DEVELOPMENT AND ANALYSIS OF FAULT TOLERANT ALGORITHMS FOR SPANNING TREES OF A COMPUTER NETWORK

V. Sankaran[1], M. Yamuna[2] and A.Elakkiya[3]

[1]*R&D and Industrial Cell, Dhanalakshmi College of Engineering, Chennai (India)*

[2, 3]*School of Advanced Sciences, VIT University, Vellore, (India).*

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. In this paper we introduced few new techniques which can reduce the execution time in choosing a minimum cost spanning tree when a fault occurs. Different types of possible faults are listed and their corresponding minimum cost spanning trees are pre-generated and kept at all possible sources and destinations. Quick sort algorithm is suggested to sort the link costs which will take the minimum execution time. For a given fault, an optimum minimum cost spanning tree is chosen which will vary minimally from the minimum cost spanning tree of the fault-free network and this alternate tree is readily used to transmit packets in the occurrence of the fault. This technique will minimize the delay time when a fault occurs and this will improve the quality of the communication.

Keywords Broadcasting, Fault Tolerant Networks, Minimum Cost Spanning Trees, Multicasting, Unified Communication, VoIP

I INTRODUCTION

Unified communication is a technology which is a combination of Voice over Internet (VoIP) [1], 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) [2] 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.

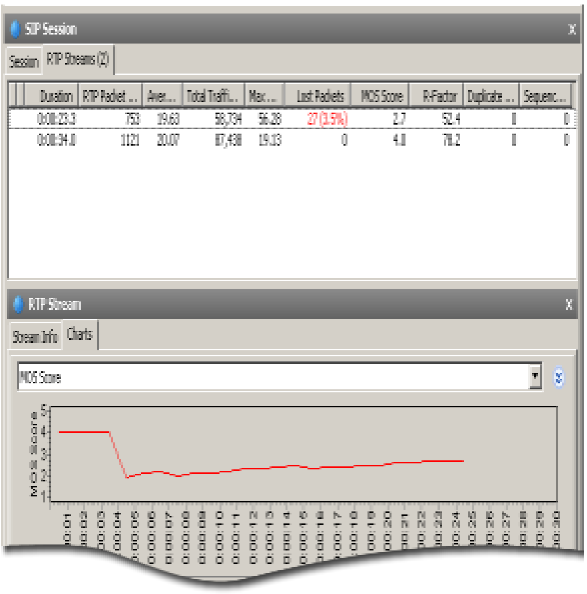


Fig 1: MOS Graph

II 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.

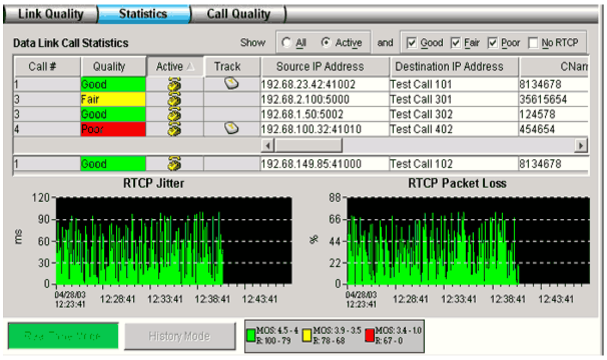


Fig 2: Monitoring tool displaying jitter and packet loss

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.

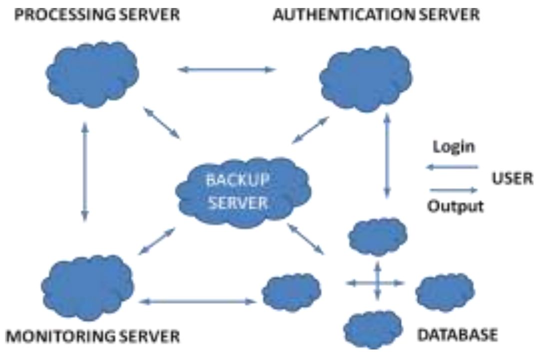


Fig 3: Servers in clouds for real time monitoring

III FAULTS AND GENERATION OF MINIMUM COST SPANNING TREE FOR BROADCASTING

3.1 Efficient Algorithm for a Fault Tolerant Network

Efficient broadcasting increase the overall quality metrics as it is described in previous section as the packet drops get reduced when the path is more free from the fault. In internet network, a broadcasting is done from single source to all other nodes in a selected sub-network and for this purpose the minimum cost spanning tree is derived in real time based on varying cost factors of a network. For much of the network the issue is not in varying cost but the occurrence of a fault which disconnects the spanning tree and hence the broadcasting message never reaches some part of the destination network. Since in broadcasting network acknowledgement is not defined, the loss of message becomes evident and the purpose of broadcasting gets defeated.

To avoid such scenario, whenever a fault occurs a new spanning tree is generated and used and the optimality is compromised for the speed. In [4] technique is suggested to minimize the impact of cost deviation on occurrence of a fault. It is given below:

Optimization Methodology

Step 1:

a. The broadcasting source S be a single node or multiple nodes which is used to broadcast the desired messages.
b. The source set S generates the Minimum Cost Spanning Tree (MCST) based on available facts with it. Let the MCST be denoted by "T" and its total cost be C(T) = C_T

Step 2:

a. With the use of prior events, a data base is created where this data base has all possible faults that may occur in near future. These faults have high probability of occurrence because of various factors.

b. For each Fault "F_k", a MCST "T_k" is pre-generated with cost "C_{Tk}".

Step 3:

a. A fault "F" occurs and this information reaches the source set "S".

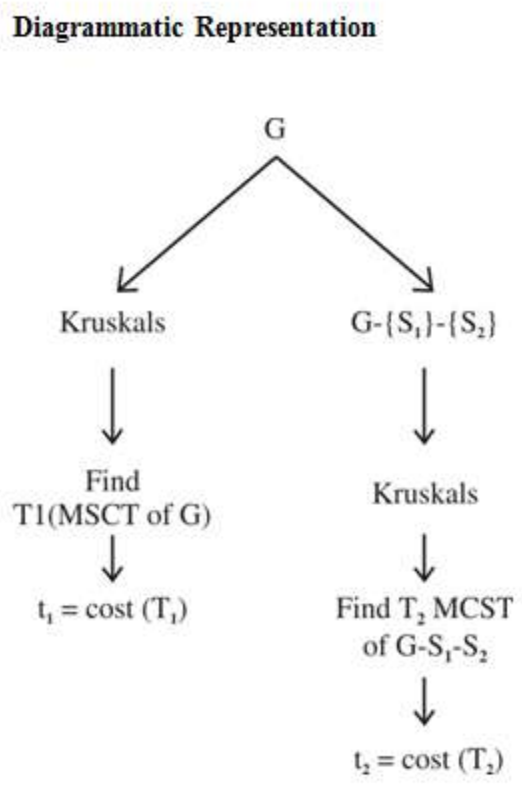
b. Instead of regenerating the MCST for this fault, the source pick the MCST from the database and uses it for broadcasting.

When the network is huge the time generating the MCST is very high and saving this time at the source will optimize the total cost for broadcasting. As in other algorithms in networking, this method is not error-free and it is a greedy solution, that is, gives an efficient algorithm which gives near optimum solution.

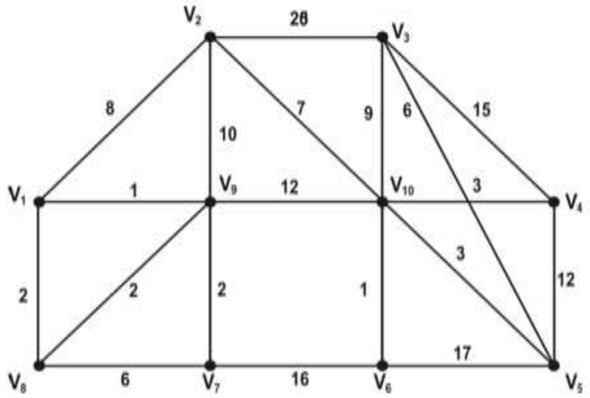
Also as explained at end of this section, the source S can choose a desired MCST based on "preferences set", a set of preferences of nodes and links that can either be most preferred or most disliked depends on various factors like cost of that node or link, vulnerability and time zones of the geographical location .

3.2 Quick Sort in Minimum cost Spanning Tree Algorithm

The following displays the algorithm at top level on sorting.



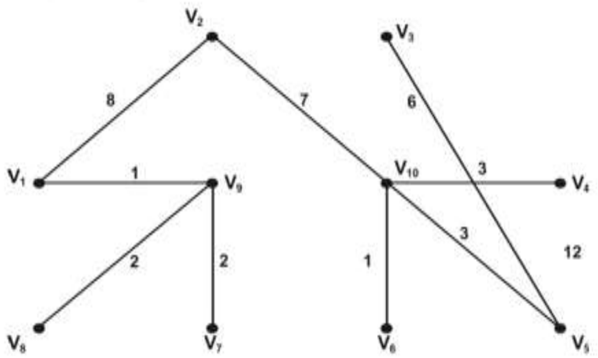
An example explained below will explain the algorithm.



Graph Network

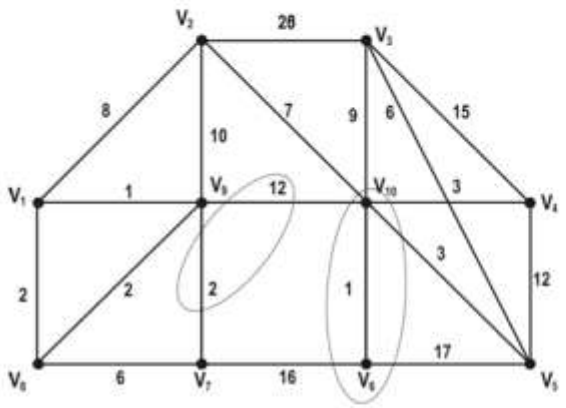
MCST is found for above network either with the use of Kruskal or Prim's algorithm and the MCST is given below:

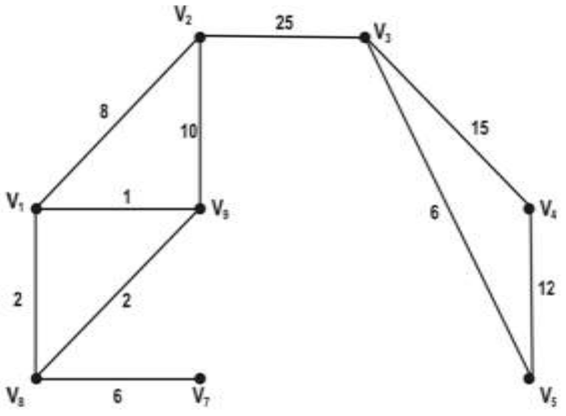
Minimum cost spanning tree without fault of G



Here the cost of minimum cost spanning tree is 33.
i.e., Cost (T_1) =33

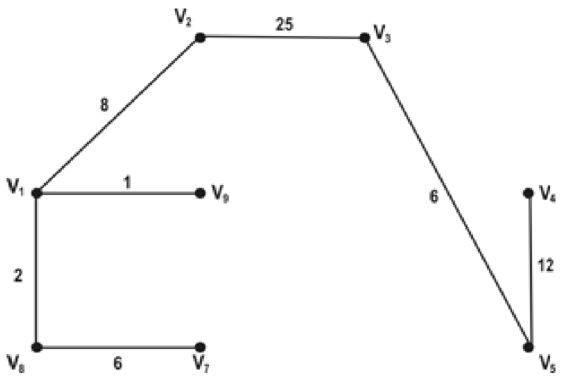
When the fault occur the resulted graph and spanning trees are shown below:





G-{S_1}-{S_2}

Minimum Cost Spanning Tree of G – {S_1} – {S_2}



Here the cost of minimum cost spanning tree of G-{S_1}-{S_2} is 60

Where S_1 and S_2 are the faults in G

i.e., Cost (T_2) = 60

$\Rightarrow C(T_1) < C(T_2)$

$\Rightarrow$ Increase in cost when fault occurs

The resultant MCST is considered for broadcasting by considering various preferences which can vary with respect to time.

Graph operations used in this paper are referred from the book [4] .

IV 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 paper we introduced an algorithm which can be applied for a network where the faults occur frequently and also the network is heavily used for broadcasting.

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.

REFERENCES

[1] http://www.ijcaonline.org/volume21/number9/pxc3873481.pdf

[2] http://advances.uniza.sk/index.php/AEEE/article/viewFile/542/732

[3] A. Elakkiya, Development and analysis of a fault tolerant algorithms for spanning tree of a computer networks, M. Phil diss., Auxilium College, Vellore, Tamil Nadu, (2012).

[4] Narsingh Deo, Graph Theory, PHI, 23rd Indian reprint (2002).