

Study of QoS Management Techniques for Voice Applications

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Abstract—In recent years, Voice over IP has influences on global telecommunication. Voice and multimedia applications delivery services remain a major and challenging task for network researchers and designers. VoIP packets are sensitive to packet losses and transmission delay to assure correct communication. Quality of Services provisioning for coexisted and traffic flows in networks is still demanding problem. This paper discusses QoS management techniques for VoIP applications. Some important techniques like flow classification, load balancing policies, congestion control and packet loss recovery are studied. Finally, three routing policies like Single path routing, phase routing and multipath routing with forward error correction are simulated using Ns2 simulator and the results are analyzed to identify the best routing policy.

Keywords— Multipath routing, MPLS, QoS, Two phase routing, VoIP.

I. INTRODUCTION

VOICE over IP (VoIP) enables the voice and other multimedia real time sessions over IP-based networks with low cost. The critical problem is performing load balancing and assuring Quality of Service (QoS) for VoIP applications. Quality of VoIP applications is affected by basic network behavior, codec type, packet loss, delay and jitter. [1] QoS are managed by variety of mechanisms such as flow classification, load balancing, routing policies, admission control and traffic shaping, etc [2]. By classifying flows, it enables the users to learn more about the traffic and identify characteristics that are common to a set of flows. Due to this, the operator is able to manage flows from one class differently from another.

Common QoS Mechanisms for VoIP network and other real time applications are best effort services, integrated services (IntServ), differentiated services (DiffServ) and traffic engineering MPLS network [3]. 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 based on IETF RSVP (Resource Reservation Protocol) for QoS between end user nodes. Diffserv works among intermediate nodes on L2-L7 layers of OSI. The most

important feature of “coloring” the IP traffic is the DSCP (Diffserv Code Point) field in the IP packet header. 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 aggregate. MPLS-TE networks provide the QoS level of service and best meets the VoIP needs of voice users.

The ingress interface needs classification, marking, and shaping (e.g. FIFO, FQ-Fair Queue, WFQ-Weighted Fair Queue, WRED-Weighted Random Early Detected, “tail-drop”, Q-Low Latency Queuing), while the egress interface needs congestion avoidance, policing, and shaping tasks, etc. The voice is classified as the traffic with the highest priority. Modeling packet switched networks is mainly characterized by manipulating only the packet arriving time series as a stochastic process [4].

QoS requirements of VoIP include packet loss, delay, and delay jitter. The current H.323 and SIP frameworks support some kind of interfaces to QoS management, but they do not provide functional QoS management mechanisms. To provide the QoS for multimedia applications several important techniques including packet classifier, buffer management, scheduling, loss recovery, and error concealment techniques, admission control, resource provisioning, traffic engineering, and connection management techniques are identified. Admission control plays a critical role in QoS for VoIP [5].

II. RELATED APPROACHES

A. Flow classification and Congestion Control Techniques

Traffic flows are categorized into distinct classes in order to control and manage highly aggregated Internet traffic flows efficiently. It also has the potential to support ISP provides to assure the required QoS for different users. Furthermore, real-time traffic classification is the core component of emerging QoS-enabled products and automated QoS architectures. To classify the flows in AQM queues an Active queue management (AQM) for non responsive traffic is proposed [6]. It is based on statistic measurement on the incoming traffic rate and the AQM packet loss rate. This scheme is implemented by estimating the average periodic cycle of the responsive traffic using a traffic estimator. Then, a wavelet de-noising filter is applied to remove the non-responsive traffic bursts before they enter the AQM queue. A wavelet de-noising filter uses a threshold function to remove the high peaks of traffic changes at different time scales. The thresholds should be set such that the resultant de-noised traffic is exactly the same as the responsive traffic. The

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accuracy of de-noising is measured through the Mean Square Error (MSE) between the packet loss according to the de-noised traffic and the estimated packet loss according to the responsive traffic, in each periodic cycle. Extra buffer space is used to allow non-responsive traffic passing the router, available buffer size puts an additional constraint to how much nonresponsive traffic can be filtered before packet drop happens.

To regulate the flows edge routers keep track of incoming flows and their arrival rates [7]. Edge routers classify incoming packets, compute flow-specific arrival rates and rate-limit of individual flows. Core routers use the Random Early Detection (RED) algorithm for queue management and generate source quenches on packet drop to advice sources to reduce their sending rates. Then edge routers control these flow using token-bucket based regulators. The congested routers in ERUF generate rate limited source quenches in response to packet drops. These source quenches are used by edge routers to detect congestion (or incipient congestion) further downstream. The main disadvantage of using source quenches is that it adds traffic in the reverse direction of the original flow, which increases congestion along paths that have multiple congested routers in both directions.

B. Load Balancing and Traffic Splitting Techniques

Link failure or system failure leads to load unbalanced situation. Traditional routing approaches take considerable long time to update its routing table and to recalculate the routes to take path diversion. Therefore load balancing becomes an important criterion. Load balancing can be adaptive or non adaptive. Adaptive load balancing policies use real time system state information based on various metrics like bandwidth, delay, free available memory, to take load balancing decisions [8]. Non adaptive or static load balancing policies do not use real time system state information's. To achieve load balancing and fault tolerance, traffic can be dispersed into multiple paths. Multipath 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. Traffic is distributed in single path, two simultaneous paths or multipath concurrently [9].

SMLDR (Shortest Multipath Labeled Distance Routing) is an on-demand loop free multipath routing protocol [10]. 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.

Several traffic splitting algorithm like flow based, packet based, S-Hash splitting and Bin-based splitting were proposed to split the traffic flow into multiple paths. FLARE is flowlet based splitting technique [11] uses periodic pings to estimate the delay between multiple paths and uses a hash table that maps flow into paths. Each table entry contains two fields called time and path id. When a packet arrives, FLARE computes a hash of the source IP, destination IP, source port and destination port. This hash is used as the key into the

flowlet table. By identifying the forwarding equivalence class of the traffic aggregate traffic flows are divided [12] on a per-packet basis at the ingress of an MPLS cloud and dispersed into multiple paths by adding flow label. This is achieved by inserting splitting id in the TTL field of the MPLS header along with the sequence number in the label filed at the ingress router. Fig.1. shows the splitting of MPLS Header



Fig.1. Splitting MPLS Header

The combined use of the differentiated services (DiffServ) and multiprotocol label switching (MPLS) technologies is envisioned to provide guaranteed quality of service (QoS) for multimedia traffic in IP networks[13,14]. The traffic bandwidth requests are estimated to provision the LSP and allocate resources to satisfy the QoS of the requests. This approach includes LSP setup, LSP capacity allocation and LSP routing. Markov decision Processes is used as the basis of the LSP setup algorithm, Kalman filter theory is used for the LSP capacity allocation and the LSP routing relies on the Stochastic Comparison theory.

Multi-homed Stream Control Transmission Protocol (SCTP) and Multi-Protocol Label Switching Traffic Engineering (MPLS-TE) mechanisms are used to perform load balancing [15-17]. Mohamad Chaitou et.al [18] have proposed fast reroute extensions to RSVP-TE to support the protection of multipoint to multipoint (MP2MP) MPLS TE-tunnels. An MP2MP TE-LSP is a bidirectional TE-LSP which connects a set of leaf nodes that can act indifferently as sender or receiver. The main motivation is to reduce the number of TE-tunnels (TE-LSP: TE-Label Switched Path) needed to maintain multipoint connectivity between edge routers [19]. Reducing the number of TE-LSPs reduces the number of control plane and data plane states that are present on a node. A state denotes the data information which is must be stored at a node to maintain a TE-LSP. The number of states needed to maintain a TE-LSP on a router must be equal to the number of neighbors of this router that are crossed by this TE-LSP.

Reduction in the number of TE-LSPs also reduces the length of MPLS tables in the data plane. Number of TE-LSPs also reduces the length of MPLS tables in the data plane. Fast Re-Route (FRR) mechanism is a used to protect MPLS-TE tunnels in the case of link and/or node failure. An MP2MP bypass TE-tunnel connecting a set of nodes around the protected element is used to protect an element (link or node) of a primary MP2MP TE-tunnel. A bypass TE-LSP connecting the upstream node of the protected element, called Point of Local Repair (PLR), to the downstream node of the protected element, called Merge Point (MP), is used to encapsulate the primary P2P TE-LSP during failure. The node of the primary MP2MP TE-tunnel upstream to the protected element, called the Upstream Protecting Node (UPN), selects the MP2MP bypass TE-tunnel to be used for the protection.

C. Packet Loss and Congestion Recovery Technique

There are several receiving end Packet Loss Concealment (PLC) techniques [20] are available. They are Frame Estimation, Frame Substitution, Frame stretching, Frame Delaying (or) Adaptive play out. Forward error correction (FEC) is media-specific or media-independent packet loss Concealment technique [21]. Naofumi Aoki et.al [22] have proposed a Packet Loss Concealment Technique for VoIP using Steganography. In this technique, sender side information that improves a receiver waveform (reconstruction technique), is transmitted by using steganography, so that its datagram is completely compatible with the conventional format of VoIP. The two way Pitch waveform replication (PWR) technique is used for transmitting the data from sender to the receiver. The two-way PWR takes into account the waveform reconstruction from the forward frame of the gap as well as the backward one. The performance of the two-side PWR is enhanced by using sender-based side information. Even though the proposed technique requires more complicated procedure, its look-ahead delay, which can be managed by the buffer at the receiver, is still the same as that of the conventional two-side PWR.

K.Maheswari et.al [23] have proposed a sender and receiver based repair technique to overcome the difficulties faced by VoIP. It is an evaluation of speech quality by applying improved Packet Loss Concealment (PLC) algorithms with an end-to-end bidirectional transmission control scheme. The channel condition is estimated before transmission at the sender side. If the network is busy, then the transmission is halted from the source otherwise the packets are transmitted to the destination. The destination sends acknowledgment to the sender. The encoded audio samples are collected into packets and transmitted over the network. An artificial voice reconstruction delay is introduced at the receiver. Receiver maintains network delay and readjusts the amount of buffering between talk spurts. The amount of buffering is enough to receive most of the packets, but some will always arrive too late to be played back and hence, can be considered lost. The lost packets are replaced by copies of last received packets. The limitation faced here is the buffer must have a copy of the last packet.

Teck-Kuen Chua et.al [24] have proposed loss-recovery techniques which is used by several designers to mitigate the undesired effects of packet loss. Audio quality is degraded by losing packets and losing voice packets in the network. Some of the loss-recovery techniques use sender based procedures, and others use receiver-based procedures. In sender-based loss-recovery techniques, the sender assumes an active role to help the receiver recover lost data or improve QoS when packet loss occurs. Commonly, sender-based techniques are independent of receiver-based techniques, so designers can employ both types of loss-recovery methods simultaneously.

The analysis of the bandwidth requirements, buffering delays, and perceptual sound qualities have been made. The analysis result shows that every sender-based loss-recovery technique is not suitable for real-time interactive VoIP communications. The result also shows that the interleaving

technique does not achieve any consistent or significant improvement in an environment of lost packets. The pFEC technique with $n = 2$ is effective for improving the audio quality in random- loss and network-loss environments. But pFEC technique fails to improve the audio quality at all when losses of multiple consecutive packets occur.

RTCP's feedback based adaptive FEC called s proposed [25]. The USF algorithm receives feedback from the receiver that consists of the number of packets lost before reconstruction, number of packets lost after reconstruction. This value will be used by the USF algorithm to calculate the percentage of packets lost after reconstruction. It also gets an additional value like the total number of packets lost in loss bursts. This value will be used to account for packet loss bursts in the network. The amount of redundancy added to the packet stream by the USF algorithm depends on the current value of the combination number. The value is initialized to 0 at the start of a voice session.

III. COMPARISON OF ROUTING POLICIES

Various levels of Quality of Service (QoS) can be provided within a network by using different routing policies. This article examines the three approaches like single path routing, two phase routing and multipath routing. To achieve QoS, Vandermonde Matrix Level FEC mechanism is combined with the above routing policies and the efficiency is compared.

The QoS Performance relies on selecting best routing paths that have sufficient resources for all connections to be admitted and efficient resource utilization. Single path routing is the traditional approach. Generally single path routing suffers from convergence delay, as it takes considerable long time for the update messages to propagate throughout the network and for the routers to re-calculate routing paths. In the convergence period, packets may be delayed, dropped, or fall in temporary routing loops. By doing so, the significant performance of on-serving applications will definitely be degraded. Therefore VoIP applications require a special packet forwarding architecture.

Twophase routing is an alternate approach proposed by [26]. In Phase1, the maximum incoming traffic flows are divided and forwarded to intermediate node called as Rendezvous Point (Intermediate node). The decision of RP does not depend on the destination node. RP node is selected by evaluating link weight. In phase2, the traffic flows are routed to their respective destinations. The two phase routing strategy depends on the ingress-egress capacities T_i , C_i and the traffic split ratios f_j . Fig.2. shows an view of two phase routing.

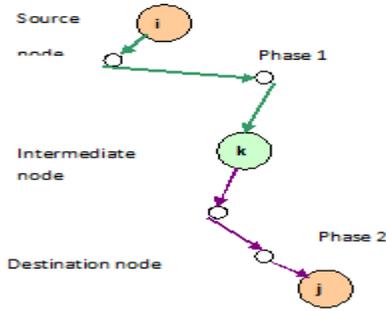


Fig. 2 Two Phase Routing

Multipath dispersion splits and disperses the packets simultaneously in all available paths. Hanoach [27] proposed a multipath dispersion model to study the effect of packet dispersion on the quality of Voice over IP applications. MultiProtocol Label Switching network creates multiple label switched path between sender and receiver. The required number of paths which satisfies the traffic engineering constraints can be selected. The packets are dispersed into these paths using a round-robin dispersion algorithm. The destination endpoint, receive and synchronize packets arriving from parallel paths and manage the jitter buffer optimally in order to reduce delay to a minimum and handle out-of-order packets.

Vandermonde packet Level FEC [28] can be used to recover from packet erasure. Packet-level FEC works by adding another error-recovery packet for every N packets that are sent. This FEC packet contains information that can be used to reconstruct any single packet within the group of N. If one of these N packets happens to be lost during transfer, the FEC packet is used on the far end to reconstitute the lost packet. This eliminates the need to retransmit the lost packet, which dramatically reduces application response time and improves network efficiency.

IV. SIMULATION RESULTS

In this single path routing, twophase routing, MPLS multipath routing techniques have been simulated using the network simulator-2 (Ns-2) [29]. The simulation topology in consists of 10 nodes as shown in Fig.3. It has one sender (ingress node) and 1 destination (egress nodes). The link bandwidth is assigned as in Fig.3. and link delay is set as 10ms.

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, Encoder & Decoder: VoipEncoder, VoipDecoderOptimal.

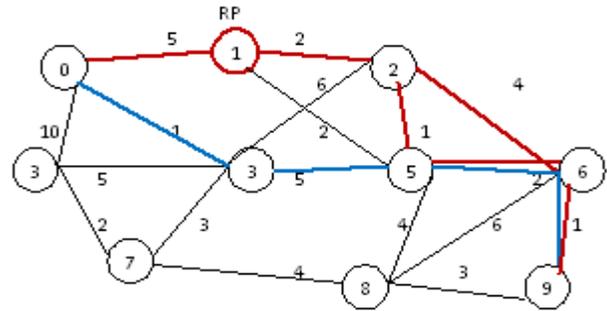


Fig. 3 Simulation Topology for Single path, two phase and Multipath Routing

The proposed strategy is evaluated on voice samples based on the two primary QoS parameters such as received bandwidth and packet loss rate. Loss rate is given by the ratio of average number of packets lost at the receivers to the average number of packets sent. Traditional single path routing selects best shortest path for dispersion. It selects path 1 as $0 \rightarrow 3 \rightarrow 5 \rightarrow 6 \rightarrow 9$. Similarly two phase routing selects node1 as RP point. It disperses the packets into two different paths from that point path 1 as $0 \rightarrow 1 \rightarrow 2 \rightarrow 5 \rightarrow 6 \rightarrow 9$ and path2 as $0 \rightarrow 1 \rightarrow 2 \rightarrow 4 \rightarrow 6 \rightarrow 9$. MPLS based Multipath routing selects multiple available paths (no of paths selected as 3), path1 as $0 \rightarrow 3 \rightarrow 5 \rightarrow 6 \rightarrow 9$ and path2 as $0 \rightarrow 1 \rightarrow 2 \rightarrow 5 \rightarrow 6 \rightarrow 9$ and path3 as $0 \rightarrow 3 \rightarrow 7 \rightarrow 8 \rightarrow 9$.

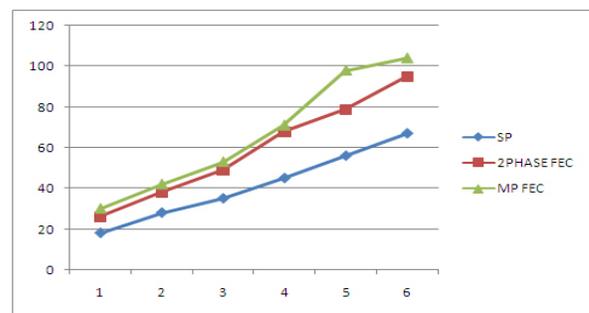


Fig.4. Time Vs Bandwidth

The packets are received at the destination node 9 and its throughput and delay are evaluated. It is shown in Fig. 4 and Fig.5. Multipath routing attains good results compared to single path and two phase routing. Load balancing becomes simplest and efficient approach With MPLS multipath routing. Since single link or node failure does not affect entire dispersion. It can be rerouted immediately to another backup path. It provides fault tolerance. But existing MPLS routing need to traffic engineered with service level requirements constraints to achieve QoS guaranteed service.

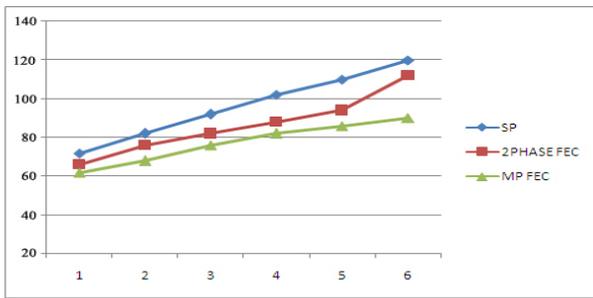


Fig.5. Time Vs Packet loss

V. CONCLUSION

This paper proposes a study of QoS management techniques for enhancing the Quality of Services for Voice over IP applications. Several QoS techniques and its approaches are analyzed. Flow classification techniques categorize the traffic flow into distinct classes to control and manage aggregated Internet traffic flows efficiently. Load balancing technique distributes the system load evenly in all paths and achieves efficient links utilization. FEC is used to recover from packet losses. Three routing policies are compared and evaluated by taking received bandwidth and delay constraints. MPLS based Multipath routing dispersion attains good throughput and less delay. In further research, Traffic engineering constraints and admission policies, QoS guaranteed path selection need to be focused along with MPLS multipath dispersion.

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