Adaptive MPEG-4 Video Transmission over PSTN Using Buffer Constraints and RTP Feedback

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Abstract

Video transmission over PSTN often suffers from some difficulties such as the bandwidth fluctuation, packet loss and transmission delay. Combining the send buffer and network feedback information, this paper brings forward an adaptive video transmission control algorithm JCBAF to timely notify the encoder to get a better match between the output bit-rate and fluctuated bandwidth. Furthermore, a smooth control strategy is proposed to inform the encoder to achieve a better tradeoff between a lower delay and a good perceptual video quality. Experimental results show that the method can effectively control delay in very low bit-rate video transmission over PSTN or other error prone channels while maintaining the video quality.

1. Introduction

As a traditional communication network, PSTN (Public Switched Telephone Network) has a wide application potential in real-time communication fields such as long-distance surveillance, education and medical treatment, video-conference for its characteristic of low cost and wide distribution. There are still many problems to transmit real-time video over PSTN for the disparity between the huge data stream and the low bandwidth. So how to efficiently transmit video over PSTN becomes an important issue in multimedia communication field.

Video Transmission is resource intensive even after efficient compression using sophisticated algorithms like MPEG-4 [1][2][3]. Moreover, under some unreliable channels like wireless network or PSTN, bandwidth fluctuation or packet loss will be inevitable because of the error prone feature. Furthermore, the mismatch between the encoder output bit-rate and the available bandwidth may lead to the overflow of the buffer, if we take transmission delay into consideration, a long end-to-end delay will become a serious problem.

An adaptive transmission control algorithm, called Joint Control using Buffer constraints And Feedback (JCBAF) is proposed in this paper. The method combines the send buffer and network feedback information together and provides timely message for the encoder to get a better match between the output bit-rate and bandwidth. At the same time, the regulation of output bit-rate should not be at the cost of drastic video quality fluctuation.

The paper is arranged as follows. Section II gives an overview of video transmission system. Section III introduces the background of this work. Problem formulation and algorithms are explained in section IV. Section V conducts the experimental results. Section VI summarizes the paper and makes some discussions.

2. Overview

It is very challenging to transmit video over traditional PSTN network under the delay restriction. While communicating between computers over PSTN, the most common way is to utilize Modems to establish a point-to-point link through serial bus, which can only achieve a very low bandwidth generally. As shown in figure 1, the transmission model indicates that the terminal can get the reconstructed video after decoding from network delivery. The video source produces raw video and puts it into the encoder, the compressed video stream is delivered to network. In this model, the network can be viewed as a black box[4], the delay and the packet loss ratio are essential parameters to reflect network state.

Figure 1: end-to-end transmission model

Even after the efficient compression of the encoder, there still exists a big gap between the output bit-rates and the low bandwidth. So some steps should be taken: firstly, extracting frames at the video source to be selectively encoded to decrease the bit-rate; secondly, permitting a
certain extent delay tolerance to guarantee a fundamental video quality.

3. Transmission Architecture

Figure 2 illustrates the architecture of video transmission system. The encoder compresses input video using MPEG-4 standard, then after encapsulated with RTP header, UDP header, IP header and PPP header, the data is delivered to the network. At the receiver side, through an inverse procedure, the data is input to the decoder for decoding and displaying. At the same time, the Qos monitor at the receiver side records the packet loss information during the specific interval and can feed such information back to the encoder periodically.

4. Problem Formulation

4.1 Buffer constraints

As the video transmission model shown in figure 1, the total delay from video source to the terminal can be formulated as follows [5][6]:

\[ D_T = D_e + D_{sb} + D_c + D_{rb} + D_d \]

In the above equation, \( D_e \), \( D_{sb} \), \( D_c \), \( D_{rb} \) and \( D_d \) denotes encoder delay, send buffer delay, channel delay, receive buffer delay and the decoding delay respectively.

As for video transmission over PSTN, once the session is established, the communication line can be treated as a fixed circuit. So the channel delay \( D_c \) is comparatively constant. Experiments show that the send buffer delay \( D_{sb} \) occupies the majority of total end-to-end delay \( D_T \). Therefore, the send buffer constraint becomes an important issue to be studied.

A suitable \( D_{sb} \) size should be defined firstly. The buffer size mainly depends on the delay tolerance of specific applications. For very low bit-rate video communication systems, the frame rate is rather low in order to effectively decrease the bit-rate, so it is inappropriate to skip more frames among the selected frames. Therefore, a certain delay tolerance can effectively avoid the mentioned problem. At the same time, the following aspects should be taken into consideration. An oversize buffer may increase the delay when the encoder produces excessive bit-rate due to drastic motion, while a small size buffer may incur frequent buffer overflow, which leads to the passive data loss and thus severe degradation of video quality.

Considering the character of surveillance systems and a proper end-to-end delay over the specific network (like PSTN), the reasonable send buffer size can be defined as follows:

\[ S_{sb} = \frac{1}{8} D_p \cdot BR_{av} \]

where \( BR_{av} \) represents the average output bit-rate, it should coincide with the available network bandwidth, \( D_p \) represents the permissible maximum delay.

\[ \alpha S_{sb} \leq (1 - \alpha) S_{sb} \]

4.2 Feedback Control

The characteristic of video transmission applications is data intensive, while some video communication systems like surveillance systems have a strict requirement of delay, which determines that we must choose connectionless transport level protocol UDP to achieve real-time. As we know, UDP does not provide the quality of service (Qos) guarantee. Based on the special requirement to transmit real-time multimedia, IETF defined RTP/RTCP protocol to transmit real-time multimedia on Internet [7][8] (see RFC 1889, RFC 1890 in detail).

RTP and RTCP are designed independent of the underlying transport and network layers. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data over network services. It does not address resource reservation and cannot guarantee Qos for real-time services. Real-time control protocol (RTCP) provides an important way to
monitor data delivery and the network state through analyzing the feedback information (Receiver Report) from the receiver, and provides necessary control functionality.

Generally, there are some parameters like delay, delay jitter and packet loss ratio to reflect the network state. Here we use the packet loss ratio calculated at the receiver as the parameter to estimate the network state.

Assuming that \( S_1, S_2, \ldots, S_t \) is the sending sequence number at time \( t \), \( R_1, R_2, \ldots, R_t \) is the receiving sequence number at time \( t \), So the packet loss ratio can be represented as:

\[
\text{Loss}_{\Delta t_{i,j}} = \frac{C^\text{empty}_{\Delta t_{i,j}} - C^\text{real}_{\Delta t_{i,j}}}{C^\text{empty}_{\Delta t_{i,j}}}
\]

where \( C^\text{empty}_{\Delta t_{i,j}} \) denotes the packet number that should receive from the sender during time interval \( \Delta t_{i,j} \), \( C^\text{real}_{\Delta t_{i,j}} = S_{ij} - S_{i-1} + 1 \)

Another parameter \( C^\text{real}_{\Delta t_{i,j}} \) denotes the packet number that really gets from the sender during time interval \( \Delta t_{i,j} \), \( C^\text{real}_{\Delta t_{i,j}} \) can be calculated through a counter at the receiver side.

4.3 Joint Control using Buffer constraints and Feedback (JCBAF)

As for real-time video communication, calculating the round-trip delay is impractical for it will conspicuously increase the network payload. So we choose the send buffer delay to indicate the mismatch between the output bit-rate and network bandwidth, and use packet loss ratio in feedback information calculated at the receiver to reflect network state.

In fact, the value of \( D_{sb} \) reflects the abound extent of the send buffer, a bigger \( D_{sb} \) means that the send buffer is facing the overflow danger. In addition, the feedback information \( \text{Loss}_{\Delta t_{i,j}} \) can indirectly represent the channel state. So \( D_{sb} \) and \( \text{Loss}_{\Delta t_{i,j}} \) can be used as parameters to notify the encoder to regulate the encode strategy to achieve joint activity between encoder and send buffer. The idea is that when the send buffer reaches the overflow value, notifies the encoder to adaptively decrease the output bit-rate. Feasible steps mainly include increasing the quantization parameters \( Q_p \) or skipping frames.

Assuming that we can get the channel bandwidth in priori, let average output bit-rate \( BR_{av} \) and the initial encoder output bit-rate equal with the network bandwidth. Denote \( L_{\text{threshold}} \) as the packet loss ratio threshold, \( D_{sb-1}, D_{sb-2} \) and \( D_{sb-3} \) (\( D_{sb-1} < D_{sb-2} < D_{sb-3} \)) as 3-level threshold values to control the send buffer. The value of \( D_{sb-1}, D_{sb-2} \) and \( D_{sb-3} \) can be calculated by the follows:

\[
D_{sb-1} = 8 \cdot \alpha \cdot S_{sb} / BR_{av} \\
D_{sb-2} = 4 \cdot S_{sb} / BR_{av} \\
D_{sb-3} = 8 \cdot (1 - \alpha) S_{sb} / BR_{av}
\]

Algorithm:

\[
\begin{align*}
&\text{If}( D_{sb} > D_{sb-3} \&\& \text{Loss}_{\Delta t_{i,j}} > L_{\text{threshold}} ) \\
&\text{Frame Skip}; \\
&Q_p = Q_p_{-\text{min}}
\end{align*}
\]

\[
\begin{align*}
&\text{If}( D_{sb-1} < D_{sb} < D_{sb-2} ) \\
&Q_p_{+} = 2; \\
&B = B_{av} +
\end{align*}
\]

\[
\begin{align*}
&\text{If}( D_{sb} < D_{sb-1} ) \\
&Q_p_{-} = 2; \\
&B = BR_{av} -
\end{align*}
\]

4.4 Packetization Strategy

Appropriate packetization strategy can achieve a good transmission performance. RTP provides the sync solution of multi-stream, including video and audio, or multi video and audio. After adding the RTP header to the payload, then the UDP header, IP header will be added sequentially until the data is delivered to the data link layer. As we known, different networks have different limit about the length of pass-through data, which is called path maximum transfer unit (\( pMTU \)). As for the traditional PSTN network, the digital link is established through serial lines using PPP protocol. Specifically, here the \( pMTU \) is equal to the \( MTU \) of PPP links, which is a very small value compared with a frame length.

Dapeng Wu etc.[3] proposed an efficient and robust packetization algorithm in sync layer for internet transport. But unlike the dramatic bandwidth fluctuation over internet, the PSTN can get a relatively stable available network resource, so the packet loss ratio can be controlled in a low level. So we define the packetization algorithm:

\[
pMTU = pMTU - \text{UDP Header} - \text{IP Header};
\]

\[
\begin{align*}
&\text{If}(\text{find the VOP_start_code}) \\
&\text{if}(\text{the VOP_len} < pMTU) \\
&\text{packet_len} = \text{VOP_len};
\end{align*}
\]

...
else if(remaining_len < pMTU )
    packet_len = remaining_len;
else do
    { packet_len = pMTU ;
      remaining_len = VOP_len - packet_len;
    }while(find the next VOP_start_code);

5. Experiments

We test our proposed algorithm with a surveillance system (AVSS), which adopts JCBAF as the transmission and control algorithm. The encoder is based on MPEG-4 verification model. Some important parameters for the test are as follows. Resolution: CIF 352*288. Frame rate: 3frames/sec. Frame sequence: I 1P 2P 3P… 27P output bit-rate: 30kbps. The values of some parameters appeared in the paper are listed as follows: \( D_p = 10s \), BR=30kbps, \( \alpha = 0.1 \), \( S_{ab} = 40\, kbyte \), \( L_{threshold} = 0.25\% \), \( pMTU = 296\, byte \). The sequence is an indoor scene alternating with motion. Figure 4 shows the results of send buffer delay under usual method and JCBAF. At the same time, the corresponding frame length is listed in Figure 5.

![Figure 4: send buffer delay](#)

![Figure 5: frame length](#)

6. Discussions

Combining the send buffer information and network feedback information, an adaptive video transmission control algorithm JCBAF is proposed in the paper to get a better match between the encoder output bit-rates and bandwidth. Furthermore, the encoder can achieve a better tradeoff between lower delay and a good perceptual video quality under this regulation strategy. We also develop a simple but effective packetization algorithm in this paper. Experimental results show that the method can effectively control delay in very low bit-rate video transmission over PSTN, and may also suitable for other error prone channels as well.

It should be pointed out that when some drastic motion appears, the encoder will increase its quantization parameter or even skip frames to maintain the output bit-rate at a low level, which may inevitably lead to the decrease of video quality. Further job should be done in rate control aspect because it is a valuable issue to study how to combine the network and encoder well in very low bit-rate video transmission.

References


[7] RFC1889, Internet RFC/STD/FYI/BCP Archives

[8] RFC1890, Internet RFC/STD/FYI/BCP Archives