An MPLS Based Load Balancing Technique for VoIP Flows

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Abstract: The tremendous growth of internet increase the enormous pressure on internet service providers to ensure quality of service for real time applications such as voice over IP and video on demand. These applications are very sensitive to delay, jitter, packet loss and needs to be engineered. In this study, a multi protocol label switching based load balancing technique is proposed to enhance the quality of service for VoIP applications. The proposed study is implemented with an effective flow classification technique and prioritizes the voice packets based on their flow arrival rate and bandwidth utilization. The link failure or system failure in IP network causes load unbalanced situation and increases the congestion in the network. During this load unbalanced condition both responsive and unresponsive VoIP flows are rerouted into congestion free constraint based least loaded multiple paths. The non VoIP data flows are routed using Stream Control Transmission Protocol Multihomming technique. Simulation results are reported to show the efficiency of the proposed technique.

Key words: Multipath routing, MPLS, VoIP, QoS, unresponsive flows, load balancing

INTRODUCTION

Voice over Internet Protocol (VoIP) or Internet Telephony emulates toll services with lower communication cost. It enables the telephone calls over Internet protocol (IP) based data networks with a suitable Quality of Service (QoS) and low cost. The VoIP necessitates more quality of service requirements. It also demands high availability of resources, good quality of voices, low delay and packet loss like traditional telephone but VoIP suffers from limited bandwidth capacity (Bandung *et al.*, 2007).

Basic network characteristics such as packet loss rate, delay and jitter and bandwidth capacity are considered as QoS constraints for VoIP (Singh *et al.*, 2012). Common QoS Design networks are best effort service, integrated service and differentiate service (Skinnemoen *et al.*, 2005). Differentiated services (Diffserv) combined with Multi-Protocol Label Switching (MPLS) protocols can be used to enhance the QoS, reliability and traffic engineering. MPLS establish multiple Label Switched Paths (LSPs) between ingress and egress nodes to improve the network performance (Al-Irhayim *et al.*, 2000; Krile and Kresic, 2009). MPLS-Traffic Engineering (MPLS-TE) with QoS techniques can emulate layer 2 service over a packet infrastructure.

The MPLS packet forwarding technique reduces the processing overhead at the intermediate nodes (Mellah and Abbou, 2006) and enables Traffic Engineering-Label Switched Paths (TE-LSPs) to alleviate un even load distribution (Chang *et al.*, 2009). MPLS utilizes several label distribution protocols like Resource Reservation Protocol (RSVP), Border Gateway Protocol (BGP) and Label Distribution Protocol (LDP) for label mapping and forwarding. RSVP avoids delay and data loss by reserving required bandwidth for the voice connections (Moltchanov, 2008). LDP provides extended neighbor discovery to establish peer relationship with non neighboring routers.

Conventional IP network always choose shortest available path for routing and does consider the network load conditions. This can lead to network load unbalanced situation and result in increase in congestion and packet losses and the network performance is degraded (Cidon *et al.*, 1999). Specifically inadequate network resources and unbalanced traffic distribution can cause congestion in the network (Susitaival and Aalto, 2004). Generally, there are two types of losses namely random losses and burst losses. Poor channel allocations can cause random losses. Link or node failure or severe channel destruction causes Burst losses. In addition to these factors, congestion can be reasoned by factors such as limited buffer, bandwidth and processing capability at network routers (Ash *et al.*, 1981).

Packet loss in VoIP degrades the audio quality. Sender or receiver based packet loss recovery techniques like Forward Error Correction (FEC) or Packet Loss Concealment (PLC) techniques have been proposed (Haywood and Peng, 2008). These loss-recovery mechanisms use the retransmission or additional data during data transmission to reconstruct the lost packets.

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This technique may require more network bandwidth and additional resources (Chua and Pheanis, 2006). Traffic classification and analysis on large-scale network links is used to identify its characteristics for traffic management and traffic engineering. It is used to plan QoS for applications, admission control, traffic shaping and resource provisioning. Real-time traffic classifications become one of the core component of emerging QoS enabled network design (Nguyen and Armitage, 2008; Kambourakis *et al.*, 2006).

To manage the unresponsive VoIP flow and to alleviate congestion, this study proposes a load balancing technique in MPLS network. Proposed technique finds congestion free least loaded multiple paths using heuristic multipath discovery algorithm. Core router use Rainbow Fair Queuing (RFQ) mechanism to differentiate packet of unresponsive flows in order to regulate the flows during load unbalanced condition.

MATERIALS AND METHODS

Literature review: Fair bandwidth allocation in routers can reduce the complexity of managing the unresponsive flows. To reduce the processing overhead and allocate fair bandwidth, Core-Stateless Fair Queuing (CSFQ) and Rainbow Fair Queuing (RFQ) have been proposed (Stoica *et al.*, 2003; Cao *et al.*, 2000). Bothe CSFQ and RFQ algorithms maintain Per-flow state information only at the MPLS edge router. CSFQ uses complicated scheduling and buffering algorithms when compared to RFQ. RFQ can differentiate packets based on their priority and use simple threshold-based dropping procedure to preserve the high priority packets during congestion.

An Active Queue Management Algorithm for Controlling Unresponsive Flows and Per-flow Scheduling is described by Santhi and Natarajan (2010). It supports Explicit Congestion Notification (ECN), to strengthen the robustness of Internet against unresponsive flows. Packet arrival rate and the link capacity are estimated to predict the congestion and penalize the unresponsive flows. It uses deficit round robin hashing mechanism to assign flow into individual queue. Also, it maintains probability for each active flow. Packets get dropped depending on its probability ratio when the buffer size reaches to maximum value.

To achieve load balancing a Linear Incremental Programming model with Proactive Failure Recovery (PFR) is proposed (Zhang *et al.*, 2008). Traffic flows are divided among multiple paths optimally to avoid congestion. The failure links are identified and the flows affected by the failures are re-allocated to multiple alternative backup paths incrementally without disturbing flows.

Traffic-splitting algorithm called FLARE operates on bursts of packets and avoids reordering. Since, packet based splitting require packet reordering. FLARE splits

the traffic based on flow given by traffic layer. This technique uses periodic pings to estimate the path delay and uses a hash table that maps flow into paths. The token counting algorithm is used to avoid congestion (Kandula *et al.*, 2007).

Route discovery are based on the parameter like link disjoint, node disjoint, zone disjoint, non disjoint or set of disjoint paths that satisfy the given QoS. The routing takes place from the source node or from the intermediate nodes. Traffic is distributed in single path, two simultaneous paths or multipath concurrently (Tsai and Moors, 2006). To perform load balancing in MPLS network Multipath adaptive packet dispersion is proposed (Banu and Ramachandran, 2012). The incoming traffic flow is classified as VoIP and non VoIP flows. Using the trigger handler, the adaptive packet scheduler reroutes the flow into the best shortest multiple paths.

Proposed load balancing technique: The steps involved in the proposed technique are depicted in Fig. 1. It is
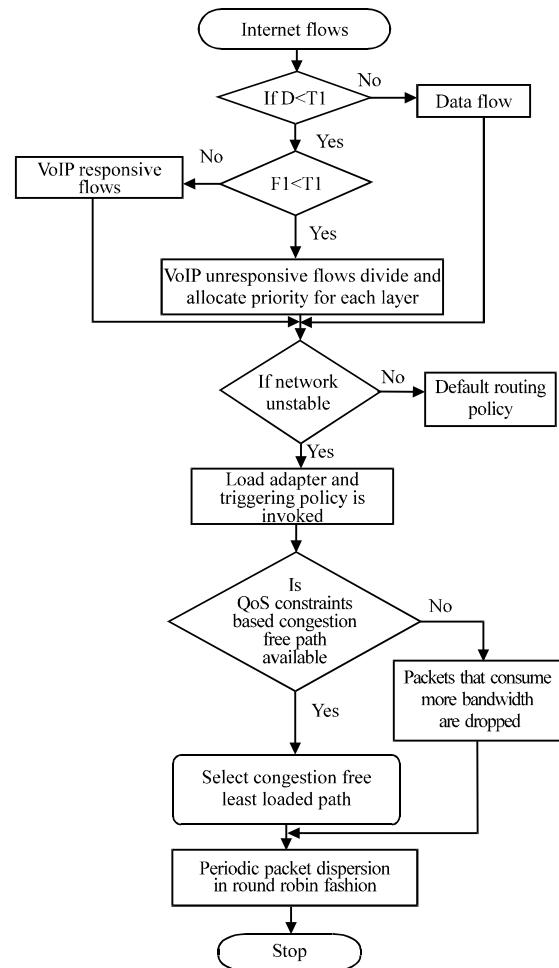

Fig. 1: Steps involved in the proposed system

mainly dealt with flow classification and effective triggering policies to perform load balancing and alleviate congestion. As a first step, internet flows are classified and unresponsive VoIP flows are identified. During load unbalanced condition, periodic round robin packet dispersion scheduler resides in a router that routes responsive and unresponsive VoIP packets into multiple QoS guaranteed parallel paths. If the congestion free least loaded (bandwidth guaranteed) paths are not available then the low priority packets from unresponsive flows are dropped to balance the load and preserve the high priority VoIP flows. The non VoIP data flows are routed using Stream Control Transmission Protocol (SCTP) Multihomming with grouping based multipath selection technique. The overview proposed routing modules are shown in Fig. 2. Flow classification and priority marking modules are dealt with MPLS label edge router and multipath routing modules are dealt with MPLS core router.

Flow arrival rate estimation and flow classification: Internet traffic flows are classified as VoIP responsive, non responsive flow and normal data flow by using edge-to-edge delay probing with flow arrival rate estimation. Delay probing mechanism is used at the ingress router to identify the flow that requires high bandwidth and posses high of active flows. Ingress router of each LSP periodically spacket losses ends a back to back (a probe pair) probe packets on all the forward LSP at regular time interval to various destinations. Probe packets are time stamped at the receiver to estimates the delay available bandwidth. By using the sequence number of the arrived packets and the gap between them the number of lost packets for specific time period is estimated.

Difference between the actual loss rate and probe packet loss rate, D is calculated and compared with a threshold value T1. Flow arrival rate of each flow is

Fig. 2: Overview of routing modules

measured using exponential averaging method. If the Value of D exceeds the threshold T1 and flow arrival rate exceeds threshold value T2 then the flow is classified as nonresponsive VoIP flows. If the value of D exceeds the threshold T1 and flow arrival rate is less than threshold value T2 then the flow is classified as responsive VoIP flows. Otherwise it is marked as normal data flows.

The VoIP flows are reported on flow basis or aggregate fashion. Number of flows to be reported exceeds a threshold value then the feedback is carried in an aggregate fashion else reported individually for a traffic class.

Let a^x_i and l^x_i be the arrival time and length of the packet x of flow i. The Flow Rate (FR) of flow i is estimated as given in the Eq.1:

$$FR_i^{new} = \left(1 - e^{-A_i^x/K}\right) \frac{l_i^x}{A_i^x} + e^{\left(-A_i^x/K\right)} FR_i^{old} \tag{1}$$

where, $A_i^x = a_i^x - a_i^{x-1}$ and K is a constant. $e^{-A_i^x/K}$ is an exponential weight that gives consistent value for bursty traffic without considering inter arrival time differences.

The loss ratio (L_{Ri}) at each node n_i along P at the interval t can be calculated as given in the Eq. 2:

$$L_{Ri}^t = PL/R \tag{2}$$

Where:
PL = The number of packets lost
R = The estimated arrival rate of the packet

Then, the total loss ratio at destination can be calculated as given in the Eq. 3.

$$L_R^t = \sum L_{Ri}^t \tag{3}$$

The actual traffic flows are transmitted for the very small duration of t sec via ingress router and flow arrival rate are marked as label. The actual loss ratio (L_{Ract}) at each node along R1 at the interval t can be estimated similarly as Eq. 2. Then, the total actual loss ratio at destination can be calculated as given in the Eq. 4:

$$L_{Ract}^t = \sum L_{Racti}^t \tag{4}$$

At egress router, the difference in loss ratios can then be estimated as given in the Eq. 5:
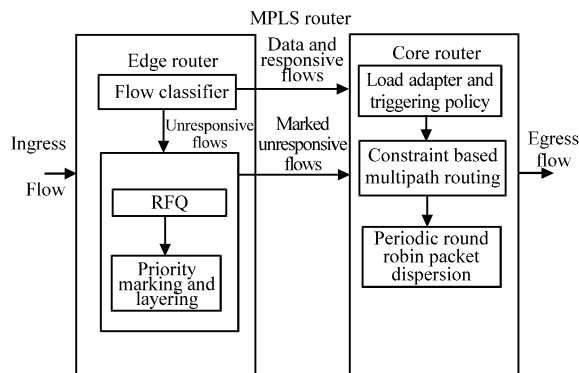
$$D = L_{Ract}^t - L_R^t \tag{5}$$

If loss ratio D<threshold T1 then the flow is classified as data flow. If D>T1 and flow rate FR_i>T2 then flow is marked as unresponsive VoIP flow else they are classified as responsive VoIP flows. The unresponsive flow rates are reported depending on flow basis or aggregate fashion.

Priority marking: The unresponsive flows that devour more bandwidth are marked and prioritized using Rainbow Fair Queuing (RFQ). When a flow enters the network, the ingress router measures its flow rate and divides the unresponsive flow into multiple thin layers. Each layer is marked and identified by a color. Flows with arrival rate (FR) will be assigned colors with 0...y, y is a value that satisfies:

$$\sum_{i=0}^{y} c_i \geq FR$$

Assigned color value is passed in the packet header as labels. Flow rates can be allotted for each layer using non linear encoding. This determines which layer a packet should belong to. The rate c_i of layer i as defined as in the Eq. 6:

$$c_i = \begin{cases} \dfrac{u^{INT(i/v)-N/v}}{v}MFR \\ \dfrac{u^{1-N/v}}{v}MFR \end{cases} \quad (6)$$

Where $v \leq i \leq N$ and $0 \leq i < v$

where, N denotes the total number of colors (layers), variables u and v represents the block structure and MFR stand for maximum flow rate in the network.

Where and when the network is in unbalanced condition, the core router looks for the layer (flows) that consumes more bandwidth. Then, it reroutes the flow through congestion free least loaded multiple paths. If the core router does not find any better congestion free path then it drops the low priority packets from that unresponsive flow to balance the load.

Load adapter and triggering policy: The load adapter checks for system load balancing state. When queue occupancies of the active users exceed 75% of the total available buffer space then the system is in unbalanced state. When the network is in unbalanced condition, triggering policy is invoked to override the default routing policy with multipath routing. It disperses the flow into congestion free QoS guaranteed multiple parallel paths through periodic round robin scheduling.

Congestion free least loaded paths are found using heuristic multipath discovery algorithm. If the core router does not find any better congestion free path then it

drops the some low priority packets from the lower color layer to balance the load and preserves the high priority flows. Bandwidth and delay are the QoS constraints considered in this proposed approach.

The non VoIP data flows are dispersed simultaneously on multiple paths using SCTP concurrent multipath transfer. SCTP multihoming allows an association between two end points with multiple IP addresses (multiple interfaces) for each end point. By estimating the available bandwidth on all paths, the best path for transmission is selected.

Multipath discovery algorithm: Multipath routing involves traffic splitting along two or more disjoint paths which satisfies the bandwidth and delay requirement of each type of flow depending on the type of traffic. To find multiple paths from the Source (S) to the Destination (D), the proposed system utilizes heuristic multipath algorithm.

Consider the network as communication graph G = (V, E) where V set of nodes and E denotes the set of links. The quality of each link Li $\in$ E is determined by two factors namely link bandwidth (bw) and delay (De). The Available bandwidth (Abw) and delay (Ade) of link L_i is determined. The delay and bandwidth of Path (P) are represented in Eq. 7 and 8, respectively:

$$Abw(P_i) = \min_{i \in [1...n-1]} \{bw(n_i, n_{i+1}...)\} \quad (7)$$

$$ADe(P_i) = \sum_{i=1}^{n-1} De(n_i, n_{i+1}...) \quad (8)$$

Consider MP includes set of multiple paths such that MP = {P_1, P_2, P_3 ...P_n}. Let N be the number paths considered in the network. Then, the cumulative value of bandwidth (Abw) and (Ade) are given, respectively in following Eq. 9 and 10:

$$Cbw(P_i) = \sum_{i=1}^{N} bw(P_i) \quad (9)$$

$$Cde(P_i) = \max_{P_i \in MP} \{De(P_i)\} \quad (10)$$

Heuristic multipath discovery algorithm makes use of preflow (Cerulli *et al.*, 2008) push algorithm, shortest path algorithm to find multiple paths between source and destination with required bandwidth and low delay. Let ML be the max flow argument in per flow algorithm. Execute pre flow push algorithm to compute ML such that ($ML_{bw} \geq$ Abw) && (ML_{de}<ADe) and add all possible paths to form max-flow graph G' = (V', E'). From G' shortest

best paths are selected for dispersion. Packets of particular flow f, are dispersed over these paths using the periodic round robin dispersion.

RESULTS AND DICUSSION

The proposed system is simulated with network simulator Ns2. The simulation topology consists of sender (ingress) and destination (egress) node. These nodes connected to 20 MPLS enabled Label Switched Routers (LSRs). Different link bandwidth and delay are assigned for the 10 paths. Both CBR and VoIP traffic with random exponential loss rate of 0.05. The performance of the proposed system is compared with MPLS routing by considering CSFQ queuing policy (Stoica *et al.*, 2003). Several parameters used for simulation shown in Table 1. It defines traffic type, transport protocol, packet size, buffer size, link bandwidth and delay, type of traffic Source, codec used for encoding and decoding the VoIP traffic, no of frames per packet in data link layer and incoming flow rate. Performance of the proposed system is evaluated by measuring the QoS metrics like throughput, delay and packet loss rate for different simulation scenarios.

Although, several multipath routing algorithm is proposed (Santhi and Natarajan, 2010; Zhang *et al.*, 2008; Kandula *et al.*, 2007) for congestion avoidance or to reduce congestion. The routing decisions are made after the occurrences of congestion by using congestion notification signal, failure link identification, etc. Core routers (Stoica *et al.*, 2003) need to calculate the dropping probability for each incoming packet routers and need to be updated with it per flow state information periodically. An Active Queue Management Algorithm (Santhi and Natarajan, 2010) require ECN notification to identify congestion and then dispersion decision is carried out but ECN notification has to reach the sender in time. During this period packet losses can increase. Generally IP routing utilizes one shortest path for dispersion. Other paths in the network are not utilized since, the cost of the paths is high compared with the selected path. This leads to increase in congestion (Cidon *et al.*, 1999).

Table 1: Simulation parameters for proposed system

Parameters	Values
Traffic type	Exponential
Transport protocol	UDP and SCTP
Packet size	512 bytes
Buffer size	150 bytes
Link bandwidth	200 Mb
Link delay	10 msec
Traffic source	VoIP
VoIP codec	GSM.AMR
No. of VoIP frames per packet	2
Flow rate	250-1000 Mb

The proposed system avoids these issues by implementing load adapter and constraint based multipath routing with priority marking. To balance the load and avoid congestion, congestion free least loaded paths are selected to disperse the packets. During heavily load simple RFQ threshold mapping policy drops the low priority packets by preserving the high priority flows to control the flow rate. Different FEC scheme can be implemented along with proposed system to recover from packet losses that occur due to congestion, link failure, packet or frame error. By utilizing the available paths efficiently, the link bandwidth utilization is improved. Simulation experiment is carried out for three different scenarios.

Case 1: Bottleneck bandwidth plays an important role for packet dispersion. So, the performance of the proposed system is evaluated for different bottleneck bandwidth rates varying from 250-1000 Mb at regular intervals.

Case 2: Generally incoming flow rates will vary for different type of flow. In this experiment flow rate is varied from 250-1000 Mb at regular intervals.

Case 3: In this system load is taken as fixed value, flow time is varied form from 0.5-4.5 sec at regular time intervals.

Simulation results for throughput analysis: Simulation experiments are conducted individually for both CSFQ and proposed system. When the bottleneck bandwidth is increased from 250-1000 Mb at regular intervals different type of VoIP flows with packet size of 512 bytes is admitted in the ingress links. After implementing the three simulation exercises received throughput are given in Table 2. It shows that throughput for proposed system is increased when compared with CSFQ for all three simulation cases.

Figure 3-5 shows the performance comparison of throughput parameter for all three cases of simulation scenarios. At initial stage of throughput comparison both the algorithm results in similar performance but when the bottleneck bandwidth is increased gradually the throughput of the proposed system increase proportionally and its performance increases to 9%. For various flow rates proposed system improves throughput by 8%. During fixed load condition for various time intervals throughput values are measured and it shows that the proposed system attains 1.04% throughput increases.

Table 2: Throughput analysis of simulation scenarios

Bottleneck bandwidth (Mb)	Case 1: Throughput (Gb sec^{-1})		Flow rate (kb)	Case 2: Throughput (Gb sec^{-1})		Flow time (kb)	Case 3: Throughput (Gb sec^{-1})	
	CSFQ	Proposed		CSFQ	Proposed		CSFQ	Proposed
250	1.664	1.960	250	2.144	2.156	0.5	0.144	0.168
						1.0	0.240	0.252
500	3.760	4.124	500	4.288	4.272	1.5	0.256	0.264
						2.0	0.256	0.264
750	2.368	6.272	750	5.104	6.300	2.5	0.256	0.264
						3.0	0.256	0.264
1000	4.336	8.416	1000	4.304	8.408	3.5	0.254	0.264
						4.0	0.252	0.264
						4.5	0.251	0.264

Fig. 3: Comparison of throughput for case 1
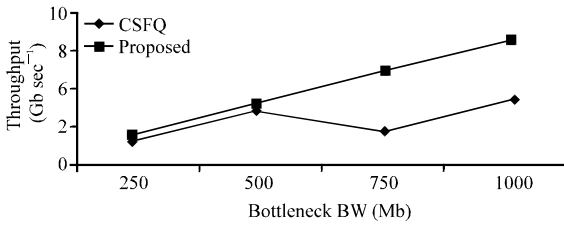

Fig. 4: Comparison of throughput for case 2
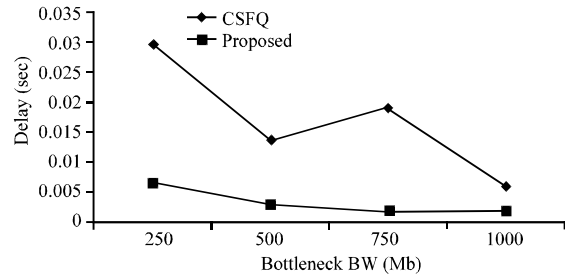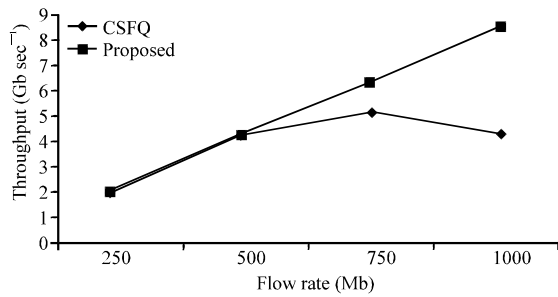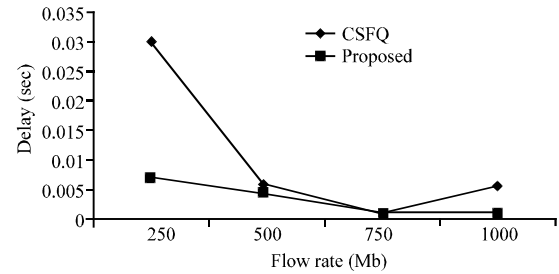

Fig. 5: Comparison of throughput for case 3
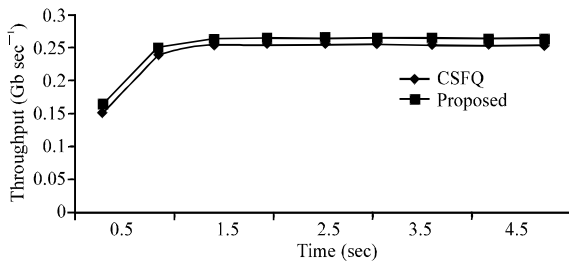

Fig. 6: Comparison of delay for case 1

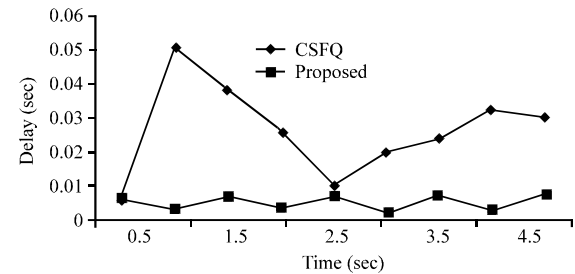Fig. 7: Comparison of delay for case 2

Fig. 8: Comparison of delay for case 3

Simulation results for delay analysis: Delay is measured as the one-way delay experienced by packets between source and destination. Generally when the load increased congestion is also increased propositionally and hence there will be heavy delay. For three different simulation exercises received delay is noticed and given in Table 3. The proposed system decreases the end to end delay during high load, flow rate and for fixed load scenarios.

Figure 6-8 shows the performance comparison of delay parameter for all three cases of simulation scenarios. When the bottleneck bandwidth is increased regular at intervals the proposed system decreases the delay proportionally and its performance increases to 18%. For various flow rates proposed system improves performance of delay by 9.5%. During fixed load

Table 3: Delay analysis of simulation scenarios

Bottleneck bandwidth (Mb)	Case 1: Delay (Gb sec^{-1})		Flow rate (kb)	Case 2: Delay (Gb sec^{-1})		Flow time (kb)	Case 3: Delay (Gb sec^{-1})	
	CSFQ	Proposed		CSFQ	Proposed		CSFQ	Proposed
250	0.029728	0.006728	250	0.030000	0.007000	0.5	0.006	0.007
						1.0	0.050	0.003
500	0.013728	0.002728	500	0.006000	0.003000	1.5	0.038	0.007
						2.0	0.026	0.003
750	0.019061	0.001395	750	0.000395	0.000911	2.5	0.014	0.007
						3.0	0.020	0.003
1000	0.005728	0.001404	1000	0.005728	0.000554	3.5	0.024	0.007
						4.0	0.032	0.003
						4.5	0.030	0.007

Table 4: Packet loss analysis of simulation scenarios

Bottleneck bandwidth (Mb)	Case 1: Packet loss rate		Flow rate (kb)	Case 2: Packet loss rate		Flow time (kb)	Case 3: Packet loss rate	
	CSFQ	Proposed		CSFQ	Proposed		CSFQ	Proposed
250	0.688623	0.627942	250	0.000000	0.005535	0.5	0.000000	0.0000
						1.0	0.000000	0.0000
500	0.46347	0.40093	500	0.003717	0.015668	1.5	0.011976	0.0023
						2.0	0.013043	0.0023
750	0.685775	0.155172	750	0.133152	0.010678	2.5	0.010274	0.0023
						3.0	0.011268	0.0023
1000	0.446939	0.008483	1000	0.451020	0.009892	3.5	0.009592	0.0023
						4.0	0.010417	0.0023
						4.5	0.009225	0.0023

Fig. 9: Comparison of packet loss for case 1

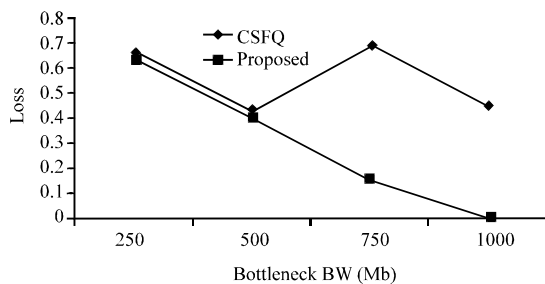Fig. 10: Comparison of packet loss for case 2

Fig. 11: Comparison of packet loss for case 3

condition, the simulation analysis shows that the proposed system attains 23% throughput increases.

Simulation results for packet loss analysis: Network delay and congestion etc can cause packets losses at the application level. Even when the bottleneck (congestion) increases the proposed system decreases packet loss rate compared with CSFQ. Table 4 shows the packet loss rate analysis of all simulation scenarios. The proposed system decreases the packet loss rate during high load, flow rate and for various simulation times.

Figure 9-11 shows the performance comparison of packet loss parameter for all three cases of simulation scenarios. The simulation results shows that the proposed system improves the performance of packet loss rate by 33, 11 and 26%, respectively.

Summarizing all the simulation results it is observed that the proposed system has superior results for all three
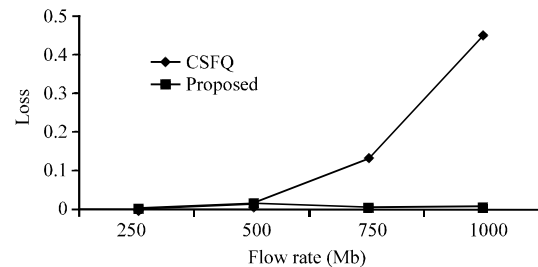
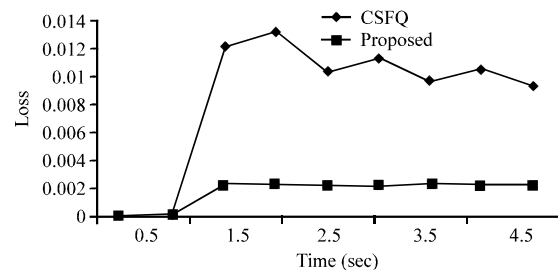QoS parameters of VoIP services. IT attains good throughput with less delay and packet drop when compared with CSFQ strategy.

CONCLUSION

In this study, an MPLS based load balancing is proposed for VoIP flows. Initially, VoIP flows are

differentiated as responsive, unresponsive and dataflow using delay probing techniques. During network unbalanced load condition, triggering policy is invoked to override default routing policy with multipath routing policy. The data rates of unresponsive flow are estimated and flows are marked based on their data rates using Rainbow Fair Queuing (RFQ) mechanism. To perform multipath dispersion and alleviate the problem of congestion, the core router looks for congestion free paths and reroutes the flows into best multiple paths that satisfies the given QoS requirements. Otherwise, it drops the some of the low priority packets from those unresponsive flows. By various simulation results, it is shown that the proposed technique attains good throughput with less delay and packet drop. Due to this improvement the proposed dispersion algorithm can be incorporated in the MPLS router to take automatic dispersion decisions based on the current network conditions to perform efficient load balancing. As future research, this strategy can be extended into interactive video streaming applications.

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