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24/7 CALL CENTER SOLUTION: BUSINESS PURPOSE CALL CENTER SYSTEM WITH ASTERISK PABX

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Abstract- The purpose of this research is to propose a call management system suitable for businesses using Asterisk PABX (Private Automatic Branch Exchange). Asterisk is an open source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. It supports IP (Internet Protocol) PBX systems, VoIP (Voice Over IP) gateways. The features that could be incorporated with asterisk specific for businesses are IVR (Interactive voice response) with voice recognition, efficient call queuing system, Dual tone multi frequency (DTMF) click to call, CRM (customer relationship management system), dialer and SMS gateway. Asterisk is free and open source there for it is highly customizable. Therefore, it can be easily integrated into the solution that the business wants. This research paper explains about how a call center system can be created using Asterisk PBX to provide communications solution for businesses and how the above mentioned features could be incorporated into Asterisk PBX.

Index Terms- VoIP, DTMF, SMS, Dialer, click to Call, CRM, IVR

I. INTRODUCTION

Call management is an essential part of a modern business which is needed satisfy and retain its customers therefore businesses need an efficient method to handle customer calls. Currently businesses use call centers to handle customer calls which are expensive and not customizable. These are also not suitable for all businesses. Currently call centers use cloud system to store data which could result in loss of data if the cloud is hacked and the response time of the system could drop when the cloud traffic is high. These issues could result in a reduction in number of customer as well as loss of customer reputation for many businesses.

The solution for this problem can be resolved using the proposed system. The system mainly focuses on handling business customers using VoIP and DTMF. Most of the countries allow free VoIP calls (toll free numbers) which allow customers to call the business without being charged for the call. Some businesses might require voice response to calls rather that DTMF since it might not be suitable for certain business where the call attendant might not be able to press the buttons of the phone. Most businesses look forward to increase their profits by retaining their customers doing campaigns and promotions. In this proposed system, it can be done easily with the use of Dialer and SMS. Dialer is an application which can automatically make calls to selected customers with prerecorded promotion messages. SMS is an application which can send bulk SMSs to selected customers automatically. Businesses would also need to keep track of its customers and a CRM system could be used for that. With CRM being integrated into Asterisk, features such as pop-up screen containing client information, Call Record/Monitor, Speed Dial, management of user information, customer calling history, call recordings, and statistics management could be implemented. IVR is a technology which allows a computer to communicate with humans through the use of DTMF tones input via keypad or voice input and this can be implemented along with the proposed solution. This will allow the callers to listen to choose specific option such as language or department when they call the business. Queueing is a technology used to distribute incoming calls to specific agents in the organization. From the caller's perspective, without virtual queuing they have only two choices: wait until an agent becomes available, or hang up and try again later. Queuing systems allow customers to receive callbacks instead of waiting in a queue. Click to call is a technology which allows the customers to make calls instantly to the organization by simply pressing a button found on the website of the organization. With all of the above mentioned features, the proposed system could be implemented and how these features could be implemented with asterisk will be discussed in detailed in this research paper.

Section ii of the research paper describes about the previous work done by others. Section iii contains details about our approach on how the system will be implemented by using Asterisk PABX as well as different components that will be needed to implement the solution such as VoIP, IVR, Call queues, SMS, Dialer, CRM and Click to call. Section iv contains details about the conclusion that was derived from the research. Section v contains details about future work that could be done based on the research paper.

II. Background And related work

Predictive dialing system is a software system used to control and manage the outbound dialing process. The working concept is that the back server dials the phone numbers in batches according to some preset criteria. If the dialing result is Positive Voice, the system will immediately assign this line to an idle agent, and the agent will talk to the customer directly. If the result is Busy Signal or No Answer, the system will record the dialing result, schedule the numbers to be re-dialed in future calling cycles according to the re-dialing strategy. Then it will dial the next number. If the result is Wrong Number, the number will be removed from the system. The outbound technology of call center plays aa important role in CRM and marketing. Accurate Dialing-keep an idle agent to make sure that all the successful outbound dialing can be served by an available agent. A predictive dialing system includes two core modules: The Dialing Control Server and the Application Server. The Dialing Control Server has the capabilities of dialing customers and routing effective calls to the agents. The Application Server can be logically divided into several layers, such as system service, data service, API interface and system application. The Design of the system includes the following features such as Monitor management which includes Agent group management (managing agent groups and agents), Customer information management, Call list management (each job should be assigned to a specific agent group), Report form management (Create reports), Monitor management (monitor the working situation of workers) and Agent-side functions such as Receive incoming calls and Post processing where agents can perform post processing of the call data during or after the conversation. The running environment and development tools used in the system are SQL Server: SQLServer2000® SP3, Oracle 8.15, and IIS 5.0 [1].

A CRM model based on VoIP can allows people who need help desk services while surfing a web page, to request a customer service orally by a traditional phone or VoIP software. The customer service agent can retrieve the web-surfing status of the customer through the internet link, if necessary. In this way, the customer can be promptly and correctly served by personalized service. The proposed model consists of a PSTNM.323 gateways (A standard for multimedia communications over local area network), an Intranet and a cluster of customer service operators' PCs. The system will also use the Real-time Transport Protocol (RTP) to transmit voice packets. The system also supports Predictive Dialing, Skill-based Routing (an added value to ACD that not only equitably distributes incoming calls among a group of agents to yield a better level of customer services, but also routes incoming calls to the "best qualified" operator to handle a caller's needs) and Quality of service (necessary to ensure that the quality of voice transmission will not be affected by varying network traffic conditions). When a new telephone call comes into the gateway, the gateway routes the call to the Automatic Call Distributor (ACD). The ACD finds the most appropriate agent to serve the customer. The database server stores the information on the customer's history with the company. The directory server maps the agent's name to the IP address of the workstation the agent is currently working. The web server is integrated with the call center in the system. Also, the web-surfing status of the customer can be transferred to the agent when necessary. In this way, the customer can receive a personalized service [2].

Due to the current shift from a product-oriented to customer-oriented business strategy, it has become vital for companies to include a customer relationship management system within their company and companies who focus on customers through electronic medium should focus on including an Electronic Customer Relationship Management to keep their customers satisfied. E-CRM includes all the activities needed to obtain, fabricate and preserve customer relationship throughout e-business processes. Vital CRM concepts including customization, personalization, cultivating active customers, many-to-many marketing are either made possible or made easier to put into practice with e-CRM tools. In addition to its enabling function, e-CRM is also assumed to be more expedient, more interactive, more proficient and offering an advanced level of customization. One of the first applications of e-CRM has been in VoIP services. Call centers can implement CRM to call centers by installing a VoIP Gateway and an Internet Call Manager, and adding software to the existing automatic computer telephony integration (CTI) application, call distributor (ACD), and agent stations. The Internet Call Manager provides the Call control function between the existing call center systems and Web callers. The VoIP Gateway changes VoIP calls into circuit-switched calls and directs them to a conventional ACD, where it is queued for an agent. All call records and customer information are stored in Billing Database and Customer Database respectively. These databases will be used by different functions such as Customer Care, Call Provisioning, Billing and Accounting [3].

The remote channel is a common medium over which numerous clients seek assets. Since there are numerous clients, it is essential to distribute this common asset in a reasonable way among the clients. Further since the accessible range is constrained, it is likewise essential to proficiently utilize the channel. In any case, the time-shifting nature of the remote environment, combined with various channel conditions for various clients. Postures critical difficulties to achieving these objectives. Besides. The absence of accessibility of channel and landing measurements further confuses the arrangement which would be to allot assets (time spaces, recurrence, power, and so on.) at a base station to numerous contending streams, where every stream is planned for an alternate beneficiary, the channel conditions might be time-changing and diverse for various recipients. It is outstanding that fittingly picked line length based approaches are throughput-ideal while different arrangements in view of the estimation of channel measurements can be utilized to dispense assets genuinely, (for example, relative decency) among contending clients. Subsequently, a blend of line length based booking at the base station and clog control executed either at the base station or toward the end clients can prompt reasonable asset distribution and line length dependability. Association between the end-to-end clog controller and the nearby line length-based scheduler strangely brings about a relatively reasonable portion of the administrations. Besides, utilizing virtual lines, the cushion levels are kept low and henceforth the deferrals experienced by the streams are additionally low [4].

Practically all organizations are occupied with giving data and help to existing and imminent clients. As of late, the diminished expenses of broadcast communications and data innovation have made it progressively sparing to solidify such data conveyance capacities, which prompted the development of gatherings that spend significant time in taking care of client telephone calls. For most by far of these gatherings, their essential capacity is to get phone calls that have been started by clients. With the improvement of call center operations administration scientists can profit by enhancing the path in which the pressure amongst proficiency and nature of administration is displayed. Verifiably, most research accessible as needs be focus operations has compared administration quality with client holding up times. Notwithstanding, there are various studies that exhibit that clients put a high esteem on different measurements of their experience, including components, for example, first call determination and saw operator competency, and in addition less substantial measures, for example, pleasantness and amicability. In that capacity, there is a requirement for viably demonstrating administration quality in a way steadier with these client values. Potential to essentially enhance the route in which call focuses are overseen since there is still huge work to be done on customary call focus operations administration issues, including both hypothetical and observational exploration. Determining models will keep on playing a vital part in operations, serving as a basic contribution for both asset procurement and asset organization choices. What's more, there is an open door for expanded joining of estimating, procuring, staffing, planning, and steering choices, at last prompting better asset use and lower client holding up times [5].

Call focuses, or their contemporary successors contact focuses, are the favored and pervasive path for some organizations to speak with their clients. The "lining perspective" of call focuses is both regular and helpful. In like manner, lining models have served as common standard bolster devices for call focus administration. Be that as it may, the advanced call focus is a complex socio-specialized framework. It hence appreciates focal components that test existing lining hypothesis as far as possible, and past. Most call focuses additionally bolster Interactive Voice Response (IVR) units, likewise called Voice Response Units (VRU's), which are the mechanical renditions of voice-mail, including the potential outcomes of co-operations. Yet, all the more for the most part, a present pattern is the expansion of the call focus into a contact focus. The last is a call focus in which the customary telephone utility is upgraded by some extra multi-media client contact stations, usually VRU, email, fax, Internet or talk (in a specific order of commonness). Lines in administration operations are regularly the field where clients, administration suppliers (servers, or operators) and administrators set up contact, to mutually make the administration experience. Process-wise, lines play in administrations much the same part as inventories in assembling [6].

Electronic Private Branch Exchange (EPBX) System is an existing system for the communication by means of allocated extension numbers to the receiver. It needs extra manpower and announces difficulties at the time of installation. The main weakness of using EPBX that to shift that system is difficult. Hence IVR with reproduced vocal or messages which enable button press and voice can be used. Also the system includes GSM Gateway tools to connect to the IVR server over the Wi-Fi from smart devices. In IVRs GSM Gateway consuming facility of implanting SIM. With the number delivered by SIM we can make calls. This method helps to progress traditional telephone structure to create it easier and flexible. Legato, Maletsabisa Molopo, et al. Established an Interactive Voice Response system (IVRs) with voice extensible markup language (VXML) and asterisk. In this structure uses voice recognition to synthesis (text to speech) or pre-recorded audio to reply to the client. VXML is used in contact call centers to center. The system consists of one or more smart phones an IP Private Bank Exchange (PBX) server and optionally a VOIP Gateway to attach to existing PSTN lines. The IP PBX server gatherings in a same manner to a proxy server: Smart telephone users, being either soft telephones or hardware-based telephones, inventory with the IP PBX server, and when they wish to generate a call they request the IP PBX to found the connection. The IP PBX has a directory of all telephones/clients and their equivalent Smart telephone address and thus is able to attach an internal call or route an external call via also a VOIP gateway or a VOIP service provider. But there are boundaries in VXML and also lesser efficiency. Also VXML wanted a telephony stage in command to allow IVR. The research group has obvious to use Asterisk to implement a suitable structure, later was able to provide Interactive Voice Response which makes structure independent of human interference and aids to deliver better productivity and client support [7].

A Structure which can be utilized to connect allocated number extensions to the receiver has been established which efforts with the use of Interactive Voice Response system (IVRs) and has a reproduced vocal sound or messages. The IVR structure provides applicable replies in the form of vocal. A GSM Gateway is used to attach a sim by simplifying the mobile IVR for the structure. The number delivered by SIM and make calls. The main weakness here is high rate for phone wireless structure. The planned system is an open source, lesser rate and numerous calls go to without a moment's delay [13].

Short Message Service (SMS) has confirmed to be popular universal in sending and receiving alphanumeric messages on phones, and is maintained by all hand phones from the initial GSM phone handsets to the modern smart phones [8][9]. A lone SMS message has the extent constraint of 160 characters and is supplied within a short time in general. SMS messages can also be supplied universally. In accumulation to the common messaging and distribution submissions, there is a huge amount of research literature available on specialized SMS submissions for example diabetes management [10] [11], asthma control, weight control [12], phone payment, and other fitness care applications, to name a rare. There is a huge response of same types of less volume, but vastly customizable combined communication facilities that require particular forms of integration of SMS messaging with vocal, VoIP, and web applications.

Building on Structure earlier extensive knowledge in developing custom SMS combined solutions in the service provider manufacturing, a structure was build structure which was directed towards the place market. The obtainable stage is an extensive multi-tenant stage that allows fast facility of bendable services to numerous providers for resell to their client bases. The kinds of services series from humble outbound SMS distribution to any inventive SMS combined applications. The substructure and stage have been built mostly with standard apparatuses and open source software. Application programming interfaces (HTTP APIs) have been developing to enable internal structure communications and external correlation with client' systems. A modify database has been made to support all the structure configuration, multi-tenant organization, real-time service processes, and billing behavior.
A SMS unit for Asterisk was established by Adrian Kennard, and is an implementation of the ETSI requirement for phone line SMS, ETSI ES 201 912. Phone line SMS is initial to be available in different parts of Europe, and is accessible from BT in the UK. However, Asterisk would agree gateways to be made in other places for example the US, and use of SMS skilled phones such as the Magic Messenger. SMS works using analogue or ISDN lines.

Transfer messages from an Asterisk box can be utilized for a selection of explanations, including announcement from any checking structure, email topic lines, etc. Getting messages to an Asterisk box is classically utilized just to email the messages to somebody correct - we email and texts that are received to us through numbers to the appropriate creature. Established messages could similarly be utilized to switch submissions, be able to competitions, votes, and post items to IRC, whatever thing. Expending a terminal such as a magic messenger, an Asterisk box could ask as a message center sending messages to the terminal, which will beep and pop up the message (and remember 100 or so messages in its memory).

The customary techniques for communication are tested and attractive each day. WebRTC is by all accounts a portion of this variation. Statement designates idea of WebRTC (Web Real-Time Communication) or up and coming period of online communication. The real impartial of this paper is to get a plan without limits media transmission management i.e. WebRTC and study its particular quality in conveying a more adjustable method for communication by authorizing web programs to provide constant communication as a module of its central aptitudes. This paper additionally covers the project, altered settlements and some of its result in the IT field of the before mentioned improvement [14].

WebRTC is an API enrolled by World Wide Web consortium (W3C) that backings program to-program submissions for vocal calling, constant video talk and P2P file sharing without units or stranger programming's i.e. by straightforward Java Script APIs. It is underway by Google, to make a standard based endless media Engine into the superior portion of the reachable programs. The W3C and IETF personifies a WebRTC API that permits a web application running on any device, through the endangered access to the info peripherals, to trade continuing media and data with a distant assembly in a distributed method. Theoretically, WebRTC and HTML5 could authorize the similar change for constant that the first program talented for data. So also, WebRTC may authorize to idea the program at the server where the facts or applications lives [14].

Voice over IP (VoIP) skill is rapidly organized. The elasticity (reduced communication hardware) and cost productivity (by using the existing IP) are the key issues luring initiatives to change to VoIP. Some security complications may surface with the extensive placement of VoIP. This object presents a summary of VoIP structures and its security problems. This briefly describe simple VoIP architecture and its central differences compared to PSTN. Next, basic VoIP protocols utilized for gesturing and media transport, as well as defense apparatuses are described. Finally, present and potential VoIP bouts along with the methods that have been adopted to counter the bouts are conferred. VoIP provides an elasticity of value-added and modified services for defining customized results. Security problems in VoIP are unique and multifaceted. The VoIP organization can be visualized as three layers; end client tools, network mechanisms, and a gateway to the traditional mobile network [15].

This aids in appreciative existing capabilities and to recognize breaks in addressing the many threats and weaknesses present in VoIP systems. The principal goal was to make a roadmap of existing work in safeguarding VoIP, towards falling the start-up strength required by other researchers to initiate research in this universe. A secondary goal was to classify gaps in existing research, and to aid inform the security community of challenges and occasions for further work [16].

Most of the research previously done has been focused on providing communication explanation for businesses and other establishments consuming several dissimilar systems and none of it actually utilizes asterisk as the telephony switching PBX. Hence, none of the existing structure is identical to the proposed result. The structure planned in this research paper will effort on optimizing Asterisk PBX to effort with businesses by implementing features for example CRM, IVR, Dialer, SMS, Queuing and Click to call.

III. WRITE DOWN YOUR STUDIES AND FINDINGS

The proposed application is a web application specially created to improve communication aspect of modern businesses. Every business has a department which is specially allocated for communication aspect of the company.

As a solution for this, the telecommunication providers introduced a method called call hunting (method of distributing phone calls from a single telephone number to a group of several phone lines). This method was unsuccessful since multiple calls cannot be taken at the same time. Later a method called e1(E1 is a digital transmission link with a total transmit and receive rate of 2.048 Mbps) lines was introduced as a solution for this issue but this was not suitable for most businesses since it was expensive and not able to shift easily.

Therefore, a solution was needed and a protocol called SIP (Session Initiation Protocol) was Introduced to overcome communication issues faced by businesses. This is a popular protocol used by most companies nowadays.

The solution proposed, uses SIP protocol and Asterisk virtual PBX which is created by Digium.

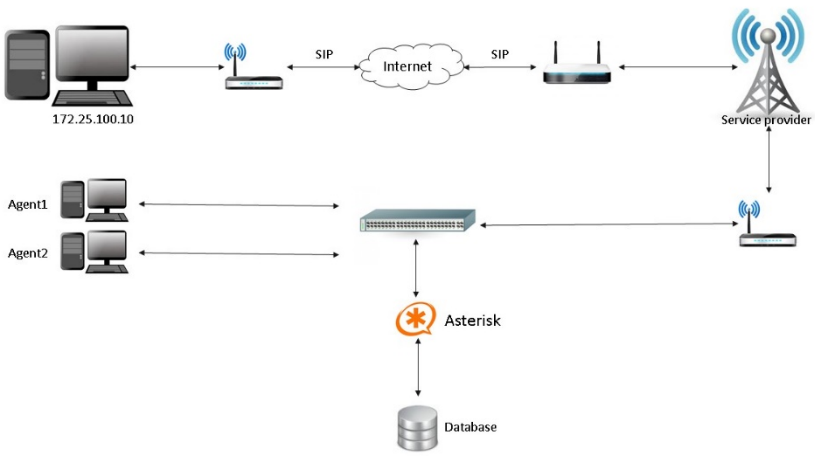

Figure 1: Architecture Diagram

What are the main telecommunication requirements in a business?

- Handle customer information
- Maintain customer satisfaction
- Quick response to customers
- Campaigns and promotions

The proposed system satisfies all above mentioned requirements and the methodology to implement the system is explained with the use of several diagrams and graphs.

The following were found as main requirements of business communication.

(A) Configuration GUIs (Graphical User Interfaces)

(B) VoIP (Voice Over IP)

(C) IVR (Interactive Voice Response)

(D) Queuing

(E) SMS (Short Message Service)

(F) Dialer

(G) CRM (Customer Relationship Management)

(H) Click to Call

(A) Configuration GUIs:

The research group has designed GUI for the satisfaction of the business and the efficiency of their works. Also for the configuration purposes GUIs are helpful.

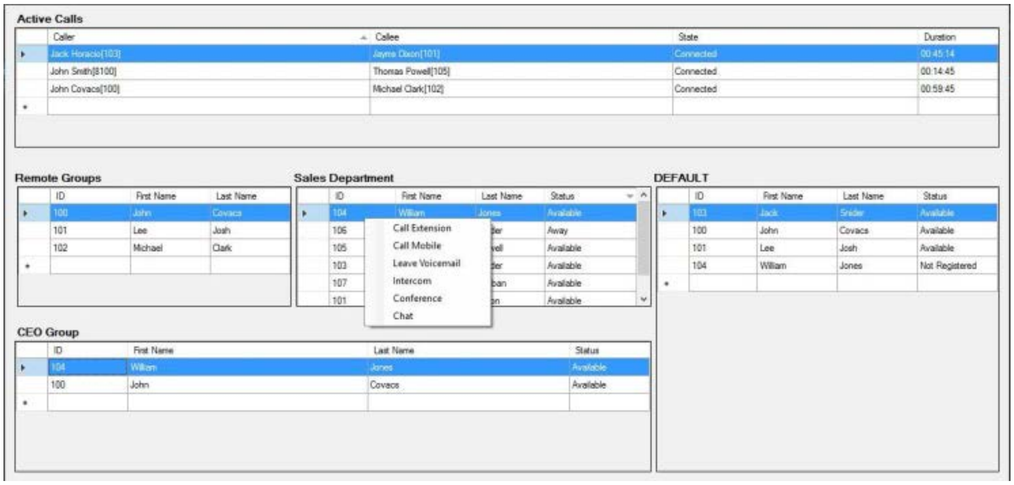

Figure 2: Main GUI

(B) VoIP:

VoIP is used to generate calls via SIP between telecommunication clients. The proposed system generates VoIP calls through asterisk making sure that a high quality is maintained and has a low cost. SIP communication mainly uses four types of codecs which are G.711, G.729, G.723.1 and GSM 06.10. Asterisk supports all four types of codecs. G.711 has two variants, A-Law and u-Law. A-law is being used in Europe and in international telephone links, u-Law is used in the U.S.A. and Japan. The proposed system uses both variants of. It uses a logarithmic compression which squeezes each 16-bit sample to 8 bits, thus it achieves a compression ratio of 1:2. The bitrate is 64 Kbit/s for one direction, so a call consumes 128 Kbit/s. G.711 is the same codec used by the PSTN network, hence it provides the best voice quality. However, it consumes more bandwidth than other codecs. Compared to all other codecs, G.711 is more affordable, which was the reason why it was chosen to be the codec to be used in the system.

Table 1: Maximum bandwidth requirements of different Codecs

Minimum Bandwidth Requirements				
			PPP	
Codec	Voice Bit Rate	Speech Sample Time	RTP	Compressed RTP (cRTP)
G.711	64Kbps	20 msec	83Kbps	68Kbps
G.711	64Kbps	30 msec	77Kbps	67Kbps
G.729A	8Kbps	20 msec	27Kbps	12Kbps
G.729A	8Kbps	30 msec	20Kbps	11Kbps

(C) Interactive voice response:

SIP calls which are received by Asterisk will be initially forwarded to IVR. The main purpose of IVR is to handle incoming calls but can also be used to handle outgoing calls.

```
[general]

[globals]

QUEUE_100=sales
QUEUE_101=support

[default]

exten => 1011,hint,SIP/1011
exten => 1012,hint,SIP/1012

exten => 2004,1,Goto(conference,s,1)
exten => 5555,1,Answer
exten => 5555,n,Background(InvalidNumber)       ; "Than
calling press 1 for sales, 2 for support, ..."
exten => 5555,n,WaitExten
exten => 2,1,Background(welcomeConference)


exten => _1XX,1,Answer()
exten => _1XX,n,Monitor(wav,,m)
exten => _1XX,n,Verbose(2,Call queue as configured in t
${EXTEN} global variable)
exten => _1XX,n,Set(thisQueue=${GLOBAL(QUEUE_${EXTEN})}
exten => _1XX,n,GotoIf($["${thisQueue}" = ""]?invalid_q
exten => _1XX,n,Verbose(2, --> Entering the ${thisQueue
exten => _1XX,n,Queue(${thisQueue})
exten => _1XX,n,Hangup()
```

Figure 3: Dial plan for number pattern checking

This is the dial plan which will be used to route the call. All incoming calls will be checked using its first number and assign it to its dial plan (_1XX). This can be used to match any pattern of number using the asterisk dial plan. Thereafter the call will be recorded automatically. Thereafter calls will be routed to queues. Implementation about queues will be discussed in section (D).

Number matching formats.

Businesses can receive calls from any part of the world. Therefore, the system needs to be able to allow all calls from any region.

- County wise (international)
- Area wise (Local)

```
[incoming-Local]
exten => 07XXXXXXXX,1,Dial(SIP/sip-phone_1,60)
exten => 07XXXXXXXX,n,Hangup()

exten => 03XXXXXXX,1,Dial(SIP/sip-phone_2,60)
exten => 03XXXXXXX,n,Hangup()

exten => 01XXXXXXX,1,Dial(SIP/sip-phone_3,60)
exten => 01XXXXXXX,n,Hangup()

[outgoing-Local]
exten => s,1,Dial(SIP/${EXTEN}@provider)
exten => s,n,Hangup()

[incoming-International]
exten => s,1,Dial(SIP/sip-phone_4,60)
exten => s,n,Hangup()

[outgoing-International]
exten => s,1,Dial(SIP/${EXTEN}@provider)
exten => s,n,Hangup()
```

Figure 4: Dial plan for different carriers and regions

(D) Queuing

Queuing concept is used in call centers. Call centers use an Automatic Call Distributor (ACD) to distribute incoming calls to agents. ACDs hold queued calls in First in, First Out order until agents become available. From the caller's viewpoint, without effective queuing they have only two choices: wait until an agent resource becomes available, or suspend up and try over later. From the call center's viewpoint, a lengthy queue results in many uncontrolled calls, repeat attempts, and customer dissatisfaction. Queuing systems allow customers to receive callbacks instead of waiting in an ACD queue.

Asterisk Call Queues support the following features

- Incoming calls being located in the queue
- Members that answer the queue (extensions or customers that login as agents)
- An approach for how to switch the queue and split calls between associates
- Music played while waiting in the queue
- Notices for associates and callers

Queues are defined in queues.conf file in Asterisk.

Calls are distributed among the members handling a queue with one of several approaches, defined in queues.conf

- ringall: ring all accessible channels until one answers (default)
- roundrobin: take turns ringing each accessible interface
- leastrecent: ring interface which was least in recent times called by this queue
- fewestcalls: ring the one with fewest ended calls from this queue
- random: ring accidental interface
- rrmemory: round robin with memory, recall where we left off last ring pass

extensions.conf includes information of extension numbers allocated to each queues

exten => 28,1,AgentLogin(1001)

exten => 29,1,Queue(queue1)

queues.conf file holds information about the Queue assets such as call distribution method, waiting time and callback as well as information about which agent is allocated to which queue as publicized below

[queue1]

member => Agent/1001

agents.conf file contains details about agents

[agents]

agent => 1001,4321,Wayne Kerr

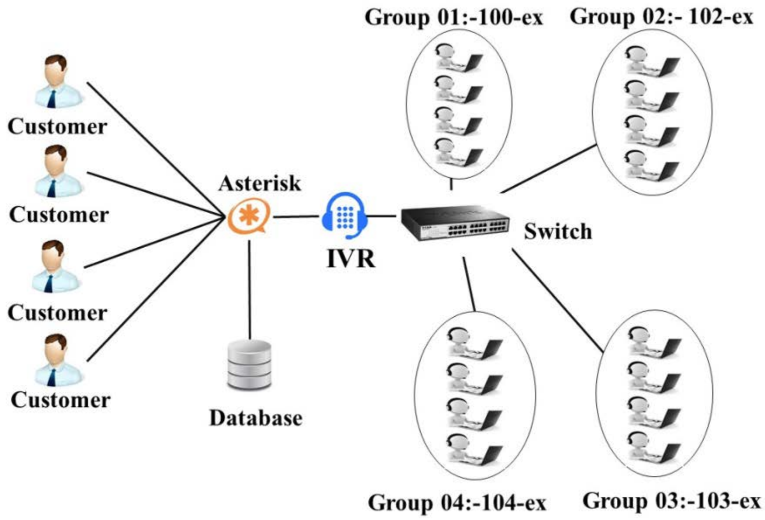

Figure 5: Architecture diagram for Queues

(E) SMS (Short Message Service)

Short Message Service (SMS), or texting is very common communication technique between mobile phones. A message can be sent between two phones, and normally comprises of 160 characters. There are ways in which various types of data can be preset in a text message such as ring tones, and minor graphic, etc. Text messaging is being used for voting and spam, and also competition.

Sending a message involves the mobile phone contacting a message center (SMSC) and passing the message to it. The message center then contacts the destination mobile to deliver the message. The SMSC is accountable for storing the message and trying to send it until the endpoint mobile is available, or a timeout.

Landline SMS works in mostly the same way. You would normally have an appropriate text capable landline phone, or a separate texting box such as a Magic Messenger on your telephone line. This sends a message to a message center your telco provides by making ordinary call and sending the data using 1200 Baud FSK signaling according to the ETSI spec. To receive a message the message center calls the line with a definite calling number, and the text capable phone answers the call and receives the data using 1200 Baud FSK indicating. This works mainly well in the UK as the calling line identity is sent before the initial ring, so no phones in the house would ring when a message arrives.

Sending messages from an asterisk can be used for a selection of reasons, containing notification from any monitoring systems, email subject lines, etc.

Receiving messages to an asterisk box is usually used just to email the messages to someone appropriate -we email and texts that are acknowledged to our direct numbers to the suitable person. Received messages could also be used to control and manage applications, competitions, votes, and post items to IRC, anything.

Using a terminal such as a magic messenger, an asterisk could request as a message center sending messages to the terminal, which will beep and pop up the message

Sending/Receiving SMS using Asterisk Management Interface

Sending SMS using Asterisk CLI

```
wat send sms
<span><number><message>
```

Receiving SMS using Asterisk CLI

```
--Span1: SMS received HelloWorld
```

[smsmtrx]

exten = _X.,1,SMS(${EXTEN}|a)

exten = _X.,2,System(someapptohandleincomingsms ${EXTEN})

exten = _X.,3,Hangup

[smsmorx]

exten = _X.,1,SMS(${EXTEN}|sa)

exten = _X.,2,System(someapptohandlelocalsms ${EXTEN})

exten = _X.,3,Hangup

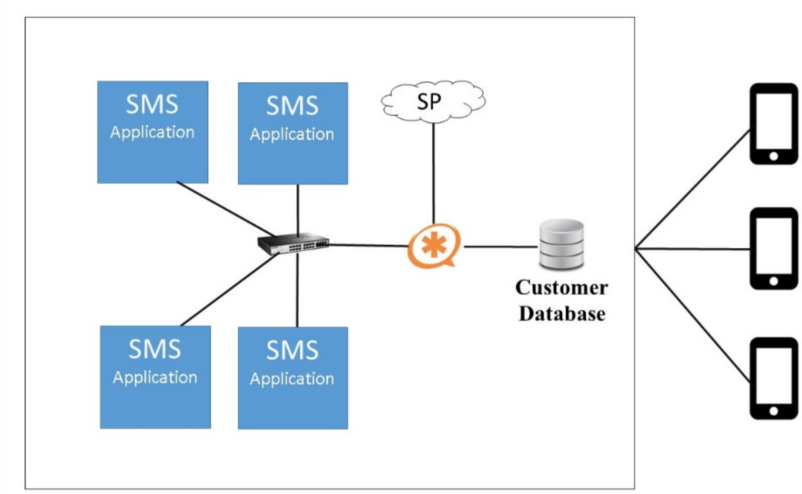

Figure 6: Architecture Diagram for SMS Handling

(F) Predictive Dialer

A predictive dialer is an outgoing call handling system designed to maintain a high level of utilization and efficiency of cost in the contact center. The dialer automatically calls a list of telephone numbers and displays the unnecessary calls such as answering machines, busy signals, and then connects a waiting representative with the customer.

There are loads of various vendors of predictive dialer software program across the globe, and every predictive dialer has its pros and cons.

The concept of predictive dialer is an important aspect for any business. With the use of predictive dialer, activities like marketing, campaigns and promotions can be done automatically without any human interaction. Therefore, the profit of the organization can be improved by implementing a predictive dialer.

Predictive dialers are commonly used by service providers. They will offer this as a service to their customers. Therefore, most businesses are used to outsourcing their call management department function to these service providers. Therefore, as a solution to this, the predictive dialer function could be integrated with Asterisk call center engine Since Asterisk engine highly supports outbound call management.

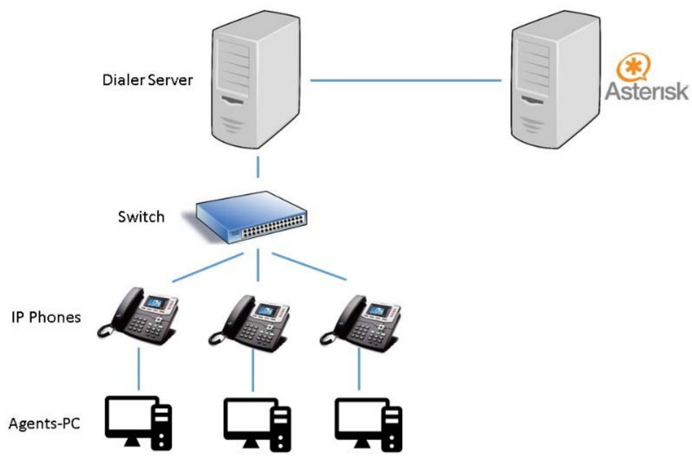

Figure 7: Architecture diagram for dialer operation

(G) CRM (Customer Relationship Management)

CRM allows a company to interact with its customers with the goal of improving its relationship with its customers, retaining its customers and increasing the overall profit of the company. CRM also allows the agents who are facing the customers to have details such as personal information, past history, concerns and buying preferences of the customers when they face them

To implement customer relationship management to the system and application called asterCRM could be used. This would allow businesses to keep track of their customers, log activities, and simplify business p. It is an open source application that bridges Asterisk and CRM. Many functions can be implemented with the use of asterCRM such as a pop-up screen containing client information, Call Record/Monitor, Speed Dial, management of user information, customer calling history, call recordings, and statistics management.

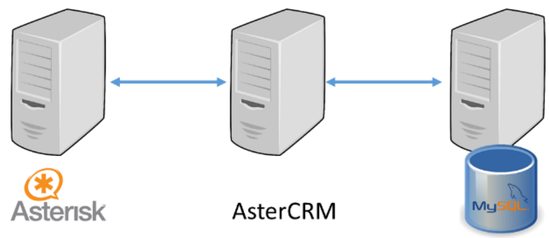

Figure 8: Architecture diagram for CRM with Asterisk

In order to implement asterCRM, MySQL 4.1 or above needs to be installed and the needed databases and tables needs to be created in it. AsterCRM source files needs to be downloaded from astercc.org and unzipped. In order to asterCRM to be able to connect to MySQL, the manager,conf file needs to be edited as shown below.

```
enabled=yes
port=5038
bindaddr=0.0.0.0
secret=astercc
read=system,call,log,verbose,command,agent,user
write=system,call,log,verbose,command,agent,user
deny=0.0.0.0/0.0.0.0
```

Figure 9: Configuration of AsterCRM (Source: http://inhibitz.ucoz.ru/_ld/0/17_Asterisk1.6.pdf)

astercrm.conf.php which is found in /var/www/html/astercc needs to be modified to fit the configuration.

To Start the asterCRM service Modify/opt/asterisk/scripts/astercc/eventsdaemon.pl to fit the configuration and start eventsdaemon. In order to access asterCRM using the web Browser, http://YOUR-WEB-SERVER-ADDRESS/astercc/astercrm needs to be visited.

The following are some screenshot of asterCRM interface.

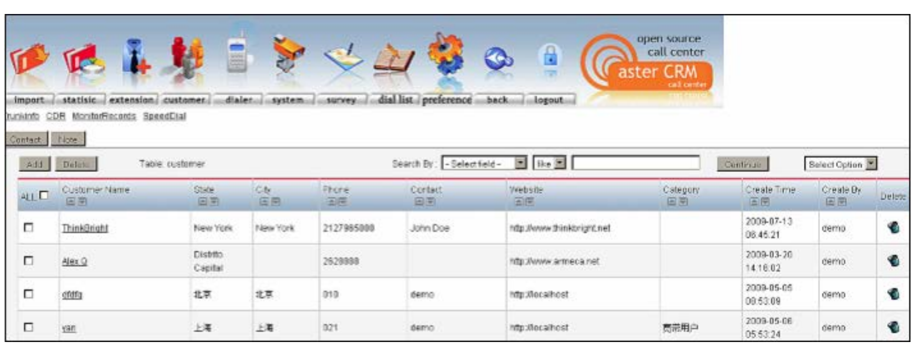

Figure 10: Interface of AsterCRM 1 (Source: http://inhibitz.ucoz.ru/_ld/0/17_Asterisk1.6.pdf)

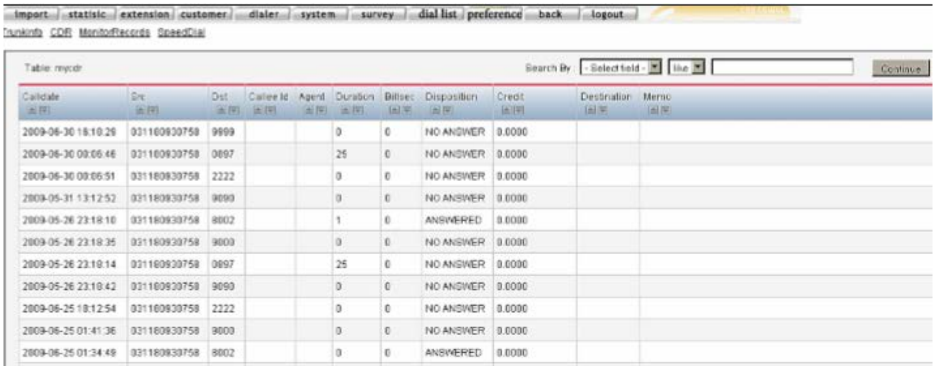

Figure 11: Interface of AsterCRM 2 (Source: http://inhibitz.ucoz.ru/_ld/0/17_Asterisk1.6.pdf)

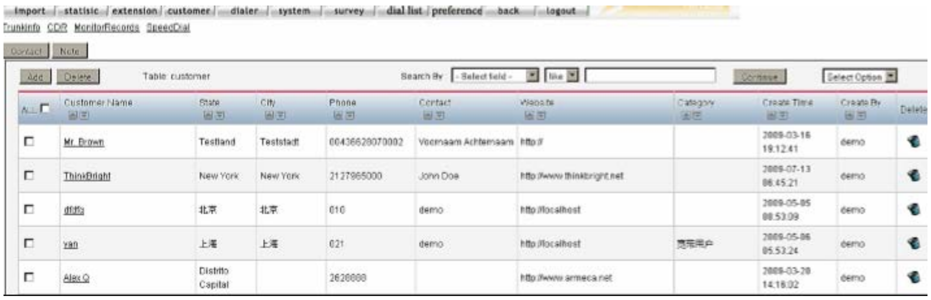

Figure 12: Interface of AsterCRM 3 (Source: http://inhibitz.ucoz.ru/_ld/0/17_Asterisk1.6.pdf)

(H) Click to Call

There are lots of ways to build "Click-to-call" functionality, which allows you to dial a phone number without truly dialing it. The convention for these scripts is to tell Asterisk to call your extension, wait for you to answer the call, and when you do, start a new call to the endpoint number.

Click to call is one of the most modern methods used for communication. This is popular nowadays since modern business mostly function online and attract customers though web based systems. Most of the customers prefer to communicate with the business through web based systems since these are more user friendly than traditional communications methods.

Every single modern business has their own websites to market themselves with the customers. All customers prefer to solve their issues as easy and simple as possible. This would allow the customer to make enquiries with the business without ever having to visit the office premises. Customers will be able to solve their issues without having any cost.

The method to implement this along with the asterisk call management system is described below using diagrams.

Figure 13: Interface for Customer Inquiries

Figure 14 Click to call button provided in the web site

This is this a sample interface that the customer will be able to use under inquiries. Traditionally customers will only be able to view the contact details when they select the inquiries link. With Click to call being implemented, the customers will be able to make a VoIP call as automatically as soon as they press the call us now button. The benefit of using this is that the customer will be able to contact the business without being charged.

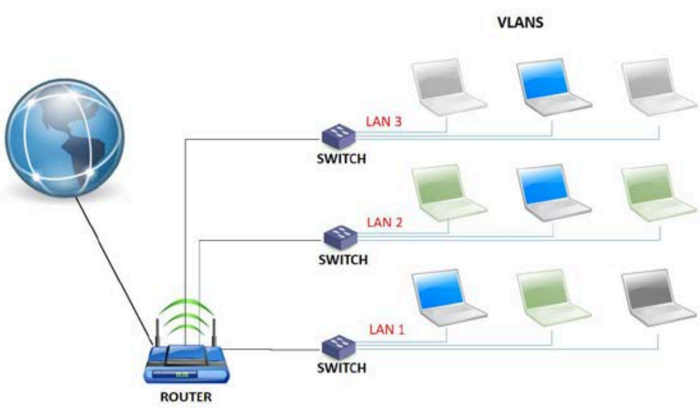

Figure 15: Architecture diagram for click to call

Sample Dial plan (Call Flow)

The following figure shows the inbound and outbound calls which are coming through the web to the proposed system, how it is handled and the call flow.

```
[general]

[globals]

QUEUE_111=sales
QUEUE_112=support

[default]

exten => 1011,hint,SIP/1011
exten => 1012,hint,SIP/1012

exten => 2004,1,Goto(conference,s,1)
exten => 5555,1,Answer
exten => 5555,n,Background(InvalidNumber) press 1 for sales, 2 for support, ..."
exten => 5555,n,WaitExten
exten => 2,1,Background(welcomeConference)

exten => _1XX,1,Answer()
exten => _1XX,n,Monitor(wav,,m)
exten => _1XX,n,Verbose(2,Call queue as configured in the QUEUE_${EXTEN} global variable)
exten => _1XX,n,Set(thisQueue=${GLOBAL(QUEUE_${EXTEN})})
exten => _1XX,n,GotoIf($["${thisQueue}" = ""]?invalid_queue,1)
exten => _1XX,n,Verbose(2, --> Entering the ${thisQueue} queue)
exten => _1XX,n,Queue(${thisQueue})
exten => _1XX,n,Hangup()a

exten => invalid_queue,1,Verbose(2,Attempted to enter invalid queue)
exten => invalid_queue,n,Playback(silence/1&invalid)
exten => invalid_queue,n,Hangup()

exten => _*10[0-1],1,Set(xtn=${EXTEN:1})
exten => _*10[0-1],n,Goto(queueLoginLogout,member_check,1)


exten => _*0[01]!,1,Verbose(2,Pausing or unpausing queue member from one or more queues)
exten => _*0[01]!,n,Set(xtn=${EXTEN:3})
exten => _*0[01]!,n,Set(thisQueue=${GLOBAL(QUEUE_${xtn})})
exten => _*0[01]!,n,GotoIf($[${ISNULL(${thisQueue})} & ${EXISTS(${xtn})}]?invalid_queue,1)
exten => _*0[01]!,n,GotoIf($[${EXTEN:2:1} = 0]?pause,1:unpause,1)

exten => unpause,1,NoOp()
exten => unpause,n,UnpauseQueueMember(${thisQueue},SIP/${CHANNEL(peername)})
exten => unpause,n,GoSub(changePauseStatus,start,1(UPQMSTATUS,UNPAUSED,available))
exten => unpause,n,Hangup()

exten => pause,1,NoOp()
exten => pause,n,PauseQueueMember(${thisQueue},SIP/${CHANNEL(peername)})
exten => pause,n,GoSub(changePauseStatus,start,1(PQMSTATUS,PAUSED,unavailable))
exten => pause,n,Hangup()


[ext-local-custom]

;isten
exten => 1000,1,Answer()
exten => 1000,n,Wait(1)
exten => 1000,n,Authenticate(007,a)
exten => 1000,n,Playback(pin-number-accepted)
exten => 1000,n,Wait(.5)
exten => 1000,n,Background(beep)
exten => 1000,n,Read(chan,,4)
exten => 1000,n,ChanSpy(SIP/${chan},q)
exten => 1000,n,Hangup()
```

```
;whisper

exten => 2000,1,Answer()
exten => 2000,n,Wait(1)
exten => 2000,n,Authenticate(007,a)
exten => 2000,n,Playback(pin-number-accepted)
exten => 2000,n,Wait(.5)
exten => 2000,n,Background(beep)
exten => 2000,n,Read(chan,,4)
exten => 2000,n,ChanSpy(SIP/${chan},qw)
exten => 2000,n,Hangup()

;listen

exten => 3000,1,Answer()
exten => 3000,n,Wait(1)
exten => 3000,n,Authenticate(007,a)
exten => 3000,n,Playback(pin-number-accepted)
exten => 3000,n,Wait(.5)
exten => 3000,n,Background(beep)
exten => 3000,n,Read(chan,,4)
exten => 3000,n,ChanSpy(SIP/${chan},qB)
exten => 3000,n,Hangup()
```

```
[inbound]
exten => 1011,1,Dial(SIP/1011)
exten => 1012,1,Dial(SIP/1012)

exten => 999,1,Ringing()
exten => 999,n,Wait(10)
exten => 999,n,Verbose(Get caller Number & Date Time)
exten => 999,n,Set(GLOBAL(uuid_in)=${UNIQUEID})
exten => 999,n,Set(GLOBAL(num_inbound)=${CALLERID(num)})
exten => 999,n,Set(GLOBAL(num_inbound_dial)=${CALLERID(dnid)})
exten => 999,n,Set(GLOBAL(context_in)=${CONTEXT})
exten => 999,n,Set(GLOBAL(caller_type_in)=${REASON})
exten => 999,n,Set(GLOBAL(dial_status_in)=${DIALSTATUS})
exten => 999,n,Set(GLOBAL(date_time_in)=${STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)})
exten => 999,n,AGI(simsyn.php,${uuid_in}, ${num_inbound}, ${num_inbound_dial}, ${context_in})
exten => 999,n,Hangup()
;exten => 999,n,Wait(5)

;exten => h,1,System(sleep 3)
exten => h,n,Verbose(Call AGI)
;exten => h,n,Wait(5)
exten => h,n,DeadAGI(simsyn.php,${uuid_in}, ${num_inbound}, ${num_inbound_dial}, ${context_in})
```

```
[outbound]
exten => 10,1,Answer()
exten => 10,n,Wait(1)
exten => 10,n,Playback(hello-world)
exten => 10,n,Set(GLOBAL(uuid_out)=${UNIQUEID})
exten => 10,n,Set(GLOBAL(num_outbound)=Asterisk)
exten => 10,n,Set(GLOBAL(num_outbound_dial)=${num_inbound})
exten => 10,n,Set(GLOBAL(context_out)=${CONTEXT})
exten => 10,n,Set(GLOBAL(dial_status_out)=${DIALSTATUS})
exten => 10,n,Set(GLOBAL(date_time_out)=${STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)})
exten => 10,n,Set(GLOBAL(caller_type_out)=${REASON})
exten => 10,n,AGI(simsyn1.php,${uuid_out}, ${num_outbound}, ${num_outbound_dial}, ${context_out})
exten => 10,n,Wait(1)
exten => 10,n,Hangup()

[click-to-call-inbound]
exten => 1011,1,Dial(SIP/1011)
exten => 1012,1,Dial(SIP/1012)

exten => 10001,1,Ringing()
exten => 10001,n,Wait(10)
exten => 10001,n,Verbose(Get caller Number & Date Time)
exten => 10001,n,Set(GLOBAL(uuid_in)=${UNIQUEID})
exten => 10001,n,Set(GLOBAL(num_inbound)=${CALLERID(num)})
exten => 10001,n,Set(GLOBAL(num_inbound_dial)=${CALLERID(dnid)})
exten => 10001,n,Set(GLOBAL(context_in)=${CONTEXT})
exten => 10001,n,Set(GLOBAL(caller_type_in)=${REASON})
exten => 10001,n,Set(GLOBAL(dial_status_in)=${DIALSTATUS})
exten => 10001,n,Set(GLOBAL(date_time_in)=${STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)})
exten => 10001,n,AGI(simsyn.php,${uuid_in}, ${num_inbound}, ${num_inbound_dial}, ${context_in})
```

Figure 16: Dial Plan

Call Center Summery Wallboard:

The summery wallboard can be used to view the summery of the call center activities by the employees of the organization to get an idea of the overall performance level of the agents.

Figure 17: Summery Wallboard

SIP Phone:

The following image shows the interface of the sample SIP phones that would be used by the agents of the call center.

Figure 18: SIP Phone interface

IV. CONCLUSION

The system which is proposed in this research paper can be used for any businesses without any issues. Every business has a primary requirement to improve its growth and in order to improve its growth, the business need to maintain a high customer satisfaction. The system that we propose will help all business to achieve that. This system can be implemented without much cost and can be used by anyone easily. The only cost will be the initial cost that is spent to implement the system. All aspect of marketing of the company can be handled by this system without any cost.

V. FUTURE WORK

As a future work, this system can be implemented under three different categories such as basic, medium and advanced with the advanced version containing more advanced features and the basic system will contain only the basic functions and this basic version can be released to the public as an open source project. Afterwards, the feedback from the public can be listened to and the system could be further improved based on it.

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