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A Review of Voice over IP Mobile Telephony Using WIFI

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Abstract— Voice telephony over mobile is currently supported at a cost using service provider such as GSM, or using IP service provider at cheaper cost. The purpose of this research is to design and implement a telephony program that uses WIFI in p2p (Peer -to- Peer) or WLAN (Wireless Local Area Network) as a means of communication between mobile phones at no cost. The system will allow users to search for other individuals within WIFI range and to establish free p2p 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 WIFI connectivity is available. The system will use a novel algorithm to convert mobile number into IP address and use it as a mean for contacting other mobile over p2p or AP using WIFI technology. The software will use a correlation between current address books available in mobile phones to convert phone numbers into IP addresses. The system will allow user to make voice conversation, sending SMS (Short Message Service) as well as MMS. Inbox and outbox services, message delivery reports, and message drafts will be used for SMS and MMS management. The current system will only allow for one call per connection, and no call waiting, or conference calls.

Keywords-VOIP; Peer to peer; SIP

I. INTRODUCTION

The support of telephony services over mobile phone has been used everywhere using technology such as GSM (Global System for Mobile) and 3rd Generation mobile telecommunication 3G , but at high cost. On the other hand, IP telephony try to reduce the cost for supporting this service over mobile phone, but it is facing difficulties since the same feature is supported on desktop and laptop at lower complexity. The challenge is to provide the ame service over mobile phone at no cost, as it has been described in this paper. 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 not require users to register 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. 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 [1]. P2p 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 [2]. 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 [3].

Currently, servicing IP addressing in traditional networks are managed by two technologies, the DNS (Domain Name System), and DHCP (Domain Host Configuration Protocol) [4]. DNS Servers resolve human friendly domain names to IP addresses for computers and resources on the Internet globally. DNS keeps website addresses consistent regardless of the physical location or routing protocol [4]. DHCP helps to make automatic network configuration, IP address allocation, for network devices. Whenever a new device is connected to the network the device will request for an IP address from the server, which will allocate the address to the networked device for a specific time period , where dynamic network addressing is used [5][6]. The DNS mechanism cannot be applied to p2p Ad hoc network, and therefore a better solution should be used such as the one s paper, which is based on WIFI technology. The first stage of developing the voice 802.11 (Wireless Fidelity) application problem of developing a method that could addresses to mobile devices on the flinter action and central management [7]. GSM mobile phones (Subscriber Identification Modules) cards t users uniquely in GSM networks.

A telephone call is between two parties - the calling party (or caller) and the called party(or callee) who are connected by one or more switches at various carrier companies' exchanges. These switches form an electrical connection between both end-users, and their setting is electronically determined by pulses or tones generated by the dialed number. When a connection is established, and caller and called subsequently go in speech, their voices are transported as analogue and digital signals between the switches in the network. In order to successfully realize this process, the telecom exchange companies are charged for this. Each time a number is dialed, each of these companies sees it as an attempt, which may either be successful or a failure. They make a very small profit margin for each successful call but rely on the minutes generated by the huge amounts of successful calls in order to make a noticeable profit.

II. RELATED WORK

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. 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 cost effective 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 has implemented the system using JAVA which is platform independent.

Communication has been of prime importance to man since ancient time. Various methods have been deployed communication. In early days of voice transfer PSTN networks were used. These consisted of Private Branch Exchange office owned by service providers. These were wired network a copper line connected subscriber home to local office. Local offices were further connected in hierarchies order. Switching was done through hardware like trunk lines processors. Setting up was tedious, time consuming, and costly affairs. Copper wires need to lay down to customer premise. Switching offices need to be setup and hierarchical network backbone need to be established to route calls. All these required lot of time and efforts also this network could not support video traffic.

Wired LAN was later employed to transfer voice and video over local area network. It consisted of many configurations like 802.3, 802.4 etc. As it was wired systems connected through it lacked mobility. Also configuring LAN required time. Wires need to be setup to individual PC's. Troubleshooting and maintaining this network was great trouble. Also topology like star could bring whole network down if central hub fails. The wired media in LANs are dominated by a variety of UTP and STP to support a range of local data services from several Mbps up to over a gigabit per second within 100 meters of distance. The early LANs were operating on the so called thick cable to cover up to 500 meters per segment. IN the LAN applications fiber lines mostly serve the backbone to interconnect servers and other high-speed elements of the local networks.

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. 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.

III. PROPOSED WORK

The proposed work includes the following.

1. The application on implementing SIP-based VoIP applications for Smartphone OS such as Android mobile.
2. The purpose of this application is to implement a telephony program that uses WIFI in Peer to-Peer or WLAN (Wireless Local Area Network) as a means of communication between mobile phones at no cost. The system will allow users to search for other individuals within WIFI range and to establish free peer to peer connection for voice communication and also for file transfer and chatting.
3. Voice over Internet Protocol is used for communication of two persons by sending voice packets in a real time fashion. Various protocols are involved in implementing VoIP.
4. The tasks are divided into two. The major task is to establish a session between the two communicating parties. The protocols involved in establishing the session are called as Control plane protocols. Session Initiation Protocol and H.32 are some of the control plane protocols. These protocols are also called as signaling protocols as they are used to establish sessions between the users. Due to various advantages which are offered by Session Initiation Protocol (SIP), it has been majorly adopted by the telecommunication industry. One of the main advantages of SIP is that it is human readable and is less complex when compared with H.323 which is mainly binary. So, in this application we implemented SIP as our signaling protocol.

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IV. VOICE OVER INTERNET PROTOCOL

The basic idea of our approach is to make voice call. Describes our mobile VOIP and SIP protocol. In the beginning, the mobile A and mobile register itself to the server for the service. Both A and B mobile have the unique IP address and ID. When mobile a tries to call B mobile A sends the request to server where sever check the IP address of mobile B and sends the IP address of mobile B to mobile A for peer to peer connection. After receiving the IP address of mobile B, mobile A makes the peer to peer connection for the voice call using protocols VOIP and SIP.

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V. IMPEMENTATION

Wi-Fi phone works by accessing wireless Internet connections such as a wireless router in your home or office, or Wi-Fi hotspots around the globe. You can access open Wi-Fi hotspots quickly and easily, as well as various secure hotspots. Voice over Internet Protocol (Voice over IP, VoIP) is a family of technologies, methodologies, communication protocols, and transmission techniques for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms frequently encountered and often used synonymously with VoIP are IP VoIP is available on many smartphones and Internet devices so that users of portable devices that are not phones may place calls or send SMS text messages over 3G or Wi-Fi. On the receiving side, similar steps (usually in the Reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the Original voice stream. Telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone. It works at many hotels, airports, coffee shops, and more! This innovative wireless Internet phone provides portable, high-quality and low-cost plans.

Fig Voice over internet protocol

SIP-based VoIP Service

The session initiation protocol (SIP) is the Internet standard signaling protocol for setting up, controlling, and terminating VoIP sessions1. SIP-based VoIP services require infrastructure support from entities such as SIP registrars, call proxies, and so forth (see Fig. 1) – we collectively refer to these entities as SIP servers. A SIP registrar associates SIP users (e.g., names or identities called SIP URIs) with their current locations (e.g., IP addresses). A SIP call proxy assists users in establishing calls (called dialogs in the SIP jargon) by handling and forwarding signaling messages among users (and other SIP servers). In practice, a physical host (SIP server) may assume multiple logical roles, e.g., functioning both as registrars and call proxies. SIP is a text-based request-response protocol, with syntax very similar to HTTP. SIP messages are of type either request or response. The method field is used to distinguish between different SIP operations. The most common methods include REGISTER (for user registration), INVITE, ACK, BYE, CANCEL (these four used for call set-up or tear-down), SUBSCRIBE, and NOTIFY (for event notification). Response messages contain a response code informing the results of the requested operations (e.g., 200 OK). The FROM and TO fields in an SIP message contain respectively the SIP URIs of the user where a request message is originated from (e.g., the caller of a call) or destined to (e.g., the callee of a call).

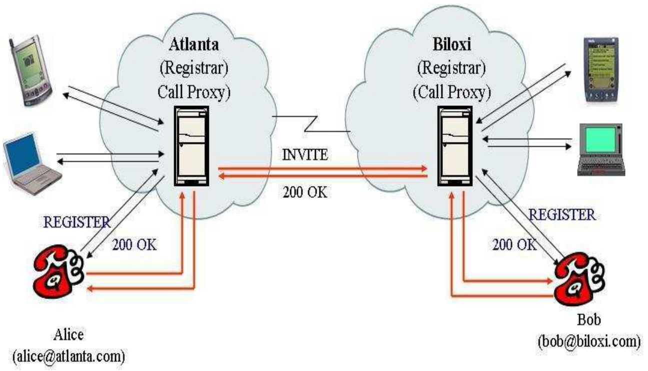

Fig SIP servers and clients

in the case of a REGISTER message, both FROM and TO typically contain the SIP URI of the user where the request is originated. Other important fields include VIA and various identifiers and tags to string together various transactions

CONCLUSION

It can be possible to design a application for Smartphone OS such as Android mobile by which we can communicate with other using SIP-based VoIP. The purpose of this application is to implement a telephony program that uses WIFI in Peer to-Peer or WLAN (Wireless Local Area Network) as a means of communication between mobile phones at no cost. The system will allow users to search for other individuals within WIFI range and to establish free peer to peer connection for voice communication and also for file transfer and chatting.

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. Future work will focus of the development of automated algorithm to develop IP collision avoidance and correction for large Mobile digit numbers. Furthermore, a complete solution to establish communication channel between 2 mobile phones using p2p, or through AP, and optional GSM will be developed. In addition, 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.

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