Design and Implementation of Compatible VoIP

Nisarg Brahmbhatt, *Student Member, IEEE*, Pankaj Mann, *Student Member, IEEE* and Abhishek Rawat

***Abstract*—In this paper, we propose a simplified way of voice communication over Wireless Local Area Network (WLAN) using Internet Protocol, without using internet connectivity at any end in enclosed system. A voice signal is converted into an electrical signal using a microphone and this analog signal will be converted into a digital signal using appropriate quantization level and sampling rate. This digital data is then transmitted over the WLAN network using internet protocol in certain defined packet size, i.e. 8 bit digital packet. At the receiver's end, packets passed through WLAN are received as is, which are combined and converted back to the transmitted analog signal from transmitter's end. These signals can be combined into sound signal through a speaker connected with receiving computer. This method of communication over internet protocol is generally called VoIP, i.e. Voice over Internet Protocol. This method can replace the current PBX telephonic network inside organizations. The demonstration of this mode of communication has been successfully done on institute WLAN with the VoIP server.**

Keywords: VoIP; IoT; Raspberry Pi; Voice Communication; Pulse Code Modulation.

I. Introduction

The modern era of communication started with morse code, single line communication, then wired telephones. Several such telephones were multiplexed on a single wire, and a network was established. After that, era of wireless communication via cellular phones started. Now, Internet of Things (IoT) is going to be the new modern era of telecommunication. [2]

IoT basically means copious amounts of data, analyzing data, make it meaningful and also share it for its users. For instance, a company wants to place a call to its office in another city and in meantime also wants to exchange data, both can be achieved simultaneously with the help of VoIP with IoT and thus, reducing all the heavy telephonic bills and also we get to exchange information for free. Also, VoIP in IoT is a platform that can serve us with information from devices for humans to get involved in with this data.

Therefore, what VoIP does for people's and businesses' bottom lines in relation to telephonic calls and services, the IoT is performing the same with knowledge. Information we had no way of knowing prior to the IoT can now be known, studied and, the hope is, understood and acted upon. [1]

Nisarg Brahmbhatt is an Undergraduate Student with Department of Electrical Engineering, Institute of Infrastructure Technology, Research and Management, Ahmedabad, 380026. email: nisarg.brahmbhatt@iitram.ac.in

Pankaj Mann is a Graduate Student at Indian Institute of Managment, Lucknow, Uttar Pradesh 226013. email: mann56pankaj@gmail.com

Dr. Abhishek Rawat is an Assistant Professor with Department of Electrical Engineering, Institute of Infrastructure Technology, Research and Management, Ahmedabad, 380026. India. email: arawat@iitram.ac.in

When we speak, acoustic waves are created which propagates in air to enable the hearing sense. A mic can convert these waves into electrical signals as it works on the principle of electromagnetic induction. The voice signal incident on the diaphragm, which vibrates due to the pressure difference in the wave. A coil of wire is attached to the other end of the diaphragm, which slides smoothly on a magnet. The diaphragm vibrates, moving the coil over the magnet, producing current in the coil due to electromagnetic induction. This current when plot against time is electrical representation of sound wave.

Human sound listening capacity comes under the range of frequency between 300 Hz to 3 KHz. Traditionally, voice communication is done via wires, where various analog signals are multiplexed and transmitted to a distant location, and at the receiver, a speaker produces the sound, which is an acoustic wave, sensed by ears. The working principle of speaker is similar to mic, here the electrical signal is given to the coil, which moves the coil over the magnet to vibrate the diaphragm that will produce sound.

The proposed method involves voice communication through an existing Internet Protocol (IP) network. Since voice signal is analog in nature, in order to transmit it through internet protocol, digital data packets are required. Pulse code modulation is used to convert it into digital packets and then transferred through IP. Two different computer systems are used, so that the duplex mode of communication can be established. The WLAN and LAN network of our institute is used as the channel, without the use of any internet connectivity at any point in the enclosed system.

This paper proceeds as Section II describing formulation of VoIP system and hardware which is used, Section III describes about the pulse code modulation that is used in sending voice signal through communication channels. In Section IV, voice data transmission is explained, Section V analyzes the result and finally Section VI draws some conclusions of the paper.

II. VoIP System Foundation

The proposed VoIP module is comprised of, Broadcom Processor Raspberry Pi, a mic, earphones, USB sound card and a Wi-Fi network. For powering the Raspberry Pi, portable mobile phone charger was used. The basic VoIP system is as shown in Figure 1.

A. *Raspberry Pi 3 model B*

Raspberry Pi 3 model B (from here on, it shall be referred as RPi) is the latest version of Raspberry Pi which is

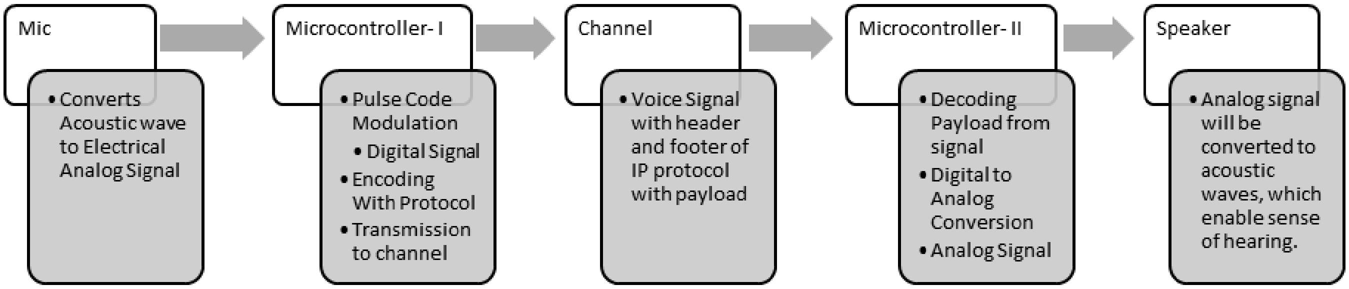

Fig. 1. Block Diagram of a basic VoIP system.

low power, single board credit card sized computer [3]. It is manufactured in several board configuration through licensed manufacturing agreements with Newark element 14, RS Components and Egoman [4]. It is packed with a 1.2GHz quad-core ARM Cortex-A53 processor, with 1GB of RAM at 900MHz. It comes with inbuilt Wi-Fi and Bluetooth adapters. The foundation provides Debian and Arch Linux ARM Distributions for download [5], and promotes Python as the main programming language. RPi 3 includes a slot for memory card, in which operating system (DEBIAN with integrated VoIP server libraries, in this case) is installed and programmed libraries are written for speech signal transmission, receiving and managing.

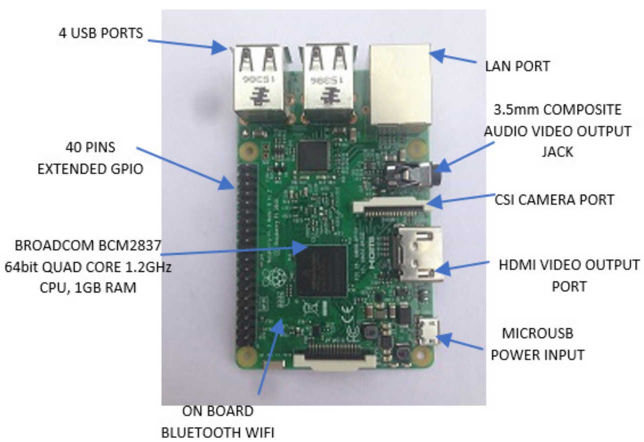

Fig. 2. Top view of Raspberry Pi 3 model B.

This has an output of 3.5mm audio video composite output port, through which we connect our speakers with earphones or a USB mic may be required due to unavailability of earphones. Fig. 2 shows a top view of RPi, that weve used. It has a HDMI port for multimedia output which is generally used to connect RPi to an external display monitor, used while programming or debugging the micro-controller. It has 40 GPIO (General Purpose Input-Output) pins which will not be used in our model however, we can use them while designing a fully functioning network phone, and these can be used to connect to different devices, or to disconnect or connect during an ongoing transmission. Thus, making a RPi board that can function as a handheld computer device as well as a mobile handset for placing calls, but here, not as a mobile handset to place calls. It is used as a server to handle call requests and manage traffic through it's platform.

A server for handling calling requests and registry of calls and user authorization access keys to use this feature, are hosted by the same RPi, viz. by making a connection with existing institutional WLAN infrastructure. Enabling a static IP address to the VoIP server created, that also gives administrative access to a known entity to handle access permissions for different users via a '.html' page, giving easy access to the administrator for switching control options and monitoring traffic on the server with an external monitor.

III. Pulse Code Modulation

For high speed voice communication relying on analog modulation is not enough. All the signals that exists in nature are analog, for transmitting digital signals we need to convert these analog signal to a digital signal. Pulse code modulation is one such technique.

Generically this process is known as sampling. Voice signals are typically of frequency 300Hz to 3400Hz, and in order to gather sufficient information from the analog signal, sampling frequency is determined by the Nyquist Sampling Theorem, which states that the frequency of sampling should be at least twice the highest frequency component. So to be on the safer side we sample at 8 KHz, i.e. every 125 µs.

Once we got a sampled signal, we assign each of the sample a binary code, this could be difficult considering infinitely many amplitude levels. To tackle this problem, we use different levels of quantization, and assign a binary code to each of this level. The number of levels is decided with bit width of the binary code. We adapted 8-bit binary code, so a total of 256 quantization levels. It may happen that the amplitude of sampled signal will lie between two quantization levels, and the signal is rounded up to the nearest whole level. The difference between the sampled level and quantization level is called quantization distortion. This is equivalent to the noise. As we increase the bit-width of the digital signal, we increase the number of quantization levels results in decrease of the quantization noise. In voice communication, 8-bit digital signal is more than enough, unlike with digital music, where we need to have quantization noise as low as possible, and thus we increase the bit-width of the digital signal and hence the size of audio will increase. Now consider Figure 3, showing pulse code modulation over a sinusoidal signal, with 256 quantization levels and 8-bit message signal.

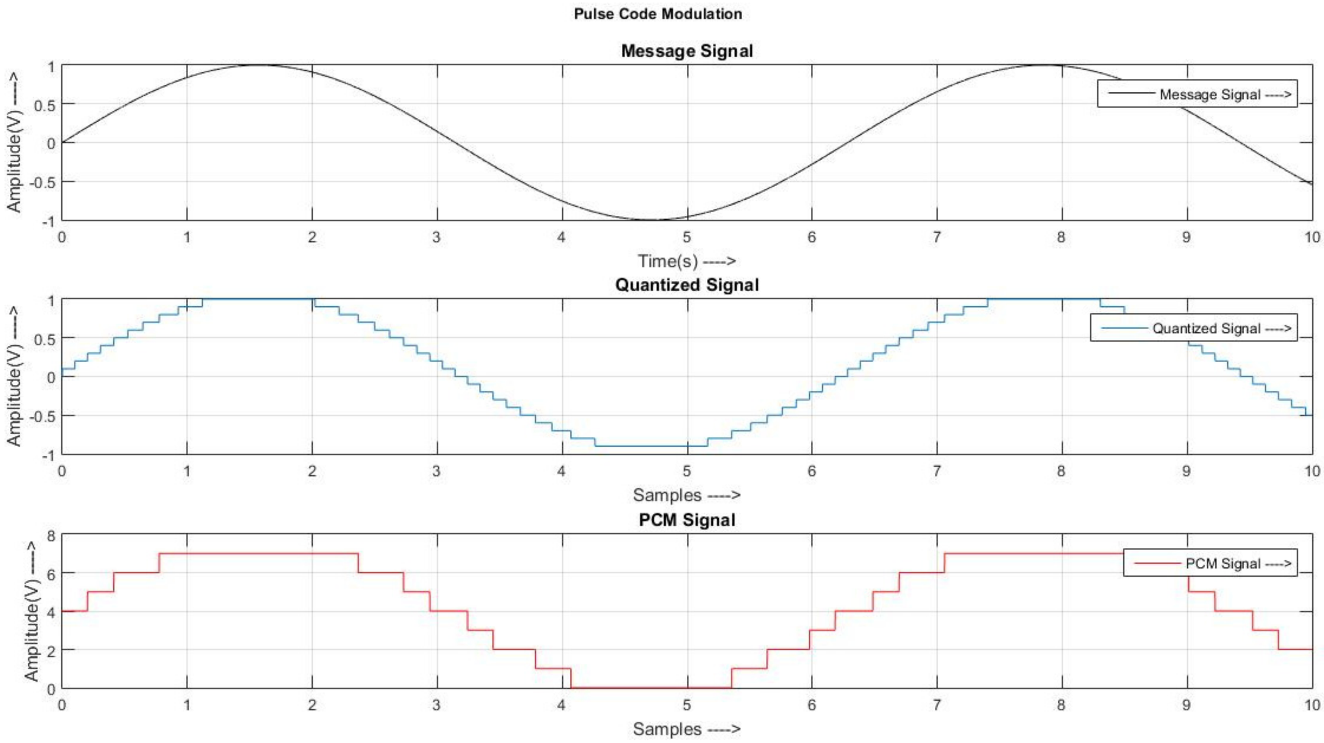

Fig. 3. Pulse code modulated sine wave with 256 quantization levels.

IV. Voice Data Transmission

An internet protocol defines a uniform format for host addresses. An IP address is such a component, it is a unique naming scheme which is required by all the components connected to internet protocol. The message on internet goes through a sequence of machines this sequence is defined by the IP protocol. It defines a packet of digital size 8-bit. A packet generally consists of three main parts, header, payload and the footer. The actual data is in the payload and the header is a metadata which is required by the machine to make sense from payload. These rules enables effective communication, among other tasks a protocol aids in routing, flow control and arbitration of data. Generally the header and footer of a packet contains control and information fields but sometimes the control information is calculated based on the data so in such cases metadata is placed in footer.

In our proposed setup, VoIP server is coordinating information of users with different user codes. For example, *abc* user is sitting on a LAN connected computer in lab 1, and *xyz* user is sitting in lab 2 connected with a different WiFi access point (but on the same WLAN infrastructure) from RPi server, then voice data transmission takes place from LAN to WiFi with the help of VoIP server embedded in RPi. Here, RPi server takes voice data packets from LAN PC via a very lightweight SIP [7] client and transfers them to WiFi PCs lightweight SIP client that reads the voice data packets to voice signal and then simultaneously from WiFi PC to LAN PC with the help of VoIP server on RPi via WLAN and LAN. This to and fro, viz. duplex of voice data packets is continuous till the termination of call from either end [8]. The observed QoS for this specific mode of communication was inferred to be of good satisfaction level.

V. Result Analysis

The Python libraries are used for converting the voice signal to digital data '*most VoIP Library*' is used for transfer of data [6]. Here, a local server was started on RPi, and both the clients are connected to it on WiFi (*abc* user) and LAN (*xyz* user). Specific IP addresses are assigned to these clients in VoIP server on RPi. For making the call, IP of address or the alternate domain name of the receiver is required. The local VoIP server (connection shown in Figure 5) has the current status of all the devices connected to it. When one of the authorized device tries connects to the other authorized device, a request is sent to the server and based on the status of the recipient, server reverts back with a message to the caller. If the recipient is not available the call is not made and if the recipient is available and accepts to connect through, the call is connected to receivers end, and effective communication without use of telephonic PBX or internet is now successful [9], [10]. During an ongoing call the status of both the devices will be shown as busy by the server. The powering of RPi and connections made to USB monitor and keyboard is shown as in Figure 4. The IP address of VoIP server on RPi is as shown in Figure 5.

Key parameters like Received Signal Strength Indicator (RSSI), Noise, Transmission (Tx) Rate, Physical layer of the OSI model (PHY Mode), Modulation and Coding Scheme Index (MCS Index) of the WiFi connection with shortest distance from WiFi access point are analyzed as shown in

Table I. Also, we have placed client device behind or near the signal dampening elements in lab, i.e. walls and tables.

Fig. 4. RPi server on institutional WiFi in lab 1, powered by a standard mini-USB charger pin, with HDMI monitor & USB keyboard and mouse.

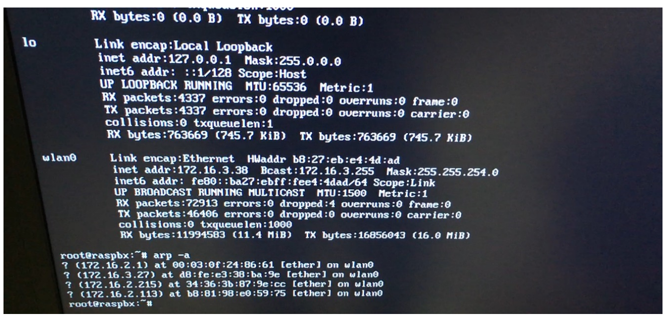

Fig. 5. IP address (172.16.3.38) of RPi VoIP server in lab 1.

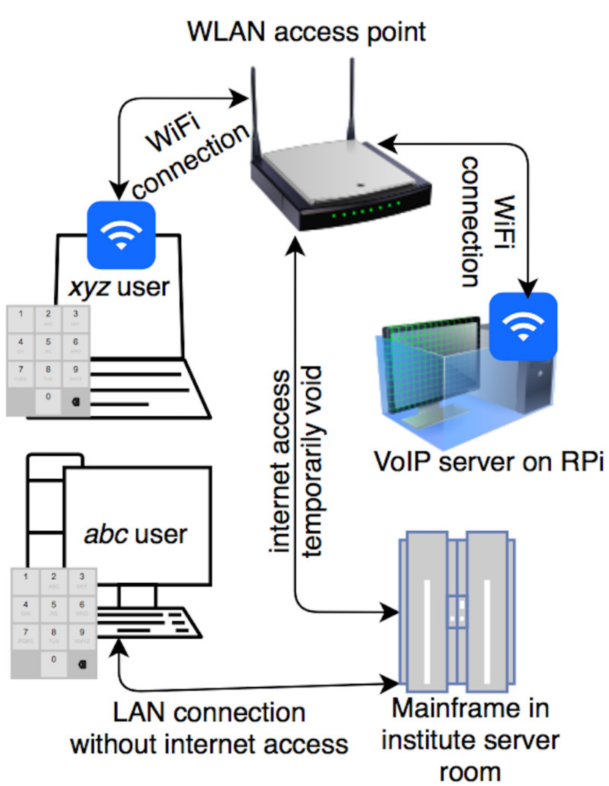

Fig. 6. Pictorial representation of implemented VoIP module.

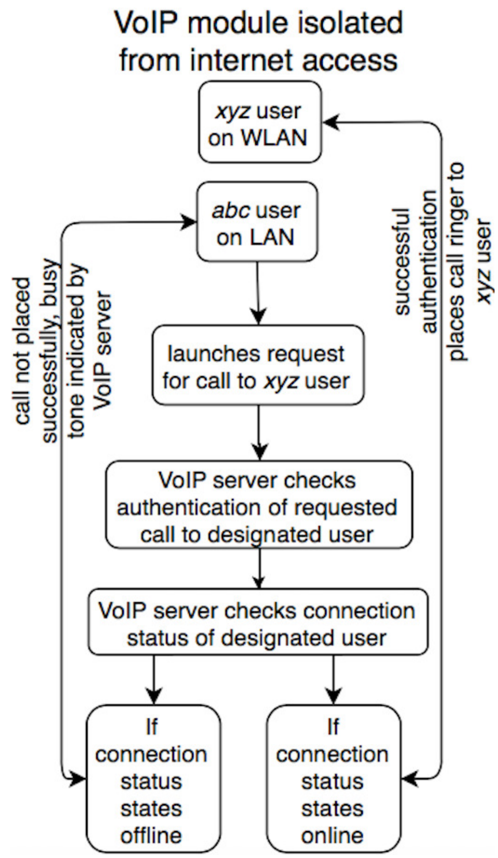

Fig. 7. Flowchart of call request from user *abc* to user *xyz* in module.

A pictorial representation of the implemented and tested VoIP module is shown in Figure 6. It clearly states that the module is isolated from internet access. Also, a flowchart, as shown in Figure 7, depicts a call request from *abc* user, on LAN, to *xyz* user, on WLAN, through VoIP sever on RPi. When authentication of requested calls are placed, the VoIP server will authenticate both the addresses, viz. the recipient and the sender authorization key as VoIP server is preset with the user domain names and their IP addresses, making the system a little more secure in terms of placing calls and no other recipient on network is able to trap the duplex communication between the two users.

TABLE I
PARAMETERS OF WIFI FOR CALL CONSISTENCY ON NETWORK

Channel Freq.(G), BW (M) (Hz)	RSSI (dBm)	Noise (dBm)	Tx Rate (Mbps)	PHY Mode (802.11)	MCS Index	Dist. from WLAN access pt. (m)
2.4, 20	-76	-72	56	n	11	21
2.4, 20	-68	-84	78	n	12	19
2.4, 20	-61	-88	105	n	13	16
2.4, 20	-59	-91	119	n	14	13
2.4, 20	-51	-93	131	n	15	10
2.4, 20	-46	-96	137	n	15	7
2.4, 20	-34	-99	145	n	15	4
2.4, 20	-19	-99	145	n	15	1

VI. Concluding Remarks

VoIP telephony based on local IP network could be a best solution for institutional communication without the use of internet or telephonic PBX. We can also make such specific module to communicate using VoIP such as IP telephones, and then take it to a larger scale. We know that voice communication is free over internet, to make our server online, there will be no incremental cost, and it will also work even in the absence of network or emergencies.

Initially, we prepared a small VoIP system using Raspberry Pi 3 model B and inferred the traffic analysis, without internet connectivity, in wireless and LAN mode. For the implementation, we consider a number of issues that may arise during the testing phase of this network, such as data packet loss due to noise, received signal strength indicator (RSSI in wireless mode). This type of module can be helpful in designing smart home or large scale industrial communication links.

References

[1] K. Rohloff, D. B. Cousins and D. Sumorok, "Scalable, Practical VoIP Teleconferencing With End-to-End Homomorphic Encryption," in *IEEE Transactions on Information Forensics and Security*, vol. 12, no. 5, pp. 1031-1041, May 2017.

[2] A. H. Ngu, M. Gutierrez, V. Metsis, S. Nepal and Q. Z. Sheng, "IoT Middleware: A Survey on Issues and Enabling Technologies," in *IEEE Internet of Things Journal*, vol. 4, no. 1, pp. 1-20, Feb. 2017.

[3] Cellan-Jones, Rory, "A £15 computer to inspire young programmers." BBC News: http://www.bbc.co.uk/blogs/thereporters/rorycellanjones/2011/05/a_15_computer_to_inspire_young.html, 5 May 2011. Retrieved 2 Jul. 2016.

[4] "About the Licensed manufacturing deal": http://www.raspberrypi.org/about. Retrieved 2 Jul. 2016.

[5] Raspberry Pi Downloads: http://www.raspberrypi.org/downloads. Retrieved 2 Jul. 2016.

[6] Python most voip library: http://most-voip.readthedocs.io/en/latest/python_docs/#python-most-voip-library. Retrieved 24 Jul. 2016.

[7] Session Initiation Protocol (SIP): www.cs.columbia.edu/sip/papers.html. Retrieved 23 Oct. 2016.

[8] SIP Basic Call Flow Examples: www.ietf.org/rfc/rfc3665.txt?number=3665. Retrieved 20 Nov. 2016.

[9] Mark A. Miller, P.E., "Voice over IP Technologies: Building the Converged Network", M and T Books, ISBN: 0-7645-4907-3. pp 211-245.

[10] Galitizine, Greg. "Pulling Together - Interoperability through Open Standards." Internet Telephony (Oct. 2000), pp 62-66.