A feasibility study of powerline communication technology for digital inclusion in Brazilian Amazon

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ABSTRACT

In the current national scene, many actions point at projects of digital inclusion and citizenship. In this context, providing access technologies as a requisite for the implementation of these actions is primordial. In this way, many innovative experiences have been presented in the past few years. This paper presents a study on the Powerline Communication – PLC technology; as a proposal for a feasible access network for Brazilian Amazon. First, the characteristics of the PLC technology are studied from an implanted indoor prototype at Federal University of Pará. The measures used in this prototype serve as input for a created model, from which it is intended to study the system more widely, considering factors such as: scalability, reliability and the physical characteristics.

Keywords: Power line communication, VoIP, digital inclusion

1. INTRODUCTION

The Brazilian Amazon has particular characteristics that motivate a great number of researches in all knowledge areas and general interest of the entire world. Despite the natural appeal, the under development of the region promotes many actions to diminish the existent problems. One of then is the poor communication infrastructure, not only last mile communication, but even indoor, which increases the digital exclusion. The Federal University of Pará, particularly, the High Performance Networks Planning Laboratory, has being conducting its efforts on communication technologies researches (e.g. Digital Subscriber Line - DSL and Powerline Communication - PLC) with an approach strongly directed to the peculiar characteristics of the Amazon region, toward the digital inclusion, as [1] and [2].

In this context, actions such teleeducation stands out due to joint use of technology and education, contributing to digital and social inclusion of people. An example is the possibility of training a group of people, in different classrooms, in the same time, with only one teacher. This paper presents a feasibility study of powerline communication technology applied to digital inclusion with a case study of VoIP.

There have been many researches on VoIP over consolidated technologies. In [15] was made a performance evaluation of VoIP over 3G-WLANInternetworking system focused in the performance of VoIP with IPSec tunnel in 3G-WLAN internetworking system, the maximum number of VoIP connections in a single 802.11b Access Point, the handoff latency due to terminal mobility and the impact of terminal mobility on the VoIP performance. In [16] the interest is in verify the QoS in terms of MOS value on VoIP over WLANS. [17] presents a performance evaluation of VoIP over satellite networks under different link and traffic conditions based on laboratory experiments. [18] study the performance of VoIP over GPRS in terms of capacity of voice connections compared with a traditional circuit switch.

The characteristics of PLC technology are shown in section 2. Section 3 shows the main characteristics of VoIP, which have a straight influence on quality of the transmitted voice. The performance evaluation of VoIP over PLC is presented in section 4 and a case study in section 5.

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2. POWERLINE COMMUNICATION

The PLC technology uses the existing electric infrastructure to provide the data communication. This technology has shown itself sufficiently competitive in the broadband access market, disputing with other more traditional, like DSL. PLC counts with the advantage of having a sufficiently wide infrastructure and with relatively little cost associated [3].

PLC, however, as well as other data communication technologies, possesses some inconveniences. The physical environment is very hostile for the data, once that it was not conceived for this purpose, thus there are many properties of the power systems which influence negatively the high speed communications (losses in the cable, propagation in multiple paths and the noise, for example) [4].

A way to reduce the impact of the transmission environment in the communication is the application of efficient modulation methods, such as OFDM (Ortogonal Frequency Division Multiplexing), besides mechanisms for error correction such as FEC (Forward Error Correction) and the ARQ (Automatic Repeat reQuest) [5].

However, independent of the adopted techniques, a factor to be observed and understood is the noise in the communication.

2.1 Noise Characteristics

Given the physical characteristics of the conductor and the nature of the electric energy, this environment is not best indicated for data communication. The propagation of the signal through the power transmission line provokes an attenuation and a delay in the signal, which increases with the distance and the frequency. This attenuation depends on the impedance characteristic (Z_L) and on the propagation constant (γ) of the transmission lines. In accordance with [13] and [1], these two parameters are resistance functions R, conductance G, inductance L, and capacitance C, per unit of length, which depend on the frequency.

Based on the studies presented in [6] and [7], two types of noises can be adopted: (a) deep noise, formed by deep colored noises, short band noises and asynchronous periodic impulsive noises in relation to the channel frequency; and (b) impulsive noise, formed by asynchronous impulsive noises and synchronous periodic impulsive noises in relation to the channel frequency. Having the latter a great probability to cause bit errors or in bits sequence in a of high speed data communication [4].

With the purpose to characterize the typical traffic, as well as the essential characteristics of this type of communication regarding the communication channel and the noises, a PLC prototype, which will be described in section 3, is being tested.

2.2 PLC channel model

The topology of the power distribution networks considerably differs from the traditional communication networks, such as twisted pair, coaxial cable or optic fiber. Numerous reflections of the signal are received and occur mainly due the junction of cables of different impedances.

The PLC channel has the characteristic of being selective to the frequency, this if due to the multipath characteristic of the power transmission networks.

Given the hostile characteristics of PLC channel (multipath, selective to the frequency and the impulsive noise) it is necessary to apply an efficient modulation technique.

For allowing the transmission of data through many independent sub-carriers, the OFDM modulation technique has been a good alternative for the data transmission on the PLC channel. Once that only a small percentage of sub-carriers will be reached by the effect of frequency selectivity of the canal, error correction codes can be used to correct the data of the few wrong sub-carriers, as well as new sub-carriers can be used to substitute the wrong ones.

The OFDM modulation characteristics regarding the impulsive noise and the multipath are presented in details in [8].

3. VOIP

The use of voice over IP (VoIP) comes being one of the great goals of investments for solution suppliers and users of telecommunication in last and next years. As the name implies, VoIP refers to calls that traverse networks using Internet Protocol (IP). This technology came to change the way we communicate on the Web, as well as the way we do business, it opens a new horizon for combining voice and data in the same terminal equipment of user, increasing the number of applications to be accessible through the same media for transmission and diminishing the costs of communication when compared with the conventional interurban calls.

VoIP can facilitate tasks that may be more difficult to achieve using traditional phone networks, like many VoIP packages include PSTN features that most telcos normally charge extra for, or maybe be unavailable, such as integration of data, voice and fax, sound grading, video telephony, unified messaging and web-based call centers, which are not available in traditional telephone networks [9]. It also allows users to travel anywhere in the world and the incoming phone calls can be automatically routed to your VoIP phone, regardless of where you are connected to the network.

The first process in an IP voice system is the digitization of the speaker's voice. Then occurs the compression of the signal, after that the system examines the digitalized data to discards any packets that do not contains a speech. Second, some algorithms and sophisticated codecs are employed to reduce the amount of information that must be sent and compression the voice streams respectively. Just remembering that a normal telephone uses circuit switching for phone calls, which involves routing of your call through the switch at your local carrier to the person you are calling. Packet switching on the other hand is more efficient in transmitting data since a packet is sent from one system to another, having several advantages over circuit switching. The key difference between packet and circuit switching is that the first one allows numerous telephone calls to inhabit the space occupied by only one in a circuit-switched network.

VoIP is best understood as a collection of the protocols that make up its mechanics. Those protocols are loosely analogous to the PSTN (Public Switched Telephone Network), which is broken down into three categories: access, switching, and transport. Three categories of protocols are relevant to VoIP: signaling, routing and transporting.

3.1 H.323 Protocol

The H.323 is an ITU-T standard [10] that does not guarantee QoS and it was developed before the emergence of VoIP, but it is the most widely used VoIP standard. This protocol the base technology for the transmission of real-time audio, video and data communication over packet based networks. It specifies the components, protocols and procedures that provide multimedia communication over packet based networks (include LANs and WANs).

It addresses call control, multimedia management, bandwidth management and interfaces between LANs and other network. The entities in the H.323 topology are the endpoints, also called user terminals, gateways (GW), multipoint control units (MCU) and gatekeeper (GK).

The gateway perform the translation of signaling and media exchange between H.323 and PSTN endpoint [11], for example, a VoIP gateway provides translation of transmission formats and signaling procedures between a telephone switched circuit gateway network (SCN) and a packet network. In addition, the VoIP gateway may perform speech transcoding and compression, and it is usually capable of generating and detecting DTMF signals [12].

Although MCUs, though listed separately, are in practice part of a gatekeeper or a high speed computer that acts as a terminal serving one or more users. And the gatekeeper, that is used for admission control, address translation function, manage network bandwidth, allocate voice, video and data, handle routing for multiple H323 gateways, provide for charging and billing and support network security and subscriber authentication.

On the other hand to set up a call using the H.323 is really complexity and takes a long time. First, the protocol uses multiple roundtrip messages to establish signaling and control for any call between two terminals. Moreover, H.323 requires that TCP connections be used to carry the messages, requiring an additional roundtrip exchange. The recently released version 3 is an improvement and includes both a "fast connect" procedure that effectively consolidates the Q.931 messages exchange between terminals, and a tunneling procedure that lets H.245 share a single TCP connection with Q.931 [13].

3.2 Quality of Service on VoIP applications

When packets are lost and delayed in any point of the transmission through the network, VoIP users will see a drop-out of voice.

What is more noticeable in case of congestion, that some packets could be lost or delayed in the network (one of major challenges is the provision of Quality of Service, currently IP does not offer any quality of QoS guarantees to the transmitted data), also a percentage of packet are discarded due to errors that occurs during the transmission. These problems results in a substantial deterioration of voice quality. Because VoIP systems are time sensitive and cannot wait for retransmission, more sophisticated error detection and correction systems are used to create sound to fill in the gaps [9].

So, there are some things to consider:

- Latency: Delay for packet delivery;
- Jitter: Variations in delay of packet delivery;
- Packet loss: Too much traffic in the network causes the network to drop packets;
- Burstiness of Loss and Jitter: Loss and Discards (due to jitter) tend to occur in bursts;

Jitter causes strange sound effects, but can be handled to some degree with "jitter buffers" in the software. Packet loss causes interrupts. Some degree of packet loss won't be noticeable, but lots of packet loss will make sound lousy.

To correct the packet loss it can be used some codecs algorithms, but they are effective only if a single packet is lost during the period of transmission. Today there are many codecs available for digitizing speech, such as ITU-T G.711 (PCM), G.723 (MP-MLQ), G.726 (ADPCM), G.728 (LD-CELP) and G.729 (CS-ACELP).

A very important factor affecting voice quality is the total network load, what was said previously. When the congestion in the network is high, jitter and frame loss will increase. So to get measurements of the network load, as well as the number of collisions, many different tools are available (protocols analyzers). To gauge the effect of priorities, it is possible to use the latency and loss application to ensure that priorities are configured correctly and indeed voice is given precedence over data traffic.

Another factor used to determine the quality of transmitted speech is the benchmark called Mean Opinion Score (MOS). With MOS, a number of listeners judge the quality of a voice sample on a scale of 1 to 5, where 1 is lowest perceived quality, and 5 is the highest perceived quality [16]. The average of the scores provides the MOS for that sample. Table 1 shows the rating scheme used by listeners to judge the voice sample. Each codec implementation has a Mean Opinion Score, the MOS of G.711, codec used in the experiments, is 4.1.

Table 1. Mean Opinion Score (MOS)

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

4. PERFORMANCE EVALUATION

To perform the study of VoIP over PLC a PLC prototype was used in conjunction with RADCOM© Performer Analyzer, which has a VoIP module that gives values such as jitter, packet loss and MOS. In sub-section 4.1 the testbed is presented.

4.1 Testbed of VoIP over PLC

The PLC Indoor (PLC/LPRAD) prototype is located in the dependences of the Laboratory of Electric Engineering and Computing - LEEC of the UFPA, in which is inserted the Laboratory of High Performance Networks Planning - LPRAD - and has the purpose of supplying information regarding the nature of the noise (physical layer), supply typical information the superior layers, load characterization and performance measures of the system (e.g. trustworthiness and scalability). The information obtained in this prototype is serving as entry parameters for the simulations of PLC network in the NS-2 [14].

The sub-station of the LEEC, after the transforming exit from medium to low voltage, possess two circuits, one for the building illumination, feeding of the air conditioners and for some outlets in building walls, and the other, for the power keys located in the building rooms, from which each room mounts its internal power system.

In this sub-station a PLC base station was installed (HE), which is connected to the three phases of the circuit that feeds the power keys of the rooms; this way, all the rooms of the LEEC will possess internet connectivity points through the PLC network. This circuit was chosen given the fact of being wider than the first one.

In LPRAD, there is a PLC MODEM with a machine that is the transmitter, and in another lab of LEEC there is a set PLC MODEM/ RADCOM© Performer Analyzer. The Figure 1 shows the VoIP over PLC prototype.

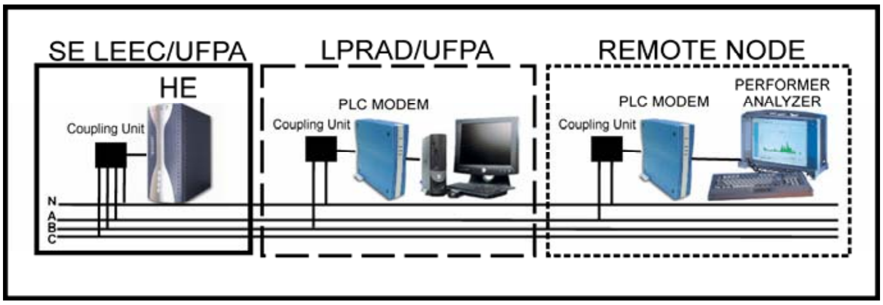

Fig. 1. VoIP over PLC Prototype Topology.

4.2 VoIP over PLC experiment

The experiment was performed using the prototype, previous described, with four remote nodes, used one by one, located in different points of the LEEC, two in the first floor (Power System Laboratory and Digital Electronics Laboratory) and two in the second floor (LPRAD and Laboratory of Applied Electromagnetism - LEA). To generate the VoIP calls, a software called CallGen323 was used, this software was developed by the OpenH232 Project and is available in [21]. The main objective of the OpenH323 Project is "to create a full featured, interoperable, open source implementation of the ITU-T H.323 teleconferencing protocol that can be used by personal developers and commercial users without charge."

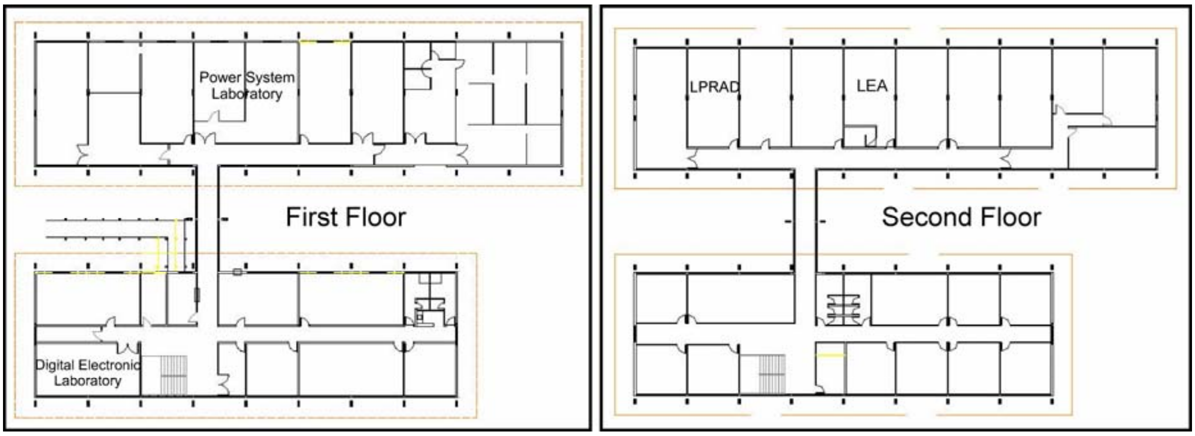

Fig. 2. LEEC Building

Each VoIP call had duration of three minutes, repeated twenty times, and total audio bytes of 81% and silence suppression of 20.38%. Each experiment takes one hour to be completed. The voice file used was obtained in [19]. All the metrics relative to VoIP calls were captured by MediaPro, a software module of RADCOM© Performer Analyzer. More information about MediaPro and RADCOM© Performer Analyzer can be found in [20].

The values obtained from the experiments are presented in Tables 2, 3, 4, 5 and figures 3, 4, 5 and 6.

Table 2. Obtained values - LEA

Metric	Value
Average packet loss (%)	0.1
Average jitter (ms)	4.1
Bandwidth (Kbps)	64.727
MOS	4.18

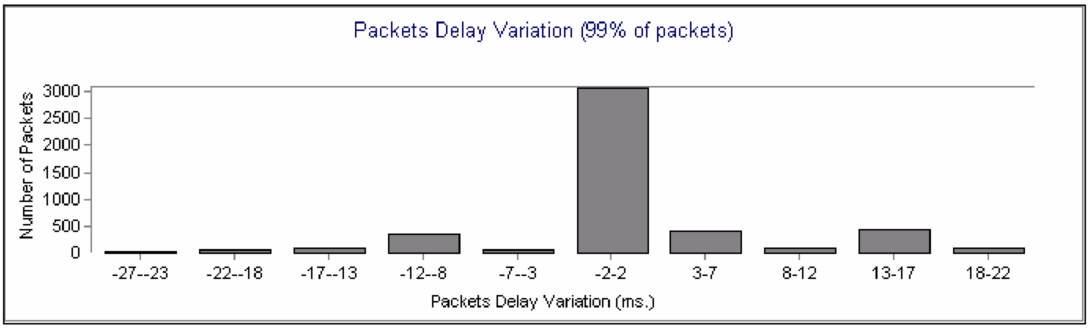

Fig. 3. Jitter obtained - LEA

Table 3. Obtained values - LPRAD

Metric	Value
Average packet loss (%)	0.2
Average jitter (ms)	4.2
Bandwidth (Kbps)	64.673
MOS	4.17

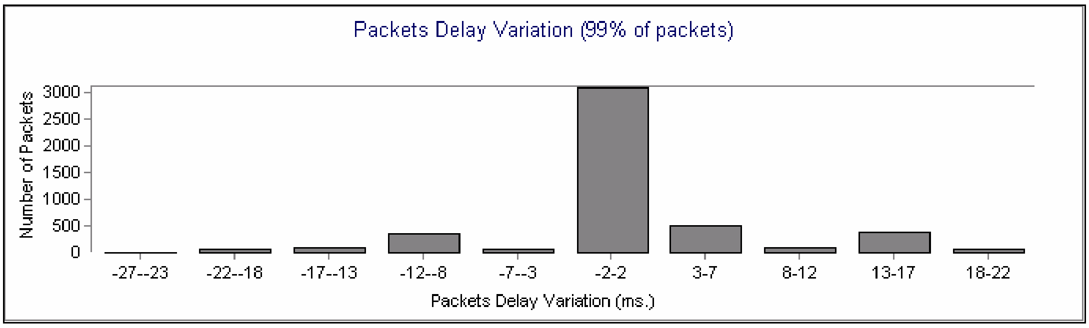

Fig. 4. Jitter obtained - LPRAD

Table 4. Obtained values – Power Systems Laboratory

Metric	Value
Average packet loss (%)	0.1
Average jitter (ms)	5.3
Bandwidth (Kbps)	64.436
MOS	4.08

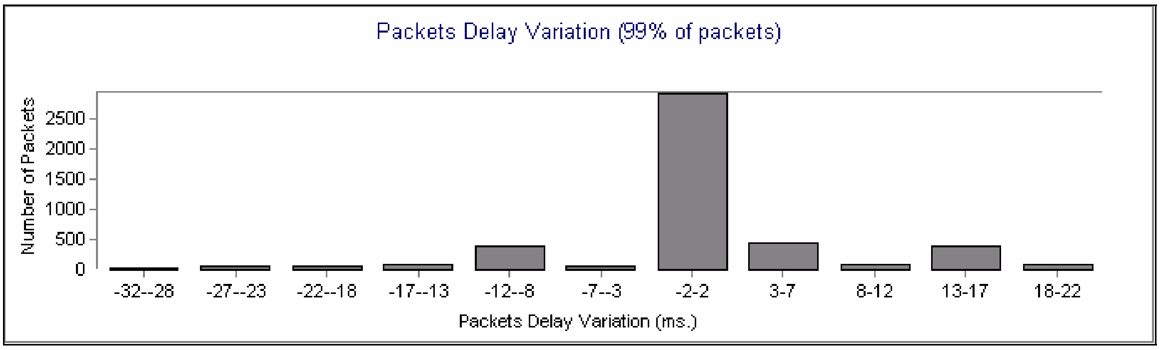

Fig. 5. Jitter obtained - Power Systems Laboratory

Table 5. Obtained values – Digital Electronic Laboratory

Metric	Value
Average packet loss (%)	0.0
Average jitter (ms)	4.8
Bandwidth (Kbps)	64.779
MOS	4.19

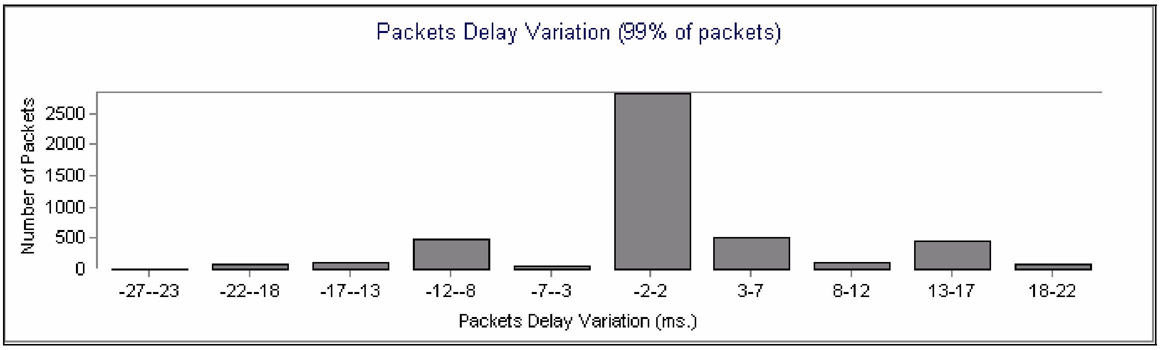

Fig. 6. Jitter obtained – Digital Electronic Laboratory

Based on the values observed in the prototype, a simulation model was constructed for the NS-2. The simulator allows error models to be created, this way, specific characteristics of the given network can be inserted, in what refers to the loss of packages. This is particularly interesting in the ratio where the physical characteristics of PLC network can be inserted in the model. With this, a more complete analysis can be made, enclosing from the inferior to the highest layers.

The error model uses basically two parameters: the percentage of packages loss and its probability distribution.

5. CASE STUDY: TELEEDUCATION

In order to verify the feasibility of the PLC technology for digital inclusion purposes, a fictitious scenario based on an idea of a Regional Training Center (CTR), located in main cities of Pará. In these cities will be concentrate all the efforts to training teachers, community people, health agents and so on, of neighboring cities. These kinds of actions are common in Amazon due to its particular characteristics.

A CTR is composed by three computer laboratories, each one with fourteen computers, two classrooms with one computer connected to Internet and the administration area.

The classrooms A and B are destined to video conferencing lectures; and the A, B and C laboratories are destined to internet access and distance learning classes. Figure 7 shows a PLC-based infrastructure of a CTR.

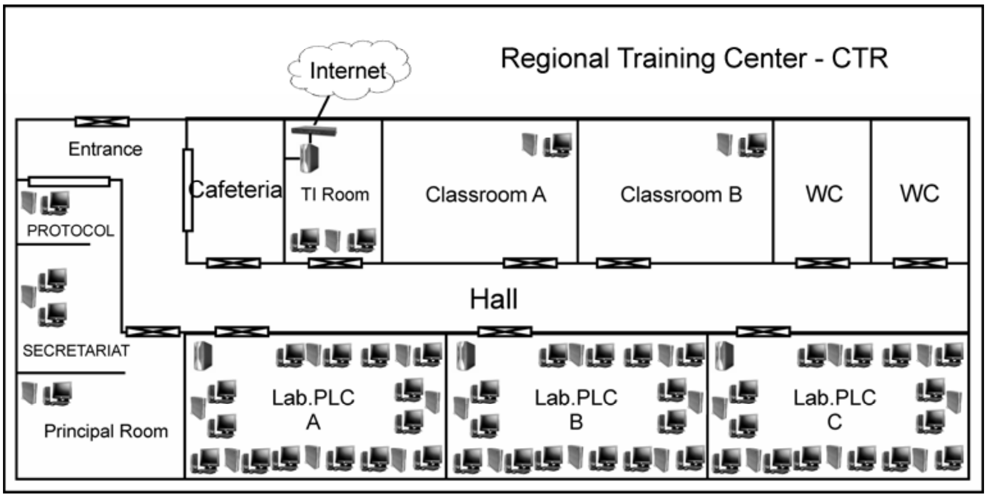

Fig. 7. CTR infrastructure

Two scenarios were proposed for scalability system analysis: (a) one computer on laboratory A is running VoIP application, and the other ones are running typical HTTP applications over Ethernet; (b) the second case is similar to the first one, but running over PLC network.

The simulations were performed considering VoIP using H.323 protocol.

To used model, the input data set was obtained from a trace file captured by RADCOM© Performer Analyzer during the experiments.

Table 6. Simulation values – PLC

Metric	Value
Throughput (Kbps)	60.73360
Blocking probability (%)	3.95

Table 7. Simulation values – Ethernet

Metric	Value
Throughput (Kbps)	63.27342
Blocking probability (%)	0.0

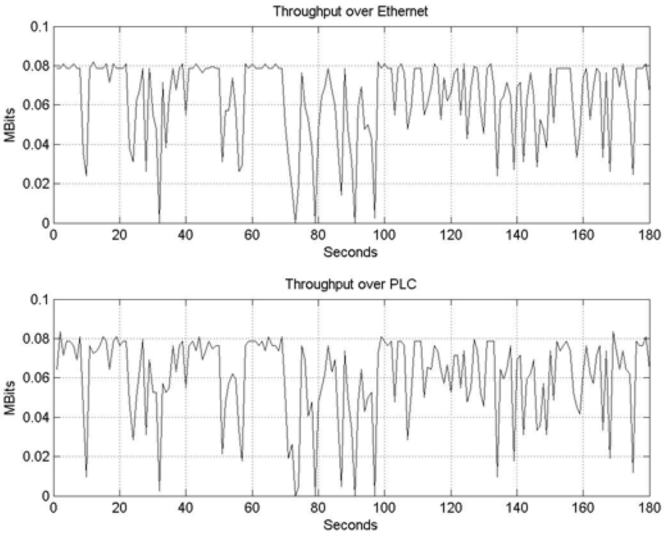

Fig. 8. VoIP throughput over PLC and Ethernet.

Although the PLC network presents a certain loss of packages due to the noise in the line, the transmission rates were satisfactory, not causing a significant degradation of the transmission in relation the Ethernet networks and the throughput observed in PLC simulation is near to the value obtained in experiment.

6. CONCLUSION

Observing the simulation results, it is possible to conclude that has a great viability in the use of PLC technology for action of digital inclusion in similar scenarios to the presented one in this paper, that basically characterize typical scenarios of the Amazon region. However, it is necessary to evaluate which applications will be used in the network; therefore an increase in the system's load can compromise the quality of the diverse applications that are running in the network. This paper is derived from a research line that investigates the viability of certain technologies under specific and singular conditions of the Amazon. For the factors presented in this paper and other diverse experiments carried out, it was observed that PLC technology has a great potentiality to provide communication in regions with "poor" telecommunication infrastructure, due to its existing infrastructure and the ubiquity of access points. As continuation of this research, it is being currently implemented in the LPRAD-UFPA an NS-2 module, PLC_NS, that incorporates characteristics of the environment to the traditional package routing evaluation made by that simulator.

7. ACKNOWLEDGMENT

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