PPU College Of Engineering and Technology

The Home of Competent Engineers and Researchers

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Electrical and Computer Engineering Department

Communication Engineering Program

Graduation Project

Follow me

"Landline/mobile bridge via internet "

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جامعة بوليتكنك فلسطين

الخليل-فلسطين

كلية الهندسة والتكنولوجيا

دائرة الهندسة الكهربائية والحاسوب

اسم المشروع

Follow me

Landline\mobile bridge via PPU net

أسماء الطلبة

أنغام سلهب — بيان الهشلمون

رزان الجعبري — صفاء الحداد

بناء على نظام كلية الهندسة والتكنولوجيا و إشراف ومتابعة المشرف المباشر على المشروع وموافقة أعضاء اللجنة الممتحنة تم تقديم هذا المشروع إلى دائرة الهندسة الكهربائية والحاسوب وذلك للوفاء بمتطلبات درجة البكالوريوس فس الهندسة لتخصص هندسة الاتصالات والالكترونيات .

توقيع المشرف

....

توقيع اللجنة الممتحنة

....

توقيع رئيس الدائرة

....

Dedication

To our prophet Mohammed peace be upon him.

To who gave us life and hope and passion for knowledge and reading.........our parents

To the man who learned us how to research, how to dream , to our supervisor Eng. Ayman Wazwaz.

To our teachers... ...

To our colleagues...

To Pbx programmer Abd Albaset Jabari ..

Acknowledgment

This report is the first and foremost a collective effort, which was not successful without these cooperative efforts. All praise and thanks to Allah, which we estimated to complete this work.

And many thanks to our supervisor Eng.Ayman Wazwaz who had always been a beacon of science through his advice and supervision which invaluable.

Thanks also to everyone who contributed to the completion of this work.

Abstract

In this graduation project, a cost effective and practical method of communication is proposed. The focus is on resolving the situation that is faced when the person you are trying to reach through the landline is not available to answer. It is common in the large companies to use the GSM (Global System for Mobile Communications) network to contact the mobile of the desired subscriber. Despite the fact that GSM can do the job, but it is expensive..

Instead, in our project, a free cost effective and more practical method of third party (the system) communication is suggested and discussed. This is briefly the coexist of landline connections to the mobile throw GSM system by the well known internet protocol 'VoIP', which will enable the application of this concept even in the small scale of homes or small offices, schools and universities.

We built our system using all the needed hardware components and software programs, and we created away to interface them with each other, to completely achieve the system objectives in efficient way.

الملخص

في زمن التطور والتقدم التقني المتسارع في مجال الشبكات أصبح هناك اهتمام في عملية توفير الوقت واختصار المسافات لتسريع عمليات التبادل واتخاذ القرار والتحكم بالأعمال بشكل أفضل .ومن هذا الباب توجه كثير من أصحاب الشركات ورجال الأعمال نحو هذه التقنيات لتضمن لهم عملية إدارة سهلة لأعمالهم.

إحدى التقنيات التي ظهرت في هذا المجال هي تقنية نقل الصوت عبر بروتوكول الانترنت VoIP والتي سنستخدمها في مشروعنا بإذن الله . عن طريق تحويل مكالمات الهاتف الأرضي الفائتة إلى الأجهزة المحمولة بدون تكلفة باستخدام هذه التقنية مقارنة بنظام الاتصالات النقالة GSM باستخدام شبكة الاتصال اللاسلكية Wi-Fi .

وهنا تم شرح خصائص و ميزات هذه التقنيات و كيفية ارتباطها بالمشروع . و التخطيط لما سيتم تنفيذه في الفصل القادم حيث سوف نقوم بتطبيق المشروع في الطابقين الرابع والخامس , مبنى B واد الهرية بالارتباط مع مقسم الجامعة .

لقد قمنا باستخدام جميع الاجزاء والبرمجيات الازمة لاتمام المشروع , وقد قمنا بعمل ربط بين هذه الأجزاء بطريقـــة متكامله , لتحقيق أفضل كفاءة وعمل للمشروع.

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Chapter Six :
Results and conclusion

Figures

Tables

Abbreviations

A

ACK Acknowledgement

A\D Analog to Digital

AP Access Point

ATA Analog Telephone Adapter

C

CO Central Office

CDR Call Detail Reports

D

D\A Digital to Analog

DHCP Dynamic Host Configuration Protocol

DNS Domain Name System

DTMF Dual-Tone Multi-Frequency

E

EXT Extension

F

FXO Foreign Exchange Office

FXS Foreign Exchange Station

H

HTTP Hyper Text Transfer Protocol

I

IEEE Institute of Electrical and Electronics Engineering

IP Internet Protocol

IVR Interactive Voice Response

IM instant messaging

L

LAN Local Area Network

M

MMS Multimedia Messaging Service

O

OS Operating System

P

PBX Private Branch Exchange

PCM Pulse Code Modulation

PLC Packet Loss Concealment

POE Power Over Ethernet

POTS Plain Old Telephone Service

PT Proprietary Telephone

PSTN Public Switched Telephone Network

Q

QoS Quality of Service

R

RJ Radio Jockey

S

SIP Session Initiation Protocol

SMS Short Messaging Service

V

VoIP Voice over Internet Protocol

W

WEP Wired Equivalent Privacy

Wi-Fi Wireless Fidelity

WLAN Wireless Local Area Network

1

Chapter 1

Introduction

1.1Overview

In this chapter, we will provide a brief summary and description of the system . Starting with objectives, we will show the major and minor goals that we are targeting to achieve by the end of this project. In addition, the physical hardware and software requirements needed to accomplish will be listed along with the proposed approach . at the end of this chapter, related works will be reviewed with a short description of the additions and advantages that the proposed system will add over what has been done before.

1.2Objectives (aim of the project):

Fast and easy communications are needed for growing communities, where internet is becoming a primary component inside our life. The use of internet for faster, easier and cheaper connections has always been a dream and by the advancement of telecommunications it is used more in the different applications. The goal of our project is to reduce missed calls by people and enhance the communications between them by re-converting convectional landline to landline calls to a mobile connection using the advanced phone appliances such smart phones. The system will enable free calls between the landline office and smart phone within the network, improving the quality of the voice and the security of connections. Thus the primary and secondary goal of this work can be summarized as follows:

- Achieving cheaper communication using VoIP applications .
- Achieving faster and easier communication method that is more practical and less complexity.

- Enhancing the commutations security by using an authenticated application .

1.3 Requirements:

In order to design and implement the targeted system , the following hardware and software requirements will be needed:

Hardware:

- Mobile smart phones.
- Landline phones.
- PBX(Private Branch Exchange).
- ATA(Analog Telephone Adapter)

Software:

- Elastix server
- Apache TOMCAT server.

1.4 Challenges :

- In our system we need IP PBX which is not available so we will built IP PBX that is compatible with the available analog PBX .
- There is a need to built an android VoIP application that achieves the authentication .

1.5 Assumption :

- The project will be apply to the landlines in offices that receive calls converted by PBX.
- Each office owner may own smart phone supplied with the software for VoIP calls

.

1.6 Approach:

Our system will convert the missed landline calls to VoIP calls such that after a predefined delay the PBX will transfer the call "analog signal " to a trunk that is connected with the Elastix server that will forward the call to the desired SIP phones.

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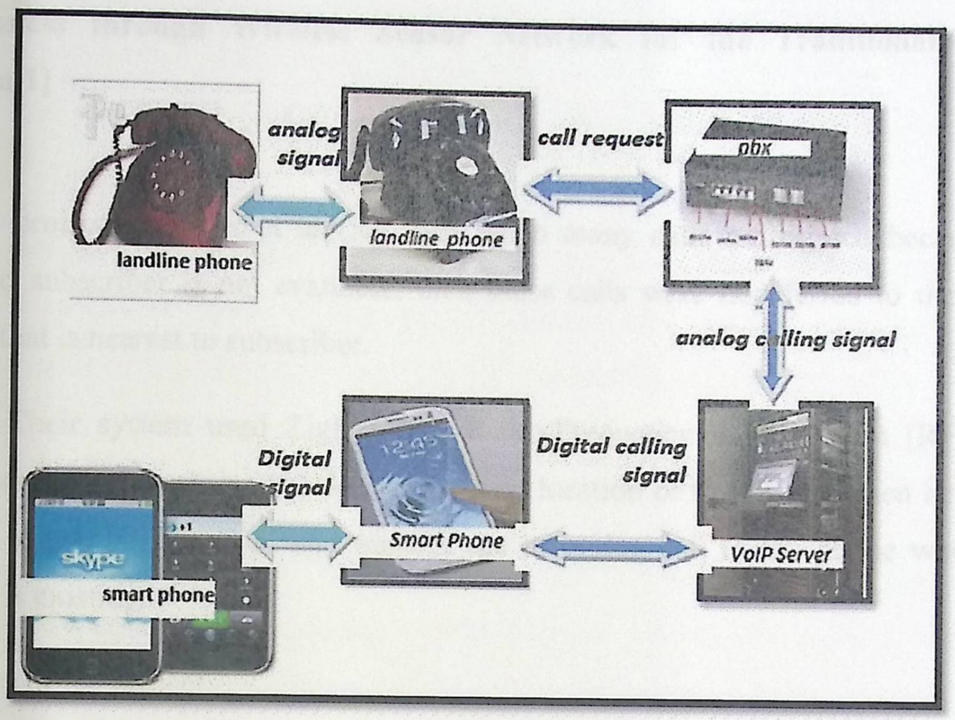

Fig 1.1 Block diagram

1.7 Literature review (Related works):

The advancement of telecommunications sector has been the focus of many researchers around the world, because of its importance to the economies and communities. As will be shown next, many of them have studied the process of transferring calls from one phone to another within a network. Although good results have been achieved, some problems still have to be solved such as voice quality, security level, and price of the transferring technology and the complexity of the system. In addition to summarizing some of the related papers, we will discuss the differences and additions of our project over these projects and works.

1.7.1 An Intelligent Auto Call-Transfer Service Based on Location-Awareness through Wireless Sensor Network for the Traditional Phone System[1]

in this project presented a service, that when many calls are missed because the targeted subscriber is not available, then these calls were transferred to the office phone that is nearest to subscriber.

Their system used ZigBee and Radio-Frequency Identification (RFID) to build positioning system, that detects the new location of the person when he or she moves to another location, and transfer the incoming call to the phone where the person is existing.

Features of the project:

- Low power consumption because ZigBee uses the intensity of the received signal to detect the new location.
- Supports mesh communications and provides multipath to enhance stability of the system.
- It can be used in different types of applications.

Differences to our project :

- Our project will not support the positioning system by ZigBee and RFID.

1.7.2Voice over IP mobile telephony using WIFI p2p[2]

The purpose of this paper is to make communication between mobile phones at no cost using WIFI.

When a mobile user wants to make a call with other mobile user , both mobile numbers will be converted into IP number using NOVEL algorithm , if the called number within the range then a virtual connection will be established between the user using VOIP , if no P2P connection can be done then the calling mobile phone will check if the connection can be done through AP.

If the called number not within the range and not covered by any WIFI a message will appear to the calling mobile number asking if he / she is willing to continue the call through GSM .

Differences to our system :

- They use NOVEL algorithm to convert mobile number into IP number .
- Our system will depend on PBX more than wireless network.

1.7.3 PSTN VoIP Application Support System Design using mobile Short Message Service (SMS): Case Study of PSTN VoIP Missed Call Notification to mobile phone by SMS[3]

This paper aims to remove restrictions that relate to PSTN VoIP through notification of missed calls to subscribers of landlines by sending a SMS to his mobile phone, Principle of the system design through the use of Asterisk softswitch which converts the call in case of answer and send SMS notification if there is no answer, The process begins with " capture signaling packet from Asterisk softswitch " And save files temporarily until the files are analyzed by PERL language , then take decision either forward the call or send SMS to the mobile phone number that saved in database layout in the case of no answer.

They are also aspire in addition to the notification of missed calls, to check PSTN billing, join PSTN number with mobile phone number ,build PSTN VoIP application that support system design from web interface and mobile application.

Features :

The system allows PSTN subscribers to control its PSTN number easily using SMS from their mobile phones

Differences from our projects :

- In our project if no answer ,the landline call will transfer to a mobile phone

1.7.4 IMVoIP Graduation project [4]

This project aims to facilitate communication between the management of the enterprise with officials sections staff by sending messages to sections where officials connects the officials with the administration.

The idea of the project is to set up a local network within the organization through the use of SMS over internet protocol that rely on a server Elastix where users can send and receive messages through mobile devices in order to avoid the use of servers free like Hotmail, which does not have the integrity and confidentiality.

Differences from our system :

- In our system we will raise the level of security by building an authenticated android application.

1.8 Project schedule

- Stage 1 :Select the idea
  Determine the idea of the project, the motivation and the main objective we intend to achieve.

- Stage 2: Preparing for the project
  In this stage ,more and deeper determination of the tasks ,and steps we want to perform, is done.

- Stage3: Project analysis
  In this step, a study of the all possible design options to determine our own design

- Stage 4:Determine the project requirement
  After determine our design scheme we specify all the needed requirement for the user and the system ,software and hardware. and try to bring them to be ready for the implementation stage .

- Stage 5:Studing the principles
  This stage of the project is necessary to study the Wi–Fi ,VOIP server and programming languages .

- Stage 6: Documentation writing
  Documenting the project will begin from the first stage to the last stage.

- Stage 7:make hardware available .

  In this stage the needed hardware mobiles and main server's PC will be brought for the next step.

- Stage 8 : build up the software.

  The programming of the Elastix server and the code of the android application is started and will be downloaded to the mobile.

- Stage 9: operate the system.

  In this stage we achieve the goals of our system by testing the calls.

- Stage 10:Writing documentation .

  The documentation will continue from the first stage to the last one in parallel.

Table 1.1 Timing plane for the first semester

Task/week	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
S1															
S2															
S3															
S4															
S5															
S6															

Table 1.2 Timing plane for second semester

Task/week	16	17	18	19	20	21	22	23	24	25	26	27	28
S7													
S8													
S9													
S10													

Cost of the project :

Components	Price ($)
FXS card (4 port)	250$
FXO card (4 port)	250$
IP phone	200$
Adapter	5 $
RJ 11 splitter	10 $
Core 2 duo	Available
Galaxy s1	Available
(8 port)ATA	483$

1.9 Scenario

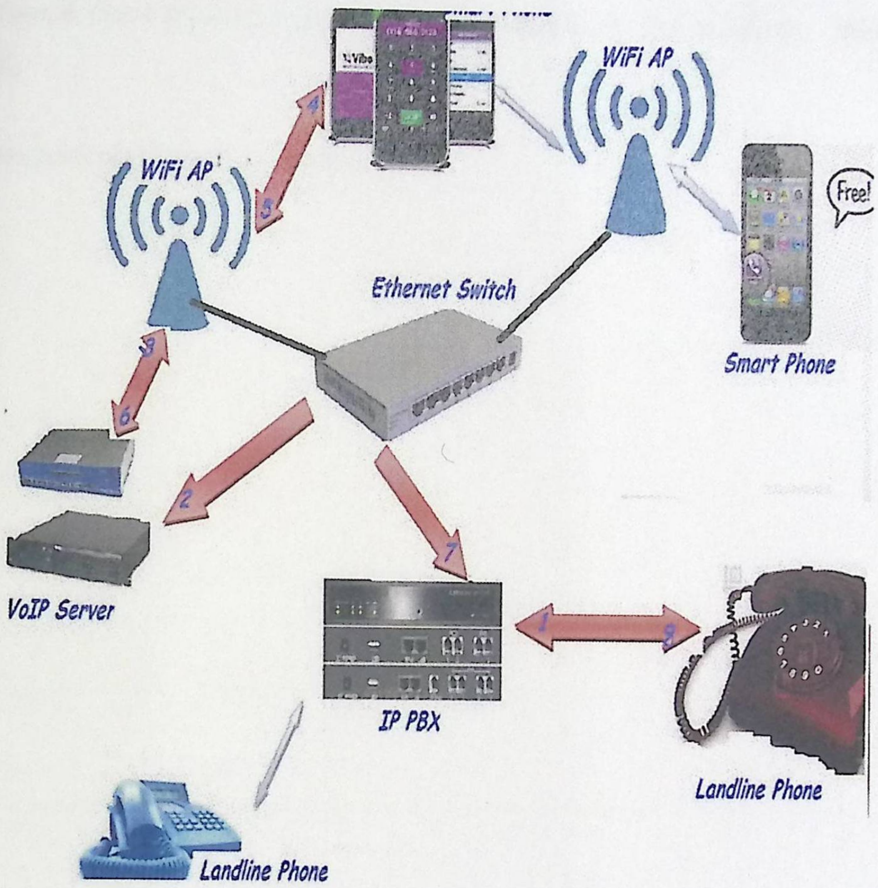

Fig 1.2 System Components

Scenario Steps :-

To set up a call the caller subscriber sends request (analog signal) to the PBX , which turn it into called subscriber. If there is no answer the call(analog signal) will be forwarded to the VoIP Server through twisted pair to get the user name, VoIP server route the call request(Digital signal) to the Wi-Fi access point and it route the request (Digital signal)to the Smartphone, then the Smartphone send accept (Digital signal) to the Wi-Fi access point .which route the setup call signal (Digital signal)to VoIP server and in turn pass the setup call to the PBX via twisted pair ,PBX forward the call setup signal (analog signal) to the landline phone to initiate the call.

Details and cases are explained in section 3.4

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Details and cases are explained in section 3.4

2

Chapter 2

Theoretical Background

2.1Overview.

2.2 VoIP server .

2.4 Smart Phones.

2.4 PBX (Private Branch Exchange).

2.5 FXO and FXS

2.6 IP phone

2.1Overview

In this chapter we will discuss the various technologies that will be used in our project, where each technology will be briefly explained and viewed in terms of advantages and disadvantages. First VoIP technology is discussed showing its importance, positives and negatives. Besides, we will explain using g.711 code to improve voice quality in our system. Further, we will write about SIP as a protocol widely used in the smart phones and servers to enable establishing a VoIP call. After that, PBX technology types are listed with particular focus on the "configuration" type, which will be utilized in our project, and it's construction and building. The smart phones are then compared with traditional old calling devices. Additionally, the programming languages that will be used are discussed and explained. Finally, the Wi-Fi standards and operating systems are added and their advantages and disadvantages are shown.

2.2 VoIP server

2.2.1 What is VoIP?

VoIP (Voice over Internet Protocol) has gained a wide acceptance recently and has become one of the main stream communication technologies.

VoIP is a transmission technique and another way of making phone calls. Its main function is to transmit the voice communications and multimedia field across internet. For instance, if you have a computer with a microphone, a speaker and an internet connection you can communicate by VoIP and also you can make it if you have a mobile and a home phone.

There are some terminologies related to communication over internet protocol such as voice over broadband, internet telephony and IP telephony.[5]

2.2.2 VoIP design

To start a VoIP call, there are group of steps that happened including call signaling and setup, digitization of the analog voice signal, encoding, packetization and transmission of this signal as an IP packet. On the receiver side, similar steps are followed(usually in the reverse order) such as receiving and decoding the IP packets, converting digital signals to analog ones which will reproduce the original voice stream as in figure 2.1

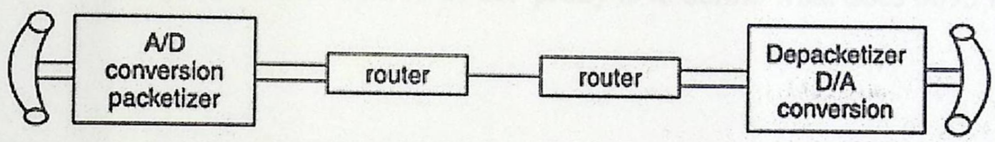

Fig 2.1 Elements of basic operation of VoIP[6]

2.2.3 The protocol of the VoIP (SIP)

What is SIP?

SIP (Session Initiation Protocol)network engineers made a clear distinction between two phases of voice call: setup and transfer. Both use different protocols to transfer the voice packets between two phones.

SIP is a call setup protocol for IP-based communications, it operates at the application layer it is designed to setup multimedia sessions between group of callers in addition to simple telephone system, also SIP is used to setup video, audio multicast and instant messaging conferences.

SIP uses the real time transport protocol in the VoIP to signal and setup traditional telephone calls over IP infrastructure in order to send the voice between callers .SIP session involves client requesting and a server, when the request is received, the SIP) server sends response to the user to indicate the availability of the session.

SIP proxy:

If an SIP phone wants to call another phone at the extension 0095 for example, then the phone translates 0095in to an IP number and sends the call request to its SIP proxy, the main function of SIP proxy is to define what does 0095 mean.[7]

SIP Message :

SIP message is a text protocol with a syntax similar to http protocol (hyper text transfer protocol).

There are two types of SIP messages:

- Requests: which defined the nature of the request, there are different types of SIP request messages such as (such as not such that)register ,invite ,ACK (acknowledgement), cancel, bye and options .
- Responses: there are different types of responses such as: Provisional (1xx), success (2xx), redirection (3xx), server error (5xx)and global failure.

2.2.4 Features (advantages) of the VoIP

- The most important feature of VoIP is the ability to make free phone calls.
- It can be hold either in video or audio calls.

- It is a cheap communication method that can be easily implemented..
- It can establish call conference from each phone in the network.
- VoIP is portable, the mobility and portability is another advantage of VoIP over traditional telephones.

2.2.5 Disadvantages of the VoIP

- Bad voice quality.
- Bandwidth dependency.
- Lack of enough security

.

2.2.6 VoIP voice quality:

One main drawback in making VoIP calls as compared with landline calls is the quality of the voice. Where a satisfactory connection is when you can make a decent call without suffering from echo, noise and delays.

The quality of the VoIP depends on broadband connection, provider service, hardware and the destination of the call.

The data (voice) must be compressed and transmitted, then decompressed and delivered, and all this has to be done in a short amount of time (milliseconds), if it takes more than this time , because of the slow connection and the breakdown of the hardware, then the quality of the VoIP call will be reduced.

To solve this problem, we should do the following:

- Improve the rooming service.
- Improve the mobility.
- Enhance the coverage.

* Enhance the service of the VoIP, broadband connection and utilize high quality hardware.

2.2.6.1 G.711 codec

G.711 is pulse code modulation (PCM) standard for internet protocol (IP) private branch venders,G7.11 packetize the analogue signal at high data rate exceeds 64kilobits per second (kbps) ,G.711 is commonly used in VoIP application to increase quality of the VoIP by giving the priority to the voice over multimedia and so on. G.711 offers new technology to the world it was called packet loss concealment (PLC) which minimize the bad effect of the dropped packets.

There are a group of coding algorithms necessary for the network transformation to packetized voice , the table 2.1 shows the comparison between G.711 codec and other codec's.

Table 2.1 comparison between G.711 codec and other codec's [6]

Coding algorithm	Voice bit rate (Kbps)	Voice frame size (bytes)	Header (bytes)	Packets per second	Packet bit-rate (Kbps)
G.711 8-bit PCM	64	80	40	100	96
G.723.1 MPMLQ(1)	6.3	30	40	26	14.6
G.723.1 ACELP(2)	5.3	30	40	22	12.3
G.726 ADPCM (3)	32	40	40	100	64

G.728 LD-CELP(4)	16	20	40	100	48
G.729a CS-ACELP (5)	8	10	40	100	40

2.2.6.2 Bandwidth dependency:

To make a VoIP call, we must have at least 512 kbps download speed and 128 kbps upload speed for one suite VoIP to make a decent VoIP call.

2.2.7 Security:

Due to possible interference in IP addresses over the global internet network, the level of security in when making VoIP connections is weak point and has to be enhanced. Due to this slow security level, hackers might be able to hack into personal files, steal information or disable large companies' operating systems. In our project we will try to overcome this problem and solve it and this will be shown later.

2.2.8 The relation between VoIP and our project:

The VoIP gate way allows telephone calls to be completed through the IP network ,VoIP server routes the call request (Digital signal) to the Wi-Fi access point to get the IP number of the called subscriber, and after the smart phone send accept (digital signal) through Wi-Fi access point the VoIP server transmits the call setup to the PBX.

2.3 Smart Phones

2.3.1 What is a smart phone ?

Smart phones the term given to phones that have become operate an operating system as a small computer, Which allows browsing the Internet , e-mail and computer applications as well as telephone services such as communication, SMS and camera .

There are many companies producing smart phones with different operating systems such as Symbian OS , Blackberry RIM , Apple Iphone OS, Google Android . Smart phone operating systems differ from computer operating systems where each device its own operating system distinct from other systems by applications offered and the functions are performed .

2.3.2 Smartphone features:

- Increase connectivity.
- Instant access to information via the Internet.
- Became used today to pay the bills and restaurant reservations from the phone's web browser. [8]
- You do not need to additional burden because users carry cell phones anyway.
- Reduce the number of technological devices carried by businessmen and doctors and engineers.
- Many applications provide entertainment and recreation.

2.3.3 Smart phone limitations:

- Small Screen Size.
- Awkward Keyboard Size.
- High Cost.
- Website Access.
- Work-Life Balance.

2.3.4 Android operating system :

Recently observed increasing number of Android devices including HTC, Motorola, Sony Ericsson and Samsung , this is due to that Google provides to companies Android Linux based Operating system code on an open-source It also has applications store contains a thousand application ,Android integrated with Google services such as calendars, Google contacts, email out-of-the-box ,cool maps program with Google's Street View and directions YouTube app, wirelessly purchase songs ,It also supports background applications so that you can run more than one application at the same time[9] .

To accommodate the need to increase the usage and flexibility in the Android operating system , Google opened many programming languages to create applications and the most commonly used is Java language which working on Dalvik Java virtual machine .

Android is an open-source software stack for mobile device led by Google ,it was created to reduce the restriction and control of other operating systems to make it an open source and open platform to create a variety of application and games for all the user of the system and then marketed and distributed , so its operates hundreds of millions of mobile device in more than 190 countries around the world , android based on the Java language to write application codes, Java language features as the easiest and simplest C and C++ , although it derives a large part of the C and C++ syntax.

No longer GSM network is the only method of communication between mobile phones , after the development of smart phones that support VoIP calls through applications provided by Android operating system , Where it can make a call via the Internet connection free of charge except the payment for the use of the Internet ,The most important VoIP applications Skype , Fring , Line 2 , Tango , Viper and Google Voice etc[10] .

VoIP (voice over Internet Protocol) : is a protocol for communication via the Internet , transmission techniques includes voice communications and multimedia sessions over Internet Protocol (IP) networks, in order to make voice over IP call each phone require IP number (address) which consist of four numbers separated by dots, Because it is difficult to deal with this type of numbering it's assigned to a common name in order to facilitate the process of dealing.

IP telephony is a term associated with VoIP which it's Principle of work : " The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream. Even though IP Telephony and VoIP are terms that are used interchangeably, they are actually different; IP telephony has to do with digital telephony systems that use IP protocols for voice communication, while VoIP is actually a subset of IP Telephony. VoIP is a technology used by IP telephony as a means of transporting phone calls ." [11]

2.3.5 Smart phone in our system

The basic function of the smart phone in our project is receiving calls transferred from landline phones by VoIP application this is in order to implement our scenario which is : " when the caller subscriber sends a request (analog signal) to the PBX , which turns it into the called subscriber if there is no answer the call (analog signal) will be converted to(digital signal) by the FXO unit , then forwarded to the VoIP Server through Ethernet switch to get the office owner IP address so the VoIP server routs the call to the wanted smart phone through LAN\ WLAN. The smart phone has a VoIP application to initiate the call , so it will acknowledge the call request to the VoIP server through LAN\WLAN , the digital signal will be converted to analog signal by the FXO unit to be forwarded to the PBX which switch the call to the landline phone ."

For this purpose, each landline phone owner needs smart phone loaded by VoIP application

2.4 PBX (Private Branch Exchange)

2.4.1 What is PBX ?

A PBX is a switch station for telephone systems. It consists mainly of several branches of telephone systems and it switches connections to and from them . it allow user to share phone lines instead of having a dedicated phone line for each user , so each user is able to have a phone without the high cost of various dedicated phone lines. Private branch exchanges used analog technology originally [12].

The PBX consists of :-

- **Endpoints**

Endpoints consist of ways that allow user to access the PBX system. They include telephones, fax machines, modems, PDAs and computers. PBX allow the transmission of the analog and digital information. This includes voice and data transmission. Within the system, features are offered such as call waiting tones. These endpoints are connected to each other through phone wires.

- **Gateway Interfaces**

Gateway interfaces are a key component to PBX systems, which will allow user to connect to outside location just like the original telephone works ,Without these gateway interfaces, calls outside of the system cannot take place.

- **Switchboard**

The main purpose of it is to find the path to complete the call from the user to the endpoint .And it can work manually or by using computer . Most systems today are all automatic, meaning no one is needed to transfer calls. The switchboard interconnects all phone lines, internally and externally. It is make connecting to the extension that the user trying to call.

- **Control Processor**

It is the most important part in the system where the control is done in it . Most PBX systems now use a computer application such as Microsoft, Unix or Linux. These applications are programmed to meet the company's needs. These applications then direct the system where to go and what to do.

2.4.2 Advantages of a PBX

- Calls can be answered with a conditional greeting , directed, and transferred automatically
- The most important benefit is reducing the cost
- Processes high call volumes efficiently and quickly
- Automatic call forwarding when extensions are busy

- A dial-by-extension directory for incoming callers
- Ring groups for departments
- Call recording
- Call logs
- Music on hold
- Call waiting
- Conference calling
- Call screening
- Voicemail to email

2.4.3 How a PBX Telephone System Works

- Incoming PBX Calls

PBX allows the transfers of calls from a public switched telephone network to a private switched network. The incoming calls is routed through a private switching system to a telephone with a private number .[13]

- Outgoing PBX Calls

Internal telephones on a PBX network enters a code, usually 0 or 1, to access an outside line. Once the private line is connected to an outside line, the caller dials the call normally.

- Internal PBX Calls

Calls between line in the PBX work similarly to calls between private lines on a traditional public switched network. But the calls on the PBX are run over private, internal telephone lines and switches.

2.4.4 Types of PBX

- Analog PBX

It uses a "Phone Box" that connects several phones using phone lines and it send voice and signaling information as actual analog sound. The information never gets digitalized. Each hope will listen to the call before it directs it further, and it need a VOIP gateway to connect exchange.

- Digital PBX

Digital PBXs encode analog sound into a digital format that can be more easily processed by a computer Once the sound is encoded using a standard industry audio codec, G.711. The digitized information is sent on a channel using circuit switching. Digital PBXs can also support analog trunks.

Circuit switching sets up an end-to-end open connection. It leaves the channel open for the length of the call and for the callers' exclusive use. A VoIP gateway is required to connect Exchange . [14]

- Hybrid PBX

A hybrid PBX is generally provide a digital PBX with IP PBX capabilities this hybrid approach allows a customer to run a mixture of digital and IP-based communication . This PBX is popular because it allows users to have the best of both worlds. Older phones without a network chip can still be used, but they can also be upgraded to include network chips if the funds for an upgrade become available.

- IP PBX

IP PBXs are designed to carry voice over IP (data) networks. The IP phone contains a Network Interface Card so it is part of the network. The phone converts

2.5.4.1 How to Program a PBX

- Log in to the PBX administration portal with a computer and connectivity software. Most manufacturers supply this software for the PBX with purchase of the system.
- Set up the dial plan for the system. The dial plan tells the system what the extension numbers are to begin with and how to get an outside line, feature access codes and the number for the attendant console. Set up the number 9 to gain an outside line and 0 for the operator console.
- Configure the system parameters and feature access codes. System parameters set up music on hold, caller id, system alerting and various call treatment options. Feature access codes are numbers you can dial to access system features such as conferencing, speed dialing, call park, call coverage, call pickup and call transfer among others. Use one to three digit numbers and make sure they do not match any portions of numbers you have planned for extensions.
- Set up Class of Restrictions and Class of service tables .These tables set up groups of permissions to assign to users. The permissions are features such as long distance dialing, call forwarding availability, priority calling and others. Restrict long distance calling to only those users that needs access with these tables.
- Configure the trunks and outbound routing. The trunks are the phone lines used for inbound and outbound calls connected to the local exchange carrier, sometimes referred to as the telephone company. Set them up in groups. Trunk groups are trunks or phone lines that service the same types of calls. Configure them in local dialing and long distance groups. Set up the routing so the outbound calls travel out the least expensive path.

2.5.2 Wi-Fi advantages and disadvantages

Much like everything else in the world, Wi-Fi has its own advantages and disadvantages. By means of the growing popularity of small portable devices ,Wi-Fi proves to be more beneficial by minimizing expensive deployment , provides user mobility and supporting high bandwidth and Quality Of Service (QOS) . In addition new protocols for the QOS and power saving mechanisms make Wi-Fi more suitable for latency sensitive applications such as voice and video services .

A few challenges need to be considered since Wi-Fi networks are limited range , varies according to the frequency band used in which the 2.4 GHz band (802.11b/g) has a greater range than the 5GHz band (802.11a). [17]

An additional delay with Wi-Fi is regarded as the overload of access points (AP's) , Wi-Fi can handle ten of communicating clients simultaneously , however many 802.11b and 802.11g access points default to the same channel ,leading to a congestion on certain channels.

2.5.3 (IEEE) 802.11 standards

There are several specifications in the 802.11 family developed by the IEEE :

1) 802.11a : which applies to a Wireless LAN (WLAN) and provides up to 54Mbps in the 5GHz band uses encoding scheme such that OFDM rather than FHSS or DSSS .
2) 802.11b : which applies to a Wireless LAN (WLAN) and provides up to 11Mbps in the 2.4GHz band uses only DSSS.
3) 802.11g: which applies to a Wireless LAN (WLAN) and provides up to 54Mbps in the 2.4GHz band uses for short distance transmission .
4) 802.11e: this standard that defines the QOS support for LANs with two types : prioritized QOS and parameterized QOS .

5) 802.11n: this standard adds MIMO to increase data throughput through spatial multiplexing and increase the range , the real speed is 100Mbps , so up to 4-5 times faster than 802.11g .
6) 802.11i : which enhances the wireless network with the security that is stronger and better suited to voice .
7) 802.11r: which called Fast Basic Service Set transition that supports fast handoff between access points , specifically in order to enable VoIP roaming on a Wi-Fi network with 802.1X authentication [18].

A Wi-Fi enhanced device can connect to the Internet when it is within the limited range .

802.11b and 802.11g access points have arrange of 32m indoors and 95m outdoors but IEEE 802.11n can more than double the range .

Wi-Fi in the 2.4GHz frequency has slightly better range than Wi-Fi in the 5GH frequency that is used by 802.11a .

2.5.4 Wi-Fi main issues

- Wi-Fi Security

Wi-Fi technologies are not inherently secure . The security can be divided into two categories :

1) Solutions built into Wi-Fi standards .
2) Solutions independent of Wi-Fi , but used to support the standard .

The solution with security built into Wi-Fi standard include techniques such as Wired Equivalent Privacy (WEP) , Temporal Key Integrity Protocol (TKIP) . The solutions that use additional security is not built into Wi-Fi standards include techniques such as IP security (IPSec) , Secure Sockets Layer (SSL), point to point tunneling protocol (PPTP) encryption .

- Wi-Fi interference

Wi-Fi faces an interference with all devices working on the same band .a minimum distance of about 5m is recommended to avoid such interference . The spectrum for the 2.4 GHz band is divided into 11 channels (1-11) , Wi-Fi signal occupies 5 channels , any two channels number that differ by 5 or more such as 2 and 7 don't overlap . The channels 1 ,6 ,11 are the only group of 3 non over lapping channels in the U.S .

- Wi-Fi mobility

In addition to connecting to one access point it also needs to be able to disassociate from it current access point. The 802.11r standard is currently being developed that allows for fast hand-offs. The standard allows a user to connect and negotiate security and QOS settings before dissociating with its current access point. This allows a mobile user to maintain service as access points are changed.

2.5.5Voice meets Wi-Fi

Communicating using voice over Wi-Fi is something many people are paying close attention to . taking the advantages of a decreased communications cost and easy connections . where VOIP consists of the hardware and software that enables people to use the Internet as a transmission medium used for telephone calls .

Wi-Fi based VOIP service is the wireless version of this technology that is designed to work on wireless devices .[19]

The voice applications such as VOIP will increase the number of connections into the hundreds which will make it very easy for an access point to be over loaded if some of the signals are not passed off to an access point with less devoted resources .

Quality of Voice is a complex issue and is being addressed in at least two ways. In the standard 802.11e priority queues are utilized to help ensure the throughput of Voice and other multimedia traffic. Voice quality is also determined by the codec that is chosen and implemented. The codec is important, because the sampling and the compression ratio can drastically decrease the quality. A score called MOS is used to determine the quality of telephony .

2.5.6 The major wireless network components:

- **Router**

The Enterprise Router is basically a Layer-3 Network device that connects disparate networks. It acts as a gateway between the LAN and the WAN networks and the Internet Leased Lines/ MPLS Circuits/ Managed Leased Lines/ Broadband networks are all terminated on the router. Some Routers support additional modules for secure connectivity to other branches through VPN, Intrusion Prevention and Content Filtering etc. Routers have WAN ports and LAN ports to connect WAN and LAN connections respectively, and some of them have built in Wireless/ VoIP capabilities.

- **Wireless Access Points**

The Wireless Access Points contain built in radios which provide wireless signals for connecting certain network devices that has an in-built wireless adapter. Basically these access points send wireless signals that can be interpreted by the wireless enabled network clients for communicating the data/ information over the wireless medium. Their job is just to collect these signals, convert them in to wired signals and send it over the LAN network for the wireless controller to interpret them and take appropriate action. They generally have a coverage range of 20-30 meters indoor and 80-100 meters outdoor and each device can connect to more than 15 wireless devices within their coverage area. They operate in the 2.4 and 5 Ghz frequency spectrum.

- Network Endpoints/ Devices

There are various network devices/ endpoints connecting to the LAN via edge switches wireless access points. Some of them include PC/ Laptop/ PDA etc for data connectivity, IP Phones, Cell Phones/Wi-Fi Phones, Soft Phones for voice connectivity, IP Surveillance Cameras/ IP Video Conferencing devices for video over IP. There are also network based accessories like network printers, MFP's (Multi-Function Printers), Scanners etc. connecting to the enterprise computer network.

- IP Telephony Server

The IP Telephony Server provides the call control functions (voice switching) for the telephony operations in an enterprise network. Since the IP Phones connect to the computer networks, these IP Telephony Servers provide centralized administration and connectivity to PSTN Lines to all the IP Phones/ VOIP devices over the network including the assigning of extension/ DID numbers and IVR (Interactive Voice Response).

- Edge Switches

The Edge/ endpoint switches are basically Layer-2 switches that provide direct connectivity to the various network devices like PC's, laptops, Wireless Access Points etc using the Copper UTP cables. They come in various configurations including 8 Port/ 16 Port, 24 Port, 48 Port etc. They support 10/100 Mbps as well as 10/100/1000 Mbps connectivity to the various network devices. Some of them even support POE (Power Over Ethernet) for electrical power required for operation of certain network devices (like Wireless Access Points, IP Phones etc) and some of them could be stacked to each other for providing a single management interface/ combined backplane for multiple such edge switches.

2.6 FXO and FXS

2.6.1 FXO (Foreign Exchange Office):

It receives POTS (plain old telephone service) typically from a Central Office of the Public Switched Telephone Network. And it provides on-hook/off-hook indication (loop closure/non-closure) at the FXS's end of a telephone circuit. [20]

How it works ?

A telecommunication line from FXO port must be connect to an FXS port to work in a correct way and vice versa ,when we connect FXO port on the analog telephone to the FXS port we will receive FXS service from the telephone company and we will hear the dial tone when we pick up the phone .The scenario becomes more interesting when we add additional network elements , such as a Private Branch Exchange (PBX) or a Voice-over-IP gateway or router .As example we can connect the FXO interface on a phone to the FXS port supplied by a PBX , multiplexer, or Voice-over-IP gateway or router.

PBX connection

When we connect the PBX to the Telco Central Office, the (FXS) lines must be plugged from the phone company into FXO ports on the PBX. The FXO ports on the PBX provide on hook/off-hook indication (loop closure) to the local Telco network

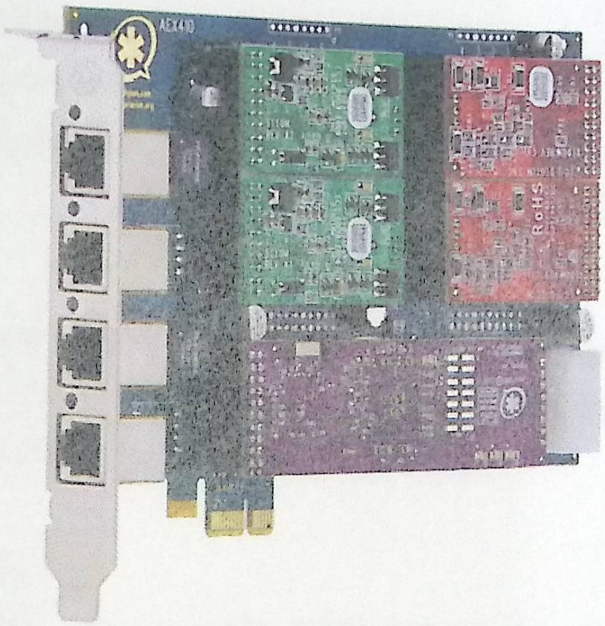

Figure2.3(4 port) card

There is many ports in the FXO and FXS cards up to 24 ports , in the figure 2.3 4-port card .The modular nature of the cards allows you to mix and match between line (FXO) and station (FXS) interfaces, giving you exactly the ports you need.

2.6.2 FXS (Foreign Exchange Station)

It delivers POTS service from the local phone company's Central Office (CO) and must be connected to subscriber equipment (telephones, modems, and fax machines). FXS is interface points to the subscriber[20]

It provide these service :

- Dial tone
- Generates ring voltage
- Battery current .

An FXS interface utilizes an FXO protocol to detect when the terminating device (telephone) goes on-hook or off-hook, and can send and receive voice signals. An FXS interface provides service at the station end of a foreign exchange line.

PBX Connections

When you connect a PBX to analog phones, you plug phone cables into FXS ports on the PBX. The FXS ports on the PBX provide POTS service, including battery current, ring voltage, and dial tone to the phones.

2.7 IP phones .

IP phone uses voice over IP(VoIP) technologies ,instead of using the ordinary system public switch telephone network (PSTN) ,and this allowing telephone calls to be made over an IP network such as the internet .

The phones uses Session Initiation Protocol (SIP) or skinny client control protocol(SCCP) as control protocol .

Elements of VoIP phones :

- Physical Hardware
- DNS client
- STUN client
- DHCP client (not commonly used)
- Signaling stack (SIP, H.323, Skinny, Skype, or others)
- RTP stack
- Codec

2.7.1 Advantages of VoIP phones

- Save a lot of many , International calls are much more expensive Since VoIP uses the Internet as backbone, the only cost you have when using it is the monthly Internet bill to your ISP.
- With VoIP, you can setup a conference with a whole team communicating in real time.
- more calls can be handled on one access line because VoIP packets data during transmission .
- Preserving user name/ number when choosing a different service provider (not widely supported).
- Call forwarding, call waiting, voicemail, and caller ID .

2.7.2 Disadvantage of VoIP phones

- Needs Electric Power
- Emergency Calls
- Sound Quality And Reliability - Some VoIP providers have problems with sound quality and reliability.
- Requires Internet access to make calls outside the local area network (LAN).

2.7.3 The relationship between IP phone and our project :

IP phone will be utilized instead of normal landlines in the case where the extensions inside the PBX are limited.

3

Chapter three:

Design concepts

3.1 Overview

In this chapter, we will show the main design of our system with more details and explanations. Through an informative block diagrams, system main flow chart , and we will demonstrate the main components and elements that constructs the project idea and discuss their relations with each other and with the overall work as shown in fig 3.1 .

3.2 Main block diagram

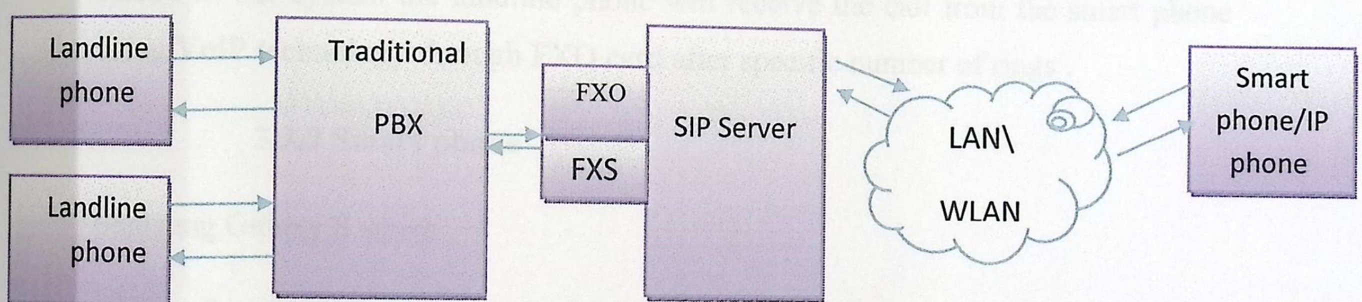

Fig 3.1 Main block diagram .

3.3 The main components of the project :

Hardware components :

- Landline phones .
- Smart phones .
- PBX
- VoIP server (Elastix).
- FXS
- FXO

3.1 Overview

In this chapter, we will show the main design of our system with more details and explanations. Through an informative block diagrams, system main flow chart , and we will demonstrate the main components and elements that constructs the project idea and discuss their relations with each other and with the overall work as shown in fig 3.1 .

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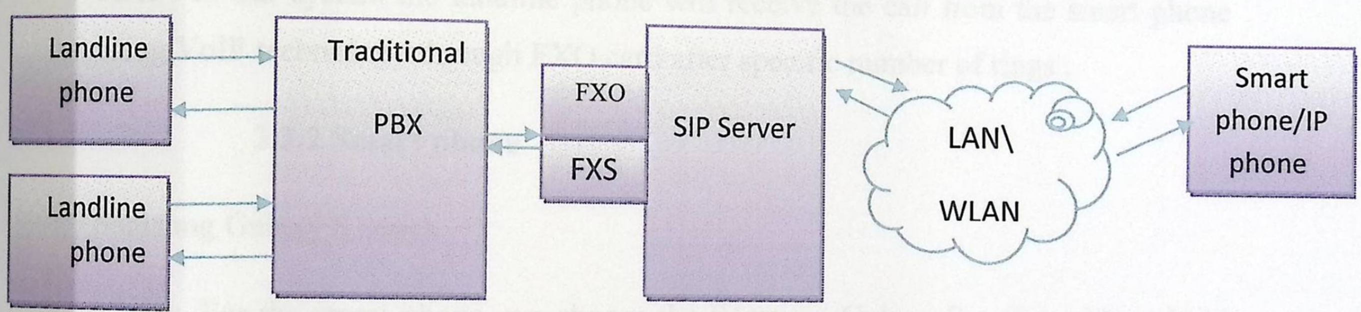


Fig 3.1 Main block diagram .

3.3 The main components of the project :

Hardware components :

- Landline phones .
- Smart phones .
- PBX
- VoIP server (Elastix).
- FXS
- FXO

- Wi-Fi
- Analog Telephone Adapter (ATA)
- Proprietary Telephone (PT).

Software components :

- Android VoIP application.
- Apache TOMCAT server.

3.3.1 Landline phone

Is a transceiver device that sends and receives analog signal to and from the PBX . In our system the landline phone will receive the call from the smart phone using VoIP technology through FXO card after specific number of rings .

3.3.2 Smart phone

Samsung Galaxy S series

For the smart phone, we choose the Samsung Galaxy S series with android operating system since it is fair enough for our project .

3.3.3 Private Branch Exchange (PBX)

In our system we intended to use Panasonic KX-TD1232 but because of the lack of the free extensions we will use the analog Panasonic KX-TES284.

<u>Panasonic KX-TD1232 PBX Features</u> :

- This PBX has 2 Digital Super Hybrid Systems are connected together to double the capacity of the system .
- The 2 systems function as one , each system provides 32 line and extended by a card with 16 line to get an overall system with 96 line , and 8 outside lines from the Public Switched Telephone Networks (PSTN) .
- It is supplied by 32 volt .

- Non-blocking PCM time switching

<u>Panasonic KX-TES284 PBX</u> :

This PBX has 24 internal extensions and 8 external lines , provides 5 (volt) line voltage without PSTN .

We will use this PBX in our system and connect 6 extensions with the ATA because of the difference in the voltage between the PBX line voltage and the FXO voltage in order to handle with IP PBX that forwards the VoIP calls .

In case of Panasonic KX-TD1232 PBX provided with PSTN lines , the 6 extensions will be connected to the FXO units directly since the line voltage of the PBX is compatible with the FXO voltage.

3.3.4 PT.

Is a device that is used in the programming of (PBX), it contains additional buttons and LEDs for signals for some external and internal lines, through which direct calls can be performed without the need of providing numbers. In order to connect the PT to the (PBX) pair of wires are required: the first is called tip and ring for heat transfer, while the second is data 1 and 2 for signals exchange.

3.3.5 VoIP server

Is a Personal Computer Core 2 Duo With Elastix software which is a collection of an Open Source products and tools compiled together to become an integrated IP PBX. We choose Core2Due since it is sufficient in our project , it provides 2.8GHz CPU and 4GB RAM where the minimum Elastix hardware requirements are :

- CPU x86 with a 500MHz CPU and 256MB RAM.

- 10/100 Mb/s network card

3.3.5.1 The IP PBX Components

Four main components needed to set up VoIP PBX:

- Elastix, the Asterisk powered IP PBX
- The soft phones
- The VOIP gateway (FXO ,FXS) services that lets us call other VOIP users and people on the PSTN.
- LAN / WLAN network with broadband access, a router and switches .

The Elastix server has a public IP which is 195.3.190.19 , a private IP 10.10.75.195, gateway 10.10.75.254 . The server is plugged with FXO/FXS units in order to handle with an IP PBX .

3.3.6 Foreign Exchange Station FXS .

FXS Call Procedures

1) Call Initiation

An FXS device initiates a call by presenting ring voltage over the line to the attached FXO device. (FXS devices cannot pass dialed digits.) [20]

2) Call Reception

An FXS device receives a call by

1) Detecting the line has been seized (the attached telephone–FXO device–has gone off hook)

2) Receiving Dual-Tone Multi-Frequency (DTMF) digits indicating how the call should be routed.

3) Line Power

FXS devices supply approximately 50 volts DC power to the line. During an emergency, FXO devices can use FXS line voltage for power in order to remain operable in the event of a local electrical power failure.

4) FXS Call Clearing

Under normal case an FXS device does not initiate call clearing. Instead, FXS devices rely on the two parties at each end of the call to recognize the call has ended (by hearing the line go quiet); then the FXO device at each end clears its segment of the call.

How it works in our project ?

We will connect the FXS to the PBX Central Office (CO) lines by the interface RJ-11 in the case when a SIP phone calls a landline directly .

3.3.7 Foreign Exchange Office (FXO)

1) Call Initiation

An FXO device initiates a call by

1) Going off-hook to seize the telephone line.

2) Dialing the Dual-Tone Multi-Frequency (DTMF) digits, which identify the destination to be called.[20]

2) Call Reception

An FXO device receives a call by

1) Detecting the ring voltage supplied by the FXS device .

2) Going off-hook to answer the call.

How it works in our project :

We will plug the FXO in the server in order to handle with an IP PBX in a case when a landline calls the SIP phone directly.

3.4 Generic process

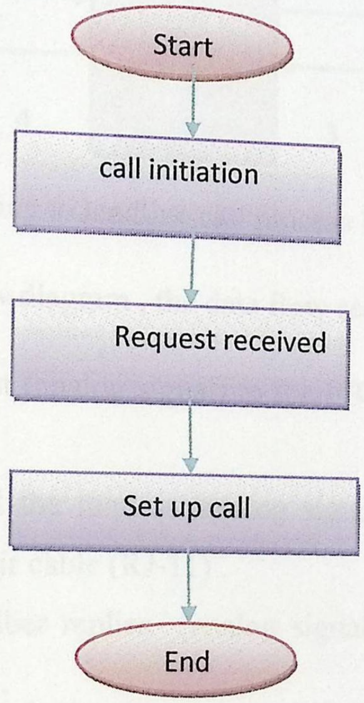


Fig 3.2 call setup process (generic)

The whole system can be divided into 4 cases :

- Landline to landline call.
- Landline to smart phone call
- Smart phone to landline call .
- Smartphone to smart phone call .

3.4.1 Landline to landline call.

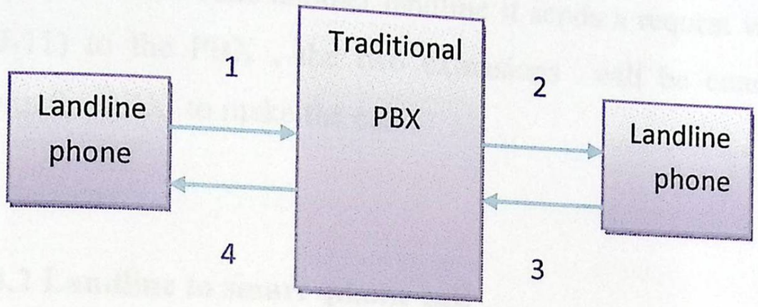


Fig 3.3 Landline to landline call process Block diagram

As numbered in the data flow diagram , the data flow sequence is:

1. Send a call request (analog signal) to the PBX through twisted pair cable (RJ-11).
2. The PBX forward the request (analog signal) to the called subscribe through twisted pair cable (RJ-11) .
3. The called subscriber replies (analog signal) to the PBX through the twisted pair cable .
4. Call setup is done .

The data sequence flow chart :

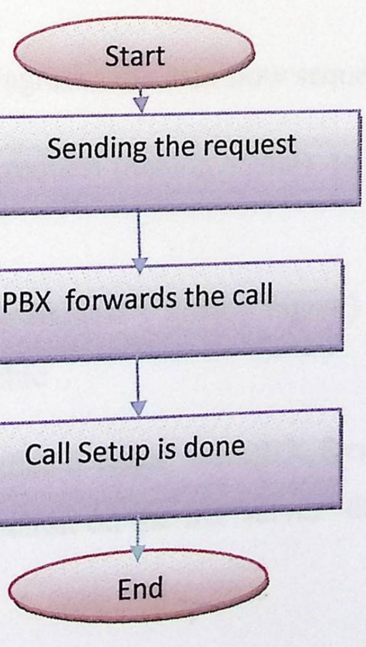


Fig 3.4 Landline to landline call process flowchart

When a landline calls another landline it sends a request via twisted pair cable (RJ-11) to the PBX , the two extensions will be connected together internally in the PBX to make the call .

3.4.2 Landline to smart phone call

3.4.2.1 In Panasonic KX-TD1232 PBX with PSTN lines

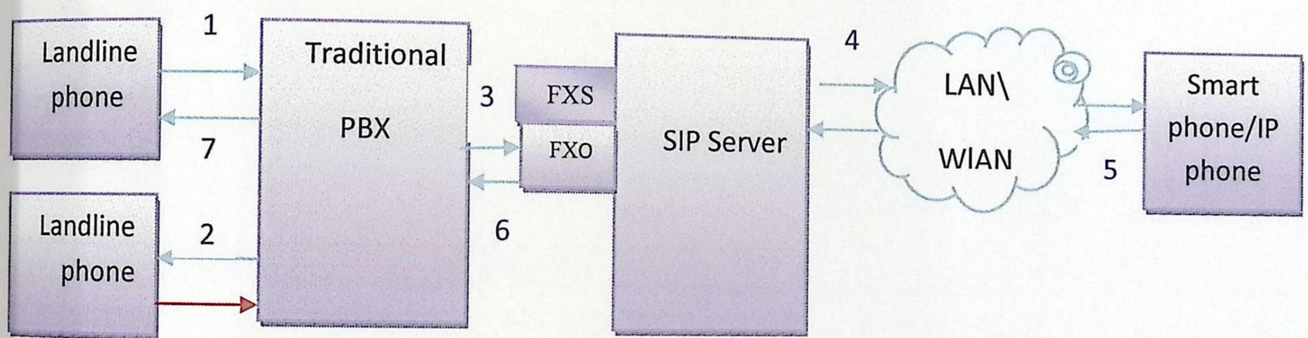


Fig 3.5 Landline to smart phone call process with PSTN Block diagram .

As numbered in the data flow diagram , the data flow sequence is:

1)The landline sends a call request (analog signal) to the PBX through twisted pair cable (RJ-11) .

2) The PBX forwards the request (analog signal) to the called subscriber through the twisted pair cable .

3) If there is no answer after 3 rings the PBX forwards the request (analog signal) to the FXO connection on the SIP server through twisted pair cable (RJ-11) .

4) Depending on the callee' status if it's registered or not , the call will be forwarded through Wi-Fi to the wanted smart phone.

5) The called subscriber replies (digital signal) to the SIP server through LAN/WLAN

6) In SIP Server after digital to analog conversion by the FXO , replies (analog signal) transferred to the PBX through twisted pair cable .

7) Call setup is done .

The data sequence flow chart :

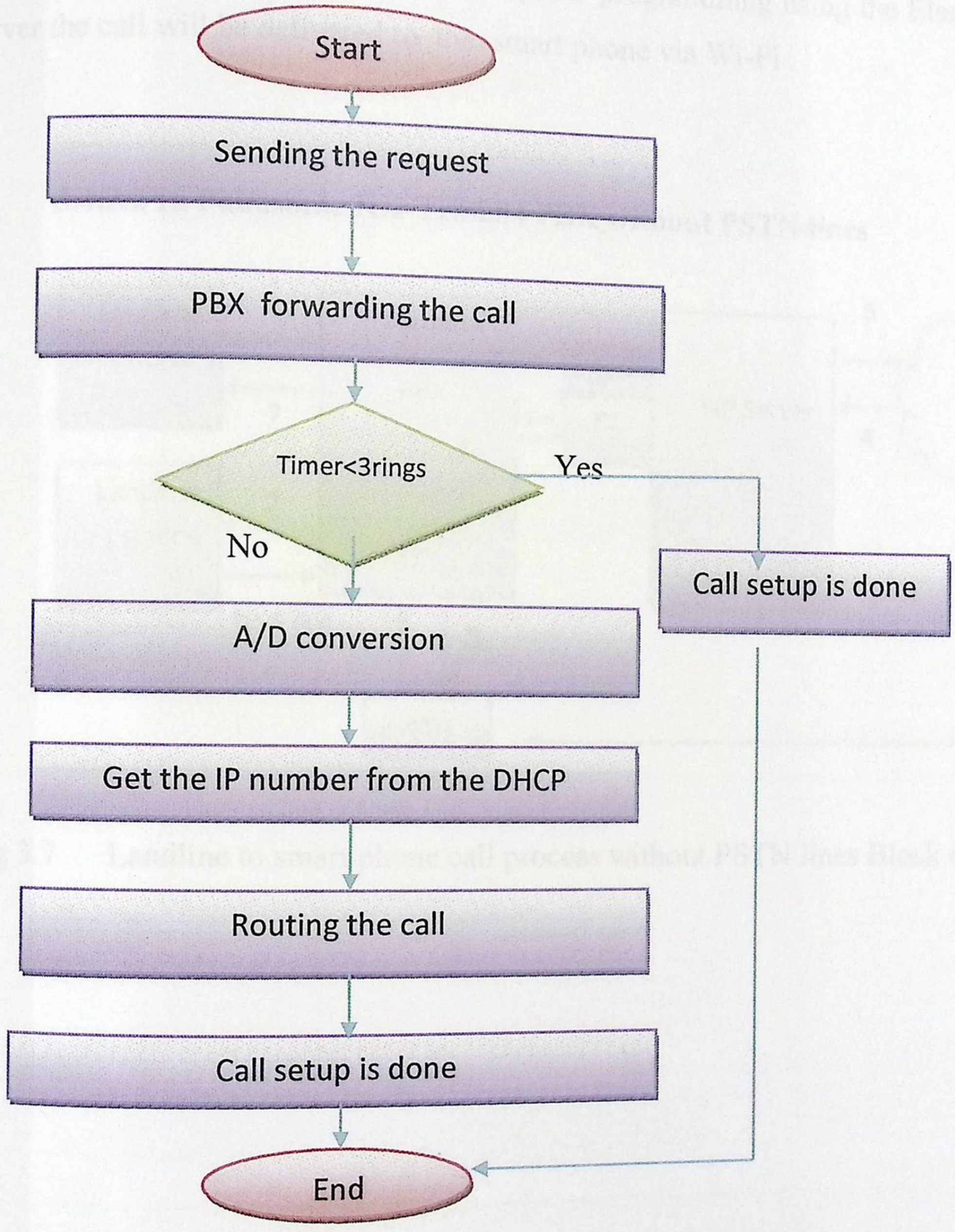


Fig 3.6 Landline to smart phone call process with PSTN lines flowchart.

When there is a missed call after 3 rings , the PBX will transfer the call to the FXO in the server through twisted pair cable (RJ-11) , the FXO will convert the analog signal to digital signal ,and by a prior programming using the Elastix in the server the call will be delivered to the smart phone via Wi-Fi .

3.4.2.2 In Panasonic KX-TES284 PBX_without PSTN lines

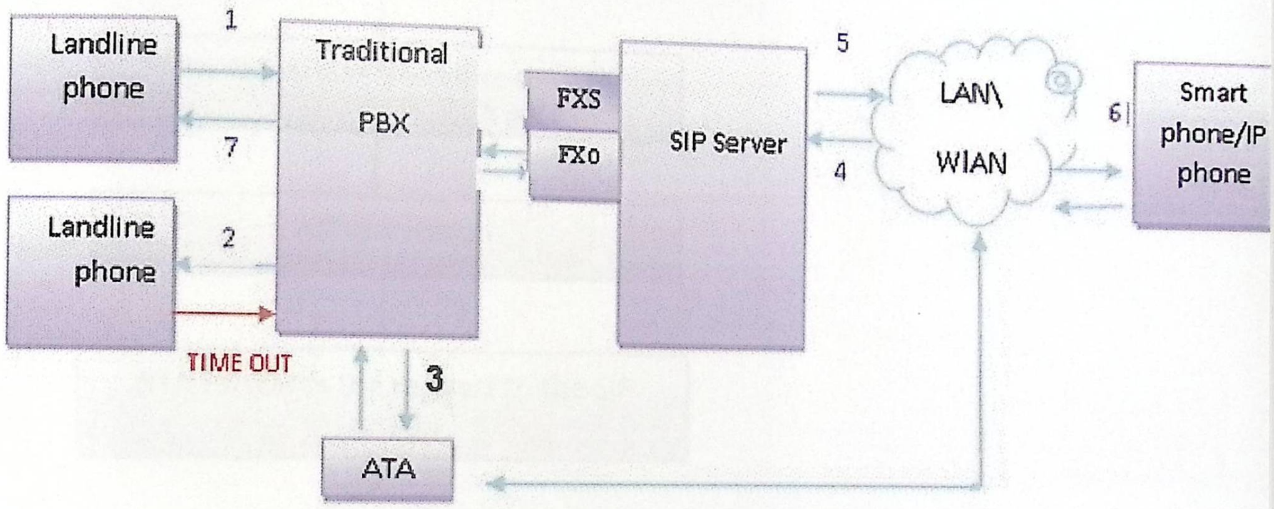


Fig 3.7 Landline to smart phone call process without PSTN lines Block diagram

The data sequence flow chart :

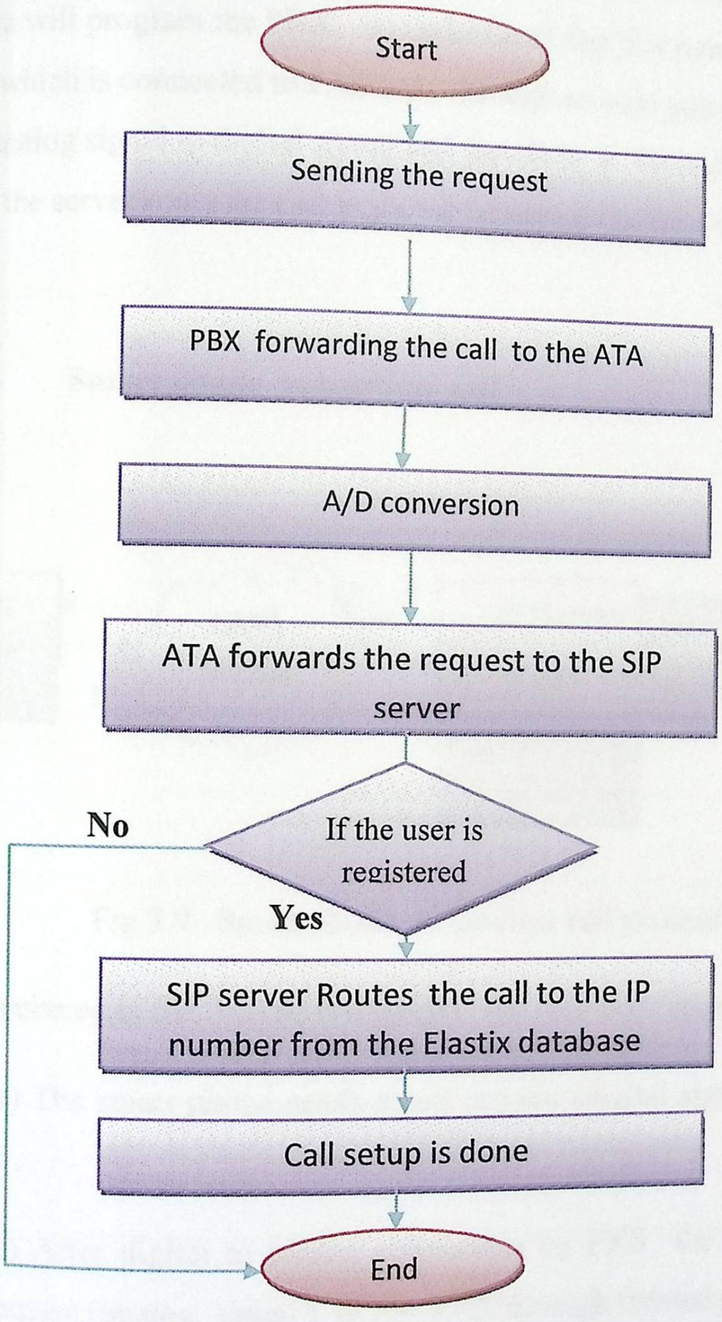


Fig 3.8 Landline to smart phone call process without PSTN lines flowchart

When we want to make a VOIP call from a landline to a callee' smart phone in direct way without calling his landline because we know already that he has left his office , we will put a specific number such that 9 before the smart phone number , so we will program the PBX to understand that this number means take an external line which is connected to FXS unit through twisted pair (RJ-11) that will convert the analog signal to digital signal and forwards it through Wi-Fi to the server and then the server routs the call to the smart phone via Wi-Fi to make the call .

3.4.3 Smart phone to landline call .

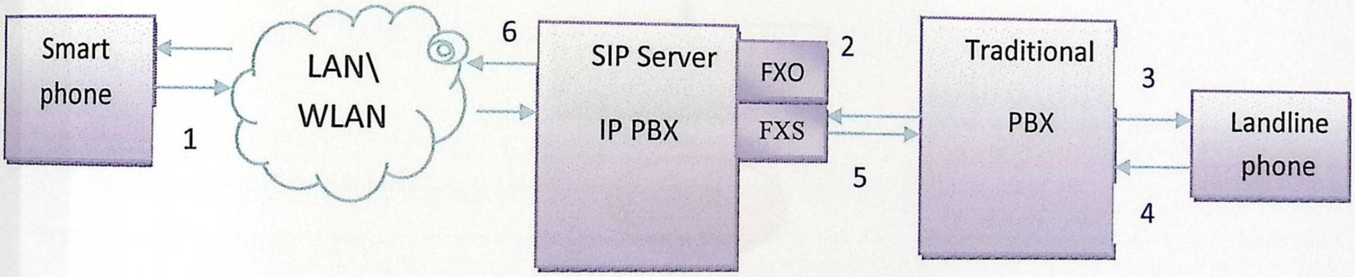


Fig 3.9 Smart phone to landline call process Block diagram

As numbered in the data flow diagram , the data flow sequence is :

1) The smart phone sends a call request (digital signal) through Wi-Fi to the server .

2) After digital to analog conversion by FXS the SIP server forwards the request (analog signal) to the PBX through twisted pair cables.

3) The PBX forwards the request (analog signal) to the landline through twisted pair cables.

4) The landline replies (analog signal) to the PBX through twisted pair cables

5) The PBX replies (analog signal) to the SIP server through twisted pair cables

6) Call setup is done.

The data sequence flow chart :

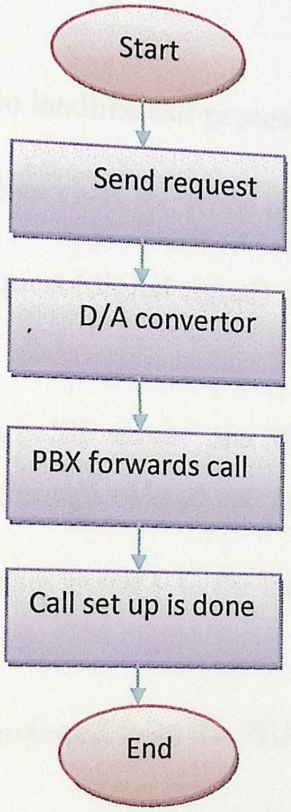


Fig 3.10 Smart phone to landline call process flowchart

3.4.4 Smart phone to smart phone call

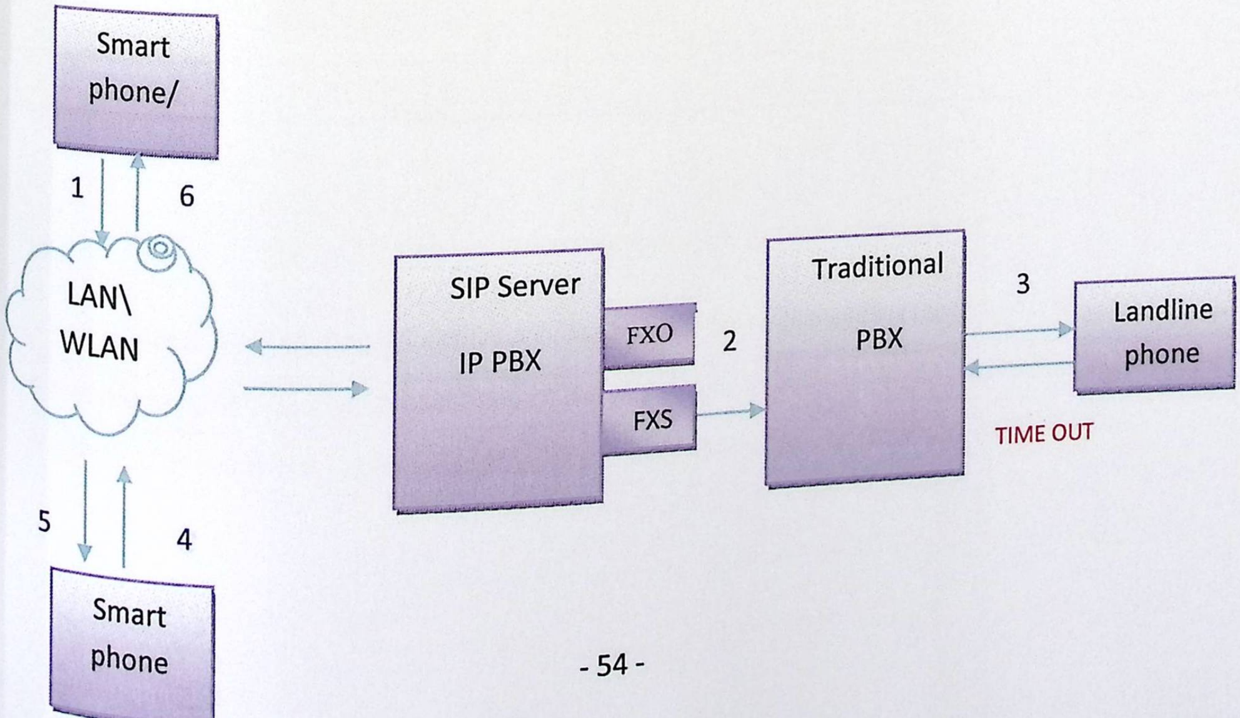


Fig 3.11 Smart phone to landline call process Block diagram

As numbered in the flow chart , the flow chart sequence is :

1) The smart phone sends a call request (digital signal) to the SIP server through Wi-Fi .

2) After digital to analog conversion in SIP server ,the SIP server forwards the request (analog signal) to the PBX through twisted pair cables.

3) The PBX forwards the request (analog signal) to the landline through twisted pair cables .

4)If there is no answer the request transferred from the PBX to the SIP server through twisted pair cable .

5) In SIP server after analog to digital conversion ,the server gets the IP number of the smart phone from the DHCP depending on the callee' username and password then routes the call through Wi-Fi to the wanted smart phone.

6)Call setup is done.

The data sequence flow chart :

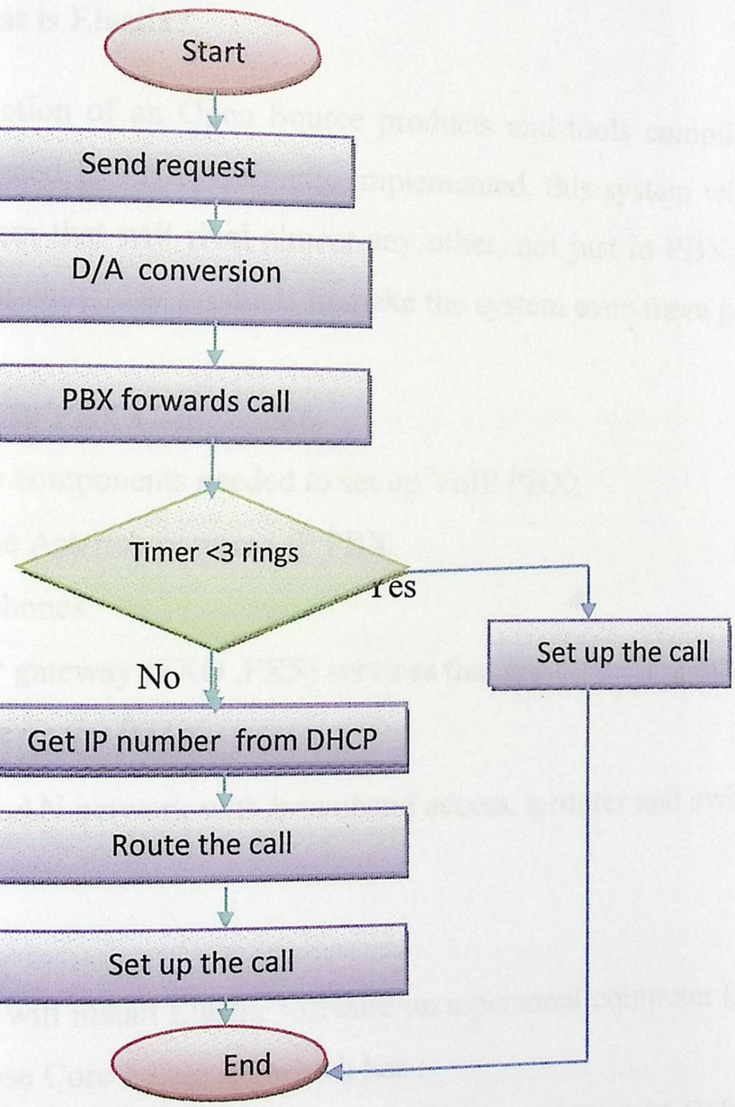


Fig 3.12 Smart phone to smart phone call process flowchart

When a smart phone wants to make a call to smart phone it sends a request to the server via Wi-Fi , in the server FXS convert the signal to analog and then forward it to the PBX through twisted pair cable and when the time is out the PBX transfer it to the SIP server then analog to digital conversion by FXO and rout it via Wi-Fi to the smart phone .

3.5 Elastix

3.5.1 What is Elastix?

Is a collection of an Open Source products and tools compiled together to become an integrated IP PBX. Correctly implemented, this system will provide you with a PBX system that will rival almost any other, not just in PBX functions, but ability to integrate with other products to make the system even more powerful.

3.5.2 The IP PBX Components

Four main components needed to set up VoIP PBX:

- Elastix, the Asterisk powered IP PBX
- The soft phones
- The VOIP gateway (FXO ,FXS) services that allows calling other VOIP users and people on the PSTN.
- LAN / WLAN network with broadband access, a router and switches .

IP PBX

In our project we will install Elastix software on a personal computer (PC) to run the IP PBX. We choose Core 2 Duo PC which has :

- Processor : Intel (R) core (tm) is CPU 760 @ 2.8 GHz,2.93 GHz.
- 4GB RAM .
- 32 bit operating system.

Phones/Landlines

We use SIP phones which support the VOIP service . We will run the softphone like 3CX on another PC.

VOIP gateway (FXO , FXS)

To communicate with others on the PSTN network we need to obtain a VOIP gateway. In our project we will use 4-ports FXO an 4-ports FXS.

LAN / WLAN

We will pick a static IP address to the IP PBX that is in the 5th floor (10.10.75.195) table 3 below :

Table 3. 1 : Elastix database

Floor number	Extension	Username	Password
Floor 4	229	Dr.Ramzi Quasma	123aaa
	227	Ibrahim Abu Shukur	123bbb
	219	Ahmad Qdemat	123ccc
	219	Ali Amro	123ddd
	219	Shehda Zahda	123eee
Floor 5	254	Ayman Wazwaz	123fff
	254	Dr.Murad Abu Subeih	123ggg
	254	Dr.Osama Ata	123hhh
	255	Dr.Amal Duweik	123iii
	255	Elayan Abu Gharbia	123jjj
	255	Naseem Qteet	123kkk
	226	Dr.Ghandi Manasra	123lll
	226	Abdalla Irman	123mmm
	226	Khalid Dghamin	123nnn

We set the (Domain Name System) DNS Server to10.2.0.11
And the default gateway to 10.10.75.254

Access point

Access point (AP) in the project acts as an intermediary between the server and the mobile phone and we use it to send the request from the mobile to the server and vice versa.

3.6 Quality of Service (QOS) of the VoIP

3.6.1 What is Quality of Service in the VoIP ?

In general, quality can mean many things in networking . In the VoIP, quality simply means being able to listen and speak clearly with continuous voice, without echo and unwanted noise. Quality of service in the VoIP depends on the following factors:[21]

- Latency(delay).
- Bandwidth .
- Jitter.
- Packet loss.

•Delay : Too much delay on a voice call can make the quality of the call Unacceptable. all voice communications have some amount of delay. Typically, with VoIP, optimum call quality includes an end-to-end delay of less than 150ms.

• Jitter : Jitter is variations in delay of packet delivery. If traffic (packet delivery) over a connection is constantly delayed at 100 ms, no issue occurs. However, if for the first portion of the call there is short delay (e.g., below 5ms), followed by a period of long delay (e.g., over 300ms), the jitter become an important issue, because of the jitter the voice packets of the same flow are not received at the same time therefore , jitter buffer are introduced to reduce the jitter effect and make the conversation smoothly, it puts the packets in a queue and this buffer queue can be fixed or adaptive . .

• Bandwidth: The amount of the bandwidth identifies whether a call will work correctly or not, VOIP Bandwidth consumption naturally depends on the codec used.. The codec selected for use over a specific line is dictated by the amount of available bandwidth and the number of active calls required.

• packet loss : Small amounts of loss (< 1%) over the connection is Unimportant ,but if this loss becomes larger ,then significant loss in voice quality occurs.

3.6.2 How to achieve QoS

The QoS configurations will be such that they appropriate voice over other data types. VOIP quality of service depends on the parameters that we Previously mentioned like a delay, jitter and packet loss. VOIP QoS is improved by controlling the values of these parameters to be within the acceptable range. VoIP uses codec G.711 to improve the quality of the voice in the Bandwidth (one direction) NEB = 87.2 Kbps [22].

3.6.3 WLAN and QoS

802.11e is a proposed enhancement to the 802.11a and 802.11b wireless LAN (WLAN) specifications. It offers quality of service (QoS) features, including the prioritization of data, voice, and video transmissions.

3.7 VoIP application

There is a need to build an android application to improve the system security by connecting this application with a database .

3.8 Authentication

Like any other system in the world ,our system must be secured so we expect to connect our Elastix server with our university server (trusting center) which is secure and we want to relate each extension with the instructor' username , mail and password in the two servers so that each call must be forwarded from Elastix server to the trusting center to insure the authentication .

4

Chapter 4

Detailed Design

4.1 Overview :

This chapter will focus on the software and hardware design and implementation of each part of the system independently .

We talked about the type of the PBX that we will use in the project and the free extension that we can use it from our university PBX and how to program the PBX using PT device also we talked about the features of the FXS and FXO module We show the complete hardware component connection in figure 4.1 .

4.2 scenario :

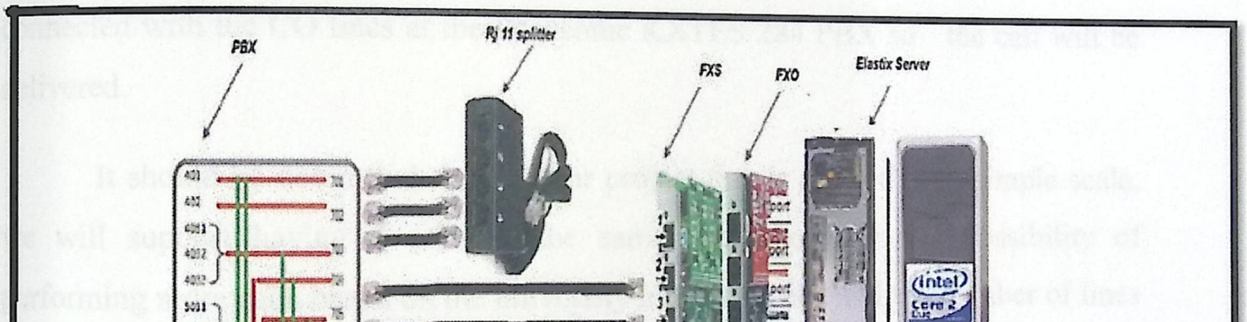


Fig 4.1 General scenario

General scenario

In this section we are describing the final scenario of our system, which includes all the tools that will be used and the way of connecting them together such that the goals of the system are achieved.

In our system ,at the first case if there is a landline missed call after 3 rings the call will be forwarded by the ATA to the SIP Elastix sever, which in turns forwards the call to the desired SIP extensions through Wi-Fi .

In the second case when the SIP phone calls the landline phone directly the call will be received by the SIP server and then forwarded by the FXS units that is connected with the CO lines at the Panasonic KXTES 284 PBX so the call will be delivered.

It should be noted that through our project that is applied on a simple scale, we will support having 4 calls at the same time, however the possibility of performing more calls based on the university network speed and the number of lines will be analyzed and presented.

4.3 Panasonic KX TD 1232 hardware

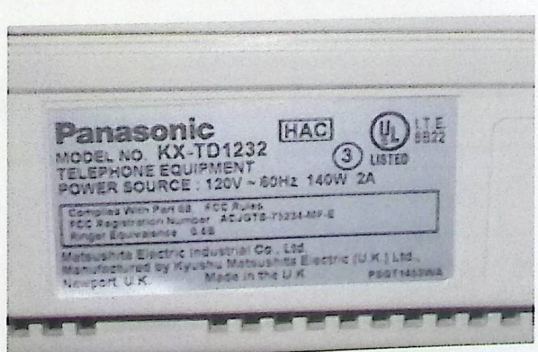
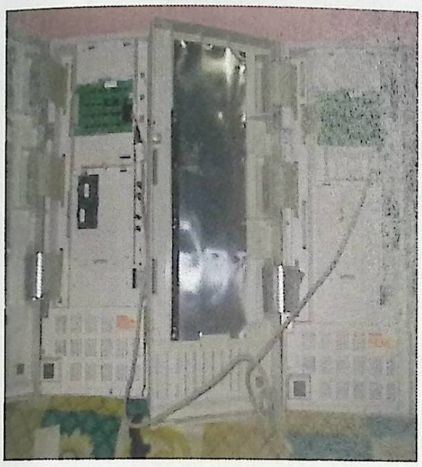


Fig 4.2Panasonic KX- TD 1232

- Power source : 120v – 60Hz , 140W , 2A.

.



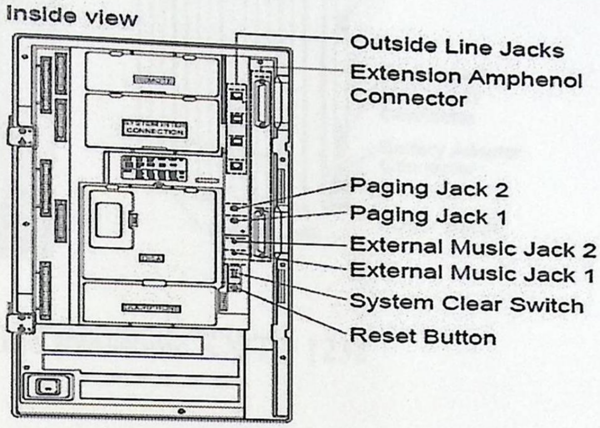


Fig 4.3 Internal view

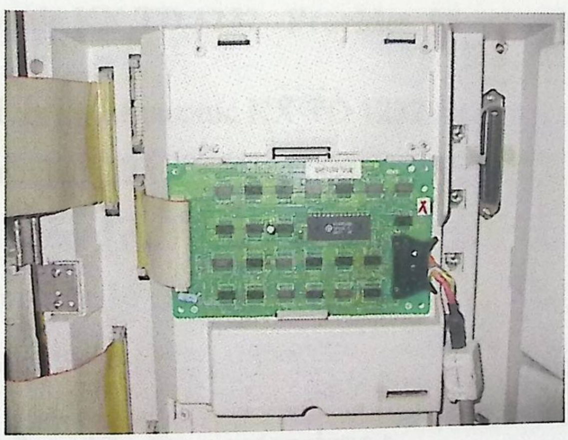


Fig 4.4Panasonic KX TD 1232 cabinet

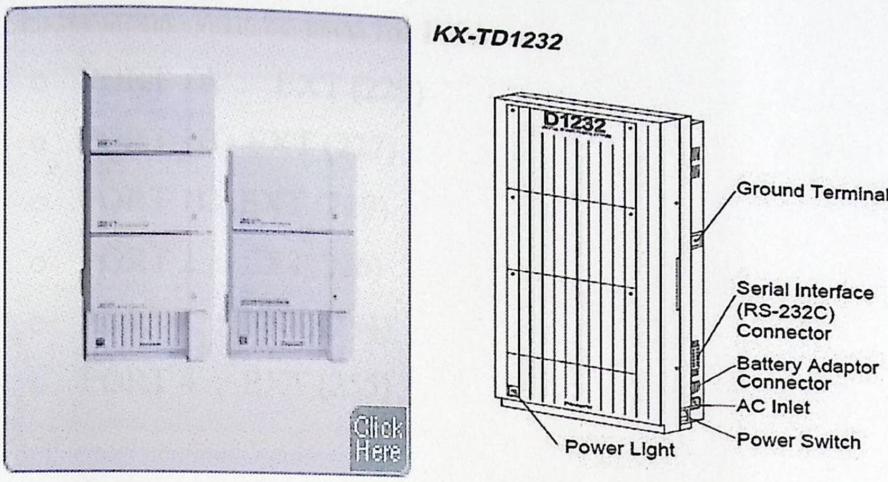


Fig 4.5 Hybrid Panasonic KX TD 1232

4.3.1 Hybrid Panasonic KX TD 1232

Hybrid Panasonic KX TD 1232 Provides 96 extensions:

- Two Hybrid Panasonic KX TD 1232 .
  each provide 32 extensions .
- Two Extended card
  each provide 16 extensions

4.3.2 12 - Free extensions :

- 6 Extensions will be used for FXS .
  - PORT 1A : EXT (229)
  - PORT 1B : EXT (227)
  - PORT 1C : EXT (219)
  - PORT 2 : EXT (226)
  - PORT 3 : EXT (254)
  - PORT 4 : EXT (255)

- 6 Extensions Will be used for FXO.
    - PORT 1A : EXT (229)
    - PORT 1B : EXT (227)
    - PORT 1C :EXT (219)
    - PORT 2 : EXT(226)
    - PORT 3 : EXT (254)
    - PORT 4 : EXT (255)

4.3 Panasonic KX TD 1232 software

There is a way to control the PBX by a Proprietary Telephone (PT) :

4.3.1Proprietary Telephone

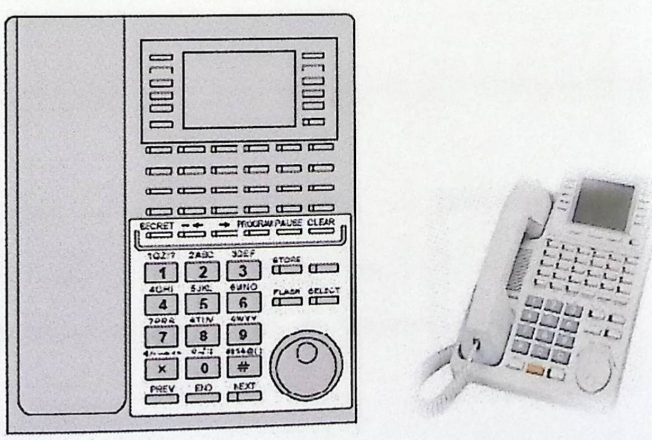


Fig 4.6 proprietary telephone for Panasonic KX TD 1232

4.3.1.1 How to control PBX using PT

We will control the PBX through digital orders :

- **Extensions number set :**

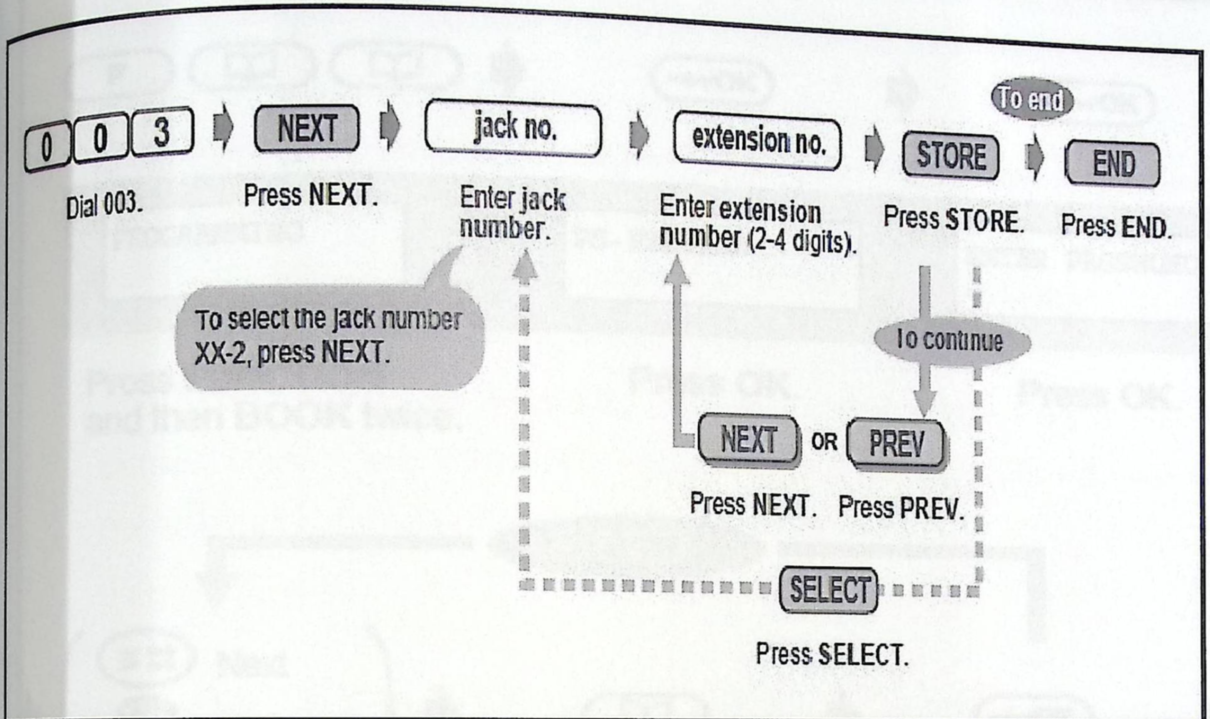


Fig 4.7 Adding extensions steps[23]

- **How to Specify the number of rings before convert it to the smart phone ?**
  - Display : (Auto ANS delay = 4 rings)
  - Item select the number of rings before transfer it .
  - Selection :select 1 ring , or add it manually

Shown in the figure 4.8

Through changing the initial setting :

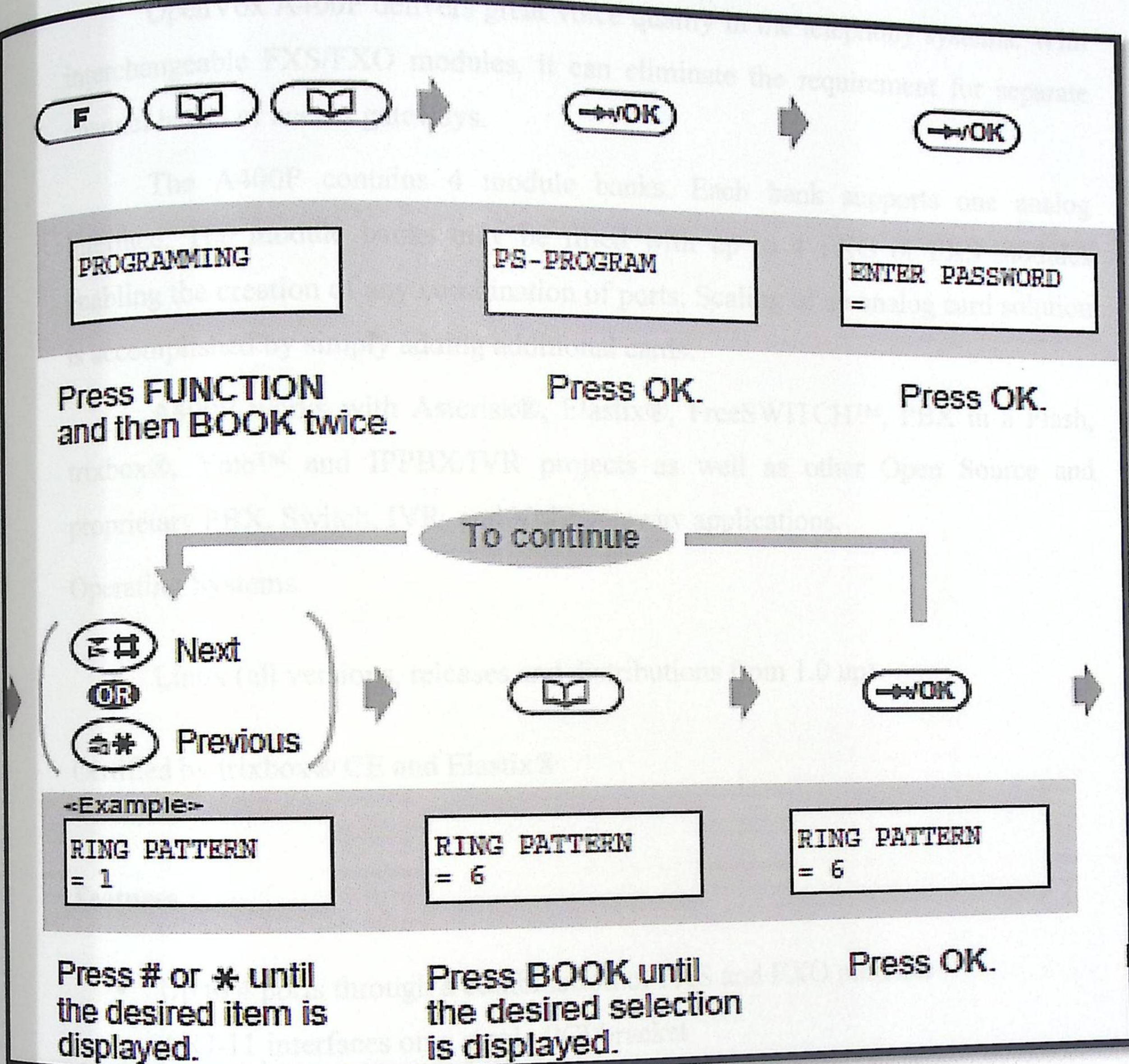


Fig 4.8 Specify delay rings steps[23]

4.4 FXO and FXS modules

4.4.1 FXO module

Product ID: A400P04

Manufacturer: OpenVox Communication Co. Ltd.

OpenVox A400P04 Analog PCI Card; 4-Port FXO

OpenVox A400P delivers great voice quality in the telephony systems. With interchangeable FXS/FXO modules, it can eliminate the requirement for separate channel banks or access gateways.

The A400P contains 4 module banks. Each bank supports one analog interface. The module banks may be filled with up to 4 FXO or FXS modules enabling the creation of any combination of ports. Scaling of an analog card solution is accomplished by simply adding additional cards.

A400P works with Asterisk®, Elastix®, FreeSWITCH™, PBX in a Flash, trixbox®, Yate™ and IPPBX/IVR projects as well as other Open Source and proprietary PBX, Switch, IVR, and VoIP gateway applications.

Operating Systems

- Linux (all versions, releases and distributions from 1.0 up)

Certified by trixbox® CE and Elastix®

Features :

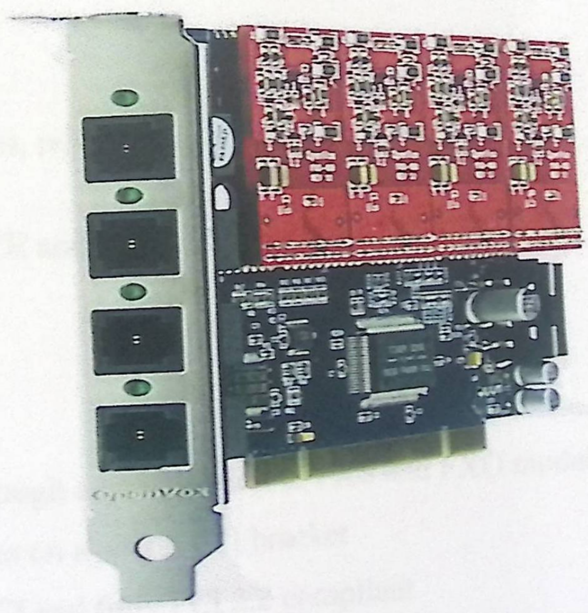
- Up to 4 ports through a combination of FXS and FXO modules
- 4 RJ-11 interfaces on a single PCI bracket
- 32 bit 33MHz PCI and fully PCI 2.2 compliant
- 32 bit bus master DMA data exchanges across PCI interface at 132 Mbytes/sec
- Autosense compatibility with 5 V and 3.3 V PCI busses compatible
- Firmware accelerate I/O access to achieve high stability
- Easy to use.
- Full software and hardware compatible.
- Certified by TrixBox CE and Elastix.

Applications:

- Channel Bank Replacement / Alternative
- Small Office Home Office (SOHO) applications
- Small and Medium Business (SMB) applications
- Gateway Termination to analog telephones/lines

Services and Features

- Caller ID and Call Waiting Caller ID
- ADSI Telephones
- Loop start Signaling Support



4.9 FXO module[24]

4.4.2 FXS module

Product ID: A400P40

Manufacturer: OpenVox Communication Co. Ltd.

OpenVox A400P40 Analog PCI Card; 4-Port FXS

OpenVox A400P delivers great voice quality in the telephony systems. With interchangeable FXS/FXO modules, it can eliminate the requirement for separate channel banks or access gateways.

The A400P contains 4 module banks. Each bank supports one analog interface. The module banks may be filled with up to 4 FXO or FXS modules enabling the creation of any combination of ports. Scaling of an analog card solution is accomplished by simply adding additional cards.

A400P works with Asterisk®, Elastix®, FreeSWITCH™, PBX in a Flash, TrixBox®, Yate™ and IPPBX/IVR projects as well as other Open Source and proprietary PBX, Switch, IVR, and VoIP gateway applications.

Operating Systems

- Linux (all versions, releases and distributions from 1.0 up)

Certified by TrixBox® CE and Elastix®

Features:

- Up to 4 ports through a combination of FXS and FXO modules
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- 32 bit 33MHz PCI and fully PCI 2.2 compliant
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- Firmware accelerate I/O access to achieve high stability

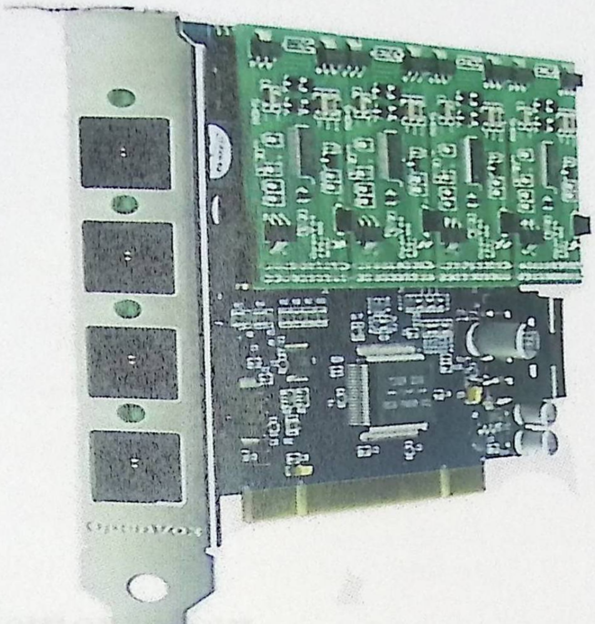


Fig 4.9 FXS module[24]

5

Chapter 5
System Implementation

5.1 Overview

In this chapter, Elastix settings and PBX programming will be mentioned and described , also the code of the android application and the TOMCAT server which is related with the database .

5.2 Installation and preparing the system

At this stage, all system parts must be ready to be tested at the server side and the mobile side .

5.2.1 Preparing Elastix Server

The first thing in the configuration is the general settings such as:

Country Indications: Israel

Security settings : Allow Anonymous Inbound SIP Calls

Online Updates : Enable check for updates and adding email update angam.salhab@gmail.com

5.2.1.2 Extensions

In Panasonic KX-TES 284 with PSTN lines we have six extensions VOIP service provided , these extensions connected to an FXO ports (A1600P card) as shown in fig(5.1). This requires to establish SIP extensions to receive VOIP calls. In case of Panasonic KX-TES 284 without PSTN lines, we had to connect it with ATA DAG 1000 which forwards the call to the real SIP extensions via virtual ones as shown fig(5.2).

1- Connection using PSTN lines :

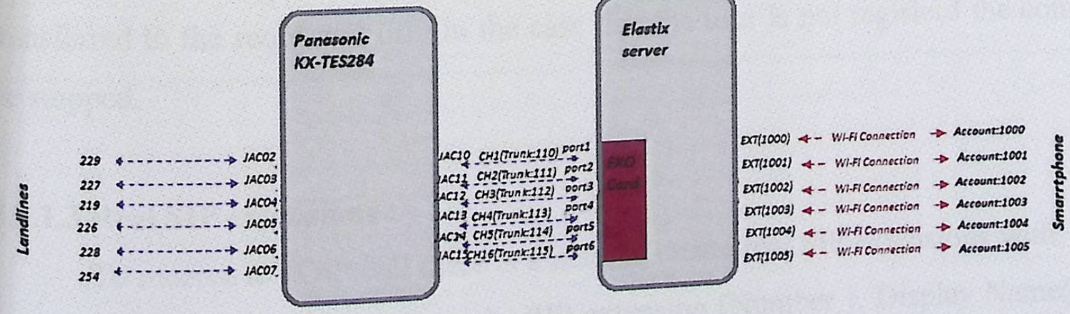


Fig 5.1 The system with PSTN lines

In the case of Panasonic KX-TES 284 when there is a missed call to extension 229 The panasonic is programmed to direct the call to a trunk such as 110, which represnets a direct connection with one of the ports inside the FXO card .

Once the extension 1000 is registerd in the ealstix server the call will be transferred to the requested user,In the case that the user is not registerd the connection will be stopped.

2- Connecttion without using PSTN line .

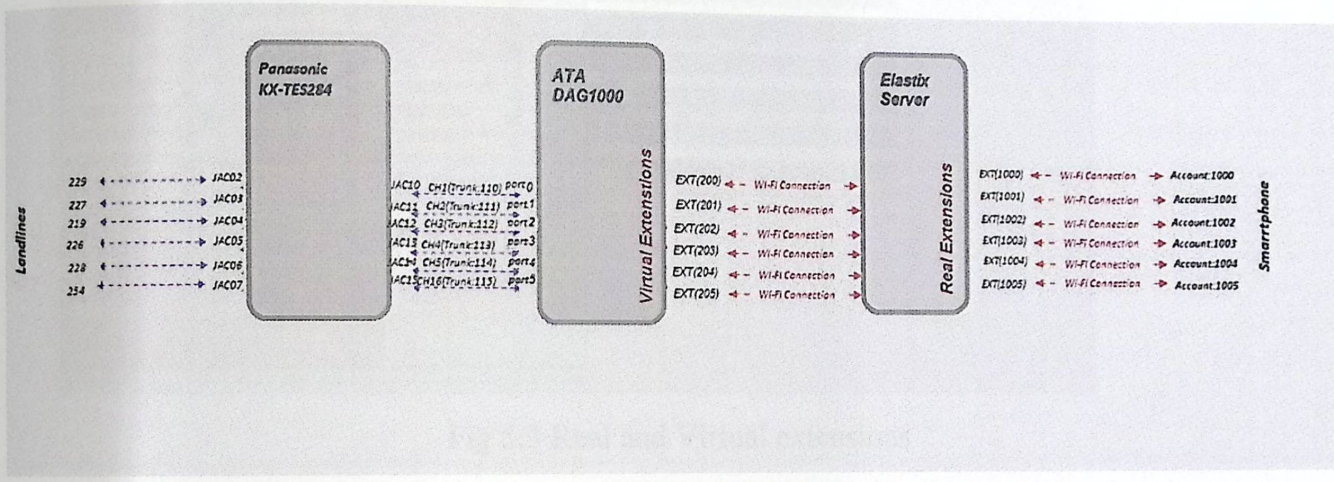


Fig 5.2 The system without PSTN lines

In the case of Panasonic KX-TES 284 when there is a missed call to extension 229 The panasonic is programmed to direct the call to a trunk such as 110 that is connected directly to ATA which is an interface between the server and the Panasonic . it has virtual extensions such as 200 which acts as an interface between the sip extension 1000 and the trunk 110. Once the extension 1000 is registerd in the ealstix server the call will be transferred to the requested user,In the case that the user is not registerd the connection will be stopped.

5.2.1.3 Real SIP extensions :

To receive a VOIP call there is a need to create real SIP extensions with user name , password and a number as followed : SIP extension (Number), Display Name(username) , CID Num Alias (SIP ext. NO),Secret (Password) , Nat(enable) , Dial (sip/ext.NO) ,Port:5060 , Status for voicemail:(enabled) and Email Address (username@ppu.edu)

We create (1000,1001,1002,1003,1004,1005,1006) Real SIP extension as shown in fig (5.3)

5.2.1.4 Virtual Extensions

In the case of Panasonic KX-TES 284 without PSTN lines there is a need to create virtual SIP extensions with user name ,password and SIP extension (Number). We create (200,201,202,203,204,205,206) Real SIP extension as shown in fig(5.3)

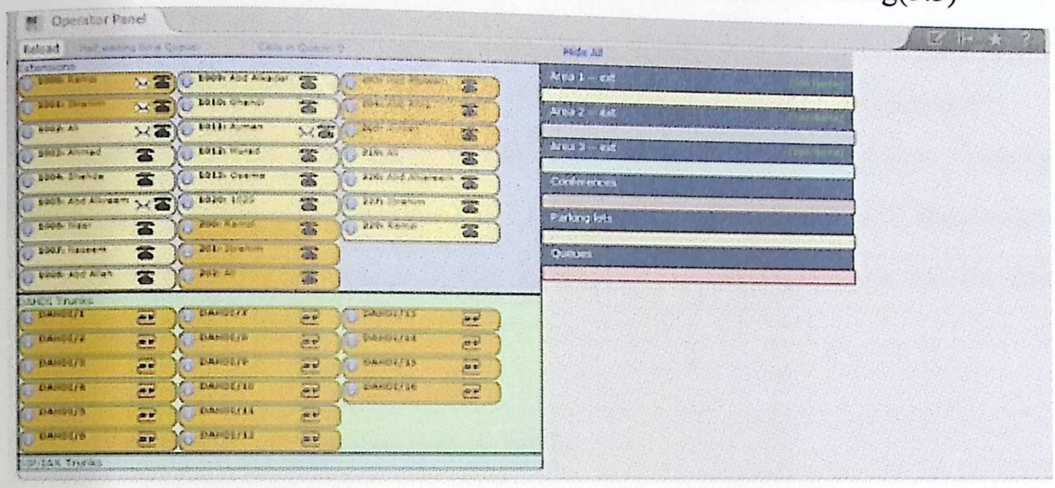


Fig 5.3 Real and Virtual extensions

5.2.1.5 ZAP channel DIDs

Once the ATCOM -1600P/16"ATCOM AX1600P/800P Board17"(MASTER) is installed the card must be detected using zapata.conf commands as shown in Fig 5.4 we note that when the cards are detected it appears in a green color .

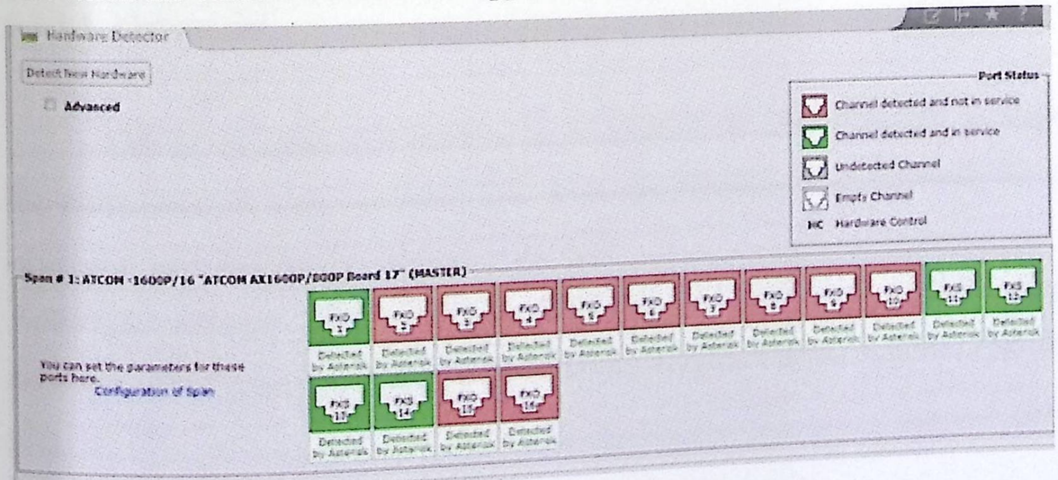


Fig5.4 Hardware Detection.

To handle calls coming in through the Zaptel channel we need to define the channel for each port that connected with the specific trunk by determining the DID of the channel through the ZAP channel DIDs as followed :

Channel : detected channel , Description : username , and channel DIDs : ZAP trunk number

5.2.1.6 Creating ZAP Trunks :

ZAP Trunks is physical connection between KX-TES 284 and FXO (1600P) cards , so that in Elastix server there is a need to create a ZAP trunk as followed :

Create ZAP trunk (DAHDI compatibility mode) with Trunk name : username ,Outbound Caller ID : trunk number CID options : Allow any CID Maximum Channels : (simultaneous calls),Dialed Number Manipulation before sending out this trunk .We set the match pattern as XXX

5.2.1.7 Inbound routes (incoming call):

In the absence of the PSTN lines in the Panasonic , where the behavior of incoming calls from all trunks is being handle When an incoming call from Panasonic is received, asterisk needs to know where to direct it. It can be directed to an extension, Digital Receptionist (IVR), for this purpose Inbound Route needs to be set up.

Description : Username , DID Number :trunk , Set destination : < SIP extension > name.

5.2.1.8 Follow me

One of the cases that needs to be resolved is the case of having no response by one of the extensions. In order to solve this issue, the Elastix server forwards the call to another extension such that the connection is not lost.

Follow me settings: Ring strategy : ringallv2 //to call the main numbers then the other numbers , Ring time :7sec// three rings , Extension quick pick : Enable , Play music on hold : ring

Follow me list: main SIP extension

Followed SIP extension

5.2.1.9 Voicemail

We created a system that enables the user to record a voice message using the landline that can be sent to the other user through his email.

This system consists of a domain (@ppu.edu), email list as detailed below.

Account Name	Used Space	Reconstruct MailBox
ahqdemat@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
ali-amro@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
armana@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
aymanw@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
daud@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
ghandi@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
ibraheem@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
murads@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
nassim_eng83@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
nizar@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
oata@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
ramzi@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
szahda@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct
zaro@ppu.edu	0 KB / 30 KB (0.00%)	Reconstruct

Fig 5.5 users emails

5.2.1.10 Smartphone to landline (FXS port)

To make a call from a smart phone to a landline directly, we created real ZAP extensions in the elastix server , which is physically connected between the FXS ports (A1600P card) and the Panasonic KX-TES 284 CO lines.

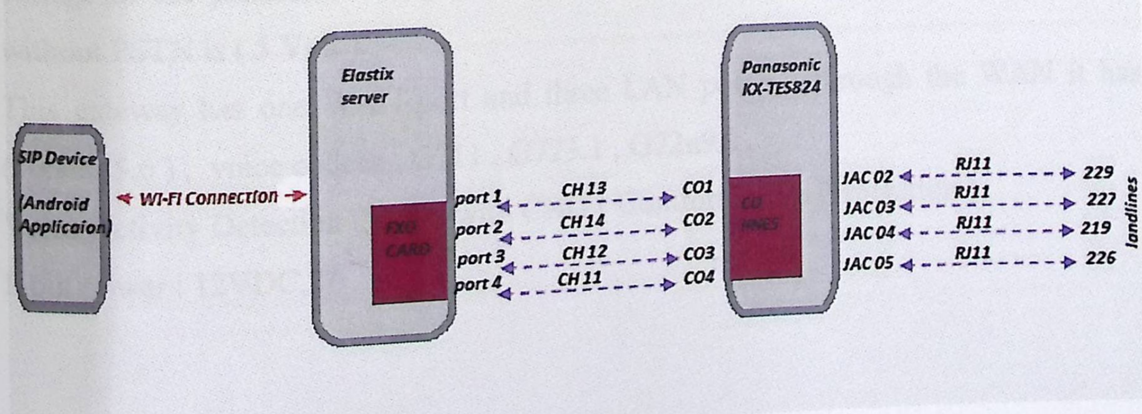


Fig 5.6 Smartphone to landline

We create the ZAP extension as followed : User Extension: landline number , Display name : username, Chanel :detected channels (11,12,13,14) .

5.2.1.11 Openfire

Openfire is an instant messaging (IM) and group chat server that is written in Javaand it is normally used as an instant transfer system through internet protocol " VoIP". It was created inside our system to allow authenticated chat between users, the users should register with a password , username and an IP for the elastix server by using spark application .

5.2.2 Panasonic KX-TES 284 Programming

5.2.2.1 PBX Programming

The PBX extensions are programmed using the digit order (006) to identify the (prefix=2) for the Office numbers and (prefix=1) for the trunks numbers , (712 + ext. no. + #) to make the Call forwarding from JACS to Trunks that connected with FXO PORTS , (*350) to enable the CO lines that connected with FXS PORTS.

5.2.2..2 DAG1000- 8FXO ATA

The need of this gateway appears because of the absence of the PSTN lines since the A1600P FXO card has input voltage (48-62 Volt) which is compatible with the line voltage of the panasonic that provided with PSTN . knowing that Panasonic line voltage without PSTN is (5 Volt).

This gateway has one WAN port and three LAN ports , through the WAN it has an IP (10.10.75.6) , voice codecs : G711 , G723.1 , G72n9A .

Voice Activity Detection (VAD) with CNG (Comfort Noise Generation) .

Input power : 12VDC,2A

How to connect the ATA with Panasonic PBX ?

The ATA consists of 8 FXO units , each PBX trunk is connected with one of the 8ports which has a virtual extension to be an interface to the real SIP extensions .
In the system parameter we identify the SIP user ID, Authantication ID , Authantication password and offhook Auto-dial : (real SIP extension) for each port .
As a result , the ports are regesterd successfully as in fig (5.7)

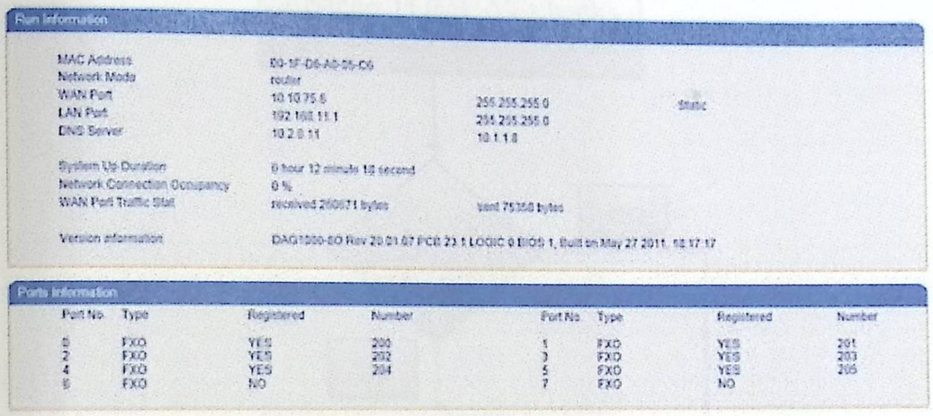


Fig (5.7) This figure shows the type , status and the number for each port .

5.2.3 Android Application

Since the security is a major requirement and objective in our project we developed an android application that's only allows for registered users with a certain username password and IP (Elastix server IP) to login and use the system.
To achieve these goals, we developed an android application by calling the VOIP software (3cx version 3.0), by Generating an interface that contains text fields to insert username and password ,which is sent to the authenticated server in order to ensure the validity of the input information by checking the database as shown in fig 5.8 .

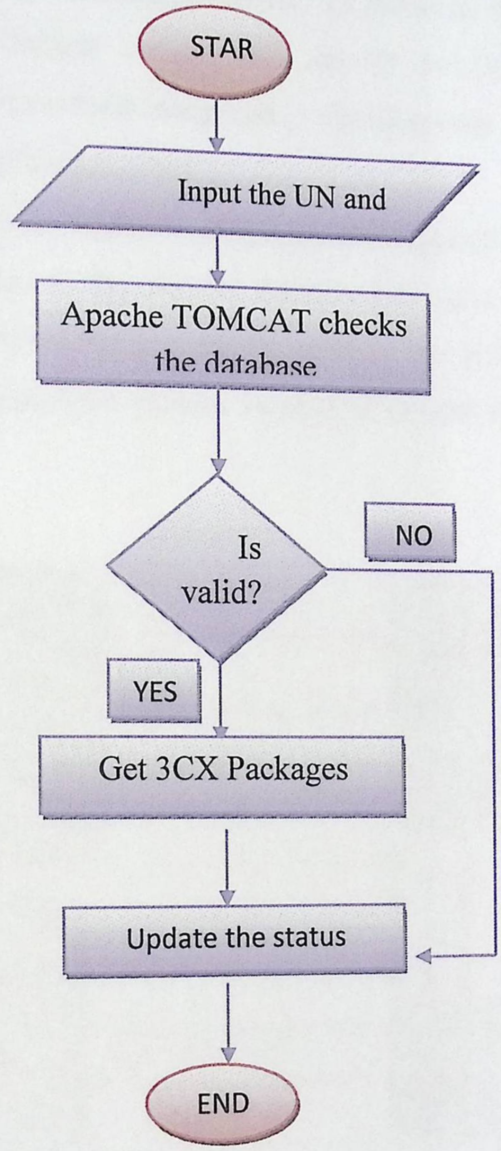


Fig 5.8 Authentication Process

5.2.3.1 Servlet

A servlet is a Java programming language class used to extend the capabilities of a server. Although servlets can respond to any types of requests, they are commonly used to extend the applications hosted by web servers, so they can be thought of as Java Applets that run on servers instead of in web browsers.

The servlet Provides dynamic content such as the results of a database query .The system contains a database with username and password . To make the database available , it will be related to the Apache Tomcat server with an IP (10.10.75.42) that is called by NetBeanswhich is an open-source integrated development environment ,supports development of all Java application types .

In our system the user registers in the android application with a username and a password, that acts as input to the Apache tomcat web server, which in turn checks the database and make the verification process by sending the 3CX Packages in case of valid input or terminates the registration process in case of invalid input as shown below in Fig 5.8.

.

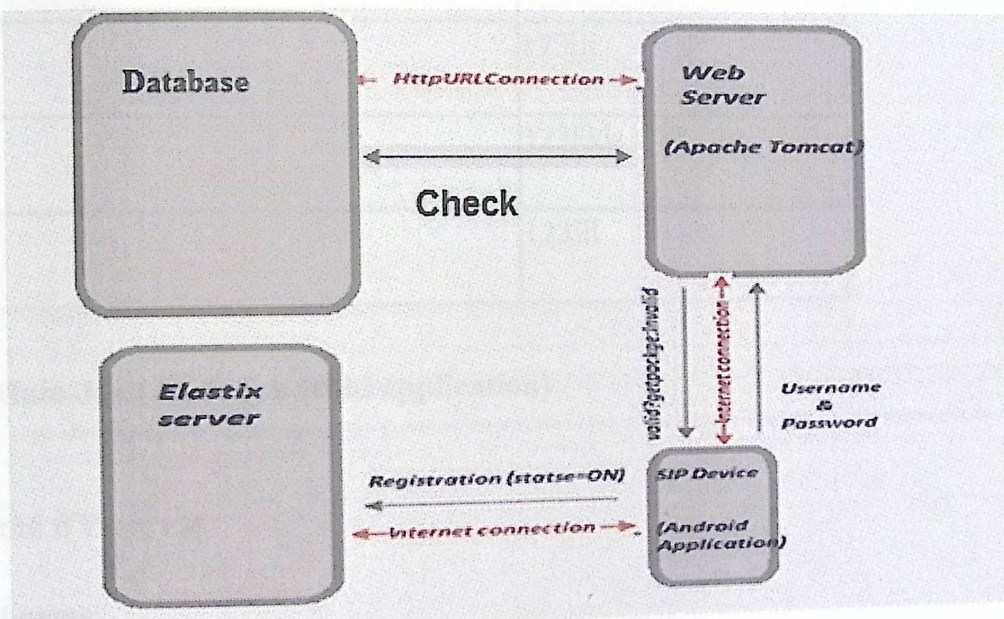


Fig 5.9 The Verification Process

5.2.3.2 Database

Table 5.1: The database

Un	Pwd
Ramzi	123aaa
Ibraheem	123bbb
Ali	123ccc

Un	Pwd
AbdAlkareem	123ddd
Ayman	123eee
Murad	123fff
Abd Allah	123ggg
Osama	123hhh
AbdAlkader	123iii
Ghandi	123jjj
Shehda	123kkk
Ahmad	123lll

5.2.3.4 Main Java Code (android application)

```
public void f(View v){

EditText name;

name = (EditText) findViewById(R.id.name);

string s = name.getText().toString();

EditText password;

password = (EditText) findViewById(R.id.password);

String  pwd = password.getText().toString();

try{

URL url = new URL("http://10.10.75.42
:8092/Authantication/check?un="+s+"&pwd="+pwd);
```

```
HttpURLConnection con = (HttpURLConnection)
;)url.openConnection(

String str = readValue(con.getInputStream)();

if(str.equals("Invalid));

Toast.makeText(this,str,
Toast.LENGTH_SHORT).show();

else{final PackageManager pm =
getPackageManager();

Intent i;

i=pm.getLaunchIntentForPackage
("com.tcx.sipphone");

if (i != null)

startActivity(i);

else

Toast.makeText(this,"3CX is not Installed",Toast.LENGTH_LONG).show();

}

}

catch(Exception n){

Toast.makeText(this,n.getMessage(), Toast.LENGTH_SHORT).show();}

public String readValue(InputStream in){

BufferedReaderbr=null;

Str ="";

try{

br =new BufferedReader(newBufferedreaderinputstreamreader(in));
```

```
while((line = br.readLine())!=null}(str = str  + line;}
returnstr;}
catch(Exception e){return e.getMessage();}
finally{
if (br!=null)
try{
br.close();
} catch (IOExceptione ){
//TODO Auto-generatedcatch block
e.printStackTrace();
```

2.3.5 Netbeans Java Code

```
public class check extendsHttpServlet {
protected voiddoGet(HttpServletRequest request, HttpServletResponse response)
throwsServletException, IOException {
booleanisValid=false;
try {
    String username,password;
username=request.getParameter("un");
password=request.getParameter("pwd");
Class.forName("sun.jdbc.odbc.JdbcOdbcDriver");
    String db="mydb";
    String dbURL="jdbc:odbc:"+db;
```

```
        String result="";
        Connection con =DriverManager.getConnection(dbURL,"","");
        Statement s=(Statement)con.createStatement();

s.execute("SELECT pwd FROM T1 where un = '"+username+"'");
ResultSetrs =s.getResultSet();
if(rs.next()){
            String dbpassword=rs.getString("pwd");
if(password.equals(dbpassword)){
isValid=true ;
            }
else {
isValid=false;

            }
                }
else {
isValid=false;
}
if(isvalid){
response.getwriter.write("valid");
}
else
```

```
response.getwriter.write("invalid");

}

catch (Exception ex) {

ex.printStackTrace();

else

response.getwriter.write("invalid");

catch (Exception ex) {

ex.printStackTrace();

response.getWriter().write("Error:"+ex.getMessage());

    }

  }}
```

5.2.3.6 Mobile application

An application has already been built using the NetBeansenvironment , to use this application it must be installed to the mobile and to perform this we do the following :
The first is to connect the mobile phone (Galaxy S) by the USB cable or Bluetooth connection to the PC at which the workspace of the NetBeans is created. Then the following file must be copied to the mobile SD card .
C:\workspace\VoIPsystem\bin\auth.apk
Opening the ".apk" file , a message will appear at the screen of the mobile as shown in fig (5.10) , shows the name of the application and the permutations needed for it to work probably.

In our application we need to access the username and password to get the 3CX application , and we need a full internet access for the communication between the mobile (client) and PC (server) while posting data.
Clicking the install button to allow the installation of the application as shown at figure (5.11)

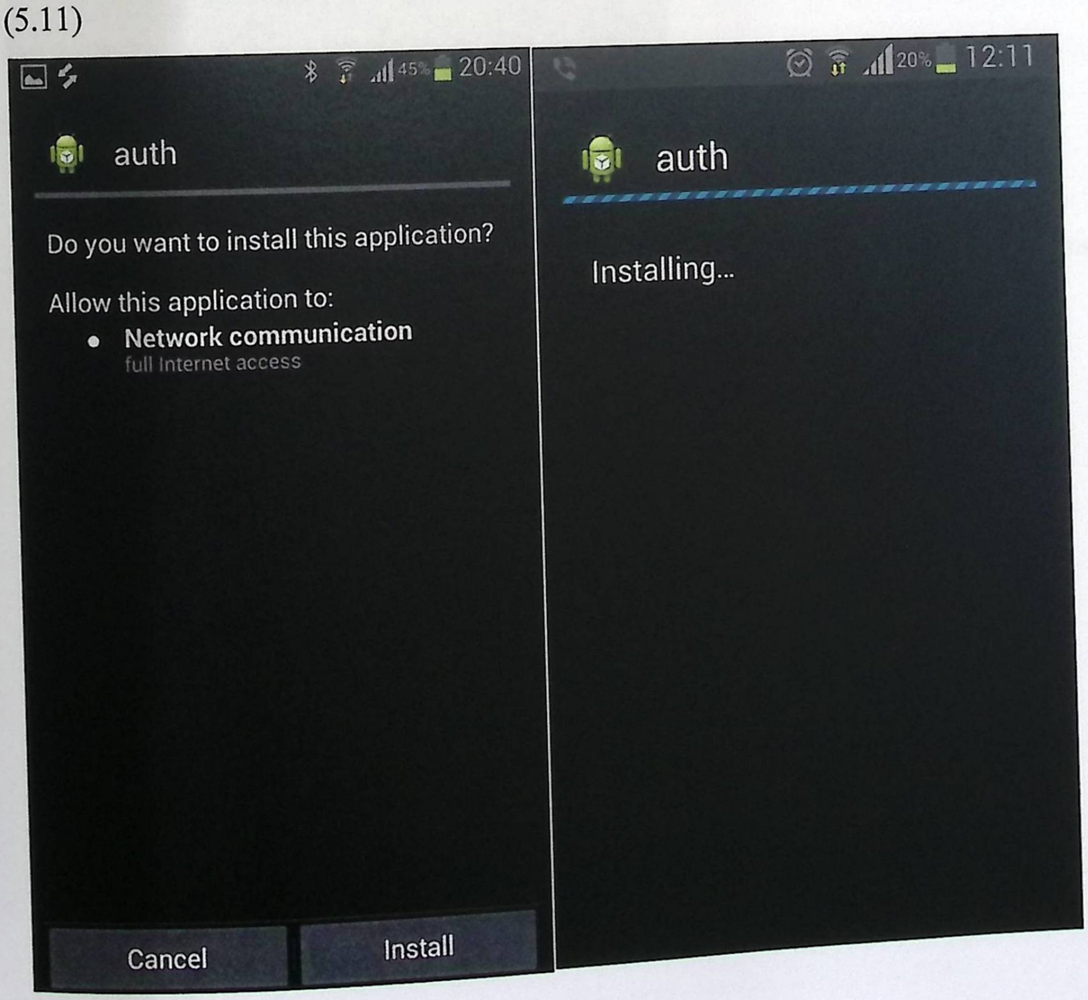


Fig 5.10 Opening .apk file Fig 5.11 Installing application

When the installation is completed as shown in the figure 5.12 , click the button to start the application.

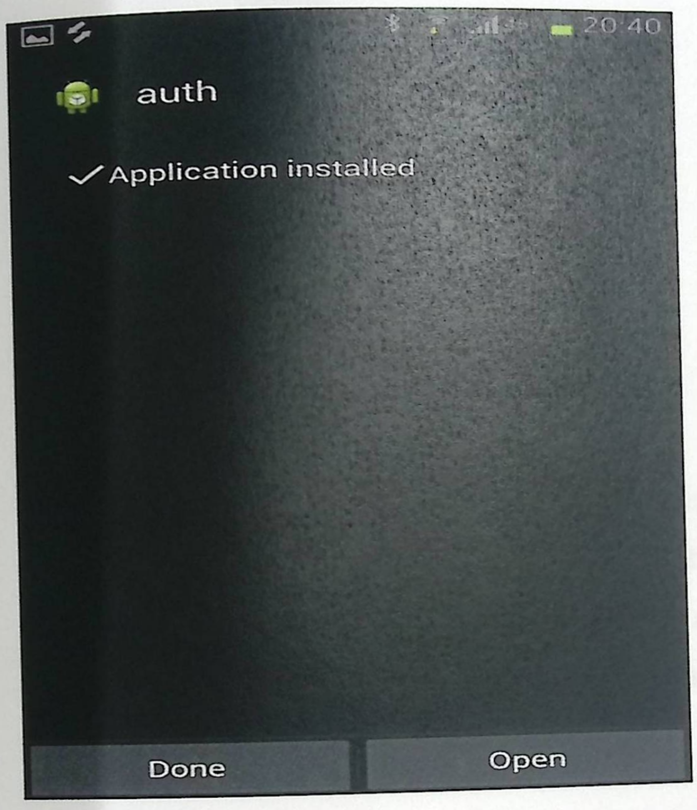


Fig 5.12 Installation is completed

Now, the application on the mobile is ready to be used by the user. The user must enter its information to send it to the Apache TOMCAT server .

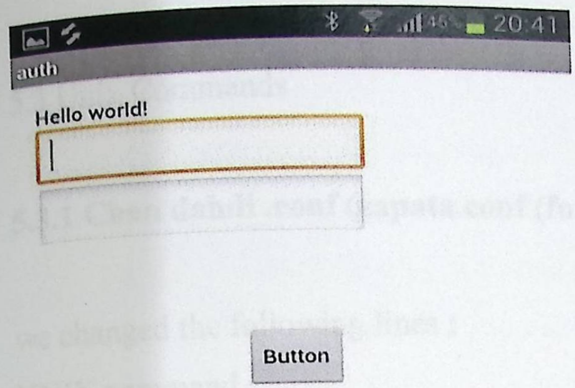
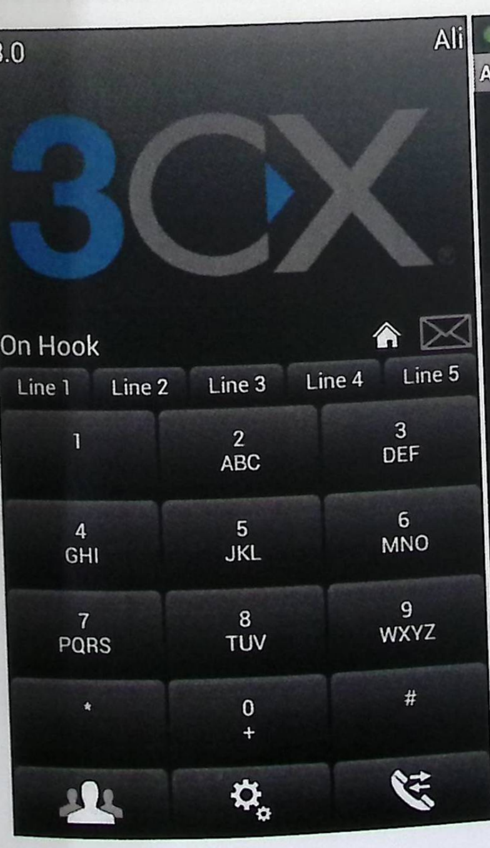


Fig 5.13 Registration

After the registration is completed with a valid username and password , the application will call 3CX packages and open it . as shown in fig 5.14 , then we must set the accounts with the username , password , extension number , the external IP and internal IP as shown in fig 5.15.



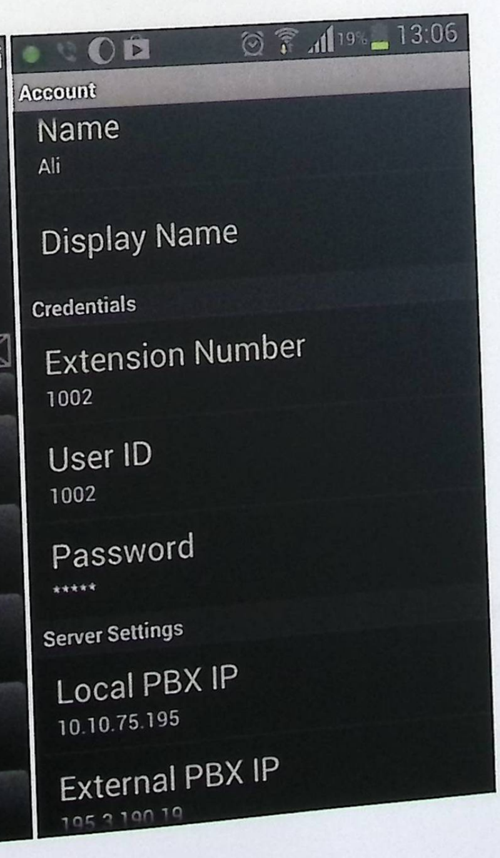


Fig 5.14 3CX application

Fig 5.15 Account settings

5.3 Unix Commands

5.3.1 Chan dahdi .conf (zapata.conf (for ZAPTEL users only)

we changed the following lines :

UNIX command

context=from-pstn>> context = from-zaptel ;

busydetect=yes;

busycount=3;   // This is to ensure that the line is hung up if there is no answer or busy after 3 busy tones)

echotraining = yes ;

echotraining = 800 ;   //We added it to ensure that zap trunk working properly.

5.3.2 SIP_NAT.CONF

The only other .conf file that requires attention is the sip_nat.conf .

Our system by default allows internal users to register inside the local network using private IP 10.10.75.195. However in order to improve the system more, we worked on allowing users from outside the network to be able to connect to the network using public IP 195.3.190.19.

We inserted commands in sip_nat.conf as shown next.

UNIX commands

sip_nat.conf

nat=yes

externip=<195.3.190.19> or

localnet=10.10.75.195/255.255.255.0

externrefresh=10

5.3.3 Openfire Unix Command

In order to activate IM, we need to add asterisk files to the openfire such as Asterisk –IM openfire pluggin and Kraken IM Gateway .

UNIX commands

```
root@ ~10 ] #cd/opt
root@ ~10  opt] #cd openfire
root@ ~10 openfre] #cd plugin
root@ ~10 plugins] #cd astersik-im
root@ ~10 astersik-im ] #cd database
root@ ~10 database ] #cd nano asterisk-imhsqldb.sql
```

5.3.4 ZAP Hardware Detection Command

Once the ATCOM -1600P/16"ATCOM AX1600P/800P Board17"(MASTER) is installed the card must be detected using the following lines at the end of zapata.conf :

```
#include zapata-channels.conf
#include zapata_additional.conf
```

6

Chapter 6

Results and conclusion

6.1 Overview.

6.2 Elastix Result .

6.3 Roaming .

6.4 Conclusion .

6.5 Challenges .

6.6 Future Work Recommendation .

6.1 Overview

In this chapter we included the results when making calls and we will mention what we achieved in this project and the conclustion for all things that we have done,also we will talk about the challenges that we faced and ending with recommendation needed for the future work .

6.2 Elastix Results

6.2.1Call Detail Reports (CDR) :

Reports are essential in the operation of a PBX .Including the record of all incoming and outgoing calls , the channels used , the status and duration of calls as shown in the following table (6.1)

Table (6.1) Call Detail Reports

Date	Source	Ring Group	Destination	Src. Channel	Account Code	Dst. Channel	Status
2013-05-21 16:08:56	201		0598017477	Local/0598017477@from-internal-0000000e;2		DAHDI/2-1	ANSWERED
2013-05-21 16:08:49	201		1001	SIP/201-000002be		Local/0598017477@from-internal-0000000e;1	ANSWERED
2013-05-21 16:08:28	1002		227	SIP/1002-000002bd		DAHDI/14-1	ANSWERED
2013-05-21 16:08:18	201		0598017477	Local/0598017477@from-internal-0000000d;2		DAHDI/2-1	ANSWERED
2013-05-21 16:08:11	201		1001	SIP/201-000002bb		Local/0598017477@from-internal-0000000d;1	ANSWERED
2013-05-21 16:07:50	1002		227	SIP/1002-000002ba		DAHDI/14-1	ANSWERED
2013-05-21 16:07:47	1002		3227	SIP/1002-000002b9		DAHDI/2-1	NO ANSWEF
2013-05-21 16:07:09	1001		219	SIP/1001-000002b8		DAHDI/11-1	NO ANSWEF
2013-05-21 16:06:49	1001		219	SIP/1001-000002b7		DAHDI/11-1	NO ANSWEF
2013-05-21 16:06:41	1001		2219	SIP/1001-000002b6		DAHDI/2-1	NO ANSWEF
2013-05-21 16:05:46	1002		219	SIP/1002-000002b5		DAHDI/11-1	NO ANSWEF
2013-05-21 16:05:30	1001		219	SIP/1001-000002b4		DAHDI/11-1	NO ANSWEF
2013-05-21 16:05:01	1001		219	SIP/1001-000002b3		DAHDI/11-1	NO ANSWEF
2013-05-21 16:04:42	1001		219	SIP/1001-000002b2		DAHDI/11-1	NO ANSWEF
2013-05-21 16:03:38	1002		0598017477	Local/0598017477@from-internal-0000000c;2		DAHDI/2-1	ANSWERED
2013-05-21 16:03:31	1002		1001	SIP/1002-000002b0		Local/0598017477@from-internal-0000000c;1	ANSWERED
2013-05-21 16:02:55	1002		0598017477	Local/0598017477@from-internal-0000000b;2		DAHDI/2-1	ANSWERED
2013-05-21 16:02:48	1002		1001	SIP/1002-000002ae		Local/0598017477@from-internal-0000000b;1	ANSWERED
2013-05-21 16:01:42	1002		1001	SIP/1002-000002ac		SIP/1001-000002ad	ANSWERED
2013-05-21 16:01:05	1002		0598017477	Local/0598017477@from-internal-0000000a;2		DAHDI/2-1	ANSWERED
2013-05-21 16:00:58	1002		1001	SIP/1002-000002aa		Local/0598017477@from-internal-0000000a;1	ANSWERED
2013-05-21 15:59:49	1002		0598017477	Local/0598017477@from-internal-00000009;2		DAHDI/2-1	ANSWERED
2013-05-21 15:59:42	1002		1001	SIP/1002-000002a8		Local/0598017477@from-internal-00000009;1	ANSWERED
2013-05-21 15:59:28	1002		1001	SIP/1002-000002a7		Local/0598017477@from-internal-00000008;1	NO ANSWEF
2013-05-21 15:59:28	1002		0598017477	Local/0598017477@from-internal-00000008;2		DAHDI/2-1	NO ANSWEF
2013-05-21 15:58:53	1002		vmu1000	SIP/1002-000002a6		Local/FMGL-1001@from-internal-00000007;1	ANSWERED
2013-05-21 15:56:16	202		1002	SIP/202-000002a4		Local/FMPR-1002@from-internal-00000004;1	ANSWERED

6.2.2 Operator Panel

It displays information about the PBX activity in real time .The following figure fig (6.1)showsreal time incoming call transferred from the landline (229) to the SIP extension(1002) via the virtual extension(200). we note that the user who is registers in the Elastix server appears in an orange color and when this user doesn't answer the call he will receive a voice mail and appear as message as shown in the figure below

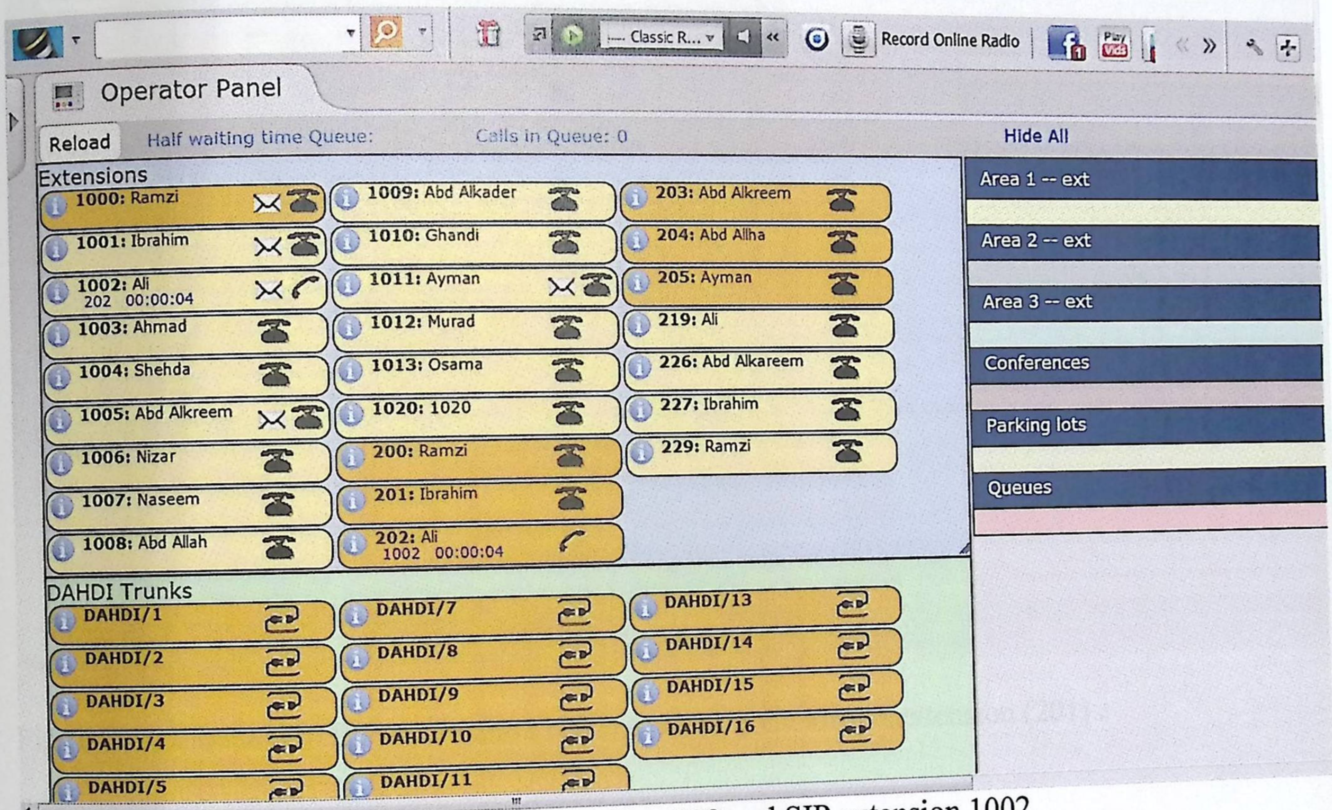


Fig (6.1).call between landline 229 and SIP extension 1002

The following figure fig (6.2) shows real time call transferred from the SIP extension (1002) to (1001) via virtual extension (201) .from the figure Ramzi,Ibrahim,& Ali are registers in the Elastix server

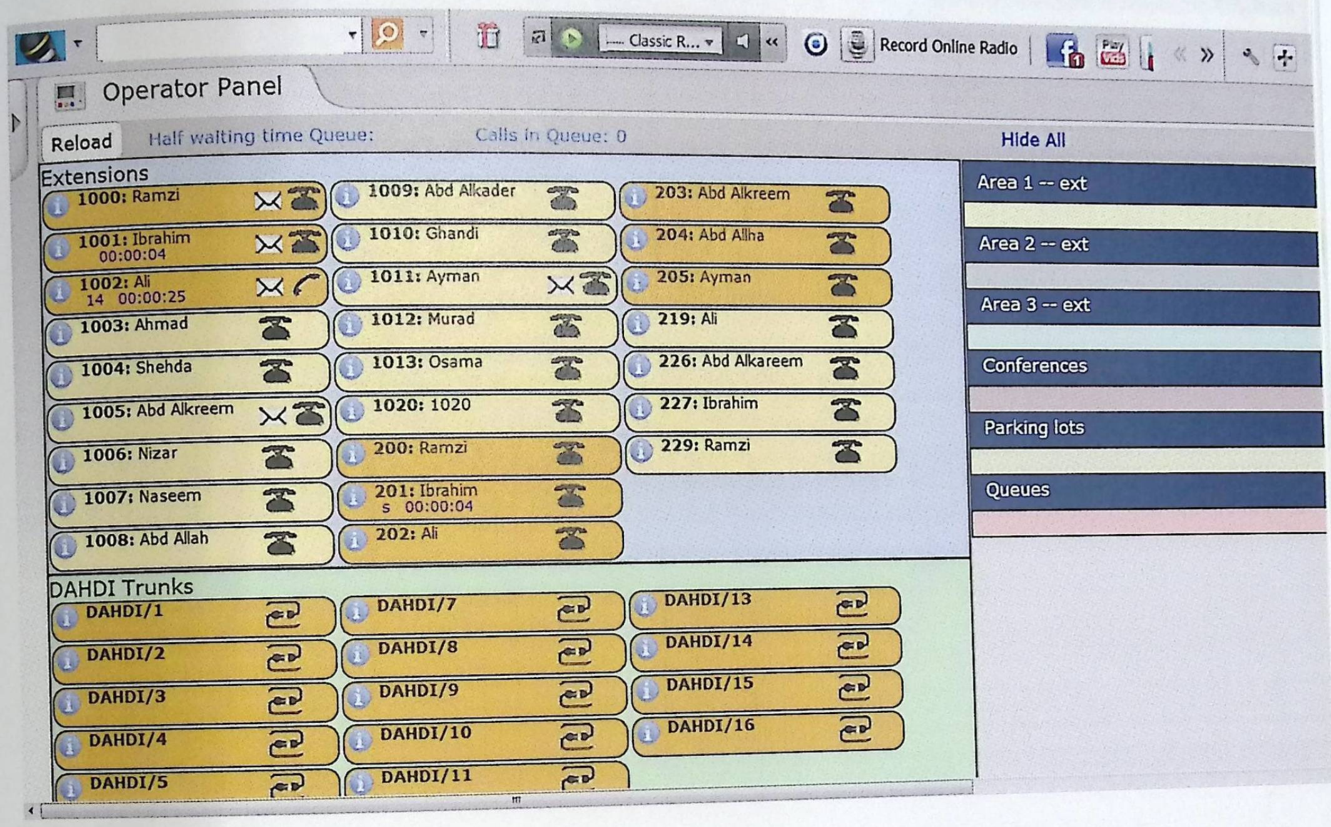
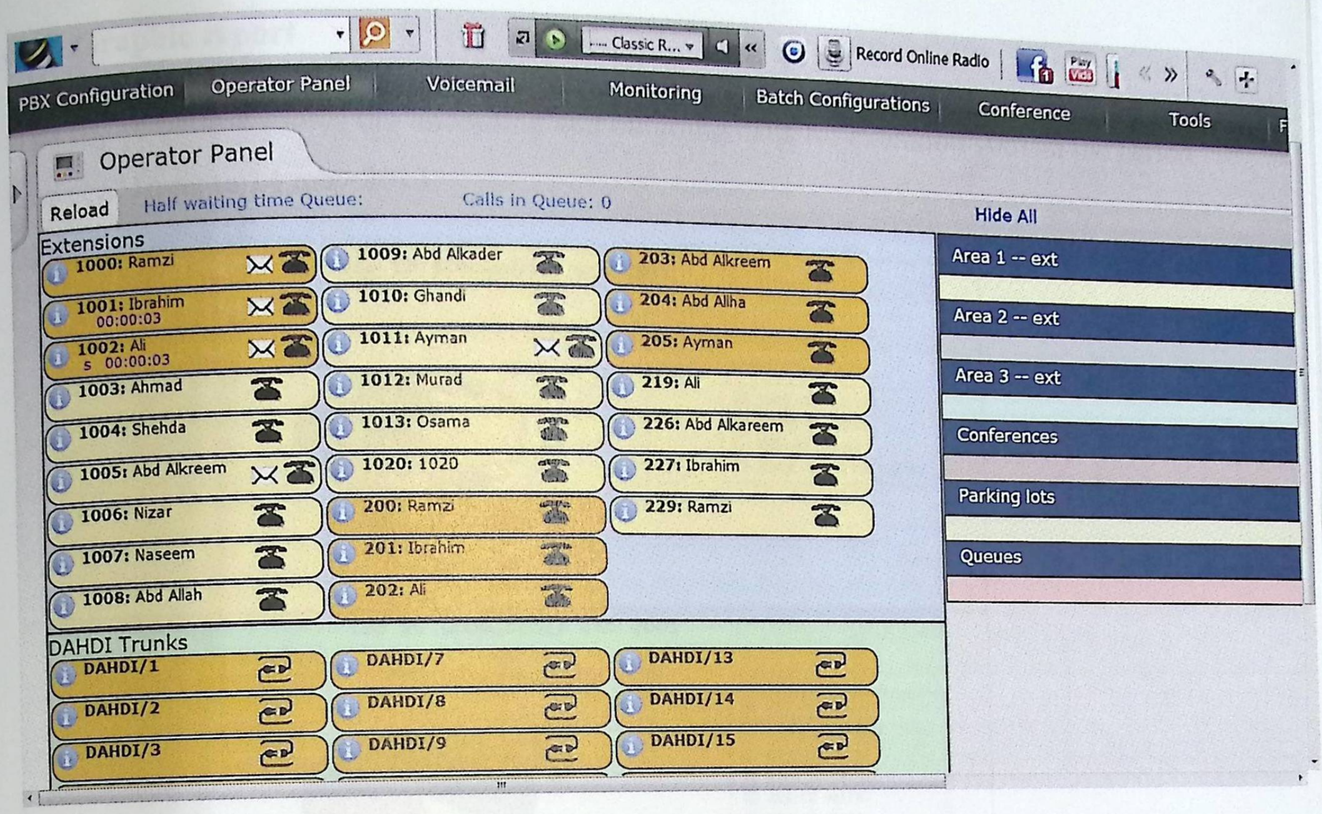


Fig (6.2) call between SIP extension (1002) to (1001) via virtual extension (201) .

The following figure fig (6.3) shows real time call transferred from the SIP extention 1001 to the SIP extension 1002.



Fig(6.3) call between SIP extensions 1001 and 1002

6.2.3 Missed calls

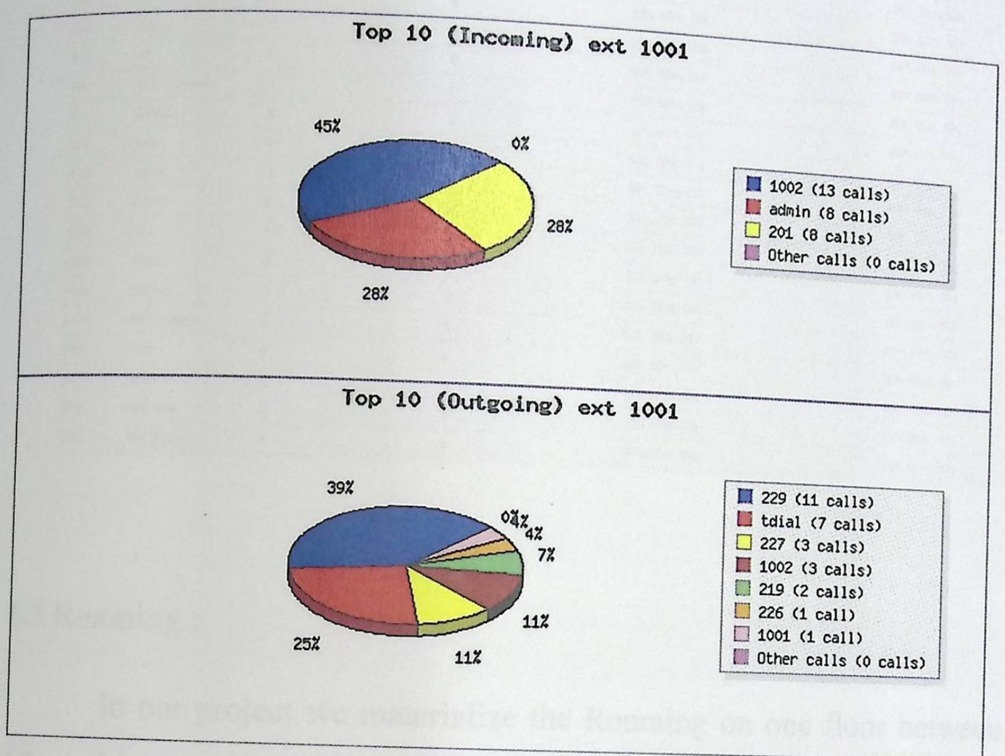
It shows the number of missed calls when the source tries to call the destination because of no answer or failed .

Table (6.2) missed calls

Date	Source	Destination	Time since last call	Number of attempts	Status
13-May-2013 14:21:54	202	FMPR-1002	21 hour(s) 19 minute(s) 12 second(s)	3	NO ANSWER
13-May-2013 14:14:49	201	tdial	21 hour(s) 26 minute(s) 16 second(s)	11	FAILED
13-May-2013 13:58:52	admin	tdial	21 hour(s) 42 minute(s) 14 second(s)	3	FAILED
13-May-2013 13:58:12	1001	FMPR-1002	21 hour(s) 42 minute(s) 53 second(s)	5	NO ANSWER
13-May-2013 13:49:06	admin	FMPR-1002	21 hour(s) 52 minute(s) 0 second(s)	1	NO ANSWER
13-May-2013 13:22:28	1001	1001	22 hour(s) 18 minute(s) 38 second(s)	1	NO ANSWER
13-May-2013 13:19:59	1001	tdial	22 hour(s) 21 minute(s) 7 second(s)	7	FAILED
13-May-2013 13:18:18	1002	tdial	22 hour(s) 22 minute(s) 47 second(s)	9	FAILED
12-May-2013 13:01:31	202	tdial	1 day(s) 22 hour(s) 39 minute(s) 34 second(s)	11	NO ANSWER
09-May-2013 12:30:59	1001	226	4 day(s) 23 hour(s) 10 minute(s) 7 second(s)	1	NO ANSWER
09-May-2013 12:30:44	1001	227	4 day(s) 23 hour(s) 10 minute(s) 21 second(s)	1	NO ANSWER
07-May-2013 13:04:02	2000000	s	6 day(s) 22 hour(s) 37 minute(s) 4 second(s)	1	NO ANSWER AND VOICEMAIL

6.2.4 Graphic report

The figure bellow shows the incoming and outgoing calls for extention 1001 in the period between 1may to 14 may 2013.



Fig(6.4) incoming and outgoing calls for extension 1001

6.2.5 Summary

It shows the number of incoming and outgoing calls and the total time for each call.

Table (6.3) summary calls

Ext.	User	# Incoming Calls	# Outgoing Calls	Total time (Incoming Calls)	Total time (Outgoing Calls)	Details
200	Ramzi	0	1	00h. 00m. 00s	00h. 00m. 05s	View
201	Ibrahim	0	29	00h. 00m. 00s	00h. 06m. 17s	View
202	Ali	0	14	00h. 00m. 00s	00h. 01m. 15s	View
203	Abd Alkreem	0	0	00h. 00m. 00s	00h. 00m. 00s	View
204	Abd Allha	0	0	00h. 00m. 00s	00h. 00m. 00s	View
205	Ayman	0	0	00h. 00m. 00s	00h. 00m. 00s	View
219	Ali	0	0	00h. 00m. 00s	00h. 00m. 00s	View
226	Abd Alkareem	0	0	00h. 00m. 00s	00h. 00m. 00s	View
227	Ibrahim	0	0	00h. 00m. 00s	00h. 00m. 00s	View
229	Ramzi	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1000	Ramzi	1	0	00h. 00m. 05s	00h. 00m. 00s	View
1001	Ibrahim	29	28	00h. 15m. 07s	00h. 03m. 45s	View
1002	Ali	13	27	00h. 00m. 01s	00h. 10m. 40s	View
1003	Ahmad	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1004	Shehda	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1005	Abd Alkreem	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1006	Nizar	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1007	Naseem	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1008	Abd Allah	0	0	00h. 00m. 00s	00h. 00m. 00s	View
1009	Abd Alkader	0	0	00h. 00m. 00s	00h. 00m. 00s	View

6.3 Roaming :

In our project we materialize the Roaming on one floor between two access points Cisco (Linksys E3000) , the first one is located in B building at 5th floor in the advanced networks lab ,the another one is also located in the 5th floor but in the control lab .

The two wireless networks have the same SSID : final project on different channels noting that the DHCP was disabled.

We make sure about the roaming by making a VoIP call using two smart phones through Elastix server and secured 3cx application, the call continues through the two access points which indicated that the rooming was achieved successfully at the same floor , and there are sufficient for good coverage in one floor of B building.

By measuring the Mean Opinion Score (MOS) which provides a numerical indication of the perceived quality from a voice codec during and after transmission and compression of the voice packets such that it has a numerical range from 1 to 5 , generally the MOS value should be higher than 3.7, in our case we get something around 4.4 as can be referred to fig (6.5) , which means good quality , knowing that the Factors that can affect the MOS performance are packet loss, Jitter, and end to end delay.

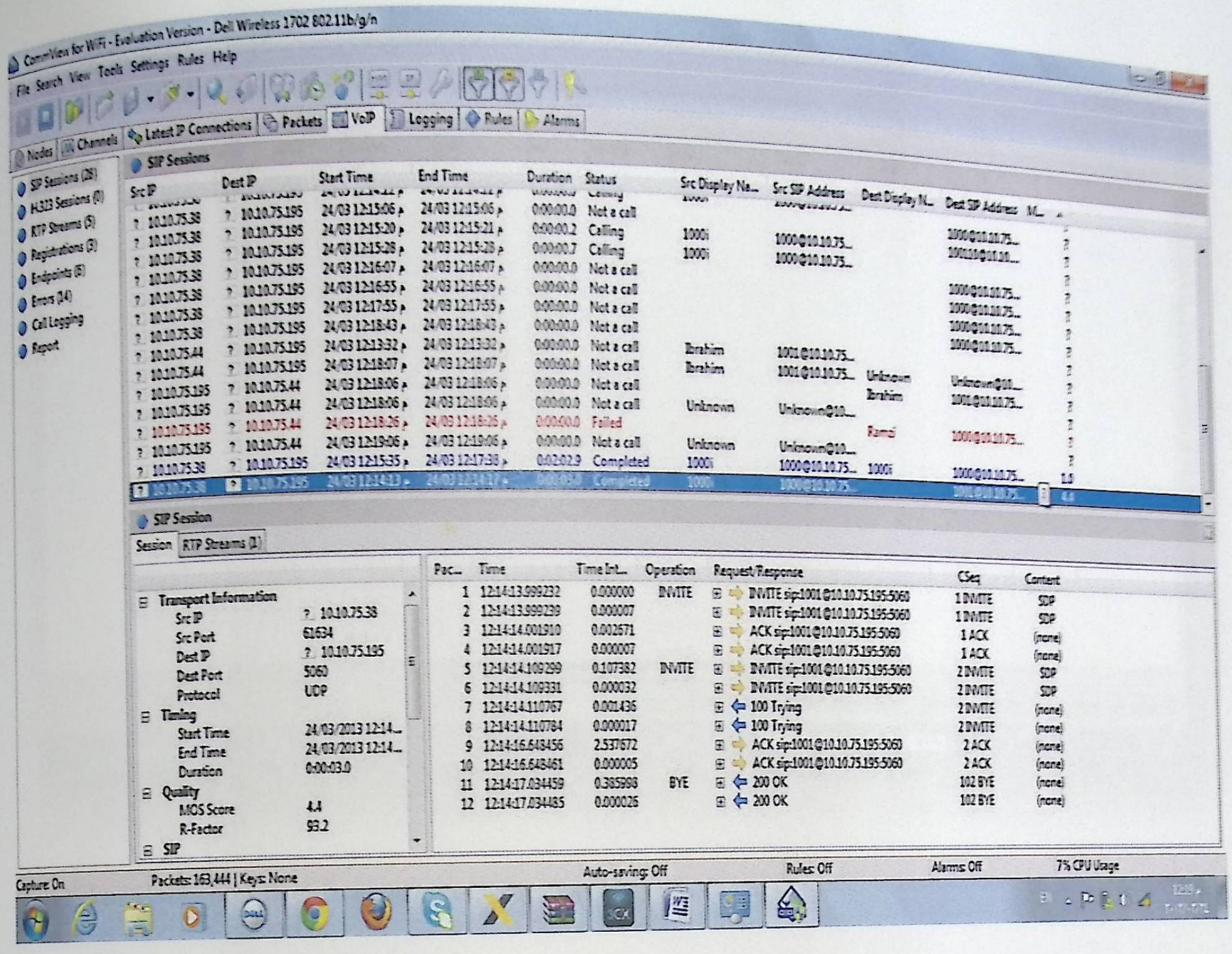


Fig (6.5) Value of MOS

By tracing the RTP streams and measuring the delay ,jitter and lost packets for a landline to Smartphone call as shown in Fig 6.7 also for a smart phone to Smartphone call as shown in Fig 6.8 ,we have a good performance for the VoIP calls since the values of the measurements are acceptable.

By measuring the propagation delay which is the amount of time it takes for the head of the signal to travel from the sender to the receiver , in our case we get it around 10 second which is acceptable to have a latency about 3 second .

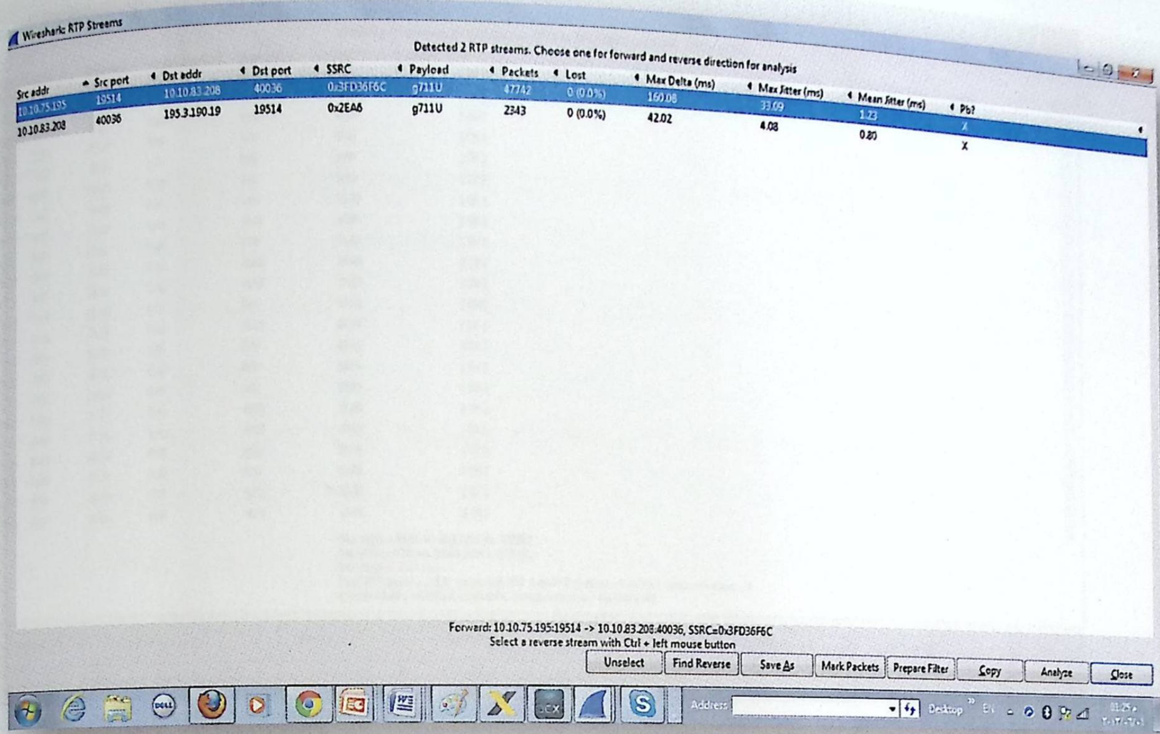


Fig (6.6) RTP stream

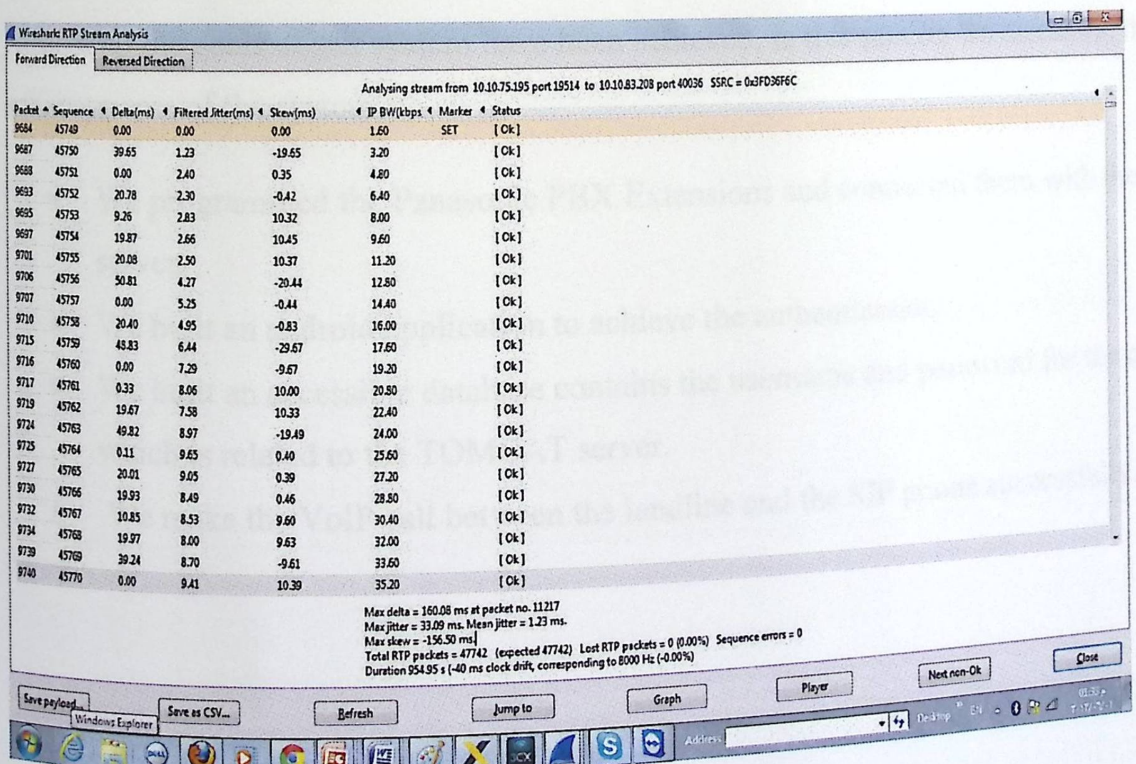


Fig (6.7) RTP stream analysis (landline to Smartphone call)

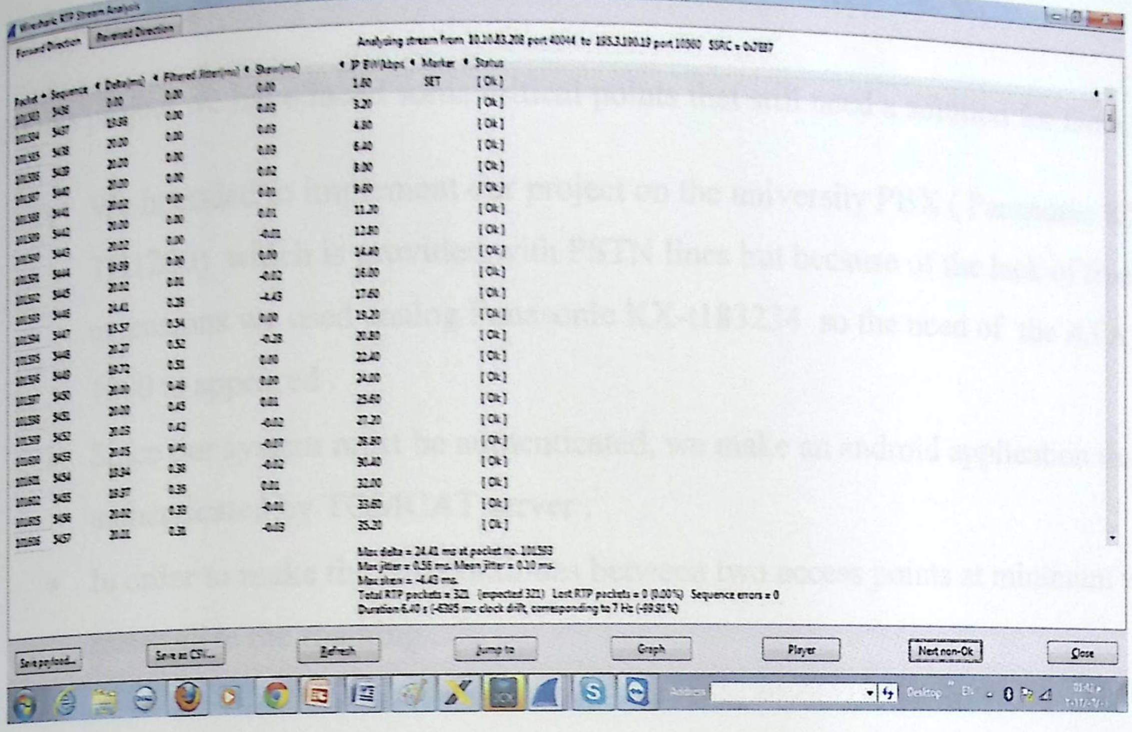


Fig (6.8) RTP stream analysis(Smartphone to Smartphone call)

6.4 Conclusion and achievement

Almost all the goals of our system have been achieved, in this section we discussed the main achievements of the system.

- We programmed the Panasonic PBX Extensions and connected them with the Elastix server.
- We built an android application to achieve the authentication.
- We built an accessible database contains the username and password for the callers which is related to the TOMCAT server.
- We make the VoIP call between the landline and the SIP phone successfully.

6.5 Challenges

At this project we have faced some critical points that still need a solution for them such as:

- We intended to implement our project on the university PBX (Panasonic KX-TD1200) which is provided with PSTN lines but because of the lack of free PSTN extensions we used analog Panasonic KX-t183234 so the need of the ATA DAG 1000 is appeared .
- Since our system must be authenticated, we make an android application that is authenticated by TOMCAT server .
- In order to make the call continues between two access points at minimum we materialize the roaming .

6.6 Future work recommendation

In this project ,there are some ideas that could be done or added to improve our system such as :

- We aspire for a freely closed socially campus that serves all of the students and university staff using authenticated VoIP .
- We hope to improve the authenticated application that provides free messages.