

## VoIP Using Power Line Communications

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### Abstract:

*This study is conducted after perceiving the lack of gathered information and relatively new research and development on the usage of Voice over Internet Protocol (VoIP) through an existing infrastructure of electrical wires known as VoIP using Power Line Communications (PLCs). The primary purpose of the PLC is the delivery of AC (50 Hz or 60 Hz) or DC electric power. PLCs system operate the largest existing network in the world, which is the electrical distribution grid and therefore can establish communication networks for data transmission. Five experimental studies are implemented on Home Area Network (HAN) to test the efficiency of the proposed system and a comprehensive presentation dealing with the different aspects of VoIP over power lines are discussed.*

**Keywords:** VoIP, PLC, Broadband PLC, Internet Protocol-Private Branch Exchange (IP-PBX), Session Initiation Protocol (SIP).

## 1 Introduction

The world is accelerating in progress. Technology did not leave anything without putting a fingerprint in it. It is found in many fields such as: telecommunications, electricity, medicine and various fields of life.

In communications when phones were discovered, one obstacle among many was faced: the large amount of wires in the road for two phones to communicate with each other. What will be the case when we have large number of phones? So, technology has provided solutions to this problem by the introduction of telephone exchange. The development continued in more than one stage even to reduce the amount of wires in road. But if we look at a construction (e.g. company) that has an internal communication system that was not established when the company was built; exterior communication wires are needed. In addition, if we find Internet network in the company, we will notice two types of scattered wires around. Technology intervened again by giving us the opportunity to transfer data through the basic existing electricity wires in any building.

Because of the lack of research and projects for this new technology, it was decided to conduct research in PLC, communication network inside a home or a building using PLC. Our system will support: Internet, audio, text, and video. Thus, we have reserved the range (3 - 30) MHz to be used for data transfer away from the power range (50 - 60) Hertz.

The proposed system design is implemented in several stages to achieve the best way to transfer the data in PLC. These stages contain the use of power line adapter, building VoIP central, and connecting the adapter with central to begin using mobile app or IP telephony to communicate and transfer data between the participants.

This system can do more tasks such as transfer of data, voice, video, and support internet services. These data transfer on the PLC through its own scope away from the power scope to guarantee their safety and not to be lost, in addition to do the modulation for data. Modulation is a signal containing information onto another signal, called a carrier, which usually has a constant and much higher frequency. The modulated carrier that is carrying the present information in the original signal can be transmitted from one place to another and the original information recovered at the destination.

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The power line adapter encodes and transmits data on the reserved band, followed by the second phase, which is the construction of central VoIP to support voice, text, and video. The third phase, connecting one side of adapter to the central and the other end connects to access point. The fourth stage, we used mobile apps or IP telephony, each participant when connected through access point, gets a static IP. A participant can communicate with anyone by knowing his/her IP. The fifth stage, the Private Branch Exchange (PBX) server can provide Internet access for the network. Therefore, Instead of using additional external wires, PLC can solve this problem by transferring the data in the lines. This technology can support transfer of text, audio, and video by using VoIP and can support Internet service.

This research aims to achieve a new communication system using VoIP and PLC. The new system can transfer the data in a secure with low cost.

## **2 Literature Review**

In this section, communication and IP-PBX are discussed carefully. In the former one, communication provides the usage of PLC system as a medium including the modulation and demodulation of transmitted and received data, respectively. Noise is considered as part of a transmitted data and needs to be classified for removal. PLC standards including advantages and disadvantages are discussed and comparison between the PLC and other technologies are summarized. In the IP-PBX subsection, an overall concept is introduced and its characteristics, applications, and the usage of VoIP are discussed.

### **2.1 Communication**

This subsection introduces the overall concept of PLC in terms of a communication medium, modem as a communication device, noise categories and noise removal ways, PLC standards, PLC applications, and PLC advantages and disadvantages.

#### **2.1.1 PLC Overview**

Power Line Communication is a technology that employs the infrastructure of electrical power distributed system as communication medium by sending data over existing power cables. PLC technology could provide the consumer with a spectrum of services such as internet, home entertainment, home automation, and enable the

electricity supply authority to efficiently manage their distribution networks in a competitive manner [1 - 3].

While the idea of sending communication signals on the same pair of wires as are used for power distribution is as old as the telegraph itself. In the 1920's at least two patents were issued to the American Telephone and Telegraph Company in the field of "Carrier Transmission over Power Circuits". In 1950 was the development of the first technique used wire transfer power to convey messages and control commands invited (Ripple Control) and were used low frequencies ranging between 0 and 100 Hz, and the purpose of lights control lighting in the streets in order to adjust consumption [4].

PLC can be broadly viewed as: Narrowband PLC and Broadband PLC as discussed by [5, 6] including a comparison summarized in Table 1.

**Table 1. Comparison between Broad and Narrow PLC**

Comparative	Narrowband PLC	Broadband PLC
Data rate	Up to 200kbps	Over 1Mbps
Frequency	Up to 500kHz	Over 2MHz
Modulation	Frequency-shift keying (FSK), Spread Frequency-shift keying (S-FSK), Binary Phase-Shift keying (BPSK), Orthogonal Frequency-Division Multiplexing (OFDM)	OFDM
Applications	Building Automation, Renewable Energy, Advanced Metering, Street Lighting, Electric Vehicle, Smart Grid	Internet HDTV Audio Gaming

### **2.1.2 Operating Principle**

The communication device used for the communication over the power lines is a Modem. It works as both transmitter and receiver, i.e., it transmits and receives data over the power lines. A power line modem not only modulates the data to transmit it over the power lines but also demodulates the data it receives from the power lines [7, 8].

### **2.1.3 Noise on Resident Power Circuit**

According to the study done by [7], the noise is categorized into four categories. Firstly noise synchronous to the power system frequency (50Hz or 60Hz). Secondly, noise with a smooth spectrum. Thirdly, single event impulse noise. Fourthly, periodic noise.

There are a few ways to reduce the noise in the communication over power lines [7, 9]:

- Implementation of Forward Error Correction (FEC) codes with interleaving can reduce the noise in category 1, 2 and 3.
- Frequency hopping with the FEC coding.
- While modulating the signal on to the power lines, television line frequencies should be avoided by using filters.

#### **2.1.4 PLC Standards**

The PLC standards were described by [2, 7] as follows:

- European Committee for Electrotechnical Standardization (CENELEC): it defines standards for allowed frequency ranges and output voltages for the communication over power lines.
- Federal Communications Commission (FCC): it standardizes the frequency ranges and transmitted power ranges for the power line communications in North America. The allowed base frequencies range from 0 to 530 KHz.
- HomePlug Powerline Alliance: it is a group of companies dedicated to improve the technology for the networking and communication over power lines standard uses a physical layer protocol (PHY) based on 128 equally divided carrier OFDMs from a frequency range of 0 to 25MHz. It uses concatenated Viterbi and Reed Solomon coding for payload data, Turbo product codes for control data and BPSK, Differential Binary Phase Shift Keying (DBPSK), Differential Quadrature Phase Shift Keying (DQPSK) or ROBO modulation with a cyclic prefix for modulation of the data.
- IEEE 1901: it is for high speed power line communications with two different physical layers, first one based on OFDM modulation and the other one based on wavelet modulation.

#### **2.1.5 Applications of PLC**

- Transmission & distribution network.
- Home control and automation.
- Entertainment.
- Telecommunication.
- Security systems.
- Automatic meter reading.

### 2.1.6 Advantages of PLC

The main advantages of PLC as described by [2, 7, and 10] as follows:

- From the economic standpoint, it is very reasonable to use a pre-installed wired network instead of running new wires.
- Use of utility-owned resources.
- Power line communications offer the potential to integrate future smart devices like home security.
- Highest reliability (for example, for protection signaling).
- PLC is appealing because there is no need to run additional wires to powered devices.
- Can be used in effective combination with broad band technologies for highest availability.

### 2.1.7 Disadvantages of PLC

- PLC is a harsh medium and data transfer through it can create a lot of problems compared with the conventional dedicated wirings. Household appliances like halogen tubes, washing machines, televisions, etc. can become prone to an unpredictable noise and interference in the transmission. Continuous plugging and unplugging of electronic devices makes power line characteristics vary constantly [7].
- Distance: Although PLC is very effective at the scale of a regular house; it is not suitable for extremely large buildings. Even those living in bigger-than-average houses are likely to benefit from using the faster 200Mbps adapters over the slower 85Mbps and 50Mbps varieties. Mansion dwellers are likely to need to install switches here and there to give the signal a boost [11].
- Security: Although significantly more secure than open wireless networks, PLC does still represent a certain degree of security risk. Certainly in any building, such as an apartment or semi-detached house, where there's a possibility of shared wiring with neighboring properties, the usual security measures should be taken, such as encryption and strong passwords. In addition, many PLC adapters are supplied with software designed to restrict the network to properly configured devices, and this should be used whenever the possibility of any risk is present [11].

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- Other interference: There have also been reports of PLC interfering with devices such as keyboards, computer mice and even fridges and other household appliances. Once again, modern adapters have incorporated technologies to overcome these problems. In particular, notching frequency filters have been developed to block out the specific frequencies that are known to cause problems. D-Link, in particular, has included programmable notch filters, so that any adapter that is causing interference difficulties can have the problem frequencies blocked [11].

Table 2 summarizes a comparison between the PLC and other technologies in terms of transportation, advantages, and disadvantages for broadband over power line (BPL), Digital Subscriber Line (DSL), cable modems, and the Integrated Services Digital Network (ISDN) [12 – 15].

**Table 2. Comparison between PLC and other Technologies**

<b>Comparatives</b>	<b>Broadband over Power Line (BPL)</b>	<b>Digital Subscriber Line (DSL)</b>	<b>Cable Modem</b>	<b>ISDN</b>
Transportation	Power transmission line	Dual phone line	Coax	Dual phone line
Advantages	<ul style="list-style-type: none"><li>– Good performance in constant contact</li><li>– Excellent geographic coverage</li></ul>	Good performance in constant contact	Excellent performance in constant contact	<ul style="list-style-type: none"><li>– Mature technology and impervious</li><li>– Available in almost every telephone centres</li></ul>
Disadvantages	<ul style="list-style-type: none"><li>– It still in the development stages</li><li>– Interference with radio signals</li></ul>	The effect distance on performance	<ul style="list-style-type: none"><li>– Limited geographic -al coverage</li><li>– Currently available in very limited places</li></ul>	<ul style="list-style-type: none"><li>– High cost</li><li>– Delivery upon request and not permanent</li></ul>

## **2.2 Introduction to IP-PBX**

An IP-PBX is a private branch exchange (telephone switching system within an enterprise) that switches calls between VoIP users on local lines while allowing all users to share a certain number of external phone lines. Customers can communicate with VoIP over IP networks [16]; so as to reduce communication costs can offset the cost of the customer. Customers can also directly contact the VoIP number of employees, so that the aims and objectives can be delivered directly to the customer. There will be no need for extra infrastructure for the telephone network and don't need to install the costly equipment's; we just need Analog Telephone Adapter (ATA) to provide this service [17, 18].

### **2.2.1 Features**

- Peer to peer phone calls.
- Peer to peer video.
- Private instant messaging.
- Voice mail.
- Interactive voice response.

### **2.2.2 Applications of IP-PBX**

- Education sector
- Business sector/ hotels
- Hospitals
- Corporate organization
- Banks

### **2.2.3 VoIP**

Voice over Internet Protocol is known as IP Phones. In general, VoIP is defined as a system that uses Internet to transmit voice packets from one place to other using IP protocol intermediaries [19]. VoIP protocol is used in VoIP transport in order that voice data can be sent properly [20]. For more details regarding VoIP technical issues, refer to [21].

SIP (Session Initiation Protocol) is a protocol multimedia issued by the group (Internet Engineering Task Force) incorporated in Multiparty Session Control Characterized SIP client-server. This means that the request is given by the client and is sent to the server. Then, the server processes the request and provides a response to the



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client. Request and response to the client request is called a SIP transaction [22].

SIP protocol is supported by a few protocols, such as Resource Reservation Protocol (RSVP) to make a reservation on the network, Real Time Protocol (RTP) and Real Time Control Protocol (RTCP) media for transmitting and know the quality of service, as well as media Session Description Protocol (SDP) to describe the session. By default, SIP uses User Datagram Protocol (UDP), but in some cases may also uses Transmission Control Protocol (TCP) as the transport protocol [23]. The components of SIP [24] are: User agent, User Agent Client (UAC), User Agent Server (UAS), Network server, Proxy server. SIP URLs are used in SIP networks are shaped like an email address user @ host where user can be any user name, phone number, or the name of the agency [25]. The SIP message consists of two parts, the request and the response [26]. The SIP request primitives are *Invite*, *Ack*, *Cancel*, and *Bye*, while the SIP response primitives are shown in Table 3 [24].

**Table 3. SIP Response Primitives**

Class Response	Type Response	Category Response
1xx	Informational	Provisional
2xx	Success	Final
3xx	Redirection	Final
4xx	Client error	Final
5xx	Server error	Final
6xx	Global error	Final

Response messages are divided into two categories, namely:

- Provisional response: is a response sent by the server to indicate the process is ongoing, but not end the call.
- Final response: Response was given that terminate SIP response code transaction SIP.

The VoIP network configuration of Phone via the Internet and Communication between IP-based devices are depicted in Figs. 1 and 2, respectively.



Fig. 1. Phone via the Internet

Fig. 2. Communication between IP-based devices

### 3 System Design

This section illustrates the transmitter and receiver block diagrams for modulates and demodulates transmitted and received data, respectively. Hardware and software requirements are described as well.

#### 3.1 Block Diagrams

The communication device used for the communication over the power lines is a modem, known as Power Line Modem. It works as both transmitter and receiver; it transmits and receives data over the power lines. The transmitter and receiver block diagrams are depicted in Figs. 3 and 4, respectively.

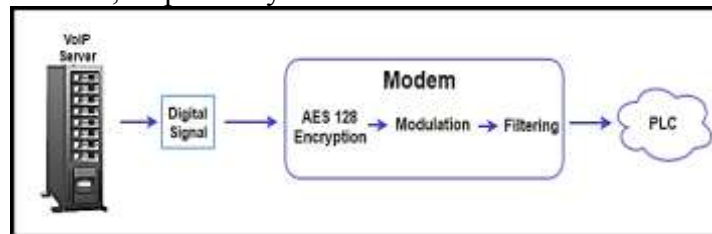


Fig. 3. Transmitter Block Diagram

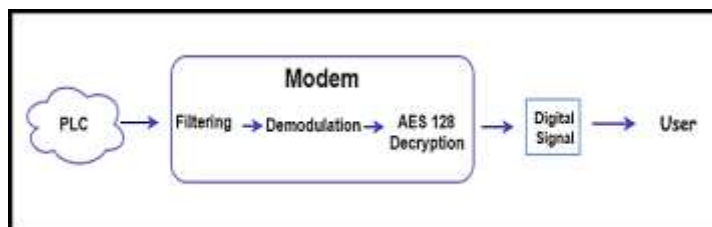


Fig. 4. Receiver block diagram

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A power line modem not only modulates the data to transmit it over the power lines but also demodulates the data it receives from the power lines by using modulation techniques. Modem uses high pass filter. Filter circuit is used to isolate the 50 Hz "low frequency" and enables high frequency "Frequency of data" to pass.






### 3.2 Requirements

To be used efficiently, the computer software needs certain hardware components to be present on a computer. In the following subsections, hardware and software requirements are precisely presented for the proposed system.

#### 3.2.1 Hardware Requirements

All hardware requirements are described precisely in Table 4.




**Table 4. Hardware Requirements**

#	Item	Description
1	 Power line adapter	<u>Model</u> : Toto link PL200 Kit <u>Features</u> : Allows users to turn their home. – It provides high speed data rate up to 200Mbps and coverage range up to 300 meters. – No need cables, using electrical wires.
2	 Access point	<u>Model</u> : The access point TP-Link TL-WR8401N. <u>Features</u> : Operates in 802.11 n/b/g standards. – It provides many functions and modes of work: AP, Repeater, Bridge (Point to Point, Point to Multipoint). – Enabling transmission speeds up to 300 Mb/s.
3	 Work station use for VoIP central	<u>Minimum requirements</u> : – Intel Dual Core E2160 1,8 GHz – Memory 1Gb / HDD 4Gb / Fast Ethernet Card
4	 Analog Telephone Adapter	<u>Model</u> : Cisco SPA112 2-Port Phone Adapter. <u>Features</u> : 2 FXS Port
5	 Analog Telephone	<u>Model</u> : Panasonic KX-TS500MXW An analog phone interprets sound as an electronic pulse. <u>Features</u> : – Electronic Handset Volume Control (6-Step) – 3-Step Ringer Selector (Off/Low/High)

### 3.2.2 Software Requirements

All software requirements are described precisely in Table 5.

**Table 5. Software Requirements**

#	Item	Description
1		<ul style="list-style-type: none"> <li>– VirtualBox is a powerful x86 and AMD64/Intel64 virtualization.</li> <li>– Available as Open Source Software under the terms of the GNU General Public License</li> <li>– VirtualBox runs on Windows, Linux, Macintosh, and Solaris hosts and supports a large number of guest operating systems.</li> </ul>
2		<ul style="list-style-type: none"> <li>– Asterisk was originally created as the engine for a PBX system</li> <li>– A PBX acts as the central switching system for phone calls.</li> <li>– Voice Messaging: once upon a time, voicemail was an optional add-on. Today, it's table stakes.</li> </ul>
3		<ul style="list-style-type: none"> <li>– Zoiper is a VoIP softphone that lets you make voice calls with your friends, family, colleagues and business partners. It's free, works on any major platform, &amp; includes a preconfigured option for registering your VoIP residential service.</li> <li>– Used the free edition.</li> </ul>

A comparison between Zoiper and other applications is summarized in Table 6.

**Table 6. Comparison between Zoiper and other Apps**

	<b>Zoiper</b>	<b>CSipSimple</b>	<b>3CX</b>
<b>Voice call</b>	Yes	Yes	Yes
<b>Video call</b>	Yes	No	No
<b>Chat</b>	Yes	Yes	No
<b>Transfer call</b>	No	Yes	Yes

### 4 System Configuration and Experimental Studies

System configuration mainly refers to the specification of a given computer system, from its hardware components to the software and various processes that are run within that system. It refers to what types and models of devices are installed and what specific software is being used to run the various parts of the computer system.

## 4.1 Building of System

This subsection describes how to configure the proposed system and what requirements are needed.

**4.1.1 Requirements:** *Virtual Box (CentOS), Asterisk PBX, PC (Windows 10 OS), Zoiper Mobile App, ATA, and Telephone.*

### 4.1.2 Working Steps

1. Install virtual Box (for virtualization operating system).
2. Install centos 5 and Asterisk PBX use virtual Box.
3. Centos need memory, hard disk and network to install as shown in Figs. 5, 6, and 7, respectively.



Fig. 5. Memory size



Fig. 6. Location and size of hard disk



Fig. 7. Select Network

4. Virtual Box allows centos to take them from the PC.
5. In installation CentOS gives IP address as depicted in Fig. 8 to operate the system. After installation, system is ready as shown in Fig. 9.

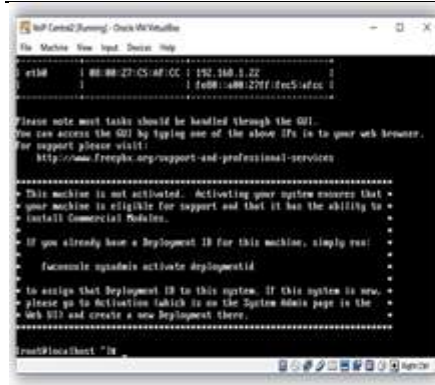


Fig. 8. IP address for BPX

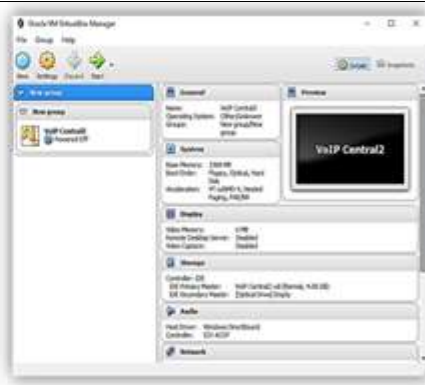


Fig. 9. IP address for BPX

6. To operate system use web browser and IP address as shown in Fig. 10.

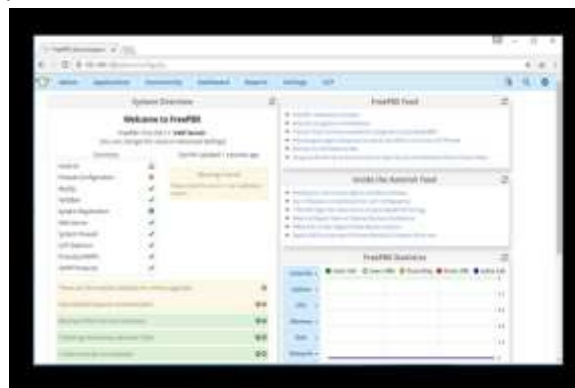


Fig. 10. BPX

7. Make extension and some settings from applications menu (From Applications > Extensions) as shown in Fig. 11.

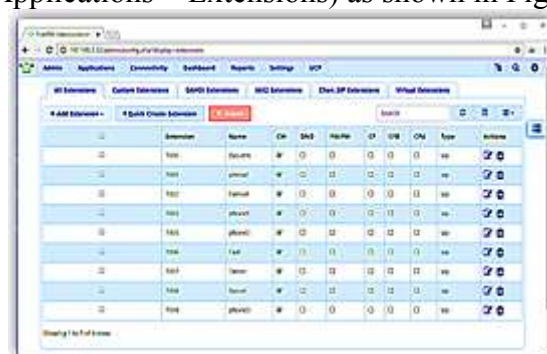


Fig. 11. Extensions

8. Configuration of ATA as depicted in Fig. 12. Enter the IP address of ATA in web browser, for example 192.168.1.10/dev.



IP Addressbook

Account name  
2000

Authentication

Host  
192.168.1.22:8080

Username  
2000

Password  
\*\*\*\*\*

Optional

Authentication user

Outbound proxy

Caller ID

Save Cancel

**Fig. 13. Zoiper configuration**

10. The following diagram (Fig. 14) shows the interconnections among system components.



## 4.2 Experiments

This section explains the details of five experiments that have been applied for implementing the proposed system. In the proposed system, there is no limitation on the calling duration between devices used in the five experiments. To ensure the quality of voice, two types of Asterisk codecs are used: u-Law and a-Law. VoIP codec selection and voice codec quality compression are discussed in detail by [27, 28].

### 4.2.1 Voice Call Mobile-to-Mobile

Requirements: Two mobiles, Zoiper app, PC for PBX, access point, valued extensions (7000, 7001), and PLC.

Working steps:

- Connect PC and mobiles at access point.
- Connect access point at PLC.
- Make sure that the mobiles are registered in the PBX.
- Mobile #1 signed with EXT. 7000 and mobile #2 signed with EXT. 7001.
- Open Zoiper app on mobile #1 and dial up the EXT. 7001.

The working steps of experiment (1) are depicted in Fig. 15.

Results:

- The call has been achieved.
- The sound is of a high quality.

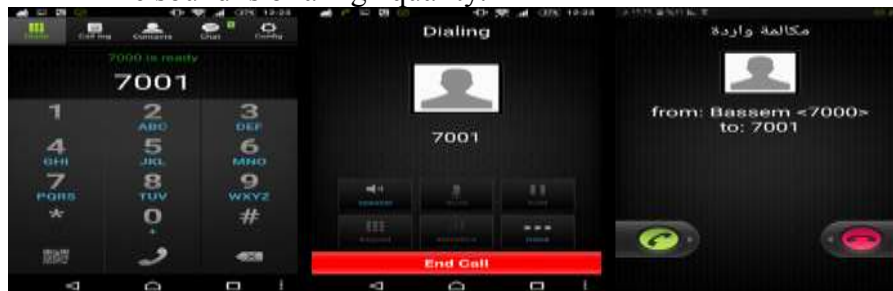


Fig. 15. Voice call between two mobiles. (a) Enter the destination number. (b) Dialing the destination. (c) Receive the call.

### 4.2.2 Video Call Mobile-to-Mobile

Requirements: Two mobiles, Zoiper app, PC for PBX, access point, valued extensions (7000, 7001), and PLC.

Working steps:

- Connect PC and mobiles at access point.
- Connect access point at PLC.



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- Make sure the mobiles are registered in the PBX.
- Mobile #1 signed with EXT. 7000 and mobile #2 signed with EXT. 7001.
- Open Zoiper app on mobile #1 and dial up the EXT. 7001.
- Select video option when call achieved.

The above-mentioned working steps of experiment (2) are depicted in Fig. 16.

##### Results:

- The video call has been achieved.
- The call has a high quality.
- Video codec H.264.
- Video capture options: size = high (640 x 480), frame per second (FPS) = 30, and bitrate = 420Kbits.



Fig. 16. Video call between two mobiles. (a) Video call from mobile #1. (b) Video call from mobile #2.

#### 4.2.3 Chatting Mobile-to-Mobile

Requirements: Two mobiles, Zoiper app, PC for PBX, access point, valued extensions (7000, 7001), and PLC.

##### Working steps:

- Connect PC and mobiles at access point.
- Connect access point at PLC.
- Make sure that the mobiles are registered in the PBX.
- Mobile #1 signed with EXT. 7000 and mobile #2 signed with EXT. 7001.
- Open Zoiper app on mobile #1.
- Select chat option in the app.
- Add a new chat with valued EXT.

Figure 17 depicts the working steps of experiment (3).

Results: The conversation was done without any problems.



Fig. 17. Chatting between two mobiles. (a) Enter the number of destination and message. (b) Receive the message in mobile #2 and reply. (c) Full chat

#### 4.2.4 Voice Call Mobile-to-Telephone

Requirements: Telephone, Mobile, Zoiper app, PC for PBX, access point, valued extensions (7000, 7003), ATA, and PLC.

Working steps:

- Connect PC, mobile, and ATA at access point.
- Connect telephone at ATA.
- Connect access point at PLC.
- Make sure that the mobile and ATA are registered in the PBX.
- Mobile #1 signed with EXT. 7000 and ATA signed with EXT. 7003.
- Open Zoiper app on mobile #1 and dial up the EXT. 7003.

The working steps of experiment (4) are shown in Fig. 18.



Fig. 18. Voice call between mobile and telephone. (a) Enter telephone number. (b) Dialing the telephone. (c) Receive the call. (d) Mobile receive call from telephone.

Results:

- The call has been achieved.
- The sound is of a high quality.
- Sound codec is a-Low.

#### 4.2.5 Voice Call Telephone-to-Telephone

Requirements: Two Telephones, Zoiper app, PC for PBX, access point, valued extensions (7003, 7005), two ATA, and PLC.

Working steps:

- Connect PC and ATA at access point.
- Connect telephone at ATA.
- Connect access point at PLC.
- Make sure that the two ATAs are registered in the PBX.
- ATA #1 signed with EXT. 7003 and ATA #2 signed with EXT. 7005.
- From telephone #1 dial up the EXT. 7005.

Figure 19 depicts the above-mentioned working steps of experiment (5).



**Fig. 19. Voice call between two Telephones. (a) Enter the number of telephone #2 and dialling. (b) Telephone #2 receive call from telephone #1. (c) Enter the number of telephone #1 and dialling. (d) Telephone #1 receive call from telephone #2**

#### 4.3 Network Statistics

The following network statistics as summarized in Table 7 are collected from the Zoiper application. The network statistics describe the characteristics of the transmitted data among different number of devices and calls. Those characteristics are codec, sampling rate, security, received packets, received bytes, received bytes payload, current received bitrate, average received bitrate, sent packets, sent bytes, and others.

To measure the reliability of the proposed system, the Current Packet Loss is calculated. As shown in Table 7, the Current Packet Loss results are equal to 0%, which means nothing lost during data transmission between sender and receiver.

**Table 7. Network Statistics**

Simultaneous Calls	Statistics (sender, receiver) During Calls	
One call (Call between two mobiles)	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 9942</i> <i>Received bytes: 2.03 MB</i> <i>Received bytes payload: 1.52 MB</i> <i>Current received bitrate: 130 kb/s</i> <i>Avg. received bitrate: 131 kb/s</i> <i>Sent packets: 6292</i> <i>Sent bytes: 1.28 MB</i> <i>Sent bytes payload: 983 KB</i> <i>Current sent bitrate: 83 kb/s</i> <i>Avg. sent bitrate: 83 kb/s</i> <i>Current packet loss: 0%</i> <i>Current received jitter: 21 ms</i>	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Total input packets: 1236</i> <i>Total input bytes: 258.0 KB</i> <i>Total input bytes payload: 193.0 KB</i> <i>Current Input bitrate: 83 kb/s</i> <i>Avg. Input bitrate: 82 kb/s</i> <i>Total output packets: 1257</i> <i>Total output bytes: 262.0 KB</i> <i>Total output bytes payload: 196.0 KB</i> <i>Current Output bitrate: 83 kb/s</i> <i>Avg. output bitrate: 84 kb/s</i> <i>Current Input loss permill: 0%</i> <i>Current input jitter: 0 ms</i>
One call (Call between mobile and telephone)	<i>Codec: a-Law</i> <i>Security: None</i> <i>Received bytes: 135.0 KB</i> <i>Current received bitrate: 83 kb/s</i> <i>Sent packets: 642</i> <i>Sent bytes payload: 100.0 KB</i> <i>Avg. sent bitrate: 82 kb/s</i> <i>Current received jitter: 0 ms</i>	<i>Sampling rate: 8 KHz</i> <i>Received packets: 649</i> <i>Received bytes payload: 101.0 KB</i> <i>Avg. received bitrate: 83 kb/s</i> <i>Sent bytes: 134.0 KB</i> <i>Current sent bitrate: 83 kb/s</i> <i>Current packet loss: 0%</i>
Two calls (The first Call between two mobiles. The second call between two telephones)	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 1183</i> <i>Received bytes: 247 KB</i> <i>Received bytes payload: 184 KB</i> <i>Current received bitrate: 83 kb/s</i> <i>Avg. received bitrate: 82 kb/s</i> <i>Sent packets: 1191</i> <i>Sent bytes: 248.0 KB</i> <i>Sent bytes payload: 186.0 KB</i> <i>Current sent bitrate: 83 kb/s</i> <i>Avg. sent bitrate: 82 kb/s</i> <i>Current packet loss: 0%</i> <i>Current received jitter: 0 ms</i>	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Total input packets: 1335</i> <i>Total input bytes: 278.0 KB</i> <i>Total input bytes payload: 208.0 KB</i> <i>Current Input bitrate: 84 kb/s</i> <i>Avg. Input bitrate: 82 kb/s</i> <i>Total output packets: 1328</i> <i>Total output bytes: 277.0 KB</i> <i>Total output bytes payload: 207.0 KB</i> <i>Current Output bitrate: 83 kb/s</i> <i>Avg. output bitrate: 82 kb/s</i> <i>Current Input loss permill: 0%</i> <i>Current input jitter: 0 ms</i>
Two calls (The first Call between two mobiles. The second call between two mobiles)	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 3008</i> <i>Received byte: 628.0 KB</i> <i>Received bytes payload : 470 KB</i> <i>Current received bitrate: 83 kb/s</i> <i>Avg. received bitrate: 83 kb/s</i> <i>Sent packets: 3044</i> <i>Sent bytes: 636.0 KB</i> <i>Sent bytes payload: 475.0 KB</i> <i>Current Sent bitrate: 84 kb/s</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 2634</i> <i>Total input bytes: 550.0 KB</i> <i>Total input bytes payload: 411.0 KB</i> <i>Current input bitrate: 83 kb/s</i> <i>Avg. input bitrate: 83 kb/s</i> <i>Total output packets: 2635</i> <i>Total output bytes: 550.0 KB</i> <i>Total output bytes payload: 411.0 KB</i> <i>Current output bitrate: 83kb/s</i>

### VoIP Using Power Line Communications

	<i>Avg. sent bitrate: 84 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 2 ms</i>	<i>Avg. output bitrate: 83kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter: 0 ms</i>
	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 7222</i> <i>Total input bytes: 1.47 MB</i> <i>Tot. input bytes payload: 1.10 MB</i> <i>Current input bitrate: 83 kb/s</i> <i>Avg. input bitrate: 81 kb/s</i> <i>Total output packets: 7386</i> <i>Total output bytes: 1.51 MB</i> <i>Total output bytes payload: 1.13 MB</i> <i>Current output bitrate: 84 kb/s</i> <i>Avg. output bitrate: 83 kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter: 0 ms</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 7291</i> <i>Received byte: 1.49 MB</i> <i>Received bytes payload : 1.11 MB</i> <i>Current received bitrate: 84 kb/s</i> <i>Avg. received bitrate: 82 kb/s</i> <i>Sent packets: 7375</i> <i>Sent bytes: 1.5 MB</i> <i>Sent bytes payload: 1.12 MB</i> <i>Current Sent bitrate: 84 kb/s</i> <i>Avg. sent bitrate: 83 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 19 ms</i>
Three calls (The first call Between two mobiles. The second call between two mobiles. The third call between two telephones)	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 871</i> <i>Received byte: 182.0 KB</i> <i>Received bytes payload : 136 KB</i> <i>Current received bitrate: 83 kb/s</i> <i>Avg. received bitrate: 85 kb/s</i> <i>Sent packets: 892</i> <i>Sent bytes: 186.0 KB</i> <i>Sent bytes payload: 139.0 KB</i> <i>Current Sent bitrate: 84 kb/s</i> <i>Avg. sent bitrate: 82 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 2 ms</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 550</i> <i>Total input bytes: 114.0 KB</i> <i>Total input bytes payload: 85.0 KB</i> <i>Current input bitrate: 83 kb/s</i> <i>Avg. input bitrate: 83 kb/s</i> <i>Total output packets: 535</i> <i>Total output bytes: 111.0 KB</i> <i>Total output bytes payload: 83.0 KB</i> <i>Current output bitrate: 83kb/s</i> <i>Avg. output bitrate: 81 kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter: 21 ms</i>
	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 5498</i> <i>Total input bytes: 1.12 MB</i> <i>Tot. input bytes payload: 859 KB</i> <i>Current input bitrate: 83 kb/s</i> <i>Avg. input bitrate: 82 kb/s</i> <i>Total output packets: 5586</i> <i>Total output bytes: 1.14 MB</i> <i>Total output bytes payload: 872 KB</i> <i>Current output bitrate: 82 kb/s</i> <i>Avg. output bitrate: 84 kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter: 0 ms</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 2641</i> <i>Received byte: 551 KB</i> <i>Received bytes payload : 412 KB</i> <i>Current received bitrate: 84 kb/s</i> <i>Avg. received bitrate: 81 kb/s</i> <i>Sent packets: 2671</i> <i>Sent bytes: 558 KB</i> <i>Sent bytes payload: 417 KB</i> <i>Current Sent bitrate: 83 kb/s</i> <i>Avg. sent bitrate: 82 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 23 ms</i>

Three calls (The first call between two mobiles. The second call between two mobiles. The third call between mobile and telephone)	<i>Codec: a-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets: 2379</i> <i>Received byte: 500.0 kb</i> <i>Received bytes payload: 374.0 kb</i> <i>Current received bitrate: 83 kb/s</i> <i>Avg. received bitrate: 70 kb/s</i> <i>Sent packets: 2892</i> <i>Sent bytes: 604.0 kb</i> <i>Sent bytes payload: 451.0 kb</i> <i>Current Sent bitrate: 83 kb/s</i> <i>Avg. sent bitrate: 83 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 0 ms</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 2485</i> <i>Total input bytes: 5190.0 kb</i> <i>Total input bytes payload: 388.0 kb</i> <i>Current input bitrate: 94 kb/s</i> <i>Avg. input bitrate: 76 kb/s</i> <i>Total output packets: 2486</i> <i>Total output bytes: 519.0 kb</i> <i>Total output bytes payload: 388 kb</i> <i>Current output bitrate: 83 kb/s</i> <i>Avg. output bitrate: 76 kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter : 0 ms</i>
	<i>Codec: u-Law</i> <i>Sampling rate: 8 kHz</i> <i>Security: None</i> <i>Total input packets: 1428</i> <i>Total input bytes: 298.0 kb</i> <i>Total input bytes payload: 223 kb</i> <i>Current input bitrate: 16 kb/s</i> <i>Avg. input bitrate: 72 kb/s</i> <i>Total output packets: 1686</i> <i>Total output bytes: 352.0 kb</i> <i>Total output bytes payload: 263kb</i> <i>Current output bitrate: 83 kb/s</i> <i>Avg. output bitrate: 72 kb/s</i> <i>Current input loss permill: 0%</i> <i>Current input jitter :0 ms</i>	<i>Codec: u-Law</i> <i>Sampling rate: 8 KHz</i> <i>Security: None</i> <i>Received packets:1737</i> <i>Received byte: 363.0 kb</i> <i>Received bytes payload: 271.0 kb</i> <i>Current received bitrate: 72 kb/s</i> <i>Avg. received bitrate: 82 kb/s</i> <i>Sent packets: 1771</i> <i>Sent bytes: 370.0 kb</i> <i>Sent bytes payload: 276.0 kb</i> <i>Current Sent bitrate: 83 kb/s</i> <i>Avg. sent bitrate: 84 kb/s</i> <i>Current packet loss: 0%</i> <i>Current jitter received: 21 ms</i>
	<i>Codec: u-Law</i> <i>Total incoming packets: 3499</i> <i>Total incoming bytes payload: 546.7 kb</i> <i>Avg. incoming bitrate: 55.2 kb/s</i> <i>Total outgoing bytes: 1126.0 kb</i> <i>Current outgoing bitrate: 83.6 kb/s</i> <i>Current incoming loss: 0.0%</i>	<i>Codec sample rate: 8 kHz</i> <i>Total incoming bytes: 731.2 kb</i> <i>Current incoming bitrate: 83.6 kb/s</i> <i>Total outgoing packets: 5388</i> <i>Total outgoing bytes payload: 841.9 kb</i> <i>Avg. outgoing bitrate: 83.4 kb/s</i> <i>Current incoming jitter: 0 ms</i>

## 5 Conclusion and Future Work

The aim of this study is to produce a communication system based on PLC with special qualities to solve the problems of multiple wires and high cost of centrals. This system can support transfer of data, voice call, video call and Internet service through electricity wires.

Power line adapter, access point, and work station are used for VoIP central, analog telephone adapter and analog telephone are used as hardware requirements, and Virtual Box, Asterisk and Zoiper are software applications. The five experiments show the robustness of the proposed system along with their associated network statistics.

The future work will be focused on how to connect the proposed system with the Public Switched Telephone Network (PSTN).

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