Pizza Party Algorithm for Real Time Distortion Management in Downlink Single-Cell CDMA Systems

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Abstract

In the interference-limited CDMA networks, the maximal number of users of real time applications can be increased by smoothly increasing the end-to-end distortions. In this paper, each user can accept a range of carefully controlled distortions. We develop a system protocol to control each user's distortion by adapting the resources like source coding rate, channel coding rate, transmit rate, and transmitted power. We want to reduce the overall distortion in downlink single-cell systems, under the constraints of users' maximal distortions and the maximal transmitted power from the base station. In order to solve such a difficult problem, inspired by an event of everyday life, we develop a heuristic and fast adaptive algorithm to allocate these resources for different users. The idea is to initialize the resource allocation with the maximal distortion for all users, and then allocate the remaining transmitted power quota first to the user who can most easily be satisfied when reducing its distortion. This user must have a high distortion, have a good channel condition, or generate small interferences to others. The allocation process is continued, until the transmitted power is used up. From the simulation results, our proposed algorithm fundamentally reduces the distortions and the necessary maximal transmitted power when the number of users is large, compared with a traditional CDMA scheme (with no distortion control).

1 Introduction

In Code Division Multiple Access (CDMA) systems, all users transmit simultaneously over the same frequency band by using different spread spectrum codes. Because perfect separation between codes is not achievable under real wireless channels, the capacity and the maximal number of users are limited by interferences. Resource allocation such as rate adaptation and power control is an important means to combat the interferences, increase the number of users, and maintain the received signals' qualities. In joint source channel coding, rate adaptation by modifying the source rates, channel coding rates, and transmit rates can adjust the source encoder's output quality and the protections for channel errors. Consequently, the reconstructed signals' qualities can be carefully controlled, according to channel conditions. Power control is a technique to maintain the received signal-to-interference-noise-ratio (SINR). So the problem is how to increase the system performance, by cleverly allocating these resources under some practical constraints.

Distortion based or downlink CDMA resource allocation is a hot topic in research literature. In [2], the authors developed a source encoding assisted multiple access protocol to selectively drop source packets and increase the system capacity during congestions.

In [3], the authors formulated the resource allocation problems for different Quality of Service (QoS) requirements. In [4], the authors presented a video transmission scheme over multi-access networks. In [5], the author tries to minimize the overall system power for uplink multi-cell CDMA systems. In [6, 7], the authors maximized the total system utility by dynamic pricing and cooperation between the mobiles and the base stations. In [8], the authors formulated the problem as a constrained optimization problem and used approximations to have a simple solution form. The existing works can be classified into two kinds. The first kind of methods uses convex or linear approximations and then applies Lagrangian methods or convex optimization methods. The problem here is that there exists a mismatch between real situations and the approximations. The second kind of methods applies nonlinear optimization programming. However the complexity grows fast as the number of users increases. Therefore, our goal is to find a fast algorithm with relatively good performance.

In this paper, motivated by everyday life, we develop a heuristic and fast resource allocation algorithm for distortion management in downlink single cell CDMA systems. We want to reduce the overall system distortion, under the constraint of maximal transmitted power from the base station and the maximal distortion for each user. If the network is lightly loaded, we will assign the minimal distortion to everybody. Otherwise, even with the maximal transmitted power, the minimal distortion cannot be achieved by everybody. Under this condition, we assign the maximal distortion to each user first. If there is transmitted power left, we will assign some extra power to the user who can be satisfied and reduce its distortion most easily. To deserve an assignment that reduces its distortion, a user must have a small rate (high distortion), have a good channel condition, or generate small interferences to others. Then we judge if the transmitted power is used up. If not, we will continue the previous step. The idea is similar to a pizza party with limited pizzas. We will let everybody eat the minimal quantity of pizzas. Then we will assign the pizzas left to kids first, then to old people and ladies, finally to young gentlemen. Here the power is similar to pizzas and different users have different appetites for powers. Because of the similarity, we call the proposed algorithm "pizza party".

The organization of this paper is as follows: In Section II, we give the system model and description about our cross-layer protocol. In Section III, we formulate the problem and develop the pizza party algorithm for the downlink distortion management. In Section IV, we present simulations studies. In Section V, we have conclusions.

2 System Model and Protocol Description

Consider N users in a single cell CDMA system. In downlink, each mobile user is subject to interferences from other users in the cell. We assume the system is synchronous and each user is assigned a unique pseudo-random code within each cell. Because of the multipath environment [10], the orthogonality may not be guaranteed. The SINR of mobile i is give by:

$$\Gamma_i = \frac{W}{R_i}\frac{P_i G_i}{G_i \sum_{\substack{k=1 \\ k\neq i}}^{N} \alpha_{ki} P_i + \sigma^2} \tag{1}$$

where W is the total bandwidth which is fixed, R_i is the i^{th} user's transmit rate, P_i is the transmitted power from the base station for mobile i, α_{ki} is orthogonality factor between

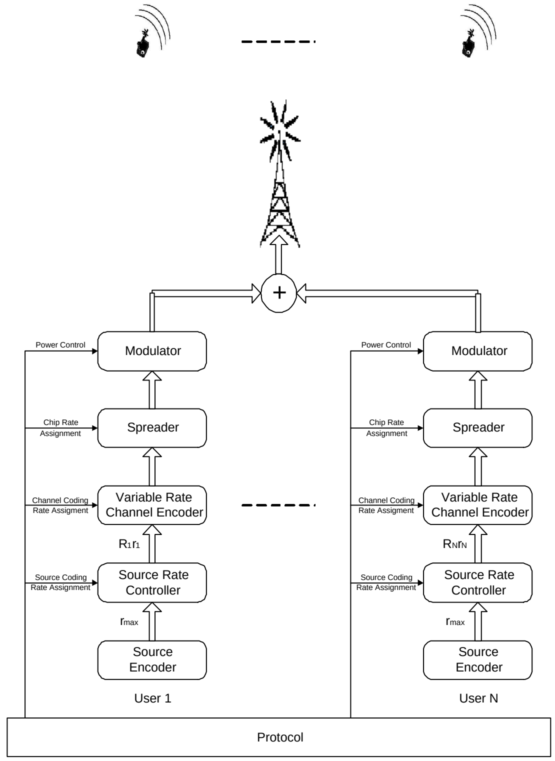

Figure 1: Block Diagram for the Proposed Protocol

mobile k and mobile i, G_i is the corresponding path loss, and mobiles' thermal noise levels are assumed to be the same as σ^2. W/R_i is the processing gain.

Fig. 1 shows the block diagram of our proposed cross layer protocol to manage the interferences by controlling the users' source rates, channel coding rates, transmit rates, and transmitted powers. The protocol is located in the base station and allocates rates and powers to all the users based on the channel conditions, in such a way that the received SINR at each user remains above a threshold. This threshold is determined by considering that the distortion due to channel-induced errors should be within a range of acceptable small values. In doing so, the protocol also considers the effects on the reconstructed signal qualities and takes into consideration the subjectively more annoying nature of channel-induced errors.

In our system, the real time source encoder has the key property that the output rate can be externally controlled. This can be implemented using either variable rate or embedded encoders. In the first case, the coder generates one bit stream for each of the possible encoding rates. Only one of these will be selected and transmitted based on the rate assignment. Using embedded encoders presents the advantage that only one bit stream is generated, making the adaptation to the rate assignment simply by dropping as many bits as necessary from the end of the bit stream. Although the "bit dropping mechanism" is exclusive to the embedded stream, we will loosely use this term to represent a reduction in the source rate, regardless of the particular source encoder implementation. We assume the source coder have the maximal output rate r_{max} bits/s and the source rate controller have the output rate r_iR_i bits/s, where r_i is the variable channel coding rate and R_i is the CDMA transmit rate. Then the data are encoded by channel coding with rate r_i. The processing gain for the CDMA spreader is W/R_i. We use BPSK modulation with power control in the modulator.

For simplicity, we assume that all the transmitted bits are equally important for error protection purposes. Because channel induced errors are more perceptibly annoying than the source encoding distortion, our design goal would be that channel induced errors would account for a small percentage of the overall end-to-end distortion. Our design will be constrained by the condition of meeting a target SINR so as to achieve the goal. Since a reduction in source encoding rate allows for an decrease in channel code rate or a decrease in transmit rate while maintaining the design goal, we note that the target SINR is a function of the source encoding rate, or equivalently, both channel coding and transmit rate. Therefore, it is possible to increase the overall end-to-end network distortion slightly and reduce the interferences greatly while meeting our design goal by clearly managing the users' source, channel coding, and transmit rates. Furthermore, we have found through simulations using different configurations of Rate Compatible Punctured Convolutional (RCPC) codes [9] that the SINR as a function of channel coding rate, when transmit rate is fixed, can be approximated accurately by

$$\gamma_i = 2^{Ar_i+B} \tag{2}$$

where γ_i is the required targeted SINR for the desired FER ($\Gamma_i \geq \gamma_i,\ \forall i$), A and B are the parameters of the error control coding scheme, and r_i is the channel coding rate.

Define $f_i(r_iR_i)$ as the distortion-rate function of the i^{th} user's source encoder transmitting at rate r_iR_i. In most well designed encoders, f_i is a convex and decreasing function. The minimum distortion occurs at maximum source rate r_{max}. Furthermore, we assume that the source encoder distortion-rate function [11], [1] is:

$$f_i = \delta 2^{2k(r_{max}-r_iR_i)} \tag{3}$$

where δ is the minimum distortion and k is a parameter depending on the encoders. This is a very general form that applies to the case of Gaussian source with squared-error distortion or when the high-rate approximation holds. In the case of realistic encoders, we find that (3) constitutes a good and tight upper bound on the real distortion-rate curve. Furthermore, the parameter k can be determined through simulations for any encoder, so that (3) can be a tight bound on the real distortion-rate operating curve. Define $D = 2^{2kr_{max}}/\delta$, the normalized distortion is given by:

$$D_i(r_i, R_i) = \frac{f_i}{\delta} = D2^{-2kr_iR_i}. \tag{4}$$

3 Real Time Distortion Management

3.1 Problem Formulation

In practice, the transmitted power from the base station is bounded, because there exists an implementation limitation and co-channel interferences should not be introduced too much to other cells. When the system is lightly loaded, each user could have the minimal distortion and the necessary total transmitted power could be still less than the maximal transmitted power available from the base station antenna. When the system becomes more loaded, even with the maximal transmitted power, the system cannot let every user have the minimal distortion. Under this condition, we need to have a graceful distortion control: Some users, with bad channel conditions and who introduce too many interferences to others, will sacrifice their performances slightly and increase their distortions in a controlled way. By doing these, the system will use the limited transmitted power to reduce interferences, optimize the overall system performance, and increase the total number of users. The problem is to decide who will sacrifice and how the users increase their distortions.

First, we assume the transmit rate R_i is fixed and we only modify the channel coding rate r_i. Later, we will show how to modify both R_i and r_i. We want to minimize the overall system distortion, under the constraint that each user's distortion is smaller than a maximal acceptable distortion and the overall transmitted power $P_{sum} = \sum_{i=1}^{N} P_i$ from the base station is bounded. The problem is formulated as:

$$\min_{r_i} \sum_{i=1}^{N} D_i \tag{5}$$

$$\text{subject to} \begin{cases} \text{Distortion Range:} & 1 \leq D_i \leq D_{max}, \forall\ i, \\ \text{Transmitted Power:} & P_{sum} \leq P_{max}, \end{cases}$$

where P_{max} is the maximal transmitted power and D_{max} is the maximal acceptable distortion. In this paper, for simplicity, we assume all users have the same D_{max}.

The problem in (5) is a nonlinear nonconvex problem and there might be many local minima. It is very difficult to solve it by Lagrangian method or nonlinear integer programming. Moreover, the computation complexity will grow quickly with the number of users increases. In order to fit the real CDMA system with large number of users, we need to develop a fast algorithm with relatively good performances.

3.2 Pizza Party Algorithm

The intuitive idea to develop a fast algorithm comes from an everyday life event. For example, in a pizza party with limited available pizzas, if the number of people is small, everybody will have enough food and there might be some pizzas left. However if the number of people is large and there is no way that everybody will be well satisfied, it is necessary to decide how to allocate the pizzas. One possible solution is to let everybody eat the minimal pizzas. (We assume there are enough pizzas for this requirement.) Then we will let kids eat one more slice of pizza, because they eat less and are easy to be happy. If there are any pizzas left, we will give one slice per time to the people who can be satisfied easily then. (Probably older people will get pizzas next, then ladies, and finally young males.) By allocating pizzas in such a way, we can use the limited pizzas to let the overall people's satisfaction high.

By using the same idea above, we can view the overall transmitted power as pizzas and the user's distortion as the index for hunger. In order to decide who is easy to be satisfied, we need to find the differential of the overall transmitted power with respect to each user's distortion. First we need to have a simple approximation for P_{sum}. Define

$$T_i = \frac{2^{Ar_i+B} R_i}{W} = \frac{P_i G_i}{G_i \sum_{k \neq i} \alpha_{ki} P_i + \sigma^2}. \tag{6}$$

If the processing gain is large, i.e., W/R_i is large, T_i is a small number. We know $\alpha_{ki} < 1$. So $\alpha_{ki} T_i$ is also a small number. We can have

$$P_{sum} = \mathbf{1}^T [\mathbf{I} - \mathbf{F}]^{-1} \mathbf{u} \approx \mathbf{1}^T [\mathbf{I} + \mathbf{F}] \mathbf{u} = \sum_{i=1}^{N} \frac{\sigma^2 T_i}{G_i} + \sum_{i=1}^{N} \sum_{j \neq i}^{N} \frac{\sigma^2 \alpha_{ji} T_i T_j}{G_j} \tag{7}$$

where $\mathbf{1} = [1 \dots 1]^T$, $\mathbf{u} = [u_1, \dots u_N]^T$ with $u_i = \sigma^2 T_i / G_i$, and

$$[\mathbf{F}]_{ji} = \begin{cases} 0 & \text{if } j = i, \\ \alpha_{ji} T_i & \text{if } j \neq i. \end{cases}$$

We find the gradient of the overall transmitted power with respect to each user's distortion. The gradient can be written as a function of the following three differentials:

$$g_i = \frac{\partial P_{sum}}{\partial D_i} = \frac{\partial P_{sum}}{\partial T_i} \frac{\partial T_i}{\partial r_i} / \frac{\partial D_i}{\partial r_i} \tag{8}$$

where

$$\frac{\partial P_{sum}}{\partial T_i} = \frac{\sigma^2}{G_i} + \sum_{j \neq i}^{N} \frac{\sigma^2 \alpha_{ji} T_j}{G_j}, \tag{9}$$

$$\frac{\partial T_i}{r_i} = \frac{A R_i 2^{A r_i + B} \ln 2}{W}, \tag{10}$$

$$\frac{\partial D_i}{\partial r_i} = -2kDR_i 2^{-2k r_i R_i} \ln 2. \tag{11}$$

So the final gradient can be written as:

$$g_i \;\; = \;\; C 2^{(A+2kR_i) r_i} \left(\frac{1}{G_i} + \sum_{j \neq i}^{N} \frac{\alpha_{ji} T_j}{G_j} \right) \tag{12}$$

where C is a negative constant. The absolute value of g_i is determined by the three factors: current rates (the term before the parentheses), channel gain (the first term inside the parentheses), and interferences to others (the second term inside the parentheses).

If P_{max} is large enough for every user in the cell to have the minimal distortion, we assign $D_i = 1$ to everybody and there is might be some overall transmitted power left.

If P_{max} is not large enough for every one to have the minimal distortion, we will initially assign $D_i = D_{max}$, $\forall i$. If the power is still not enough, it means that there are not enough "pizzas" to satisfy the group's minimal needs and we report an outage. If there is some power left, we will see who will be most easily to be satisfied by determining the gradient $\partial P_{sum} / \partial D_i$. If the absolute value of the gradient is small, that means this user is a "kid" who can eat little and become happy. For this user, from (12), the current rates is low (i.e. the distortion is high), channel gain is good, or interferences to others are small, consequently this user deserves a smaller distortion. In other words, this user can reduce its end-to-end distortion while creating the smallest strain on the available resources. So we assign a higher r_i to this user to let the distortion become small. Then we estimate the gradient and assign the rate again. We continue this process, according to the order of the gradients, until the transmitted power is used up. By doing this, we reduce the distortions by consuming the minimal resources step by step.

On the whole, the proposed adaptive algorithm is given in Table 1. As we have mentioned before, (5) is extremely difficult to solve by traditional methods in which the complexity grows fast with the number of user N increasing. In our proposed algorithm, the complexity lies in computing the gradients in (12) and calculating the overall transmitted power in (7). So the complexity is $O(N^2)$ and the proposed algorithm can be easily implemented in practice.

Table 1: Pizza Party Algorithm

1. Initialization: If everybody can get D_{min}, then allocate the powers and stop; else allocate D_{max} to everybody. If $P_{sum} > P_{max}$, report an outage.
2. Repeat: • Calculate $\lvert g_i \rvert$ • Increase the rate of the user with smallest $\lvert g_i \rvert$. • If $P_{sum} > P_{max}$, return the previous rate allocation and break.
3. Rate and Power Assignment.

3.3 Joint Consideration with Transmit Rate

In this subsection, we will consider the case where the transmit rate R_i can also be adapted. We will show that there is no need to adapt both the transmit rate and the channel coding rate, as long as $A \ln 2 \geq 1$. The new gradient is developed by adapting the transmit rate only.

If both R_i and r_i are adapted for resource allocation, we try to find out how to select R_i and r_i. We want to minimize the distortion, under the constraint that the demand for the transmitted power is fixed, i.e., $T_i = C'$, where C' is a constant. The problem is:

$$\min_{r_i, R_i} D 2^{-2kr_iR_i} \tag{13}$$

$$\text{subject to } T_i = R_i 2^{Ar_i+B}/W = C'.$$

Write the Lagrangian function as:

$$J = D2^{-2kr_iR_i} + \lambda(R_i 2^{Ar_i+B}/W - C'). \tag{14}$$

where λ is the Lagrangian multiplier. The solution is $r_i = \frac{1}{A \ln 2}$. Astonishingly, r_i is a fixed value and there is no need to adapt r_i, if $A \ln 2 \geq 1$. In our simulation setup, this condition is always held. Therefore we can adapt R_i only. The gradient of the overall transmitted power with respect to the distortion is now given by:

$$g_i = \frac{\partial P_{sum}}{\partial D_i} = \frac{\partial P_{sum}}{\partial T_i} \frac{\partial T_i}{\partial R_i} / \frac{\partial D_i}{\partial R_i} = C'' 2^{\frac{2kR_i}{A \ln 2}} \left(\frac{1}{G_i} + \sum_{j \neq i}^{N} \frac{\alpha_{ji} T_j}{G_j} \right). \tag{15}$$

where C'' is a negative constant. We can use the same algorithm in Table 1 by varying the transmit rate instead.

4 Simulation Results

We focus our study on real time voice communications. We use eighteen sequences, both male and female speakers, from the NIST speech corpus [12]. We encoded these sequences using the GSM AMR (Advance Multi-Rate) Narrow-band Speech Encoder [13]. This encoder operates with 20 ms frames, 5 ms look-ahead and includes an error concealment mode. Of the eight possible encoding rates: 12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15 and 4.75 kbps, we used only the six highest ones.

To determine the end-to-end distortion, we choose a perceptually weighted log-spectral distortion measure calculated by numerical approximation of the function

$$SD(\hat{A}(f), A(f)) = \sqrt{\int |W_B(f)|^2 \left| 10 \log \frac{|\hat{A}(f)|^2}{|A(f)|^2} \right|^2 df} \tag{16}$$

where $A(f)$ and $\hat{A}(f)$ are the FFT-approximated spectra of the original and the reconstructed speech frames, and $W_B(f)$ is the subjective sensitivity weighting function [14]:

$$W_B(f) = \frac{1}{25 + 75(1 + 1.4(f/1000)^2)^{0.69}}. \tag{17}$$

This distortion is measured on a frame-by-frame basis and then averaged over all frames, including outliers to further capture the effects of channel errors. We choose this measurement not only because of its good mathematical properties, but also because of its good correspondences to subjective measure [15]. We report a normalized distortion measure, which is computed as the ratio of the spectral distortions to that of the speech sequence encoded at the highest rate (12.2kbps) without channel noise.

Also, for the proposed system, we assumed BPSK modulation. For RCPC channel coder, we choose a memory 4, puncturing period 8, mother code rate 1/4 (variable rate in our system) RCPC code in [9] decoded with a soft Viterbi decoder. The total bandwidth W is 1.5616MHz. We assumed a channel affected by normalized Rayleigh fading (average power loss equal to 1), and normalized path loss (with propagation constants assumed equal to 1) with a path loss exponent equal to 3. The cell radius is 500m. α_{ki} is assume to be the same for all the users and is set to 0.9. Background noise level was assumed equal to 10^{-6}. $k = 3.3 \cdot 10^{-5}$. $r_{max} = 12.2$kbps.

One important point worth of noticing is that the constraint on the channel induced errors not only is necessary for (5) but also is advantageous, because it assures that the increase in distortion is smooth, controllable, and predictable. This is because the dominant process is the reduction in source encoding rate, thus the system behavior follows the rate-distortion curve. Channel induced distortion is kept at a sufficiently small value by appropriately setting the rates and powers. In contrast, this is not the case for the traditional CDMA approach where the increase in distortion is a consequence of the uncontrolled increase in channel-induced errors. In this case, the system behavior is much less predictable, because the random process of errors in the channel will dominate, and distortion is more subjectively annoying.

In Fig. 2 (a), we show the distortion as a function of SINR for six possible operating modes, where each mode is characterized by the pair (source encoding rate, channel code rate). Without adaptive source coding, each user's distortion has to follow a specific curve. With adaptive source coding, each user can follow the minimum of different curves, so that the distortion can be greatly reduced. Fig. 2 (b) shows an example of the simulations we conducted to find an approximation for the target SINR-channel coding rate function. The figure shows the target SINR, in $\log_2$ scale, as a function of the source encoding rate, where the channel induced errors are less than 3% of that of the corresponding source encoding distortion (channel induced distortion contributes 3 % to the end-to-end distortion). The figure confirms that (2) is a good approximation.

First, we fix the transmit rate at 24.4kbps and processing gain at 64. In Fig. 3 (a), we show the normalized distortion vs. the number of calls with different transmitted powers for the proposed scheme. The figure also includes, for comparison purposes, results for an

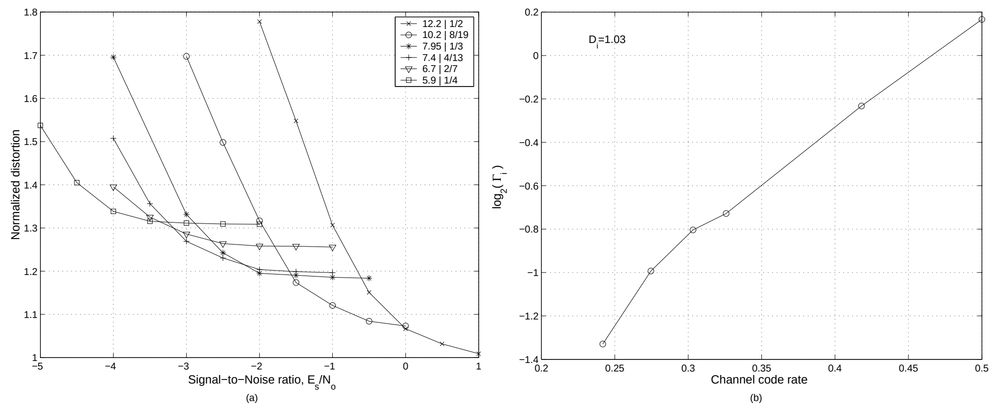

Figure 2: (a) Rate Distortion Curves (b) Required SINR vs. Rate

equivalent traditional CDMA system that shares the same configuration as our proposed scheme but operates without changing mode. For the case of this traditional system, all calls operates in the (12.2 Kbps, 1/2) mode. From these results, we can draw several conclusions. When the number of users is small, all the schemes with different powers works the same. This is because there is enough power for everybody to have the minimal distortion. When the user number is increased, our scheme can reduce the normalized distortion fundamentally, when compared to the traditional system. This is because our scheme controls the distortion smoothly by adapting the source and channel coding rates. In particular, if, for example $P_{max} = 350$, we can see that our system can support 30 users with 6 % less distortion, 40 with 12 % and 50 users with 37 % less distortion. When the transmitted power is increased, the distortion will be reduced. In Fig. 3 (b), we compared the normalized distortion as a function of the maximal available power for a fixed number of users in the system ($N = 30$, $N = 40$, and $N = 50$) that represents different network loading conditions. It shows our system can deliver the same level of average end-to-end distortion by a much lower maximum transmitted power. We also show the case where we modify the transmit rate only and fix the channel coding rate as $\frac{2}{7}$. In this case, we have a slightly performance loss when P_{max} is small.

5 Conclusions

In this paper, we develop a protocol to smoothly control each user's distortion by varying the source coding rate, channel coding rate, transmit rate, and transmitted power. We develop a fast algorithm to reduce the system overall distortion under the maximal transmitted power and maximal user's distortion constraints, according to different users' current rates, channel conditions, and interferences to others. Compared with the traditional scheme, our scheme can greatly reduce the distortion and the required transmitted power, which, in turn, will increase the maximal number of admissible users.

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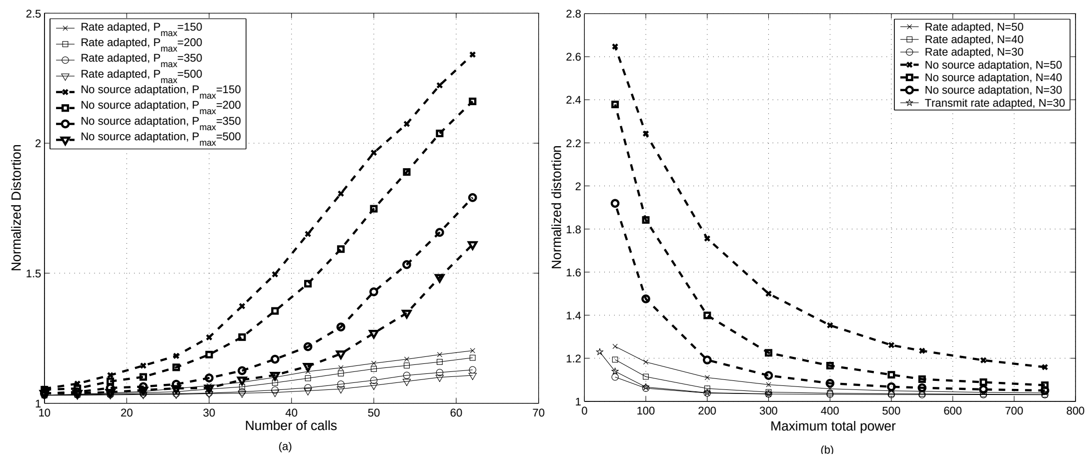

Figure 3: (a) Norm. Distortion vs. No. of Calls (b) Norm. Distortion vs. P_{max}

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