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Intra-Networking (Establishing a Private Branch Exchange)

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***Abstract**: One of the major problems today in the enterprises of information technology is the transfer of information to the different sectors of the management. In this process it may face severe security issues and have to pay more for expenses equipment. By using the intranetworking systems, it can be negotiated. The main purpose of the intranetworking is to provide a medium for networking i.e., communication. By using this intranet technology communication is cheaper and easier than the previous equipment used. It also provides better internal communications and helps in creating better workplace for community where information storage is secured. Establishing a Private branch exchange (PBX) is a solution for this. This paper explains how a PBX can be established by any user, who requires it using an open source software called Asterisk.*

***Index Terms**: Asterisk, Communications, Intranetworking, PBX, VOIP.*

I. INTRODUCTION

These days most corporations have their own internet which can be accessible only to those within the company. The main aim of an intranet is it often provides a wide range of information and services to an organization that are unavailable to public, unlike the internet. Intranet can be accessed from internet but with restrictions as it is designed to a private space. For example, if in a company it is doubtful that an employee is able to access all the information regarding the company, so that employee can be restricted by giving him only access to certain areas of internet which is relevant to his job. Therefore, to communicate over intranet a PBX (Private Branch Exchange) is supposed to be established which uses the protocol named Voice Over Internet Protocol (VoIP) and an open source communications toolkit which powers IP PBX systems i.e. Asterisk[1]. A PBX is a telephone system within an enterprise that switches calls between enterprise users on local communication lines. VOIP (Voice Over IP) is the protocol that is used by PBX. It helps in transmission of voice and multimedia content over Internet Protocols (IP). This technology allows voice calls over internet instead of the regular telephone lines. In order to establish a PBX server with Ubuntu as the operating system and an open source software named 'Asterisk" for defining the users of an enterprise in the software. A PBX acts as the central switching system for phone calls within an enterprise. A traditional PBX is made up of two key elements: lines and stations. The lines, sometimes called trunks, are connections to the global public switched telephone network (PSTN). Stations are simply mobile phones or other endpoint devices like fax machines, modems, etc. The main aim of a PBX is to provide shared access to limited resources. Rather than having a separate communication lines for each phone, an enterprise could share a small pool of communication lines across a much larger pool of stations. Early telephone systems are some example of the switchboards i.e., they connect the calls from one department to another depending upon the type of the call they received. Today, a business phone system is much more than just a simple switch. Addition to the technologies like automated attendant, voice messaging, call queuing and multi-party conferencing can also be done using the intranet[5,6]. Outside connectivity is now available over the Internet in the form of SIP trunks or other VoIP services.

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VOIP:

The ability to channel calls through the internet's data network is known as Voice over Internet Protocol. VoIP makes voice and data networks converge: users had access to the internet, analog phone calls, and VoIP phone calls all through the same line. The VoIP system also allowed companies to integrate their communication with other applications. VoIP works on these sequential steps:

1. Analog telephone calls are converted to digital signals.
2. The digital signals are translated into Internet Protocol (IP) packets.
3. The IP packets are converted back to telephone signals, and received by a telephone on the other end.

By VoIP users could video chat, share data, instant message and many more [3].

Key IP PBX Features

VoIP Ready: The world is moving away from legacy PSTN lines and towards VoIP. Make sure your PBX can support IP stations (phones) and IP trunks (service). SIP is the current de facto standard, so don't buy a phone system that doesn't support it. Voice Messaging: Once upon a time, voicemail was an optional add-on. Today, it's table stakes. Look for PBXs that can forward voicemail messages to your email as attachments. If possible, look for IP phones that support visual voicemail. Mobility: Most businesses have at least some road warriors who spend much of their time out of the office. Make sure your PBX supports mobility features like Find Me / Follow Me, remote IP extensions and fixed / mobile convergence.

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Conferencing: To cut down on travel costs, teleconferencing is one of the best ways. Make sure your phone system has native support for true multi-party conferences (not just basic three-way calling). Reporting: If you can't measure it, you can't manage it. Make sure that the PBX you pick includes basic call history reporting features**.**

Asterisk as a PBX: Asterisk **-** software implementation of a telephone private branch exchange (PBX). It allows telephones interfaced with a variety of hardware technologies to make calls to one another, and to connect to telephony services, such as the public switched telephone network (PSTN) and voice over internet protocol (VoIP) services. The asterisk software includes many features available in commercial and proprietary PBX systems: voice mail, conference calling, interactive voice response and automatic call distribution. Asterisk supports several standard VoIP, including Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP) and H.323. Asterisk supports most SIP telephones, acting both as registrar and back-to-back user agent. Asterisk **-** the engine for a PBX system was created originally (in fact, many refer to it as the Asterisk PBX) and includes all of the components necessary to build a powerful, scalable business phone system. These include advanced features that usually cost extra on a commercial phone system: things like voicemail, automated attendant, call queueing, conference bridging, parking, paging, and intercom calling. Asterisk is a software implementation of a telephone private branch exchange (PBX). It allows telephones interfaced with a variety of hardware technologies to make calls to one another, and to connect to telephony services, such as the public switched telephone network (PSTN) and voice over Internet Protocol (VoIP) services. Its name comes from the asterisk symbol "*".

SIP (Session Initiation Protocol)

SIP is a signalling protocol used to establish a "session" between 2 or more participants, modify that session, and eventually terminate that session. It has found its MAJOR use in the world of IP Telephony. The fact that SIP is an open standard has sparked enormous interest in the telephony market, and manufacturers shipping SIP-based phones have seen tremendous growth in this sector. The SIP Protocol is text-based, and bears significant resemblance to the HTTP protocol. The messages are text-based, and the request-response mechanism makes for easier troubleshooting.

The actual data transmission is done by the Transmission Control Protocol (TCP) or the User Datagram Protocol (UDP) on layer 5 of the OSI model[7,8]. The Session Description Protocol (or SDP) controls which of the protocols is used. The SIP messages describe the identity of the participants in a call, and how the participants can be reached over an IP network.. Encapsulated inside the SIP messages we can sometimes also see an SDP declaration.

SDP (Session Description Protocol) will define the type of media channels that will be established for the session – typically this will declare which codecs are available, and how the media engines can reach each other over an IP network. In TCP/IP networks, one of the main protocols is TCP. Dealing with packets will be handled by IP protocol, whereas, connection establishment between two hosts and data stream exchange is enabled by TCP. Data delivery and order of packet delivery as sent by transmitter are guaranteed by TCP.

Programs that run on different computers on a network uses the Internet Protocol suite named as User Datagram Protocol (UDP). UDP a connectionless protocol is used to send short messages called datagram and are unreliable.

The parameters of streaming media communications can be best described using a Session Description Protocol (SDP). SDP is used for describing multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation.

II. Literature Survey

Until the late 1960's all the companies and businesses were dependent upon the public telephone networks to connect within the office or to call a co-worker in the office. Then soon the companies invested on the Switchboard PBX the first-generation PBX. The switchboard PBX consists of switchboards with cord circuits which were manually operated by the switchboard operators. In this process, a small number of phone calls from a large number of phone calls will be shared. The second-generation PBX was Automatic PBX, here all the process is automated without the help of the switchboard operators. The third generation PBX was Electro-Mechanical Switching PBX, in this type of PBX an electric pulse triggered the switches to physically flip until the call was fully connected. The fourth generation of the PBX is the Electronic Switched PBX, these systems gradually replaced all the manually handled systems, these systems sometimes also called as the solid-state systems. The fifth generation PBX were the TDM PBX, this time division multiplexing systems allowed many signals or calls or multiple strains of information to be carried over a single path. This TDM PBX has allowed the companies to transmit from the analog to digital transmission. These were designed using CPUs, operating systems, hard drive, RAM. In early 1990s, as the internet rolled around, internet merged into the PBX and the seventh generation PBX is IP PBX, this uses the different internet protocols to build a PBX. This also offered the VoIP protocols for the multimedia transmission. This now turned out to be the IP and Hybrid PBX with using small amount of hardware and introducing variants of applications into them which can be connected to the smart devices.

There are many papers and projects done till date on the Asterisk PBX server but this paper introduces a new application along with the existing voice and video calls, which can be obtained by the help of asterisk in the server. The application generates the call logs i.e., all users all call details present within the server (date, time, from where to whom, duration). Along with this the server will be able to give the user all the alerts (in the form of a voice message and a text message from the server) for the calls which the user missed by using the details of the caller.

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III. METHODOLOGY

IP PBX is a full complete telephony system that provides calls over the internet or IP data network (Figure 1). All the information is sent through the data packets over the data network. An IP PBX consists of one or more SIP configured phones, an IP PBX server (with VoIP gateways).

IP PBX server works similar to the Proxy server. User Clients, using either the smart gadgets or desk phones register with the IP PBX established server. When the users try to contact with the other users IP PBX establishes the connection between the users which are trying to contact. This is much easier to install configure than the usual phones as the IP PBX runs as a software.

IP PBX can be managed easily using GUI, allowing to use easily and tune the phone system. The cost of the calls can be cut down to a very great extent. It also doesn't require any wiring.

IP PBX delivers the best customer service automatically by recording all the call details. It is possible with the IP PBX as the software is installed once can be used as many times as one needs.

Figure 1: IP PBX System

To make the audio and video calls possible, initially it is required to construct a asterisk server for that, it is required to install the asterisk which is an open source software and all the required library files which are required for it. Then in the sip.conf and the extensions.conf file, the users are defined and some of the codecs are enabled so as to make the calls as shown in figure 2 & figure 3 . For making the calls through the smart devices an application is to be downloaded i.e., either Csip Simple (only for calls and messages) or VoIP by antisip (for calls, video calls and messages). Once the application is downloaded users must be registered on the phones. Thus, once it is done audio and video calls can be enabled.

To store all the caller Ids, call log, caller details, a database server is being created using the DBMS (Database management system) software MySQL. A database is a structered collection of data. In this we store all the call details. The MySQL software is an open source. It is a relational database i.e, stores the data in separate tables, rows, columns, rather than storing all the data in a file or a text format. This detail can be stored as per required. Different tables, rows, columns can be created and even removed. It is very flexible to append or alter anything required. MySQL is a "Structured Query Language", which is a Language-specific API that hides the SQL syntax. The MySQL Database Server is very fast, reliable, scalable, and easy to use. It works in client/server or embedded systems

IV. HARDWARE AND SOFTWARE REQUIREMENT

A computer with Ubuntu operating system, asterisk software installed [4] with all the required user defined in the files of the sip.conf and extensions.conf. For establishing the database install the MySQL database and allow the MySQL database for required permissions, services and default data in database[2]. After the installation process of the MySQL server run this database server through the root thus it takes to the MySQL database where all the required additions can be done for the call to register.

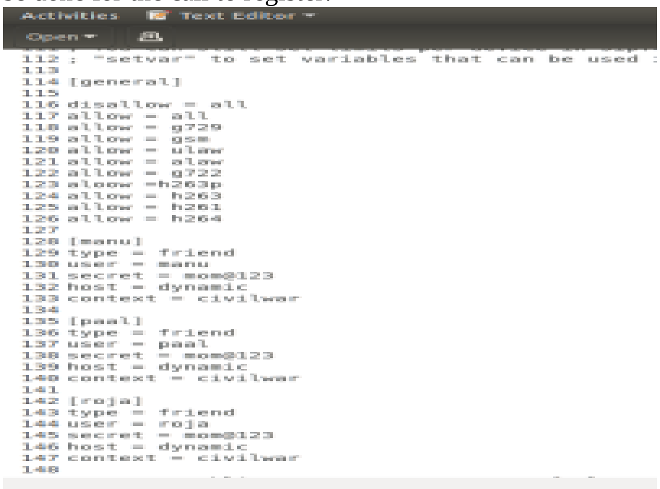

Figure 2:

Figure 3:

Now for storing the data create a new database using the command "create database mca;". Here mca is the database name (which can be anything). To see all the databases present use command "show databases;". To use the database the command is "Use mca;", thus it changes the database.

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Now for creating tables (rows and columns) use command "Create table list;", this creates table with the content in it. To see what the table contains use command "describe list;" (here list is the name or also called as the variables). Data insertion into the tables, can be done using the command "insert into list ('the data differentiating each with a comma (',')');". To see all the inserted items, use command "select *from list;".

For finding the unanswered calls and busy calls create another database and required tables in it. After defining all the tables, the database should be linked in order to get all the call details with the asterisk server. Usually all the call logs are saved in the cdr_custom.conf , but from all the logs only the unanswered and busy calls are required so to obtain those unanswered and busy calls create a new csv file (shown in figure 4) with name simple.csv and make the changes like removing ' ; ' and adding the command:

",${CSV_QUOTE(${CDR(disposition)})},
${CSV_QUOTE(${CDR(start)})} "

at the end and save the file. When the calls are made depending upon the type of the call the data is stored.

The process until now i.e., creation of database will consist of all the unanswered and busy calls, but every time when there is a change in the logs, the database doesn't get updated by itself. So, to copy the updated busy responses and unanswered calls, the server requires some custom programs with logic to copy only the data which is updated. Also, for the voice message from the asterisk server about the missed calls notification, the program code as to be implemented. All the program codes with the logic are saved as .php files in "/var/lib/asterisk/agi-bin/". Also create a temporary file for the MySQL database (Figure 5) to remember the memory position where it ended so as to continue from that position; this is done by creating a text file in var/tmp/simplefloc.txt and inserting default value '0'. Along with that create a file with hello-world.txt in the home folder so that it acts as the temporary buffer for the details to be copied into the MySQL database as asterisk cannot be directly be copied into the server's database.

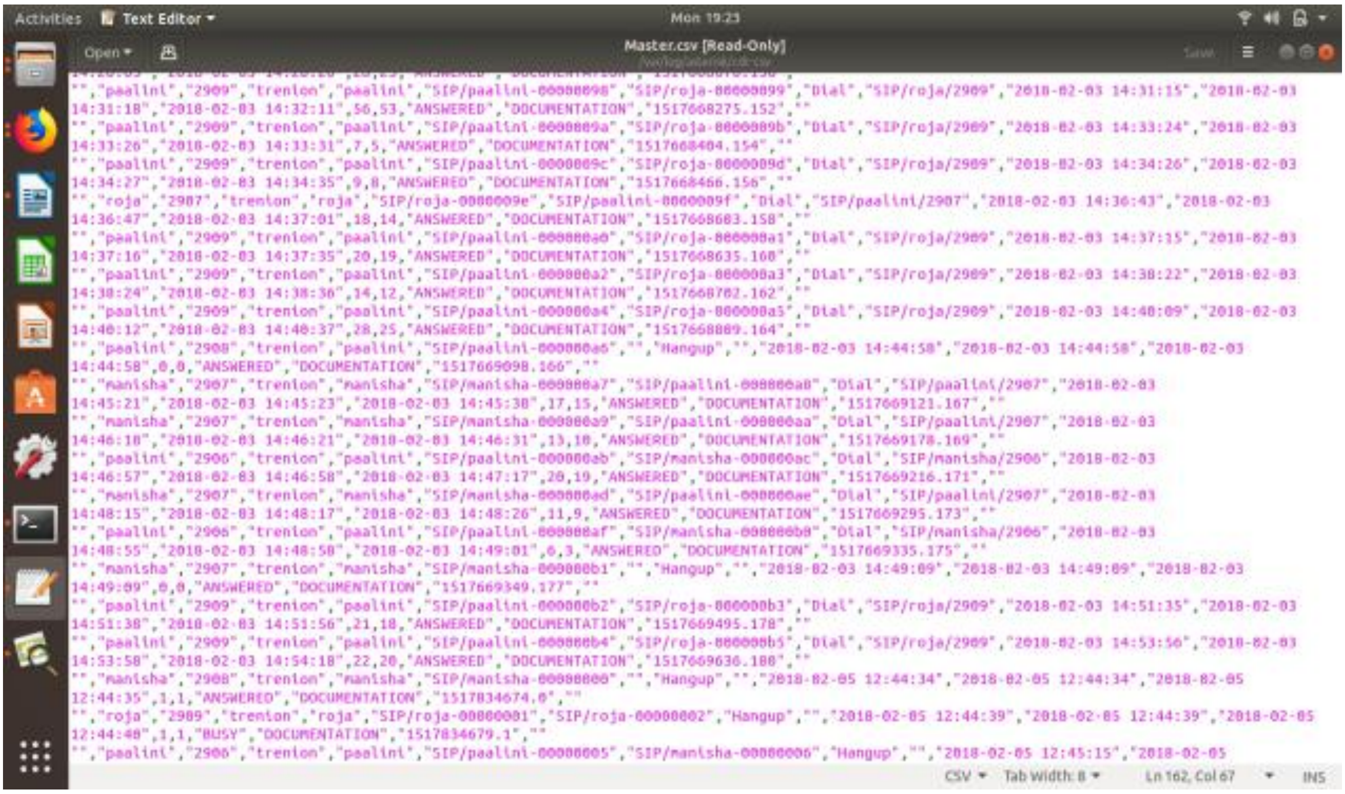

Figure 4: screen shot showing master.csv file

Once the setup is done the created program codes with the logic are to be run in two different terminals with the command "cc-o 'filename' (mysql_config –cflags)filename.c$(mysql_config-libs)". The above command compiles the two programs present in the two different terminals. Once after compilation the two programs are to be runned using the command ". /filename.c" in the two terminals (command window). Thus, now the asterisk server is ready to alert the user whenever the user disconnects or connects into the network and make calls. But, as the call alert much contain (according to the aim of the project) voice message the server much have the voice packages to install the voice passages use the commands "sudo apt-get install festive, sudo apt-get espeak". Thus, reloading the server, it is ready for performing the task. Let the programs be mca1.c and mca2.c for understanding.

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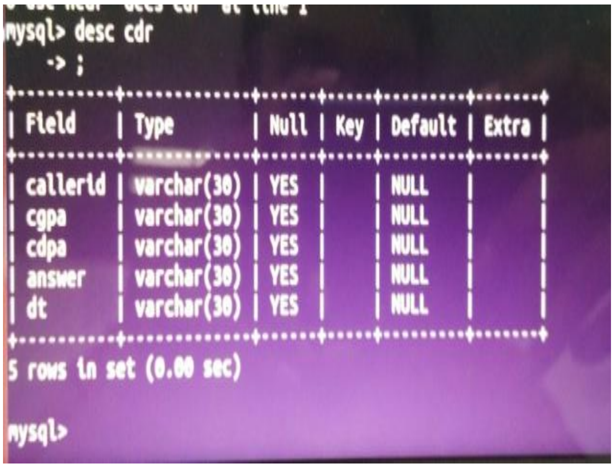

```
mysql> desc cdr
    -> ;
+----------+-------------+------+-----+---------+-------+
| Field    | Type        | Null | Key | Default | Extra |
+----------+-------------+------+-----+---------+-------+
| callerId | varchar(30) | YES  |     | NULL    |       |
| cgpa     | varchar(30) | YES  |     | NULL    |       |
| cdpa     | varchar(30) | YES  |     | NULL    |       |
| answer   | varchar(30) | YES  |     | NULL    |       |
| dt       | varchar(30) | YES  |     | NULL    |       |
+----------+-------------+------+-----+---------+-------+
5 rows in set (0.00 sec)

mysql>
```

Figure 5:

V. FLOWCHART & RESULTS

The figure 6 and figure 7 below explain the process flow that takes place between asterisk and MySQL. In first flow chart after initiating parameters such as users and their respective user ids a connection is requested to MySQL server. If there occurs any error during the process, connection is rejected and error message is sent else MySQL server makes an attempt of using mca database.If this attempt is successful then open simple.csv and simplefloc in read mode and print the change by checking the file modification time but if the attempt fails then update and store the values of simplefloc to buffer and make the attempt again. Hence the call log is updated.In the next flow chart, once the user's registrations are verified if any modifications don't exist no action takes place but if modifications exist a voice message is to be delivered but first verification of username to the number stored in user data and the calling party address is obtained. Then the information of missed call is sent in the format of message. After the message is delivered the cdr is deleted. Hence a missed call is alerted. Upon doing all the required processes as stated above, asterisk will be able to make audio calls, video calls, and also give the missed call alerts when the user misses the call or when the users are out of the reach of server (Figure 8 to 12). The following are some of the screen shots of the IP PBX working process.

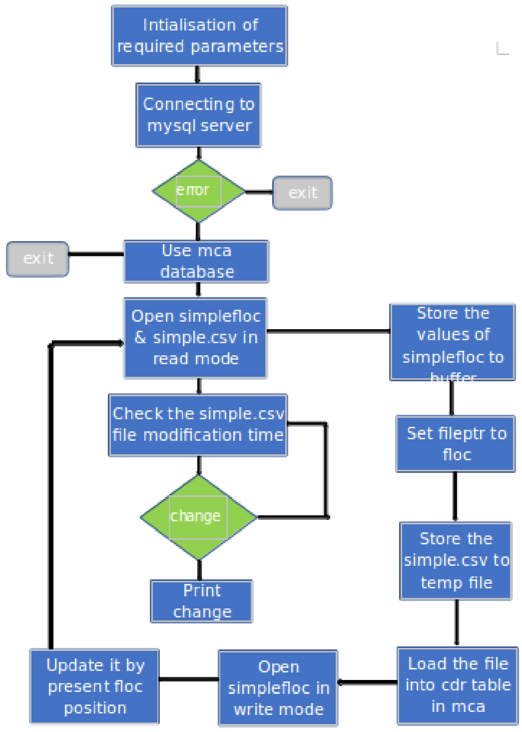

Figure 6: Process flow

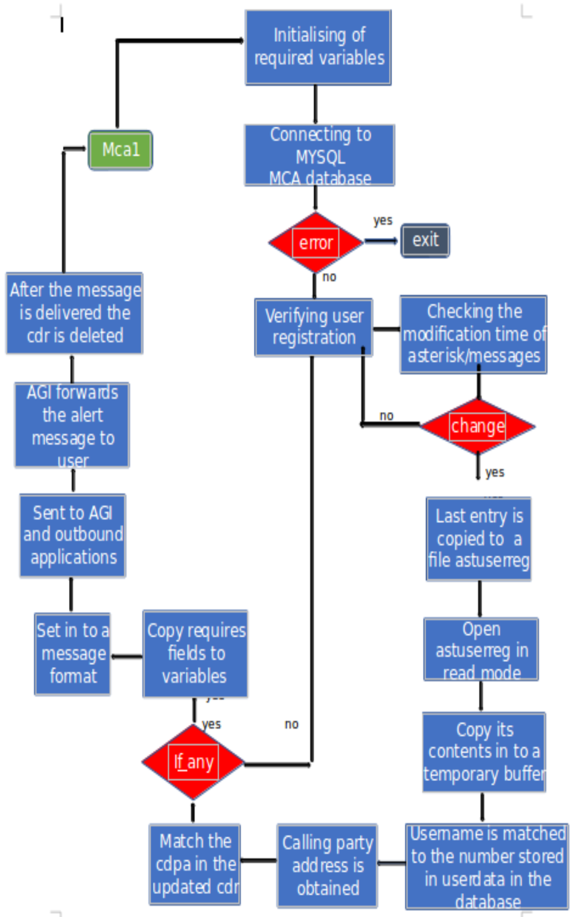

Figure 7: Process Flow

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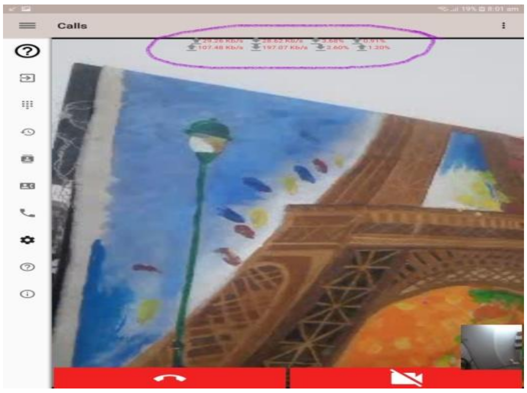


Figure 8: Call Alert

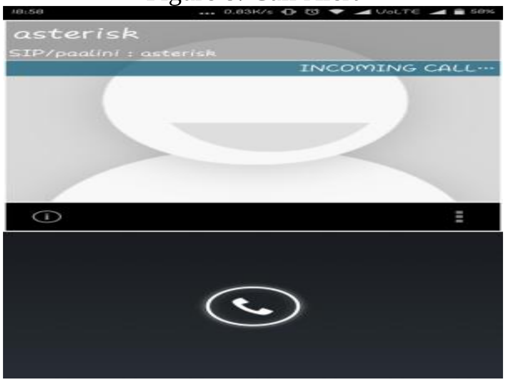


Figure 9: Incoming Call Alert

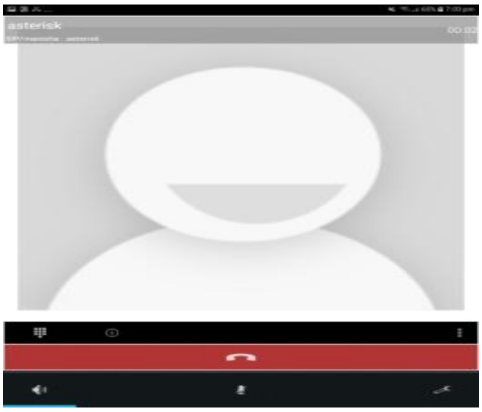
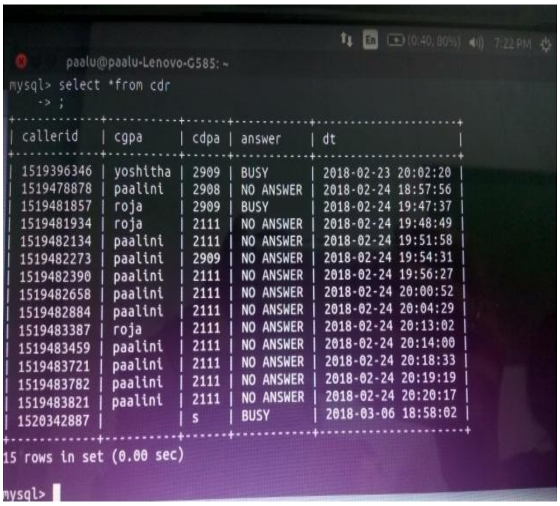


Figure 10: Missed call Alert from the Server

Figure 11: Message from the Asterisk server.


paalu@paalu-Lenovo-G585: ~

mysql> select *from cdr
-> ;

callerid	cgpa	cdpa	answer	dt
1519396346	yoshitha	2909	BUSY	2018-02-23 20:02:20
1519478878	paalini	2908	NO ANSWER	2018-02-24 18:57:56
1519481857	roja	2909	BUSY	2018-02-24 19:47:37
1519481934	roja	2111	NO ANSWER	2018-02-24 19:48:49
1519482134	paalini	2111	NO ANSWER	2018-02-24 19:51:58
1519482273	paalini	2909	NO ANSWER	2018-02-24 19:54:31
1519482390	paalini	2111	NO ANSWER	2018-02-24 19:56:27
1519482658	paalini	2111	NO ANSWER	2018-02-24 20:00:52
1519482884	paalini	2111	NO ANSWER	2018-02-24 20:04:29
1519483387	roja	2111	NO ANSWER	2018-02-24 20:13:02
1519483459	paalini	2111	NO ANSWER	2018-02-24 20:14:00
1519483721	paalini	2111	NO ANSWER	2018-02-24 20:18:33
1519483782	paalini	2111	NO ANSWER	2018-02-24 20:19:19
1519483821	paalini	2111	NO ANSWER	2018-02-24 20:20:17
1520342887		s	BUSY	2018-03-06 18:58:02

15 rows in set (0.00 sec)

mysql>

Figure 12: Log history in that saved in the server

VI. Mathematical Analysis

The maximum number of users using the application concurrently depends upon mainly the Access point or router model, bandwidth, throughput, coverage and interference. Theoretically most of the access points or the routers claim to support 255 users by a single device, but however it is not possible to support such a huge number of users by a single device. Each user in the network are allotted with some particular bandwidth, however users use their network occasionally so the router shifts the available unused bandwidth to another user among the network. The number of users in a network are directly proportional to the bandwidth of the network. The maximum number of the users handled by the server ∞ The maximum strength of the router. For example, if a server is having a network speed of 100Mbps and let us consider that on average in a server any user uses up to 10Mbps then the server can accommodate up to 10 devices with that strength. (100/10 = 10).

$$Maximum\ number\ of\ users = \frac{Network\ speed\ of\ the\ server}{Average\ speed\ required\ by\ a\ single\ user}$$

VII. CONCLUSION

By estimating the number of users and the widespread of the company a strong router is to be established and the database has to be updated with all the current working employees in order to communicate through intranetworking. Hence, a secure communication can be established by implementing a PBX server based on VOIP and SIP protocols using Asterisk software**.** There are many open source PBX server software, but are unknown. A personal laptop or Desktop can also be configured into a server and a simple private communication network can be established. This makes the networking system cheaper and easily accessible.

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Dr. N Dinesh Kumar is working as Associate professor and Head Electronics & Communication Engineering department at Vignan Institute of Technology & Science, Deshmukhi, affiliated to JNTUH, Hyderabad. He had about 17 years of teaching experience and about 1yr 9 months of software development experience. To his credit he had 62 reputed peer reviewed International/national journals and conference papers. He is member of many learned societies and professional bodies such as IEEE, IETE, IACSIT, IAENG, SDIWC, WASET, ISOI. Research topics of interest are Wireless communications, Embedded systems and Energy Harvesting. Completed two consultancy projects and Presently working on Energy harvesting research project funded by ANURAG, DRDO.

Ms. Manisha Sai Pulaparty pursuing final year B.Tech in Vignan Institute of Technology and Science. During her four years in UG course she had worked on different projects namely Automatic Parking Lot system, Low Cost HD surveillance camera, Automatic Solar tracking System, Maze Solving robot, Intra-networking (Establishing a Private branch exchange) and EMG controlled Prosthetic Arm. She had been a student member of IEEE and also Student branch chair of IEEE of the College. She has participated in National level Project Expos and paper presentations and won many prizes. Research topics of interest are Embedded Systems.

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