Efficient Load balancing techniques for VoIP applications

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Abstract: VoIP applications require more reliable quality of service guaranteed packet switching techniques. Load balancing is an efficient optimization technique to improve QoS. It moves the traffic from congested links to alternative paths in the network. To perform load balancing and improve link utilization Efficient Load balancing techniques for VoIP applications is proposed for Multiprotocol Label Switching networks. The proposed MPLSMR algorithm classifies the flows and finds the multiple paths for a source and destination pair. To utilize all the available paths efficiently, this algorithm first finds an intermediate node k, from the intermediate node k multiple paths which satisfy the given QoS constraints are discovered using grouping based multipath selection algorithm. The incoming flow is splitted into these paths and the packets are dispersed in weighted round robin fashion.

Keywords: MPLS, VoIP, Load balancing, Multipath routing, QoS, traffic Split.

1. INTRODUCTION

Voice over Internet Protocol (VoIP) transmits voice packets over data network. The audio signal is transformed into digital form by an analog-to-digital converter and voice data is packetized and encoded prior to transmission of the signal at the sender is shown in Figure.1. Encoding - decoding is done by Coder-decoder (CODEC) that transform sampled voice data into a specific network-level representation and vice versa. Most of the codecs are defined by standards of the International Telecommunication Union, the Telecommunication division (ITU-T). Each of them has different properties regarding the amount of bandwidth, encoding standards and delay it requires. Table.1. lists some of the important encoding standards defined by the International Telecommunications Union (ITU) [1].

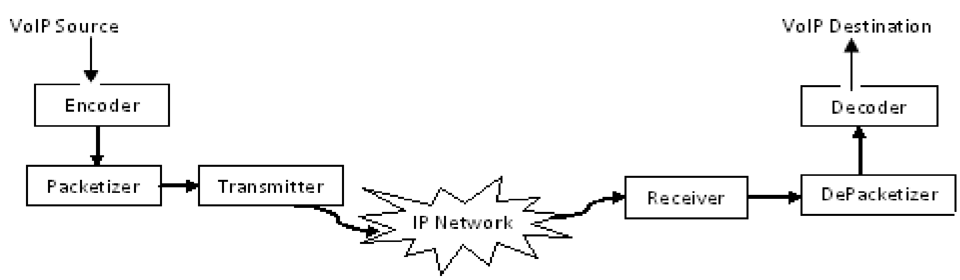

Fig.1. VoIP System

Receiver decapsulates and decodes the received VoIP packets. Decoding may include dejittering, error correction and packet loss concealment. The digital data is then converted to analogue form again and played at an output device. Several standard protocols are used for transporting voice packets in IP networks is shown in Figure. 2. The most commonly used protocols are H.323, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP), H.248, RTP, RTCP, RTSP and RSVP. H.323 is the ITU standard signaling protocol for establishing VoIP connections over packet switched networks. SIP is an Internet Engineering Task Force (IETF) standard text based open protocol for establishing VoIP connections.

SIP is similar to H.323 and HTTP but specifically designed for the Internet to set up a session over the Internet. MGCP is the first protocol developed by the IETF to signal control information between VoIP network components. It is a master - slave protocol that allows a central coordinator (Call Agents) to monitor and instructs communication events and the gateways. They send media to specific addresses to establish and maintain VoIP communication path between endpoints and tear downs the path after communication.

ITU Standard Codec Type	Encryption-Decryption	Bandwidth (kbps)	Delay(ms)
G.711	PCM	64	<1.00
G.721	ADPCM	32,16,24,40	<1.00
G.728	LD-CELP	16	~2.50
G.729	CS-ACELP	8	~15.00
G.723.1	Multi Rate	6,3,5,3	~30.00

Table. 1. ITU standards CODEC types

Application layer	Voice			
	RTP	RTCP	SIP	H.323
Transport Layer	UDP			
Network Layer	IP			
Data Link layer	802.3			

Fig. 2. VoIP protocols

All H.323 terminals use H.245, Q.931 Registration Admission Status (RAS) and Real Time Transport Protocol (RTP) to communicate with an endpoint in some other network. H.248 is supported by both the IETF and ITU. It is also used to signal control information between VoIP network elements. RTP is a standard IETF transport protocol for real time applications that transport the audio/media portion of VoIP communication. RTCP is a control protocol and works in conjunction with RTP.

Real Time Streaming Protocol (RTSP) is a client-server protocol that provides remote control services over the delivery of real-time audio and video streams. A VoIP server provides playback, fast forward, reverse, and absolute positioning or recording services for the media streams. RTSP acts as the network remote controller. Resource Reservation Protocol (RSVP) developed by IETF reserves the resources in each node along the direction data path for specific IP traffic streams.

User Satisfaction	E-Model-R	MOS
Very Satisfied	90	4.3
Satisfied	80	4.0
Some Users Dissatisfied	70	3.6
Many Users Dissatisfied	60	3.1
Nearly All Users Dissatisfied	50	2.6
Not Recommended	0	1.0

Table. 2. MOS and E- Model values

Many Researches on VoIP aims to provide more reliable, timely and fair QoS guaranteed packet switching techniques. Quality of voice is numerically measured in telecommunications using the Mean Opinion Score (MOS). The MOS test is also called as Absolute Category Rating (ACR) test. [2] The European Telecommunications Standards Institute (ETSI) developed the E-model as a computational transmission rating model to address qualitative issue to ensure that users will be satisfied with end-to-end transmission performance. The model integrates transmission rating factor (R-value), as the measure of voice quality Comparing the MOS scale and E-model provides a reference as to what is considered acceptable. Table. 2. shows a comparison MOS and E-Model.

The three main factors that mainly affect VoIP quality are delay, jitter, and packet loss. Latency (delay) is the mouth-to-ear overall delay. Several factors describe latency is as follows. Each compression algorithm has certain built-in delay referred as CODEC Latency. Packet creation delay, queuing delay is referred as waiting time in a queue for transmission. The time a network device takes to make the forwarding decision is known as packet forwarding delay or network delay. Voice Packet data must be delivered in a timely manner in order to ensure user satisfaction. Jitter can be defined as one-way delay variation and influences quality if it exceeds a maximum value. Most users recognize round-trip delays when they exceed 250 ms. Network congestion, timing drift, or route changes can cause Jitter and may differ for each packet.

Packet loss occurs when packets sent are not properly received for playback by the other end. Overloaded links, buffer size in the receiving device, collisions, congestions in link, physical layer errors etc, can cause packet losses. When the packet losses arises the receiver either introduces gaps in playback or tries to recover from this error by using Packet loss concealment , forward error correction techniques.

Common approaches for designing VoIP network and other real time applications are best effort services, integrated services (IntServ), differentiated services (DiffServ) and traffic engineering MPLS network. Best effort service means that each user gets a fair share of the available network resources with no promise of QoS guarantees like delay, throughput and jitter. IntServ is developed to provide customized support for different service classes by making advanced reservation of the required resources. It supports two types of service classes: guaranteed service allows receiver of the flow to specify maximum allowable end to end delay and ensures bandwidth availability. The second class is controlled load provides unloaded or minimum loaded path to a flow. If the network cannot provide the required bandwidth the session is not allowed.

DiffServ is designed as an alternative to IntServ to classify and manage network traffic to provide QoS. It provides services to aggregates classes and defines per hop behavior. A single flow receives the same QoS as of all the other flows in that aggregates. Per flow state information is not maintained in router. This simplifies the routing overhead. DiffServ makes use of DiffServ field in the IP header to assign a packet to aggregate class. It also makes use of Service level agreement (SLA) between provider and subscriber. MPLS- TE networks provide the QoS guaranteed level of service and best meets the VoIP requirements of voice users.

MPLS is an advanced packet-forwarding technique uses encapsulated fixed length labels to make high speed forwarding decisions [3]. Routers in MPLS network are referred as Label Switching Routers (LSRs). The router by which a packet enters the MPLS is called the ingress LSR, and the one by which it leaves the MPLS is called the egress LSR. Label switched path (LSP) is the routing path that starts with ingress Label switch router and terminates at egress LSR. The ingress nodes add the label for each incoming packet. The labels are detached from the egress border node, whenever the packet leaves the MPLS domain. With MPLS, the processing overhead required for routing at the intermediate nodes could be reduced, thereby improving their packet forwarding performance. In addition, the MPLS-Traffic Engineering (MPLS-TE) approach [4] allows setting up explicitly routed Traffic Engineering-Label Switched Paths (TE-LSPs) whose paths satisfy a set of traffic engineering constraints, including bandwidth, throughput etc. The example of MPLS network with an LSP from LER1 to LER5 for the flow f1 from server s1 to destination host h1 is shown in Figure. 3.

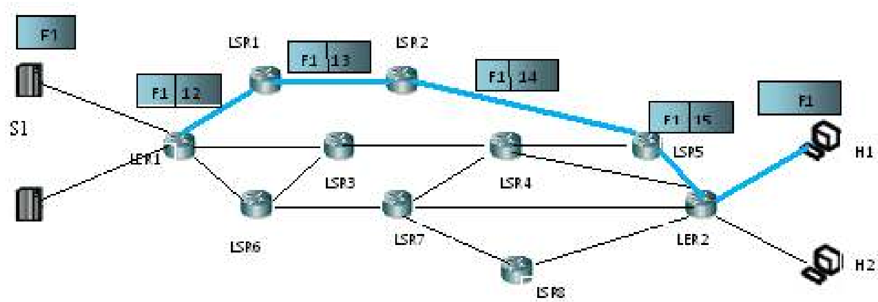

Fig. 3. MPLS Network

Load balancing moves the traffic from congested links to other parts of the network. It is an optimization problem when its traffic demands are known. Link or node failures introduce additional overhead to reconfigure routing tables and find an alternative path for packet dispersion. The objective of traffic aware routing and load balancing are to avoid congestion while the traffic is routed from a router to another. To perform load balancing and improve link utilization the proposed technique utilizes the Multiprotocol label Switching (MPLS) network multipath routing technique.

2. RELATED STUDY

To perform link utilization Multipath Adaptive Forwarding Equivalence Class (MAFEC) algorithm is proposed [5]. This MAFEC transmit the same-label-traffic into multiple available LSPs (label switch path). This technique tries to avoid routing oscillation and aims to increase efficient usage of network bandwidth. Fish network model is used to analyze and find multiple paths link-disjoint paths. The packets are transmitted on all available shortest LSP to perform link utilization.

Efficient Bandwidth Estimation Management is proposed for VoIP Concurrent Multipath Transfer [6]. Grouping-based Multipath Selection algorithm is utilized for selecting the multiple paths for packet dispersion. The bandwidth for each path is computed using Westwood approach. Then the best paths that statisfy the required QoS are selected and the packets are dispersed into multiple paths using SCTP protocol. SCTP allows multiple paths to be established between the same source and destination nodes. This technique improves link utilization and performs load balancing.

Bandwidth management technique for Multiprotocol Label Switched networks is proposed [7]. The technique identifies multiple paths between source and destination. Packets are dispersed into one shortest path and the remaining is considered as backup paths. The path failure or node failure utilizes the backup paths and bandwidth sharing among these paths increases network resource utilization.

A reliable multipath routing for IEEE 802.16 wireless mesh networks is proposed [8] to perform interference load aware (ILA) routing. The source node selects multiple paths for the destination node and a value is associated with each path by evaluating QoS constraints. One path is notified as primary path and the alternate path is chosen from the set of multiple paths which has the second minimum value. In case of failure in alternate path, then source node chooses second alternate path that has next minimum value and this process continues. This technique tries to avoid congestion and packet losses.

Adaptive load balancing policies use real time system state information based on various metrics like bandwidth etc to take load balancing decisions. To minimize the packet loss and to achieve load balancing a lightweight SIP load balancing scheme is proposed [9] that aims to provide services towards call transmission, link failures, device failures etc. The proposed load balancing scheme are evaluated on real time services on the Web Servers. Balancing the load of SIP transactions can affect the following factors such as availability, redundancy & QoS.

An alternate approach to perform load balancing and to minimize the packet losses are discussed [10]. The incoming packets are dispersed over multiple paths, in contrast to the traditional single path routing approach. This study also focused on the network performance by calculating the loss rate and loss burstiness on the applications. NLR metrics have been analyzed for various packet dispersion strategies over memoryless (Bernoulli) loss model or bursty (Gilbert) loss model to minimize the packet loss rate.

To perform load balancing in MPLS network Multipath adaptive packet dispersion is proposed [11] . The incoming traffic flow is classified using the packet probing mechanism. The trigger handler is proposed to check the balanced load in the system using buffer occupancy threshold value. If the network condition becomes unbalanced, the adaptive packet scheduler routes the flow into the best shortest multiple paths.

Hong Li et al [12] proposed a mechanism to select the optimal multipath with adaptive playback scheduling at the receiver. The resulting VoIP performance is evaluated with R factor. The R-factor is a Score defined in ITU – TE model. It measures the subjective voice quality under many mouth-to-ear impairments. Adaptive playback scheduling is a technique that is used at the receiver to adjust the playback delay and playback loss for the received voice signal.

SMLDR (Shortest Multipath Labeled Distance Routing) is an on-demand loop free multipath routing protocol[13]. It modifies the labeled distance routing protocol. The shortest path is calculated based on the distance. The routing table entries are ordered on the basis of the limiting distance. SMLDR finds multiple shortest paths with equal cost distance. These paths are considered as alternate path to perform link failure recovery.

Analysis of several multipath algorithms is performed by taking the Various routing algorithm, route discovery methods, routing choices, traffic distribution, packet allocation granularity, route maintenance and its applications are studied [14]. Generally route discovery can use one of 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.

A randomized distance vector routing protocol (RDVRP) is used to distribute the data traffic randomly over all available paths to a destination in the network to perform load balancing. Multiple loop-free next-hops is found for a destination and updated with the forwarding table (FIB) at each router. Keeping multiple entries will increase the memory needed for FIB. The RDVRP describes the changes in the routing tables. It also describes the next hop calculation and population methods by considering the properties like convergence time, inconsistencies during the convergence period etc. The performance of the RDVRP is compared with SEGAL algorithm. SEGAL is a distance-vector and loop-free routing protocol uses a shortest-distance path for each pair of source and destination nodes[15].

Congestion-Triggered MultiPath routing (CTMP) scheme modifies the existing Path Vector (PV) routing to compute class-c paths and includes additional congestion-related information along with the routing information [16]. The ECI (Explicit Congestion Indication) and MPI (Multipath Indication) are one-bit flags stored in the routing table for each destination. The ECI bit is set to 0 by default to indicate no congestion and 1 to indicate congestion on the path to the destination. The MPI bit has a default value of 0 and is set to 1 if a class-c multipath route is established to the corresponding destination. The path capacity and utilization can be computed by a given node in a similar way as the path vector in PV routing. The packet dispersion takes place to these multiple paths.

Multipath Distance vector routing algorithm MDVA uses a set of loop-free invariants to prevent the count-to-infinity problem caused by Distributed Bellman-Ford (DBF) algorithm [17]. Each node maintains a table to store the successor set, the feasible distance, the reported distance and the shortest distance for the successor set. The table also stores the set of waiting neighbors in a diffusing computation. Each node also maintains a neighbor table for each neighbor with their distance metrics. The link table stores the cost of adjacent link to each neighbor. Using these values the count to infinity is solved and multiple paths have been discovered.

3. PROPOSED SYSTEM

To achieve load balancing MPLS based multipath routing is proposed. In MPLS multiple LSPs between source and destination is established. Packets belonging to a single flow can be routed through these multiple LSPs. To find multiple paths between source and destination pair a MPLSMR (MPLS - multipath routing) algorithm is proposed.

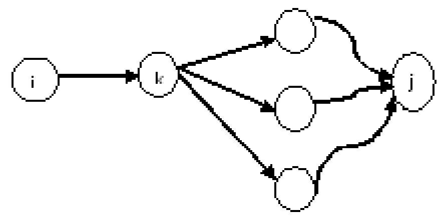

Fig. 4. Multipath from intermediate node to destination

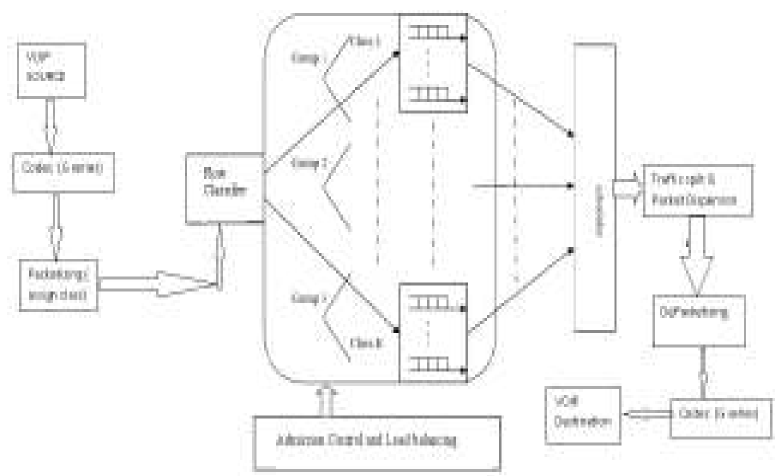

Fig.5. Multipath Load Balancing System

MPLSMR is based on distance vector bellman ford algorithm. To utilize all the available paths efficiently, this algorithm first finds an intermediate node k, from the intermediate node k multiple paths which satisfies the QoS constraints are discovered. The packets are dispersed into these multiple paths with the required QoS. An example for multipath routing is shown in Figure.4.

Multipath routing involves flow classifier, path discovery (scheduler), traffic distribution, splitting and Load balancing. The proposed system is shown in Figure.5.

Steps involved in the proposed Multipath routing

Multiprotocol Label Switching Multipath Routing Algorithm (MPLSMR) in MPLS network G= (V,E)

For a given source and destination node i, j € V

- Classify the incoming flow f into appropriate group
- Find an intermediate node k for the given source node i
- Find all available shortest paths from k to destination j , referred as P {k, j}
- Calculate bandwidth (bw) and delay (d) for the P {k, j} shortest path set.
- Group these paths using the calculated bandwidth
- Select the path which has higher bandwidth and minimum delay in each group
- If the number of paths selected are greater than the required paths then select the best path from the list
- Split the flow among these paths
- Disperse using weighted round robin dispersion.

3.1. Route lookup and Path Discovery

Path discovery is the process of determining the available paths for a source-destination pair. In MPLSMR the multiple paths are found from intermediate node to utilize all the available multiple paths efficiently for the same source – destination pairs. From the available multiple shortest paths the number of link disjoint loopless paths for transmission are selected based on the QoS constraints. The maximum bandwidth and minimum delay paths are the constraint selected for dispersion. Link disjoint paths have no common links. A link failure will not affect entire the dispersion only that particular link is affected. The packets for that link are redirected to another path.

In a distance vector routing protocol, each node maintains a routing table and a distance vector. The Protocol learns routes to an IP destination address from its own locally connected networks and from routes received from neighboring routers. A shortest path neighbor to each destination is calculated and maintained in the table. Each node has incomplete knowledge of the network topology, and knows only its neighboring nodes. From these neighbors, the node chooses a closest neighbor to each destination. Each node periodically sends its distance vector to each of its neighbors to inform the routing updates. MPLSMR protocol find shortest path and records all the other alternative shortest paths and does not discard it. The routing protocol maintains all possible next hops.

First MPLS-MDVR finds an intermediate router for each flow. Then multiple paths from that intermediate router are found using path discovery procedure. Using Grouping-based Multipath [10] Selection more than one path can be selected to transmit data simultaneously by identifying the correlations between multiple paths. The flow splitter splits the flow among these paths.

In this multipath selection technique, the entire shortest path from a source to destination is grouped based on their target Bandwidth and delay. The bandwidth and delay for each for paths need to be identified for each path. Assume bandwidth and delay is known and grouping is carried by assigning the initial representative bandwidth Bxy which has an identical source or destination address path. A new bandwidth path is only compared to its representative bandwidth of the group to determine whether it should join the group or not. However, if there is no representative bandwidth in the group, the new bandwidth is not joined to that group. Finally, if the new bandwidth cannot be joined to any existing group, a new group can be created with the bandwidth as representative bandwidth. The path with the highest bandwidth is selected from each group. If the number of required path by the application is less the selected path then the path with maximum bandwidth and minimum delay are selected within the selected paths.

3.2. *Load balancing*

Flow classifier classifies the incoming flows and places the packets in the appropriate queue based on their QoS requirement. The load adapter is used to check the system load balancing condition. System load balancing can be measured by using different triggering policies like Maximum Queue Length Threshold (MQLT), Buffer Occupancy Threshold (BOT), and Periodic Mapping (PM). The proposed approach uses BOT policy to measure the system load balancing. When queue occupancies of the active users exceed a threshold value, load adapter is invoked. Queue occupancy threshold can be set to the ¾ of the total available buffer space. When the system is in unbalanced state and the triggering policy invokes multipath routing and the packet are routed into multiple congestion free least loaded path. Otherwise the flows are routed normally by using default routing policy.

3.3. *Traffic Splitting and scheduling*

A multipath protocol can forward traffic using only the path with the best metric. The remaining paths from the set of available paths are considered as backup paths. Or utilize all the available paths concurrently. The number of paths can be selected based on the choice of path reliability, disjointness, available bandwidth and delay. In QoS routing, a subset of paths are selected if the combined metric (bandwidth and delay) satisfies the QoS requirement. In the proposed system multiple paths are utilized in a weighted round-robin fashion. Flow-Based traffic Splitting approach is based on the technique of solving the maximum multicommodity flow problem [18]. This approach requires per flow state information to be calculated. For example for distributing the packets from source i to destination j, first the intermediate node k is selected. Then multiple paths from node k to j are discovered. The packets 1,2 are splitted and dispersed via path1, packet 3 is dispersed via path2, packet4,5 are dispersed via path3. Traffic splitting is shown in Figure.6.

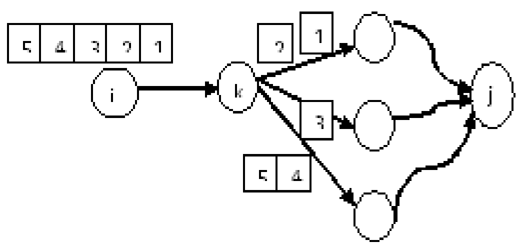

Fig.6. Traffic splitting for multiple paths

The traffic flow split algorithm works as follows.

Start with initial weights W (e).

1) Compute intermediate nodes k for which W(e) is maximum. Identifies paths P_i from source node i to intermediate node k and paths Q_j from intermediate node k to destination node j.

2) For a traffic at intermediate node k, the incoming traffic rate on path Pi is Ri and the traffic rate on path Qj is Cj . Compute total traffic flow rate f (e) on link e as given in Eq.1.

$$f(e) = \sum Ri + \sum Cj \quad (1)$$

3) Compute the maximum value α for the traffic split ratio for intermediate node k for sending flow along paths (Pi,Qj) , as given in Eq.2.

$$\alpha = \min(ue/f(e)) \quad (2)$$

u is the constant value selected for traffic split.

4) The split ratio α is associated with node k.

4. SIMULATION RESULT

Simulation setup: The proposed multipath routing is simulated using the network simulator-2 (Ns-2) [19]. This routing scheme can be implemented in the network by forming variable bandwidth paths between the nodes. The simulation topology consists of 10 nodes is shown in Figure.7. It has one sender (ingress node) and 2 destinations (egress nodes) with 3 junction nodes. The different link bandwidth and link delay is assigned for all the links.

VoIP traffic for the ingress node with the following specifications is taken for routing. Packet size: 1000kb, Traffic Model: Exponential, VoIP codec: GSM.AMR, Number of VoIP frames per packet: 2, Rate: 5Mb.

The proposed strategy is evaluated based on the two primary QoS parameters such as throughput and delay for MPLS Multipath Routing compared with single path routing. Multipath routing provides several advantages than single path routing. It provides Fault tolerance, since the redundant information can be routed to the destination via alternative paths. When a link becomes over utilised or when there is congestion in network multipath routing performs load balancing by diverting traffic through alternate paths. The algorithm splits the traffic into multiple streams for effective bandwidth aggregation and for End-to-end delay minimization.

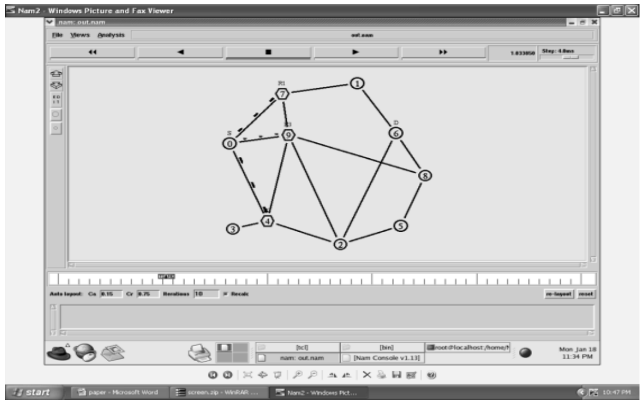

Fig. 7. Simulation for MPLS Multipath Routing

Simulation of the proposed work is carried out by varying the load as 1500, 2000, 2500, 3000, 3500, 4000, 4500, 5000 bytes and the delay is noticed in sec. The delay is reduced for the proposed MPLS MR (MPLS- Multipath routing) algorithm than single path algorithm is shown in Figure.8. The delay is noticed for various time intervals from .5 seconds to 4 seconds for both multipath and single path algorithm and the end to end delay is reduced in the MPLSMR algorithm is shown in Figure. 9. Similarly throughput is measured for the same time intervals and the MPLSMR algorithm increases the throughput ratio. It is shown in Figure. 10.

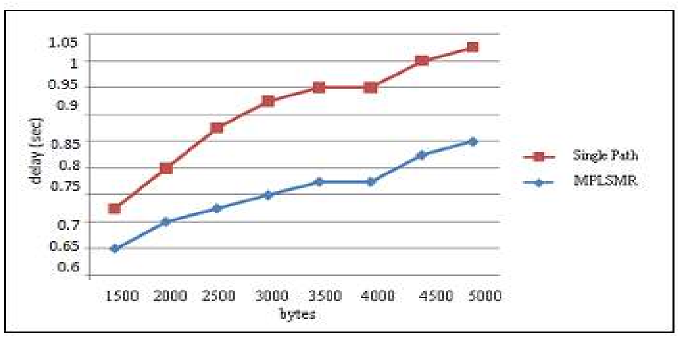

Fig.8. Load (bytes) Vs Delay (Sec)

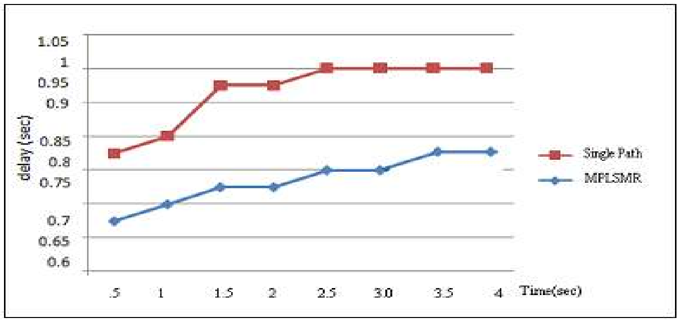

Fig. 9. Load (bytes) Vs Delay (Sec)

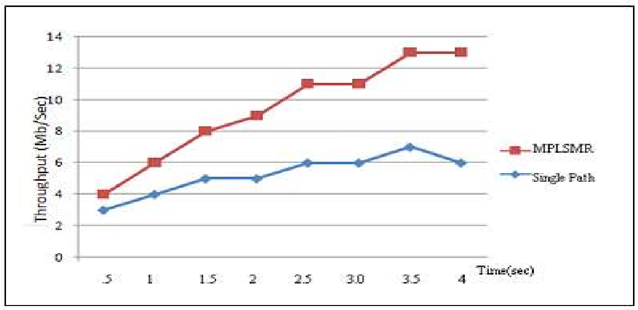

Fig. 10. Load (bytes) Vs Delay (Sec)

5. CONCLUSION

In this paper Efficient Load balancing techniques for VoIP applications is proposed which classifies the flows and finds the multiple paths for a source and destination pair using MPLSMR algorithm. To efficiency is achieved by utilizing all the available paths from the intermediate node. Traffic splitter splits the incoming flow into multiple paths which satisfies the required QoS and the packets are dispersed in weighted round robin fashion. It provides Fault tolerance, load balancing, bandwidth aggregation, link utilization and End-to-end delay minimization. The proposed system achieves high throughput and less delay when compared with single path routing.

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