Mobile Telephony Communication using WIFI P2P

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Abstract—The purpose of this paper is to design and implement a telephony program that uses Wi-Fi using Peer-to-Peer communication between mobile phones without any cost. The system will allow users to search for other individuals within the Wi-Fi range and to establish free P2P voice connections, or to establish virtual connection through Access Points (AP) in case of unavailability of the mobile device in the same Wi-Fi. In addition, we are using a novel algorithm to tackle the previous fundamental problem of designing Ad hoc and p2p telephony using Wi-Fi which will be independent of the service provider database. This algorithm always gives unique IP address for corresponding mobile number and vice versa. Thus, providing real feel of telecommunication without changing the behavior of user interface with the traditional mobile services. Further we are using SIP (Session Initiation Protocol) to create, modify and terminate multimedia sessions between two participants. This proposed paper will allow one call per connection, and no call waiting or conference calls. It can be further extended to support SMS (Short Message Service) as well as MMS using the same novel algorithm.

Keywords—WIFI p2p, AP, mobile telephony

I. Introduction

The main objective of this paper is to present how Peer-to-Peer based services can be efficiently realized in nextgeneration mobile networks. Currently, GSM and IP service provider provides services over mobile phones but at cost. Servicing IP addressing in traditional networks are managed by two technologies such as DNS and DHCP. They try to reduce the cost for supporting these services over mobile phones. Two approaches are suggested in this paper to meet the objective of having free telephony services over mobile phones. These are the use of WiFi technology over AP and WiFi over p2p (peer-to-peer). In addition a novel algorithm has been invented to tackle the first fundamental problem of designing Ad hoc and p2p telephony using WiFi, which will not rely on any central database, and will require users toregister to any service. This can be achieved through executing an algorithm to map a mobile number to a unique IP address that can be used to establish p2p connection to any other mobile phone running the same algorithm. For converting mobile number into IP address and vice versa. Here, IPv6 is used because IPv4 range is not sufficient for unique addresses.

The advances of VoIP and Intenet telephony in general have come a long way since their inception. Most recently, the "next big thing" has been to merge WiFi with VoIP, producing one of the oddest acronyms you'll ever see. VoWiFi, or Voice over Wireless Fidelity, simply means a WiFi based VoIP consists of the hardware and software that enables people to use the Internet as the transmission medium for telephone calls, VoWiFi is the wireless version of this technology that is designed to work on wirless devices such as a laptop or PDA. Some may wonder why a person or organization wouldn't simply use a cell phone for mobile communications, but again business and organizations can take advantage of a decreased



communications cost while having a mobile system that offers more reliable coverage indoors and higher voice quality than traditional cellular service with VoWiFi. Along with added benefits to business and those with a need for wireless communications, VoWiFi also opens up the door for a whole new market of consumer products such as a standalone VoWiFi handheld. Many cellular phone companies such as Nokia and Motorola have already announced dual-mode cellular phones that will support seamless roaming from WiFi to cellular networks when WiFi is unavailable to a caller. That is one of the biggest challenges facing VoWiFi roaming access. A WiFi access point offers a communication range upto 90 meters (commonly called a hotspot), and continuous conversations would mean that the caller must stay within an area of overlapping hotspots, or as already suggested, have a VoWiFi dual-mode phone that would switch to a regular cellular phone transmission when the caller moves out of a hot spot range.

Voice over WiFi telephony is a challenging research topic. The system is transparent to the user, where a user needs only to dial the required phone number the same way of using the normal mobile phone. Following the mobile conversion to IPv6, the software applied at the mobile phone will try to establish p2p connection to the dialed mobile phone using the same algorithm. If no p2p connection can be made, then the calling mobile phone will check if virtual connection can be possible through AP using the same mechanism in p2p to establish communication channel with the called mobile phone. If no wireless connection using WIFI is possible between the 2 mobile phones, then a message would be displayed on the calling mobile phone notifying its user to proceed with the call using GSM technology or abort the call. The IP collision problem for mapping mobile to IP has been avoided since there is a unique mapping resulting in a unique IP for each mobile number. The work presented in this paper is a first step for developing a p2p voice to voice communication between 2 mobile users using the WIFI network which is based on 10 mobile digit numbers.

п. Voice over WiFi (VoWiFi)

VoIP is the technology that allows the transmission of audio files by transmitting them into data packets across the Internet. By integrating Wireless and VoIP a new generation of audio telecommunications has been birthed. By having a VoIP service and a wireless connection, you can enjoy the best of both worlds- wireless VoIP connections. Wireless networks are activated by what is known as a hotspot. A hotspot is an area where there is an access point. Wireless connections are basically created by radio signals. An access point is where the network has established their main signal. It is possible for wireless users to basically jump from hotspot to hotspot and utilize various networks access points.

The combination of Wireless and VoIP has led to the invention - VoWiFi technology. VoWiFi stands for Voice over Wireless Fidelity. Many people are choosing the freedom that is offered by VoIP. VoIP can offer nearly free or free long distance phone calls. Since VoWi-Fi operates from hotspot to hotspot or network-to-network, you may think that there are roaming charges involved. There are no roaming charges involved with VoIP. So you can take your VoWiFi phone from hotspot to hotspot, maintaining your connection (provided you easily go from hotspot to hotspot) absolutely free.

ш. Modules

A module specifies the functionality of the system. Each module describes some specific task of the system. Generally software is made up of the different modules. This project divided into following modules:

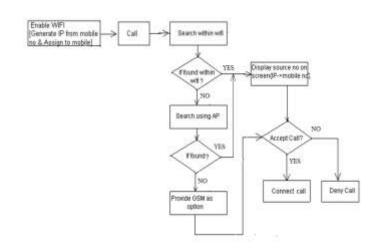
A. Enable WIFI

When the WIFI supported mobile will come into WIFI region then we have to enable WIFI of that mobile. At that time IP address is generated from mobile number by using novel algorithm. That IP address is assigned to that mobile.

B. Call

When caller dials the number (mobile number of receiver), at that time IP address is generated from dialed number and

and the broad cast request for connection.



c. Search

For connection different searches are used. Initially caller mobile will search receiver mobile is present in WIFI region directly. If receiver is not found in that region then it will take the help of AP (access point). With the help of AP the caller mobile will search receiver mobile. If in both cases receiver mobile is not found then system provides GSM option for connection.

D. Display number on screen

When the receiver mobile is found then at receiver side source number is displayed on the screen with ring. That source number is generated from source IP address using novel algorithm. Novel algorithm is used for convert mobile number into IP address and IP address into mobile number.



E. Provide GSM option

If receiver mobile is not found within WIFI directly or with the help of AP then this new option is provided to the caller. This is provided for call as like the way of tradition call. In that call method charges are applied by GSM services provider. This option is enabled means receiver is not within the source's WIFI network.

F. Connection

If receiver mobile is found to caller then number is displayed on the receiver screen then receiver mobile user has options like accept call or reject call. If he accept the call then the connection will established between caller and receiver mobile. If that connection within WIFI network then it is free of cost communication and if it is using GSM service then charges are applied by GSM service provider.

G. Reject Call

Reject call means break connection or avoid connection. When phone rings and call is rejected means connection is not

established and when call is rejected while the communication is going on is the break in connection. Connection can be closed by either side as caller or receiver side.

IV. Algorithm for mapping

There are two algorithms which are used to mapping between IP address and mobile number. For MobToIP (string number) input is mobile number and output is unique IP address and for IPToMob (string address) input is IP address and output is mobile number.

A. MobToIP (String mobilenumber)

Mobile number should be 10 digit integer and I P address is 16 Hexadecimal integer. Conversion of mobile number to IP address is done by this algorithm. Steps of this algorithm is as follow:

/* Variables:

IPaddress - is character array and initialized as

IPaddress \leftarrow [,,0", ,,0", ,,0", ,,0", ,,:"... ,,:" ,,0" ,,0" ,,0" ,,0"] Mobileno - is character array and it will take input as mobile number */

MobToIP(string mobilenumber)

```
{
```

```
1. i←0
```

{

```
2. read mobile number character by character
```

3. while(mobile number has not finished)

```
i. if(IPaddress[i]=0) then
```

{

a. Copy mobile number character into IPaddress[i]b. Read next character of mobile number

b. Read next character of mobile number

```
}//end if
```

ii. Increment i

```
} //end while
```

4. Return IPaddress }//end function

B. IPToMob (String Address)

```
This algorithm converts IPaddress into mobile number. It is
required to display on receiver' s mobile screen.
/* Variables:
IPaddress as input
Mobileno is initialized to null
String1, String2←null
IPToMob(String address)
{
 1. String1←Reverse of IPaddress,
 2. i←0
 3. While(String1 is not null)
    {
     i. If (String1[i] = 0' \text{ or } String1[i] = ':') then
         a. String1[i]←'*'
        Else
         b. Break
        }
    }//end of while
 4. String2←Reverse of String1
 5. i←0
 6. i←0
 7. While(j<=length of mobile number)
     i. If(String2[i]!= '*' and String2[i]!=':')
         a. String1[j]←String2[i]
         b. Increment j
      Else if(String2[i]='*')
        a. String1[j] \leftarrow '0'
        b. Increment j
       ļ
     ii. Increment i
 8. String1[j]←null
```

9. Mobileno←String1

10. Return Mobileno

10. Ketuili w

v. Why this particular algorithm?

A software solution was developed in order to convert mobile numbers to IP addresses and vice versa. In this program development, it is possible to map the mobile numbers to a valid IPv6 address, and therefore there is no need for DNS lookup. In addition, there is no need for complicated



hashing and addressing protocols because the IP-to-Mobile algorithm would produce a unique number used as input, leading to a unique output hexadecimal IPv6 address. The outputted address is then allocated to private IP within a specified range in order not to conflict with other devices in the same wireless range.

vi. Working with wifi

The issue while working with WiFi is the radio selection in a multi-radio device. Mobile devices are nowadays equipped with several radios that support packet data communications. In addition to their long-range cellular radios (e.g., GSM and 3G/WCDMA), they often have short-range radios (e.g., Bluetooth), and medium range radios (e.g., IEEE 802.11). The mobile device should be able to select the best radio according to the situation, i.e., use a long-range radio when the device is on the move, and use a short or medium-range, highbandwidth radio when it is in stationary.

vп. Sip

SIP, the session initiation protocol, is the IETF protocol for VoIP and other text and multimedia sessions, like instantmessaging, video, online games and other services. SIP is very much like HTTP, the Web protocol, or SMTP. Messages consist of headers and a message body. SIP message bodies for phone calls are defined in SDP – the session description protocol. SIP offers all potentialities of the common Internet Telephony features like:

- 1) Call or media transfer
- 2) Call conference
- 3) Call hold

Since SIP is a flexible protocol, it is possible to add more features and keep downward interoperability. SIP can be regarded as the enabler protocol for telephony and voice over IP (VoIP) services. The following features of SIP play a major role in the enablement of IP telephony and VoIP:

A. Name Translation and User Location:

Ensuring that the call reaches the called party wherever they are located. Carrying out any mapping of descriptive information to location information. Ensuring that details of the nature of the call (Session) are supported.

B. Feature Negotiation:

This allows the group involved in a call (this may be a multi-party call) to agree on the features supported recognizing that not all the parties can support the same level of features. For example video may or may not be supported; as any form of MIME type is supported by SIP, there is plenty of scope for negotiation.

c. Call Participant Management:

During a call, participant can bring other users onto the call or cancel connections to other users. In addition, users could be transferred or placed on hold.

D. Call Feature Changes:

A user should be able to change the call characteristics during the course of the call. For example, a call may have been set up as 'voice-only', but in the course of the call, the users may need to enable a video function. A third party joining a call may require different features to be enabled in order to participate in the call.

E. Media Negotiation:

The inherent SIP mechanisms that enable negotiation of the media used in a call enable selection of the appropriate codec for establishing a call between the various devices. This way, less advanced devices can participate in the call, provided the appropriate codec is selected.

viii. Constraints on mobile

- 1) The first constraint of mobile environment is the limited network bandwidth available for the mobile device.
- The second constraint is the limited computational power in mobile devices. Thus the peer-to-peer application should not have computationally intensive algorithms or use large data structures.
- 3) The 3rd constraint is the limited battery capacity of the mobile device. By limiting bandwidth use and computationally intensive algorithms in the application the battery can be conserved.

As we consider these technical constraints in the design of mobile peer-to-peer application, we should select a mobile peer-to-peer architecture that creates a minimal signaling load on the mobile peer, uses no complex algorithms or data structures, and has efficient protocol coding.

IX. Future work

Future work will focus of the development of automated algorithm to develop IP collision avoidance and correction for large mobile. The number of businesses using voice over wireless local area networks (VoWLAN) is set to triple over the next two years. The release of the voice API by the giant mobile manufacturers will enable future complete working system based on the architecture initiated by this research work. In addition, Integration of video, data, audio, and web browsing capabilities on wireless phones. Concentrate on the security issues.

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References

- Ghassan Kbar, Wathiq Mansoor, Aryan Naim, "Voice Over IP mobile telephony using WIFI p2p", 2010 Sixth International Conference on Wireless and Mobile Communications
- [2] B. Regis J, and D. W. Gregory, "Voice and Data Communication", 4rth ed. California, Berkeley: McGraw Hill, 2001
- [3] GSM, http://www.gsmworld.com/technology/gsm/index.htm
- [4] WIFI IEEE 802.11, http://www.wifinotes.com/IEEE-802.11.html
- [5] Mark A. Miller, P.E., "Voice over IP Technologies: building the converged network". Wiley-dreamtech India Pvt. Ltd.
- [6] www.wifinotes.com/
- [7] www.p2psip.org

