Mobile Multipath Cooperative Network for Real-Time Streaming

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Abstract. Access links are often times the bottlenecks of wireless wide area networks (WWAN). The prevalent use of multimedia applications on mobile devices introduces an ever increasing traffic load on WWAN access links, leading to traffic congestion and unsatisfactory user experiences. In this paper, we introduce a mobile multipath cooperative network. In the system, multiple paths are dynamically established among cooperative devices over WWAN and WLAN so that multiple descriptions of a multimedia stream can be transported over distinct end-to-end paths between two mobiles. As a result, the capacities of multiple wireless access links can be utilized to enhance quality of experience of a multimedia application. We also introduce a MDC rate adaptation algorithm that jointly adapts the source coding rates among multiple paths. Our lab experiments show that real-time streaming with multiple description coding benefits significantly from the proposed cooperative network and subsequently enhance the quality of user experience for multimedia applications.

Keywords: Multiple description coding, mobile multimedia service, network architecture, overlay network, quality of experience, real-time streaming.

1 Introduction

The amount of digital information created and replicated in the world is expected to grow to 35 trillion gigabytes in the next 10 years [1]. The primary driver for this exponential growth is multimedia fueled by social media networks and the increasing desire of users to share content, be it a personal video of your cat or a digital photo taken at a social gathering. More than 70% of the data generated this year will be from individuals at home, at work or on the go [1]. Multimedia capture devices on mobiles play a major role in creating this user generated content (UGC) due to their inherent mobility. Mobiles are available anytime, anywhere, when the user wants to capture or access content.

The computing and multimedia processing capabilities in mobile devices has increased several fold over the past couple of years, particularly with respect to video formats (resolution, frame rate). For example, VGA at 30 fps video recording was prevalent until about 2010, while upwards of 8M pixel cameras, HD (up to 1080p60)

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and 3D stereo video capture is becoming common going forward. In addition, the increasing memory capacity leads to a large amount of personal content to be stored on the phones. Consequently, mobiles become the primary sources of UGC with no efficient means to share the content instantly at its captured quality.

The primary means of sharing this content today is through enterprise servers. IDC (International Data Corporation) data shows that nearly 75% of the digital world today is a copy. The double digit annual growth in the amount of data exacerbated by high fidelity capture devices not only burdens the enterprise servers but makes this conventional means of sharing unsustainable.

QoE (Quality of Experience) is further compromised by limited capacity on radio access links in wireless networks. This makes it difficult for users to instantly share their mobile multimedia contents over cellular networks. As shown in Fig. 1, the uplink capacity of WWANs (both 3G and 4G) is significantly lower than that of the downlink. The capacity of a cellular access link is also subject to constraints, such as the maximum bit rate allowed for a user according to its service subscription, the number of simultaneously active users, and channel condition fluctuation.

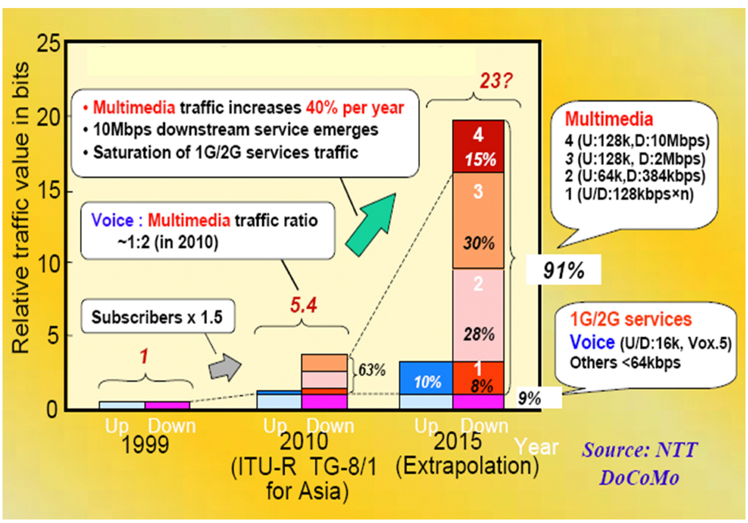

Fig. 1. Forecasted traffic vs. cellular capacity increase

To increase the access link capacity, a mobile device may contract a neighboring cooperative device and utilize the access link of the cooperative neighbor to support its traffic. Subsequently, multiple paths can be established between a pair of source and destination devices to increase the capacity and the reliability of the transport for the multimedia application. This concept is shown in Fig. 2.

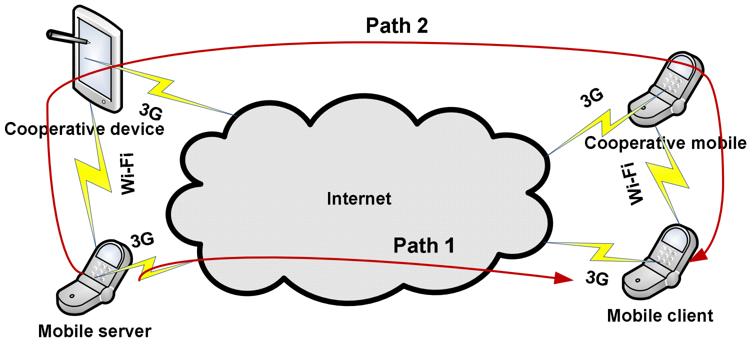

Fig. 2. Multiple access links are used by a multimedia application to increase the capacity and the reliability of its network access

In Fig. 2, a mobile server can stream the data simultaneously on two disjoint paths to the mobile client; one is a direct path while the other traverses across cooperative devices and utilize the capacity of their access links.

Based on the concept illustrated Fig. 2, we introduce in this paper a mobile multipath cooperative network. The proposed system comprises a mobile multipath overlay network and a new method of MDC with rate adaptation. In the mobile multipath overlay network, neighboring devices cooperate to provide multiple wireless access links for a mobile device. As a result, a multipath transport facility is provided between two remote mobiles. The new method of MDC with rate adaptation adjusts the source coding rate of the content upon capacity fluctuation and synchronizes the deliveries of multiple descriptions over different paths.

The proposed system can be deployed over the existing IP network. It leverages the proliferation and availability of multiple mobiles in vicinity. It eases the impact of information explosion on enterprise content servers in the cloud and subsequently opens the door to new mobile driven multimedia services. Many service models can be conveniently built on top of the proposed architecture. Using Multiple Description Coding (MDC) [5] over two different paths can provide a quality improvement of a QVGA video stream from 300 Kbps at 15fps and 28dB PSNR to 600Kbps at 30fps and 30dB PSNR. High quality multimedia content sharing can also be enabled when cooperate devices distribute individually obtained descriptions among each other.

In Section 2, we introduce the mobile multipath overlay network architecture and its protocols. In Section 3, we present the MDC module that utilizes the mobile multipath overlay network for real-time streaming. In Section 4, we introduce the multipath rate adaptation system. In Section 5, we present our experiment results that demonstrate the performance gain provided by our proposed system. In section 6, we give conclusions.

2 Mobile Multipath Overlay Network Architecture

In this section, we introduce the mobile multipath overlay network for transporting multiple descriptions of a multimedia content. Several algorithms have been proposed to transport multiple descriptions of a multimedia content across the network [2][3][4]. These algorithms focus on selecting multiple paths in an infrastructure based content distribution network and utilizing multiple access links directly available at a mobile handset. Different from those approaches, in our proposed overlay network, neighboring devices communicate among one another to provide multiple wireless access links for a mobile device.

We call a traffic source as a source and a traffic destination as an aggregator. The cooperative device contracted by a source is called a source helper. The cooperative device contracted by an aggregator is called an aggregator helper. A device may recruit multiple helpers. A multimedia content is encoded into multiple descriptions, each traversing one path in the overlay network to reach the aggregator. Each description can be rendered at an aggregator independently. And the rendered quality increases incrementally based on number of descriptions received.

The data plane of the multipath overlay network introduces an overlay switching layer between the application and the underlying transport protocol. The function of the overlay switching layer is to route the traffic from the source to the aggregator over overlay network helpers. An illustration of the multipath overlay network data plane is shown in Fig. 3. We use RTP as an example for the application layer.

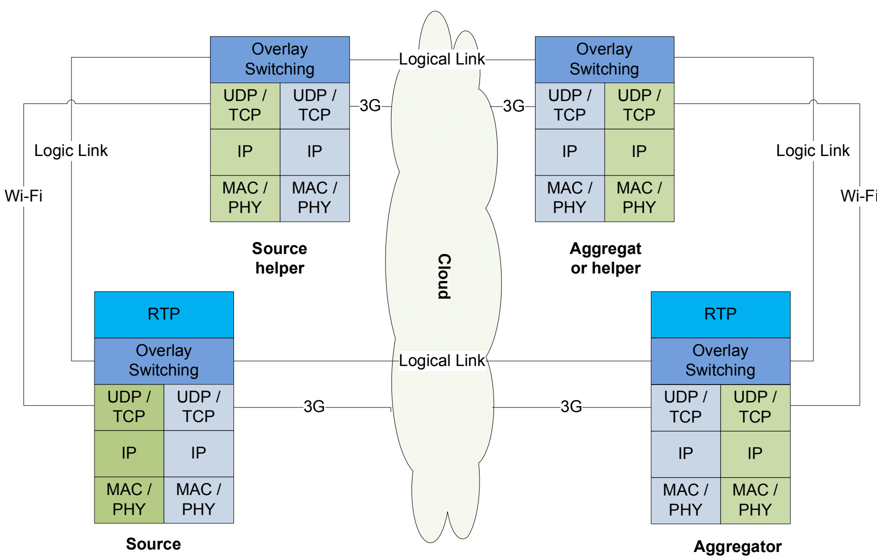

Fig. 3. The data plane of the multipath overlay network

The overlay switching layer employs a label switching architecture similar to that in MPLS [6]. The label switching architecture reduces the routing processing complexity and enables differentiating per hop behaviors for different streaming sessions. Each packet is tagged with a label when travelling from one overlay device to another. The logical link between two overlay devices on an established overlay network path is called a segment. The label tagged on a packet is uniquely assigned by the downstream overlay device of that segment during the session establishment phase (whose details are in Section 2.1). An overlay device determines the QoS treatment it should give to an incoming packet based on the label. When sending the packet downstream, an overlay network device should swap the label to another label assigned by the downstream overlay device on the successive segment.

A device may function as source/aggregator/helper and may support one or more sessions for a service and one or more services.

The control plane of the multipath overlay network is used to establish, replace and release a data plane path between a source and an aggregator. The multipath overlay network uses TCP for transporting overlay network control plane messages. The control plane does not include the logic link between a source helper and an aggregator helper. During the signaling phase, the source and the aggregator can contract its neighboring cooperative devices to function as helpers. The selection of helpers is based on local P2P device and service discovery procedures that take into consideration physical channel capacities and processing capabilities.

In the following, we briefly describe the protocols to establish and maintain a multipath transport for a streaming session in the mobile overlay network.

2.1 Multipath Management Protocol

In Fig. 4, we show the multipath establishment procedure.

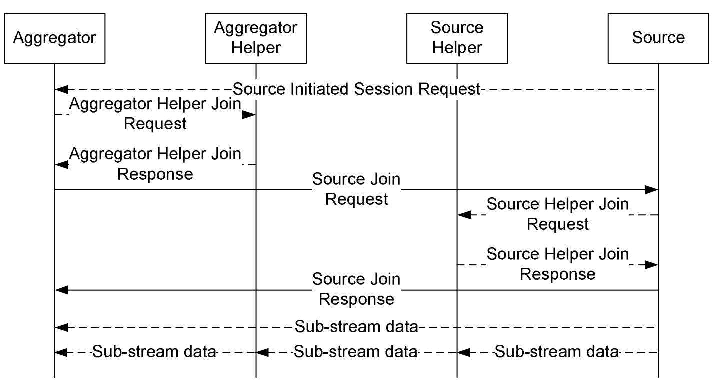

Fig. 4. Multipath establishment for a streaming session. Helper Join Request/Response may be optional.

When a mobile device intends to download data, it acts as an aggregator. When a mobile device intends to upload data, it acts as a source. If a mobile device intends to upload data, it needs to first send a session request to an aggregator to bootstrap the multipath establishment procedure.

The multipath establishment procedure is as follows. An aggregator, if needed, sends a request to a cooperative device to contract it as an aggregator helper. The request contains a designated label that should be tagged to the packet of this streaming session sent towards the aggregator on this path segment. An aggregator helper sends a response back to the aggregator with a designated label that should be tagged to the packet of this streaming session sent from the source side to the aggregator helper on upstream path segment. Subsequently, the aggregator requests the designed source device to join the session and provides the label to be used on the direct path and the label to be used on the alternative path to reach the aggregator helper. The source device determines whether a source helper is needed to send the alternative description of the same streaming session based on its access capacity. If a source helper is contracted to cooperate, the source should give the label to be used for forwarding the traffic towards the aggregator helper to the source helper. And the source helper should give the label to the source for sending packets of this streaming session to the source helper. At the end, the source sends a response to the aggregator. The source can send a description without completing the path establishment for another description.

One significant advantage can be provided by the multipath overlay network is that, when one path is down, the streaming can still be supported by another path. To help ensuring the QoE of streaming over multipath, a seamless path replacement procedure is introduced. In Fig. 5, we show the multipath replacement procedure.

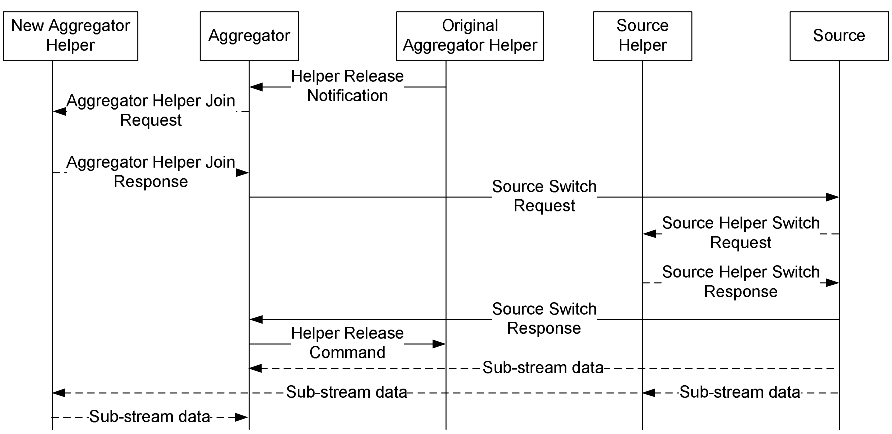

Fig. 5. Multipath replacement for a streaming session

A beneficiary device, the aggregator or the source, should contract another cooperative device to replace the original helper if the original helper is determined to be unsuitable for the delivery of a description of the multimedia stream.

3 Multiple Description Coding Module

Multiple Description Coding (MDC) [5][7] is proposed as an alternative to layered coding for streaming over unreliable channels. The goal of the MDC is to create several independent descriptions that can contribute to one or more characteristics of video, such as spatial or temporal resolution, signal-to-noise ratio, frequency content etc. These independent descriptions can be sent over different paths and aggregated at the receiver.

MDC is different from Scalable Video Coding (SVC) [8] [9] in that SVC generates a base layer and one or more enhancement layer bit streams. The base layer is required for the stream to be decoded. Enhancement layers provide improved quality. In MDC, all descriptions are independent and hence the more descriptions the decoder receives, the better the quality. For scenarios where the packet transmission schedules can be optimized in a rate-distortion sense, layered coding provides a better performance. The converse is true for scenarios where the packet schedules are not rate-distortion optimized [10] [11].

Besides increased fault tolerance, MDC allows content providers to send descriptions of a stream without paying attention to the client. Receivers that can't sustain the data rate only subscribe to a subset of these descriptions, thus freeing the content provider from sending streams at lower bit rates. The robustness of MDC comes from the fact that it is highly unlikely that the same portion of the same picture is corrupted in all the descriptions. Various experiments have shown that the MDC is very robust and the quality is acceptable even at high loss rates.

We utilize MDC in H.264 in a number of ways as those described in [12] to achieve better QoE, including Group Of Pictures (GOP) level MDC, Frame Level MDC, Slice Level MDC, Macro Block (MB) Level MDC and Region Of Interest (ROI) and/or Persistence based MDC. Each one of the options has its own advantages and disadvantages.

MDC in combination with Rate Adaptation is used to improve the performance of the proposed MDC system. This is due to the fact that the rate adaptation makes a more efficient use of the instantaneous network capacity. MDC enables real-time rate adaptation through Selective Combining at frame or slice level using the above methods to achieve an average 3dB improvement in PSNR with perceivable visual quality enhancement. Multiple descriptions also decrease the peak-variations in a given sequence and improve the peak to average ratios.

4 Multipath Rate Adaptation Algorithm

4.1 Rate Adaptation Algorithm

Many algorithms [13][14][15][16] have been introduced to optimize MDC data transmission over multiple paths. These algorithms either focused on optimizing the distribution the coded data among different paths or adjust the data transmission rate of the coded content. Instead, our proposed rate adaptation algorithm adjusts the source coding rate of an individual description so that the highest feasible quality of a description can be delivered without jeopardizing delay and delay jitter experience of the real-time streaming. The proposed rate adaptation system also performs throttling and seeking forward operations on each path to synchronize multiple descriptions of the same multimedia content.

The rate adaptation system operates on feedback information, including aggregator's buffer occupancy status, buffer difference status among multiple paths, packet loss ratio, network delay, delay jitter and throughput on individual paths.

The buffer occupancy report of each individual path sent to the source identifies the buffer state including overflow, underflow, or normal. If buffer overflow occurs, the source throttles the transmission of the corresponding description data so that the delivery of the description can be in synchronization with the overall playback of the stream. Buffer underflow signals potential network congestion. If underflow occurs, the source reduces the source coding rate in an attempt to restore buffer normal buffer status. If underflow persists, the source performs seeking forward to synchronize delivery of the description with the overall playback of the stream.

It is critical to maintain synchronization among multiple descriptions. In doing so, the aggregator calculates a reference buffer occupancy level which is the average buffer size of healthy receiving buffers. The buffer difference of each individual buffer from the reference level is then calculated. The buffer difference status is reported back to source and identified as early, late, and normal. If description early occurs, the source throttles the transmission of the corresponding description. If description late occurs, the source first reduces the source coding rate to restore the description normal status. If description late persists, the source performs seeking forward.

The source also monitors traffic performance metrics, including packet loss ratio, throughput, delay and jitter. Desirable ranges of these traffic performance metrics are calculated. The desirable range of throughput is set based on the transmission data rate. The desirable ranges of delay and delay jitter are set to be vicinity of the current calibrated network delay. The calibrated network delay needs to be calculated carefully, as the delay may vary based on the adjusted source coding rate. To avoid oscillation, the calibrated delay is initially set based on the result obtained by burst transmissions at the session initialization. The calibrated delay is then calculated as a low pass filtered version of the instantaneous delay. The source reduces the source coding rate when any of the performance metrics deteriorates with respect to their desirable ranges. The source increases the source coding rate when the throughput and the delay outperform with respect to the desirable ranges, since serves to restore the high source coding rate after temporary network congestion events.

The proposed rate adaptation system also takes into account the scene change. When the multimedia streaming operates at a scene that generates a very low data rate, the system cannot effectively obtain the capacity information based on feedback. The data rate instructed by the rate adaptation shall remain at the previous level, when the intensity of the scene goes down. When the multimedia content is switched from a lower data rate generating scene to a higher data rate generating scene, the source coding rate should be allowed to go beyond the rate instructed the rate adaptation system, since the instructed data rate does not necessarily represents the capacity of the path. This manner, a scene change opportunistically explores the capacity of the network.

4.2 Overlay Feedback System

A feedback system is designed to report the buffer status and traffic performance to the source. The data flows of the feedback system are shown in Fig. 6.

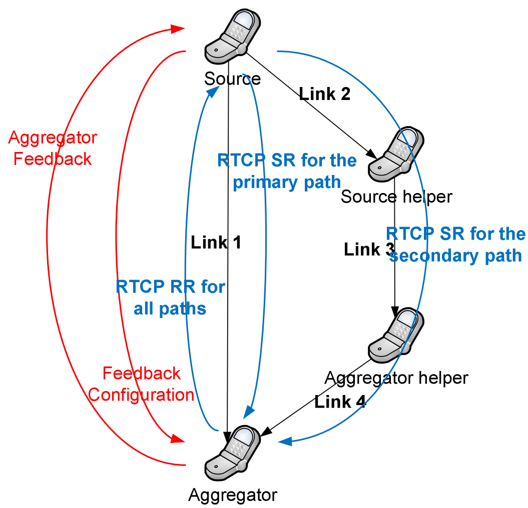

Fig. 6. Data flows of the feedback system

An overlay network control plane message, called Feedback Configuration is sent whenever necessary from the source to the aggregator to configure the aggregator to perform feedback, including feedback time interval and the buffer difference window.

RTCP is used to feedback delay, jitter, and packet loss ratio. To measure the delay, RTCP Sender Reports (SR) need to be sent to the aggregator on each individual path tagged with transmission timing information. The RTCP Sender Report (SR) for the secondary path utilizes the overlay network path. To remove the necessity of establishing a reverse overlay network path, all RTCP Receiver Reports (RR) are sent from the aggregator to the source directly. RTCP only provides round trip time information of a path. The delay of the primary path can be estimated as half of its round trip time. However, the delay of an alternative path has to be calculated based on the delay of the primary path. Denote the round trip time for the ith path by $RTT(i)$. The primary path is the 0th path. The delay of the ith path $D(i)$ is estimated as follows:

$$D(0) = RTT(0)/2,$$

$$D(i) = RTT(i) - D(0), \forall i > 0.$$

An overlay network control plane message, called Aggregator Feedback, is sent periodically to the source to report the buffer occupancy status, the buffer difference status, and the receiving throughput.

4.3 Performance of the Rate Adaptation System

We demonstrate the effectiveness of our rate adaptation system in Fig. 7. The study is done by writing a rate adaptation system simulator.

In the simulator, a movie trailer's data are streamed down from a source device to an aggregator device over two paths. The frame rate is 30fps. The movie trailer roughly goes through a scene change every 4 seconds and has 6 source coding levels of each description. The source data rate varies between 200Kbps to 3Mbps.

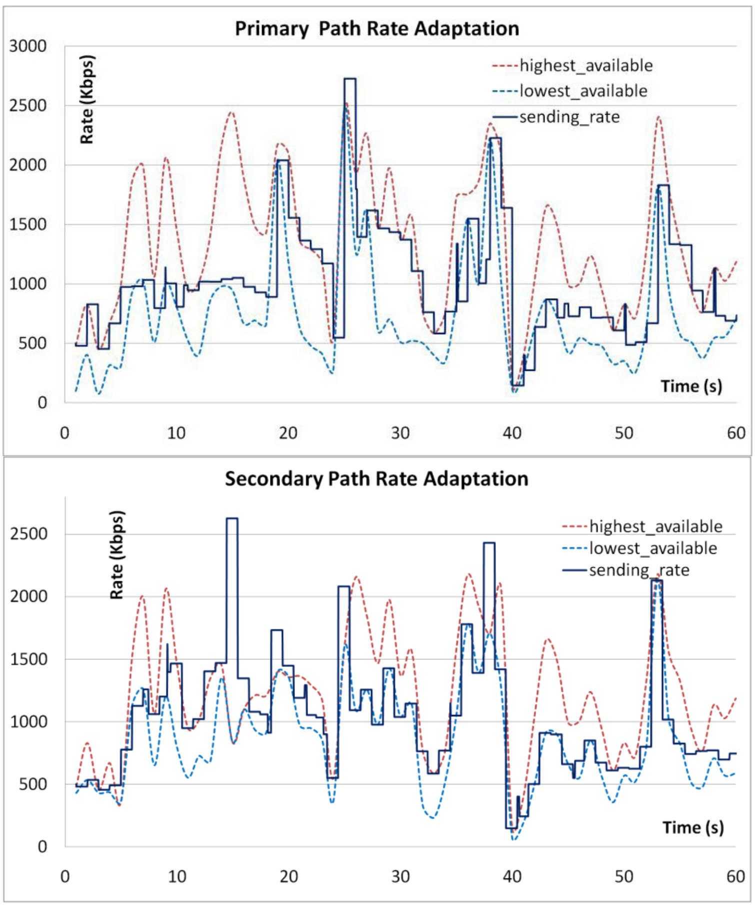

Fig. 7. Capacity utilization of individual paths under source coding rate adaptation

The primary path has a latency 100ms. The secondary path has a latency 500ms. The network capacity of each path varies between 2Mbps and 500Kbps. The feedback system signals the statistics described in Section 4.2 to the source every 0.5 second.

In Fig. 7, the source coding rate over time is plotted against a capacity region identified by the highest available rate and lowest available rate. The highest available rate is equal to the minimum of the network capacity and the highest available source coding rate of the scene. The lowest available rate is equal to the minimum of the network capacity and the lowest available source coding rate of the scene. As the network capacity varies and the source rate of the scene varies, a desirable source coding rate should be close to the highest available rate curve, yet not degrading traffic performance seen at the receiver. The rate adaptation maintains the two descriptions to be synchronized and provides 80% capacity utilization with less than 30ms delay jitter on average.

5 Experiment Results

We investigated the end-to-end distortion performance of the proposed system and compare the results with a single description coded bitstream.

The clips used for testing are movie trailers, including Golden Eye (GE), Despicable Me (DM) and Sorcerer's Apprentice (SA). The DM and SA clips are of WVGA (800x480) resolution while the GE clip is of VGA (640x480) resolution. The GE clip is a fast moving scene which has High Texture and High Motion. The SA clip is a Medium Motion and High Texture clip with frequent scene changes. The DM is a cartoon trailer that has fast scene cuts, High Texture and a combination of Low Motion and High Motion scenes.

First, the sequences are encoded using an un-modified Joint Model 17 (JM) encoder using the constant Quantization Parameter (QP) mode. Then the JM 17 code is modified to include the necessary changes required for the MDC system, including but not limited to inserting the SEI messages and having the QP values read from a file. All the clips are again encoded using this modified encoder and the results are investigated. All other settings for both the encoders are kept as close to the same as possible.

The lab system setup is as shown in Fig. 8. In the set up, we have a laptop acting as an MDC aggregator in the cloud, a mobile acting as a MDC source and a mobile server, and a cooperative mobile acting as a source helper. The source and its helper are connected using a Wi-Fi connection, while both of them access the cloud via their 3G access links. The source intends to upload its data, hence streams a description directly to the aggregator and streams another description via the helper to the aggregator.

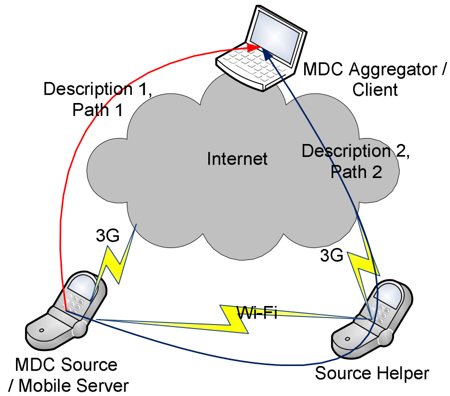

Fig. 8. Lab System Set Up

In a simple experiment, a video sequence is encoded using two slices, the first slice with a Quantization Parameter (QP) value of 25 and the second slice with a QP value of 35. This sequence is called Description 1 (D1). The same sequence is encoded again using two slices, but now the first slice is encoded using a QP value of 35 and the second slice is encoded using a QP value of 25 (Description 2, D2). Fig. 9 shows the example. When the client receives both the descriptions D1 and D2, highest quality slices among available descriptions are selected for decoding and hence obtains a better video quality equivalent to QP 25 coded frames.

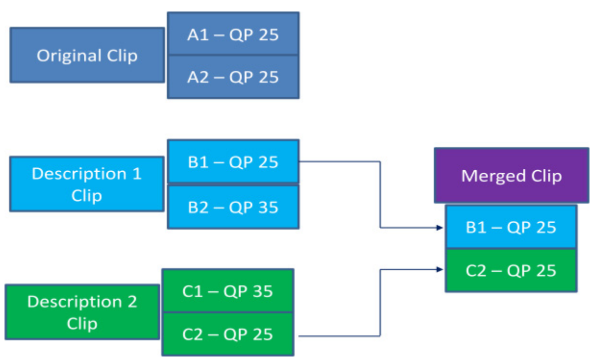

Fig. 9. MDC Encoder – Slice Level Example

Various experiments have been conducted in our labs. For each experiment, we calculated the bit rate and the PSNR of the luma (Y) component and averaged across the total number of frames in each sequence. The first set of experiments shows a typical use case scenario in which each description is of acceptable quality and the MDC selective combined clip shows a marked quality improvement. The second set of experiments shows the capability of the system to adapt to the varying capacity of individual paths. Table 1 shows the results from both configurations.

Table 1. Performance of the Slice Level MDC System

Content	Description 1, D1		Description 2, D2		Combined (D1 U D2)		
	Bit Rate (Kbps)	PSNR (dB)	Bit Rate (Kbps)	PSNR (dB)	PSNR (dB)	ΔPSNR (dB)	ΔRate (%)
Typical Configuration : QP 25 / QP 35							
GE	732	35.9	702	35.4	38.6	2.7	22.1
DM	754	38.3	858	39.3	42.4	4.1	26.7
Edge Configuration : QP 25 / QP 50							
GE	618	27.9	577	26.8	39.6	11.7	4.6
SA	729	28.9	750	27.6	45.8	16.9	6.8

Another advantage of using the proposed MDC system is the reduction in the peak frame size. This helps reduce traffic congestion in the network since peaks are not only reduced but staggered in time when descriptions are transmitted over two paths. Fig. 10 shows the distribution of frame sizes for the SA sequence at different QP settings. As it shows, if the sequence is encoded using QP 25 (blue curve) across the frame, the individual frame sizes have peaks that are quite large compared to the average frame size. This is due to the fact that the scene changes cause I-frames to have much larger size.

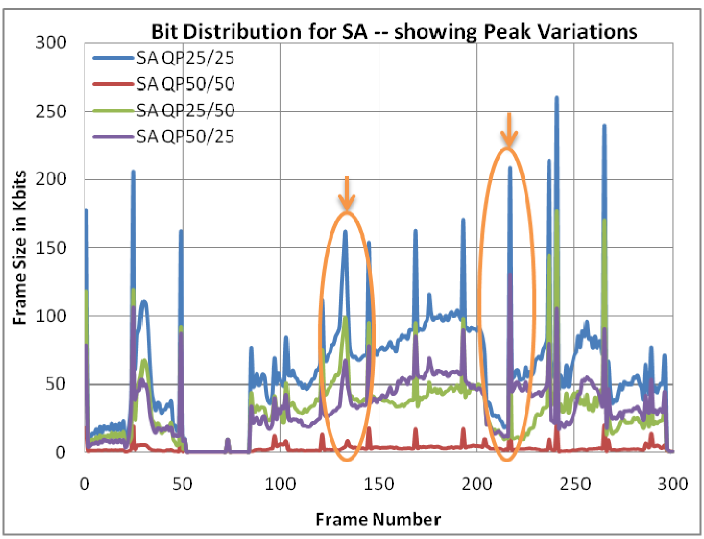

Fig. 10. Bit Distribution for Sorcerer's Apprentice, showing peak reduction

In the case of the proposed MDC scheme, by encoding half of the frames at a QP value of 25 and the other half at QP value of 50, the peaks are distributed across both descriptions. Fig. 11 and Fig. 12 show the quality of the decoded video at the client side. The image on the right shows the quality when both descriptions are received and the left side shows the quality of the video when only one description is received and decoded. Fig. 11 shows the I-frame quality and Fig. 12 shows a succeeding P-frame quality. The figures show that the quality is acceptable even if one of the descriptions is received and the quality improves when both the descriptions are received and decoded.

Fig. 11. Visual Video Quality for an I-frame (L) D1 only (R) D1 U D2

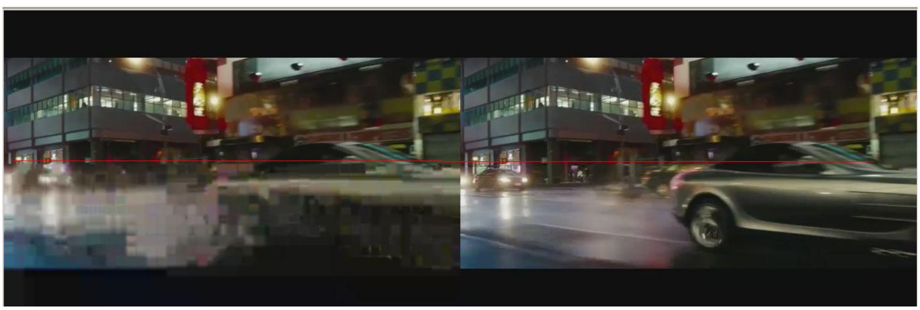

Fig. 12. Visual Video Quality for a Succeeding P-frame (L) D1 only (R) D1 U D2

As shown in Table 1, when both descriptions are received and decoded, the typical increase in Luma (Y) PSNR as compared to the single description case is about 3dB for the typical configuration with a 24% increase in the total bit rate and is about 13dB for the edge configuration case with a 5% increase in the total bit rate. We have also computed the VQM (voice quality metric) scores and the SSIM (structural similarity) values for the merged sequences and for the individual sequences and noticed that the increase is consistent with the visual observations.

From experiments conducted in our lab and from the data presented here, it could be observed that the proposed system consisting of just two descriptions can greatly increase the quality of the video in the case of High Motion and High Texture sequences. In general, it is known that these types of sequences usually require high coding rates and also have high peak-to-average ratios since the frames changes much more frequently. Hence by using the proposed system, we could not only reduce the peaks but also increase the overall quality of the bitstream.

6 Conclusions

In this paper, we introduce a mobile multipath cooperative network for real-time streaming. The system can be deployed over the existing IP network. The architecture enables a mobile device to access the cloud using multiple access links including those provided by its neighboring mobile devices. Different streaming sessions served by the same mobile device can be given distinct per hop QoS and routing treatments. As a result, the network access capacity, the reliability and subsequently the quality of experience for a multimedia streaming originated from or destined to a mobile device are significantly enhanced. We also introduced a new method of MDC with rate adaptation that performs source coding rate adjustment and synchronization for multiple descriptions across different paths. Our source coding rate adaptation system is shown to well utilize the network capacity and maintains synchronization of multiple descriptions over different paths without inducing undesirable delay and delay jitter.

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