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Open Voice Mobile Telephony using Wi-Fi

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Abstract: The concept of this system is to provide telephony service that is transferring voice from one mobile to another mobile using Wi-Fi. Currently telephony service over mobile is supported at cost using service provider such as GSM, and also IP service provider at cheaper cost. The purpose of this system is to provide free telephony service that uses Wi-Fi technology over access point. The system does not require any Internet connection. The system allows users to search for other those within Wi-Fi range to establish free virtual connection through wireless Access Points. The system uses IP address which is map from mobile number and use it as a mean for communicating with other mobile over Access Point using Wi-Fi technology.

Keywords: Access Point, IP, Wi-Fi.

I. INTRODUCTION

A. NEED

There are two reasons:

I. In GSM and 3G mobile telephony same cost required for short as well as long range distance [8].

II. In Skype & Gtalk etc, they required registration, high speed Internet & also have to pay tariff after first some limited calls.

In our project we provide the telephony service over mobile phone at no cost.

B. BASIC CONCEPT

This system provides to meet the objective of having free telephony services over mobile phones .These are the use of WIFI technology over Access Point and will not require users to register to any service. This can be achieved through dialing the mobile number to map a mobile number to a unique IP address that can be used to establish virtual Wi-Fi connection to any other mobile phone running in same Wi-Fi network. Ad hoc network is an IEEE 802.11 communication network that establishes contact with multiple stations in a given area network without the use of access points or server[4]. Wireless Access point networks help extending the range of fixed wireless networks and give rise to flexible architectures to adapt to geography of users, information, and signal transmission in a locally optimal manner.

This mobile telephony software lends itself to be a completely distributed system in terms of architecture. Distributed computing architecture is described as a number of autonomous processing systems that are interconnected by a computer network and that cooperate in accomplishing the assigned tasks.

II. LITERATURE SURVEY

A. RELATED WORK DONE

This work done is many telecommunications companies and other organizations have been switching their legacy phone infrastructure to a VoIP network, which reduces costs for lines, equipment, manpower, and maintenance. Wireless LAN was employed to remove some of shortcoming of wired LAN. Setting up WLAN was easy and less time consuming. Also systems connected through wireless LAN can be mobile. There is no need to draw costly copper cable to each PC. It is easy to maintain and troubleshoot this system. Popular digital wireless transmission techniques can be divided into three categories according to their applications. The first category is pulse transmission technique used mostly in IR applications. The second category is basic modulation techniques widely used in TDMA cellular, as well as a number of mobile data networks. The third category is spread spectrum systems used in the CDMA, as well as WLANs operating in ISM bands [5]. The main advantage of using wireless LAN is that it provides the ability to change the network infrastructure of an organization easily and without the need for expensive re-routing of cable or the installation of new cable runs. A WLAN can be configured in two basic ways: Peer to peer (ad-hoc mode) and Client-server (infrastructure networking).

The ad-hoc mode consists of two or more PCs equipped with wireless adapter cards, but with no connection to wired network. It can be used to quickly and easily setup a WLAN where no wired infrastructure is available, such as at a conference center or off-site meeting location. The client-server configuration typically consists of multiple PCs using wireless links to communicate with a central access point that is itself connected by cable to the backbone of the wired network [5]. In the past, the goal of telecom engineers was to provide better services at whatever costs. The costs were then being levied on the customer.

To this end, only the rich could afford these services. Over the years, there have been changes to this situation. The industry is driving to the positive direction where better services are being provided at very low charges to the customer. In addition, telecom companies have in recent years experienced a significant increase in number, which has led to a high level of competition amongst them. At the same time, the number of customers has also grown tremendously. Thus, there is the need for better management of resources such as optimization of the quality of the services they provide to these and other carrier customers. Trade-offs need to be made between costs, quality and priorities [8].

B. EXISTING SYSTEM

There are currently systems like Skype, Gtalk, which are useful for low cost communication. Skype for example allows free call to first fifty contacts. If we wish to have more than fifty contacts on same identity we need to pay tariff to Skype. In case we don’t wish to pay than we need to open new account with new identity. For companies second solution is not recommended. Also server of Skype, Gtalk are not accessible to administrator. For using these services we need to have access to net connection. It could be a costly affair for small companies. Installation and maintenance of wired LAN is tedious and costly affair.

Comparatively installation of WLAN is simple and quicker. Maintenance required is also less. Comparatively it is easier to troubleshoot. Hence we propose a wireless system for audio and video calls. The motive behind system is to enable the costeffective voice and video communication. We have designed a client server model based system to implement it. Our server being accessible to administrator it is easy for him to have control over system. Also there is no need for internet connection for working of this system. We have implemented the system using Android platform.

III. SYSTEM ARCHITECHTURE

A. WORKING

We have developed voice transferring from one mobile to another mobile using Wi-Fi see the fig 1.1.The basic concept behind this voice transferring is Open Voice communication Using Wi-Fi.User1 dial the mobile number of user2 then request send to the access point. Access point maps the dialed mobile number to IP Address and search for the connection. If the user2 is within the Wi-Fi range and user2 accepts connection request then virtual connection is establish between user1 and user2.

The system will allow users to search for other individuals within Wi-Fi range and to establish free voice connections, or to establish virtual connection through Access Points (AP) as well as giving the option to user to use GSM in the case of no Wi-Fi connectivity is available. The software will use a correlation between current address books available in mobile phones to convert phone numbers into IP addresses.

The first step of the system is to resolve the technical issue regarding mapping of the mobile user's phone number to a unique IP address in order to avoid IP collision, centralized control, and user configuration.

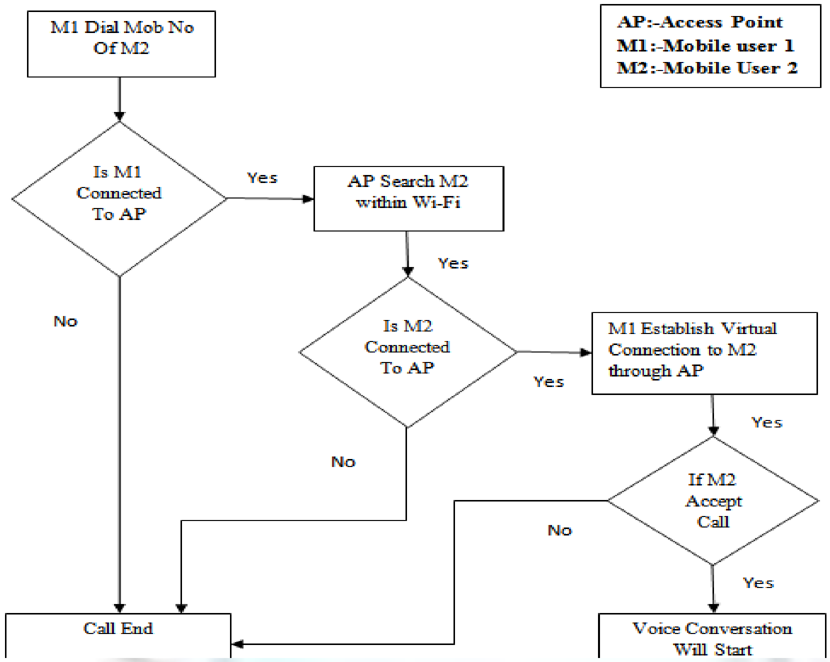

Fig 1.1: Mobile Communication Using WI-FI

IV. IMPLEMENTATION

A. SOCKET COMMUNICATION

Socket is a communication endpoint of a communication link managed by the transport Services. From the connection view, there are 2 types of socket communications, connection oriented and connectionless socket communications. For the first type, a connection must be set up before transmission. It is more reliable than connectionless socket communication. TCP is the transport protocol for the connection-oriented socket communication. For the second type, there are no connection setup, packet acknowledgement and retransmission services. UDP is the transport protocol for the connection-oriented socket communication. So, we choose TCP as our transport protocol. From the synchronization view, there are also 2 types of socket communications, asynchronous and synchronous socket communications. For the first type, the execution of the client application is not suspended while waiting for the server to return a response. For the second type, the execution of the client application is suspended until the server returns a response. For our real time voice communication, we choose asynchronous socket communications. There are 2 types of socket interfaces, Berkeley Socket Interface for UNIX system and Windows Socket. It is an interface, not a protocol. So, we can use different socket interface in the 2 ends of the communication links. In our project the both ends of the communication link.

V. VOICE TRANSMISSION

A. PULSE CODE MODULATION (PCM)

Analog transmission is not particularly efficient. When the signal-to-noise ratio of an analog signal deteriorates due to attenuation, amplifying the signal also amplifies noise. Digital signals are more easily separated from noise and can be regenerated in their original state. The conversion of analogue signals to digital signals therefore eliminates the problems caused by attenuation. Pulse Code Modulation (PCM) is the simplest form of waveform coding. Waveform coding is used to encode analogue signals (for example speech) into a digital signal[7]. The digital signal is subsequently used to reconstruct the analogue signal. The accuracy with which the analogue signal can be reproduced depends in part on the number of bits used to encode the original signal. Pulse code modulation is an extension of Pulse Amplitude Modulation (PAM), in which a sampled signal consists of a train of pulses where each pulse corresponds to the amplitude of the signal at the corresponding sampling time (the signal is modulated in amplitude). Each analogue sample value is quantized into a discrete value for representation as a digital code word. Pulse code modulation is the most frequently used analogue-to-digital conversion technique, and is defined in the ITU-T G.711 specification. The main parts of a conversion system are the encoder (the analogue-to-digital converter) and the decoder (the digital-to-analogue converter). The combined encoder/decoder is known as a codec. A PCM encoder performs three functions:

- sampling
- quantizing
- encoding

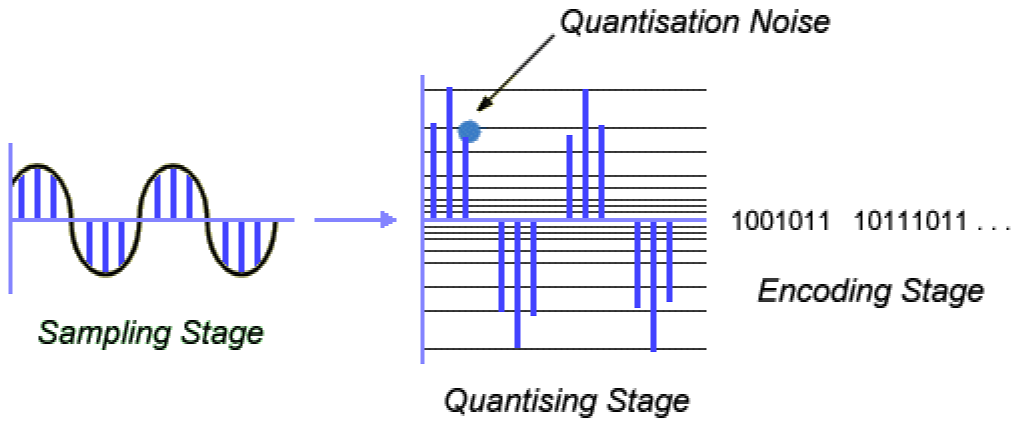

Fig 1.2: Stages in the analogue-to-digital conversion process

The human voice uses frequencies between 100Hz and 10,000Hz, but it has been found that most of the energy in speech is between 300 Hertz and 3400 Hertz - a bandwidth of approximately 3100 Hertz. Before converting the signal from analog to digital, the unwanted frequency components of the signal are filtered out. This makes the task of converting the signal to digital form much easier, and results in an acceptable quality of signal reproduction for voice communication. From an equipment point of view, because the manufacture of very precise filters would be expensive, a bandwidth of 4000 Hertz is generally used. This bandwidth limitation also helps to reducealiasing - aliasing happens when the number of samples is insufficient to adequately represent the analog waveform (the same effect you can see on a computer screen when diagonal and curved lines are displayed as a series of zigzag horizontal and vertical lines).

B. SAMPLING

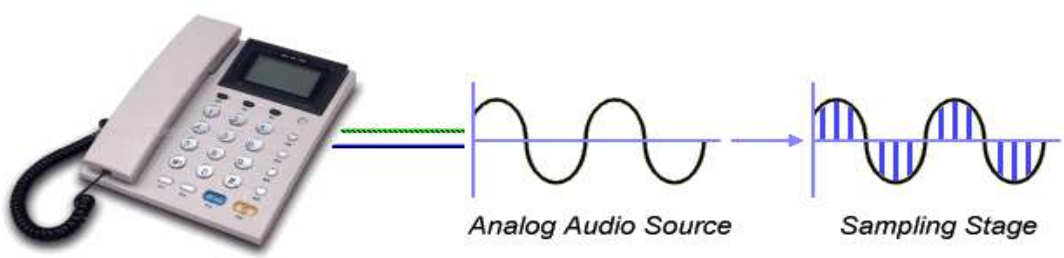

Fig 1.3: Sampling the analogue signal

Sampling is the process of reading the values of the filtered analogue signal at discrete time intervals (i.e. at a constant sampling frequency, called the sampling frequency)[7]. A scientist called Harry Nyquist discovered that the original analogue signal could be reconstructed if enough samples were taken. He found that if the sampling frequency is at least twice the highest frequency of the input analogue signal, the signal could be reconstructed using a low-pass filter at the destination. The first step in converting analog voice signals into digital is called sampling. The voice signal is sampled 8,000 times per second and each sample can be encoded in 8 bits. This produces a bit stream of 64,000 bits per second. This many samples are sufficient to reproduce the original sound accurately. The process of converting one sample into 8 bits is called “quantizing” because the infinite possible values of a voice sample must fit into one of 256 discrete values available for the digital byte (2^8=256). This process is called Pulse Code Modulation (PCM). The device that produces a digital signal from an analog one is called a codec, which is an abbreviation of code/decode. Normally a codec is embedded in a microchip called a digital signal processor (DSP). PCM produces a 64 kbps stream of data with excellent voice quality.

This process allowed long distance calls to be places on the T1 lines of the telephone company for transmission. One voice call takes up one channel, not a very efficient scheme. With VoIP, we want to cram as much voice data into as little digital signal as possible. And instead of diverting our digital voice signal directly onto a T1 line, we need to packetize it and send it over an IP network.

C. ENCODING

Encoding is the process of representing the sampled values as a binary number in the range 0to n. The value of n is chosen as a power of 2, depending on the accuracy required. Increasing n reduces the step size between adjacent Quantization levels and hence reduces the Quantization noise. The down side of this is that the amount of digital data required representing the analogue signal increases.

D. NOISE

In any communication system, the received signal will consist of the transmitted signal, attenuated as it has propagated along the transmission media and suffering from some distortion due to the characteristics of the system. In addition, unwanted signals (or noise) may occur between the transmitter and the receiver which are added to the transmitted signal. Noise is the main factor that limits the performance of a communications system.

VI. FEATURES AND APPLICATIONS

A. FEATURES

1. Unlimited Free Calling
2. No Internet connection
3. End User Can Make Call Using Same GSM Mobile Number.
4. User Friendly Interface.

B. APPLICATIONS

1. University/College campus
2. Organization campus area
3. Hospital campus area
4. Mining area
5. Five star hotels area

VII. FUTURE ENHANCEMENT

We have plans to modify this application for more practical and commercial use. Hence, we have listed some of the points as future enhancements as below:

- Video Conferencing Chat.
- Voice Conference
- File Sharing
- Text Messaging
- Automatic Conversion of Mobile number to IP Address & Vice-Versa.

VIII. CONCLUSION

Using Open Voice Mobile Telephony application it is possible to make a free voice call within Wi-Fi local area network. This voice communication doesn’t required Internet connection. User can call to another user within Wi-Fi range using same GSM mobile number.

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