

IMPLEMENTATION OF VOICE CALL TRANSFER SERVICE BETWEEN SMART PHONE AND TABLET THROUGH WI-FI

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Abstract. *Communication through voice call leads to significant growth in technology in distant areas where two or more people from opposite ends of world will connect. This research describes a case study of voice call transfer service. This research aims at designing a system that will allow Android users to communicate over Wi-Fi. This design is able to transfer voice of incoming telephone caller over Wi-Fi network at real time through UDP. It uses client/server architecture: Server for receiving telephone call and transferring voice (one user) and client for receiving incoming caller voice and enables communication with server. Architecture designed could be used on Android smart phones with telephony enabled and tablets with telephony not enabled. Outcome of this research will allow users to communicate on real time at no cost. Proposed design gives cost effective, reliable and real time voice communication over Wi-Fi. It provides good and comfort experience to users in emergency situation where user cannot effort cost for telephone call. Proposed design is useful for educational organizations, construction buildings, shopping malls and hospitals which point to new possibilities for voice communication.*

Key words: *voice call, voice call transfer service, real time, client/server architecture, incoming caller, telephony.*

1. INTRODUCTION

Usage of Wi-Fi enabled mobile phones to access Internet is increasing day-by-day. It would be very extraordinary experience to provide communication in secure manner with off-shelf device without using Internet [1]. Proposed research implements design without having any infrastructure, direct communication between devices in which one device was telephony enabled and other device was telephony not enabled. It offers low cost and fast installation for direct wireless voice communication.

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In a business organization or university campus, employees need to communicate with each other very frequently which consumes large amount of budget or cost to pay Internet bill or telephony bill. So it is a big issue which needs to reduce cost of phone call or Internet data flow usage. In a fire accident when a large building was fired, without network, a way in which stranded people can communicate with people outside has an emergent issue. Not only for fire accident, all stranded people caused by disaster, i.e. fire, earthquake, tsunami, volcanoes and debris flow need to communicate with outside in an emergent way while regular communication facilities are destroyed such as phone facilities or Internet facilities. Handling these emergent circumstances are very critical [2].

Concept of voice communication over Wi-Fi is processed by wireless communication through 2.4 GHz free channel [3]. Wireless communication infrastructure represents core for information sharing between connected devices [4]. Router can be used to enable Wi-Fi on devices to enable communication within network range. Proposed research uses WLAN for implementing required design. Wireless LAN technology has ability to change network infrastructure of an organization without expensive re-routing of cable or installation of new cable [5]. Voice over Internet protocol (VoIP) is wireless communication protocol processes packet based IP communication to carry digitized voice [6]. Hence IP addresses are very important because socket programming of UDP processed for voice communication through IP addresses and ports. So that Internet based server is not necessary in proposed research [7].

1.1. Client-Server model

If there are several computers and resources which are available to other resources to establish connection between them, it is called as network. In network some devices are receiving information from others and some are sending information to others. Devices which receive information is called "client" and devices which send information is called "server".

Proposed research implemented client-server model [8] for voice transferring between telephony caller and Wi-Fi user. It can be implemented by smart phone as server and Android tablet as client. Proposed research implements real time voice call transfer service between smart phone and tablet through Wi-Fi with no cost from telephone and Internet service providers. Figure 1 shows basic idea of proposed research implementation. As depicted in figure, telephony subscriber needs to establish connection with another subscriber (smart phone user) through telephone network. Connection establishment is shown by text-field displaying caller's identity; input recipient number and call recipient is notified about an incoming call. After receiving call, wireless connection is established and call receiving subscriber registered to Wi-Fi network and updated with other registered users in network range. Then telephony not enabled user (Tablet user) get connected with smart phone user through their IP addresses. Then captured voice of incoming caller from smart phone is transferred to tablet at real time and voice of tablet user is sent to smart phone [9]. Hence real time communication between incoming caller and tablet user.

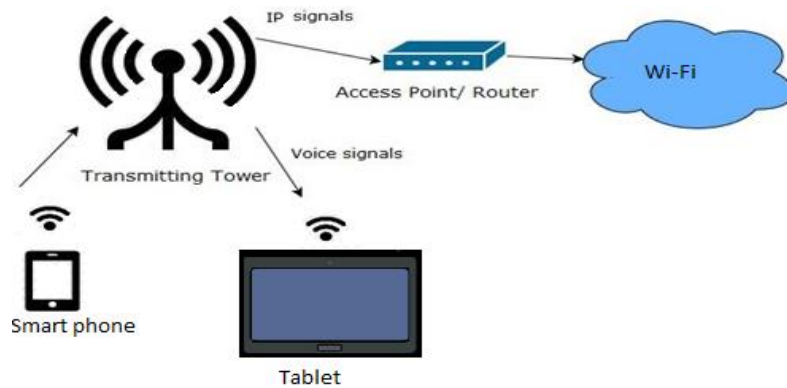


Fig. 1 Idea of proposed application

1.2. Voice codec

Figure 2 describes process of voice codec for wireless LAN. As depicted in figure; first speech signal has to be digitized at sender before transmitted over packet switched networks. Reverse process has been performed at receiver. Digitalization process consists of sampling, quantization and encoding. Different types of encoding techniques in wireless networks are G.711, G.729 and G.723.1. Then encoded speech is packetized into packets of equal size. Each packet consists of headers and payload for certain duration depends on codec deployed in application [10].



Fig. 2 Process of voice codec for wireless LAN

1.2.1. G.711 codec

In wireless networks, G.711 encodes telephone audio signal of 64 kbps with sample rate 8 kHz and 8 bits per sample. In IP network, voice is converted into packets with durations of 5, 10 or 20 ms of sampled voice, and these samples are encapsulated in packet.

1.2.2. G.723.1 codec

G.723.1 codec belongs to Algebraic Code Excited Linear Prediction (ACELP) family and has two bit rates associated with it: 5.3 kbps and 6.3 kbps. Encoder consists of Voice Activity Detection and Comfort Noise Generation (VAD/CNG) and decoder is capable of accepting silence frames. Encoder operates on speech frames of 30 ms corresponding to 240 samples at sampling rate of 8000 samples/s and total algorithmic delay is 37.5ms. It offers good speech quality in network impairments such as frame loss and bit errors. It is suitable for VOIP applications.

1.2.3. G.729 codec

G.729 codec belongs to Code Excited Linear Prediction coding (CELP) family and uses Conjugate Structure-Algebraic Code Excited Linear Prediction (CS-ACELP) model. This was specially designed for wireless applications at fixed 8 Kbit/s output rate but it does not include channel coding. It works on frame of 80 speech samples (10ms) and acquires look-ahead delay of 5ms. Total algorithmic delay is 15ms.

Figure 3 shows architecture of voice call transfer service through Wi-Fi. Proposed architecture was divided into two phases: In first phase, incoming call is detecting and receiving, registration of users. Out of registered users, one is acting as server and other is acting as client. Connection between registered users was achieved by connecting them to same Wi-Fi router. Each user is added to Wi-Fi network through specific ports assigned by programmer. Therefore, there is no need of using any server. Both devices will be connected with their IP addresses and users are added or removed through packets originated from UDP port. In second phase, voice of incoming caller is captured and transferred through wifi from smart phone to tablet. UDP based socket programming will be used to implement this.

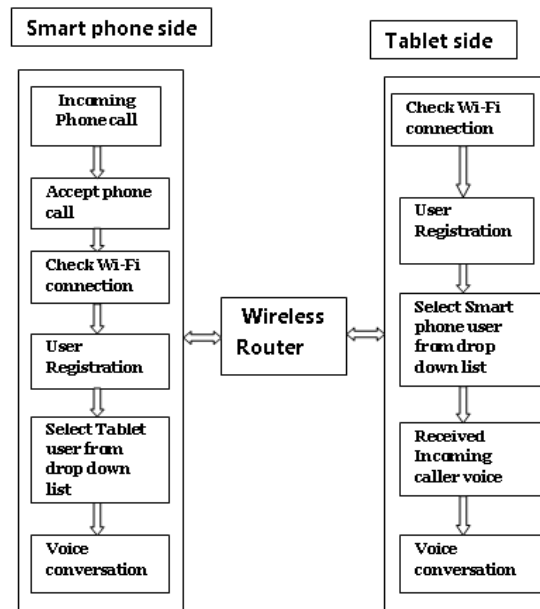


Fig. 3 Architecture of proposed application

Figure 4 shows real time audio encoding and audio decoding. It implements cost less voice communication. Real time communication is established between them through specific ports assigned by programmer. Steps in Audio encoding are record voice, encode, storing in buffer, packet framing and transfer through port. Steps in audio decoding are receiving through port, storing in buffer, reframe packet, decoding and playing voice.

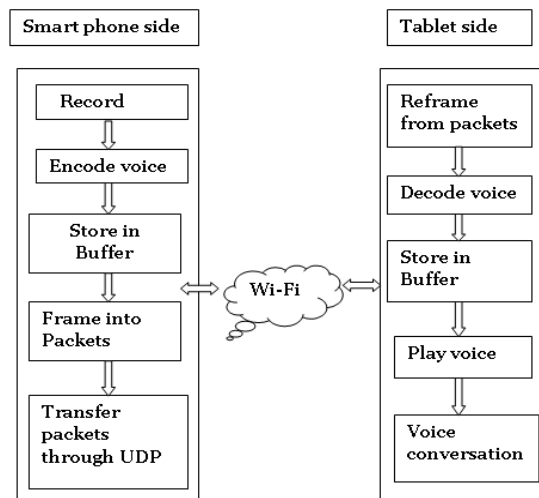


Fig. 4 Process of real time audio encoding and audio decoding

2. RESEARCH BACKGROUND

Latest smart phones feature with Wi-Fi enabling facility. Number of smart phone users for Wi-Fi service has been increasing more and more every year [11]. Usage of Wi-Fi eliminates cost consumed by service providers for short-distance calls. Wi-Fi calls combine voice and data into single signal by digitizing raw voice signals. Converting this combined signal into IP packet and sent through Wi-Fi which replaces existing telephony network [12]. This entire process is facilitated by VOIP.

In current days, voice telephony over mobile trends as growing technology because of cost consuming telephony service. Wi-Fi allows data and voice transmission within its coverage area. Voice over Internet protocol (VOIP) provides communication between Wi-Fi connecting users through Internet. VOIP is a process of exchanging voice between caller and callee through Wi-Fi/Internet. Transforming voice in the way telecommunication evolves makes it more advantageous. Increasing demand of service with its broadband infrastructure causes VoIP to develop economical IP phone liked equipment [13]. IP phone was used as interface between telephony network and IP network. But only drawback is IP phones are fixed type. Voice communications through VOIP are more delay sensitive than error sensitive [14]. To resolve this issue, a real time voice protocol UDP is developed [15].

UDP is a non-reliable wireless communication protocol suited best for real time voice processing. In transport layer, it worked as signaling protocol for IP based applications. It provides peer to peer packet based communication. It converts voice data into packets and communicated to router through Wi-Fi channel and send to device within the Wi-Fi range. UDP reforms communication from circuit based switched network to IP packet based switched network. UDP has network applications in Domain Naming System (DNS), Simple Network Management Protocol (SNMP), Dynamic Host Configuration Protocol (DHCP) and Routing Information Protocol (RIP) [16]. Due to high speed and less impact on lost packet, UDP is suitable for real time voice streaming applications. Existing research possess real time audio, audio streaming applications from one peer to another [17]. UDP

is used in tunnels which create virtual link for direct connection between two locations that are distant in physical network topology.

Voice processing of real time communication is done by encryption at sender and decryption at receiver through digital signals [18]. Previous research explains voice codecs for voice processing. International telecommunications union (ITU) implements voice codec G.711 and G.7xx for audio compression and de-compression. Each codec has different packet size and hence performs differently. Research in [19] describes features of voice codec G.711. It possesses high bit rate (64Kbps) as per ITU standard which is preferable for digital telephony and IP networks. Its quality of voice is high with low processor time. But it has drawback of higher bandwidth utilization. Some of the researches suggest graphical user interface (GUI) mode for voice processing applications. In [20], authors explained dual channel wireless model. It provides high reliability with low delay. By using this model, same packet was broadcasted by multiple transmitters. Hence packet-loss was decreased compared with single channel transmitter [21].

Till date, existing technologies support chatting, video chatting and calling from one device to another over Internet. Communication protocol like VoIP works with help of IP based protocols which makes effective communication with less jitter. Such a system makes WiFi subscribers call each other with no cost over Internet. Proposed paper implements an application by using free source facility and standard for providing free voice call transfer without Internet over Wi-Fi within LAN network from smart phone to tablet. It not only saves money on calling but also provides an effective way for communication to utilize resources in an effective manner. Proposed paper uses WLAN communication medium to provide cost less voice call transfer facility within Wi-Fi covered area [22].

3. METHODOLOGY

Wi-Fi router is acting like switch to detect devices active in Wi-Fi LAN range. Proposed application presents detail implementation of voice call transfer service through Wi-Fi using socket programming. In this work, Android platform has been chosen for developing and implementing mobile application with Java in Android as programming language. It uses required Android APIs to implement proposed application. Android mobile application named "DST_MLRIT_VOICE_CALL" has been developed for proposed research. Different modules are created, developed and running for smart phone and tablet. App contains three sections: Java code, XML code, Android manifest. These three sections inter connected each other to user requirements in Android GUI [23].

Software package using Android studio will be developed for receiving call, automatically acquiring IP address of tablet user and transferring voice from smart phone to tablet [24]. JAVA is used as backbone of programming because it is platform independent. JAVA incorporates number of features from object-oriented language. JAVA includes extensive libraries for multimedia, networking, multi reading, graphic, database access. It has unique attributes that are embedded into its design features makes software developers to design most web applications. Embedded design features include OOP, platform independence, high performance, multithreading and dynamic linking made programming complex applications rather simple and straightforward [25-26].

Application started with program file MainActivity.java. It describes main functions of app. Java programming of app starts with method onCreate(). It sets up all variables and

elements initially for application. It provides buttons declaration and onclick methods of buttons. In GUI, main user interface layout file `activity_main.xml` provides start-up screen. Then, `MainActivity.java` will call `contact_manager.java` for connection handling process. In this way, all programming files link elements of front end with back end [27-28].

Proposed research designs client server based model to implement voice call transfer service. Proposed implementation designed as mobile application in Android studio. Figure 5 shows flow chart for voice call transfer service programmatically. Telephony API in Android makes app continuously in listening state for incoming phone calls. Whenever incoming call was detected, smart phone register itself with name in Wi-Fi network and sends request to other users connected to same network. Datagram socket API makes easy for user registration with add name and remove name parts using UDP packets.

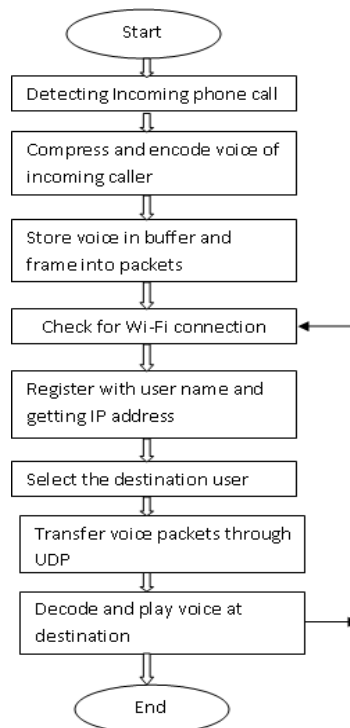


Fig. 5 Design flow of implementation of voice call transfer service

Name entered by user identified with its device IP address through Wi-Fi manager API. Router forwards this request in form of packets through UDP based socket programming. Update button on screen updates all registered within network. UDP socket continuously updates add and remove requests within Wi-Fi network. When telephony caller (smart phone user) wants to transfer callee voice to another user (tablet user) in same Wi-Fi network, application continues to runs by choosing user from drop down list on smart phone.

Both smart phone and tablet are Android devices. Proposed application first detects incoming call. Incoming voice is received by smart phone with support of Android telephony API. Voice from microphone of smart phone is taken for processing to

destination. Sampling rate is kept at 8 Kbps. 16 bit PCM is used for sample voice. Buffer is used to store the sample voice from PCM. Voice codec Api compress and encode voice data to transmit over low bit rate IEEE 802.15.4 standard. Voice encoder converts voice signal into frames and decoder converts frames into voice signal. Single frame contains raw voice data of 20 ms voice signal. Processed frames are given to encoder, which compresses and returns 38 bytes of encoded voice frames. UDP converts these frames into packets. UDP socket based communication is to be used for smart phone to communicate with tablet through Wi-Fi. Smart phone will detect destination device. Destination device is identified with its IP address through auto discovery method.

Figure 6 shown process of voice transferring between two users. Audio manager API uses read method to send encoded data over Wi-Fi. Real time communication is made by thread running for pre defined time interval reads raw voice from microphone. Socket programming provides packet based communication between devices through Wi-Fi. UDP datagram protocol provides packet based communication for tablet and smart phone. As and when Android devices starts transferring voice call, voice codec compresses and encodes user voice into frames and stored into buffer. Data in buffer is converted into packets by UDP and sent over Wi-Fi to another user [25].

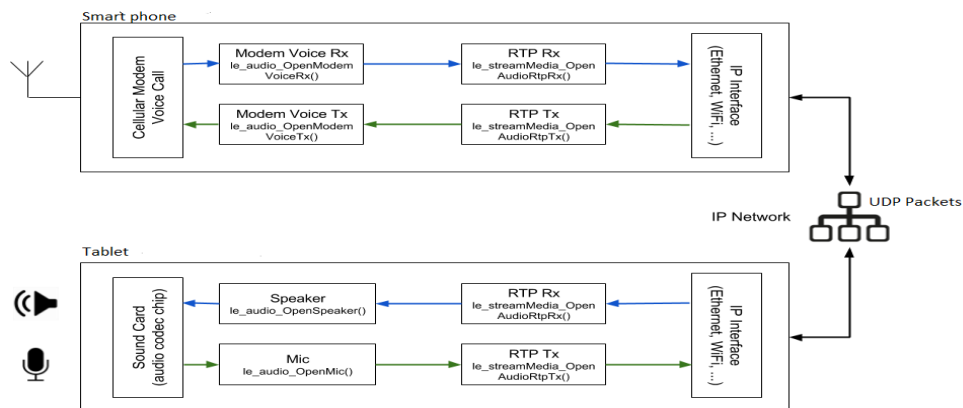


Fig. 6 Design flow of voice transferring between two users

At receiving end, tablet received voice call through Wi-Fi notified as incoming call received from smart phone. Received voice packets are converted into frames and stored in receive buffer. Decoder in codec decodes frames in buffer into voice. Decoded voice is played on tablet GUI at real time. In the same manner, voice from tablet transferred to smart phone at real time.

4. RESEARCH RESULTS

Proposed research is designed as mobile application named “DST_MLRIT_VOICE_CALL”. It was installed on Samsung galaxy smart phone and Lenovo Android tablet which are connected to same Wi-Fi. Designed application installed successfully on both devices. Voice of incoming caller is transferred between smart phone and tablet successfully via Wi-Fi without consuming any cost from telephone and Internet service providers.

It is verified on Samsung galaxy phone and Samsung tablet. Experiment have been conducted using Android studio tools like import and run on Samsung galaxy smart phone as

server and Samsung tablet as client and is quite easy and quick set up. Figure 7 (a) shows deployment of mobile app in smart phone. App launching page is shown in figure 7 (b), Incoming phone call received on smart phone shown in figure 7 (c).

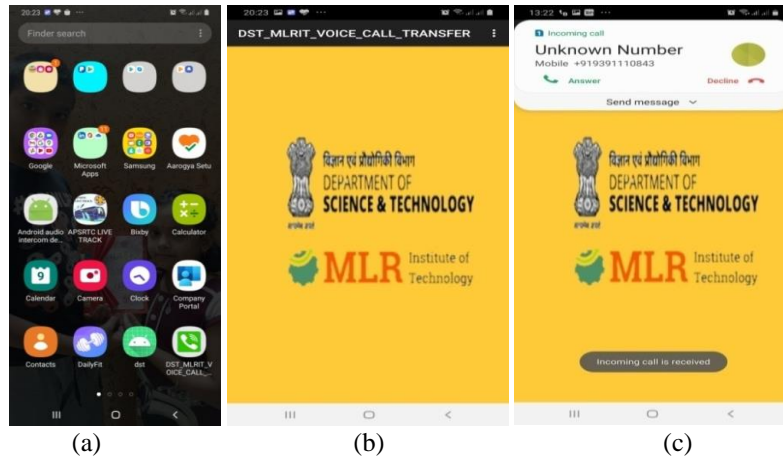


Fig. 7 Pages of mobile application on smart phone (a) Application icon (b) Application launching window (c) Receiving incoming call

Application has process permissions for wifi connectivity, phone and audio. As conversation started, screen appears is shown in figure 8 (a). After starting conversation from caller, with wifi permission, users need to be registered. User registration screen with registered user name “sai” is shown in figure 8 (b). After user submit name, app automatically get IP address of registered user. Update button will update all registered users. After updating, registered users list will be displayed. Screen showing all registered users is shown in figure 8 (c). After user registration, received incoming call will be transferred to registered user. For transferring call, socket based programming is used. When transferring call, packets are sent and received through UDP. Transferring voice from smart phone through Wi-Fi is shown in figure 8 (d).

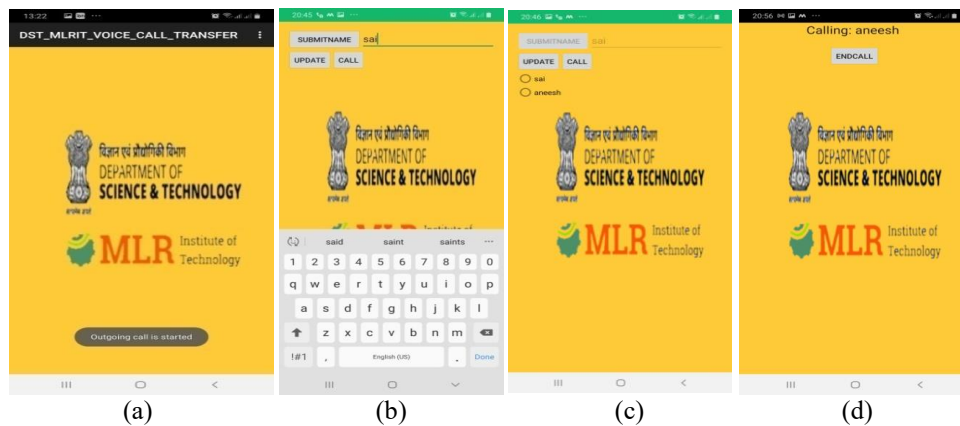


Fig. 8 Pages of mobile application on smart phone (a) Start of outgoing call (b) User registration page (c) Updating users page (d) Transferring incoming caller voice page.

When voice call is transferring from smart phone to tablet, tablet user needs to be registered on same network in application. User registration screen on tablet is shown in figure 9 (a). For proposed application, user with name “aneesh” is registered. After submitting name, user can click on submit name. Update button in screen updates all registered users. List of registered users shown below update button. Screen shot of update of users list is shown in figure 9 (b).

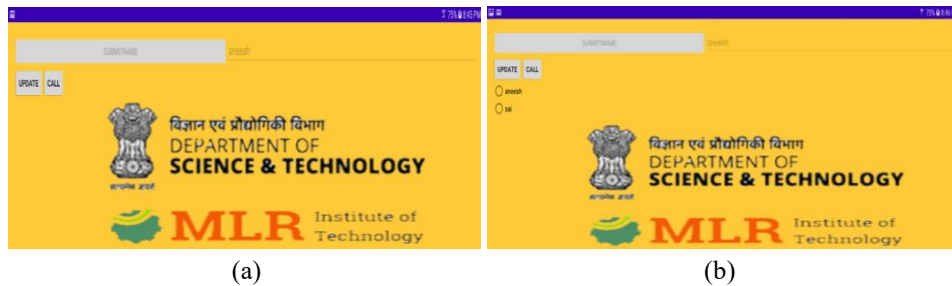


Fig. 9 Pages of mobile application on Tablet (a) Launching application and user registration (b) Updating users

After user registration, user gets incoming voice call from smart phone user which is voice transferring from smart phone. Voice call transferring from smart phone to tablet is shown in figure 10 (a). Accept button is used to accept call screen is shown in figure 10 (b). After completing their conversation, end call button is used to end call.

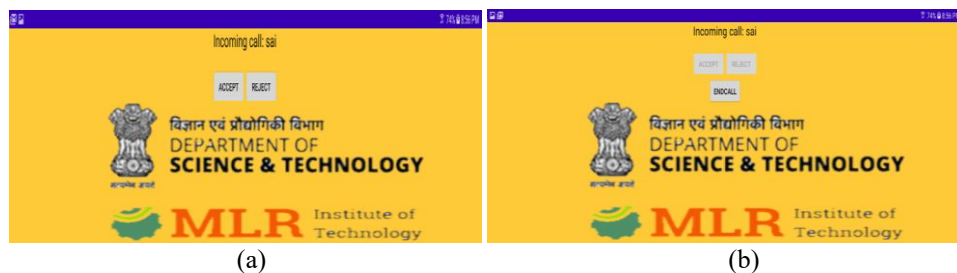


Fig. 10 Pages of mobile application on Tablet (a) Receiving incoming caller voice from smart phone (b) After accepting conversation

5. CONCLUSION

Main motive behind proposed research is to enable cost less and server less real time voice call transfer service from smart phone to tablet without using Internet. It was implemented as mobile application in client-server model. Using Androidapk, proposed implementation was installed in devices as app. Android APIs are used for software model of this design. Wi-Fi service and UDP are used to provide real time voice communication between incoming caller and tablet through smart phone.

Major challenges at sending side of proposed research are capturing received voice from microphone, stored it in buffer and transfer voice to destination device at real-time.

At receiving side, tablet programmed to receive and send voice through UDP ports. As UDP does not reserve extra bandwidth, it would not slow down network. Therefore, it resembles quality of system that can be designed for voice communication over Wi-Fi.

Nowadays, trend moves towards telecommunication with virtual office/class rooms that create legitimate business deals for voice communication. Proposed research design, implements and test on devices and achieves cost effective, server less and reliable voice communication over Wi-Fi. Advantages of proposed model are server less, Internet less and cost less real time communication service without changing infrastructure of organization and without using expensive re-routing of cable or installation of new cables. It is implemented designed and tested for one to one device. In future, it creates opportunity to other developers and researchers to extend this service from one device to more devices.

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