*Sindh Univ. Res. Jour. (Sci. Ser.) Vol.49(1) 195-200 (2017)*

**SINDH UNIVERSITY RESEARCH JOURNAL (SCIENCE SERIES)**

# Simfree Communication using Rasberry Pi+ Based Base-station for Disaster Mitigation

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Received 13th April 2016 and Revised 20th January 2017

**Abstract:** The telecommunication facilities are normally not available in flood and earthquake effected areas. Under such circumstances, telecommunication company free (telecofree) and simfree communication is required which can work in all situations (independent of telecommunication companies). In this research work, Subscriber Identity Module (simfree) communication was done by using raspberry pi B+ as base station. The device was operated at 2.4 GHz frequencies using Wi-Fi enabled android mobile phones. The android devices were connected with centralized mini base station. The base-station did the user authentication to communicate in between two devices and for conference call services. The base station provided the facility for communicating the text, voice and video data in between the android cell phones without using any SIM. This research helped in designing and implementing the future adhoc networks for the military communication and disaster mitigation

**Keywords:** Base Station, Raspberry Pi, Software Defined Radio, Audio Conference, Disaster Mitigation

## 1. INTRODUCTION

The mobile communication facilities were destroyed and not available (Sayeeda, 2014) (Secretariat, 2009) after 2005 earthquake in Azad Kashmir and 2010 flood in Khyber Pakhtun Khuwa (KPK). It creates lot of hurdles in rescue and disaster mitigation activities. In these situations, it was felt that there must be some communication facilities independent of telecommunication companies (telecos). For this purpose, licensed band communication was not suitable and it was observed that licensed free frequency bands can be more helpful for this propose. It was also observed that communication devices must have the ability to work SIM (Subscriber Identity Module) free, protocol free, modulation free which can communicate at any desired frequencies using some programming techniques. If all of these tasks are successfully obtained, then mobile communication is possible in between rescue workers in emergency situations.

In hilly areas, the population and number of mobile users are very small. In these areas, it is commercially not viable for the telecommunication companies to provide the communication facilities. Only solution available is satellite communication which is very rare and costly for poor Pakistani people (Sonkar, 2012). Mitola presented the idea of software defined radio (SDR) in his MS dissertation in KTH Sweden in 1996 (Mitola, 1995). Software define radio (SDR) has a programmable board which can be programmed to fulfill the required communication protocols. The communication parameters like frequency bands, air interface protocol can be upgraded doing some python based programming and by using GNU library without replacing hardware module. SDR is a secure and efficient solution for the multiple bands & multi-mode functionalities problems for wireless communication systems. These features make SDR a best candidate for Disaster (1995). These radios are very popular for military and civil applications. American JTRS radio is the best example of SDR based military communication systems ( Place, October 2000,) (Donald *et al.*, 2006) (Donald *et al.*, 2007).

The raspberry pi is an ARM processor based embedded device that can plugged in with any HDMI supported liquid crystal display (LCD) monitor or television (TV) and workable with normal standard Universal serial bus peripheral (USB) supported keyboard and mouse. It is a small (credit card) size device that enables users for all levels/ages to expand its computing functionality and features. It boots from an attached standard (SD) memory card. It's a cheapest, simplest and user friendly device (Karankumar, 2014). The user can write programs in multiple programming languages like python, scratch and Perl. Raspberry Pi is made on integrated chips. It has its own random access memory (RAM), CPU, GPU, USB controller. It has 40 general purpose input output (GPIO) pins which can be used to perform additional functionalities on input and output signals. The raspberry pi is commonly used in automation industry for monitoring purposes. 2.4 GHZ is an unlicensed wireless industrial scientific and medical (ISM) band. It can be easily used for testing and small

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level wireless setups such as Adhoc networks, Point to point wireless connectivity for medium level data rate. It can be used for the data rate up to 150 mega bits per second (MBPS). 2.4GHZ devices are cheapest than other radio devices to perform lab work and medium level deployments (Terms and Definitions, 2009). The present research work is based on VoIP protocols for SIM free calling system using raspberry pi B+ and 2.4 GHz wireless USB transceiver. In this research, the idea of calling system was implemented using Linux, Asterisk and Free PBX system. The research work is subdivided in to two sections. Section A covers the introduction of whole idea. Section B covers the system model. Section C presented the results. Conclusions are covered in section D. In the end future work and references are mentioned..

## 2. MATERIALS AND METHODS

FreePBX system is compatible with IP phone application with windows and android phone devices. They are compatible with popular manufacturer devices like physical SIP phones. The displayed applications on screen allow the user to directly control the functionalities though settings of android phones. The applications are voicemail, follow ME , call parking, hot desking, Don't disturb, CRM, login / logout are the few examples of these applications.

| Frequency Range | | Bandwidth | Center Frequency | Availability |
|---|---|---|---|---|
| 6.765 MHz | 6.75 MHz | 30 kHz | 6.780 MHz | Subject to local acceptance |
| 13.553 MHz | 13.567 MHz | 14 kHz | 13.560 MHz | Worldwide |
| 26.957 MHz | 27.283 MHz | 326 kHz | 27.120 MHz | Worldwide |
| 40.660 MHz | 40.700 MHz | 40 kHz | 40.680 MHz | Worldwide |
| 433.050 MHz | 434.790 MHz | 1.74 MHz | 433.920 MHz | Region 1 only |
| 902.000 MHz | 928.000 MHz | 26 MHz | 915.000 MHz | Region 2 only |
| 2.400 GHz | 2.500 GHz | 100 MHz | 2.450 GHz | Worldwide |
| 5.725 GHz | 5.875 GHz | 150 MHz | 5.800 GHz | Worldwide |
| 24.000 GHz | 24.250 GHz | 250 MHz | 24.125 GHz | Worldwide |
| 61.000 GHz | 61.500 GHz | 500 MHz | 61.250 GHz | Subject to local acceptance |
| 122.000 GHz | 123.000 GHz | 1 GHz | 122.500 GHz | Subject to local acceptance |
| 244.000GHz | 246.000 GHz | 2 GHz | 245.000 GHz | Subject to local acceptance |

**Table1: ISM band frequency allocation.**

## 3 EXPERIMENTAL WORK AND DISSCUSSION

In this research work different protocols implemented to communicate between client end devices like raspberry pi.

### *1) Implementation of system*

Currently three major protocols exist in VoIP telephonic systems that are widely used in telecommunication systems. These protocols are H.323, SIP and MGCP. SIP protocol was used for our communication systems.

### *2) SIP (Session Initiation Protocol)*

SIP is an IETF standard protocol for interaction of user level devices with sip bases serer media control devices, multimedia elements like as voice, games, text chat, video calling, and virtual reality. SIP protocol has dynamic functionality like one side communication, plenary unicast and multicast which will be unavoidably and involve the initiator. The SIP protocol is described in RFC-3261 standard. It is application level protocol of signal processing for setting up, terminating and modifying the real time session of communication between client and media gateway device (SIP gateway) network. SIP also support the single-media and multi-media sessions including conference base callings between multi users (not limited to VOIP calls only). The SIP can be used as text base protocol. It is alternate to H.323 and resemble with HTTP. SIP has more popularity than H.323 protocol due to its advance and easy to implementation of the session (A white paper by Siemens Communications, 2004).

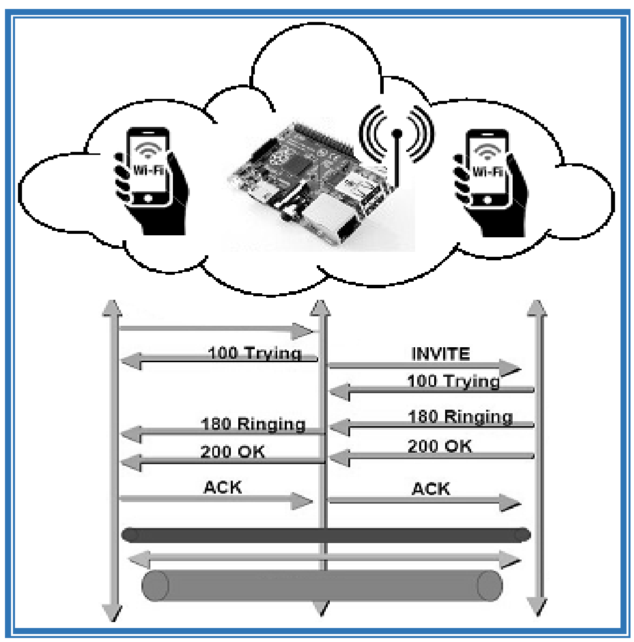


**Fig. 1 Session initiation protocol**

### *3) Voice communication protocol*

In this practical implementation, the GSM, μ-Law, A-Law voice protocols were used for to communicate in between client end devices . These protocols were selected due to their high quality voice, low data rate. Multiple protocol selection for voice communication will be helpful in case if user ends devices support only one specific protocol. These protocols were selected dynamically by the asterisk voice module upon call initiation. Too many voice communication protocols are available that make the best media stream as per application requirement and network capability. These implementations are based on compressed, narrowband speech. In this research work, GSM-EFR is used to code speech signal.


**Fig.2. incoming call processing**

### *4) Video Communication Protocol*

Currently, Asterisk dumps the content of Real-Time Transport Protocol (RTP) packets including some timing information in H.263, H.263p files (OlleE, 2013) (Arjan, 2005). In this research work, H.263p codecs handles (coding/decoding) the conversion of analog signal to digital signals, and vice versa. H.263p is a successor of H.26x as a standard of video codec. H.263p has hybrid functionality like intra frame transform coding and inter frame compression. By comparing H.263p previous codec's allows ratio of high compression (Anurag *etal*, 2014 and Pandy et al, 2014). H.263p supports five resolutions CIF, QCIF, 4CIF and 16CIF) have a motion compensation capability; it will allow some optional technique coder to adopt technique of spatial redundancy as advantage to enhance robustness next to the loss of data in tx channel.


**Fig. 3. Video Conversation between two users**

### *5)Text Messages Implementation*

For the regular telco messaging between users using the SIP gateway we have modified some configuration in dial plan settings because Asterisk comes with a channel independent dial plan for handling SIP messaging. To enable this feature through the process of using SMSing with a dial plan configuration. Asterisk uses the *Message* AST_MSG_QUEUE channels to do all SIP Method MESSAGE related processing (Bryant, 2010). SMS() is a feature which will be used in case if users are unable to communicate on long distance or low signal area co communicate with other users. In case of emergency, they will use SMS service to deliver messages.

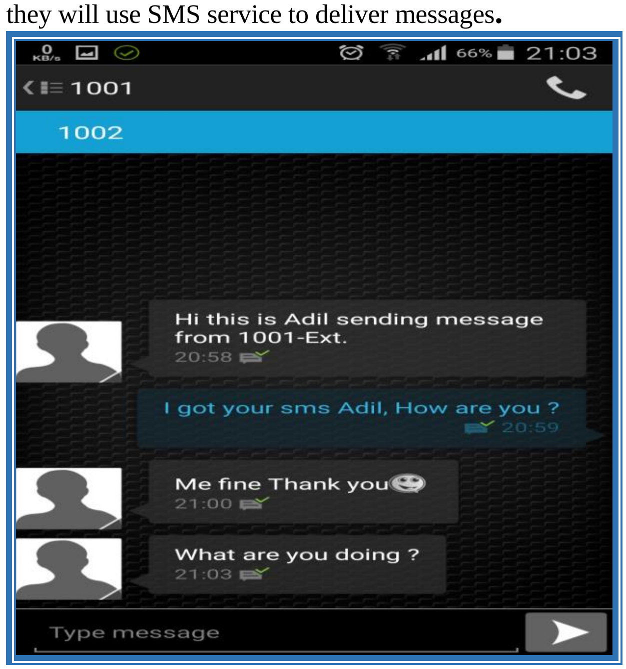


**Fig. 4.Text message Conversation between two users**

***6) Data Processing in System***

Nowadays, voice networks like PSTN are utilizing Digital switching to establish a link between caller and receiver**.** When connection only offer limited bandwidth it does possible to accept quality level without the extra effort of complicated encoding algorithm**.** There are 3 essential components in VoIP (Coder/Decoder) codec, Using voice codec's , first the Voice converted to a digital data stream and encapsulated into network data packets, then digital signals are compressed, encoded into a predetermined format (Sonkar *etal*, 2012)**.** A process is performed in terms of packetization means distributing fragmenting to encoded voice signal into equal size of data packets **(**Sonkar *etal*, 2012)**.** There are two steps of handling VoIP data; call setup and voice call data processing (Jianqiang 2007). During the process of call setup the person who wants to call someone can discover and the action of communicating the recipient to build a voice call data session, for example, Adil Dials yasir's SIP address 1001@example.com on his (IP phone/ Android phone). 1st the request can be contacted to Raspberry Pi server to find and make call session between them. Then, a call connection will be initiated between Amir's phone and Adil's phone, so it can negotiate necessary limit or boundary for voice call data transmission. Figure 3 represent the hierarchy of voice data in VoIP systems.

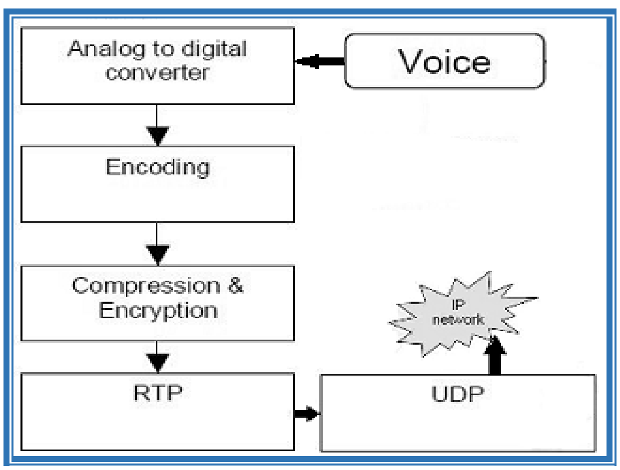


**Fig. 5. Basic Voice flow in VOIP system**

***7) VoIP QoS (Quality of Service)***

For VoIP QoS (Quality of Service) is important factor. Due to multiple users the VoIP provide the voice quality as it was presented by traditional telephone networks (PSTN). Jitter mainly affects the QoS in VoIP telephony networks, packet loss & latency (delay variation) **(**Olusegun, 2011)**.** Network data delay, the setup of VoIP is time critical for transmitting the data from source to destination**.** Time delay (TTL) is very serious issue in VoIP communication systems; the average latency rate for household calls is 150ms for single side (one-way) voice traffic (Karie, 2004). Suppression limitize the specific amount of security which can be included into the VoIP communication network. exceptionally its 20ms to 50ms will be left for security implementations back when encoding and traveling might be take 110-135 (R-Barbieri, 2002) (Jianqiang)**.** Jitter will refer as invariable data packets delay. Intermittent disconnection audio data streaming, jitter is more tending to the cause of QoS than the actual time delays themselves. Most of the time jitter mostly caused by a low data bandwidth in VoIP systems **(**Jianqiang)**.** Voice over IP packet loss is behavior that differs from one's own. the packets of voice data is very small having a payload of only between 10 to maximum 50 bytes, which is approximately 12.0 to 62.5ms. Consequently, irregularly one packet loss is not so important. Which is the reason that's why prefer first TCP is a connection oriented protocol and the UDP is connectionless protocol instead for Audio/Video communication, (Digium, 2012)**.** The QOS is highly sensitive to packet network impairments. The details of the parameters which affect the QoS are explained briefly in the given paragraphs.

**a) Delay**

Delay is a packets transfer (Call initiation) total time between two calling users, speaking words, voice, text messages or hearing them at other end also its categorized in 3 sections, A-Network delay, B-Source delay C-delay at receiver end.

**b)Jitter**

Jitter cause variable delay in packet-switched communication. This is due to variable processing time at routers and different path followed by the packets. It introduces extra buffering delay and packet loss and end-to-end quality degradation in communication.

**c)Packet Loss**

Packets data transfer over the IP network based communication can be corrupted or damaged during the transfer between two nodes. It is known as Packet Loss. In IP networks, data packets are discarded mostly on late arrival at receiver due to jitter. Additional packets may be overflowed or discarded. In this case, Jitter buffer or router buffer will be fully loaded. it may cause more data loss and effects transmission (QOS).

**d)Echo**

Echo occurs when a call initiator hears his own reflected voice after his talk on phone's microphone during the VoIP call session. Echo is a reflection of transferred voice signals from the other end; it is the one of the weird issue in (PSTN) & VoIP networks.

**e)Throughput**

Throughput is the rate of successfully data transferred over a communication media during the specific interval of time. It can be measured in units of information processed in a specific given time.

**f) PDD (Post Dialing Delay)**

PDD is the time between the call start and starting ringing at the receiver end. It can be considered as the time from the call invite user sent by the VoIP group to the RINGING message from the termination. The PDD can be measured in packet data network (VoIP) with simplest method starting from call INVITE stage up to destination status tone. It is also in Figure 6 and 7.

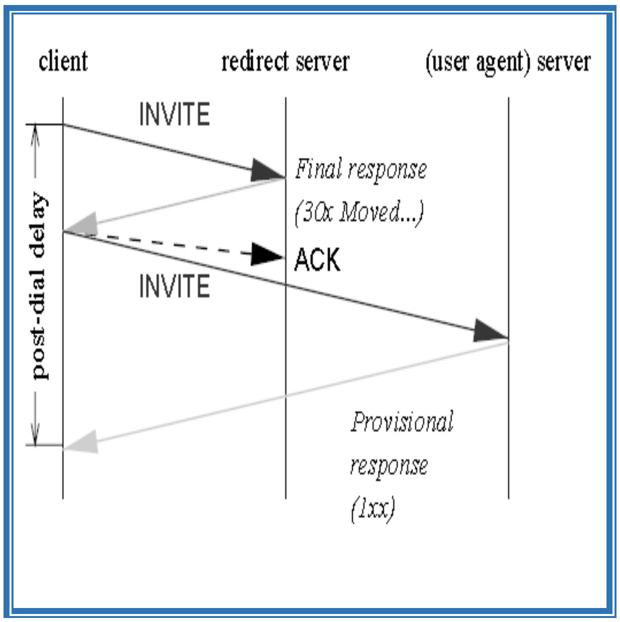


**Fig.6. Ack: return for call initiation (RTP )**

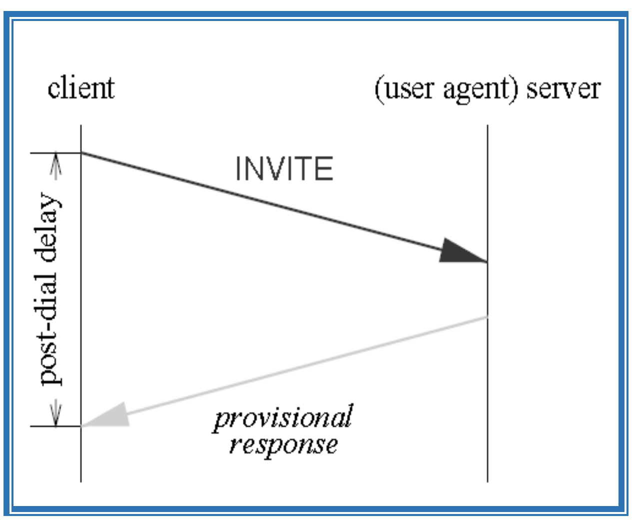


**Fig. 7. Invite process for session initiation**

These results were obtained during implementation phase.

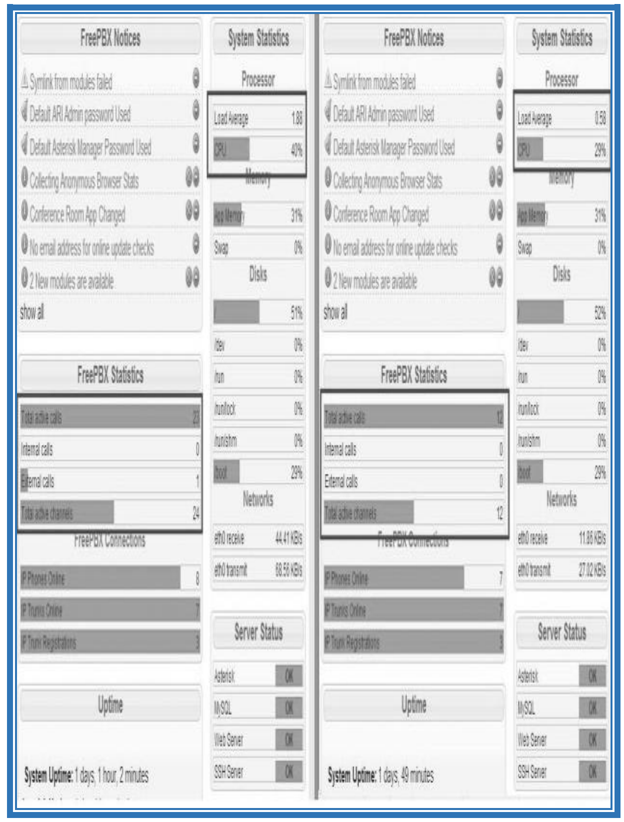


**Fig.8. Administrative panel**

## 4. CONCLUSION

The results were drawn on the basis of implementations & analysis done in the research work. This research work was a practical demonstration of building a functional raspberry pi calling system that served in disastrous situation well. Voice communication is a fast growing technology in commination field. It offers real time communication to achieve reliable and high-quality voice communication over the IP network. It was a real challenge to communicate the voice using credit card device like raspberry pi.

The designing a high quality calling system using asterisk PBX system that includes the best codec selection for voice and video and instant messaging on a same raspberry pi chip. In this research, the connection establishment with client devices to raspberry pi base station was explored. The calls from endpoint users was established by connecting them to raspberry pi base station and accept calls from those endpoints. The communication was done for a distance of 200m with the help of 200mwatts wifi device.

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