IJISET - International Journal of Innovative Science, Engineering & Technology, Vol. 2 Issue 4, April 2015.

www.ijiset.com

ISSN 2348 – 7968

Analysis Of Qos In Asterisk Based Communication System

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Abstract**—***QOS refers to Quality of Service. Asterisk is an open source framework used for implementing communication systems. We have implemented multimedia communication system using Asterisk. This paper depicts the overall performance characteristics of an asterisk system which we analyzed in our project. For a good communication, the performance characteristics are vital to be examined. Tests were conducted which lasts for a few minutes. The Test provides technical call information.*

Keywords: Asterisk, QOS, Performance, Multimedia Communication.

I. INTRODUCTION

Success of a product depends on the quality, grade and cost. Among the three, Quality is the most important factor which determines whether the product is good or not. After manufacturing a product the first job is to test whether the product is giving the required output which is called testing. Then comes the Quality of Service. Likewise we have implemented a prototype of Multimedia Communication system using Asterisk Platform.Jitsi is a SIP Communicator, which has an inbuilt feature of showing technical call info.

II. ABOUT THE PROJECT

Asterisk is open source. It implements communication in software in place of hardware. This lets new features to be rapidly added with slight effort. Asterisk uses Linux operating system such as CentOS, Debian and RaspbianWheezy. Our Project uses Asterisk which is installed in Raspberry Pi. Raspberry Pi is a small computer on which the Asterisk will be installed.

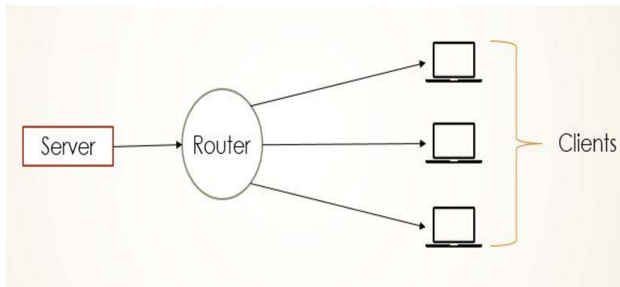

Fig 2.a Block Diagram of Implemented System

III. HARDWARE AND SOFTWARE PACKAGES USED

Hardware Used:

- Raspberry Pi mode B+ (Server)
- IP Time G104M Wireless Router

Software Packages Used:

- Asterisk 1.8
- Zaptel 1.4.12.1
- LibPRI 1.4.12
- Jitsi 2.6.5390

IV. OUTPUT

I. Placing a Call:

ISSN 2348 – 7968

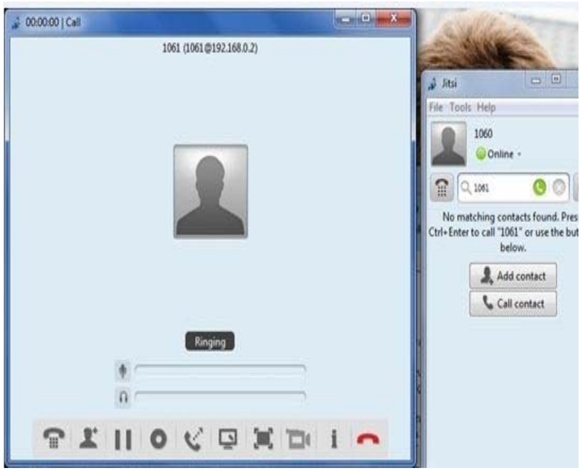

FIG 6.a User 1060 placing a call

II. Receiving a call:

FIG 6.b User 1061 receiving a call

III. Call Connected:

FIG 6.c Call connected

V. QOS PARAMETERS

The Call Quality depends upon the below listed parameters

i. Bandwidth:

It is the measure of rate of data transfer or thoughput, measured in bits per second.

ii. Loss Rate:

It is the rate of packets which failed to reach the destination.

iii. RTT:

RTT is Round Trip Time which defines the time taken for a packet to travel from a specific source to a specific destination.

iv. Jitter:

Jitter is the deviation of latency from packet to packet.

VI. QOS ANALYSIS

i. Audio Call:

Call information :
Identity : 1060@192.168.0.2 (SIP)
Signalling call transport : UDP

1061@192.168.0.2 :
Call duration : 00:01:47

Audio info :
Media stream transport protocol : UDP / SRTP (Key exchange protocol: ZRTP AES-CM-128/DH3K)
Codec / Frequency : GSM / 8000 Hz
Local IP / Port : 192.168.0.7 / 5018
Remote IP / Port : 192.168.0.4 / 5018
Bandwith : ↓ 18 Kbps ↑ 13 Kbps
Loss rate : ↓ 0% ↑ 0%
Packets decoded with FEC : 0
Packets currently being discarded : 0%
Number of discarded packets : 19 (0 late, 11 full, 8 shrink, 0 reset)
Adaptive jitter buffer : enabled
Jitter buffer : ~80ms; currently in queue: 2/8 packets
RTT : 12 ms
Jitter : ↓ 6 ms ↑ 6 ms

ii. Video Call:

Call information :
Identity : 1060@192.168.0.2 (SIP)
Signalling call transport : UDP

1061@192.168.0.2 :
Call duration : 00:01:47

Video info :
Media stream transport protocol : UDP / SRTP (Key exchange protocol: ZRTP AES-CM-128)
Video size : ↓ 640 x 480 ↑ N.A.
Codec / Frequency : H264 / 90000 Hz
Local IP / Port : 192.168.0.7 / 5020
Remote IP / Port : 192.168.0.4 / 5020
Bandwith : ↓ 331 Kbps ↑ 773 Kbps
Loss rate : ↓ 0% ↑ 0%
Packets decoded with FEC : 0
Packets currently being discarded : 0%
Number of discarded packets : 0 (0 late, 0 full, 0 shrink, 0 reset)
Adaptive jitter buffer : enabled
Jitter buffer : ~200ms; currently in queue: 0/21 packets
RTT : 276 ms
Jitter : ↓ 6 ms ↑ 6 ms

VII. CONCLUSION

Tests have been conducted. The audio quality appears to be a bit echoed, this can be rectified by enabling the adaptive jitter buffer, after that there was a significant improvement in call quality. Also different codecs shows different characteristics.

ACKNOWLEDGEMENTS

The Images posted in this paper was extracted from our other paper '*Implementation of Real-Time Multimedia Communication using Asterisk Platform*' which is yet to be published.

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