IPv6 Performance Analysis, Transition Strategies and Considerations for VoIP Deployment

Paschal A. Ochang, Phil Irving

***Abstract*—** The transition into IPv6 is gradually evolving due to the numerous advantages it offers, but integration with popular telephony applications like Voice over Internet Protocol (VOIP) has been lengthy. The implementation of VOIP technologies by organisations has been rapid due to the numerous advantages it offers in terms of low cost and the ability to transfer voice over an internet protocol network, but most of the implementation has been with IPv4 due to the fact that IPv4 is already implemented by most networks. This article looks at the support IPv6 provides for VOIP based on the enhanced features it possesses while keeping in view the performance analysis of VOIP deployment with IPv6 and other characteristics that affect their integration. Furthermore the paper analyses different deployment strategies that can be used to deploy and implement VOIP with IPv6 and how to maintain interoperability between VOIP IPv6 networks and VOIP IPv4 networks.

***Index Terms*—**Delay, Jitter, Latency, IPv4, IPv6, Performance Analysis, QoS, Translation, Tunelling, VoIP.

I. Introduction

The internet protocol version 6 (IPv6) was developed as the next generation protocol with the view of replacing the IPv4 [18] due to the limited number of addresses provided by the IPv4. Although the 32bit address of the IPv4 could be extended theoretically, the IPv6 provided a 128bit addressing scheme therefore providing an increased number of addressable nodes [2]. Considering the influx of IPv6 into current network technologies, its effects on one of the promising technologies called Voice over Internet Protocol (VOIP) has to be considered. Reference [18] pointed out that VOIP is actually gaining popularity and based on this popularity it could serve as the major component of network traffic in the future. Putting all this into consideration a performance analysis is necessary in other to determine the possibilities of integrating IPv6 and VOIP, keeping in view the characteristic of both technologies and the merits and demerits they tend to provide due to their integration, and if possible recommendations can be made for further deployment. Section 2 of this paper gives a brief overview of VOIP and IPv6 while section 3 gives an overview of the features IPv6 possess that makes it a good choice for deployment with VOIP. Section 4 takes a look at the performance analysis of VOIP due to its integration with VOIP and other factors affecting their integration. Section 5 analyses the deployment of IPv6 together with VOIP while keeping in view transition mechanisms from IPv4 VOIP systems to IPv6 VOIP systems in other to maintain interoperability between the different IP versions. Section 6 gives recommendations for the deployment of VOIP using IPv6 based on the overall analysis of the paper.

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II. Background

A. IPv6

The internet protocol version 6 (IPv6) was developed by the IETF as the modern successor to the IPv4 [5], with a major aim of providing a robust addressing scheme with address sizes of 128bit (2128) as compared to the IPv4 32bit (232) address sizes thereby increasing the address sizes by fourfold [3]. Reference [3] pointed out that some changes were made to the IPv4 datagram header in other to produce the IPv6 datagram header, these changes are as follows

- The header size of IPv4 was increased from 40 bytes to 80 bytes to give the IPv6 an 80 bytes header.
- New fields were added mainly for traffic class and flow label thereby enhancing Quality of Service (QoS).
- The Time To Leave (TTL) replaced the hop limit.
- The Type of Service field, Flags, Fragment, Identification, IP header Length and Header Checksum fields were also removed.

These changes as shown in fig. 1 below has given the IPv6 the ability to offer various advantages and benefits into the world of networking and communications. Reference [8] pointed out that the IPv6 offers

- Better support for real-time traffic because of the label flow field in its datagram which enables routers to recognise the end-end flow that packets belong to
- Scalability due to the fact that it provides a high number of unique node addresses
- Plug and play support for network devices therefore one does not have to configure network parameters like the gateway, sub network mask, non-published dynamic IP addresses or any other parameters

- More enhanced and mobility mechanisms
- Better routing and addressing hierarchy
- Optimisation due to the fact that it removes obsolete IPv4 characteristics
- Extensibility because it accommodates new extensions and new options.

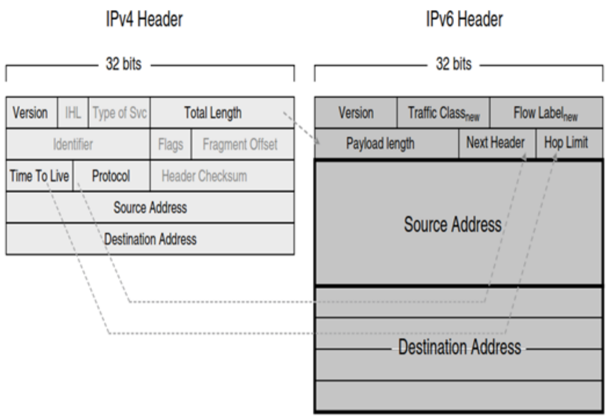

Fig. 1. A comparison of the IPv4 and the IPv6 headers.

Although the IPv6 offers all the above mention benefits its implementation and acceptance level has actually been on the slow side, as of the month of October 2011 core IPv6 routing tables contained less than 7,500 entries while core IPv4 routing tables had almost 400,000 entries [10].

B. VoIP

Voice over Internet Protocol (VoIP) can be described as IP Telephony which enables digitised voice streams to be transported over traditional data networks or IP based networks such as the internet [6]. VOIP adds value to existing telephony devices and reduces cost in terms of long distance calling between organisations and can be implemented on any IP based network. VOIP uses various signalling protocols to set up and tear down calls, these include Skinny Client Control Protocol (SCCP) by Cisco, IAX by Digium, Session Initiation Protocol (SIP) which was developed the IETF and H.323 which was developed by the ITU, although the SIP protocol and the H.323 protocol standout as the major contenders [15]. VOIP uses codec standards to convert speech from its analogue form to a more digitised manner that can be transferred over the network; therefore the perception of the end user on the voice quality received depends on the codec used [11].

III. IPV6 SUPPORT FOR VOIP

The new features added to the IPv6 datagram makes it a suitable protocol for handling VOIP traffic, this is justified by the addition of a flow label and traffic class fields as shown in fig.1 which provide better support for real time traffic and better quality of service [8]. The flow labelling method enables IPv6 to label packets which require special handling by routers [2]. A Flow can be a sequence of IPv6 packets carrying VOIP traffic sent by a particular source which can be a soft phone or User Agent (UA) to a particular destination node. The traffic class field of the IPv6 header can be used in such a way that an originating node or a forwarding router can actually identify or distinguish the priority of a given IPv6 packet which might be packets carrying real time audio traffic, and based on the priority the packet can be forwarded to the destination UA thereby enhancing Quality of Service. The IPv6 implements three types of addresses that are suitable for next generation communication and multimedia systems [3], this include the

- Unicast address type
- Muticast address type
- And the Anycast address type

The Unicast address is used to identify or represent a single interface within the scope of the unicast address type, this could be VOIP (SIP) soft phone or handset deployed and in an IPv6 environment.

A multicast address is a scheme or a mechanism implemented by the IPv6 that allows multiple VOIP interfaces to be identified in other to deliver a packet to the identified interfaces. The multicast mechanism is very useful for supporting extended VOIP functionalities like audio conferencing and bridging through the internet. The multicast addressing mechanism can also be used to support the Internet Protocol Multimedia Subsystem (IMS) and IPTV in terms of features such as program distribution [8].

The anycast address mechanism is used to send an IPv6 packet to the nearest node identified by an address based on a potential set of members or receivers, this method can be used to support group distribution of VOIP Voicemail [8].

IV. PERFORMANCE ANALYSIS OF VOIP AND IPV6 INTEGRATION

The implementation of VOIP will be a driving force in the promotion of IPv6 [17], but before further integration can occur, the performance of VOIP on IPv6 needs to be considered. Various factors affect the overall user experience and performance of VOIP on the IP network and most of this factors have been experienced on the IPv4, although the IPv6 is a more advanced version of IPv4 most of the native IP characteristics are inherited.

A. Mean Opinion Score (MOS)

The Mean opinion Score (MOS) is a method used in telephony in other to determine the quality of a network from a human perspective. A tester or a user can rate the audio quality of a network using a rating scheme of 1 to 5, where 1 is used to signify the lowest audio quality perceived and the highest audio quality perceived is indicated by 5. This method can also be applied in measuring the quality of the IP network and its performance in terms of VOIP. The IPv6 has a larger packet size compared to the IPv4 and this may lead to overhead [18]. Some research have shown that with significant background voice traffic there is a significant loss of packets and reduction of audio quality leading to a low MOS. Reference [18] pointed out that when background voice traffic was injected into the network at around 50Mbps there was no significant loss of packets and the MOS score between IPv6 and IPv4 remained at 4.41 but when the background audio traffic was around 100Mbps there was about 5% loss in packets on the systems running IPv6 and the MOS score reduced to 4 while in the systems running IPv4 there was no significant loss in packets and the MOS score was still maintained at 4.41, but with further injection of audio traffic even the IPv4 started dropping packets and the MOS score reduced. Clearly this shows that most of the audio codec still support more of the IPv4 but still calls for the need for packet loss handling techniques, and the IPv4 datagram does not have the ability to provide these techniques. Therefore the IPv6 stands out as the protocol that has the ability to provide a solution to the issue of packet loss in VOIP due to its quality of service support.

B. B. Quality of Service (QoS)

The performance of IPv6 to a large extent depends on its ability to provide a good QoS when deployed with VOIP. The performance of VOIP suffers from underlying factors such as jitter, latency and delay and this happens as a result of the fact that they is always a possibility of packet loss when packet switching or handing over occurs [11]. Previous research has pointed out some of the factors that trigger low QoS factors and they have been various implementations to identify and reduce the impacts of these factors using IPv6

1) Latency

Latency can be described by the amount of time it takes VOIP traffic to reach its destination in the network [9]. The amount of latency experienced by VOIP on IPv6 and can depend on the type of codec used; the G7.11 codec is mostly used because it offers simplicity low delay and excellent audio quality [11]. Latency at the Application layer of the OSI model is not only dependent on the internet protocol version used but also on the operating system used, [9] demonstrated this by generating VOIP traffic using both TCP and UDP while varying different codec on different operating systems running IPv6 and IPv4 respectively, and windows server 2003 operating system had a high latency with all different codecs used, while windows XP and windows7 operating systems had very low latency. This calls for operating system and codec considerations when integrating VOIP and IPv6.

2) Jitter and Delay

The amount of variation in latency and delay of VOIP traffic may result in service degradation causing jitter [12]. Although there are no significant differences in the mean and maximum jitter when comparing IPv6 to IPv4 [18], this is dependent on bandwidth and other link factors. VOIP is foreseen as an alternative to the public switched telephone network (PSTN) in the future, this has led to its implementation in business organisations and campuses where user agents (UA) are mobile. The mobility of users in a network led to the development of Mobile IPv6 (MIPv6) by the IETF [13] which is defined in RFC 3775. When a mobile user agent in VOIP architecture roams from one access point (AP) to another (layer 2 handoff) or from one subnet to another (layer 3 handoff) [6] they might be loss of packets depending on the handover speed [14] resulting in jitter. Fast Handovers for Mobile IPv6 (FMIPv6) was designed by the IETF to handle traffic delay by anticipating a layer 3 handover and handling portions of the handover before the actual handover occurred. Reference [6] showed that on implementing FMIPv6 on a network although they were transmission delay which was as a result of routing, they was no loss of IPv6 packets during handoff.

V. DEPLOYMENT OF IPV6 AND VOIP

The deployment and implementation of IPv6 is gradually on the increase with most of the Asian countries spearheading the deployment of IPv6 due to their internet economic drive and the United States Department of Defence (DOD) decision in 2003 to migrate the network of the pentagon to IPv6 [4]. The missing pieces to the puzzle still remain the deployment of IPv6 maintaining interoperability, scalability and performance.

A. Transition Strategies

Transition in this scenario can be viewed as migration into native IPv6 which is a stage that is not currently achieved. Transaction can be viewed in four stages, this include: (a) IPv4 only networks, (b) IPv6 networks connected through IPv4 networks, (c) Interconnection of IPv4/IPv6 networks, (d) IPv4 networks connected through IPv6 network, (e) IPv6 only networks [5].

B. Transition Mechanisms

Various Transition Mechanisms has been designed by the IETF in other to handle various scenarios, due to the fact that this research paper is based on the considerations for the deployment of VOIP on IPv6 the transition mechanisms will be analysed in such a way that the user agents (UAs) running native IPv6 addresses will be able to communicate with other UAs running native IPv6 and considering heterogeneous scenarios the IPv6 UAs will also be able to communicate with UAs in IPv4 networks and vice versa

1) Dual Stack or Dual IP mode

This is a mechanism that can be used to deploy VOIP and IPv6 while also considering interoperability with native IPv4 networks. As shown in fig. 2 below in a dual stack method all UAs, routers and hosts in the network provide support for both IPv4 and IPv6 [8]

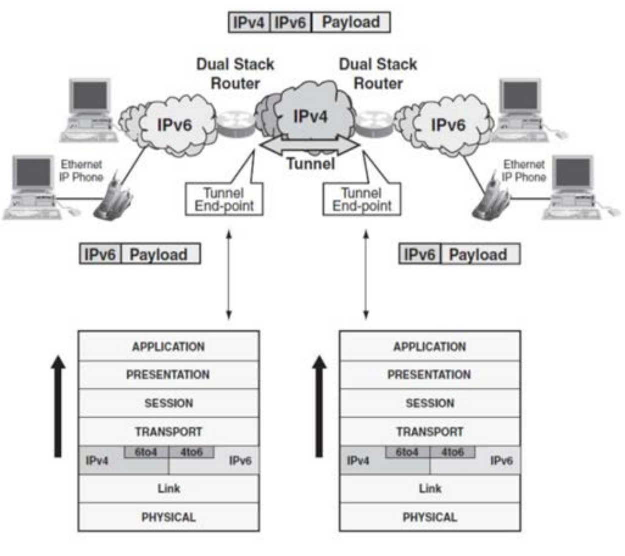

Fig. 2. Dual Stack or Dual IP method of implementing VoIP with IPv6

Most SIP UAs already support dual IP mode. For SIP communication to occur in this method the corresponding SIP UAs can be configured with both IPv4 (SIPv4) and IPv6 (SIPv6) addresses [17] and the corresponding Domain Name Service (DNS) entries are made on a DNS server if a DNS server is implemented. Furthermore the SIP header carries the corresponding IP address of the calling UA and the receiving UA can receive the call based on the IP address on the SIP header [4].

2) Tunnelling

As long as core networks and most internet service providers remain natively IPv4 enabled, the only method of connecting an IPv6 UA to another IPv6 node directly is by tunnelling the traffic [4]. This mechanism is used to encapsulate IPv6 packets carrying VOIP traffic in an IPv4 header so that the IPv6 packet can be routed through an IPv4 network to an IPv6 network cloud or node.

3) Translation

The deployment of IPv6 and VOIP (VOIPv6) will go a long way to resolve the perceived address shortage in VOIPv4; however a SIPv6 UA cannot communicate directly with a SIPv4 UA without the implementation of translation [17]. IPv6 to IPv4 translation in VOIP involves the translation of SIP messages and translation of the Real-time Transport Protocol (RTP) packets [1]. Translation can be achieved by implementing a SIP application level gateway (SIP ALG) [3] and Network Address Translation and Protocol Translation (NAT-PT) gateway, this method is justified by Chen and Wu (2005) who implemented a SIPv6 translator which contained a NAT-PT gateway and a SIP-ALG gateway in other to implement translation. In this method the SIP signalling packets are translated from one IP version to the other by the SIP-ALG and the NAT-PT is responsible for translating the RTP packets which contain the encapsulated audio stream. In fig. 3 below we propose a SIPv6 translator architectural design and prototype. The prototype design builds on the theory that with a SIPv6 translator UAs deployed with IPv6 addresses can communicate with IPv4 VoIP networks and also with PSTN phones through a PSTN gateway even though the PSTN gateway is an IPv4 gateway.

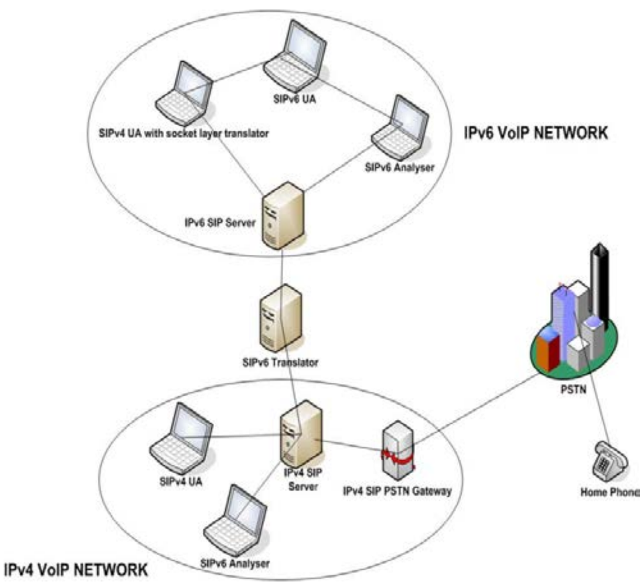

Fig. 3. Translation method of implementation IPv6 with VoIP by using a SIP translator

Another approach can be used to carry out translation by implementing a SIP gateway [7] or SIP proxy server to act as a back to back user agent (B2BUA). The SIP gateway can be a logical entity containing a proxy gateway and a media gateway. The SIP gateway is responsible for translating the SIP signalling messages between a native SIPv6 UA and SIPv4 UA while the media is responsible for forwarding RTP streams to the proxy server who controls the media sessions [4].

VI. RECOMMENDATION

IPv6 has been recommended by the IETF as a protocol with next generation capabilities but the adoption of the protocol has been delayed due to most of the existing infrastructure already supporting native IPv4. Some of the low rate in deployment of IPv6 by organisations has been due to the low knowledge base and awareness on the performance analysis of VOIP on IPv6 networks. The understanding of the support IPv6 can provide to VOIP can go a long way to convince organisations into the deployment of IPv6 with VOIP. The deployment strategies discussed in this paper gives a clear map on how to carry out a transition from the existing VOIP IPv4 architecture into an IPv6 VOIP based communication architecture while maintaining backward compatibility. Therefore we propose a practical implementation of the theoretical architectural designed proposed in fig 3. VOIP technology is already been adopted by mobile service providers and considering the daily increase in the number of users the depletion of IP addresses continue to become feasible, therefore it’s a good measure to implement VOIP and IPv6 as a future remedy to Quality of Service (QoS) issues and IP address depletion.

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