SENSOR-EFFICIENT SPATIAL PROCESSING OF MULTIPLE CO-CHANNEL DIGITAL SIGNALS

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ABSTRACT

An adaptive space-time processing system for dynamic spatial channels in a CDMA mobile communications network is presented. The proposed system consists of an adaptive sensor array that estimates the desired directions and waveforms, followed by adaptive linear equalizers that refine the waveform estimates by compensating for the mobile radio uplink channel. Capacity gains are achieved through code reuse, which is made possible by performing the spatial processing *after* the codes have been decorrelated with one another. Simulation results for a two sensor array in a variety of interference scenarios are presented in the form of bit error rate curves.

1. INTRODUCTION

The number of users in mobile communications systems has increased dramatically in the last few years. Since the frequency spectrum available for such systems is limited, future increases in the number of users will significantly burden system capacity. One promising alternative to increasing the number of base stations is spatial processing with sensor arrays at existing base stations [1, 2, 3]. Spatial processing offers the capability of forming beams in the direction of a group of users in receive and possibly transmit mode, thus reducing the level of co-channel interference and potentially increasing system capacity.

In typical mobile communications scenarios the number of users will almost certainly be larger than the number of elements in a sensor array. Moreover, the received signal from each user often consists of a number of coherent multipath reflections. The large number of wavefronts propagating towards the array makes determining the optimal beamformer weights for a particular user a difficult task. The approach taken here is to simplify the problem somewhat by employing code reuse within the system. Since the base station assigns the codes, the number of users with a specific code would presumably be known. Furthermore, by beamforming after the received signals have been correlated, the problem is simplified to discriminating spatially among users with the same code. SNR gains relative to the uncorrelated interference are also realized.

The angle of arrival (AOA) and signal waveform estimation technique used is a so called "sensor-efficient" algorithm. The term sensor-efficient was originally used in wideband array processing [4] to describe algorithms that were able to localize a number of sources equal to or greater than the number of elements in the array. Recent theoretical results [5, 6] have shown that it is possible to estimate AOAs in a sensor-efficient manner for narrowband signals as well. A sensor-efficient algorithm for narrowband signals was developed in [7] for the case where the number of sources is equal to the number of sensors. The same procedure is used here for two sources and two sensors because of its relative simplicity and to demonstrate that large arrays are not a necessity for achieving capacity increases via spatial processing.

2. UPLINK CHANNEL MODEL

Typically, the *downlink* propagation model for a mobile communications system assumes that multipath components of the signal of interest are uniformly distributed around the mobile with respect to AOA [8]. This propagation model is justified because the mobile is frequently surrounded by local scatterers. However, in an urban macrocellular environment, the base station is usually elevated above surrounding objects and, hence, relatively free of local scatterers. Therefore, plane waves at the base station tend to arrive from one general direction. For this reason, the *uplink* propagation channel can be modeled as an omnidirectional signal generated at the mobile and reflected by a ring of scatterers that surround the mobile [2, 8]. The scatterers are assumed to be uniformly distributed around the mobile, and the radius of the ring is assumed to be much less than the distance from the base station to the mobile. As the geometry of Figure 1 indicates, the received signal at the base station is a sum of the multipath reflections that occupy a sector of width $d\psi$.

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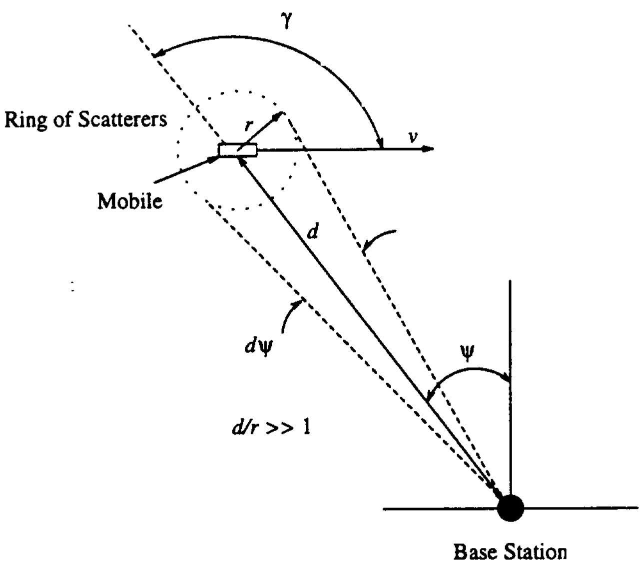

Figure 1: The geometry at the base station.

The number of scatterers is assumed to be large enough that the central limit theorem applies. The received signal will therefore have a Rayleigh distributed envelope and a uniform phase distribution between $-\pi$ and π. A power spectral analysis was conducted in [8] for the signal received at the base station and a geometry similar to that depicted in Figure 1. The result was the following approximate expression for the power spectral density of the base station received signal:

$$S(f) \simeq 3K\left[\left(\frac{r}{d}\right)^2 (f_m^2 - f_\gamma^2) - (f - f_\gamma)^2\right]^{-\frac{1}{2}}. \quad (1)$$

Where K is a constant that depends on the power of the transmitter, distance, and antenna gain, r is the radius of the ring of scatterers, d is the distance from the base station to the mobile, f_m is the Doppler shift, defined as velocity v divided by wavelength λ, and $f_\gamma = f_m \cos\gamma$, where γ is the angle between the direction of vehicle travel and the bearing of the mobile to the base station.

For simulation purposes, the mobile radio channel can be modeled by a time-varying transversal filter whose complex coefficients are filtered white Gaussian noise sequences [9]. Figure 2 illustrates the procedure. For the uplink channel model, the fixed filter, $B(z)$, that acts upon the input white Gaussian noise, $u_i[k]$, is chosen to approximate the fading spectrum given by (1). This simulation technique is a useful one because it acknowledges the relationship between the assumed geometry, vehicle speed and the fading spectrum. Moreover, a variety of fading scenarios are easily simulated by simply changing the coefficients of $B(z)$. In the simulations conducted here, a 3rd-order bandpass filter was used with center frequency equal to f_γ and cutoff frequencies that corresponded to the maximum and minimum expected Doppler frequencies.

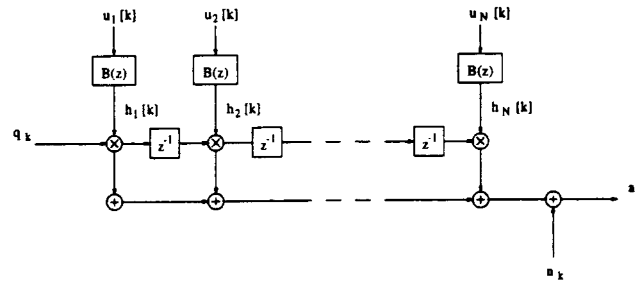

Figure 2: Baseband model for time-varying channel. Transmitted symbols q_k are filtered by a time varying transversal filter whose taps are filtered white Gaussian random sequences, $u_i[k]$. The fixed filter $B(z)$ is chosen to approximate the desired fading spectrum, resulting in a received sequence a_k. The received signal is also corrupted by additive noise n_k.

3. SENSOR-EFFICIENT AOA ESTIMATION

Sensor-efficient AOA estimation for constant modulus narrowband signals was addressed in [5, 7]. Observed results have shown that the technique in [7] works well for slowly varying modulus signals too. The algorithm is briefly summarized below for the case of two sources and two sensors. The array data is expressed in matrix notation as:

$$\mathbf{Y} = \mathbf{SA}. \quad (2)$$

Where $\mathbf{S}$ is the 2×2 matrix of steering vectors

$$\mathbf{S} = [\ \mathbf{s}_1 \ \mid \ \mathbf{s}_2\ ], \quad (3)$$

$\mathbf{A}$ is the $2 \times N$ matrix of signal amplitudes and inter-snapshot phases,

$$\mathbf{A} = \begin{bmatrix} A_1 e^{j\alpha_{1,1}} & A_1 e^{j\alpha_{1,2}} & \cdots & A_1 e^{j\alpha_{1,N}} \\ A_2 e^{j\alpha_{2,1}} & A_2 e^{j\alpha_{2,2}} & \cdots & A_2 e^{j\alpha_{2,N}} \end{bmatrix}. \quad (4)$$

Where N is the total number of snapshots to be processed. The algorithm begins by guessing AOAs for each of the sources. With this estimate of $\mathbf{S}$, the matrix of amplitude and phase estimates is simply,

$$\widehat{\mathbf{A}} = \mathbf{S}^{-1}\mathbf{Y}. \quad (5)$$

A new $\mathbf{S}$ matrix is then formed:

$$\widetilde{\mathbf{S}} = [\ \tilde{\mathbf{s}}_1 \ \mid \ \tilde{\mathbf{s}}_2\ ], \quad (6)$$

where $\tilde{\mathbf{s}}_i$ is an augmented steering vector,

$$\tilde{\mathbf{s}}_i = \left[1 \ \ \{e^{j\frac{1}{2}\widehat{\theta}_i}\} \ \ e^{j\widehat{\theta}_i}\right]^T. \quad (7)$$

The elements enclosed in brackets, {}, reflect *virtual* sensor positions and $\widehat{\theta}_i = -2\pi(d/\lambda)\sin\widehat{\psi}_i$ where d is the spacing between sensors, λ is the wavelength of the signals, and $\widehat{\psi}_i$ denotes the assumed AOA for the ith signal. The data is then interpolated by multiplying $\widetilde{\mathbf{S}}$ by $\widehat{\mathbf{A}}$:

$$\widehat{\mathbf{Y}} = \widetilde{\mathbf{S}}\widehat{\mathbf{A}}. \tag{8}$$

Note that the spacing between sensors has been cut in half by the addition of the virtual sensor, but the total array aperture has not been affected. This new spacing between array elements may introduce subtle effects on the beampattern, but does not introduce serious problems in terms of AOA estimation.

Root-MUSIC is then used on the interpolated array data to update the direction estimates and form a new $\mathbf{S}$ matrix. Then new amplitude and phase estimates are found based on the new $\mathbf{S}$. The procedure is repeated until direction, amplitude and phase estimates converge or until some predetermined maximum number of iterations has been reached. The number of iterations necessary for convergence depends on the SNR of the sources and the accuracy of the initial guesses. At startup, therefore, as many as 10-15 iterations may be necessary to obtain accurate AOA and signal waveform estimates. Over time, however, sources may be tracked easily by using AOA estimates from the previous block of snapshots as initial guesses for the next block to be processed.

Note that the rows of $\mathbf{S}^{-1}$ represent the linearly constrained beamformer weights for each of the two cases where the constraints are a unity response for one of the source directions and a zero or null response for the other. The use of linear constraints in a multipath situation is in general not a good idea since signal cancellation may occur. However, since the assumption here is that the incoming signal is confined to a relatively narrow sector (a sector much smaller than the resolution of the array, in fact), the use of linear constraints is acceptable. In a more severe multipath situation a beamformer that is able to cope with the strong coherent reflections without canceling the signal of interest would be necessary. Such a beamformer was suggested in [10].

4. SIGNAL WAVEFORM REFINEMENT

The waveform estimates given by the sensor-efficient algorithm represent optimal estimates in the sense that they have minimum variance with respect to the beamformer output. However, temporal distortions caused by the propagation channel must still be corrected. For the simulations conducted here, adaptive linear equalization was used with the well known LMS algorithm. Adaptive linear equalizers were chosen for their simplicity although maximum likelihood sequence estimation (MLSE) is known to be the optimum method for decoding the entire sequence of information bits [11]. A block diagram of the proposed adaptive space-time processor is shown in Figure 3.

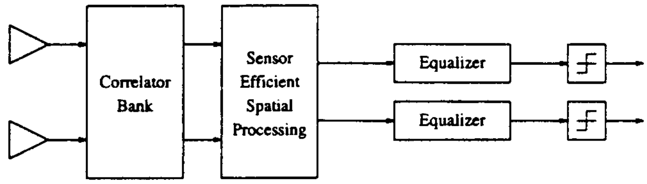

Figure 3: Block diagram of the proposed system.

5. PERFORMANCE EVALUATION

The complete algorithm as depicted in Figure 3 was implemented for two signals of interest with the following geometry. The two desired sources were modeled as a ring of 10 scatterers with an approximate radius of 73 meters. Their initial distance from the base station was 2800 meters. These dimensions yield a $d\psi$ of about 3°. For simplicity both sources were assumed to be traveling at a constant speed of 20 meters/second (about 40 mph). Furthermore, both mobiles were assumed to be traveling in a direction perpendicular to their initial bearing with respect to the base station. Source one started at an angle of 25°, while source two started at an angle of −15°. Ten seconds worth of data were simulated, which, for the geometry described above, means that both sources will have changed AOAs by about 4°. The ability of the adaptive array to track these sources over time is therefore a factor in the performance of the system.

BPSK modulation was used for each source at a bit rate of 4800 bits/second. The data was spread with Walsh codes of length 32 and the two desired sources shared the same code. The data was processed in blocks (snapshots) of 75. A flat fading model was assumed for the propagation environment necessitating the choice of a simple one tap adaptive equalizer or automatic gain control (AGC) for each signal at the output of the array. Training sequences of length 75 (the length of a block in the algorithm) were used once every 500 ms. to ensure proper tracking of the dynamic channel.

Interference sources with codes different from the desired source's code were generated with somewhat random geometries. The angular spread ($d\psi$) of each of the interferers was uniformly distributed between 1° and 3°. Distance from the base station was uniformly distributed between 1800 and 3800 meters. Vehicle speeds were uniformly distributed between 15 and 35 meters/second. Bearings with respect to the base station were uniformly distributed between −90° and 90°. Finally, a background noise level of −15 dB was present in all simulations.

Bit error rate (BER) curves were generated for a number of interferers ranging from 5 to 60 (SIRs from −14 dB to −35 dB respectively). Note that the system was assumed to employ code reuse with a reuse factor equal to two. Therefore, as many as 64 users may be supported with the length 32 Walsh codes. The results are summarized in Figure 4.

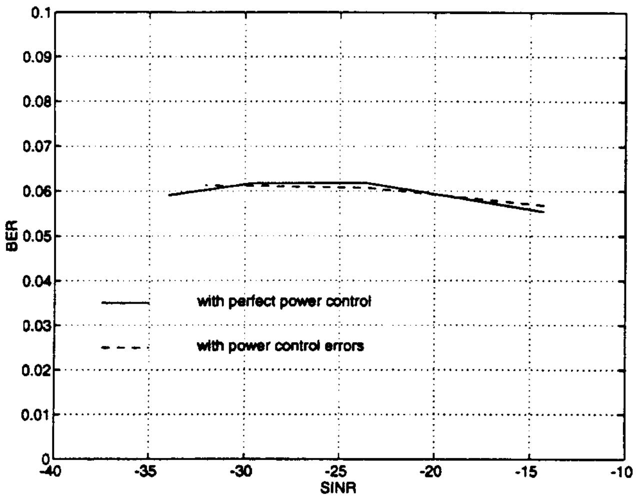

Figure 4: Bit error rates with perfect power control and with power control errors.

Ten independent simulations, each with ten seconds worth of data at a bit rate of 4800 bits/second yielded a total of 9.6e5 bits simulated for each point on the BER curve.

Although the BER is relatively high for mobile communications, there are several improvements that have not been used here but are typically present in such systems. For example, no error correction coding was used in the simulations. With error correction coding, the BERs should improve significantly. Also, the standard deviation of the direction estimates was approximately 12° for each of the simulations. Improving the variance of the direction estimates with Kalman tracking may improve the initial waveform estimates, resulting in an overall improvement in the BER. Finally, note that with a two sensor array and two desired sources, the array weights are completely specified by the constraints. The addition of just one more sensor would provide an extra degree of freedom that could be used to suppress unwanted interference.

Lastly, the idea that sensor arrays offer great potential in treating the near/far problem in CDMA systems was addressed in [3]. Currently, power control is necessary in CDMA systems to ensure that users closer to the base station are not received with significantly more power, thus blinding the receiver to users further from the base station. To demonstrate the capability of the system proposed here in the presence of power control errors, simulations were run where one of the interferers was assigned a power 10 dB higher than the other users. In a conventional CDMA system such a severe power control error would blind the receiver to other users and possibly incapacitate the entire cell. As the results in Figures 4 show, however, power control errors have little effect on the performance of the proposed system.

6. CONCLUSIONS

An algorithm whereby multiple users can share the same code in a CDMA mobile communications network has been proposed. Users with the same code are separated via spatial processing after they have been decorrelated from other users within the system. The waveform estimates from the adaptive beamformer are refined with adaptive equalizers that compensate for distortions introduced by the propagation uplink channel. In an effort to keep the simulations relatively simple, only a two sensor case with linear equalization was treated. However, by using more sophisticated equalization techniques such as MLSE with error correction coding and perhaps with the addition of another sensor, performance improvements may be realized. Investigation into these modifications is currently underway.

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