UTILIZING DIFFSERV AND SIP CONTACT HEADER FOR REAL-TIME FAX TRAFFIC ENGINEERING

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Abstract

This paper focuses on the transmission of real-time fax in IP networks. For this purpose, the best current practices, i.e., utilization of the Session Initiation Protocol (SIP) as the signaling protocol along with ITU-T T.38 recommendation, are adopted. Two traffic engineering measures: utilization of SIP contact header and DiffServ QoS architecture, are proposed to streamline the implementation of the real-time FoIP. Network simulation results show that the proposed architecture, compared to the "Best Effort" service, has much less transmission time and jitter, and packets received are in correct sequence. Moreover, SIP contact header reduces load on network's specialized resources. Therefore, this layout is a viable alternative for traditional PSTN fax.

***Keywords**: Traffic Engineering, Real-time Fax, Fax over IP, Session Initiation Protocol, QoS, Differentiated Services.*

1. Introduction

"Voice over IP" and "Fax over IP" are gradually becoming cutting-edge technologies in the telecommunications industry [1], [2]. These are meant to replace the traditional method of delivery of telecommunications services by *public switched telephone network* (PSTN) through utilization of data networks, e.g. the Internet. But, current data networks have not been designed with telecommunications services in mind. They have been optimized to carry data, which are bursty in nature. This is in obvious contradiction to the requirements of the telecommunications services. Specifically, they require a network infrastructure which is either connection-oriented in nature or at least can resemble its behaviors; therefore, it can guarantee a stream of data, free of any kind of interruption.

There are two possible approaches to accomplish faxing over the Internet protocol: Real-time and Store-and-Forward. Store-and-Forward or non-real-time usually uses E-mail capabilities to transfer fax between the end-points. In real-time approach, as the name suggests, fax is transferred in real-time manner and without delay. Real-time approach is the ultimate goal since it is the real-time faxing which makes the transition from the PSTN to the Internet-based architecture smooth. But real-time fax transmission presents a major implementation challenge in the new packet networks and that is mainly due to far more signaling, compared to voice, that is needed for end-points to negotiate fax parameters details.

In order to insure the success of VoIP networks, we need to provide support for legacy fax equipments which are attached to PSTN. The IP network should appear transparent to these equipments. For this to happen, fax gateways have been designed to translate signaling and data between fax machines in the PSTN on one side, and on the other side, the other party's fax gateway in the IP network. Signaling in V/FoIP architecture comprises session initiation, management and tear-down. Currently, there are two protocols that can provide an end-to-end solution: H.323 and Session Initiation Protocol (SIP). SIP has been adopted as the signaling protocol for real-time fax transmission in this research for reasons that will be discussed in the coming sections.

Real-time fax transmission seems more in need of high QoS compared to any other real-time application in the Internet and that is due to the large amount of signaling packets that need to be exchanged with tight time constraints. We adopt DiffServ architecture in our research for meeting the stringent QoS requirements. We also propose a measure to lessen the burden of specialized resources of the network: utilization of the SIP contact header. How this protocol header is used and what benefits it has, will be explained in Section 4. Computer simulations are utilized to study the different aspects of real-time fax transmission. Important parameters involved such as throughput, TCP congestion window of fax packets stream and received packets sequence numbers are studied.

The rest of the paper has the following organization: Section 2 briefly introduces the fundamental concepts of IP Telephony and the IP QoS architectures. In Section 3, our simulator and its developed modules and the specific simulation scenario chosen in this research are discussed. Simulation results and our final remarks are presented in Section 4 and Section 5, respectively.

2. IP Telephony Protocols & QoS Architectures

Currently, there are four signaling protocols which have grabbed the most attention in the IP Telephony industry: Media Gateway Control Protocol (MGCP), H.248/Megaco, H.323, and Session Initiation Protocol (SIP). In what follows, we give a very brief overview of these protocols.

MGCP and H.248/Megaco protocols adopt a centralized architecture. H.323 and SIP protocols, on the other hand, allow network intelligence to be distributed between endpoints and call-control devices. The call-control devices are called gatekeepers in an H.323 network, and proxy/redirect/register servers in an SIP network. The endpoint can be any device that can initiate and terminate a V/FoIP call. Of H.323 and SIP, this is the SIP that is capable of supporting user mobility by proxying and redirecting requests to the user's current location. Hence, SIP is the protocol of choice in this work.

For fax transmission, there are ITU-T Recommendations T.37 [3] and T.38 [4]; T.38 defines real-time procedures for fax support over IP networks. The legacy Group 3 fax devices attach to T.38 gateways and execute the T.30 protocol, the protocol that defines the procedures for the transmission of facsimile over the PSTN [5], in real-time. The T.38 protocol emulates the handshake procedures of the T.30 protocol on the IP network side and therefore produces the feeling of real-time fax. ITU also allows either TCP or UDP to be used as the transport protocol, as specified by Internet Fax Protocol (IFP).

Session Initiation Protocol (SIP) is an application-layer signaling protocol for creating, modifying, and terminating sessions with one or more participants [6]. SIP defines how sessions should be managed without regard to the session type and this implies the applicability of this protocol to numerous applications such as voice and fax. SIP can use either TCP or UDP as the transport layer protocol.

Up to four main SIP components may be utilized in any SIP session: 1- SIP User Agent (UA) that is the end-user device for creation and management of an SIP session. 2- SIP Registrar Server which contains the location of all User Agents within a domain. 3- SIP Proxy Server which accepts session requests made by an SIP UA, it then queries the SIP Registrar Server to obtain the recipient UA's address. The session invitation is then forwarded to the recipient UA directly if it is located in the domain covered by the Proxy Server or to the network in which the UA resides. 4- SIP Redirect Server gives the SIP Proxy Server the functionality to direct SIP session invitations to the other domains.

The SIP messages are either SIP Requests (Methods) or SIP Responses. The INVITE, REGISTER, BYE, ACK, CANCEL, and OPTIONS methods are the original six methods. An SIP response is a message created by an SIP Server/UA Server for replying to a request which a UA Client creates and falls into one of six possible SIP Response Classes. There are four types of SIP headers: general, request, response, and entity. The set of general headers includes all of the required headers in an SIP message, including the utilized Contact Header. We refer the reader to References [6] and [7] for all of the details of the SIP protocol.

Reference [8] introduces best current practices for SIP T.38 fax and SIP fax pass-through sessions. As will be described in the coming section, this research has not taken into consideration all the issues discussed in this Internet Draft due to the different focus of this project; nevertheless, Reference [8] provides a valuable starting-point for full implementation of real-time fax using SIP.

Various approaches exist for providing QoS in IP networks; one of which is the Integrated Services (IntServ) [9] and another one is Differentiated Services (DiffServ) [10]. IntServ with Resource ReSerVation Protocol (RSVP) is not recommended for use in IP Telephony because as the number of calls increases, it results in the stressing of the system.

The Internet Engineering Task Force (IETF) has developed a less complicated architecture to support DiffServ, which can scale well by aggregating traffic into classifications. The DS field in IPv4 and IPv6 is then utilized for conveying these classifications. As a result, only at network boundaries, we have to implement means for sophisticated classification, marking, policing, and shaping operations. As the packets enter a DiffServ domain, based on the DS field value, they receive a particular Per-Hop Behavior (PHB).

"Expedited Forwarding" (EF) is one of the PHBs and is suitable for real-time applications with the objective of providing a low-loss, low-latency, low-jitter, and assured bandwidth service. Another PHB, is "Assured Forwarding" (AF) which is further divided into other classes and provides a service with less quality compared to EF but a better one, compared to Internet "Best Effort". For more information regarding DiffServ, the References [11], [12], and [13] are recommended.

Multiprotocol Label Switching (MPLS) architecture is also an emerging QoS architecture which offers IP networks, the capability to provide differentiated services. MPLS separates routing from forwarding, using label swapping as the forwarding mechanism. Reference [14] provides more information regarding MPLS and IETF RFCs 3471-4 define different aspects of Generalized Multiprotocol Label Switching (GMPLS).

Authors have chosen DiffServ among these architectures in this research due to its relative maturity. We also believe that other emerging architectures also provide good results but do not enjoy the extensive work that has gone to the development of DiffServ.

3. Simulation Details

J-Sim [15], a powerful Java-based network simulation tool, is utilized throughout the implementation part of this research to explore the behavior of the developed protocol and components. In total, six modules have been developed and added to the simulator, brief descriptions of which will follow. SDPMessage, which is a class implementing SDP headers; SIPMessage, which is a class implementing SIP headers and also embeds an instance of SDPMessage in itself if the body type indicates so; SipPS, which is a class implementing an SIP proxy server; SipUA, which is a class implementing an SIP user agent and finally T38Receiver and T38Sender which act on behalf of real T.38 modules.

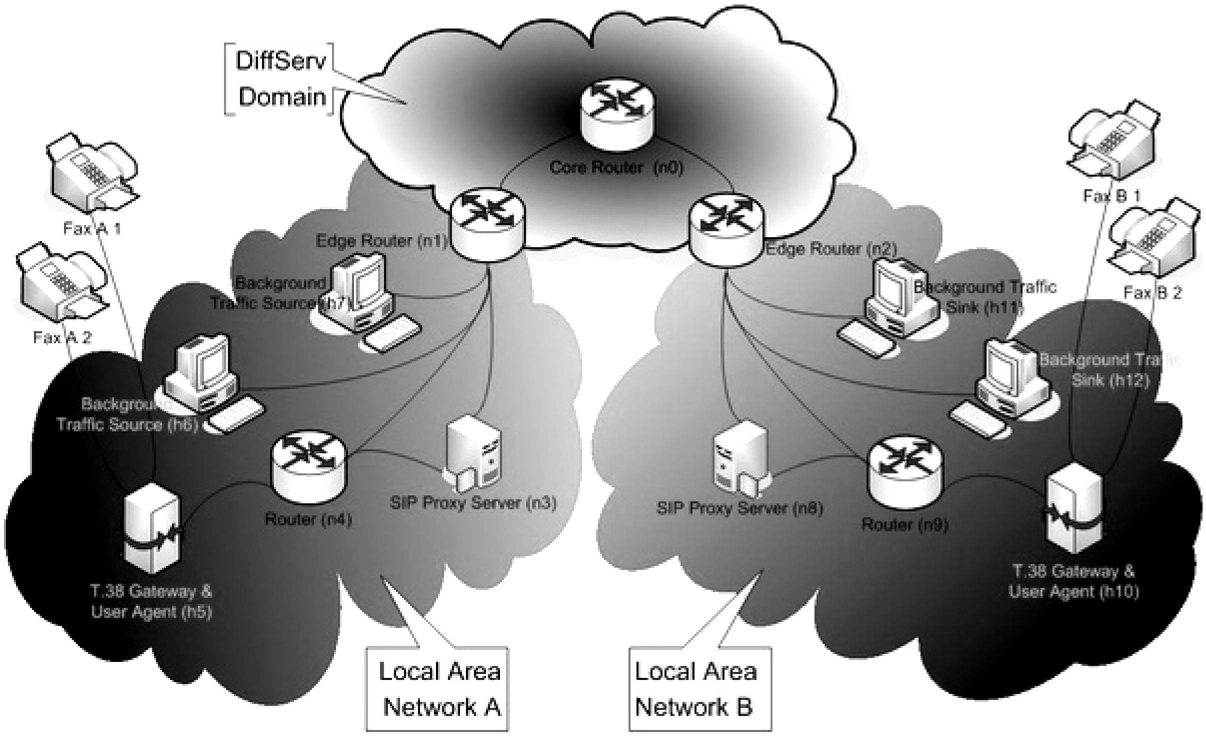

Figure 1. Network simulation scenario

We now start explaining the different aspects of the particular simulation scenario chosen in this research, the schematic diagram of which is depicted in Fig. 1. Every component of the topology is created and configured by a TCL script interface, a mechanism by which simulation scenarios are constructed and configured in this simulator. In what follows, we explain these details very briefly.

All the links (hops) present in the topology are set to have 100 ms propagation delay and 2 Mb/s capacity. Since the end-to-end delays are not measured in this simulation, the exact lengths of the links, and hence the propagation delays, are of no concern. The fax-related part of our simulation is the setting up and configuration of the T.38 modules in hosts which accommodate SIP user agents as well. The T.30 fax signaling between fax machines located in PSTN and the T.38 gateways, which are co-located with SIP user agents, are not simulated here; hence the fax calls are initiated from within the T.38 modules. As soon as the call is established between SIP user agents, the T.38 modules start the process of fax handshaking. Thereafter, they continue with fax image data transfer upon successful fax parameters details negotiation. Reference [8] can be consulted for the full implementation of SIP-T.38 interactions procedures.

We choose UDP transport for carrying SIP signaling as this is the default transport layer protocol used with SIP. On the other hand, TCP is used for fax packets transfer for better monitoring the effects of different parameters, details of which are provided in the next section. Two constant-bit-rate (CBR) background traffic sources/sinks are incorporated in the topology, which are h6-h11 and h7-h12, as shown in Fig. 1. The source rate of the former peers is almost 1 Mb/s and is twice that of the latter ones. The rationale behind such a setting will be clear in the coming section.

As depicted in Fig. 1, our topology roughly splits into three sections. It consists of two LANs and a DiffServ domain. We assume that the user agents involved are in these LANs which connect to the DiffServ domain using the domain's edge routers. The DiffServ domain magnitude and the number of components in it do not make any difference to our simulation and it can be as large as the Internet itself. One important point is that all the routers inside the domain behave exactly as our modeled core router, given that they have been pre-configured to do so. Core routers incorporate queuing to actually differentiate between the three classes of traffic, i.e., the ones that are tagged as "Expedited Forwarding-EF", "Assured Forwarding-AF", and "Best-Effort-BE". In our simulation, connection h5-h10 (user agents) is tagged as "EF", h6-h11 (Background traffic 1) as "BE", and h7-h12 (Background traffic 2) as "AF", this is done at the edge router n1. We also point out that we incorporated a 1 Mb/s bottleneck, as opposed to 2 Mb/s of other components, in the core router. This is deliberately done to simulate congestion and see the actual effect of preferential de-queuing at DiffServ domain core routers.

The simulated call flow can be understood using Fig. 2. Simulation starts by background traffic source h6. Just after that, h5, the SIP user agent, sends an INVITE to the SIP address of the other user. The rest of the call flow can be tracked using Fig. 2. The second background traffic source, h7, starts sending its packets right in the middle of the fax packets

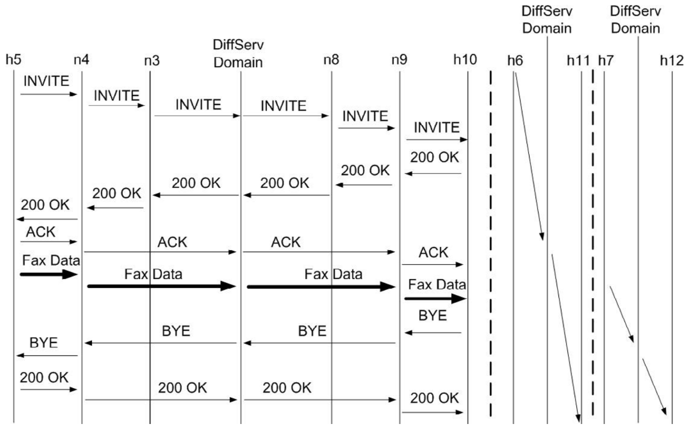

Figure 2. Simulated call flow

transfer with a DiffServ forwarding priority of "Assured Forwarding" which is higher than that of the first source but lower than the "Expedited Forwarding" priority of the fax packets. This is set so to analyze the effect of sudden over-congestion of the network on fax transfer. We point out that in our simulation, fax packets do not visit the proxy servers. This is a natural behavior, since proxy servers are highly specialized resources and we do not want to unnecessarily overload them. Hence, proxy servers will be able to handle more calls in the network.

4. Simulation Results

With the help of Network Animator (NAM) [16], packets traces can be gathered and analyzed after the simulation is finished. A snapshot of our simulation scenario analyzed using this program is provided in Fig. 3. The congestion in node 0 (core router) is visualized using the long upward buffer of the router.

4.1. SIP Contact Header Utilization

A measure that is proposed in this paper to lessen the burden of proxy servers is the utilization of the *SIP contact header.* In a normal situation and without regard to the capability provided by contact header, all SIP signaling go through the proxy servers. This is apparently unnecessary and can be avoided as soon as the SIP address of the other party is resolved into an IP address, after which, the user agent is able to contact the other party directly, bypassing the proxy servers. This happens when the initiator of the call receives its first SIP response and extracts the IP address of the other party from the contact header of this response. The call flow in Fig. 2 has already adopted this strategy.

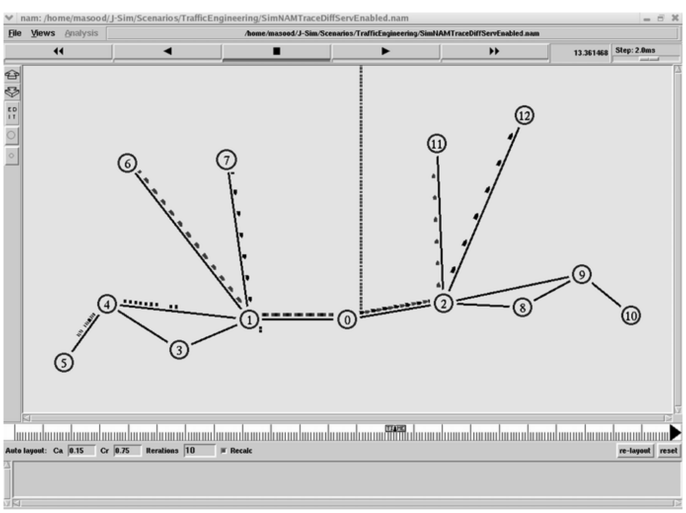

Figure 3. Network Animator (NAM) snapshot

4.2. Utilizing DiffServ for Providing QoS for Real-time Fax

We explained the tight timings associated with real-time fax message exchanges earlier. Here, we discuss and present our experiment results deriving from utilization of DiffServ architecture for real-time fax transfer. As a result of the inspection of a series of comparative diagrams, we deduce that utilization of DiffServ architecture is a viable solution for providing QoS in IP networks and is especially useful for real-time fax and its timing requirements. In what follows, we present the diagrams characterizing various traffic parameters and briefly explain the importance of each of them.

4.2.1 Fax traffic throughput. In Fig. 4, fax traffic throughput in both modes, fax traffic tagged as "Expedited Forwarding" and as "Best Effort", are depicted. One readily notices the huge difference in the duration of the fax transmission between the two modes. Fax transmission duration is five times more in

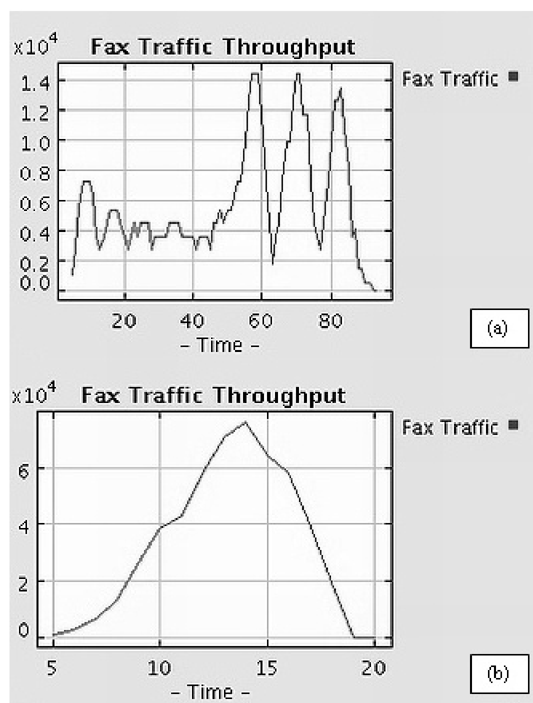

Figure 4. Fax traffic throughput. (a) Best Effort. (b) DiffServ.

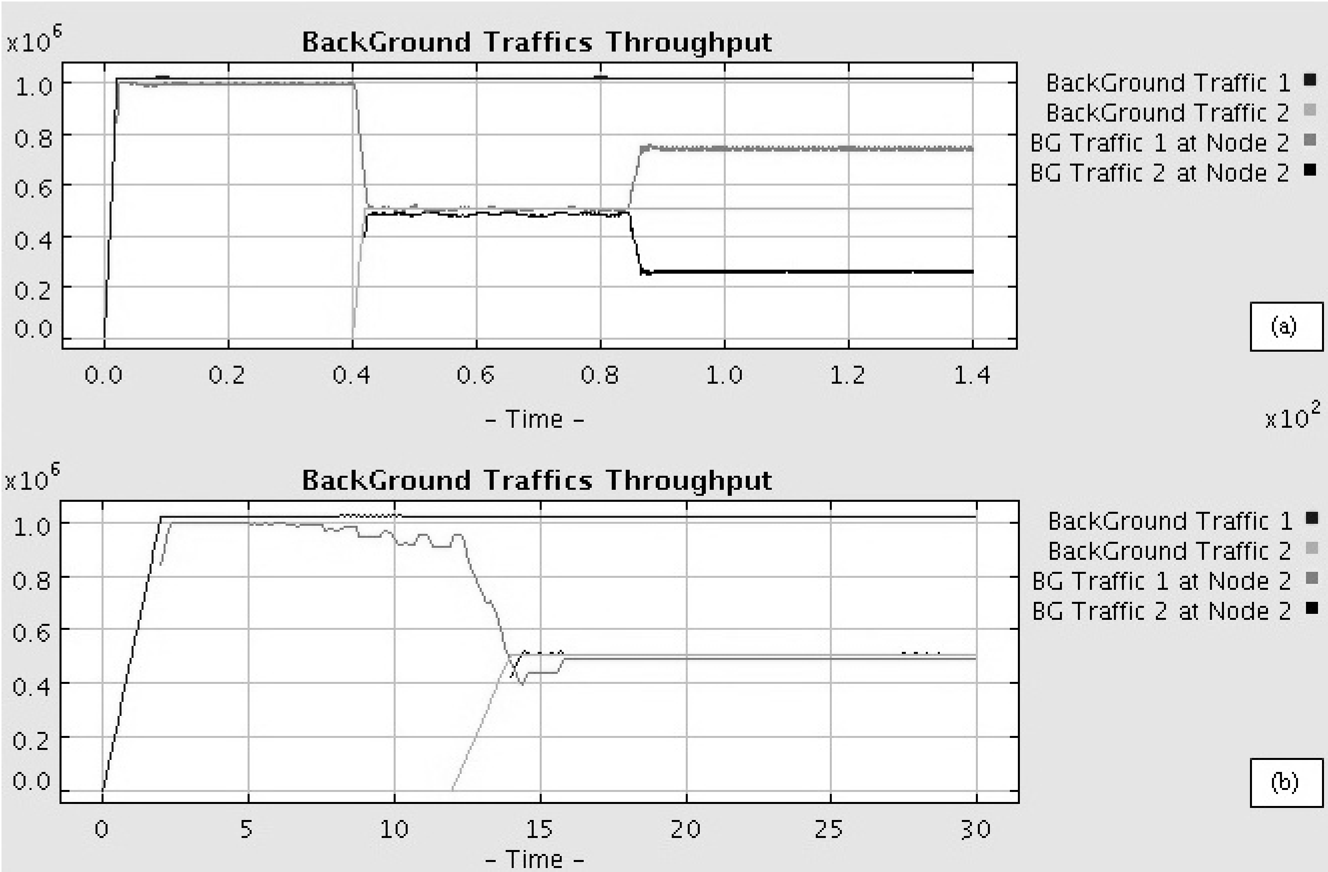

Figure 5. Background traffic throughputs. (a) Best Effort (b) DiffServ.

mode a, in which all packets are tagged as "Best Effort", than what can be achieved in mode b. Also, maximum throughput of mode a is about one-fifth of that of the mode b. It's worth emphasizing that the only difference between the two modes of simulation is the disabling of the DiffServ architecture in mode a. The second background traffic source starts in the middle of the fax transmission in both cases. Apart from the duration of the transmission, the excessive fluctuations in the case that all packets are labeled as "Best Effort", translate into jitter in the fax stream which is a significant hindrance in the success of real-time fax.

4.2.2 Background traffic throughputs. In Fig. 5, we measure the background traffic throughput of both traffic sources. This is done to monitor the effect of these traffics on the fax stream. As expected, in the case that all packets are labeled as "Best Effort", background traffic flows are hardly disturbed by the fax stream. That is because fax packets do not enjoy the preferential treatment in the core routers as is the case in the DiffServ architecture. The disturbance in the background traffic flows is apparent in the other case, shown in Fig. 5b. We also point out that since the simulated core router has a bottleneck of 1 Mb/s, as soon as the second traffic source starts, the number of packets of first source which manage to receive the node 2 drops significantly because of the over-congestion and lower tagged priority compared to the second source. So despite the fact that the rate of first source is twice that of the second one, numbers of packets of both sources that manage to receive node 2 are almost equal. But in the case that all packets are tagged as "Best Effort", as the system stabilizes, both traffic sources are treated in the core router with the same priority so the source with the higher rate has the greater number of packets at node 2.

4.2.3 Fax packets sequence number and the TCP congestion window of the fax stream. In the other two sets of figures, received fax packets sequence number and TCP congestion window of the fax stream are presented. Again, we observe a better performance when the fax packets are tagged as "Expedited Forwarding". In the packets sequence number figure, Fig. 6, in the case that all packets are tagged as "Best Effort", we observe that fax packets arrive out of sequence in some instances in time and this is a serious drawback in the real-time transmission. In Fig. 7, we notice the deficiency of flow control mechanism of TCP, its congestion window, and how it is unable to maintain a consistent window size throughout the transmission in the case that all packets are treated the same. On the other hand, it is apparent that when fax packets are tagged as "EF", TCP congestion window slowly increases in size and reaches steady-state as is the case in the uncongested networks.

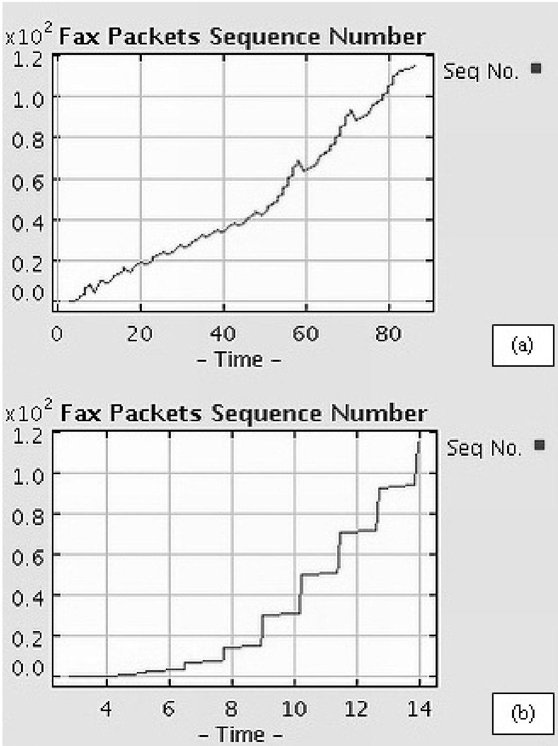

Figure 6. Received fax packets sequence numbers (a) Best Effort. (b) DiffServ.

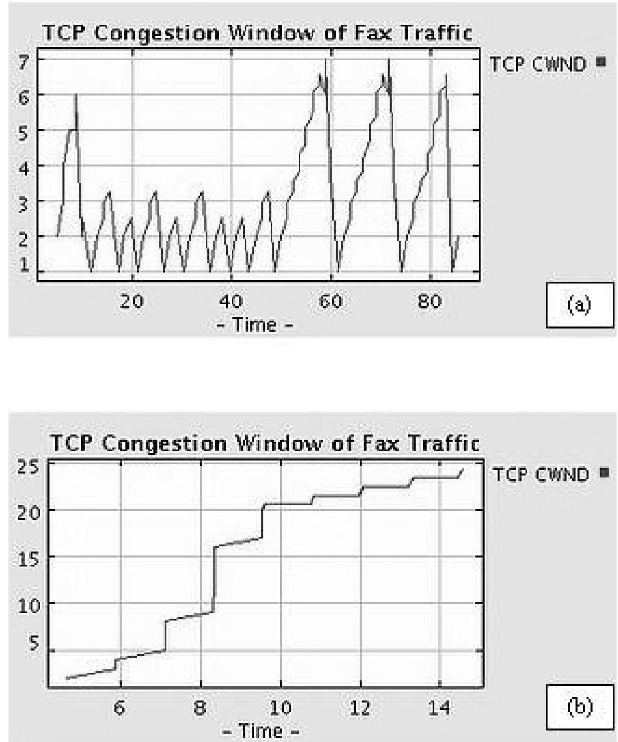

Figure 7. TCP congestion window of fax packets stream (a) Best Effort. (b) DiffServ.

5. Concluding Remarks

In this paper, we studied the transmission of real-time fax in IP networks and the related traffic engineering issues. The best current practices of real-time fax transmission, which are the utilization of the SIP as the signaling protocol along with T.38, were chosen and partially implemented. Two measures were proposed to streamline the setup and flow of fax in the network. In the setup phase, we proposed the utilization of the SIP contact header in order to reroute the SIP signaling packets away from the proxy servers and lessen their traffic loads as a consequence. The other measure was the adoption of DiffServ architecture in order to provide QoS for the transmission of real-time fax in the Internet. Through computer simulations of the network scenario, by measuring the important parameters such as throughput, TCP congestion window and fax packets sequence number, we consistently found that DiffServ nicely lends itself to providing QoS for real-time fax.

One important point is that, although we focused on fax in this paper, other streaming media can also adopt these measures and benefit from them. We also acknowledge and appreciate the other QoS architectures currently investigated and believe that these emerging architectures have the potential to deliver even more promising results in the near future.

References

[1] Bur Goode "Voice over Internet Protocol (VoIP)", Proceedings of the IEEE, Vol.90, No.9, September 2002.

[2] "Voice and fax over IP", The International Engineering Consortium, http://www.iec.org.

[3] ITU-T Recommendation T.37, "Procedures for the transfer of facsimile data via store-and-forward on the Internet", Terminals For Telematic Services, June 1998.

[4] ITU-T Recommendation T.38, "Procedures for real-time Group 3 facsimile communication over IP networks", Terminals For Telematic Services, March 2002.

[5] ITU-T Recommendation T.30, "Standardization of Group 3 facsimile terminals for document transmission", Terminals For Telematic Services, July 2003.

[6] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson. R. Sparks, M. Handley and E. Schooler, "SIP: Session Initiation Protocol",RFC 3261, IETF, June 2002.

[7] Alan B. Johnston, "SIP: Understanding the Session Initiation Protocol", 2nd Ed. Artech House 2004.

[8] Jean-Francois Mule and Jieying Li, "SIP Support for Real-time Fax: Call Flow Examples And Best Current Practices", Internet Draft, IETF, draft-ietf-sipping-realtimefax-01.txt, August 2003.

[9] R. Braden, D. Clark, and S. Shenker, "Integrated services in the internet architecture: An overview," IETF RFC 1633, 1994.

[10] D. Black, S. Blake, M. Carlson, E. Davies, Z. Wong, and W. Weiss, "An architecture for differentiated services," IETF RFC 2475, 1998.

[11] V. Jacobson, K. Nichols, and K. Poduri, "An expedited forwarding PHB," IETF RFC 2598, 1999.

[12] B. Davie and A. Charney *et al.*, "An expedited forwarding PHB," IETF RFC 3246, 2002.

[13] A. Charny *et al.*, "Supplemental information for the new definition of the EF PHB (expedited forwarding per hop behavior)," IETF RFC 3247, 2002.

[14] E. Rosen, et al., "Multiprotocol Label Switching Architecture", IETF RFC 3031, January 2001.

[15] J-Sim Simulator: http://www.j-sim.org/.

[16] Nam Network Animator: http://www.isi.edu/nsnam/nam/.