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Voice over Internet Protocol (VOIP) Based On Asterisk

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Abstract
VoIP is the emerging trend of the voice communication system. As this system uses internet protocol the voice data can be send easily on the network between source to destination. The switching method used is packet switching. In this paper it is shown that how the voice packet data will be send from source to destination using Asterisk based server and wifi based clients. As the communication is realtime the session initiation protocol SIP is used and some parameters such as packet sending and receiving, jitter and bandwidth utilization will be monitored in the server.

Keywords: SIP; Asterisk; VoIP.

1. Introduction
The asterisk is the software design for the communication of voice over the internet protocol. This technology is worldwide used in the companies for their internal voice communications. Here the Asterisk is the software used as a backbone for the voice communication. This software is installed on Linux based operating system. This system is the replacement of the old EPABX system which uses wires and the switching hardware. But because of the Internet protocol the IP based private branch exchange having more advantages over the Electronics private branch exchange. Also Wifi communication will be there so we can say that wireless in-house intercom system. This is client server architecture. In this the server is the Linux, Asterisk and the clients will be desktop or mobile phones. In desktop the system the software will be X-lite which very popular SIP client by counterpath organization is. In mobile there are so many SIP apps available. We can use any VoIP softphone in the mobiles. The Asterisk is the bundle of packages like voicemail, blacklist, call recording and interactive voice response.

Methodology

The server is the linux based computer system. In this minimum AMD system with Asterisk version 1.4.x, Linux kernel 2.4 and above, min 512 MB RAM, and 80GB hard disk is used. The client will be any laptop or mobile handset. After installation the sip.conf file of the server shall contain the users and extension.conf contains the dialplan, these number of users are configured in the clients. These files are located in /ets/asterisk of the server.

```
[6001]
username=amit
secret=123456|
type=friend
allow=alaw
context=intercom
qualify=yes
host=dynamic
```

Fig 1: sip.conf

```
[general]
static=yes
writeprotect=no
[global]
[default]

[intercom]
exten=>_6XXX,1,Dial(SIP/${EXTEN});
exten=>_6XXX,2,Hangup();
```

Fig 2: Extensions.conf

After editing these files the system is ready and the clients are able to call each other. Now depending upon the wifi area the mobile clients are having that much of area capacity to call. Also no call can transfer to other PSTN as it required voip gateways.

Server IP address is the one from where the users get registered and then calling done by using VoIP and SIP protocols. Figure 3 below shows the IP address of the server

```
Determining IP information for eth0... done.
                                                        [  OK  ]
[root@localhost ~]# ifconfig
eth0      Link encap:Ethernet  HWaddr 00:0C:29:C9:87:B9
          inet addr:192.168.0.4  Bcast:192.168.0.255  Mask:255.255.255.0
          inet6 addr: fe80::20c:29ff:fec9:87b9/64 Scope:Link
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:6031 errors:0 dropped:0 overruns:0 frame:0
          TX packets:1930 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:1483579 (1.4 MiB)  TX bytes:519132 (506.9 KiB)
          Interrupt:67 Base address:0x2000
```

After finalizing the IP address of the server the SIP client means softphone which is X-Lite for windows, Zoiper for Linux and cSipSimple for Android will be configured as a client phone system. Figure 4 below is the example of Xlite user configuration.

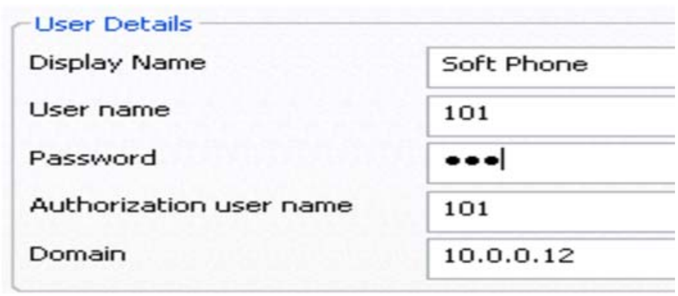


Fig 4: XLite User configuration

2. SIP Network Analysis

There are four logical types of units participate in SIP based VOIP calls: user agents (client 'UAC' or server 'UAS'), registrars, proxy and redirect servers. The user agents commence requests and they are also the final destination. Registrars maintain track of users within their assigned network domain (e.g., all users with identifiers x@ipphone.dhakacom.com register with the registrar in the ipphone.dhakacom.com domain). Proxy servers are works as an application-layer routers they forward SIP requests and responses. Redirect servers works for receive requests and then return the location of another SIP user agent or server where the user might be found. I is quite common to find proxy, redirect, and registrar servers implemented within the same program in a same server. In a classic SIP session, SIP messages originating at a user agent negotiate one or more SIP proxy servers and then reach one or more SIP user agents. While, SIP user agents can also communicate directly with each other. In fact, it is widespread that only the first request

Exchange activities along a chain of proxies, with all subsequent requests exchanged directly between the two user agents.

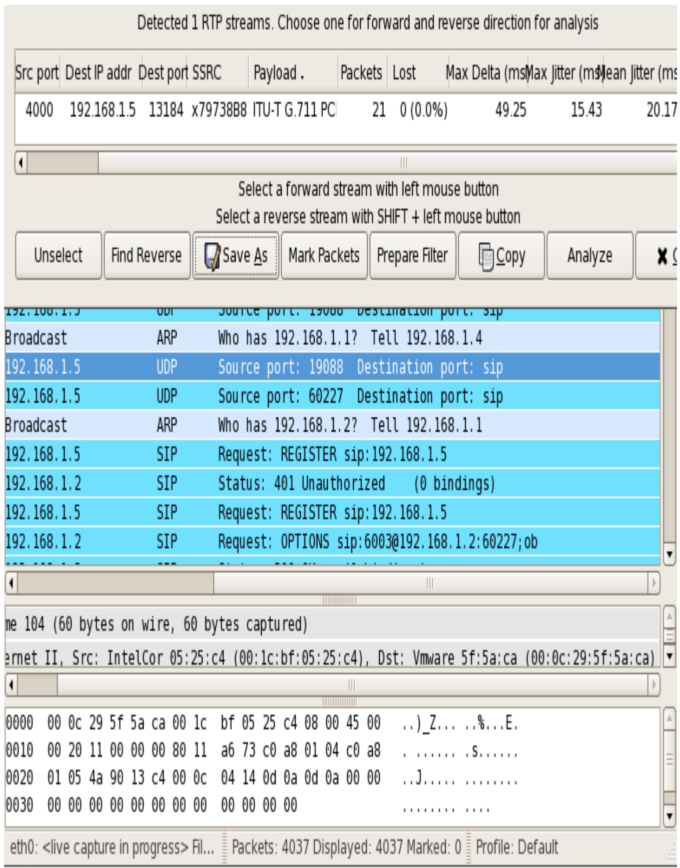
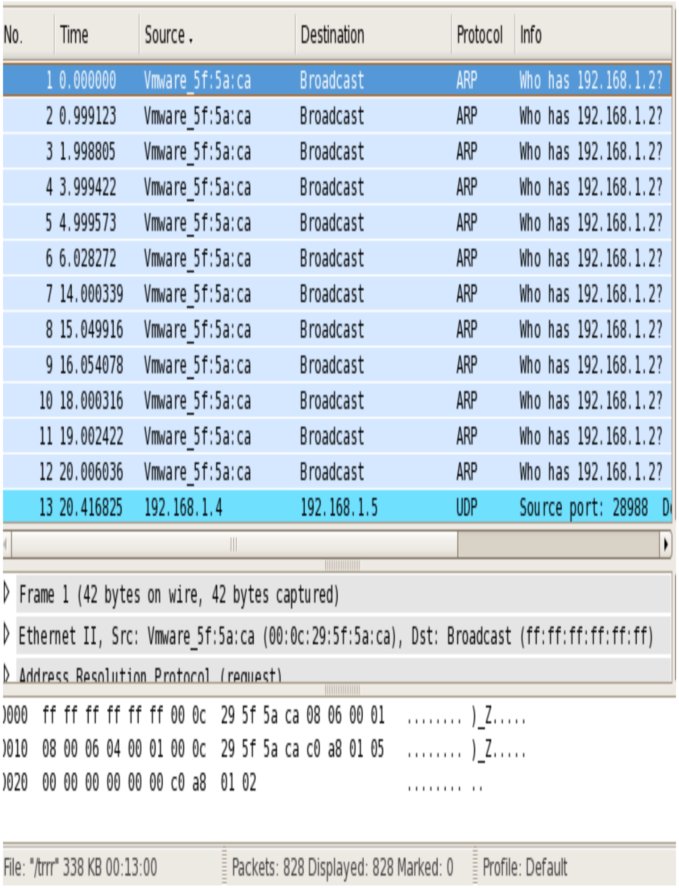


Fig 5: RTP streams

In this figure 3 the SIP protocol and RTP protocol is shown by the simulator. The port address and the IP address of the source and destination is shown. This also shows the packets loss. The single call analysis shown in the figure 5. In the figure 4 the address resolution protocol is used for the communication between two users which shows the address of the calling system and called system.



No.	Time	Source	Destination	Protocol	Info
1	0.000000	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
2	0.999123	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
3	1.998805	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
4	3.999422	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
5	4.999573	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
6	6.028272	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
7	14.000339	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
8	15.049916	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
9	16.054078	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
10	18.000316	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
11	19.002422	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
12	20.006036	Vmware_5f:5a:ca	Broadcast	ARP	Who has 192.168.1.2?
13	20.416825	192.168.1.4	192.168.1.5	UDP	Source port: 28988 D

Frame 1 (42 bytes on wire, 42 bytes captured)
Ethernet II, Src: Vmware_5f:5a:ca (00:0c:29:5f:5a:ca), Dst: Broadcast (ff:ff:ff:ff:ff:ff)
Address Resolution Protocol (request)

```
)000  ff ff ff ff ff ff 00 0c  29 5f 5a ca 08 06 00 01   ........ )_Z.....
)010  08 00 06 04 00 01 00 0c  29 5f 5a ca c0 a8 01 05   ........ )_Z.....
)020  00 00 00 00 00 00 c0 a8  01 02                     ........ ..
```

File: "/trrr" 338 KB 00:13:00 | Packets: 828 Displayed: 828 Marked: 0 | Profile: Default

Fig 6: Protocol used by the IP address

3. Network Analysis

This test were undertaken to measure the capability of existing network to handle VoIP traffic. While call is running on all the signals and real time voice data transferred using SIP and RTP protocol. The experiment is conducted in between 4 users in the wifi network. The project had to use G711 codec as project SIP p server was not properly configured for GSM codec. Though this gives less than optimum performance but was good indication what might expect inGSM implementation? GSM codec makes much smaller bandwidth consumption in comparison to G711 codec. That is why GSM implementation will be more efficient.

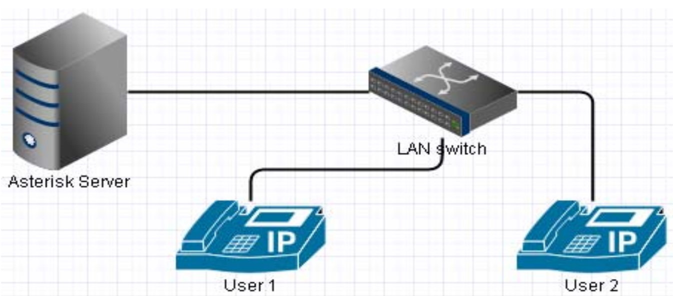


Fig 5: Network diagram

In figure 5 the network diagram shows 2 users are connected via Asterisk server. These users are connected via LAN switch and the calling will do on them. The analysis report is shown in Figure 3 and figure 4. VoIP and SIP are the two protocols are used in this. G.711 audio codec is used.

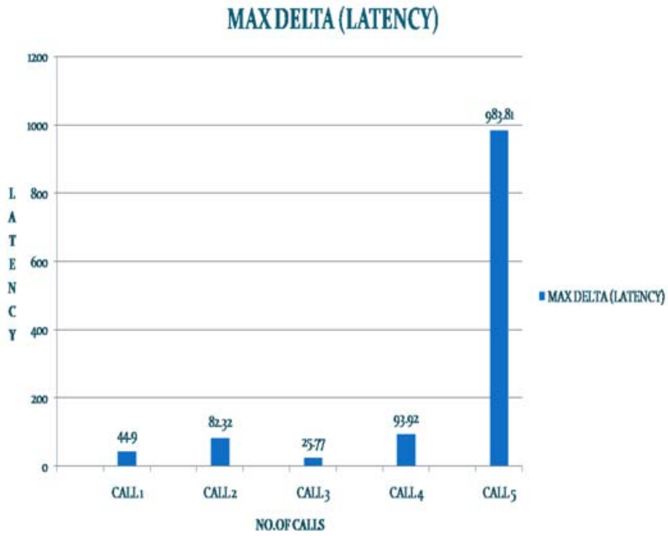


Fig 7: Latency

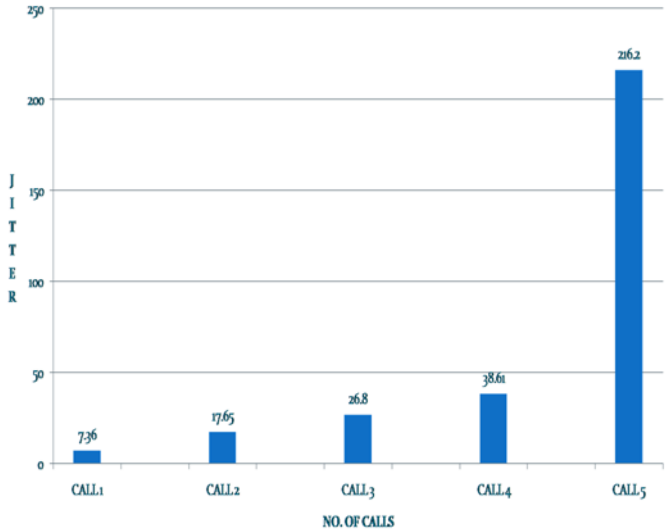


Fig 6: Jitter

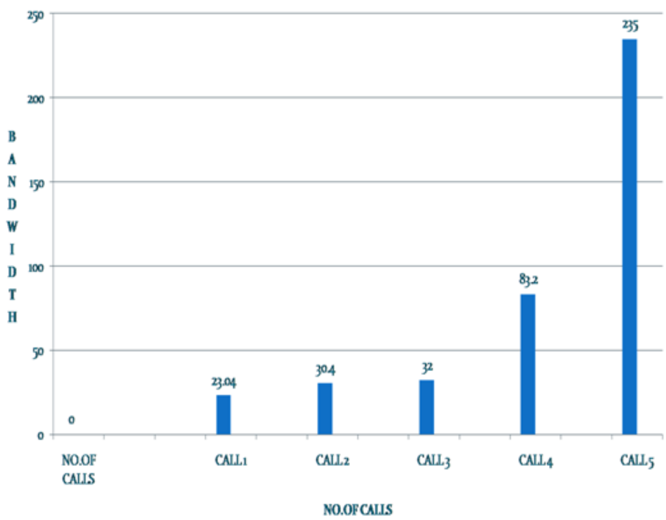


Fig 7: Bandwidth

NO. OF CALLS	IP ADDRESS OF SOURCE	IP ADDRESS OF DESTINATION	PACKET SEND	PACKET LOSS	MAX. DELTA (LATENCY)	MAX. JITTER	BAND WIDTH
CALL 1	192.168.1.3	192.168.1.7	16	0(0.0%)	44.90	7.36	23.04
CALL 2	192.168.1.6	192.168.1.7	28	0(0.0%)	82.32	17.65	30.40
CALL 3	192.168.1.9	192.168.1.7	25	2(9.0%)	25.77	26.80	32.00
CALL 4	192.168.1.7	192.168.1.1	1111	19(1.7 %)	93.92	38.61	83.20
CALL 5	192.168.1.6	192.168.1.7	792	0(0.0%)	983.81	216.20	235

Fig 8: Call chart

As per the call chart of figure 8 the Figure 5 Shows Latency of the system for different calls from call 1 to call 5. Figure 6 shows Jitter for the same calls. Figure 7 shows Bandwidth.

4. Conclusion
After Analyzing the various results and Graphs the performance parameters like Jitter, Bandwidth, Latency gives the better performance as the no. of calls increases. From the graph we have concluded that as per the number of calls increases the performance of Bandwidth also increases**.**

5. Acknowledgment
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