Voice - Video Communication on Mobile Phones and PCs' using Asterisk EPBX

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***Abstract*— This paper presents the design and implementation of Asterisk server, which serves as the local exchange for placing voice and video calls within a private Wi-Fi cloud and legacy networks, that is, with the Public Switch Telephony Network (PSTN). The model is accessible within the area of a university campus and allows only those mobile phones and PCs' to connect to the Asterisk server which are registered with the wireless network. The work proposed in this paper has an added feature for placing the voice and video calls on mobile phones hence increasing the mobility of a user. The model is successful in carrying out voice and video calls on Android supported handhelds connected with the wireless network and PCs connected with both wired LAN and wireless LAN. Further, every user is provided with his own extension number that can be used to connect to the outside world.**

Keywords-voice-video calls, wireless communications, VoIP calls, Asterisk, SIP

I. INTRODUCTION

Asterisk is a Private Branch Exchange (PBX). A PBX is a private phone switchboard, connecting to one or more telephones on one side to one or more telephone lines on the other. This is usually more cost effective than leasing a telephone line for each telephone needed in a business. Asterisk is an Open Source Linux based server that can function as a PBX. Asterisk is a framework that allows customization of modules, allowing us to create a phone system as per our own choice. Since Asterisk has a flexible architecture, it can be modified according to the requirements of a user or any organization. The call is achieved using Voice over Internet Protocol (VoIP) which costs far less than using the traditional telephony networks.

Till now, the use of asterisk has been restricted to computers or special hardware devices manufactured by Digium Inc. The aim of the paper is to elaborate a technique to develop an application for any Wi-Fi enabled cell phone to communicate to another such device using services of an Asterisk server through Wi-Fi rather than traditional telephony.

There are several handheld and desktop applications available in the market which provide free VoIP calls but are not bound within a certain limit and are not free for connecting with the PSTN exchange. The most popular among them is Skype which provides free VoIP calls for Skype-to-Skype communication but requires a paid account if a user demands to connect with a PSTN or a local telephone exchange which is generally used for intercoms to communicate among each other just like a local network interconnected with cables. The service of Skype popularly known as Skype Wi-Fi provides the same feature for a particular hotspot registered with the Skype. This service is paid and makes unaffordable for some startup companies and academic institutes to get it deployed [1].

II. APPLICATION ARCHITECTURE

The model described in the paper has three components, which are Asterisk server, clients and PSTN exchange.

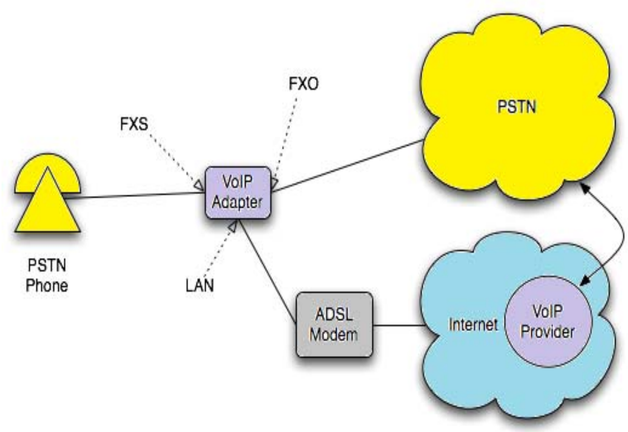

Figure 1. Actual model for placing voice and video calls using the Asterisk Server and PSTN.

III. RELATED WORK

A. *Tasks fully accomplished*

- Establishing a private and secured wireless network within the whole university campus.
- Placing of Voice and Video calls among the laptops and mobile devices registered and available within the same wireless network.
- Sending SMS among laptops and mobile phones registered with the Asterisk server [2].

B. *Tasks to be accomplished*

- Integrating the existing Asterisk Server with the university's Private Branch Exchange (PBX).

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- Establishing Voice calls among the users registered with the Asterisk server and the PSTN landline phones.
- Developing an Interactive Voice Response (IVR) system to operate the phone calls.

C. Difficulties faced

- Placing both the voice and video calls simultaneously causes delay in receiving and sending packets as the bandwidth requirement becomes a vital factor for carrying out video and audio streaming simultaneously.
- The problem of echo is found to exist when the two SIP clients running on mobile phones are in a close proximity with the server.
- If the Asterisk server is behind the Network Address Translation (NAT) and the user is outside the NAT, the user was unable to receive or send his audio to the asterisk server.
- Even if the user is well registered with the Asterisk server, the server isn't sometimes able to create a channel for a voice call for the user.

D. Attempted Solutions

- Disabling firewall at both the server and client side allowed incoming connections to get established with the Asterisk.
- Modifying the "sip.conf" file by writing "qualify=1000" for very user checks every second if the user is still reachable or not.

IV. IMPLEMENTATION

A. Softwares & Packages used

- zaptel-1.4.12.1, libpri-1.4.10.1, asterisk-1.6.2.22, asterisk-sounds-1.2.1, asterisk-addons-1.6.2.3, DAHDI Complete 2.5.0.2+2.5.0.2
- cvs, build-essential, automake, autoconf, bison, flex, libtool, libncurses5-dev, libssl-dev, libgsm1, libgsm1-dev, sysvconfig
- X-Lite, a softphone for Linux and Windows operating system
- Sipdroid, a SIP/VoIP client for Android supported mobile phones.

B. Hardware used

- 8 port FXO – FXS analog card
- Cisco routers and access points
- CAT6 & CAT5 LAN cables
- Android supported mobile phones connected with the university wireless network.
- Computers connected through wired line or wirelessly to university network.

C. Registering users with the Asterisk Server

The "sip.conf" file contains parameters relating to the configuration of sip client access to the Asterisk server. Clients must be configured in this file before they can place or receive calls using the Asterisk server. The following lines are to be written in the "sip.conf" file in the "/etc/asterisk". The users defined in this file are only allowed to connect to the Asterisk server using a Soft phone for laptops and PCs and a SIP client for handhelds [3]. A user not declared here cannot connect and place calls using the Asterisk server.

```
[utkarsh]
type = friend
secret = 08peb040
host = dynamic
?username = Utkarsh
context = trusted
callerid = utkarsh

[maqadeer]
type = friend
secret = maqadeer
host = dynamic
?username = maqadeer
context = trusted
callerid = utkarsh

[kanika]
type = friend
secret = 08peb048
host = dynamic
?username = kanika
context = trusted
callerid = kanika

[aditi]
type = friend
secret = aditi
host = dynamic
?username = aditi
context = trusted
callerid = aditi
```

D. Creating a Dial Plan

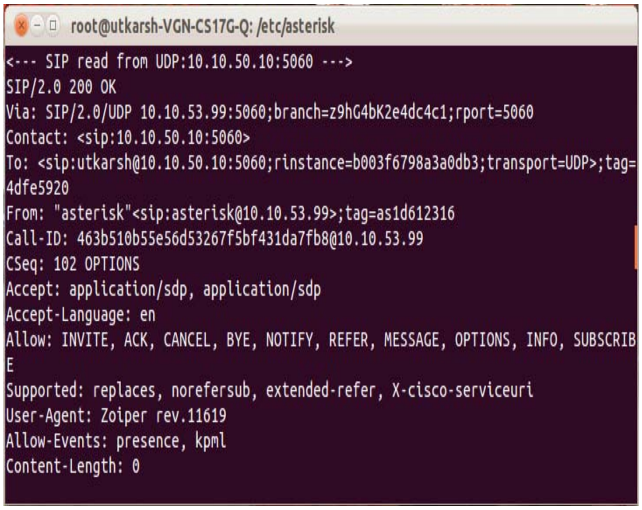
After declaring the users, now comes the task of creating a dial plan which describes the call flow and defines what actions will the Asterisk server performs when a specific number is dialed by a user. The "extensions.conf" file lays out the dial plan, bringing channels together with applications and services [4][5][6]. "extensions.conf" features extension matching logic and intelligent call routing logic. The dial plan is defined in the "extensions.conf" file present in the"/etc/asterisk" folder. The following lines provides an example of a dial plan used for implementing the work

```
[trusted]
exten => 1111,1,Dial(SIP/utkarsh)
exten => 2222,1,Dial(SIP/kanika)
exten => 3333,1,Dial(SIP/maqadeer)
exten => 4444,1,Dial(SIP/aditi)
```

E. Server logs

There are several types of logs available while a call is in active state. Writing "sipdebug=yes" in the "sip.conf" file enables viewing these logs. The following types of logs are enabled and stored for the implementation of this work:

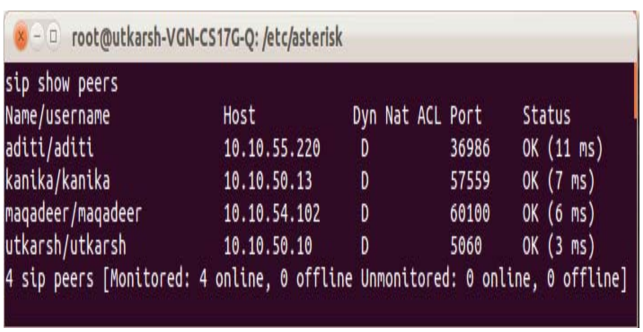
Logs when user authenticates: The security aspect of Asterisk is further enhanced, with the registered user authentication that it performs, each time user enters in the Wi-Fi cloud. In the fig 2, SIP 200 OK status code indicates successful authentication. As can be seen, the requesting IP 10.10.53.99 from port 5061 is authenticated by a registered SIP client caller id "utkarsh". Whereas, if the authentication fails, status code of form 4XX indicating error is displayed by the server. On successful completion of authentication SIP provide further message requests to clients which are listed under "Allow" attribute of authentication. In addition, further client related details like SIP client used for access, are maintained by the server.



```
root@utkarsh-VGN-CS17G-Q: /etc/asterisk
<--- SIP read from UDP:10.10.50.10:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.53.99:5060;branch=z9hG4bK2e4dc4c1;rport=5060
Contact: <sip:10.10.50.10:5060>
To: <sip:utkarsh@10.10.50.10:5060;rinstance=b003f6798a3a0db3;transport=UDP>;tag=
4dfe5920
From: "asterisk"<sip:asterisk@10.10.53.99>;tag=as1d612316
Call-ID: 463b510b55e56d53267f5bf431da7fb8@10.10.53.99
CSeq: 102 OPTIONS
Accept: application/sdp, application/sdp
Accept-Language: en
Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, MESSAGE, OPTIONS, INFO, SUBSCRIB
E
Supported: replaces, norefersub, extended-refer, X-cisco-serviceuri
User-Agent: Zoiper rev.11619
Allow-Events: presence, kpml
Content-Length: 0
```

Figure 2. Server logs when a client is registered with the Asterisk server

Viewing users registered with the Asterisk: The snapshot below shows all the SIP users currently available and registered with the Asterisk server. The Asterisk server maintains information for all the monitored and unmonitored online and offline clients. Server checks every 10 seconds if the user is still reachable or not. If the user is found unreachable, its status is set as “unreachable” else “reachable”.



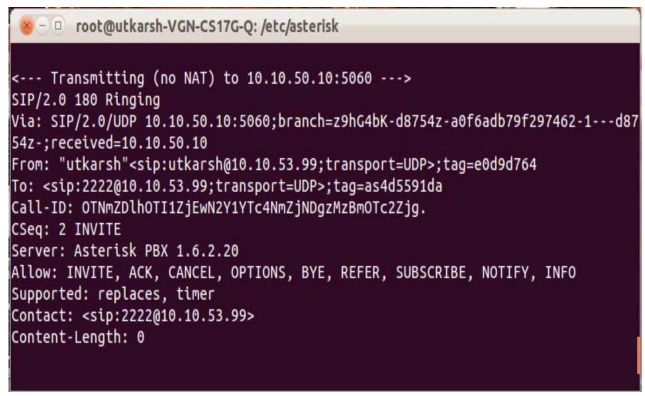
```
root@utkarsh-VGN-CS17G-Q: /etc/asterisk
sip show peers
Name/username          Host            Dyn Nat ACL Port     Status
aditi/aditi            10.10.55.220     D          36986    OK (11 ms)
kanika/kanika          10.10.50.13      D          57559    OK (7 ms)
maqadeer/maqadeer      10.10.54.102     D          60100    OK (6 ms)
utkarsh/utkarsh        10.10.50.10      D          5060     OK (3 ms)
4 sip peers [Monitored: 4 online, 0 offline Unmonitored: 0 online, 0 offline]
```

Figure 3. Server logs when a client is available in the wireless network and registered with the Asterisk server

Logs when placing an audio or a video call: When a user tries to place a call to another SIP user registered with the Asterisk server, Asterisk first authenticates the caller and checks if it is still reachable or not. After the user gets authenticated, the server now checks if the callee is also registered and reachable. After this process it searches for a free channel to allocate this call to. Meanwhile the call is in "Trying" state and the server displays its status code as 100. If a call gets established, the call is in "Ringing" state and the status code becomes “180 Ringing”. Each ongoing call is maintained by the server with a unique ID.

Client side: Calling between active registered users using laptops, is accomplished by SIP based soft clients like: X-Lite, Zoiper, DIAX etc. The users have used X-Lite to carry out the calling. To personalize the client, user addresses the server IP in the SIP account settings of the SIP client to be able to send the request to the asterisk server in the Wi-Fi cloud. Also, to enable the server to recognize the user, one configures its user id and password in the SIP settings.

Server side: As discussed earlier, a dial plan is developed in “extension.conf” file at the server to route calls. Simultaneous support to video activated by setting "videosupport" parameter in “sip.conf” to "yes" at the server side.

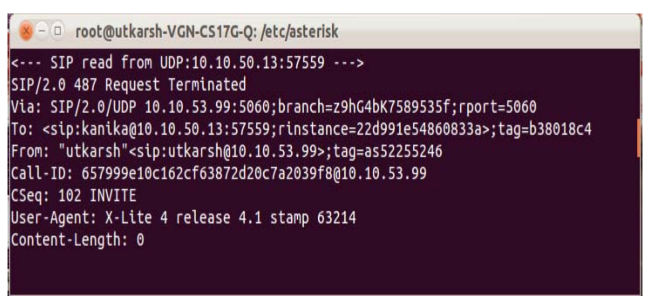


```
root@utkarsh-VGN-CS17G-Q: /etc/asterisk

<--- Transmitting (no NAT) to 10.10.50.10:5060 --->
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.10.50.10:5060;branch=z9hG4bK-d8754z-a0f6adb79f297462-1---d87
54z-;received=10.10.50.10
From: "utkarsh"<sip:utkarsh@10.10.53.99;transport=UDP>;tag=e0d9d764
To: <sip:2222@10.10.53.99;transport=UDP>;tag=as4d5591da
Call-ID: OTNmZDlhOTI1ZjEwN2Y1YTc4NmZjNDgzMzBmOTc2Zjg.
CSeq: 2 INVITE
Server: Asterisk PBX 1.6.2.20
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO
Supported: replaces, timer
Contact: <sip:2222@10.10.53.99>
Content-Length: 0
```

Figure 4. Server logs when a SIP client tries to initiate a VoIP call to another SIP user.

Logs when disconnecting an audio or a video call: If any of the SIP user, currently in a call, tries to disconnect or terminate the call, the Asterisk server first checks if the channel is still allocated to this call and then tries to free it. After successful deallocation of the used channel, Asterisk returns the status code as "487 Request Terminated".



```
root@utkarsh-VGN-CS17G-Q: /etc/asterisk
<--- SIP read from UDP:10.10.50.13:57559 --->
SIP/2.0 487 Request Terminated
Via: SIP/2.0/UDP 10.10.53.99:5060;branch=z9hG4bK7589535f;rport=5060
To: <sip:kanika@10.10.50.13:57559;rinstance=22d991e54860833a>;tag=b38018c4
From: "utkarsh"<sip:utkarsh@10.10.53.99>;tag=as52255246
Call-ID: 657999e10c162cf63872d20c7a2039f8@10.10.53.99
CSeq: 102 INVITE
User-Agent: X-Lite 4 release 4.1 stamp 63214
Content-Length: 0
```

Figure 5. Server logs when a client is deregistered with the Asterisk server

Logs when deregistering the user from the Asterisk: Once an active SIP user performs deregistering from the Asterisk server, the server first performs authentication of the requested SIP user and then attempts to deregister it. In other words, the server schedules destruction of the SIP dialog.

```
root@utkarsh-VGN-CS17G-Q: /etc/asterisk
<------------>
Scheduling destruction of SIP dialog 'YjNhN2VhM2RjZDZlOWI2NTc3OTNiODRjNGFmYTFlOW
I.' in 32000 ms (Method: REGISTER)

<--- SIP read from UDP:10.10.50.10:5060 --->
REGISTER sip:10.10.53.99;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.10.50.10:5060;branch=z9hG4bK-d8754z-d8888bc6535b7953-1---d87
54z-
Max-Forwards: 70
Contact: <sip:utkarsh@10.10.50.10:5060;rinstance=9f16b8c1fb5b5fc8;transport=UDP>
;expires=0
To: "utkarsh"<sip:utkarsh@10.10.53.99;transport=UDP>
From: "utkarsh"<sip:utkarsh@10.10.53.99;transport=UDP>;tag=08ee9503
Call-ID: YjNhN2VhM2RjZDZlOWI2NTc3OTNiODRjNGFmYTFlOWI.
CSeq: 4 REGISTER
Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, MESSAGE, OPTIONS, INFO, SUBSCRIB
E
Supported: replaces, norefersub, extended-refer, X-cisco-serviceuri
User-Agent: Zoiper rev.11619
Authorization: Digest username="utkarsh",realm="asterisk",nonce="735ae07e",uri="
sip:10.10.53.99;transport=UDP",response="ecc6dffffa4f246a48c60ec3adfd97fa",algor
ithm=MD5
Allow-Events: presence, kpml
Content-Length: 0
```

Figure 6. Server logs when a client is deregistered with the Asterisk server

F. Snapshots

The snapshot below presents a SIP client on a PC having video-voice call.

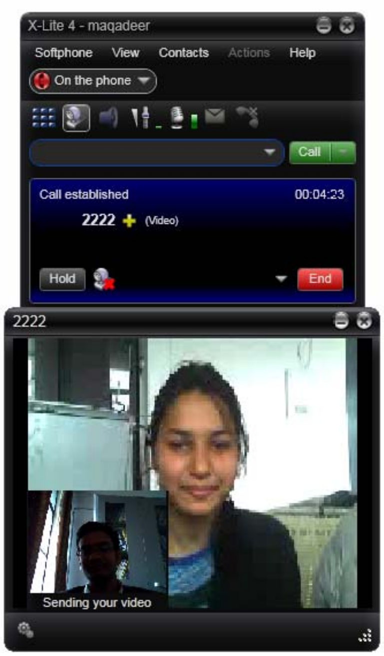


Figure 7. VoIP call between two SIP clients using PC.

The snapshot below presents a mobile SIP client having video-voice call.

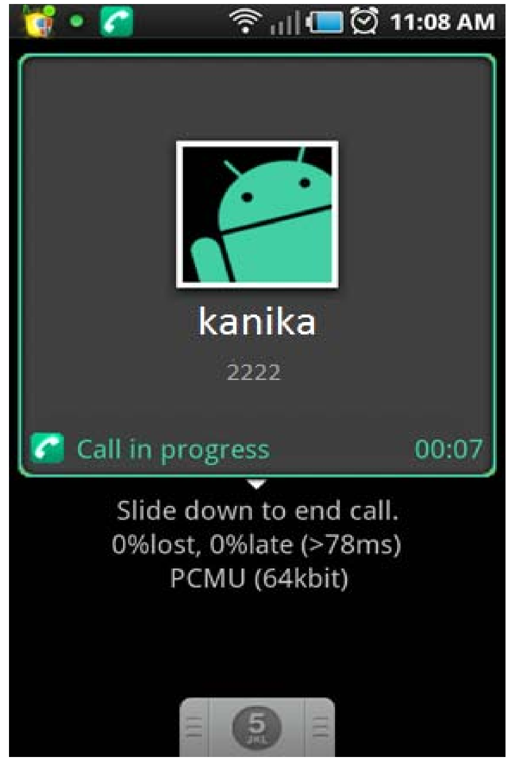


Figure 8. VoIP call between two SIP clients using mobile phones.

G. Quality of service

μ-law and a-law operate in the same way; that is they both sample audio at 8 KHz and taking into account low frequency and anti-alias filters give the standard telephony frequency response of 300Hz to 3400Hz. In a-law the audio is sampled with a 13 bit resolution and u-law with 14 bit resolution. What they then do is logarithmically compress the amplitudes so in both cases you compress the amplitude down to 8 bits. 8 bits at 8 KHz gives 64 kbit bandwidth of DSP output [7]. The reason the amplitudes are compressed to 8 bits is because the human ear is sensitive to sound in a logarithmic fashion. Most people simply won't notice the quantization used for amplitude compression for a voice conversation. When G.711 (μ-law/a-law) is received, it is expanded back to its original 13 or 14 bits [8]. This way you get up to 14 bits of dynamic range using only 8 bit values.

- Bandwidth: 64kbps
- End-to-End Delay: 140ms (average on mobile phones)
- Packet loss: 0%

V. DISCUSSION

The work has been broken down into two modules which consist of placing calls among users registered on the

Asterisk and the existing structure of the university PBX which allows wired telephones to be used as intercoms.

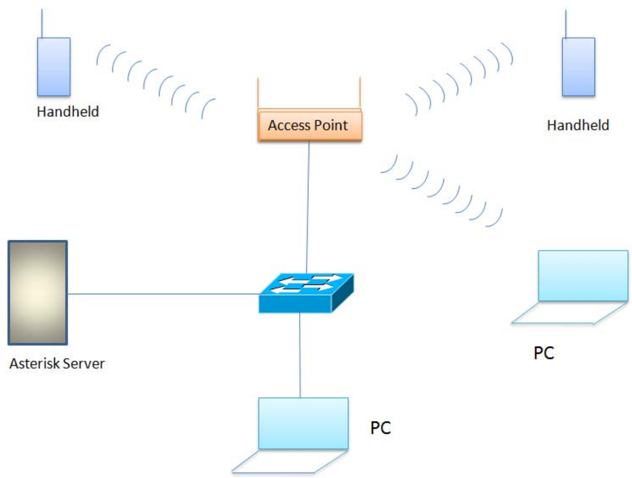


Figure 9. Actual model for placing voice and video calls using only the Asterisk Server

The Public Switched Telephone Network (PSTN), also known as Plain Old Telephone Service (POTS), is the wired phone system over which landline telephone calls are made. The PSTN relies on circuit switching. To connect one phone to another, the phone call is routed through numerous switches operating on a local, regional, national or international level.

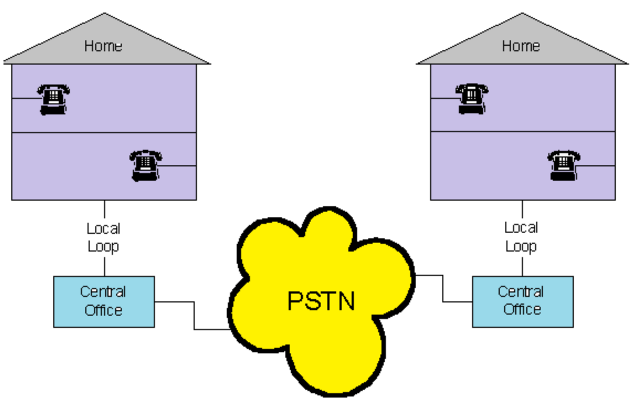


Figure 10. Actual model for placing voice and video calls using only the Asterisk Server

Support for regular POTS is provided by Digium's series of analog telephony cards. Digium's analog cards utilize separate capability modules for Foreign Exchange (FXO) trunk lines and Foreign Exchange Station (FXS) telephone sets. The analog cards are provided in four (4), eight (8), and twenty-four (24) modular port varieties for both PCI and PCI-Express slot types. An optional DSP module provides hardware-based echo cancellation for Digium's analog cards.

Support for Robbed Bit Signaling (RBS), Basic Rate ISDN (BRI) and Primary Rate ISDN (PRI) lines are provided by Digium's series of digital telephony cards. The digital cards are provided in one (1), two (2), and four (4) port varieties for both PCI and PCI-Express slot types. An optional DSP module provides hardware-based echo cancellation for Digium's digital PRI cards; digital BRI cards include on-board DSP-based echo cancellation [9].

When a telephone call is placed across a digital network, including VoIP calls, the human voice is transformed from its analog form into a digital form. This rendering of speech sound into digital form utilizes a computer algorithm to encode the signal - compression. The decoding of the digital signal is decompression. The word codec is a combination of these two functions compression – decompression [10].

All VoIP calls use some kind of codec, the most universal is a codec called G.711, available in two forms: u-law (used in North America) and a-law (used everywhere else). G.711 renders spoken voice into a data stream that utilizes 64kbit/s of bandwidth and is not computationally intensive. G.711 is also the codec used to transmit data across digital PSTN links – ISDN BRI and PRI lines.

VI. CONCLUSION

The model has been implemented in the university campus to provide free voice and video calls. It's a most effective way to diminish the large phone call bills. The service is secured and allows only the registered user to place calls. Moreover, all the calls placed using the Asterisk Server are encrypted thereby avoiding hackers to intercept an ongoing phone call.

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