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子計劃二:根據預測的無線通道情況控制即時資料的傳輸速率(I)

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Rate control for real time media based on predictive wireless channel condition

根據預測的無線通道情況控制即時資料的傳輸速率

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一, 中文摘要

關鍵字:（通道模型，自動重傳，通道預測）兩個狀態的馬可夫鍊被用來描述衰變通道。在這個模型,好狀態代表正確傳輸,而壞狀態代表位元錯誤。這兩個狀態模型的轉換機率是由通道情況推導而來,像是都卜勒頻率,平均信號雜音比及符號長度。利用這麼個模型來代表目前通道情況的目的是要用轉換機率來推測未來的通道情況。進兒預測未來的位元錯誤率及自動重傳的量。這個自動重傳的量就可以從原先已分配好的位元量減掉,以防止因重傳而使緩衝器擠滿。

ABSTRACT (keywords: channel model, ARQ, channel prediction): A two-state Markov chain is used to model the fading channel. In this model, good state indicates correct transmission and bad state indicates bit error. The transition probability of the two-state model is derived as a function of the channel condition such as Doppler frequency, average SNR and symbol timing. The purpose of the state model to represent the current channel condition is that the future channel condition can be predicted from the transition matrix. Based on this model, the future channel condition and bit error rate can be predicted and the amount of ARQ can be pre-determined. This amount of ARQ is thus subtracted from the pre-allocated target bit for real time media to pre-compensate for future re-transmission.

二, 緣由與目的

Most of the error in transmission is due to channel fading The current wireless system employs ARQ protocol to deal with erroneous packets. This kind of error concealment will increase the transmission burden and pose a problem for real time media data. The direct consequence is that the effective buffer output rate will decrease due to the retransmission. To prevent overflow, one of the method is to adapt the source coding rate according to the buffer condition. The buffer fullness reflects the channel condition prior to the current encoding moment. With such a strategy, the current coding rate will be a function of the amount of ARQ that was issued before. However, if the channel condition can be predicted in advance such that during the source coding period the transmission error rate can be estimated, the source coding rate can be accordingly adapted such that ARQ will not increase the buffer fullness to prevent future buffer overflow and frame skip.

To achieve this goal, a strategy that further reduces the bit rate allocated for each frame to be coded according to the channel prediction is proposed. This is used to pre-compensate the amount of future possible ARQ retransmission.

In the original model, the buffer update rule is as follows:

$$W = \max(W_{prev} + B_{prev} - R / F, 0)$$

if W>M, M the threshold, the next frame would be skipped. Otherwise, the frame target for the next encoding frame is

$$B_j = R / F - \Delta, where \Delta = \begin{cases} W / F & , W > Z.M \\ W - Z.M & , otherwise \end{cases}$$

by default, M=R/F, Z=0.1, we can see that Δ is used for buffer fullness adjustment. In the following, we discuss how the channel prediction is performed to further modify B_j with a two-state channel model.

三、研究方法及成果

The main point is to predict the possible number of error packet and subtract it from the current frame target bit rate B_j. We use a simplified Gilbert channel [2] for channel prediction. This model describes burst-noise using a Markov chain with two states G and B. In state G, transmission is error-free. In state B, the channel has only probability h of transmitting a digit correctly. Because we use simplified Gilbert channel, we set the parameter h as 0. To simulate burst noise, the states B and G must tend to persist; i.e., the transition probabilities b=Prob(G ->B) and g=Prob(B ->G) will be small and the probabilities 1-b, 1-g of remaining in G and B will be large. Fig. 1 is a diagram for the Markov chain.

the state probabilities at time k can be derived from the state probabilities at time t_0 recursively by

$$P^k = P^{k-1} * T = P(t_0 \mid state(t_0) = G) * T^{k-t_0} = [1\ ,\ 0] * T^{k-t_0}$$

The average number of time units the channel is in good state and bad state are denoted as $\overline{T}(G)$ and $\overline{T}(B)$ and are $\frac{1}{b}, \frac{1}{g}$ respectively, $\frac{1}{g}$ is also called as the mean burst error length or average fade duration. The stationary distribution of good state and bad state is

$$P^{\infty} = \left[P_0,\ P_1 \right] = \left[g/(b+g),\ b/(g+b) \right]$$

p_1 is the average bit error rate. To predict the future channel throughput, we use the average probability in Good state in the next k-packet interval. We define the average probability of Good state in the next k-packet intervals as $avg\{P(t \mid state(t_n) = state)\}$ and it is the average of the following equation

$$avg\{ P(t \mid state(t_n) = state)\} = \frac{1}{k} \sum_{t=1}^{k} P_0(t \mid state(t_n) = state),\ n = 0,1,2,......$$

The term (1-$avg\{P(t \mid state(t_n) = G)\}$) is the average bad state probability. Multiplied by the channel throughput, the prediction of the error bits at time t_n is obtained as

$$Bit_{retx} = \frac{R}{F} * (1 - avg\{P(k|state(t_{current}) = state)\}),\ state \in \{G,B\}$$

The error bits are subtracted from the previous B_j. That is $B_j = B_j - kBit_{retx}$. The goal is to maximize B_j, thus it is desirable that the transmission rate R is as high as possible and the error rate is as low as possible. In this paper, we aim at the establishment of the relationship between the GE model and the Raileigh channel distribution for the purpose of channel prediction..

To derive the transition probability b and g, we assume the received envelope $\sim$ =|r| as Rayleigh distribution, i.e., $f(\sim) = \frac{\sim}{\pm^2} e^{-\sim^2/2\pm^2}$ where $\pm^2$ is variance of Gaussian random process, the received SNR is $¥ = \sim^2 E_b / N_0$, its PDF is exponent distribution, i.e., $f(¥) = \frac{1}{\bar{¥}} e^{-¥/\bar{¥}}$, $¥ >= 0$ where $\bar{¥} = E[\sim^2] E_b / N_0$ is the average SNR of the received signal and E[$\sim^2$] is the average value of $\sim^2$. Since the GE channel consists of two states, we let $¥_t$ be the thresholds of the received SNR, where the channel will change state. We can calculate the stationary probabilities of the GE channel in its respective states by finding the fraction of time the Rayleigh fading channel is below and above $¥_t$ respectively. Thus

$$P^{\infty}(B) = \int_0^{¥_t} \frac{1}{\bar{¥}} e^{-¥/\bar{¥}} d¥ = 1 - e^{-¥_t/\bar{¥}} = 1 - e^{-\circ 2}$$

similarly, we have

$$P^{\infty}(G) = \int_{¥_t}^{\infty} \frac{1}{\bar{¥}} e^{-¥/\bar{¥}} d¥ = e^{-\circ 2}$$

where $\circ 2 = ¥_t / \bar{¥}$. Another related parameters are the Level Crossing Rate(LCR) and Average Fade Duration(AFD). From [3] and [4],

$$L_R = \frac{\int_0^{\infty} p(R, \&) dr d\&}{dt} = \int_0^{\infty} \& p(R, \&) d\& = \sqrt{2®} f_D \circ e^{-\circ 2}$$

$$\bar{2} = \frac{P_r[r <= R]}{L_R} = \frac{1 - e^{-\circ 2}}{\sqrt{2®} f_D \circ e^{-\circ 2}} = \frac{e^{\circ 2} - 1}{\sqrt{2®} f_D \circ}$$

the transition probability can be derived as, $g = \frac{\circ f_D T_S \sqrt{2®}}{e^{\circ 2} - 1}$, $b = \circ f_D T_S \sqrt{2®}$ where $f_D = \frac{3}{a}$ is the Doppler frequency, and T_S is the symbol duration.

Fig. 2 shows an example of packet error in a typical wireless channel for 160 frames. These error packets will cause retransmission, Without ARQ, the buffer fullness will not be affected. With ARQ, the effective throughput is reduced. Fig. 3 shows the buffer fullness condition when ARQ is used. When the buffer fullness is above a certain threshold, frame skip happens. With the above mentioned channel prediction and adaptive frame layer bit allocation, the improvement is shown in Fig.4.

四、結論

The main point is to use a two-state Markov model to represent the current channel condition. The transition matrix is a close approximation of the channel memory. From the transition matrix, we estimate the future bit error rate and the possible retransmission amount. This is equivalent to the future channel prediction. From the predicted amount of ARQ, we subtracted it from the pre-allocated target bit in real time media coding. This is to leave room for the future possible bit error and ARQ such that the retransmission ARQ will not cause transmitter buffer overflow.

Reference:

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Channel," IEEE Trans. Commun, vol. 47, no. 5, May 1999.

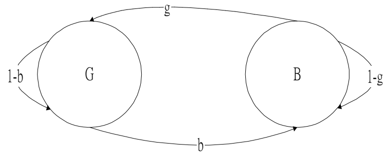

Fig.1 two state channel model

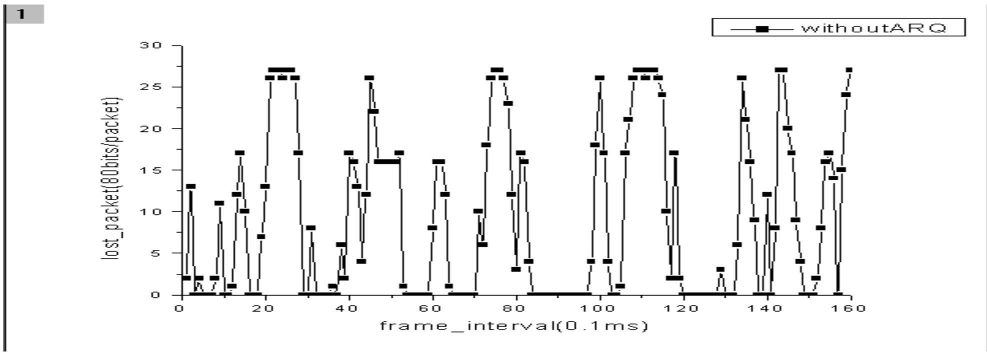

Fig 2.transmission error distribution

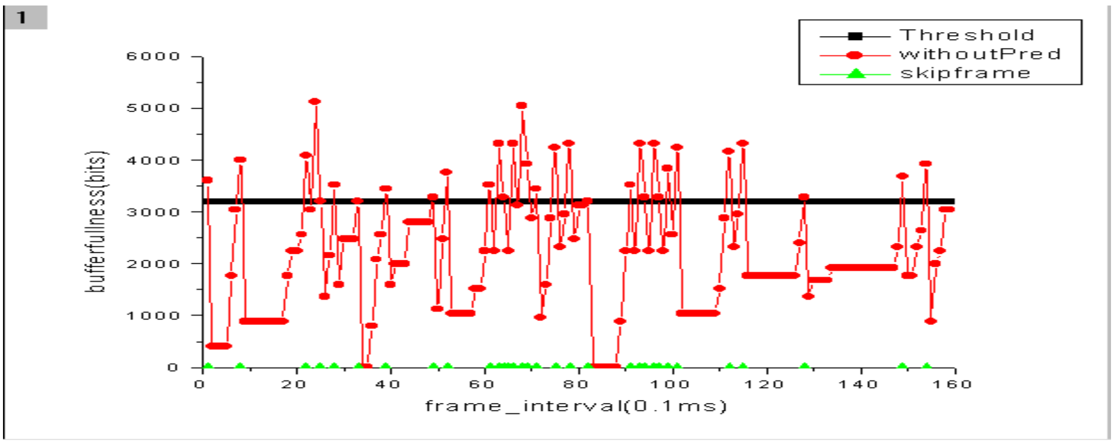

Fig 3 transmission without prediction

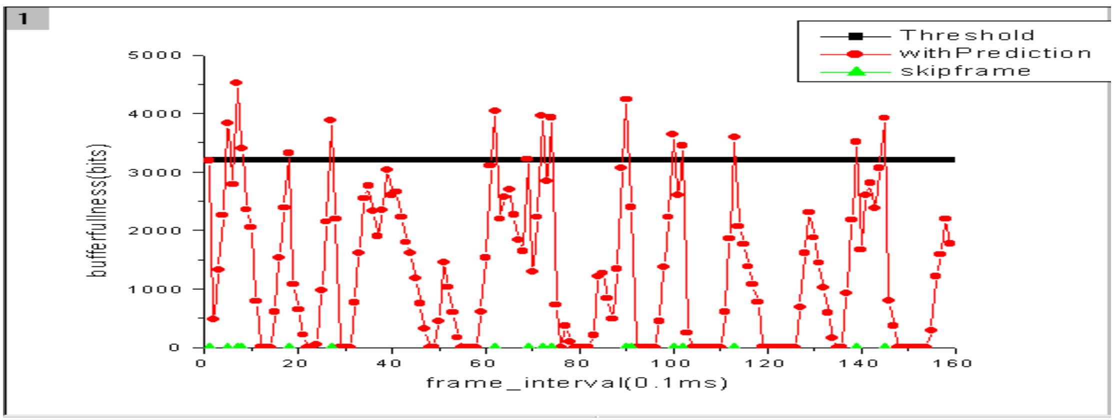

Fig 4 transmission with prediction