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Video Streaming: Single and Compound Report Transcoding Method

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Abstract: Video transcoding performs the operations such as bit rate and layout alteration to transform one compressed video stream to another. Transcoding enable multimedia devices of diverse ability and formats to restore video content on various platforms. Modern multimedia applications have become popular because of the most advanced multimedia compression techniques. Concentrated efforts on 4D Perceptual Quantization modeling (PQrc) predict the frame complexity but suffer from the time varying packet loss rate and fluctuating bandwidth. To overcome the packet loss, a Single and Compound Report Transcoding Method is presented. Moreover, an adaptive error control scheme is developed with the proposed method to initiate error flexibility features in a compressed video which provide more flexibility to the roaming clients. The proposed transcoding method divides the error control operation into single report and multiple or compound report modes according to the availability of the channel. In single report mode, Subject Prioritized Retransmission (SPR) Method is evolved to mitigate the error propagation due to packet loss in video transfer. The SPR Method resolves the retransmission schedule of a lost packet according to the packet's loss impact that is obtained in a front end encoding process. In compound report mode, Channel Alert Compound Report Transcoding (CA-CRT) Method is introduced that reduces the error propagation due to packet loss caused by transient channel switching. The experimental evaluation of single report and compound report transcoding method is compared with the 4D Perceptual Quantization (PQrc) Modeling in terms of PSNR, accuracy, packet loss rate and provides better bandwidth and results evaluate that the packet loss rate is reduced to 3.57%.

Key words: Single report, subject prioritized retransmission, channel alert compound report, video transcoding, error propagation

INTRODUCTION

Due to the simplicity of configuration and the relatively low setup cost, video transcoding is gaining more and more popularity. Among all applications, multimedia applications such as multimedia streaming, multimedia messaging, video telephony and video-on-demand are the most interesting and popular applications used in the present environment as the multimedia applications are easier to be accepted and it is very analogous to human life. Conversely, numerous characteristics such as confined bandwidth, high information, i.e., packet loss rate and unstable condition pose an enormous challenge on enabling multimedia applications. Video transport suffers from signal fading, noise interfering, network jamming and handoff which usually lead to the information loss in video.

A single bit error in a packet causes the loss of an entire packet, if the number of degraded bits goes beyond the error correction capacity. The packet loss difficulty leads to severe video and current frame quality degradation. Packet loss also leads to error diffusion on consecutive frames because of the motion predicted compensation. However, the heterogeneity of client and devices makes it very difficult to adapt the video subjects to a wide degree of different client channel condition. Especially, it is very difficult for mobile users to adapt video subjects who usually roam to various environments. To attain fault robustness while video broadcasting, a single and compound video transcoder are exactly positioned in an intermediary network. The node connected adapts to the non error flexibility streams by removing the error occurrence on compressed video streams.

There exist several channel and source coding schemes which are used in the entry medium to enhance the error robustness of outgoing video bit stream. Forward Error Correction (FEC) and Automatic Retransmission reQuest (ARQ) are the two most commonly used channel coding schemes for fault removal. In the IEEE 802.11 standard, the concept of Application Level Framing (ALF) is often used for packetization in which packets with a corrupted bit number is greater than the capability of fault alteration. In

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this way, fault packet is removed from channel. ARQ is particularly useful to combat against fault packet errors and has been adopted in several existing packet protection methods for video streaming.

The most commonly used error flexible source coding tools include data dividing, synchronization marker, Reversible Variable Length Codes (RVLC), Error Resilience Entropy Coding (EREC), Compound Report Coding (CRC), error tracking and Adaptive Intra Refresh (AIR), etc. CRC is kind of joint source and channel coder. The objective of CRC is to improve the reliability (or error flexibility) of data transmission under channel failure by employing the diversity of channels. CRC inserts added rate to create the video bit stream more flexible by removing the transmission errors.

If information are sent over multiple independent channels, the possibility of failures of all channels is greatly reduced, leading to a high possibility of receiving correct information from at least one channel. Thereby, it is easier to estimate the original information with satisfactory quality. The H.264 bit-rate control uses a 4D Perceptual Quantization modeling (PQrc) including two major encoding modules namely perceptual frame-level bit-allocation using a 1D temporal pattern and the macroblock-level quantizer decision using a 3D rate pattern. 4D Perceptual Quantization modeling (PQrc) suffers the packet loss at the varying time (Huang and Lin, 2007).

In this research, an adaptive error control scheme adaptively inserts error flexibility features into a compressed video. The compressed video in adaptive error control scheme serves as roaming clients in the entry medium. Adaptive error control scheme divides the error control operation into the single report and multiple reports based on channel circumstance judgment. In the single report mode, a subject prioritized ARQ scheme is used to mitigate error propagation due to packet loss. The application scenario is that in the existing channel, a permanent bandwidth is assigned to a portable terminal. The existing channel is not dedicated bandwidth for resending lost packets for the terminal.

SPR Method implies that the retransmitted packets will compete with the regular video packets for the constrained bandwidth resources. In the SPR-ARQ Method, the retransmission of a lost packet is scheduled by the packet's significance level. The significance level of a packet is measured by estimating the relative amount of error propagation on packet loss. In SPR Method, the encoder utilizes the motion information generated in the encoding process and the concealment alteration approximate the error propagation effect of each packet.

The measurements are performed only once and the outcome is stored in the streaming server for directing the packet retransmission scheduling and choice. To address the channel handoff problem in the compound report mode, propose a CA-CRT Method to adaptively divide the incoming bit stream into compound report which is sent to the client terminal data via diverse channels. Compared with traditional Compound Report Coding (CRC) scheme, the proposed CA-CRT Method can not only make best use of path diversity but the CA-CRT method effectively mitigates packet loss during channel switching and also avoids sacrificing the coding efficiency in normal channel conditions.

LITERATURE REVIEW

The Mean Absolute Difference (MAD) of the forecast remaining signal and the Encoder Quantization Parameter (QP) act as input parameters to encode (Mansour *et al*., 2011) using the Coarse Grain quality Scalability (CGS) feature. The cloud computing and its current trends for video transcoding in the environment is demonstrated (Chandio *et al*., 2013) but with higher complexity. The effect of video coding artifacts is examined to takes into account the salient motion (Culibrk *et al*., 2011). A dedicated transcoder from AVS to H.264/AVC with reduced resolution aims to provide a fast and consistent solution for transcoding standard-definition videos to mobile contents (Jin *et al*., 2011). A multi-stage process is introduced for accurate motion and mode mapping but the loss of packet occur on varying time range.

REMP, named Scalable REMP (S-REMP), for resourceful delivery of scalable video over IEEE 802.11n WLAN. In S-REMP, different MCS are assigning to dissimilar layers of scalable video to assure the minimal video quality to all users while providing a higher video quality to users exhibiting better channel conditions (Lim *et al*., 2012). REMP and S-REMP fails to adapt the open-source WLAN drivers and calculate the performance in real systems.

Fast rate allocation through Steepest descent (FAST) Method selects an initial (and possibly poor) solution and iteratively progress it until time is exhausted or the algorithm finishes execution (Auli-Llinas *et al*., 2011). JPEG2000-based Scalable Interactive Video (JSIV) separately compress the original video sequence frames and offer for quality and spatial resolution scalability (Naman and Taubman, 2011). JSIV fails to make use of motion compensation including hierarchical B-frames of the Scalable Video Coding (SVC) expansion of the H.264/AVC standard.

A window-level Rate Control algorithm assists with the Traditional Domain Rate-Distortion Model which accomplishes the transaction among picture quality smoothness and buffer smoothness. Meanwhile, the complexity of future frames and global statistics is not available before encoding so the flaw of traditional bit allocation with the assumption of stationary scene cannot be overcome (Xu *et al*., 2011).

The presentation of n-channel symmetric FEC based manifold account coding for a progressive mode. FEC broadcast over the Orthogonal Frequency Division Multiplexing (OFDM) networks in a frequency selective slowly unreliable Rayleigh faded environment. The expressions for the bounds of the throughput and deformation concert in an explicit closed form whereas the exact performance is given by an expression in the form of a single integration (Chang *et al*., 2011).

Broadband Wireless Access Systems (BWAS) depend, among other factors, on their capability to supervise their shared wireless resources in the most efficient way. A novel downlink packet scheduling scheme make use of realistic economic models through the use of novel utility and opportunity cost functions to concurrently gratify the diverse QoS requirements of mobile users and maximize the revenues of network operators (Al-Manthari *et al*., 2009).

A redundant representation of I, P and merge frames are encoded into multiple versions, appropriately trading off expected transmission rate with storage, to facilitate view switching but fail to perform the static view switching into our optimization framework (Cheung *et al*., 2011). An Image-Stitching Method is original utilize to at the same time encode video and depth and then a joint rate control algorithm for MVD is presented (Liu *et al*., 2011). The Joint Rate Control algorithm is absolute on three levels, specifically view level, video/depth level and frame level. In the view level, dissimilar proportions of rates are allocated for dissimilar types of views according to the pre statistical speed allocation. To overcome the packet loss rate and dissimilarity, proposed single and Compound Report Transcoding Method is developed. In summary, the contributions are:

- To mitigate the error propagation due to packet loss in video transfer using Subject Prioritized Retransmission (SPR)
- To resolve the retransmission schedule of a lost packet according to the impact of packet's loss obtained in a front end encoding process
- To reduce the error propagation due to packet loss caused by transient channel using the compound report mode, Channel Alert Compound Report Transcoding (CA-CRT)

SINGLE AND COMPOUND REPORT TRANSCODING METHOD

In this study, the packet loss due to the channel handoff on video quality degradation consisting of single and compound video frames with the description about it is presented with the help of the detailed process involved which is followed by an algorithm.

The video frames emulate an entrance medium which implements a CRC transcoder. In test scenario, each station reports to the transcoder about the packet loss rates and Round Expedition Times (RET) of the channels between the station and the mobile clients it serves. RET is possible to realize the channel handoff due to roaming between the stations and the mobile clients.

Accordingly, the entrance medium maintains a path list table that records clients reached to each base station. These paths are then ranked according to their channel conditions. The status of a path from the station to a client is updated when any new channel information about the client is received. The server does not accept any information about the client from any base station within a pre-specified timeout period. Then the clients are marked as provisionally non-connectable from base station. According to the path-list table, the entrance medium chooses to deliver the video data to the client via either only one channel (i.e., single report) with good quality or two channels (i.e., compound report) with acceptable reliable quality.

Subject Prioritized Retransmission Method: SPR-ARQ is the valuable channel coding method that mitigates the effect of error propagation due to packet loss, especially for video streaming over a packet erasure channel. Conversely, in a fixed-rate channel no additional bandwidth is set aside for resending lost packets. Subsequently the retransmitted packets also participate for the inadequate bandwidth in addition to the regular video packets. Each video packet contributes an unequal importance to the video subject, thereby leading to a dissimilar level of error propagation. Moreover, the retransmission of lost packet should be correctly planned by its importance so as to make the most of the visual quality under the bandwidth limitation. The step by process involved in the subject prioritized retransmission method is shown in the Fig 1.

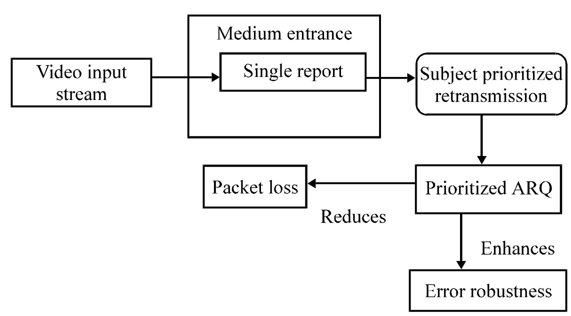

Fig. 1: Processing of subject Prioritized Retransmission Method

Subject prioritized retransmission is based on pre-analyzed packet importance information as a channel coding tool to enhance the error robustness of streaming video. The significance of the packet level is measured by estimating the loss collision value (i.e., the error propagation effect) of each packet. In SPR Method, the measurement is performed offline and the results are stored in the streaming server for guiding the packet retransmission scheduling and decision take place in the entrance medium.

To estimate the impact of each lost packet in single report mode, the pixel-level error propagation effect is characterized using two parameters namely Pixel Position Count (PPC) and Pixel-wise Suppression Error (PSE). SPR Method first calculates the pixel-level Loss Collision (LC) metric as the product of PPC and PSE by:

$$\mathrm{LC}(x, y, m) = \mathrm{PSE}(x, y, m) \times \mathrm{PPC}(x, y, m) \tag{1}$$

where, PPC(x, y, m) represents the pixel position count value for the pixel (x, y) of video frame and 'm' being reference by pixels and similarly PSE(x, y, m) denotes the pixel-wise suppression error for the pixel (x, y) with respect to the reference 'm':

$$\mathrm{PPC}(x, y, m) = \begin{cases} (x', y', m+1) \sum\limits_{\mathrm{points}(x,y,m)} \mathrm{PPC}(x', y', m+1) & 1 \le m \le M_{IG} \\ 1 & m = M_{IG} \end{cases} \tag{2}$$

M_{IG} is the image group size. The continuous video frames within an Image Group (IG) are used in the Movement Compensated Forecast (MCF) process. It is calculated recursively in Eq. 2 by summing up the individual reference counts of pixels in frame m+1 which reference to pixel (x, y) of frame 'm' in the reverse tracking order from the last frame to the first frame of a image group:

$$\mathrm{PSE}(x, y, m) = \left| f(x, y, m) - f(x, y, m-1) \right|^2 \tag{3}$$

In Eq. 3, PSE(x, y, m) denotes the norm of suppression error of pixel (x, y) of frame where f(x, y, m) is the pixel value of pixel (x, y) in frame m assuming the zero-movement error suppression scheme. The motion information to calculate the current frame's macro block-level Error Propagation (EP) is shown in Eq. 4:

$$\mathrm{EB}_{ML}(p, q) = \sum_{(x, y) \in ML_q} \left[\mathrm{LC}\left(x + MV_{x, y}\right) + MV_{x, y, m-1} \right] \tag{4}$$

Where:

p = The Macro-block Level (ML) index in a frame

(x ,y) = The pixel coordinate

q = The time index

(MV_x, MV_y) = The associated motion vector of pixel (x, y)

Finally, EB_{ML} all in one packet are summed up to estimate the packet level error propagation as follows:

$$\mathrm{EP}_p^i(n) = \sum_{q=1}^{M_{ML}} \mathrm{EP}_{ML}(p, q) \tag{5}$$

Where:

i = The packet index of a frame

M_{ML} = The number of Macro blocks Level (ML) in the packet

The packet-level ER measure is taken as the estimation of the packet loss collision value of each packet. The loss judgment of SPR Method is performed offline for pre recorded video streaming request thus resulting in no additional computational complexity while performing real-time broadcast. Besides, the loss estimation is based on the assumption that there is only a single packet loss in IG. The subject prioritized ARQ resolve the retransmission according to packets comparative significance. The experimental results show that the proposed loss estimation is reasonably accurate even when there are multiple lost packets in an IG. The algorithmic flow of the SPR Method is described as:

// Subject Prioritized Retransmission (ARQ):

If the retransmitted packet cannot meet the schedule on client side
    Do not request retransmission
Else
    Request retransmission performed
end if
If the entrance medium receives a retransmission request for p_m^{lost}
    Find p_m with the smallest loss collision in the queuing
    Regular packets with size $(p_m) \geq$ Size p_m^{lost} under the $T_d(P_m^{lost})$ constraint
end if
If p_m exists and the collision value of $p_m^{lost} > p_m$
    Retransmit p_m^{lost} and drop out p_m
Else
    Send the regular packets and ignore the retransmission
    Request for p_m^{lost}
end if

The SPR Method holds additional side information for recording the loss collision rank of each video packet within an IG. According to the packet level loss judgment, developed a subject adaptive prioritized retransmission scheme. The application scenario is that the retransmissions of lost packets from the entrance medium cause the network resource contention between the regular video packets and the protection information. Under constrained scenario, one realistic solution is to drop out some inconsequential regular packets so as to use the saved bandwidth to retransmit the significant lost

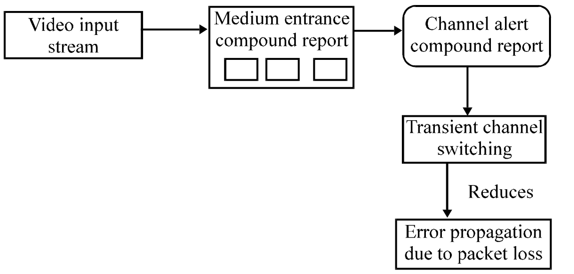

Fig. 2: Processing step of Channel Alert Compound Report Method

packets under a retransmission delay constraint. The level of importance of a packet is measured by its loss collision value defined above.

Channel Alert Compound Report Transcoding Method: The CA-CRT Method is aimed at dealing with the handoff between two or more stations separated by a certain distance. The transitions due to channel switching usually happen across the coverage boundary of the compound stations. As a result, estimating the relative qualities of the compound channels is usually sufficient for the mode decision to choose either a better condition channel or both channels with similar conditions. The process flow of channel alert compound report is shown in Fig. 2.

Channel Alert Compound Report Method initially holds the video input stream. The video input stream contains the compound report. The processing step of CR transcoder and decoder is used in the entrance medium, respectively. A low-complexity compressed domain is composed of three components namely Unpredictable Length Decoder (ULD) Index Mapper (IM) and Unpredictable Length Coder (ULC). The transcoder are replaced with an additional higher performance transcoder with a more complex CRC scheme to improve the visual quality with minimal packet loss. Each base station itself calculates the average RET of the ith channel, RET'_i and the corresponding average packet loss rate, plr'_i, from a station to the client in a sliding time interval with packets as follows:

$$RET'_i = \frac{1}{D}\sum_{j=0}^{D} RET_j^i \qquad (6)$$

$$plr'_i = \frac{1}{D}\sum_{j=0}^{D} PL_j^i \qquad (7)$$

Where:

RET_j^i = The round expedition time of the jth packet via the ith channel
PL_j^i = Corresponding status of loss

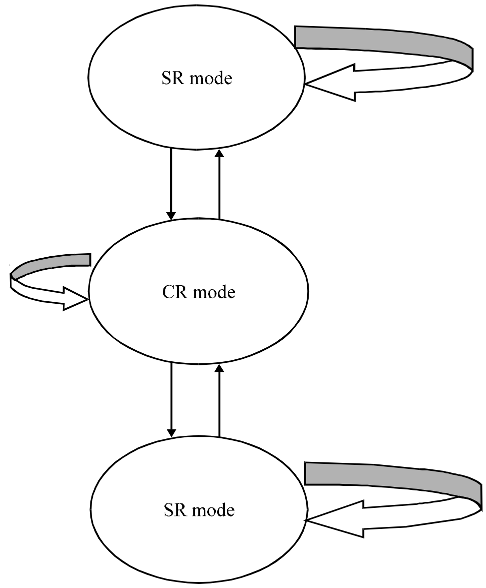

Fig. 3: Single and compound report decision scheme

The entrance medium which is currently operated in the SR mode will at the next stage either switch to the MD mode which sends video data via multiple paths or still stay in the SR mode which sends the data via the same channel. Assuming the entrance medium is originally operated in the SR mode as shown in Fig. 3, via, R_i, packet loss rate plr_j exceeds packet loss rate plr_j by more than a predetermined threshold SR. The entrance medium switches to the CR mode by transcoding the video data and sends the two descriptions via R_i and R_j, correspondingly.

If channel 'j' is not significantly better than channel 'i' in terms of plr, method will compare the RET of the compound channels. If Round Expedition Times (RET_i) is larger than RET_j, the entrance medium switch to the MD mode or else it will stay in the SR mode. The algorithmic steps of the channel alert compound report transcoding is shown as:

//Channel alert compound report transcoding:

SR (R_i): Entrance medium is operated in the SR mode via R_i
plr'_i and RET'_i: Average packet loss rate and Round Expedition Time (RET) of channel 'i'
plr_{SR}, plr_{CR}, k_{SR} and k_{CR}: Predetermined parameters for the round expedition time and packet loss rate in the SR and CR modes ($plr_{SR}>plr_{CR}$)
Start
//SR (R_i) mode
If $plr'_i-plr'_j>plr_{SR}$
    Switch to the CR mode
    Transcode the video data into compound descriptions and send the compound descriptions via several channels separately
Else
    Stay in the SR (R_i) mode
    Directly forward the video data through channel i (via R_i) without transcoding

//CR mode
End if
If plr'_i-plr'_j>plr_{CR}
Switch to the SR (R_j) mode
Forward the video data through channel j (R_j) without transcoding
Else
Stay in the CR mode
Transcoding the video data into compound report and send the compound report via several channels separately
End if

In CA-CRT Method, plr'_i-plr'_j>plr_{CR} in the CR mode indicates the condition of channel 'i' becomes much poorer such that the entrance medium needs to switch to the SR mode via R_j. Note that the lower the values of plr_{SR}, plr_{CR}, k_{SR} and k_{CR}, the higher the frequency of mode switches, thereby making it to reduce packet loss. Appropriate setting of these threshold values according to broadcast surroundings and base-station layout prevent excessive mode switches.

EXPERIMENTAL EVALUATION

Single and Compound Report Transcoding Method with SPR and CA-CRT Method is implemented in MATLAB using the YouTube Multiview Video Games Dataset. YouTube Multiview Video Games Dataset is a multi-variant dataset from UCI repository with integer and real attributes. Proposed method are measured in terms of the PSNR quality accuracy, packet loss rate and bandwidth efficiency.

PSNR quality accuracy of single and compound report mode is defined as the measure of quality of reconstruction of lossy compression codec's (e.g., for image compression). The signal in this case is the original data and the noise is the error introduced by compression. PSNR rate is measured in terms of decibel (dB). Packet loss rate is defined as the rate at which the packet are failed to reach the destination of the video stream. Packet loss rate is measured in terms of percentage (%). Bandwidth efficiency is defined as the video rate transmitted over a given specific communication system. It is measured in terms of kilo bits per second (kbp).

COMPUTATIONAL RESULTS

Single and compound report transcoding method with SPR and CA-CRT Method are compared with the existing 4D perceptual quantization modeling in measuring the PSNR quality accuracy, packet loss rate and bandwidth efficiency.

Figure 4 describes the PSNR rate based on the frame index. The frame index ranges from 50-350. Higher the rate of PSNR higher will be the accuracy level. In order to

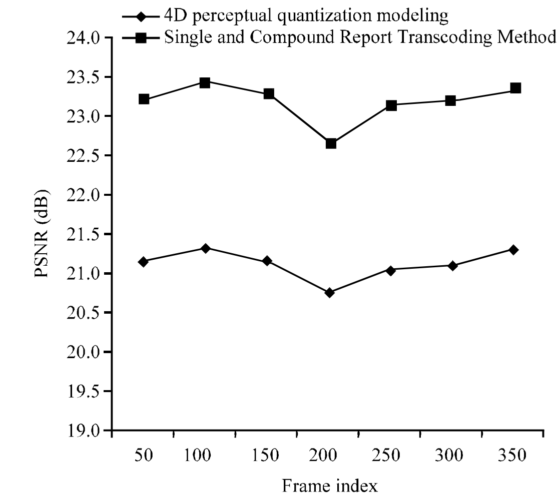

Fig. 4: Measure of PSNR

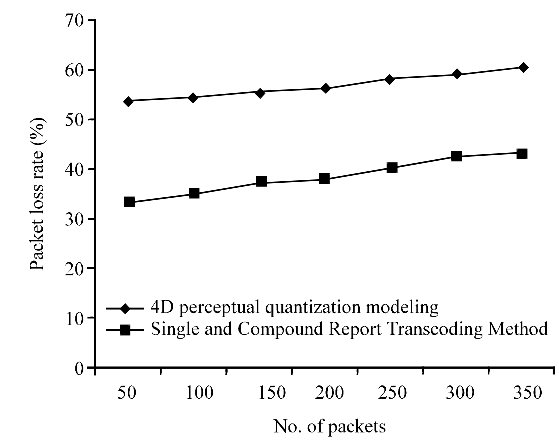

Fig. 5: Measure of packet loss rate

improve the quality accuracy of PSNR rate, uses the number of Macro blocks Level (ML) in the packet. The use of equation:

$$EP_p^i(n) = \sum_{q=1}^{M_{ML}} EP_{ML}(p,q)$$

improves the quality accuracy of the system. The PSNR rate is 5-10% improved in SPR and CA-CRT Method when compared with existing 4D perceptual quantization modeling. Figure 5 describes the packet loss rate based on the packet number. From the Fig. 5, it is evident that the rate of packet loss is lesser using single and compound report transcoding method than the existing 4D perceptual quantization modeling (Huang and Lin, 2007). This is because the packet loss rate reduces using the ER measure of each packet. The loss judgment of SPR Method is performed for pre recorded video streaming request thus resulting in no additional loss rate while

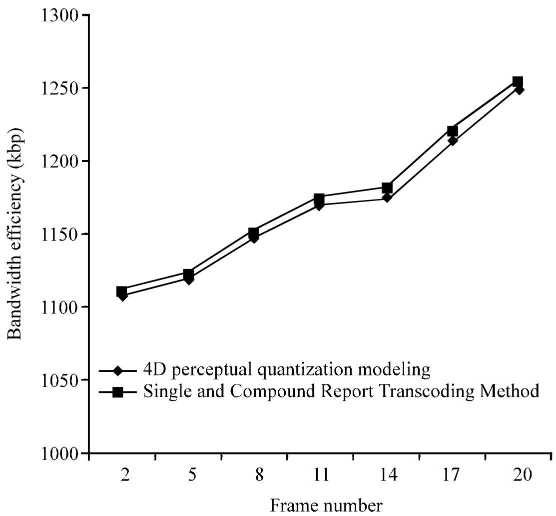

Fig. 6: Measure of bandwidth efficiency

performing real-time broadcast. Packet loss rate is 10-20% lesser in SPR Single Report Method when compared with the 4D perceptual quantization modeling.

Bandwidth efficiency is measured based on the frame number. Based on each frame, the bandwidth varies accordingly. From the Fig. 6, it is evident the average, bandwidth rate is 5-10% improved using SPR and CA-CRT Method when compared with existing 4D perceptual quantization modeling (Huang and Lin, 2007). The SPR Method implies that the retransmitted packets compete with the regular video packets for the constrained bandwidth resources. In the SPR-ARQ Method, the retransmission of a lost packet is scheduled by the packet's significance level. The significance level of a packet is measured by estimating the relative amount of error propagation on packet loss resulting in higher bandwidth rate.

As a final point, Channel Alert Compound Report transcoding scheme try to find an alternative channel with an acceptable channel condition to be used simultaneously for path diversity when detecting mobile client roaming from one station to another. The experimental results show that the proposed method achieves significant visual quality improvement over existing 4D perceptual quantization modeling methods.

CONCLUSION

To resolve the problems related to packet loss and to initiate the flexibility model for error flexibility, Single and Compound Report Transcoding Method is developed using an adaptive error control scheme. The method is comprised of two steps: single report transcoding and compound report transcoding. An efficient error control operation is provided that further divided the operation into the single report and multiple report modes according to channel circumstance. In addition, the Single Report Mode (SPR) Method mitigates the error propagation due to packet loss in video transfer and compound report reduces the error propagation due to packet loss caused by transient channel switching. The SPR Method resolves the retransmission schedule of a lost packet according to the packet's loss impact that is obtained in a front end encoding process. CA-CRT deals effectively with the handoff between two or more stations. The experimental evaluations are conducted in Matlab to obtain the effective result on single and compound report video stream. Experimental results demonstrated with the proposed scheme handle transient channel failures with improved PSNR quality, reducing the packet loss rate to 3.57% and provides better bandwidth.

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