Identifying Disaster Area using Wireless Technology

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Abstract - *The new technology has the ability to transmit a voice over Internet protocol process networks by using an Asterisk PBX. The purpose of this research is to design and implement a telephony program that uses existing WIFI in p2p (Peer-to- Peer) or WLAN (Wireless Local Area Network) as a means of communication between mobile phones and existing intercom systems at no cost so that communication can also be established on disaster area using excisting WIFI. The asterisk software will use a correlation between current address books available in mobile phones to convert phone numbers into IP addresses. Voice over Internet Protocol (VoIP) is used for voice and data communication applications. VoIP applications have acceptable quality and relatively low cost when compared with landline and cellular communication. Because of the bandwidth efficiency and low costs, businesses are migrating from traditional copper-wire telephone systems to VoIP systems. Main objective of the paper is to build a common IP-PBX system for traditional analog telephone system, digital softphone and dedicated VOIP phones using Asterisk server. Communication is established among various nodes using the Soft-routing. In order to connect the wireless nodes to this network, a wireless access point is created, through which the smart phones are connected to this VOIP network.*

Key Words: VOIP, Asterisk, WIFI, IP-PBX, WLAN

1. INTRODUCTION

VoIP is a form of communication that allows you to make phone calls over a broadband internet connection instead of typical analog telephone lines. Basic VoIP access usually allows you to call others who are also receiving calls over the internet. Interconnected VoIP services also allow you to make and receive calls to and from traditional landline numbers, usually for a service fee. [8] Some VoIP services require a computer or a dedicated VoIP phone, while others allow you to use your landline phone to place VoIP calls through a special adapter. The operating system with Asterisk is installed in Raspberry Pi. The Raspberry Pi is a series of credit card–sized single-board computers developed in the United Kingdom by the Raspberry Pi Foundation with the intention of promoting the teaching of basic computer science in schools and developing countries. Asterisk supports audio protocols such as SIP which is Session Initiation Protocol used for the audio communication. This package consists of several features such as Voicemail, Call Waiting, Caller ID, and Call Transfer and so on.

In many years ago an old telephone system was improved to a new substitute known as Private Branch Exchange (PBX). PBX system performs communication tasks such as inbound calls and outbound calls. VoIP technology was developed in 1995. Some VoIP services need only a regular phone connection, while others allow you to make telephone calls using an Internet connection instead. Some VoIP services may allow you only to call other people using the same service, but others may allow you to call any telephone number - including local, long distance, wireless and international numbers. The support of telephony services over mobile phone has been used everywhere using technology such as GSM (Global System for Mobile) and 3rd Generation mobile telecommunication 3G , but at high cost. On the other hand, IP telephony try to reduce the cost for supporting this service over mobile phone, but it is facing difficulties since the same feature is supported on desktop and laptop at lower complexity. The challenge is to provide the same service over mobile phone at no cost, as it has been described in this project. Two approaches are suggested in this paper to meet the objective of having free telephony services over mobile phones. These are the use of WIFI technology over AP (Access Point) and WIFI over p2p (Peer-to-Peer).

1.1 Asterisk

Asterisk is a software implementation of a telephone private branch exchange (PBX); it allows attached telephones to make calls to one another, and to connect to other telephone services, such as the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services. Its name comes from the asterisk symbol "*". The Asterisk software includes many features available in proprietary PBX systems: voice mail, conference calling, interactive voice response (phone menus), and automatic call distribution. Users can create new functionality by writing dial plan scripts in several of Asterisk's own extensions languages, by adding custom loadable modules written in C, or by implementing Asterisk Gateway Interface(AGI) programs using any programming language capable of communicating via network TCP sockets.

Asterisk supports several standard voice over IP protocols, including the Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP), and H.323. Asterisk supports most SIP telephones, acting both as registrar and back-to-back user agent, and can serve as a gateway between IP phones and the public switched telephone network (PSTN) via T- or E-carrier interfaces or analog FXO cards. The Inter-Asterisk eXchange (IAX) protocol, RFC 5456, native to Asterisk, provides efficient trunking of calls among Asterisk PBXes, in addition to distributing some configuration logic. Many VoIP service providers support it for call completion into the PSTN, often because they themselves have deployed Asterisk or offer it as a hosted application. Some telephones also support the IAX protocol. [7]

By supporting a variety of traditional and VoIP telephony services, Asterisk allows deployers to build telephone systems, or migrate existing systems to new technologies. Some sites are using Asterisk to replace proprietary PBXes, others provide additional features, such as voice mail or voice response menus, or virtual call shops, or to reduce cost by carrying long-distance calls over the Internet (toll bypass). Various codecs are used for compression and decompress the audio and video signals and are listed in table -1 and table -2 respectively.

Table -1: Audio Codecs used in VoIP

VOIP-SIP.ORG Codec and Bit Rate	Sample Size (Bytes)	Sample rate (ms)	MOS Quality	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.7	20 Bytes	20 ms	50	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms	—	60 Bytes	20 ms	50	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	3.61	60 Bytes	30 ms	33.3	31.5 Kbps
G.722 (64 Kbps)	80 Bytes	10 ms	4.13	160 Bytes	20 ms	50	87.2 Kbps
iLBC (15.2Kbps)	38 Bytes	20 ms	4.14	38 Bytes	20 ms	50	38.4Kbps
iLBC (13.33Kbps)	50 Bytes	30 ms	—	50 Bytes	30 ms	33.3	28.8 Kbps

Table -2: Video Codecs used in VoIP

Name	Config Value	Capability: (T)ranscoding (P)assthrough	Format Module	Distributed w/ Asterisk
JPEG	jpeg	P	format_jpeg	YES
H.261	h261	P	N/A	YES
H.263	h263	P	format_h263	YES
H.263+	h263p	P	format_h263	YES
H.264	h264	P	format_h264	YES
VP8	vp8	P	N/A	YES

1.2 TDM TO PCM Interface

A TDM to PCM Interface, also called computer– telephone integration or CTI, is a common name for any technology that allows interactions on a telephone and a computer to be integrated or coordinated. The term is predominantly used to describe desktop-based interaction for helping users be more efficient, though it can also refer to server-based functionality such as automatic call routing. CTI enables control of the phone through the PC as shown in figure 1. This allows computer software such as email programs to connect you to the telephone system with just the click of a mouse.

To operate Computer Telephony Integration, the computer needs the appropriate software to control the system phone or communicate with the telephone system. Many PBX manufacturers offer such software solutions for their devices. For communication between the telephone system and the computer software, there are standardized protocols such as CSTA (Computer Supported Telecommunications Applications). This protocol defines the type and format of the data to be transmitted without specifying the actual transport layer, thereby allowing CTI solutions to be implemented by many more manufacturers.

Fig -1: Computer Telephony Integeration

1.3 IP Telephone

IP telephony (Internet Protocol telephony) is a general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). Using the Internet, calls travel as packets of data on shared lines, avoiding the tolls of the PSTN. The challenge in IP telephony is to deliver the voice, fax, or video packets in a dependable flow to the user. Much of IP telephony focuses on that challenge. Currently, unlike traditional phone service, IP telephony service is relatively unregulated by government. In the United States, the Federal Communications Commission (FCC) regulates phone-to- phone connections, but says they do not plan to regulate connections between a phone user and an IP telephony service provider.

VoIP is an organized effort to standardize IP telephony. IP telephony is an important part of the convergence of computers, telephones, and television into a single integrated information environment. Also see another general term, computer-telephony integration (CTI), which describes technologies for using computers to manage telephone calls. VoIP phones can be simple software-based softphones as in figure 3 or purpose-built hardware devices as in figure 2, that appear much like an ordinary telephone or a cordless phone. Traditional PSTN phones are used as VoIP phones with analog telephone adapters (ATA).

Fig -2: IP Phone – Audio Codecs 320HD

Fig -3: Zoiper Softphone for PC and Android

2. PROPOSED WORK

In the early years of telephone lines, calls went through public switchboards, where operators manually directed them to the correct receivers and are known as Electronic Private Branch Exchange (EPBX) as shown in figure 4. As computers developed, an update to PBX appeared which enables the PBX to automatically route the calls to the receiver. This system is known as Electronic Private Automatic Branch Exchange (EPABX) and is shown in figure 4.

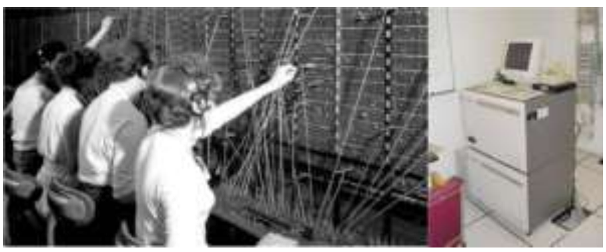

Fig -4: PBX and EPABX system

Finally, the internet came, and with it, developers created the ability to channel calls through the internet's data network and is known as Voice over Internet Protocol (VoIP). This sparked a business telephone revolution that would continue to develop through the 21st century and into the digital revolution. [8].Instead of physically connecting to the PBX with copper wiring, phones connect to the PBX over an office's Local Area Network (LAN), often leveraging the same Internet connectivity that your office computers do and the general schematic of VoIP is shown in figure 5. Generally there are two types of IP-PBX systems based on the location of server. One of them is Hosted or cloud-based PBX systems, which are the virtual equivalents of the physical equipment used by traditional business phone systems. Another one is On-Site IP-PBX, which has the server locally.

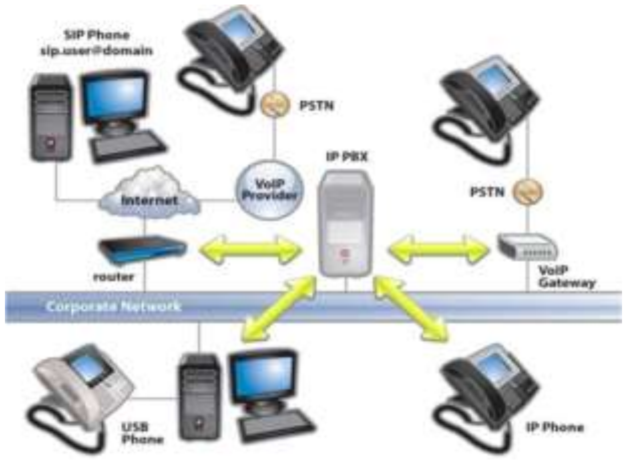

Fig -5: Basic VoIP network Layout

2.1 Problem Statement

The purpose of this research is to design and implement an intercom system that uses existing LAN (Local Area Network) or WLAN (Wireless Local Area Network) as a means of communication between mobile phones and existing intercom systems at no cost. This system will allow user to make voice and video conversation. In addition to that conference calls can also be made at free of cost. Different security services relevant for VoIP are presented and we argue that end-to-end authentication and encryption should be provided by default.

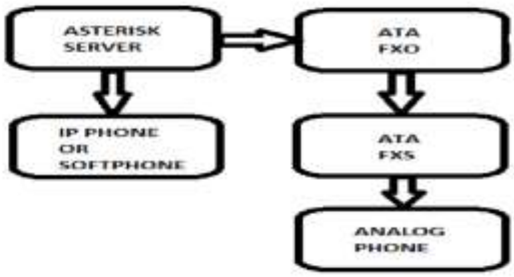

Fig -6: Block Diagram of Proposed IP-PBX system

VoIP applications have acceptable quality and relatively low cost when compared with landline and cellular Communication. Because of the bandwidth efficiency and low costs, businesses are migrating from traditional copper- wire telephone systems to VoIP systems. Main objective of the project is to build a common IP-PBX system for traditional analog telephone system, digital softphone and dedicated VOIP phones using Asterisk server as shown in figure 6. Communication is established among various nodes using the

Soft-routing. In order to connect the wireless nodes to this network, a wireless access point is created, through which the smart phones are connected to this VOIP network.

2.2 Advantages of Proposed System

i) Proposed system is highly reliable, as it is operated in LAN and WLAN. In addition to that, it can be operated using small battery as the power requirement is very less when compared to the existing system.
ii) Better sound quality can be obtained by using the CODECs for lossless compression. And also better video quality can also be achieved in video calling by using pass- through CODECs.
iii) Since the raspberry pi is fully customizable and reprogrammable, the user or organization can have the complete access and control.
iv) It is not mandatory to have the IP-phones. Instead of that, smartphones, PCs, and LAPTOPs can be used as IP-Phones using the softphone software.

3. HARDWARE DESCRIPTION

Hardware components of the proposed work explained briefly along with their specifications as below:

3.1 RASPBERRY PI 3 - MODEL B

The Raspberry Pi 3 Model B is the third generation Raspberry Pi as shown in figure 7. This powerful credit-card sized single board computer can be used for many applications and supersedes the original Raspberry Pi Model B+ and Raspberry Pi 2 Model B. [8] Additionally it adds wireless LAN & Bluetooth connectivity making it the ideal solution for powerful connected designs. Raspberry pi 3 specifications are as below:

i) 1.2GHz Quad-Core ARM Cortex-A53
ii) Broadcom BCM2387 chipset
iii) 802.11 bgn Wireless LAN and Bluetooth 4.1 (Bluetooth Classic and LE)
iv) 1GB RAM
v) 64 Bit CPU
vi) 10/100 BaseT Ethernet socket

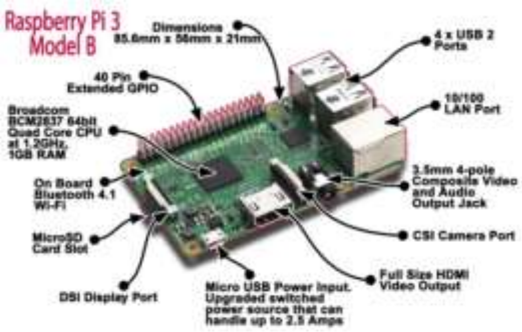

Fig -7: Raspberry pi 3 – Model B

3.3 WIFI Router

A wireless router is a device that performs the functions of a router and also includes the functions of a wireless access point. It is used to provide access to the Internet or a private computer network. Depending on the manufacturer and model, it can function in a wired local area network, in a wireless-only LAN, or in a mixed wired and wireless network.

Table -3: 802.11 physical layer standards

802.11 protocol	Release DATE	Frequency (GHz)	Bandwidth (MHz)
802.11-1997	Jun-97	2.4	22
a	Sep-99	5	20
		3.7[A]	
b	Sep-99	2.4	22
g	Jun-03	2.4	20
n	Oct-09	2.4/5	20
			40
ac	Dec-13	5	20
			40
			80
			160
ad	Dec-12	60	2,160
ah	Dec-16	0.9	
aj	Est. Jul 2017	45/60	
ax	Est. Dec 2018	2.4/5	
ay	Est. Nov 2019	60	8000
az	Est. Mar 2021	60	

3.3. IP-Phone

A VoIP phone or IP phone uses voice over IP technologies for placing and transmitting telephone calls over an IP network, such as the Internet, instead of the traditional public switched telephone network (PSTN).Digital IP-based telephone service uses control protocols such as the Session Initiation Protocol (SIP), Skinny Client Control Protocol (SCCP) or various other proprietary protocol. AudioCodes 320HD IP Phone is based on AudioCodes proprietary High Definition (HD) voice technology, providing clarity and a better audio experience in Voice- over-IP (VoIP) calls, which is shown in figure 8.

The 320HD IP Phone is a fully-featured telephone that provides voice communication over an IP network, allowing you to place and receive phone calls, put calls on hold, transfer calls, make conference calls, and so on

Fig -8: Audio Codec 320HD IP-Phone

4. SOFTWARE DESCRIPTION

4.1 ASTERISK WITH FREEPBX

Asterisk is a software implementation of a telephone private branch exchange (PBX); it allows attached telephones to make calls to one another, and to connect to other telephone services, such as the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services. Its name comes from the asterisk symbol "*".

FreePBX is a web-based open source graphical user interface (GUI) that manages Asterisk, a voice over IP and telephony server. FreePBX is licensed under the GNU General Public License version 3. It is a component of the FreePBX Distro, which is an independently maintained Linux system derived from the source code of the CentOS distribution, having Asterisk pre- installed. Figure 9 shows the FreePBX administrative panel for asterisk.

Fig -9: FREEPBX control panel

4.2 LINUX OS IN ATA

SIP (Session Initiation Protocol) is identified by IETF, use to create, change and release the session of one or several participants. These sessions are like Internet, Multimedia Conference, IP phone or Multimedia Distribution. The session participants could communicate by Multicast, Mesh Unicast, or mixture of the two. Figure 10 shows the configuration of the analog phone using the SIP protocol in the gateway web server.

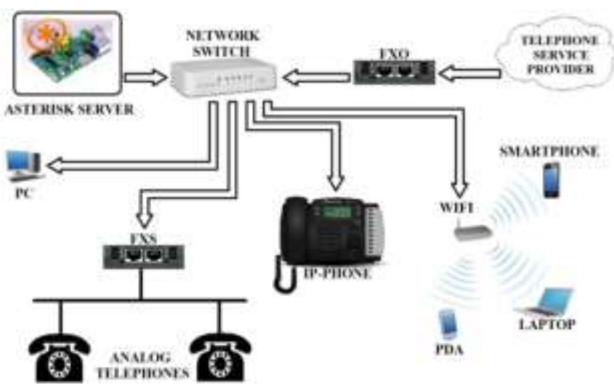

Fig -10: Easy Phone HTTP webserver

5.RESULT AND DISCUSSION

The proposed IP-PBX using the raspberry-pi is tested by running the system for continuously for 6 hours and traced the activity of the system for malfunction. In the same manner the system is tested for 3 days. But the system operates perfectly on the three days. During those testing, various wired and wireless nodes are connected to the server and various calls are made simultaneously for testing and the functionality doesn't change with time.

$$T=N/fs=64/(20MHz)=3.2\mu s$$

From the time period the frequency is 312.5 KHz. Ion Here 802.11n 20 MHz uses 64 subcarriers. Among them 48 subcarriers are data, 4 are pilots, 11 are zero-padded on either side of the spectrum, and 1 is the null DC subcarrier. The zero-padding also reduces interference between adjacent frequency channels.

The connection Layout of Proposed IP-PBX system is shown below

Fig 11: Connection Layout of Proposed IP-PBX System

By using the configuration shown in figure 11, the video call is established between the PC and the smartphone by using the zoiper softphones. When the data rate is below 256Kbps, then there is no jitter in the call. Figure 12 shows the actual hardware setup.

Fig -12: Hardware set up of Proposed IP-PBX system

6. CONCLUSION

The raspberry pi based intercom system can be installed easily along with the existing intercom or LAN or WLAN systems. This type conversion enables the user to use the smartphone as a node of intercom system with free of cost. Due to low cost, small and medium scale industries and college and school campus can afford this system with less investment. Then by installing this system in Public Wi-Fi, free calling facility can be provided to the Public.

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