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A NOVEL INTERFERENCE REJECTION SCHEME FOR DS-CDMA USING ADAPTIVE NOISE CANCELLATION

S.S.H. Wijayasuriya¹, P.G. Turner, G.H. Norton and J.P. McGeehan
Centre for Communications Research
University of Bristol
Queens Building, University Walk
Bristol BS8 1TR, United Kingdom
Tel : +44 272 303727, Fax : +44 272 255265

Abstract

This paper proposes a novel solution to the interference cancellation problem in mobile Direct Sequence Code Division Multiple Access (DS-CDMA) systems. Conventional adaptive interference cancellation techniques rely on a training sequence to update the taps of an adaptive filter or Decision Feedback Equaliser (DFE). The strategy proposed in this paper replaces this conventional optimal filtering approach by one based on the principles of Adaptive Noise Cancellation (ANC). As opposed to the conventional optimal filtering approach, the ANC approach exploits the cyclo-stationary properties of the multiple access interference, in order to model the interference generation process. Silent periods (c.f. voice activity factor) can be exploited to derive the interference model. This removes the need for a training sequence and associated system overheads. The scheme employs a Conjugate Matched Filter (CMF) to generate an interference reference input to the adaptive noise canceller. A preliminary investigation of the performance of the proposed scheme is undertaken. The ANC scheme is shown to have a significantly better performance than the conventional receiver and the DFE in the multiple access interference limited environment. It is shown that the scheme can be extended to provide blind interference cancellation in the case of acute near-far conditions.

1 Introduction

The multi-user interference in a mobile Direct Sequence Multiple Access (DS-CDMA) system is cyclo-stationary. It is imperative that this highly structured nature of the interference be exploited to model the interference generation process to a high degree of accuracy. If the interference generator is successfully characterised, it can be used to regenerate the interference which can then be cancelled from the original signal. Existing adaptive interference cancellation techniques utilise a training se-

quence to update the taps of an adaptive filter or Decision Feedback Equaliser (DFE) [1, 2], and do not exploit the cyclo-stationary properties of the interference to the full. The proposed scheme is based on the well known approach of Adaptive Noise Cancellation (ANC) which is introduced below.

Since the publication of the classic paper on the subject by Widrow *et al* [3], the principles of adaptive noise cancellation have found application in a variety of fields ranging from communications engineering to medical electronics. A block diagram for the basic ANC scheme is shown in figure 1.

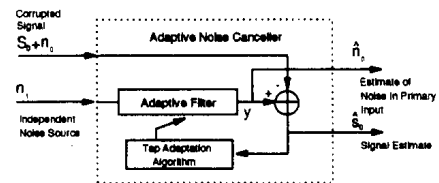


Figure 1: Adaptive Noise Cancellation

A secondary input (also referred to as the reference input) is derived from a point in the noise field where the signal is sufficiently weak so as to obtain an estimate of the noise in the primary input. In circumstances where ANC can be applied, levels of interference rejection which are difficult or impossible to achieve using direct filtering methods are often attainable [3]. Provided the desired signal is not present in any significant proportions in the secondary input, the adaptive filter will model the transfer function between the interference in the primary and secondary inputs, allowing cancellation of the interference in the primary. The filter coefficients are adjusted to minimise the Mean Squared Error (MSE) based on the error e , which is given by

$$e(k) = s_0(k) + n_0(k) - y(k)$$

¹Dr. Wijayasuriya is with MTN Networks, Sri Lanka.

$$= s_0(k) + n_0(k) - \sum_{i=1}^{M-1} w^*(i) \cdot n_1(k-i) \quad (1)$$

where $s_0(k) + n_0(k)$ is the k th primary input, $y(k)$ is the filter output, $n_1(k)$ the reference signal input and $w(i)$, the i th filter tap of the M tap transversal filter. The MSE is then given by

$$E[e(k)^2] = E[s_0(k)^2] + E[n_0(k) - y(k)]^2 \quad (2)$$

where E is the expectation operator. Under the ideal condition where the signal is uncorrelated with the noise in either input the first term on the RHS of equation (2) is not affected by the filtering process, and minimising the MSE is equivalent to minimising the second term on the RHS. From equation (2) it is clear that minimising the output power of the filter is equivalent to obtaining a Least Squares (LS) estimate of the signal $s_0(k)$. The MSE is a quadratic function of the weight vector, and an iterative search technique can be used to find the minimum of the hyper-paraboloidal error surface. The Least Mean Square (LMS) or Recursive Least Squares (RLS) algorithms can be used to achieve this end.

2 Application of ANC to DS-CDMA

The application of the principles of ANC to the interference cancellation problem in DS-CDMA is motivated by the cyclo-stationary nature of the interference. At first sight there seems to be very little similarity between the ANC case and the that of the single user receiver in DS-CDMA. The noise is replaced by multi-user interference. However, the most significant difference between the two schemes is that a secondary interference source cannot be derived by using spatial displacement. The Signal to Interference Ratio (SIR) in two spatially displaced receivers will be very similar, giving rise to almost complete cancellation of the signal.

It is proposed that the problem be translated from the spatial (three dimensional) domain to the N_c dimensional domain spanned by the code set being used (where N_c is the length of the codes). The ideas used in the spatial domain are easily translated on a qualitative basis. The vector representing the desired user's code in N_c dimensional space can be interpreted as a *look direction* along which the energy level of the signal is at a maximum. This, in fact, describes the function of the Conventional Linear Correlation Receiver (CLCR). Any other direction will suffer attenuation of the signal to varying extents. In the limit, a direction orthogonal to the *look direction* will not possess any signal components.

The analogy to the spatial case is now evident, and the problem can be rephrased in the code space as one of finding a *look direction* along which the desired signal is attenuated to a much greater extent relative to the

interfering signals. An interference estimate can then be obtained by matched filtering the received signal using a filter matched to the chosen reference direction. This matched filter will be referred to as the *Conjugate Matched Filter (CMF)* from hereon. The CMF incorporated into the proposed ANC receiver is shown in figure 2.

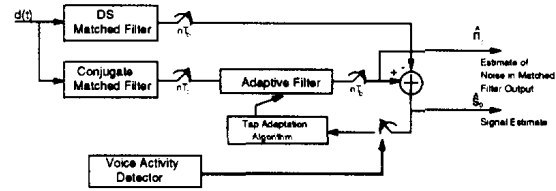


Figure 2: ANC using Conjugate Matched Filtering

A simplistic solution to waveform choice for the CMF involves using a code which is not used by any of the users in the system. The CMF is then quasi orthogonal to roughly the same extent, to all other codes. It follows that the *look direction* chosen for the reference input, does possess interference contributions from the interfering transmissions as required. This strategy however, will result in signal leakage being significant especially for a small number of interferers since signal leakage will be of the same order of magnitude as the interference components on the secondary (reference) input. The restriction to the binary alphabet for the CMF tap values is unnecessary and an alternative approach for the derivation of the CMF is proposed [4] in the next section. It is first instructive to consider the impact of signal leakage on the success of an ANC based strategy.

The requirement to minimise signal leakage can be formalised as follows. Denoting the primary and secondary input signals by X_1 and X_2 respectively, in the z domain

$$X_1(z) = M_1(z)R_s(z) + M_1(z)N(z) \quad (3)$$

and

$$X_2(z) = M_2(z)R_s(z) + M_2(z)N(z) \quad (4)$$

where $M_1(z)$ and $M_2(z)$ are the transfer functions of the matched filter and CMF respectively. $R_s(z)$ is the z transform of the desired signal component in the received signal and $N(z)$ is the z transform of the interfering signal components in the received signal. For ANC to function satisfactorily the following must be satisfied.

$$M_2(z)R_s(z) \approx 0 \quad \text{or} \quad M_2(z)R_s(z) \ll M_2(z)N(z) \quad (5)$$

3 Derivation of the CMF

The objective is clearly to derive a filter which suppresses the desired signal (thereby minimising signal

leakage), relative to the interfering signals. Such a filter can be derived empirically by adjusting the tap values until the output of the filter in response to an input consisting of random binary data modulated by the desired spreading code, is minimised. The output of the filter should be minimised over a span of chip intervals equal to the length of the adaptive filter. This objective, if achieved, will in addition help mitigate multipath effects which can also be a source of signal leakage. Since a CMF which attenuates all phase shifts of the desired users code is unlikely to be physically realisable, it would suffice to minimise over the first 5-10 chip offsets of the desired code. It follows that the adaptive filter must be of similar length. The latter predetermines that the RLS algorithm must be used, because of its superior convergence characteristics for filters possessing a small number of taps.

The minimisation is constrained by the condition that the filter tap values add up to some constant value (e.g. unity) thereby bounding the solution away from the all-zero case. The derivation of the constrained minimisation is based on the method of Lagrange Multipliers. The constraint on the optimum tap weights is written

$$\mathbf{w}_o^H \mathbf{s} = g \quad (6)$$

where g is a constant and \mathbf{s} represents the constraint on the optimum tap weight vector \mathbf{w}_o , e.g. the all one's vector $\mathbf{s} = [1, 1, \dots, 1]^T$. The mean square value of the filter output can be written

$$E[|y(n)|^2] = \sum_{k=0}^{M-1} \sum_{i=0}^{M-1} w_k^* w_i \Phi(i-k) \quad (7)$$

where $\Phi(\cdot)$ is the input auto-correlation matrix. A cost function incorporating the constraint can be written

$$J = \sum_{k=0}^{M-1} \sum_{i=0}^{M-1} w_k^* w_i \Phi(i-k) + \text{Re} \left[\lambda^* \left(\sum_{k=0}^{M-1} w_k s_k - g \right) \right] \quad (8)$$

where λ is a *forgetting factor* which attributes less emphasis to older input samples. The gradient of the cost function is written

$$\nabla_k(J) = 2 \sum_{i=0}^{M-1} w_i \Phi(i-k) + \lambda^* \mathbf{s} \quad (9)$$

Setting the gradient to zero at the minimum

$$\begin{aligned} \nabla_k(J) &= 0 \\ \Rightarrow \Phi \mathbf{w}_o &= -\frac{1}{2} \lambda^* \mathbf{s} \\ \Rightarrow \mathbf{w}_o &= -\frac{1}{2} \lambda^* \Psi \mathbf{s} \end{aligned} \quad (10)$$

where $\Psi = \Phi^{-1}$. Post-multiplying both sides by \mathbf{s} and taking the conjugate transpose

$$\mathbf{w}_o^H \mathbf{s} = -\frac{1}{2} \lambda \mathbf{s}^H \Psi \mathbf{s} \quad (11)$$

Substituting from equation (6),

$$\Rightarrow \lambda = - \left(\frac{2g}{\mathbf{s}^H \Psi \mathbf{s}} \right) \quad (12)$$

Substituting from equation (12) in equation (11) for the Lagrange multiplier,

$$\mathbf{w}_o = g^* \left(\frac{\Psi \mathbf{s}}{\mathbf{s}^H \Psi \mathbf{s}} \right) \quad (13)$$

Given that the steepest descent algorithm for the iterative update of the tap weights can be written [5]

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \frac{1}{2} \mu [\nabla(J(n))] \quad (14)$$

Equations (9) and (12) may be substituted for the gradient and the Lagrange multiplier respectively,

$$\begin{aligned} \nabla(J(n)) &= 2\Phi \mathbf{w}(n) - \left(\frac{2g}{\mathbf{s}^H \Psi \mathbf{s}} \right)^* \mathbf{s} \\ &= 2\mathbf{u}(n)\mathbf{u}^H(n)\mathbf{w}(n) - \left(\frac{2g^* \mathbf{s}}{\mathbf{s}^H (\mathbf{u}(n)\mathbf{u}^H(n))^{-1} \mathbf{s}} \right) \end{aligned} \quad (15)$$

But $y(n) = \mathbf{w}^H(n)\mathbf{u}(n)$, and substituting from equation (16) in (14) yields the tap update recursion

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \left[\left(\frac{g^* \mathbf{s}}{\mathbf{s}^H (\mathbf{u}(n)\mathbf{u}^H(n))^{-1} \mathbf{s}} \right) - y(n)\mathbf{u}(n) \right] \quad (16)$$

The recursion involves the repeated inversion of the matrix $\Phi(n) = \mathbf{u}(n)\mathbf{u}^H(n)$, where \mathbf{u} is the vector of input samples. The Sherman-Morrison Formula can be employed to perform this operation recursively.

4 Exploitation of Voice Activity

The sensitivity to signal leakage in the secondary input is a shortcoming of ANC solutions. Signal distortion, and a reduction in the maximum attainable noise cancellation arise as a result of signal leakage. In general, satisfactory performance can be obtained provided the SIR on the reference input is sufficiently low (typically less than -40 dB), the reverse situation (high SIR) should be true on the primary input. The adverse effects of signal leakage during training can be avoided by exploiting the voice activity factor in a DS-CDMA system. Many ANC schemes exploit *quiet periods* of the desired signal, to adapt the taps of the filter. Voice data systems are particularly suited to this strategy due the fact that voice activity statistics show that a single user transmission is active (on average) only 40% of the time. This can be exploited on the forward link (at which this solution is targeted) where the base station has full control of frame transmission. Frames can be interleaved so that (as far as possible) no two signals are silent during each training period (of the order of 50 symbols). The adaptation of filter coefficients is

initiated when the voice activity of the desired user is detected to be low. On the reverse link where voice activity cannot be exploited effectively the training sequence of conventional schemes can be replaced by a zero energy transmission with resulting savings in networking and synchronisation overhead. When convergence is achieved, the adaptation stops and the filter coefficients are fixed at their MSE values. The filters then act simply as a static interference regeneration and cancellation circuit. Since the signal is not present during training leakage effects are negligible.

5 Simulation Results

The ANC scheme is simulated in a mobile Rayleigh fading environment. This provides a suitable platform for an initial comparison of the receiver performance against the (CLCR), a standard decision feedback adaptive interference canceller (DFE) and a finite sequence length decorrelator - using the Sliding Window Algorithm (SLWA) [4, 6]. Figure 3 depicts the interference rejection achieved by the ANC receiver algorithm for the relatively severe multiple access interference case of 9 users using a spreading factor of 63. During the training or quiete period (the end of which is denoted by a vertical line on the trace), the upper plot shows the MSE between the interference in the primary input and that modeled by the adaptive filter. The lower plot of figure 3 shows the matched filter output for the same period of time and highlights the distortion due to MAI. Figure 3 demonstrates the interference rejection properties of the ANC receiver very well. After a training sequence or quite period of 50 symbols a significant level of rejection is achieved, the adaption is then ceased and the filter tracks the desired user's signal envelope with a comparatively low level of residual interference.

The BER performance of the ANC scheme compared to that of the SLWA, CLCR and DFE is shown in figure 4 for varying numbers of users at 25dB SNR. This figure shows that the ANC scheme significantly outperforms the DFE [4] and the CLCR receivers, with the increase in performance being most significant at higher loading. However, The near-far resistance of the SLWA maintains a relatively flat BER curve, and in this respect pays for the increased level of complexity in a multiple access limited scenario.

5.1 Blind Interference Cancellation

A blind, low complexity interference cancellation strategy is the ultimate objective of research into near-far resistant single user detectors for DS-CDMA. By removing the requirement for a silent period, the ANC scheme proposed can take the form of a completely blind scheme. The main difference from the voice activity

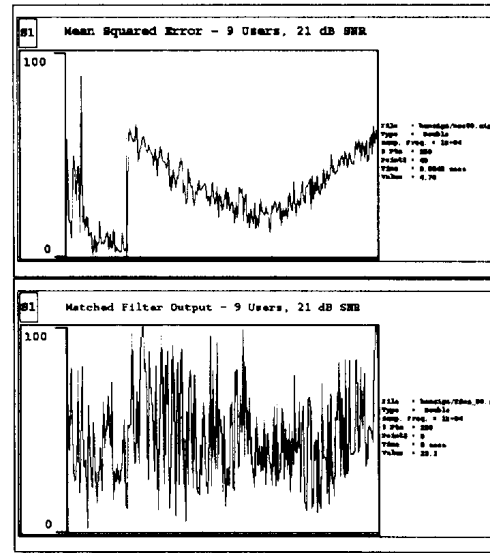


Figure 3: Interference Rejection Using ANC: 9 Users

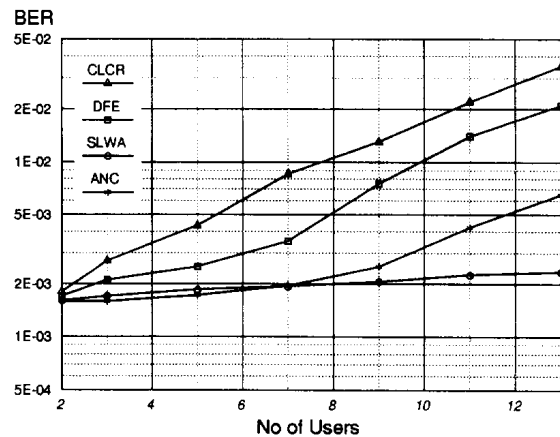


Figure 4: BER Comparison of Adaptive Techniques at 25dB SNR

based scheme is the presence of the desired signal during training, and the fact that training should be performed continuously or at regular intervals in order to track the solution. In figures 5 and 6 Blind Interference Cancellation (BIC) is shown to work in concept using a simple near-far situation. A two user scenario is considered with interfering user having average energy 40dB above that of the desired user (a Near-Far Ratio (NFR) of 40dB). The envelopes of the ANC-BIC, and matched filter outputs are shown in figures 5 and 6 respectively. The envelope of the desired signal is shown dashed. ANC-BIC is shown to recover the desired signal while the matched filter output is dominated by the interfering signal.

At this stage of the research, the success of the BIC scheme is limited by the non-ideal behaviour of the CMF, which gives rise to significant signal leakage in the reference input. Sufficiently high NFR's (low SIR's) are hence required in order to achieve significant improvements in signal recovery.

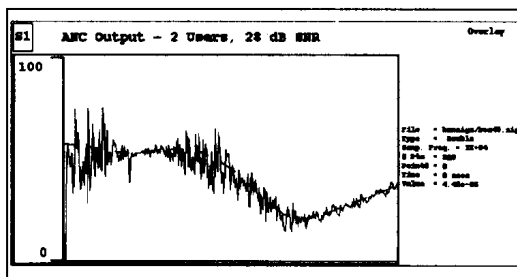


Figure 5: ANC-BIC Output: 40dB NFR

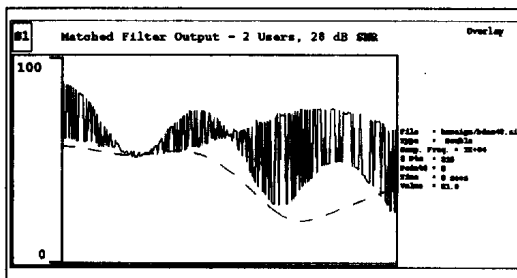


Figure 6: Matched Filter Output: 40dB NFR

6 Conclusions

A novel solution to the problem of interference cancellation in DS-CDMA systems based on the principle of adaptive noise cancellation has been proposed. The noise reference source, fundamental to the technique of ANC, is derived from a conjugate matched filter which utilises a signature waveform pseudo-orthogonal to all other users. The ANC scheme successfully rejects the

multi-user interference from the system by exploiting its cyclo-stationary properties. BER performance has been shown to be significantly better than the CLCR and DFE receivers in a multiple access limited Rayleigh fading environment. An investigation of hardware implementation issues for the CMF-ANC receiver is undertaken in [7]. From an implementation point of view the scheme can exploit the voice activity factor. Furthermore it was shown that the scheme has potential to be developed into a blind interference cancellation technique. It can be concluded that applying the ANC principles to the MAI rejection problem can yield a low complexity, near-far resistant receiver architecture for DS-CDMA systems.

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