Performance Evaluation of VoIP using Shortest-Widest and Modified Widest-Shortest QoS Routing Algorithms

Ala F. Khalifeh, and Ali H. El-Mousa

***Abstract*— Implementation of current real time services (of which one of the more important is Voice over IP) on the existing Internet faces many obstacles; among them the issue of routing. Quality of Service (QoS) routing, attempts to provide real time services with the required guarantees to achieve acceptable performance. In this paper we study VoIP routing using the Quality of Service (QSR) network simulator. We investigate the Shortest-Widest routing algorithm and the Widest-Shortest routing algorithm that uses the hop-normalized metric to provide QoS for real time applications. We show that although both algorithms have superior performance compared with the conventional delay and hop based routing algorithms, the Widest-Shortest algorithm using a modified cost metric based on the hop-normalized metric is better able to route real time traffic away from congested links thus providing better performance to satisfy real time services requirements.**

***Index Terms*— hop-normalized metric, QoS in computer networks, routing algorithms, routing metrics, VoIP.**

I. Introduction

With the current Internet, in order to effectively implement real-time applications, such as Voice over IP (VoIP) and video streaming, some changes to its current structure need to take place. One of these enhancements is the use of new routing algorithms called, QoS routing algorithms, which take real-time traffic requirements into consideration when determining the routes of the packets [1] [2]. In the area of QoS, researchers have focused on developing new algorithms capable of meeting real-time traffic requirements on one side, and having low running complexity on the other [3]–[6].

Khanna and Zinky [7] studied and identified the problems associated with the Shortest Path First (SPF) algorithm which utilized the average delay as a routing metric. They showed that it caused traffic to concentrate on links satisfying the measured delay which forced all routers to attempt to use these links while ignoring other underutilized paths. They showed that this can cause route oscillations, network congestion spread, and inefficient use of the available bandwidth. They proposed a modified cost metric which they called the hop normalized. The idea was to use a metric that will favor network performance under heavy load to route traffic away from congested links. Khanna and Zinky tested the performance of their hop-normalized metric under best-effort traffic loads, which showed that using this metric with the SPF algorithm, reduced congestion, thus improving network efficiency and showed improved network utilization with low computation overhead. This type of traffic engineering (TE) problem which aims to dynamically redistribute traffic to enhance performance during routing has received a lot of attention from the research community [8] [10].

In this paper we extend our work done previously, on the Widest-Shortest (WS) algorithm, where we explored it using different routing metrics (delay, hops, and hop-normalized) [11]. We have utilized the results of the work of Khanna and Zinky [7] and applied their cost metric to the QoS based routing protocol WS. Thus, the WS algorithm will select the path which minimizes the hop-normalized metric among those that satisfy the bandwidth requirements. We showed that using the hop-normalized metric, the routing algorithm performed very well; efficiently managing to distribute traffic such that real-time flows avoided congested links. As a result, we obtained improved throughput, end-to-end delay and jitter. Here, we study the Shortest-Widest (SW) routing algorithm which was suggested to be used with the QoS Open Shortest Path First (OSPF) protocol as proposed by Wang and Crowcroft [12]. This algorithm selects the path with the largest available bandwidth. If several paths exist with as large a bandwidth, the one with the smallest hop count is selected. Our aim is to investigate this algorithm using the same network architecture used in [11] and using the same traffic flows. We used the same simulator and recorded the same network parameters. We then performed a comparison to see how both the WS using the hop-normalized metric and the SW performed under the same conditions.

The rest of the paper is organized as follows: Section II presents a brief technical background to the problem. Section III presents our approach to the problem and the network model we utilized for our simulations. Section IV presents and discusses the simulation results obtained. Section V is a conclusion and presents further ideas for future work.

Manuscript received March 18, 2007.
Ala F. Khalifeh is with the Electrical and Computer Engineering Department at the University of California-Irvine, CA USA (e-mail: akhalife@uci.edu).
Ali H. El-Mousa is with the Computer Engineering Department, Faculty of Engineering & Technology, University of Jordan, Amman, Jordan. (Phone: ++96265355000 ext. 23000, fax: ++96265355588, e-mail: elmousa@ju.edu.jo)

II. Technical Background

This section covers QoS metrics, real time traffic characteristics for various sources, and routing algorithms.

A. Quality of Service Metrics

In order to have a good VoIP service, it is extremely important that such a service meets some QoS regulations and standards such as: end-to-end delay (latency), packet loss, and delay variation called jitter delay.

1) *End-to-End Delay:* Delays below 150 ms are acceptable for most applications [13].

2) *Mean Opinion Score (MOS):* MOS is a subjective method of quality assessment [13]. Test subjects rank the voice quality using the following scale:

5 – Excellent, 4 – Good, 3 – Fair, 2 – Poor, 1 – Bad

Using this scale, an average score of 4 and above is considered as toll-quality.

3) *Packet Loss:* For real-time services such as VoIP the Mean Opinion Score is sharply affected by packet loss [13].

4) *Jitter Delay:* Jitter is introduced by variable transmission delay over the network. To reduce this variation, some buffering techniques at the receiver side are used.

B. Real-Time Traffic Characteristics

1) *Video Real-Time Traffic:* Bandwidth requirements for network multimedia applications can range anywhere from 100 Kbps to 70 or 100 Mbps [14].

2) *Real-Time QoS Service Models:* The approach that Differentiated Services use for resource allocation is to aggregate traffic rather than individual flows, so resources are allocated to individual classes. While in IntServ, resource reservation is based on per-flow behavior, so IntServ can provide better control on QoS than DiffServ [14]. Therefore, we adopted the IntServ architecture model.

3) *Voice over IP Traffic:* Studies of telephone users have demonstrated that the average talk-spurt is exponentially distributed and lasts between 0.4-1.2 seconds followed by an exponentially distributed silence period of 0.6-1.8 seconds in length [13]. We have used the same voice traffic generation model.

4) *VoIP Bandwidth Requirement:* In the simulation, the bandwidth of VoIP traffic is selected according to the rates standardized by the International Telecommunication Union (ITU) [13] shown in Table I.

Table I VoIP ITU recommendations

No.	Coding Standard	Compression Algorithm	Bit Rate Kbps
1	G.711	PCM	64
2	G.729	CS-ACElP	8
3	G.723.1	ACELP	5.3/6.4

C. QoS Routing Algorithms

Studies have shown that in order to support QoS requirements, current routing protocols need to consider more than one single metric, so that the routing algorithm will be able to find a path that satisfies multiple constraints [4].

Wang and Crowcroft [12] studied the complexity of taking more than one metric into consideration in the routing process and obtained the following result: To get a feasible and efficient QoS routing method, the chosen metrics should be orthogonal to each other to remove any redundant information between them.

According to the above rules, it is clear that any two or more of delay, delay jitter, hop-count, and loss probability in any combination as metrics are NP-complete. The only feasible combinations are bandwidth and one of the four (delay, loss probability, hop-count and delay jitter).

Currently, Best Effort (BE) traffic uses the shortest path algorithm, which depends on minimizing a certain cost function (metric) mainly the delay. Khanna and Zinky [7] found that the correlation between successive measured delays is high when a network is lightly loaded, but that the predictive value of measured delays declines sharply under heavy traffic loads. This leads to the problem that the delay shortest path algorithm will not select the shortest-delay path. To solve this problem, they proposed a modified metric called the Hop-Normalized metric. The main idea behind this metric is to normalize the link cost in terms of hops.

So when a link reports a cost, the cost is relative to the costs of alternate links. Thus when a link reports its cost, it is not done in an absolute fashion; rather it reports it relative to the cost of other alternate links. The hop-normalized cost is a function of the delay and not a delay. The reported cost values of the links will reflect the true image for the traffic load conditions and congestion of the whole network links. This way the congested links could be avoided. A detailed exposition to the hop-normalized is given in [7].

III. Our Approach

In this part, we will discuss the simulation model used and our approach in evaluating the performance of real-time traffic using the WS and the SW algorithms.

A. Simulation Models, Settings and Scenarios

We have implemented our simulation using the Quality of Service Routing Simulator (QSR) [15]. This is designed to study QoS routing mechanisms.

Both types of traffic sources, simple and real-time traffic loads are supported in our simulations. Many parameters of these workloads can be adjusted such as the transmitting rate and the period of duration of traffic production and pauses. Moreover, both packet size and starting time of traffic flows can be determined.

B. Network and Traffic Models

This section describes the architecture of the network used during simulations. Also, it presents the details of the traffic models utilized.

1) *Workloads and Traffic Sources:* The different traffic loads used in the simulation throughout the scenario are shown in Table II.

2) *VoIP Workloads:* Several VoIP workloads are used. The source/destination nodes are chosen such that there are many alternative routes between them (Fig. 1). Nodes 2, 5 and 6 are located on the edges of the network.

Table II Scenario traffic loads

No	Traffic Type	Source	Destination	Bit Rate (Kbps)	Talk Period (sec)	Silence Period (sec)	Arrival Time (ms)
1	VoIP	2	5	64	0.6	0.58	0.2
2	VoIP	2	5	64	0.9	0.77	0.15
3	VoIP	2	5	64	0.89	0.56	0.1
4	VoIP	2	5	64	0.57	0.85	0.5
5	VoIP	2	5	64	0.82	0.99	7.5
6	VoIP	2	4	64	0.43	0.58	0.8
7	Video Traffic	2	5	512	NA	NA	NA
8	HTTP Traffic	2	4	2048	NA	NA	NA
9	HTTP Traffic	1	6	512	NA	NA	NA
10	HTTP Traffic	7	5	512	NA	NA	NA
11	FTP Traffic	1	5	200	NA	NA	NA

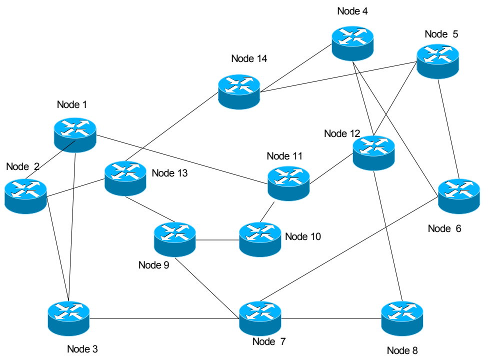

Fig. 1 NSFNET backbone network topology

This allows for many alternative routes and paths between these nodes. According to the ITU recommendation [13], the 64 kbps bit rate is chosen since this is the highest one that is supposed to give the highest quality.

3) *Video Traffic:* One video traffic load is chosen, between node 2 and node 5. These nodes are located at the edge of the network. The bit-rate used to simulate video traffic is 0.5 Mbps, which successfully simulates important applications [16] such as video conferencing or distance learning applications.

4) *HTTP Traffic Sources:* It is important to try to have maximum performance of real-time traffic with minimum impact on the BE traffic [17]. To study this aspect, three HTTP traffic loads are studied, the source/destination nodes for them are chosen such that they will cover many routes and occupy many links in the network. The bit-rate for source 8 is deliberately chosen higher than the bandwidth of the T1 links which might be part of its path between node 2 and node 4. Congestion can occur in that link and so its effect on real-time traffic loads can be studied.

5) *FTP Traffic:* We included also an FTP traffic source between nodes 1 and 5 as part of the background traffic used in the simulation.

IV. Simulation Results and Analysis

In studying the performance of VoIP and video traffic sources; the following performance measures are used: throughput at the receiving end, end-to-end delay, and jitter. Also, the Mean Opinion Score (MOS) is measured. The study cases are summarized in Table III.

Table III Study cases

Case No.	Description	Justification
Case 1	Real-time flows will use the WS routing algorithm using the hop-normalized metric as its cost metric.	To study the performance of real-time traffic using the hop normalized metric with the WS algorithm
Case 2	Real-time flows will use SW Path algorithm	To study the performance of real-time traffic using the SW path algorithm

For convenience, we have chosen to show the obtained results in detail for some of the traffic sources. Thus the results for two real-time sources are presented fully. We have chosen one of the five VoIP traffic sources, the one that is transmitted from node 2 to node 5 i.e. source number 1. Also, the video source between node 2 and node 5 is also selected for detailed study, i.e. source number 7.

To be able to compare between the WS and the SW algorithms, the results of the two cases are first plotted on the same graph. Later, detailed individual results and discussion are provided to explain the different aspects obtained. Full analysis in terms of performance graphs is done for the following real-time flows: VoIP source No. 1 and video source No. 7. Regarding BE traffic, the throughput of HTTP source No. 8 is shown.

A. Throughput

1) *Real-Time Traffic Loads:* Figs. 2-3 show the throughput obtained for VoIP traffic source No.1 and the video traffic source No. 7, respectively.

2) *Non Real-Time Traffic Loads:* Both real and non real-time traffic sources are examined, Fig. 4 shows the throughput for the best-effort traffic sources No. 8.

B. End-to-end Delay

The delay is another important issue that should be considered when evaluating the performance of real-time applications. Figs. 5-6 show the delay for VoIP and video sources

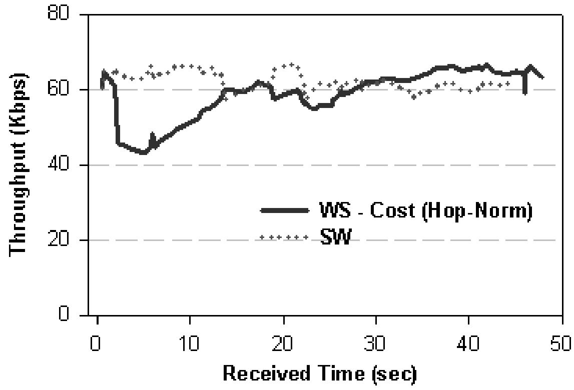

Fig. 2 Throughput for VoIP traffic source 1

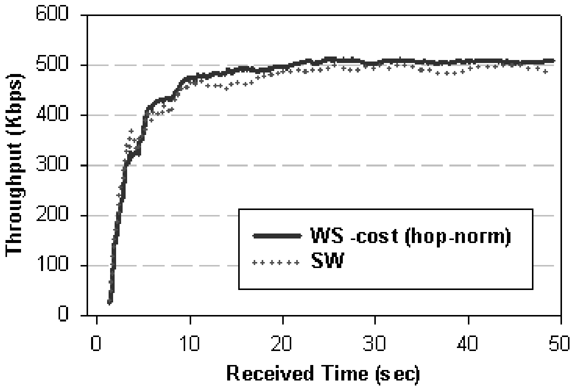

Fig. 3 Throughput for video traffic source 7

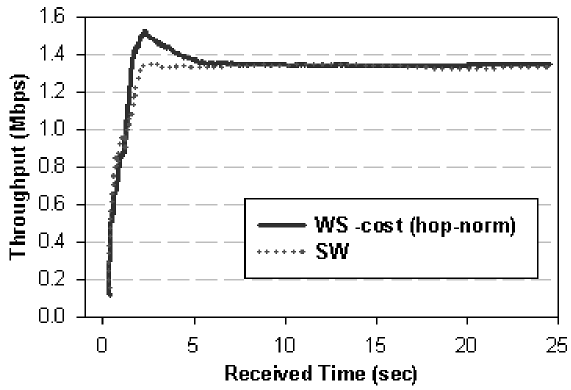

Fig. 4 Throughput for BE traffic source 8

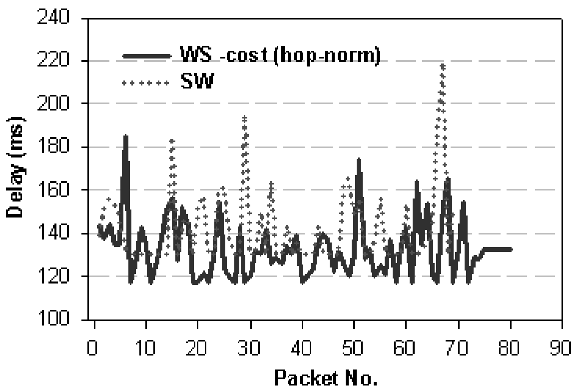

Fig. 5 End-to-End delay, VoIP source 1

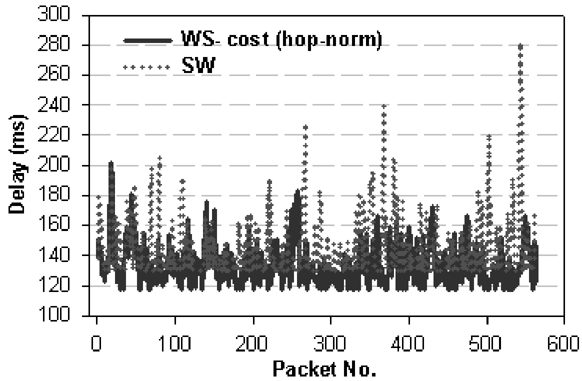

Fig. 6 End-to-End delay for video source 7

C. Jitter

Delay variation or what is called Jitter, significantly affects the quality of real-time services. Below, the jitter for the VoIP and video sources are shown through Figs. 7-8.

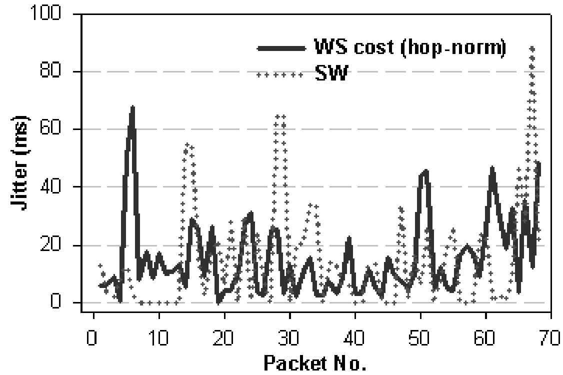

Fig. 7 Jitter for VoIP source 1

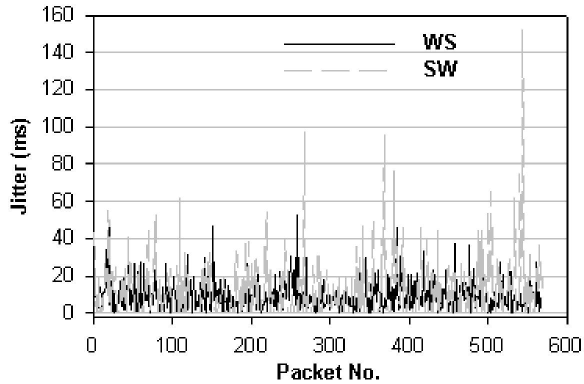

Fig. 8 Jitter for video source 7

D. Results Analysis and Discussions

Case 1: WS Algorithm Using Hop-Normalized Metric: This algorithm routes traffic flows away from congested links. So, in our case, the real-time traffic may be routed for example to link 2-13, link 13-9, link 9-10, link 10-11, link 11-12 and finally to either link 12-5 to reach node 5 or to link 12-4 to reach node 4. These routes are longer in terms of hop-count, but have higher bandwidth. Another suggested route which real-time flows may select is through link 2-3, link 3-7, link 7-6 and finally link 6-5. This route may be suitable to the real-time sources whose destination node is node 5. It is difficult to precisely predict how the routing occurs, but we note that there are many alternative routes, which may be different in terms of hop-count, and links bandwidth.

1) *Throughput Analysis:* The simulation results are very consistent with the operation of the algorithm using this metric. The throughput for BE traffic source 8 increases significantly to reach 1.4 Mbps, which is equal to 90.3% utilization of its link, which indicates that most of the real-time flows avoided the links that are used by the BE source No. 8 and used other links that do not suffer from congestion. Consequently, BE traffic throughput increases significantly. Regarding the throughput of real-time sources, table IV shows the average value for the obtained bandwidth and the performance evaluation for each source.

Table IV Throughput and Evaluation, Case 1

Source No.	Requested BW Kbps	Obtained BW Kbps	Packet lost%	MOS out of 5	Performance
1	64	59.2	7.5 %	4.25	Good to Excellent
6	64	57.8	9.3 %	4.1	Good
7	512	457.5	10.6 %	3.9	Good

2) *End-to-End Delay Analysis:* The end-to-end delay for the real-time flows decreases to a value below 150 ms, which is very acceptable and will result in very good quality as seen in Figs. 5-6. The average delay for VoIP source No.1 is 132.9 ms, and for the source No. 6 is 133 ms and finally for the video traffic source is 133.8 ms. It is noted that in some packets, for example in packet number 12 for VoIP source No.1, the delay may reach up to 200 ms. For these kinds of packets, it is better to discard them than delay the rest of the packets. This is usually done at the receiver's buffer.

3) *Jitter Analysis:* The jitter delay is uniform and low. For example, as we can see in Fig. 7 for source no. 1, the jitter delay encountered by almost two thirds of the packets lies within the 10 ms region which is quite small. Also, generally most of jitter has the same values and these are very close, consequently, de-jittering can be successfully implemented to remove these delay variations on the received packets. The jitter delay encountered by the video traffic is higher than 10 ms, as shown in Fig. 8. However, most of these packets encounter a delay within 20 ms, and since the end-to-end delay is about 133.8 ms, which is lower than 150 ms, de-jittering techniques can be used successfully to remove these variations.

Case 2: Shortest-Widest Path Algorithm: This algorithm selects the path with the largest available bandwidth. If several paths exist with as large a bandwidth, the one with the smallest hop-count is chosen. This algorithm is also attractive and efficient, in our case, since link 2-3 has higher bandwidth than the other two links link 1-2 and link 2-13, the routing algorithm selects link 2-13 which is the "widest" in terms of the available bandwidth, then at the next node; node 13, this algorithm selects link 13-9, which has 3 Mbps, which is greater than the 1.5 Mbps of link 13-14. So, the real-time traffic will not share the 1.5 Mbps link connecting nodes 13 with node 14. That leads to higher throughput for the best-effort traffic. Again the algorithm is repeated until it reaches the destination node, the selected path is link 2-13, link 13-9, link 9-10, link 10-11, link 11-12 and finally link 12-5 or link 12-4 depending on the traffic destination.

1) *Throughput Analysis:* Fig. 4 shows the throughput for best-effort traffic source No.8. It is obvious that the throughput for this source reaches up to 1.4 Mbps, which is equal to 90.3 % utilization for the shortest path link between node 2 and node 4. Moreover, the average throughput for the real-time sources is very near to the required bandwidth. Table V shows average values for the obtained bandwidth and the performance evaluation for each source.

Table V Throughput Evaluations, Case 2

Source No.	Requested BW Kbps	Obtained BW Kbps	Packet lost%	MOS out of 5	Performance
1	64	61.9	3.2 %	4.7	Excellent to Good
6	64	60.18	5.9 %	4.1	Good
7	512	457.5	7.08 %	4.3	Good to Excellent

2) *End-to-End Delay Analysis*: Since the real-time traffic sources avoided using the congested links, the end-to-end delay encountered by these sources was acceptable and within the recommended bound. Figs. 5-6 show how the delay for this algorithm is near to the delay curves for case 1. Overall, the curves show that the delay produced by the WS algorithm is less. In this case, the average delay for VoIP source No. 1 is 140 ms, and for the video source it is 143.5 ms. All these values are below the recommended 150 ms, which results in acceptable delay and performance.

3) *Jitter Analysis:* Figs. 7-8 show that the jitter delay is within the range of 20 ms for VoIP source No.1. For the video traffic it is within the range of 20 ms. In fact, it is noticeable that the jitter values are almost the same, and the variation in the delay is not large. Some packets encounter high variation in the delay, usually, such packets are ignored at the receiver side.

E. Comparative Analysis between WS Using the Hop-Normalized Metric and SW

Careful examination of the results obtained for both algorithms as shown in Figs. 2-8 reveal the following:

- Regarding throughput, both algorithms provided good to excellent performance with the SW algorithm providing slightly better performance than WS as demonstrated in Fig. 4 for best effort sources 8.
- Figs. 5-6 demonstrate that the WS algorithms provided significant improvement in end-to-end delay for all cases studied as compared to SW. As mentioned before, this is quite important for real-time traffic.
- Regarding jitter, again the WS algorithm produces significantly less jitter than SW. This is rather important as this allows efficient use of de-jittering techniques.

The above seems to suggest that the WS utilizing the hop-normalized metric is a better choice since it has superior overall QoS performance. We decided to further investigate the throughput aspect of performance.

F. Shortest-Widest Path Draw Back

From the above results, the performance of real-time services, using the SW algorithm was good and attractive. But this algorithm may not always avoid the congested links. In our work, some links were set in a way that their bandwidths were higher than other links, so the congested links that carry the best-effort-traffic source No. 8, were avoided using this algorithm. But suppose that all of the links' bandwidths were similar, i.e. 1.554 Mbps, and suppose that we have one of the real-time flows that want to transfer traffic from node 2 to node 5. In this case, the algorithm will check first the three outgoing links connected to node 2, i.e. link 2-13, link 2-1, and link 2-3. Since all of the three links have the same bandwidth, it will choose the node which leads to the shortest path between node 2 and node 5, so it will choose node 13, and the algorithm will repeat again. Accordingly, the shortest path between node 2 and node 5 is chosen for that real-time source, and it will be routed to that path. To investigate this plausible scenario, we have repeated the simulation using the same bandwidth setting for all the links. We set all the links bandwidth to 1.554 Mbps and we made the video source, which has a bit-rate of 0.5 Mbps, request a path between node 2 and node 5. We kept the same best-effort background sources. We used the SW algorithm and the WS algorithm with the hop-normalized metric. The throughput of the best-effort traffic source No. 8 was monitored. The results obtained are shown in Fig. 9.

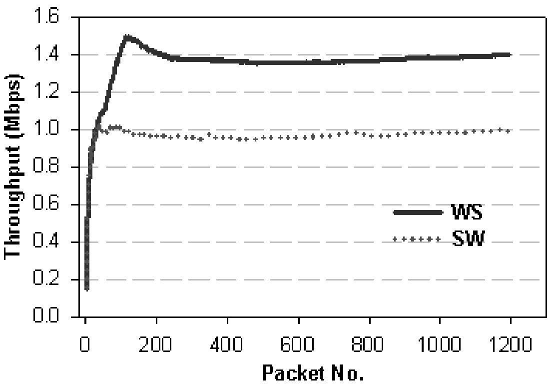

Fig. 9 Throughput for BE traffic source no. 8

From Fig. 9, the throughput of the best-effort traffic is around 1 Mbps when utilizing SW which proves that the video traffic did reserve 0.5 Mbps along that path so the best-effort traffic utilized the residual bandwidth. However, when the WS algorithm is used, the congested link that carried the best-effort traffic between node 2 and node 8 was avoided and the video traffic was routed using other links. Therefore, the best-effort traffic utilized most of the available link bandwidth (about 1.4 Mbps). This demonstrates that the WS with the hop-normalized is better equipped to handle congested links which arise under heavy loads.

V. CONCLUSION

QoS routing is considered one of the important factors that affect the end-to-end delay of real-time streaming applications such as VoIP. To have an efficient algorithm, the routing metrics used in determining the best path should reflect the traffic congestion on the outgoing links, so the algorithm can avoid routing real-time traffic to these congested links. Both the WS with the hop-normalized metric and the SW were tested for performance in a loaded network. Both algorithms exhibited very good performance but we have found that when using the WS algorithm that uses the hop-normalized metric, the system was better able to avoid the congested links in the routing decision. Also, the WS showed better end-to end delay, lower jitter, and high throughput for real-time flows.

For future work, we are looking to extend our simulation to more complex networks using a wider range of traffic sources including bursty ones. Also, we will examine the performance of the WS modified algorithm under the Diffserv architecture.

REFERENCES

[1] Bo Li, Mounir Hamdi, Y. T. Hou, Dongyi Jiang, and Xi-Ren Cao, "QoS enabled voice support in the next generation internet: issues, existing approaches and challenges", *IEEE Communications Magazine,* vol. 38, 2000, pp. 54-61.

[2] James. F. Durkin, *Voice-Enabling the Data Network: H:323. MGCP, SIP, QOS, SLAs, and security*, 1st ed., Cisco Press, USA, 2003.

[3] Douglas S. Reeves and Hussein F. Salama, "A Distributed algorithm for delay-constrained unicast routing", *IEEE/ACM Transactions on Networking*, vol. 8 issue 2, 2000, pp. 239-250.

[4] Turgay Korkmaz and Marwan Krunz, "Routing multimedia traffic with QoS guarantees", *IEEE Transactions on Multimedia*, vol. 5 No. 3, 2003, pp. 429-443.

[5] Fernando Kuipers, Turgay Korkmaz, Marwan Krunz, and Piet Van Mieghem, "Overview of constraint-based path selection algorithms for QoS routing", *IEEE Communications Magazine*, vol. 40 issue 12, 2002, pp. 50-55.

[6] Fernando Kuipers, Turgay Korkmaz, Marwan Krunz, and Piet Van Mieghem, "Performance evaluation of constraint-based path selection algorithms", *IEEE Network*, vol. 18 issue 5, 2004, pp. 16-23.

[7] Atul Khanna, and John Zinky, "The revised ARPANET routing metric, *ACM SIGCOMM Computer Communication Review*, vol. 19 issue 4, 1989, pp. 45-56.

[8] Jeon Hahnearl; Kim Sung-Dae; Kim Young-Joon; Kim Hyung-Taek; and Lee Jai-Yong, "Performance analysis of QoS routing with network dependant delay cost"; *Proceedings 15th International Conference on Information Networking,* 2001, pp. 689-693.

[9] Antoine B. Bagula, "Hybrid routing in next generation IP networks", *Computer Communications*, vol. 29, issue 7, 2006, pp. 879-892.

[10] Yanxing Zheng, Turgay Korkmaz, Wenhua Dou, and Jing Tian, "Highly responsive and efficient QoS routing using pre- and on-demand computations along with a new normal measure" *Computer Networks*, vol. 50, 2006, pp. 3743-3762.

[11] Ala. Khalifeh, and Ali. El-Mousa, "QoS routing of VoIP using a modified widest-shortest routing algorithm", *Proceedings of the ACS/IEEE International Conference on Computer Systems and Applications (AICCSA '2007)*, to be published, May 2007.

[12] Z. Wang, and J. Crowcroft, "Quality of service routing for supporting multimedia applications, *IEEE journal on selected areas in communications*, vol. 14 issue 7, Sept. 1996, pp. 1228-1234.

[13] Hassan Mahbub, Nayandoro Alfandika, and Mohammed Atiquzzaman, "Internet telephony: services, technical challenges, and products", *IEEE Communications Magazine*, vol. 38, Issue 4, 2000, pp. 96-103.

[14] Cisco Systems, (2006). Designing Internetworks for Multimedia [Online]. Available: http://www.cisco.com/univercd/cc/td/doc/cisintwk/idg4/nd2013.htm,.

[15] P. Zhang , and R. Kantola, *QoS Routing Simulator: User's Manual*, Version 2.1, 2002.

[16] M. Baldi, and Y. Ofek, "End-to-End delay of videoconferencing over packet switched networks", *Proceedings of the Conference on Computer Communications (IEEE Infocom)*, Vol. 3, March 1998, pp. 1084 - 1092

[17] S. Chen , and K. Nahrstedt , "An Overview of Quality of Service Routing for Next-Generation High-Speed Networks: Problems and Solutions", *IEEE Network,* vol. 12, issue 6, Nov./Dec. 1998, pp.64–79.

ISBN:978-988-98671-5-7