3. Simulation results

The ns-2 simulator was used to evaluate the proposed scheme on a 14-node NSF network topology (Fig. 2(a)) where each link had sixteen 10 Gbps channels in each direction. The load was distributed among the ingress-egress nodes uniformly so that when the proportion of video traffic was for example 10%, all ingress nodes sent 10% video traffic and 90% BE traffic. Burst aggregation was based upon a size threshold, in which the maximum burst size was set to 500 KBytes. Size threshold was chosen to avoid performance degradation in case of utilizing time out aggregation technique with multiple burstifiers [13]. The core nodes were bufferless and the wavelength continuity constraint was observed. The OBS control plane supported JET (Just Enough Time). Data bursts were scheduled with the LAUC-VF (Latest Available Unscheduled Channel with Void Filling) channel scheduling protocol with control bursts scheduled by LAUC. All routes were established by a shortest path routing algorithm with the number of hops as the metric.

Five different 10-second CIF (352×288) standard test sequences were encoded using H.264/AVC (Advanced Video Coding) reference software JM16.1 [17] with 30 frames per second. Video data partitioning was implemented in the encoding process in order to facilitate priority provisioning over the OBS network through duplication at the ingress node of “A” packets, which are the most important. The encoded H.264 videos were then converted into trace files which were fed into the OBS network. The simulations were run many times to obtain 99% confidence intervals, and for each run, the received H.264 file was derived from the trace file which was produced. The H.264 videos were then decoded to obtain a YUV video format. The resulting PSNR (Peak Signal to Noise Ratio) was measured by using the EvalVid framework [18] to compare this new file with the original lossless YUV video.

As mentioned earlier, “A” packets are smaller than other packets, although they are crucial to the quality of the resulting video. Because the aim of this paper is to minimize the increase in load when the proportion of video traffic increases, three different scenarios were investigated in which the proportion of video traffic was 10%, 30%, and 50% respectively. Figure 2(b) compares the Basic Cloning Scheme (BCS) [19], CCS, and ES. Figure 2(b) shows that for BCS and CCS, changing the type of video sequence (Akiyo, Stefan, Paris, Foreman, and Mobile) doesn’t affect the increase in load—this is because the videos are sent at a similar rate (1 Mbps for each video sequence). However when applying ES, the type of video does have a slight effect because each video sequence has a different proportion of “A” packets, therefore the duplication of “A” packets varies slightly among different videos as shown in Fig. 2(b).

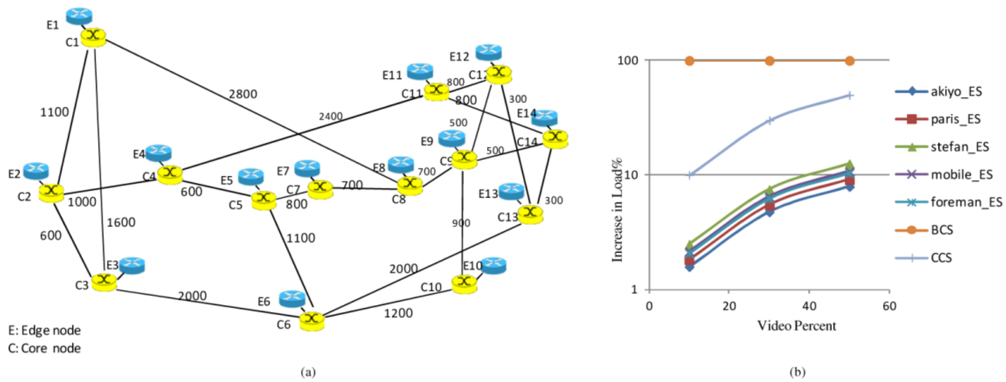

Fig. 2. (a) 14-node NSF network topology (b) Extra load added due to applying different schemes

Figures 3(a), (b), and (c) show the PSNR values for the Akiyo, Stefan, and Mobile video sequences respectively. As the proportion of video traffic increases, the added load increases,

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thus increasing the loss rate. Figures 3(a), (b), and (c) compare the PSNR values when streaming a video without ES against the case where ES was implemented with a video percentage equivalent to 50% of the total traffic which is the worst-case scenario for the ES scheme. Figures 4(a), (b), and (c) compare PSNR values with 10%, 30%, and 50% video traffic, with video traffic from Akiyo, Stefan, and Mobile video sequences. Figures 3(a), (b), and (c) show considerable improvement in PSNR for the different video sequences. Moreover, the proposed scheme exhibits very robust performance; it clearly outperforms other schemes in the worst-case scenario, when video composes 50% of the total traffic. Figure 4 compares the PSNR values for three video sequences when the video comprises 10%, 30%, and 50% of the total traffic, showing that as the percentage increases there is little effect on the received video quality. Clearly the objective has been attained of alleviating the performance degradation as the proportion of video traffic increases.

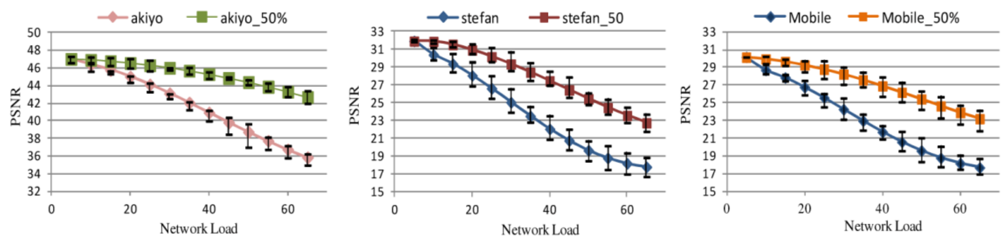

Fig. 3. Video quality, regular streaming versus worst-case ES scheme video streaming for (a) Akiyo (b) Stefan (c) Mobile video sequences

Figures 5(a), (b), and (c) show the loss rate of Akiyo, Stefan, and Mobile video sequences respectively. With ES, there is a slight increase in the loss rate because of the increased network load, however bear in mind that this loss is distributed across the whole streamed video which includes the original video plus the duplicated "A" packets. However as shown in Fig. 5, video streamed with ES will has a much lower loss rate for "A" packets, which results in improved video quality. The way duplication of "A" packets is carried out plays a

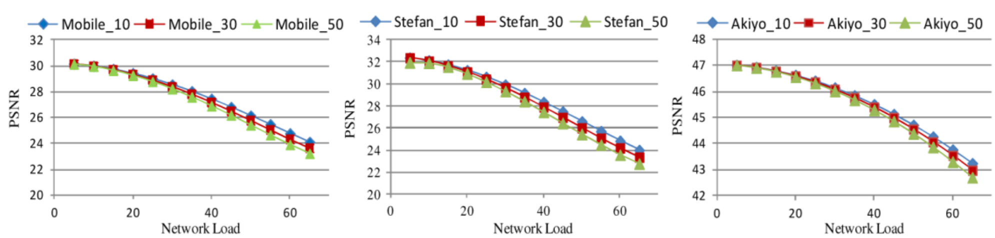

Fig. 4. Video quality comparison of (a) Akiyo, (b) Stefan, (c) Mobile video sequences in three scenarios; 10%, 30%, and 50% video rates ES streaming

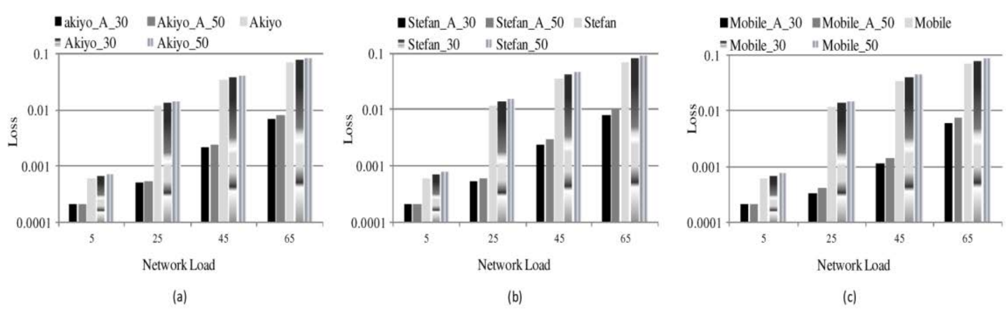

Fig. 5. Video loss rate comparison; regular streaming, 30% ES, and 50% ES streaming for three different video sequences (a) Akiyo (b) Stefan (c) Mobile

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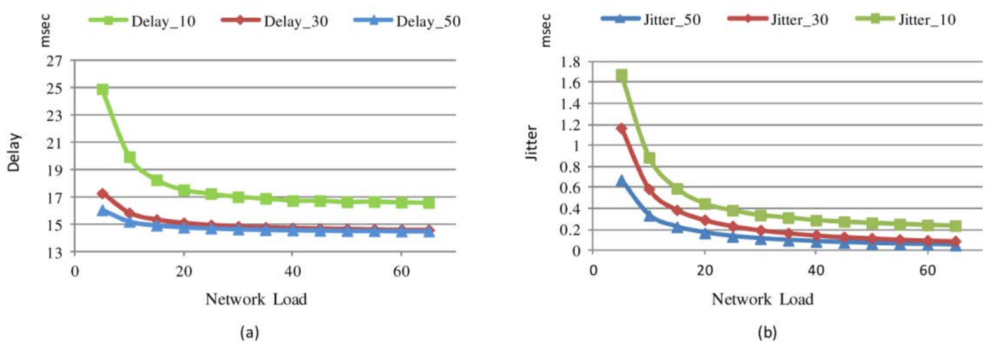

Fig. 6. (a) Average End to End Delay and (b) Average Jitter

very important role in reconstructing a video with higher quality. If one burst is dropped video packets may be lost, however the video can still be reconstructed because its complement is in another burst (due to the use of Buffer 1 and Buffer 2). Figure 6(a) shows the average ETE delay for video traffic from the five sequences, demonstrating that with the size threshold aggregation technique, ETE delay reduces as incoming traffic increases.

Figure 6(b) shows the mean jitter value for the video sequences. Like ETE delay, jitter increases as the proportion of video traffic increases. However, the maximum average jitter is 1.7 ms when the proportion of video traffic is 10% and the network load is 5%. At higher loads and with the same proportion of video traffic, jitter stabilizes and settles to below 0.35 ms.

4. Conclusions

This paper presented a novel ingress node design which delivers enhanced streamed video across OBS networks. The proposed scheme is called the Enhanced Scheme, in which the most important packets arising from video data partitioning encoding are switched between two buffers in the ingress node in order to facilitate better video reconstruction.

Extensive ns-2 simulations were conducted to evaluate the Enhanced Scheme, showing its efficiency where an average improvement of 5 dB in video quality was obtained in the worst-case scenario with medium and high network loads. The proposed scheme improves the quality of high-priority traffic (i.e. video traffic) without significantly affecting the best effort traffic loss rate, which increased by only 12% in the worst-case scenario. The end-to-end delay was below 17 ms with a network load of greater than 20% while the average jitter value was less than 0.35 ms under the same network load.

Future work may evaluate the suggested scheme for scalable video streaming, in which a H.264/SVC codec will be used instead of the H.264/AVC. Instead of duplicating “A” packets at the ingress node, duplication of the base layer will be investigated, and performance will be evaluated in that case.

Acknowledgments

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