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# Performance of Various VoIP Vocoders using Wireshark with Asterisk PBX

R.Chinna Rao, D.Elizabath Rani, S.Srinivasa Rao

***Abstract: A private branch exchange (PBX) is implemented by an Asterisk software. In conjunction with appropriate telecommunication hardware interfaces and network applications, Asterisk is employed to establish and manage the telephone calls between telecommunication endpoints, like customary telephone sets, destinations on the general public switched telephone network (PSTN), and devices or services on voice over internet Protocol (VoIP) networks. A Vocoder may be a system of coders and encoders that is employed to scaleback the bandwidth over the restricted use of bandwidth necessities and restricted capability channels in real time needs. This paper presents Performance of Various VOIP Vocoders using wireshark with Asterisk PBX.***

***Keywords*** **: Vocoders, Asterisk PBX, VOIP, Wireshark.**

## I. INTRODUCTION

The VoIP is more scalable, has the ability to reduce costs and allows the integration of new features. In a data network perspective, the VoIP has become the voice in "only" one more application. Regarding a traditional PBX system, a VoIP PBX allow beyond existing functions, new ones, like connect employees that work from home with office PBX on broadband connections, easily connect offices in different geographic locations, offer voicemail integrated with email account. These features are performed using the Internet or through a private IP network.

In this paper is examined in some detail, issues that should be taken into consideration to implement a VoIP PBX system, more specifically the Asterisk PBX, as well as one of the signaling protocols must be used in VoIP and RTP (Real Time Transport Protocol), which helps to ensure the transmission of packets in real time applications and SDP (Session Description Protocol) a protocol that provides a standard representation for describing streaming media initialization parameters.

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The Asterisk PBX is a cost free and open source software for Linux, which supports multiple protocols (SIP, SCCP, H.323, IAX2, etc). The Asterisk allows both the use of phones in software and VoIP telephony devices.

A point to point call means that there is a user (calling party) that calls to another user (called party) and establish a telephone call. A conference call is a telephone call in which the calling party could have more than one called party that could enter or exit at any time in a conference, since conference call is active. In an interactive voice response, or IVR, Asterisk PBX as the ability to answer the call and interact with the callers with pre-recorded audio to further direct callers on how to proceed.

In different scenarios is analyzed the performance of Asterisk PBX depending on the type of sound encoding used, number of simultaneous calls and number of calls per second. Parameters such as: the load on the network, use of computer resources (CPU and memory) and response time of the Asterisk PBX, are measured.

## II. VOICE OVER INTERNET PROTOCOL (VOIP)

Voice over IP, or VoIP, can be defined with a set of technologies and components that together provide to a user the possibility of establishing voice calls through the Internet or other packet-switched networks. Through VoIP we can get several additional services beyond voice calls, such as voicemail, fax, IVR (Interactive Voice Response), conference calls, instant messaging, etc. In order to allow the existence of these services there are set several protocols, such as, SIP (Session Initiation Protocol).

To analyze the performance of VoIP software, such as the Asterisk PBX, we must understand what are the network conditions that may adversely affect the performance for VoIP. These are mainly the latency, jitter and packets loss. The latency is the time that a packet takes to travel from one point A to point B network. The jitter is the variation of latency on a range of packages. The packet loss is the number of packets sent from a given point of the network that never reach the point B. Due to the audio in VoIP be sent in real time by UDP, a packet that is received out of order is discarded. [1][7].

In VoIP, codec (a combination of Coder - decoder), can be defined as an algorithm used to encode and decode voice streams. Codecs are used to convert an analog voice signal to digitally encoded version and vice versa. There are several ways to encode and decode the streams of voice. Codecs can vary in the sound quality, the bandwidth required, the computational requirements, etc. [5]

The asterisk is cost free open source software for Linux, developed by Mark Spencer. This software gives full functionality of a PBX (Private Branch eXchange)

traditional and provides additional features such as IVR (Interactive Voice Response). The Asterisk is essentially a VoIP PBX that supports multiple protocols used in VoIP (SIP, SCCP, H.323, IAX2, etc.).

Asterisk allows the use of software phones and VoIP telephony devices. With Asterisk is also possible inter-operate with the traditional telephone systems, in this case additional hardware is needed.

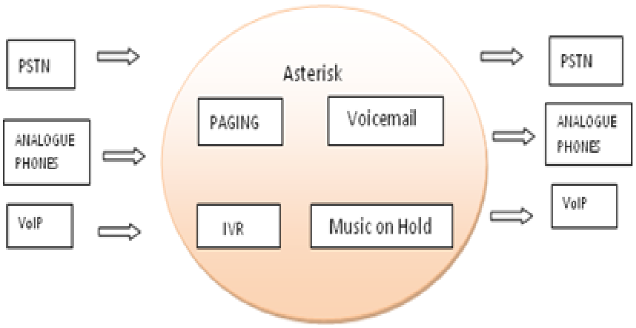


**Fig.1. Structure of Asterisk**

## III. ANALYSIS AND TRAFFIC CAPTURE SOFTWARE

One way of assessing the performance of the PBX Asterisk, is using tools, usually in software, which allows the capture of packets (sniffers). The use of these kinds of tools can also diagnose errors, producing statistics and have a graphical view of traffic flow data (for the PBX Asterisk, the data traffic mostly VoIP) in network. A software tool for pturing and analyzing traffic data is the Wireshark The Wire shark has a simple and intuitive raphical interface, and also be used in a ext-only mode.To test the SIP ormance in Asterisk PBX there are several software tools, such as SiPP [16], software developed by HP and its main features are: SIP basic scenarios included, ability to create complex SIP scenarios by changing the files in XML format, producing a set and UDP transport, possibility the scenario of point to point calls, when a user (calling party) makes a call to another user (called party), the Asterisk PBX routes the call to a called party that as been set previously and "negotiate" the codec the his want to use. Asterisk PBX acts as an intermediary in the call. The aim of this test is to understand what the maximum umber of simultaneous calls that the Asterisk PBX supports, and what impact when it is necessary to make transcoding, that is, make possible the communication between two terminals even if they do not use the same voice codec.

To evaluate the performance of point to point calls, load tests were conducted. That was used 5 computers connected in network through an 8 port 10/100 Mbps hub.The first series of test was performed based on PBX Asterisk installation of a computer with a 2.6 GHz Pentium Celeron processor. Then tests were performed using a weakest computer, a Pentium 350 MHz.The configurations of the remaining elements of network are: a computer that generates SIP calls, using the Sipp application in order to load the Asterisk PBX. There was another computer that acts as SIP call receiver, running the Sipp application.

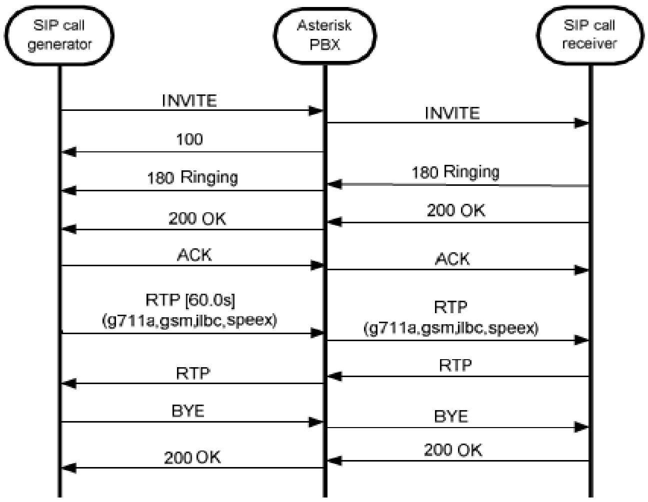


**Fig.2. SIP and RTP flow messages, used in point to point calls**

Finally there is also a computer that captures all traffic on the network. Figure 1 illustrates the network configuration used and the characteristics of each computer. When a caller (SIP call generator) makes a call the extension 222333444, that was routed to the recipient (SIP call receiver), then the caller transmits RTP voice traffic for 60 seconds, which may be encoded in several ways: G711a, GSM, iLBC or Speex. The recipient receives the voice traffic and echoes it back to the sender.

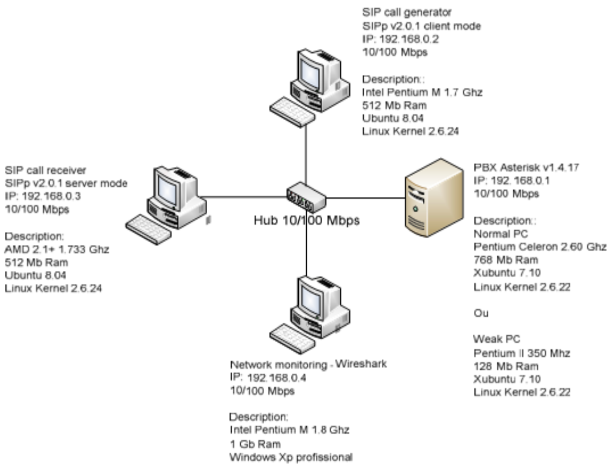


**Fig.3. Asterisk PBX**

The Asterisk PBX allows the codec used by the sender to transmit the voice traffic to be different from that recipient use/support, in this case the Asterisk PBX is responsible for transcoding of codec. In load tests, are generated three calls per second. The variable parameter is the number of simultaneous calls which serves to test the limit for which there is no retransmission of messages due to timeout, for lack of response from the Asterisk PBX

## IV. G.711 ALAW VOCODER

The Toll-Quality audio tt 64 kbit / s is delivered by the G.711 narrowband audio codec.

This codec can also transmits 300–3400 Hz audio signals which are sampled at a speed of 8,000 samples per second, whose sensitivity is of 50 parts per million (ppm). The following screenshots show G.711 Alaw Vocoder simulation results using WIRESHARK.

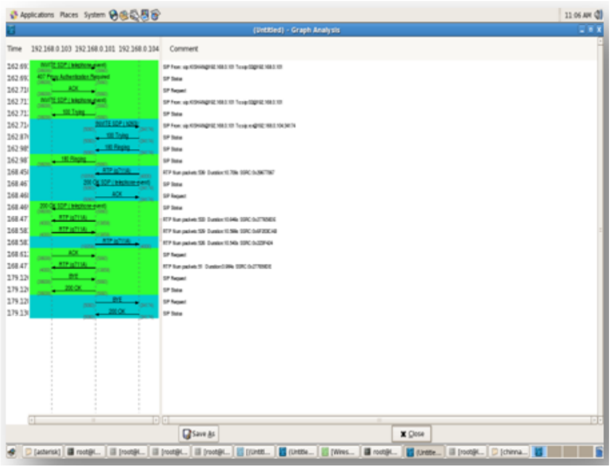

**Fig.4. ALAW_VOIP**

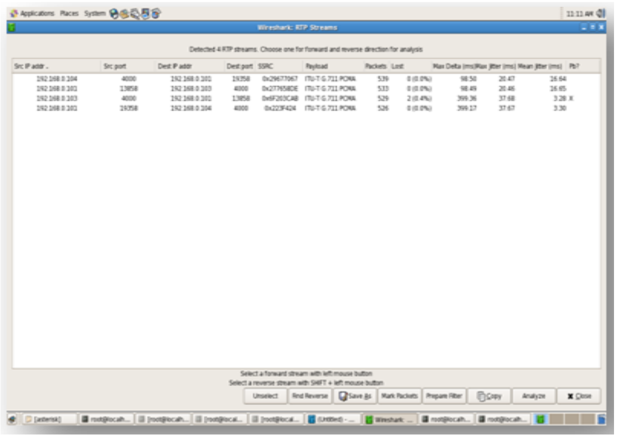

**Fig.5. ALAW_RTP**

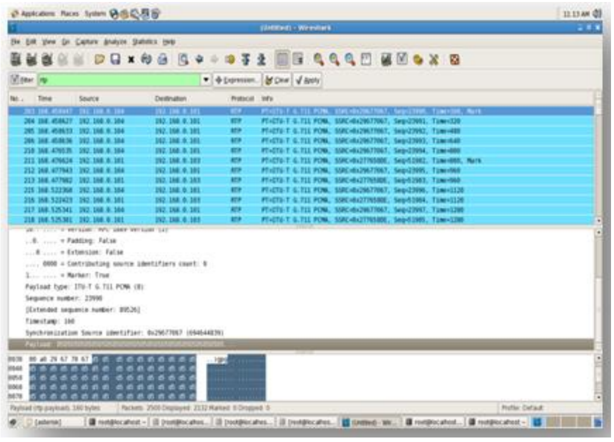

**Fig.6. Structure of Asterisk**

## V. G.711 µLAW VOCODER

The narrowband audio codec G.711 is a ITU standard codec with high bit rate (64 Kbps). It is the native language of the modern digital telephone network. Although formally standardized in 1988, the G.711 PCM codec is that the gramps of digital telephone. Fictional by Bell Systems and introduced within the early 70's, the T1 digital trunk used AN 8-bit pulse code modulation (PCM) encryption theme with a sample rate of 8000 samples per second. This allowed for a (theoretical) most voice bandwith of 4000 rate**.** A T1 trunk carries twenty four digital PCM channels multiplexed along. The improved European E1 commonplace carries thirty channels. There square measure 2 versions: A-law and U-law. U-law is native to the T1 customary utilized in North America and Japan. The A-law is native to the E1 customary utilized in the remainder of the planet. The distinction is within the methodology the analog signal being sampled. In each schemes, the signal isn't sampled linearly, however in an exceedingly exponent fashion. A-law provides a lot of dynamic vary as against U-law. The result's a less 'fuzzy' sound as sampling artifacts square measure higher suppressed.

Using G.711 for VoIP can provide the most effective voice quality; since it's an equivalent codec utilized by the PSTN network and ISDN lines, it sounds rather like employing a regular or ISDN phone. It conjointly has all-time low latency (lag) as a result of there's very little to no want for buffering, that prices process power. The draw back is that it takes additional information measure than different codecs, up to eighty four Kbps together with all the UDP and scientific discipline overhead. However, with increasing broadband bandwith, this could not be a tangle. The following screenshots shows the simulation results of G.711 Ulaw Vocoder using WIRESHARK.

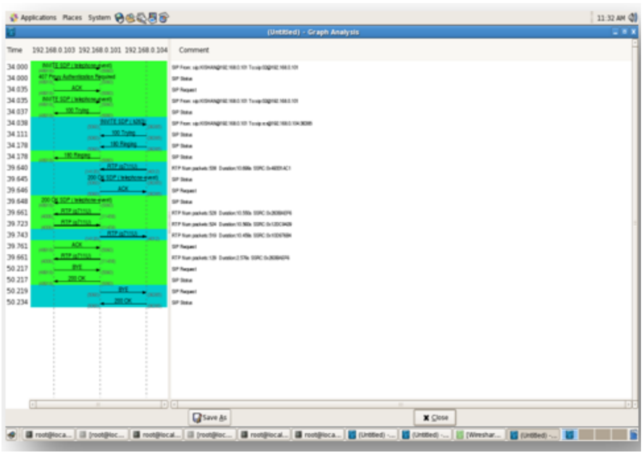

**Fig.7. ULAW_VOIP**

## VI. ILBC VOCODER

ILBC (Internet Low Bitrate Codec) is a high-complexity speech codec appropriate for strong voice communication over IP. ILBC is that the initial codec ever to be standardized by the IETF (RFC3951 and RFC3952) and is selected by Cable Labs as a compulsory element of Packet Cable voice-over-cable telephone systems. iLBC, uses a block-independent linear-predictive coding (LPC) algorithm and has support for two basic frame lengths: 20 ms at 15.2 kbit/s and 30 ms at 13.33 kbit/s.Other standard low bit rate codecs create use of dependencies between speech frames, leading to error proliferation once packets ar lost or delayed.

In contrast, iLBC encoded speech frames ar independent. This distinctive technology provides iLBC robustness against packet loss and delay.

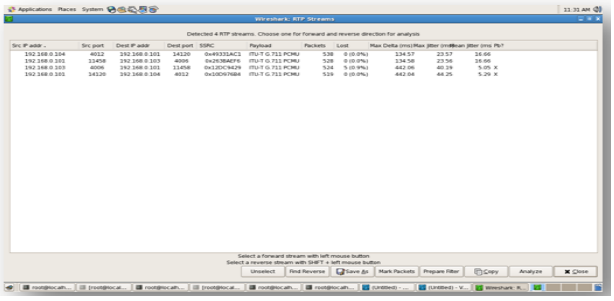

Fig.8. Fig. 8. ULAW_RTP

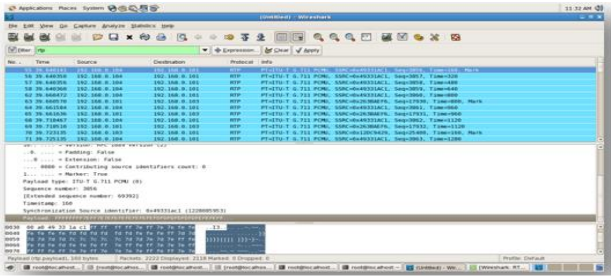

Fig.9. ULAW_WIRE SHARK

The following screenshots shows the simulation results of iLBC Vocoder using WIRESHARK.

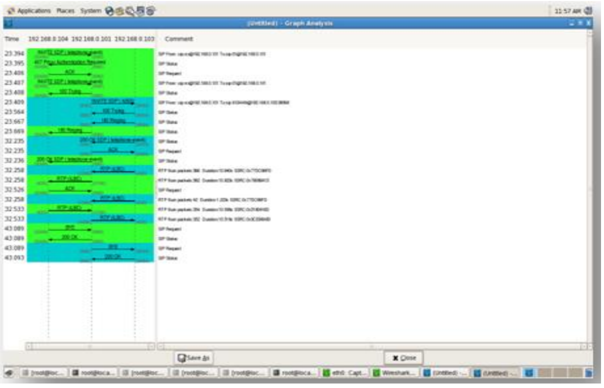

Fig.10. ULAW_WIRE SHARK

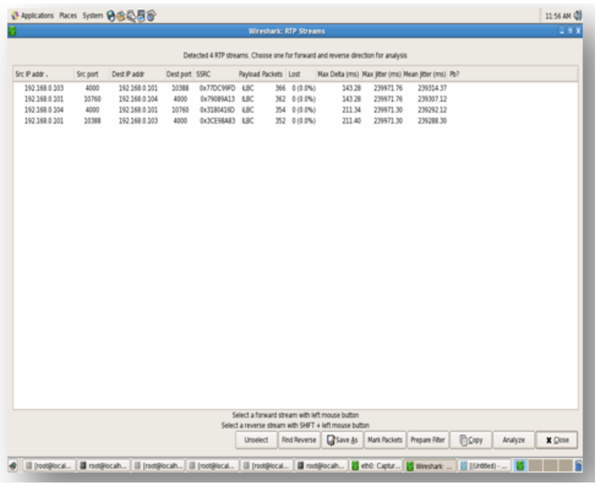

Fig.11. iLBC_RTP

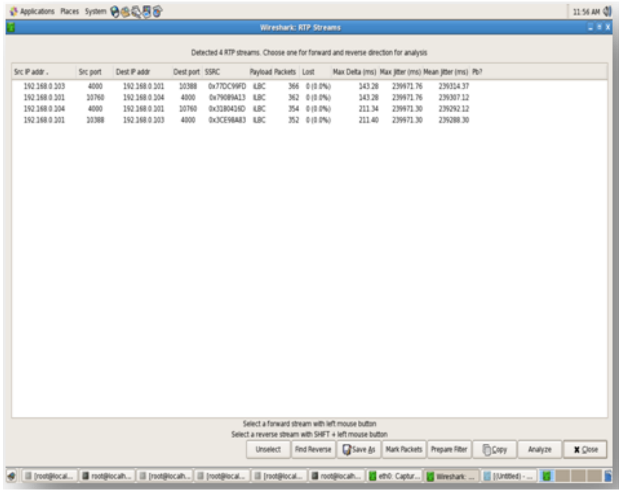

Fig.12. iLBC_WIRE SHARK

## VII. GSM VOCODER

The Full Rate (FR or GSM-FR or GSM 06.10 or typically merely GSM) was the primary digital speech coding standard utilized in the GSM digital mobile phone system. It uses linear predictive coding (LPC). The bit rate of the codec is 13 kbit/s, or 1.625 bits/audio sample (often padded out to 33 bytes/20 ms or 13.2 kbit/s). the standard of the coded speech is quite poor by trendy standards, however at the time of development (early 1990s) it had been an honest compromise between process quality and quality, requiring solely on the order of 1,000,000 additions and multiplications per second. The codec continues to be wide employed in networks round the world. bit by bit francium are going to be replaced by increased Full Rate (EFR) and adjustive Multi-Rate (AMR) standards, which give abundant higher speech quality with lower bit rate.The following screenshots shows the simulation results of GSM Vocoder mistreatment WIRESHARK.

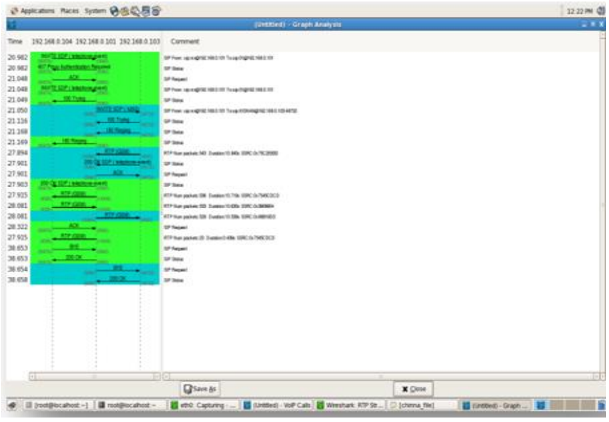

Fig. 13. GSM_WIRE SHARK

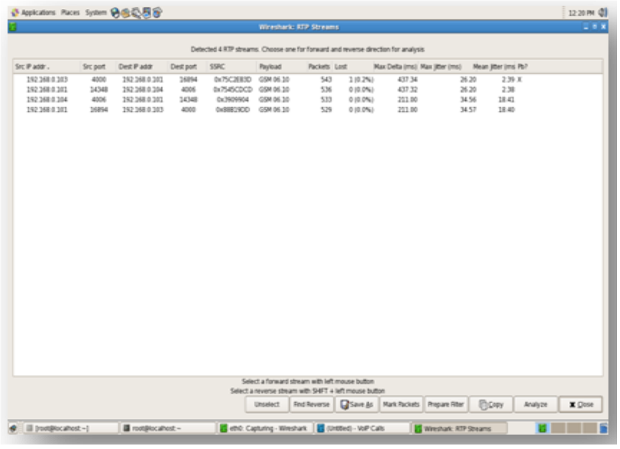

**Fig.14. GSM_WIRE SHARK**

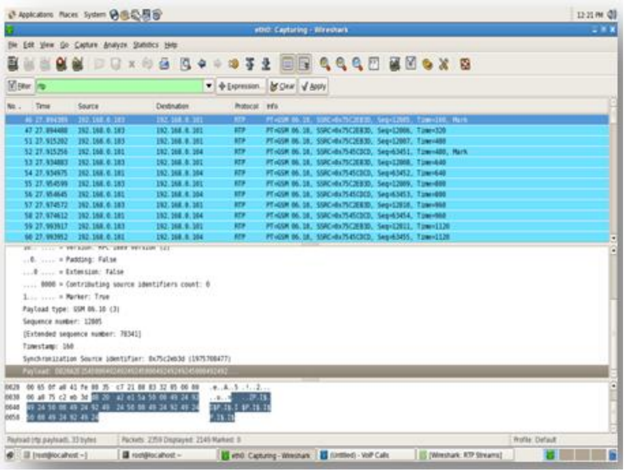

**Fig.15. GSM_WIRE SHARK**

## VIII. SPEEX VOCODER

Speex is AN audio compression codec specifically tuned for the replica of human speech and additionally a free software speech codec which will be used on VoIP applications and podcasts. it's based on the CELP speech coding formula. the subsequent screenshots shows the simulation results of SPEEX Vocoder exploitation WIRESHARK.

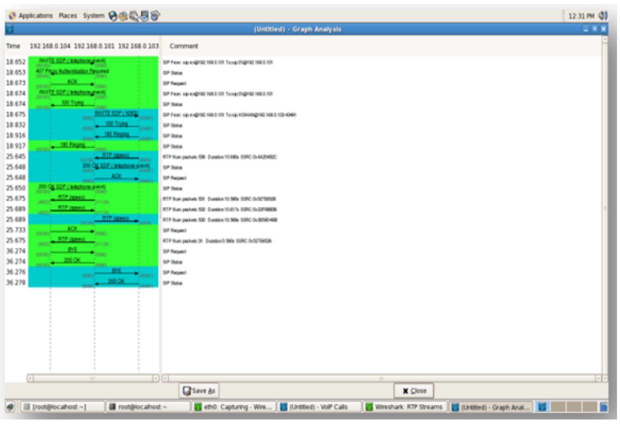

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**Fig.16. SPEEX_VoIP**

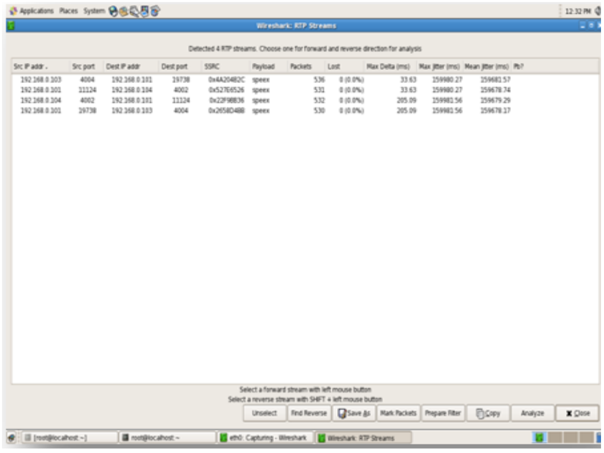

**Fig.17. SPEEX_RTP**

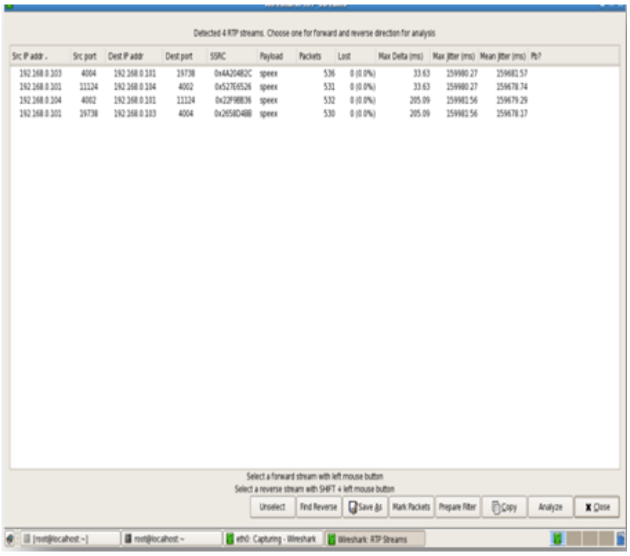

**Fig.18. SPEEX_WIRE SHARK**

## IX. RESULT AND DISCUSSION

Various Vocoders performance in terms of the percentage of packets of data saved is evaluated using Wireshark simulator with the help of Asterisk PBX. In this evaluation we found that the ILBC vocoders with Wireshark shows performance about 18.83% percentage of packets of data saved which is more compared to the other vocoders. The complete data showing their performances are listed in below table 1.

**Table 1: Performance of Various Vocoders with Wireshark simulator**

| Type of Vocoder | Original packets of data to be transmitted | Actual packets of data transmitted | Percentage of packets of data saved |
|---|---|---|---|
| G.711 ALaw | 2500 | 2132 | 14.72 % |
| G.711 µLaw | 2222 | 2118 | 4.69 % |
| G.722 | 2458 | 2173 | 11.59 % |
| GSM | 2359 | 2149 | 8.91 % |
| ILBC | 1774 | 1440 | 18.83 % |
| SPEEX | 2356 | 2134 | 9.42 % |

## X. CONCLUSIONS

Various Vocoders performance is evaluated using Wireshark simulator. In this evaluation we found that the ILBC vocoders with Wireshark shows performance about 18.83% percentage of packets of data saved which is more compared to the other vocoders. More research work can be done in improving the performance of vocoders and thereby optimizing the bandwidth also which is essential in mobile networks.

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