# A Novel Scheme for Streaming Multimedia to Personal Wireless Handheld Devices

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**Abstract** — *Delivering streaming video over wireless is an important component of many interactive multimedia applications running on personal wireless handheld devices. Such personal devices have to be inexpensive, compact, and lightweight. Wireless channels have a high channel bit error rate and limited bandwidth. Delay variation of packets due to network congestion and the high bit error rate degrades the quality of video at the handheld device. Multimedia application use a buffer at the handheld device smooth out the delay variation and improve the quality of streaming video. However, the buffer size has to be kept small in order to reduce the size, weight and power consumption of handheld devices. In this paper, we propose a novel selective retransmission scheme for multimedia transmission over wireless networks. Our scheme is based on retransmission of only the most important information in a video in order to achieve a high quality of video. Our objective is to develop a simple cost effective scheme which offers an acceptable video quality over a noisy wireless channel using a small buffer size at the handheld device. We have developed an analytical model to determine the networking requirements, video quality as a function of video compression parameters and network error conditions, and optimally dimension the buffer at the handheld device.*

**Index Terms — Wireless handheld device, Streaming video, Personal Digital Assistant, Handheld device desgin.**

## I. INTRODUCTION

The transmission of multimedia over wireless channels to mobile users is becoming a research topic of growing interest [1-6]. With the emergence of small wireless handheld devices (such as PDAs), it is expected that interactive multimedia will be a major source of traffic to these handheld devices. These devices could be carried by users inside buildings when they are connected by wireless Local Area Network (LAN) or in vehicles when they will be connected to the cellular network, such as GPRS [7-8].

Wireless transmissions use radio as the transmission media. Radio links connect users to base stations which are connected to routers using wired links as shown in Figure 1. The wireless segment provides mobility to a user while using the network. In contrast to wireline transmission links where the bandwidth can be easily increased and the channel quality can be guaranteed, the bandwidth of a wireless channel is limited because of spectrum allocation and physical limitations. The transmission quality of radio is easily affected by environments such as buildings, moving objects, and atmosphere, etc. Moreover, because of the mobile nature of the users, the access point of a mobile user changes continuously. All these factors in wireless networks give rise to issues such as effective bandwidth allocation, high channel bit error rate, and user handover.

Multimedia is an emerging service that integrates voice, video and data in the same service. Networked multimedia applications include remote education, video-on-demand, tele-shopping, home games, entertainment, etc. Because of traffic characteristics such as high bit rate, video will be the dominant traffic in multimedia streams and hence needs to be managed efficiently [9]. For efficient utilization of network resources, video must be compressed to reduce its bandwidth requirement. Although there exist several compression techniques, MPEG [10] is one of the most widely used compression algorithms for networked video applications.

A wireless handheld device, such as Personal Data Assistant (PDA), can integrate voice, video, and data in one device. In contrast to data, multimedia can tolerate a certain level of error. Therefore, although a wireless network has a high bit error rate when compared to a wireline network, it is possible to cost effectively transmit multimedia over wireless networks with acceptable quality of service (QoS).

The authors in [11] studied the go-back-$W$ retransmission scheme with window size $W$ for the experimental studies to evaluate the effectiveness of video transmission over a wireless channel. They also considered the receiver buffer size for single retransmission. Their experimental results have shown that retransmission without FEC can improve the quality of video transmission. However, their studies did not present detailed theoretical model and analysis of their retransmission scheme. Since the effectiveness of a retransmission scheme depends on network parameters such as channel bit error rate, channel transmission rate, and video parameters such as the video frame size, it is very important to study the effect of these parameters on the effectiveness and efficiency of retransmission schemes. Moreover, to sustain continuous video display at the wireless handheld device (receiver), continuous monitoring of the network status is required to prevent transmission collapse. Finally, during the time taken for retransmission, the handheld device can only decode the buffered data upto the errored frame. To prevent starvation at the handheld device (resulting in frozen display), the device buffer must have a minimum fill level to sustain

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continuous display during retransmission. The minimum fill level thereafter decides the minimum handheld device buffer size. The longer the retransmission time, the larger the minimum fill level. Since the retransmission time depends on the channel bit error rate, the minimum device buffer size depends on the channel bit error rate. The authors in [12] have reviewed several error resilient video coding methods for MPEG-2 transmission over wireless networks. They found that in a WAN testbed, the wireless packet losses occur independently.

The high bit error rate of wireless channels raises a number of challenging issues for multimedia transmission over a wireless network. Such issues include techniques to ensure a continuous display at the handheld device in spite of errors in the received video. Currently, errors are compensated mainly by link layer Forward Error Correction (FEC) which employs redundant bits to passively correct errors at the handheld device. Although FEC can correct some errors without retransmission, it needs extra bandwidth to transmit a lot of redundant bits for correction. This lowers the utilization of bandwidth, which is expensive for wireless channels. Furthermore, implementing FEC requires a lot of computing power at the handheld device, thereby increasing the cost and power consumption of the handheld device. Additionally, complicated error concealment algorithms will have to be implemented at the handheld device to handle different error types [13]. All of the above issues may prevent cost effective wide scale commercial deployment of wireless multimedia receiver if FEC is used at the handheld device. Therefore, new schemes for handling errored packets at the handheld device are required for widespread deployment of interactive multimedia applications on handheld devices using noisy wireless links.

In this paper, a new *selective retransmission scheme* is proposed for multimedia transmission over noisy wireless channels in order to ensure acceptable video quality at the handheld device and allow the design of a cost effective handheld device. Since multimedia can tolerate a certain level of error, with little effect on picture quality, our scheme sets a threshold for the number of errored packets, and to decide whether retransmission is required. To prevent the handheld device from starvation (i.e. ensuring a continuous display), a buffer is required at the handheld device.

Our *objective* is to develop an efficient error handling scheme for wireless transmission of multimedia to a handheld device, and to show that the scheme results in a cost effective device. The *significance* of our scheme is that it does not require powerful computing capability and complicated error correction algorithm to be implemented at the handheld device. The handheld device will require simple hardware and thereby reduce the cost of the handheld device. Our scheme also offers a method to monitor the status of the transmission system to check if the conditions to maintain transmission are satisfied. Our performance measure is maintaining a continuous display at the handheld device

The rest of the paper is organized as follows. In Section II, we propose the system model and our proposed selective retransmission scheme. Analysis of a multimedia wireless system using our proposed scheme is performed in Section III. In Section IV, the minimum handheld device buffer size is derived. Numerical results obtained from our analysis are given in Section V, followed by conclusions in Section VI.

## II. System Model and Proposed Retransmission Scheme

In this section, we describe the system model and the operating principle for our selective retransmission scheme. The model will be used in Section III to determine values of network parameters and the size of the receiver buffer to sustain continuous display at the handheld device. As shown in Figure 1, the system consists of a multimedia server and receivers connected by wireless links and routers.

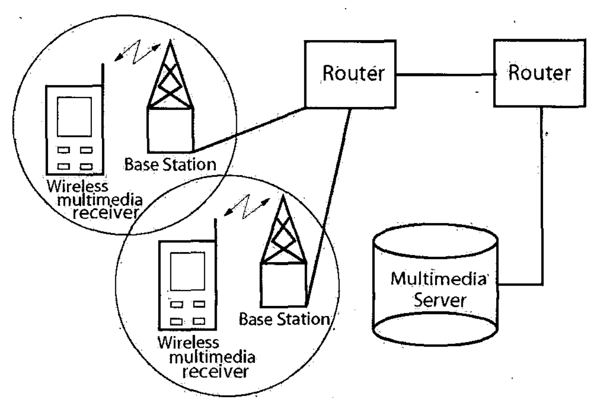


**Figure 1. A wireless handheld device connected to a multimedia server through routers and base stations.**

MPEG compressed video consists of I, P and B frames. The frames are grouped to form a special structure called Group of Picture (GoP). Each GoP includes an I frame followed by a number of P and B frames. A GoP is denoted by *MmNn* which represents a total of $n$ frames in the GoP with $m-1$ number of B frames between the I/P or P/P frames. At the decoder, error in a received I frame will affect all the frames in the GoP, while error in a received P frame will affect the quality of the subsequent P and B frames. An error in a received B frame only affects its own quality. Therefore, I frame is the most important, while the B frame is the least important. To maintain the quality of video at the handheld device, *we propose the following retransmission scheme based on errors in the received video stream at the* handheld device.

- If the number of errored packets in an I frame is higher than an acceptable error threshold for the I frame, the packets belonging to the I frame are discarded, and the I frame is retransmitted from the multimedia server;

- If the number of errored packets in a P frame is higher than the acceptable error threshold for the P frame, the P frame is discarded and is retransmitted;
- Errored packets in B frames are discarded and are not retransmitted.

In the next section, we develop a mathematical model to analyze our proposed scheme, and to derive the network requirements to sustain a continuous video display at the handheld devices. The model will also be used to determine the minimum size of the buffer at the handheld device. In our discussion, we assume:

- The wireless channel has a bit error rate which is independent of the traffic pattern and the channel transmission rate;
- Each bit has equal probability of being errored during transmission. Moreover, each packet has equal probability of having errors;
- The channel has fixed round trip time between the server and handheld device;
- The amount of bandwidth available to the application changes over time depending on the level of network congestion in the wired part of the network, and other users competing for the wireless bandwidth.

## III. SYSTEM ANALYSIS

In this section, we develop a mathematical model to analyze our proposed retransmission scheme. First, we define the following variables.

### A. Notation

- $\rho$ : Channel bit error rate of the wireless link;
- $S_I$, $S_P$, and $S_B$ Average size of I, P and B frames respectively of MPEG video;
- *N*: Number of bits in the payload of a packet;
- $N_I$, $N_P$ and $N_B$: Average number of packets belonging to I, P and B frame in a GoP;
- $d_I$, $d_P$: Threshold for the acceptable number of errored packets in I and P frames respectively;
- *p*: Probability that a packet has error;
- $W_I$: Probability that an I frame is errored given that an error has occurred;
- $W_P$: Probability that a P frame is errored given that an error has occurred;
- $P_I(d_I)$: Probability that an I frame needs to be retransmitted, called *I frame transmission failure probability*;
- $P_P(d_P)$: Probability that a P frame needs to be retransmitted, called *P frame transmission failure probability*;
- $T_d$ :Fixed round trip time (FRTT) from server to handheld device;
- $T_r$: Average successful retransmission time for an errored frame, called the *average system recover time interval*. This is the average time required for successful retransmission;
- $\tau_f$: Average time interval between consecutive frame errors, called *average system failure time interval*;
- $\lambda$ : Channel transmission rate;
- $\mu(t)$ : Multimedia playback rate at time *t*;
- $C_{\min}$ : Minimum receiver buffer size.

### B. Average System Failure and Recovery Time

Since the probability of a bit being in error is $\rho$, the probability that the payload of a packet has no error is $(1-\rho)^N$. Therefore, the probability (*p*) that a packet is in error can be expressed as:

$$p = 1-(1-\rho)^N \tag{1}$$

The probability $P_I(d_I)$ that an I frame needs to be retransmitted is the probability that the number of errored packets in the frame is higher than the threshold $d_I$. Therefore, $P_I(d_I)$ can be expressed as:

$$P_I(d_I) = 1-\sum_{i=0}^{d_I}\binom{N_I}{i}p^i(1-p)^{N_I-i} \tag{2}$$

where $\sum_{i=0}^{d_I}\binom{N_I}{i}p^i(1-p)^{N_I-i}$ represents the probability of the I frame having upto $d_I$ errored packets. Similarly, $P_P(d_P)$ can be expressed as:

$$P_P(d_P) = 1 - \sum_{j=0}^{d_P} \binom{N_P}{j} p^j (1-p)^{N_P - j} \quad (3)$$

Given that a frame is in error, the probability that the frame is an I frame is proportional to the size of the I frame in a GoP. Therefore, $W_I$ can be expressed as:

$$W_I = \frac{S_I}{S_I + S_P\left(\frac{n}{m} - 1\right) + S_B \frac{m-1}{m} n} \quad (4)$$

where the denominator represents the size of a GoP. Similarly, we can obtain the expression for $W_P$ as:

$$W_P = \frac{S_P}{S_I + S_P\left(\frac{n}{m} - 1\right) + S_B \frac{m-1}{m} n} \quad (5)$$

If an I frame is in error, the average time $T_r^I$ for successful retransmission is expressed as:

$$T_r^I = \left(T_D + \tau_s^I\right)\left(1 - P_I(d_I)\right) + 2\left(T_D + \tau_s^I\right) P_I(d_I) \times \left(1 - P_I(d_I)\right) + \dots\dots \quad (6)$$

$$= \frac{T_D + \tau_s^I}{1 - P_I(d_I)}$$

where $\tau_s^I = \frac{S_I}{\lambda}$ is the time required to transmit an I frame into the wireless channel. Similarly, $T_r^P$ for successful retransmission of a P frame can be expressed as:

$$T_r^P = \frac{T_D + \tau_s^P}{1 - P_P(d_P)} \quad (7)$$

where, $\tau_s^P = \frac{S_P}{\lambda}$ is the time to transmit a P frame into the channel.

Therefore, the average system recover time $(T_r)$ is the statistical sum of the I and P frames recover times $T_r^I$ and $T_r^P$ with weights $W_I$ and $W_P$ respectively. Therefore, $T_r$ can be expressed as:

$$T_r = T_r^I W_I + T_r^P W_P \quad (8)$$

Because a wireless channel has significant error rate, let's say that an error happens after time $\tau_f$ which is defined as the average system failure time. Since the acceptable error thresholds are $d_I$, and $d_P$ for I and P frames respectively, the average system failure time is:

$$\tau_f = \frac{\min(d_I, d_P)}{\lambda_p} \quad (9)$$

where, $\lambda$ is the available bandwidth of the connection.

An interactive multimedia application at the handheld device can perform VCR like interactive functions such as stop, playback, fastforward (FFW), and fastbackward (FBW) which are represented by a state transition diagram as shown in Figure 2. In the FFW/FBW state, the handheld device will consume data at a higher speed than the playback state. Therefore, in response to a FFW/FBW request, the sever will request higher bandwidth than is required in normal playback. The bandwidth, $\lambda$, required by the server is decided by the level of interactivity of the handheld device and the available bandwidth.

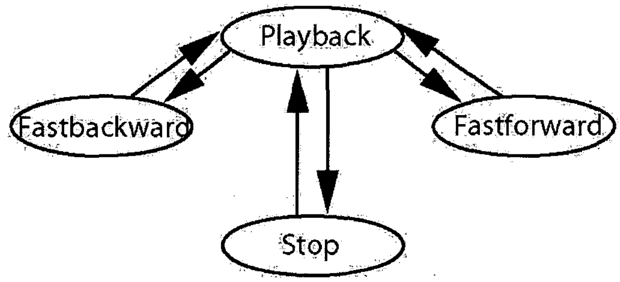


**Figure 2. The interactive operation state at the receiver.**

Let's denote the stop, playback, FFW and FBW states of the receiver by 0, 1, 2, 3, and the state transition probability from state $i$ to state $j$ (where $0 \le i, j \le 3$) is represented by $\theta_{ij}$. If stationary state probability vector for the receiver is $X = (X_0, X_1, X_2, X_3)$, $X$ can be obtained by solving the following equation [14]:

$$\begin{aligned}
X_0 &= \frac{1}{1 + \theta_{0,1}/\theta_{1,0}(1 + \theta_{1,2}/\theta_{2,1}) + \theta_{1,3}/\theta_{3,1})} \\
X_1 &= \frac{\theta_{0,1}}{\theta_{1,0}(1 + \theta_{0,1}/\theta_{1,0}(1 + \theta_{1,2}/\theta_{2,1}) + \theta_{1,3}/\theta_{3,1}))} \\
X_2 &= \frac{\theta_{0,1}\theta_{1,2}}{\theta_{1,0}\theta_{2,1}(1 + \theta_{0,1}/\theta_{1,0}(1 + \theta_{1,2}/\theta_{2,1}) + \theta_{1,3}/\theta_{3,1}))} \\
X_3 &= \frac{\theta_{0,1}\theta_{1,3}}{\theta_{1,0}\theta_{3,1}(1 + \theta_{0,1}/\theta_{1,0}(1 + \theta_{1,2}/\theta_{2,1}) + \theta_{1,3}/\theta_{3,1}))}
\end{aligned} \quad (10)$$

For interactive multimedia, the expected channel transmission rate $E[\lambda(t)]$ is therefore expressed as the statistical sum of the playback rate, fastforward rate, and fastbackward rate with weights $X_1$, $X_2$, and $X_3$.

$$E[\lambda(t)] = [X_1 + k(X_1 + X_3)]\mu(t) \quad (11)$$

where, $k$ is the FFW/FBW speed factor implying that the channel transmission rate is $k$ times that of the normal playback rate.

### C. System Model and Dynamics

If the handheld device receives a frame with the number of errored packets exceeding the threshold, the server will retransmit the corresponding frame by switching to the retransmission state. Therefore, the status of the system can be divided into two states as shown in Figure 3:

- State 0: the *normal state* in which multimedia is continuously received at the handheld device;
- State 1: the *retransmission state* in which the system is retransmitting a frame. The handheld device consumes data upto the errored frame and then waits until the successful arrival of the retransmitted frame.

From Figure 3, the state transition matrix $M$ is expressed as:

$$M = \begin{bmatrix} 1-\dfrac{1}{\tau_f} & \dfrac{1}{\tau_f} \\ \dfrac{1}{T_r} & 1-\dfrac{1}{T_r} \end{bmatrix} \tag{2}$$

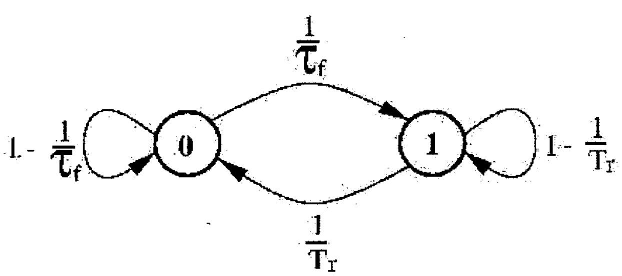


**Figure 3. The state diagram of the client-server type wireless multimedia transmission system.**

Let $V = (V_0, V_1)$ denote the long term stationary state vector, which satisfies the stationary equation:

$$V = VM \tag{13}$$

By solving the stationary equation, the steady state probabilities are obtained as:

$$V_0 = \frac{1}{1+\alpha} \tag{14}$$

$$V_1 = \frac{\alpha}{1+\alpha} \tag{15}$$

where $\alpha = T_r/T_f$. In order for the system to work satisfactorily, it should be at state 0 most of the time, i.e., $\alpha << 1$. This implies:

$$\frac{T_d + \tau_s^I}{1-P_I(d_I)}W_I + \frac{T_d + \tau_s^P}{1-P_P(d_P)}W_P << \frac{\min(d_I, d_P)}{\lambda_p} \tag{16}$$

If Det($M$)=0, there is no non-trivial steady state solution, i.e., the system does not have any long term stationary state. In this case, the system will not work properly; this gives the critical condition that the system will be down:

$$T_r \tau_f = T_r + \tau_f \tag{17}$$

Because $T_r$ and $\tau_f$ are related to channel bit error rate, channel transmission rate, multimedia frame size, GoP structure, acceptable error threshold, FRTT, etc., by monitoring and dynamically adjusting these parameters during multimedia transmission, we can satisfy acceptable quality of service and prevent the video transmission from collapse.

## IV. MINIMUM RECEIVER BUFFER REQUIREMENT

For MPEG video, the decoding of P and B frames in a GoP depends on the I frame in that GoP. The decoding of a B frame, however, depends only on the P frames preceding and following the B frame. Therefore, an error in an I or P frame propagates to all other frames in the GoP. The decoder at the handheld device will deplete the buffered data until the damaged frame is recovered. Assuming that the system failed at time $t_1$, to sustain a continuous display at handheld device, the amount of buffered data $C(t_1)$ must satisfy:

$$C(t_1) = \int_{t_1}^{t_1+T_r} \mu(t)dt \tag{18}$$

By using the average expression, let $E[\mu(t_1)]$ denote the average value of $\mu(t)$ for $t_1 \le t \le t_1 + T_r$. Then $C(t_1)$ can be expressed as:

$$C(t_1) = T_r E[\mu(t_1)] \tag{19}$$

The minimum amount of buffered data also sets the minimum handheld device buffer size

$$C_{\min} = T_r E[\mu(t_1)] \tag{20}$$

By substituting the expression for $T_r$ into Equation (20), we get:

$$C_{\min} = \left( \frac{T_d + \tau_s^I}{1-P_I(d_I)}W_I + \frac{T_d + \tau_s^P}{1-P_P(d_P)}W_P \right) E[\mu(t_1)] \tag{21}$$

From Equation (21), the minimum handheld device buffer size is decided by the playback rate, the FRTT, the link rate, and the frame failure probability.

## V. NUMERICAL RESULTS

In this section, we present results to evaluate the performance of our proposed scheme and show its effectiveness in multimedia transmission over wireless networks. The average system failure time interval versus the level of user interactivity is shown in Figure 4. As expected

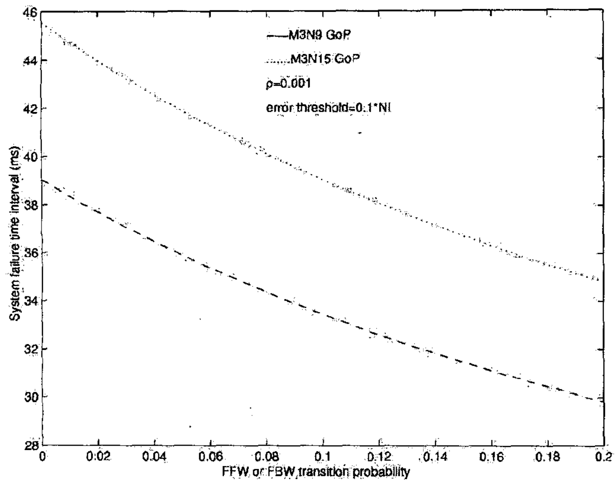


**Figure 4. Average system failure time interval versus the receiver's level of interactivity.**

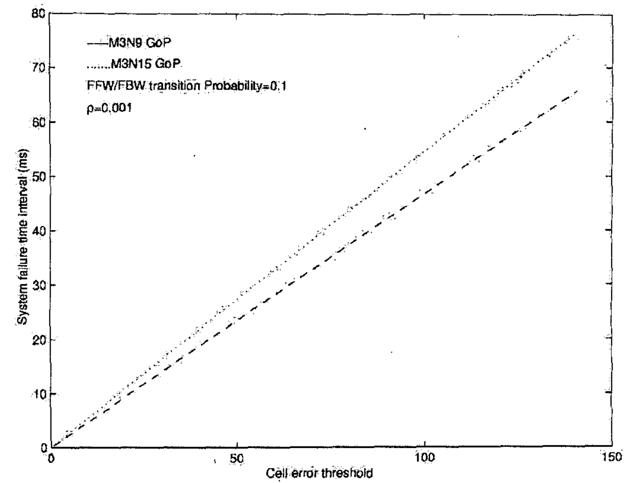


**Figure 6. Average system failure time interval versus packet error threshold.**

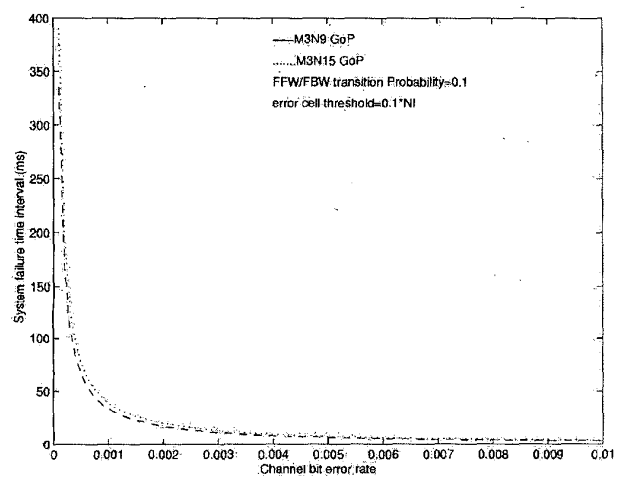


**Figure 5. Average system failure time interval versus the channel bit error rate.**

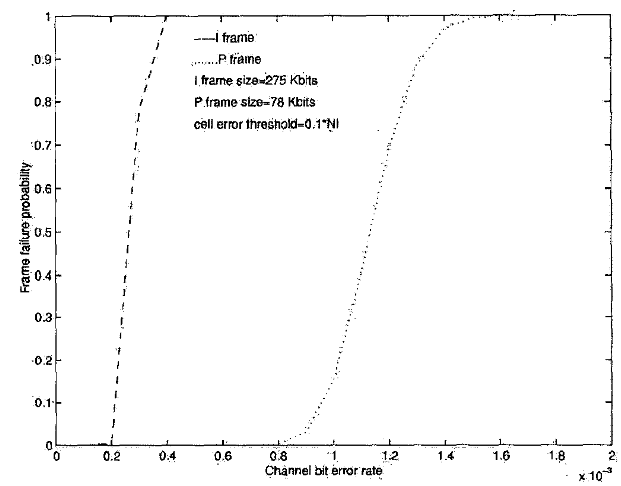


**Figure 7. I & P frame failure probability versus channel bit error rate.**

from Equations (9) and (11), with an increase in the fastforward/fastbackward probability of the user, the required channel transmission rate also increases. This results in more data bits being transmitted in unit time, i.e. higher probability of error happening in unit time. Therefore, the average system failure time interval decreases.

The average system failure time interval versus the channel bit error rate is shown in Figure 5. As pointed out in Equation (9), the average system failure time interval is inversely proportional to the channel bit error rate.

The average system failure time interval versus the packet error threshold is shown in Figure 6. The average system failure time interval linearly increases with an increase in the packet error threshold that depends on the QoS acceptable by the user.

The I frame and P frame failure probability versus channel bit error rate is shown in Figure 7. As the channel bit error rate increase, the frame failure probability increase abruptly. Since the size of an I frame is much larger than that of a P frame, the I frame failure probability is much more sensitive to the channel bit error rate than the P frame.

The I frame and P frame failure probabilities versus packet error threshold are shown in Figure 8. As the packet error threshold increase, the frame failure probability decreases abruptly. This enables us to properly choose the packet error threshold to compensate for the channel bit error rate which in turn reduces the number of retransmissions and increases the network bandwidth utilization.

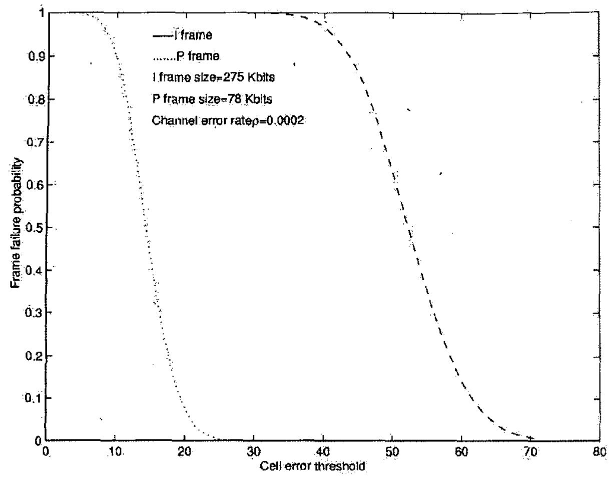


**Figure 8. I & P frame failure probability versus the packet error threshold.**

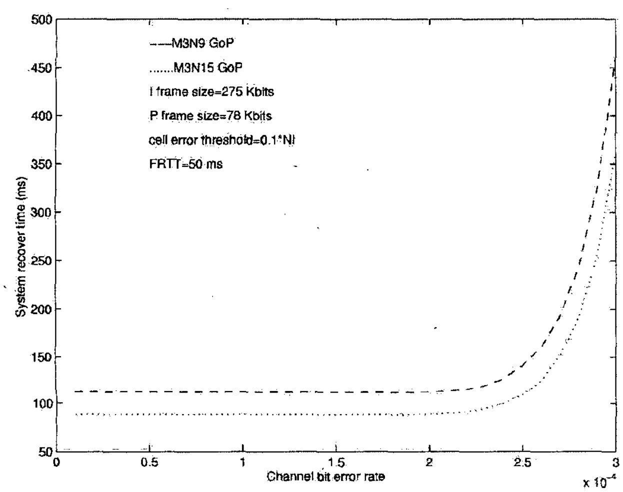


**Figure 9. Average system recover time interval versus channel bit error rate.**

The average system recover time interval versus channel bit error rate is shown in Figure 9. When channel error rate is relatively small, the average retransmission time is almost constant. As the channel bit error rate increases to some area, the average retransmission time increases sharply.

The system recover time interval versus handheld device's interactive level is shown in Figure 10. As the interactive level increase, the channel transmission rate increase, the time to inject a frame into network decrease, the average retransmission time decrease.

The average system recover time interval versus the I frame size is shown in Figure 11. As the I frame size increase, the probability of I frame transmission increases with an associated increase in time to inject an I frame into network. Therefore, the average retransmission time will increase.

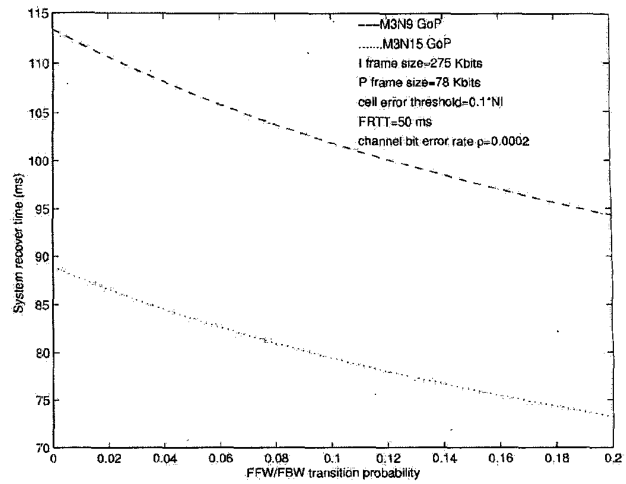


**Figure 10. Average system recover time interval versus user's level of interactivity.**

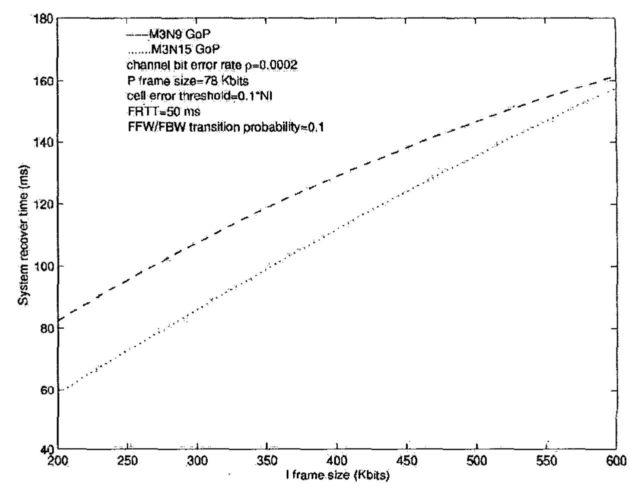


**Figure 11. Average system recover time interval versus I frame size.**

The minimum receiver buffer size versus channel bit error rate is shown in Figure 12. As described in Equation (20), the minimum receiver buffer size is proportional to the average system retransmission time. Therefore, when the channel error rate is low, the minimum receiver buffer size is almost constant. As the channel bit error rate increases, the minimum receiver buffer size increases abruptly in order to maintain a continuous display at the handheld device.

The minimum receiver buffer size versus the user's level of interactivity is shown in Figure 13. Since interactivity only affects the channel transmission rate, its effect on receiver buffer size is relatively small.

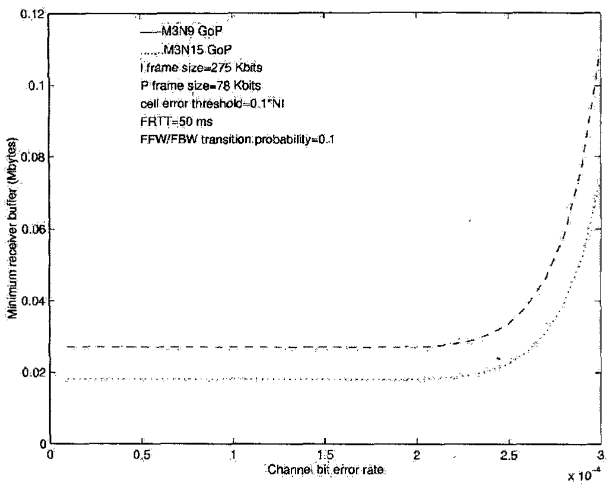


**Figure 12. Minimum receiver buffer size versus channel bit error rate.**

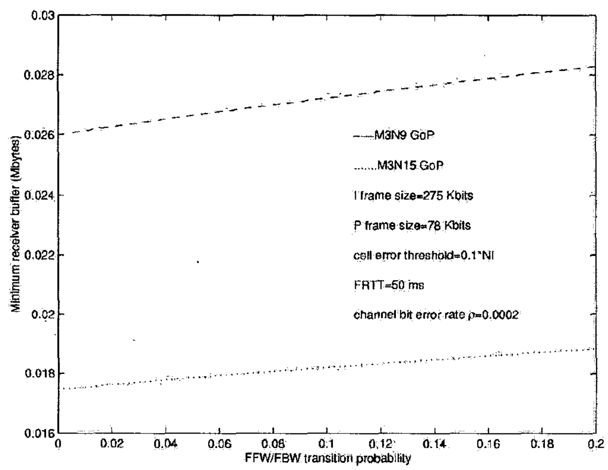


**Figure 13. Minimum receiver buffer size versus handheld device's level of interactivity.**

The minimum receiver buffer size versus the I frame size is shown in Figure 14. As the I frame size increases, the frame transmission failure probability increases, the time needed to inject the frame into the network increases, and the required channel transmission rate increases resulting in an increase in the minimum receiver buffer size as shown in Equation (21).

## VI. CONCLUSIONS

In this paper, we have *proposed a new selective retransmission scheme for interactive multimedia application over a noisy wireless channel to a handheld device.* We analyzed the system requirements and minimum receiver buffer size for providing acceptable QoS to the user by maintaining a continuous video display at the handheld device.

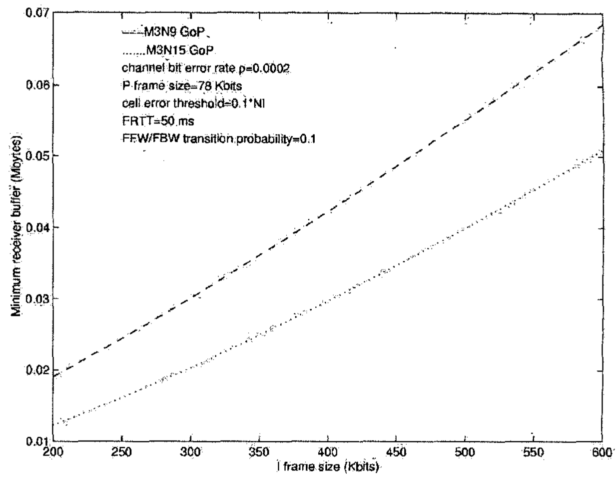


**Figure 14. Minimum receiver buffer size versus I frame size**

We have used average system failure time interval and average system recover time as our performance measures. From our results, we conclude that the average system failure time interval will decrease with increase in the channel error rate and level of user interactivity. The I frame has a much higher transmission failure probability than the P frame for a given channel bit error rate.

The *average system recover time interval has tight relationship with the channel bit error rate and the frame sizes of video, but has little effect on the level of user interactivity.* By choosing an acceptable packet error threshold for a given channel error rate, our proposed selective retransmission scheme requires a small buffer to cost effectively transmit multimedia over a wireless channel, with bit error rate less than $10^{-4}$, to a handheld device.

We have also investigated the *effect of coding* on the performance of our proposed algorithm. We have shown that an MPEG video using an M3N9 GoP has a shorter average system failure time interval than a video using an M3N15 GoP under the same system conditions. Moreover, M3N9 GoP has a longer average system recover time and larger handheld device buffer requirements when compared to an M3N15 GoP. We conclude that for multimedia transmission over a noisy wireless channel to a handheld device, a large GoP structure (such as M3N15 GoP) will have better performance than a small GoP structure (such as M3N9 GoP).

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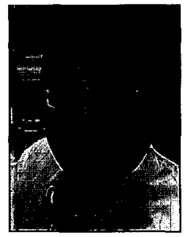

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