Palestine Polytechnic University

College of Engineering and Technology

College of Information Technology and Computer Engineering

Bachelor Thesis

Graduation Project

Intercom System Based on VoIP and GSM

(24 hour at home)

Project Team

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2013

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.... Makes life easier

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Acknowledgment

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Abstract

The computers nowadays is available in every property as well as the internet network, we attempt to turn the internet into a new level rather than browsing and e-mailing. We will use the availability of the internet to reach the property owner everywhere after he leaves his property (home for example).

The absence of the property owner is sometimes annoying to some visitors in difficult and emergency circumstances, so we will employ the internet to reach him everywhere using the intercom located at the front door.

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We will build a server that manages the VoIP call and the GSM call (depends on the availability of any of the two technologies) and provide extra features just like monitoring the property or control the gates from outside.

الملخص

اصبح البيت الفلسطيني في الآونة الاخيرة بيتاً عصرياً و حديثاً يحتوي على التكنولوجيا المختلفة التي تفيد في جميع مجالات الحياة، واصبح الحاسوب يعتبر من اساسيات الحياة و لا يوجد تقريباً أي بيت يخلو من جهاز الحاسوب. علاوة على ذلك اصبحت شبكة الانترنت متوفرة بشكل كبير في البيوت و ستمسي جزءاً أساسيا من البيت، بالتالي سوف نستخدم هذه الشبكة المتوفرة و ننقلها الى مستوى آخر – عدا التصفح و تبادل البريد الالكتروني - نوظفه من اجل تسهيل الاتصال و التواصل بين الاشخاص.

إن غياب اصحاب البيوت و الشأن عن اماكن تواجدهم يعتبر مزعجا نوعا ما الى بعض الزوار خصوصا في الحالات الطارئة. تقوم فكرة المشروع على توظيف شبكة الانترنت المتوفرة في البيوت من اجل الوصول الى صاحب البيت او الملكية عن طريق جهاز " الانتركم " أثناء خروجه من البيت عند الحاجة الماسة الى التواصل معه.

سيتم ربط المنزل بالجهاز المحمول الخاص بصاحب البيت عبر شبكة الانترنت و شبكة GSM المتوفرة و تحويل الاتصال من جهاز " الانتركم " على مدخل البيت الى الجهاز المحمول و ذلك عند خروجه من المنزل. بالتالي نكون قد انشئنا اتصالا بين الزائر و صاحب البيت اينما كان.

سنقوم ببناء خادم يدعم الاتصال عن طريق الانترنت و تنظيم عملية الاتصال إما عن طريق الانترنت أو عن طريق شبكة GSM ، بالإضافة إلى توفير خدمات اخرى حسب احتياجات صاحب الشأن مثلا مراقبة المنزل و الملكية عن طريق الكاميرا الموجودة هناك أو التحكم بالبوابات عن طريق الجهاز المحمول .

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Abstract

The computers nowadays is available in every property as well as the internet network, we attempt to turn the internet into a new level rather than browsing and e-mailing. We will use the availability of the internet to reach the property owner everywhere after he leaves his property (home for example).

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Dedication

To those who nobly give their lives to our beloved Palestine

….

To those who paid their freedom for Palestine

…

To our teachers, our constant source of inspiration

Our fathers

…

To the source of love and happiness

Our lovely mothers

…

To the flowers of the earth and the stars of the sky

Our sisters and brothers

…

To our friends, beloved and teachers

To everyone who helped us in this project

…

To you we dedicate this project

Table of Content

Table of Figures

List Of Table **Page**

1

Chapter One

Introduction

1.1 Overview

1.2 General Idea about the project

1.3 Project Objectives

1.4 Motivations

1.5 Requirements

1.6 Literature Review (Related Projects)

1.7 Time Schedule

1.8 General system block diagram.

1.9 Report Content

1.1 Overview

Nowadays the wireless communication systems and there technologies is widely used such as the IP network and it's technologies like VoIP (Voice over IP) and GSM (Global System for Mobile communication) network technology, so we can use the availability of the internet access to design an intercom system that will use the VoIP technology as well as the GSM.

In one hand the absence of the property owner (house for example) is sometimes annoying to people, so we will reach the owner or the needed person anywhere and make a call with him, in the other hand people nowadays are going toward the IP technology to save time and money and to easily reach each other.

1.2 General Idea about the project

The main idea is – in one hand - to design and build an intercom system based on VoIP and GSM to reach the house owner through VoIP if he had internet access or through the GSM network if he hadn't internet access and so answer the intercom using his mobile phone virtually from anywhere.

In the other hand, establishing and setting up the Elastix server (see chapter 2.4) which will give us a wide range of useful options to deal with daily life issues.

1.3 Project Objectives

- Easily answer the intercom from virtually anywhere with some of Ambiguity reply to intercom even if you are not at home.
- Solve daily life issues due to the absence of the needed person.
- Achieve security through answering the intercom anywhere and anytime, and events recording.

1.4 Motivations:

- The absence of the house's owner is annoying at some difficult circumstances such as an emergency case if he was a doctor.
- Any property guard went to inspection tour and a goods are needed to be delivered while he is in his patrol away from the front gate, the guard can be reached using the gate intercom.
- This type of communication is quick and efficient and, with no range limit.

1.5 Requirements

Hardware:

- Mobile phone (endpoint) that supports the SIP VoIP protocol.
- Intercom circuit (door panel unit).
- Elastix server (server that will control the system)
- FXS and FXO cards.
- DoorBell Fon.
- Controller (optional)
- GSM module (GSM modem , PIC controller)

Software:

- Elasxtix software.
- 3CX mobile phone software
- C programming software.

1.6 Literature Review (Related Projects)

1.3.1 SIP-enabled IP Outdoor Intercom

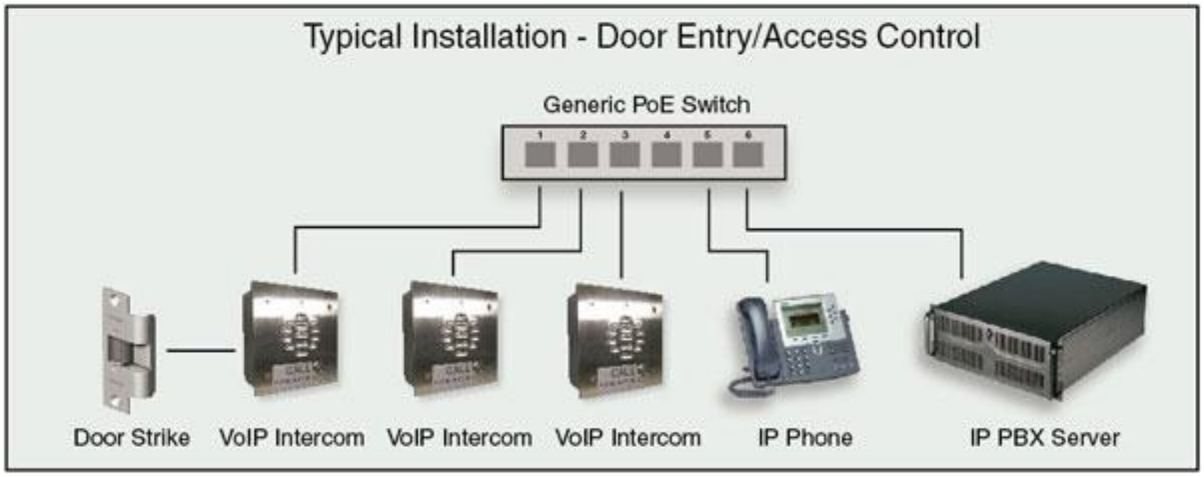


Figure 1.1: Cyber Data VoIP intercom product

Cyber Data is one leading IP endpoint company that designs and manufactures innovative SIP-enabled IP Paging and Two-way Communication/Access Control devices that are PoE (Powered over Ethernet 802.3af or 802.3at). The endpoints in here can be a wired IP phones or other inter-coms in the house.

Our design is intended to have a GSM module beside the VoIP technology if there is no internet access at the endpoint the owner can receive a GSM call instead of VoIP call , also our system well reach a mobile phone – not a wired IP phone only - anywhere (wirelessly).

1.3.2 VoIP Intercom/Roip 302M with GSM module

DBL Technology Limited is a leading manufacturer in developing and manufacturing advanced VoIP products for worldwide market. This product support:

Radio over IP (RoIP) technology is to convert the audio and Push To Talk (PTT) signals in a radio terminal into IP packets and then transmit the data via the IP networks. The challenge in this technology is to insure that the audio is transmitted in real time and the Push To Talk (PPT) control signal is transmitted immediately and reliably.

In this project, we will make the VoIP technology and the GSM technology operate alternatively unlike the VoIP Intercom/Roip 302M with GSM module which operate only in one way, also we intend to merge the intercom circuit with such a VoIP system.

1.3.3 Vedix GSM door intercom system

Figure 1.2: Vedix GSM door intercom product

Vedix Security Ltd. is company that manufactures and distributes a GSM door intercom system a worldwide. This system is designed to work on the same technology as mobile phones. It enables a call to be made from an entry point (Door, gate etc), to any telephone number (mobile or land line).

We intend to -in our project- build a system that support two technologies – VoIP and GSM – to work alternatively not only a one technology such as GSM, also the GSM module we intend to build cost less than the product provided by Vedix company.

1.7 Time Schedule

First semester (table 1.1)

- **Stage 1 :select the idea**

Determine the idea of our project.

- **Stage 2 :preparing for the project**

In this stage, we've become deeper for the steps we want to do.

- **Stage 3 :project analysis**

In this stage, a study of all possible design options to determine our own design.

- **Stage 4 :determine the project requirements**

After determine our design scheme, we specify all the needed requirements for the user and the system, software and hardware. And try to bring them to be ready for the implementation stage.

- **Stage 5 :studying the principles**

This stage of the project is necessary to study the VoIP, GSM availability in our project and how we will apply their principles in the system design.

- **Stage 6 :documentation**

Documenting the project will being done in parallel with the previous stages.

Second semester

- **Stage 7 :make the hardware available**
- **Stage 8 :build up the system**
- **Stage 9 :testing the system**
- **Stage 10 :writing documentation**

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

Table 1.1: Time planning for first semester

Week / Task	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
S7															
S8															
S9															
S10															

Table 1.2: Time planning for second semester

1.8 General system block diagram

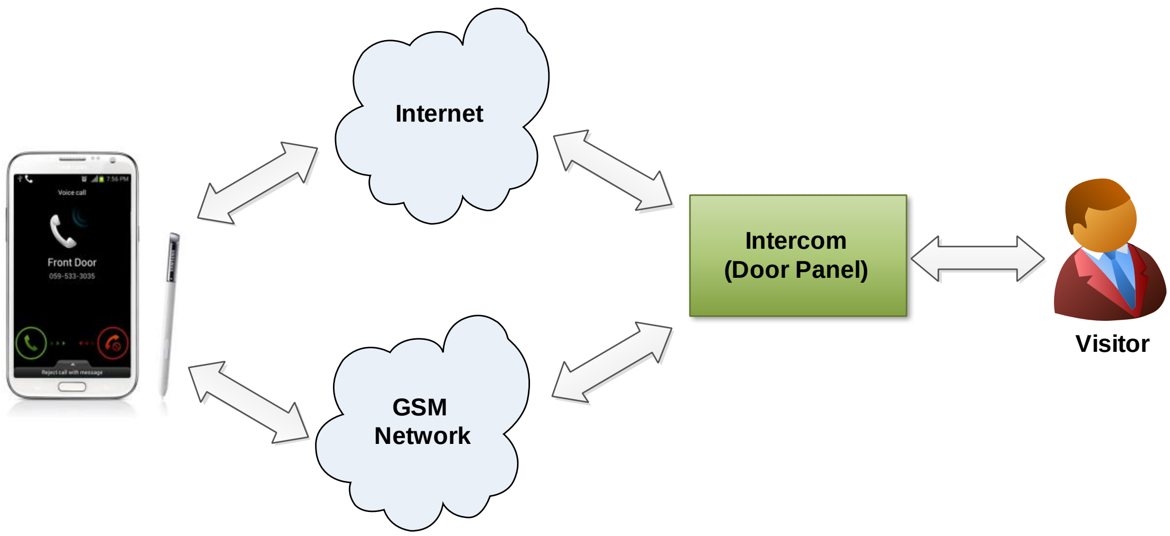


Figure 1.3: General block diagram

Figure 1.3 shows that the general block diagram for our graduation project, which includes the visitor, door panel, Internet network, GSM network and the endpoint which is the mobile phone. In the next chapters we will introduce the used technologies and how we will design our system to achieve the project goals and objectives.

1.9 Report contents

Chapter one

“Introduction”, this chapter gives an introduction about the project, its overview, objectives, motivation, related works and project time plan.

Chapter two

“Theoretical background”, this chapter introduces a background related to the main concept and technologies of the project. It’s a background of intercom, VoIP, GSM and Elastix.

Chapter three

“System design”, this chapter describes the project objectives, a general block diagrams and explains how the system works.

Chapter four

“Detailed system design”, this chapter describes the project detailed system design steps, such as installing and learning the project software, implementing and installing the hardware parts of this project.

Chapter five

“Testing and Monitoring”, the project part testing and monitoring calls.

Chapter six

“Conclusion and future work”, this chapter includes the results of the system, suggestions and improvements to the system which could be used in future work.

2

Chapter Two

Theoretical Background

2.1 Intercom

2.2 VoIP technology

2.3 GSM technology

2.4 Project Software (Elastix)

2.1. Intercom

2.1.1 Definition

Intercoms are electronic communications systems which consist of fixed microphone/speaker units (door phone) that connect to a central control device, enables home owners to communicate directly between rooms, the front door or the entire house through the Intercom Stations located in each room. Is a stand-alone voice communications system for use within a building or small collection of buildings, functioning independently of the public telephone network. There are two basic types of products: hard wired and wireless.

Traditional intercom systems are composed entirely of analogue electronics components but many new features and interfacing options can be accomplished with new intercom systems based on digital connections. Video signals can be carried as well as voice.

The following figure describes the main component of intercom systems:

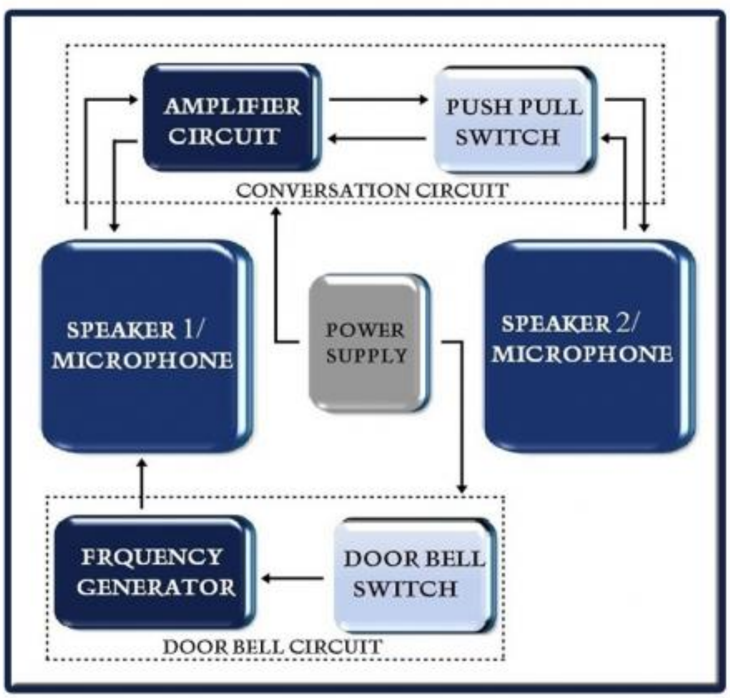


Figure 2.1: intercom system block diagram

2.1.2 Types

1) **Wired intercom** systems are more easily installed when a home is under construction, though it is possible to retrofit a wired system if you aren't intimidated by installing behind-the-wall wiring. Wired intercoms can also be installed as part of a whole-house, structured media plan, which might include a security/surveillance system and home automation. [1]

With a wired intercom system, the master station, or control panel, is hardwired into your electrical system and the low-voltage wires that support the substations in rooms throughout your home or at your front door or entry gate are run behind the walls or outside from each substation to the master station. Despite the relative complexity of retrofitting your home for a wired intercom, there are several advantages to a wired system, including: [1]

- A wired system is closed. No one outside the system, such as someone with a similar intercom or other device operating on the same radio frequency, can listen in.
- A wired system operates with little interference from outside sources such as a microwave or other wireless device, which can disrupt wireless signals.
- A wired system offers high-quality sound throughout the home. There are no dead spots where an intercom doesn't work and no quality loss for substations that are farther away from the master station.

2) **Wireless intercom** unit uses radio frequencies to transmit intercom signals. This type of intercom is less expensive than hardwired and carrier-current intercom systems. Interference may cause a problem, depending on the distance and the brand of the wireless intercom system. Typically, most wireless devices work straight out of the box without any installation required. Most of the indoor/outdoor wireless intercom systems have a portable receiver that allows you to conveniently use the device from various locations. This comes in handy at times such as when you are outside in the backyard and a visitor arrives at your front door. In addition to ease of installation, other advantages to wireless intercoms include: [2]

- Wireless substations are completely portable. Move them from room to room anywhere an outlet is located. This portability means you can find out who's at the door without getting up from your chair to go to the master station, for example.
- Depending on the range of your system, and most wireless systems operate at a range of 1000 to 1500 ft. from master station to substation, you can use a wireless intercom in an outbuilding, such as a detached garage or workshop, allowing you to stay connected to home even when outside.

2.1.3 Components

Whether wired or wireless, the main components of an intercom system are the same: [1]

The master station

Is the control panel. It contains the electronic circuitry for the intercom system, and may have the power switch for the entire system as well as radio or other music distribution controls. Some basic systems only allow you to talk from a substation to the master station, rather than from substation to substation. Keep this limitation in mind when you are shopping for a system.

Substations

Also called slave or remote stations, consist of a speaker, which can deliver voice and music and act as a microphone during a reply, and a switch to transfer from listen to talk modes.

Keep the following in mind when purchasing substations:

- Check the range on wireless substations if you're planning to install one in an outbuilding or at a gated entry. The range of wired substations is limited only by how far you are willing and able to run the wires connecting it.
- Outside substations, often installed at an entry door and including a doorbell feature, allow you to know who is ringing the bell without having to open the door.

Most stand-alone intercom systems are audio only, meaning you can talk and listen from station to station, but not see the person with whom you're talking. However, video systems are becoming more affordable and available. A video intercom includes a camera, with a monitor at the control panel, which allows you to see who's at the front door or entry gate, while you're talking with them, or you can keep an eye, as well as an ear, on your child's room or on an ill family member.

2.1.4 Features

Voice activated systems allow you to answer a call from the front door or another room without having to leave your chair or interrupt your task to push a button. Voice activated systems are also helpful for people confined to a bed, who can call for help without having to get up or struggle to reach a button. [1]

2.2. VoIP

2.2.1 Introduction:

The possibility of voice communications traveling over the Internet, rather than the PSTN, first became a reality in February 1995 when Vocaltec, Inc. introduced its Internet Phone software. Designed to run on a 486/33-MHz (or higher) personal computer (PC) equipped with a sound card, speakers, microphone, and modem (see Figure 1), the software compresses the voice signal and translates it into IP packets for transmission over the Internet. This PC-to-PC Internet telephony works, however, only if both parties are using Internet Phone software.

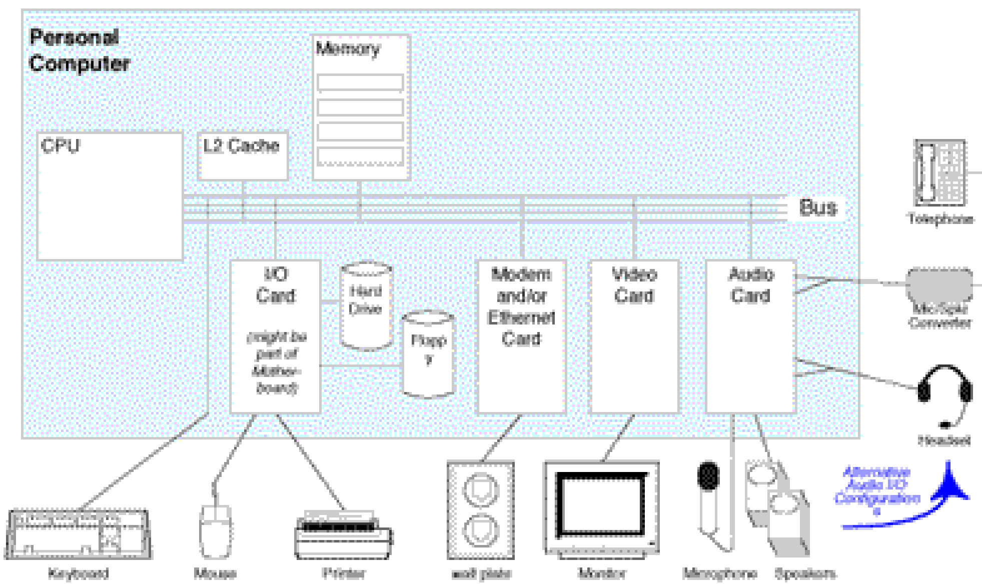


Figure 2.2: PC Configuration for VoIP

2.2.2 Definition:

VoIP (Voice over Internet Protocol) , a category of hardware and software that enables people to use the Internet or your own internal network as the transmission medium for telephone calls by sending voice data in packets using IP rather than by traditional circuit transmissions of the PSTN.

“commonly refers to the communication protocols, technologies, methodologies, and transmission techniques involved in the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. “

Other terms commonly associated with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, IP communications, and broadband phone.

2.2.3 How does VoIP Work?

The main idea of VoIP is convert the voice into digital signal that travels over the internet. When you calling a phone number, the signal is converted to a regular telephone signal before it reaches the destination. VoIP can allow you to make a call directly from a computer, a special VoIP phone, or a traditional phone connected to a special adapter. Also if you have wireless in a public region like cafes, airports, etc then you can connect to the internet and use VoIP service. [4]

2.2.4 Advantages and disadvantages of VoIP

Advantages

Lower cost

In general phone service via VoIP costs free or less than equivalent service from traditional sources.

In the most case, users see VoIP phone calls as FREE. Just they paid money for the internet service. While, using VoIP over this service may not involve any extra charges, so the users view the calls as free. Examples are Skype, Viber.

Increased Functionality

VoIP provides some offer that are difficult to be available with traditional phone networks, or are available but only for an additional fee. You may also be able to avoid paying for both a broadband connection and a traditional telephone line. Incoming phone calls are automatically routed to VoIP phone wherever you plug it into the network. Everywhere you have internet connection then you can receive incoming calls.

Simple and Scalable Infrastructure

The infrastructure of this technology can be simply build due to its scalability of adding new components easily without any difficulty. The process of converting the voice into signals is based on software rather than hardware which make it easy to alter and maintain the whole system. The hassles of separate cabling for telephone systems can also be avoided by using this technology.

Disadvantages

If you're considering replacing your traditional telephone service with VoIP, there are some possible differences:

1-Some VoIP services don't work during power outages and the service provider may not offer backup power.

2-Not all VoIP services connect directly to emergency services .

3-VoIP providers may or may not offer directory assistance/white page listings.

2.2.5 VoIP Protocols

First of all the VoIP technology is intended to resolve some key issues to ensure that this technology can be relied on and to become popular. Some of them are discussed below [3]:

- **QoS (Quality of Service):**

The VoIP technology must provide a real time guarantees and a good voice communications acceptable for the user.

1. The delay must be less than a threshold value.
2. Echo cancellation.
3. Packet prioritization (give the voice packets high priority).
4. Error detection and correction.

- **Interoperability**

This means that products from different venders must be able to operate together such as the interoperability in GSM network. So standards must be agreed and devised to achieve interoperability. Some of these standards are H.323 and Tiphon from ITU.

- **Security**

As we are using the IP network, the security issues exist. In a public Internet, the packets can be routed through any router and can be intercepted by anyone. Security can be obtained by using encryption and tunneling. Tunneling: to establish a secure tunnel between two gateways or gatekeepers.

- **Integration with Public Switched Telephone Network(PSTN)**

An important issue to make the two networks (IP and PSTN) appear to be one network to the end-user and easy to manage by operator.

- **Scalability**

VOIP systems need to be flexible enough to grow to large user market and allow a mix of private and public services.

2.2.5.1 H.323 standard

The ITU-T (International Telecommunications Union) developed this standard for multimedia conferencing over LANs and later was extended to cover Voice over IP. The H.323 standard includes point to point communications and multipoint communications. Interoperability can be achieved if the different venders use this standard on their product and applications. [4]

H.323 Architecture

The figure 2.3 shows the main elements of the H.323 protocol, these elements include terminals, gateways, gatekeepers, and multipoint control units (MCU).

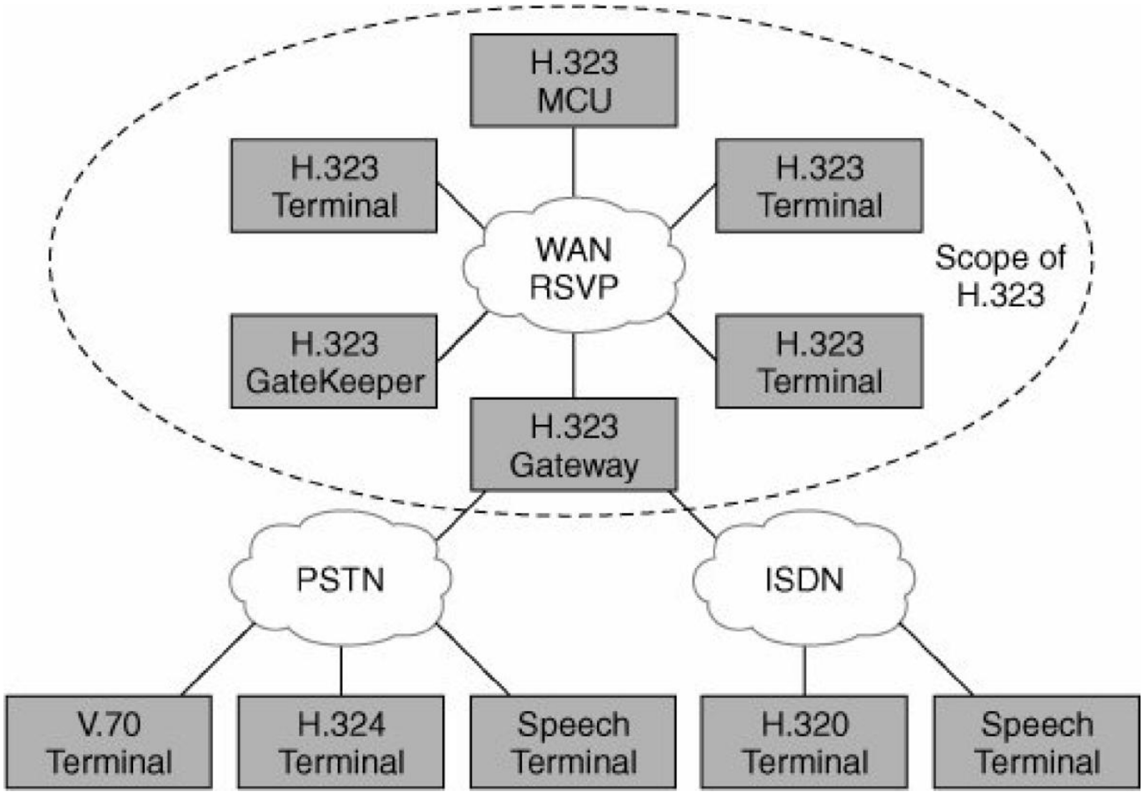


Figure 2.3: H.323 architecture

Terminal

This provides point-to-point or multipoint conferencing for audio and, optionally, video and data communication.

Gateway

Interconnect to Public Switched Telephone Network (PSTN) or ISDN networks for H.323 endpoint interworking. It translates between audio, video, and data transmission formats as well as communication systems and protocols.

Gatekeeper

An optional function, the gatekeeper provides pre-call and call-level control services to H.323 endpoints and also provides admission control and address translation services for terminals or gateways.

Multipoint Control Units (MCU)

Supports conferences between three or more endpoints in a multipoint conference. The MCU consists of a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP). MCs transmit the capability set to each endpoint in the multipoint conference. The multipoint processor (MP) receives audio, video, and/or data streams and distributes them to endpoints participating in a multipoint conference, as shown in figure 2.3.

2.2.5.2 Session Initiating Protocol (SIP)

It is an application layer control and signaling protocol for creating, modifying and terminating sessions with one or more participants. SIP is a text-based protocol that is similar to HTTP and Simple Mail Transfer Protocol (SMTP). SIP is a peer-to-peer protocol, which means that network capabilities such as call routing and session management functions are distributed across all the nodes (including endpoints and network servers) within the SIP network. It provides the following services: [4]

1. User Location: the capability of determining the location of the end user.
2. Call Setup: establishment of session parameters for the parties who are involved in the session.
3. User Availability: determination of the willingness of the end user to engage in communication.
4. User Capabilities: determination of the media capabilities of the devices that are involved in the session.
5. Call Handling: the transfer, modification and termination of calls.

SIP Architecture

The figure 2.4 shows the main elements of the SIP protocol. The SIP network typically comprises the following devices: [5]

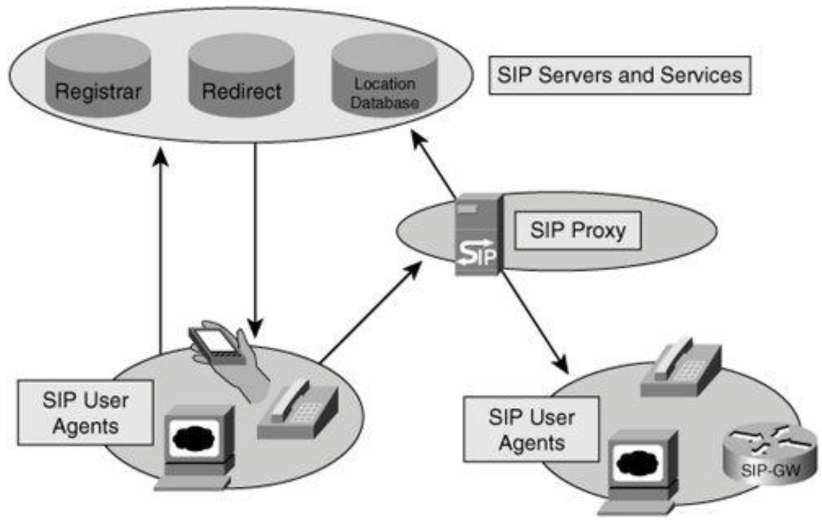


Figure 2.4: SIP system elements

SIP user agents (Clients and Servers)

User Agent Client (UAC) is used to initiate SIP request and accept SIP response. Example of UAC is a phone that supports SIP protocol (Samsung Galaxy S2).

User Agent Server (UAS) is used to receive requests and return response. Example of UAS is a SIP phone accepting an invite request.

Register server

Receives update about the current location of the users and save these updates into the location database.

Redirect server

Determines the next-hop server on receiving requests and returns the address of the next-hop server to the client instead of forwarding the request.

Proxy server

On receiving requests, forwards them on- behalf of the UAC- to the next-hop server, which has more information about the location of the called party.

SIP Messages

There are some messages are used for communication between clients and SIP servers. These messages are divided into two categories, SIP REQUESTs and SIP RESPONSEs.[4]

SIP REQUESTs

1. INVITE: invite a user to participate in a session or modify the characteristics of a previously established session.
2. ACK: used only with INVITE requests, this means that the UAC has received the INVITE requests.
3. OPTIONS: to inform the UAS about the UAC call capabilities.
4. BYE: terminate a session or ongoing call.
5. CANCEL: cancel a request, such as cancel a sent INVITE request to UAS.

SIP RESPONSEs

The responses are generated by the proxy server or the UAS to indicate the status of the SIP request sent by UAC. SIP responses are numbered from 100 to 699 and grouped as 1xx to 6xx indicates the following:

- 1xx: indicates progress by the server.
- 2xx: indicates successful processing of the SIP request.
- 3xx: indicates that the SIP request needs to be redirected to another UAS for processing.
- 4xx, 5xx or 6xx: indicate failure in processing of the request.

In the following table there are some examples of the SIP responses and there indications:

Class of response	**Status of code**	**Indication**
informational	100	Trying
	180	Ringing
Success	200	Ok
Redirection	300	Multiple choices
	380	Alternative server
Client error	400	Bad request
	404	Not found
	408	Request time-out

Table 2.1: SIP response table

SIP message structure

A SIP message consists of the following:

- A start-line.
- One or more header fields.
- An empty line indicating the end of header fields.
- An optional message body.

The following table is an example of a SIP request message, because the SIP is a text-based protocol that is similar to HTTP and Simple Mail Transfer Protocol (SMTP), the SIP messages is not in binary representation. [4]

INVITE sip:bob@proxy.company.com SIP/2.0	Request Line
Via: SIP/2.0/UDP ph1.company.com:5060;branch=z9hG4bK83749.1 From: Alice <sip:alice@company.com>;tag=1234567 To: Bob <sip:bob@proxy.company.com> Call-ID: 12345601@ph1.company.com CSeq: 1 INVITE Contact: <sip:alice@ph1.company.com> Content-Type: application/sdp Content-Length: ... ……. Message body ……	SIP message header Blank line

Table 2.2: SIP request component

SIP message header conveys the signaling and routing information for the SIP network elements. The header fields consists of a field name followed by a colon (:) and the value of that field.

The following table describes the function of the key SIP headers:

SIP header	Explanation
From	Describes the identity of the SIP request initiator.
To	Indicates the requested receiver of the SIP request.
Call-ID	Identifies a series of SIP messages.
Cseq	Identifies order, sequences SIP requests and differentiates between new messages and retransmission messages. Defined by integer value.
Via	Indicates the path of the SIP request message.
Contact	Identifies where the UA wants to receive a new SIP request.
Content-Type	Identifies the type of the message body.
Content-Length	Identifies the size of the message body (decimal size).

Table 2.3: SIP header key function description

SIP operation

The UAs (callers and callees) in VoIP networks are identified by addresses. The following figure describes the main steps to initiate a SIP call: [6]

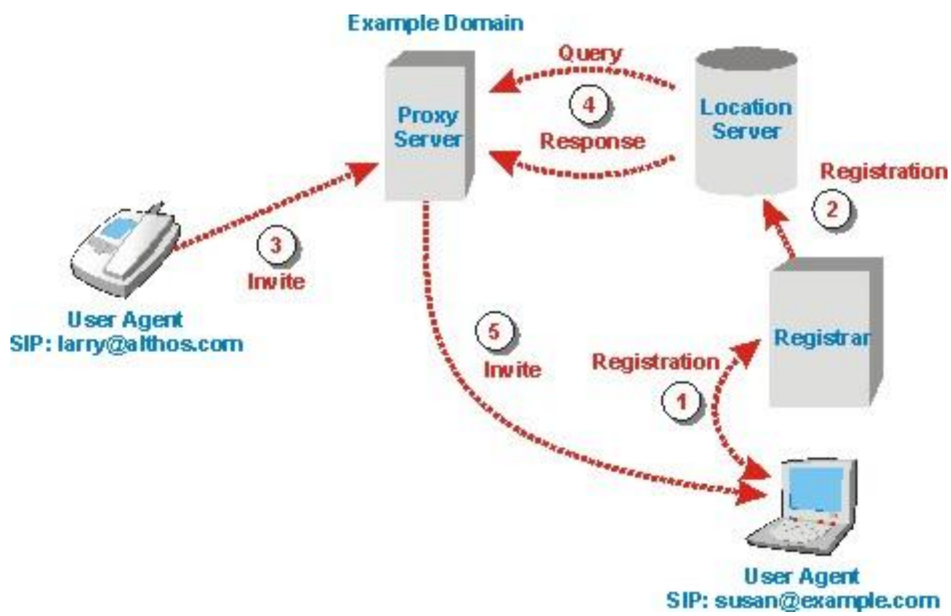


Figure 2.5: SIP operation

The SIP protocol perform its operation in six main steps, these steps are:

1. SIP addressing.
2. Locating a SIP server.
3. SIP transaction.
4. SIP invitation.
5. Locating a user.
6. Changing an existing session.
7. Terminating a session.

SIP addressing

The UAs in the network must be identified by an address, the address here in the form of URL (username@host).

SIP server locating

The client can either send the request to a SIP proxy server or it can send it directly to the IP address and port corresponding to the Uniform Request Identifier (URI).

SIP transaction

The client can send requests to the located SIP server. SIP transaction is the process where the requests and the responses triggered by the request sent to the located SIP server.

SIP invitation

The following flow diagram shows the invitation process:

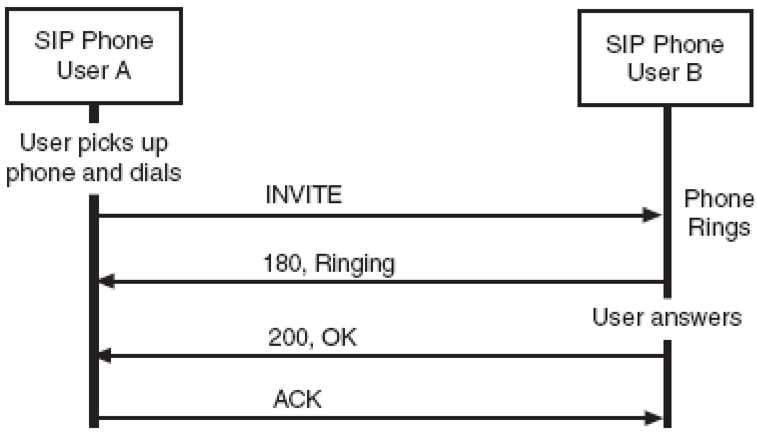


Figure 2.6: SIP invitation session

User A sent an INVITE request to user B to join a session or establish a two party conversation. User B send a 180 code (ringing) and an ACK (code 200) if it answers the requested call. Then user A confirms that it has received that response by sending an ACK request.

Locating a user

User B may change its location many times, these locations, must be dynamically registered to the SIP server. There is a server in the SIP system called Location Server this server is responsible for generating a list of all possible locations and pass them to the SIP server.[5]

Changing an existing session

Sometimes we need to change an existing session parameters due to some errors of fault in preceding the call. The main parameters remain unchanged (such as Caller ID) but the changing done by re-issuing the INVITE message using the same Call ID but a new body to convey the new information.[5]

Terminating a session

When user B wants to terminate ongoing call or a session, it sends a BYE request to user A. User A send back an ACK (code 200) response to terminate the session as shown in the following flow diagram:

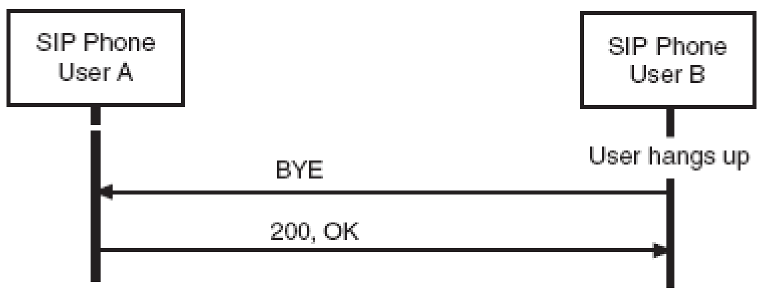


Figure 2.7: SIP session termination

The following call flow shows how to initiate a SIP session and terminating it:

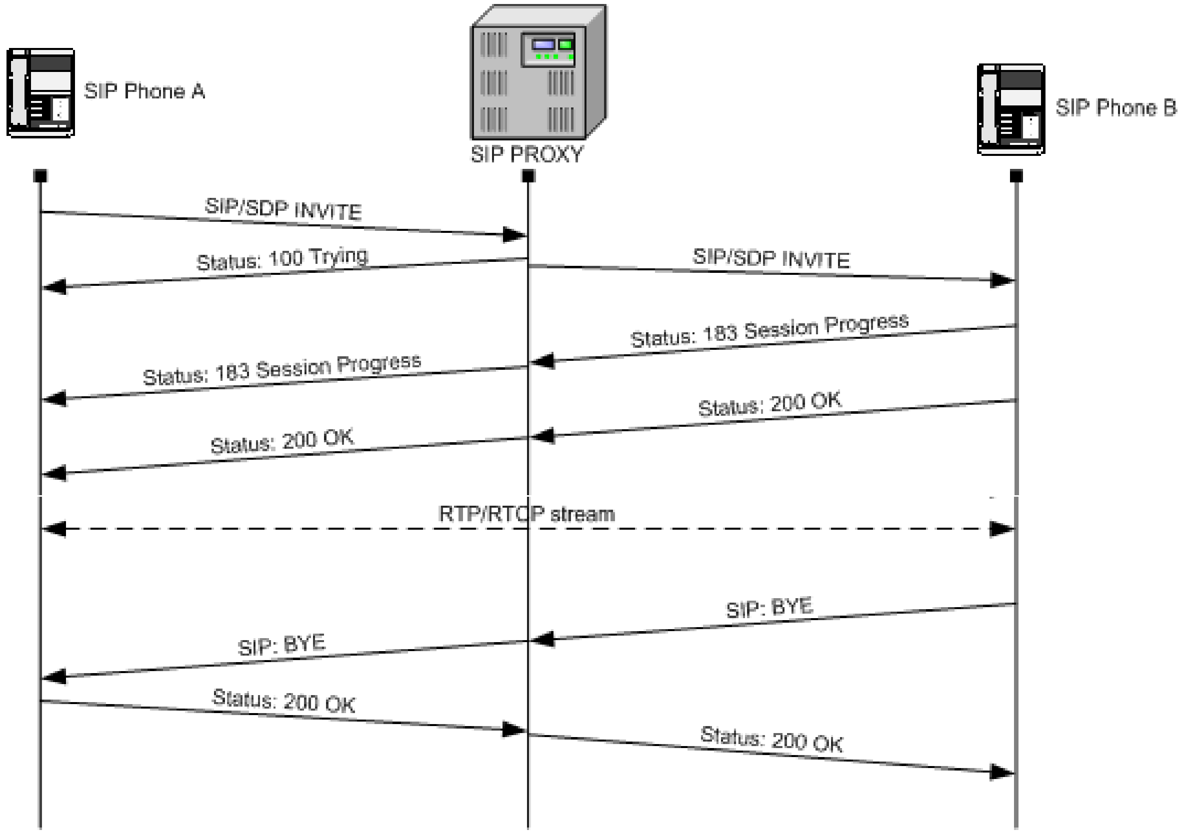


Figure 2.8: SIP Call Flow diagram

2.2.5.3 Supporting protocols

There are some protocols that work in parallel with the SIP protocol to provide an interactive real-time multimedia session. These protocols are needed by the SIP agents or application because of the following: [7]

- To describe the characteristics of a session the characteristics include whether a session is an audio or video session, what codec are used, what the media source is, and what the destination addresses are.
- To handle media these protocols control and transmit audio/video packets for a session.

- To support functions Needs include AAA for authentication, authorization, and accounting; Resource Reservation Protocol (RSVP) for reserving network resources; RTP/RTCP is used for transporting real time data, RTSP for controlled delivery of streams, SAP (Session Announcement Protocol) for advertising multimedia sessions and SDP (Session Description Protocol) for describing multimedia sessions; Domain Name System (DNS) for hostname-to-IP address resolution; and Transport Layer Security (TLS) for preventing eavesdropping, tampering, or message forgery.[4]

2.2.6 Comparison between H.323 and SIP

In the VoIP part in our project we will use the SIP protocol in the needed equipments and in the call process rather than H.323 protocol because the SIP protocol takes many advantages on H.323. These advantages are listed in the comparisons table below: [8]

	H.323	**SIP**
Complexity	Complex	Simple protocol
Representation Form	Binary representation	Textual presentation
Compatibility	full backward compatibility	Doesn't require full backward compatibility
Scalability	Not very scalable	Highly scalability
Signaling	Complex signaling	Simple signaling
Loop Detection	Difficult	Easy

Table 2.4: H.323 and SIP comparison

2.2.7 VoIP applications and devices

Nowadays there are many applications that use VoIP as a communication way between people, and there are many providers that provide the VoIP technology.

In order to use the VoIP services we must have an access to the IP network and an end-point that support VoIP protocols like the IP telephone. [7]

The use of Wi-Fi enabled (compatible with 802.11a/b/g/n standards) IP phones in the enterprise enables users to be connected from anywhere within the enterprise, from public hotspots, and from their home offices. The use of Wi-Fi phones affords all the productivity gains as well as better audio quality than cellular systems. Wi-Fi phones enable users within an enterprise to receive calls made to their extensions, without being tied to their desks.

Some of the Wi-Fi phones can also allow users to switch between VoIP and cellular phone networks. A person may initiate a VoIP call using his phone at the office. As he walks out of the office, the call gets seamlessly switched over to a cellular network. When he comes back within the enterprise network, the call gets moved back to the enterprise VoIP network.

The advantage is much greater flexibility in mobile communication as well as a potential cost savings by shifting the call minutes from a cellular network to an enterprise network. [4]

2.3. GSM Technology

GSM stands for Global System for Mobile Communications. Just like computers, mobile phones have evolved over time. There were first generation mobile phones in the 70's, there are 2nd generation mobile phones in the 80's and 90's, and now there are 3rd generation mobile phones which we call as 3G phones. So this GSM is called a 2nd generation, or 2G communications technology.

One of the most important conclusions from the early tests of the new GSM technology was that the new standard should employ Time Division Multiple Access (TDMA) technology. This ensured the support of major corporate players like Nokia, Ericsson and Siemens, and the flexibility of having access to a broad range of suppliers and the potential to get product faster into the marketplace. After a series of tests, the GSM digital standard was proven to work in 1988. [9]

2.3.1. GSM Radio Spectrum

The ITU, which manages the international allocation of radio spectrum, allocated the 890-915 MHz bands for the uplink (mobile station to base station) and 935-960 MHz bands for the downlink (base station to mobile station) for mobile networks in Europe. Since this range was already being used in the early1980s by the analog systems of the day, the CEPT had the foresight to reserve the top 10 MHz of each band for the GSM network that was still being developed. It should be noted that the World Radio-Communications Conference (WRC) in 1992 identified frequency bands for FPLMTS (Future Public Land Mobile Telecommunications Systems), which is in fact the original name of IMT-2000 (UMTS).42 The existing second-generation bands for second-generation GSM services consist of spectrum between 862 and 960 MHz and the totality of the GSM 1800 band 1710 - 1880 MHz [9]

2.3.2. Network Components

GSM networks are made up of Mobile services Switching Centers (MSC), Base Station Systems (BSS) and Mobile Stations (MS). These three entities can be broken down further into smaller entities; such as, within the BSS we have Base Station Controllers, Base Transceiver Stations and Transcoders. [10]

Mobile Stations within the cellular network are located in "cells"; these cells are provided by the BSSs. Each BSS can provide one or more cells, dependent on the manufacturers equipment.

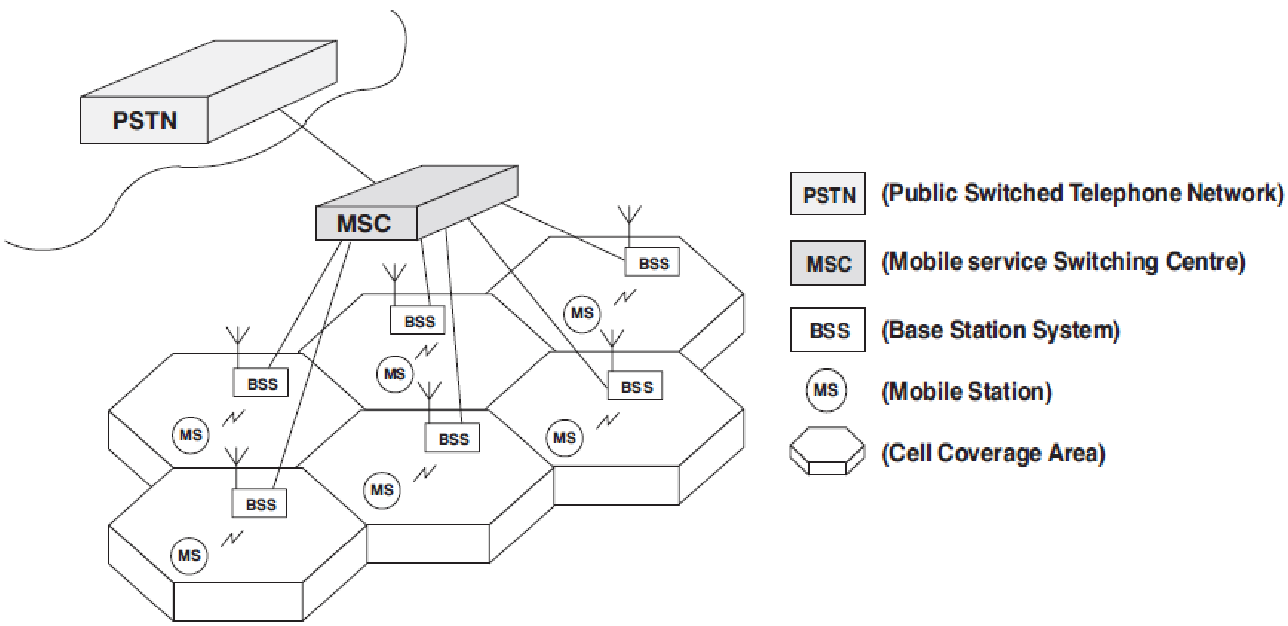


Figure 2.9: GSM network component

Each network component is designed to communicate over an interface specified by the GSM standards. This provides flexibility and enables a network provider to utilize system components from different manufacturers. For example Motorola Base Station System (BSS) equipment may be coupled with an Ericsson Network Switching System. [10]

The principle component groups of a GSM network are:

- **The Mobile Station (MS)**

This consists of the mobile telephone, fax machine etc. This is the part of the network that the subscriber will see.

- **The Base Station System (BSS)**

  This is the part of the network which provides the radio interconnection from the MS to the land-based switching equipment.

- **The Network Switching System**

  This consists of the Mobile services Switching Centre (MSC) and its associated system-control databases and processors together with the required interfaces.

  This is the part which provides for interconnection between the GSM network and the Public Switched Telephone Network (PSTN).

- **The Operations and Maintenance System**

  This enables the network provider to configure and maintain the network from a central location

2.3.3 Features of GSM Technology

There are many features associated with GSM technology due to which it is bar far the most leading mobile communication technology in the world today. GSM technology facilitates with high speed integrated data, voice data, fax, mail, voice mail and mostly used SMS feature. GSM also make sure that all the communication made between networks are secured and protected from intruders and frauds.[11]

Here are the main features of the GSM network:

- **Compatibility**

GSM has been specified and developed by many European countries working in co-operation with each other. The result is a cellular system which has been implemented throughout Europe and many parts of the world.

- **Noise Robust**

The noise which interferes with the current system may be produced by any of the following sources:

- A powerful or nearby external source (a vehicle ignition system or a lightning bolt, perhaps);
- Another transmission on the same frequency (co-channel interference).
- Another transmission “breaking through” from a nearby frequency (adjacent channel interference);
- Background radio noise intruding because the required signal is too weak to exclude it.

In order to combat the problems caused by noise, GSM uses digital technology instead of analogue.

By using digital signals, we can manipulate the data and include sophisticated error protection, detection and correction software. The overall result is that the signals passed across the GSM air interface withstand more errors (that is, we can locate and correct more errors than current analogue systems). [10]

- **Flexibility and Increased Capacity**

GSM equipment is fully controlled by its software. Network re-configurations can be made quickly and easily with a minimum of manual intervention required. Also, since one carrier can support eight users, expansion can be made with less equipment.

GSM networks also offer the flexibility of international roaming. This allows the mobile user to travel to foreign countries and still use their mobiles on the foreign network.

- **Use of Standardized Open Interfaces.**

The equipment in each of the analogue cellular networks tends to be produced by one manufacturer. This is because the equipment is only designed to communicate with other equipment made by that manufacturer.

The situation is very different with GSM, where standard interfaces such as C7 and X.25 are used throughout the network. This means that network planners can select different manufacturers for different pieces of hardware. Competition between manufacturers is therefore intense in the GSM market and manufacturers must ensure they support the latest developments at a competitive price. [10]

- **Improved Security and Confidentiality.**

Security figures high on the list of problems encountered by some operators of analogue systems. In some systems, it is virtually non-existent and the unscrupulous were quick to recognize this. With some of the “first generation” systems, it has been estimated that up to 20% of cellular phone calls are stolen.

With GSM, both the Mobile Equipment (ME) and Mobile Subscriber are identified. The ME has a unique number coded into it when it is manufactured. This can be checked against a database every time the mobile makes a call to validate the actual equipment.

The subscriber is authenticated by use of a smart card known as a Subscriber Identity Module (SIM), again this allows the network to check a MS subscriber against a database for authentication. [10]

- **Flexible Handover Processes.**

Handovers take place as the MS moves between cells, gradually losing the RF signal of one and gaining that of the other. [10]

When GSM was specified a great deal of thought went into the design and implementation of handovers. Although the GSM system is more complicated than analogue in this area, the flexibility of the GSM handover processes offer significant improvements which provide a much better quality of service to the subscriber. GSM provides handover processes for the following:

- Quality (uplink/downlink).
- Interference (uplink/downlink).
- RF level (uplink/downlink).
- MS distance.
- Power budget.

2.4 Project Software (ElastiX)

Elastix is an Open Source Software to establish Unified Communications. About this concept, Elastix goal is to incorporate all the communication alternatives, available at an enterprise level, into a unique solution. Elastix (as shown in figure 2.10) brings together IP PBX (Private Branch Exchange), email, IM, faxing and collaboration functionality. It has a Web interface and includes capabilities such as a Call Center software with predictive dialing. [12]

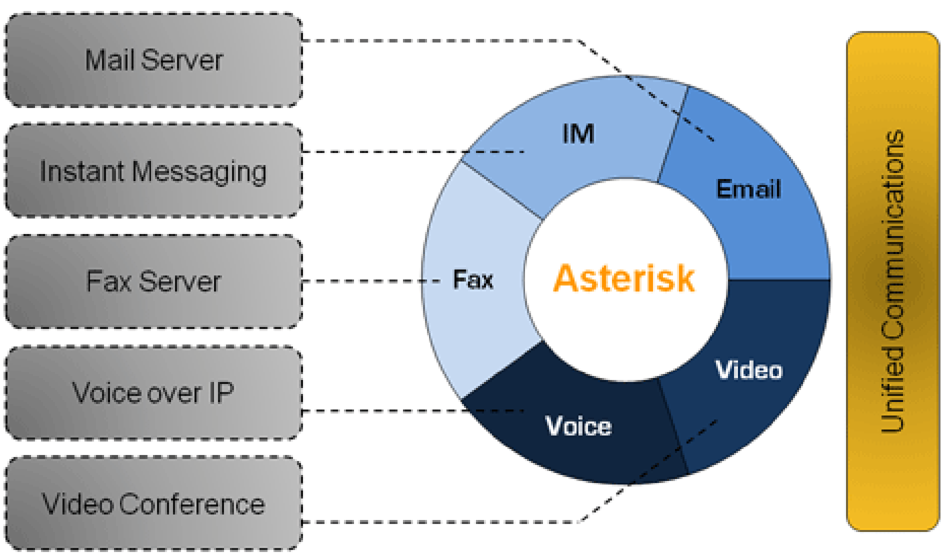


Figure 2.10: Elastix Communication media

Elastix is a collection of “best of breed” 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. Some of the major components that make up Elastix are: [13]

- **Flash Operator Panel**, a screen-based operator's console
- **Conferencing** control application
- **freePBX®** (embedded and standalone) a web User Interface tool for Elastix.
- **A Maintenance system**, also part of Elastix, which provides low level interfaces to some components and real time system information
- **OSLEC** - Software Based Echo Cancellation

Each of these proven products is written and maintained by separate companies or entities, and in many cases small and large companies are using these products in production.

Elastix developers have written a Web Interface that allows you to access these programs, so that they, in general, look like one complete product. Elastix have also written certain software such as reporting programs, Hardware detection, Network Configuration, Software Update Module, Backup Restore module, User Management and many more modules themselves. [13]

Elastix it's capable of establish an efficient environment for organizations with the addition of many features that allows to integrate other locations of companies to centralize business and take it global. A user in corporation located in South America for example shares the same functionalities of another located in Asia besides having direct internal communication. Some of the basic Features of Elastix include: [12]

- Voicemail
- Fax-to-email
- Support for soft phones
- Web Interface Configuration
- Virtual conference rooms
- Call recording
- Least Cost Routing
- Extension Roaming
- PBX Interconnection
- Caller ID

3

Chapter Three

System Design

3.1 Overview

The internet nowadays is nearly available in every house, building, organization and company. We will design our system based on this availability of the IP network and take the internet into a new level rather than browsing and e-mailing; we will apply the VoIP technology to easily reach the house owner anywhere.

In this chapter we will introduce and discuss the system design to achieve our goals in reaching the house owner using VoIP in first or GSM alternatively.

The following flowchart shows the main idea of our system:

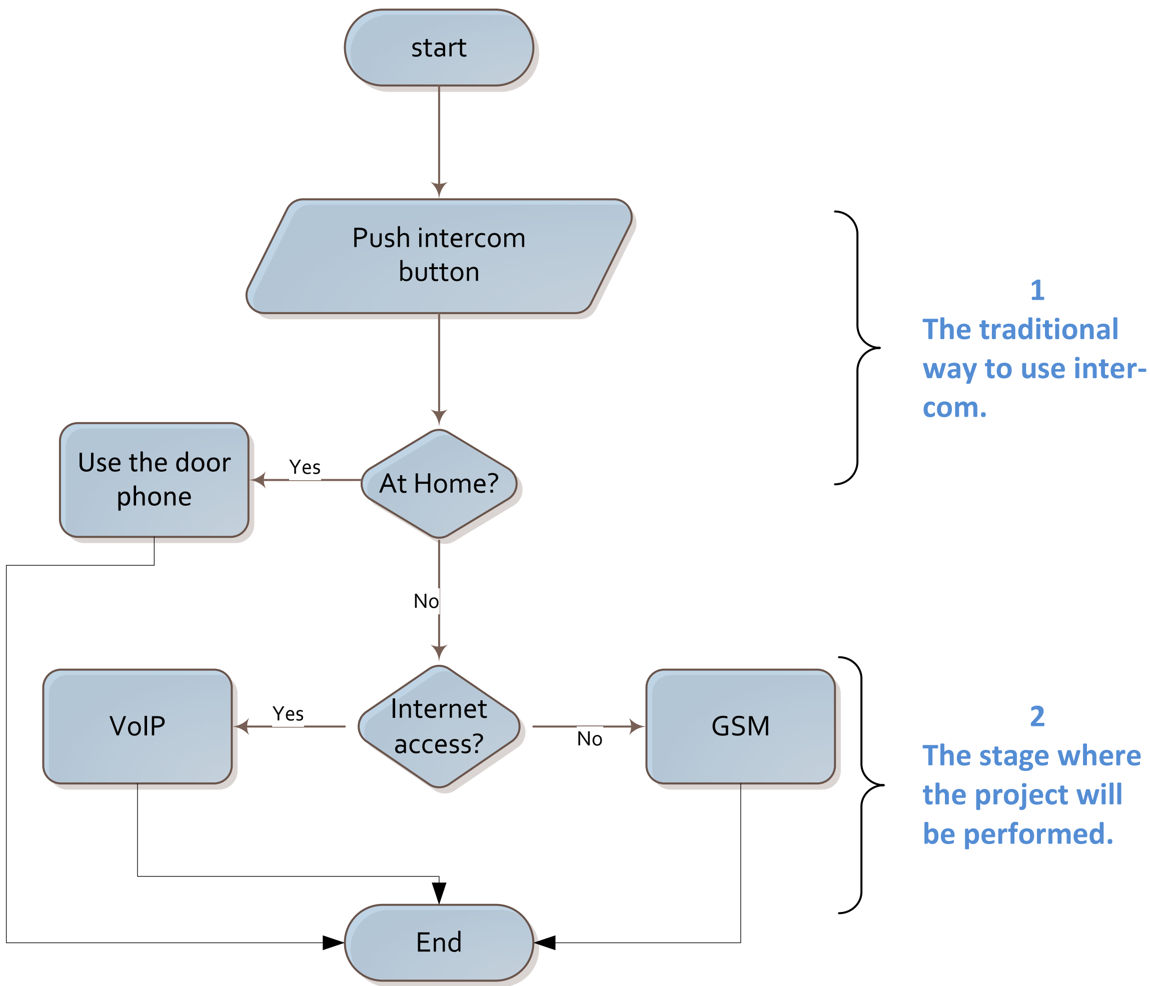


Figure 3.1: System behavior flowchart

As shown in figure 3.1 flowchart if the property owner (house owner) is at home he will use the intercom system and answer it from the handset located inside the home.

If the owner is not at home, then we will try to reach him anywhere using VoIP or GSM network (depending on his connection status). To achieve the second stage of the flowchart we need to setup and design a system that will call the owner anywhere he is.

Deep inside the system design first we have to introduce the main component of the intended designed system in second stage. The following figure (figure 3.2) shows the main component of the system.

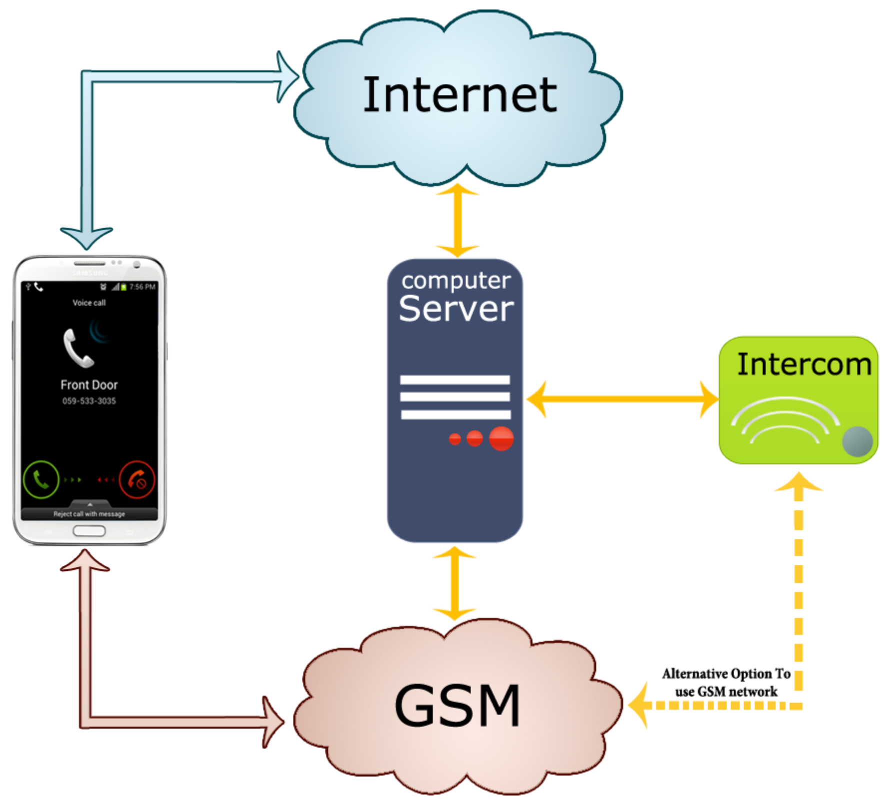


Figure 3.2: System Behavior

The main components of the system are:

- Intercom block
- Server
- Endpoint (Owners mobile phone or computer)

The next block diagram shows the main components and the sub components of the system and the design options (in gray-color) to reach the owner using the GSM network. The options to reach the owner via GSM network will be discussed later in section 3.4.

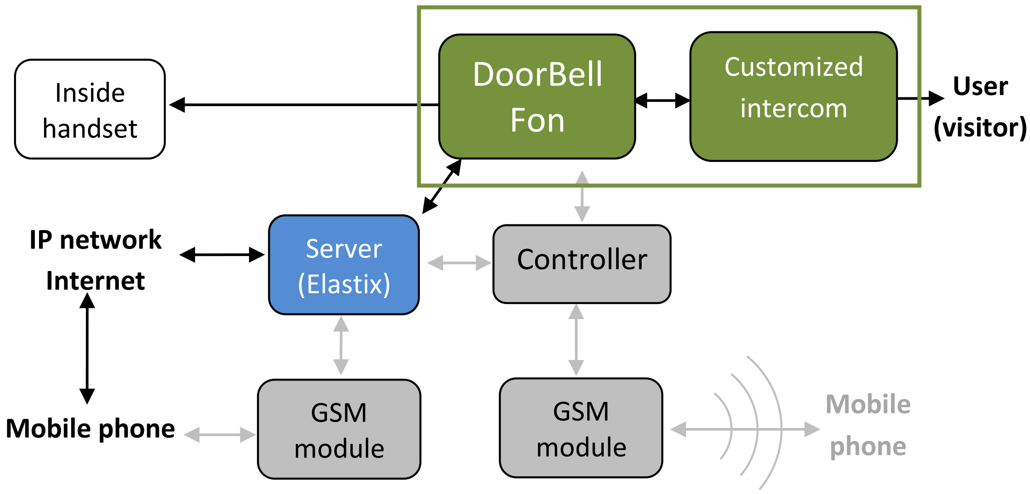


Figure 3.3: System block diagram

3.2 Intercom block

The first main component of the system is the intercom block, which will provide an interface between the visitor and the house owner. The intercom block and its components are described in the figure 3.4.

The intercom circuit (customized intercom) will provide an interface between the visitor and the rest of the system; it will receive the voice signal through a microphone sensor and convert it to its similar electrical signal.

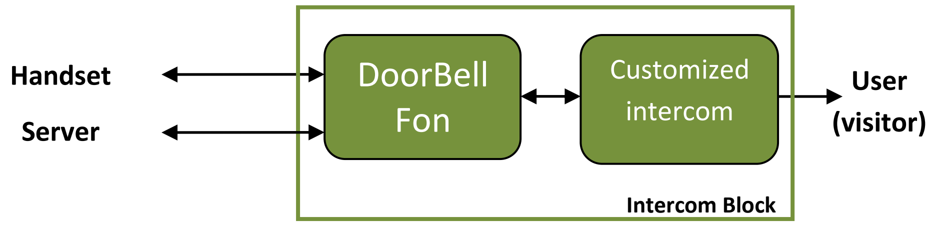


Figure 3.4: Intercom Block Diagram

The DoorBell Fon controller is the brain of this intercom system. The controller is designed to connect between the public telephone system and telephones within home. To accommodate this purpose, it has two RJ14 jacks, one that connects to the telephone line coming into home and the other that connects to a handset, or even multiple handsets, within your home. In addition to the two jacks, there is a wire connection panel that can accommodate two different doorbells. Each doorbell is connected via a negative and a positive lead to the controller so we can answer the intercom using the telephones in homes.

The customized intercom circuit shown in the figure 3.5, it was build from analog phone circuit with relays and PIC microcontroller(PIC 18F2550).

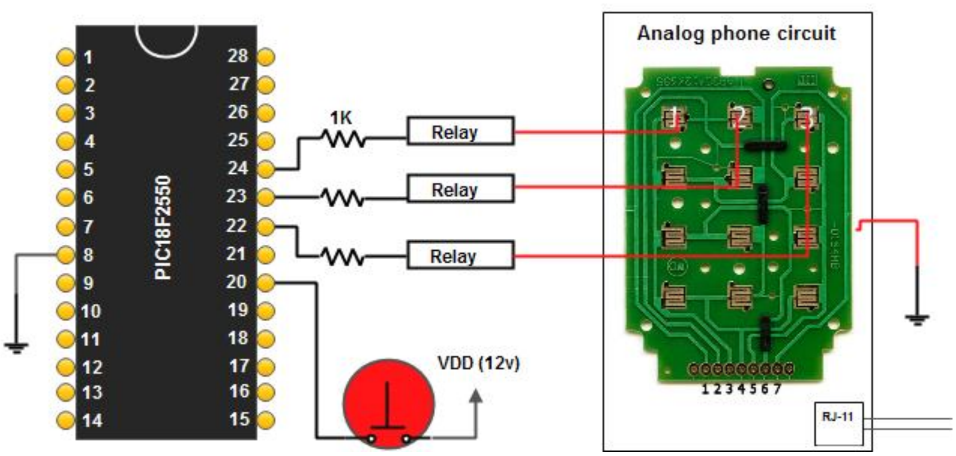


Figure 3.5: Customized Intercom

From analog circuit a three numbers connected with relays as well as a PIC controller, after pushing the key the controller will activate a relay to open the line and after one second a second relay will close number one circuit on the keypad and so on for numbers two and three.

3.3 The server

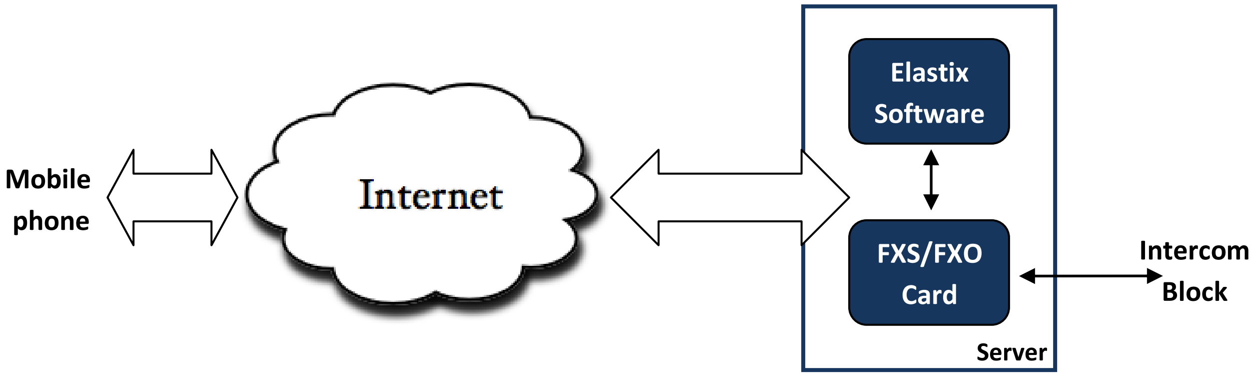


Figure 3.6: Server block diagram

The server is a PC with a FXS/FXO (foreign exchange station / foreign exchange office) card (PCI cards) and the Elastix software that will handle the audio routing process to its final destination (endpoint; houses owner mobile phone or maybe a computer).

FXS/FXO Card Functions

FXS/FXO (foreign exchange station / foreign exchange office) card (PCI cards) have 4 ports (channels) can be programmed to be output or input or both.

- FXS (Foreign Exchange Station) is an interface which delivers battery, and provides ringing. The FXS interface initiates and sends a ringing voltage to the FXO interface. The FXS will provide and sends the ringing voltage (SIP INVITE message) to the FXO circuit at the card.
- FXO (Foreign Exchange Office) will forward the ringing voltage to the desired destination (Extension) already determined by the Elastix Software. The FXO (Foreign Exchange Office) after receiving the ringing voltage (SIP INVITE message) will forward this message to the endpoint as programmed in Elastix software.
- The used card in this project will be programmed inside the software configuration files to match the project requirements and objectives and to match the server specification (see chapter 4 for detailed information).

Software

Elastix software is open source software with a WUI (web user interface) must be given an IP address (static IP address) from the local network of the server." Elastix system is assigned a dynamic IP address on your Network, which will last about 7-8 Days (normally). After that time period, it will request a new address. In most cases it is allocated the same address, unless you have a busy network, where in some cases it may allocate that address to another device". [11]

Elastix server will be setup with the required hardware (FXS/FXO card) and creating a profiles or extensions for the desired endpoints, this profile contains a username and a password and the server (Elastix) IP (domain name). These profiles represent registration, addressing and authentication of the endpoints at the server.

Because Elastix is open source software, we will need to program and modify some of it's files to meet our provided services requirements.

Client (endpoint)

The endpoint (mobile phone) will have SIP client software this software is available and a free cost on the phones stores (3CX phone is an example for a free software for Android and IOS). This kind of software will be configured to match the profile that was assigned in Elastix. If the endpoint was computer also there are a programs and softwares that support the SIP call and also can be configured to match the assigned profile in Elastix.

Note: the FXS/FXO output port can be connected to any desired extension to achieve more options like house control options (see chapter 3.7)

3.4 Through GSM

If the house owner doesn't have an access to the internet with his mobile phone or can't receive VoIP calls, we can reach him and initiate a call with him suing the GSM network.

The system design has two options to reach the owner through the GSM network; these options are shown in figure 3.3 in grey color, one option is a GSM module attached to the intercom block and the other option is a GSM modem after the server block.

3.4.1 Option 1

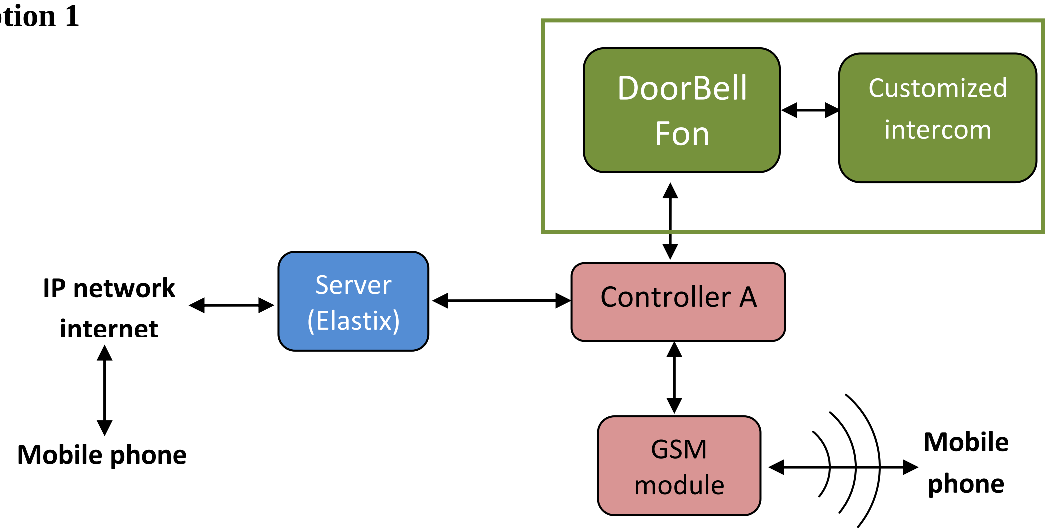


Figure 3.7: Through GSM (option 1)

After pushing the intercom call button the signal will go the controller; this controller will examine the status of the returned signal from the server (SIP RESPONSE) and decide if the GSM module will be activated and run a GSM call or there is an IP call and no need for the GSM call. This controller will be programmed to achieve this goal.

3.4.2 Option 2

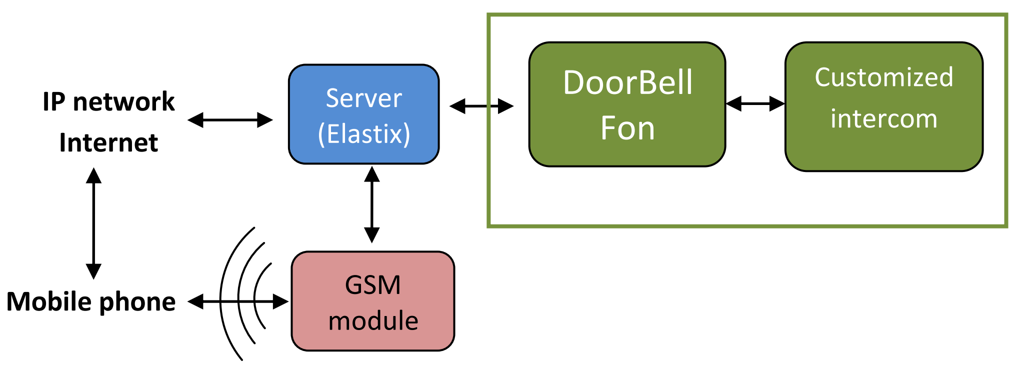


Figure 3.8: Through GSM (option 2)

Another option to reach the owner via GSM network is to attach the GSM module (described in figure 3.8) to the output of the server (FXO output port) and then the module will initiate a GSM call to the final destination if there was a SIP RESPONSE message from the mobile phone to the server indicate that the mobile phone is not responding.

Option 2 has advantages over option 1; these advantages are:

- Lower cost
- More efficiency
- High reliability
- High flexibility

The GSM module block shown in figure 3.8 is a GSM modem, controller and a matching circuit. The GSM modem accepts a SIM to subscribe to any operator exists. The Controller is to handle and initiate a GSM call to a saved phone number. The matching circuit is used to provide the interface between the server and the rest of the GSM module (see chapter 4). The following block diagram show the infrastructure of the GSM module.

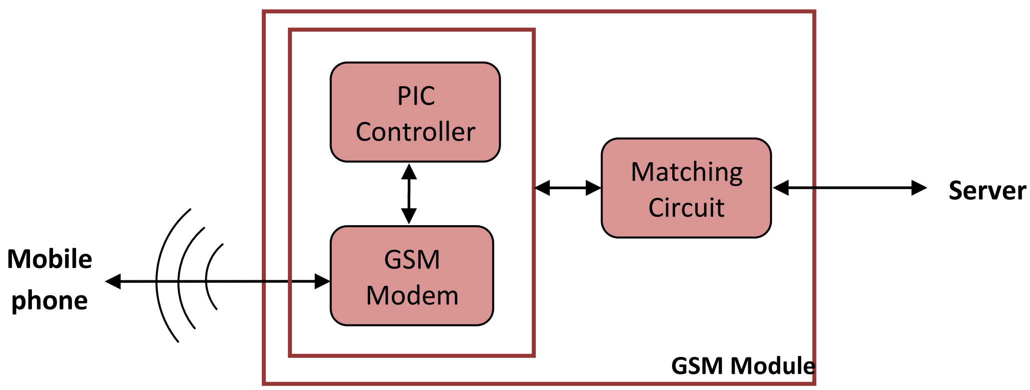


Figure 3.9: GSM module block diagram

The PIC controller (18F2580) will control the connection between the server block output (FXS/FXO card output) and will control the GSM call initiating and forwarding and the destination phone number to be reached and called.

The GSM modem (SIM 300 Module) accepts a SIM card, and operates over a subscription to a mobile operator, just like a mobile phone. From the mobile operator perspective, a GSM modem looks just like a mobile phone. This modem has the following features:

- Highly Reliable for 24x7 operation with Matched Antenna
- Status of Modem Indicated by LED
- Simple to Use & Low Cost
- Quad Band Modem supports all GSM operator SIM cards

The matching circuit -described later in chapter 4- is used for the following functions:

- Split the voice signal coming from the server (microphone and speaker).
- Provide signal to initiate the call.
- Keep the connection on until hang up by user.

3.5 Use-case diagram and scenario

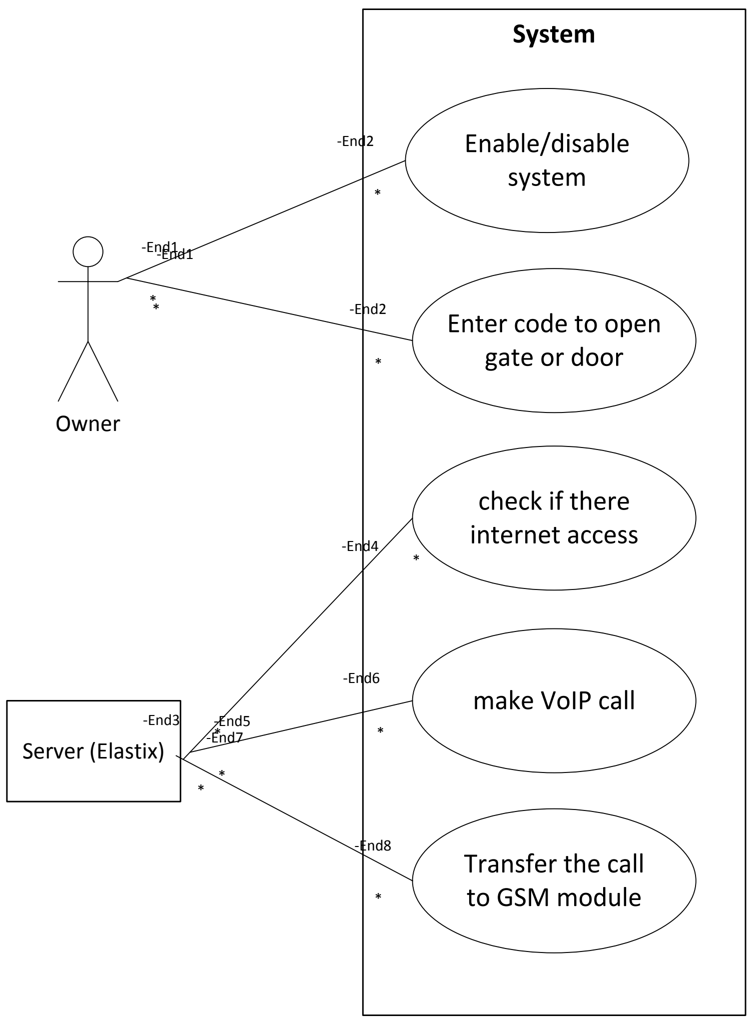


Figure 3.10: use-case diagram

Use-case scenario

- 〉 Use-case: Enable/Disable system

Primary actor: Owner

Goal in context: To enable system to answer intercom from end point (mobile) when the owner leaves the home with some of ambiguous, or disable system to answer intercom by traditional way from hand phone.

Precondition: turn switch on to enable system, turn switch off to disable system.

Trigger: the owner leaves the home and decides to answer intercom from his or her mobile.

Scenario:

1. Owner turn on the switch to answer the intercom from his mobile.
2. Owner turn off the switch to answer the intercom by traditional way.

Exception:

1. Owner leaves the home and forget to turn the switch on.
2. Power outages.

- 〉 Use-case: Enter code to open gate or door.

Primary actor: Owner

Goal in context: Enter code from owner's mobile to open gate or door when the owner away from home.

Trigger: the owner away from home and need to open the gate or door to someone known.

Scenario:

1. Owner away from home and system was enabled.
2. Owner knows who on the door and decides to open the door or gate for known one.

3. Owner sends code from his mobile to open the door or gate.

Exception:

1. Power outages.
2. Enter incorrect code.

- Use-case: check if there internet access or not

Primary actor: Server (Elastix software)

Goal in context: Check if there is internet access both in server and end point to make VoIP call, and if not to make GSM call.

Precondition: the system was enabled.

Trigger: the server receive signal from intercom means that the system is enabled, and owner will answer intercom from his mobile.

Scenario:

1. The system is enabled, the bell is ring.
2. Sends signal to server.
3. Server check if there internet access, make VoIP call.
4. Server check if no internet access sends signal to GSM module to make GSM call.

Exception:

1. Error in server or in Elastix software.
2. Power outages.

- Use-case: Make VoIP call

Primary actor: Server

Goal in context: To make VoIP call when system was enabled, and there is internet access in both server and end point (mobile).

Precondition: System was enabled, and the server check and found there is internet access.

Trigger: the owner enabled the system, have internet access on his mobile, and decide to answer the intercom from his mobile through internet.

Scenario:

1. server receive signal from intercom, means system is enabled.
2. Server check if there is internet access.
3. If yes, through FXS/FXO server make VoIP call.

Exception:

1. Power outages.
2. Internet connection interrupted even in server or end point.

- Use-case: Transfer the call to GSM module

Primary actor: Server

Goal in context: To transfer the call to GSM module when no internet access on server or end point.

Precondition: turn switch on to enable system, server found that no internet access.

Trigger: the server found there was no internet access, and decides to transfer the call to GSM module to make wireless call.

Scenario:

1. The server found no internet access.
2. Elastix software transfer call command from VoIP to GSM module.
3. GSM module make call to the mobile number(owner's number) saved in Pik- controller

Exception:

1. GSM network connection interrupted.
2. Power outages.
3. Error from server (or Elastix).

3.6 How System Works

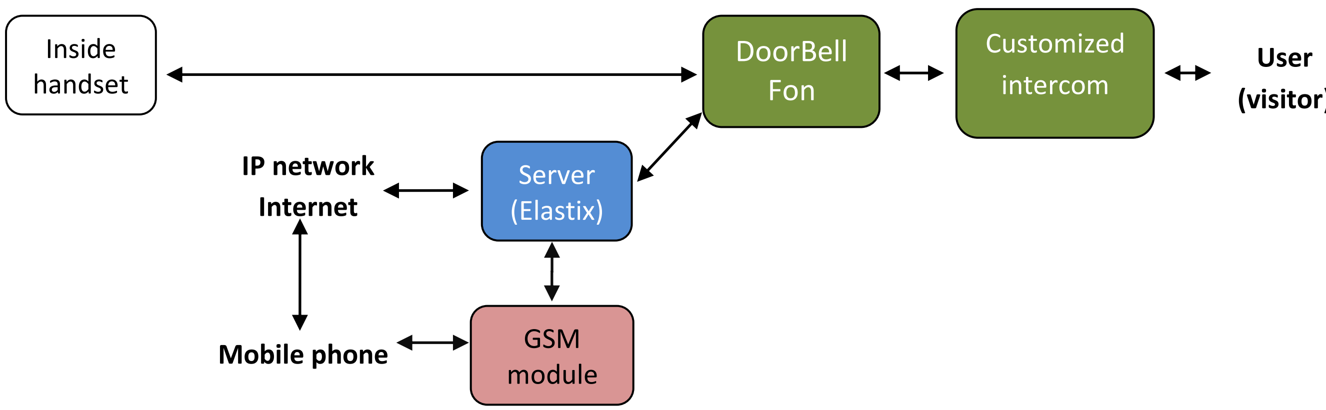


Figure 3.11: System block diagram

1) When the visitor come to the home and rings the bell of customized intercom.
2) The inside handsets will ring (through Elastix).
3) If no one answers or after a predetermined period of time the ringing signal will go to another extension(VoIP call).
4) The server will check if the end user has an internet access or not to do VoIP or GSM call.
5) If there is an internet access the server will decide to make a VoIP call.
6) If the end user does not have internet access in his or her mobile the server will make GSM call through GSM module.

3.7 Extra features

The flexibility and the availability of the internet can add extra features to this project after implementing the server and installing the Elastix software. One advantage of Elastix is that we can add many extensions (depends on the employed network, hardware and server specification) attached to FXS/FXO card as shown in figure 3.12.

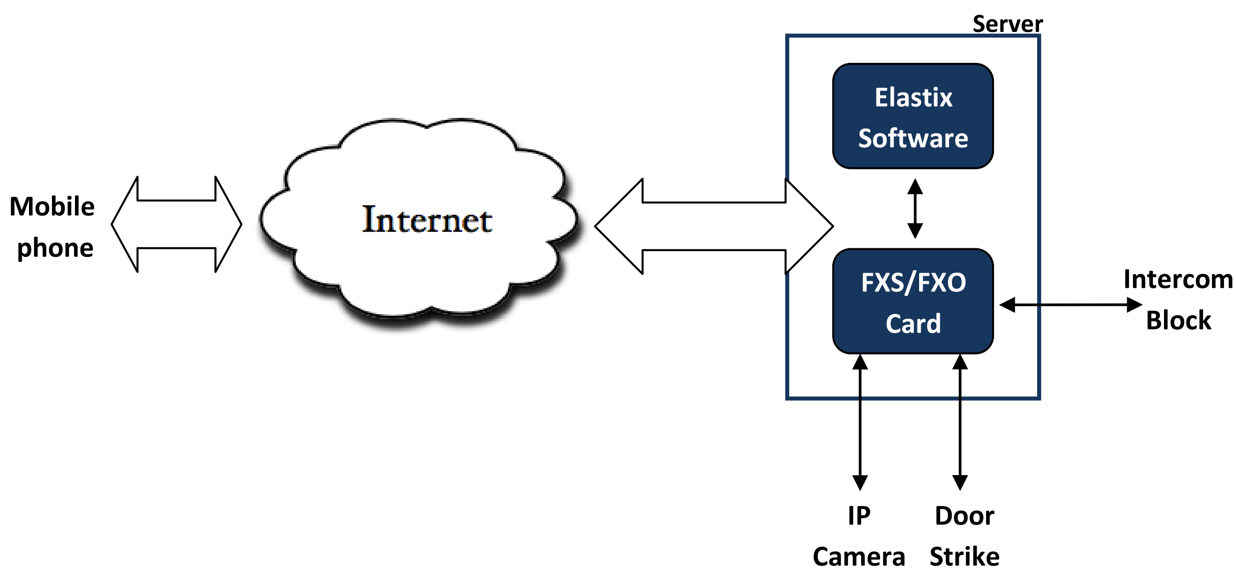


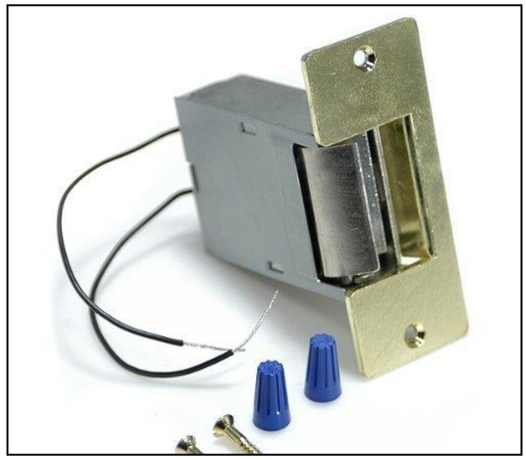
Figure 3.12: Extra features added

For example we can use IP camera as endpoint to reach it and make a connection (live view) with it from anywhere or an electronic door strike to remotely control any gate or door in houses. To achieve such features, first we need to program the Elastix software to meet our specification to such needed extensions.

VoIP calls gives us a REQUESTs and RESPONSEs when establishing a call between two endpoints, we can use these REQUESTs and RESPONSEs and employ the provided signals and voltages by them in these extensions.

From the property owner side, these extensions are endpoints must be configured with high level security to the server so no one reach them. The following two pictures show the IP camera and the electronic door strike (figure 3.13).

(a) IP camera (wireless or wired)



(b) Electronic door strike.

Figure 3.13: (a) IP camera and (b) Electronic strike

4

Chapter Four

Detailed System Design

4.1 Introduction

4.2 Elastix server.

4.3 Hardware design.

4.1 Introduction

In this chapter we will introduce the detailed system design of each part of the system independently; the system design will include 2 levels:

- **Level one**: Elastix server
  - Map our requirements
  - Plan server security
  - Set-up Elastix and Configure Elastix PBX.
  - FXS/FXO card installation
  - End point managing
- **Level two**: hardware design.
  - Intercom block
  - Through GSM (option 2)
  - Integration

4.2 Elastix server

4.2.1 Map our requirements

For level one we will start to map and determine our system requirements and compare the requirements with the facilities we already have like a LAN with internet access. For the purposes of our project we will need:

- 2 extension using Softphone (we will use these 2 extension for initial testing)
- 1 extension using mobile phone.
- 1 extension using GSM module.
- 1 extension using intercom block.

4.2.2 Plan server security

Once the above are set-up, we will plan the server security which it’s an important part and can’t be over stated. There are five major steps to achieve security:

1) **Don’t accept SIP authentication requests from all IP addresses**. Use the “permit=” and “deny=” lines in sip.conf to only allow a reasonable subset of IP addresses to reach each listed extension/user in sip.conf file.

2) **Set “alwaysauthreject=yes” in your sip.conf file**. Setting this to “yes” will reject bad authentication requests on valid usernames with the same rejection information as with invalid usernames.

3) **Use STRONG passwords for SIP entities**. This is probably the most important step can take.

4) **Block AMI manager ports**. Use “permit=” and “deny=” lines in manager.conf to reduce inbound connections to known hosts only.

5) **Make your SIP usernames different than your extensions**. While it is convenient to have extension “1234″ map to SIP entry “1234″ which is also SIP user “1234″, this is an easy target for attackers to guess SIP authentication names.

4.2.3 Set-up and configuring Elastix

For stability we will install the Elastix on a dedicated server rather than using a normal computer. The used version will be Elastix 1.0. (See appendix A for further installation steps).

After installing the software on the server, using browser, connect to ***https://ipaddress/*** (e.g. https://192.168.1.100) to be presented with the Elastix initial Admin web login screen as illustrated below.

Figure 4.1: Initial admin web login screen

If correctly logged in, the System Status screen will appear. This screen is the control centre. We need to configure Elastix to match our LAN specification just like the IP address for Elastix and the suitable subnet mask. The new IP address given to Elastix must be static instead of Dynamic Host Configuration Protocol (DHCP) so we have to change it using Elastix GUI as shown in figure 4.2.

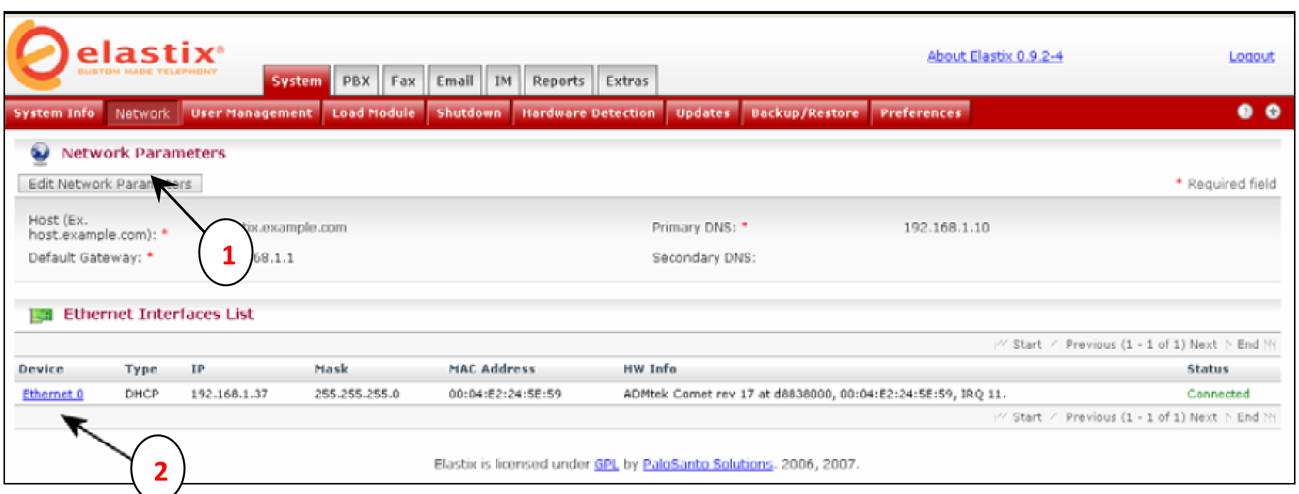


Figure 4.2: Network parameters screen

1- Selecting the Network Tab will display the following screen where you should select the network interface.
2- Here we have Ethernet card as a network interface; choose it to modify IP address, the following screen will appear.

Figure 4.3: Interface properties editing screen

This is where we will make the necessary changes:

1- Change the IP address to the one we want to use.
2- Click on the Static Radio Button.
3- Click on apply changes.

Click yes when the "Are you sure you want to apply changes" dialog box appears and we are done.

4.2.4 FXS/FXO card installation

FXS/FXO card will facilitate the connections and the calls between the home and the world outside and will be the interface between the system and any endpoint desired to be applied. Figure 4.4 shows this card and its modules attached on it.

The card we will use is the OpenVox A400E (figure 4.4) from OpenVox Communication Co.Ltd. OpenVox A400E delivers great voice quality in the telephony systems. With interchangeable FXS/FXO modules, it can eliminate the requirements for separate channel banks or access gateways. This card has advantages to use within this project:

1. It works with Elastix system in this project; Elastix Officially Certified.
2. High performance with low price.

3. Modular Design: Up to 4 FXS, FXO or mixed FXS/FXO ports per card. Each port can be set as FXO or FXS via plugging different modules in it
4. Supports PCI 2.2 or above,3.3 V and 5 V PCI slots

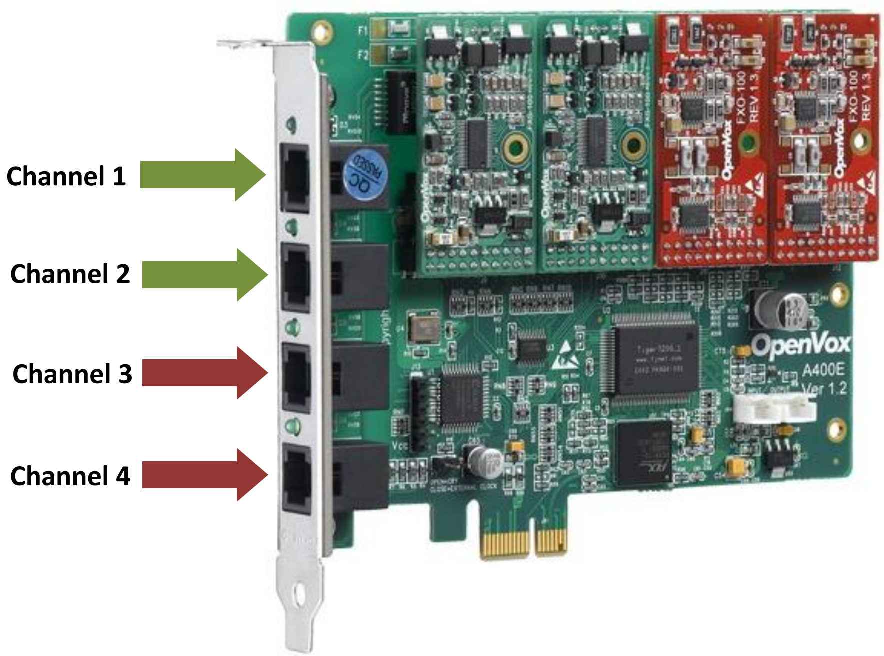


Figure 4.4: OpenVox A400E FXS/FXO Card

After attaching the card to the server PCI-E slot, we need to configure and program it to match our system (Elastix) to match the project requirement. We have to use DAHDI (Digium/Asterisk Hardware Device Interface) it’s an open source device interface technology used to control Digium and other legacy telephony interface cards.

Once the card is attached, the Elastix need to detect it like shown in the following picture

Figure 4.5: Hardware Detection menu

If the server detects the card, the following screen will appear. As shown bellow, the server detected the card with two FXS modules in service and two FXO modules not in service.

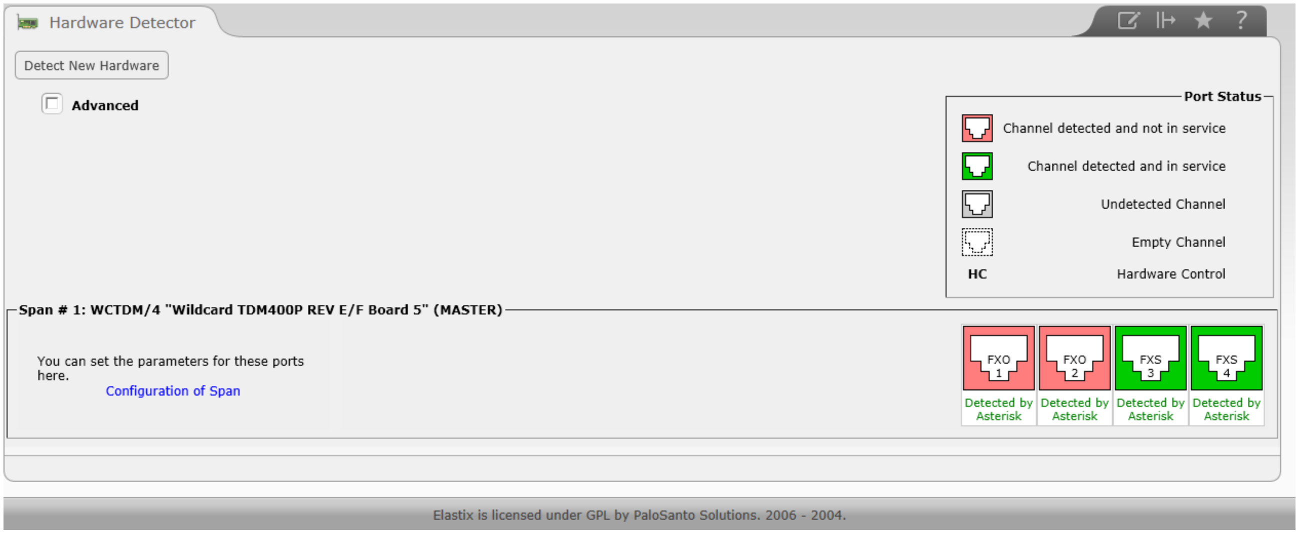


Figure 4.6: Hardware status menu

4.2.5 End point managing

After setting up the server, we need to add new extensions and profiles at the server for the end points. To create extensions, type of trunk e.g. SIP (for the mobile phone), ZAPtel (for the intercom extension), is done from the Create Extension menu illustrated below

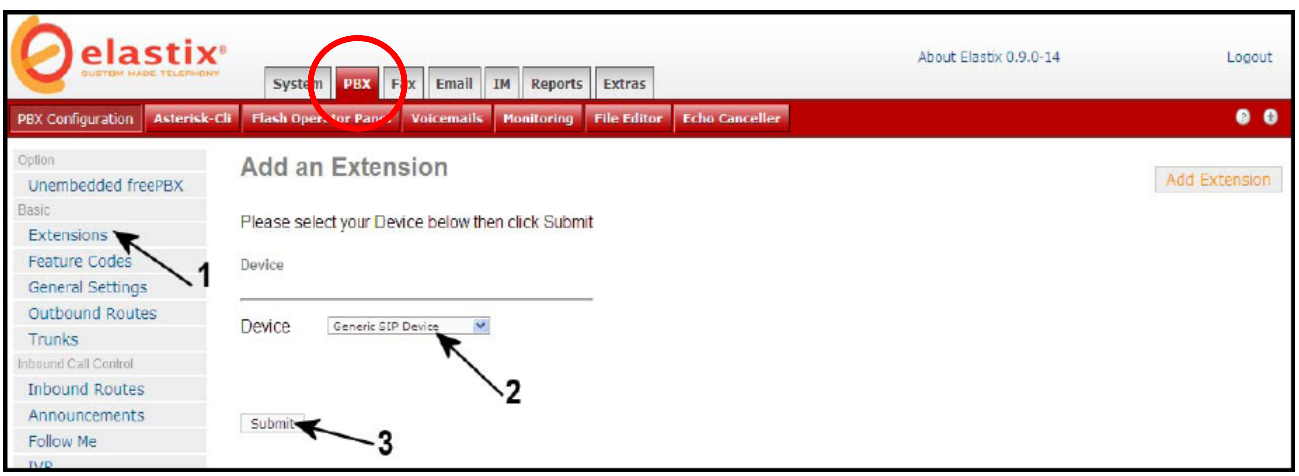


Figure 4.7: Create Extension menu

First of all we need to create two SIP profiles in order to test our server and do VoIP-to-VoIP experiment. Choose from device SIP and then submit as shown in the previous figure (figure 4.7).

Let's create two SIP extensions 2000 for mobile phone and 2001 for Softphone (software installed on a computer, Xlite). And one extension for testing the card and will be given ID 3000.

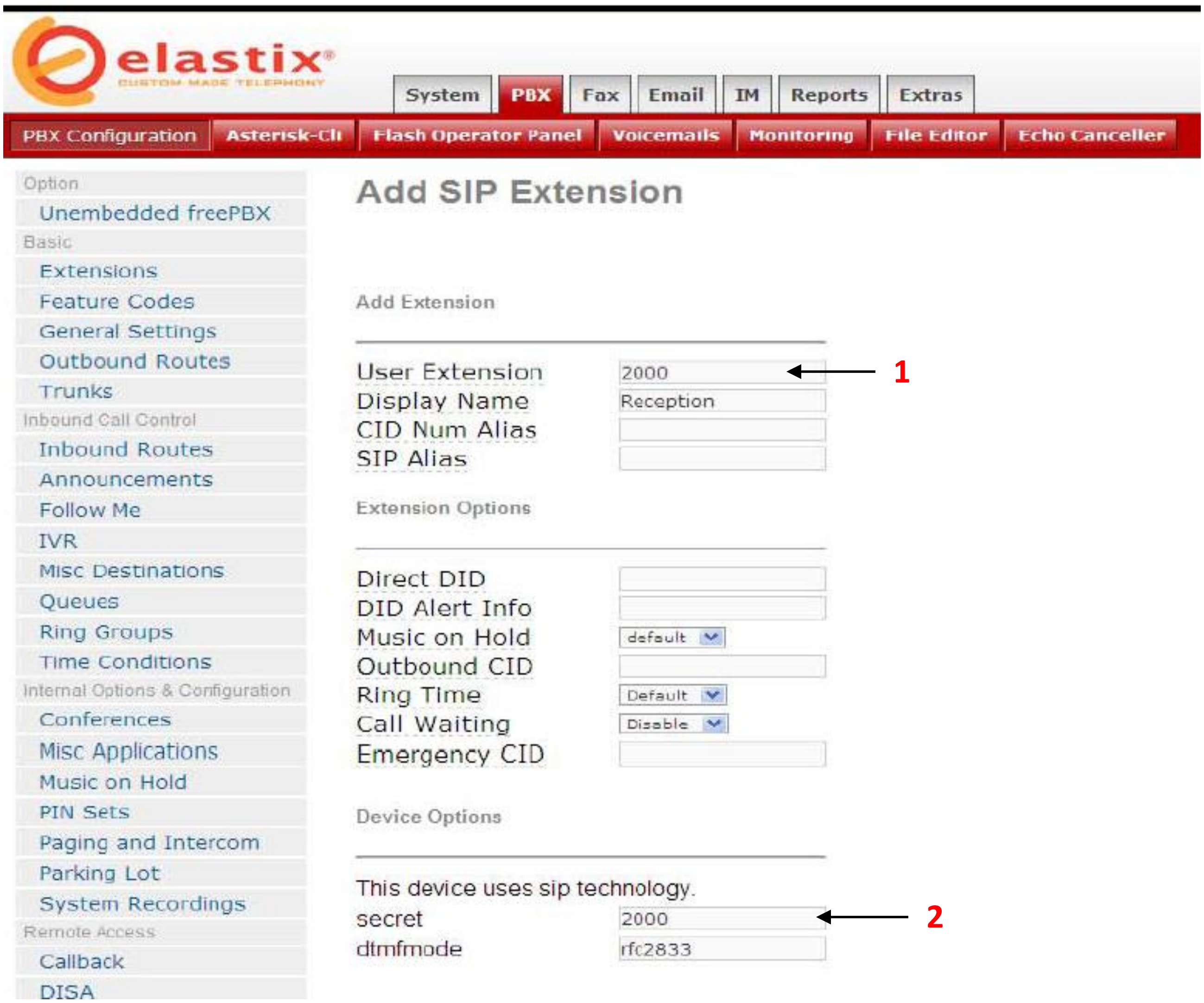


Figure 4.8: Add SIP Extension menu

This is done from the Add SIP Extension menu illustrated above in the figure 4.8. The two important parameters in the SIP extension are the User Extension and the secret. The profile 2001 is done as the same as 2000.

We will need to create another extension that is attached or connected to the server via FXS/FXO card in order to test the modules of the card so we can connect intercom to it. To create this extension, from figure 4.7 choose from the Device dropdown list Generic ZAP Device then submit.

The following menu (figure 4.9) will appear; when creating a ZAP extension the following must be done otherwise we run the risk of getting a one way audio only. This extension is given ID 3000.

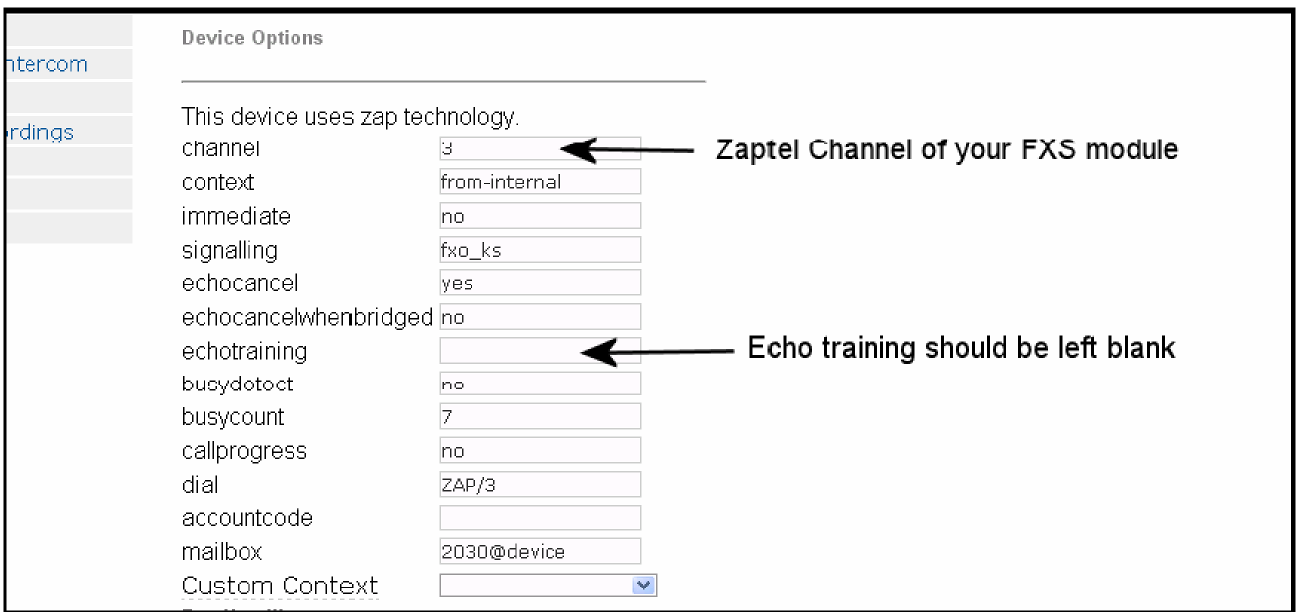


Figure 4.9: Generic ZAP Device menu

After creating the ZAP extension, we will do two experiments to test our server. These two experiments are VoIP-to-VoIP and VoIP-to-Analog phone.

Experiment No.1 VoIP-to-VoIP

The used mobile phone is Samsung Galaxy SII with android OS and the used software is 3CX. This software is available in Google store for free and supports SIP protocol. For different OS like IOS, it's also available for free in Apple store.

We need to assign the created profile on Elastix (2000) for the mobile phone as shown in the following figures 4.10.

The software home screen is shown in figure 4.10(a), to create an account choose profiles from settings, figure 4.10(b), then User and ID = 2000, Password = 2000 as the same as given in Elastix and Local PBX IP is the Elastix assigned IP.

Figure 4.10(a): Home Screen Figure 4.10(b): Settings Figure 4.10(c): Creating Account

Figure 4.10: 3CX software screens

After adding the profile information on the software, the mobile phone will automatically register to the server through the given IP (Local PBX IP) as shown in the following flowchart.

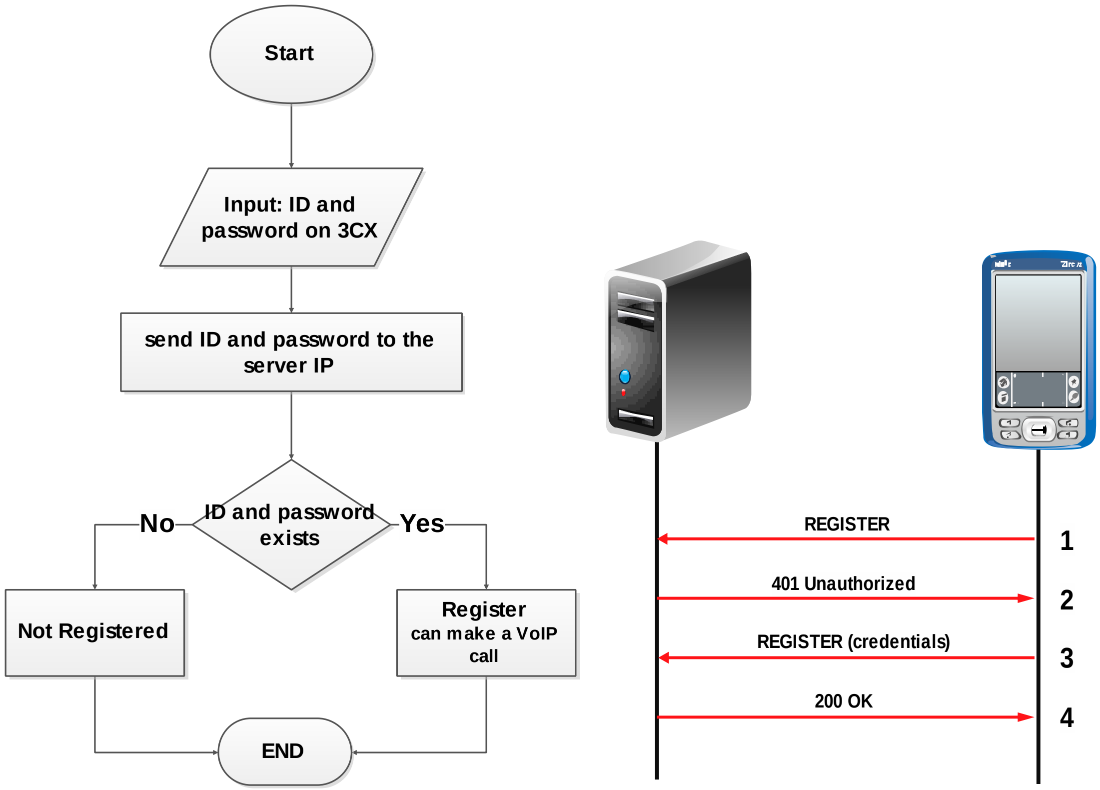


Figure 4.11: Profile authentication flowchart

When 3CX software then starts and assigned the created profile (2000), it needs authorization by the server. The following authentication flow shows the process.

1. The smart phone sends a SIP REGISTER message to local PBX IP.
2. The server returns 401 RESPONSE; this device is not registered at the server.
3. Then the mobile phone sends its credentials (ID and Password) to the server.
4. The server checks the credentials with the one saved in database; if it is registered, it sends OK RESPONSE.

The next step is to assign the profile 2001 to a Softphone. There are a number of Softphone available for use with Elastix. Some of them are free like Xlite.

When you start X-Lite 3 for the first time, the following screen shown in figure 4.12(a) will appear.

Figure 4.12(a): Xlite software

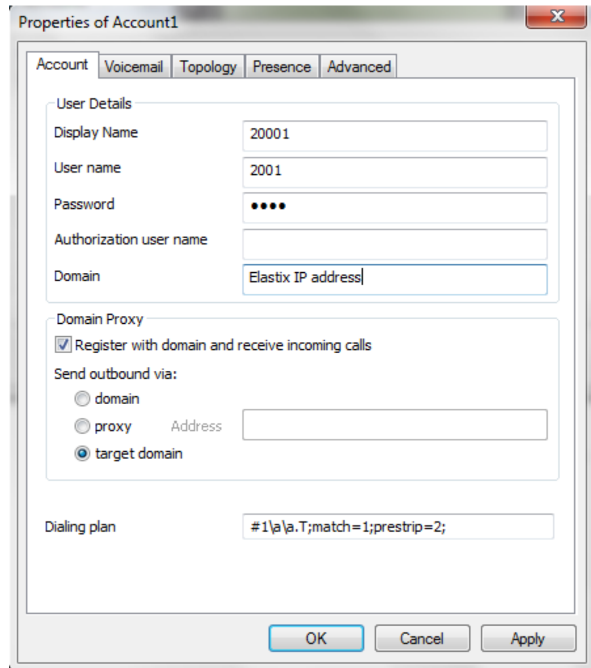


Figure 4.12(b): Profile creation screen

Figure 4.12: Xlite software

Click the Add button to get the screen in figure 4.12(b). This is where we will enter the credential (ID and password). At the various fields, add the following:

- **User Name:** 2001
- **Password:** 2001
- **Authorization User name:** 2001
- **Domain:** Elastix IP address

We are now ready to use the X-Lite Softphone after the authentication described in Figure 4.10: Profile authentication flowchart.

After assigning the profiles 2000 and 2001 for the mobile phone and the Softphone, we are ready now to make a VoIP-to-VoIP call which is done correctly.

Experiment No.2 VoIP-to-analog phone

For this experiment we need to connect an analog phone to the card (FXS module) to channel 3. This channel is assigned for a ZAP extension as shown in figure 4.8. This means that any analog phone connected to this channel is an extension has ID 3000.

When calling 3000 from the mobile phone or the Softphone, the phones ring, so our modules in service. Even the analog phone can call any extension predetermined in the server.

4.3 Hardware design

4.3.1 Intercom block

The intercom block described in chapter 3 shown in figure 4.13 has two main components; the customized intercom, DoorBell Fon.

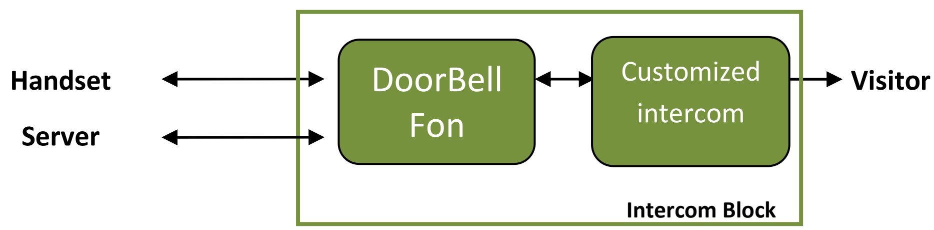


Figure 4.13: Intercom block

The door panel is the customized intercom located at the front door and the DoorBell Fon is the controller and the interface between the box and server. These parts are shown bellow.

Figure 4.14: DoorBell Fon

Figure 4.15: Door box

The DoorBell Fon controller (figure 4.14) is the brain of this intercom system. The unit is powered with standard AC electric current and connects to the door box via a pair of wires or a single pair of telephone wire.

The controller is designed to connect between the public telephone system and telephones within home. To accommodate this purpose, it has two RJ14 jacks, one that connects to the telephone line coming into home and the other that connects to a handset, or even multiple handsets, within home. In addition to the two jacks, there is a wire connection panel that can accommodate two different doorbells. Each doorbell is connected via a negative and a positive lead to the controller.

Back to the server to create the needed queue; first we need to know which port (channel) of the card is assigned to receive a voltage signal, i.e. which port is designed to accept the PSTN line (a voltage signal). This is done by going to Elastix command line to execute channels (ports) status. From the command **show dahdi channels** the following will appear

Chan	Extension	Context	Language	MusinOnHold
Pseudo				
1	Fxo	from-internal	En	
2	Fxo	from-pstn	En	
3	Fxs	from-internal	En	
4	Fxs	from-internal	En	

In the configuration above, Chan (channel) 2 has a context of **’from-pstn’**. Since we are not going to use this FXO port for standard phone calls (from-pstn) but rather for doorbell a new context needs to be created for it. We need to create a context called **from-doorbell-custom** in the **extentions_custom.conf** file. This context needs to be used by the WildCard that is in channel 2.

Before that we need to change the channel 2 status to **from-doorbell-custom**. This is done in the file ***chan_dahdi.conf.*** Change the context from **from-pstn** to **from-doorbell-custom** as shown bellow.

```
; Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1" (MASTER)
;;; line="1 WCTDM/0/0"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 1
context=default

;;; line="2 WCTDM/0/1"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-zaptel ; change to  from-doorbell-custom
channel => 2
context=default
    ;;; line="3 WCTDM/0/2"
```

Now we need to create the Now create the context in the ***extentions_custom.conf*** file.

```
    [from-doorbell-custom]
    exten => s,1,Answer()
    exten => s,2,Wait(1)
    exten => s,3,SetCIDName(Doorbell)
    exten => s,4,Queue(859|t|||60) ; Doorbell
```

The context does the following: First, it answers the doorbell. Then it sets the caller ID to ‘Doorbell’. And finally, it puts the call into a queue that will be defined then.

We will use in this project analog phone instead of the ordinary intercom handset for the following reasons:

- It's easy to integrate with Elastix
- Cost-less
- Can be used as end-point of Elastix server (VoIP client can make and receive VoIP call)

So it need a profile to use, we created a ZAP extension for it in Elastix. The profile is assigned for the DoorBell Fon thus the inside telephones.

4.3.2 Through GSM (option 2)

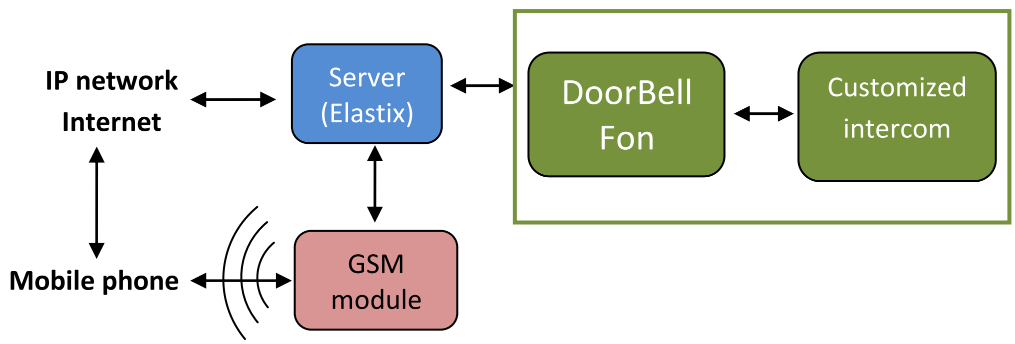


Figure 4.16: Through GSM (option 2)

The block diagram shown above-which described in chapter 3- is how to reach the owner through GSM network if he did not have an access to the internet. The detailed block diagram for the GSM module is shown bellow.

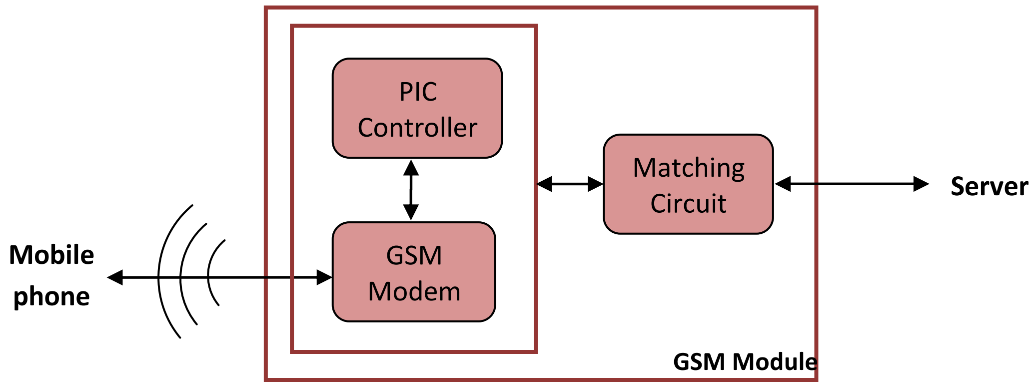


Figure 4.17: GSM module block diagram

We used this circuit and configuration because of the requirements of the project; we need to program the GSM modem using PIC to match the server and the Elastix. To reach the user using GSM, we need a specific FXS/FXO card that supports GSM modules which is high cost and cannot be used with the free Elastix versions, so we designed a matching circuit with a programmable GSM modem for the purpose.

Matching circuit

We build the matching circuit (shown in figure 4.20) because we cannot connect the module directly with the server because we need a ringing voltage and a circuit to keep the connection up between the server and the user. So we design it based on the idea of the simple analog phone system.

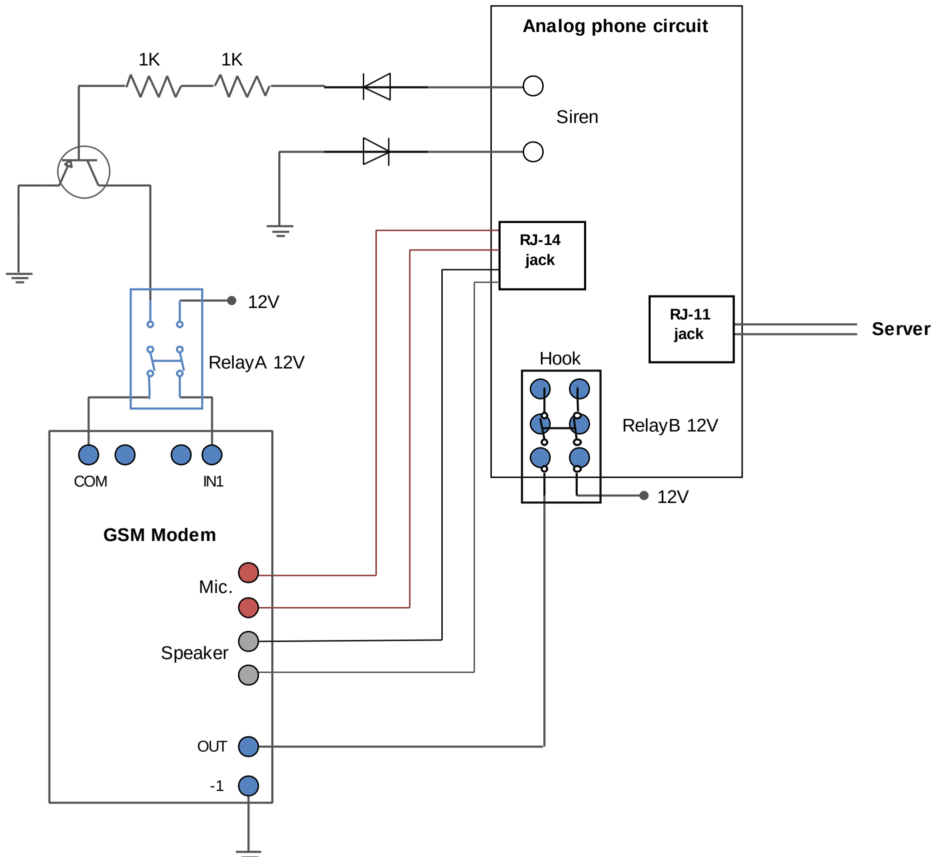


Figure 4.18: Matching circuit

The analog phone system provides 4 lines; two for the microphone and two for the speaker, which is connected directly to the GSM modem. We used the siren circuit of the phone with some modifications to activate the ringing voltage for the modem.

The adaptation consists of two diodes, 2K ohm resistor, transistor and 12v relay (RelayA). The two diodes for rectifying the analog signal comes out from the siren, the resistor to protect the transistor from high voltage, the transistor is used to activate RelayA.

When the signal (12v signal) comes from the server the siren circuit will work and the signal will go to RelayA through the diode and the resistor and the transistor; the relay will activate the GSM modem and the signal will initiate the call.

If the owner answers this GSM call, a signal will comes out from port OUT. This signal will turn on RelayB which is used close the hook up circuit on the analog phone circuit.

The analog phone circuit is then connected to the server (to FXS channel) so it needs a profile in Elastix. The GSM module will act like any extension in Elastix.

The following picture shows this circuit.

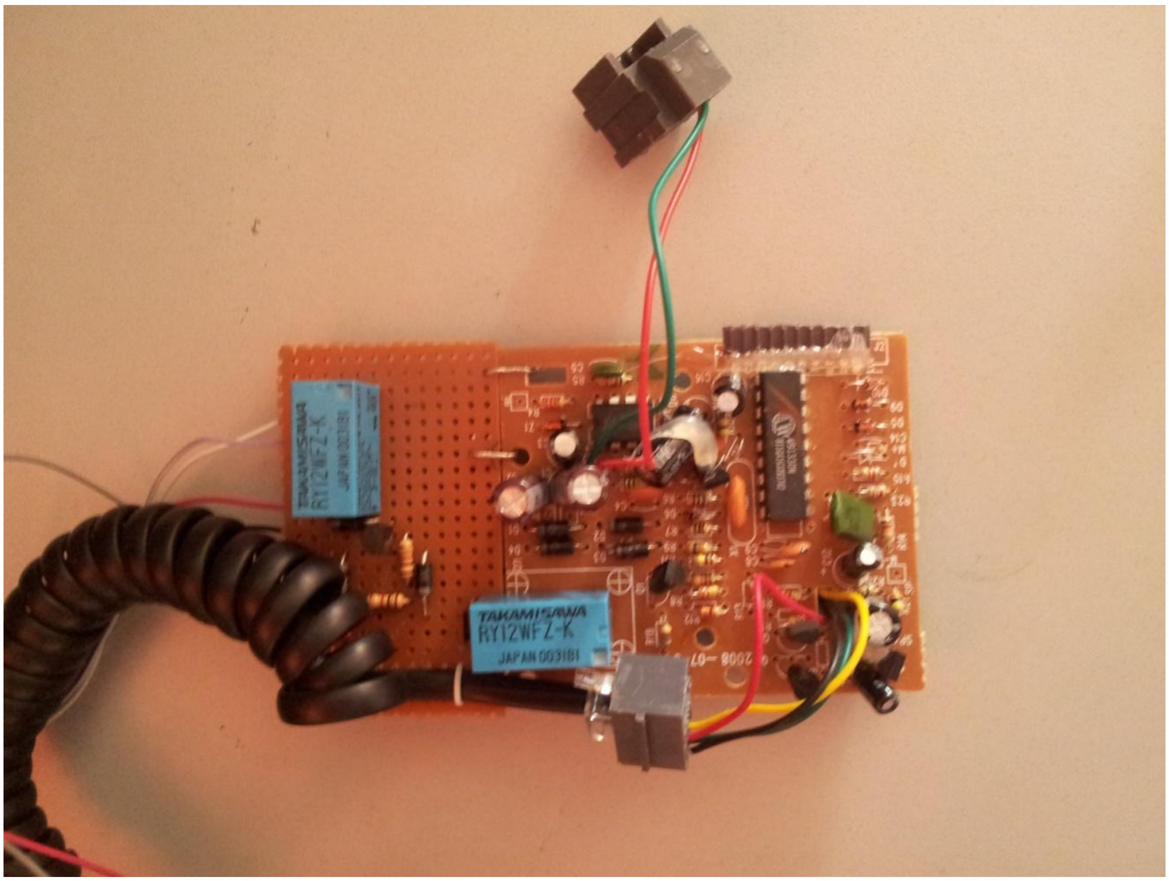


Figure 4.19: Matching circuit picture

PIC controller

Is the brain of the GSM module, the used PIC is 18F2580 shown in figure 4.22; it is used to initiate the connection on the GSM modem after receiving the ringing signal from the matching circuit. This signal will initiate a call to a saved number inside it.

It also programmed to return the credit value on the SIM and to change the used number inside it by sending SMS with a specific content to the GSM modem.

For security reason, we have programmed the PIC so no one can change the saved number by sending SMS with certain content and a password.

The code is in appendix B.

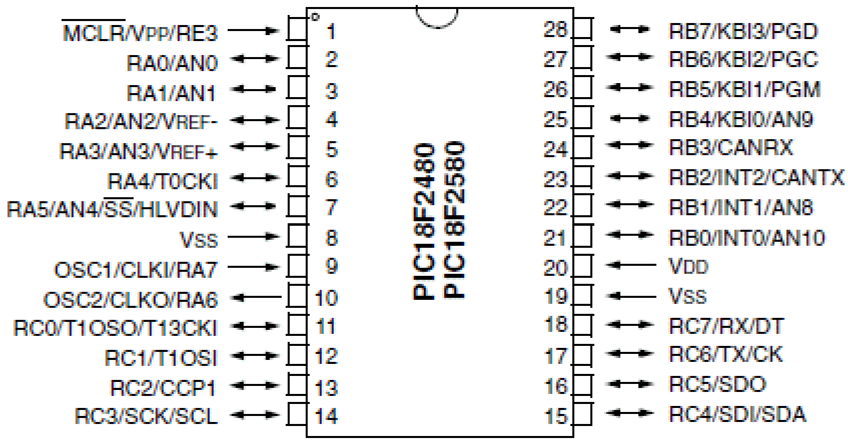


Figure 4.20: PIC18F2580 controller

GSM Modem

This GSM Modem (SIM 300) shown in figure 4.23 can accept any GSM network operator SIM card and act just like a mobile phone with its own unique phone number. Advantage of using this modem will be that we can use its RS232 port to communicate and develop embedded applications. Applications like SMS Control, data transfer, remote control and logging can be developed easily.

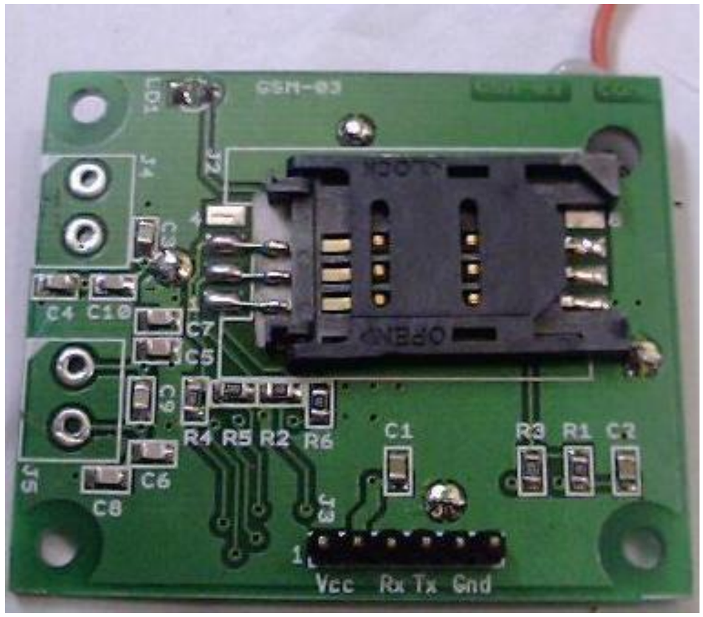


Figure 4.21: Modem (SIM 300)

The modem can be connected to any microcontroller. It can be used to send and receive SMS or make/receive voice calls when it is connected to the controller.

The following schematic diagram (figure 4.23) shows the connections between the PIC and the GSM modem. The power supply of SIM300 has to be a single voltage source of VBAT= 3.4V...4.5V. It must be able to provide sufficient current in a transmit burst which typically rises to 2A.mostly, the VBAT 8 pins are voltage input.

The microphone lines comes out from the analog phone circuit has to be connected to the SIM 300 MIC1P and MIC1N pins, and the speaker lines to SPK1P and SPK1N pins.

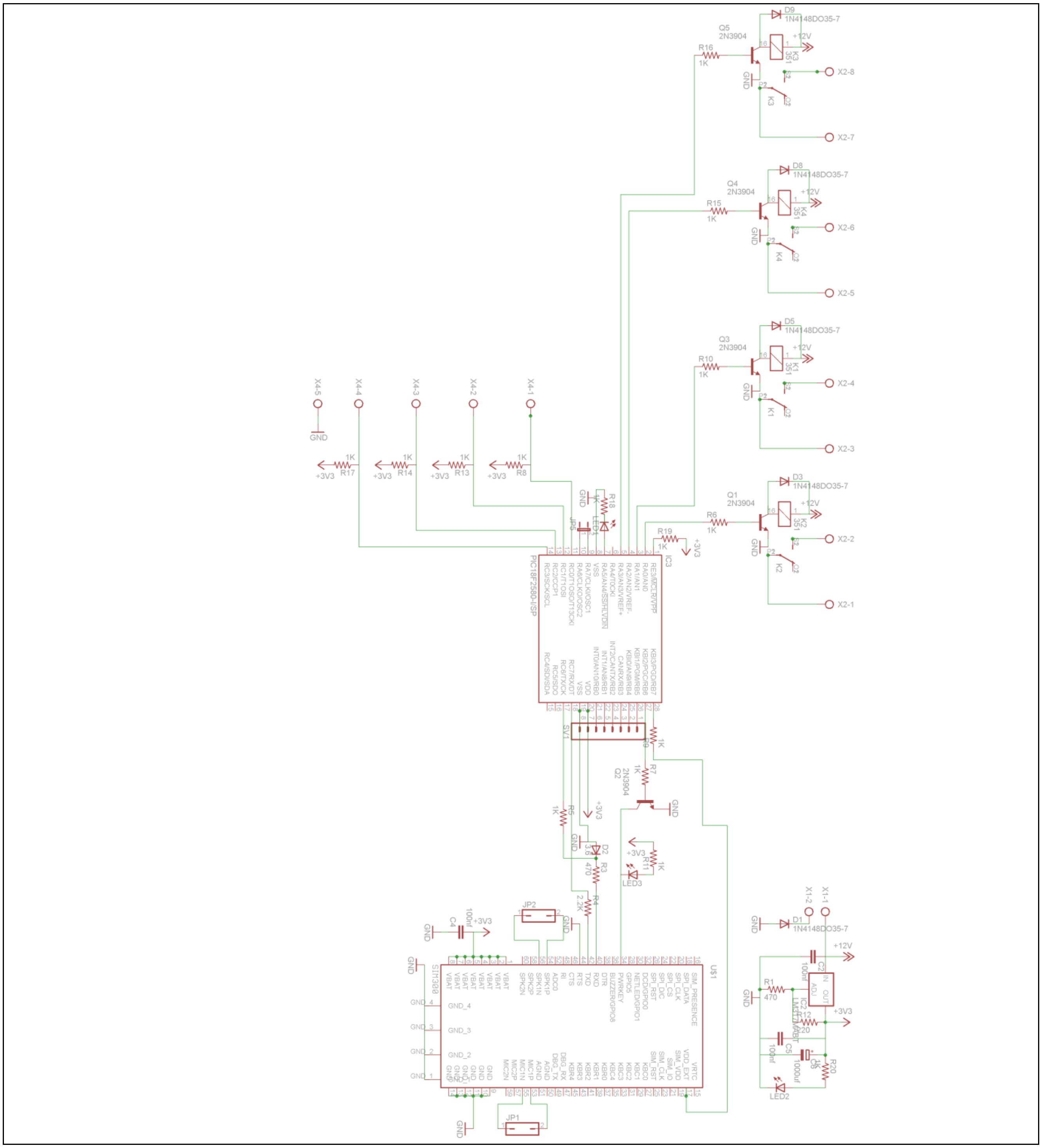


Figure 4.22: Modem (SIM 300) schematic diagram

The data receive pin RXD of the modem is connected to the data transmit pin at the PIC (RC6/TX/CK) and the data transmit pin of the modem (TXD) is connected to the data receive pin at the PIC (RC7/RX/DT). This is used for sending and receiving data between the modem and the PIC such as changing the saved number inside the PIC or return the value of the credit of the used SIM card.

For example, to change the number saved inside the PIC; first send SMS with content "1234-all" to erase the saved number, then send SMS with "1234+ new number" to save the new number. 1234 is the password of the PIC.

The following picture shows the GSM modem with the PIC controller

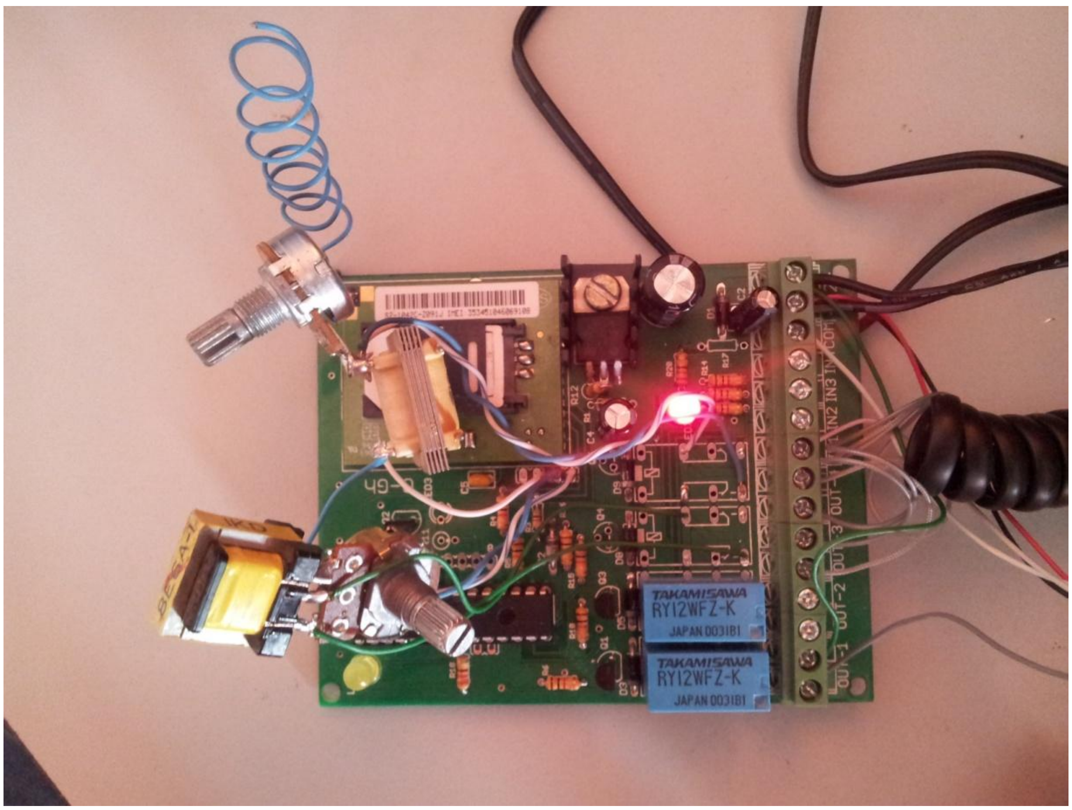


Figure 4.23: GSM modem with PIC controller

4.3.3 Integration

In this section we will connect the all blocks together using the needed interfaces. Figure 4.26 shows the final integration of all blocks and parts.

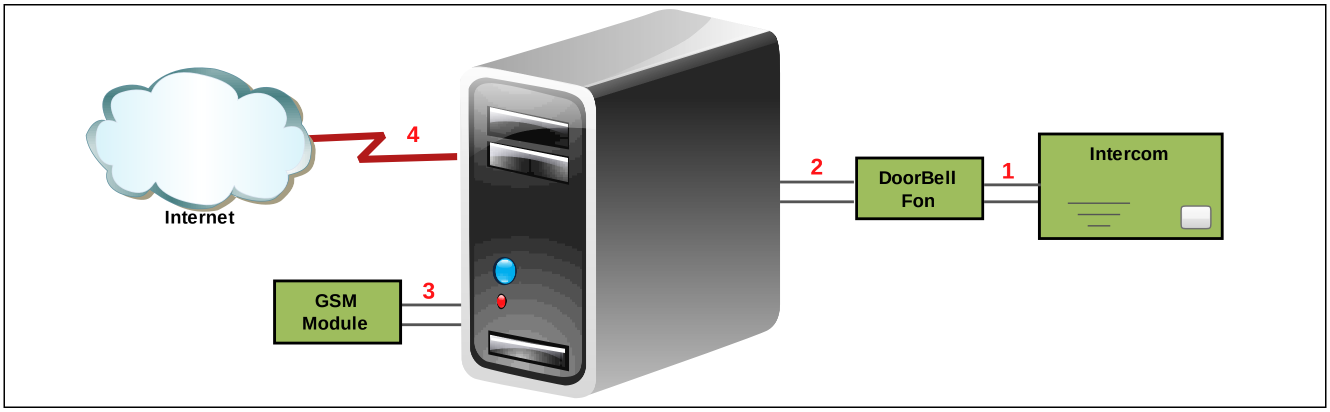


Figure 4.24: System Connections

1. The customized intercom is connected to the DoorBell Fon via two coaxial wires.
2. The DoorBell Fon then connected to the server (FXO channel on the card) via RJ11 cable with 2 RJ11 clips at either ends.
3. The GSM module then connected to the server (FXS channel on the card) via RJ11 cable.
4. The server must have internet access.
5. End-point is a smart phone.

5

Chapter Five

Testing and Monitoring

5.1 Introduction

Testing the operation of each block individually and all connected and finally monitor the calls made by Elastix and see the results.

5.2 Testing Blocks

In this section the different blocks (intercom, server and GSM module) will be tested alone before connecting them together.

5.2.1 Intercom block

Connecting the door box with the DoorBell Fon and connecting the handset to the DoorBell phone- as shown in chapter 4- is worked properly. I.e. pushing the door box call button will make the phones inside the house and the inside handset ring.

To test the connection between the DoorBell Fon and the server, the Elastix server has to detect the used FXO module (from Hardware Detection) and put it in service as shown bellow.

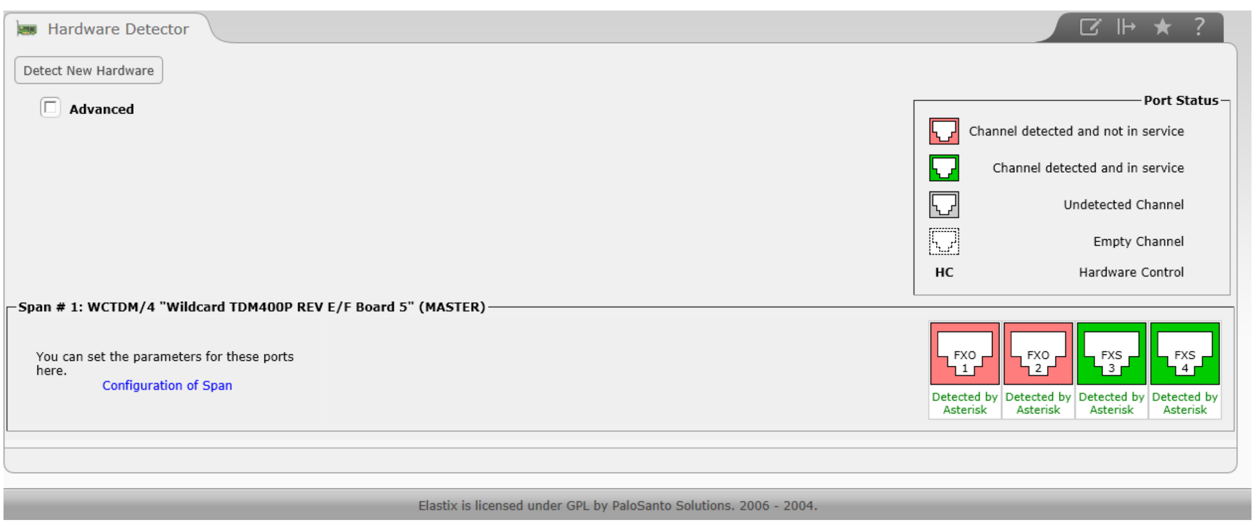


Figure 5.1: Hardware Detecting screen with FXO not in service

Before connecting the DoorBell Fon with the server, the previous figure shows that FXO on channel two is detected but not in service.

The following figure shows that FXO is detected and in service after connecting the DoorBell Fon with the server.

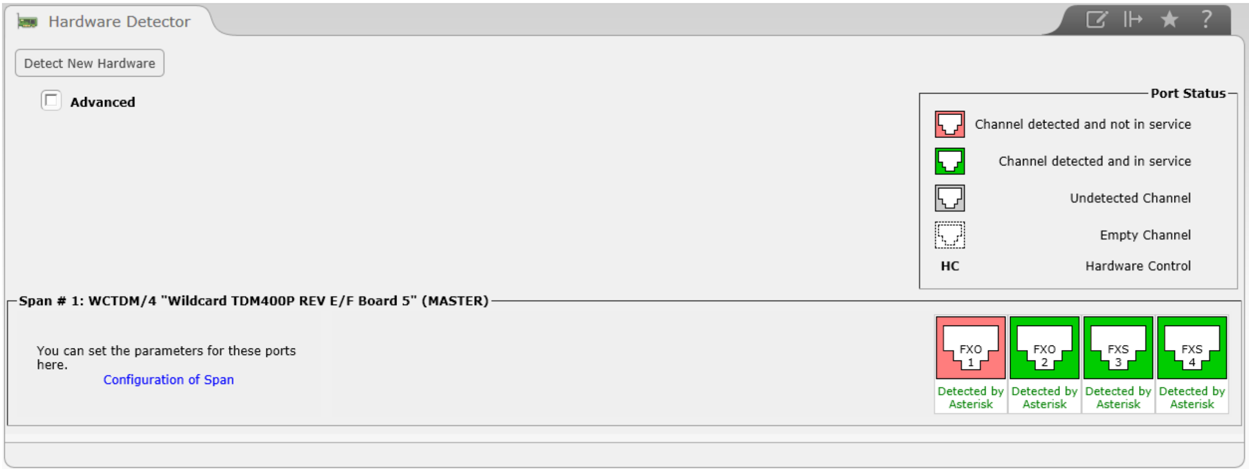


Figure 5.2: Hardware Detecting screen with FXO in service

5.2.2 GSM module

To test the module with its matching circuit, first we programmed the PIC controller for GSM call initiating and connected it to the modem, and then a PSTN line connected to the analog phone circuit in the matching circuit.

The idea from connecting the PSTN land line is that if anyone calls that PSTN number, the matching circuit should activate and initiate GSM call to a saved number in the PIC. This is worked properly. So it's ready to be connected to the server.

5.2.3 Server

This step will test the Elastix server with its features and needed functions and the VoIP calls. And manage the needed extensions in this project.

First of all, we need to create two extensions one for the mobile phone (owner) and another for the GSM module. The mobile phone extension is SIP extension with ID = 222 and the GSM module extension is ZAP extension (generic ZAP device) with ID = 333 and FXS channel = 4.

Another extension is needed for the inside handset, ZAP extension with ID = 444 and FXS channel = 3.

The following table shows the created profiles in this project.

Extension ID	Display Name	Type	Channel No.
111	Laptop	SIP	-
222	Mobile Phone	SIP	-
333	GSM	ZAP	4
444	Inside Telephones	ZAP	3

Table 5.1: Profiles list

The intercom block is connected to FXO channel (channel = 2) and it needs a queue to forward the ring to. This queue has the inside telephones profile to call (static agents = 444), as shown in chapter 4.

After creating the queue, extension 444 must have a follow me settings so that if no one answer the inside telephone or there is no one at home, the extension 444 will followed to extension 222 (owner profile).

The extension 222 also needs to have follow-me settings (if there is no internet access) inside Elastix, these settings is done as illustrated bellow. Go to extension 222 then edit follow me settings. (the same is done for extension 444)

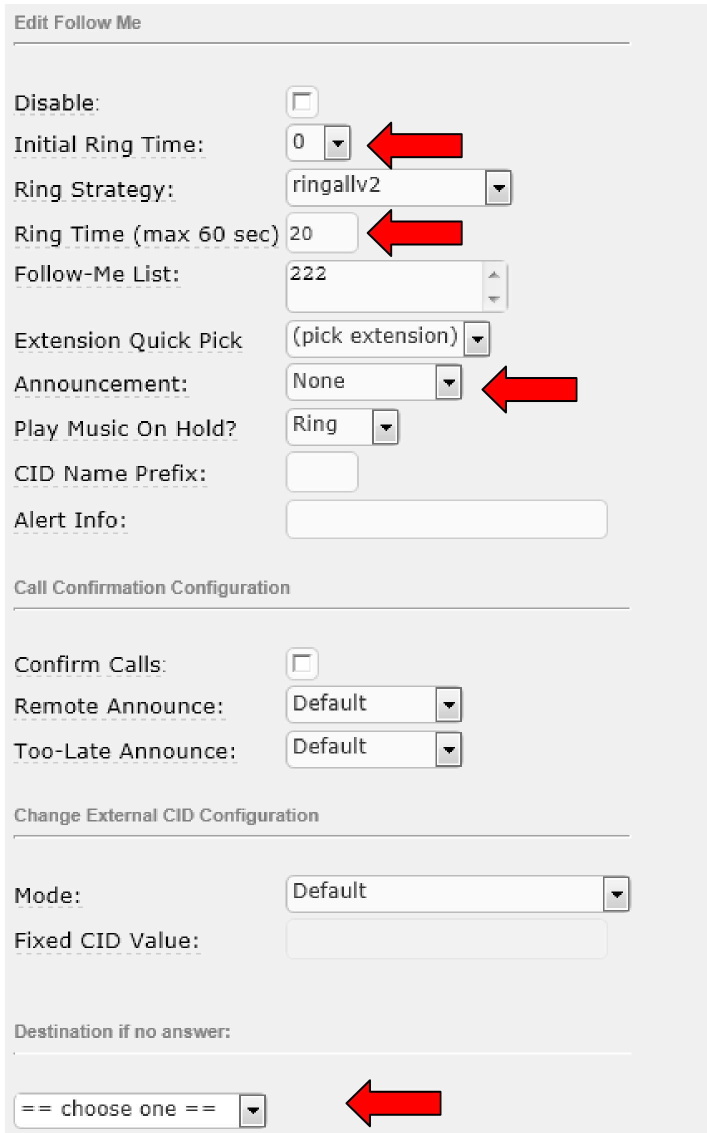


Figure 5.3: Follow-me settings

The **initial ring time** for how much rings the extension 222 will ring before forwarding the call (it depends on the user, in this project it will assigned to 0).

Ring time for how much time the extension 222 will ring before forwarding the call to the desired extension. (Also it depends on the user, but in this project it will be assigned for 30 sec).

The **announcement** is what the visitor will listen to after pushing the intercom call button.

Destination will be extension 333 (GSM module).

So the follow me process will act as shown in the figure 5.3 bellow.

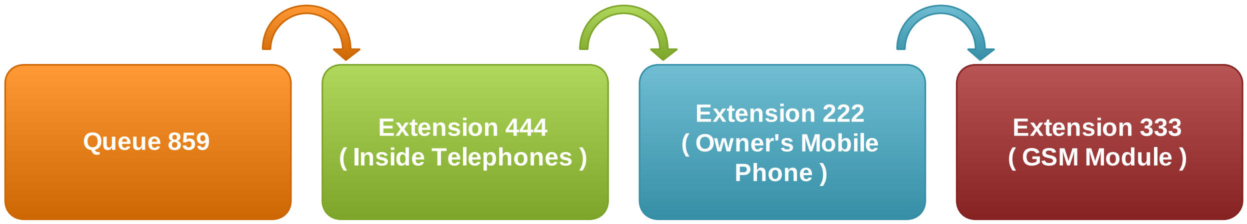


Figure 5.4: Follow-Me process

The Queue 859 will call extension 444, if no one answers or after a predetermined period of time, the call will forward to extension 222 as a VoIP call. If there isn't internet access at the owner's mobile phone, the call will forward to extension 333 and make a GSM call.

Scenario 1

When a visitor push the call button on the door box, the inside handset will ring for two rings, then the call will placed in the queue inside Elastix, this queue will forward the call to extension 222 (owner) and if he has internet access, his mobile will receive the VoIP call.

The process after placing the call in queue is shown bellow.

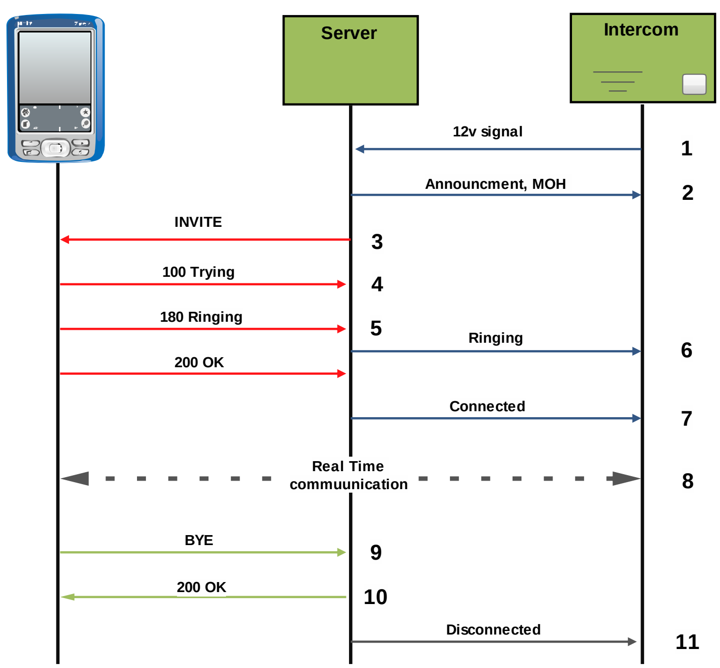


Figure 5.5: Scenario 1 call flow

1. The visitor push the call button and a 12v signal goes to FXO port at the server
2. The server will return an announcement "welcome to Palestine Polytechnic University" then music on hold.
3. Meanwhile the server will follow the queue (859) and forward the call from FXO port number 2 to the desired extension (static agent in queue) which is the extension 222 and send SIP INVITE message to it.
4. The mobile will immediately reply with the provisional response "100 Trying". This indicates that the request has been received (and thus the client does not need to retransmit it) and that it is being processed.
5. If it's registered and authorized to the server, it will send 180 Ringing.
6. The ringing signal is then delivered to the intercom.
7. And the connection has been made.
8. Real time communication.
9. Mobile phone terminated the call.
10. The server response is OK.
11. The server disconnects the line to intercom.

- **Scenario 2**

When a visitor push the call button and the call goes to extension 222; If the owner does not has internet access on his mobile, i.e. the authentication of its assigned profile (222) to Elastix had failed, the follow me setting will play the rule and place the call to the chosen destination (333) in the follow-me settings for 222.

The ringing signal then is placed in the GSM module and initiates a GSM call to the owner number saved in it.

The following call flow describes the scenario in details.

1. The visitor push the call button and a 12v signal goes to FXO port at the server.
2. The server will return an announcement "welcome to Palestine Polytechnic University" then music on hold.

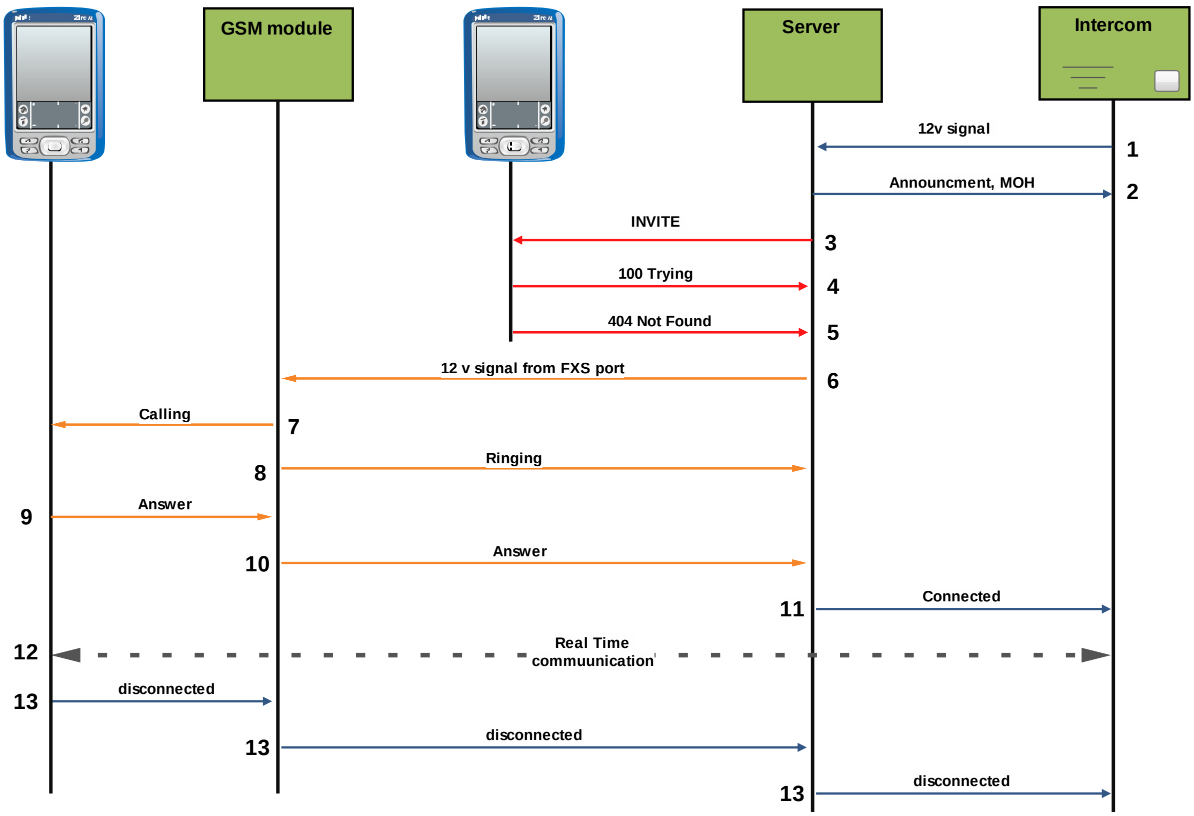


Figure 5.6: Scenario 2 call flow

3. Meanwhile the server will follow the queue (859) and forward the call from FXO port number 2 to the desired extension (static agent in queue) which is the extension 222 and send SIP INVITE message to it.
4. The mobile will immediately reply with the provisional response "100 Trying". This indicates that the request has been received (and thus the client does not need to retransmit it) and that it is being processed.
5. Because the mobile doesn’t registered to the server, it will send 404 RESPONSE; this means that the server didn’t find the client.
6. The server will forward the call to extension 333 through FXS channel number = 4 and a 12v signal will go to the GSM module and activate a GSM call.
7. The GSM module will call the owner number.
8. Ringing signal.
9. User answers the call.
10. The answer goes to the server.
11. Then to the intercom.
12. Real time communication.
13. Disconnect the call.

5.3 Results and Monitoring

Elastix provides a calls monitor function that can monitor all calls done by Elastix with the date and the length of the call. See the following calls report.

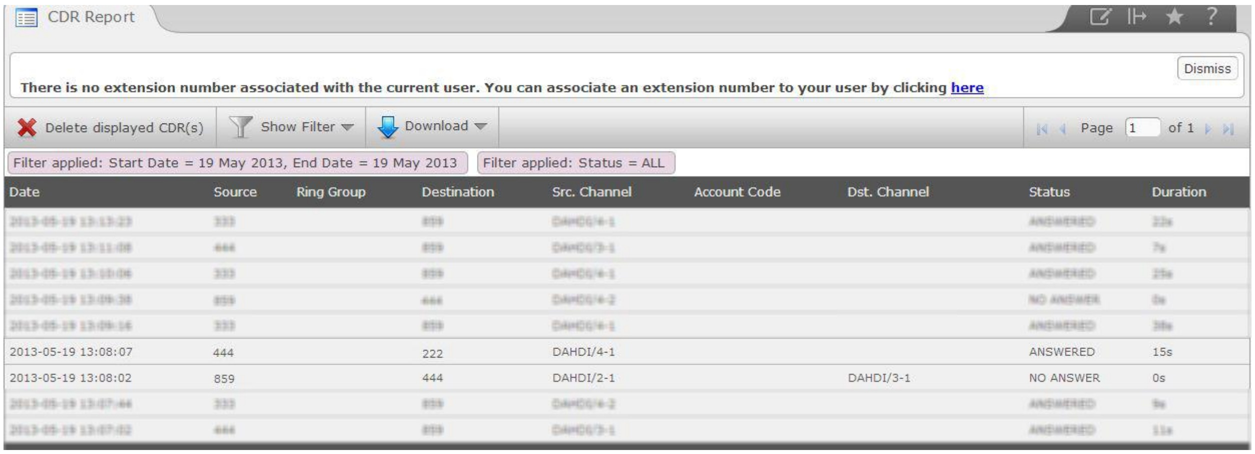


Figure 5.7: Elastix CDR report

This report shows that there was a call from 859 to 44 at 13:08:02 and no answer, after 5 seconds the call rerouted to destination 222 and the call was answered. This illustrate scenario 1.

The following calls reports illustrate scenario 2.

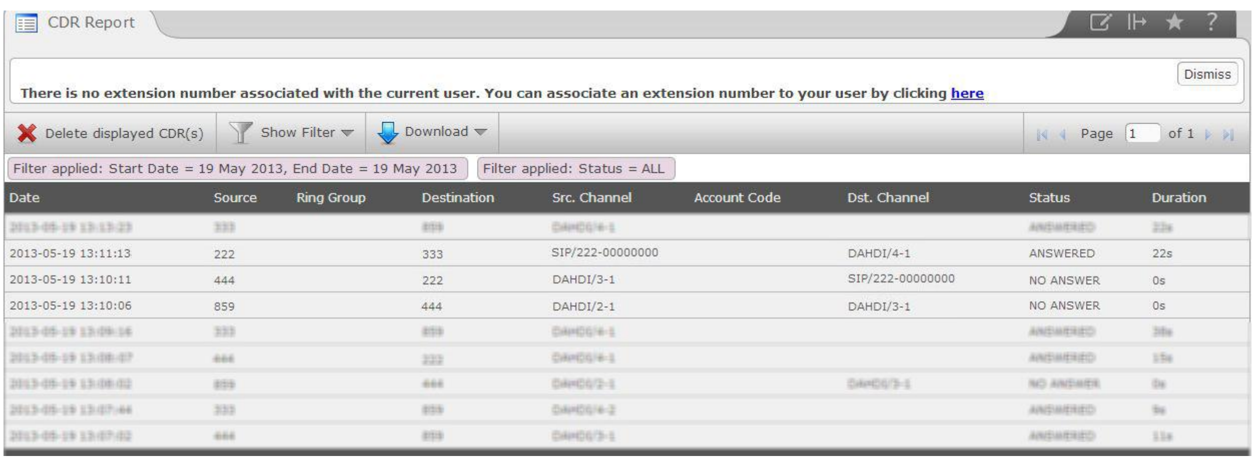


Figure 5.8: Elastix CDR report

The queue called extension 444 and after 5 seconds, the call forwarded to extension 222 (which has no access at that moment to the server) then the call forwarded to extension 333 (GSM module) to make a GSM call.

6

Chapter Six

Conclusions and Future Work

6.1 Introduction

6.2 Real Learning Outcomes

6.3 Conclusions

6.4 Problems

6.5 Future Work.

6.1 Introduction

This chapter includes the results of the system, suggestions and improvements to the system which could be used in future work.

6.2 Real Learning Outcomes

The following remarks were being outlined:

- We have learned how to combine all the knowledge that we have taken in the previous years in a practical way to help us in solving problems in our project.
- We have learned how to interface GSM modem with PIC microcontroller by hardware and software.
- We have learned how to use VoIP servers like Elastix and how to solve problems using its features and functions.
- We have learned how to solve hardware problems and how to design interfaces.
- We have learned how to work in group and under pressure.

6.3Conclusions

The system was designed, developed. Constructed and tested, and the following remarks were being outlined:

- Setting up the Elastix server was accomplished successfully and so using its functions.
- Interfacing the GSM modem with server was accomplished successfully.
- Configuring Elastix to match the project requirement was accomplished successfully.
- Reaching house owner while he is outside the house using VoIP and GSM was accomplished successfully.

6.4 Problems

During the development progress several problems has occurred most of them easy to handle and what could be described as expected problems and others were hard to solve and had to be solved with alternatives. In this chapter of the report some of the more challenging problems will be described and how they were solved.

- The availability of some of the hardware in Palestine, so we have to but it online from outside the country.
- Interfacing the GSM modem with the server was a problem but we designed a matching circuit to provide matching interfacing between the modem and the server.
- The number of the FXS modules we have is not enough to apply more features.
- Connect the intercom directly with server, which us solved by using the DoorBellFon.
- The little experience to use Elastix and we have practiced on it by self-practice by learning from tutorials.
- Noiseand echoes in the GSM modem.

6.5 Future Work

- Use analog phone instead of intercom.
- Use FXO/FXS card with more FXS modules to provide different features to the project.
- Integrate a camera at the front door to send a picture of the visitor.

APPENDIX A

INSTALLATION

A PC is a prerequisite. For stability, we suggest that Elastix be installed on a dedicated machine. This is what we are setting out to do.

INITIAL INSTALLATION

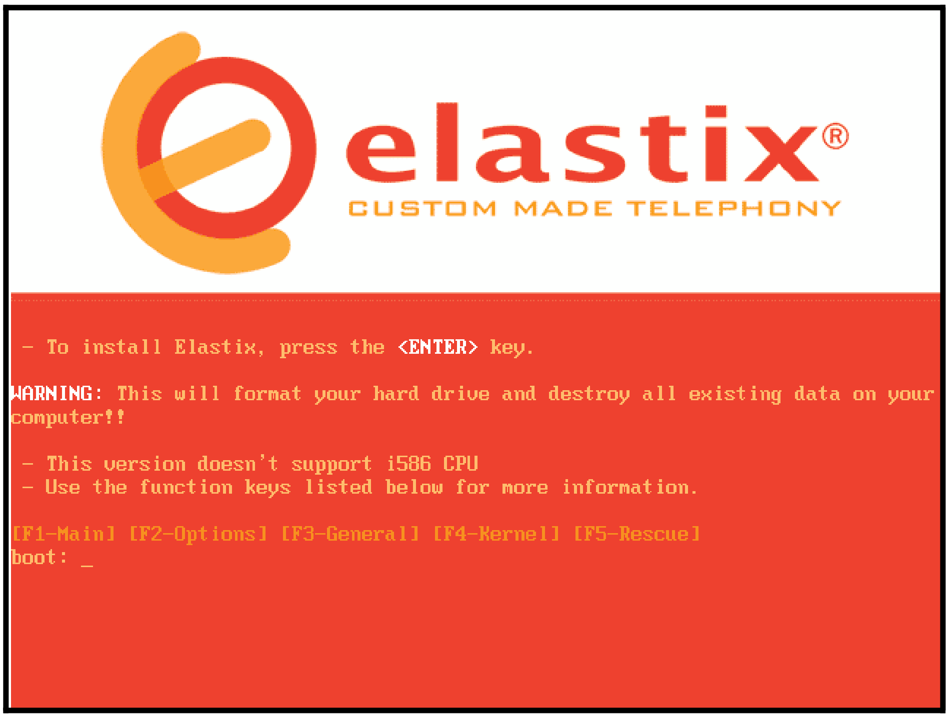
The Elastix version that we will be installing is the latest current version available today – Elastix 1.0. we will download the ISO from the Elastix website below. http://www.elastix.org

After we have completed the download,

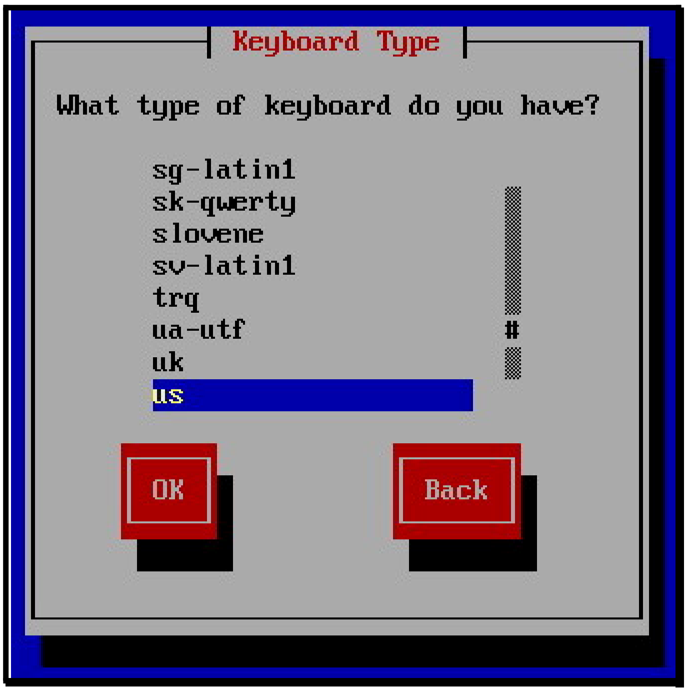
- Burn the ISO image to a blank CD.
- Ensure that PC will boot from the CD.

**NOTE: This will erase all data on the hard drives of the PC.

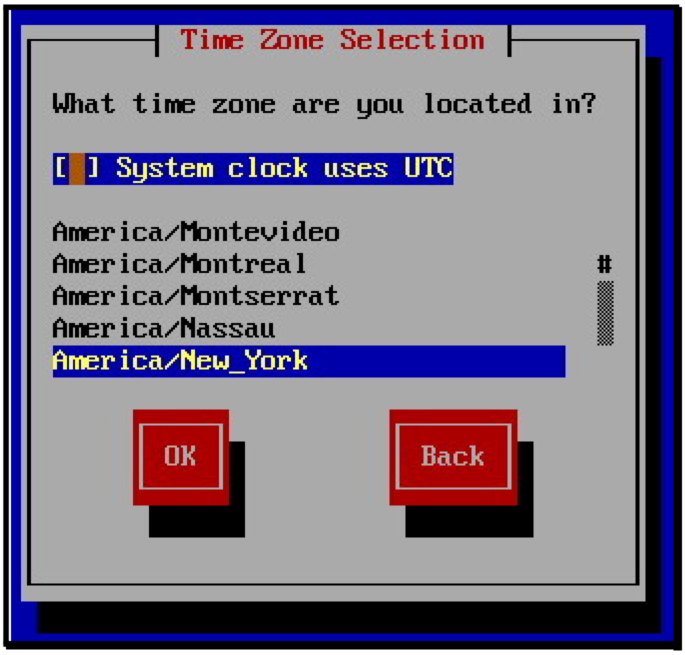
- Boot Elastix box with the CD in the CD Drive. After a few seconds, the following screen will appear. Press [Enter] to start the installation.



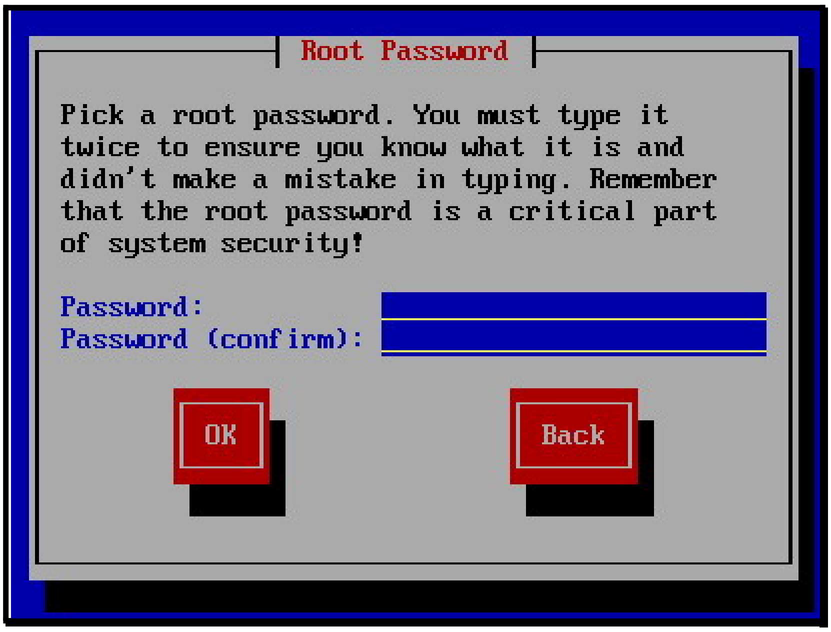
- Press the Enter key to start the installation.
- After initial system detection, we will select the type of keyboard we have, which is English.



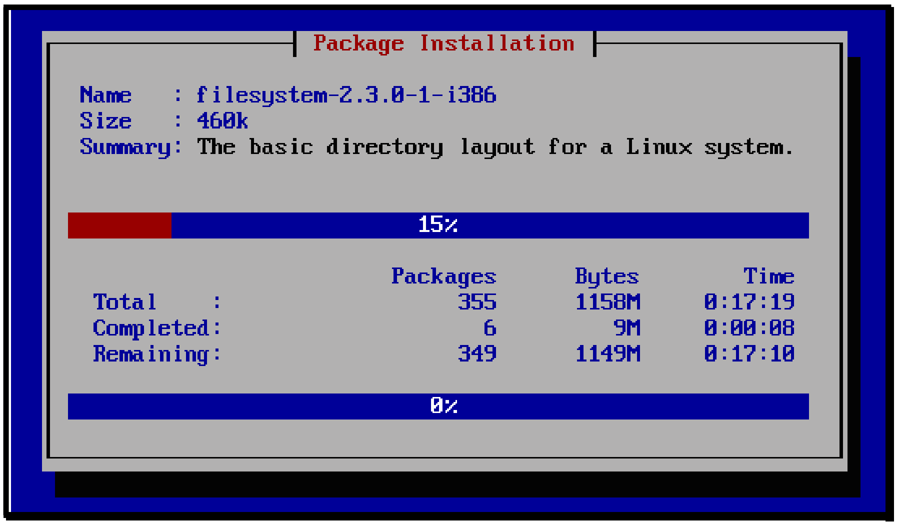
• After system hardware probing is completed, we will be asked to select the Time Zone we are in.



• Next we will be asked to enter our root password (this password is important for logging-in later).

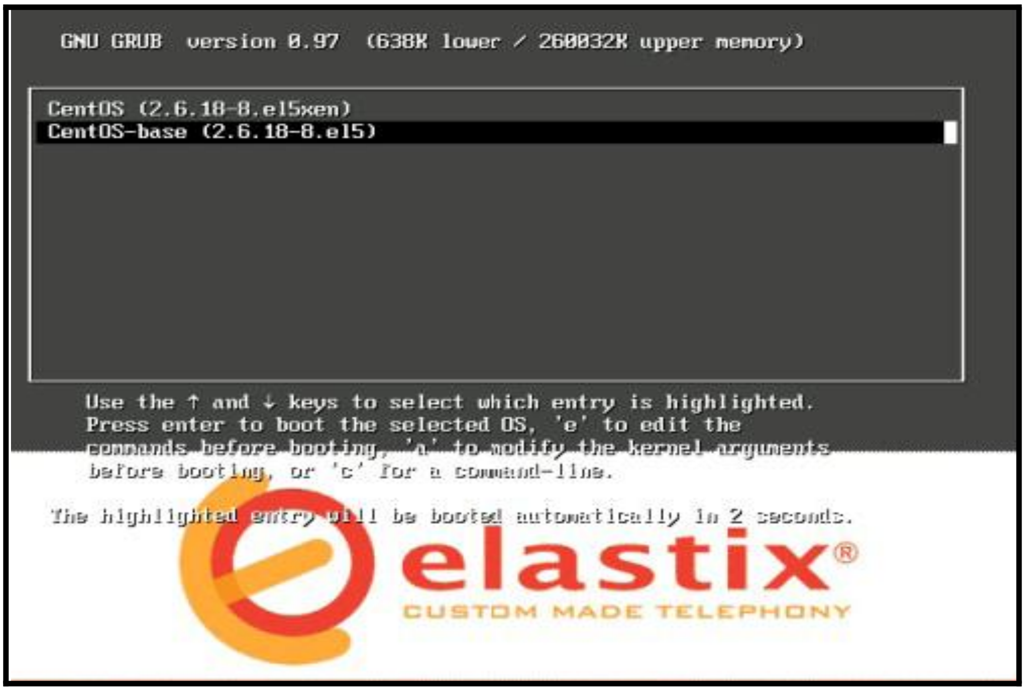


- After entering and confirming password, installation will now commence by first formatting the Hard Disk/s.
- From this point it will take about 30-45 minutes for the installation to be complete and ready for the configuration stage.
- During this stage, we will see screens similar to the following. Linux and the required files are being installed. We will wait for it to finish.



- After Linux is loaded the CD will eject. Wait for the system to reboot.
- When it reboots, we will be presented with the Elastix splash screen.
- After a moment, it will continue and we will see lots of lines of codes. This process will take a while because it is building Asterisk.
- When Asterisk build is complete, it will reboot itself.

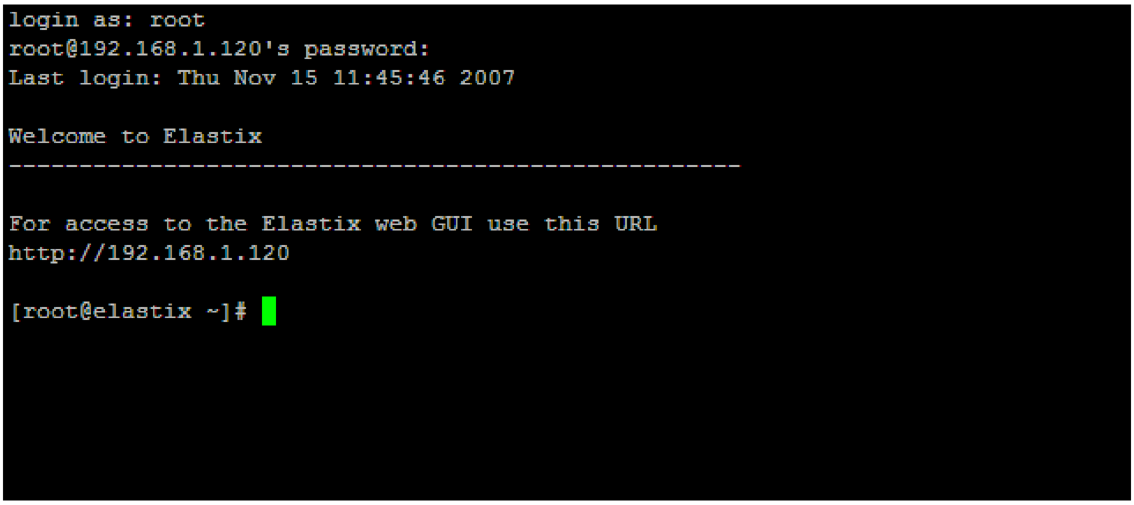
- Once rebooted, we will see the following screen where we can select the Elastix distribution versions. In this case we shall leave it at the default.



- After going through its initial startup script, Elastix is ready for configuring and make changes to the system default.

Once Elastix has been installed, we may log in to Elastix if we need to do any command line tasks.

Log in to our new Elastix (***user:*** `root`, ***password:*** `The one we gave earlier`)



Elastix uses embedded as well as stand-alone freePBX. To take full advantage of freePBX features, we may want to add extra modules that are not included in the embedded freepbx.

SET-UP ELASTIX

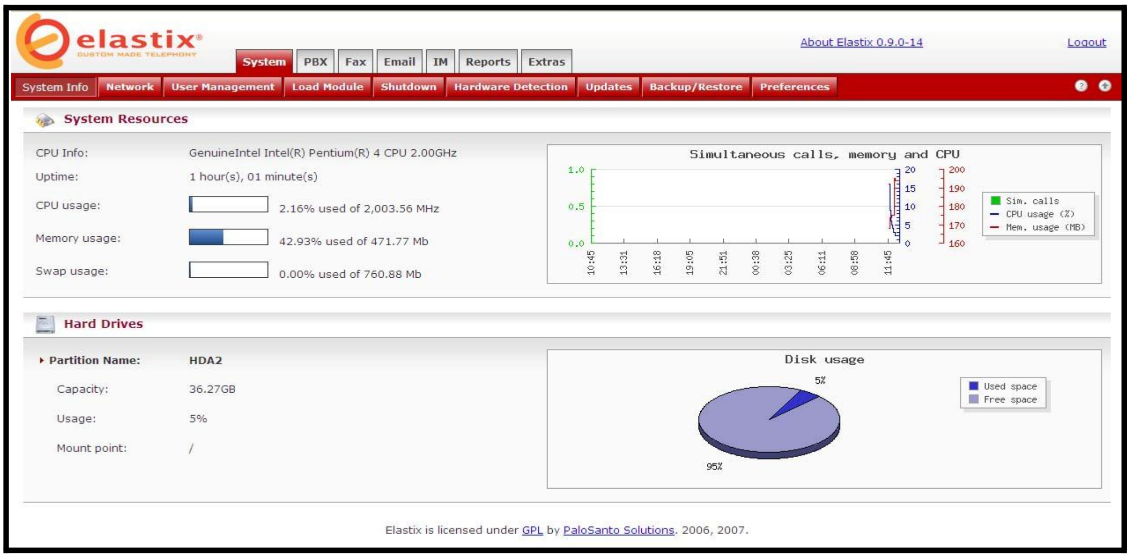
Using browser, connect to ***https://ipaddress/*** (e.g. https://192.168.1.100) to configure Elastix

We will be presented with the Elastix initial Admin web login screen as illustrated below.

Initial Admin Web Login Screen v0.90

The initial Username is ***admin*** and the default password is ***palosanto***. We can change this password later on using that User Management facility provided by Elastix.

Once logged in, the System Status screen will appear. This screen is the control centre.



The Admin Dashboard – The Elastix control centre

APPENDIX B

The PIC controller -in the GSM modem- programming code.

```
//---------PIC18F2525----------
const size=24;
char Numper[size+5][10]={0xFF};
char MS_address[2];
char SMS[160]={0xFF};
char USD[100]={0xFF};
char pass[4];
char SMSI=0,SMSIX=0;
char USDI=0;
char Inumper=0,Inumpersms=0;
char IMS=0,MS_numper=0;
char USD_Flag=0;
char Teemp[13]={0xFF};
char
char response_rcvd = 0;

short response = -1;
const GSM_OK     = 1;
const GSM_RING   = 2;
const GSM_CLIP   = 3;
const GSM_CallerID = 4;
const GSM_MS =5;
const SMSERROR=6;
const GSM_SMS=7;
const Teempsms[13]={0xFF};
char TeempsmsUSD[13]={0xFF};
char nur=0;
char callingflag=0,callcnt=0,callloop=1;
char gsm_state = 0;
GSM_USD=8;
const GSM_BUSY=9;
const GSM_NO_CARRIER=10;
const GSM_ANSWER=11;
void NO_CARRIER();
//================================

void send_at(const char *s)
{ while(*s){uart1_write(*s++);}
  UART1_Write(0x0D);
}
//================================
void SEND_SMS(const char *s)
{char i=0;
delay_ms(2000);
UART1_Write_text("AT+CMGS=\"");
for(i=0;i<13&&Teempsms[i]!=0xFF;i++)UART1_Write(Teempsms[i]);
UART1_Write('\"');UART1_Write(0x0D);
delay_ms(3000);
while(*s){UART1_Write(*s++);}
UART1_Write(26);
UART1_Write(0x0D);
```

```
delay_ms(1000);
}
//===========================
void deletsmsarry()
{char i=0;for(i=0;i<100;i++){SMS[i]=0xFF;} }
//===========================
void deletsmsnumberarry()
{char i=0;for(i=0;i<13;i++)Teempsms[i]=0xFF;}
//===========================
void deletcallernumberarry()
{char i=0;for(i=0;i<13;i++)Teemp[i]=0xFF;}
//===========================
void deletnumberarry()
{char i=0,j=0;
for(j=0;j<size;j++)
for(i=0;i<10;i++)Number[j][i]=0xFF;
}
//===========================
short checkcallernumber()
{
char i=0,j=0,flag1=0;
for(j=0;j<nur;j++)
for(i=0;i<10;i++)
{
if(Number[j][i]==Teemp[i])flag1++;else flag1=0;
if(flag1>9){j=222;flag1=0;delay_ms(200);return 1;}
}
return 0;
}
//===========================
short checksmsnumber()
{
char i=0,j=0,flag1=0;
for(j=0;j<nur;j++)
for(i=0;i<9;i++)
{
if(Number[j][i+1]==Teembsms[i+4])flag1++;else flag1=0;
if(flag1>8){j=222;flag1=0;return 1;}
}
return 0;
}
//====================================
void eewrite()
{
 char i=0,j=0;

 EEPROM_Write(0,nur);
 DELAY_MS(100);

 for(j=1;j<=nur;j++)
 for(i=0;i<10;i++)
 {
 EEPROM_Write(i+(j*10),Number[j-1][i]);
 delay_ms(1);
 }
  for(i=0;i<4;i++)
 EEPROM_Write(i+1,pass[i]);
```

```
}
//=================================



void Delet_All()
{
char i=0,nuri=0,j=0;
nuri=nur;
for(j=0;j<nuri;j++)
for(i=0;i<10;i++)
Number[j][i]=0xFF;
delay_ms(100);
eewrite();

delay_ms(500);
nur=0;
EEPROM_Write(0,0);
delay_ms(500);
SEND_SMS("Delete All Number Done");
}
//=================================
void R_ring()
{
delay_ms(100);
send_at("ATH");
delay_ms(100);
deletcallernumberarry();
}
//==================================
void deletnumber()
{
short flag=-1;
char i=0,j=0,flag1=0;
for(j=0;j<nur;j++)
for(i=0;i<10;i++)
{
if(Numpber[j][i]==SMS[i+5])flag1++;else flag1=0;
if(flag1>9){flag=j;j=222;flag1=0;}
}

if(flag>=0)
{
for(i=0;i<10;i++)
Number[j][i]=0xFF;

nur=nur-1;
for(j=flag;j<nur;j++)
for(i=0;i<10;i++)
{
Number[j][i]=Number[j+1][i];
}
EEPROM_Write(0,nur);
DELAY_MS(100);
 flag=-1;
 }
eewrite();
```

```
}
//==================================
void send_number()
{
char i=0,j=0;
delay_ms(500);
UART1_Write_text("AT+CMGS=\"");
for(i=0;i<13&&Teempsms[i]!=0xFF;i++)UART1_Write(Teempsms[i]);
UART1_Write('\"');UART1_Write(0x0D);
delay_ms(2000);
UART1_Write_text("List Numper:");
UART1_Write(0x0D);
for(j=0;j<nur&&j<13;j++)
{
for(i=0;i<10;i++)
{UART1_Write(Number[j][i]);}
UART1_Write(0x0D);
}
UART1_Write(26);
UART1_Write(0x0D);
DELAY_MS(5000);
}
//===================================
void USD_READ()
{
char i=0;
if(USD_Flag==1){
USD_Flag=0;
delay_ms(2000);
UART1_Write_text("AT+CMGS=\"");
for(i=0;i<13&&TeempsmsUSD[i]!=0xFF;i++)UART1_Write(TeempsmsUSD[i]);
UART1_Write('\"');UART1_Write(0x0D);
delay_ms(3000);
for(i=0;USD[i]!=0x22;i++)UART1_Write(USD[i]);
UART1_Write(26);
UART1_Write(0x0D);
delay_ms(1000);
}
}

//===========================================
void GETUSD()
{
delay_ms(1000);
UART1_Write_text("ATD*059#");
delay_ms(200);
USD_Flag=1;
}
//==================================
void READ_SMS()
{
char i=0;


if(SMS[0]==pass[0]||SMS[0]=='1'&&SMS[1]==pass[1]||SMS[1]=='2'&&SMS[2]==pass[2]||SMS[2]=='
3'&&SMS[3]==pass[3]||SMS[3]=='4')
{
if(SMS[4]=='+')
```

```
{
eewrite();
delay_ms(500);
else if(nur>0)send_number();
}
else if(SMS[4]=='-')
{
if(SMS[5]=='a'&&SMS[6]=='l'&&SMS[7]=='l')Delet_All();
else {deletnumber();send_number1();delay_ms(1000);}
}

else if(SMS[4]=='*')
{
for(i=0;i<13;i++)
{TeempsmsUSD[i]=Teempsms[i];}
delay_ms(1000);
UART1_Write_text("ATD*");
for(i=5;SMS[i]!='#';i++)UART1_Write(SMS[i]);
send_at("#");
delay_ms(200);
USD_Flag=1;
}
else if(SMS[4]=='s'&&SMS[5]=='c'&&SMS[6]=='a'&&SMS[7]=='n'&&SMS[8]=='1')
{
send_number();
}

 }

else if(checksmsnumber())
{
   if(SMS[0]=='A'){RA0_Bit=1;RA5_bit=1;delay_ms(1500);RA0_Bit=0;RA5_bit=0;}
else if(SMS[0]=='B'&&SMS[1]=='1'){RA1_Bit=1;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='B'&&SMS[1]=='0'){RA1_Bit=0;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='B'){RA1_Bit=1;RA5_bit=1;delay_ms(1500);RA1_Bit=0;RA5_bit=0;}

else if(SMS[0]=='C'&&SMS[1]=='1'){RA2_Bit=1;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='C'&&SMS[1]=='0'){RA2_Bit=0;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='C'){RA2_Bit=1;RA5_bit=1;delay_ms(1500);RA2_Bit=0;RA5_bit=0;}

else if(SMS[0]=='D'&&SMS[1]=='1'){RA3_Bit=1;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='D'&&SMS[1]=='0'){RA3_Bit=0;RA5_bit=1;delay_ms(1500);RA5_bit=0;}
else if(SMS[0]=='D'){RA3_Bit=1;RA5_bit=1;delay_ms(1500);RA3_Bit=0;RA5_bit=0;}

}
delay_ms(100);
deletsmsarry();
delay_ms(10);DELET_SMS();
deletsmsnumberarry();
}
//===========================================
 void setupAT()
 {
 char i=0;
   for(i=0;i<10;i++)
 {
 send_at("AT");
 delay_ms(1000);
```

```
 if (response_rcvd==1&&response==GSM_OK) break;
 }
 DELAY_MS(1000);
 response_rcvd=0;
 send_at("ATE0");
 delay_ms(500);
 send_at("AT+CMGF=1");
 delay_ms(500);
 send_at("AT+CLIP=1");
 delay_ms(6000);
 send_at("AT+CMGDA=\"DEL ALL\"");
 delay_ms(1000);

 }
 //==================
 void poweron()
 {

 if(RB7_Bit==0)
 {
// while(RB7_Bit==0)
 {
 RB6_Bit=1;
 DELAY_MS(2300);
 RB6_Bit=0;
 DELAY_MS(5000);
 }
 setupAT();
 }
 }
//====================
void setup()
{

PIE1.RCIE   = 1;
INTCON.PEIE = 1;
INTCON.GIE  = 1;

ADCON1 = 15;            // Configure all ports with analog function as digital
CMCON  = 7;            // Disable comparators
UART1_Init(9600);

TRISA=0;
TRISB=0b10000000;
TRISC=0b10001111;

LATA = 0;
LATB = 0;
LATC = 0;

delay_ms(1000);
RA5_bit=1;
 deletsmsnumberarry();
 deletcallernumberarry();
 deletsmsarry();
delay_ms(500);
deletnumberarry();
eeread();
```

```
delay_ms(500);

}
//====================
 void interrupt()
{
char tmp;
char i,j;

  if (INTCON.F1==1)
  {
  INTCON.F1=0;  // clears  (external interrupt)
  }
//---------
 if(PIR1.RCIF==1)// If there is something in RX buffer
{
PIR1.RCIF=0 ; //Clear flag
tmp=RCREG ;
 switch (gsm_state) {
      case  0:{if (tmp == 'O')gsm_state = 1;
              else if (tmp == 'R')gsm_state = 10;
              else if (tmp == 'B')gsm_state=90;
              else if (tmp == 'C')gsm_state = 20;
               else if (tmp == 'A')gsm_state = 100;break;}
      case  1:{if (tmp == 'K'){response = GSM_OK;gsm_state = 50;}else gsm_state =
0;break;}
      case 10: {if (tmp == 'I')gsm_state = 11;else gsm_state = 0;break;}
      case 20: {if(tmp=='L')gsm_state=21;else if (tmp=='A')gsm_state=95;else if
(tmp=='U')gsm_state=80;else if(tmp=='M')gsm_state=30;else gsm_state=0;break;}
      case 80: {if(tmp=='S')gsm_state=81;else gsm_state=0;break;}
      case 81: {if(tmp=='D')gsm_state=82;else gsm_state=0;break;}
      case 82: {if(tmp==0x22)gsm_state=83;else if(tmp==0x0D)gsm_state=0;else
gsm_state=82;break;}
      case 83: {if(tmp!=0x22&&USDI<100){USD[USDI]=tmp;USDI++;}
      else {USD[USDI]=tmp;gsm_state=0;USDI=0;response=GSM_USD;response_rcvd = 1;}break;}

      case 30: {if(tmp=='T')gsm_state=31;else if(tmp=='G')gsm_state=40;else
gsm_state=0;break;}
      case 31: {if(tmp=='I')gsm_state=32;else gsm_state=0;break;}
      case 32: {if(tmp==0x2C){gsm_state=33;}else gsm_state=32;break;}

      case 33: {if(tmp!=0x0D && IMS<=2){MS_address[IMS]=tmp;IMS++;gsm_state=33;}

      else if  (tmp==0x0D){MS_NUMPER=IMS;IMS=0;response=GSM_MS;response_rcvd =
1;gsm_state=0;}else {IMS=0;gsm_state=0;}break;}

      case 40: {if(tmp=='R')gsm_state=41;else gsm_state=0;break;}
      case 41: {if(tmp==0x2C)gsm_state=42;else gsm_state=41;break;}
      case 42: {if(tmp==0x22)gsm_state=43;else gsm_state=0;break;}

      case 43: {if(tmp!='+'&&tmp!='0'){gsm_state=0;response=SMSERROR;response_rcvd =
1;}else
if(tmp=='+'||tmp=='0'){Teempsms[Inumpersms]=tmp;Inumpersms++;gsm_state=44;}break;}
      case 44: {if(tmp!=0x22&&Inumpersms<13){Teempsms[Inumpersms]=tmp;Inumpersms++;}
      else if
(tmp==0x22){Inumpersms=0;gsm_state=45;}else{Inumpersms=0;gsm_state=0;response=SMSERROR;re
sponse_rcvd = 1;}break;}
      case 45: {if(tmp==0x0D)gsm_state=46;else gsm_state=45;break;}
```

```
      case 46: {if(tmp==0x0A)gsm_state=70;else gsm_state=0;break;}
      case   47: {if(tmp!=0x0D&&SMSI<30){SMS[SMSI]=tmp;SMSI++;}//20/19
                   else {gsm_state=0;SMSIX=SMSI;SMSI=0;response=GSM_SMS;response_rcvd =
1;}break;}
                else if(tmp=='+') {j++;i=0;nur++;}
             else if(tmp==0x0D||tmp=='#'){gsm_state=0;response=GSM_SMS;response_rcvd =
1;}
             break;}

      case 21: {if(tmp=='I')gsm_state=22;else gsm_state=0;break;}
      case 22: {if(tmp=='P')gsm_state=60;else gsm_state=0;break;}
      case 11: {if (tmp == 'N')gsm_state = 12;else gsm_state = 0;break;}
      case 12: {if (tmp == 'G') {response = GSM_RING;gsm_state = 50;} else gsm_state =
0;break;}
      case 50: {if (tmp == 13)gsm_state = 51;else gsm_state = 0;break;}
      case 51: {if (tmp == 10){response_rcvd = 1;}gsm_state = 0;break;}
      case 60: {if (tmp==0x22)gsm_state=61;else gsm_state=60;break;}
      case 61:{if(tmp!=0x22){Teemp[Inumper]=tmp;Inumper++;gsm_state=61;}
                else{Inumper=0;response=GSM_CallerID;gsm_state=0;response_rcvd =
1;}break;}
      case 90:{if(tmp=='U')gsm_state=91;else gsm_state=0;break;}
      case 91:{if(tmp=='S')gsm_state=92;else gsm_state=0;break;}
      case 92:{if(tmp=='Y'){gsm_state=0;response=GSM_BUSY;response_rcvd = 1;}else
gsm_state=0;break;}

      case 95:{if(tmp=='R')gsm_state=96;else gsm_state=0;break;}
      case 96:{if(tmp=='R')gsm_state=97;else gsm_state=0;break;}
      case 97:{if(tmp=='I')gsm_state=98;else gsm_state=0;break;}
      case 98:{if(tmp=='E')gsm_state=99;else gsm_state=0;break;}
      case 99:{if(tmp=='R'){gsm_state=0;response=GSM_NO_CARRIER;response_rcvd = 1;}else
gsm_state=0;break;}
      case 100:{if(tmp=='N')gsm_state=101;else gsm_state=0;break;}
      case 101:{if(tmp=='S')gsm_state=102;else gsm_state=0;break;}
      case 102:{if(tmp=='W')gsm_state=103;else gsm_state=0;break;}
      case 103:{if(tmp=='E')gsm_state=104;else gsm_state=0;break;}
      case 104:{if(tmp=='R'){gsm_state=0;response=GSM_ANSWER;response_rcvd = 1;}else
gsm_state=0;break;}

      default: {gsm_state = 0;break;}}
 }
//===============================
 void call_x(char x)
{
char i=0;
if(x<=nur)
{
UART1_Write_text("ATD");
for(i=0;i<10&&Numper[x-1][i]!=0xFF;i++)
UART1_Write(Number[x-1][i]);
send_at(";");
}
}
//==================================================
void call_in()
{
if(RC0_Bit==0){delay_ms(200);if(RC0_Bit==0){RA0_Bit=1;call_x(1);delay_ms(1000);}}
}
//========================================
```

```
void getnumber()
{
number[0][0]='0';
number[0][1]='5';
number[0][2]='9';
number[0][3]='8';
number[0][4]='5';
number[0][5]='6';
number[0][6]='5';
number[0][7]='9';
number[0][8]='9';
number[0][9]='9';
}
//================================
void main()
{

OSCCON=0x72;
setup();
poweron();
getnumber();
delay_ms(2000);
RA5_bit=0;
while(1)
{

call_in();
if (response_rcvd)
{

response_rcvd=0;
if(response==GSM_CallerID)R_ring();
else if(response==GSM_MS){delay_ms(500);NEW_SMS(); }
else if(response==SMSERROR){delay_ms(500);DELET_SMS();}
else if(response==GSM_SMS){delay_ms(500);READ_SMS();}
else if(response==GSM_USD){delay_ms(500);USD_READ();}
else if(response==GSM_BUSY){delay_ms(500);BUSY();}
else if(response==GSM_NO_CARRIER){delay_ms(500);NO_CARRIER();}
else if(response==GSM_ANSWER){delay_ms(500);NO_ANSWER();}
}


}
}
```

APPENDIX C

```
// phone PIC code

unsigned volt;
unsigned flag=0;
void main() {
 OSCCON=0x72;

 portb=0 ;
 trisb=0;

 portc=0 ;
 trisc=1;

 while(1)
 {

if(button(&PORTC,0,100,0)&&flag==0)
{
portb=1;
delay_ms(1000);

portb=2;
delay_ms(1000);

portb=4;
delay_ms(1000);

portb=8;
delay_ms(1000);

portb=0;
delay_ms(1000);
flag=1;
}
else if(button(&PORTC,0,100,0)&&flag==1)
{
portb=1;
delay_ms(1000);
portb=0;
flag=0;
}
 }
}
```

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