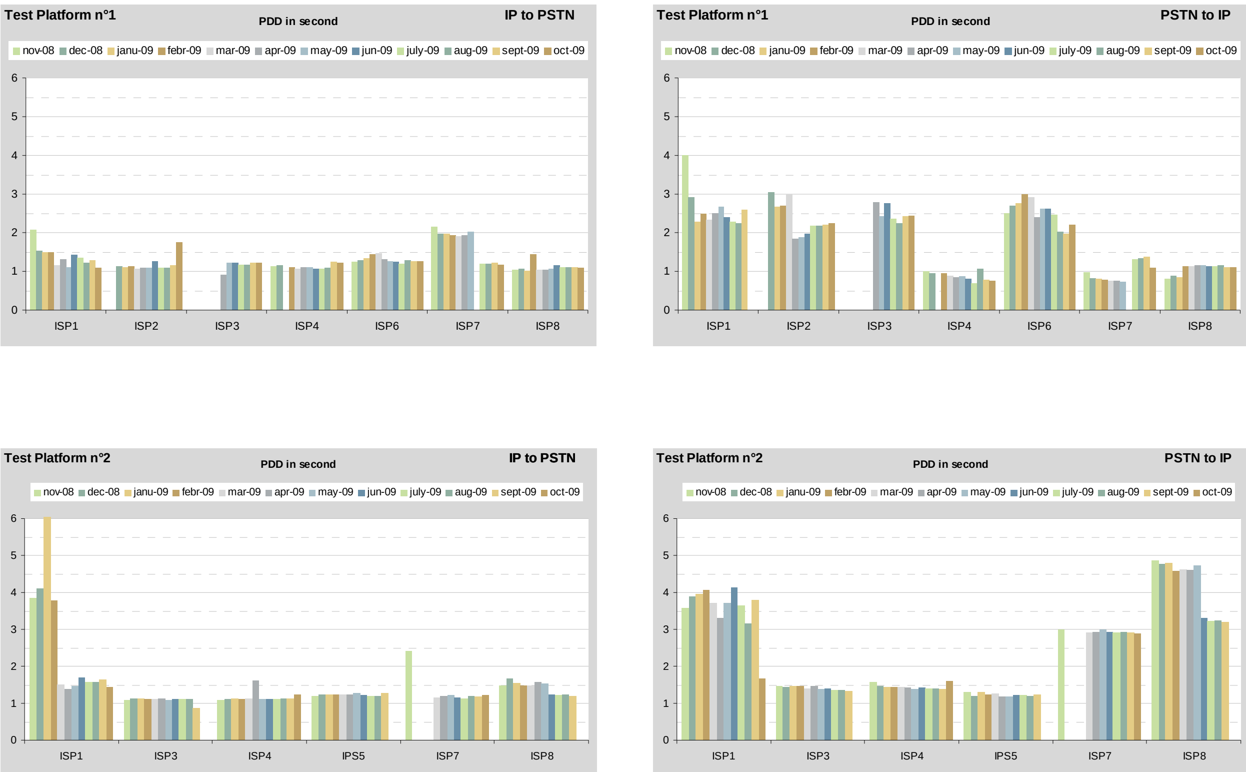


Figure B.1

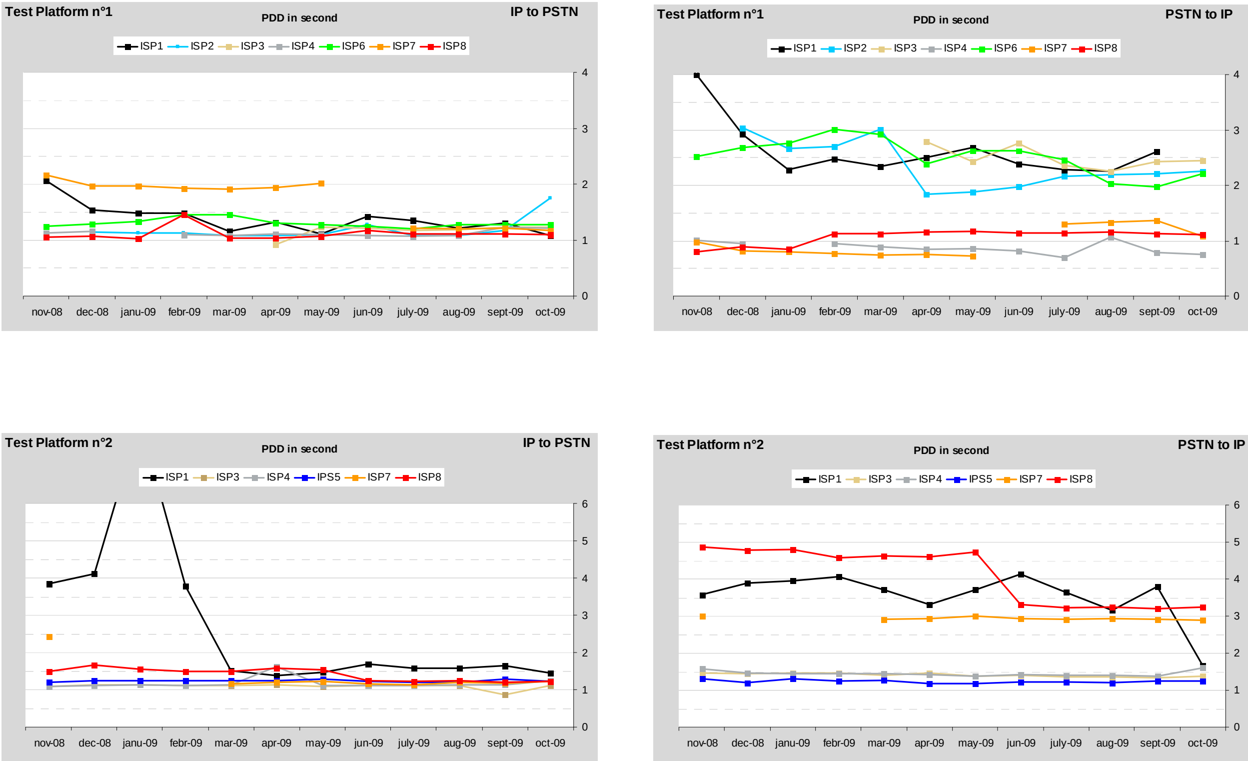


Figure B.2

Concerning the call establishment performances, a subjective study has highlighted that the users feel annoyance when the PDD exceeds 6 seconds or 7 seconds.

Except for the results obtained for the ISP1 in January 2009, in these case studies the performances are lower than 5 seconds. Globally, we notice identical performances on the two platforms with call establishment delays lower in the IP to PSTN call establishment direction than in the PSTN to IP direction.

If we examine the results in more detail, we notice that in the IP to PSTN direction there are no significant differences between the offers, as the establishment of the call is achieved within 2 seconds in the vast majority of the tests. While in PSTN to IP direction, we notice significant differences between the offers because some offers establish the call within 2 seconds and others take more than 2 seconds.

If the performances are globally very similar on the two platforms, we notice for three ISP (ISP 3, ISP 7 and ISP8) differences between the offer installed on the platform N°1 and the offer installed on the platform N°2:

- for the ISP 3, the PDD is higher on platform N°1;
- for ISP7 and 8 the PDD is higher on platform N°2.

B.3 Listening speech quality

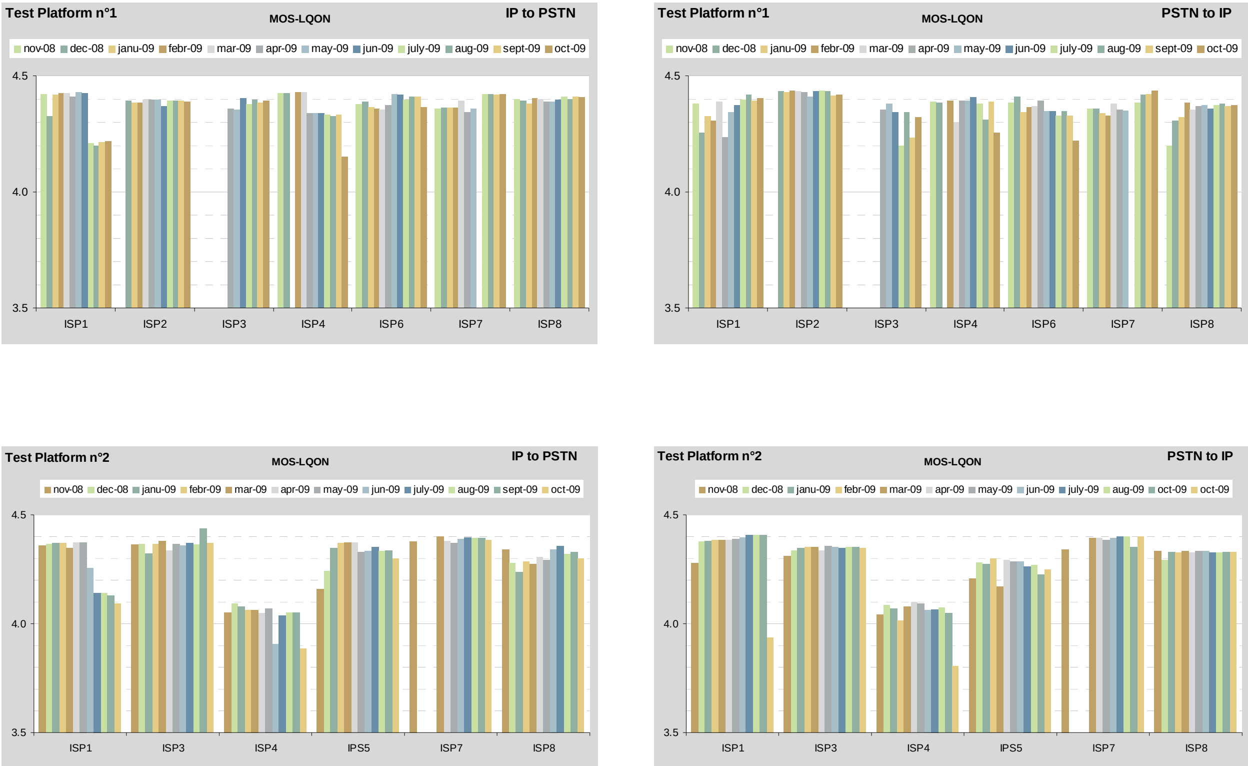


Figure B.3

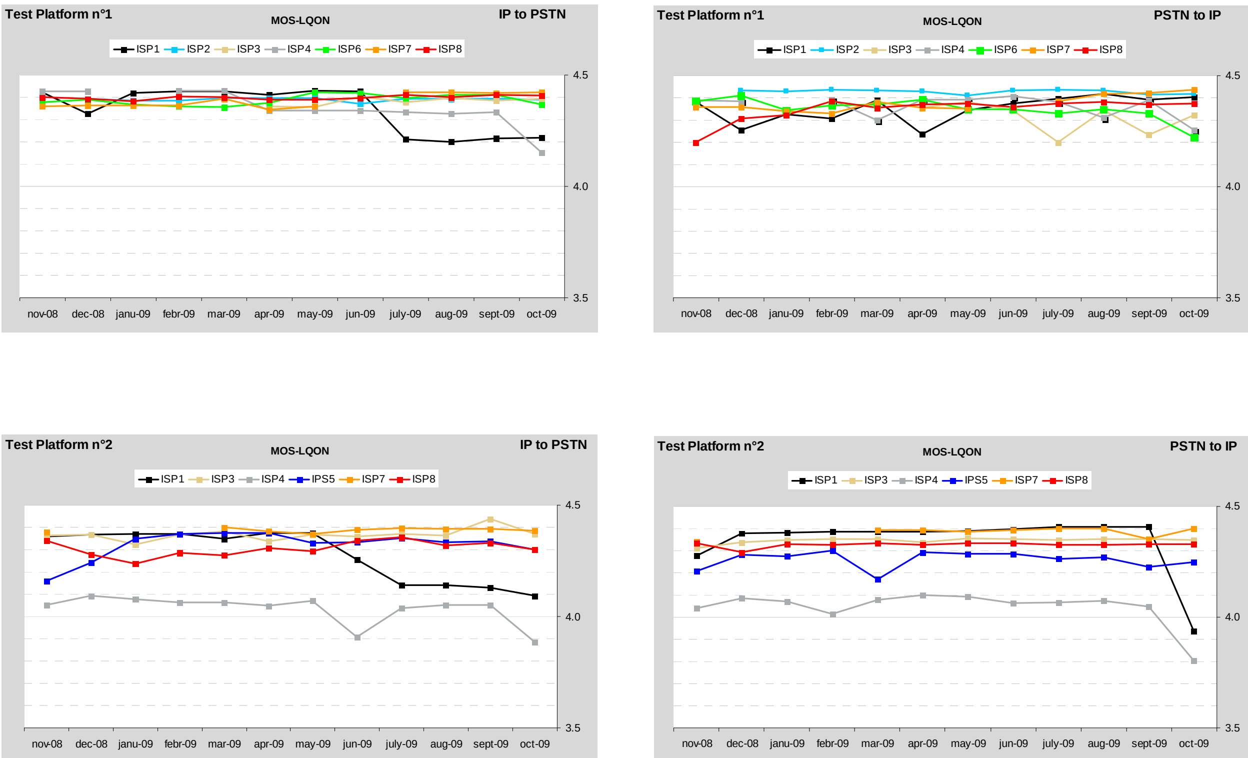


Figure B.4

Concerning the speech quality, we do not notice significant difference between the two platforms even if average MOS scores (calculated without the results obtained on the ISP4 offer) is slightly superior on the platform N°1. In the same way, we notice no difference between the performances according to transmission direction.

Globally with MOS scores higher than 4,2, we can confidently state that speech quality is good.

The most noticeable point is the difference of performance between platform N°1 and platform N°2 for the ISP4. For the platform N°1, the speech quality is characterized by a MOS score about 4,3 whereas for the platform N°2, the speech quality is characterized by a MOS score 4,0.

This is due to the different negotiated codecs. On platform N°1 the codec G.711 is used and on platform N°2 the codec G.726 32 kbps is used. This difference (for codec implementation) is imposed by the ISP. It depends on the geographical zone and on the type of ADSL option deployed (option 5 or option 3).

B.4 Listening speech quality stability

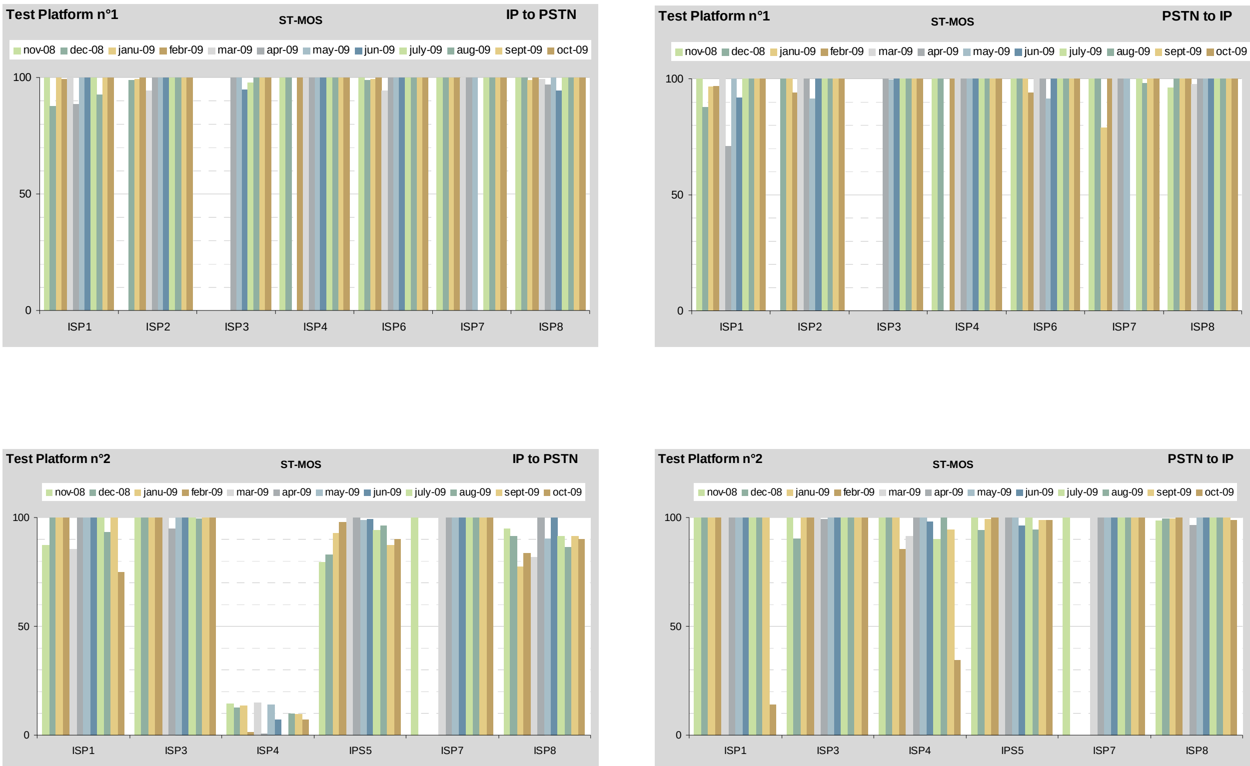


Figure B.5

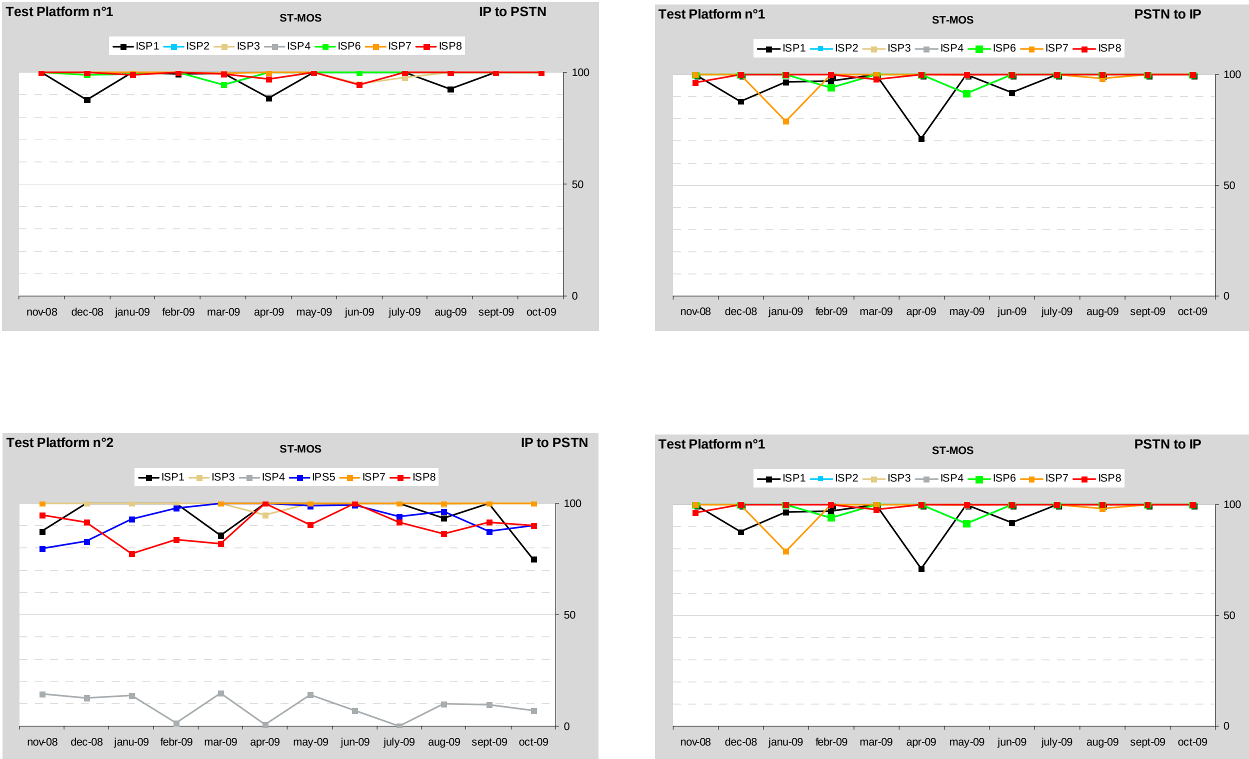


Figure B.6

Concerning the speech quality, the stability is not bad. Except for the performances of the offer of ISP4 on platform N°2, in IP to PSTN transmission direction, the stability (according to transmission way) is characterized by mean values 99 % and 98 % on platform N°1 and by the mean values 94 % to 97 % on platform N°2. Hence normally we notice that the stability of speech quality is appreciably higher on the platform N°1.

We can notice a very weak stability in the IP to PSTN transmission direction for the offer of the ISP4 deployed on platform N°2. This point is remarkable because the stability for this offer is good in the other transmission way (PSTN to IP). Besides, the offer of ISP4 deployed on platform N°1 does not present any problem of stability.

Another point worth noticing on platform N°2: in October 2009, the stability in the PSTN to IP transmission direction is highly degraded for the offers of ISP1 and ISP4.

B.5 End to end delay

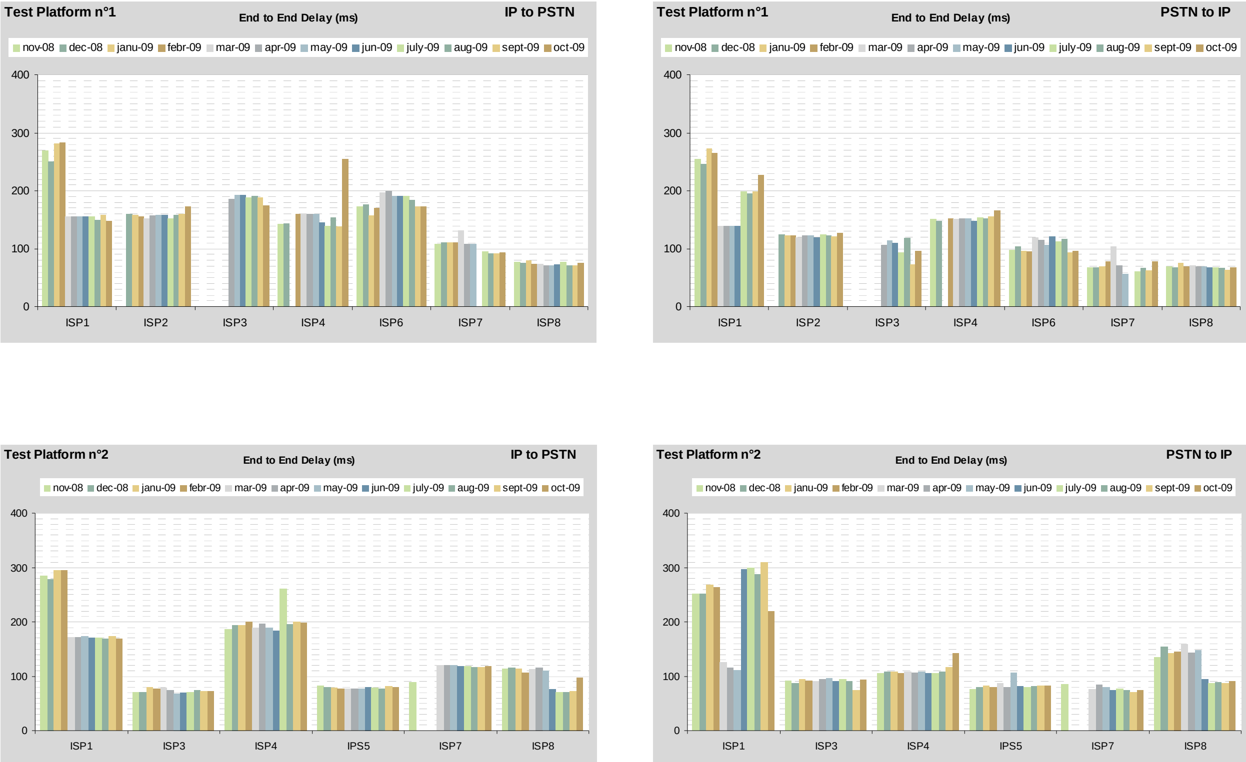


Figure B.7

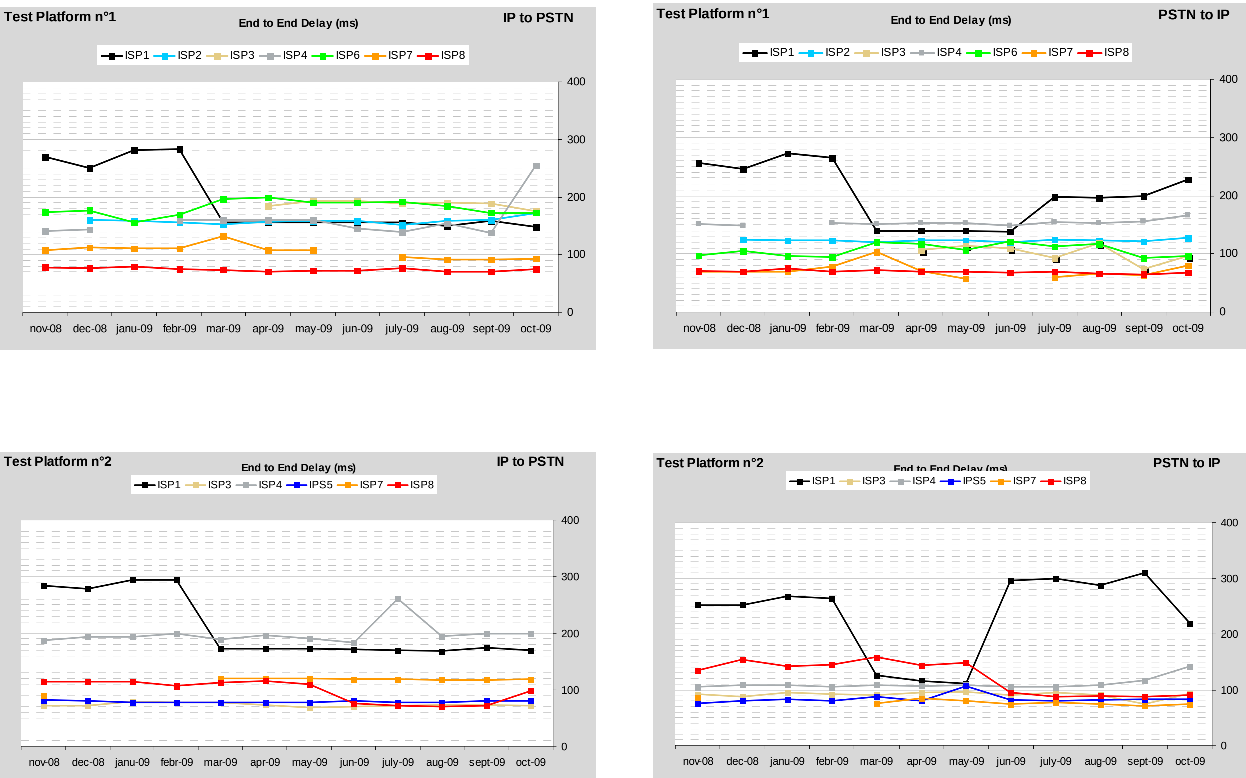


Figure B.8

Concerning the end to end delay, we notice that in the PSTN to IP transmission direction, the delay is globally lower or close to 150 ms (on platforms N°1 and N°2).

In the IP to PSTN transmission direction, the delay is globally higher than in PSTN to IP direction but it remains lower than 200 ms.

The delay for the ISP1 offer can reach 300 ms in certain cases. The last four figures clearly show that the delay varies strongly from one month to another.

B.6 End to end delay variation

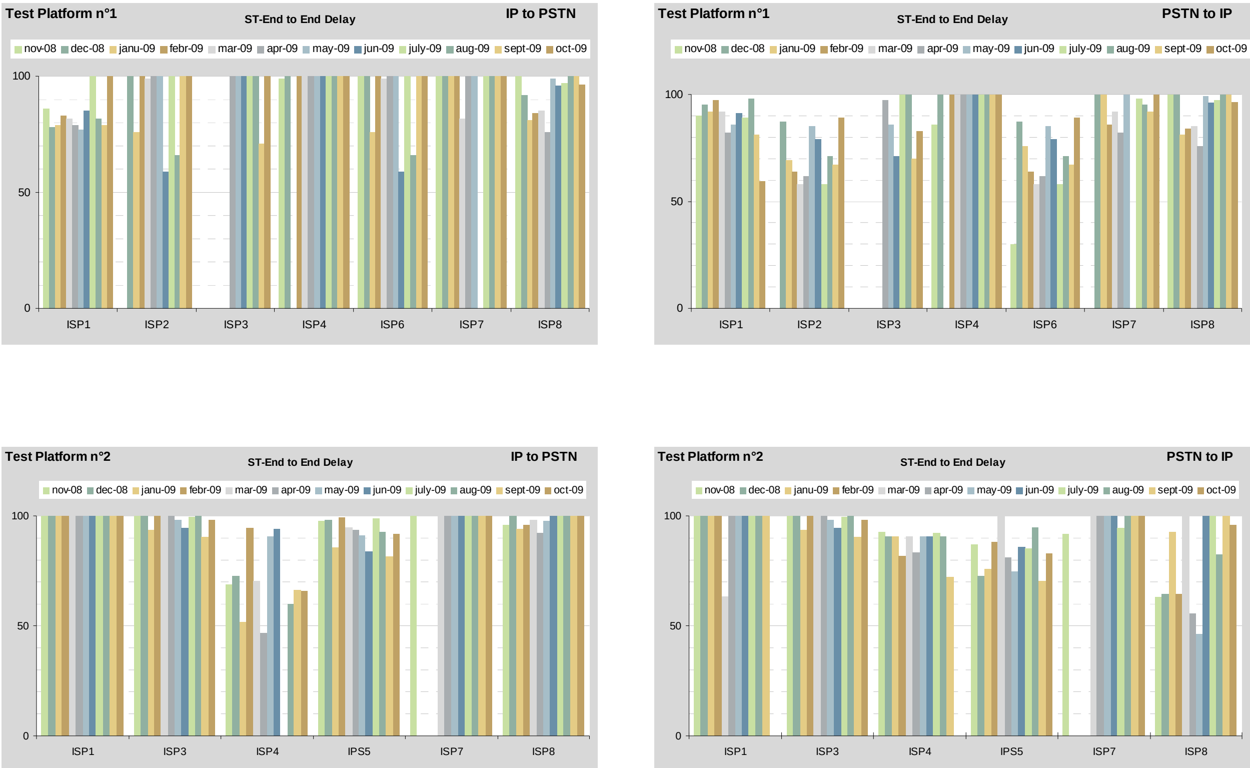
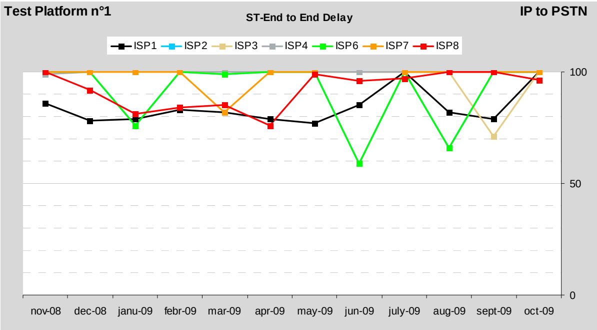
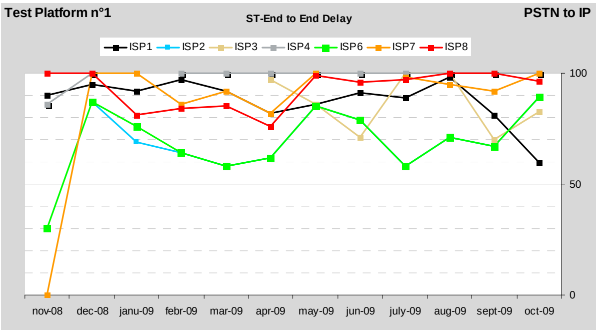
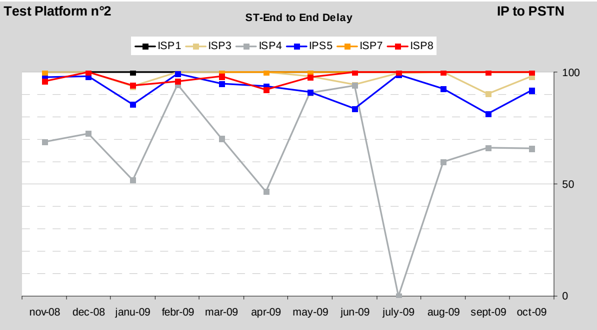


Figure B.9







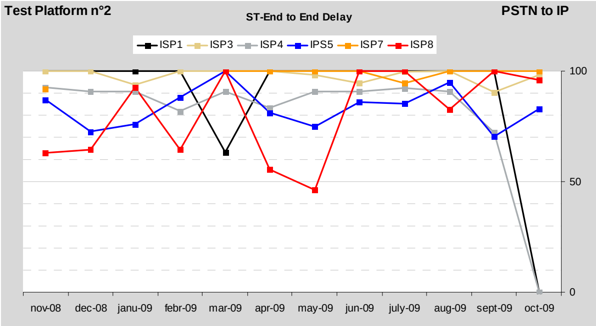


Figure B.10

Concerning the variation of delay, we notice a relative consistency of the performances between all the offers. Therse is no offer with a very weak stabilityand there is no offer with a perfect stability (indicator value equal to 100 %) in both transmission directions, throughout the whole year. On the other hand two offers (ISP1 and ISP7) on platform N°2 present a perfect stability in IP to PSTN transmission direction.

B.7 Noise level at reception

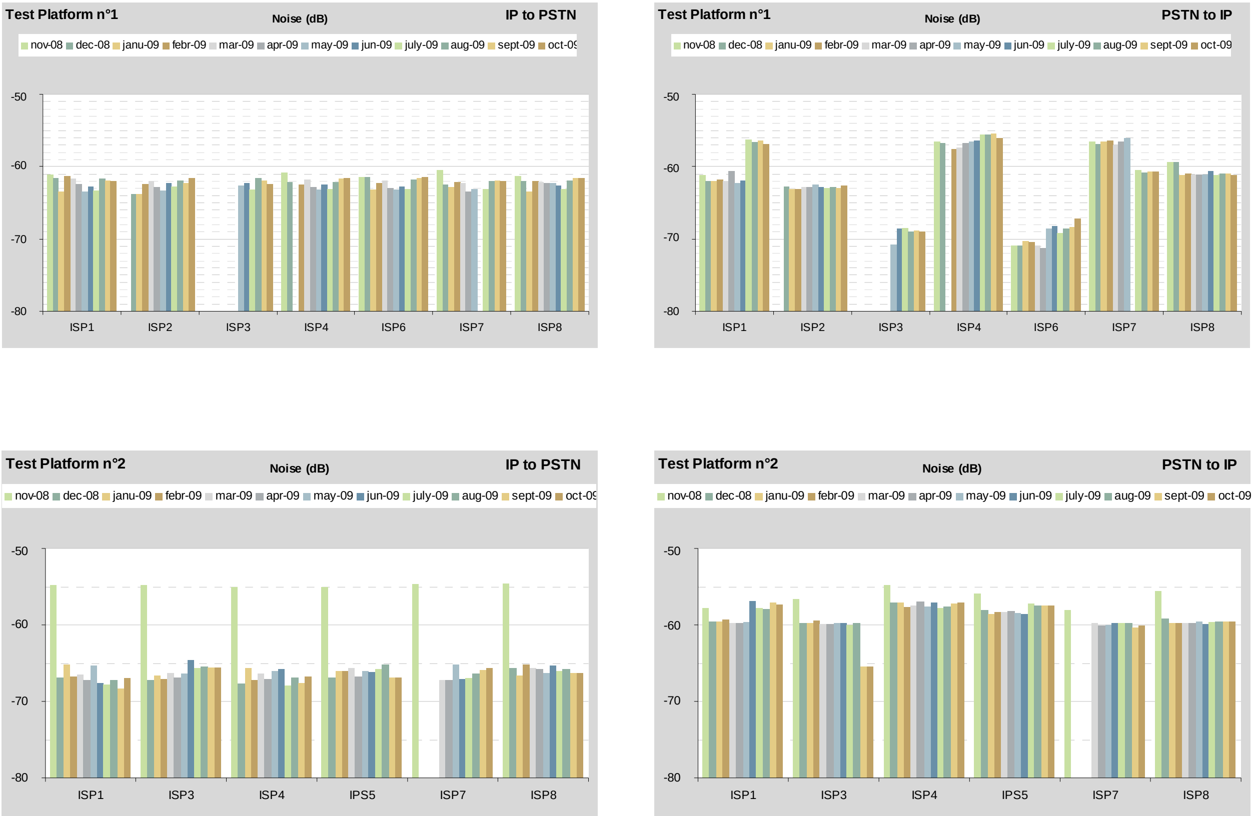


Figure B.11

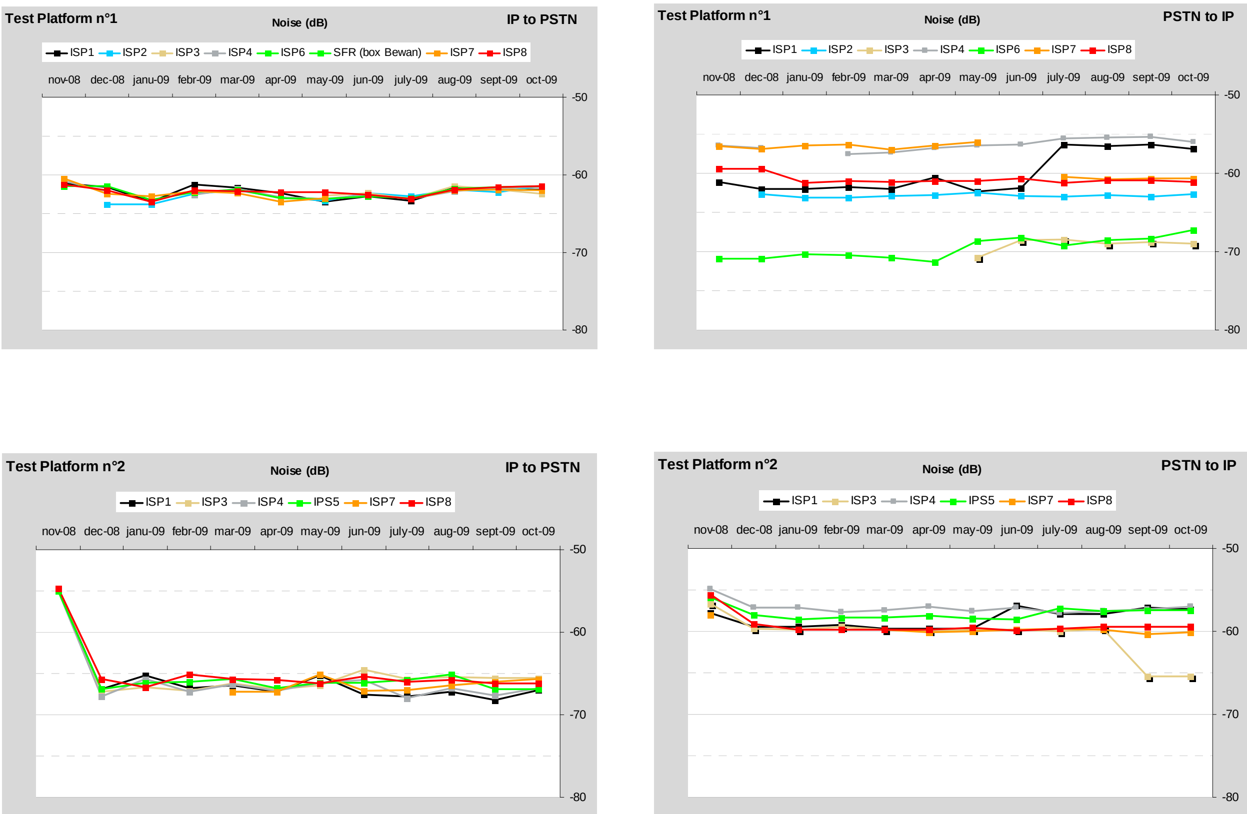


Figure B.12

On platform N°1, the noise level at the reception is relatively high with a mean level of -62 dBm on the IP and PSTN sides. On the PSTN side, the noise level at the reception is very similar for all offers. This noise level corresponds to the level of noise on the PSTN line. On the IP side, we notice some differences between the offers. The offers of ISP3 and ISP6 present a rather weak level of noise (-70 dBm on average) at the reception. The offer of the ISP4 presents a higher noise level (mean value equal -57 dBm) close to the acceptability threshold.

On platform N°2, the situation is different. On the PSTN side, the noise level at the reception is rather weak (-67 dBm on average). This performance is due to the noise level of the PSTN line. Note that in November 2008, the performances of the PSTN line were degraded with a noise level of -55 dBm. On the IP side, the noise level is consistent for the various offers, with a mean level of -59 dBm. We do not notice on this platform substantially higher or lower performance on certain offers.

B.8 Conclusion

As seen in annex A, the implementation of technological platforms on Triple Play offerings allows to obtain an overview of performance of VoIP offers proposed to residential customers. These platforms also allow to follow the evolution of the performances.

Concerning the evolution of the indicators in time, we notice that the performances are globally stable over a period of one year. However for a specific offer, we notice an improvement of the PDD and a degradation of the transmission delay.

For the same geographical area, we notice some differences between offer performance, and we can conclude that certain offers highlight specific areas of improvement.

Annex C: Performance of VoIP offers in interconnection conditions

The present document presents results obtained by tests realised to characterize IP to IP calls between different operators.

With the development of VoIP in residential context, the IP to IP traffic becomes more and more significant and the interconnection between IP networks may introduce degradation on call performance. The main objective of the tests described in this annex is to give an overview of call performance in the specific context of "IP to IP calls" in interconnection between different ISP networks.

These tests have been performed on a platform dedicated to analyses and performance measurements. This platform gives the possibility to install in a same location all the offers for residential customer from different ISP.

C.1 Context

The objective of such an experiment is to provide an overview of the performance of the Triple Play offers deployed and used by real customers in one country. A platform dedicated to technological watch of Triple Play offers is used for this test campaign. This platform gives the possibility to install in a same location all the offers for residential customers from different ISP.

It should be ensured that the offers assessed are the ones commercially available. The platform should act as real users and subscribes to the offers. Care should be taken that the ISPs cannot be aware of these experiments. ISPs do not have the possibility of adjusting (or optimizing) the functioning of this offer.

The results of this test campaign express the performance of VoIP service associated to Triple Play offers in a specific call configuration: IP to IP communication for inter-ISP connection. This configuration concerns calls established from an ISP VoIP offer to another ISP VoIP offer (see figure C.1).

In the context of development of VoIP traffic, there is an interest in estimating the performance of IP to IP calls between operators and to compare them with the IP to PSTN calls. At the beginning of the deployment of the voice over IP, the traffic was mainly IP to PSTN. With the increase of deployments of HGW (Home GateWay) in the residential context, the IP to IP traffic becomes more and more significant. The IP to IP configuration between operators is a relevant configuration to be examined and to be tested because the interconnection between IP networks is a potential source of impairment for performance of phone calls.

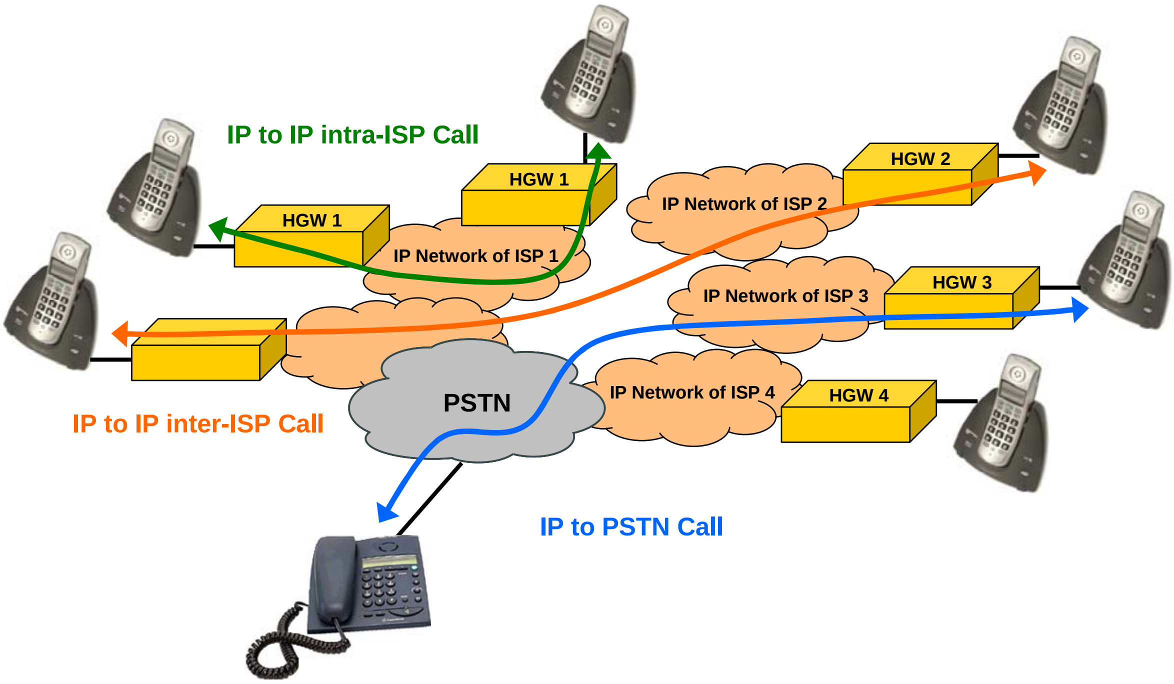


Figure C.1: Presentation of different call configurations

C.2 Platforms presentation

The technological platform is installed in a location where all Triple Play offers are available in the same conditions.

This platform is characterized by:

- Implementation of 7 offers from 7 different ISPs.
- Each offer proposes 3 services: Internet access and VoIP and when possible IPTV, but tests may be done even if only two services are available (Internet access and VoIP).
- Access to services is reached through an Home GateWay (HGW).
- Access to the network can be ADSL or cable type depending on the ISP.
- Distance between HGW and first digital equipment is about 350 meters (length of the ADSL line).
- A PSTN access line is available for the speech quality analyses (for IP to PSTN connexion).
- For VoIP services, each offer implements the codec G.711.
- Signalization protocol is not identical for all the offers from the different ISP: H.323, SIP or MGCP are implemented.

C.3 Description of the methodology

A measurement campaign was performed in January 2010 on the technological platform. An analysis of speech quality was made for each VoIP offer on the different IP to IP configurations with other operators. A test was also performed on IP to PSTN configuration. In this condition, it is possible to compare the performance between both types of call configurations.

Figure C.2 presents the overview diagram of the implemented measurement chain.

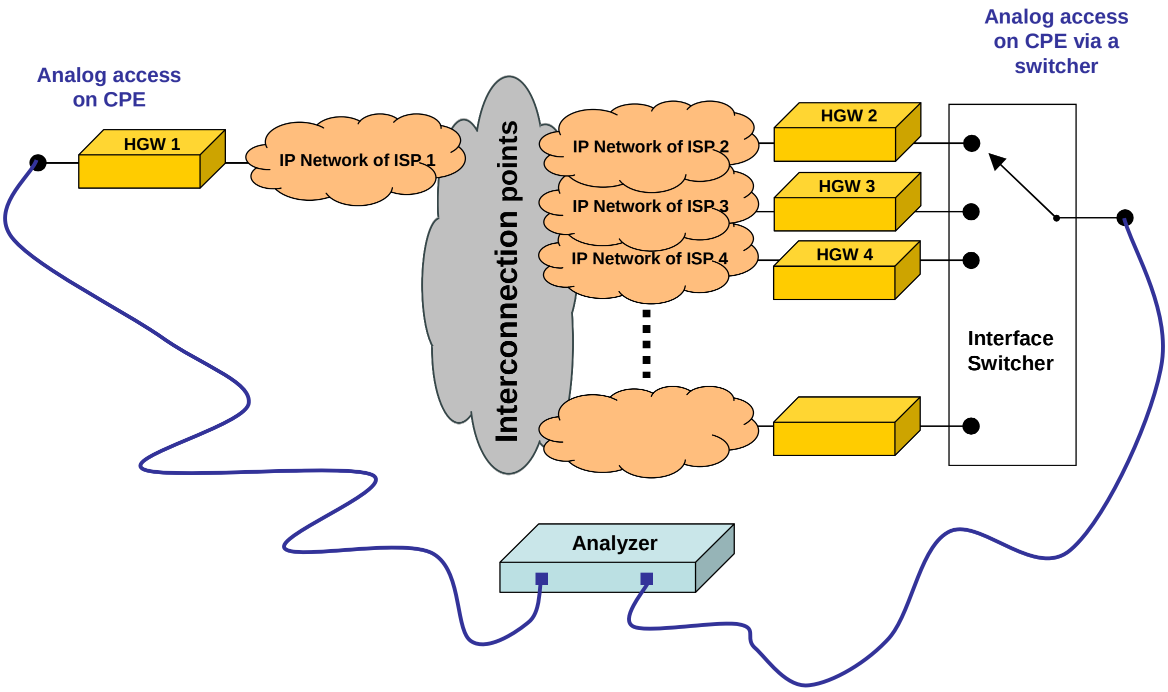


Figure C.2: Overview diagram of the measurement chain deployed for IP to IP inter-operator configuration

The analysis is made between two electrical accesses of the test communication. One of the two accesses of the analyzer is connected to the HGW of the offer under test; the other one is connected to a switch interface which allows a sequential connection to the analogue access of the other HGW. This switch interface allows to analyze sequentially the different IP to IP inter-operator configurations.

Test calls are currently always established from the HGW under test. The test calls are performed in both directions.

For each offer, the test protocol is identical:

- calibration of the measurement chain;
- for each IP to IP configuration, measurement of speech quality and delay indicators during the same call;
- for each IP to IP configuration measurement of PDD in both call establishment way (path);
- for each IP to PSTN configuration, measurement of speech quality and delay indicators during the same call;
- for each IP to PSTN configuration, measurement of PDD in both establishment way (path) (IP to PSTN and PSTN to IP).

Analyses were realized in sequence:

- analysis of offer 1 (first IP to IP configurations them IP to PSTN configuration);
- analysis of offer 2 (first IP to IP configurations them IP to PSTN configuration);
- analysis of offer 3 (first IP to IP configurations them IP to PSTN configuration) and so on.

Note that the performance of the VoIP services are determined in absence of load (without other streams associated of other applications like Internet or IPTV).

In practice, 7 offers should have been tested in this first measurement campaign on inter-ISP call configurations, however the ISP5 offer has not been made available. So results concern only the offers of 6 operators (ISP1, ISP2, ISP3, ISP4, ISP5 and ISP7).

C.4 Overview of results obtained in January 2010

Results are presented as a Pie diagram (derived from recommendation ITU-T Recommendation P. 505).

We use the Pie diagram to present, on the same figure, the performance on the different call configurations. On each P.505 figure we plot the same indicator measured in different call configurations.

These indicators are presented in reference to acceptability thresholds. These thresholds are represented by a red circle. The indicator value is green above and yellow below the threshold.

The acceptability thresholds used are:

- 3,0 seconds for PDD;
- 4,0 for MOS-LQON;
- 150 ms for end to end delay (one way delay).

The colors also allow discrimination between transmission direction (or call establishment direction):dark green (and dark yellow) for metrics associated to the path HGW under test → others HGW and light green (and light yellow) for the other direction.

For each indicator (PDD, listing speech quality and end to end delay), 7 Pie diagrams are presented: 6 diagrams by IP to IP inter-operator configuration (1 diagram by offer) and 1 diagram for the IP to PSTN configurations.

C.5 Post Dialling Delay

These results show that the call establishment performance for **IP to PSTN** and **PSTN to IP** configuration are lower than 3 seconds (average of measurement is about 1,2 seconds) for all the offers in the 2 call directions. These values are currently expected by users.

The results for **interconnection between operators** globally show small PDD increases with regard to the IP to PSTN or PSTN to IP calls. Nevertheless, we notice for some configurations a significant increase of the establishment performance. For the offers associated to IPS3, ISP 4 and ISP6 it takes more than 5 seconds (between 5,3 and 5,7 seconds) to obtain the ring back tone after dialling. It should be noted that these establishment delays are lower than those currently measured for mobile calls in roaming situation (currently delays are more than 10 seconds).

For the offer associated to IPS1, the average value of the PDD (in situation of interconnection with other operators) is lower than the average value of the PDD for IP to PSTN and PSTN to IP calls.

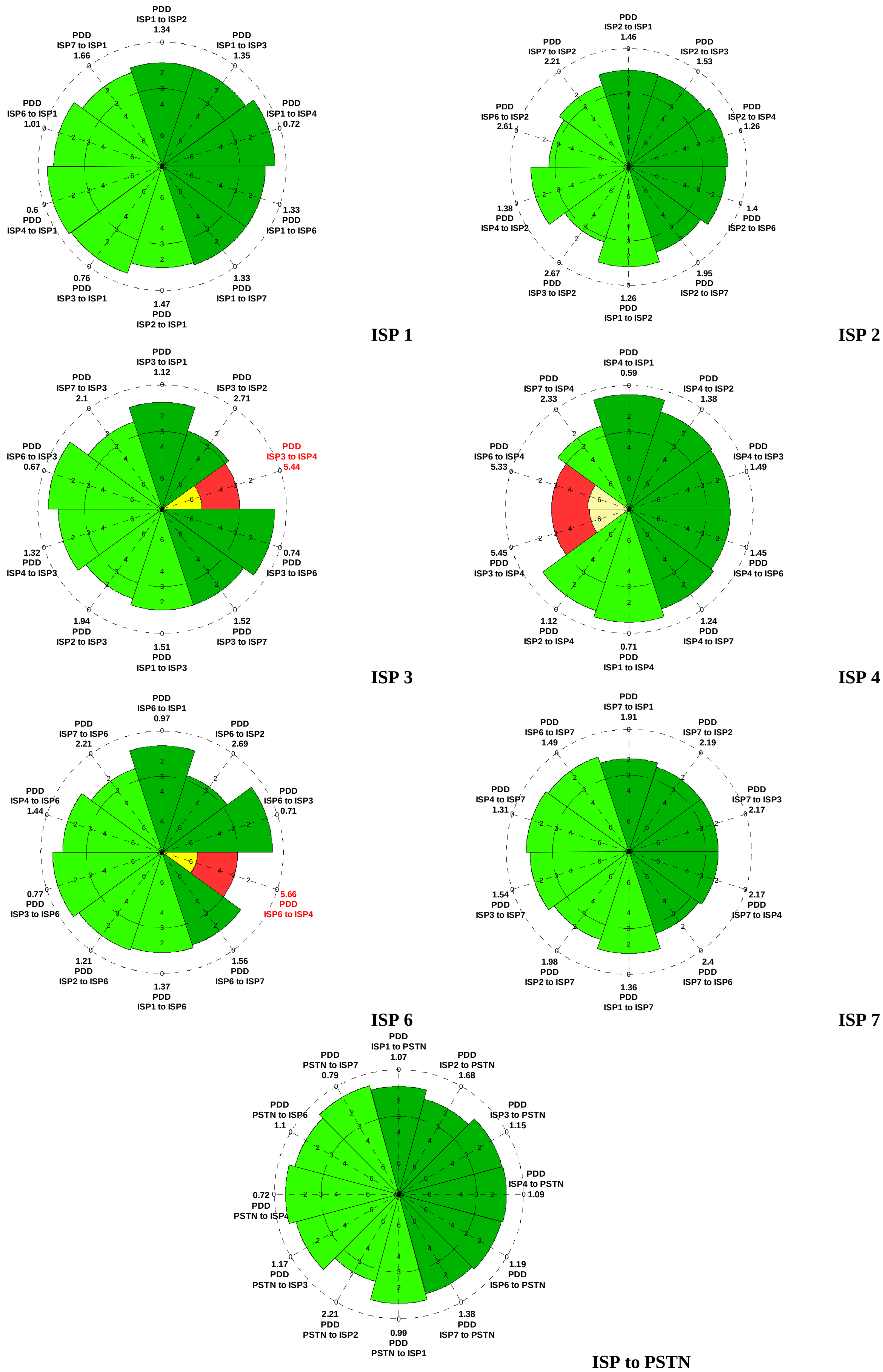


Figure C.3: Pie diagrams presenting PDD values for different call configurations

C.6 Listening speech quality

Results are presented in figure C.4.

For **IP to PSTN** and **PSTN to IP** calls, speech quality is characterized by an average MOS-LQON score of 4,4.

For **IP to IP inter-operator** calls, speech quality is characterized by MOS-LQON scores between 4,1 and 4,4.

These performances are thus in accordance with the quality level expected with codec G.711. The results also show that the transmission does not significantly degrade the quality of speech signal (it is reminded that the tests are done without load linked with other services).

It should be noted that in the particular case of the offer associated to ISP1 we notice an asymmetry between the transmission ways "ISP1 to the other operators" and "other operators to ISP1". Speech listening quality in the transmission way "IPS1 to the other operators" (with regard to the transmission way "other operators to ISP1") is 0,2 MOS lower.

This asymmetry is not observed on the performance of other offers. So globally (except for the offer of the IPS1), there is no noticed degradation on speech quality for the IP-IP calls in interconnection between operators.

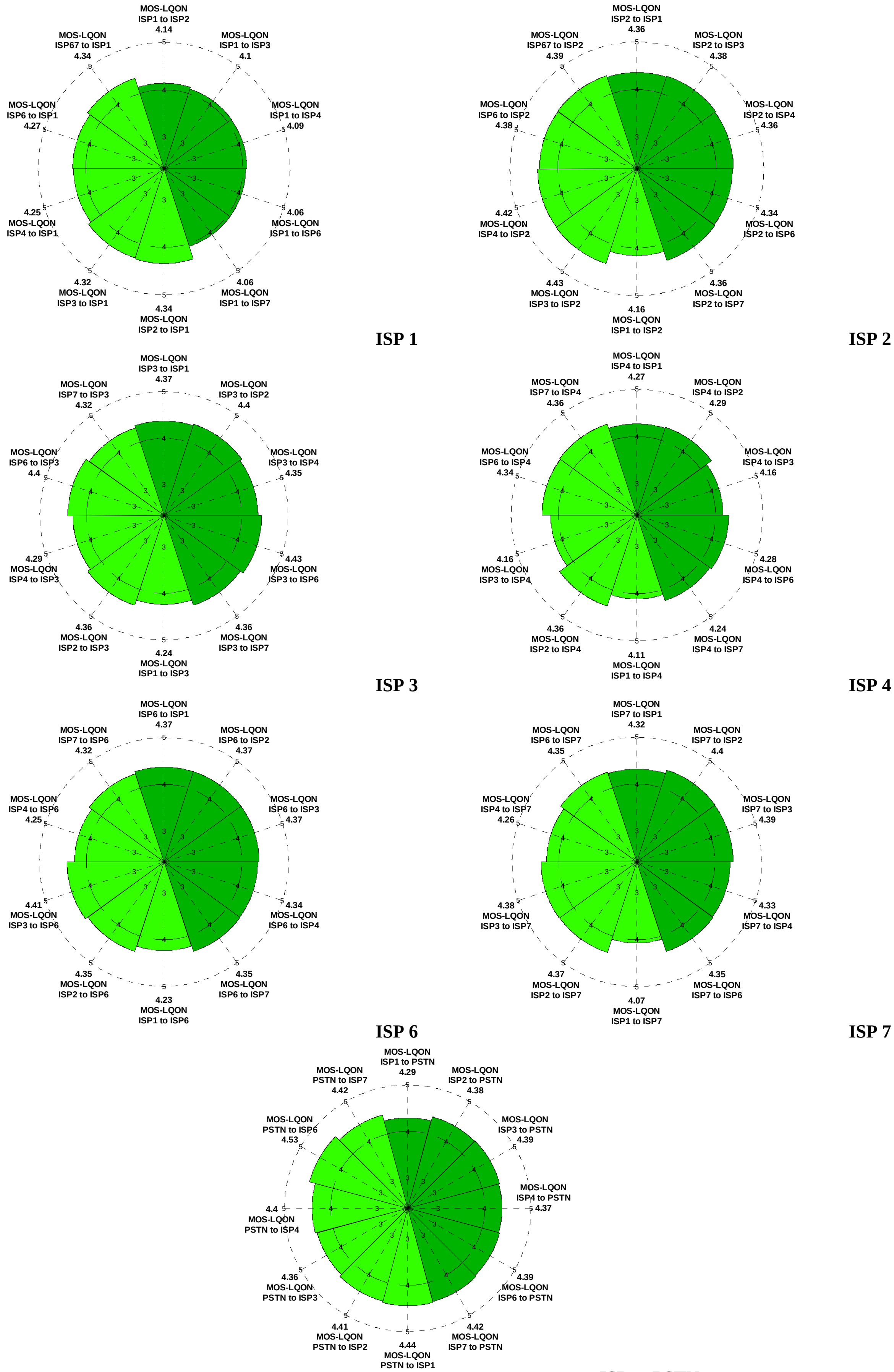


Figure C.4: Pie diagrams presenting MOS-LQON values for different call configurations

C.7 End to end delay

Results are presented in figure C.5.

For **IP to PSTN and PSTN to IP** calls, end to end delay measurements indicate that this characteristic is in all cases lower than 200 ms for all the configurations (for each offer and each transmission direction). The values of end to end delay are measured between 70 ms and 200 ms (the delay being lower than 150 ms for 6 ISP for PSTN to IP and 1 ISP for IP to PSTN).

Globally, we notice that there is an asymmetry on the delay performance between both transmission ways, IP to PSTN providing higher values than PSTN to IP:

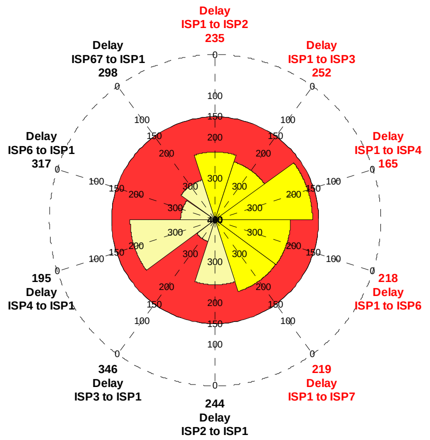
- In the PSTN to IP transmission way, the average value of end to end delay is 125 ms.
- In the IP to PSTN transmission way, the average value of end to end delay is 160 ms.

It can be seen that a transmission delay lower than 150 ms may be reached for IP to PSTN transmission way.

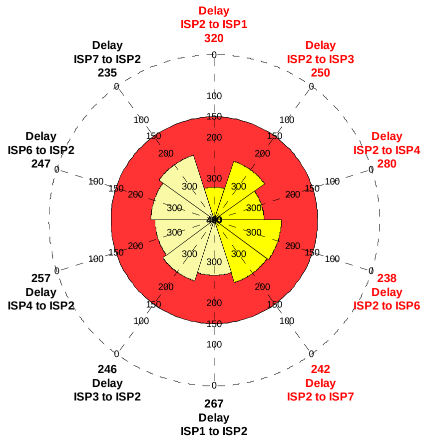
For **IP to IP** communications (inter-operators calls), the values of transmission delay are measured between 70 ms and 360 ms. Depending on the offers, the average end to end delay increases between 70 ms and 115 ms for IP to IP compared to IP/PSTN.and this performance degradation is measured in both transmission directions.

It should be noted that for the call configurations of IPS3 to ISP6 and IPS6 toISP3, the end to end delay is about 70 ms in both transmission way. These 2 VoIP offers have the same performance on calls with the PSTN, but only on PSTN to IP transmission way.

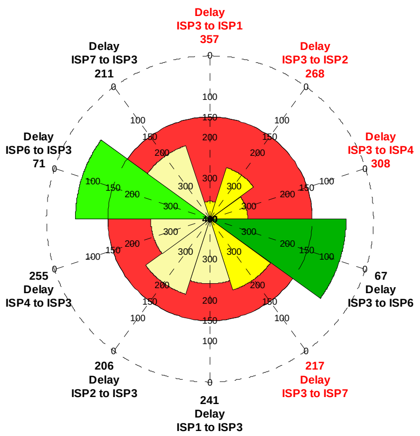
Some delays (more than 300 ms) are annoying for the interactivity of the conversation.



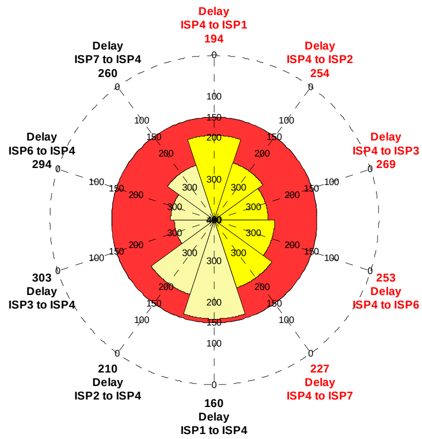
ISP 1



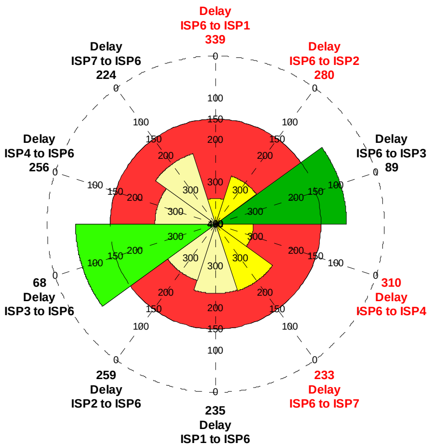
ISP 2



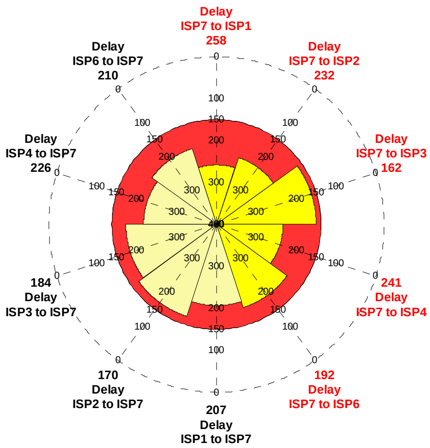
ISP 3



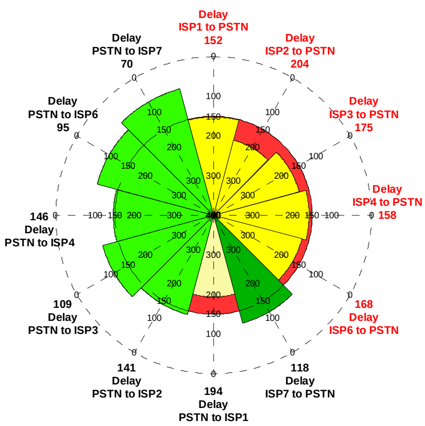
ISP 4



ISP 6



ISP 7



ISP to PSTN

Figure C.5: Pie diagrams presenting end to end Delay values for different call configurations

C.8 Conclusion

A specific test campaign has been performed on a technological platform (where the different Triple Play offers are installed) to obtain an overview of VoIP performance in the context of IP network interconnection.

The measurement results show that in comparison of IP to PSTN calls, IP to IP calls between different ISP have significantly different performances.

Concerning the PDD, there is no variation of the performances except for 3 offers for which we noted a significant increase of the establishment delay for some configurations. In these cases call attempts require more than 5 seconds before obtain the ring back tone after dialling.

Concerning speech quality, the performance is globally the same for IP to/from PSTN and IP to IP calls. There is no significant degradation of speech quality in interconnection configuration (at least without loading with other services).

Concerning the end to end delay, we noted an increase of this between 70 ms and 115 ms for IP to IP compared to IP/PSTN. Some delays (more than 300 ms) are annoying for the interactivity of the conversation. But we observe that for some offers transmission delays are lower than 100 ms. As high performance for delay is possible, improvement are to be envisaged for certain offers. As it can be seen for ISP3 and IP6 offers, shorter delays can be obtained by interconnecting them at the IP level (in this cases media gateways are not involved). If different ISPs are interconnected with TDM the end to end delay will always be significantly higher due to additional packetization and dejittering delay.

In the framework of this campaign, 3 indicators have been assessed, Post dialling delay and speech quality are acceptable in most of the configurations, but the transmission delay is too high in most of the cases of IP-IP interconnexions between operators . However, as some lower end to end delay values have been reached for some interconnection situations, these lower values should be the objective to be reached by all the operators.

History

Document history		
V1.1.1	March 2010	Publication
V1.1.2	November 2010	Publication