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# Cross-Layer Integration Approach for Improving QoS for IPv6 Based VOIP

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**ABSTRACT**

Voice over IP (VOIP) is today one of the most innovative IP based Communication Technologies in the Telecommunications industry. This has made it to enjoy a high degree of success in its application in small, medium and large scale enterprises, primarily to save cost as well as leveraging on its enhance functionalities such as mobility and scalability. Despite all its successes, VOIP still faces challenges with Quality of Service (QoS) degradation. This paper proposes a cross-layer model to effectively manage interactions in the data, network and transport layers guided by trade-off between three performance metrics that affect QoS of VOIP for an improved QoS for Voice over IPv6 (VOIPv6). The parameters taken into consideration in this proposed model are: packet loss, delay and throughput observe by the end-user.

**Keywords:** *Cross-layer Integration Manager (CLM), IPv6, QoS, Telecommunication, VOIP*

## 1. INTRODUCTION

Advances in real life network applications such as Voice Over-IP, Video Over-IP and TCP/IP based data communication services have led to considerable attention in Quality of Service (QoS) in IP networks. Voice over IP is a technology for transmitting voice calls over the Internet using data packet linked routes [1]. The main feature of this IP-based technology is that it sends conversations as IP based data packets over the internet thereby by enabling people to use the internet as a transmission medium for sending voice data in packets using IP rather than the traditional circuit transmission of the Public Switched Telephone Network (PSTN) [2, 3]. This IP based telephony brings benefits to both consumers as well as enterprise or commercial customers. One of the main reasons for embracing VOIP is to reduce both personal and commercial voice communication cost [4]. Skype and Internet Protocol Private Branch Exchange (IP PBX) are some applications that use VOIP technology and are bringing innovation in the telecommunications industry by providing high-quality and cost-effective solutions while saving cost of trunking as with the case of IP PBX which is replacing the Plain Old Telephone System (POTS) in Western Europe [5].

The following are summarized benefits of VOIP [5]:

1. Greater efficiency compared to traditional PSTN in terms of bandwidth usage.
2. Cost savings as it is setup on existing internet and LAN infrastructure.
3. Higher reliability as internet is used as packet transmission medium.
4. Supporting Innovation though integration with other applications such as email, web browser by providing services as voice delivery via email, click-to-call service on a website
5. Economic benefits to vendors of VOIP and customers alike, especially for long distance calls by avoiding access and settlement charges at least for now.

VOIP is digitized means of transmission due to its IP-based nature, which means a caller's analogue voice signal is first digitized, compressed and then encoded into digital voice stream using the CODECs (COder/DECoder) [4, 3]. Voice CODECs are standards set by the International Telecommunication Union – Telecommunication division (ITU-T), standards such as G.711, G.723 and G.729.

According to [6], the following are pending issues with VOIP QoS:

1. Lack of clarity as to where to locate QoS functionality
2. Lack of clarity on which QoS support should a protocol provide
3. No clear QoS architecture in place
4. Lack of consensus agreement on services, although there is a consensus to offer users a differentiated traffic.

## 2. QUALITY OF SERVICE (QOS) FOR IP BASED NETWORKS

QoS is service requirements that are set to guarantee performance which must be meet by the network while transporting a flow. It is the efficient use of network resources for a reliable delivery of data [7]. QoS performance guarantees could be measured using the following attributes or metrics which vary according to Service Level Agreement (SLA) and usually depends on the priority intended for a given application (voice or data) in question: bandwidth, delay (echo, talk overlap), jitter (inter-packet delay variation) and packet loss [8]. Therefore, improving QoS is hinged on reducing values of these metrics.

Internet generally uses “Best Effort” approach which is associated with IPv4, in which content of packet is not sensitive to real-time data flow [3] [7]. As demand exceed capacity, service degrades, thereby causing jitters, packet loss and delays; which a big hitch to real-time applications. [7]

## 3. QOS ISSUES WITH VOIP

Presently, VOIP uses IPv4 that is a best-effort service IP network with no built-in QoS and therefore arise several QoS issues. For example, quality of a voice-call can degrade significantly, if IP voice packets are lost or delayed at any point in the network between VoIP users [6] [5].

Users can also notice this quality degradation more in highly congested networks or over long distances. In order to address this quality issue, the next generation VOIP technology plans to use IPv6 that ensures QoS, a set of service requirements to deliver performance assurance while transporting voice traffic over the network [5].

Therefore the main aim of QoS in VOIP application is to provide some degree of certainty and management of bandwidth beyond the best-effort IP service [6] [7]. There are well established approaches in dealing with issues arising from these metrics as proposed by IETF (Internet Engineering Task Force) refer to as QoS protocols:

1. Best effort internet ( associated with IPV4, content of packet is not sensitive or important to real-time data)
2. ReSource reserVation Protocol (RSVP)
3. RSVP + Integrate Services (IntServ)
4. Differentiated Services (prioritization) (DiffServ)
5. Multi-Protocol Labelling Switching (MPLS)
6. Subnet Bandwidth Management (SBM)

Comparison of theses QoS protocols is summarized on table 1.

Delay is the main reason for packet loss during VOIP transmission at the receiving end. It is therefore, expected that reduction of delay will improve QoS of VOIP. However, the independence of the QoS protocols at different layers of the OSI model at the receiving end would tend to increase the overall transmission delay as shown on figure 1 [7]. An approach integrating the different QoS protocol at different layers of the OSI model will help in reducing the values of QoS of VOIP performance metrics. The concept of cross-layer integration is to allow the use of each QoS protocol at different OSI model layer as well as communicating and exchange of information with each other starting from the bottom layer to the higher layer to better achieve QoS.

The purpose of this study is to propose a cross-layer integration approach for VOIPv6 QoS support. The cross-layer interaction is managed between the data, network and transport layers by applying the different IETF QoS protocols in dealing with the metrics that affect QoS of VOIP in an integrated manner.

**Table 1: Comparison BEST-Effort, RSVP, RSVP (IntServ), DiffServ, MPLS and SBM [6] [7]**

| QoS Protocol | Attributes |
|---|---|
| Best-Effort | Associated with IPv4<br>Not sensitive to content of packet to real-time data<br>No guarantees, just connectivity<br>No isolation<br>Service scope is end-to-end<br>No setup<br>Highly scalable |
| Resource Reservation - RSVP | Most complex of all QoS protocols<br>Biggest departure from “Best-effort” approach<br>Provides highest level of QoS in terms of service guarantee, resource allocation & detail of feedback to QoS-enabled applications on per flow basis |
| RSVP + IntServ | Guaranteed as close as to a dedicated circuit<br>Controlled load equivalent to best-effort under unloaded scenario<br>per flow guarantee<br>per flow isolation<br>per setup<br>Not scalable (each router maintain per flow state) |
| DiffServ prioritization | Applied on per traffic flow aggregates<br>Services scope is at domain<br>A predefined Per Hop Behaviour (PHB) is applied to every service class<br>Long term setup |

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| | |
|---|---|
| | Scalable (edge router maintain per aggregate state, core router per class state |
| MPLS | Aimed at simplifying routing processes<br>For establishing fixed bandwidth routes<br>Traffic is marked and used to determine next router hop and not priority |
| SBM | Is a top-to-bottom QoS approach<br>Applies to OSI layer 2 (Data link layer) which makes QoS enabled on LAN<br>Provided all traffic passes through at least one SBM enabled switch |

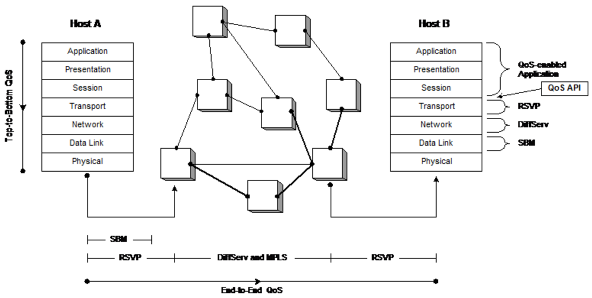


**Figure 1: QoS Architecture [7]**

## 4. INTERNET PROTOCOL VERSION 6 (IPv6)

Both IPv4 and IPv6 define data communication from one computer to another computer over the internet network layer. IPv6 is documented on different RFCs (Request for Comments) stating from RFC 2460 of IETF. IPv6 addresses the main issue with IPV4 which is exhaustion of IPv4 in the nearest future [5] [9]. IPv6 has a very large address space and consists of 128 bits as compared to 32 bits in IPv4. Therefore, it is now possible to support $2^{128}$ unique IP addresses, a substantial increase in number of computers that can be addressed with the help of IPv6 addressing format [5].

In addition, this addressing format will also eliminate the need of network address translation (NAT) that causes several networking problems in end-to-end nature of the Internet, simplifies header, improve support for extensions and headers, better support for authentication and privacy, as well as allowing a 20-bit flow labelling [5]. Problem associated with NAT is hiding multiple hosts behind pool of IP addresses and VOIP through the internet do not work smoothly with NAT [5].

By allowing packets belonging to a particular traffic flows to be labelled using flow labelling, real-time data stream packets in a network can be prioritize [7]. IPv6 implements QoS with the help of classification and marking (of IP packets) to ensure a reliable VOIP infrastructure. With the help of classification and marking technique, the network can identify packets or traffic flows and then can assign certain parameters within the packet headers in order to group them [6]. In order to implement QOS marking, IPv6 provides a traffic class field (8 bits) in the IPv6 header. Table 2 defines the function of fields in IPv6 header, while Figure 2 shows the structure of IPv6 header in a graphical form. IPv6 does not by itself provide QoS, it relies on the network router to make logical decisions based on the data provided header. In general, IPv6 has in it the best characteristics of IPv4 in addition to enhanced capabilities. IPv6 and QoSIPv6 or Internet Protocol version 6 is the next generation protocol for internet [5].

**Table 2: IPv6 Fields Length and their Functions [5]**

| IPv6 | Length | Function |
|---|---|---|
| Version | 8 bits | Identifies the version of the protocol. For example, for IPv6, the version is 6 |
| Class | 8 bits | Intended for originating nodes and forwarding routers to identify and distinguish between different classes or priorities of IPv6 packets |
| Flow label | 20 bits | Defines how traffic is handled and identified. A flow is a sequence of packets either sent to a unicast or a multicast destination. This field identifies packets that require special handling by the IPv6 node |
| Payload length | 16 bits | Identifies the length, in octet, of the payload. The payload includes the optional extension headers, as well as the upper-layer protocols e.g. TCP |
| Next header | 8 bits | Identifies the header immediately following the IPv6 header. Examples of next header are:<br>0 = Hop-by-hop options<br>1 = ICMPv4<br>4 = IP in IP (encapsulation) |

| | | |
|---|---|---|
| | | 6 = TCP<br>17 = UDP<br>43 = Routing<br>44 = Fragment<br>50 = Encapsulation security payload<br>51 = Authentication<br>58 = ICMPv6<br>59 = None<br>60 = Destination options |
| Hop limit | 8 bits | Identifies the number of network segments, on which the packet is allowed to travel before being discarded by a router. The hop limit is set by the sending host and is used to prevent packets from endless circulation on an IPv6 internetwork.<br>When forwarding an IPv6 packet, IPv6 routers must decrease the hop limit by 1, and must discard the IPv6 packet when the hop limit is 0 |
| Source address | 128 bits | Identifies the IPv6 address of the original source of the IPv6 packet. |
| Destination source | 128 bits | Identifies the IPv6 address of the intermediate or final destination of the IPv6 packet. |

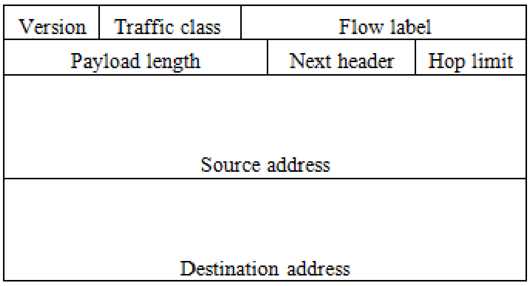


**Figure 2: IPv6 Header Structure**

IPv6 is a step and way forward towards dealing with the aforementioned QoS issues stated above most specifically the lack of consensus about QoS architecture which is preventing adequate support from protocols. Therefore a new cross-layer integrated approach is been proposed.

## 4.1 Why IPv6?

For the past 40 years IPv4 has been the underlying protocol that has makes it possible for us to connect our devices to the internet, with unique IP address computers are able to communicate and send data across to each other [9] [5]. But, with improve technology which has lead to growth of IP-based devices, there is serious concerns about IPv4 limited features, robustness, and scalability. This led to the creation of IPv6 by the Internet Engineering Task Force (IETF) with sole aim of making the internet work better [5] [9]. Table 3 is a compelling argument on why IPv6 will improve QOS in VOIP applications as well as been the future of the internet.

**Table 3: Comparison between IPv4 and IPv6 [9] [5]**

| IPv4 | IPv6 | IPv6 advantage over IPv4 |
|---|---|---|
| IPV4 addresses are 32 bit length | IPV6 address are 128 bit length | Substantially Larger address space (3.4 *$10^{38}$ unique addresses) |
| Due to IP address constraint NAT is implemented, but NAT has some inherent issues | Due to large address space NAT not required for end-to-end connectivity | Provides better end-to-end connectivity due to large address space, which is efficient & effective for VOIP applications |
| Manual configuration (static) or DHCP (dynamic) is required to configure IP addresses | Auto-configuration of addresses | Better ability for auto -configuration of devices (plug-and-play, DHCPv6 auto-configuration of address without need for a server) |
| Checksum & option fields in header, which degrade performance during forwarding of IP packets | No checksum and option fields in header, except extension headers to support more capabilities | Simplified header structure for faster routing |
| Fragmentation is done by sender and forwarding routers | Fragmentation is done by the sender only | Simplified and better routing |
| IPSec support is optional | Inbuilt IPSec support | Better security for application & networks |
| No packet flow identification | Packet flow identification is within the IPv6 header, using the Flow label field | Better Quality of Service (QOS) |
| Broadcast messages in IPv4 | A link-local scope all-nodes multicast address is used for broadcast. | Better multicast and anycast capabilities |
| Address resolution | ARP is replaced with Neighbour | Neighbour Discovery Protocol |

| | | |
|---|---|---|
| protocol (ARP) present in IPv4 to map its addresses to MAC addresses | Discovery Protocol | & auto-configuration offer better mobility feature by using Mobile IPv6 ( MIPv6). |
| In terms of Administration network renumbering and assigning of new address scheme is done manually | Network renumbering happen automatically | Offer easier administration for host & routers during switchovers or merger of networks |
| No transition mechanism defined to allow coexistence of IPv4 & IPv6 | Compatibility IPv4 and transition mechanism from IPv4 to IPv6 is incorporated into network using Dual IPv4/IPv6 stack implementations | Permit smooth transition from IPv4 |

## 5. LAYERED ARCHITECTURE VOIP NETWORK

The native network for Wired Networks is based on Layered architecture. Since this was not initially designed for Wireless Networks, it introduces inefficiencies when applied to the wireless networks [1], [10]. It has significant effect on the performance metrics of Wireless Networks and in turn poses significant issues on QoS of VoIP. This section describes briefly the layers of the layered network architecture.

**a.** **Physical Layer** - This layer is normally represented as PHY layer. It is the bottommost layer in the Transmission Control Protocol/Internet Protocol (TCP/IP) architecture. It defines the hardware technologies of a network. The QoS factors that are considered measurable in this layer are: BER (Bit Error Rate), SNR (Signal to Noise Ratio) and interference [1], [11] and [12].

**b.** **The Data Link** layer is made up of two sub layers: Logical Link Control (LLC) and Media Access Control (MAC). The LLC for the assignment of channel access for reliability in communication, while MAC handles scheduling, packet retransmission, etc. [11]. The MAC sub-layer is made up of Distributed Coordination Function (DCF) and Point Coordination Function (PCF) [10], [13].

**c.** **The Network layer** is responsible for data routing. It handles transmission of data from source to its destination.

**d.** **Transport layer** handles the delivery of data with respect to process-to-process [1]. It provides services such as congestion control and error recovery [10]. Further several protocols are available in this layer for different applications.

**e.** **The Application layer** houses protocols such as http, ftp, etc. which serve as the interface between the users and the network protocols. Figure 3 shows the TCP/IP layered Architecture for implementation in VoIP realm.

| | | | | |
|---|---|---|---|---|
| Application Layer | Voice | | | |
| | RTP | RTCP | SIP | H.323 |
| Transport Layer | UDP/TCP | | | |
| Network Layer | IPv6 | | | |
| Data-Link Layer | 802.3/802.11: LLC | | MAC | |
| Physical Layer | Ethernet/SDH/RF | | | |

**Figure 3: TCP/IP Layered Architecture for VOIP Implementation**

## 6. RELATED WORK

One of the main issue that affect VOIP is it QoS which is highly dependent on metrics such as jitters, delay, packet loss and bandwidth. Several research and studies have been done to improve voice transmission in IP based networks including [8] [2] [3] [1]. A cross-layer interaction approach to enhance QoS of VOIP over WLAN has been proposed by [1] by introducing a communication agent between the data link layer and transport layer of the TCP/IP protocol for better communication and interaction, while [2] uses QoS parameters relevant to VOIP transmission to evaluate its performance. Generally, most researchers have adopted a test bed, laboratory experimentation and simulation in an artificial setting within which relevant information and data can be generated as a means to study QoS of VOIP. Reason has been that laboratory experimentation usually permits an observation of the dynamic behaviour of the monitoring system (or its sub-system) under controlled conditions, but the limitation is getting only approximated results compared to real-life situation.

Authors in [1] proposed a cross-layer approach that incorporates ARQ (Automatic Repeat reQuest) scheme between the Transport and Data Link layer employed for increasing wireless reliability. The major limitation of this technique is that it does not take acknowledgement (ACK) generated at the receiver side into consideration, because, locally generated ACK does not contain vital fields such as advertised window for the sender. An intended solution to the limitation of technique in [1] was proposed in [14]. In [14], cross-layering approach is used which locally generates TCP ACK thereby reducing the duplication of acknowledgments at Data link and Transport layers. It introduces agents at the MAC layer of mobile station for updating TCP ACK header at the AP (Agent AP).

## 7. CROSS-LAYER INTEGRATION FRAMEWORK


**Figure 4: VOIP and Performance Metrics Trade-off Triangle**

In order to achieve a balanced trade-off among the three metrics with improved QoS of VoIP, there is need to implement mechanism among the MAC, Network and Transport layers which can provide minimum packet loss, minimum delay and maximum throughput in fairness to VoIP QoS.

### b. Cross-Layer integration between MAC, Network and Transport Layers

Cross Layer Interaction Manager (CLM) handles the interactions in the Media Access Control (MAC) layer, Network layer and Transport Layer. Moreover, retransmitting lost packets from AP instead of retransmitting them from the sender side would reduce delay and at the same time hide the occurrence of lost packets from transport layer which then does not need to apply congestion control whenever there are packets lost layer, and if it is a voice packet error, then it will relay the event to Transport layer. Accordingly, it will adjust voice packet related parameters, such as window size, compression ratio, etc. to encounter the error. This will reduce the delay and improve the overall TCP throughput, since TCP will now know when to segregate the error types and when to apply the congestion mechanism. The proposed framework is depicted in figure 5.

### a. Trade-off Triangle

The proposed framework is guided by the trade-off triangle shown in Figure 4. The three performance metrics that consequentially affects the QoS of VoIP are indicated. Traditionally, reducing packet loss will mean moving towards increasing delay which negatively affects the VoIP QoS. In the same vein, reducing delay means moving toward increasing packet loss and decreasing throughput, this also negatively affects the QoS of VoIP.

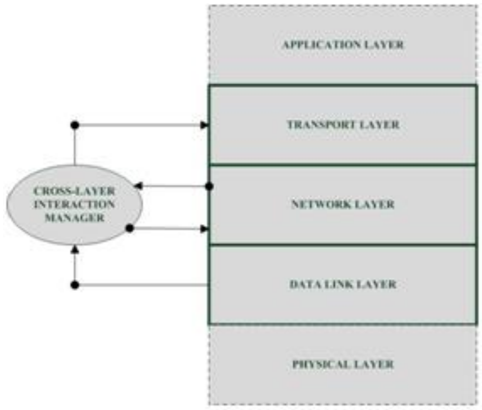


**Figure 5: Proposed cross-layered integration framework**

## 8. CONCLUSION AND FUTURE WORK

This paper has presented the basic features of VOIP, QoS and IPv6. Issues associated with VOIP and QoS as well as reasons as to why IPv6 will help in addressing QoS issues in VOIP as against IPv4 have been highlighted. We also proposed a Cross-layer integration framework, Cross-Layer Interaction Manager (CLM) aiding with the handling of interactions between the data (MAC), network and transport layers with the aim of improving the QoS in IPv6 based VOIP. Future work will involve experimentation and simulation study to implement the proposed framework on existing simulation packages such as Network Simulator-2 (NS2), autoVOIP™, and OPNET.

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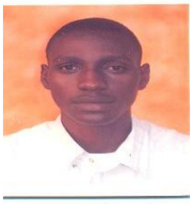

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