INEXPENSIVE TELEPHONE ACCESS TO MOBILE USERS USING VOIP

Jan Newmarch and Ning Xia

School of Network Computing, Monash University

Abstract
Voice over IP is becoming a popular choice for organisations to use internally and externally. Most VoIP systems are large-scale in nature and do not address the needs of mobile users. This project reports on a low-cost system to support a mobile user while maintaining a fixed phone number. It uses VoIP to combine the best features of mobile phones and internet data transport.

1. Introduction

People are becoming accustomed to carrying a variety of devices with them, particularly if aspects of their work involve mobility: mobile phone, pager, PDA, laptop, etc. They use a variety of network connections, such as GPRS, WiFi, Bluetooth, and sometimes fixed connections such as ethernet or dialup. Some of these devices are converging and containing a number of different functions, such as the blueberry which can manage email and phones, and third generation mobile phones which can browse web pages as well as deal with voice and image messages.

Connections through the cellular network are typically quite expensive compared to local phone calls or even to long-distance calls. These costs become significantly higher when compared to data transmission costs through a local network (free) or the internet.

The principal advantages of a mobile phone are that it gives a fixed phone number and is (nearly) always in reach wherever the user is located. A mobile phone gets its value due to the widespread cellular network, which ensures that a person can be reached wherever they are.

The spread of wired and wireless networks means that a person with a suitable device such as a laptop or PDA will increasingly be accessible via the internet. This raises the possibility that computing devices connected to the internet can be used instead of mobile phones, with a corresponding reduction in call costs.

This paper discusses a project in which we seek to combine the advantages of a mobile phone with the advantages of cheap data networking. We give the advantage of fixed phone number by using a fixed phone connection, typically the phone point in a user's office or home. This is connected by modem to a PC on the network, and then voice over IP [2] is used to connect to the person's laptop or PDA. Essentially, the laptop or PDA becomes a mobile "softphone" connected to the PSTN network through a fixed access point.

This is considerably cheaper than using a commercial PSTN/VoIP gateway which typically connects an organisation's switchboard into the IP network - we are using a simple voice modem to do this connection on an individual basis.

The structure of this paper is as follows. In the next section we discuss current VoIP systems and how they fail to meet the requirements outlined above. This is followed by a quick overview of some relevant technologies. We then discuss our system design and implementation and conclude with a summary of results. The major contribution of this paper is that it shows that it is possible to build lightweight VoIP systems that are more flexible than large corporate systems, and in particular can meet the needs of users who are mobile within the organisation.

2. PSTN/VoIP Systems

Current VoIP systems are usually implemented on an organisational basis. In all cases the intent is to allow PSTN and internet systems to be combined through some sort of gateway as in Figure 1 [4]. The systems use differing technologies depending on scale. For example, a CISCO IP gateway costs about US$25,000 while an IP PBX is often US$150,000.

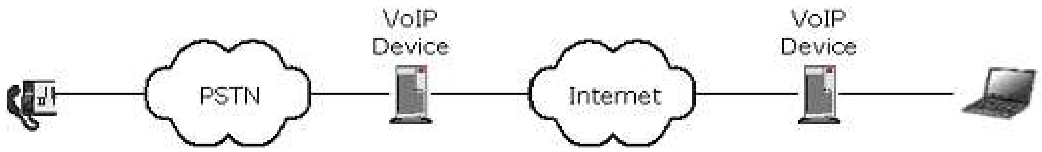

Figure 1 Current VoIP systems

All current systems are designed for an organisational scale and aim to allow a company to realize VoIP and POTS communication functions, not for a common user. Thus costs on all of above systems are expensive. For example, a Cisco IP gateway is about US$25,000 and IP PBXs price are often US$150,000. These organisational systems are designed to replace or augment existing phone systems, often by replacing the voice network with a data network that is required anyway for the corporate network. The systems are not oriented to the mobile user and there are no mechanisms in place to support mobility.

3. Voice modems

A modem is a communication device between DTE (Data Terminal Equipment) and analog telephoneline [3]. Currently, most high speed modems support data and voice transmission. To communicate with a modem, users can use Hayes AT Commands [3]. In this project, a NetComm Wave V.92 Data/Fax/Voice modem is used, which supports TAM voice commands ('+' voice commands) following IS-101 Voice Control Interim Standard for Asynchronous DCE. Modems seem to be inconsistent in their support for these standards.

4. System design

The parameters controlling our design were

4.1 Low cost

Currently, most VoIP systems are expensive and only suitable for corporations. Most of the cost of installing and running a VoIP system is spent on VoIP devices and upgrade of previous equipment. A recent report shows that cost is still a main barrier for companies to adopt VoIP system [3]. Therefore, low-cost is the first issue considered in this system.

While current systems provide a gateway between the organisational PBX and the IP network, we want a modem plus software gateway to function at an individual level. Hence, a user only needs to buy a modem which is about $100, while a voice gateway could be about US$25,000.

4.2 Mobility

This project arose from a user requiring a single fixed phone number while being mobile among a number of offices. In each of these offices a connection to the internet could be made (but typically on different subnets), and calls to the fixed line in the base office need to be sent to wherever the user happens to be at the time. Similarly, calls from the user need to be sent out through this one fixed line.

In concept, this is similar to the support for mobile devices offered by IPv6: packets sent to a base station address are forwarded to the current actual address. In this case, we are restricted to an existing IPv4 network. In a real sense, we are copying the IPv6 mobility mechanism into an application layer solution.

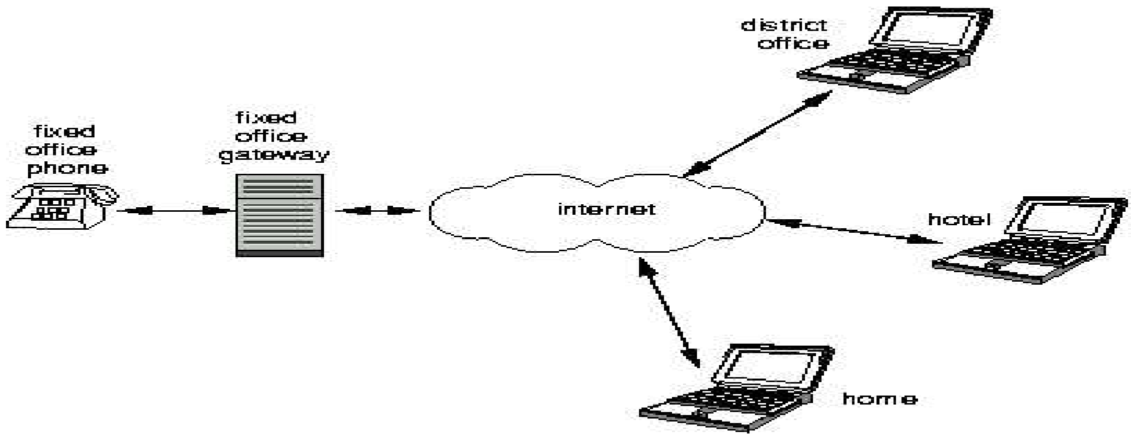

Figure 2 Mobility Supported by the System

The system is also able to offer other conveniences such as voicemail. Instead of being recorded by the organisational PBX, it will need to be recorded on an internet device, to be accessible from anywhere.

4.3 Decentralised structure

To realize the decentralized structure feature of this system, a gateway only serves a single user and binds with a telephone line. As mentioned above, a modem is a suitable device to be used in this system because it costs less than common VoIP devices. While this provides a fixed entry point into the network for each modem, each entry point should be independent and not based on a centralised structure. This requires no central infrastructure, which typically leads to expensive systems that take time to put in place.

4.4 Gateway and user decoupled

Since the system adopts a decentralized structure, the gateway and user can be anywhere on the Internet. Therefore, the way to locate a users current location becomes a important problem to be solved. A gateway's position will be fixed, while the user's position is mobile. Hence, a user will need to report and update his current location to its gateway. SIP [5] defines a register method which allows a user to register his current location or notify the gateway if the user moves to a new place. The register mechanism solves the users real-time position problem perfectly. These considerations lead to the design shown in Figure 3

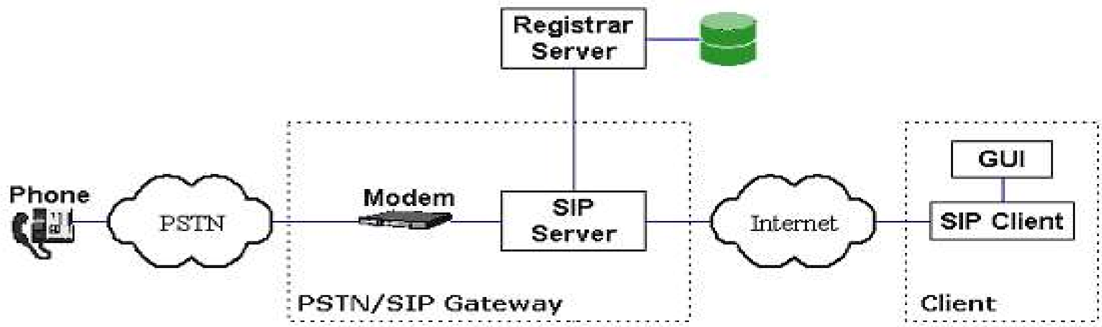

Figure 3 Modem Based Light-weight VoIP System

5. Implementation

Separate modules were written to handle the modem, act as registrar and supply user-side interface (extended softphone). All modules were written in Java. The modem module uses the serial port commAPI to control the modem by AT commands and receive commands from it, and additionally to communicate with the registrar and client. The registrar acts as a SIP server and additionally maintains information about the client's current location in the nework. This information is leased to prevent it getting too stale. The client-side module controls the sound card to make and receive calls, and also interacts with the registrar and modem .

When a call comes into the modem it checks to see if the user's location is currently registered with the registrar. The registrar will then attempt to set up a SIP session between the modem and client. Voice data is transmitted acrosss the network using RTP using G723.1 over the Java Media Framework. Java Sound is used for dealing with the soundcards.

In addition, a voicemail server was also set up, and this interacts with the other modules so that a SIP sesion can be set up with the modem if the user is not available, and the user can later access voice data on this server using JMF.

6. Results and Evaluation

System usability: The system has been tested and works on Windows XP and on RedHat Linux 9.0. Some brands of soundcards are not supported by Java Sound 1.4.1 on Linux, which may restrict use.

Cost saving: An economical lightweight VoIP system is this project's aim, so cost saving for a single user is an important point to achieve. There are only two devices required on the server side: a modem and PC. On the client side, only a digital terminal such PC or laptop is required, preferably with headphones and a reasonable quality microphone. Compared to normal VoIP solutions, using a voice gateway and digital terminal on both sides, the cost saving is substantial. However, this would not scale up to an organisational solution.

Mobility: Multiple-platform support and a registry mechanism satisfies the mobility requirement. Users can move from one place to another place without worrying about missing their telephone calls. Users only need to change their registered location and then they can answer their calls at their new location. Moreover, calls can be recorded as voicemails when users are unable to answer calls.

Voice quality: An acceptable quality of voice is expected in this system even though the system is just for common users use. The quality of voice during two way RTP communication between the software gateways is clear and not too noisy under normal situations. Some extreme conditions are not supported well such as transmitting music because G.723 is not well-suited to music or sound effects.

7. Conclusion

Nowadays, VoIP technology draws more attention than before due to its superior features of cost saving and other advantages. Most VoIP systems, however, are expensive and designed for large corporations. To offer mobile users an economical and convenient VoIP system, a modem based lightweight VoIP is discussed in this paper. The project takes advantage of current readily available equipment and lowers change or upgrade fees. In details, the system uses current existing telephone line, PC, a Data/Voice modem and a software PSTN/SIP gateway instead of a dedicated hardware gateway. Therefore, the total cost of the system is low. The system adopts a decentralized structure to provide mobility to users. The gateway in this system can be set up anywhere on the Internet with a connection to a telephone line. Also, a user can use this system from anywhere which offers network connectivity.

8. References

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