# ISSUES IN VOICE OVER MPLS AND DIFFSERV DOMAINS

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## ABSTRACT

The enormous growth of the Internet has encouraged more applications, users and services to be deployed. New protocols such as RSVP, MPLS and Diffserv promise to provide quality of service, through reservation of resources or differentiation of traffic. This paper studies the performance of voice in terms of delay and jitter under varying load conditions when an integrated environment of MPLS and Diffserv is provided.

*Keywords: VOIP, MPLS, Diffserv, IP-QoS*

## 1.0 INTRODUCTION

Internet has seen enormous growth in both the number of users and in the demand for new services from applications. There has been a growing interest in migrating from circuit-switched network telephony service onto an IP-based packet switched network infrastructure. The success of IP-based telephony shall depend on whether it can provide the customer with the required QoS or not. This service requires stringent bounds on end-to-end packet delay, jitter, and loss.[1]

New protocols such as MPLS and Diffserv are soon going to be realized due to the major advantages associated with them such as QoS, gigabit forwarding, network scaling, traffic engineering, and scalable multi-class of services. MPLS as an advanced forwarding scheme converges the connection-oriented forwarding techniques and the Internet routing protocols. Diffserv can provide scalable multi-class services in IP networks and solves the scalability problem of RSVP, offering QoS on aggregations of flows, in contrast to RSVP, which is per-flow basis.

### *1.1 Voice Over IP*

Voice over the Internet is the ability to make telephone calls over IP-based data networks with a suitable quality of service (QoS) and low cost. Carrying traditional telephone traffic over the IP network has both opportunities and challenges. VOIP differs from the traditional voice telephony because it uses data network, resulting in its lower cost. The VOIP system can be expanded easily by expanding the network or adding additional ports to the VOIP gateway and the voice data (in data form) is processed easily by the PC. A *Gateway* interconnects the Voice source to the network through the local exchange, forwards the voice to the destined subscriber across the Internet, and sends the incoming calls to the corresponding extension via the PXB. VOIP delivers real-time and two way synchronous voice traffic over the Internet or Intranet[2].

The IP telephony provides a number of benefits as compared to the Public Switched Telephone Network (PSTN) such as: integration of voice, data and fax, sound grading, video telephony, unified messaging, low-cost voice calls, real-time billing, remote teleworking, enhanced teleconferencing, etc. It may face many technical challenges such as: loss, delay, and jitter[3]. Internet telephony has caught the world's attention despite the inferior quality for many of these connections. Many companies have introduced products that improve and commercialize the technology. New protocols such as Diffserv and MPLS are being introduced with additional features like QoS, reliability and traffic engineering that improve the performance of voice transmission over the IP network. Unlike data, VOIP is more sensitive to delay than loss, thus sufficient bandwidth must be guaranteed to the voice application. Some protocols that decrease voice transmission delay by giving high priority to the voice than data traffic have been introduced. Resource reSerVation Protocol (RSVP) provides for the routers to reserve the required bandwidth for the voice connections, and Real-time Transport Protocol (RTP) uses synchronization to prevent delays and protect data against loss[4].

### *1.2 Differentiated Services*

The current Internet delivers only best effort to all traffic[5]. Services differentiation is desired to accommodate heterogeneous application requirements and user expectations, and to permit differentiated pricing of Internet services. The differentiated services share the network with best effort traffic instead of replacing it. This is desirable economically, since the same network can be used for both kinds of traffic.

The two new protocols, Diffserv [RFC 2474] & MPLS [18][11] pioneered by the IETF, are to offer Quality of

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services to IP traffic. Diffserv can provide packets with a preferential treatment using different code-points in their header. MPLS enables greater control over routing in packet networks. Using both of them can provide differentiated services with greater routing control [6]. The IETF differentiated services framework [19] defines a number of mechanisms for differentiating traffic streams within a network and providing different levels of delivery service to these streams[7]. The Diffserv architecture is composed of a number of functional elements implemented in network nodes to provide per-hop forwarding behaviors, packet classification function, and traffic conditioning functions such as metering, marking, shaping and policing. This architecture achieves scalability by implementing complex classification and conditioning functions only at network boundary nodes, and by applying Per-Hop Behaviors (PHB) to aggregates of traffic which have been appropriately marked using the DS field in IPv4 or IPv6 [DSFIELD]. PHB permits a reasonably granular means of allocating buffer and bandwidth resources at each node among competing traffic streams. Per-application flow or per-customer forwarding state need not be maintained within the core of the network.

Differentiated service, in contrast to Integrated service, deals with aggregates of flows. Each Per Domain Behavior (PDB) receiving identical treatment by a node, is designed by a particular value of the Differentiated Services byte in the IP header, the DS byte. The Diffserv domain is composed of network elements under a common administration, with a common relationship between the DS code-point and actual handling of the PDB by the nodes of the Domain[8]. The boundary nodes perform traffic conditioning operations including shaping and policing on the PDB's, while the intermediate nodes of the domain processes a PDB in accordance with the appropriate PHB.

Network nodes that implement the differentiated services enhancement to IP use the DS byte in the IP header to select a PHB as the specific forwarding treatment for that packet [RFC2474, RFC2475]. Exploiting controlled sources such as TCP is the basis of the Assured Forwarding (AF)[8]. The AF PHB group provides delivery of IP packets in four independently forwarded AF classes. Each AF class in each DS node allocates a certain amount of forwarding resources (buffer and bandwidth). Within each AF class, an IP packet can be assigned one of the three differentiated levels of drop precedence, which determines the relative importance of the packet within the AF class. A DS node does not reorder IP packets of the same microflow if they belong to the same AF class, whether they are in or out of profile[9].

The second Diffserv forwarding techniques is Expedited Forwarding Per-Hop Behavior (EF PHB). This is considered the premium service; whereby it minimizes losses, latency, and jitter and achieves assured bandwidth and end to end service through DS Domain. It behaves like a point to point connection or "Virtual Leased Wire"[10].

The EF PHB provides a node's conditioner such that the aggregation maximum arrival rate is less than the aggregation minimum departure rate through boundary conditioner performing policing and shaping, so that aggregate's arrival rate at any node is always less than that node's configured minimum departure rate[10]. The EF traffic should receive this rate independent of the intensity of any other traffic attempting to transit the node. Thus the EF PHB ensures no (or very small) queues, which in turn causes no loss, latency and jitter. The weighted round robin scheduler can be used to assign the shared output bandwidth to EF queue[10].

Peak bit rate is awarded to a specific flow or aggregation of flows within the contracted rate to guarantee the availability of the contracted bandwidth as long as the user' traffic is within the contracted rate. On the other hand, if user exceeds the peak rate, the traffic will be discarded. First-hop routers set the premium bit of those flows that match a premium service specification after performing traffic shaping on the flow to smooth all traffic bursts before they enter the network. A compliant router along the path make two levels of priority queuing, the high priority queue for the premium service and the low priority queue for the normal best effort traffic. This results in two "virtual networks" along the domain, one which appears like a virtual leased wire where the packets experience almost no queuing delay, and identical best-effort virtual network with buffers designed to absorb traffic bursts.

*1.3 The suitability of EF for carrying voice*

End-to-end connection and the absence of delay in the network are the basic requirements of voice applications. As it has been mentioned above, the EF ensures no queue (or very small queue) for some aggregate. Since the aggregates do not see any queue at the nodes while they traverse in the network, they do not face loss, latency and jitter[10]. Premium service promises an end-to-end quality of services through DS domains which appears to the endpoints as "virtual wire", where a flow's bursts may queue at the shaper at the edge of the network, but thereafter only in proportion to the in-degree of each router. It's not like best effort traffic, does not need queue management, because it's very regular traffic patterns and small or nonexistence queues. Based on these characteristics, the EF service is a best solution for running voice over differentiated services.

Jacobson [5] has defined a "Premium" service that is provisioned according to peak capacity profiles that are strictly not oversubscribed and that is given its own high-priority queue in routers. A premium service protects the network from congestion by shaping traffic flows at the nodes. However, the premium service reduces the capacity of the best effort internet by the amount of bandwidth allocated to it, but it's not like a standard telephone line, it gives the chance to best effort traffic if the capacity is not being used by it.

The intermediate routers do forwarding path decisions separately and more simply than the setting up of the service agreements and traffic profile. Pushing the complexity and most of the processing to the boundary[5] decreases delay in the domain and the only actions that need to be handled in the forwarding path are to classify a packet into one of two queues on a single bit and to service the two queues using simple priority. The premium service solves the jitter problem, by building the virtual leased wire within its domain.

*1.4 Voice over MPLS*

There are currently two techniques of building core IP networks. The first is where the core of the network is based on the datagram routers; and second is where the network of datagram routers operates over an ATM core[11].

MPLS emerged as a refined solution to meet the bandwidth-management and service requirements for next generation IP-based backbone networks. Addressing issues related to scalability and routing, MPLS gives a versatile solution to solve the problems faced by present-day networks in terms of speed, scalability, QoS management, and traffic engineering. It provides connection-oriented (label based) forwarding based on IP routing and control protocols.

Generally, in MPLS, data transmission occurs on Label-Switched Paths (LSPs). The path computation for an LSP may seek to satisfy a set of requirements associated with the LSP, taking into account a set of constraints imposed by administrative policies and the prevailing state of the network[12] - which usually relates to topology data and resource availability. Hence in short, computation of an engineered path that satisfies an arbitrary set of constraints is referred to as "constraint based routing".

MPLS enhances source routing and allows for certain techniques, used in circuit switching[12], in IP networks. Constrained-based Routing-Label Distribution Protocol (CR-LDP) is a simple, scalable, open, non-proprietary, traffic engineering signaling protocol for MPLS IP networks. CR-LDP provides mechanisms for establishing explicitly routed LSPs. CR-LDP is defined for the specific purpose of establishing and maintaining explicitly routed LSPs. Optional capabilities provide for negotiation of LSP services and traffic management parameters over and above best-effort packet delivery including bandwidth allocation and setup and holding priorities. CR-LDP works equally well for Multi-service switched networks, router networks, or hybrid networks. This applicability statement does not preclude the use of other signaling and label distribution protocols for the traffic engineering application in MPLS based networks. Service providers can deploy whatever signaling protocol meets their needs. In particular CR-LDP and RSVP-TE are two signaling protocols that perform similar functions in MPLS networks.

## 2 Integration of MPLS and Diffserv

*2.1 MPLS/Diffserv*

MPLS and Diffserv operate at different layers in the protocol stack. Diffserv operates at the network layer, while MPLS is often viewed as operating between the link layer and the network layer. They do not work naturally together, because MPLS was designed without taking QoS in consideration, but the two are complementary techniques that can be implemented in an IP QoS network to implement an end-to-end QoS solution. When used together, Diffserv provides the standardized QoS mechanisms and MPLS provides routing techniques increasing the network resource optimization and providing traffic engineering. An MPLS domain uses MPLS signaling protocols to establish a label switched path to forward data through a common path. The ingress LSR labels the packets, and the LSRs along the LSP forward the packets to the next hop. In Diffserv, the ingress router classifies the packets and then marks them with the corresponding DSCP. The intermediate routers use PHB to determine the scheduling treatment and drop probability for each packet[13].

The two protocols are independent of each other. However, associating an MPLS flow with a particular DSCP solves the problem of implementing an end to end service such as Voice over IP. The DSCP of the packet can be determined from the label when the MPLS packet is received[13]. MPLS makes the DS more reliable and faster due to its path-oriented feature. With the MLS/Diffserv techniques, separate classes of services supported via separate LSPs are routed separately, and all classes of service supported on the same LSP are routed together[14].

*2.2 Traffic Engineering (TE):*

The mapping of traffic flows onto an existing physical network topology is called traffic engineering[15]. To support extremely rapid growth rates and maintain a reliable infrastructure for mission-critical applications, an organization has to balance the traffic load on the various links, routers, and switches in the network so that none of them are over utilized or under utilized. The congestion in conventional IP networks occurs due to selecting the shortest path calculated by the Interior Gateway Protocol (IGP). Traffic Engineering, which is provided by MPLS, solves the above problem by selecting the less congested path instead of the shortest path, leading to the optimum and reliable utilization of the network infrastructure[16].

Traffic engineering is typically done today in IP over ATM networks using manual configuration. Traffic engineering is difficult to accomplish with datagram routing. Some degree of load balancing can be obtained by adjusting the metrics associated with network links[17]. However, there is a limit to how much can be accomplished in this way. Furthermore, in networks with a large number of alternate paths, balancing the traffic on all links is difficult to achieve solely by adjustment of the metrics used with hop by hop datagram routing. MPLS allows streams from any particular ingress node to any particular egress node to be individually identified. MPLS therefore provides a straightforward mechanism to measure the traffic associated with each ingress node to egress node pair[17]. In addition, since MPLS allows efficient explicit routing of LSPs, it is straightforward to ensure that any particular stream of data takes the preferred path.

The suitability of MPLS for Traffic Engineering can be attributed to many factors[5]. A good implementation of MPLS can offer significantly lower overhead than other alternatives for Traffic Engineering.

## 3 Voice Over MPLS/Diffserv Architecture

It is observed that the premium service (EF) of Diffserv is not enough alone to ensure the quality of service. Some research results have shown the failure of EF PHB in meeting the delay and jitter targets if an EF packet arrives on an output link on which a large BE packet is already in the middle of being transmitted.

Therefore it is important that some type of prior reservation be made for transmission of real-time voice across a domain. MPLS provides the mechanism that can be used to separate the BE traffic path from EF and AF traffic path and to install explicitly routed LSP's (Label Switched Paths).

We have set up voice over MPLS/Diffserv using NS-2 simulator and its associated modules. We are in the process of designing rules and specifications for an architecture to support voice in an MPLS and Diffserv domain. These rules will dictate the formation of LSP's carrying voice calls and aggregation of LSP's into virtual trunks across the domain. The specifications are supposed to standardize the treatment of voice calls in the IP networks and to make it easy to perform traffic engineering in a domain that deals with voice calls.

## 4 Simulation and Results

A network model based on a single bottleneck links was developed. The network model, shown in Figure 1 has been studied using Network Simulator (NS-2) package. Attaching a different number of sources and destinations, simulating voice traffic with EF flows, the network working performance is evaluated. The two intermediate links between the ingress (C0) and egress (C1) are assigned as a Diffserv domain with a bandwidth of 1Mbps and a delay of 1ms, while the access links to sources and destinations are assigned a bandwidth of 1Mbps and a delay of 0.1ms. The explicit route of MPLS domain is C0- C1 - C2. The node-conditioner performing policing and shaping is implemented at first LSR or ingress. The MPLS has several LSPs; each carries the traffic of multiple sources. One of the flows between the sources and a corresponding destination is taken as a representation to study the delay variation, jitter and loss.

Using this model, both MPLS and Diffserv are implemented together in the same network, providing insight on the performance of the two protocols in terms of delay, jitter and loss. The schedulers' parameters at nodes C0, C1, and C2 are set as follows: Queue size of EF traffic is 90, queue size of BE traffic is 10, the allocated bandwidth for each EF class is 1, the allocated bandwidth for each BE class is 9, and the threshold for Weighted Round Robin is 10,000.

The network was studied with different loading conditions. At the beginning a lightly loaded network was monitored, whereby fifteen CBR sources (each generating traffic at 64 Kbps, equivalent to one voice channel) were connected to the ingress, passing the packets through the Diffserv/MPLS domain to fifteen destinations, connected to the egress. The network performance reflected no jitter or loss. The delay is very small running uniformly at 0.02236s, as shown in Fig.2. The abscissa of the figure represents the number of packets sent at different simulation time between 0-5s. The ordinate is the delay. The graph showed the delay remains constant all over the 5s. This was expected, as the total EF bandwidth requirement is slightly lower than the aggregate bandwidth of the Diffserv domain network.

A similar setup was tested using a 1000 Kbps FTP source in addition to the fifteen EF sources. The performance, shown in figure 3, was somewhat different at the beginning as it reflected an initial jitter, before decreasing to a constant level comparable to the first test causing the delivery delay constant (0.02236) and jitter to almost zero. The initial higher delay is due to the earlier presence of BE traffic. This can be explained as at the beginning, the FTP source was sending packets at the same time with other CBR sources. After an initial slow down of EF traffic forwarding at the Ingress LSR (C0), this effect started decreasing because the TCP agent protocol of the FTP source controls itself whenever sees that the network is busy.

EF normally arranges to pass through Diffserv domain without much delay, jitter or loss. However, increasing the number of CBR sources beyond the capacity rate of the Diffserv link ought to cause congestion, which results in a jitter, and eventually packet loss. Such a case was examined using 18 and 20 CBR sources with a total bandwidth of 1.152Mbps and 1.28Mbps respectively, exceeding the link bandwidth of 1Mbps. The delay shown in Figures 4 & 5, started increasing causing jitter up to 1,18828s (whish is associated with the maximum capacity of the buffer), after which it changes very small. This happens after the buffer of the C0 fills up when the node starts to drop. Increasing the number of sources will cause the buffer to be filled up in a shorter duration causing packet-drops at an earlier time. Furthermore, the congestion slows the forwarding process at the ingress, which in turn increases the delay and produces jitter. After the buffer of C0 has been filled up, C0 starts to drop the packets which are coming from low-priority sources, making the network to be stable and the delay to be constant.

## 5 Conclusion and Discussion

The voice architecture over MPLS/Diffserv domain can be represented as in figure 6. Based on the simulation results, it can be concluded that the MPLS/Diffserv domain can support p number of voice applications efficiently, utilizing the line capacity fully even if all the sources start to send at the same time as long as the following condition is satisfied:

$$C_{trunk} \geq \sum_{K=1}^{p} S_k = \sum_{j=1}^{n} \sum_{i=1}^{m} Uji$$

Whereby m is the number of microflows, n is the no. of paths, S is the source bandwidth, U is the utilization, and C is the capacity.

As an example, the topology in Figure 1 supports 15 voice sources (traffic from sources 1-7 traverses LSP1 and traffic from sources 8-15 traverses LSP2) efficiently with very small delay and no jitters. Violating the above condition decreases the performance of the architecture in terms of loss and delay if the sources are continuing to congest the network. The effect of BE traffic has no effect on the steady-state performance of voice traffic, and the transient delay shall die out within a short time duration.

We are working to define a standard mechanism of establishing voice calls over the IP network in an MPLS and Diffserv domain. There are many issues pending including the rules to extend such a mechanism across several independent domains, fault tolerance, and rules dealing with link re-adjustment scenarios as per the ISP policies and the overall effect of this voice architecture over the load management of the domain.

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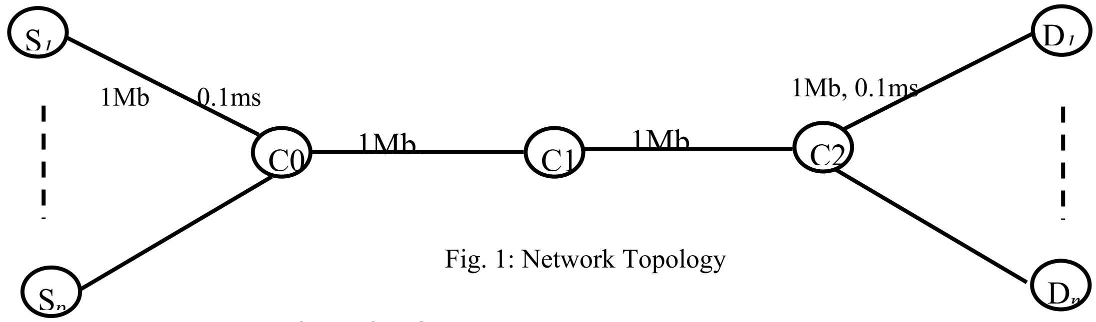



Fig. 1: Network Topology

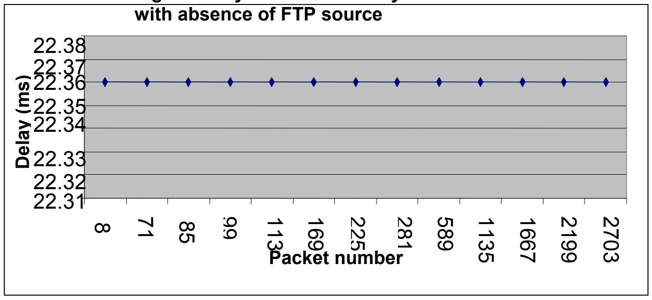



**Fig. 2: Delay variation & Delay from S9-D9 with absence of FTP source**

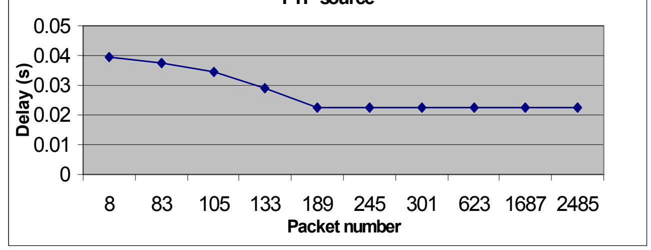



**Fig. 3: Delay variation & Delay from (S9-D9) with existence of FTP source**

Fig. 4: Delay variation & Delay from (S9-D9) with when the network is congested

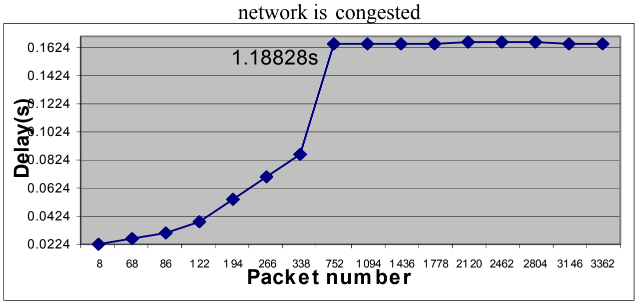
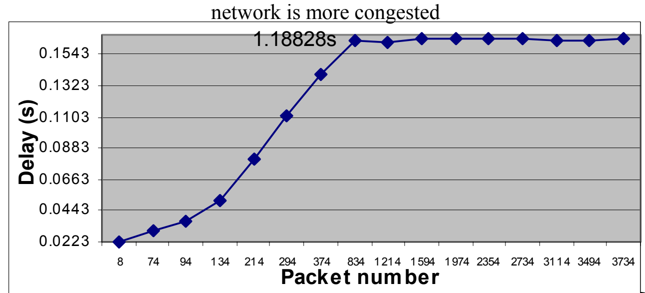



Fig. 5: Delay variation & Delay from (S9-D9) with when the network is more congested



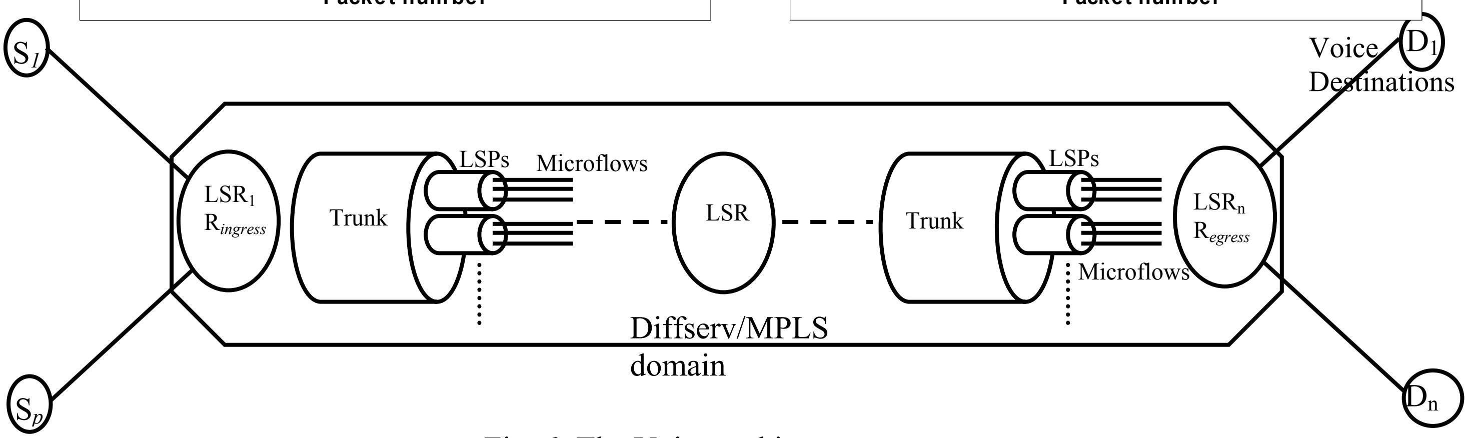




Fig. 6: The Voice architecture over MPLS/Diffserv domain