Analysis of voice quality over infrastructured WLAN with Distribution System

Edgar da Silva Arrais*, Claudio de Castro Monteiro*

* *Federal Institute of Education, Science and Technology, AE 310 South Avenue 05 LO, South Master Plan, Palmas, Tocantins - Brazil.*

edgararrais@gmail.com, ccm@ifto.edu.br

Abstract— **This assignment presents a study about voice transmission in WLAN networks based on IP protocol (VoIP). The assignment presents an statistical analysis of the QOS (Quality of Service) metrics of these networks for VoIP transmission. We considered to make calls on a scenario where WDS (Wireless Distribution System) exists. The objetive is to analyse the impact that the WDS may bring on in voice transmission. The results found are compared between the two environments and presented in tabular and graph form.**

Index Terms—**WLAN, VoIP, WDS**

I. Introduction

The traffic generated by real time multimedia applications in mobile devices grows exponencially day-by-day on the internet. It is common to find such devices connected to the internet through WLAN, 3G, WiMAX networks among other wireless access technologies. With mobile users increasingly thristy for content, connection speed and largely for the network availability anywhere using the available means on their devices, we observe one of the greatest challenges faced by the network managers, the service quality in WLAN networks.

According to jiwire [1] indicators the number of Wi-Fi hot-spots tripled around the world in four years. This raise is justified due to the popularization of the mobile devices like smartphones and tablets. The most part of these users, which corresponds to 55% of the mobile users surveyed seeks the wireless access as a way of fast connections with a minor cost per data traffic. The real time multimedia applications detached in [1] are the on demand video transmissions, and voice over IP (VoIP).

Attend this raising demand of applications is one of the challenges faced by the managers, preserving the minimum quality of the services provided by these networks in any point of the network. The number of access points (AP) controlled can raise significantly according to the coverage area and the number of users to be integrated to these networks. One of the commonly used technologies to realize the WLAN networks distribution is the WDS (Wireless Distribution System). WDS allows the interconnection among Aps without cables, through the own AP's wireless interface. Such characteristic is extremely interesting when it is intended to expand the network to regions where the cabling is unable to reach. With WDS function active the AP starts to realize the function of repeater of the clients signal to the central AP connected to the main network (DS). Within this the WLAN signal coverage is expanded according to the installation of other Aps with WDS. In contrast one disadvantage is that the reduced bandwidth due to the sharing with the connected users and a little latency inserted on the network by WDS.

Real-time applications like VoIP have characteristics of delay intolerance, packet loss, lack of bandwidth, voice compression, therefore it is necessary the use of mechanisms to assure the Quality of Service (QoS). Many other aggravating already known can be detached when the use of VoIP is related with the transmission means, mainly when the used mean is the wireless pattern.

This assignment has as an objective analyse the impact of VoIP implementation in WLAN networks, with the use of WDS through call analysis. To reach the research's objective, It was necessary to create two WLAN environments. In both environments we executed VoIP calls and each collect was processed with the use of a packet analyser. Using statistic tools it was possible to obtain the QOS's metrics involved. The intention is to appreciate the impact caused by the use of VoIP applications in environments with WDS.

To evaluate such impacts we realized tests with the purpose of obtaining the QOS's metrics: delay, outflow, jitter and loss. From the obtained results, we expect to conclude that such metrics have different values in environments using WDS, what may compromise the quality of service. This assignment is organized as follows. In section 2 it is described the bibliographic study. In section 3 it is described the study's

Manuscript received July 30, 2012. This work was supported in part by the IFTO - Federal Institute of Education, Science and Technology of Tocantins.

E. S. A. is with the IFTO - Federal Institute of Education, Science and Technology of Tocantins, BR Brazil (+55 (63)-3229-2200; e-mail: edgararrais@gmail.com).

C. C. M. is with the IFTO - Federal Institute of Education, Science and Technology of Tocantins, BR Brazil (+55 (63)-3229-2200; e-mail: ccm@ifto.edu.br).

proposal and in section 4 it is described the used tools, the process of the environments's implementation, methodology and data analysis. In section 5 are exposed the results. In section 6 are presented the research's future assignments.

II. Related Works

Local wireless networks (WLAN) are a viable and economic alternative for the conventional networks based in cabling. The great number of mobile devices with wireless interface offered in the market grows each day, as well as its search due mainly to the drop of the prices of electronic components. Today it is possible to find many devices with wireless communication interface due to its setup facility, low costs of implementation and the mobility that this interface offers to the device.

WLANs provide the same basic functionalities of cabling networks (LAN), but combining mobility with access speed that results in the creation of new habits, relations and ways of working. A good example is on the major use of services that demand real time requirements like voice applications over IP and video on mobile devices. The use of multimedia services (voice and video) may demand resources reservations and availability to have a good functioning and provide a good use experience to the user. Initially WLAN networks, based on the 802.11 pattern were not projected for this kind of traffic. For the multimedia services to function properly, the associated use of the packets priorization techniques may be necessary.

The pattern WLAN 802.11b and 802.11g can support theoretically 200 and 900 VoIP sessions respectively, but due to the inclusion of a great number of headers this number is reduced drastically to only a few sessions. The headers added to the payload consist in the real time protocols (RTP), UDP, IP and headers of the MAC layer 802.11. According to [2], the 802.11g pattern presents the best performance for the VoIP traffic in relation to the 802.11a and 802.11b patterns, in both types of CBR (Constant Bit Rate) and VBR (Variable Bit Rate) traffic. A comparative study was realized using the patterns 802.11a and 802.11g and 802.11b where it was illustrated the maximum number of sessions and broadcasting rates on the Nominal and Multiplex-Multicast (M-M), for both types of traffic CBR and VBR.

As an alternative to raise the number of supported sessions in WLAN networks, it was proposed the use of the M-M regime to send packets where the data load reduction was provided for the Multicast packet sending regime. In this mode the headers UDP, IP and RTP are compressed and rewritten in a mini-header and after this combined in a single packet Multiplex-Multicast, where right after it is sent through the WLAN to the IP Multicast address. Among the three patterns analysed in [2] the 802.11g had the best performance compared to the others, with VBR traffic and M-M regime, the difference among the others showed to be well significant in the number of VoIP sessions supported and in the bitrate reached. Using VBR traffic the flux rate is variable and exist the packet send economy with periods of silence, what ensure a better audio quality transmitted.

Other important factor related in the use of WLAN networks can be described as an interference of many other networks very close, producing the interference among them. When there are many other WLAN networks in a small geographic area the number of supported sessions is reduced even more, reaching values between 1.63 and 10.34 sessions using only the GSM codec 6.10. According to [3] one of the main culprit by the reduction of the VoIP session numbers in areas with multiple WLAN networks is the mutual interference of CSMA (Carrier Sense Multiple Access). Many transmission sources using the same frequency or very close frequencies end up suffering interference into one another. This way the percentage of lost packets can raise what is not interesting to any intolerant to losses application, including VoIP.

To weaken the number of sessions it was proposed in [3] an admission call control based on a graphic model of conflicts, where it was possible to raise in 52% the number of VoIP sessions using the pattern 802.11b and 802.11g, what represented the number of 2.48 and 14.14 sessions. It was also proposed the use of three orthogonal channels to reduce the lateral interference, what raised the supported sessions capacity in the 802.11b and 802.11g patterns to 7.39 and 40,91. With the use of CoTDMA (coarse-grained time-division multiple-access) in a partnership with CSMA was possible to obtain an improvement of 35% in the used patterns, that resulted in 10 sessions to 802.11b and 58 sessions to 802.11g.

Other relevant factor to the VoIP is the voice quality transmission in IP networks evaluation. The evaluations can be done through subjective methods, where people do the listening evaluation, or in an objective form these based on mathematic models. In [4] it is referred the recommendation ITU-T P.800 as one of the methods and procedures used for this kind of evaluation. It consists basically in computing an average opinion of the listener in a scale from 1 to 5. This scale is known as MOS (Mean Opinion Score). The MOS scale is the following: 1 – for a horrible quality, 2 – for bad, 3 – for a reasonable, 4 – referring to a good quality, and 5 – for an excellent quality. In [5] it is presented many objective methods, with emphasis for the E-model presented as an effective method on the voice quality. The E-model is an evaluation model based on a concept where “psychological factors on a subjective scale are auditory”, that is factors gauged from the call quality loss may be computed generating a scalar model named R factor. The R factor scale has a variation from 0 to 100. The R factor [5] is calculated from the equation (1).

$$R = (Ro - Is) - Id - Ie + A \tag{1}$$

Ro is the noise signal relation; according to the recommendation the pattern value is 94.77. **Is** is the factor for the voice quantization, and its pattern value is 1.41. ***Id*** corresponds to the delay degradation. ***Ie*** represents the signal distortion caused by the codecs, loss and buffers. For the codec

G.722 used in this study we assumed the value of the recommendation like being 13. And finally ***A*** corresponds to the quantity of degradation that the user is willing to accept. In this study we assumed the value 10, that in the recommendation refers to the quality in celular networks with mobility. The articles and recommendations studied indicated alternatives to fix or improve the performance of the use of VoIP on WLAN networks. In this assignment, accordingly described previously it will be analysed the use of WDS as a way of WLAN network expansion, thus seeking to prove its performance in relation to the VoIP and the induced impact. All collected samples in this study were made in real environments, without the use of simulators. No QOS policy is implemented in both environments. As a method of the quality call's evaluation in this study will be used the E-model.

III. PROPOSAL

The objective of this study is to analyse from a statistic point of view the variables (random), latency, jitter, packet loss and the throughput measured in a non-intrusive way in WLAN networks making use of a VoIP application. We proposed also to realize an objective measurement according illustrated in picture 1 about the speaking quality, obtaining the prediction of the quality call based on the parameters of delay using the E-Model. In this study the evaluation will be realized through experiments in real environments, the first in a WLAN network without the use of WDS, and second in a WLAN network with WDS at the Federal Institute of Education, Science e Tecnology – IFTO.

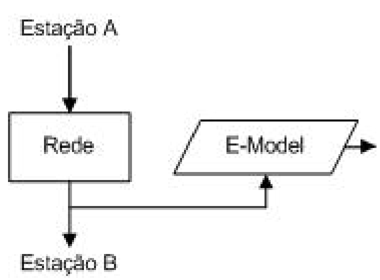

Fig. 1. Non-Intrusive Measurement Objective

The size of the collected sample in this environment will be realized per point, the size of the sample will be applied in both environments for statistic analysis purposes. Addresing the study of random variables, in these environments we adopt an average of the metrics found as a comparative value, because these values can be different according to the network demand, time, background traffic among other variables that may interfere in the result. A pre-sample will be processed in the second environment of this study with the objective to serve as initial parameter for the size of the sample in both environments, this way avoiding waste of resources and operating costs with unnecessary collection. Knowing that the second environment tends to be to most degraded considering that the first does not exist the background traffic, that is, without any network use at the time of the VoIP calls, it will be realized a pre-sample with the objective of knowing the average values and average diversion pattern for each one of the analysed variables. From the calculation of the sample size (2) we seek to know an ideal sample size for both environments with 95% trustyworthy of finding the real average values for quality metrics of service in voice transmissions under IP networks.

With the sample size defined, with all the samples collected, processed and with its respective graphics we have the capacity to analyse the impact caused by the use of WDS in WLAN networks in VoIP transmissions. The metrics found will be analysed, compared and discussed, this way we intend to get to a conclusion of the benefits of the use or not of the WDS as a way of expansion of a WLAN network and mainly to point out the positive and negative points of the use of the technology.

A. Materials

The materials used in this study are the ones related on table I. The access points used were the 3COM 7760[11] model, referred now like WDS1 and WDS2. The operating system used on the stations was Ubuntu 10.10 32 bits [6] installed on the two laptops Asus Eee PC 1015PEM [7] referred now as station A and Station B. The multimedia applicative that allowed the accomplishment of the calls between stations A and B was the Jitsi [8]. The codec used in both environments was the G.722. In the analysis phase, it was used the statistic software R [9] to execute the calculations and the Gnuplot [10] to illustrate the results with graphics.

TABLE I
MATERIALS

	VERSION
Computer	Asus Eee PC 1015 PEM
System Operational	Ubuntu 10.10 Maverick Meerkat
Softphone	Jitsi 1.1
PBX	Askozia Embedded 2.0.4
Codec	G.722
Sniffer	Wireshark 1.8.0
Statistical Software	R-2.15.1
Graphics	Gnuplot 4.6.0
Access Point	3Com 7760 a/b/g PoE

On the PBX two SIP counts were added with identification 101@200.129.176.50 and 102@200.129.176.50 to configure on stations A and B. The packet analyser was installed on station B with the objective of realizing the capture in parallel to the call answering, accordingly illustrated in picture 1. The WLAN networks in both environments do not have any security key set. The distribution of the IP addresses to the stations was realized through DHCP protocol.

B. Environment 1

The first environment proposed was the scenario where the

WDS is not implemented. Picture 2 illustrates the environment created to collect the samples. The environment was configured in a typical form, being this a BSS (basic service set) composed by an access point pattern 802.11g connected directly through the port uplink to the PBX VoIP. The A and B stations were associated to this BSS to make use of the VoIP application.

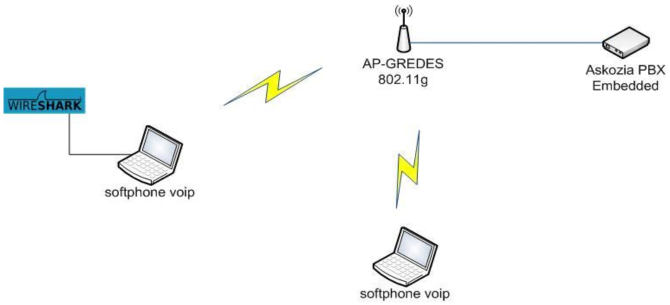

Fig. 2. Environment 1 without a WDS.

In this environment were realized calls between the two stations and captured the exchanged packets during the collected sample. The packets were saved in an arquive of the own wireshark, where each file corresponds to a realized call. After collecting the files were exported to the CSV format (Comma-separated values). Right after the files were imported on the R software to realize the statistical analisys. The calculations that were realized are based in the QOS's metrics: latency, jitter, packet loss and outflow.

The stations remained without mobility in the moment of the realization of the calls. The distance between A station and the access point was about 4 meters and between B station it was 3 meters.

C. *Environment 2*

The second environment proposed was based in a WLAN network with WDS existing at IFTO. This environment is composed by two access points (AP) forming an ESS (Extended Service Set) connected through its wireless interfaces. Picture 3 illustrates this environment, as well as two stations that were configured to this network to make calls.

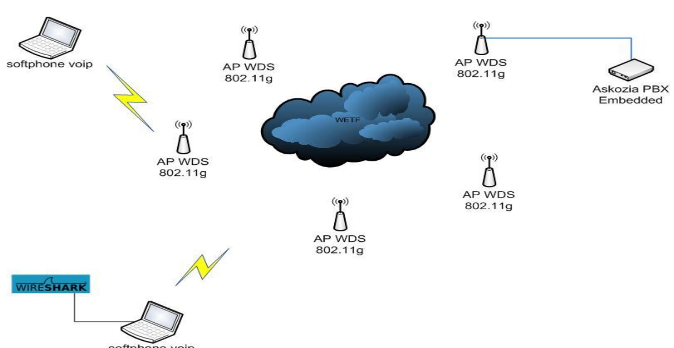

Fig. 3. Environment 2 with WDS.

The location of the stations in relation to the access points is defined on table II. The collection points were named with P1 for the collection point 1 and so on until the collection point P5. Both access points were named in the image as WDS1 and WDS2. All the collections in this environment were realized in a space outdoors. On table II it is illustrated the distance in meters in relation to the access point at the moment of collection and the number of necessary jumps until DS (distributed system).

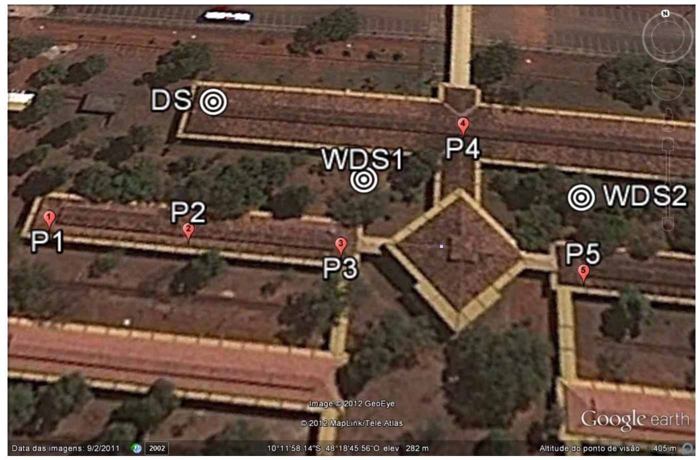

Fig. 4. Collection Points.

TABLE II
DISTANCE ACCESS POINTS

Checkpoint	AP	Distance	Hops
P1	DS	46,44	0
P2	WDS1	41,96	1
P3	WDS1	14,76	1
P4	WDS1	31,07	1
P5	WDS2	16,30	2

IV. Methodology

The study was realized in three steps: data collection, processing and analysis of the results.

A. *Collection*

The collection was realized in both environments through the accomplishment of the VoIP calls between two stations without mobility to fixed distances in pre-fixed points at IFTO. The points were locations where there is concentration of users from the network analysed, being these usually near to the classrooms or areas of free access. The collection began executing a pre-sample calls in the environment. It was adopted a fixed number of twenty calls, with the objective of indicating the number of necessary calls to be realized to a trustiful degree of 95% (3). Each call made had the average duration of one minute between the stations. Below follows the formula for the sample calculation (2) used.

$$n = \left(\frac{Z\frac{\alpha}{2} \times \sigma}{E} \right)^2 \tag{2}$$

Where, ***n*** corresponds to the number of individuals on the sample, $Z\frac{\alpha}{2}$ is the critical value that corresponds to the wished trust degree. σ is the swerve from the metric's population pattern studied and ***E*** corresponds to the maximum estimated error. The pre-sample realized of 20 calls was realized to define the size of initial sample to be realized in both environments in this study. With the pattern swerve of the calls and the maximum error admitted it was possible to find a sample size of 60 calls, according table II, for the second environment.

TABLE III
PRE-SAMPLE

	MEAN	STANDARD DEVIATION	ERROR	SAMPLE SIZE
Delay	36.53	9.04	2.3	60
Jitter	0.006	0.0008	0.06	42
Throughput	85.62	0.01	3	53

The metrics delay resulted in a sample size of 60 calls, being this the size adopted, inasmuch it comprehends the sample generated by the other metrics. The packet loss wan not analysed inasmuch it generated a null loss, taking its invalid sample for the calculation of the sample size.

B. Analysis

The data analysis was realized with the use of the statistic software R and the shell script language. This processing had as an objective to provide the average central tendency and the pattern dispersion measurement of each service quality metric broached in this study. In the analysis was also realized the objective measuring of the call quality with the found results. With the first environment properly mounted the first collection of 60 calls can be realized. The calls were always made with station A as the origin. In each call, the wireshark software was executed with the purpose of capturing all the received and sent packets through the WLAN interface. In this environment, were generated 60 calls through one-day period in different time. All the capture files were saved in a directory for later analysis.

With the use of filters on wireshark, it was possible to realize the relevant data export used for the statistical calculations. They were: packet size, average time, protocol, ip origin address, ip destination address and RTP packet sequence. With the exported data of each call, it was realized the import of these files to the statistic software R. With the information of the imported calls we realized an exploratory analysis through descriptive average statistics, swerve-pattern in each one of the imported files, as in general we obtained the graphics of all made calls. With the calls from the first environment saved and converted to one of the formats accepted by the R software, we imported the 60 files realizing the calculations of the metrics delay, jitter, packet loss and outflow. At the end of the analysis, we obtained 60 averages of each metric. The data obtained through the R tool will be illustrated in section 5. To estimate the population average of the metrics, it was used a break of 95% of trust (3).

$$\mu = \pm z \frac{\sigma x}{\sqrt{n}} \tag{3}$$

Where, ***μ*** corresponds to the average of the population, ***z*** the number of pattern swerves in the table of normal distribution, σx corresponds to the pattern swerve of the population and ***n*** the number of elements in the sample.

V. RESULTS

The analysis of the calls presented in this study show the impact provoked use of VoIP in WLAN networks with the use of WDS. In the first environment the average values found are the values expressed in table IV.

A. Environment 1

TABLE IV
AMBIENT 1

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	19.98	19.99	20	0.002
Jitter	0.002	0.006	0.008	0.0008
Throughput	85.55	85.62	85.67	0.01
Packet Loss	0	0	0	0

In this environment we note the values of the metrics suitable to voice transmission, inasmuch the values attend the maximum requirements according to the recommendations of ITU-T. The delay average value found on the calls is within the maximum value accepted for the VoIP according to ITU-T – G.114's recommendation [12]. The variation of the delay is quite below of what is recommended, as well as the other metrics favoring this way the environment for the VoIP transmission. It is important to remember that this environment was analysed initially as a reference to be compared with the results of the second environment where the network's distribution is done using WDS. The variation of the delay in each call in this environment was relatively small, according to the illustration of picture 5. The small delay's variation can be justified by the circumstance of the environment being in use only by the two stations, that is, it did not occur any background traffic at the moment of the collection.

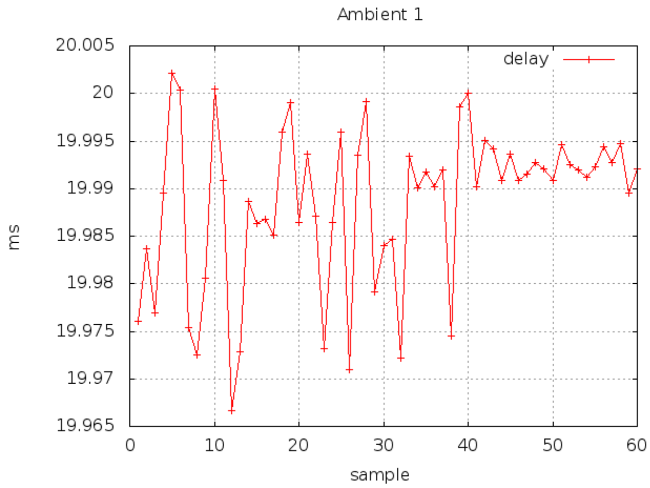

Fig. 5. Delay Mean

The other metrics also had little variation on the realized calls sample. The jitter's average is expressed on picture 6.

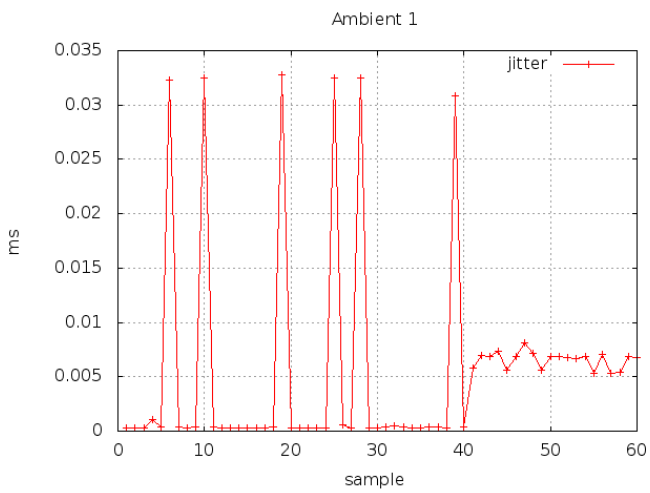

Fig. 6. Jitter Mean

The average jitter on this environment was kept around 0.006 *ms* that is the time's variation in the packets delivery was very small. The loss found in this environment was null, as expected and confirmed. The average throughput was 85.62 bits/s was found in the sample and the same way of other metrics they remained constant and coherent to the environment without representing grand variations, according to picture 7.

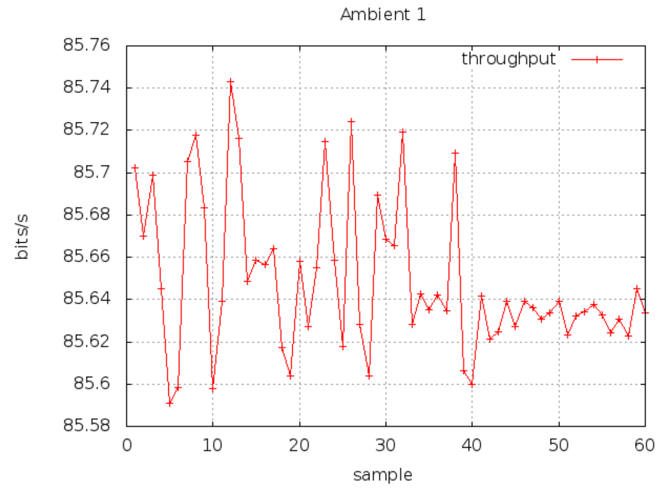

Fig. 7. Throughput Mean

To estimate the metric's average population collected in this study we used a gap of trust of 95%. Applying the formula (2) we achieved.

TABLE V
CONFIDENCE INTERVAL

	Lower Endpoint	Mean	Upper Endpoint	μ
Delay	19.9935	19.9942	19.9949	0.0007
Jitter	0.0063	0.0065	0.0067	0.0002
Throughput	85.6213	85.6246	85.6278	0.0032
Packet Loss	0	0	0	0

The quality call measurement was realized in this study using the E-model. Initially it is necessary to know the scalar factor (R-factor) that varies from 0 to 100. The R factor is related with MOS as described on section 3 methodology. In this first environment we obtained the average factors.

TABLE VI
FACTOR R AND MOS

	Factor R Mean	MOS
Ambient 1	70.36	3

All the calls made in the first environment had a quality average prediction on the scale MOS 3 that suggest the calls had a reasonable quality, where some users would be unsatisfied with that quality. This value was expected due to used codec in the G.722 study, being for this, expected a MOS on scale 3.

B. Environment 2

In the second environment collection the average values of the metrics are the values expressed by collection points. In the tables below are illustrated the resulting average values from five collection points.

TABLE VII
AMBIENT 2 - P1

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	19.98	20.51	20.96	0.31
Jitter	0.001	0.01	0.05	0.01
Throughput	81.66	83.46	85.66	1.27
Packet Loss	0	0.11	2.09	0.45

The point P1 was located roughly 46,44 meters of distance from the access point. A station was associated to WDS1 at the moment of the calls. The average delay in this environment was 20,51ms. Compared to the average delay in the first environment this corresponds to an increase of 2,60% on the average delay. The jitter metric had an increase of 172% in this collecting point compared to the environment without the WDS. The throughput had a loss of 2.52% when compared to the average throughput of the first environment. The metric packet loss had an average value of 0.1%. In the first environment we did not have packet loss.

Fig. 8. Delay P1.

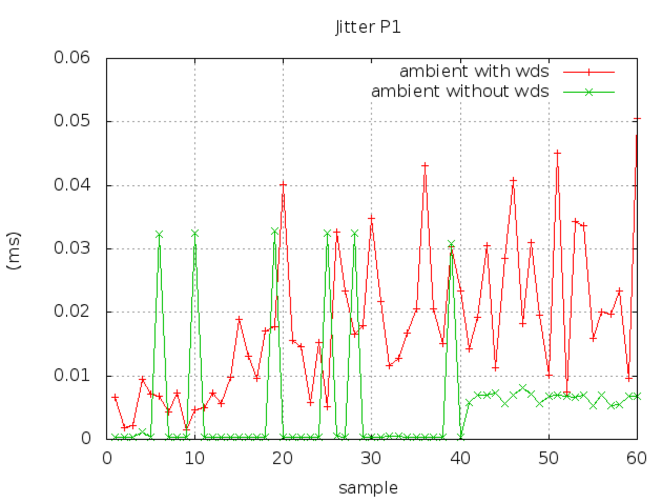

Fig. 9. Jitter P1.

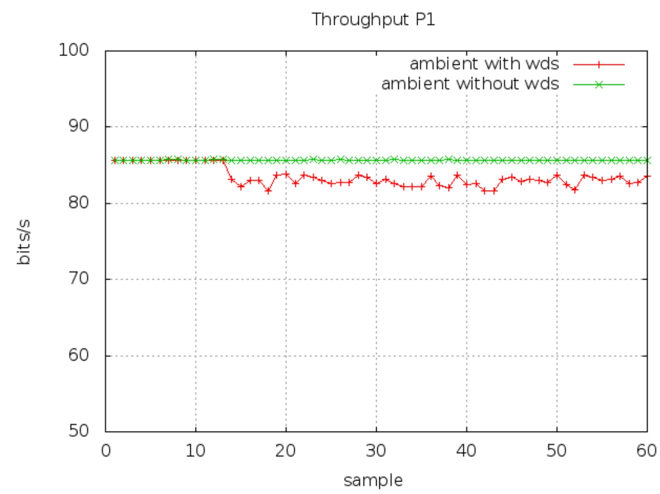

Fig. 10. Throughput P1.

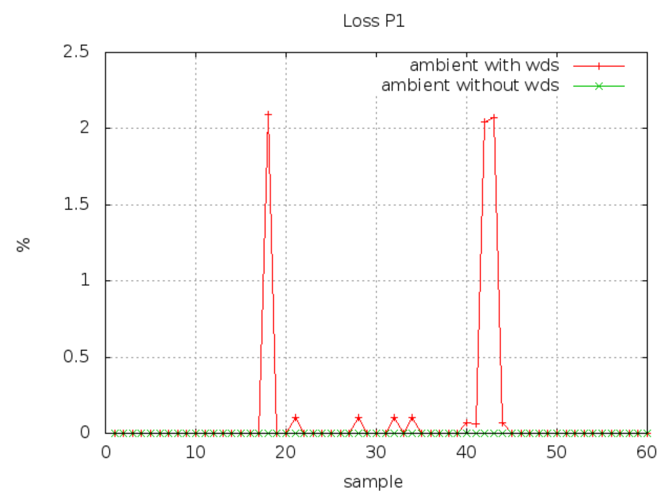

Fig. 11. Packet Loss P1.

Estimating the real average of the metrics population to a gap of 95% of trust in P1, we obtained:

TABLE IIX
CONFIDENCE INTERVAL

	LOWER ENDPOINT	MEAN	UPPER ENDPOINT	μ
Delay	20.4374	20.5159	20.5943	0.0784
Jitter	0.0149	0.0179	0.0209	0.0029
Throughput	83.1437	83.4662	83.7888	0.3225
Packet Loss	0.0010	0.1136	0.2284	0.1147

Measuring the quality of the realized calls in this point, we obtained:

TABLE IX
FACTOR R AND MOS

	FACTOR R MEAN	MOS
Ambient 2 - P1	69.84	2

Comparing the MOS obtained in the first environment with MOS of the P1 point was found the R factor equal to 69.84, resulting in a inferior scale of the realized calls in the environment without WDS. Such fact can be justified due to the station being associated to the WDS1, where there is a jump between the access points to reach the PBX.

The second collection point, P2 was at a rough distance of 41.96 meters from the access point WDS1. The metric's average is illustrated on table X.

TABLE X
AMBIENT 2 – P2

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	19.98	20.29	21.48	0.39
Jitter	0.0006	0.009	0.038	0.007
Throughput	79.68	84.38	85.66	1.61
Packet Loss	0	0.0051	0.17	0.02

The metric's delay had an increase of 1.51% in the average when compared to the environment 1. Jitter had an increase of 44.27%. Throughput had an average of 1.45% inferior to the obtained in environment 1. Packet loss had an increase of 0.005% in this point.

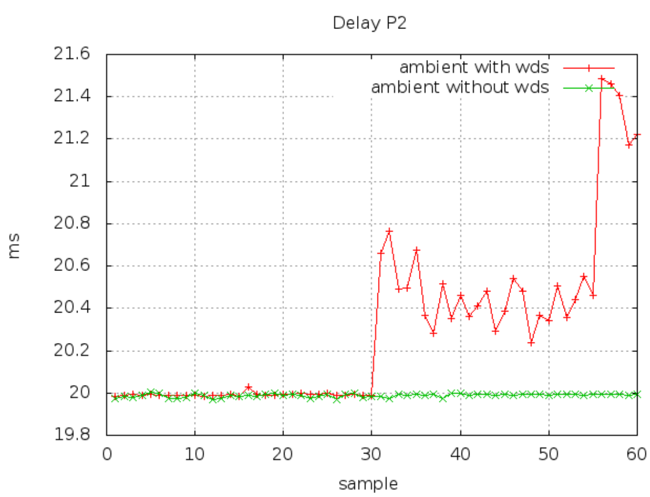

Fig. 12. Delay P2.

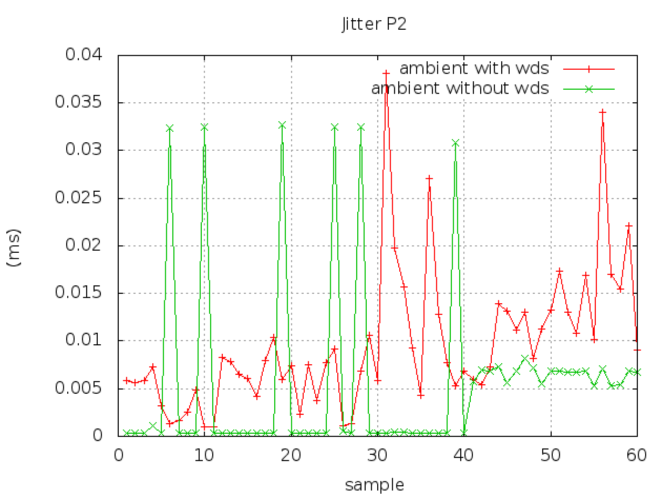

Fig. 13. Jitter P2.

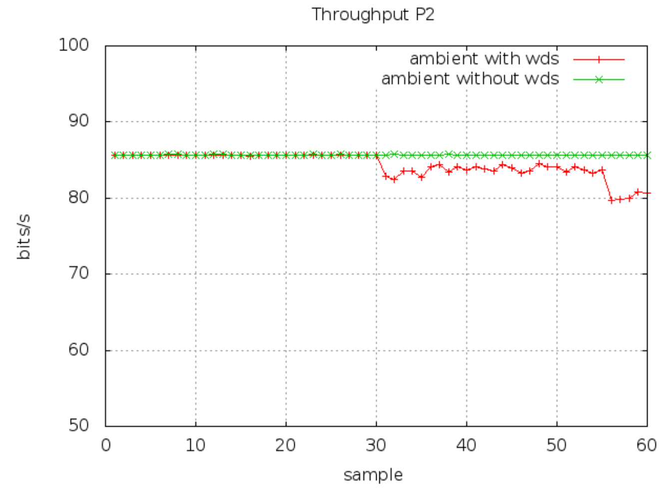

Fig. 14. Throughput P2.

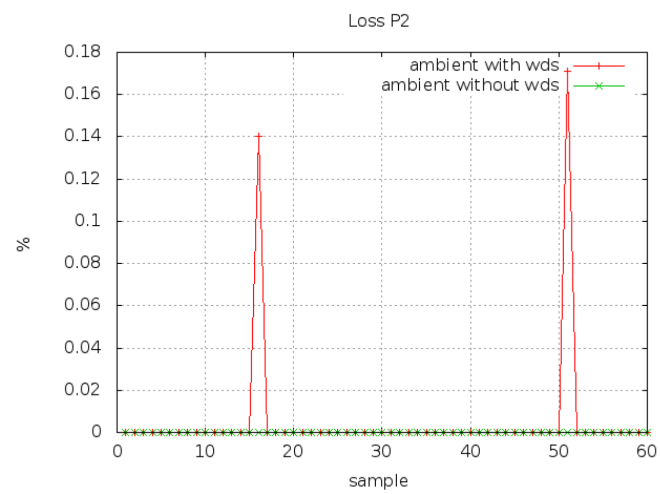

Fig. 15. Packet Loss P2.

Estimating the population average of the metrics in P2, we obtained:

TABLE XI
CONFIDENCE INTERVAL

	LOWER ENDPOINT	MEAN	UPPER ENDPOINT	μ
Delay	20.1957	20.2966	20.3974	0.1008
Jitter	0.0076	0.0094	0.0113	0.0018
Throughput	83.9722	84.3800	84.7877	0.4077
Packet Loss	0.0019	0.0051	0.0123	0.0071

Measuring the quality of the calls realized in this point, we obtained:

TABLE XII
FACTOR R AND MOS

	FACTOR R MEAN	MOS
Ambient 2 – P2	70.06	3

The MOS obtained in P2 was the same obtianed in P1. The rising of the average of delays and jitter can be justified too by the aditional jump now realized in this collection point to reach the PBX. This jump inserted a significant delay and jitter in the calls, this way the MOS was obtained due to the scale for the obtained value 3 comprehend values between 70 and 80 for the R factor.

The collection point P3 was at a distance of 14.73 meters from the WDS1. The average of the metrics is expressed on table XIII.

TABLE XIII
AMBIENT 2 – P3

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	20.32	20.64	21.31	0.18
Jitter	0.000004	0.014	0.034	0.007
Throughput	80.33	82.94	84.21	0.75
Packet Loss	0	0.16	4.78	0.76

In this point the average delay had an increase of 3.24% compared to the first environment. Jitter had an increase of 125.05%. Throughput had an average of 3.13% less when compared to environment 1. Packet loss had an increase of 0.16%.

Fig. 16. Delay mean in P3.

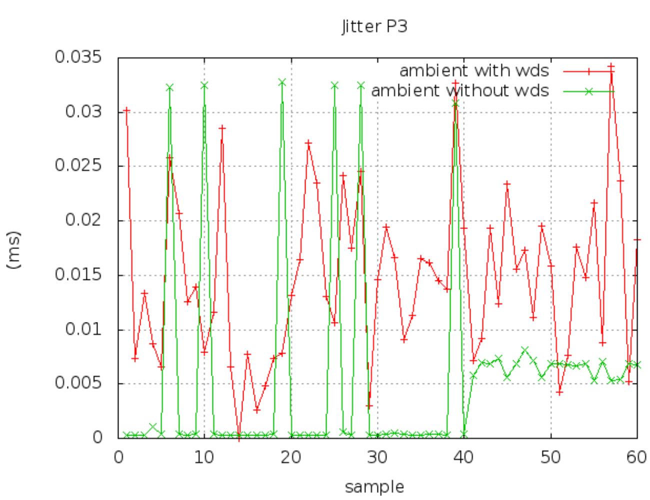

Fig. 17. Jitter mean in P3

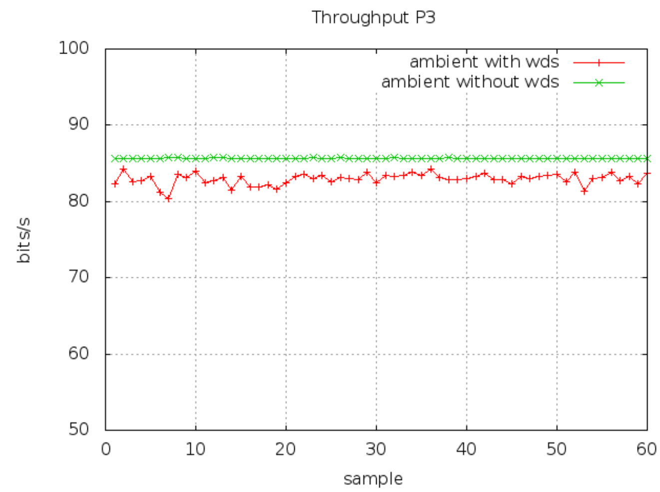

Fig. 18. Throughput mean in P3.

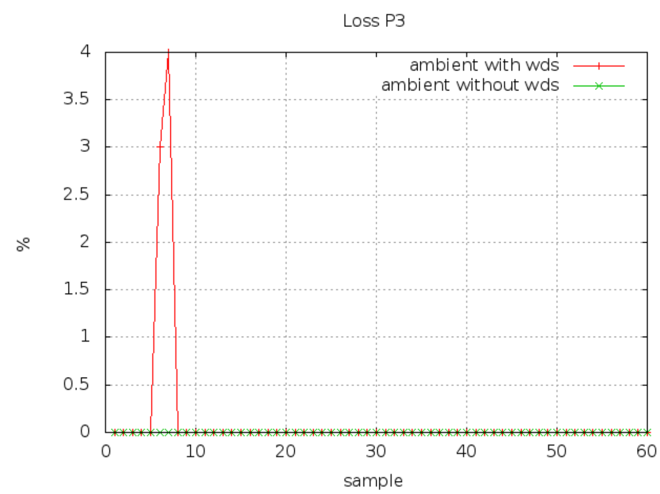

Fig. 19. Packet loss mean in P3.

Estimating the average population of the metrics in P3, according to table XIV, and calculating the MOS we obtained:

TABLE XIV
CONFIDENCE INTERVAL

	LOWER ENDPOINT	MEAN	UPPER ENDPOINT	μ
Delay	20.5945	20.6424	20.6904	0.0479
Jitter	0.0128	0.0148	0.0167	0.0019
Throughput	82.7518	82.9425	83.1332	0.1907
Packet Loss	0.0262	0.1683	0,3629	0.1946

TABLE XV
FACTOR R AND MOS

	FACTOR MEAN	MOS
Ambient 2 – P3	69.71	2

In P3 the collection was realized near to the access point. In this point we obtained a MOS equal 2. This loss in performance can be justified by the raise of the average delay and jitter, what degraded the quality of the call. The fourth collecting point P4 is located at 31.07 meters from the access point WDS1. In this point we obtained the following averages:

TABLE XVI
AMBIENT 2 – P4

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	19.99	20.60	22.10	0.42
Jitter	0.003	0.016	0.041	0.009
Throughput	77.44	83.10	85.64	1.66
Packet Loss	0	0.03	2.05	0.26

In P4 the average delay had an increase of 3.07% in relation to the first environment. Jitter had an increase of 153.21%. Throughput an average of 2.94% less when compared to environment 1. The packets loss in this collection point was 0.034%.

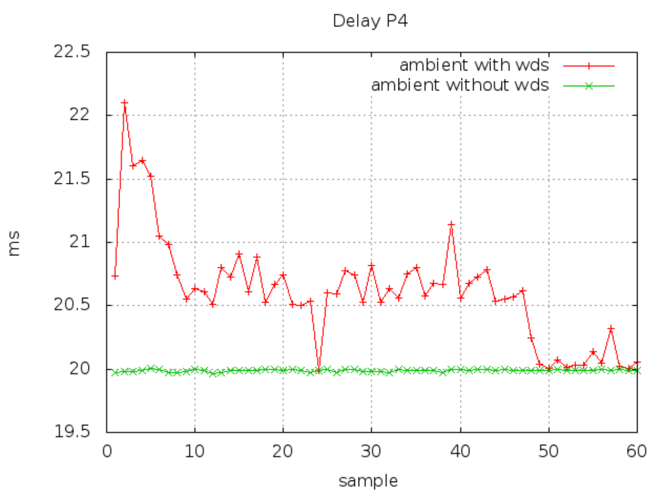

Fig. 20. Delay mean in P4.

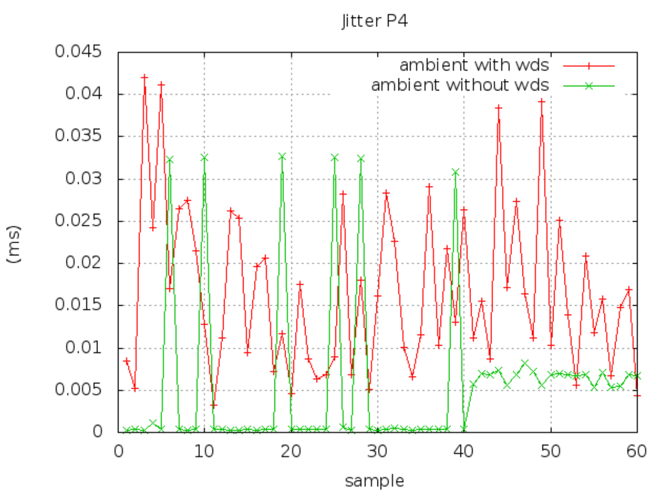

Fig. 21. Jitter mean in P4.

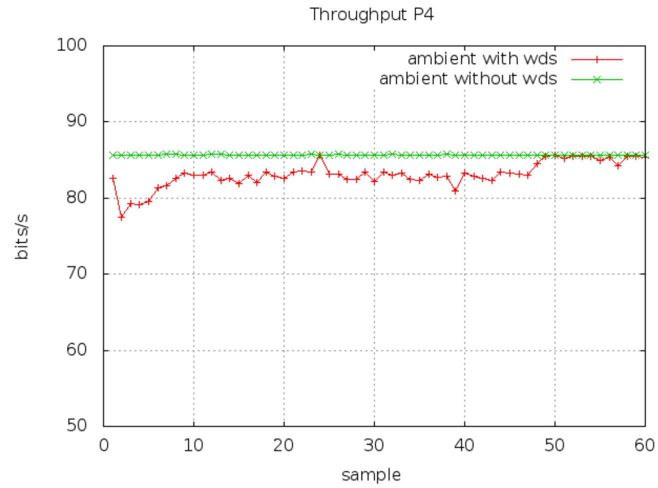

Fig. 22. Throughput mean in P4.

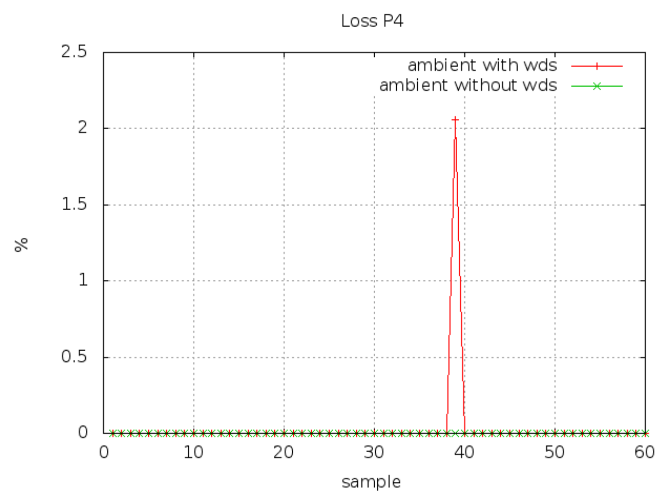

Fig. 23. Packet loss mean in P4.

Estimating the population average of the metrics in P4 (see table XVII), and calculating MOS (see table XVIII) we obtained:

TABLE XIV
CONFIDENCE INTERVAL

	LOWER ENDPOINT	MEAN	UPPER ENDPOINT	μ
Delay	20.5025	20.6091	20.7156	0.1065
Jitter	0.0142	0.0166	0.0191	0.0024
Throughput	82.6813	83.1035	83.5256	0.4221
Packet Loss	0.0328	0.0342	0.1013	0.0671

Measuring the quality of the calls made in this point, we obtained:

TABLE XVIII
FACTOR R AND MOS

	FACTOR R MEAN	MOS
Ambient 2 – P4	69,75	2

In the collection point P4 we obtained a R factor equal to 69.75. This value led the MOS to be measured on a scale like 2, what turns the quality call bad for many users. The collection point P5 was at a distance of 16.30 meters from WDS2. In this collection the packets transmitted through the station makes use of two jumps to reach DS.

TABLE XIX
AMBIENT 2 – P5

	MIN	MEAN	MAX	STANDARD DEVIATION
Delay	20,54	22,78	28,65	1,84
Jitter	0,00001	0,028	0,063	0,012
Throughput	59,74	75,57	83,32	5,62
Packet Loss	0	0,98	24,97	3,42

In P5 the average delay had an increase of 13.97% in relation to the first environment. In this collection we computed the worst performance for the metric's delay. Jitter had an increase of 326% compared to the first environment. Throughput had an average of 11.78% less than when compared to the first environment. Packet loss was computed in 0.98%.

Fig. 24. Delay mean in P5.

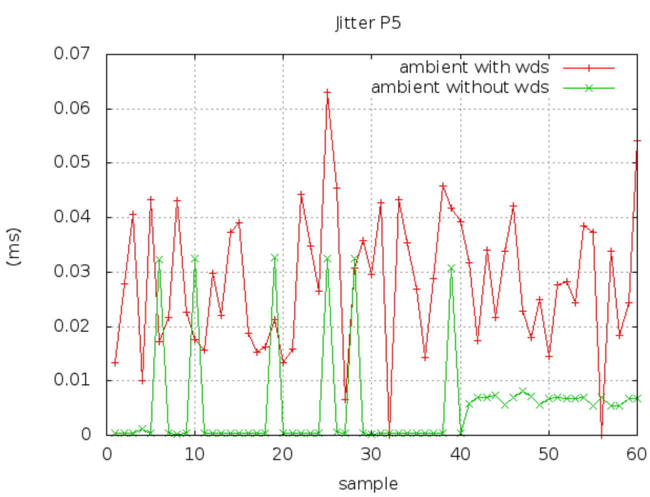

Fig. 25. Jitter mean in P5.

Fig. 26. Throughput mean in P5.

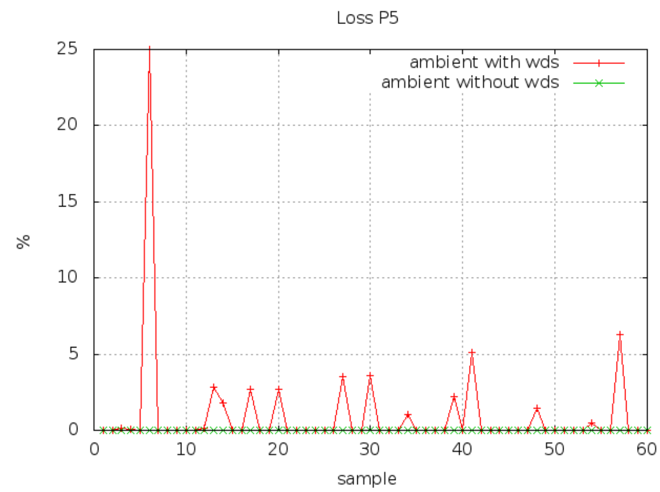

Fig. 27. Packet loss mean in P5.

TABLE XIV
CONFIDENCE INTERVAL

	LOWER ENDPOINT	MEAN	UPPER ENDPOINT	μ
Delay	22.3216	22.7876	23.2537	0.4660
Jitter	0.0248	0.0280	0.0313	0.0032
Throughput	74.1502	75.5734	76.9966	1.4232
Packet Loss	0.1170	0.9831	1.8492	0.8661

TABLE XXI
FACTOR R AND MOS

	FACTOR MEAN	MOS
Ambient 2 – P5	67,57	2

In P5 was observed the worst performance among the five collection points. This fact can be justified due to the transited packets among the stations make use of the 2 jumps using the access points WDS2 and WDS1 to reach the PBX. This additional jump influenced directly in the outflow's performance, increase of the delay and jitter, as well as the loss began to have a considerable value at this point.

VI. Final Considerations

In this study we analysed the use of VoIP applications in WLAN environments with and without using WDS. Through the execution of calls in two environments we observed and computed significant differences on the analysed metrics (see table XXII). In both environments analysed we found the metrics favorable for voice transmissions on IP networks.

TABLE XXII
All Ambients

	Delay	Jitter	Throughput	Packet Loss
Ambient 1	19,9942	0,0065	85,62	0
Ambient 2 P1	20,5159	0,0179	83,46	0,113
Ambient 2 P2	20,2966	0,0094	84,38	0,005
Ambient 2 P3	20,6424	0,0148	82,94	0,168
Ambient 2 P4	20,6091	0,0166	83,10	0,034
Ambient 2 P5	22,7876	0,0280	75,57	0,983

We observed in the environment where WDS is practiced the difference in the computed metrics. These metrics have different values when the collection point is using other points to take the packets to the DS. In our study the collection point P5 was the only one analysed in what the packets transmitted need to pass through two access points to reach the DS. It is possible that exists a relation between the number of jumps and the use of WDS. We observed the different metrics in environments with a number of jumps superior to 1. These metrics were observed on point P5 where the delay's average suffered a considerable growth. The other metrics also had its average values altered. Jitter had an increase and throughput suffered a loss. Packet loss came close to 1%.

VII. Future Assignments

As future assignments we aim yet to realize the measurement in other 5 points scattered through IFTO. In these additional points we expect to find metrics that degrade even more the VoIP applications. These 5 points use a higher quantity of jumps between the access points to reach the DS. As a future assignment it remains to use a multiple regression model to estimate the MOS using the variables of QOS looking for a relation between the analysed variables.

Other assignment to be realized in this same environment is to make calls with the stations in movement. We tried to observe the variations of the metrics at the moment of the change of BSS, looking to compute the delay and the influence of it in the VoIP applications.

References

[1] Jireware, "JiWire Mobile Audience Insights Report 2012 ". Available: http://www.jiwire.com/

[2] M. I. Anis and M. Z. Khan, J. Inan, R. Azmatullah, "Comparison between the ieee 802.11 x standards of voip using the cbr and vbr voice schemes". In Radio and Wireless Symposium, 2008 IEEE. IEEE, 2008.

[3] A. Chan and S. C. Liew. "Performance of voip over multiple co-located ieee 802.11 wireless lans". IEEE Transactions on Mobile Computing. 2008.

[4] L. Sun. "Speech Quality Prediction for Voice over Internet Protocol Networks". 2004 198 f. Thesis (Doctor of Philosophy in Computing, Communications and Eletronics) Faculty of Thecnology – University of Plymouth, Plymouth – United Kingdom.

[5] ITU-T The E-Model, a computational model for use in transmission planning. Recommendation G.107. Geneva – Switzerland. 2005.

[6] Canonical. Ubuntu 10.10. [Online]. Available: http://www.ubuntu.com

[7] AsusTeK Computer Inc. [Online]. Available: http://br.asus.com/Eee/Eee_PC/Eee_PC_1015PEM

[8] Jitsi Communicator. [Online]. Available: https://jitsi.org

[9] R. The R Project for Statistical Computing. [Online]. Available: http://www.r-project.org/

[10] Gnuplot. [Online]. Available: http://www.gnuplot.info/

[11] 3Com.[Online. Available: http://h20000.www2.hp.com

[12] ITU-T G.114. One-way transmission time. Recommendation. 05/2003.

Edgar S. Arrais. Graduated in of Technology in Computer Networks at University Lutheran Center of Palmas (2009). Pos graduate in telematics at the Federal Institute of Education, Science and Technology of Tocantins (2012). He is currently Manager of Networks - Department of Public Safety Tocantins. He has experience in Computer Science with emphasis in Computer Networks, acting on the following topics: embedded system, software, open source, hardware and management.

Claudio C. Monteiro. Graduated in Data Processing Technology from the University of Amazonia (1990), Masters in Computer Science from Universidade Federal da Paraíba (1997). PhD in Electrical Engineering at the University of Brasilia - UNB (2012). In addition, is teacher at the Federal Institute of Education, Science and Technology of Tocantins (IFTO - palmas.ifto.edu.br), leader of the Research Group on Networks Computers (Gredes - gredes.ifto.edu.br). He has experience in computer science, with emphasis on wireless networks, network protocols, QoS / QoE and operating systems.