Multiplexing VBR Video and Training Sequences on Wireless Fading Channels

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Abstract

A time slotted multiple access scheme that shares bandwidth between source and training signals is proposed for the transmission of variable bit rate (VBR) video signals over non-ideal channels. The error performance is examined using finite state Markov chains that model both the VBR source traffic and the time variation of fading channels. The characteristic time-scale of the fade duration, the step-size parameter of the LMS equalizer training algorithm, and the number of training sequences allocated per time slot jointly influence error performance. The consideration of physical layer characteristics in the design of higher layer protocols enhances the performance of conventional application layer based preventive and reactive error control schemes. Simulation results for fast and slow fading channels show that the proposed channel access scheme can lead to an order of magnitude decrease in the bit error ratio if the combination of training overhead and step-size parameter are chosen judiciously with respect to the fading time-scale. In addition, the multiplexing efficiency of VBR encoded video is shown to be useful for limiting the channel access delays when training overhead is increased.

I. Introduction

Secure and timely wireless transmission of multimedia signals between mobile terminals is an important and challenging problem for military communications. The performance of real-time packet transmission schemes over wireless channels is governed by the traffic characteristics of the source signal, the channel properties and channel access protocols. Third generation TDMA protocols are designed to support bandwidth-intensive multimedia applications at a peak rate of 5 Mbps. The TDMA scheme is susceptible to intersymbol interference and detection errors arising from multipath propagation and slow fading. To contend with these problems, TDMA time slots are interlaced with training symbols between data sequences. These symbols enhance performance when multipath is present. The number and the placement of training symbols relative to the data sequences have been shown to impact the performance of channel equalizers in controlling error [1]. For stationary time-invariant channels the number of training symbols required per time slot will depend on the channel delay spread and the type of equalizer used. In the case of fading channels the number of training slots may be varied to match the state of the channel thereby maintaining a fixed error performance for a given signal to noise ratio (SNR).

From the application layer perspective, channel access protocols must bound the channel access delay and packet loss resulting from detection errors and overflow rates to be well within the tolerable values. For VBR video traffic, the delay and loss performance measures will be influenced by both the traffic characteristics and the dynamics of the access protocol. For packet video transmission, error control may be applied at the application layer by transmitting redundant information through forward error correction (FEC) codes or through automatic repeat request (ARQ) retransmission of errored packets. The FEC and ARQ schemes represent extreme cases in that the former is a preventive scheme that incorporates the worst case channel model and the latter represents a reactive scheme that assumes the minimum channel information. In this work, we examine the potential of adaptive equalizers in controlling error by changing the number of training sequences allocated in response to a channel state. The channel bandwidth is therefore shared between the source encoders and channel equalizers. The trade-off in increased delays for the video source is compensated using variable bit rate encoding. Decision feedback equalizers are implemented to examine the error performance for wireless channels characterized by multipath and slow fading.

II. Transmitter, Channel and Source Models

A. Transmitter Model

A video source model of MPEG-2 encoded variable bit rate video is used to simulate a channel-encoded bit-stream input sequence. These bits when present, are input to a BPSK modulator and transmitted over a wireless channel. At the receiver, the received signal is processed by a filter matched to the symbol shaping pulse and then input to a decision-feedback channel equalizer.

The receiver samples the matched filter output at the symbol rate. The equalized signal is then passed through a maximum likelihood detector.

We consider the TDMA access scheme which is part of the IMT-2000/UMTS standard. The TDMA schemes require equalization techniques at the receiver to combat the problem of inter-symbol interference that arises due to signal arrivals generated from multipath transmission. The IMT-2000 TDMA standard specifies a frame time of 4.615 msec and a slot time of 288 microseconds. The number of slots per frame is 16. The standards fix the training sequence in every slot to be 27 bits and the training sequence is interspersed between two data bursts, each of maximum length 342 bits. In this study we examine how this fixed slot structure may be changed to support a variable bit rate transmission signal and variable length training sequences in a multipath and fading channel.

B. Channel Model

We consider wireless channels characterized by the effects of shadowing and multipath fading. Shadowing is caused by changes in the physical features of the propagation channel and result in a slow variation in the mean envelope of the received signal with time. Multipath arrivals are caused by reflection and scattering of a single transmitted radio signal by structures in the propagation path. The variations in amplitude, phase, and propagation delay among the different arrivals distort the transmitted signal causing fades of upto 20 *dB* below the mean power level. The time and space varying characteristics of the propagation path can be captured by channel impulse response models [2,3]. The channel impulse response considered in this work is of the form,

$$h(t) = \sum_{k=0}^{N-1} a_k(t)\ \delta(t - \tau_k) \tag{1}$$

where $a_k(t)$ are the randomly varying complex amplitudes and τ_k are arrival-times of the $k-th$ multipath component. The impulse observation time and application time are given by t and τ respectively. The number of multipath components N is assumed to be a constant. In this work we model the multipath amplitudes a_k as an exponentially decaying profile, with time constant set equal to the delay spread. Such models have been recommended by the PCS Joint Technical Committee for use in both indoor and outdoor channels [4].

Slow fading effects in the channel are represented using two received power levels. A two state Markov chain is used to model transitions between these levels. By varying the average holding time in each state of the Markov chain, one can examine the performance effects over a range of time-scales of the fading process. Next we provide a brief overview of the discrete time Markov chain model of VBR video.

C. VBR Video Source Model

An MPEG-2 video encoder generates a bitstream which is modeled at the video frame level. MPEG-2 frames can be of type intra (I), predictive (P) or bi-directional (B). Here we consider only the I and P frame types, with the I frames being generated at scene changes. The I frame bit rates contribute to the largest amplitudes whereas the P frames transmit the differential information in successive frames and result in a distribution of moderate valued frame rates. VBR video is encoded using a fixed quantizer step-size and therefore represents constant quality video with time. The bit rate variability and the temporal correlation between successive video frames must be considered to evaluate the performance in a queue. These features have been shown to be adequately captured by a finite state Markov chain model [5].

The VBR video source model considered here is represented by an I state discrete-time, Markov chain, with a transition probability matrix $\mathbf{P}_V$ and a rate vector $\mathbf{R}_V : \left[r_1, r_2 \ldots, r_I \right]$. The rate r_i represents the number of bits generated per video frame when the process is in state i. The last state I represents the intraframe state and the remaining states correspond to the P frames. The diagonal elements of $\mathbf{P}_V$ are dominant with the exception that $p_{II} = 0$. This structure results due to an immediate transition from an I frame to a P frame. This diagonally dominant structure signifies that VBR video is characterized by strong short-term correlations. The parameters for this Markov chain are obtained empirically by analysis of a MPEG-2 stream from the *Blues Brothers* movie [5]. This data is modeled with $I = 17$ states.

III. Multiplexing VBR Video and Training Symbols

As new wireless standards push the limit on channel capacity to support high bit rate services, flat fading channels will be transformed to frequency selective channels, causing a marked increase in inter-symbol interference. The degradation in system performance due to detection errors is therefore an important issue in such systems. The use of a channel access scheme that maximizes bit error performance without significant sacrifice of channel utilization is considered. This scheme seeks to balance, by increased equalizer training, the deleterious effects of impulse response and signal to noise ratio variations due to channel fading. Increasing the number of training bits during high fade intervals results in improved detection at the cost of reduced channel capacity and increased delays for a fixed rate data source. However, for a variable bit rate source, with capacity allocated at a value between its average and peak rates, the channel can be shared efficiently with training bits. The adjustment of the number of training bits in a TDMA slot can be accomplished adaptively at the transmitter using radio channel state information sent by the receiver over a control channel. The improvement in bit error performance and the lower data transmission rate during poor channel conditions serves to diminish packet errors.

The successful application of the proposed scheme is influenced by the performance of equalizer in terms of its adaptability to changing channel conditions. The channel delay spread, the fade level and the fade duration are three basic parameters that influence performance. A decision feedback equalizer (DFE) is considered in this study. The least-mean-squares (LMS) algorithm is chosen for adaptively training the tap coefficients. This algorithm is preferred due to its simplicity and computational economy.

The time-variant channel is modeled using a two state Markov chain, with a transition matrix $\mathbf{P}_C$ and an associated fade level vector $\mathbf{R}_C$: $\left[f_1, f_2 \right]$. The $f_i, i = 1, 2$ represent the fade amplitude levels normalized by the rms amplitude of the fading signal. We will assume that state 1 represents the non-fade state with $f_1 = 1$ and f_2 represents the amplitude of the fade state, when the amplitude falls below this value. Using Rayleigh distributed statistics for the amplitude envelope variations, theoretical expressions for the average fade durations τ_{f_i} and level crossing rates have been determined in [6] as a function of the normalized fade levels f_i and a given Doppler spread. We wish to examine the impact of $\{ f_i, \tau_{f_i} \}$ on the equalizer performance. Next an access protocol is proposed that takes into consideration the state of a channel during transmission and varies the number of training symbols provided per slot in proportion to the fade level of the state.

A. Bit Error Performance

The performance of the LMS algorithm depends on the multipath delay spread, the number of taps of the equalizer M, the step-size parameter μ, the fading characteristics and the spectral characteristics of the vector input to the equalizer. In this presentation, to focus on the main results, we fix the multipath delay spread to be restricted to a delay spread of one symbol and consider a fixed number of taps of the equalizer $M = 5$ that is found sufficient to model this level of inter-symbol interference. We focus in particular, on the influence of the change in μ and the change in spectral characteristics of the input caused due to a variation in the fading characteristics.

A basic problem with the LMS algorithm is the self-noise generated due to noisy characteristics in the estimation error. This causes a deviation from the optimal coefficient values. The self-noise being a function of the input power, tends to decrease when the channel goes into a fade state. By selecting smaller values of μ, the error due to self-noise may be further reduced. This comes at the cost of reduced convergence rates and poor tracking performance of the equalizer. This source of error, typical for time-variant channels causes the estimated values to lag the optimum values in time [7]. However, by providing an increased number of training slots between data transmission, one can attempt to compensate for the effects of tracking error.

In the following simulation results, the parameters of the fading channel model are chosen as : $f_i : (1.0, 0.6)$, representing a mean fade level of $-4.436\ dB$ when the channel is in the fade state. The average durations t_{f_i} spent in each of these states may be varied by adjusting the diagonal transition probabilities of the Markov chain in the fading channel model. We represent these durations in units of TDMA slots and consider first the case $t_{f_i} : (30, 10)$, where the channel sojourns an average of 30 and 10 slot times when it enters the normal and the fade state respectively. Based on the channel state in a particular TDMA slot, the number of training bits allocated is varied. When in the fade state, *ts* number of training symbols are transmitted. In the normal state, the standard value of 27 training symbols are retained.

In Fig. (1) we depict the bit error ratios (BER) as a function of peak signal to noise ratio (PSNR) in decibels for three choices of the length of the training sequence denoted as $ts : [27, 100, 200]$. The two sets of curves represent the errors for $\mu = 0.1$ and $\mu = 0.001$. It is evident that the combination of increased training symbols and reduced value of μ provides a significant improvement in the bit error rate. For $\mu = 0.1$, an increase in length of training sequence has no effect on the BER which stabilizes to a limit around 10^{-3} for large values of SNR. But with μ reduced to 0.001, a significant improvement in performance is observed at the high SNRs by increasing *ts* to 100 and 200 symbols. For $PSNR = 25$ dB, the BER drops from approximately 5.E-05 (ts=27) to 1.4E-06 (ts=100) and finally to 7.E-07 for *ts* equal to 200 symbols per TDMA slot.

It is however important to note that the level of improvement in the error performance is a strong function of the average fade durations. To examine this, we fix the $PSNR = 25$ dB, select an intermediate $\mu = 0.01$ and vary the average holding time in the fade state from 2 to 30 slots. The BERs for the three training lengths are depicted in Fig. (2a). The relatively small values of $t_{f_2} < 10$ represent channels where fades occur for short durations and the time between these fades is large. We examine the error performance for $ts = 200$. The increased *ts* and moderate value of μ positions the system to converge to sub-optimum values in the fade state. However, due to short fade durations, the overall error which typically occurs when the system is in the fade state is minimized. As t_{f_2} is increased, the process spends larger durations in the fade state and coupled with the convergence to sub-optimal values for large training sequences, results in increased error rates. A comparison of the results for $ts = 100$ and $ts = 200$ show that for $t_{f_2} > 10$, a reduced number of $ts = 100$ produces improvement over $ts = 200$, simply because the process is not allowed to deviate much further from the optimal value for $ts = 100$. For larger values of t_{f_2}, it is seen that no performance gain can be achieved by increasing the number of training symbols for a fixed μ. Any further performance improvement can only come by reducing the value of μ. This result for $\mu = 0.001$ is depicted in Fig. (2b). The turning points in the error for a fixed *ts* occur at higher values, since the convergence to sub-optimal state is slowed down in time due the selection of a smaller μ.

B. Video Access Delays

The performance improvement that can be achieved with increase in number of training symbols comes at the cost of increased channel access delays for the video source. We will use fluid flow analysis [8] to derive the waiting time distributions for the VBR video source. It is assumed that the source bit-stream enters a buffer with infinite waiting room. The VBR source model simulates a MPEG-2 encoder at a frame rate of $1/24^{th}$ of a second. The average and peak rates of the video source are approximately, 1.45 and 2.93 Mbps. The bits wait for channel access and are serviced at the channel rate on a first-in first-out basis . The available channel capacity is assumed to be 2.6 Mbps. When no additional training sequences are multiplexed beyond the standard specification a capacity of $C = 2.4$ Mbps is available for data transfer. This leads to a 60 % utilization factor. As the length of the training sequence allocated per TDMA slot is increased, the capacity allocated to the video is reduced, leading to increased utilization factors. By increasing the training symbols from the standard 27 per slot to 200 during the fade durations, the utilization increases from $\rho_{\min} = 0.6$ to $\rho_{\max} = 0.8$. The distribution of waiting times for channel access is determined as follows.

The buffer occupancy is assumed to be a continuous random variable x. The video input into the buffer is represented using a continuous time Markov generator Q_V obtained from P_V. In the steady state, the delays in the buffer are represented in terms of the cumulative probability distributions $F_s(x)$ $s = 1, 2, .. I$,which represents the conditional probabilities when the source is in state s. These distributions can be shown to be solutions of

$$\frac{\partial \mathbf{F}}{\partial x} \mathbf{D} = \mathbf{F}\, Q_V \qquad (2)$$

where $\mathbf{F}$ is a row vector of the distributions $F_s(x)$ and $\mathbf{D}$ represents a diagonal matrix with elements $d_{ii} = r_i - C$. The solution to Eq. 2 follows that of an eigenvalue problem. The set of coefficients for the I modes are obtained using the boundary conditions at $x = \infty$ and $x = 0$. The cumulative distribution function of the delay $F(x) = \sum_{j=1}^{I} F_j(x)$.

Fig. (3) presents the results for the complementary delay distribution $1 - F(x)$ for the cases where the number of training symbols used were 27, 100 and 200, corresponding to utilizations of 0.6, 0.67 and 0.8 respectively. The horizontal axis represents the waiting times in milliseconds. As expected the waiting times increase with the utilization factor. However, the increase in the number of training symbols from 27 to 100 yields a moderate increase in the delay from 25 to 50 milliseconds at the 99.90^{th} percentile and from 45 to 85 milliseconds at 99.99^{th} percentile of the probability distribution function. These delays are reasonable for wireless environments. Although the waiting times increase to significantly larger values at the 80 % utilization level, the main advantage of the proposed method arises when the multiplexing efficiency of variable bit rate signals is considered. That is, when more than one video source is multiplexed on to the channel at the same levels of utilization, the resultant smoothing of the instantaneous bit rate variations leads to better in queue performance. Since wireless channels are limited in capacity, for the video rates considered in this paper, the multiplexing of at most two video streams is possible if fading is present. With improvement in low bit-rate coding techniques an increased number of sources can be be multiplexed on wireless channels. Fig. (4) depicts the waiting times at the 99.90^{th} percentile value of the waiting time distribution as a function of the utilization factor ρ. The results illustrate the cases when one and two sources are multiplexed. A significant increase in performance is observed due to multiplexing. The channel access delays under multiplexing are tolerable even at the 80 % utilization level. The multiplexing efficiency arises due to a reduction in the rate variability of the multiplexed video stream.

IV. Conclusions

Markov chain models of VBR source video and channel fading dynamics are used to explain the dependence of BER on the length of the training sequences, step-size parameter μ and characteristic time-scales *ts* of the fade durations. The self-noise of the LMS training algorithm is found to be the dominant contributor to the detection errors. To control this, sufficiently small values of the step-size should be considered. In combination with longer training sequences, this technique allows bit error ratios at a fixed SNR to drop over two orders of magnitude. The selection of μ and *ts* is also shown to be strongly dependent on the average durations of the fades. The increased channel access delays of the video source are shown to be controlled using VBR encoding and through the multiplexing efficiency of these signals.

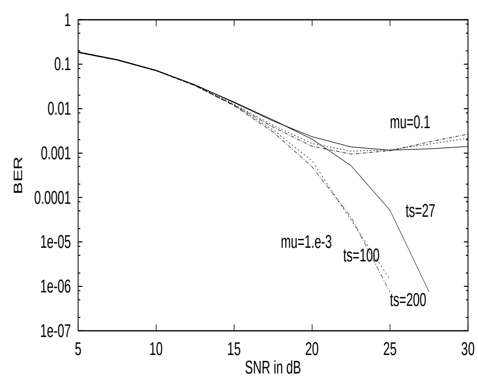

Figure 1: The BER for changing μ and training sequence lengths.

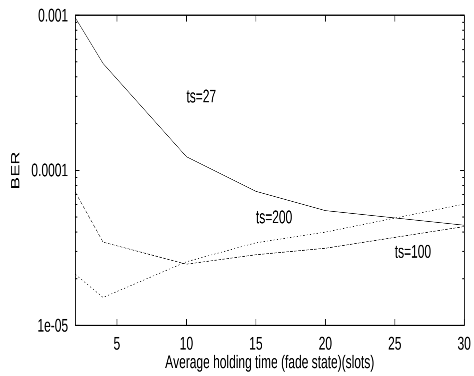

Figure 2(a): BER for PSNR=25 dB as a function of average holding times in the fade state. $\mu = 0.01$.

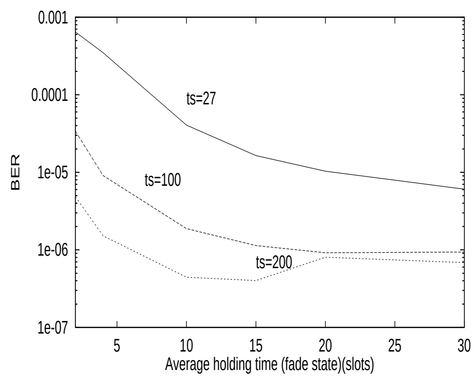

Figure 2(b): BER for PSNR=25 dB as a function of average holding times in the fade state. $\mu = 0.001$

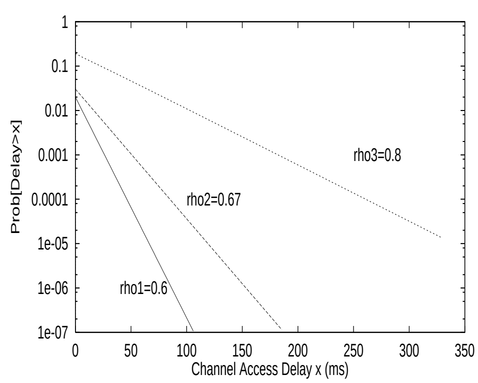

Figure 3: Complementary delay distributions for video source for varying utilization factors.

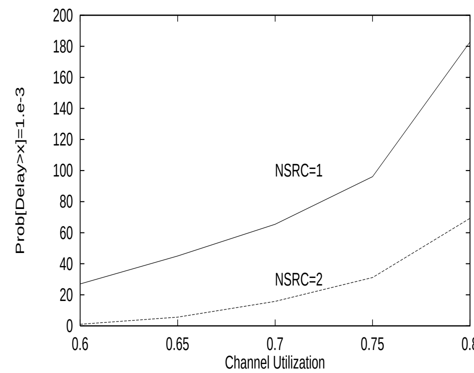

Figure 4: The channel access delays at the 99.9^{th} percentile as a function of utilization.

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