CLOUD TELEPHONY TECHNOLOGIES

Project report submitted in partial fulfillment of the requirement for the degree of

BACHELOR OF TECHNOLOGY
IN

COMPUTER SCIENCE ENGINEERING

By

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UNDER THE GUIDANCE OF

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DECLARATION

I hereby declare that the work reported in the B-Tech thesis entitled "CLOUD TELEPHONY TECHNOLOGIES" submitted at Jaypee University of Information Technology, Waknaghat, India is an authentic record of my work carried out under the supervision of Mr.Jitendra Kumar. I have not submitted this work elsewhere for any other degree or diploma.

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Date: 24/5/2019

CERTIFICATE

It is certified that the work contained in the project report titled, “Cloud Telephony Technologies” by “Abhin Bajaj” enrollment number 151469 has been carried out under our supervision and that this work has not been submitted elsewhere for a degree.

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Senior Manager

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ABSTRACT

Real Time Data Services Private Limited (RTDS) is a global information technology, consulting and outsourcing company serving clients in multiple locations across the globe. RTDS commands a substantial reputation, especially in cloud hosting of accounting software, business applications, and contact centre solutions.

RTDS relies on industry experts such as Microsoft, Intuit, VMware, Bill & Pay, Method CRM, Salesforce and more for its technology support. RTDS has its data centres located in Chicago, Dallas, Seattle, and Phoenix to serve businesses over the globe.

RTDS offers a range of business support services to its multiple global clients. The 3 Brands of RTDS are -

1. Ace Cloud Hosting (ACH)
2. Servetel
3. The Real PBX(TRP)

Servetel, a product of Servetel Communications Pvt. Ltd., is a leading provider of telephony services for the businesses based in India Some of the services offered by Servetel include Toll- free numbers and call centre software (such as predictive diallers, manual dialler, call monitoring system, etc.). These solutions are delivered over the cloud technology.

The Real PBX is a business communication solution owned by The Real PBX Pvt. Ltd. Based in London and enjoys business support services from RTDS. The company offers various cloud-based solutions to its clients all over the world.

TRP in its offerings presents Hosted PBX, Toll-free number, Local Number, Dialer Hosting, Dialler Support, and brings down the requirements for the local hardware at the client’s end by providing the entire communication setup over the cloud.

This project covers the analysis and comparison of various Cloud Telephony Technologies in RTDS. We are trained in various portals created by the company using the cloud technology and various programming languages.

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Chapter 1

Introduction

Real Time Data Services Private Limited (RTDS) is a worldwide data innovation, counseling and re-appropriating organization serving customers in various areas over the globe. It directions a significant notoriety, particularly in cloud facilitating of bookkeeping programming, business applications, and contact focus arrangements. RTDS has its server farms situated in Chicago, Dallas, Seattle, and Phoenix to serve organizations over the globe. RTDS helps making more extravagant and progressively important client encounters by utilizing its industry-wide experience, profound innovation aptitude, far reaching arrangement of administration and vertically adjusted plan of action.

RTDS is all around situated to be an accomplice and co-pioneer to organizations in their change venture, in recognizing new development openings and in encouraging their invasion into new markets. RTDS depends on industry specialists, for example, Microsoft, Intuit, VMware, Bill and Pay, Method CRM, Salesforce and morefor its technology support.
The organization at that point works through these three sister organizations giving cloud communication answers for customers everywhere throughout the world. The equivalent will be talked about underneath in the report.

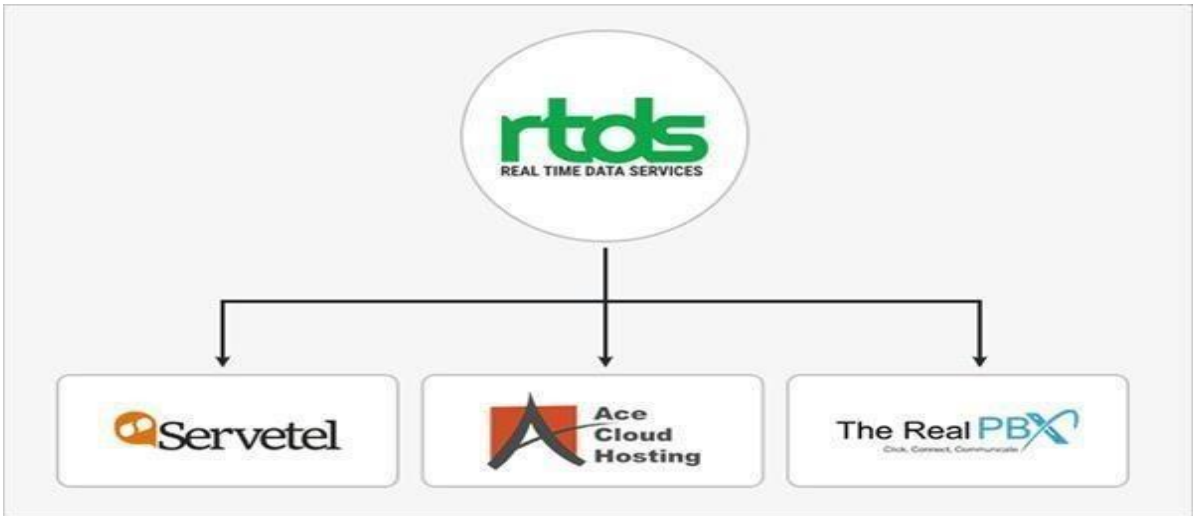

Figure 1 : RTDS and its sister companies.

The organization at that point works through these three sister organizations giving cloud communication answers for customers everywhere throughout the world. The equivalent will be talked about underneath in the report.

Chapter 2

Literature Review

2.1 What is Cloud?

The cloud is regularly used to allude to a few servers associated with the web that can be rented as a feature of a product or application administration. Cloud-based administrations can incorporate web facilitating, information facilitating and sharing, and programming or application use.

'The cloud' can likewise allude to distributed computing, where a few servers are connected together to share the heap. This implies as opposed to utilizing one single incredible machine, complex procedures can be dispersed over different littler PCs.

One of the benefits of distributed storage is that there are many conveyed assets going about as one regularly called combined capacity mists. This makes the cloud exceptionally tolerant of shortcomings, because of the dispersion of information. Utilization of the cloud will in general decrease the production of various renditions of records, because of shared access to reports, documents and information.

2.1.1 What are the benefits of a cloud server?

- A cloud server gives the business client dependability and security on the grounds that any product issues are disconnected from your condition. Other cloud servers won't affect on your cloud server and the other way around. In the event that another client over-burdens their cloud server, this will have no effect on your cloud server, dissimilar to with physical servers.

- Cloud servers are steady, quick and secure. They evade the equipment issues seen with physical servers, and they are probably going to be the most steady choice for organizations needing to keep their IT spending plan down.

- Cloud servers give a quicker support of your cash. You'll get a bigger number of assets and a quicker administration than you would at a comparative cost of physical server. A cloud-facilitated site will run quicker.

- You get versatility with cloud servers. It is simple and fast to update by including memory and circle space, just as being progressively reasonable.

2.1.2 What is Cloud Telephony?

As mechanical changes began topping the business, the wires, equipment, and programming wound up out of date. With the cloud, a business winds up free from the establishment, IT staff or framework upkeep cost. The specialist organization currently handles all the setup, upkeep and redesign of the communication framework on the cloud. As buyers of these administrations, business just needs an interface to associate with the communication server to begin.

Cloud communication is where specialized gadgets are facilitated over specialist co-op's premises and are made available to the clients through the cloud. Further, a web interface is furnished to the clients to associate with this cloud-facilitated framework.

Cloud Telephony offers numerous favorable circumstances, some of them being-

- Lower Cost of Ownership

Most organizations spare around 15-30% on operational costs. The requirement for assets is brought down as the entire set up is virtualized which straightforwardly influences the general spending of the venture. The compensation per-utilize model makes it progressively reasonable for the little and medium venture organizations to adjust.

- Reliable with High Uptime

Different servers are kept in server farms to deal with any sudden specialized disappointment. On the off chance that one framework fizzles, another arrangement of the framework naturally has its spot to begins the administrations with no time slack.

- Easy Scalability

State of no business stays steady – each business will undoubtedly develop or recoil. To stay aware of the regularly evolving need, cloud communication administrations comes in all respects conveniently. Any no. of clients/channels can be included or expelled from the server according to the prerequisite.

- Always Updated

Programming industry continues growing and including new highlights. To match up your correspondence framework with these changes, specialist co-ops continue refreshing the product and equipment at the backend without influencing the client's use. Keeping your administrations refreshed without contributing any additional exertion or time is the real advantage of cloud communication.

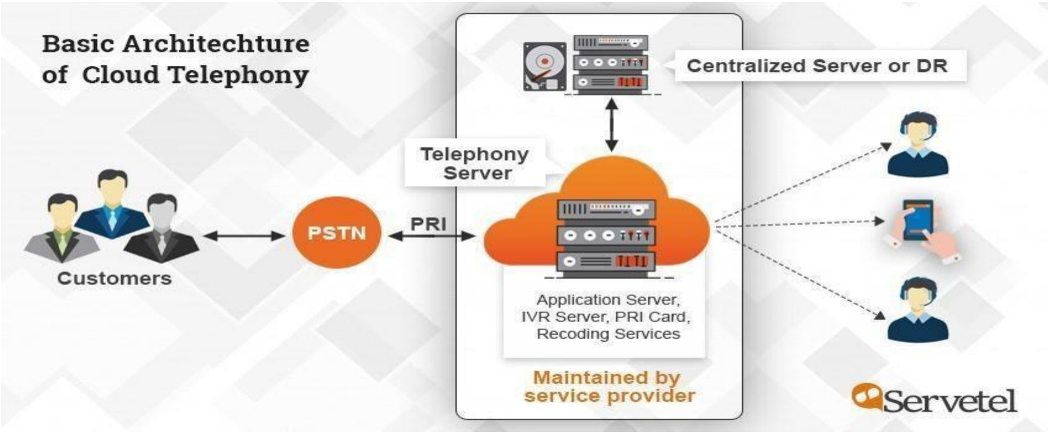

Figure 2 : Cloud Telephony Working Concept

At the point when a client dials the call — it is first diverted to Hosted Public Branch

Exchange (PBX) on the cloud. This exchange of information is made through a Primary Rate Interface (PRI) line which is associated with the communication server with a PRI card (PRI line is a media transmission standard utilized on an ISDN (Integrated Services Digital Network) to convey voice, video, information, and other system benefits between the client and the system).

This facilitated PBX (which is given by the specialist co-op) stores the important telephone numbers and data to choose steering structure for each call. All your office telephone augmentation is associated with the facilitated PBX, so every telephone call arriving on PBX chooses its best appropriate operator in the wake of perusing through these the set needs in the framework.

Call exchange between the operators is nearly a quicker procedure as every one of the specialists are associated with a similar umbrella of facilitated PBX.

2.1.3 Why Should Business Go for Cloud Telephony?

Organizations can use various highlights to secure their regular procedures and benefit making, yet there are not many key highlights of the cloud communication administrations which makes it the best correspondence mode for the business

Phone message

Phone message is a strategy for putting away voice messages electronically. This element is especially gainful in the situations where every one of the specialists are occupied on other line or are inaccessible because of different reasons. The message is put away as a voice recording on the neighborhood stockpiling — which is later recovered by the specialists and reacted.

1. Call Forwarding

This component permits the entrepreneur or specialists to stay accessible for the calls— on the off chance that they are not physically show in the workplace. You can use the cloud administrations to advance the approaching calls to your number. Store a rundown of sending numbers in the framework to consequently advance your calls to those numbers as indicated by the set need. The guest stays immaculate by the entire sending process.

2. Automatic Attendant

Intuitive Voice Response (IVR), is a shrewd component which causes the business to naturally visit/answer the call with a recorded welcome message. The programmed specialist replaces the manual assignment of visiting and exchanging the calls. This element spares a ton of human costs which used to be spent on enlisting a committed asset to go to the approaching calls.

3. Call Recording and Monitoring

When the business is set up, the subsequent stage lies in the support which must be accomplished by continually observing your procedures. Call recording is one such office which enables business to record each approaching and active call contingent on the business necessity. This element helps both the clients and the business in keeping up the credibility of the discussion and the business forms.

4. Ease of Use with Desktop, Web and Mobile Applications

To use all the above highlights, an interface is given by the cloud specialist co-op. To extend the utilization and adaptability of utilization, this interface is effectively open through the work area and web stages. Many specialist organizations have now created applications for simple openness of administrations for the operators. With the versatile applications, specialists/administrators can whenever access and screen their approaches their fingertips.

5. Integration with Other Systems

There are many cloud communication virtual products which increases the value of cloud communication forms by giving the joining adaptability. You can incorporate your CRM, industry-explicit administration framework or other programming into your cloud communication frameworks to expand your use.

2.2 TRAI guidelines

Specialized ASPECTS OF INTERNET TELEPHONY

A. Definition and Meaning

Global Telecommunication UnioN Telecommunication Standardization Sector (ITUT) Study Group 2 (SG2) issued the accompanying clarification of the expression "IP Telephony": "IP is a shortened form for Internet Protocol. It is an interchanges convention created to help a bundle exchanged system. The convention has been created by the Internet Engineering Task Force (IETF). IP communication is the trading of data basically as discourse that uses a system known as Internet Protocol."

The a wide range of 'flavors' of IP Telephony give, to shifting degrees, elective methods for beginning, transmitting, and ending voice which would some way or another be conveyed by the open exchanged phone organize (PSTN). While the development of IP Telephony is frequently connected with the ascent of the Internet itself, value that IP Telephony regularly does not include the open Internet by any means – but instead just its hidden innovation, the Internet Protocol suite.

All around, there are fundamentally two strategies for voice transmission over IP systems; in light of sort of IP organize utilized. At the point when voice is transmitted over open Internet, it is named as Internet Telephony. Correspondingly, when voice is transmitted over oversaw IP systems, it is named as Voice over IP (VoIP).

Customary communication utilizes circuit-exchanging innovation while VoIP utilizes bundle exchanging. In circuit-exchanged systems, arrange assets are committed to the circuit amid the whole discussion, and the whole data pursues the equivalent devoted way. In parcel exchanged systems, the message (voice information) is broken into bundles, every one of which 8 can take an alternate course to the goal, where the parcels are recompiled into the first message. In that capacity, bundle changing should be a substantially more productive and financially savvy method for sending voice messages and information.

IP Telephony is utilized as a conventional term for some methods for transmitting voice, fax and related administrations over parcel exchanged IP based systems. Web Telephony is a type of IP communication, which utilizes Internet Protocol (IP) for transmitting IP parcels over Internet cloud. The fundamental advances associated with beginning an IP phone call are transformation of the simple voice sign to computerized position (parallel information) at endorser premise itself and pressure/interpretation of the information into IP bundles for transmission over the Internet. The procedure is turned around at the getting 9 closes.

The correspondence ordinarily happens continuously. Along these lines, the fundamental distinction between IP Telephony and typical communication is that while in ordinary communication, circuit-exchanging innovation is utilized (especially in the entrance arrange), though IP Telephony depends on parcel exchanging innovation. According to display administration models three primary sending situations for IP Telephony are conceivable:

PC-to-PC Internet Telephony:
In this situation, the calling and called parties both have PCs or comparative gadgets that empower them to associate with the Public Internet (allude Figure1). Both end-clients can build up correspondence (Data or voice correspondence) just by earlier time obsession, as they must be associated with the Internet in the meantime and utilize perfect programming. Directly, huge quantities of VoIP applications are accessible on Internet to make PC-to-PC Internet Telephony conceivable.

The Internet Service Provider (ISP's) job in such situation is restricted to giving access to the Internet. The ISP organize is straightforward to such application utilized by the supporters. The voice application utilized by the client is straightforward for the ISP, which takes no particular measures to ensure the nature of the voice administration however only of the utilization of a voice application by means of the Internet.

Today, PC proportional gadgets like tablets or cell phones are accessible, which can likewise run such programming supporting Internet Telephony. This kind of Internet Telephony is allowed under existing ISP permit. It is likewise viewed as a 'Preposterous' (OTT) application administration.

For this situation, the calling and called parties are the two supporters of the open phone arrange (fixed or portable) and utilize their phone set for voice correspondence in the typical manner.

There are two techniques for conveying by methods for two normal phone sets by means of an IP based system. At least one media transmission players have built up passages that empower the transmission of voice over an IP based system in a manner that is straightforward to phone clients. What we have for this situation isn't the Internet however an "oversaw" IP arrange, for example a system which has been dimensioned in

such a route in order to empower voice to be conveyed with a satisfactory nature of administration.
Such sorts of calls are not Internet Telephony calls. Figure 3 below illustrates such a scenario.

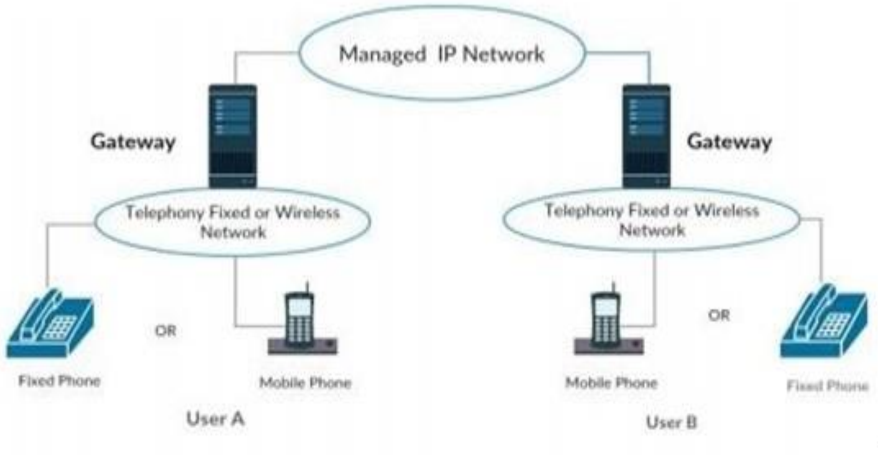

Figure 3 : Phone to Phone over IP

In this situation, the portals and oversaw IP system could have a place with various players, contingent upon whether we are taking a gander at:

a) The absolutely inner utilization of VoIP inside the system of a solitary phone administrator, which possesses and deals with the whole task, taking care of the two clients An and B;

b) The arrangement of a long-remove voice administration by a long-separate administrator utilizing VoIP innovation (clients An and B for this situation having a place with various systems), where case the entire task has a place with and is overseen by such a long-remove administrator.

The present administrative structure takes into consideration this sort of communication and is typically named as VoIP on the grounds that the open Internet isn't coming into picture.

PC-to-Phone or Phone-to-PC Internet Telephony: In this situation, one of the clients has a PC by which he interfaces with the Internet by means of an entrance arrange and an ISP while the other client is an 'ordinary' supporter of a fixed or cell phone network(refer Figure 3).

PC-to-Phone In this situation User An absolute necessity utilize the administrations of an ISP to get associated with the Internet by means of the system of his ISP. When associated, he utilizes the administrations of an Internet Telephony specialist co-op (ITSP) working a door which guarantees access to the point that is nearest to the phone trade of the called supporter. It is this entryway that will deal with the calling gathering's call and all the flagging identifying with the phone call at the called party end.

Client A runs programming (Dialler) introduced at his PC (Equivalent gadget) to dial the quantity of the client B. It ought to be noticed that the ITSP gives a single direction PC-to-telephone administration and does not oversee endorsers accordingly; truth be told, the PC supporter utilizes the ITSP's administrations exclusively for active calls. It 12 ought to likewise be noticed that the ITSP has an overseen IP organize, along these lines guaranteeing a specific nature of administration for voice similarly as the door nearest to the called supporter, and that the ITSP additionally deals with the interconnection with the last's phone administrator. The arrangement of Internet access and arrangement of Internet Telephony administration might be finished by a similar specialist co-op or by various specialist co-ops, which means in this manner that the ISP and ITSP might be same or unique. Telephone to-PC For this situation, the calling party is the communication client and the called party is the PC or equal gadget client. Since a communication client can basically dial an E.164 number to achieve the called party, at that point by one way or another the PC client ought to have an E.164 number by an IP communication administrator.

Chapter 3

Methodology

3.1 Introduction

A system port is a procedure explicit or an application-explicit programming develop filling in as a correspondence endpoint, which is utilized by the Transport Layer conventions of Internet Protocol suite, for example, User Datagram Protocol (UDP) and Transmission Control Protocol (TCP).

A particular system port is distinguished by its number normally alluded to as port number, the IP address in which the port is related with and the sort of transport convention utilized for the correspondence.

A port number is a 16-bit unsigned whole number that ranges from 0 to 65535. Any systems administration procedure or gadget utilizes a particular system port to transmit and get information. This implies it tunes in for approaching bundles whose goal port matches that port number, or potentially transmits active parcels whose source port is set to that port number. Procedures may utilize various system ports to get and send information.

You can have a sum of 65,535 TCP Ports and another 65,535 UDP ports. At the point when a program on your PC sends or gets information over the Internet it sends that information to an IP address and a particular port on the remote PC and gets the information on a normally arbitrary port without anyone else PC. In the event that it utilizes the TCP convention to send and get the information, at that point it will interface and tie itself to a TCP port. On the off chance that it utilizes the UDP convention to send and get information, it will utilize a UDP port

														6 5 5 3 1	6 5 5 3 2	6 5 5 3 3	6 5 5 3 4	6 5 5 3 5

Figure 4: IP address with Port

3.2 PRI Lines

PRI Line is a Digital Trunk Line that is given by a Telecom specialist co-op which empowers clients behind a PBX to make 30 synchronous approaching/active calls utilizing a solitary line.

- It is a type of ISDN (Integrated Services Digital Network) line whic h is a media transmission standard that empowers customary telephone lines to convey voice, information and video traffic, among others.

• A PRI circuit comprises of two sets of copper lines ending on a modem from a specialist co-op premises to the client premises. It utilizes multiplexing/demultiplexing systems to convey more than one divert in a solitary circuit. There are two normal types of PRI lines
E1 (which convey 30 directs in the two sets of copper lines, normal in Europe, India) and T1 (which convey 23/24 diverts in the two sets of copper lines, regular in United States).

• Each direct in a PRI line gives 64 Kbps to information transmission.

Points of interest of PRI Lines:

Not at all like the crude methods for correspondence utilizing the BRI lines which is an abbreviat particle for Basic Rate Interface with only 2 channels to utilize, the PRI lines with 30 channels have some more noteworthy focal points also.

In the event that thirty separate simple trunks are taken rather than one PRI line: -

• The cost of ending all the thirty simple trunk lines winds up higher than ending one PRI line.

• There would be thirty rentals to be paid rather than one merged lower rental for a PRI line.

• Some simple trunks may be utilized increasingly (uneven conveyance of calls) and a few lines might not have even crossed the free calls limit.

• Terminating 30 simple trunks in a PBX additionally requires all the more free spaces/cards than the one opening generally involved by one or even two PRI trunk cards.

1. Direct Inward Dialing: For each PRI line, the specialist organization would give more around 100-500 numbers which can be utilized by outcasts to call the expansion straightforwardly, rather than experiencing the PBX Auto-orderly.

2. Caller ID: Since every one of the augmentations have their own number, this interesting number will be shown in the telephones that they are calling to. Some call focus applications depend on the one of a kind guest ID number for separation of administrations.

3. It is conceivable to offer both voice and information in the PRI line. Some specialist co-ops have dynamic contributions where information is transmitted in every one of the channels that are free (not involved by voice) at that given purpose of time.

4. Call chasing (Where the consider arrives in any channel that is free, rather than the called number explicitly – For instance, if there is one load up number yet various individuals are bringing in the meantime and still a channel is dispensed to them with simple lines, on the off chance that one number is occupied, they have to bring

in another number physically) is conceivable naturally with a PRI association, yet for the simple trunks this office should be stretched out by the specialist organization and empowered on the PBX, including extra expense on occasion.

5. PRI lines can be utilized for voice network, information availability, video conferencing, faxing, and all the above should be possible at the same time as well (on various channels).

6. PRI lines are start to finish computerized lines and consequently the clearness is vastly improved than simple trunk lines.

7. Since they are advanced lines, PRI lines are increasingly solid and inconvenience shooting is additionally simpler with them. They are for the most part on a fiber center ring and henceforth there is some repetition.

8. It is more diligently to take advantage of computerized lines and tune in to the discussions.

9. There are adaptable charging alternatives accessible with the vast majority of the PRI specialist organizations. The charging can be unified or conveyed (office insightful, and so on.).

10. PRI lines set aside lesser effort to build up calls then simple trunk lines.

11. Some specialist organizations offer adaptable plans where rather than the full 30 channels, they give and charge to just 20 channels and so on. This makes PRI lines increasingly efficient for littler organizations.

PRI Cards

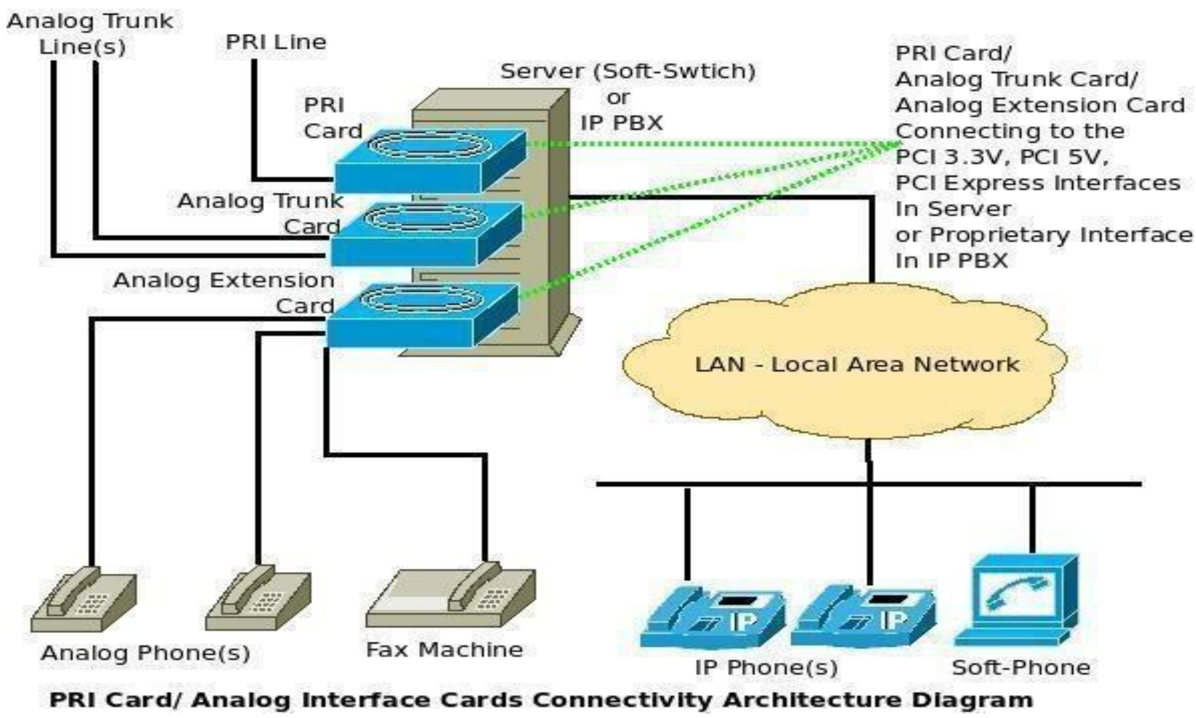

Figure 5: PRI card

A PRI Card is utilized to interface PRI lines to IP PBX/IP Telephony Server with the goal that all the IP Phones/Analog telephones (augmentations) can make active calls or get approaching calls utilizing it.

In an Analog/IP PBX, one needs to obtain a specific PRI Card that fits into one of the unfilled openings of the PBX so as to associate the PRI Line. This Card is for the most part restrictive to the particular PBX merchant. Yet, with Soft-Switches (that run utilizing standard server equipment), one can buy a conventional PRI Card to interface/associate the PRI line.

The PRI lines can be introduced utilizing the PCI space of the normal motherboard of a Central Processing Unit or the CPU. PCI Slot is a Peripheral Component Interconnect (PCI) opening is an associating device for a 32-bit PC transport. These devices are incorporated with the motherboard of PCs and gadgets so as to take into account the expansion of PCI gadgets like modems, arrange equipment or sound and video cards.

The conventional PRI Cards are for the most part merchant impartial and they are embedded into PCI 3.3V/PCI 5V/PCI Express (vacant) Slots in the server. There are distinctive PRI Cards for each kind of PCI interface.

One PRI Card can have 1,2 or 4 Slots to associate with 1,2 or 4 PRI lines. Some PRI Cards accompany reverberation undoing modules (at additional expense) so as to decrease the reverberation created when Digital Signals are changed over to IP and the other way around. It is prescribed to purchase PRI Cards alongside Echo Cancellation modules.

3.3 Session Initiation Protocol

SIP (Session Initiation Protocol) is a flagging convention, generally utilized for setting up, interfacing and separating correspondence sessions, regularly voice or video brings over the Internet. Taste is an institutionalized convention with its premise originating from the IP people group and as a rule utilizes UDP or TCP. The convention can be utilized for setting up, adjusting and ending two-party (unicast), or multiparty (multicast) sessions comprising of at least one media streams. Adjustments can incorporate changing IP addresses or/or ports, welcoming more members, and including or erasing the media streams.

Taste is an application layer control convention that underpins five pieces of making and halting interchanges. It doesn't give administrations; in this way, it acts with different conventions to give these administrations, one of which is ordinarily RTP that conveys the voice for a call. The five pieces of setting up and ending calls that SIP handles are:

- User Location: Determines where the end framework is that will be utilized for a call.

- User Availability: Determination of the ability (accessibility) of the called party to participate in a call.

- User Capabilities: Determination of the media and parameters which will be utilized for the call.

● Session Setup: Establishment of the session parameters from the two gatherings (ringing).

● Session Management: Invoking the administrations including exchange, end, and adjusting the session's parameters.

SIP depends on a solicitation/reaction exchange model where every exchange comprises of a solicitation that conjures a specific technique or capacity on the server and at any rate one reaction.

Taste normally sends these messages in UDP (User Datagram Protocol) on port 5060, with 5061 utilized for a second line on a two-line ATA. Incorporated into the welcome, when setting up a call, are parameters depicting precisely what structure the sound or video will utilize.

RTP will ordinarily be carried on a port from a scope of ports, in all probability somewhere in the range of 10,000 and 20,000, which are then allocated to every endpoint after an arrange and acknowledgment of a specific port on each side.

Taste is likewise used to keep the ATA gadget enrolled with the supplier by correspondence with their server. This happens when the ATA gadget or IP-telephone is first connected and a short time later normally on a preset interim. Data that is passed incorporates the IP address where the ATA can be found and other data that keeps the server refreshed with any data that may have changed since the last enrollment.

As appeared certain data is sent alongside an Invite which begins the procedure to set up a call session. That call session is normally voice sent by means of RTP (Real-time Transport Protocol). The Real-Time Transport Protocol (RTP) is an Internet convention standard that determines a path for projects to oversee constant transmission of media information, with VoIP is normally voice, yet could be video, too.

Chapter 4

Implementation of Servetel

4.1 Introduction

Servetel takes its PRI lines from different operators (Airtel, Vodafone etc.) and implements the call flow through SIP unlike the traditional PRI system of the PSTN network.

To implement the portal for Servetel, Asterisk an open source framework for building communications applications has been used. It runs on Linux, BSD and OS X and allows you to build a PBX given sufficient Linux and telephony know how. Asterisk does voice over IP in four protocols and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, IAX, SIP, H.323 (as both client and gateway), MGCP (call manager only) and SCCP/Skinny. Check the Features section for a more complete list.

Asterisk needs no additional hardware for Voice-over-IP, although it does expect a non- standard driver that implements dummy hardware as a non-portable timing mechanism (for certain applications such as conferencing). A single (or multiple) VOIP provider(s) can be used for outgoing and/or incoming calls (outgoing and incoming calls can be handled through entirely different VOIP and/or telecom providers).

The Servetel Panel is equipped with all such features with a backend governed by Asterisks.

The user is given a panel to operate the features included in it and can login through the credentials created by the system upon verification of the shared contact information of the client.

Glance at the Portal

Servetel demo page static link: https://www.servetel.in/demo/

The Servetel portal has two kinds of accessibility options.

1. The User Interface
2. The Admin Interface

The Servetel portal is accessible to users after they login through the unique username and password for their account. The same is created using the 5-digit client ID and the email they provide during authentication. The password is encrypted with 128 bits. This means that to crack the code the computer needs to run 2^{128} different combinations in order to break the key.

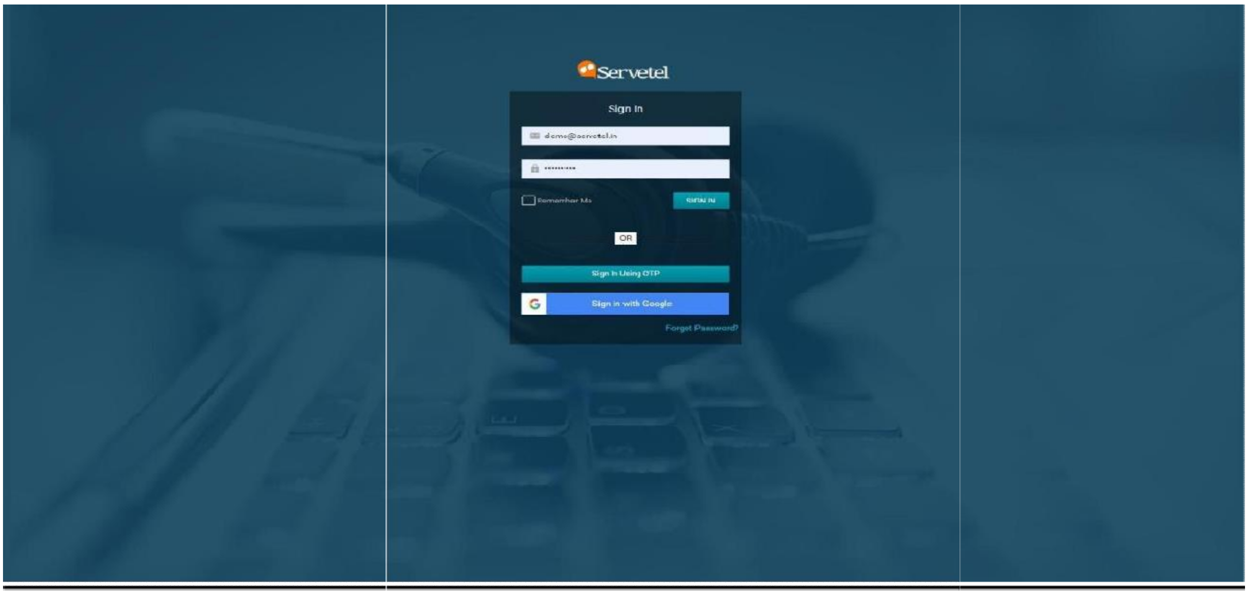

Figure 6: Login Page

The facility of creating their own account is provided to the clients wherein they can get the system generated credentials immediately after verification of contact information via OTP on email and mobile phone using the link shared below

Servetel sign-up trial link: https://customer.servetel.in/signup-trial

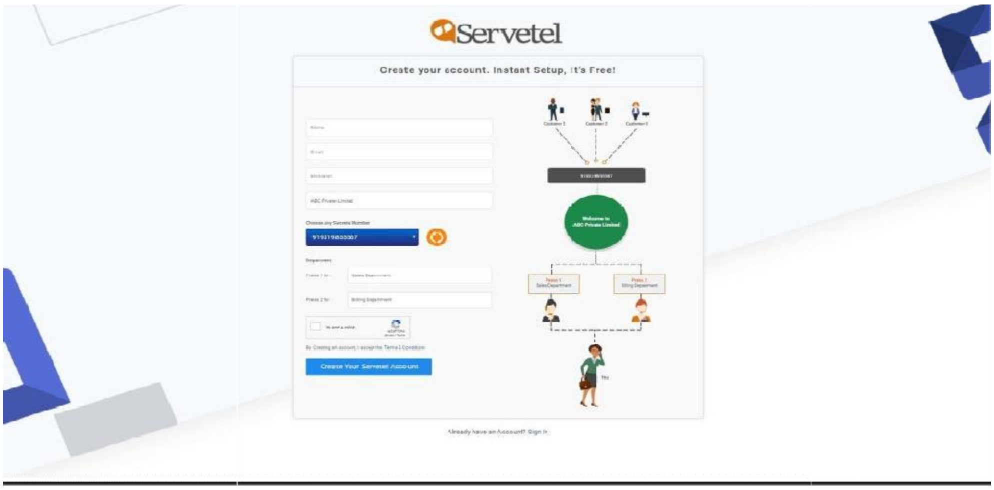

Figure 7: Trial setup Page

The portal has a user-friendly interface where all facilities of the panel are one click away. The same is shared below.

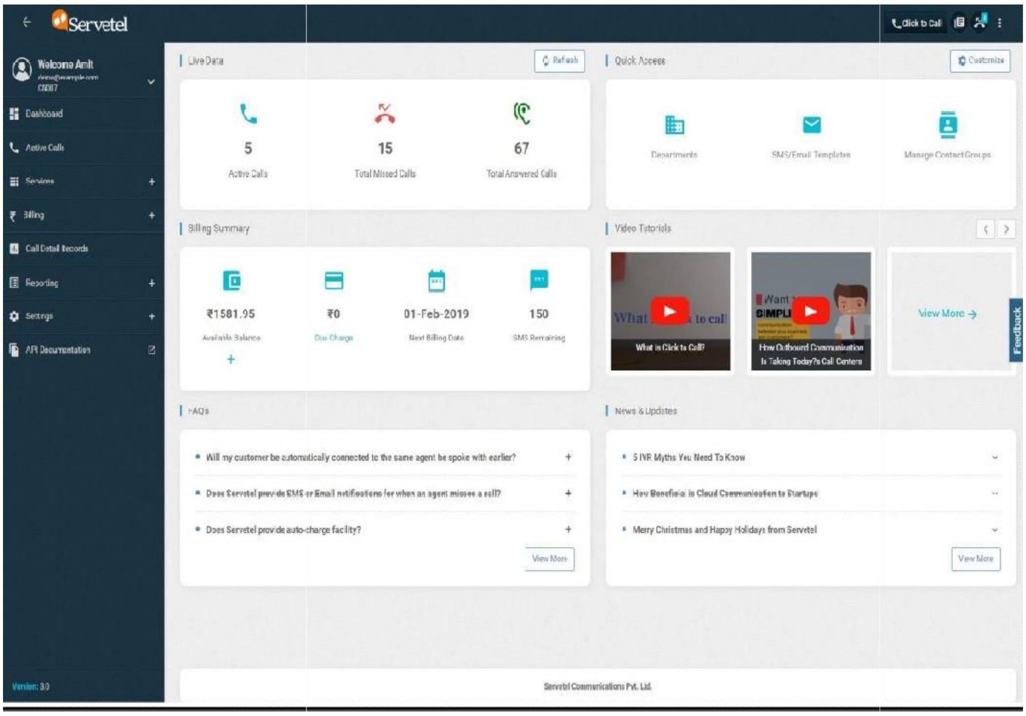

Figure 8: Customer Main Portal

There are various other pages of the panel only accessible to the admin team or us who can monitor every account and make changes in the as required. The accounts are monitored over fraudulent use as well to maintain the decorum of technology over a provided service by Servetel.

The admin panel can monitor the call traffic from all running customer accounts to maintain system stability and agility. Each call that lands on the server using the PRI line is recorded through the LINUX codes of Asterisks. The call can be monitored on real time basis and the status of the same can be altered manually as well if required.

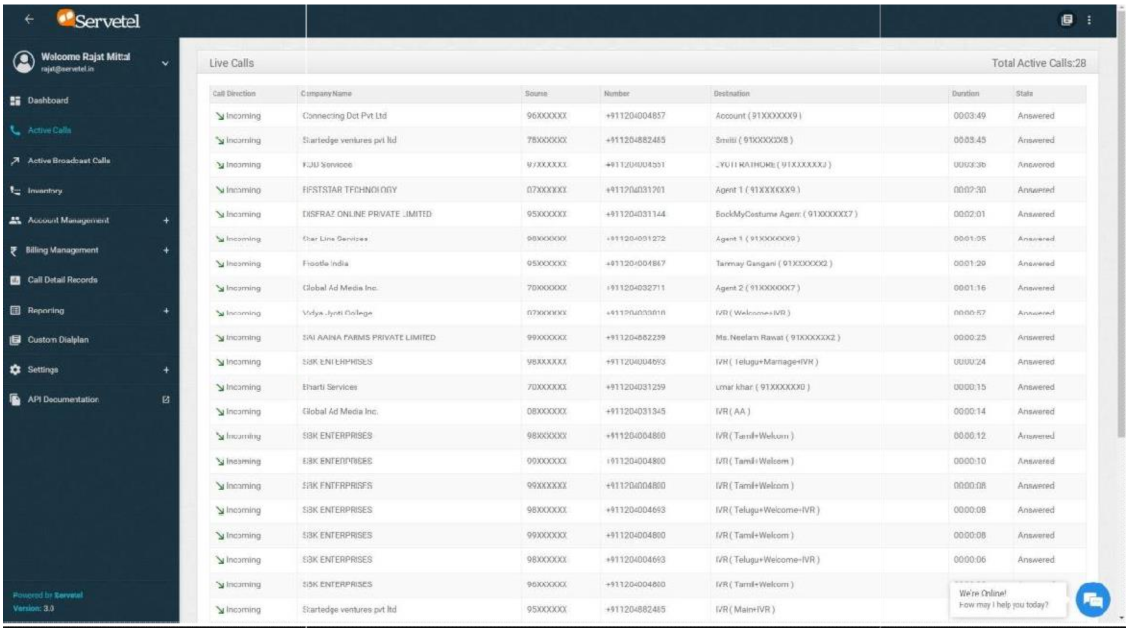

Figure 9: Customer Main Portal: Active calls

The panel then can help us view all the Call Detail Records which a confidential piece of information for any firm and hence, even the admin side of the panel hides the number of the caller to maintain Privacy of data in compliance with the GDPR or General Data Protection Regulations.

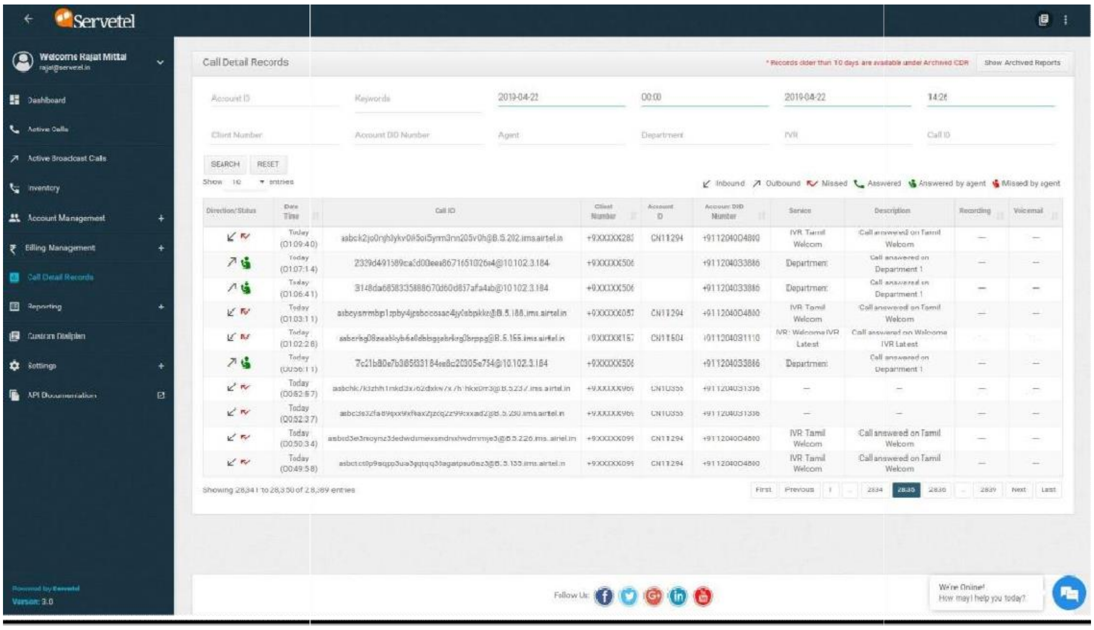

Figure 10: Customer Main Portal: CDR

The admin access allows viewing each account for their activity and the frequency of usage. The data collected helps in further research and R&D of the panel for improving performance and the customizations according to the needs.

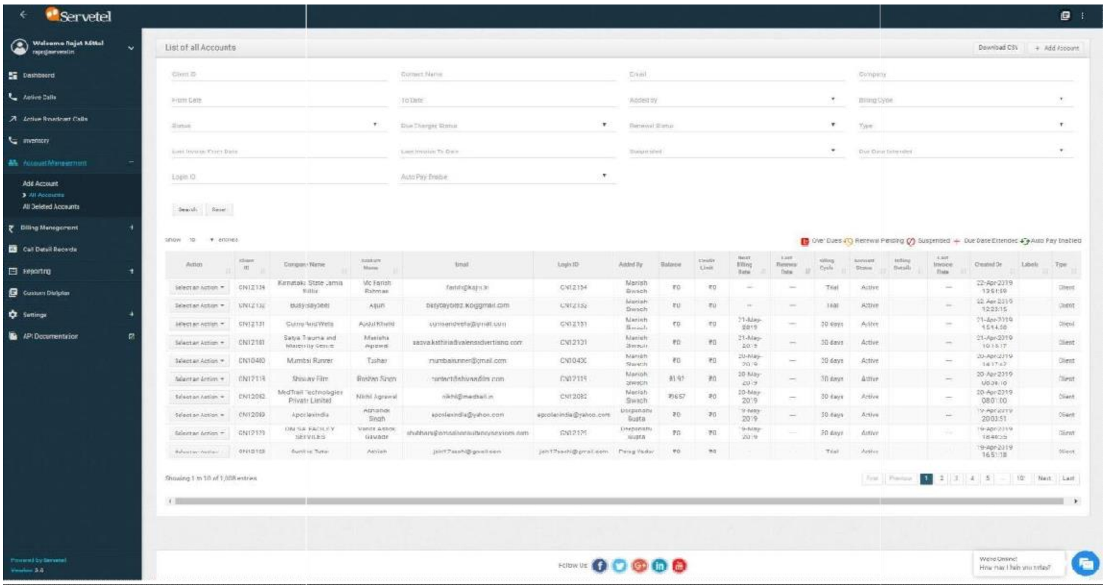

Figure 11: Customer Main Portal: Admin/Reseller accounts access

Each number procured by the company is maintained in automated inventory which changes the status of the number as a client selects it for use. The numbers can be ported to other provider or can be ported to us from other providers using the No Objection Certificate or the NOC. These numbers are allocated uniquely. If the number is released by one user in case of no future use, the number goes into quarantine status where the number is then given back to the vendor to avoid clash of contact information of the two customer companies for Servetel.

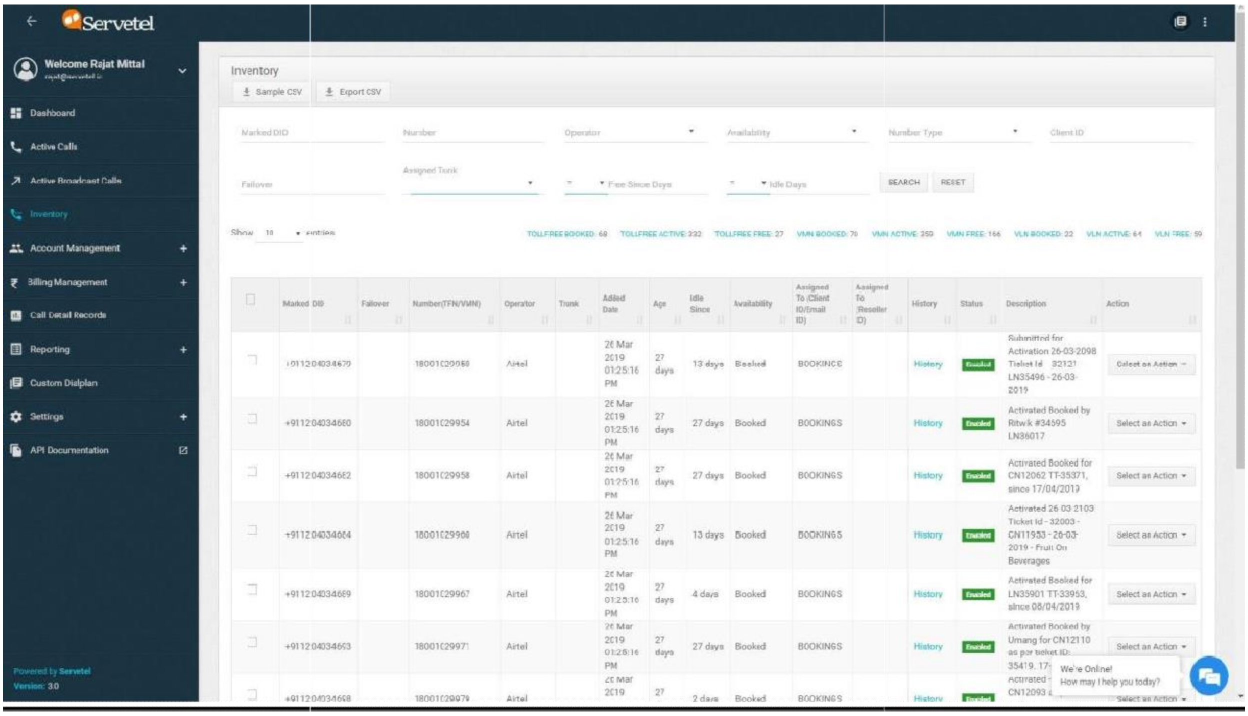

Figure 12: Customer Main Portal: Admin Inventory

There have come multiple cases where businesses feel the need to integrate their calling solutions with their existing interface of either a website or a mobile application. Servetel has the dedicated modules for API or Application Programming Interface integration into any website or mobile application. The Amazon interface is one such example.
https://servetel.readme.io/docs

Chapter 5

The Real PBX!

5.1 Introduction to Hosted PBX

Facilitated PBX, otherwise called Virtual PBX and Cloud PBX is an exceptionally versatile, IP-based correspondence administration that courses brings over the Internet and permits multi-area organizations to streamline their correspondence and lessen the capital consumption.

Virtual PBX arrangement utilizes a solitary terminal to deal with all phone business lines without making any change in the current setup. There is no compelling reason to build up another physical set up, virtual PBX administration offers a fitting and-play framework that encourages you structure modified call steering answer for guiding calls from any area to any division or specialist's augmentation whether it is an individual portable, work area telephone or work area. Versatile clients can take influence of cell phone applications to supplant cell phone's ID with organization's guest ID and in-house workers can utilize VoIP work area telephone or softphones to make and get calls by means of cloud PBX.

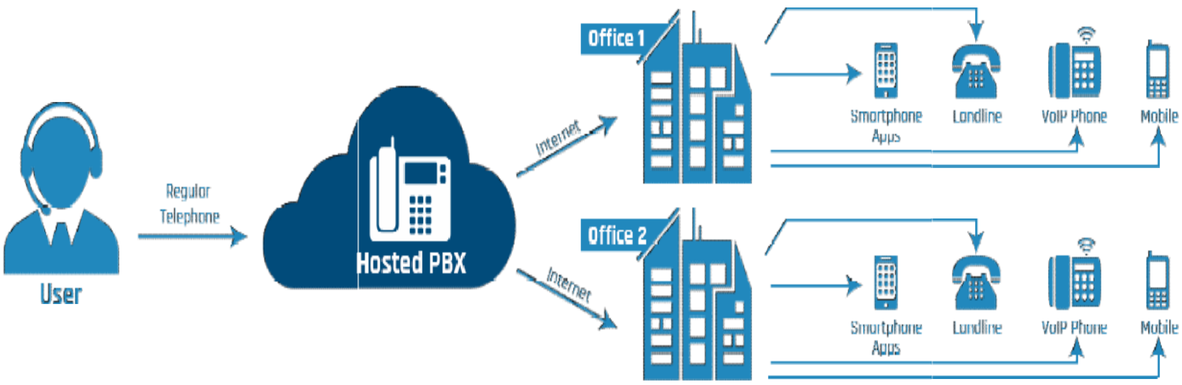

Figure 13: PBX working

5.2 Features of PBX

The Real PBX's Hosted PBX service is an all-in-one solution to meet all the communication needs without having to incur installation and maintenance cost and operational inefficiencies of the traditional hardwired phone.

Calling Features

- Call Blast +
- Call Return +
- Call Hunting +
- Call Waiting +
- Call Parking +
- Call Transfer +
- Call Blocking +
- Repeat Dial +
- Call Recording +
- Unlimited Calling +
- Call Forwarding +
- Call Conferencing +
- Find Me/Follow Me +
- Group Call Pick-up +

Call Management & Reporting

- Caller ID +
- Dial by Name +
- Music on Hold +
- Unlimited Extensions +
- Do Not Disturb +
- Conference Bridge +
- Call Voice Tagging +
- Simultaneous Ringing +
- Call Distribution (ACD) +
- Call Queue Management +
- Call Routing (Time of Day) +
- Call Monitoring/Barge In +
- 3rd Party Conferencing +

Figure 14 : PBX features list

The sound sign are simple in nature which are first changed over into computerized information parcels and after that are transmitted over the rapid broadband system. At the getting side, the advanced sign are again changed over once more into simple voice signals. IP PBX calls can be made in 3 different ways:

By means of ATA – It is the least complex and most effortless approach to make VoIP PBX calls. ATA is a simple to computerized converter, which enables you to interface your standard telephone to your web association with use VoIP PBX. It changes over discourse signals into computerized information to be transmitted over the web.

By means of IP Phone – These are particular telephones that look simply like ordinary telephones, with a handset, catches, and support. Be that as it may, IP telephones use RJ-45 Ethernet connectors rather than standard RJ-11 telephone connectors. They have all the equipment and programming on board to interface your telephone straightforwardly to the broadband system to make IP PBX calls.

Through Computer to Computer – This is positively the least expensive and most straightforward approach to make business PBX calls. A few facilitated PBX specialist co-ops offer ease administrations and programming for this kind of PBX framework. Mouthpiece, speaker and rapid web association is all you need, and you are prepared to make calls.

All the previously mentioned highlights are given by The Real PBX through various entries which fill in as the interface between the clients and the specialist organizations and are setup over servers situated at Data Centers in different urban communities everywhere throughout the world.

5.2.1 Toll Free Forwarding portal

The Toll-Free Forwarding (TFF) Portal is provided by The Real PBX to the clients who want the calls landing on their Toll-Free Number to be forwarded to the Toll Free Numbers of their agent. The client charges the customer for the calls that they intake.

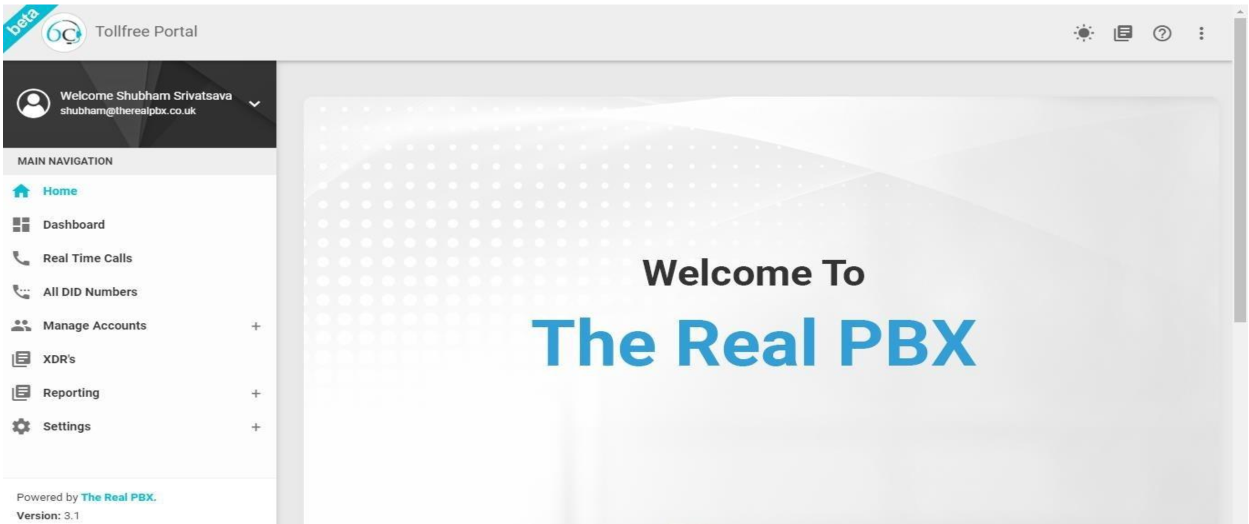

Figure 15: PBX Forwarding portal

We, as the admins of the panel provide the access of the panel to the client after creating their accounts on the portal and applying the desired routing of the DIDs to the server of the TFF portal.

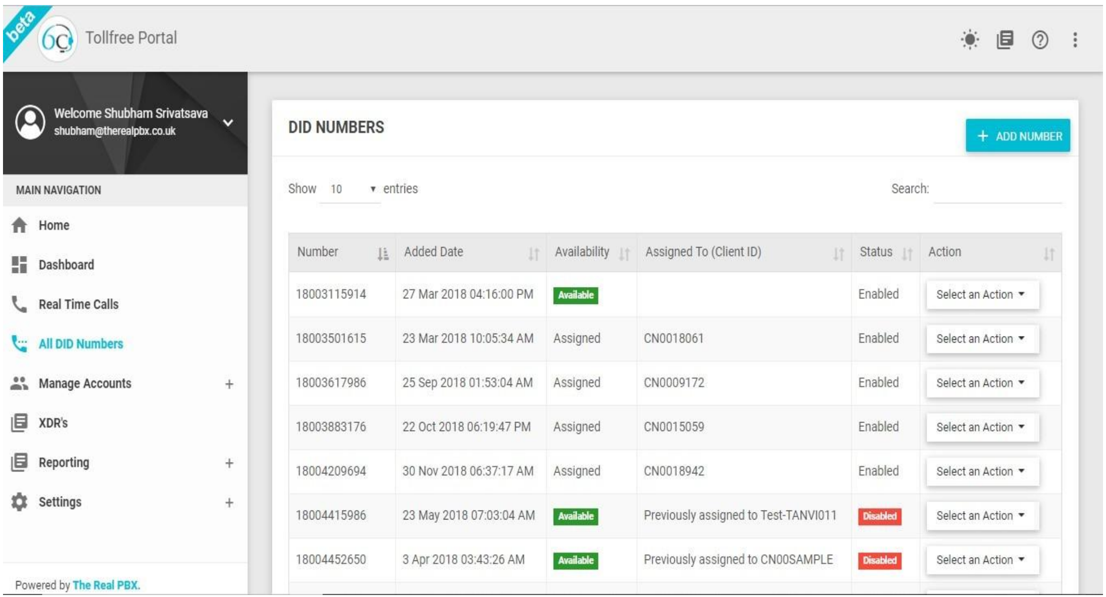

Figure 16: PBX Forwarding portal: Admin access DID inventory

The DID when after getting added to the portal must be enabled and assigned to the client so that he gets to access it. The number when added to the portal goes to the inventory which now has 2000 DIDs from which the calls get forward to over more than 10,000 destinations.

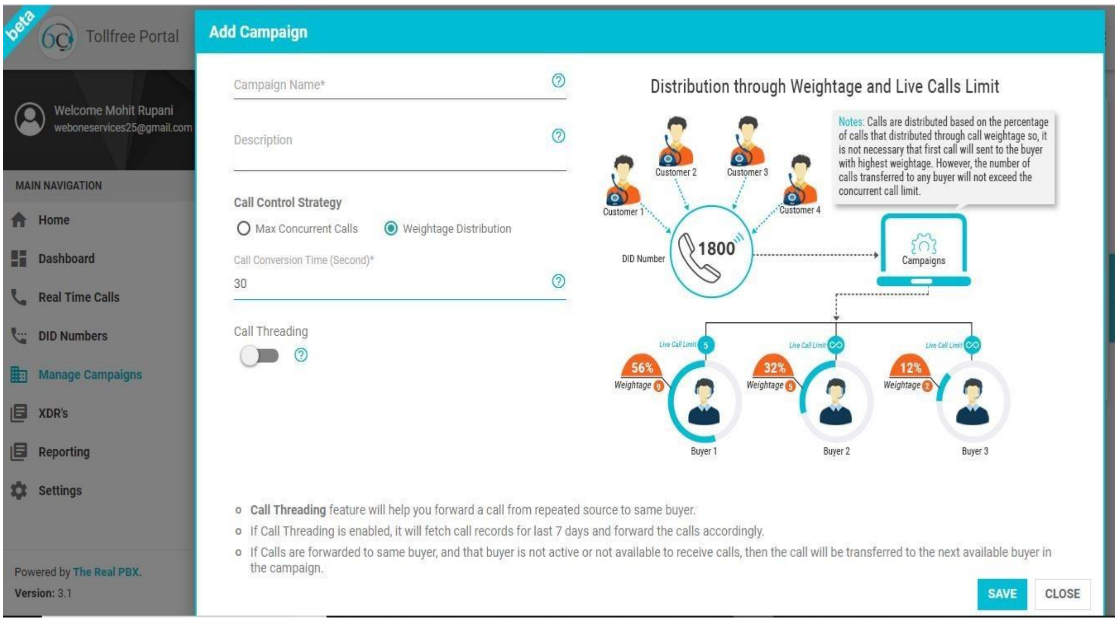

Figure 17: PBX Forwarding portal: Campaign Creation

After gaining the access of the portal, the client creates a campaign which decides the strategy through which the calls would be forwarded to the buyers. The 2 strategies which are provided by the portal are-

1. Max Concurrent Calls-Calls under this strategy would be forwarded to the most prioritized buyer until his/her max call limit is reached. Once the first buyer in on the maximum of his concurrent calls, the next calls are forwarded to the second buyer according to the same strategy.
2. Weightage Distribution-Calls under this strategy would get forwarded to the buyers according to the percentage weightage of the buyer set by the owner of the account.

The account owner has the rights to enable/disable a buyer or a campaign as per his wish. The owner also decides the “call conversion time” which determines if the call that has been forwarded to the buyer should be considered as successful or not.

The panel then can help us view all the Call Detail Records of all the calls that were forwarded to the buyers along with the destination and stats of the call. The call records are confidential piece of information for any firm and hence, even the admin side of the panel hides the number of the caller to maintain Privacy of data in compliance with the GDPR or General Data Protection Regulations.

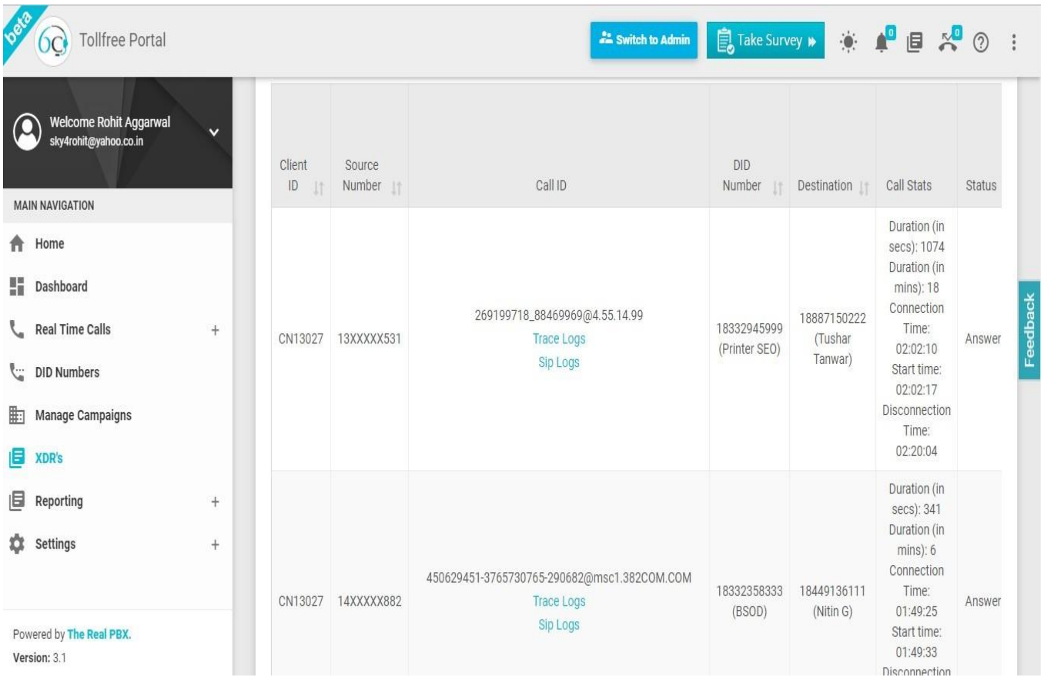

Figure 18 : PBX Forwarding portal: CDR

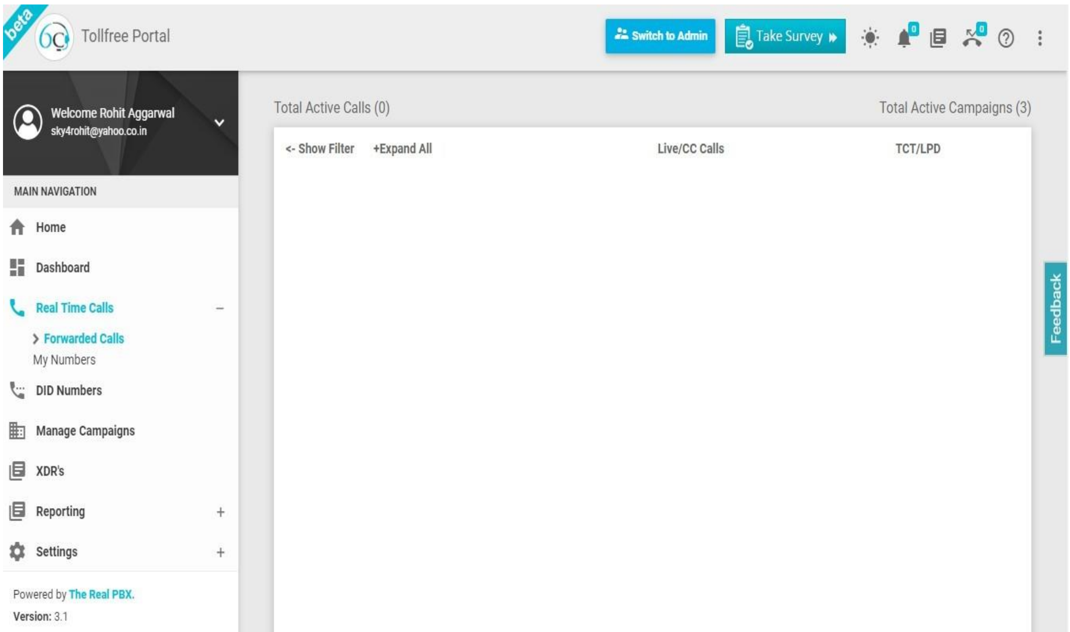

Figure 19: PBX Forwarding portal: Active calls

Under the live calls, the owner of the account can see all the calls that are being forwarded to the buyers under real time. The owner can also monitor under which campaign are the calls getting forwarded as well as the number of calls that are being forwarded to a particular buyer assigned to the campaign.

5.2.2 MyPBX Portal

The “MyPBX” portal is one of the latest portals developed by The Real PBX which has integrated all the past technologies and services and brought them under one roof.

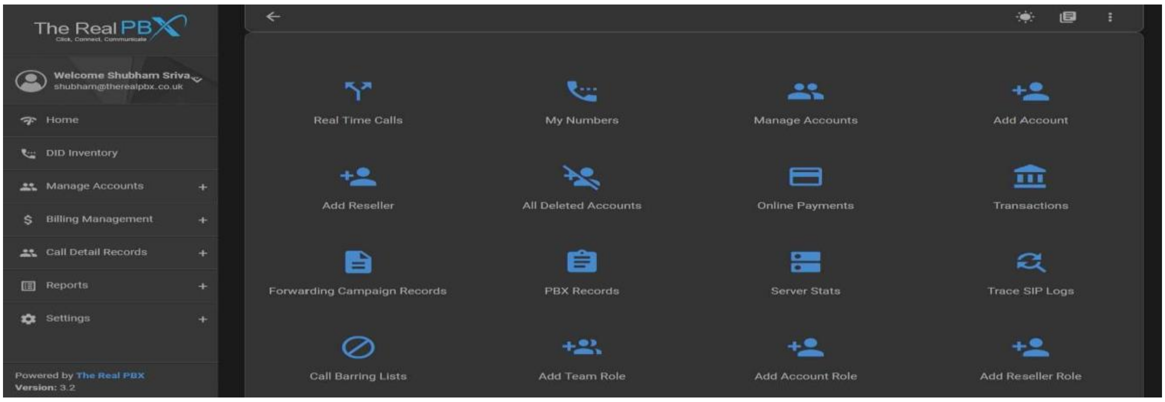

Figure 20 : PBX MyPBX portal

Apart from managing the calls, the customer can also manage their billing and payments made regarding the account.

Figure 21 : PBX MyPBX portal: Features

In addition to editing the destination and call strategies, the customer also has the access to features such as setting up the active hours, setting a time group for a particular destination, setting up an IVR that plays before the calls land on the desired destination as well as listening to voicemails left by the caller.

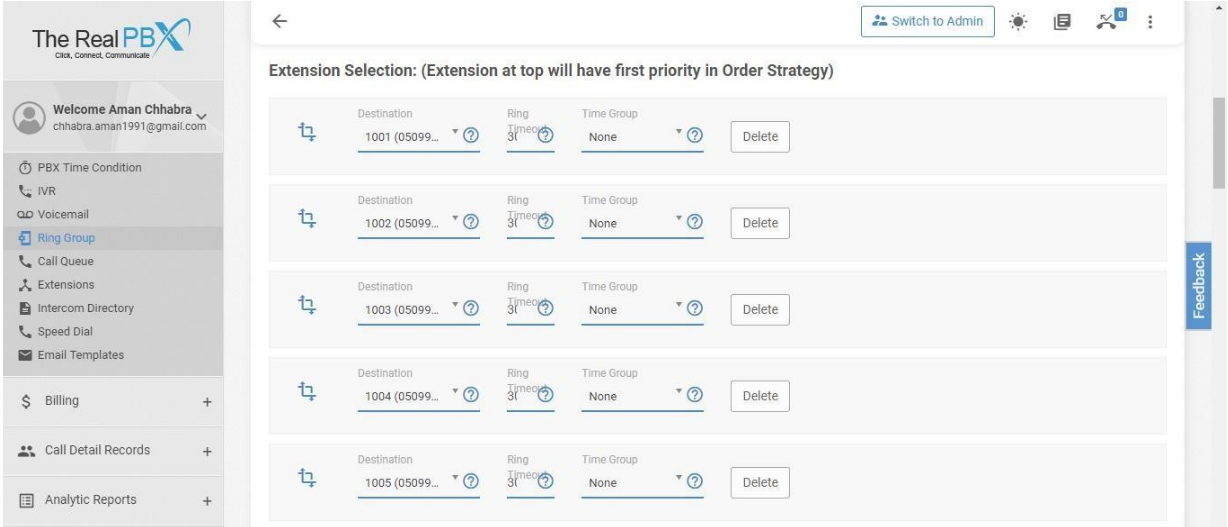

Figure 22: PBX MyPBX portal: Setup

The "Ring Group" feature allows the calls on a TFN to land on extensions, provided by us, as per the call strategy set under the same. The call strategies available are as follows:

1. Simultaneous
2. Round Robin
3. Random Order
4. Order By

The "Call Queue" feature, much like the Ring Group, has an added advantage. If a call queue has been set up for an account, the calls would remain in the queue and not get disconnected even if all the agents are busy. The call stays in the queue till an agent becomes available.

Figure 23 : PBX MyPBX portal: Real Time calls

Under the Real Time calls, the account owner can see the active calls in their ring group or call queue in the real time. He can also monitor the agents that are connected on a call as well as all the other detail of the same.

Figure 24 : PBX MyPBX portal: CDR filters

One of the most powerful tools that MyPBX has provided to its customers is the flexibility of the parameters through which he can check his call records. The customer can pull up the call reports number wise, extension wise, ring group wise as well as call queue wise.

MyPBX has provided the customers with one of the most user-friendly interface in the industry and its popularity and increasing rapidly in the market.

MyPBX Portal- Dialer- https://pbx826.therealpbx.com:8443

Basic flow of working of dialer is discussed below. (Due to the company privacy policy and security resones of customers information the pictures of portal were not allowed to be used. However to explain the working of this portal this main dashboard can be used as reference.)

Create User – The customer is provided with a login credential of admin role. Customer can make agents as per the plan purchased from the company and use the dialer.
Create campaign – For a particular objective of the customer supposedly a marketing initiative a campaign is created.

Create List- The leads that will be dialled from the dialler are converted into a specific format of excel ad are uploaded as lists on the main portal.
Create Inbound/dial plan – The Dial plan is created according to customer need. Main point to discuss in inbound routing/dial plan is the dial ratio. The dial ratio is the number of calls dialled per agent available. However, at a time only one call is connected.
Set carrier – The TFN (purchased from our company) is set to be the carrier and the calling is started automatically.

Basic Example of dialler can be seen in daily life in Dish TV channel offers which are informed from time to time whenever any new channel launches the calling is done according to the data (Leads).

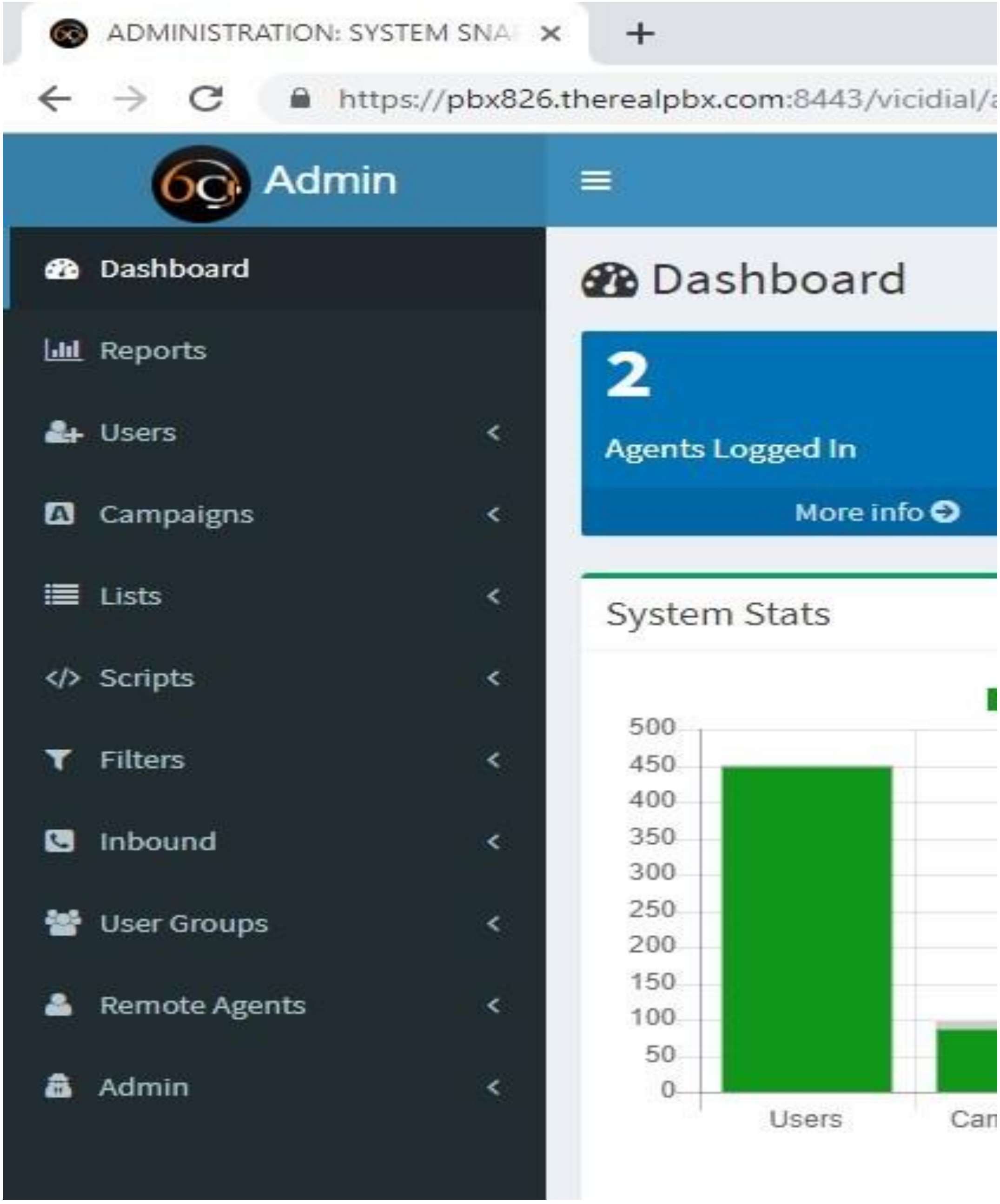

Figure 24 : PBX Dialer portal : Main Portal

Chapter 6

Elastix Portal - Work Flow

Let us try and unravel the mystery involved behind the demo that was provided at the start of our presentation.

Let us unlock the marvels of VOIP –

1. A US/UK number is procured.

2. This number is not associated with any sim card unlike in India and this can be hosted on either a server or a softphone.

3. Now a call is made from an India Mobile Number to this number (this mobile number must be configured on the portal of Elastix and the server) this is a missed call.

4. This call triggers the server and the server now makes a call from your mobile number (that was configured on the portal).

5. As I pick the call, the server would ask me to authenticate myself by asking for a preset password o log in. This is a feature to ensure two way authentication.

6. Now I get a dial tone on my phone indicating that I have logged into the server.

7. Now I press 2277 followed by the country code and the mobile number. Here I am instructing the server to make calls on my behalf using any display ID that I specify (can either be numeric, alpha-numeric or alphabetic.

8. The call will now keep ringing until the person being called picks up or I manually disconnect the call from my end.

9. I also have the freedom to specify Ring- Groups and Ring strategy as per need.

Chapter 7

Results and Discussions / Analysis

Case 1 – Servetel - The Automated Delivery Scheduling

One of the most concerning issues looked by the coordinations business is the expense caused because of client inaccessibility amid the conveyance. Computerized conveyance rescheduling is utilizing a telephone framework to reschedule orders. Utilizing Servetel, you can guarantee your client is accessible for conveyance before the conveyance specialist achieves the location. You can likewise reschedule the conveyance naturally to locate the following best time your client is accessible.

How it functions?

1. The official looks at the bundles to be conveyed around the same time from the center point. Registration data is caught in framework

2. Customer data associated with a conveyance is utilized by Servetel to trigger a robotized call

3. Servetel dials out the client's number and associates the client to the conveyance official

4. Based on the client's reaction, you can convey or re-plan the conveyance so as to spare time and cost.

Points of interest

1. Improves Operational Efficiency - Using innovation to affirm a client's accessibility prompts a more noteworthy exactness and proficiency inside an organization.

2. Reduce Costs - Checking the accessibility of the client preceding conveyance guarantees no cash is squandered because of client inaccessibility.

3. Enhanced Customer Experience - Ensuring a client gets their item at the time that works best for them prompts a superior client experience.

4. Dispute Resolution - All the calls directed by means of Servetel are followed and recorded. In this way, on the off chance that there is a debate, it very well may be settled effectively.

5. No More Manual Data Updating - No compelling reason to refresh the information got from clients on rescheduling physically. Servetel's framework naturally takes inputs entered.

Case 2 – MyPBX- The Automated Surveys and Feedback

Consistent ongoing criticism from the clients is urgent for any business. it has played a benchwarmer job in light of the fact that most organizations don't have the assets to call up the clients physically. Here's when robotized input and studies can work for you. Utilize computerized IVR calls to finish the criticism circle, SMS to acquire input with respect to quality, missed calls to enlist support, etc.

How it functions?

1. Calls are conveyed to the numbers from the database given by the client or activated naturally after an administration is checked finished by the work force.

2. An outbound call is activated and a predefined IVR call stream is played to the client.

3. Based on the information entered by the client, the keen IVR framework will dominate or the call gets separated.

4. The reactions entered by the clients can be traded to a Google spreadsheet with joining or sent continuously to an endpoint determined by the organization.

Favorable circumstances

1. Reduce Costs - There is positively no cost consumed by the client. For a business, robotized calls spare time and cash when contrasted with traditional review or input techniques.
2. Customised Questions - Integrate with your Customer Relationship Manager or CRM to pose customized inquiries dependent on your client's activities and to get a more prominent reaction rate.
3. Analysis - The data gathered can be easily processed, analysed and stored for future use.

Chapter 8

Conclusion

8.1 Summary and Conclusions

Communication is the backbone of the business industry. That is how people get to express their problem and get solutions for the same.

Cloud Telephony System simply transfers your data to cloud. It is easy and affordable solution that helps especially in boosting business. All the cloud telephony system is hosted by some third party over internet. These applications can be operated from "Cloud" irrespective of the physical boundaries.

It is cost-effective, secure, and trustworthy system that eases the burden and smoothens the flow of work. Ever since cloud technology has emerged, businesses have reaped numerous benefits and relished too.

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