Max-Packet Generation Process For Removing Skew In Network Multimedia Communication- A Mathematical Model

S.Kandar[a], K.Roy[b], M.Barman[c], C.T.Bhunia[d]
[a,b,c]*Haldia Institute of Technology*
P.O.-HIT(W.B, India), PIN-721657
E-mail: shyamalenduk@yahoo.com
[d]*SMIEEE, Senior Associate ICTP, Italy, Director, BITM, Bolpur(W.B)*
E-mail:ctbhunia@vsnl.com

Abstract. Multimedia data are sensed by human. These types of data are error tolerable to some extent but delay intolerable. To provide multimedia services with a guaranteed QoS is a research challenge. The two important parameters that degrade QoS are jittering & Skew. Researchers studied different methods for reducing jittering & skew. Earlier investigations attempted to use buffer management and introducing variable delay in delivering buffer to meet the challenges of jittering and skew. In the current work, we investigate Max-packet (Large packet made of data of different services) to reduce the effect of skew.

Keywords: Skew, Max-Packet, Combined Packet, Only Video packet.

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1. INTRODUCTION

In order to achieve some guaranteed QoS, two issues that are paramount importance particularly for continuous bit rate (CBR) services are: jitter and skew [1-4]. Jitter is caused due to the variable delay that occurs during transmission through network between the packets of a particular service, say only for audio or only for video. Jitter can be removed by providing a compensation buffer at the receiving side.

Skew refers to the variable delay between the two (or more) corresponding packets of two (or more services) during transportation in the network.

At the transmitter side say for two media components Audio & Video, Pi is the packet for Video and pi is the packet for Audio. At the time of transmission at the transmitter side say T is instant of transmission of K th sample of media1(Video). So in the transmitter side T will also be the instant of transmission of K th sample of media2(Audio). At the receiver side say at T/ instant of time K th sample of media1 (Video) is received. Say at T// instant of time the K th sample of media2 (Audio) is received. Skew will occur when T/ ≠ T// [Fig: 1].

If skew occurs, at the receiver side there will be a mismatch between audio & video. This will affect the user in realizing the multimedia data. This type of mismatch occurs due to different types of delay in the network. In order to remove skew and jittering, different researchers [7-8] investigated different techniques. From implementation point of view, these techniques are hard to provide guaranteed QoS. Basically they are supposed to provide a pre defined perceived Quality of Service.

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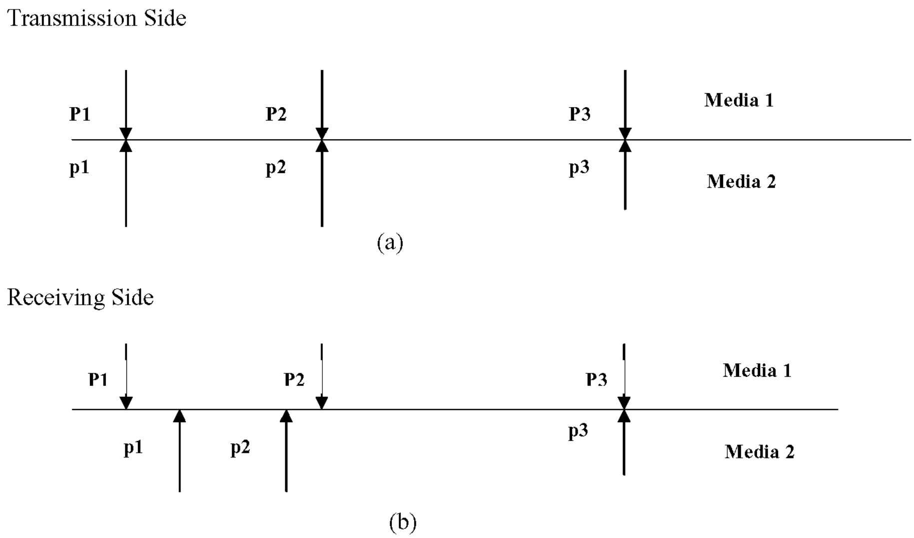

FIGURE 1. Occurrence of Skew

2. DISCUSSION ABOUT THE PROPOSED MAX-PACKET

Skew occurs due to the variable delay between the two (or more) corresponding packets of two (or more services) during transportation in the network. To remove skew we propose the concept of Max packet. Our goal is to provide guaranteed QoS. If we group the samples of two or mode media components generated with in a same time instant in a single packet then at the receiver side the variable delay between two or more corresponding packets of two or more services will not appear during transportation in the network. If we consider two media components as voice & video then the corresponding data of two Medias generated within a same time instant will be in the Max-Packet. The concept of Max-packet was studied elsewhere [5]

If a multimedia data consists of only audio & video then with in a same time instant it is seen that size of video data is much more than the size of audio data. For video the least frequency of a video signal is 33.4 MHz, where as for audio the maximum audible audio frequency is 20 KHz. For some instant of time the size of data for both of the components can vary. So the Max-Packet will be of variable length. In a Max-Packet there are more than one media components [For our case it is two, Audio & Video], so there must be a some extra bits in between two media data in the Max-Packet to differentiate between the two media components at the receiver side. As in the Max-Packet data of more than one media exist there must be some bits to indicate each type of media data within the packet.

In multimedia there are 6 types of multimedia data [Text, image, Graphics, Audio, Video & Animation], but those can be grouped into 4types [Text, Image, Audio, Video]. So two bits are sufficient to identify the media components [00, 01, 10, 11]. Yet we shall take 3 bits means 8 combinations. Say for 000 for Audio, 001 for Video, 011 for Text, 111 for Image. This bit pattern will be known to both sender & receiver side.

Flag(1 byte)	Address(1 to several byte) [Depends on the number of nodes	Control bit (1 or two byte)	Voice (Variable Length)	Stuffed bit	Video (variable length)	Frame check sequence(2 to 4 byte)	Flag(1 byte)

FIGURE 2. Variable Length MAX-Packet

3. MAX-PACKET GENERATION PROCESS

On a laboratory scale Audio signal of frequency 20 KHz and video signal of 40 MHz are taken. These signals are sampled separately just at Nyquist rate (40 KHz for Audio and 80MHz for video) followed by conventional process of PCM (Pulse code modulation).Then the samples are needed to be quantized. In order to maintain the synchronization between audio and video signals the quantization level (**Q_l**) must be kept same for both of the signals. After quantization, the signals will be digitized. These digitized values will be put in packets. To remove skew the digitized values of the samples of audio & video with in a given time instant will be put in a single packet. For developing the packet, one media will be taken as Master and another will be taken as Slave. In our work Audio is taken as Master and Video is taken as Slave. In our research there is a frequency difference of 2 X 10^3 times to video signal than that of audio signal. That means if two audio samples are taken at time instant t_1 & t_2 , then for these two audio samples there will be $(t_2 - t_1)$ X $(2 * 10^3)$ number of video samples provided $t_2 > t_1$. If a combined packet is produced by taking the audio and video samples between t_1 & t_2, the packet size will be huge. By using the large packet the benefit of packet switching will not be obtained.

This can be handled in another way. One combined packets will be made from the quantized values, obtained from the samples at time instant **t_1** ; and in time instant **t_2** another combined packet will be made. Between **t_1 & t_2** there will be (n-1) number of quantized slave sample, i.e. only video samples. Where n= **C_f video/C_f audio**, **C_f** stands for carrier frequency. m number of video samples will be grouped to form a Only Video packet. For (n-1) video samples number of packets will be

$$\left\lceil [(n-1)/m] \right\rceil \qquad (1)$$

For each packet [Combined or only video] there will be a header of 1 byte i.e. 8 bit. For combined packet the header is 00000000 & for only video packet the header is 11111111. Between two combined packets the only video packets will be numbered to do synchronization at the receiver side. There will no packet containing only audio data.

4. BLOCK DIAGRAM FOR MAX-PACKET GENERATION

From the discussion in the previous topic it is clear that the total process is divided into several stages.

a) Sampling of Audio signal
b) Sampling of Video signal
c) Combined packet Formation
d) Only video packet Formation

a) Sampling of Audio signal

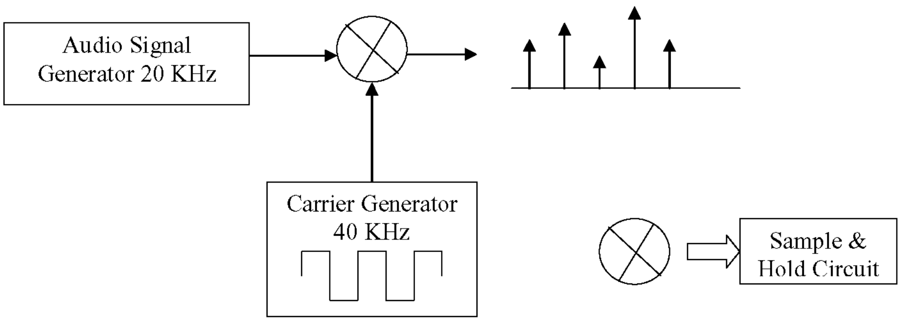

FIGURE 3. Sampling process of Audio signal

b) Sampling of Video signal

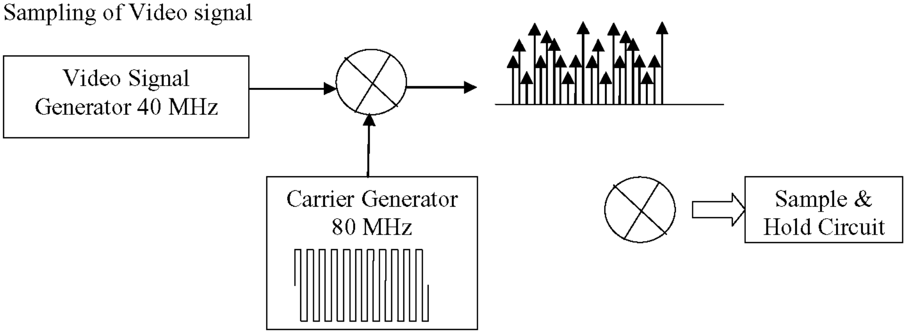

FIGURE 4. Sampling process of Video signal

c) Combined packet Formation

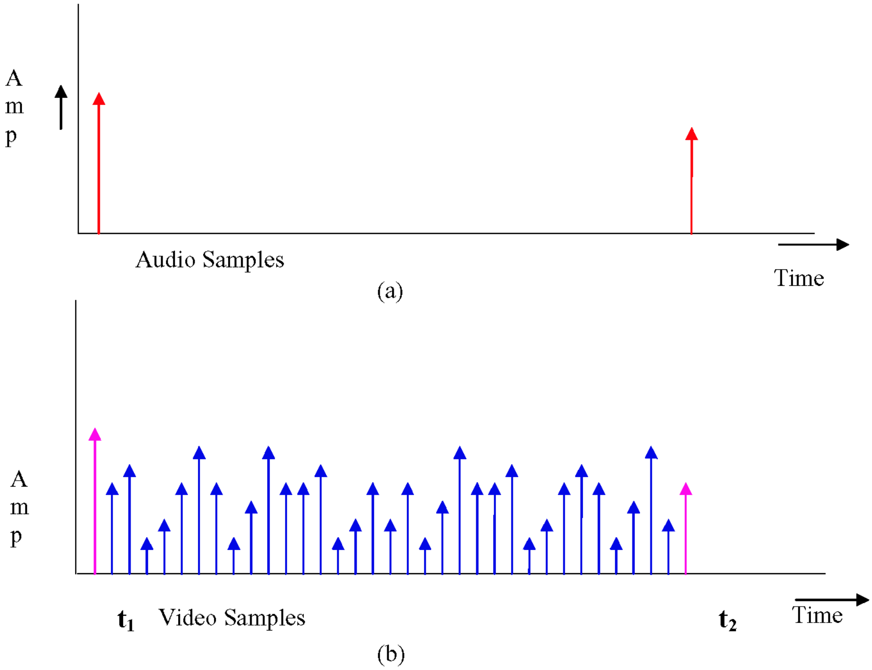

FIGURE 5. Combined packet Formation

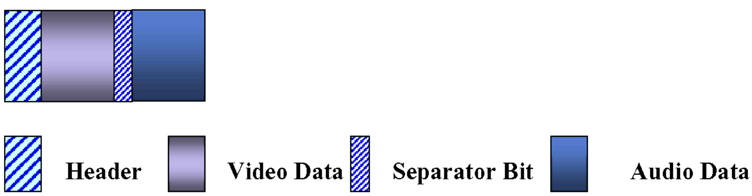

FIGURE 6. Combined Packet containing Video & Audio data

d) Only Video Packet Formation

FIGURE 7. Only Video Packet

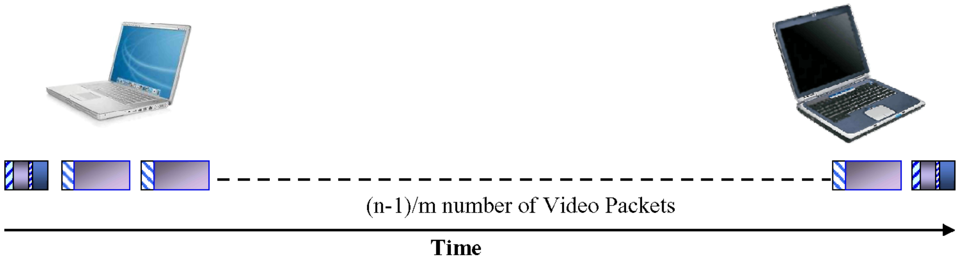

FIGURE 8. Packet Transmission from Source to Destination

5. CALCULATION OF OVERHEAD BITS

Overhead bits means how many number extra bits are sent with the original data at the time of packets transmission from sender to receiver. Overhead bits are added in the form of header of the packet or any stuffed bits. This calculation is necessary because number of overhead bit increases means size of the data increases means the transmission time from source to destination increases.

Let the multimedia data is transmitted within a time period T.

After a certain time period audio and video data are grouped and forms a combined packet. Let such type of first packet is formed at the time instant t_1 and the second such type of packet is formed at the time instant t_2.

So total number of combined packets within the time period T is $T/(t_2-t_1)$.

For each such type of combined packet the overhead bits are in the formed of Headed (h) and Separator bit (s).

So number of overhead bits for combined packets are

$$[T/(t_2-t_1)]*(h+s)\,] \qquad (2)$$

Within the time period $(t_2 - t_1)$ there are

$\lceil[(n-1)/m]\rceil$ number of Only Video packets. Where n= quantization level and m= number of video samples in a single Only Video packet.

Total number of Only Video Packets within the time period T is

$$[T/(t_2-t_1)] * \lceil[(n-1)/m]\rceil \qquad (3)$$

For each Only Video packet overhead bit is header(h). So total number of overhead bits for Only video packets is

$$h*[\{T/(t_2-t_1)\} * \lceil[(n-1)/m]\rceil\,] \qquad (4)$$

So, during transmission of the total multimedia data the overhead bits are

$$[\{T/(t_2-t_1)\} * (h+s)] + h * [\{T/(t_2-t_1)\} * \lceil [(n-1)/m] \rceil]$$

$$= [T/(t_2-t_1)] * [h+s+h * \lceil [(n-1)/m] \rceil] \qquad (5)$$

We like to make it future work a comparative gain of QoS with respect to overhead bits. The overhead bits provide a trade off with expected QoS.

6. CONCLUSION

We have proposed a concept of Max packet, which is proposed to eliminate effect of skew. By this technique skew will be totally removed but due to network delay there may occur jittering of video packets. We believe if the technique is implemented with other techniques namely accelerating & deaccelerating or buffer management technique of removing skew & jitter; we may derive it as a sound technique which will successfully remove skew & jittering in network multimedia communication and will provide Quality multimedia service. The proposed technique is under practical implementation & testing.

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