LINK ADAPTATION TO CHANNEL INTERFERENCE USING MULTI-RATE SOURCE AND CHANNEL CODING FOR CDMA MOBILE COMMUNICATIONS

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ABSTRACT

This paper investigates the benefits of link adaptation according to the propagation conditions using multi-rate vocoders and channel coders. A multi-rate source and channel coding (MRC) system trades-off the source codec bit rate against channel codec bit rate according to the prevailing channel conditions. MRC systems continuously reconfigure transmission resources to optimally combat channel degradations and hence, maintain a reasonable quality of service even in marginal channel conditions. The aim is to maintain a satisfactory level of speech quality over a wide range of channel conditions with minimum resources. This paper shows the benefit of MRC system particularly in crowded cells of high channel interference.

1. INTRODUCTION

In recent years, the rapid growth of mobile communication systems and the emergence of a universal mobile telecommunication system (UMTS) concept have highlighted the importance of this market. Increasing numbers of users have to be supported in these systems. Unfortunately, the allocated frequency bandwidth is limited for these services. This has emphasised the need for an efficient use of the transmission medium by using low bit rate (LBR) speech coding algorithms, the most efficient of these being model-based coders. The corruption of these models' parameters often results in high subjective degradation of the received speech quality. Forward error control (FEC) strategies must therefore be used to limit the degrading effects of channel errors on the output speech and to assure the integrity of each user's communications.

In practice, FEC are deployed to maintain an average quality of service for 90 to 99% of the time for worst channel conditions. This situation is similar to the allocation of a fade margin in link budget calculations for modulated channels. When channel conditions exceed the worst designed case, conventional FEC techniques fail, leading to service interruption. In better channel conditions which predominate most of the time however, less number of FEC bits are needed to maintain the output quality. Thus, the allocated redundancy for FEC subtracts from the overall channel bit rate which could have been used for the speech coder to enhance the quality and/or allocate more users into the system.

To avoid this redundancy during good channel conditions, link adaptation which reconfigures transmission resources can be used [1, 2]. This adaptation can optimally combat channel errors and hence, maintain a reasonable quality of service even in marginal conditions. One such link adaptation techniques employs multi-rate source and channel codecs (MRC) by trading-off the gross bit rate of each traffic channel between the speech and channel codecs according to the prevailing channel conditions. Thus, when channel conditions worsen, the transmitter reduces its speech rate and increases its channel coding rate. This introduces more coding gain and so enhances the receiver's ability to mitigate the higher degradation. Whilst the lower rate speech codec is expected to produce reduced quality speech, the perceived degradation is less than that of a fixed rate system employing FEC bits that are not adequate enough to combat sever channel conditions. Thus, a multi-rate coded system is capable of maintaining acceptable quality of service and preserving the integrity of the communication link even in marginal channel conditions. Whilst intermittent, such marginal channel conditions are quite common in real systems. These are encountered for example just prior to hand-off; during deep shadowing such as is experienced when a communicating mobile suddenly goes behind a building or hill; during the power ramp-up phase of mobiles joining the system; and in overcrowded cells in peak traffic.

This paper presents an overview of the MRC system configuration and its performance in crowded mobile cells which result in high users' interference. This paper is organised as follows: section 2. presents a brief description of the MRC concept and system configuration. The speech and channel codecs are briefly described in 3. Then in 4. a CDMA verification environment is presented which is designed to assess the MRC system. The rate switching algorithm and network protocol are described in 5. Finally, section 6. details some experiments and simulation results.

2. MRC CONCEPT, OPERATION AND CONFIGURATION

Figure 1 depicts a schematic block diagram of the MRC system. It presents a CDMA-based communication link with multi-rate voice and channel coders. The MRC system operates with a gross rate of 13kb/s achieved in four modes (0,1,2 and 3) corresponding to a four-rate vocoder with corresponding four-rate channel coder as shown in table 1.

The transmitter Initially operates with mode 1 and depending on the prevailing channel conditions it switches

Mode of Operation	0	1	2	3
Speech Coding Rate (kb/s)	9.6	8.0	6.4	4.8
Channel Coding Rate (kb/s)	3.4	5.0	5.6	8.2
Gross Rate (kb/s)	13.0			

Table 1. MRC operational modes

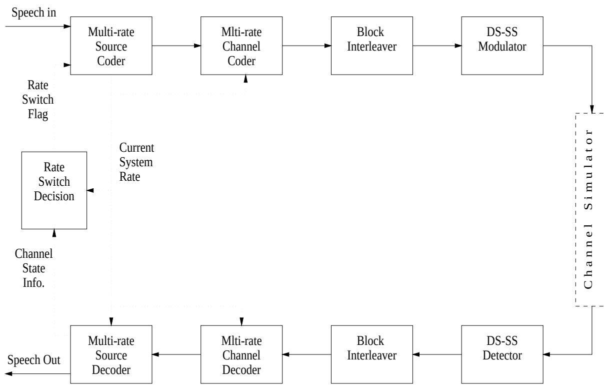

Figure 1. MRC-CDMA schematic block diagram

mode or continues in its current mode. The rate switching, detailed in section 5., is carried out from one mode to another by increments or decrements of one, in order to smooth the quality transition between different rates. As the mode changes from 3 to 2, 2 to 1, or 1 to 0, the vocoder rate increases at the cost of reduction of channel coding redundancy. Since the channel coding gain is proportional to the amount of redundancy bits, MRC operates at four different Eb/I_0 giving the system more flexibility in combating interference and fading. This implies that the lower vocoder rate modes can operate in more degraded channels.

Ideally, the system operates in mode 0 in good channel conditions. It takes the system therefore 20ms to switch from the initial mode 1 to mode 0 unless the channel conditions degrade whence a lower rate vocoder is activated until the conditions improve again. This also implies a reduction in the speech quality as a result of switching to a lower vocoder rate. Fig. 2 depicts the output speech quality variation as a result of switching the rates. The figure shows speech output relative segmental SNR of the modes (0,1), (1,2), and (2,3). Each second mode is injected randomly into the original mode with 10,20,40,50 % rate switching. The results show that the variation in quality as a result of switching is not very significant and changes slowly with the rate of injection. However, it was shown in [3] that modes 0 and 1 prevail for most of the time thus predominating the quality.

3. MULTI-RATE SOURCE AND CHANNEL CODECS DESCRIPTION

The multi-rate speech codec is based on Pulsed Residual Excited linear prediction (PRELP) [4], a CELP-type algorithm with an algebraic codebook. This codec has important features of low complexity, gradual quality differential between rates, and continuity in codec filter memories during rate changes [5]. The codec is based on a 20ms LPC frame length. The configuration for each rate assignments is summarised in Table 2. Associated with the speech codec

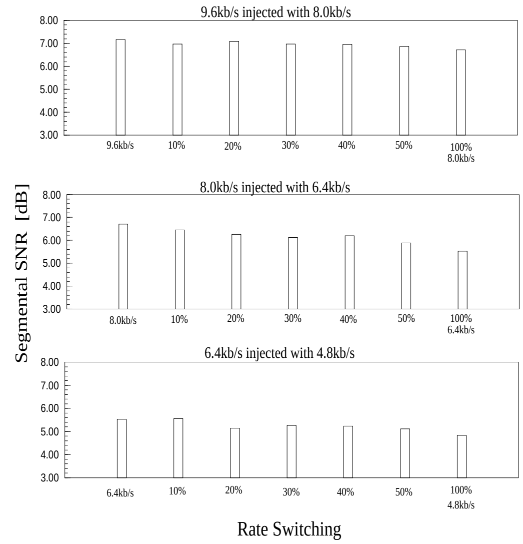

Figure 2. Performance of multi-rate codecs with rate switching

is also a voice activity detector (VAD) which can be enabled during experiments involving discontinuous transmissions (DTX).

Rate kb/s	Spectral Envelope bits/frame	Frame rms	LTP lag	LTP gain	CB Index	CB gain
9.6	37	6	45	25	45	30
8.0	28	6	36	20	40	24
6.4	28	4	28	20	32	16
4.8	28	6	18	10	22	12

Table 2. Bit Assignments Table for MRC vocoders

Subjective tests were carried out for each mode and the 16kb/s G.728 as shown in Table 3. This gives an indication of the subjective quality that each mode carries.

Rate kb/s	G.728	9.6	8.0	6.4	4.8
MOS score	4.0	3.95	3.8	3.6	3.3

Table 3. Subjective Test Results

The multi-rate channel codec used applies unequal error protection (UEP) and is based on a $\frac{1}{2}$ rate convolutional code and $\frac{2}{3}$ rate and $\frac{3}{5}$ rate punctured convolutional codes of constraint length 7. For each rate, the error sensitivity of each transmitted bit was determined [5]. The error sensitivities were then used to classify the bits for UEP by the punctured convolutional codes. For each rate, an inner CRC is used to detect lost frames at the output of the Viterbi decoder. This information is then signalled to the speech decoder to enable lost frame reconstruction.

4. CDMA VERIFICATION ENVIRONMENT

A dynamic link adaptive simulator (DLS) based on a CDMA environment was developed [6] to assess the MRC system in a realistic environment. In this system, an adaptive link between the mobile station (MS) of interest and a base station (BS) is simulated incorporating power control. The effects of terrain, shadowing, multipath fading, power control errors, neighbouring cells, and users' mobility, traffic, and interference on this link are computed.

Initially, the DLS assumes a given number of users in the cell distributed randomly. The output of a simple traffic model simulating the MS's entering and leaving the cell of interest is fed to the interference model. The DLS updates the locations and speeds of each MS according to a linear mobility algorithm. The DLS creates a database of information that includes number of users, each user's location relative to the BS, each user's speed, direction of roaming (towards or away from BS), transmit power, and power control error. This information is updated at every closed loop power control clock edge. This information assists the DLS to compute the intra-cell interference power as shown in 4.1.

The radio propagation model used in the DLS uses the Hata propagation model [7]. This model computes the long-term mean value of the path loss. It incorporates a Log-normal long-term fading with a standard deviation of 8 dB which represents the effect of different shadowing obstacles. The short-term fading component is calculated according to a Rayleigh distribution function that is dependent on the mobile's velocity. The multipath model is based on a set of 6-tap channel impulse responses positioned differently according to the terrain type [8]. The DLS can be configured for different types of terrain (urban, suburban, rural) and can be adapted for changes in the BS and the MS's antenna heights and gains.

The BS measures the received powers of the signal transmitted by the MS's and sends power control command to increase/decrease their power. Since diversity reception exploited by the use of the RAKE receiver reduces the envelope variation due to fading, deep fading typically encountered in narrow band transmission is eradicated. Therefore, it is reasonable to assume that the envelope variation after power control is due to the power control error whose values follow a certain distribution function [9]. This can be shown to be a log-normal distribution with a 1-3 dB standard variation depending on received signal-to-interference ratio (SIR) estimation accuracy, power control step size, and the frequency of power control commands. The closed loop power control algorithm implemented in this simulation is similar in principle to that of IS-95 with the ability to reconfigure its parameters.

A simple Markov loss teletraffic model is utilised to increase/decrease number of users. Also, a mobility model that computes mobile location with respect to the BS is used [6]. The handover algorithm used in the DLS system is simple hard handover. When a mobile crosses the border of the cell of interest it is dropped.

4.1. Interference Modelling

In a typical multi-user CDMA model [10], the output of the correlation receiver Z constitutes of $Z = D + M + I + N_0$ where D is the desired component, M is the multiple access interference, I is the multipath interference component, and N_0 is the AWGN noise term. The multiple access component can be represented using standard Gaussian approximation (SGA) based on the central limit theorem assuming that all users' interference are mutually independent random variables and tend towards normal distribution. This method is often accurate in applications where there are a large number of simultaneous users on the channel [11]. In practice, the multiple access interference is dependent on the relative time and phase offsets of the interfering signals as well as the aperiodic autocorrelation function of the signature sequences. A more accurate improved Gaussian approximation (IGA) can be obtained by finding the distribution of the variance of the multiple-access interference over all possible combinations of desired signature sequences and interfering signal delays and phases [12].

Power control error should be considered in the calculation since it significantly increases the interference [13, 9]. For the application in this paper, however, we are only interested in the effect of the users' interference on the MRC system when the number of simultaneous users is large. Therefore, the SGA model was used to model the interference as derived in [14]. In addition, a fixed factor is used to represent the neighbouring cells interference [15]. This factor can be adaptive to other cell load and users' handover.

5. RATE SWITCHING ALGORITHM

Initially, the MRC system operates in mode 1. The rate switching model is activated at the receiver according to the prevailing channel conditions. This is sensed by the total BER of the received data at the base station (BS). The BS monitors the BER by counting the number of corrected errors and lost frame decisions from the channel decoder. When the number of errors exceeds a threshold, the receiver issues a command to the transmitter to switch to a higher mode. This is done on a frame of 20ms bases. The commands are issued to increment or decrement the modes. When the the MRC is operating at mode 3, any increment command will be ignored. Similarly, when it is operation on mode 0, any decrement command is ignored. The transition between modes is conducted consecutively to smooth the transition in the output speech quality.

The operating rate can be indicated to the receiver using two bits that are punctured in the traffic bit stream using symbol puncturing similar to the power control commands that are used in IS-95 [16]. In our simulation, we send the rate switch commands on a separate data channel subjected to an error rate of 0.05%.

6. RESULTS AND CONCLUSION

We present here the performance of the MRC system under the DLS using initial number of users of 30 distributed randomly in the cell of interest. The power control commands and the rate switching command error rate are assumed to be 0.05%. We assume that voice activity of each user is 40%.

Figure 3 presents a comparative performance of the MRC system using rate switching versus a single rate system which employs the 8.0kb/s vocoder. Both systems were tested under the same conditions of the DLS. The figure depicts the segmental signal-to-noise (SegSNR) of both systems versus the Ec/I_0. The Ec/I_0 varies at the base station according to the channel conditions. Though objective

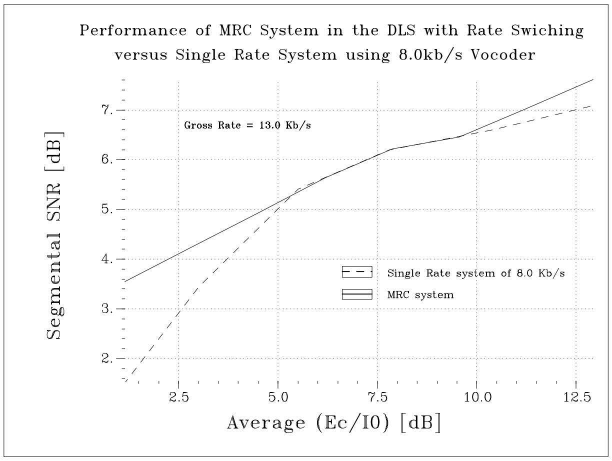

Figure 3. Quality of Speech Output of the MRC system

measurements do not in general provide good indications of absolute speech quality for parametric codecs such as PRELP, it is reasonable to use such measures to indicate relative differences between variations of the same codec [17]. The figure shows that for channel conditions with Ec/I_0 below 5.0 dB the 8.0kb/s system drops the output speech quality dramatically. The MRC on the other hand maintains a reasonable quality at low Ec/I_0 by switching to a lower rate. When the system is operating with Ec/I_0 between 5.0 and 9.5 dB the 8.0kb/s and the MRC system performs the same SegSNR (5.2-6.5 dB) which indicates that the MRC is operating in the 8.0kb/s mode. Finally, when channel conditions are best (i.e. Ec/I_0 is above 9.5 dB), the MRC system operates at its highest rate, 9.6 kb/s, giving a SegSNR of up to 7.6 dB and performing better than the 8.0kb/s system.

In conclusion, the MRC system is particularly useful with severely degraded channel conditions and can maintain the output speech quality with minimum resources, particularly when the interference is high in a crowded cells.

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