Link Adaptation and Power Control for Streaming Services in EGPRS Wireless Networks[1]

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Abstract: Using the MPEG-4 Advanced Audio Coder (AAC) music as an example of streaming applications, we investigate the improvement of error performance for the streaming service by link-adaptation and power-control techniques in an Enhanced General Packet Radio Services (EGPRS) cellular network. A low packet error rate and variability are essential in providing short error-burst length so that error concealment techniques can be effectively applied to music packets. In this paper, we study the effects of a combined link adaptation and power control scheme (referred to as the *error-based* schme) for achieving a target error rate and reducing error variability. By simulation, we compare the error performance of the error-based scheme at both the EGPRS block and AAC frame level with another adaptation algorithm (referred to as the *throughput-based* scheme) with a goal of maximizing overall network throughput. It is found that when offered with similar traffic load, the former scheme can provide noticeable improvement of music quality over the throughput-based scheme. To achieve similar AAC frame error rate, our results also show that the error-based scheme can increase the link throughput over the throughput-based scheme by 66.7% in one of our examples. These results reveal that by aiming at required error targets and thus reducing error variability, the error-based scheme for link adaptation and power control are helpful in improving quality and capacity for streaming applications.

Keywords: Adaptive Kalman filtering, adaptive modulation, error analysis, information rates, land mobile radio cellular systems, music, power control, radio communication.

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1. INTRODUCTION

With the promise of the third generation (3G) wireless networks [1], wireless data services will become very popular soon. It is expected that the networks will initially provide best-effort data services such as wireless Internet access. As the systems continue to evolve, a variety of *streaming* data applications such as encoded music, compressed video and audio programming will be supported in the future. So, it is important to investigate techniques for supporting such streaming services. In particular, we shall examine in this paper how link adaptation and power control can improve the error performance for delivering MPEG-4 Advanced Audio Coder (AAC) [2] coded music in the Enhanced General Packet Radio Services (EGPRS) network [3].

In one extreme, voice service in the EGPRS network has a very stringent delay requirement, which precludes the possibility of retransmission in case of packet error. In addition, voice also requires a very low packet error probability to maintain satisfactory quality. In the other extreme, many popular data applications such as web browsing do not have tight delay requirements. As a result, retransmission based on Automatic Repeat reQuest (ARQ) procedures is applicable. Streaming applications such as music delivery lie somewhere between the two extremes. Specifically, the streaming nature of the applications requires the packet error probability be less than a certain target, which is usually less stringent than that for the voice service. In addition, the variability of error rate from packets to packets (i.e., distribution of errors) should be low to maintain service quality. Most streaming applications allow data buffering with a modest delay before presenting it to users, thus a limited number of retransmissions become feasible. As long as the packet error rate and error variability are reasonably low, error concealment algorithms can be applied to the buffered data to reduce the impacts of erroneous and missing packets on service quality.

The MPEG-4 AAC offers high quality audio with data rates in the range of 32 to 128 Kbps for two-channel stereo. In the EGPRS network, depending on the radio condition, the data rate for each time slot (of a total of 8 slots per GSM TDMA frame [4]) can range from about 11 to 65 Kbps. Evidently, the high end of the data rate is adequate for the MPEG-4 music. However, given the EGPRS constraints, various approaches are needed to improve the music quality and capacity. For this purpose, several techniques including ARQ, dynamic packet assignment, packet shuffling, SINR-based link adaptation, and random packet discarding have been proposed and examined in our previous work [5]. In this paper, we investigate how the error performance for MPEG-4 music can be further improved by link adaptation and power control with a goal of achieving a target error performance.

Let us briefly discuss how this work is related to existing work on the subject of link adaptation and power control. The main idea of link adaptation is to adapt the modulation and coding levels according to the channel and interference conditions. For best-effort data services, the link adaptation algorithm can be designed to maximize the overall network throughput [6, 7]. However, for real-time services, different link-adaptation algorithms may be needed to deliver their target error performance (e.g., [8, 9]). Unlike the methods in the latter references where the link is adapted according to the measured error statistics, the one we propose here for streaming services is simply based on the relationship between error rate and *signal-to-interference-plus-noise ratio (SINR).* The operations of this method are similar to those in [6, 7], but our main goal is *not* to maximize the network throughput. As to be discussed in detail below, the objective of our link-adaptation algorithm is to achieve a target error probability needed for the music service. The chosen target is non-zero but must be below a certain value so that limited retransmission and error concealment techniques can work effectively to maintain music quality.

Power control is a well-known technique for combating and managing interference in cellular networks (e.g., [10, 11]). The basic idea of power control is to adjust transmission power to a level just enough for achieving a desirable performance. Combining power control with link adaptation is a difficult problem and an open area of research. Specifically, without knowing the transmission power in priori, one cannot predict the SINR, which is needed for choosing the appropriate modulation/coding level. In turn, without knowing the modulation level, the transmission power cannot be adjusted accordingly. The approach we study in this paper assumes that the maximum transmission power is used for predicting the SINR as follows. The Kalman-filter method proposed in [12] is employed to continuously track the (co-channel) interference power. The signal path gain between a transmitter and its associated receiver can be also estimated based on the control channel operations in the EGPRS network. Thus, the SINR can be predicted from the path gain, the maximum power, and the predicted interference power. The estimated SINR is used as an input to adapt the link. Once the modulation level is selected, the power is further re-fined to meet the SINR target associated with the selected modulation level. In aiming at the given SINR targets, the power control in effect helps reduce the variance of error rates. As to be discussed further below, this characteristic is quite important in maintaining the perceived music quality.

Using simulation models, we study how a combined link adaptation and power control scheme for achieving a target error rate, which is referred to as the *error-based* scheme, can enhance the error performance at the EGPRS block and the AAC frame level. In particular, we compare the error performance of the error-based scheme with another adaptation algorithm for maximizing the overall network throughput, which is referred to as the *throughput-based* scheme. In addition, we also examine the capacity gain by the error-based scheme for a given AAC frame error rate.

The rest of this paper is organized as follows. In Section 2, we present the relevant aspects of the EGPRS and AAC design under consideration. In Section 3, we describe two existing techniques, ARQ and packet shuffling, that are employed to enhance music quality in this work. Then, we present the link-adaptation and power-control algorithm in use in Section 4. Simulation results to quantify the improvement of error performance by the algorithm are presented and discussed in Section 5. Our conclusions are given in Section 6.

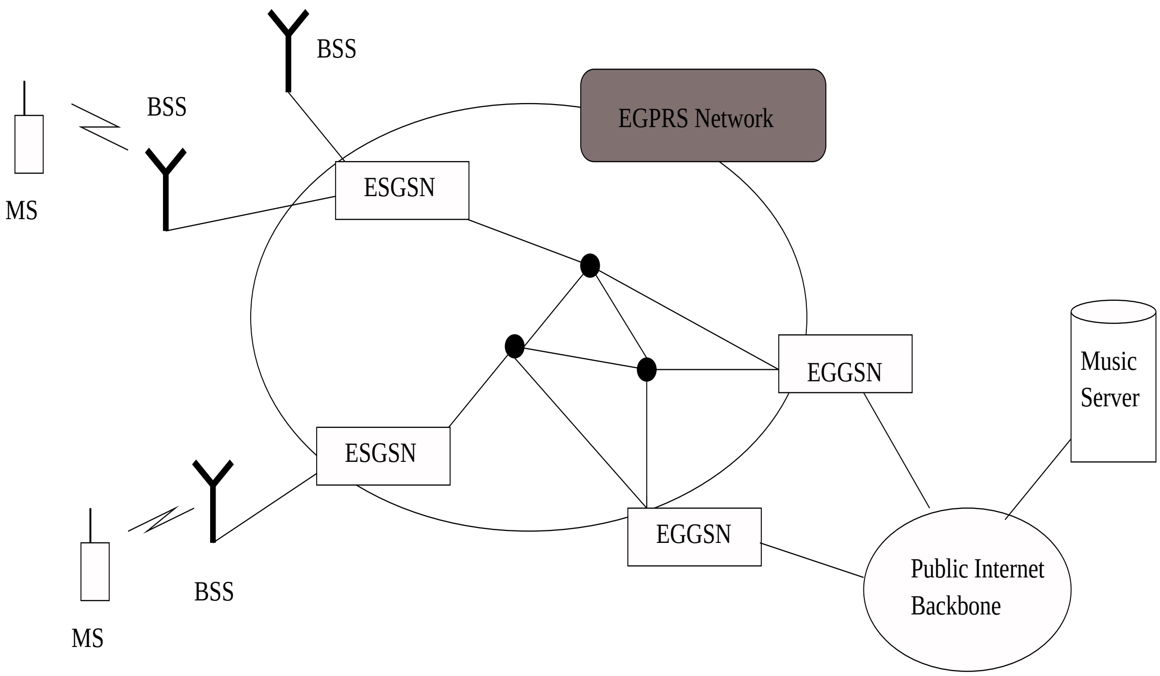
2. EGPRS NETWORK AND AAC MUSIC

2.1 System Architecture

Figure 1 shows a simplified architecture of the system under consideration. As shown in the figure, AAC music servers are attached to a data network, e.g., the Internet, and the EGPRS network uses the Enhanced Gateway GPRS Support Node (EGGSN) to act as a logical interface with the external data network. The EGPRS network supports transport of IP packets and consists of a set of Enhanced Serving GPRS Support Nodes (ESGSN's), which are mobility-aware routers, and base-station systems (BSS's) connected to the serving nodes. Music data requested by mobile user (or mobile station, MS) is delivered from the server, through the serving nodes and the associated BSS, and finally transmitted to the users over the radio link.

The air interface for the EGPRS network is based on TDMA, packet-switched radio technology with 200 KHz channels. The air interface, which is our focus, is also known as the *Enhanced Data rates for GSM Evolution (EDGE).* For this reason, the terms EGPRS and EDGE are used interchangeably in this paper. To transport data between the EGPRS serving nodes and the mobile users, two types of bearers: packet and circuit-switched bearers can be used, which are targeted for "connection-less" and "connection-oriented" data transfer, respectively. The two types of bearers have different physical and link layer structures, and cannot be easily combined on the same radio channel. Thus, for integrated music and data services, we consider the transmission of music over the packet-switched bearers in this paper. The topic of using circuit-switched bearers for music services will be addressed in our subsequent work.

Figure 1. The System architecture



The EGPRS employs a link-adaptation algorithm to adapt the modulation and coding level (which is referred to as *transmission mode* below) for each link according to its radio and interference conditions. Specifics of the adaptation algorithms under consideration are presented later. We assume that for each link, the adaptation occurs once every 100 msec. Information bits are grouped into EDGE radio blocks, each of which can be transmitted in four bursts (i.e., in the same time slot of four consecutive TDMA frames). Depending on the transmission mode, the number of information bits varies from one block to another. As a result of such adaptation, the data rate of a link can vary from about 11 to 65 Kbps. In most cases, one AAC frame (packet) cannot fit into one radio block. Therefore, AAC frame segmentation at the base station and frame re-assembly at the mobile terminal is necessary.

2.2 Parameters of MPEG-4 Advanced Audio Coder

The basic idea behind perceptual audio coders such as MPEG-4 Advanced Audio Coder (AAC) is to hide quantization noise below the signal-dependent masking thresholds of the human auditory system [2, 13]. With AAC, the audio stream is divided into subbands using a filter bank, which uses a 1024 point Modified Discrete Cosine Transform (MDCT). Quantization noise (step size) is set separately in each subband to fall below the masking threshold. In addition, correlation between audio samples is used to remove redundancies.

The raw data produced by the AAC encoder is designed to be parsable so that it can be used with any data transport mechanism, including digital cellular air interfaces such as the EDGE system. The perceived audio quality is a function of this AAC raw data rate, which depends on the audio sampling rate and the compression rate. Typical AAC parameters are summarized in Table I, assuming an average compression ratio of 16:1, i.e., from 16 to 1 bit per sample on average and two audio channels for stereo. The AAC frame lengths for each channel is variable from frame to frame, depending on the music source, with an average of 1024 bits per frame (exactly 1024 samples per frame times an average of 1 bit per sample) per channel. Since each AAC frame is generated based on 1024 samples, the time period between two successive AAC frames is determined by the audio sample rate. For example, when the sample rate is 48 KHz, the frame time is 21.3 (=1024/48) msec as shown in the table.

The AAC frames that use the 1024 point MDCT (so-called long frames) are used when the music is stationary over the frame transmission time. So-called short frames are used when the audio changes rapidly within the frame transmission time, in which 8 consecutive 128 point MDCTs are used, thus sacrificing frequency resolution to capture the time dynamics. These short frames cannot be predicted from surrounding long frames, and thus can be designated high priority AAC frames.

Table I. AAC Data Rate and Music Quality

Highest audio freq. (KHz)	Audio sample rate (KHz)	Bits per sample per channel	Stereo data rate (Kbps)	Quality	AAC frame time (msec)
24	48	1	96	CD	21.3
16	32	1	64	FM	32
12	24	1	48		42.7
8	16	1	32		64
6	12	1	24	AM	85.3

The *n*th consecutive AAC frame in a given channel has a limited useful lifetime, comprising the startup (playout) delay plus *n* times the time per frame. If it is delayed beyond its lifetime, the AAC frame has to be dropped. For a given data rate, the audio quality is determined by the number and distribution of gaps in the AAC frame sequence at the receiver, where the gaps are caused by frames that cannot arrive on time due to transmission error. As a result, the distribution of AAC frame error rate plays an important in defining the music quality.

Of course, the gaps in AAC frames may be concealed using error mitigation techniques in which a prediction or interpolation algorithm is used to estimate the missing frame and thus "fill the gap." The details of the error mitigation algorithms lie beyond the scope of this paper. The performance of the error concealment schemes is best when only one or just a few consecutive AAC frames are missing. For most types of music it may be difficult to mitigate a gap of more than 3 AAC frames (192 msec at 32 Kbps) and still maintain audio quality. MPEG-4 AAC includes additional error robustness tools, which allow partially damaged AAC blocks to be reconstructed, as well as a more finely scalable data rate.

3. Existing Techniques in Use

We apply two techniques proposed in our previous work [5]: a) link-layer retransmission by use of an ARQ procedure, and b) packet shuffling, to improve music quality in this paper.

3.1 ARQ Retransmission

The EGPRS network provides an option of using link level Automatic Repeat reQuest (ARQ) to create a reliable pipe for data transfer over the radio link. We assume that the network can allow up to 3 retransmissions for each EDGE (radio) block in this study.

3.2 Packet Shuffling

In order to increase the effectiveness of any error concealment techniques, the AAC frames may be shuffled or interleaved at the server or at an intermediate node. The assembled AAC frames are then re-ordered by the receiver and placed in the playout buffer. The purpose of this shuffling and un-shuffling is to effectively spread out bursts of AAC frame errors over the radio link. Note, however, that as long as frame numbers (or a similar index) is included in the AAC frame, it is not necessary for the mobile to even be aware of the shuffling or interleaving. This is very desirable, as it implies that this feature can be activated without changing or re-programming the mobile terminals.

The specific shuffling technique proposed and simulated is based upon a convolutional interleaver [14, 15]. Two integer parameters, N and B, specify this interleaver. The values that we use are $N = 6$ and $B = 1$. If $T(k)$ denotes the position at which frame k is transmitted, then we have:

$$T(k) = k + (k \bmod N) NB. \tag{1}$$

The following table illustrates this interleaver ($N = 6$, $B = 1$):

Table II. Convolutional Interleaving for $N = 6$, $B = 1$.

k	0	1	2	3	4	5	6	7
$T(k)$	0	7	14	21	28	35	6	13

Therefore, the order of transmission of the frames is 0, 7, 1, 14, 8, 2, 21, 15, 9, 3, 28, 16, 10, 4, 35, 29, 17, 11, 5, 42, 36, ... It is easy to show that using this technique, two consecutive frames are separated by at least NB frames after the shuffling. (This statement is not applicable for a few initial frames, i.e., when the interleaver begins operation. However, this should have a minimal impact on performance for reasonable sized files.) This is why this technique is very effective in shortening gaps in the received AAC frame sequence that result from a burst of over-the-air transmission errors. On the other hand, this shuffling introduces a delay in the transmission of some frames, and therefore comes at a price of reducing the effective playout delay (or equivalently, the buffer size). Therefore, the choice of interleaver should provide a suitable trade-off between the frame error spreading and the reduced buffer size.

4. NEW LINK-LEVEL TECHNIQUES

4.1 Link-Adaptation Algorithm

We consider a setting where the adaptation algorithm adapts each link to one of six modulation/coding levels (or six modes), which represents an early proposal for the EGPRS network. As pointed out above, link adaptation can be designed to maximize network throughput as studied in [6, 7]. Specifically, the interference power is tracked and estimated to predict the SINR performance prior to the the link-adaptation decision (a method for such SINR prediction is discussed in the following subsection). Based on the predicted SINR, the modulation level is chosen according to Table III.

Table III. SINR Thresholds for Maximizing Throughput.

mode level	1	2	3	4	5	6
lower SINR	-	4	10	11	13	21
upper SINR	4	10	11	13	21	∞

For example, when the predicted SINR is greater than 11 dB, but less than or equal to 13 dB, mode 4 is chosen for the transmission. Thresholds in the table are adopted from Figure 1 of [16] for maximizing network throughput and the adaptation is thus referred to as the *throughput-based* scheme. We shall use this as a baseline case for comparison.

It is helpful to explain briefly how thresholds in Table III are obtained. For a given modulation level operating at a specified SINR condition, its bit error rate (BER) can be estimated for a particular channel model (e.g., Rayleigh channel). Since the data rate is fixed for each modulation level, its throughput, which equals data rate times one minus the BER, at the given SINR can be computed. As intuitively expected, for each modulation level, the throughput and SINR exhit a saturation-curve relationship. That is, when SINR is increased from a small value, the associated throughput increases quickly. As SINR continues to increase, the rate of increase in throughput decreases. As SINR goes beyond a certain value, throughput increases only marginally. Such SINR values for different modulation levels are typically different. With link adaptation, one can overlay the throughput-SINR curves for all modulation levels in a single plot. Then, it is evident that at a given SINR, the modulation level that yields the maximum throughput should be used for transmission. As a result, the relationship between the maximum throughput and SINR for such link adaptation is the upper envelop of the individual throughput-SINR curves for all modulation levels. Thresholds in Table III are thus obtained. It is interesting to point out that the same link-adaptation thresholds apply, independent of traffic-load conditions. However, when the traffic load increases, interference increases, thus decreasing the SINR. Consequently, it is likely to use low modulation levels for transmission, resulting in a low network throughput.

Now, let us examine another set of thresholds for link adaptation by considering error performance. We shall use a set of typical block error probabilities as a function of SINR for various transmission modes shown in Figure 2. For a given error target, one can determine the selection thresholds for link adaptation from this set of curves. As an example, the thresholds in Table IV are obtained to yield an approximate error rate of 15%.

Figure 2. EDGE Block Error Rate Vs. SINR

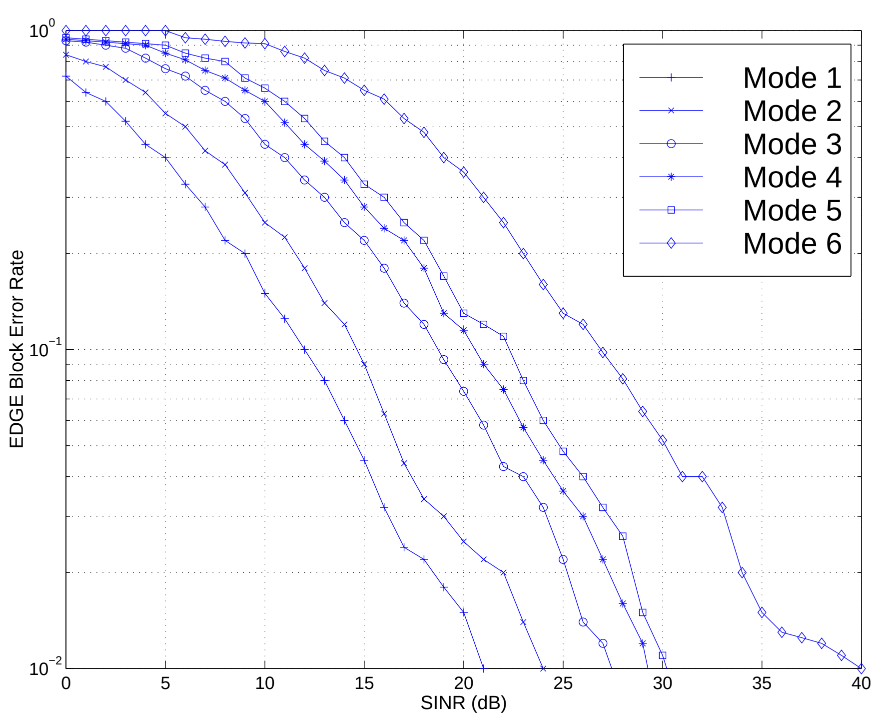


Table IV. SINR Thresholds for Error Performance.

mode level	1	2	3	4	5	6
lower SINR	-	10	13	17	19	20
upper SINR	10	13	17	19	20	∞

Since the thresholds are chosen to target at a specific error performance, the adaptation scheme is called the *error-based* link adaptation method. Note that the key difference between throughput and error-based adaptation is that the thresholds in Table III are lower than those in Table IV. As a result, one would expect that higher transmission modes are used more often by the throughput-based adaptation than by the error-based method. As our performance results show, the throughput-based scheme typically yields a higher throughput than the error-based approach. On the other hand, latter provides lower error probability than the throughput-based method does. As will be discussed later, this is a desirable characteristic of the error-based adaptation for real-time services such as streaming music.

4.2 SINR Prediction and Power Control

We apply the Kalman-filter method [12] to predict interference power for predicting SINR and adjusting transmission power. Using this method, each terminal continuously measures the interference power for its assigned radio channel (e.g., the same time slot of the consecutive TDMA frames), regardless of whether or not the base station associated with the terminal is transmitting data and whether or not the data being transmitted is intended for the terminal. The measurements are fed into a Kalman filter to predict future interference power. Let $\tilde{i}_n$ and $\tilde{I}_n$ denote the predicted interference power for frame n in mW and dBm, respectively. The time and measurement update equations for the Kalman filter for interference predictions are:

$$\tilde{I}_{n+1} = \hat{I}_n \tag{2}$$

$$\tilde{P}_{n+1} = \hat{P}_n + Q_n \tag{3}$$

$$K_n = \tilde{P}_n(\tilde{P}_n + R_n)^{-1} \tag{4}$$

$$\hat{I}_n = \tilde{I}_n + K_n(Z_n - \tilde{I}_n) \quad (5)$$

$$\hat{P}_{n+1} = (1 - K_n)\tilde{P}_n \quad (6)$$

where $\tilde{I}_n$ and $\hat{I}_n$ are the a priori and a posteriori estimate of interference power in dBm for frame *n*, $\tilde{P}_n$ and $\hat{P}_n$ are the a priori and a posteriori estimate of error variance, K_n is the Kalman gain, and Q_n and R_n are the variance for the process noise (i.e., the change of interference power of one frame relative to the previous one) and the interference measurement noise, respectively.

Let Z_n and $\overline{Z}_n$ be the measured interference power for frame *n* and the average interference over the last *W* frames prior to frame *n*, respectively. Hence, we have

$$\overline{Z}_n = \frac{1}{W}\sum_{i=n-W+1}^{n} Z_i. \quad (7)$$

We can estimate Q_n by

$$Q_n \approx \frac{1}{W-1}\sum_{i=n-W+1}^{n} (Z_i - \overline{Z}_n)^2. \quad (8)$$

Note that R_n depends on the actual error characteristics of interference measurements. However, to illustrate our ideas without considering details of the error characteristics, we assume for simplicity in this paper that R_n is given by

$$R_n = \eta Q_n \quad (9)$$

where η is a given constant, typically between 0 and 1. Readers are referred to [12] for the details of the Kalman method.

With the predicted interference $\tilde{i}_n$ in mW from (2), one can then estimated the SINR for frame *n* by

$$SINR_n = \frac{p_{\max} g_n}{\tilde{i}_n} \quad (10)$$

where $p_{\max}$ is the maximum power and g_n is the signal path gain between the transmitter and the intended receiver for frame *n* (which can be estimated and known to both transmitter and receiver by use of control channel associated with handoff purposes in the EDGE system). Then, the predicted $SINR_n$ is used in the link adaptation procedure to choose the transmission mode discussed above.

In addition, the predicted interference $\tilde{i}_n$ is also used to control transmission power p_n for frame *n*:

$$p_n = \min\left[\frac{\gamma^* \, \tilde{i}_n}{g_n},\ p_{\max}\right] \tag{11}$$

where γ^* is the SINR target associated with the adapted transmission mode (which is chosen to be the upper SINR value in Table IV for the adapted mode, with an exception that for mode 6, γ^* is set to be 25 dB). By aiming at the SINR target, the power control not only attempts to achieve the associated error rate, it also helps reduce the variance of error rates among users or terminal locations.

We note that since $p_{\max}$ is used in (10) for link adaptation, p_n by (11) is identical to $p_{\max}$ at the beginning of a packet transmission, except when the highest mode is selected for transmission. When the latter happens, p_n is lower than $p_{\max}$. In general, the transmission mode and power can be re-adjusted subsequently at different times. So, it is possible that p_n in (11) is no longer close to $p_{\max}$, but is rather adjusted according to changes of $\tilde{i}_n$ and g_n during the rest of the packet transmission. In case transmission mode and power are re-adjusted by (10) and (11) synchronously, p_n will be close to $p_{\max}$ for the entire packet transmission, except that p_n is lower than $p_{\max}$ for the highest mode. Section 5 considers such a case and our results show that transmitting at power lower than $p_{\max}$ for the highest mode can indeed deliver significant performance improvement. This is so because a good fraction of packets are transmitted using the highest mode for the reasonable network setting under consideration. When the highest mode is chosen, the radio link is likely to be good. Thus, transmission at power level lower than $p_{\max}$ in such cases reduces interference to neighboring cells without affecting the signal quality, thus improving overall network performance. We remark that transmission power other than the maximum $p_{\max}$ could be used for predicting SINR in (10), which may yield further performance gain, and this represents a topic for future investigation.

It is also worth noting that the power control represents a closed-loop control that requires exchange of control information between the receiver and the transmitter. Such information exchange can be made possible by including pertinent information in appropriate control messages in the EDGE system. For downlink transmission of music data, the receiving terminal continuously measure interference power and send the measurements to the BSS in the control messages. The computation associated with the Kalman filter can be performed by the base station. Finally, the base station transmits at the power level p_n in Eq.(11) for each frame *n*. Readers are referred to [17] for the exchange of power-control information between receivers and transmitters.

5. PERFORMANCE STUDY

5.1 Simulation Model

We simulate an EGPRS network with a total of 37 cells in a traditional hexagonal layout. Each cell is divided into three sectors, each of which is served by a base-station antenna at the center of the cell. The 3-dB beamwidth of each base-station antenna is 60 degrees, while mobile terminals have omni-directional antennas. The radiation pattern for the base-station antenna is assumed to be a parabolic shape. That is, a 3 dB drop occurs at the 30-degree angle and any direction beyond a threshold angle in clockwise or anti-clockwise direction suffers a fixed attenuation relative to the gain at the front direction, which is called the *front-to-back* (FTB) ratio. We assume the FTB ratio is 25 dB for the base-station antenna. Frequency reuse factor of 1/3 (i.e., the available spectrum is divided into 3 frequency sets and they are re-used in every cell) is assumed. Each radio link between a terminal and a BS antenna is characterized by a nominal propagation model with an exponent of 3.5 and lognormal shadow fading with a dB standard deviation of 6. Cell radius is assumed to be 1 Km and the median path loss at 100m from the cell center is set to -73 dB. Thermal noise at each receiver is fixed and equal to -116 dBm, which corresponds to the 200 KHz GSM channel with 5 dB noise factor. Transmission power can vary between 1 to 30 dBm. Since our focus here is to study the performance improvement by the error-based adaptation and power control, only one downlink channel (i.e., use of one of 8 time slots in the GSM TDMA frame) shared by all music and data users, is considered in our model to reduce simulation time.

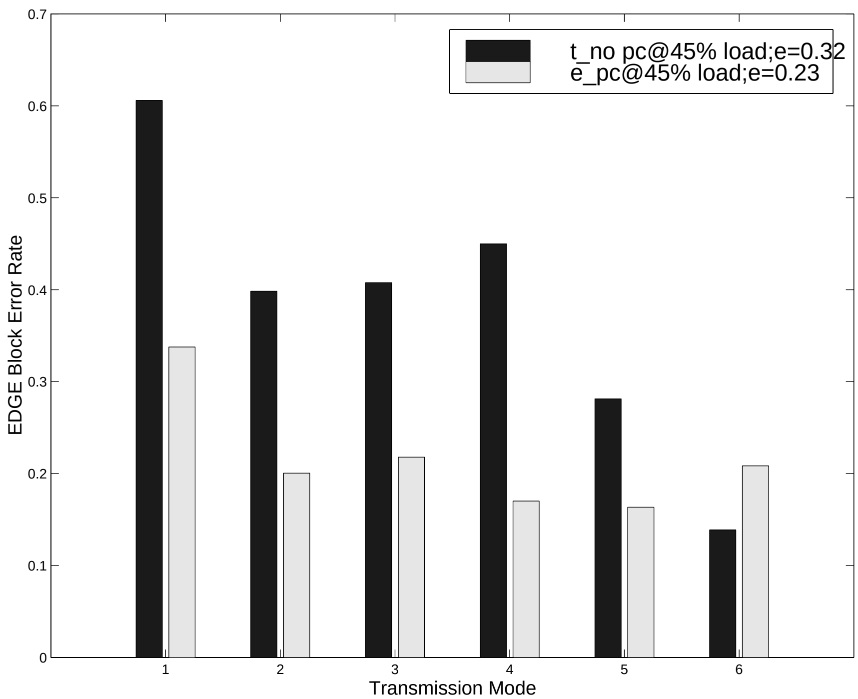
In reality, a terminal can move from place to place while receiving music data. To study the improvement of music quality due to our link-level techniques, we have to examine the aggregated error improvement for a large number of moving terminals under different traffic load conditions, which requires a huge amount of simulation time. Instead, we use an alternative approach as follows. Each sector is populated with 100 terminals randomly, each of which selects the BS that provides the strongest signal power. We assume that all terminals remain at the fixed locations. To capture the effects of mobility, terminals in the same sector take turns in a random fashion to be the recipient of the data being transmitted by the BS. After sending data to a terminal for an average of 0.6 second, the channel either remains idle for a random period of time or starts to send data to another randomly selected terminal in the sector. As a result, the channel alternates between transmitting and being idle and each transmission period corresponds to data transmission to a specific terminal. The idle time has a geometric distribution and its average is determined to match a given traffic load of the channel. The 0.6-second duration is chosen because the time constant for changes of shadow fading for moving terminals and interference power in the packet-switched network is on this order of magnitude. Specifically, based on the EDGE data rates for the single time-slot operation, the transmission time for sending a typical IP packet is about 0.6 second on average. This way, we can use the overall error statistics for all terminals to approximate the expected error performance for a randomly selected, moving terminal.

We note that although a channel alternates between transmitting and being idle according to the specified traffic load as described above, the accuracy of interference estimation by the Kalman filter, (2) to (6), is adequate for our purposes because of "averaging" effects due to existence of many close interferers in the tight frequency reuse of 1/3.

A radio block is the basic transmission unit of the EDGE system. For each block transmission, the SINR is measured at the receiving end, which in turns depends on the path loss, shadowing and interference power. The SINR measurement is rounded to its closest integer in dB and the block error state (good or bad) is determined based on the SINR value the adapted transmission mode, and the corresponding error probabilities in Figure 2. (Note that Rayleigh fading is included in the link performance curves.) When power control is used, the SINR target γ^* in (11) is chosen to be the upper SINR values for the adapted transmission mode in Table IV, with an exception that for mode 6, the target is set to be 25 dB. The initial transmission mode and power are chosen according to (10) and (11) at the beginning of each packet transmission. The mode and power are re-adjusted every 5 slots throughout the rest of the packet transmission.

As discussed earlier, each AAC frame is divided into a number of EDGE blocks, each of which is sent over the radio link. Each AAC frame has a useful lifetime and a frame is in error if any of its blocks either arrives too late, or is received in error after retransmission for a pre-specified number of times. The ultimate, perceived music quality is characterized by the AAC frame error rate (FER). In particular, we use the FER over a one-second sliding window as a measure of quality. In addition to considering AAC FER, we also obtain results to show the performance improvement by the error-based adaptation and power control at the EDGE block level. They are needed because the latter provides additional insights into the performance improvement. As discussed above, an error at the EDGE level may be mitigated by the ARQ procedure before presenting the associated AAC frame to the user. In addition, packet shuffling at the AAC frame level help randomize the errors among various AAC frames so that error concealment techniques can be effective in maintaining music quality in case of missing frames. The effects of error concealment are not examined in this study.

Figure 3. EDGE Block Error Rate Vs. Transmission Mode

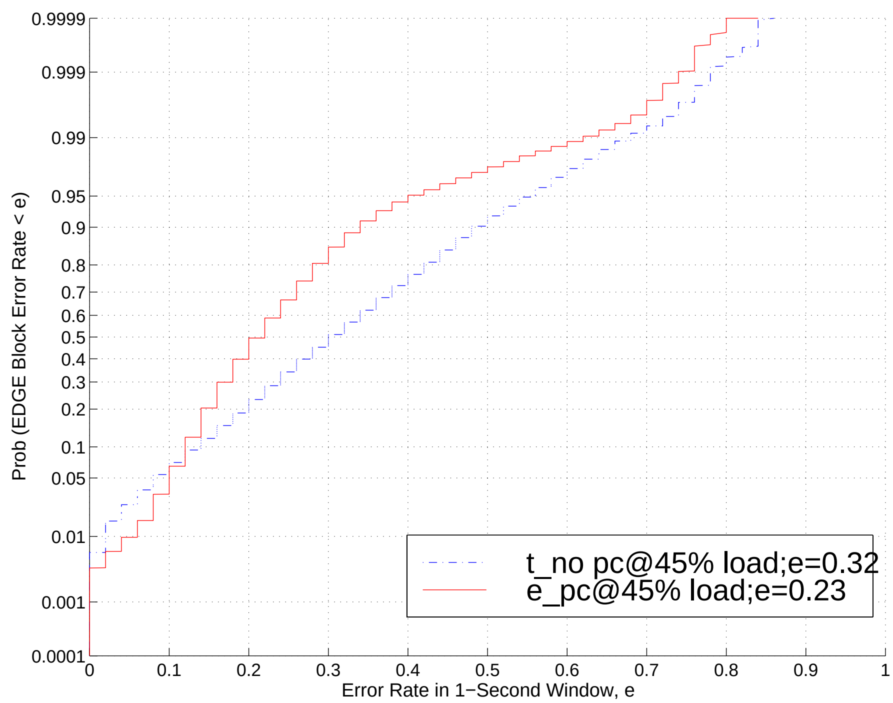


5.2 Numerical Results & Discussion

Figure 3 shows the EDGE block error rate for all six transmission modes for the throughput-based adaptation with no power control and the error-based adaptation with power control, which are labeled for brevity as "t_no pc" and "e_pc" cases in the figure, respectively. This figure assumes that the downlink channel is loaded with 45% offered traffic, excluding ARQ retransmissions. One can make two observations from these results. First, the EDGE block error rate for the throughput-based and error-based scheme is about 32% and 23%, respectively. Although not shown in the figure, their corresponding average throughput is 31.9 and 26.3 Kbps, respectively. Evidently, the throughput-based scheme indeed can offer a higher throughput than the error-based method. However, the latter provides better error performance than the latter. For real-time services with delay constraints, only limited ARQ retransmission is feasible. In such cases, as shown below, better error performance for the error-based method yields better music quality, despite its slight disadvantage in terms of throughput relative to the throughput-based scheme. Second, the error rates for various transmission modes under the throughput-based scheme have a higher variability than those for the error-based scheme. As intuitively expected, except for mode 1 (i.e., the most robust modulation and coding), the error rate for all other modes generally fall between 15 to 20% under the error-based adaptation as the SINR thresholds are selected to yield a target error of 15% in Table IV. The relatively higher error rate for mode 1 is due to a fact that it is used for transmission even when the link quality is extremely poor. In this case, there is no "guarantee" of error performance for the lowest mode. For the throughput-based scheme, the error rate for transmission modes 5 and 6 are noticeable lower than that for other modes because when the high modes are used for transmission, the link quality is usually good for correct reception. This disparity results into a high variability of error rate for various modes. In essence, the error-based scheme has achieved the goal of reducing the variance of error rate. As to be discussed later, such improvement at the EDGE block level can help maintaining the music quality at the AAC frame level.

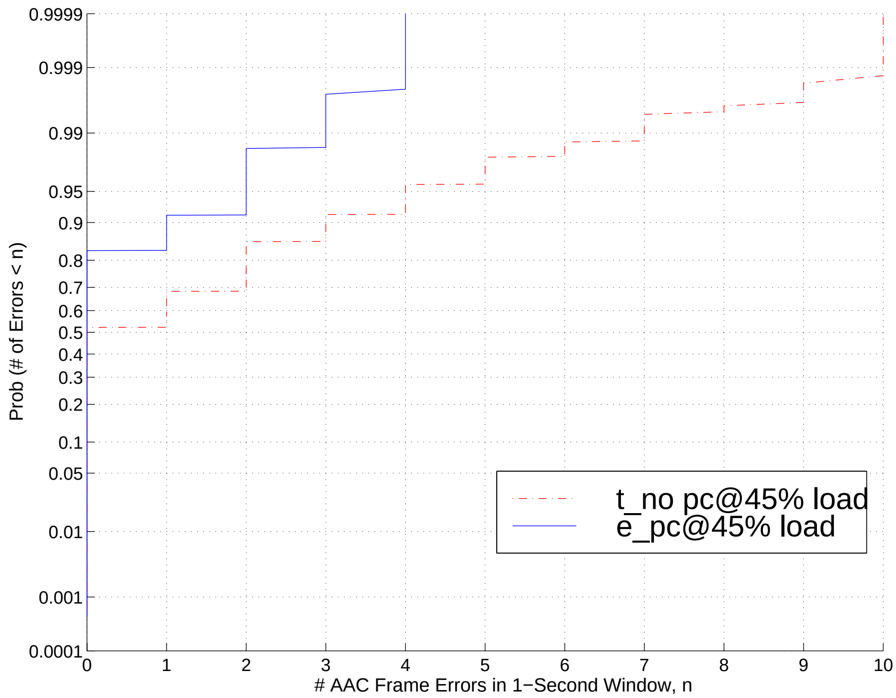
Figure 4 presents the cumulative distribution function (CDF) of EDGE block error rate over a sliding window of 1 second with the traffic load of 45%. As shown in the figure, except for a percentile of less than 5%, the error-based adaptation with power control yields better error performance at the EDGE block level than the throughput-based adaptation with no power control. Evidently, the cross-over at about 5 percentile error rate is due to a fact that the power control limits the transmission power for few terminal locations with excellent link quality, while the throughput-based scheme continues to transmit at full power to those locations. As a result, the error rate at those locations is lowered for the throughput-based scheme. Nevertheless, the associated link quality at those locations is already excellent. One can afford to degrade their error performance a little bit by transmitting at less than the maximum power level.

Figure 4. CDF of EDGE Block Error Rate with 45% Load



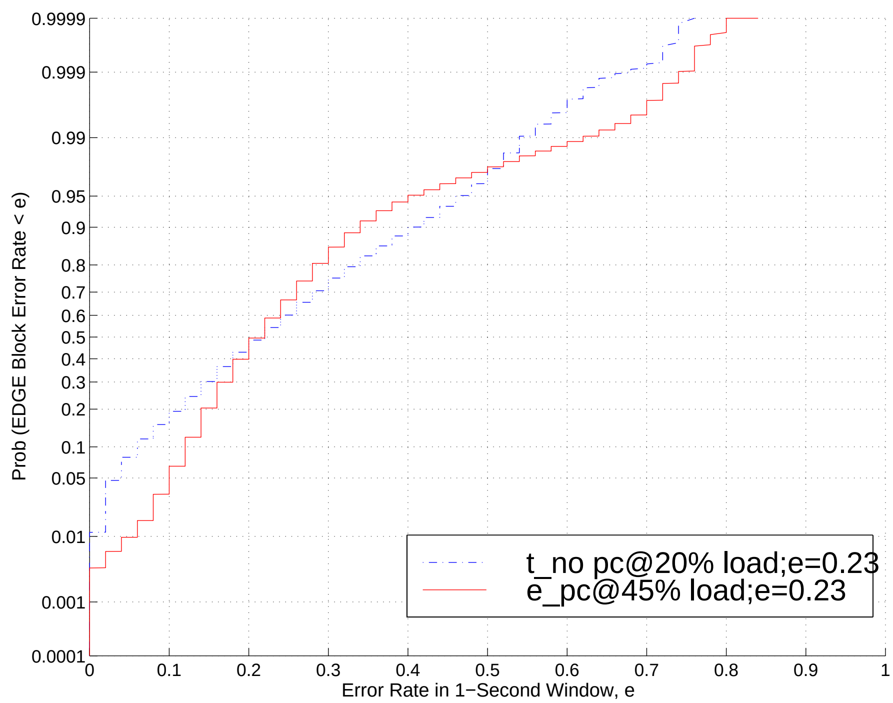
Recall that each AAC frame is segmented into a number of fully and partially filled EDGE blocks. A reception error for an EDGE block may be mitigated by the ARQ procedure and packet shuffling helps randomize the distribution of AAC frame errors. Corresponding to Figure 4, the CDF for the number of AAC frame errors over a sliding window of 1 second is presented in Figure 5. These results reveal a substantial error improvement for the error-based scheme over the throughput-based scheme at the AAC frame level. Specifically, the 91st percentile of the number of AAC frame errors in the one-second window is reduced from 3 for the throughput-based scheme to 1 by the error-based scheme. Such a reduction in error rate translates into a noticeable improvement of music quality.

Figure 5. CDF for Number of AAC Frame Errors with 45% Load



While Figures 3 to 5 present the error improvement for the error-based scheme for a given traffic load, a question that needs to be addressed is that: how does the improvement translate into an increase of network capacity? We attempt to provide some insights into this issue in Figure 6 and 7. In particular, we compare the CDF for the EDGE block error rate for the throughput-based scheme with original traffic load of 20% to that for the error-based scheme with 45% traffic in Figure 6. Coincidentally, despite the difference in offered traffic, both schemes yield an error rate of about 23% averaged over a relatively long period of time. However, in terms of number of EDGE block errors in the one-second sliding window, the CDF's in the figure show that depending on the error percentile, one scheme can perform better or worse than the other. Nevertheless, one can observe that for 50th or lower percentile, and 97th or higher percentile, the error-based scheme yields a higher block error rate than the throughput-based scheme does. The opposite is true between 50th to 97th percentile.

Figure 6. CDF of EDGE Block Error Rate with 20/45% Load

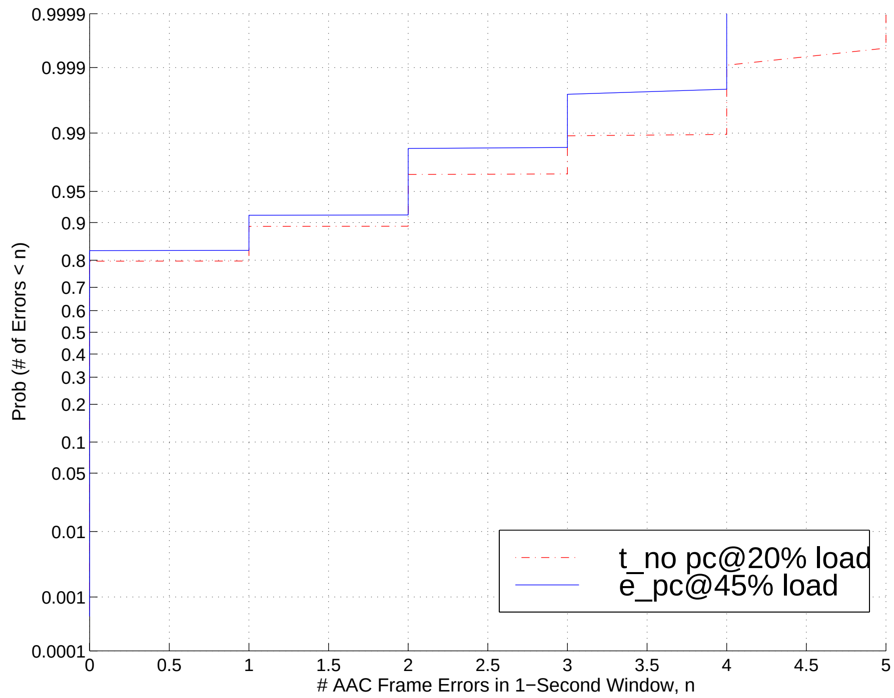


While the comparison of EDGE block error rate in Figure 6 does not yield a clear conclusion, the CDF's for the number of AAC frame errors in a one-second window in Figure 7 should provide a better understanding. Results in the latter figure reveal that the error-based scheme yields slightly better AAC frame error rate than the throughput-based scheme. Specifically, the probability of having at most 2 AAC frame errors in a one-second window is about 98% and 96% for the error and throughput-based scheme, respectively. Therefore, to provide similar music quality, the error-based scheme can effectively support an offered traffic load increased from 20% for the throughput-based scheme to 45%. Although not shown in the figure, the associated throughput (i.e., average data rate times link utilization) is about 10.2 and 17 Kbps for the 20% and 45% load cases, respectively. Thus, the error-based scheme in effect yields a throughput increase of 66.7% over the throughput-based scheme to achieve the similar music error performance.

It is worthnoting that there is a tradeoff among buffer size, ARQ retransmission, packet error rate and throughput. Specifically, if a large buffer can be used to buffer music packets before playout to receiving users, further advanced techniques can be applied to conceal packet errors and a high degree of retransmissions can be tolerated. In this case, the main design concern is not packet error performance. Instead, one should maximize the data throughput, in hope of further enhancing music quality and/or supporting additional users. On the other hand, when the playout buffer size and thus retransmission are limited, as has been in this study, packet error performance should be the primary concern and data throughput is a secondary one. Given an error performance requirement, our results reveal that the error-based scheme outperforms the throughput-based scheme in terms of meeting the target error performance as well as improving overall throughput.

One could choose a set of SINR thresholds higher than those in Table IV for the error-based scheme. In such a case, most of the packet transmissions will be carried out using low modulation levels. Thus, the overall throughput is lowered (which can eventually affect the music quality), although further improving packet error performance. As a result, the tradeoff among error performance, throughput and degree of retransmissions is also changed. With very limited retransmission for music services, our results show that the SINR thresholds with a target packet error rate of 15% in Table IV appear to be able to achieve a balanced tradeoff between error and throughput performance for the EGPRS wireless network. Evidently, to identify the optimal SINR thresholds for other real-time applications with different error rate and retransmission requirements remains an open research topic.

Figure 7. CDF for Number of AAC Frame Errors with 20/45% Load



6. Conclusions

In this paper, we continue our investigation into the possibility of embedding streaming data services in an EGPRS network. Specifically, we have examined how a combined link-adaptation and power-control scheme for achieving a target error rate and reducing error variability can help improve the quality of MPEG-4 AAC coded music. By simulation techniques, we have compared the error performance at both the EDGE block and AAC frame level of this error-based scheme with the throughput-based adaptation algorithm for maximizing network throughput. Our results reveal that the error-based scheme can provide noticeable improvement of music quality over the throughput-based scheme when offered with similar traffic load. This is so because with the limited ARQ retransmission, the error-based scheme aims at achieving a satisfactory packet error rate and helps reduce the error variability for various transmission modes and users. Specifically, with 45% offered traffic load for a 1/3 frequency reuse, the 91st percentile of the number of AAC frame errors in the one-second sliding window is reduced from 3 for the throughput-based scheme to 1 by the error-based scheme. To achieve similar AAC frame error rate, our results also show that the error-based scheme can increase the radio link throughput over the throughput-based scheme by 66.7% in one of our examples. Based on these results, we conclude that by aiming at required error targets and thus reducing error variability, the error-based scheme for link adaptation and power control are helpful in improving quality and capacity for streaming applications.

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REFERENCES

[1] Special Issue on "The Evolution of TDMA to 3G," *IEEE Personal Communications Magazine,* Vol. 6, No. 3, June 1999.

[2] M. Bosi, et al., "ISO/IEC MPEG-2 Advanced Audio Coding", *J. Audio Eng. Soc.,* vol. 45, No. 10, October 1997, pp. 789-812.

[3] A. Furuskar, et al., "EDGE: Enhanced Data Rates for GSM and TDMA/136 Evolution," *IEEE Personal Communications Magazine,* Vol. 6, No. 3, June 1999, pp. 56-66.

[4] T. S. Rappaport, *Wireless Communications: Principles and Practice,* New York: IEEE Press and Prentice Hall, 1996.

[5] K. Chawla, P. F. Driessen, and X. Qiu, "Transmission of Streaming Data Over an EGPRS Wireless Network," *Proc. of IEEE Veh. Tech. Conf.,* Tokyo, Japan, May 2000.

[6] X. Qiu and J. Chuang, "Link Adaptation in Wireless Data Networks for Throughput Maximization Under Retransmissions," *Proc. of IEEE ICC'99,* Vancouver, Canada, 1999, pp. 1272-1277.

[7] J. Chuang, X. Qiu and J. Whitehead, "Data Throughput Enhancement in Wireless Packet Systems by Improved Link Adaptation with Application to the EDGE System," *Proc. of IEEE Veh. Tech. Conf.,* Amsterdam, The Netherlands, Sept. 1999.

[8] K. K. Leung and L.-C. Wang, "Integrated Link Adaptation and Power Control for Wireless IP Networks," *Proc. of IEEE Veh. Tech. Conf.,* Tokyo, Japan, May 2000.

[9] K. K. Leung and L.-C. Wang, "Controlling QoS by Integrated Power Control and Link Adaptation in Broadband Wireless Networks," *European Trans. on Commun.*, Vol. 11, July-Aug 2000, pp. 383-394; early results in *Proc. of IEEE PIRMC'99*, Osaka, Japan, Sept. 1999, pp. 1174-1180.

[10] Special issue on power control, *Wireless Networks*, 4 (1998) 3, Zvi Rosberg and Jens Zander (Ed.).

[11] S. Ulukus and R. D. Yates, "Stochastic Power Control for Cellular Radio Systems," *IEEE Trans. on Communications*, Vol. 46, June 1998, pp. 784-798.

[12] K. K. Leung, "A Kalman-Filter Method for Power Control in Broadband Wireless Networks," *Proc. of IEEE Infocom'99*, New York, NY, March 1999, pp. 948-956.

[13] http://www.mpeg.org/MPEG/aac.html

[14] J. L. Ramsey, "Realization of Optimum Interleavers," *IEEE Trans. Inform. Theory*, Vol. IT-6, No. 8, May 1970, pp. 338-345.

[15] E. K. Hall and S. J. Wilson, "Convolutional Interleavers for Stream-Oriented Parallel Concatenated Convolutional Codes," *Proc. of ISIT*, August 1998, p. 33.

[16] J. C.-I. Chuang, "Improvement of Data Throughput in Wireless Packet Systems with Link Adaptation and Efficient Frequency Reuse," *Proc. of IEEE Veh. Tech. Conf.*, May 1999, pp. 821-825.

[17] J. Chuang, K.K. Leung, X. Qiu, S. Timiri and L.-C. Wang, "Power Control for Wireless Packet Voice with Application to EDGE System," *Proc. of IEEE Globecom 2000*, San Francisco, CA, Nov. 2000, pp. 213-219.

Figure and Table Captions