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Low Cost Video Conferencing over VOIP Using TDM to PCM Interface

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ABSTRACT

As the cutting edge phone systems started to come to fruition, numerous privately owned businesses choose to actualize their own telephone utility. So they could deal with calls interior to the association. Voice communication over versatile is as of now upheld at a cost utilizing specialist co-op, for example, GSM, or utilizing IP specialist co-op at less expensive cost. The reason for this examination is to plan and actualize a communication program that utilizations existing WIFI in p2p (Peer-to-Peer) or WLAN (Wireless Local Area Network) as a methods for correspondence between cell phones and existing radio frameworks at no cost. The reference mark programming will utilize a relationship between present address books accessible in cell phones to change over telephone numbers into IP addresses. The framework will enable client to make voice and video discussion. The present framework will take into account one call for every association, and no call pausing. Notwithstanding that telephone calls can likewise be made at free of cost. Distinctive security administrations applicable for VoIP are displayed and we contend that conclusion to-end validation and encryption ought to be given of course. Voice over Internet Protocol (VoIP) is utilized for voice and information correspondence applications. VoIP applications have adequate quality and generally minimal effort when contrasted and landline and cell correspondence. On account of the transfer speed effectiveness and low costs, organizations are moving from conventional copper-wire phone frameworks to VoIP frameworks. Fundamental goal of the paper is to construct a typical IP-PBX framework for customary simple phone framework, computerized softphone and devoted VOIP telephones utilizing Asterisk server. Correspondence is built up among different hubs utilizing the Soft-steering. Keeping in mind the end goal to associate the remote hubs to this system, a remote access point is made, through which the advanced mobile phones are associated with this VOIP arrangement.

1. INTRODUCTION

VoIP is a type of correspondence that enables you to make telephone brings over a broadband web association rather than run of the mill simple phone lines. Essential VoIP get to for the most part enables you to call other people who are likewise accepting brings over the web. Interconnected VoIP benefits additionally enable you to make and get calls to and from customary landline numbers, as a rule for an administration charge. [8] Some VoIP administrations require a PC or a committed VoIP telephone, while others enable you to utilize your landline telephone to put VoIP calls through an uncommon connector. The working framework with Asterisk is introduced in Raspberry Pi. Indicator bolsters sound conventions, for example, SIP which is Session Initiation Protocol utilized for the sound correspondence. This bundle comprises of a few highlights, for example, Voicemail, Call Waiting, Caller ID, and Call Transfer et cetera.

The test is to give a similar administration over cell phone at no cost, as it has been depicted in this venture. Two methodologies are proposed in this paper to meet the target of having free communication benefits over cell phones. These are the utilization of WIFI innovation over AP (Access Point) and WIFI over p2p (Peer-to-Peer).

1.1. ASTERISK

Reference bullet is a product usage of a phone private branch trade (PBX); it enables joined phones to make calls to interface with other telephone utilities, for example, the Public Switched Telephone Network (PSTN) and Voice over Internet Protocol (VoIP) administrations. Its name originates from the reference bullet image "*". The Asterisk programming incorporates numerous highlights accessible in restrictive PBX frameworks: voice message, gathering calling, intuitive voice reaction (telephone menus), and programmed call appropriation. Clients can make new usefulness by composing dial design contents in Asterisk's very own few expansions dialects, by including custom loadable modules written in C, or by executing Asterisk Gateway Interface (AGI) programs utilizing any programming dialect equipped for conveying by means of system TCP attachments.

Reference bullet underpins a few standard voice over IP conventions, including the Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP), and H.323. Indicator bolsters most SIP phones, acting both as enlistment center and consecutive client specialist, and can fill in as a passage between IP telephones and the Public Switched Telephone Network (PSTN) by means of T-or E-bearer interfaces or simple FXO cards. Numerous VoIP specialist organizations bolster it for call fulfillment into the PSTN, frequently in light of the fact that they themselves have sent Asterisk or offer it as a facilitated application. A few phones likewise bolster the IAX convention. [7]

Different codecs are utilized for pressure and decompress the sound and video flags and are recorded in table I and table II individually.

Table I: 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	4.3	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 II: 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

Notwithstanding VoIP conventions, Asterisk bolsters customary circuit-exchanging conventions, for example, ISDN and SS7. This requires fitting equipment interface cards, advertised by outsider merchants. Every convention requires the establishment of programming modules.

1.2. TDM to PCM Interface

A TDM to PCM Interface, additionally called computer– phone mix or CTI, is a typical name for any innovation that permits connections on a phone and a PC to be incorporated or composed. The term is dominatingly used to depict work area based cooperation for helping clients be more effective, however it can likewise allude to server-based usefulness, for example, programmed call directing. CTI empowers control of the telephone through the PC as appeared in figure 1.

Fig. 1.Computer Telephony Integration (CTI)

1.3. IP Telephone

IP communication (Internet Protocol communication) is a general term for the advancements that utilization the Internet Protocol's bundle changed associations with trade voice, fax, and different types of data that have customarily been persisted the devoted circuit-exchanged associations of the Public Switched Telephone Network (PSTN). Utilizing the Internet, calls go as parcels of information on shared lines, keeping away from the tolls of the PSTN. VoIP is a sorted out push to institutionalize IP communication. IP communication is an imperative piece of the merging of PCs, phones, and TV into a solitary coordinated data condition. VoIP telephones can be straightforward programming based softphones as in figure 3 or reason manufactured equipment gadgets as in figure 2 that seem much like a standard phone or a cordless telephone.

Fig. 2. IP Phone – Audio Codecs 320HD

Fig. 3.Zoiper Softphone for PC and Android

1.4. Wireless Integration

A Wireless Network Interface Controller (WNIC) is utilized for remote system joining which associates with a remote radio-based PC organize, instead of a wired system, for example, Token Ring or Ethernet. A WNIC, much the same as different NICs, chips away at the Layer 1 and Layer 2 of the OSI Model. This card utilizes a radio wire to convey by means of microwave radiation with different speeds as appeared in figure 4 and range as in figure 5. A WNIC in a work station is generally associated utilizing the PCI transport. [3]

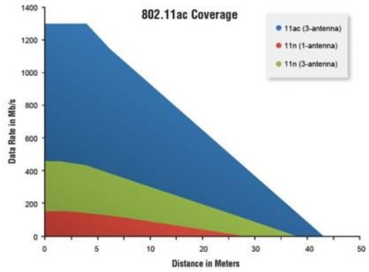

Fig. 4.Data rate vs. Distance graph for 802.11acn

Fig. 5. Dual Band 802.11acn WNIC

2. PROPOSED WORK

In the early years of phone lines, calls experienced open switchboards, where administrators physically guided them to the right beneficiaries and are known as Electronic Private Branch Exchange (EPBX) as appeared in figure 6(left). As PCs built up, a refresh to PBX showed up this empowers the PBX to naturally course the calls to the beneficiary. This framework is known as Electronic Private Automatic Branch Exchange (EPABX) and is appeared in figure 6(right).

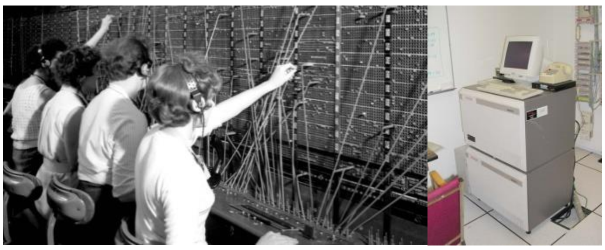

Fig. 6. PBX and EPABX system

In traditional system, the phones are connected using the copper wire system. In this system, the phones are connected using the available wifi facility inside an organization and the general schematic of VoIP is shown in figure 7. [8]

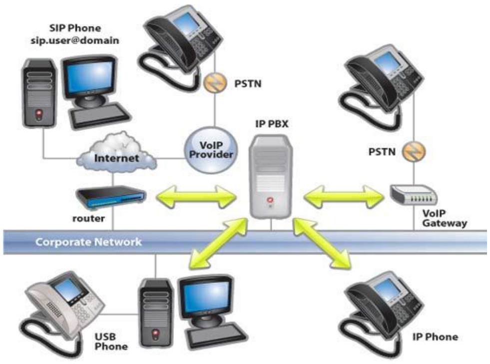

Fig. 7. Basic VoIP network Layout

2.1. Problem Statement

The reason for this examination is to outline and actualize a radio framework that utilizations existing LAN (Local Area Network) or WLAN (Wireless Local Area Network) as a methods for correspondence between cell phones and existing radio frameworks at no cost. This framework will enable client to make voice and video discussion. Notwithstanding that phone calls can likewise be made at free of cost. Distinctive security administrations important for VoIP are exhibited and we contend that conclusion to-end confirmation and encryption ought to be given as a matter of course.

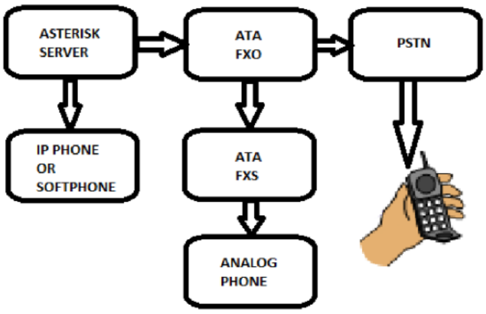

Fig. 8. Block Diagram of Proposed IP-PBX system

Principle goal of the undertaking is to manufacture a typical IP-PBX framework for conventional simple phone framework, computerized softphone and devoted VOIP telephones utilizing Asterisk server as appeared in figure 8. Correspondence is set up among different hubs utilizing the Soft-directing. With a specific end goal to interface the remote hubs to this system, a remote access point is made, through which the advanced mobile phones are associated with this VOIP arrange.

2.2. Advantages of Proposed System

1. Proposed framework is very solid, as it is worked in LAN and WLAN. Notwithstanding that, it can be worked utilizing little battery as the power necessity is less when contrasted with the current framework.
2. Better sound quality can be gotten by utilizing the CODECs for lossless pressure. And furthermore better video

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3. Since the raspberry pi is completely adjustable and reprogrammable, the client or association can have the total access and control.
4. It isn't compulsory to have the IP-telephones. Rather than that, cell phones, PCs, and LAPTOPs can be utilized as IP-Phones utilizing the softphone programming.

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 era Raspberry Pi as appeared in figure 9. This intense Master card estimated single board PC can be utilized for some applications and supersedes the first Raspberry Pi Model B+ and Raspberry Pi 2 Model B. [8] Additionally it includes remote LAN and Bluetooth availability making it the perfect answer for effective associated plans.

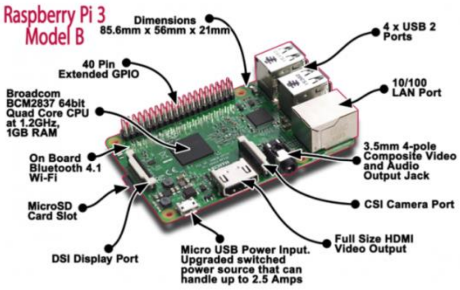

Fig. 9. Raspberry pi 3 – Model B

3.2. HYBERTONE HT842T FXO ATA

FXS Series Gateway is the execution Gateway, which created by HYBERTONE Co. FXS Series portals worked in H.323 and SIP conventions. Exceedingly solid line recognition keeps the line hanging passing in the biggest tent. Helpful and commonsense capacity of a broken system escape, when disengage the system, or VOIP logon falls flat, FXS to say the chance to bounce specifically to the outside PSTN. It can forward outside Caller ID Number under the SIP convention, which is an imperative capacity of PBX application. Super Echo Cancelation Algorithm and Balanced Circuit influence line to resound least. Low value, Stable and execution are pronoun of the FXS Series items; it is the main selection of PXS Manufacturers, Call Center and System Integrators. Figure 10 demonstrates the HT842T simple connector.

Fig. 10. Analog Telephone adapter

3.3. WIFI Router

A remote switch is a gadget that plays out the elements of a switch and furthermore incorporates the elements of a remote access point. Double Band Wireless Routers have the capacity to transmit on the 5 GHz and 2.4 GHz remote band. 2.4GHz however more far reaching in utilization (each of the 802.11b and g gadgets keep running on 2.4GHz just) has just 3 non-covering channels for transmission, which are swarmed because of a ton of meddling gadgets other Wi-Fi get to focuses, microwave broilers, cordless telephones, Bluetooth gadgets, child screens, and so forth all make for a boisterous domain which increment obstruction and corrupt the execution. Wi-Fi or WiFi is an innovation for remote neighborhood with gadgets in light of the IEEE 802.11 principles. Table III shows different 802.11 principles alongside their transmission capacity.

Table III : 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 3.7[A]	20
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	

TP-Link's Archer C50 accompanies the cutting edge Wi-Fi standard – 802.11ac as appeared in figure 11, 3 times speedier than remote N speeds and conveying a consolidated remote information exchange rate of up to 1.2Gbps. With 300Mbps over the 2.4GHz band and 867Mbps remote speeds over the perfectly clear 5GHz band, the Archer C50 is the predominant decision for consistent HD gushing, web based gaming and other transfer speed serious errands. Synchronous Dual Band offers you the adaptability of two devoted systems – with different circumstances the data transmission for your requirements.

Fig. 11. TP-link Archer C50 – 50 user Wi-Fi router with Dual band

3.4. IP-Phone

A VoIP telephone or IP telephone utilizes voice over IP advancements for putting and transmitting phone brings over an IP arrange, for example, the Internet, rather than the customary open exchanged phone organize (PSTN).

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The 320HD IP Phone is a completely included phone that gives voice correspondence over an IP arrange, enabling you to put and get telephone calls, put approaches hold, exchange calls, make phone calls, et cetera.

Fig. 12. AudioCodec 320HD IP-Phone

4. SOFTWARE DESCRIPTION

4.1. ASTERISK WITH FREEPBX

FreePBX is an electronic open source graphical UI (GUI) that oversees Asterisk, a voice over IP and communication server. FreePBX is authorized under the GNU General Public License variant 3. It is a segment of the FreePBXDistro, which is an autonomously kept up Linux framework got from the source code of the CentOS dispersion, having Asterisk pre-introduced. Figure 13 demonstrates the FreePBX managerial board for reference mark.

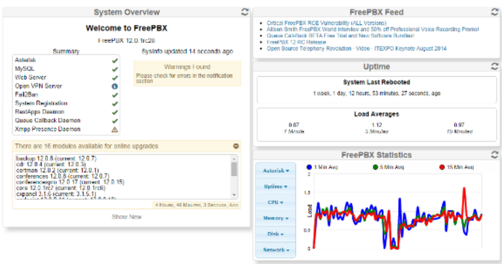

Fig. 13. FREEPBX control panel

4.2. LINUX OS in ATA

Figure 14 shows the configuration of the analog phone using the SIP protocol in the gateway web server.

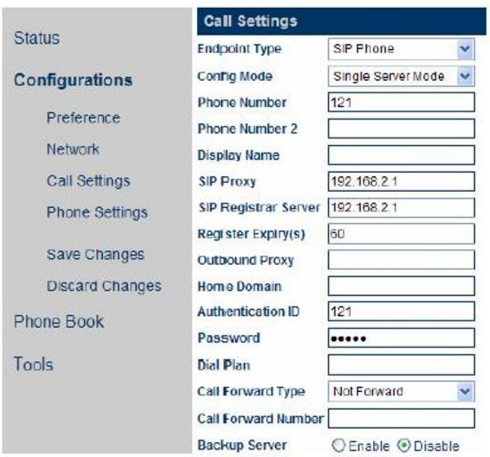

Fig. 14. Easy Phone HTTP webserver

4.3. IP-PHONE CONFIGURATION

In order to connect the IP-Phone with this asterisk, it has to be configured with an IP-address along with the phone extension. This IP-Phone can be configured using LCD and keypad on the Phone or by using the configuration webpage. Figure 15 shows the configuration by LCD and figure 16 shows the configuration by web page.

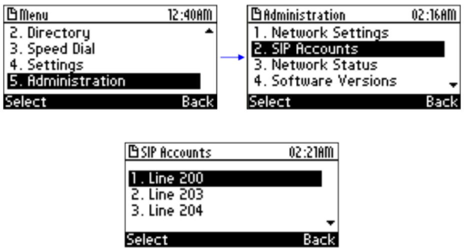

Fig. 15. IP-Phone LCD-based Management

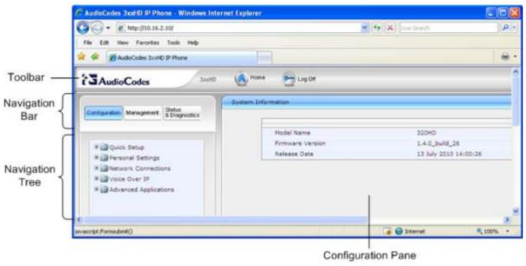

Fig. 16. IP-Phone WEB-based Management

4.4. OZEKI VoIP Softphone

A delicate telephone is a product program for making phone brings over the Internet utilizing a universally useful PC, instead of utilizing committed equipment. A softphone enables you to make and get VoIP telephone calls from and by PCs, workstations, or other registering gadgets. Generally, the interface looks like a telephone with catches speaking to the keys, which you squeeze utilizing the mouse to dial as appeared in figure 17. For talking and tuning in, a headset and amplifier do the trick.

Fig. 17. Ozeki VoIP SIP Softphone

A run of the mill softphone work with Ozeki SDK is appeared in figure 18.

Fig. 18. Softphone build with Ozeki SDK

5. RESULT AND DISCUSSION

The proposed IP-PBX utilizing the raspberry-pi is tried by running the framework for ceaselessly for 6 hours and followed the movement of the framework for glitch. In a similar way the framework is tried for 3 days. While testing the video calls, the most extreme rate that the framework can endure is 324kb/s. This reasons the rate of information relies upon the Wi-Fi dongle utilized, which is 802.11n and has the transmission capacity of 20MHz in the band of 2.4GHz. This conclusion likewise fulfills with the Sampling hypothesis as beneath.

T=N/fs=64/(20MHz)=3.2µs

From the day and age the recurrence is 312.5 KHz. Here 802.11n 20 MHz utilizes 64 subcarriers. Among them 48 subcarriers are information, 4 are pilots, 11 are zero-cushioned on either side of the range, and 1 is the invalid DC subcarrier. The zero-cushioning additionally diminishes impedance between nearby recurrence channels.

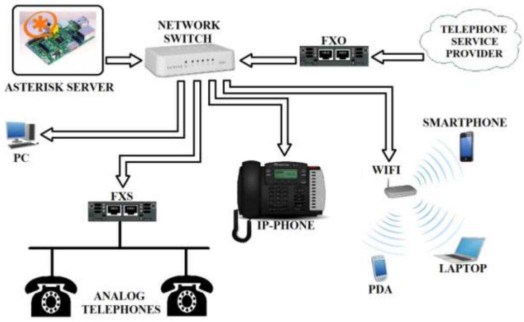

Fig. 19. Connection layout of proposedIP-PBX system

By utilizing the arrangement appeared in figure 19, the video call is set up between the PC and the cell phone by utilizing the zoiper softphones. At the point when the information rate is underneath 256Kbps, at that point there is no jitter in the call. Figure 20 demonstrates the genuine equipment setup.

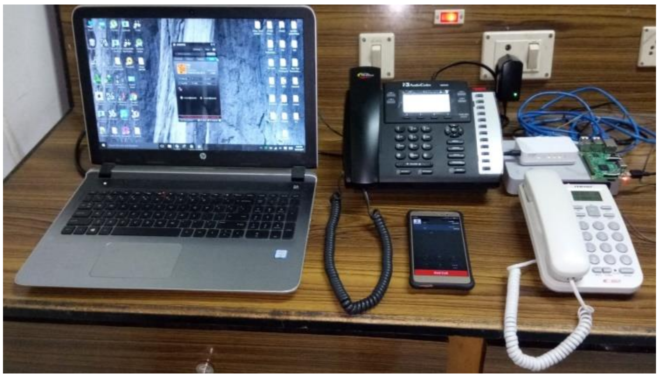

Fig. 20. Hardware setup of proposed IP-PBX system

6. CONCLUSION

The raspberry pi based radio framework can be introduced effortlessly alongside the current radio or LAN or WLAN frameworks. This compose change empowers the client to utilize the cell phone as a hub of radio framework with free of cost. Because of ease, little and medium scale businesses and school and school grounds can bear the cost of this framework with less venture. At that point by introducing this framework in Public Wi-Fi, free calling office can be given to the Public.

7. FUTURE SCOPE

When every one of the regions are introduced with this IP-PBX radio framework as appeared in figure 21, voice and video calls can be set up with no jitter and at free of cost. By utilizing this office, Emergency Communication Services (ECS) can be given by utilizing the Wi-Fi and under-ground fiber optic links. Since, amid catastrophe, the GSM system may fall flat, and around then, the individual in threat can be effortlessly imparted utilizing this framework and number lives can be spared. Since video call can likewise be built up, First AID can be given by specialists over video conferencing for individuals whom are disengaged, because of debacle.

Fig. 21.Wi-Fi based ECS system using IP-PBX

REFERENCES

[1] Ashwini S. Gawarle (2017), "Design a Free Voice Calling System Using Raspberry Pi", International Journal of Research in Engineering and Applied Sciences (IJREAS).

[2] Chunhui Yuan and Hongli Zhao (2016), "Implementing VoIP Voice Communication System based on Soft-switch Technology", 2016 International Conference on Cyber-Enabled Distributed Computing and Knowledge Discovery.

[3] Hussein Al-Saadawi and AsafVarol (2017), "Voice over IP Forensic Approaches: A Review", 978-1-5090-5835-8/17/$31.00 ©2017 IEEE.

[4] Nikhil A.karotiya, Nikhil P.Wyawahare and Dr.S.L.Haridas (2016), "Review paper on Point to Point Communication with the use of Power over Ethernet based on VOIP system on Asterisk", International Conference on Advances in Electrical, Electronics, Information, Communication and Bio-Informatics (AEEICB16).

[5] Pramila.B Bamnote (2016), "Design and implementation of wifi based intercom system using Arm11", International Research Journal of Engineering and Technology (IRJET).

[6] Rahul C. Vaidya and Prof. S.S. Kulkarni (2012), "Voice over IP Mobile Telephony Using WIFI", International Journal of Scientific & Engineering Research Volume 3, Issue 12, December-2012.

[7] Ryota Nishimura, Koji Sugioka, Daisuke Yamamoto, Takahiro Uchiya, and Ichi Takumi (2014), "A VoIP-based Voice Interaction System for a Virtual Telephone Operator Using Video Calls", 2014 IEEE 3rd Global Conference on Consumer Electronics (GCCE).

[8] Serge Fabrice Mbianda Ngongang, Navid Tadayon and Georges Kaddoum (2016), "Voice over Wi-Fi: Feasibility Analysis", 2016 Advances in Wireless Optical Communications (RTUWO).