An Efficient Content based Routing Algorithm for a Unified Communication Cloud Network

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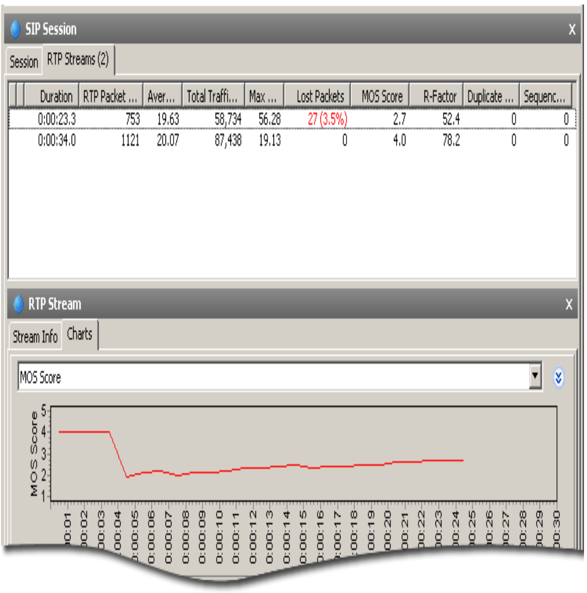
ABSTRACT
Unified communication consisting of Voice, Video and Data play an important role in present day internet based business communications. Quality of this communication is highly essential and even though this quality is directly depend on the bandwidth of the network, it get impacted by various other factors like protocol compliance of the end media devices, real time occurrence of delay at each router and packet drops. These factors are captured as metrics and clouds of database servers and processing servers are used to store these metrics to do real time monitoring of the quality of calls. In this paper few new quality metrics like buffer queue size allocated for ports and time delay at each router for routing are introduced and an efficient algorithm is developed which will use these new metrics to execute an effecting routing decisions for communication packets based on content type of these packets which will result in to an optimized way of storing the metrics values in the cloud database servers. Any efficiency in storing these metrics in cloud servers and real time quality evaluation will result into the efficient filling up of the quality of service(QOS) bits which can be used for optimized packet-drop free routing.

General Terms
Cloud Computing, Unified Communication, Routing.

Keywords
VoIP, Unified Communication, Content based routing, Queuing Theory, Database Servers, Processing Servers,

1. INTRODUCTION
Unified communication is a technology which is a combination of Voice over Internet(VoIP), Video communication and Data communication through an IP network. Since this technology removes the necessity of physical presence of team members for any meetings in same hall, this becomes a very high effective cost saving tool for business establishments. This technology is currently growing with internet and rapidly undergoing continuous changes in methodology and implementation.

VoIP and Video communication uses SIP(Session Initiation Protocol) and RTP(Real Time Protocol) for initial handshake and voice packet transmission respectively. SIP servers establishes the connection between caller and callee. Different CODECs(Coding and Decoding) like G711, G723 are used for compression and coding of the voice packets. VoIP quality is measured by the metric MOS(Mean Opinion Score) which vary between 1 and 5 where 1 stands for poor quality and 5 stand for best quality. Figure 1 shows a sample MOS graph. Various researches were done on calculation of MOS and optimization of the same [2], [16].



. Fig 1: MOS Graph

2. MOS FORMULA AND METRICS
MOS is calculated based on two or three metrics related to packet transmission. Specifically it is a function of jitter and packet loss where jitter is the delay in transmission of consecutive packets at destination compared its starting point and packet loss is defined as number of packets that doesn't reach the destination per second. Since MOS is a caller and callee based opinion metric, there are several formulas used by the service providers to assess the quality of VoIP calls. The benchmarking of the calls are also based on the CODECs that are used for communication. Figure 2 shows a monitoring tools that displays the metrics like jitter and packet loss.

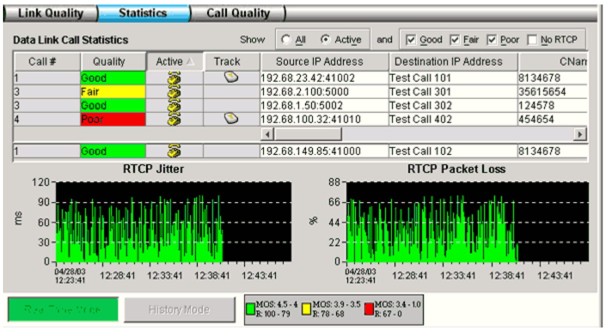


Fig 2: Monitoring tool displaying jitter and packet loss

The metrics like jitter and packet loss are continuously monitored by software tools by sniffing the VoIP traffic at source and destination. These metrics are collected on real time basis and stored in data base servers in cloud. A reporting software application that is running in a processing server calculates the MOS using these metrics and report it to the user for real time monitoring of the quality. This is either done by static pdf reports or through a monitoring application. Figure 3 shows various cloud and servers that are used for real time monitoring.

Various software companies providing VoIP monitoring developed good number of monitoring tools and specifically the Cisco[7] and Broadsoft [6].

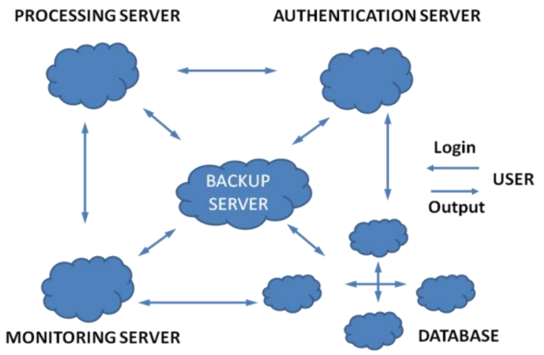


Fig 3: Servers in clouds for real time monitoring

3. CONTENT BASED ROUTING AND NEW METRICS FOR QUALITY

3.1 Buffer Allocation

Routers and Switches play a major role in routing these communication packets from source to destination. Each router has ports that receives these packets and send the packets to the destination. Thousands of packets are received and sent through these ports per second and large amount of memory is used as queues to store these packets for process. Queuing theory is used to manage these queues of packets to minimize the waiting time for every packet. An algorithm is introduced to optimally allocate the buffers to these queues to reduce the dropping of packets and this algorithm can be implemented in real time and the queue length can be managed dynamically so that the quality of the communication can be maintained at the benchmarked level. Figure 4 explains the basic functionality of the buffer in a router.

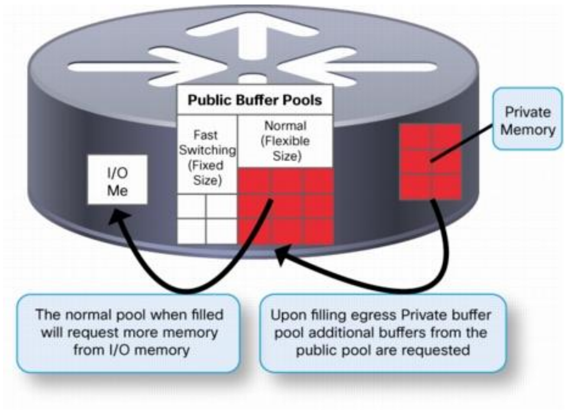


Fig 4: Routing buffer

There are various queuing theory techniques used [5, 8, 10] by routers to allocate buffers to the ports and either they may be of fixed or variable size or combination of both depend on router's operating system and Quality of Service(QoS) configuration.

3.2 Content Based Routing Algorithm

Compared to VoIP, the Video calls requires larger buffer size as 50 packets per second is for first and second requires four times of it for each input and output stream. Since both uses the same RTP protocol, we need a special check while handshaking is done to differentiate voice and video calls. Once the stream id is identified, the monitoring tool can track these traffic differently at source and destination and also at each port in the routers. The following algorithm is designed based on the fact that a video call from same destination of a voice call can be allocated a larger buffer to minimize the time delay in processing.

Let P1, P2, .., PN be the Ports with queue buffer size S1, S2,…, SN. Let F1, F2, …, FN be the size of the queues respecitively that get filled by packets

Step 1 Let P1 receives a voice call packets for forwarding. Let this call be called as C1. This is made between the source "SOURCE 1" to the destination "DESTINATION 1" Let this call packets uses the queue Q1 with size S1 without any drop in the packets.

Step 2. After a time "t" seconds, let the port P1 receives a video call packets for forwarding. Let this video call be called as V1. This is made between the source "SOURCE 2" to the destination "DESTINATION 2" Let this call packets uses the queue Q2 with size S2 without any drop in the packets at initial stage.

Step 3: Continuously check the value E2 = S2 – F2. It is observed that the queue Q2 gets filled by frequently and the value E2 reaches 0(zero) often. Hence there can be a drop of packets in the video call.

Step 4: Router find a port that manages and routes any voice calls, namely port P1 is found. Here any simple queuing theory models can be used to find a port that has more empty space queue for most of the time that can be used to reallocate it to the queue that are full most of the time.

Step 5: Find a port that has queue which has empty part. Here the Port P1 has queue with most part of the queue is empty as Voice packets are slow and number of packets is just 50 per sec compared to large number of packets per second for video. Allocate a part of memory buffer from port P1 to Port 2 and add it to the Queue Q2.

Step 6. Execute Step 2 to Step 5 continuously for all ports, by considering the nature of the packets like Voice or Video as one of the parameter which will give solution in shorter time than considering just the size of the queues alone.
Step 7. Also in Step 6 consider ports that are dealing with calls of same source or same destination as routing of such packets are going to be same for same source or destination which will also save time and in turn this will reduce the waiting time of packets in each port.
Step 8. Collect the values of free space available in queue buffer of each such ports together with total time delay or waiting time for each packet. Send these metrics to the database servers that are already storing the jitter and packet loss metrics.

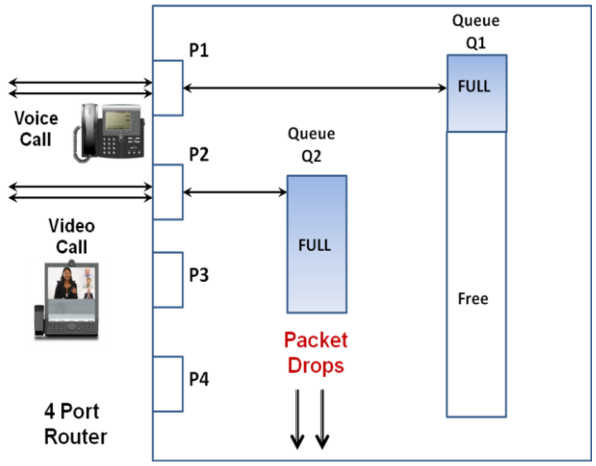


Fig 5. Packet drops as queue is full

Figure 5 and Figure 6 shows the steps of the algorithm.

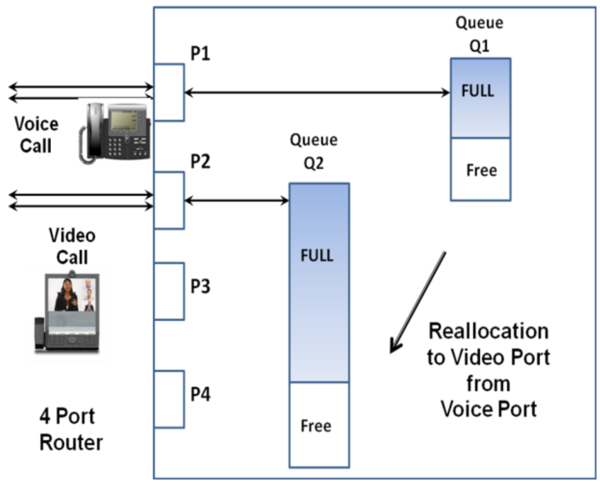


Fig 6. Queue get reallocated based on content of the traffic

This algorithm considers the nature of the communication packets that reaches the ports to decide on reallocation of buffers in addition to considering fullness and free part of the queues. This method will give a reallocation process which will last for long time as we try to pre-determine the flow speed at each port.
The following recursion formula is used to find the MOS value and this formula is referenced slightly differently in various research work and also varies while it is implemented by the service providers.

*Step1: Effective Loss = Average Loss + (Jitter*2) + 10*
Setp2: R = Average Loss - (Effective Loss/40)
*Step3: r = R - (Packet Loss*2.5)*
*Step4: MOS = 1 + 0.035*r + 0.000007 * r * (r-60)*(100-r)*

Figure 7 explains a simple relation between the packet loss and MOS for a 60 seconds call where the MOS is low whenever the packet loss is high together with the high jitter.

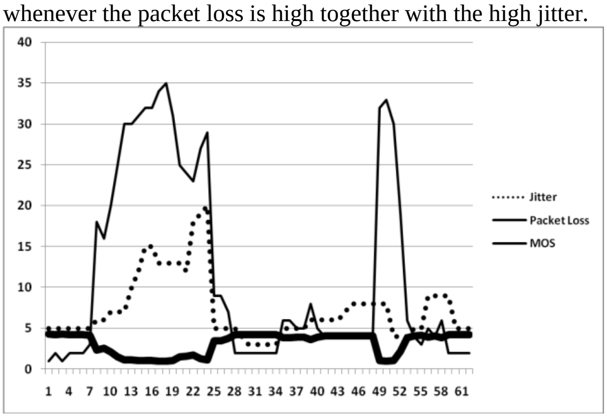


Fig 7. High packet loss and low MOS

4. CONCLUSION AND FURTHER AREA FOR RESEARCH

To increase the quality level of communication, we need to reduce the dropped packets. Primarily the packets are dropped because of the queues in ports of router get full often. In this project a new method is introduced to reallocate buffers of sufficient size to required queues based on the sources of the communication that get started. Priority is also considered for Video calls compared to Voice calls when they are made from same source and possibly to same destination. Also the metrics related to the free part of the queue buffer of each port together with the waiting time are to be measured and stored along with the jitter and packet loss. This will help to find the correlation between MOS value and delay at ports at a particular time.
Currently the computer network is working based on IPv4 protocol which uses 32 bit IP address. Technology is going towards the protocol IPv6 which uses 128 bit address where the packets sizes may also increase including the protocol header size. This concept will completely change the requirement of new algorithms in routers to handle with keeping the overall time delay that is going to introduced in network due to IPv6. This area is a promising domain for further research.
Also the concept of Router and Switches are merged together and present day network devices are having both routing and switching features where new algorithms can be developed for such devices.

5. ACKNOWLEDGMENTS

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