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 Are Network) as a means of communication between mobile phones at no cost. 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. Different security services relevant for VoIP are presented and we argue that end-to-end authentication and encryption should be provided by default. For media protection we evaluate the possibility of using either SRTP or IPsec, and we examine several alternatives of how a secure VoIP call can be established. The solution we suggest is based on SRTP for media protection, S/MIME and MIKEY for end-to-end authentication and keying, and TLS for hop-by-hop protection of SIP messages.

Keywords-VOIP; Peer to peer; SIP

1.INTRODUCTION

Voice over Internet Protocol (VoIP) is a form of communication that allows you to make phone calls over a broadband internet connection instead of typical analog telephone lines. Basic VoIP access usually allows you to call others who are also receiving calls over the internet. Interconnected VoIP services also allow you to make and receive calls to and from traditional landline numbers, usually for a service fee. Some VoIP services require a computer or a dedicated VoIP phone, while others allow you to use your landline phone to place VoIP calls through a special adapter.

VoIP is becoming an attractive communications option for consumers. Given the trend towards lower fees for basic broadband service and the brisk adoption of even faster internet offerings, VoIP usage should only gain popularity with time. However, as VoIP usage increases, so will the potential threats to the typical user. While VoIP vulnerabilities are typically similar to the ones users face on the internet, new threats, scams, and attacks unique to IP telephony are now emerging. Voice over Internet Protocol (VoIP) is a technology for communicating using "Internet protocol" instead of traditional analog systems. Some VoIP services need only a regular phone connection, while others allow you to make telephone calls using an Internet connection instead. Some VoIP services may allow you only to call other people using the same service, but others may allow you to call any telephone number - including local , long distance, wireless and international numbers. 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 . 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. 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.

Currently, servicing IP addressing in traditional networks are managed by two technologies, the DNS (Domain Name System), and DHCP (Domain Host Configuration Protocol). DNS Servers resolve human

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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. 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. 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. 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 callee 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.

2. LITERATURE SURVEY

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.

Voice over Internet Protocol (VoIP) has been popular in the telecommunications world since its emergence in the late 90s, as a new technology transporting multimedia over the IP network. In this book, the multimedia (or rich media) includes not only voice, but also video, instant message, presence data, and fax data over the IP network. Today people commonly make phone calls with IP phones or client software (such as Skype or iChat) on their computer, or send instant messages to their friends. This gives them convenience and cost savings. 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. 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.

3. PROPOSED WORK

The proposed work includes the following.
The application on implementing SIP-based VoIP applications for Smartphone OS such as Android mobile.
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.
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.
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.
This application on implementing SIP-based VoIP applications for Smartphone OS such as Android mobile. 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.

4. 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.
VoIP converts the voice signal from your telephone into a digital signal that can travel over the Internet. If you are calling a regular telephone number, the signal is then converted back at the other end. Depending on the type of VoIP service, you can make a VoIP call from a computer, a special VoIP phone, or a traditional phone with or without an adapter. In addition, new wireless "hot spots" in public locations such as airports, parks and cafes allow you to connect to the Internet, and may enable you to use VoIP service wirelessly. If your VoIP service provider assigns you a regular telephone number, then you can receive calls from regular telephones that don't need special equipment, and most likely you'll be able to dial just as you always have. VoIP may offer features and services that are not available with more traditional telephone services. If you use VoIP, you can decide whether to pay the cost of keeping your regular telephone service. You can also use your computer and VoIP service at the same time. You can also take some VoIP services with you when you travel and use them via an Internet connection.

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Fig:- VOIP

H.323, a protocol suite defined by ITU-T, is for voice transmission over internet (Voice over IP or VOIP). In addition to voice applications, H.323 provides mechanisms for video communication and data collaboration, in combination with the ITU-T T.120 series standards. H.323 is one of the major VOIP standards, on a par with Megaco and SIP. H.323 is an umbrella specification, because it includes various other ITU standards. The components under H.323 architecture are terminal, gateway, gatekeeper and multipoint control units (MCUs). Terminal represents the end device of every connection. It provides real time two way communications with another H.323 terminal, GW or MSU. This communication consists of speech, speech and data, speech, and video, or a combination of speech, data and video. Gateways establish the connection between the terminals in the H.323 network and the terminals belonging to networks with different protocol stacks such as the traditional PSTN network or SIP or Megaco end points. Gatekeepers are responsible for translating between telephone number and IP addresses. They also manage the bandwidth and provide a mechanism for terminal registration and authentication. Gatekeepers also provide services such as call transfer, call forwarding etc. MCUs take care of establishing multipoint conferences. An MCU consists of a mandatory Multipoint Control, which is for call signaling and conference control, and an optional Multipoint Processor, which is for switching/mixing of the media stream and sometimes real-time transcoding of the received audio/video streams

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. 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.

5. 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.

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8. References

B. Regis J, and D. W. Gregory, “Voice and Data Communication”, 4rth ed. California, Berkeley: McGraw Hill, 2001.

F. Borko, and M. Ilyas, “Wireless Internet Handbook Technology, Standards, and Applications”, 1st ed. NY, New York: McGraw Hill, 2000.

O. Tammer, and P. Valduriez, “Principles of Distributed Database Systems”, 2nd ed. New Jersey: Prentice Hall, 1999.

] C. Douglas, “Internetworking with TCP/IP Volume 1 : Principles, Protocols, and Architecture”, 3rd. Upper Saddle River, NJ: Prentice Hall International, 1995.

Hac, and Anna, “Mobile Telecommunication Protocols for Data Networks”, 1st. London: John Wiley & Sons LTD, 2000.

J. Distefane, and A. Ronan, “Guide To Wireless Enterprise Application Architecture”, 1st ed. NY, New York: John Wiley & Sons Inc, 2002.

William Lee, “Wireless and Cellular Telecommunications. 3rd”, New York, NY: McGraw Hill, 2006

ISSN 2229-5518

S. Gast, and Mathew, "802.11 Wireless Networks the Definitive Guide", 1st. California: Orielly, 2000.
Taylor D. Edgar, "TCP/IP Complete", 1st. New Jersey: McGraw Hill, 1998.
A. Forouzan Behrouz, "TCP/IP Protocol Suite", 3rd. NY: McGraw Hill, 2006.
D. Sanjoy, C. Papadimitriou, and U. Vazirani, "Algorithms", 1st. New York: McGraw Hill, 2008.
Yuan Michael, "Enterprise J2ME: Developing Mobile Java. Applications", 1st. Upper Saddle River, NJ: Prentice Hall, 2004.
GSM, http://www.gsmworld.com/technology/gsm/index.htm.
WIFI IEEE 802.11, http://www.wifinotes.com/IEEE-802.11.html.