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Technical Report

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

Reference

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Contents

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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 presents the results obtained on 2 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.

2 References

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2.1 Normative references

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Not applicable.

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

[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 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
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 customer. 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 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 two technological watch 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 part 1 of this multi-part document [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.

5 Platform presentation

The technological watch 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. If we position these 2 measurement points in the area categories defined in EG 202 765-2 [i.1], platform N°1 is a measurement point in category N°2 and platform N°2 is a measurement point in category N°1.

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.
- Access to services is obtained by 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 VoIP services, codec G.711 is implemented on each offer.
- Signalization protocol is not identical on all the offers: depending of ISP, H.323, SIP or MGCP are implemented.

The platform N°2 is characterized by:

- Implementation of 6 offers concerning 6 different ISP.
- Each offer proposes 2 services: Internet access and VoIP.
- Access to services is obtained by an Home GateWay.
- For all offers, only ADSL technology is deployed to access to the network.
- Distance between HGW and first digital equipment is about 2 000 meters (Length of the ADSL line).
- A PSTN access line is available for the speech quality analyses.
- For VoIP services, 2 codecs are implemented: G.711 and G.726 32 kbps.
- Signalization protocol is not identical on all the offers: depending of ISP, H.323, SIP or MGCP are implemented.

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

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

6 Presentation of test conditions

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

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

6.1 Indicator description

The determined 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	Note 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 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 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.

Test calls are always established from IP to PSTN except when PDD indicator is determined. In this case test calls are performed in both directions (IP to PSTN and PSTN to IP).

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.

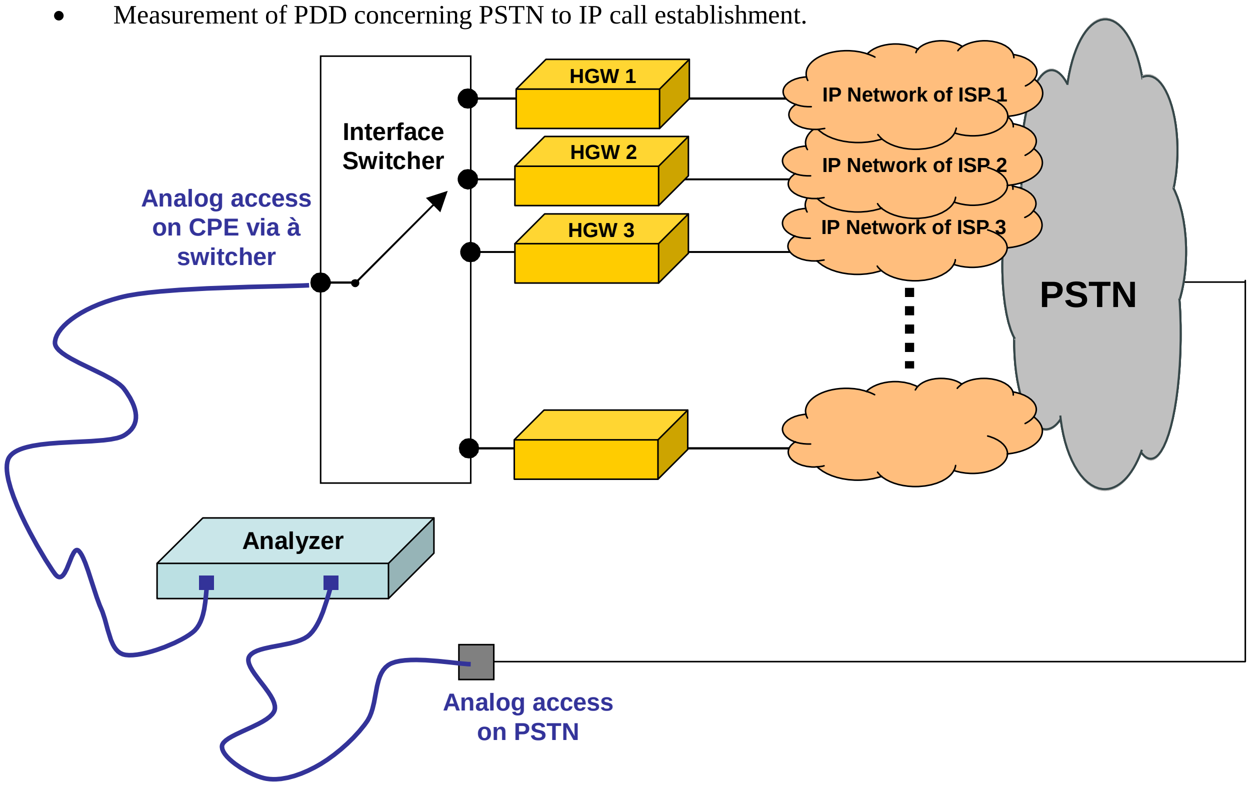


Figure 1: Overview diagram of the measurement chain deployed on both platforms of follow-up

Analyse 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 others 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.

6.3 Pie diagram presentation

An interesting presentation of the results is used within the framework of this activity; it is the Pie diagram (recommendation 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, 12 indicators are presented on a Pie Diagram (ITU-T recommendation P.505 [i.5]). 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.

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

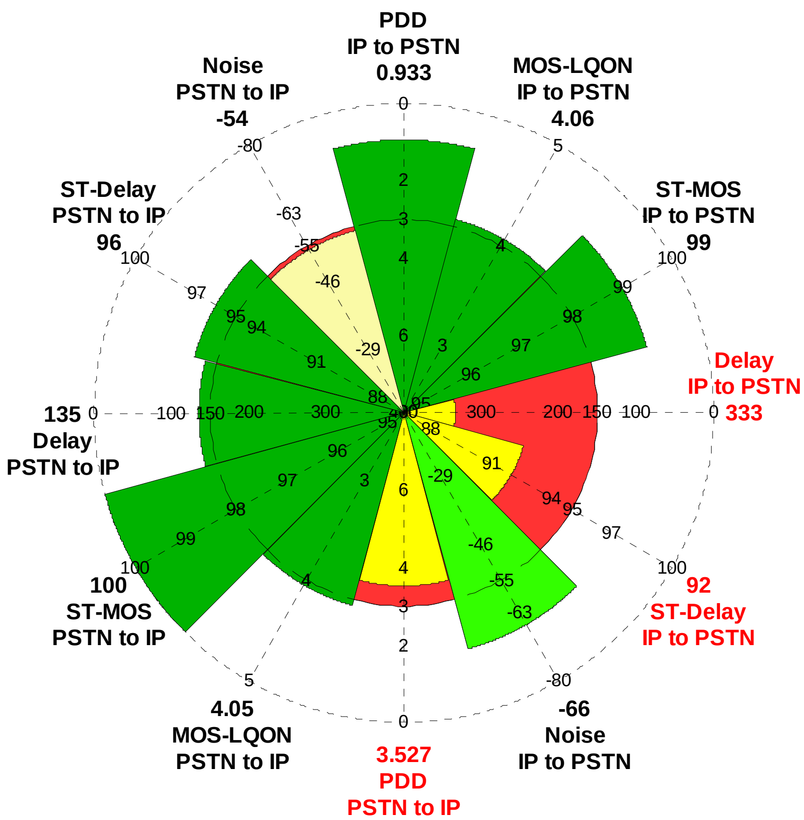
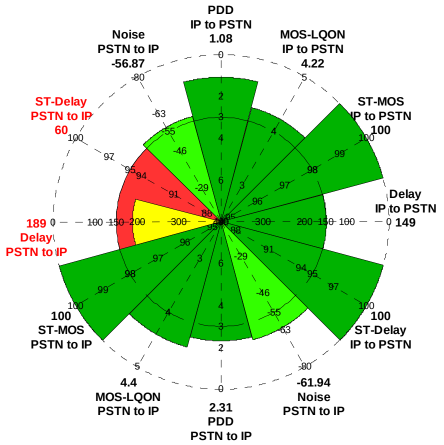


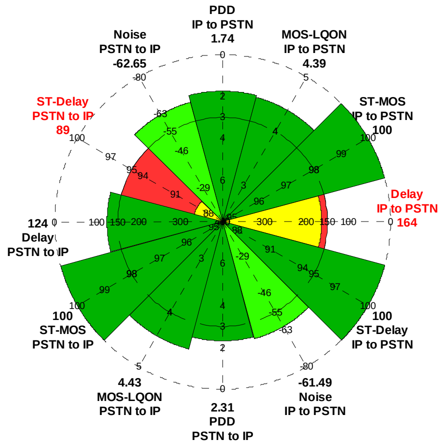
Figure 2: Example of Pie diagram with indicators determined within the framework of the two platforms of the VoIP offers

7 Overview of results obtained in October 2009

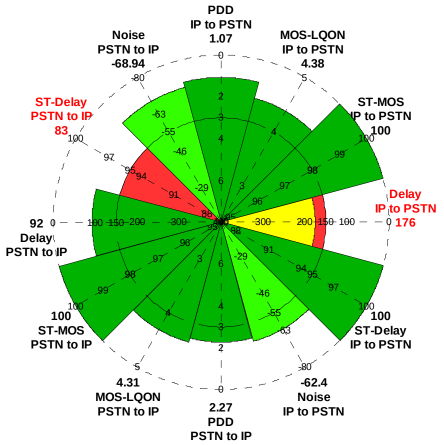
Pie diagrams obtained for October 2009 on Platform N°1 are presented here.



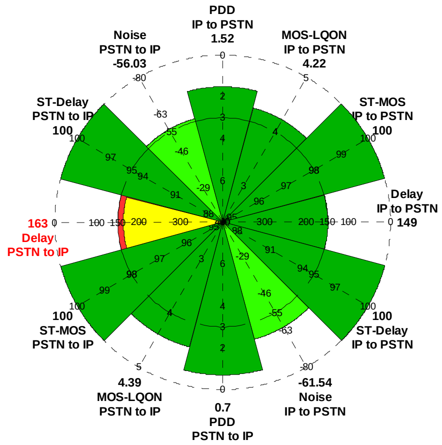
ISP 1



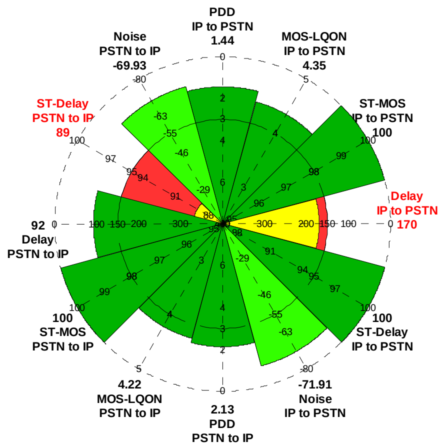
ISP 2



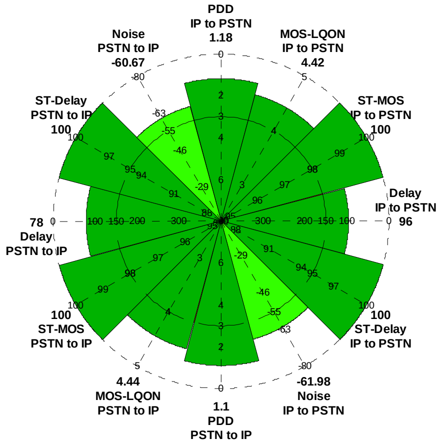
ISP 3



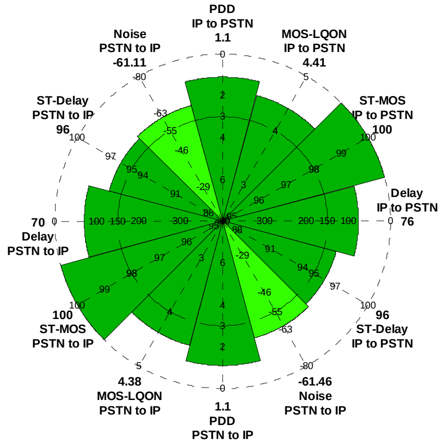
ISP 4



ISP 6

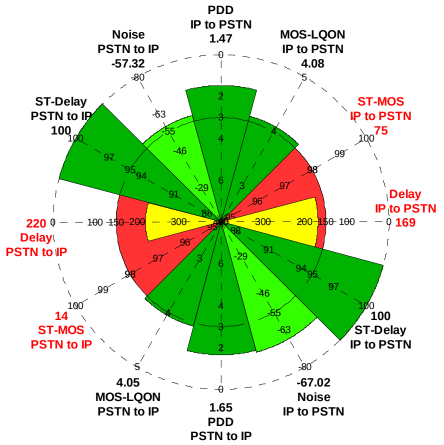


ISP 7

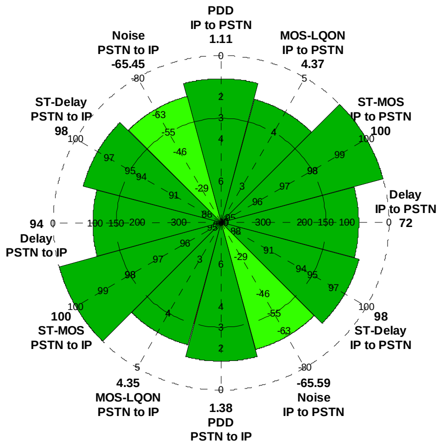


ISP 8

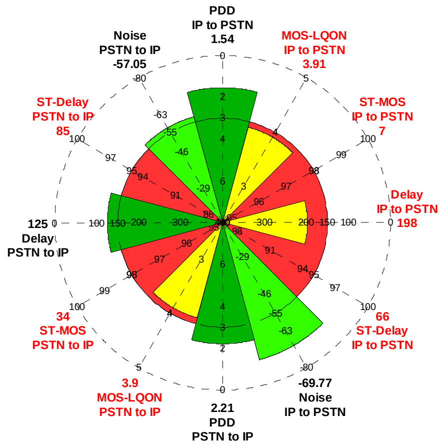
Pie diagrams obtained for October 2009 on Platform N°2 are presented here.



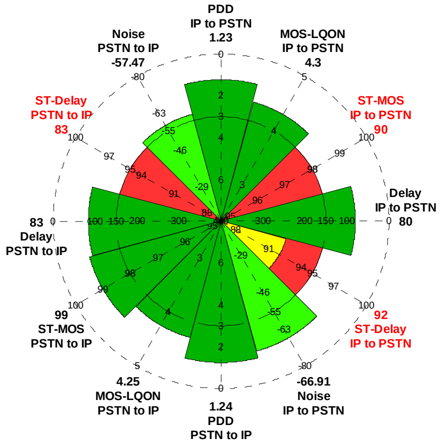
ISP 1



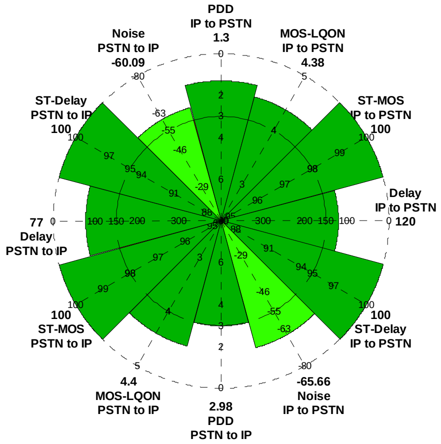
ISP 3



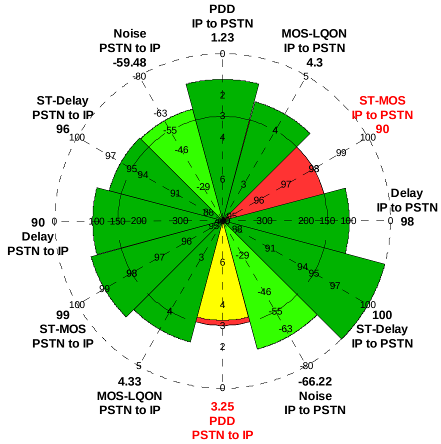
ISP 4



ISP 5



ISP 7



ISP 8

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.

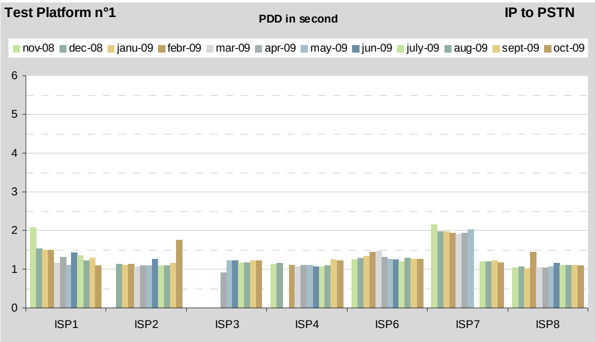
If we compare the results obtained on the two platforms, we notice that the performances are slightly superior on the platform N°1. We also note that the offer ISP7 presents very correct and very similar performances on both platforms. On the other hand we also note that the offer ISP4 presents different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

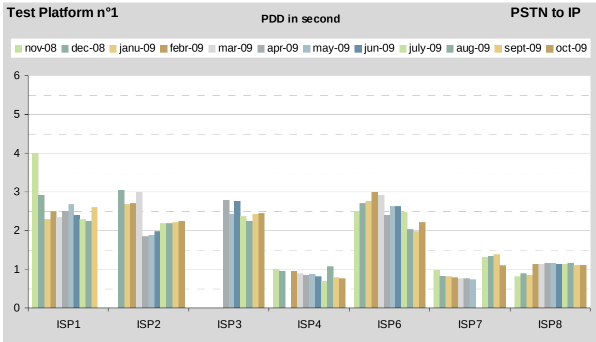
8 Overview of results obtained during one year

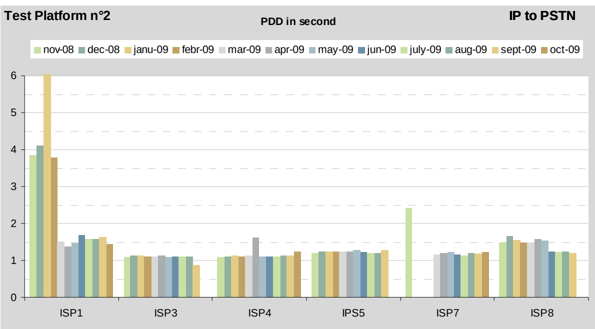
This clause presents a synthesis of the results obtained over one year, from November 2008 till October, 2009. The results refer to both platforms and the performances are presented indicator by indicator.

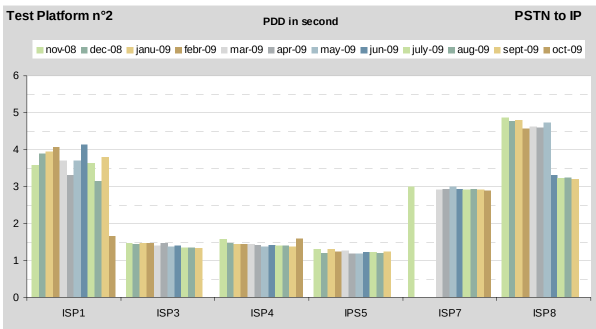
For each metric, the synthesis is presented by 8 graphs where are represented both platforms and both transmission directions (or call attempt direction for PDD).

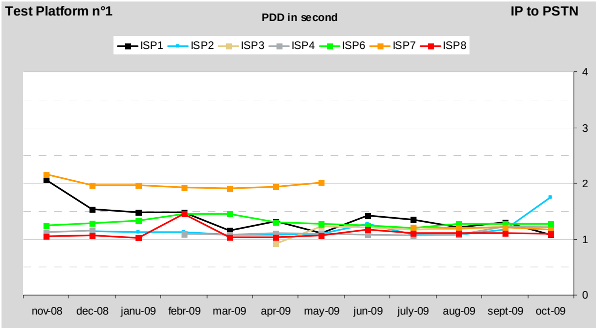
8.1 Post Dialling Delay

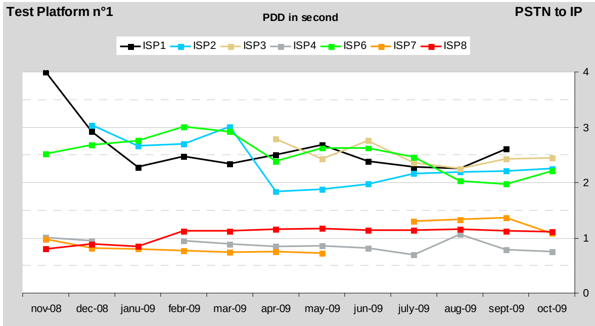


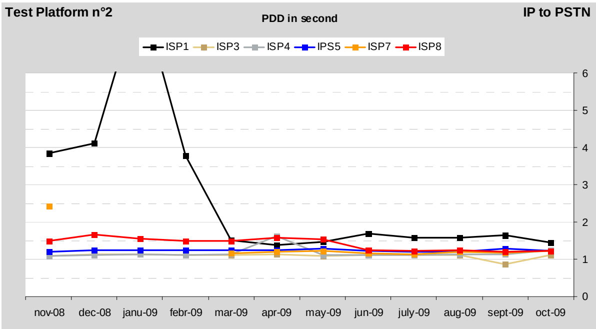


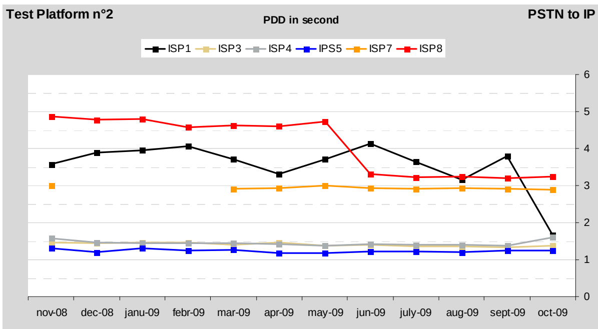












Concerning the call establishment performances, a subjective study has highlighted that the users feel annoyance when the PDD exceeds 6 or 7 seconds.

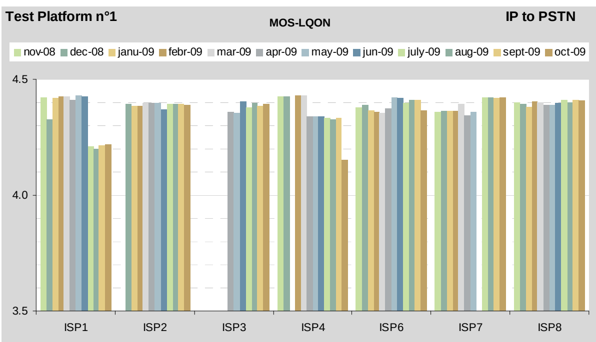
Except for the results obtained for the ISP1 in January 2009, in these case studies the performances are lower than 5 seconds. Globally, we notice identical performances on the two platforms with call establishment delays lower in the IP to PSTN call establishment direction than in the PSTN to IP direction.

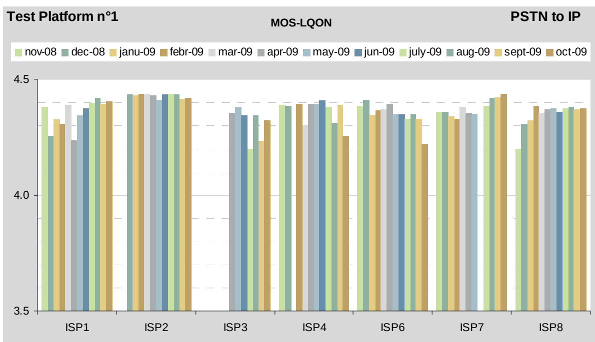
If we examine the results more in detail, we notice that in the IP to PSTN direction there are no significant differences between the offers, as the establishment of the call is achieved within 2 seconds in the vast majority of the testsWhile in PSTN to IP direction, we notice significant differences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

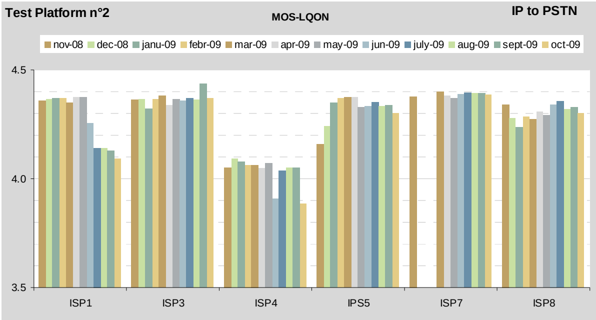
If the performances are globally very similar on the two platforms, we notice for three ISP (ISP 3, ISP 7 and ISP8) differences between the offer installed on the platform N°1 and the offer installed on the platform N°2:

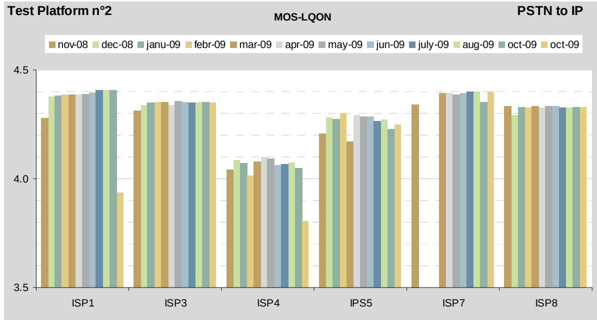
- for the ISP 3, the PDD is higher on platform N°1;
- for ISP7 and 8 the PDD is higher on platform N°2.

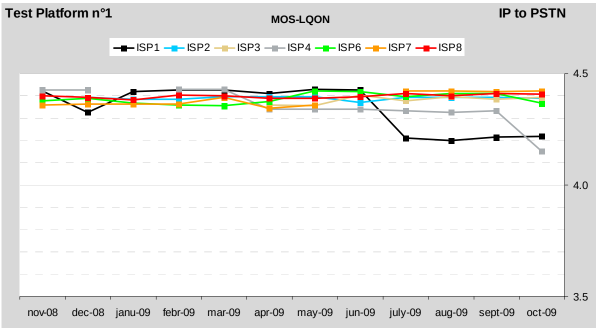
8.2 Listening speech quality

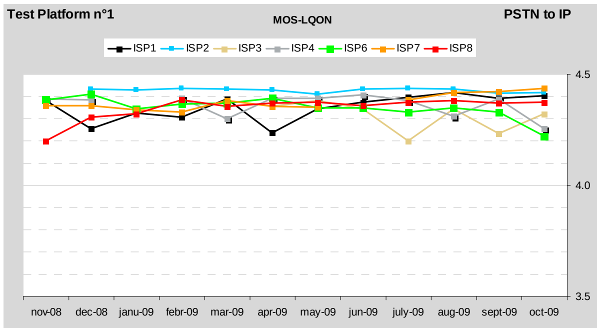


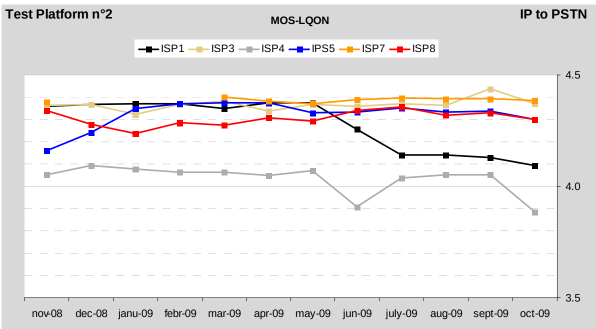


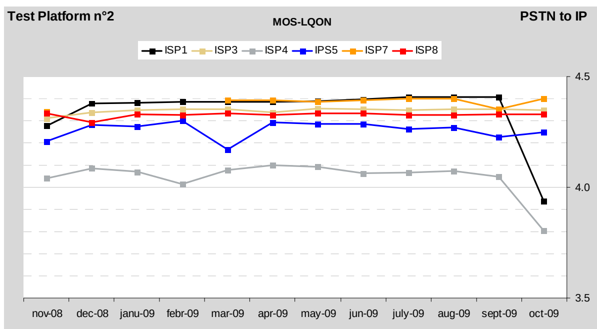












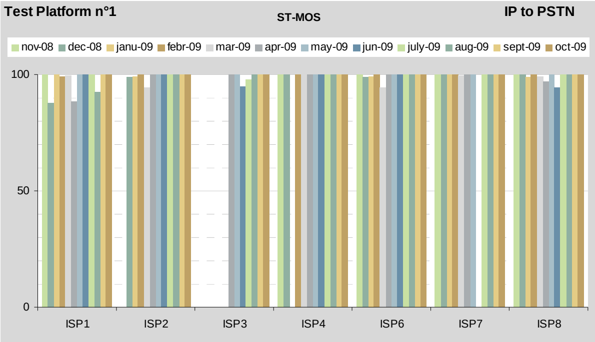
Concerning the speech quality, we do not notice significant difference between the two platforms even if average MOS scores (calculated without the results obtain on the ISP4 offer) is slightly superior on the platform N°1. In the same way, we notice no difference between the performances according to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

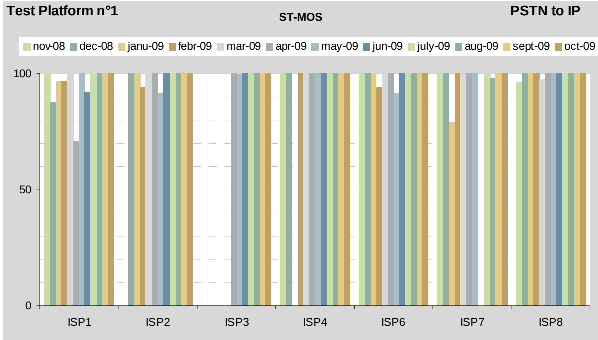
Globally with MOS scores higher than 4,2, we can confidently state that speech quality is good.

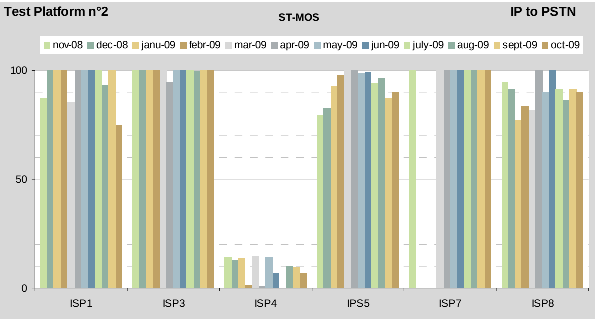
The most noticeable point is the difference of performance between platform N°1 and platform N°2 for the ISP4. For the platform N°1, the speech quality is characterized by a MOS score about 4,3 whereas for the platform N°2, the speech quality is characterized by a MOS score 4,0.

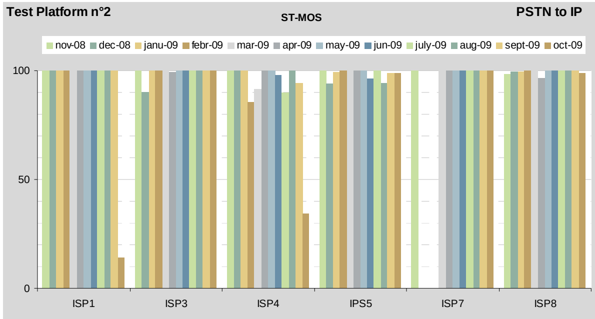
This is due to the different negotiated codecs. On platform N°1 the codec G.711 is used and on platform N°2 the codec G.726 32 kbps is used. This difference (for codec implementation) is imposed by the ISP. It depends on the geographical zone and on the type of ADSL option deployed (option 5 or option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

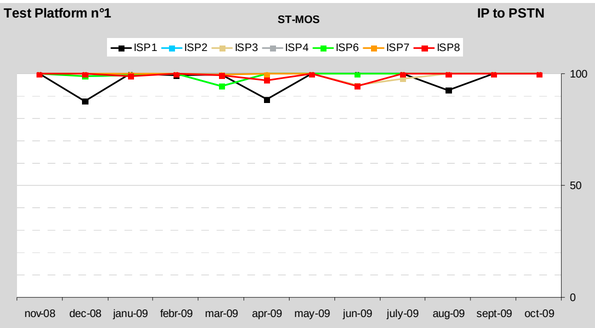
8.3 Listening speech quality stability

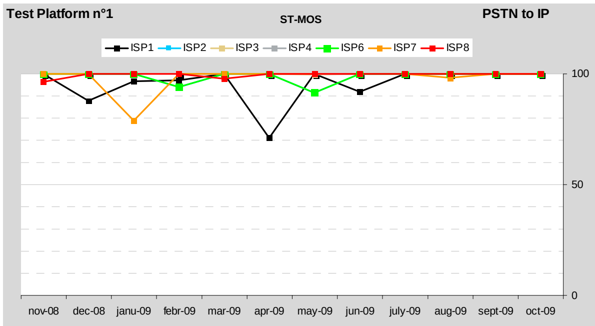


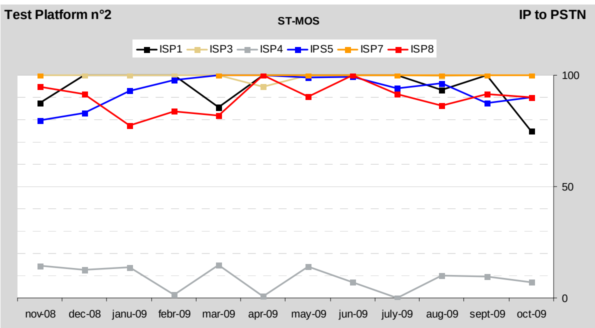


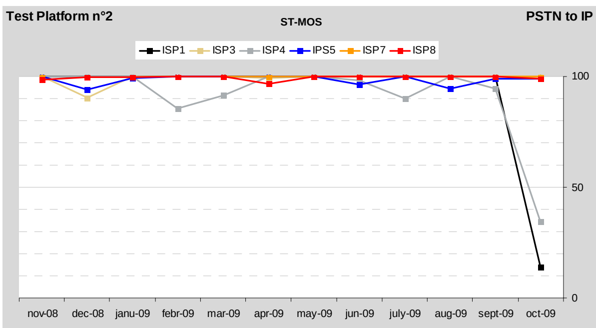










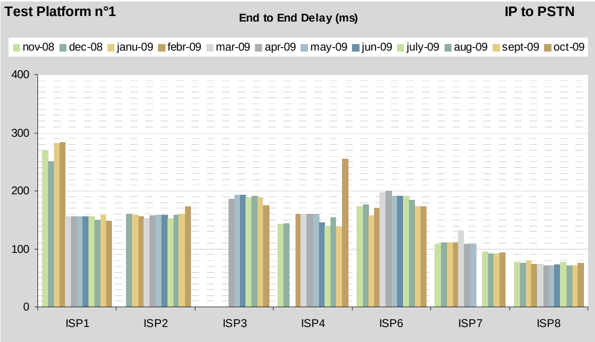


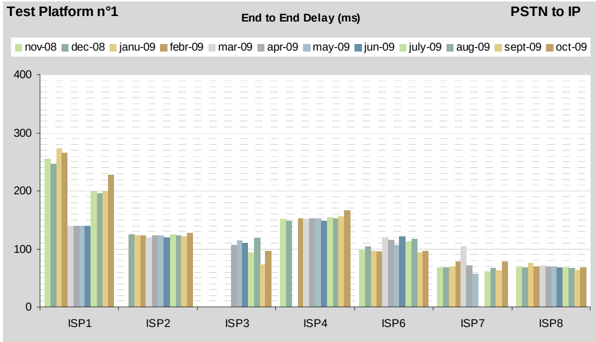
Concerning the speech quality, the stability is not bad. Except for the performances of the offer of ISP4 on platform N°2, in IP to PSTN transmission direction, the stability (according to transmission way) is characterized by mean values 99 % and 98 % on platform N°1 and by the mean values 94 % to 97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

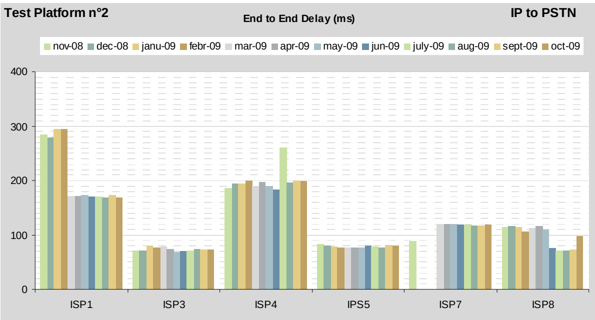
We can notice a very weak stability in the IP to PSTN transmission direction for the offer of the ISP4 deployed on platform N°2. This point is remarkable because the stability for this offer is good in the other transmission way (PSTN to IP). Besides, the offer of ISP4 deployed on platform N°1 does not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

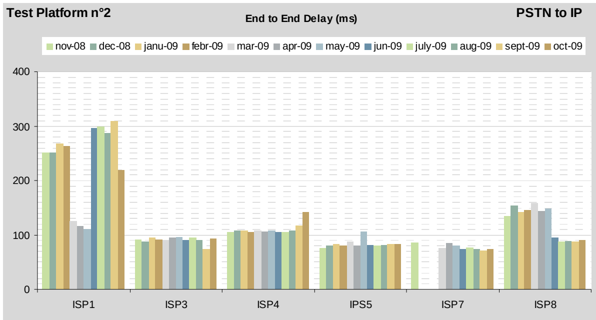
An other point worth noticing on platform N°2: in October 2009, the stability in the PSTN to IP transmission direction is highly degraded for the offers of ISP1 and ISP4.

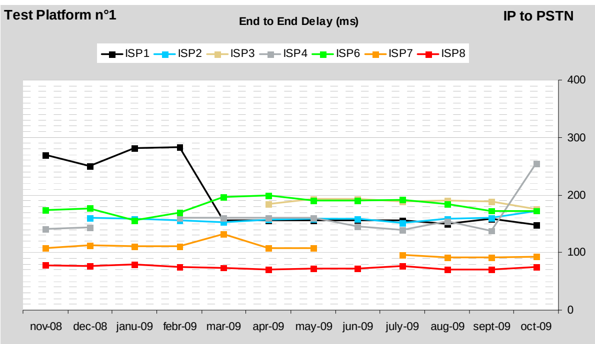
8.4 End to end delay

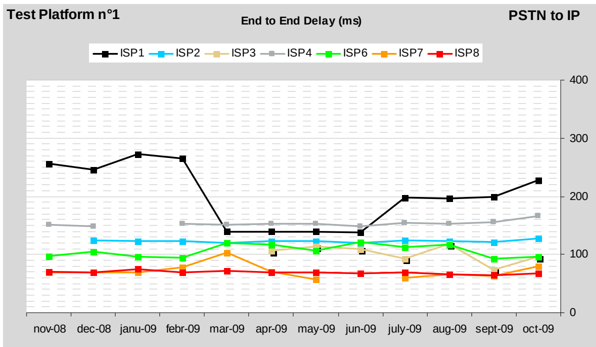


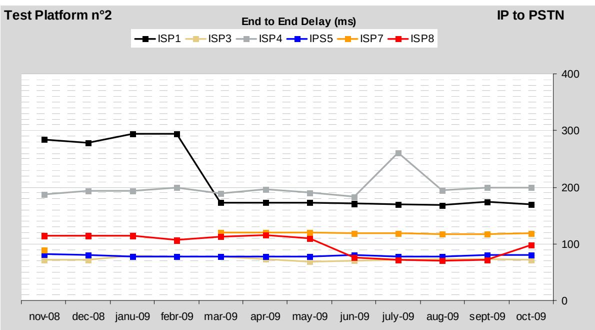


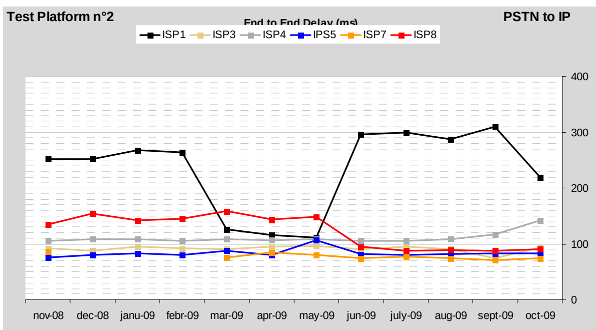










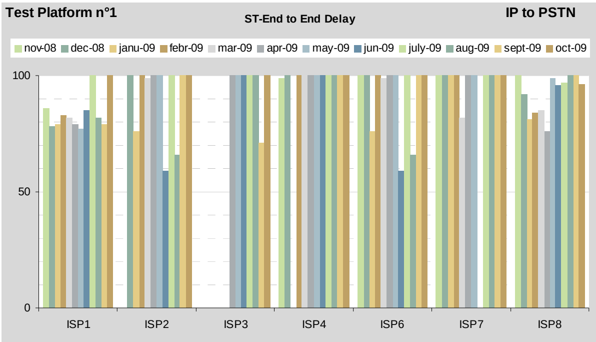


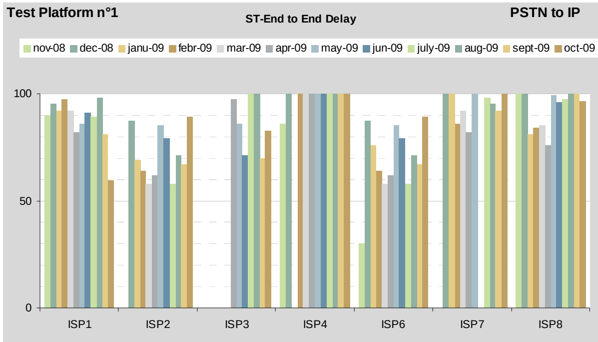
Concerning the end to end delay, we notice that in the PSTN to IP transmission direction, the delay is globally lower or close to 150 ms (on platforms N°1 and N°2).

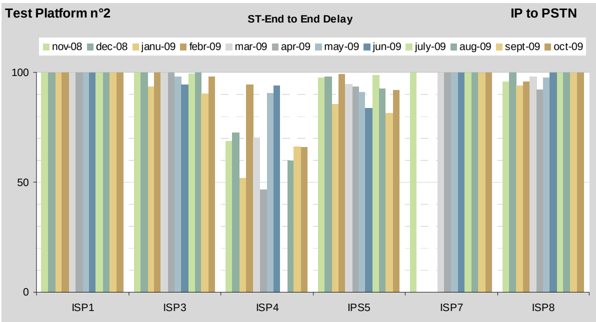
In the IP to PSTN transmission direction, the delay is globally higher than in PSTN to IP direction but it remains lower than 200 ms.

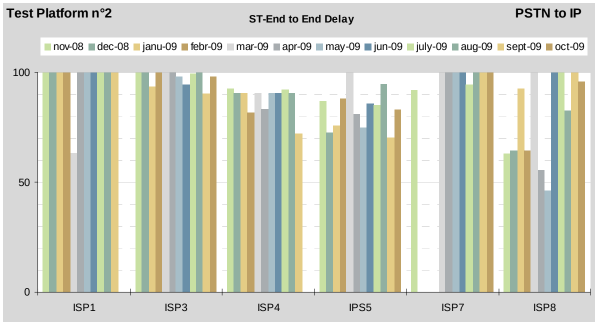
The delay for the ISP1 offer can reach 300 ms in certain cases. The last four figures clearly show that the delay varies strongly from one month to another.

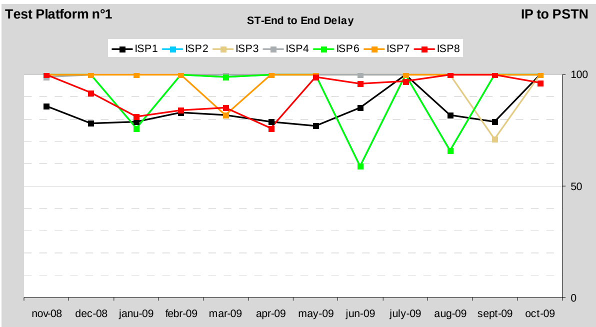
8.5 End to end delay variation

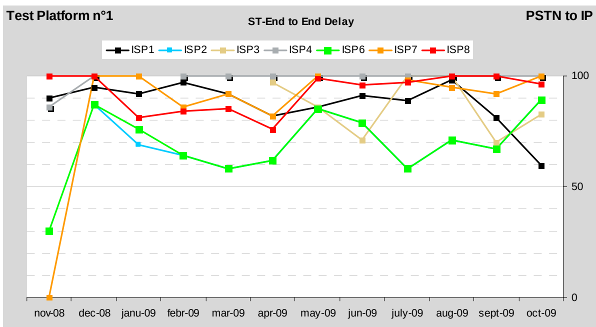


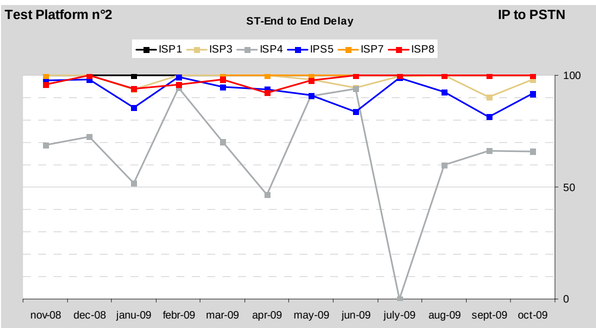


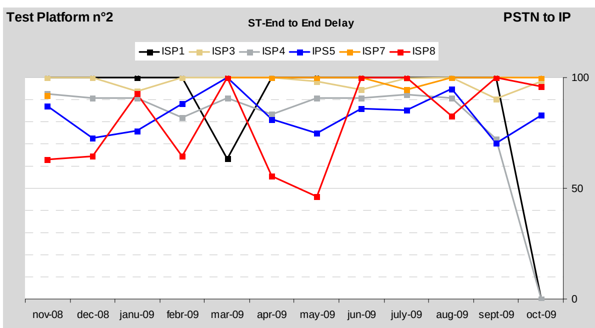






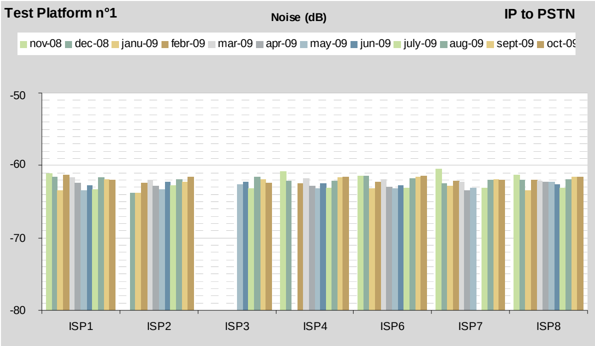


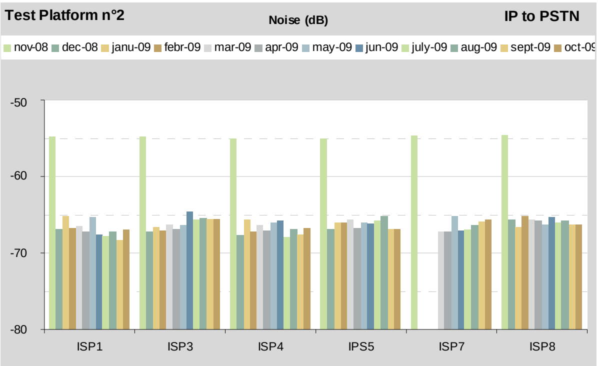


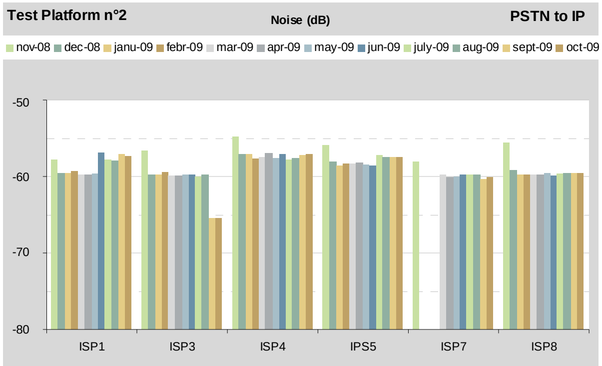


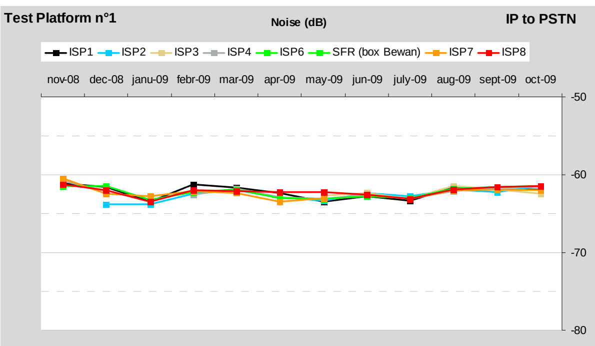
Concerning the variation of delay, we notice a relative consistency of the performances between all the offers. These is no offer with a very weak stabilityand there is no offer with a perfect stability (indicator value equal to 100 %) in both transmission directions, throughout the whole year. On the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

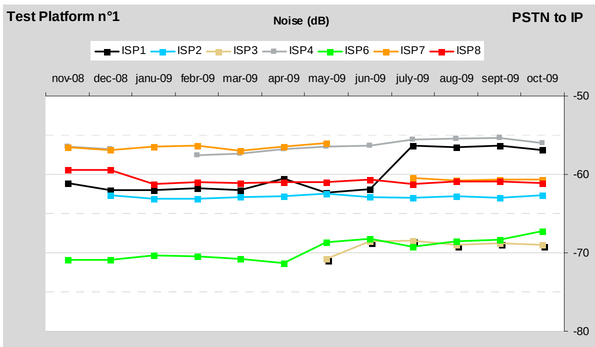
8.6 Noise level at reception

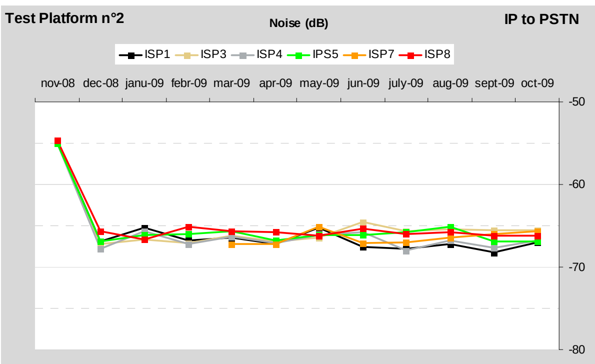


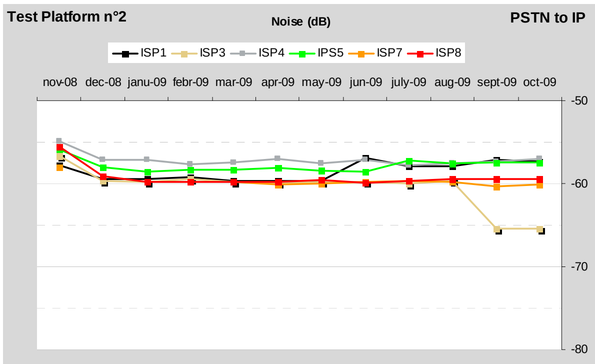












On platform N°1, the noise level at the reception is relatively high with a mean level of -62 dBm on the IP and PSTN sides. On the PSTN side, the noise level at the reception is very similar for all offers. This noise level corresponds to the level of noise on the PSTN line. On the IP side, we notice 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.

9 Conclusion

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.

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.

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.

History

Document history		
V1.1.1	March 2010	Publication