Palestine Polytechnic University

College of Engineering & technology

Electrical and Computer Engineering Department

Graduation Project

Implementing Voice Over Internet Protocol (VoIP) In PPU

Project Team

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Abstract

This project aims to apply the technique of communication via the Internet on the campus of the Palestine Polytechnic University with relying on the infrastructure and communications network at the University from exchanges , telephone lines and fast Internet lines etc… .

The main objective of the project is to reduce the high costs of communications at the university through the study of the options available and recommend the best.

We use in this project a modern technology device called Analog Telephone Adapter (ATA), which is used for the transport of voice over the Internet.

Finally in our project we seek to reducing the calling cost in the PPU campus to 90% of the current cost .

ملخص

يهدف هذا المشروع إلى تطبيق تقنية الاتصال عبر الانترنت في حرم جامعة بوليتكنك فلسطين بالاعتماد على البنية التحتية وشبكة الاتصالات الموجودة في الجامعة من مقاسم وهواتف وخطوط انترنت سريعة الخ...

إن الهدف الأساسي للمشروع هو تقليل التكاليف الباهظة للاتصالات في الجامعة من خلال دراسة الخيارات المتاحة والتوصية بأفضلها .

نستخدم في هذا المشروع تقنية حديثة تعتمد على جهاز Analog Telephone Adapter (ATA) والذي يستخدم لنقل الصوت عبر الانترنت .

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Abbreviations

ADSL: Asymmetric Digital Subscriber Line

ATA: analog telephone adaptor

CPU: Central processor unit

DTMF: Dual-tone multi-frequency

FXS : Foreign EXchange Station

HFAI: Hands-free answer on intercom

http: Hypertext Transfer Protocol

IAX2: Inter-Asterisk Exchange

IP internet protocol

ISP: Internet service providers

LAN: local area network

LED : Light-emitting diode

MG: Media Gateways

MGC: Media Gateway Controller

MGCP: Media Gateway Control Protocol

ODM: original device manufacturer

OS : operating system .

PBX : Private branch exchange

PC : personal computer .

PDA : Personal digital assistant

PPU Palestine Polytechnic University.

PSAP: Public-safety answering point

PSTN :Public switched telephone network

QoS : Quality of Service

RSVP Reservation Protocol

RTP Real-time Protocol

SCTP: Stream Control Transmission protocol

SIP: Session Initiation Protocol

Telco : telecommunication companies

TFTP : Trivial File Transfer Protocol

TUI: Telephony User Interface

UDP : User Datagram Protocol

UPS: Uninterruptible Power Supply

VAIL: VoIP Application Interface Layer

VOIP : voice over internet protocol

CHAPTER ONE

INTRODUCTION

Introduction

1.1 General Description Of VOIP

VoIP (voice over IP) is an IP telephony term for a set of facilities used to manage the delivery of voice information over the Internet.VoIP involves sending voice information in digital form in discrete packets rather than by using the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.

Voice-over-Internet protocol (VoIP) is a protocol optimized for the transmission of voice through the Internet or other packet -switched networks. VoIP is often used abstractly to refer to the actual transmission of voice (rather than the protocol implementing it).

"Some cost savings are due to utilizing a single network to carry voice and data, especially where users have underused network capacity that can carry VoIP at no additional cost. VoIP-to-VoIP phone calls are sometimes free, while VoIP calls connecting to public switched telephone networks (VoIP-to-PSTN) may have a cost that is borne by the VoIP user.

Voice-over-IP systems carry telephony signals as digital signals, typically reduced in data rate using speech data compression techniques, encapsulated in a data-packet stream over IP."[5]

1.2 Project Objectives :

1- Reduce the cost of calling in Palestine Polytechnic University through implementing the VoIP technology. .

2- Testing the VoIP over the internet in PPU depending on the infrastructure from calling network and Private Branch Exchange (PBX).

3- In this project an important theoretical study will be done about the VoIP technology available to be implemented in our university and its options.

4- Solving the problem of the communication in the university community by implementing this project in the different buildings.

1.3 Literature Review

We searched the Internet and libraries for any documents related to the idea of our project, but we find that the available information and data are scarce, in most commercial and non-scientific and no direct relationship with the basic idea of our project

1- In Palestine Polytechnic University (PPU) Our idea was raised previously, but did not find any document about that project.

2- We find an implementation of nearly similar the idea of our project its used pure software computer such that Skype without uses the telephone its Proposal by: The Clarkson University Networking Club which named Clarkson University Campus Wide Voice over IP (VoIP) Implementation.

"The main objective of Clarkson University Campus Wide Voice over IP (VoIP) Implementation is to connect the campus together using VoIP (Voice over IP), thus allowing communication between campus residents easier and more efficient. By implementing this technology they will be able to easily connect to anywhere on campus, and potentially off campus, using this VoIP technology.

Faculty, staff, and students will all share the same central server where all of the VoIP connections will be made. Also, to deal with security and safety, it is an essential plan to keep a safe environment within the campus. With the implementation of VoIP across campus, users can have the comfort of knowing that there is access to help – Campus Safety, the local police, the hospital and the 911 emergency number –available in each dorm room, rather than having to seek help from a less reliable, alternative source, especially if there is no phone line activated or available in their current location."

1.4 Risk analysis

In this section we discuses about the anticipated risks that might affect on the project scheduling or the quality of the project, as below.

- Risk identification.
- Risk analysis.
- Risk planning.
- Risk monitoring.

The following table shows the possibility of accruing of each of above risks.

ID	Possibility	Effects
1	Low	Catastrophic
2	Moderate	Serious
3	High	Tolerable
4	Low	Catastrophic

Table 1.1 possibility risks.

Project Analysis

As any human project or work there is several problem or risk can affect on the project track, and can happen while we work on this project we put an estimated risk.

ID	Risks
	Technology risks
1	Fail in provider server.
2	Fail in some of hardware equipments, because of problems of electricity.
3	Problems in internet connection .
	Organization risks
	As we a group of students of graduation project and don't work in an organization, so there's no organization risk.
	Requirements Risks
4	The electrical and computer department asks for new objective to the project that can require new project design and rework it from beginning.

Table 1.2 Risks definition

Reduction Strategies

To reduce the risk that may be occur, we can avoid the failing in some hardware equipment and problem in internet connection by adding another external line with a new Analog Telephone Adapter connected to a new internet line.

On the other hand we continue permanently with electrical and computer department and brief them on what we are doing to avoid the risk of adding extra objective.

1.5 Time Plan

The following table defines the main task in the project:

T1	Project Definition	1 Week
T2	Collecting data	2 Weeks
T3	Analyses	8 Weeks
T4	Design	8 Weeks
T5	Testing	4 Weeks
T6	Documentation	16 Weeks

Table 1.3 Time scheduled Table

The time of the project is scheduled over 16 week; table 1.2 shows how the work scheduled over these weeks:

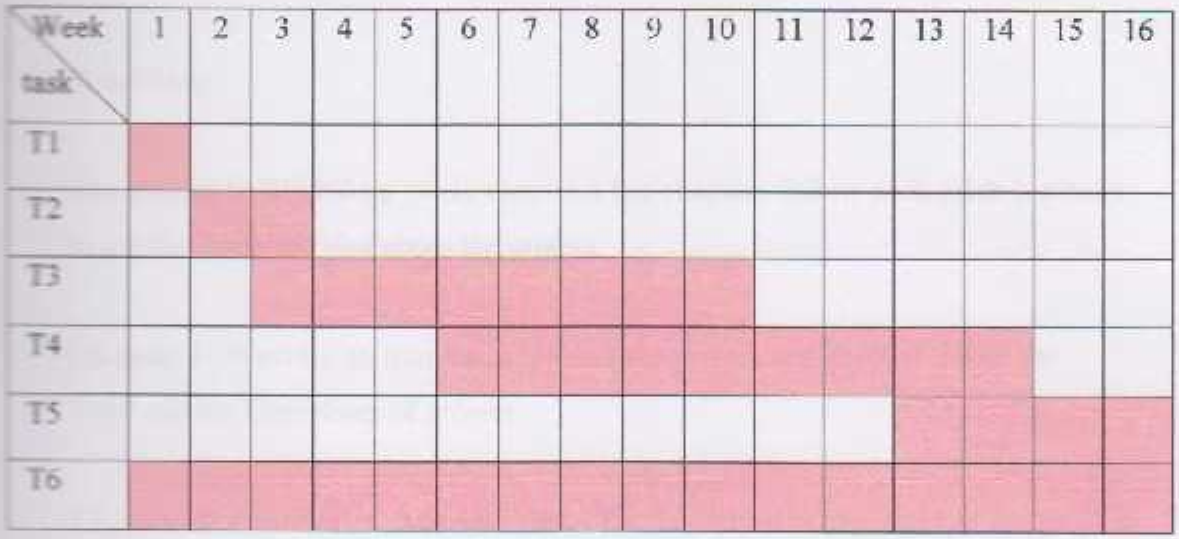


Table 1.4 Schedule Table

1.6 Budget

Equipment	Cost($)	Total ($)
Using the internet	50$	
2 * Phone	30*2 =60$	
2 * Phone adapter "intercall"	120*2= 240$	
Two ADSL Lines	2*70=140$	
Printing the final copy of chapters	50$	**540$**

Table (1.5) Budget

1.7 Road Map

The project is divided up to six chapters; the chapters follow each other logically to get the complete idea about the project

Chapter 1: Provides an introduction about the project and abstract about the VoIP and the Objectives of project.

Chapter 2: Provides an overview about the theoretical background of the project and the hardware we needed to implement the project in the PPU campus.

Chapter 3: Provides a conceptual design and the detail objective of the project . We dealt with way of implement the project and draw the block diagram.

Chapter 4: Detail description and characteristics of the each block in the block diagram of the project.

Chapter 5: Testing and Implementation.

Chapter 6: Conclusions and Recommendation.

CHAPTER TWO

Theoretical Background

2.1.1 Background Of VoIP

Internet telephony or Voice Over IP (Internet Protocol) is where you use your data channels - frequently broadband connections - for voice communications.

VoIP is very cost-effective. There are two charges made for traditional phone calls. First you pay for the exchange line rental and secondly for the actual call. Compare this with data charges. Normally you pay for Internet access at a flat monthly rate irrespective of the time that you use. Hence VoIP calls are effectively 'free'.

The service uses an Internet data connection to connect a standard telephone device to another similar device, or to the public switched telephone system, in order to connect to any telephone in the world. The technology has been used for several years by business customers, primarily to reduce the cost of international and long-distance calls.

While providing some important cost-savings to consumers, current VoIP technology does not provide for the transmission of the caller's physical location along with the voice call. If the caller dials the police station 100, there is no inherent protocol within the VoIP technology for routing the call to the nearest PSAP, or to display the caller's location, telephone number or other information.

2.1.2 VoIP phone software Vs hardware

If using a software based soft-phone, calls can only be placed from the computer on which the soft-phone software resides. Thus with a soft-phone the caller is typically limited to a single point of calling. When using a hardware based VoIP phone-device phone-adapter it is possible to connect traditional analog phones directly to a VoIP phone-adapter without the need to operate a computer. The converted analog phone signal can then be connected to multiple house phones or extensions, just as any traditional phone company signal can be connected. A second VoIP hardware configuration option involves the use of a specially designed VoIP telephone which incorporates a VoIP phone adapter directly into the phone itself, and this phone adapter is connect to the internet which also does not require the use of a computer .

2.2 Project Interaction With The Surrounding Environment :

The applications of our project have a great effect with Surrounding Environment. In today's world the communications play major role in the lives of many people in different areas of health, education, economy , industry and in government areas :

1- Government

Municipalities and other governmental entities tend to have multiple departments and locations running on disparate – and often incompatible – technology infrastructures. This can, and often does, pose significant maintenance and support challenges.

The adoption of IP telephony creates a single virtual telephony environment for various departments spread throughout the city – or country. By unifying voice and data technology infrastructures, maintenance and support are greatly simplified. Furthermore, a single network can provide virtually every city employee with traditional telephone services as caller ID, call forwarding, voice mail, and advanced directory service – plus advanced IP telephone capabilities such as auto attendant, "follow me" messaging, message forwarding to off-system users, and centralized directory integration.

2-Education

Traditionally paper-based, school districts are turning to enhanced IP telephony systems to replace tedious handwritten tasks. Attendance, hall passes, scheduling, and some security features are integrated into applications that help save time and money while simplifying the administrative process.

Universities can also leverage the flexibility of Pro Curve IP Telephony solutions to help drive revenue to offset additional infrastructure costs. By deploying IP phones into dormitory rooms throughout the campus, a university can derive advertising revenue from a host of locally supported student service businesses ranging from pizza delivery shops, bookstores, and more.

3-Healthcare

The healthcare industry depends on innovation to provide the best possible service to its ultimate customer, the patient. Pro Curve has helped create tools that give doctors, nurses, pharmacists, and medical staff access to scheduling, medical records, and lab results through IP phones or other mobile devices, such as PDAs or tablet PCs.

A converged data and voice network allows healthcare providers to access, manipulate, and archive voice, text, and displayed information in ways that help cut costs and enhance productivity. For example, after a patient visit, doctors can link verbal dictation to patient records for immediate updates. Invoices can be submitted using voice technology, saving doctors and other hospital staff time and reducing administration expenses.

4- Finance

By combining multiple network infrastructures into a single IP-based network, financial institutions can consolidate disparate infrastructures and reduce communications costs. At the same time, those savings can be transferred back to customers in the form of more competitive products such as Internet banking and bill pay, and offering value-added services such as wealth management and online trading and trade monitoring. Using IP telephony solutions, banks can move all channels into the branch, reducing manual cash handling by tellers, and providing an additional delivery mechanism for enhanced customer service. The result: simpler operations, lower maintenance and support costs, and greater agility.

5- Manufacturing :

The manufacturing industry is using IP telephony to speed up time to market and to balance supply and demand. The technology lowers the risk of unpredictable demand, uncertain availability, and fluctuating prices for direct materials by speeding up the communication process between distributors and suppliers. Industries like automotive, retail, pharmaceuticals, and high-tech are realizing dramatic improvements in inventory, service levels, supply and demand variability, and distribution channels through converged voice/data systems. Manufacturing organizations also realize savings in long-distance rates and can leverage multiple mobile communication and productivity solutions. Instead of expensive PBX equipment at every office site, centralized IP telephony solutions provide cost-effective central management and support.

2.3 Functionality

VoIP can facilitate tasks that may be more difficult to achieve using traditional networks that have been typically used historically:

- Ability to transmit more than one telephone call down the same broadband-connected telephone line. This can make VoIP a simple way to add an extra *telephone line to a home or office by using another ATA devices – which has* unique address – connected to the same network "ADSL line" and other traditional telephone.

- Many VoIP packages include PSTN features that most telecommunication companies (telco) normally charge extra for, or may be unavailable from your local telco, such as 3-way calling, call forwarding, automatic redial, and caller ID.

- VoIP can be secured with existing off-the-shelf protocols such as Secure Real-time Transport Protocol. Most of the difficulties of creating a secure phone over traditional phone lines, like digitizing and digital transmission are already in place with VoIP. It is only necessary to encrypt and authenticate the existing data stream.

- VoIP is location independent, only an internet connection is needed to get a connection to a VoIP provider, for instance call center agents using VoIP phones can work from anywhere with a sufficiently fast and stable Internet connection.

- Reliability: Data lines have never been as reliable as the traditional telephone land line. With the increasing availability and reliability of broadband this situation is changing. However, many organizations do not want to 'put all their eggs in one basket' if both data and voice communications are mission critical.

2.4 Quality of Service

One of the key issues is quality of service. Call quality can vary dramatically. This is usually down to the bandwidth available for voice calls. Voice calls take up more bandwidth than data transfers and are more susceptible to sampling problems. Hence calls where words are truncated or missing are not uncommon. Variations in service can also occur during the course of the day as more and more people log onto data services and less bandwidth is available for voice. Sufficient bandwidth for all traffic is the key.

Many Factors that effect on the QoS such as :

- Dropped packets: The routers might fail to deliver (drop) some packets if they arrive when their buffers are already full. Some, none, or all of the packets might be dropped, depending on the state of the network, and it is impossible to determine what will happen in advance. The receiving application may ask for this information to be retransmitted, possibly causing severe delays in the overall transmission.

- Delay: It might take a long time for a packet to reach its destination, because it gets held up in long queues, or takes a less direct route to avoid congestion. In some cases, excessive delay can render an application, such as VoIP or online gaming unusable.

- Jitter: Packets from the source will reach the destination with different delays. A packet's delay varies with its position in the queues of the routers along the path between source and destination and this position can vary unpredictably.

- Out-of-order delivery: When a collection of related packets is routed through the Internet, different packets may take different routes, each resulting in a different delay. The result is that the packets arrive in a different order than they were sent. This problem requires special additional protocols responsible for rearranging out-of-order packets to an isochronous state once they reach their destination. This is especially important for video and VoIP streams where quality is dramatically affected by both latency and lack of isochronicity.

- Error: Sometimes packets are misdirected, or combined together, or corrupted, while en route. The receiver has to detect this and, just as if the packet was dropped, ask the sender to repeat itself.

2.5 Theoretical Background Of The Project Components

1-Traditional phone: the telephone handles two types of information: signals and voice, at different times on the same twisted pair of wires. The signaling equipment consists of a bell to alert the user of incoming calls, and a dial to enter the phone number for outgoing calls. A calling party wishing to speak to another telephone will pick up the handset, thus operating the switch hook, which puts the telephone into active state or off hook with a resistance short across the wires, causing current to flow. The telephone exchange detects the DC current, attaches a digit receiver, and sends dial tone to indicate readiness. The user pushes the number buttons, which are connected to a tone generator inside the dial, which generates DTMF tones. The exchange connects the line to the desired line and alerts that line.

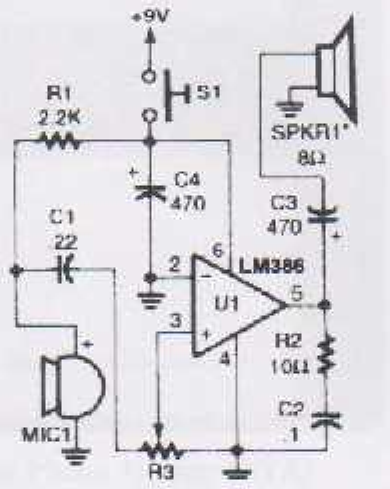


Fig 2.1 Traditional Phone Circuit


Fig 2.2 Traditional Phone

The phone we use is a normal phone which achieve the goal of connect with other phone and it can be built on the PBXs , the main feature of most normal phone compatible with the PBX and any normal phone with normal characteristic specially the heat and frequency can we used with our project

2- **ATA** - The simplest and most common way is through the use of a device called an (ATA) analog telephone adaptor The ATA allows you to connect a standard phone to your computer or your Internet connection for use with VoIP.

Fig 2.3 Analog Phone Adapter (ATA)

The ATA is an analog-to-digital converter. It takes the analog signal from your traditional phone and converts it into digital data for transmission over the Internet. Providers like InterCall "BESTip" are bundling ATAs free with their service. You simply crack the ATA out of the box, plug the cable from your phone that would normally go in the wall socket into the ATA, and you're ready to make VoIP calls. Some ATAs may ship with additional software that is loaded onto the host computer to configure it; but in any case, it is a very straightforward setup.

2- **PPU Internet (ADSL Internet) :**

Our device need a high speed internet link to connect to the provider to obtain a high quality voice with minimum delay "real time nearly" .

According to the infrastructure of the Internet in the PPU they used a firewall, which does not allow using VoIP port, so as to protect the network from viruses and attacks.

Therefore, we prefer using the Internet ADSL lines in each building, otherwise we must open the VoIP port in the firewall to allow us to use the infrastructure of the Internet in the university.

4-The PPU PBX :

Short for private branch exchange, a private telephone network used within an enterprise. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.

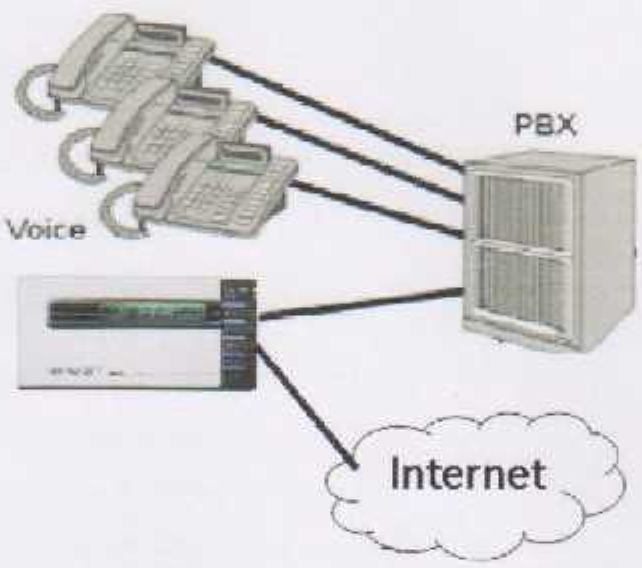


Fig 2.4 Private Branch Exchange (PBX)

Most medium-sized and larger companies use a PBX because it's much less expensive than connecting an external telephone line to every telephone in the organization. In addition, it's easier to call someone within a PBX because the number you need to dial is typically just 3 or 4 digits.

A new variation on the PBX theme is the Centrex, which is a PBX with all switching occurring at a local telephone office instead of at the company's premises.

In the PPU where are many PBX effuse over different building with different type , distribute as following:

1- In WAD ALHARIA : there is an old tow PBX 32/12 connect to gather with 64 internal line and 3 external line used.
2- In ABU ROMMAN : there is tow modern PBX one in the building "56/12" and other in IT Center "32/12" connect to gather internally .
3- In the CENTER BUILDING there is "32/12" PBX used 24internal and 6 external .
4- In ABU KTELA : there is an "32/12" PBX used 32 internal and 2 external .

CHAPTER THREE

PROJECT CONCEPTUAL DESIGN

3.1 Voice over IP Protocols:

3.1.1 Voice over IP (VoIP) Protocols Description

Of course, VoIP uses the Internet Protocol (IP), but there are many more protocols that it uses in order to successfully transfer data. First, voice quality is extremely sensitive to the amount of delay, packet loss and bandwidth available in the system. Because of this, VoIP uses the Real-time Protocol (RTP) for voice transmission because it is important that there is no delay in voice transfer (that it is received in real-time). The RTP protocol basically "rides" on top of User Datagram Protocol (UDP) and provides sequence numbers and timestamps necessary for the ordering of packets at the receiver. To limit "jitter" (variability in delay) on the receiver end, the packets are buffered and played back at a constant rate. Furthermore, to ensure voice quality and manage the number of simultaneous calls that are going on, the Reservation Protocol (RSVP) can be used to reserve bandwidth through the network. The VoIP architecture consists of a Media Gateway Controller (MGC) that supervises calls and services from end to end. The MGC has Media Gateways (MG)s that are the connection between the Public Switched Telephone Network (PSTN) and the IP network (IP network - Media Gateways - PSTN). These gateways actually create, modify and destroy connections as instructed by the MGC. The controllers and gateways interact over a control plane via the MEGACO Protocol (RFC 3525), previously the Media Gateway Control Protocol (MGCP).The media controllers interact with their peers using the Session Initiation Protocol (SIP), which is a text-based messaging protocol whose roots are in Hypertext Transfer Protocol (HTTP). (There is another protocol that can be used instead of SIP, named H.323. SIP is used to initiate communication sessions between users (its messages are session-specific).The media gateways first receive the initial DTMF signals from the user and convert them to SIP messages for the IP-based application servers to understand. They then convert the voice payload that follows to RTP packets to be used by the media processors.

3.1.2 General Protocols In VoIP:

- ✓ Internet Protocol (IP)
- ✓ Real-time Protocol (RTP)
- ✓ User Datagram Protocol (UDP)
- ✓ Reservation Protocol (RSVP)
- ✓ Media Gateway Control Protocol (MGCP)
- ✓ Session Initiation Protocol (SIP)
- ✓ Hypertext Transfer Protocol (HTTP)
- ✓ H.323 Protocol
- ✓ Stream Control Transmission Protocol (SCTP)

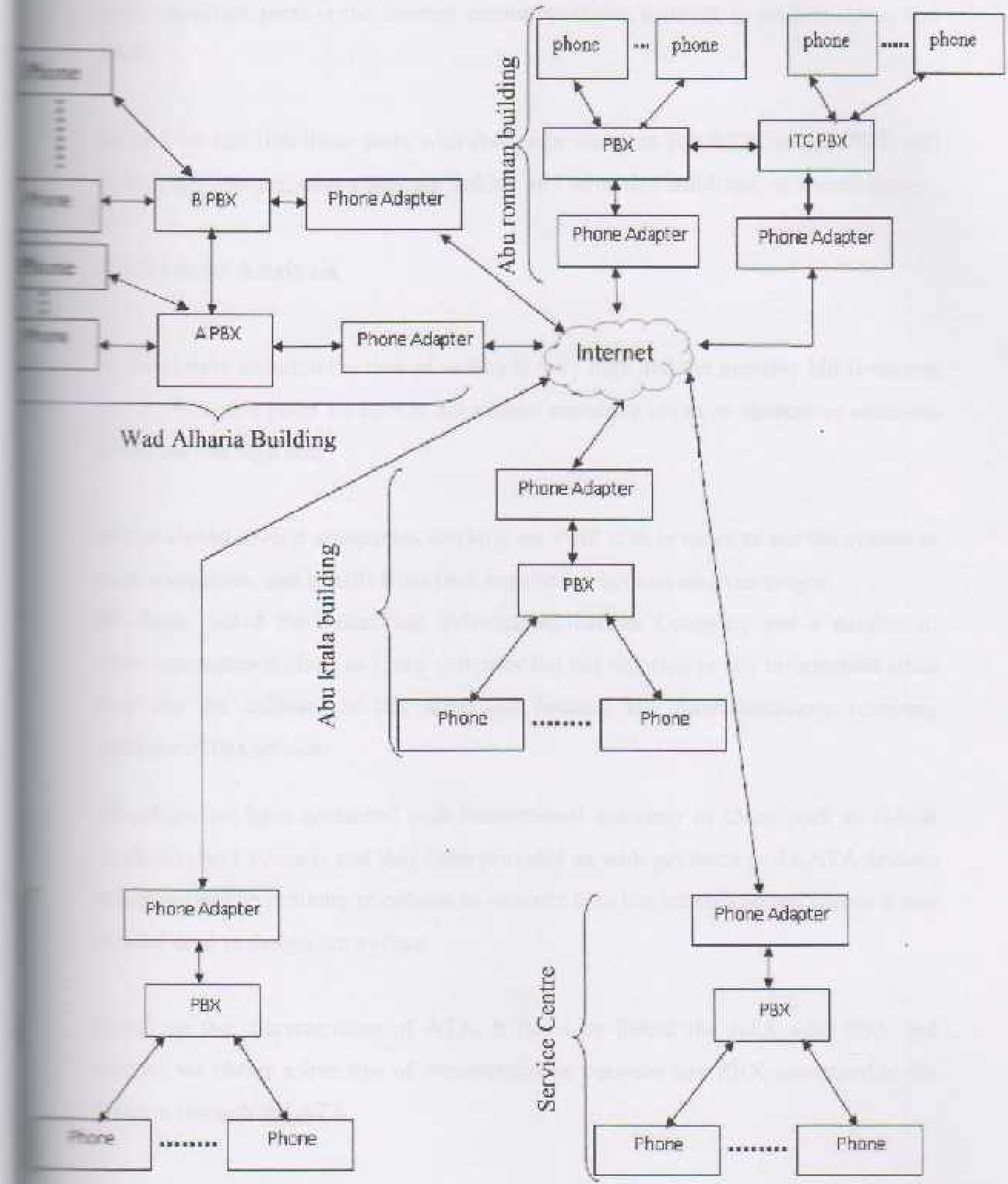
3.1.3 User Datagram Protocol (UDP):

User Datagram Protocol (UDP) is one of the core protocols of the Internet protocol suite. Using UDP, programs on networked computers can send short messages sometimes known as datagrams (using Datagram Sockets) to one another. UDP is sometimes called the Universal Datagram Protocol. The protocol was designed by David P. Reed in 1980.

UDP does not guarantee reliability or ordering in the way that TCP does. Datagrams may arrive out of order, appear duplicated, or go missing without notice. Avoiding the overhead of checking whether every packet actually arrived makes UDP faster and more efficient, for applications that do not need guaranteed delivery. Time-sensitive applications often use UDP because dropped packets are preferable to delayed packets. UDP's stateless nature is also useful for servers that answer small queries from huge numbers of clients. Unlike TCP, UDP is compatible with packet broadcast (sending to all on local network) and multicasting (send to all subscribers).

Theirs several real time application is used this protocol to decrease the delay on it. In our ATA (intercall) its used this protocol to decreasing the delay to minimum delay.

3.2 Project Design Block Diagram :



Project includes several important parts scattered in the building of the university, the most important parts is the internal communications network in each building and PBX.

So that we can link these parts with free lines we must put ATA in each PBX and linking the Internet, where they are linking all University buildings, as shown above.

3.3 System Analysis

In the current situation the cost of calling is very high and the monthly bill is exceed 800 $, from this point we start in our project searching about an alternative solutions to reduce this high cost .

We've visited several companies working on VoIP area in order to see the system in their companies, and benefit from their experience in designing our project
We have visited the Palestinian Telecommunications Company and a number of other companies such as al-Jenan company but not reported in any information since they use the software in this area, and because the communications company defiance of this service.

Therefore, we have contacted with international company in Qatar such as NASR Technologies Company and they have provided us with guidance to the ATA devices which is used in ordinary telephone to connect it to the Internet so we choice it and depend on it to design our system.

Based on the characteristics of ATA, it could be linked the ATA with PBX and thereby we obtain a free line of communication between any PBX connected to the Internet through the ATA.

- In Wad Alharya we find that the number of phone is sixty four in both building and two PBX.

- In Abu Rumman building there are sixty phone in both the building and IT center, with one PBX in each building.
- In Abu Ktella building the all internal thirty two internal lines are in used, with one PBX.
- Center Building in eyen sara street there are twenty internal lines.
- In Service Center in side of the traffic light in eyen sara we find that there are seven internal lines .

Draw from the foregoing that the total was 183 phones , and therefore we studying the different scenarios available and recommend the better.

3.4 Design Options:

So there are many scenarios to implementing our project.

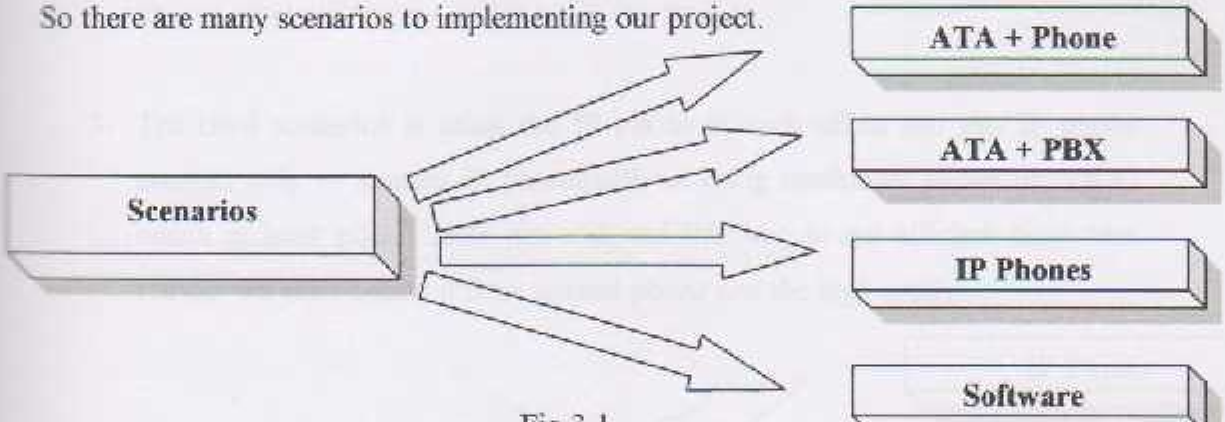


Fig 3.1

The first scenarios is using ATA 10I. connected to normal phone in each office, but we found it is very costly since there are a huge number of office in the university which mean more and more ATA L10 devices needed, in this case its budget is more than the current cost using traditional calling through ten years so we didn't prefer this scenario.

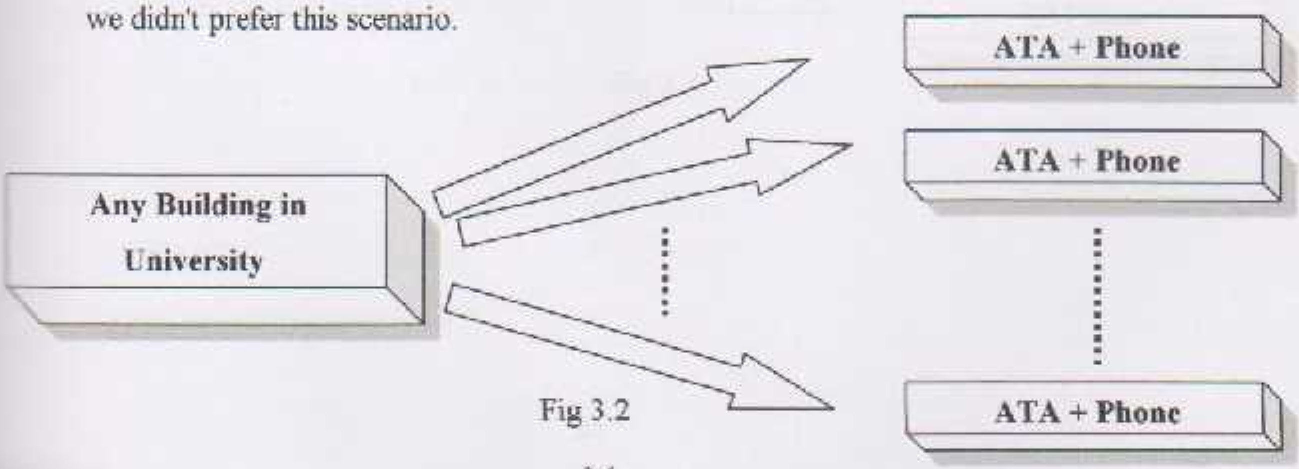


Fig 3.2

2- The second scenario is connecting the ATA with the PBX so we need only one ATA in each PBX ,can be extended to the max external line in the PBX if we need using another ATA L10 in each external line and this is the best way and the minimum cost we can approach.

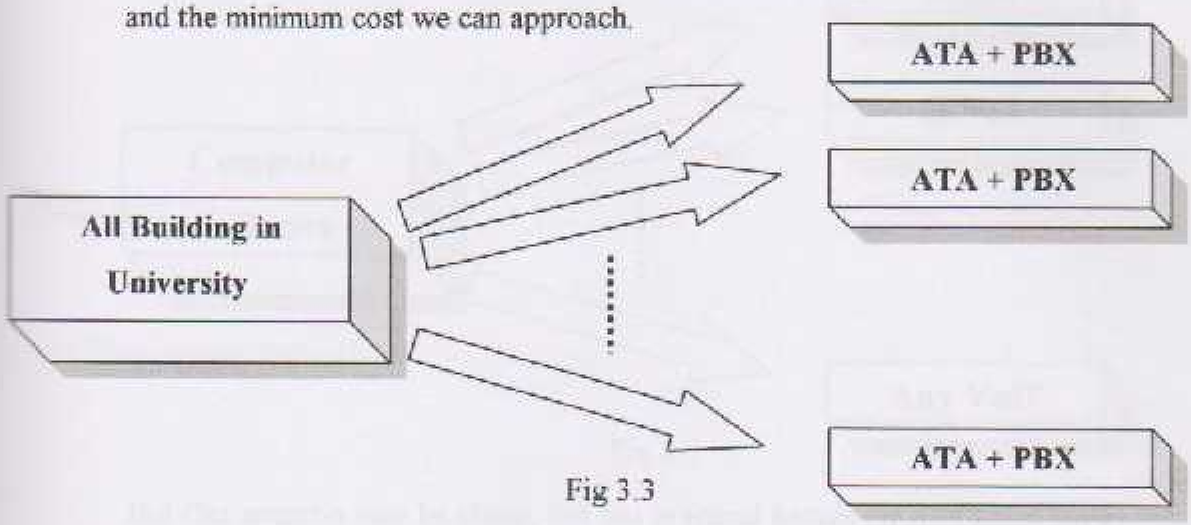


Fig 3.3

3- The third scenarios is using the IP phone in each office and this IP phone connect only to another IP phone(without using traditional phone or PBX) which at least priced 200$ per unit and this way is not efficient since two reason: we can't calling it from normal phone and the high costly.

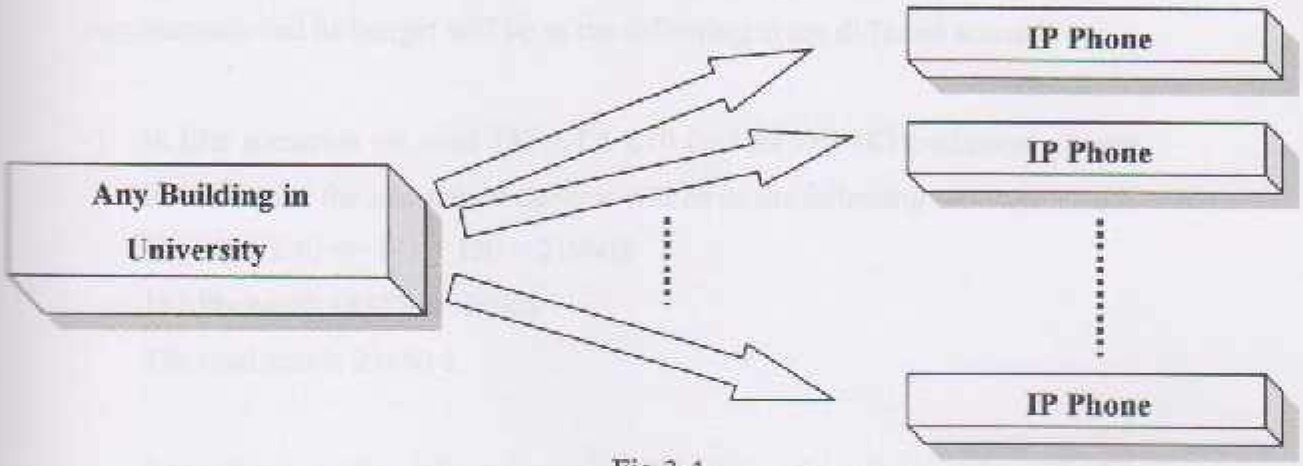


Fig 3.4

4- The fourth scenarios is using the software on the computer: here we need to installing ready software on each computer in each office with using a head phones and MIC .

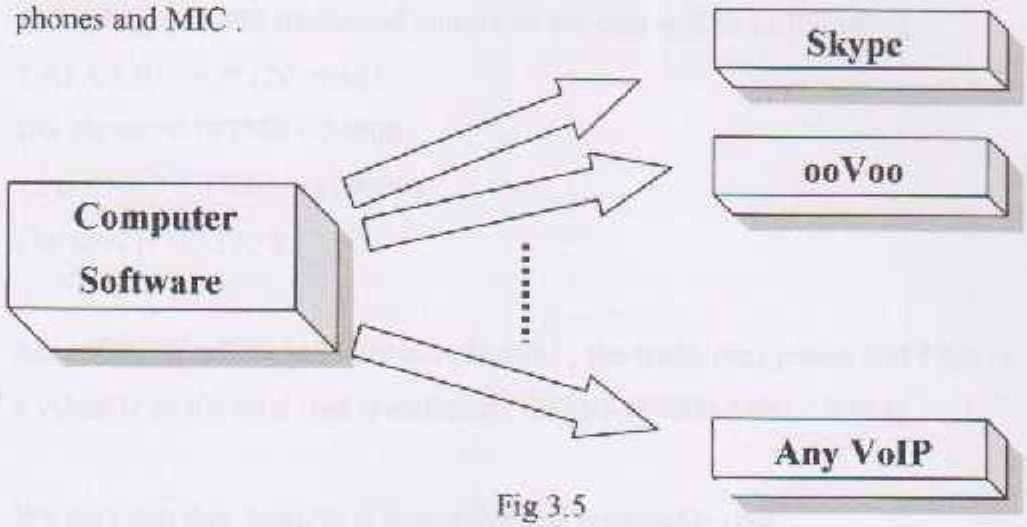


Fig 3.5

But this scenario may be cheap, but not practical because it is requires highly skills users in dealing with these software and it is not effective unless the computer is in operation.

3.5 The Scenarios With System Requirements:

We apply these scenarios on the offices in the building of the university then the requirements and its budget will be as the following in the different scenario :

1- In first scenarios we need 183 ATA L10 devices and 183 traditional phones in addition to the internet, so the cost will be as the following:

183 ATA L10 => 183 * 120 = 21960$

183 Phone => 183*30 = 5490$

The total cost is 27450 $.

According to calling infrastructure in PPU , the traditional phone is a valuable so the total cost is estimate (27450 -5490= 21960$)

We note in this scenario that it is too expensive.

2- In the second scenarios we will use the ATA L10 to connect the PBX in the different building which mean that we need seven ATA L10 in addition to seven PBX and 183 traditional phones so the cost will be as following :

7 ATA L10 => 7*120 =840$

183 phone =>183*30 – 5490$.

7 PBX –> 7 * 17000 = 119000$

The total is 125330 $.

According to calling infrastructure in PPU , the traditional phone and PBX is a valuable so the total cost is estimate (125330-119000-5490 = 840 $)

We note that this scenario is acceptable and reasonable cost

3- In the third scenarios we use 183 IP phones , in this case we can't use the PBX so we must using IP phone in each office , then the cost will be as the following :

183 IP phone => 183*200 = 36600 $

We note in this scenario that it is more expensive.

4- In the fourth scenario we need 183 unit , each unit consist of PC , MIC , Headphone and VoIP software which often free to download.

So the Expected cost per unit is 1200$ (PC) + 5$ (MIC)+10$(Headphone) so the total cost equal to : (1200+5+10) * 183 = 222345 $

But the PC is available in each office so the total cast is 222345-(183*1200)= 2745 $.

Scenarios	Estimated cost
First scenario	21960$
Second scenario	840$
Third scenario	36600$
Fourth scenario	2745$

Table 3.1 summary comparing

We can deduce from this that the second scenario is better because it costs less and more meaningful in terms of performance. With known that the calling cost between two ATA is zero$.

3.5.1 Hardware:

We faced many problems in the designing the hardware; the main problems we didn't find any data sheet about the phone adapter or IP phone so we can't built it, also we need a provider.

In our project we used ATA 10L to connect the centrals in our campus with internet to make a free external line between them in order to reduce the cost of external calling. ATAs are used by many VoIP companies selling a telco-alternative VoIP service, where the device is used to replace a user's connection to a traditional telephone company. When sold in connection with a VoIP service, the ATA is often locked so it cannot be used with a competing service, and the user can only partly change its configuration.

FXS to Ethernet gateways: The most common ATA is a box with at least one Foreign EXchange Station used to connect a conventional telephone, and an Ethernet jack used to connect the adapter to a LAN. Using such an ATA, it is possible to connect a conventional telephone to a remote VoIP server. The ATA communicates with the server using a protocol such as H.323, SIP, MGCP, SCCP or IAX, Since the ATA communicates directly with the VoIP server, it does not require any software.

Fig 3.6 ATA communicates directly

3.5.2 Software:

One main option is to use software installing on computer, this way needed fluently uses of computer and good skills in dealing with these software.
There are many VoIP ready software we can use it

To make VoIP communication possible throughout the campus, each campus instructors will need a softphone, to connect to the central VoIP server. Softphones are simply software that can be installed on a computer, we would recommend softphones for campus use because each instructor already have access to a computer in their office with using head phones and MIC and each instructor has own account on the software and there are many free software available on the web for download. In our project, we examined different softphone to find the best possible communication for users in different building. Below is a list of some possible software and vendors that were looked at.

3.5.2.1 Skype Internet Telephony[8]

Skype is becoming popular, but is still relatively new in its development.

Fig. 3.7 Skype

Skype has Windows, Mac OS X, Linux, and Pocket PC clients. Its clients use end-to-end encryption to make calls to all over the world. It is free of charge to call other registered users of Skype and they have recently come out with "SkypeOut", which allows the user to call existing phone numbers using the telephone network.

Security features

Secure communication is a feature of Skype; encryption cannot be disabled, and is invisible to the user. Skype reportedly uses non-proprietary, widely trusted encryption techniques: RSA for key negotiation and the Advanced Encryption Standard to encrypt conversations. Skype provides an uncontrolled registration system for users with absolutely no proof of identity. This permits users to use the system without revealing their identity to other users. It is trivially easy, of course, for anybody to set up an account using any name; the displayed caller's name is no guarantee of authenticity.

3.5.2.2 ooVoo Internet Telephony[9]

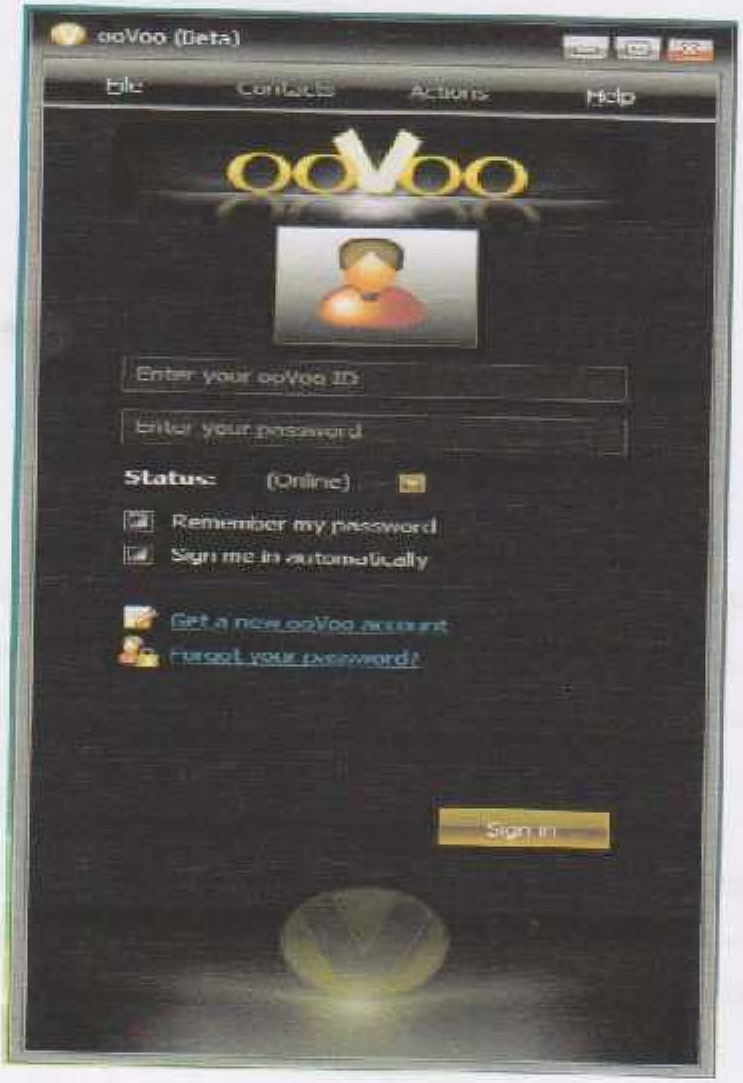


Fig. 3.8 ooVoo

ooVoo is a VoIP tools like skype and it allow you to create e-mail to connected to the world.

ooVoo takes that to a whole new level. Video conferencing with up to six people. All you need is a webcam, a headset with a mic and the free download of ooVoo.

3.5.2.3 SJPhone Softphone[10]

Fig. 3.9 SJPhone Softphone

SJPhone, from SJ Labs, The phone works with any PC, PDA, or IP Phone. It also supports both SIP and H.323 protocols. It is also supported across a large range of operating systems including MS Windows XP, 2000, 98/ME, Linux and MAC OS X.

one can communicate with another via computers throughout the local area network (LAN). Everyone using SJPhone on the network can view who is currently on the service and click on the user to speak with them. SJPhone is free to download and use.

CAPTER FOUR

DETAILED TECHNICAL

PROJECT DESIGN

4.1 SUBSYSTEM DETAILED DESIGN

4.1.1 Traditional phone:

In General Traditional tele-phone shared the following features:

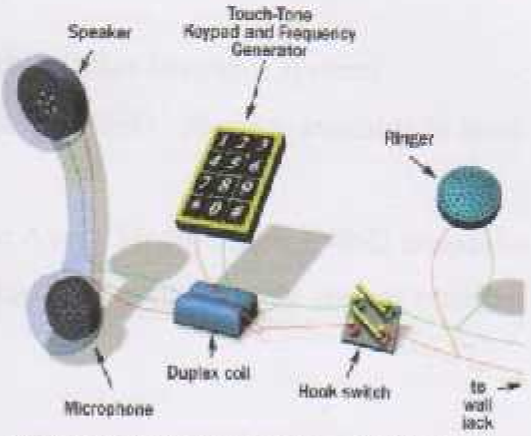


Fig 4.1 General Traditional Telephone

- Displays caller information for incoming calls--the name and/or number of external callers as well as the extension and co-worker name for internal calls that you receive, so you don't have to have a separate Caller ID box
- 6 Button Digital Telephone
- 6 programmable call appearance / feature buttons with LED
- 2-line x 24 character Liquid Crystal Display
- Identifies internal and external callers (subject to availability) without Caller ID port or seperate display

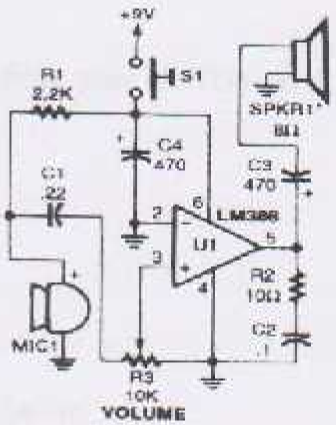


Fig 4.2 Traditional telephone circuit

- 12 programmable feature-only buttons without LED
- Lets you program unassigned line buttons for easy access to system features such as autodial
- Conference up to 3 internal parties plus 2 external
- Hands-free conversation with built-in speakerphone
- Hands-free answer on intercom (HFAI)
- Fixed-feature buttons for Conference, Speaker, Transfer, Redial Hold and Mute
- Built-in headset jack, in addition to handset jack

- Message waiting light
- 8 ringing patterns for easy recognition in open offices
- Volume control
- Wall mountable
- For use with the Merlin Magix and IP Office telephone systems.
- The new Lucent Avaya Merlin Magix 4406D+ phone is available in black or white.
- Single pair wiring lets you use the Avaya Merlin Magix 4406D telephone with your existing premises wiring, saving you the big cost of rewiring your office.

4.1.2 Analog Telephone Adaptor (ATA 10L)

BESTip(intercall) ATA

Requirements: Before making any Internet call from your BESTip ATA, you need the following items:

1. A Touch-tone phone set.

2. A 110/220V AC electrical outlet.

3. A valid Internet connection, either broadband or dial-up.

4. An analog phone line

Connection:

According to the diagram below for the typical connection.

Fig 4.3 ATA Typical Connection

How to enter or change setting

Use the buttons in the front panel for menu function navigation. Go to the option you want to setup. Then follow steps described below to enter or change the setting.

1. Press [● OK] one time, the title will start flashing. You may enter new setting from phone keypad now.

2. Press [● OK] button again. The title will stop flashing. The new value has been store in memory temporarily. You may go to the next setting.

3. After all the settings has been setup, please press [▲ UP] one or two times until the screen displays "Update Setup". All the setting will be saved permanently.

Please refer to instructions in the next page for "Quick Setting" operation.

Buttons & Phone Keypad

- Before making phone call, you must enter your Internet information with your touch-tone
- telephone keypad and BESTip's buttons.
- Refer to the Typical Phone Keypad and Character Set below to enter characters.

Features and Benefits

- One independent phone lines (FXS) .
- One 10/100BaseTX LAN port (xDSL/cable modem, wireless, PC, etc.)
- Superior voice quality using various QoS mechanisms
- Allows incoming/originating calls over VoIP and PSTN lines via one phone
- Firewall .
- DHCP server client and relay
- NAT server to enable connection of phones and PC while using one IP address
- Integrated web server for easy provisioning
- Auto provisioning and automatic configuration with TFTP and HTTP to aid large installations.

4.1.3 PPU PBX (Private branch exchange)

A Private Branch eXchange (PBX) is a telephone exchange that serves a particular business or office, as opposed to one that a common carrier or telephone company operates for many businesses or for the general public.

PBX System Components

1. The PBX's internal switching network.
2. Central processor unit (CPU) or computer inside the system, including memory.
3. Logic cards, switching and control cards, power cards and related devices that facilitate *PBX operation*.

4. Stations or telephone sets, sometimes called lines.
5. Outside Telco trunks that deliver signals to (and carry them from) the PBX.
6. Console or switchboard allows the operator to control incoming calls.
7. Uninterruptible Power Supply (UPS) consisting of sensors, power switches and batteries.
8. Interconnecting wiring.
9. Cabinets, closets, vaults and other housings.

PBX Functions

Functionally, the PBX performs four main call processing duties:

1. Establishing connections (circuits) between the telephone sets of two users. (e.g. mapping a dialed number to a physical phone, ensuring the phone isn't already busy).
2. Maintaining such connections as long as the users require them (i.e. channeling voice signals between the users).
3. disconnecting those connections as per the users requirement.
4. Providing information for accounting purposes (e.g. metering calls).

In the PPU where are many PBX effuse over different building with different type

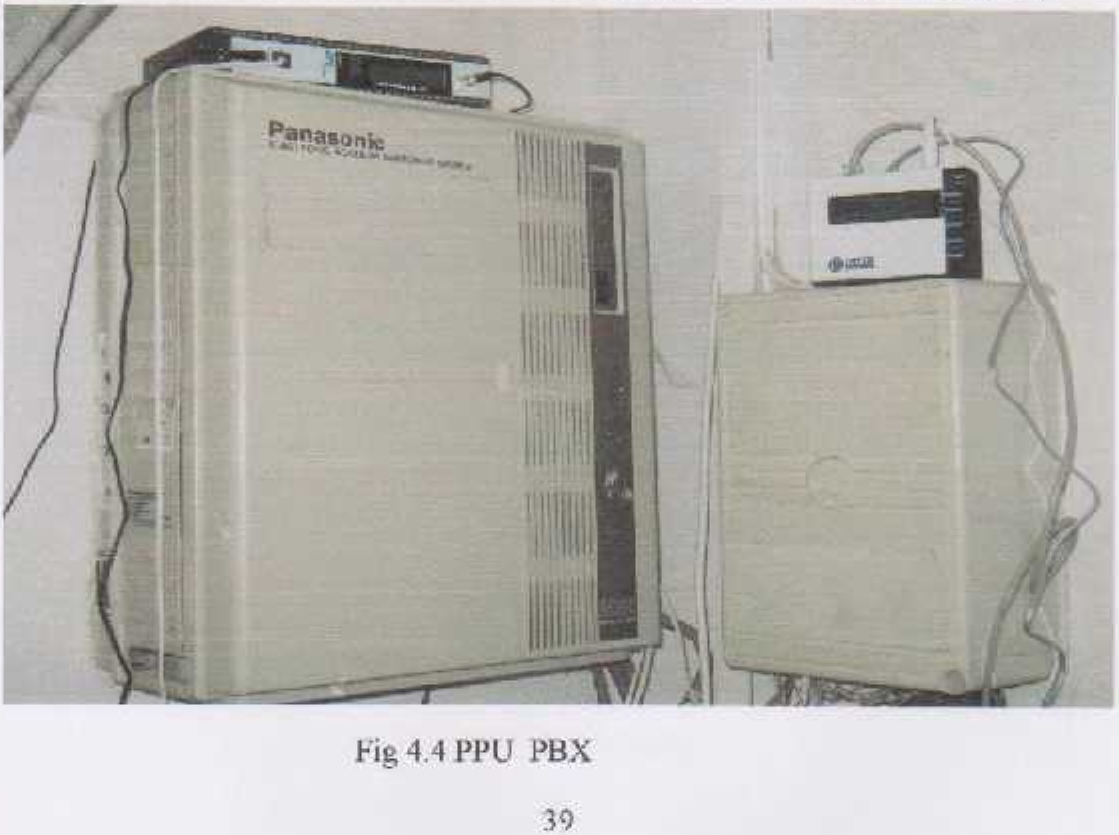


Fig 4.4 PPU PBX

CAPTER FIVE

TESTING AND IMPLEMENTATION

5.1 Component testing

The testing process is done through two stage:

1- First stage is the subsystem test , In this part of the testing process, we connect the ATA L10 on the Internet and with traditional phone on the other hand, in the same way we connect the other device in different place and then we have a process of communication between them. We were able to calling through them with high quality and competitive traditional calling.

Fig. 5.1 First Stage Testing

2- The second stage is integrated testing using PBX , here we connect one ATA L10 in Services PBX and another one in Main center PBX and using ADSL lines and make successful calling between two different office through it.

Fig. 5.2 Second Stage Testing

CAPTER SIX

CONCLUSION AND RECOMMENDATIONS

6.1 Problems and difficulties that we faced in the project

We have faced many difficulties during our work on this project, these problems are summarized as follows:

- Lack of data sheet to build the ATA device.
- The need to service provider
- The small number of companies operating in VoIP area.
- Preventing Palestinian Telecommunications Company of companies and institutions from using this technique, resulting in the non-diffusion, and the lack of workers in this area.

6.2 Conclusions :

We have succeeded in conducting the process of testing between the two buildings of the university : between the service building in Eyne Sarah with the main building, achieve line between them is free and highly efficient voice

The system can be implemented in a simple way by normal people in their home to make the international calling with very low cost.

6.3 Recommendations For Future Work :

1- We recommend to apply this project in all Palestine Polytechnic University buildings.

2- We hope to generalize this project and apply it in the companies and huge organization both public and private as the same.

3- Increased attention to this subject in the university through courses because the general trend in international calling is in this direction

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3- Raake, Alexander, "Speech Quality of VoIP: Assessment and Prediction" , John Wiley & Sons, Ltd ,2007.

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5- http://en.wikipedia.org/wiki/VoIP

6- http://www.mybestip.com

7- http://www.arab-eng.org

8- http://www.Skype.com

9- http://www.ooVoo.com

10- http://www.sjlabs.com/sjp.html

11- http://www.hp.com/rnd/pdf_html/IP_telephony_whitepaper.htm

12- http://www.fcc.gov/voip/

13- http:// www.windowsfordevices.com/files/misc/VoIP_WhitePaper.doc

14- http://searchunifiedcommunications.techtarget.com.

Appendix A

BESTip ATA COMBO Quick Start Manual

Requirements:

Before making any Internet call from your BESTip ATA, you need the following items:

1. A Touch-tone phone set.
2. A 110/220V AC electrical outlet.
3. A valid Internet connection, either broadband or dial-up.
4. An analog phone line

Connection:

Please refer to the diagram below for the typical connection.

Buttons & Phone Keypad

Before making phone call, you must enter your Internet information with your touch-tone telephone keypad and BESTip's buttons.
Refer to the Typical Phone Keypad and Character Set below to enter characters.

Typical Phone Keypad and Character Set

KEY PAD	Character Set
1	1 [illegible]
2	2 abcABC
3	3 defDEF
4	4 ghiGHI
5	5 jklJKL
6	6 mnoMNO
7	7 pqrsPQRS
8	8 tuvTUV
9	9 wxyzWXYZ
0	0 .,@"space"
*	
#	#

How to enter or change setting

Use the buttons in the front panel for menu function navigation. Go to the option you want to setup. Then follow steps described below to enter or change the setting,

1. Press [● OK] one time, the title will start flashing. You may enter new setting from phone keypad now.
2. Press [● OK] button again. The title will stop flashing. The new value has been store in memory temporarily. You may go to the next setting.
3. After all the settings has been setup, please press [▲ UP] one or two times until the screen displays "Update Setup". All the setting will be saved permanently.

Please refer to instructions in the next page for "Quick Setting" operation.

Switch between Broadband and Dial-Up mode

1

Press ▼ DOWN button at standby state to enter **setup mode.**

2

Press ▶ NEXT button until you see **System Setting** option. Press. ▼ DOWN to enter **System Setting** menu.

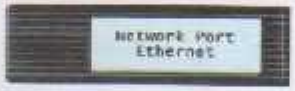
The first option is "Network Port", it is setup to "Ethernet" for broadband by default. It can be changed to "modem" for dial-up connection.

3

If the mode display in the screen is already the mode you want, then press ▼ DOWN and go to next page for following setup. Otherwise press ● OK button to modify it.

4

Press ▶ NEXT or ◀ BACK button to switch network port selection between "Ethernet" and "modem".



(for broadband)

(for dial-up)

5

When decision made, Press ● OK button. Then, press ▲ UP two times to save the new setting. Then please go to step "1" at this page and do the setup procedure for setting the device properly.

Broadband Mode - Setup Quick Configuration

1 Press [▼ DOWN] button at standby state to enter **setup mode**.

`BESTip ATA COMBO` / `xx:xx:xx`

2 Press [▶ NEXT] button until you see **System Setting** option. Press [▼ DOWN] to enter **System Setting** menu.

`Function Select` / `System Setting`

3 Subject to your broadband environment there are three options up to your choice in **IP Policy: Static IP, DHCP,** and **ADSL PPPoE**. Please make sure the one you choose meet your requirement. Press [▶ NEXT] or [◀ BACK] to navigate between the three options, Press [▼ DOWN] to make your choice.

DHCP:

`IP Policy` / `DHCP`

[▼ DOWN]

Skip to option "Station ID"

ADSL PPPoE:

`IP Policy` / `ADSL PPPoE`

[▼ DOWN]

`PPPoE Login Name`

[▼ DOWN]

`PPPoE Password`

[▼ DOWN]

`PPPoE Type` / `1` — Normally set to 1

[▼ DOWN]

`PPPoE Timeout` / `0` — Idle time to shut down Internet connection. 0 means always online.

Skip to option "Station ID"

Static IP:

`IP Policy` / `Static IP`

[▼ DOWN]

`My IP Address`

[▼ DOWN]

`Netmask IP Addr.`

[▼ DOWN]

`Router IP addr.`

[▼ DOWN]

`Primary DNS`

[▼ DOWN]

`Secondary DNS` — Enter Secondary DNS IP address or Leave it blank

4 Enter **Station ID.** Normally set to 1

`Station ID` / `1`

5 Press [▼ DOWN] to enter **Gatekeeper. AUTO** is applicable most of the time.

`Gatekeeper` / `AUTO`

6 Press [▼ DOWN] to enter **Country Code.** Enter your telephone country code

`Country Code`

7 Press [▼ DOWN] to enter **Area Code.** Enter local telephone domestic area code

`Area Code`

8 Press [▼ DOWN] to enter **IDD Prefix.** It is used for dialing international call. "00" in most cases

`IDD Prefix` / `00`

9 Press [▼ DOWN] to enter **2 nd IDD Prefix.** Leave it blank unless you are in USA.

`2nd IDD Prefix`

10 Press [▼ DOWN] button to enter **DDD Prefix.** It is used for dialing domestic and local call. "0" in most cases

`DDD Prefix` / `0`

11 You may skip the rest of options. Press [▲ UP] to return to top of **System Setting** menu.

`Function Select` / `System Setting`

12 Press [▲ UP] button to save configuration data in memory

`Update Setup` / `Please Wait ....`

13 It will return to standby state again after updated. Please restart the device to make setup changes effective.

`BESTip ATA 20L` / `xx:xx:xx`

Start Making VoIP Call

The device is ready to use in standby state when the blue ***ON LINE*** *led on. You can pick up phone and dial Internet phone call now.*

You may check up your credit by pressing [▲ UP] in standby state. It will show balance left in your account. You can make phone call in this state, too.

`Point xxx` / `Ready for Dial`

Dial-UP mode - Setup Quick Configuration

❶

Press ▼ DOWN button to enter **Quick Setting**.

❷

Press ▼ DOWN button to enter **PBX Dial Prefix.** Enter code the access trunk if the device is connected with PBX.

❸

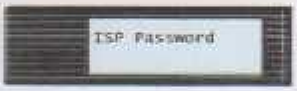
Press ▼ DOWN button to enter **ISP Phone**. Enter ISP phone number.

❹

Press ▼ DOWN button to enter **ISP Login Name**. Enter ISP Login Name.

❺

Press ▼ DOWN button to enter **ISP Password**. Enter ISP Password.



❻

Press ▼ DOWN button to enter **ISP PPP Type**. Choose ISP PPP Type using ► NEXT or ◄ BACK , use "Type 1" normally.

❼

Press ▼ DOWN button to enter **ISP Idle Timeout**. Choose ISP Idle Timeout using ► NEXT or ◄ BACK , "0" is no idle timeout.

❽

Press ▼ DOWN button to enter **Primary DNS**. Enter DNS IP address. You may leave it blank.

❾

Press ▼ DOWN button to enter **Secondary DNS**. Enter DNS IP address. You may leave it blank.

❿

Press ▼ DOWN button to enter **Country Code**. Enter telephone country code, "886" for example.

⓫

Press ▼ DOWN button to enter **Area Code**. Enter telephone area code, "2" for example.

⓬

Press ▼ DOWN button to enter **IDD Prefix**. Enter IDD prefix code when dial international call,"00" for example.

⓭

Press ▼ DOWN button to enter **IDD Prefix**. Enter DDD prefix code when dial domestic and local Call, "0" for example.

⓮

Press ▼ DOWN button to back to the beginning of **Quick Setting**.

⓯

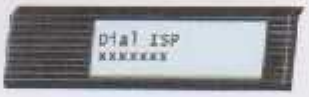
Press ▲ UP button to save configuration data in memory.

⓰

It will display the standby screen again after the configuration data has been saved.

Start Making VoIP Call

Press ▲ UP button to make the device start dial to ISP.



It will display the ITSP balance and "Ready for dial" after the device is connect to Internet. You can pick up the phone And dial Internet phone call now.

Appendix B

STip International Calling Rates (In US$ per Minute)

try	Code	Rate $
nistan	93	0.39
a	355	0.13
a Mobile	355	0.31
a	213	0.19
n Mobile	213	0.23
can Samoa	1684	0.15
rra	376	0.07
rra Mobile	376	0.44
a	244	0.25
a Mobile	244	0.28
lla	1264	0.30
ctica	6721	1.90
ua and Barbuda	1268	0.22
ntina	54	0.05
ntina Mobile	549	0.15
nia	374	0.15
nia Mobile	374	0.29
a	297	0.29
nsion Island	247	0.90
alia	61	0.04
alia Mobile	61	0.18
ia	43	0.03
a Mobile	43	0.28
aijan	994	0.19
aijan Mobile	994	0.24
mas	1242	0.07
ain	973	0.14
gladesh	880	0.07
ados	1246	0.29
rus	375	0.35
ium	32	0.03
ium Mobile	32	0.33
e	501	0.35
n	229	0.19
n Mobile	229	0.25
muda	1441	0.08
tan	975	0.29
ia	591	0.18
ia and Herzego	387	0.19
ia and H Mobile	387	0.39
wana	267	0.18
wana Mobile	267	0.25
il	55	0.06
il Mobile	55	0.29
sh Virgin Islands	1284	0.19
ei Darussalam	673	0.08
aria	359	0.05
aria Mobile	359	0.49
ina Faso	226	0.22
ina Faso Mobile	226	0.29
ndi	257	0.15
bodia	855	0.12
eroon	237	0.16
eroon Mobile	237	0.29
ada	1	0.03
e Verde	238	0.38
man Islands	1345	0.25
ral African Rep.	236	0.45
d	235	0.22
e	56	0.05
e Mobile	56	0.25
na	86	0.03
mbia	57	0.07
mbia Mobile	57	0.19
noros	269	0.49
go	242	0.19
k Islands	682	1.50
ta Rica	506	0.11
e Divoire	225	0.29
e Divoire Mobile	225	0.39
atia	385	0.06
atia Mobile	385	0.37
a	53	1.50
rus	357	0.04
rus Mobile	357	0.09
ch Republic	420	0.04
ch Republic Mob	420	0.29
m. Rep. of Congo	243	0.54
mark	45	0.03
mark Mobile	45	0.33
go Garcia	246	1.90
bouti	253	0.58
minica	1767	0.25

Country	Code	Rate $
East Timor	670	0.75
Ecuador	593	0.36
Egypt	20	0.19
Egypt Mobile	20	0.17
El Salvador	503	0.23
Ellipso Satellite	8812	5.83
Equatorial Guinea	240	0.39
Eritree	291	0.39
Estonia	372	0.05
Estonia Mobile	372	0.39
Ethiopia	251	0.35
Falkland Islands	500	0.98
Faroe Islands	298	0.34
Fiji	679	0.39
Finland	358	0.06
Finland Mobile	358	0.24
France	33	0.04
France Mobile	33	0.26
French Guiana	594	0.11
French Guiana Mobile	594	0.47
French Polynesia	689	0.39
Gabon	241	0.19
Gabon Mobile	241	0.28
Gambia	220	0.47
Garuda Satellite	8823	6.00
Georgia	995	0.11
Georgia Mobile	995	0.23
Germany	49	0.03
Germany Mobile	49	0.29
Ghana	233	0.11
Ghana Mobile	233	0.22
Gibraltar	350	0.07
Gibraltar Mobile	350	0.50
Global Networks Sat	8828	5.31
Globalstar Satellite	8818	4.77
Globalstar Satellite	8819	6.57
Greece	30	0.03
Greece Mobile	30	0.35
Greenland	299	0.96
Grenada	1473	0.35
Guadeloupe	590	0.07
Guadeloupe Mobile	590	0.54
Guam	1671	0.04
Guatemala	502	0.22
Guinea	224	0.37
Guinea Bissau	245	0.98
Guyana	592	0.48
Haiti	509	0.39
Honduras	504	0.29
Hong Kong	852	0.04
Hungary	36	0.05
Hungary Mobile	36	0.32
Iceland	354	0.04
Iceland Mobile	354	0.39
India	91	0.10
Indonesia	62	0.15
Indonesia Mobile	62	0.19
Inmarsat Atlantic Ocean	87	8.00
Iran	98	0.19
Iraq	964	0.09
Iraq Mobile	964	0.19
Ireland	353	0.03
Ireland Mobile	353	0.38
Iridium Satellite	8816	9.59
Israel	972	0.04
Israel Mobile	972	0.16
Israel - Palestine	972	0.39
Israel - Palestine Mobile	972	0.42
Italy	39	0.03
Italy Mobile	39	0.29
Jamaica	1876	0.37
Japan	81	0.05
Japan Mobile	81	0.22
Jordan	962	0.05
Jordan Mobile	962	0.15
Kazakhstan	721	0.17
Kazakhstan Mobile	721	0.24
Kenya	254	0.19
Kenya Mobile	254	0.27
Kiribati	686	0.98
Kuwait	965	0.19
Kyrgyzstan	996	0.18
Laos	856	0.11

Country	Code	Rate $
Lebanon	961	0.16
Lebanon Mobile	961	0.29
Lesotho	266	0.46
Liberia	231	0.49
Libya	218	0.39
Liechtenstein	423	0.10
Liechtenstein Mobile	423	0.76
Lithuania	370	0.10
Lithuania Mobile	370	0.32
Luxembourg	352	0.04
Luxembourg Mobile	352	0.34
Macau	853	0.08
Macedonia	389	0.29
Macedonia Mobile	389	0.49
Madagascar	261	0.36
Malawi	265	0.19
Malaysia	60	0.07
Maldives	960	0.39
Mali	223	0.29
Malta	356	0.16
Malta Mobile	356	0.49
Mariana Islands	1670	0.10
Marshall Islands	692	0.60
Martinique	596	0.07
Martinique Mobile	596	0.48
Mauritania	222	0.35
Mauritius	230	0.26
Mayotte Island	269	0.43
Mexico	52	0.12
Mexico Mobile	521	0.35
Micronesia	691	0.39
Moldova	373	0.19
Moldova Mobile	373	0.29
Monaco	377	0.07
Monaco Mobile	377	0.39
Mongolia	976	0.19
Montenegro	382	0.29
Montenegro Mobile	382	0.49
Montserrat	1664	0.29
Morocco	212	0.23
Morocco Mobile	212	0.38
Mozambique	258	0.17
Mozambique Mobile	258	0.39
Myanmar	95	0.49
Namibia	264	0.16
Namibia Mobile	264	0.38
Nauru	674	1.40
Nepal	977	0.25
Netherlands	31	0.03
Netherlands Mobile	31	0.29
Netherlands Antilles	599	0.19
New Caledonia	687	0.49
New Zealand	64	0.04
New Zealand Mobile	64	0.39
Nicaragua	505	0.37
Niger	227	0.19
Nigeria	234	0.17
Niue	683	0.97
Norfolk Island	6723	1.50
North Korea	850	0.78
Norway	47	0.03
Norway Mobile	47	0.29
Oman	968	0.28
Pakistan	92	0.07
Palau	680	0.65
Palestine	970	0.39
Palestine Mobile	970	0.42
Panama	507	0.08
Panama Mobile	507	0.19
Papua New Guinea	675	0.75
Paraguay	595	0.25
Peru	51	0.29
Philippines	63	0.19
Philippines Mobile	63	0.24
Poland	48	0.03
Poland Mobile	48	0.15
Portugal	351	0.03
Portugal Mobile	351	0.33
Puerto Rico	1787	0.04
Qatar	974	0.29
Reunion Island	262	0.15
Reunion Island Mobile	262	0.50
Romania	40	0.06

Country	Code	Rate $
Rwanda	250	0.19
Saint Kitts and Nevis	1869	0.38
Saint Lucia	1758	0.38
Saint Vincent	1784	0.38
Samoa	685	0.77
San Marino	378	0.06
San Marino Mobile	378	0.39
Sao Tome and Principe	239	1.90
Saudi Arabia	966	0.16
Saudi Arabia Mobile	966	0.24
Senegal	221	0.26
Senegal Mobile	221	0.35
Serbia	381	0.19
Serbia Mobile	381	0.39
Seychelles	248	0.22
Sierra Leone	232	0.19
Sierra Leone Mobile	232	0.39
Singapore	65	0.03
Slovakia	421	0.08
Slovakia Mobile	421	0.30
Slovenia	386	0.06
Slovenia Mobile	386	0.47
Solomon Islands	677	1.50
Somalia	252	0.69
South Africa	27	0.09
South Africa Mobile	27	0.29
South Korea	82	0.03
South Korea Mobile	82	0.09
Spain	34	0.03
Spain Mobile	34	0.25
Sri Lanka	94	0.16
Sri Lanka Mobile	94	0.19
St. Helena	290	1.95
St. Pierre and Miquelon	508	0.36
Sudan	249	0.16
Suriname	597	0.38
Swaziland	268	0.25
Sweden	46	0.03
Sweden Mobile	46	0.29
Switzerland	41	0.04
Switzerland Mobile	41	0.45
Syria	963	0.29
Taiwan	886	0.04
Taiwan Mobile	886	0.12
Tajikistan	992	0.14
Tajikistan Mobile	992	0.16
Tanzania	255	0.25
Thailand	66	0.04
Thuraya Satellite	88216	1.25
Togo	228	0.24
Tokelau	690	1.18
Tonga	676	0.34
Trinidad and Tobago	1868	0.19
Tunisia	216	0.25
Tunisia Mobile	216	0.35
Turkey	90	0.09
Turkey Mobile	90	0.29
Turkmenistan	993	0.19
Turks and Caicos	1649	0.37
Tuvalu	688	0.84
Uganda	256	0.19
Ukraine	380	0.15
Ukraine Mobile	380	0.21
United Arab Emirates	971	0.25
United Kingdom	44	0.03
United Kingdom Mobile	44	0.25
Uruguay	598	0.14
Uruguay Mobile	598	0.36
USA	1	0.03
Uzbekistan	998	0.12
Vanuatu	678	1.30
Venezuela	58	0.05
Venezuela Mobile	58	0.26
Vietnam	84	0.12
Wallis & Futuna Islands	681	1.00
Yemen	967	0.24
Zambia	260	0.10
Zambia Mobile	260	0.25
Zimbabwe	263	0.14
Zimbabwe Mobile	263	0.58