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Voice Quality of VoIP in High Availability Environment

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Abstract- The development of telecommunication technology specified the Internet Protocol (IP) based technology for the next generation network. Voice over Internet Protocol (VoIP) has been introduced to overcome future telephony demand. However, these rapid changes encountered some issues, and the most critical is how to provide the services availability and reliability equally to circuit based telephony. Virtualization is widely used not only for hardware efficiency and maintenance, but also for High Availability support. Virtualized environment provides the ability among servers to migrate or replicate into another machine, even when they are running their services, which is known as Live Migration. In this paper, the voice quality of VoIP service when running on the High Availability system in virtualized environment is studied and examined. The objective analysis by using quality of services (QoS) attributes is conducted as well as the subjective analysis using Mean Opinion Score (MOS). The work utilizes Xen® Hypervisor with modified Remus extensions to provide the High Availability environment. Remus approach using checkpoint based is deployed to copy the primary server to the backup server. A range of 40ms – 900ms has been applied as time interval of checkpoint. The results show that the mean jitter is 9,98 ms, packet loss 3,12% and MOS 3.61 for Remus 400ms checkpoint. MOS with different checkpoint time interval is also presented.

Keywords: VoIP, High Availability, virtualization.

I. Introduction

The telecommunication services demand has been increased year by year. The telecommunication technology is also improved by utilizing the Internet Protocol (IP) based technology as the next generation network. Voice over Internet Protocol (VoIP) is introduced to overcome future telephony demand. However, these rapid changes encountered some issues, in which several researches have been conducted to make sure the IP telephony could provide the performance availability and reliability equally to the circuit based telephony.

One of the most significant discussion is how to provide a high carrier-grade for VoIP to overcome the public telephony availability. Those carrier grades have to reach 99.999 percent of the time, well known as five nine rules, which means those networks must have the maximum 5 minutes downtime over a year.

Virtualization is widely used not only for hardware efficiency and maintenance, but also for High Availability support. Research on the virtualized based servers have been introduced to provide high carrier-grade in an IP based network. Virtualized environment provides the ability among servers to migrate or replicate the contents into another machine, even while they are still running their services. This is known as Live Migration. Virtualization support a continuous real-time replication/live migration of running servers for ensuring High Availability. Since whole machine is replicated, there is no need to modify the system or software, and the risk of data inconsistency is minimized. One of virtualization hypervisor that able to do live migration is Xen® Hypervisor [1].

In this paper, the impact of continuous live migration process using Xen® with Remus for high available environment of VoIP to the voice quality of VoIP is investigated and examined.

II. Related Work

The work in [2] presented a brief description how to do the continuous live migration processes in LAN environment. It provided an application transparent solution for high availability services. The application transparent solution means the client will not aware if the server has changed due to a fail over. This approch provide a significant improvement for High Availability services. This implementatios are suitable for general purposes packet data such as web service, mail service or FTP service.

The High Availability for VoIP service has been studied in [3]. This work proved that a general High Availability technique mentioned above is not suitable for voice and any real time services. The most significant issue is that the process generate very high jitter, latency and packet loss. The work in [4] proposed classification of packet and ensuring User Datagram Protocol (UDP) and Real Time Protocol (RTP) packets would not be buffered. This technique would help to decrease latency and jitter for VoIP service when using the continuous live migration in order to provide a High Availability service. Those research works only with one VoIP call during measurement. Therefore, it needs to expand the research to simulate the real server load by using call generator. Beside the objective voice quality analysis using the QoS attributes i.e., jitter, delay and packet loss, this work also utilises the subjective analysis using Mean Opinion Score (MOS).

III. VOICE OVER IP (VOIP)

Internet Protocol (IP) is a packet switching technology that divides messages into several packet segments and encapsulate these packets by adding some information before sent them into the network. IP is defined by The Internet Engineering Task Force (IETF) RFC 791 [5] on September 1981 with a revision in RFC 1349 [6] and RFC 2474 [7].

VoIP is a technology that enables communication between end stations using IP based network. This emerging technology is widely adopted and competed with traditional circuit based switching telephony. Adoption and implementation of VoIP have increased in the past few years and certainly VoIP technology will replace traditional telephony. European Telecommunications Network Operators Association (ETNO) reported the number of VoIP users have been increased more than 400 percent in the past 5 years in Europe, which means 1 out of 4 customers used to manage VoIP in Europe [8]. This trend has kept increasing as depicted in figure 1.

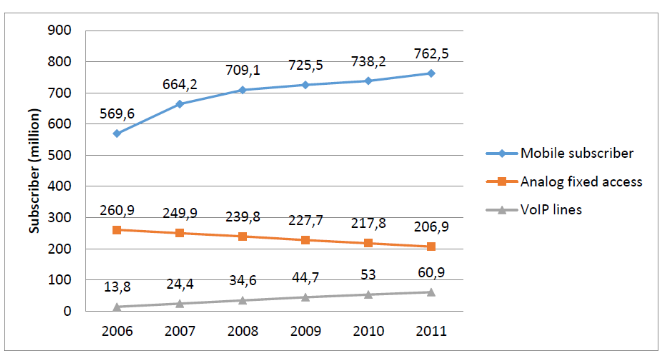

Fig 1. Various voice services take up growth in Europe [8]

VoIP is implemented by using a protocol suite to define how the process works. There are two main signaling protocol suites of VoIP i.e., the H.323 protocol defined by International Telecommunication Union – Telecommunication Standardization Sector (ITU.T) [9] and the Session Initiation Protocol (SIP) defined by The Internet Engineering Task Force (IETF) RFC 2543 in 1999 [10] and revised by RFC 3261 in 2002 [11]. SIP is defined by IETF as a part of its multimedia data and control protocol suite. SIP works together with other IETF protocol such as Session Description Protocol (SDP), Real Time Streaming Protocol (RTSP) and Session Announcement Protocol (SAP).

IV. HIGH AVAILABILITY VOIP SERVICES AND CHALLENGE

The challenge on VoIP service is that how to provide High Availability and fault tolerant system. Availability means the ability of the server to be in a state to perform required services requested by client at any given time interval. The traditional circuit based telephony offered a reliable connection and high quality voice.

Since VoIP services use the IP, in which naturally does not provide a reliable connection, a packetization and network encoding is needed. Those processes may cause high delay, jitter and packet loss when travelled in the network. In fact, IP is not designed to deliver voice or other real time services. Telephony standard to deliver class 5 High Availability service, with only 5 minutes downtime in a whole year is difficult and will have expensive cost. Nevertheless, VoIP service must fulfil this requirement to fit the telephony standard.

Figure 2 depicts the general model of high available VoIP service as conducted in this work. Two servers are required, the first server is primary server that serve the clients, and the other one is backup server. When problem occurs in primary server, the backup server will take over with minimal downtime.

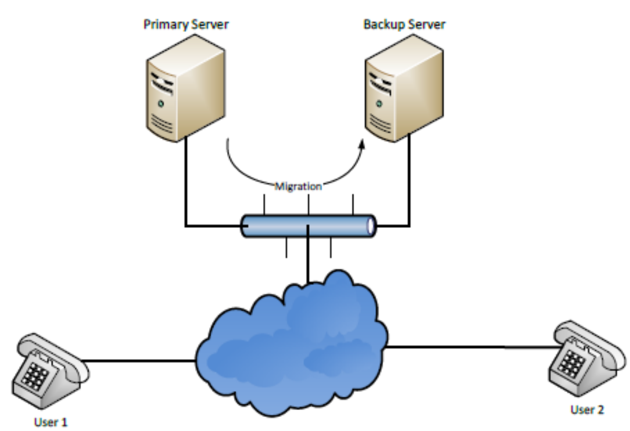

Fig 2. High Availability VoIP model

Recent developments in the field of virtualization have led to a renewed interest in how to take advantages of virtualization technology in order to support High Availability and fault tolerant systems. Virtualized environment offers High Availability and/or fault tolerance for server using continuous live migration of virtual machine between the primary server and backup server. Paravirtualization installed its hypervisor/virtual machine monitor (VMM) directly into a hardware computer without being part of the operating system. Hypervisor has responsibility to bridge the communication directly from virtual machine into hardware layer. Therefore, Paravirtualization approached has been chosen since it delivers higher performance and minimize the virtualization impact in VoIP server as hardware resources directly controlled by hypervisor layer.

The work in [2], provided a solution for transparent continuous live migration using Remus as the enhancement of Xen live migration process. Remus is a lower layer availability protection software platform, so there is no required modification of application running on top of it, therefore the availability is improved. Remus works above the hypervisor/VMM and encapsulated the protected virtual machine. Remus enhanced the live migration process by doing the live migration process in iterative ways. Figure 3 depicts the work of Remus, where each copies of primary server, called checkpoint, is sent to the backup server iteratively with user defined interval. Remus provides high availability mainly for data transfer or non realtime service. Remus buffered client request and respond it until the checkpoint's acknowledge is received by primary server.

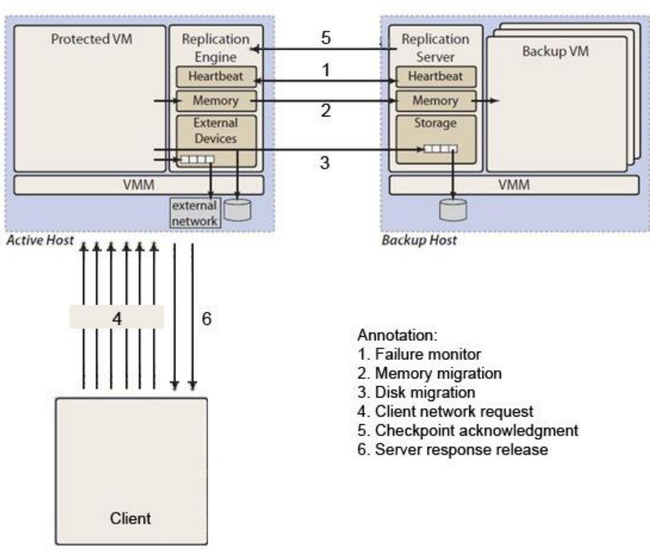

Fig 3. Remus High Avalability Works [1]

This buffering rule is not suitable for the realtime service such as VoIP since it needs packet to be transferred as soon as possible but with tolerate packet loss. Table 1 describes the comparison of network requirement for realtime and non realtime services in order to run properly. Those requirements give the reason why the general purposes of high availability works, as described in [2], do not work properly. The work in [4] enhanched the Remus by modified its buffering behaviour. The realtime protocol such as User Datagram Protocol (UDP) will not buffered by the server due to this protocol, though it is mainly used by realtime application.

TABLE I. REAL TIME AND NON REAL TIME SERVICE NETWORK REQUIREMENT

VoIP / Realtime services requirement	Non realtime services requirement
Lower Delay Lower Jitter Packet loss tolerable	Delay is tolerable Jitter is tolerable Packet loss is not tolerable

V. SIMULATION AND RESULTS

In this paper, the impact of high available service in VoIP calls quality based on works in [2] and with its enhancement works in [4] is studied and examined. Packets during VoIP call using High Available VoIP service is captured for further objective voice quality analysis using both the QoS attribute i.e., jitter, delay and packet loss, and the subjective analysis using Mean Opinion Score (MOS). The ITU – T standard in [12] describes parameters for good telephony communication i.e., the average MOS of 4.4, delay < 250 ms, jitter < 30 ms and packet loss < 5 %.

Our testbed, as depicted on figure 4, consist of Primary Server and Backup Server (IBM BladeCenter HS20 with Intel Xeon 3,4 GHz, 3Gb memory, 26 GB SCSI HDD, 2 Gigabit ethernet ports) runs the Xen virtualization (Xen 4.2.1) with xm toolstack and Remus. Both servers use Ubuntu 12.04 64 bit with 3.2.0-29-generic kernel as Domain-0 and Opensuse 12.3 with 3.7.10-1.1-xen kernel uses as Domain-U since it has suspend event channel support. One computer equipped with call generator using SIPp [13] and voice quality measurement tools, also act as packet sniffer. Additionally, FreeSWITCH [14] is used as VoIP server.

High Available VoIP server run in primary server, as depicted in figure 4, is loaded with calls using SIPp call generator with minimum CPU load 50% representing the real server work load. Another call are made using SIP Phone and the packet is sniffed to capture the data packets for further objective and subjective voice quality analyses. All call is using SIP signalling protocols and G.711 codec since it is widely used by VoIP systems. Captured Packet is analyzed using Wireshark [15] for qualitative analysis and using AQuA [16] from Sevana for subjective analysis then the result is presented.

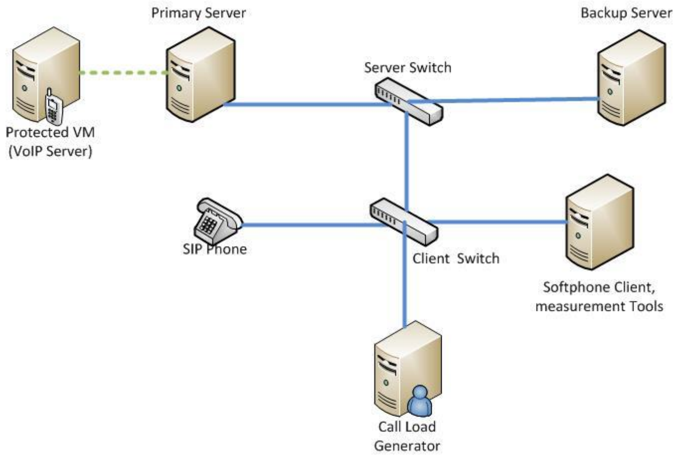

Fig 4. High level of test bed architecture.

In this result, Jitter calculation is considered to present the impact of Remus continuous live migration in VoIP communications. RTP is a protocol defined by IETF RFC 3550 [17] that responsible to provide real time data delivery service. Jitter is calculated using interarrival jitter (J) and mean deviation of the difference (D) defined for pairs packet as shown on equation 1 [17].

$$J(i) = J(i-1) + \frac{(|D(i-1,i)| - J(i-1))}{16} \qquad (1)$$

Figure 5 below depicts the jitter distributions during call without modification of Remus buffering rules that runs the high avalaibility system. Analysis of captured packet show the maximum jitter is 96,72 ms and packet loss 92,55%. The result shows that jitter level and packet loss is very high above the ITU-T standard as impact of buffering during checkpointing. The response of the server has shown its correspondence with the user's defined checkpoint interval of 400 ms.

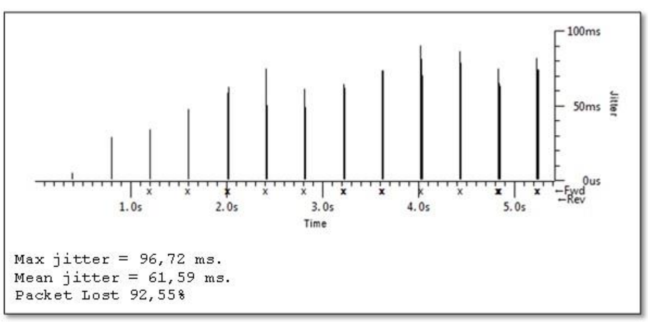

Fig 5. Jitter level in general Remus HA system

Beside a rapid increased of jitter and delay, the system also produces a high packet loss. Actually, the packets are not completely loss since buffering process holds the response during checkpointing. The packets are not valid in the received time and should be discarded. Based on work in [4], Remus buffering rules is modified to un-buffered realtime packets. Modification is performed using packet filtering and classification of the packet.

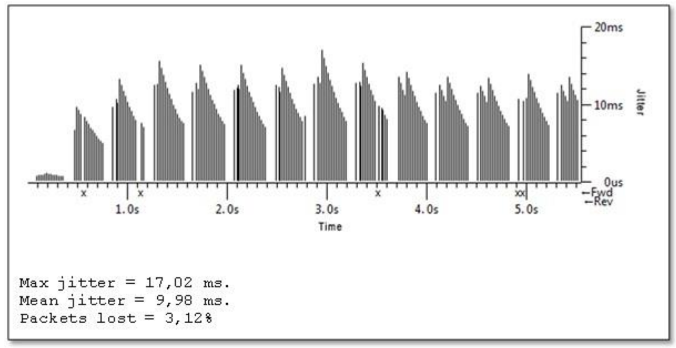

Fig 6. Jitter level in modified Remus HA system

Figure 6 shows the modification impact in High Availability VoIP service. The delay, jitter and packet loss have greatly improved. Maximum jitter is reduced from 96,72 ms to 17,02 ms and also the mean jitter from 61,59 ms to 9,98 ms. Packet loss is also significantly reduced from 92,55% to 3,12%.

Further analysis of subjective attribute using MOS is estimated using voice quality analyser software [16]. This analyser works by sending the recorded voice during the call and compared the echoed voice with original voice and estimates the MOS. This paper also proof that user defined checkpointing interval has a significant impact in voice quality, the result is shown in table 2. The most suitable checkpoint interval is every 400 – 600 ms.

TABLE II. CHECKPOINT INTERVAL IMPACT

Remus Checkpoint (ms)	MOS	Max Jitter (ms)
40	1,89	37,21
70	3,23	6,5
100	3,37	5,03
150	3,47	3,46
200	3,55	3,45
300	3,59	2,44
400	3,61	1,3
500	3,63	1,21
600	3,62	1,32
700	3,58	1,34
800	3,57	2,48

VI. CONCLUSSION AND FUTURE WORKS

This paper show that high available VoIP service could be provided using virtualization approach with modified Remus. The result of jitter, delay, packet loss and MOS of voice quality is still acceptable according to the guidelines of ITU-T. The results shows that the mean jitter is 9,98 ms, packet loss 3,12% and MOS 3.61 for Remus 400ms checkpoint. Further works should consider the optimization of packet compression and failover mechanism in Wide Area Network (WAN) due to network characteristics i.e., high delay, jitter and packet loss in WAN

REFERENCES

[1] Barham, P. Dragovic, B. Fraser, K. Hand, S. Harris, T. Ho, A. Neugebauer, R. Pratt, I. Warfield, A. Xen and the art of virtualization. SOSP ’03: Proceedings of the nineteenth ACM symposium on Operating systems principles. 2003

[2] Cully, B. Lefebvre,G. Meyer,D. Feeley, M. Hutchinson,N. Warfield, A. Remus: High Availability via Asynchronous Virtual Machine Replication. NSDI'08 Proceedings of the 5th USENIX Symposium on Networked Systems Design and Implementation. 2008

[3] Hlavacek, J. Bestak, R. Configuration of Live Migration for VoIP Applications. Mechatronika 15th International Symposium.IEEE. 2012

[4] Hlavacek, J. Bestak, R. Live Replication of Virtualized VoIP Servers. The Eighth International Multi-Conference on Computing in the Global Information Technology (ICCGI). 2013

[5] Information Sciences Institute University of Southern California. Internet Protocol. IETF RFC 791. Internet Engineering Task Force (IETF). 1981

[6] Almquist, P. Type of Service in the Internet Protocol Suite. IETF RFC 1349. Internet Engineering Task Force (IETF). 1992

[7] Nichols, K . Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers. IETF RFC 2474. Internet Engineering Task Force (IETF). 1998

[8] Annual Economic Report. European Telecommunications Network Operators’ Association (ETNO). URL: http://www.etno.eu/datas/ publications/economic-reports/etno-economicreport-2012.pdf. 2012

[9] Packet-based multimedia communications systems. ITU.T Recommendation H.323. International Telecommunication Union Telecommunication Standardization Sector (ITU.T). 2009

[10] Handley, M. Schulzrinne, H. Schooler, E. Rosenberg, J. SIP: Session Initiation Protocol. IETF RFC 2543. Internet Engineering Task Force (IETF). 1999

[11] Rosenberg, J. et all. SIP: Session Initiation Protocol. IETF RFC 3261. Internet Engineering Task Force (IETF). 2002

[12] One-way transmission time. ITU.T Recommendation G.114. International Telecommunication Union Telecommunication Standardization Sector (ITU.T). 2003

[13] http://sipp.sourceforge.net/. acessed : 4 March 2014

[14] http://www.freeswitch.org. acessed : 4 February 2014

[15] http://www.wireshark.org. acessed : 4 February 2014

[16] http://sevana.fi/. acessed : 21 march 2014

[17] Schulzrinne, H. Casner, S. Frederick, R. Jacobson, V. RTP: A Transport Protocol for Real-Time Applications. IETF RFC 3550. Internet Engineering Task Force (IETF). 2003