BLIND RECOVERY OF MULTIPLE PACKETS IN AD HOC MOBILE NETWORKS USING POLYNOMIAL PHASE MODULATING SEQUENCES

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ABSTRACT

We consider multiple packet reception (MPR) for asynchronous random access wireless mobile ad hoc networks. An interference cancellation algorithm is proposed that exploits the base-band cyclostationarity properties of the signal, which are induced at the transmitters by means of modulating the symbols with distinct polynomial phase sequences. In contrast to the method presented in [1], the proposed technique does not require knowledge of the starting time of transmission of the desired signal and can be applied to dispersive multipath channels. Also a practical way of assigning the modulating sequences via the use of a common codebook known to all nodes is proposed, and the impact on local throughput of such scheme is analyzed.

1. INTRODUCTION

A key issue in random access networks is the decrease of throughput due to collisions arising from uncoordinated transmitters. One approach to tackle this problem is to give each node the capability of receiving multiple packets simultaneously. In this paper we propose a new algorithm to accomplish such a task. We use a modulation inducing cyclostationarity (MIC) approach similar to those used in [1-3]. But, in our case, the base-band data sequence is modulated by a polynomial phase sequence (PPS). This can be viewed as introducing a watermark in the desired signal [4] or as a color code [1]. It is worth pointing out that for real signals and one transmitting antenna, the transmitted signal model in [2] is similar to the one proposed here. There are, however, some important differences. First, the algorithm in [2] is a direct blind symbol estimation method while ours is a blind equalization and signal separation method. Second, while [2] uses random modulating sequences (no synthesis procedure is given), we use PPS. This provides the application engineer with a definite framework. Third, the method in [2] requires synchronization while ours derives the synchronization from the received signal. Previous work in the area of source separation applied to the MPR problem for ad hoc networks can be found in [1] and [5]. The work in [1] uses, like the method proposed here, modulation induced cyclostationarity; however, it requires knowledge of the starting time of the desired packet. The advantage of the method to be presented here over the method in [1] is the fact that we do not require this timing, so making the proposed approach fully asynchronous.

2. SYSTEM MODEL

The reception model that we use in this paper is illustrated in figure 1. The arriving packets coming from different users are totally asynchronous (both at symbol and packet levels). The receiver will process the packets in a window by window fashion. The width of each window will be set equal to N symbols. The transmitted packets are all assumed to have a fixed length, known to the receiver, equal to J symbols.

2.1. Modulation Using Polynomial Phase Sequences

A discrete-time base-band one-way digital communications link between a given pair of nodes i and j in the network is shown in figure 2. $b_i(k)$ is the information sequence (assumed to be real valued), $s_i(k)=b_i(k)c_i(k)$ is the transmitted signal and $c_i(k)=e^{j\left[2\pi f_i k^2+2\pi\alpha_i k\right]}$ is a polynomial phase modulating sequence [6], where f_i and α_i are design parameters. Note that although $b_i(k)$ is a real sequence, the transmitted signal $s_i(k)$ is complex valued. It is possible to show that, contrary to the approach in [7], the transmitted bandwidth is not increased because the modulation is applied at baseband before pulse shaping. Still in figure 2, $\mathbf{H}_{ij}$ stands for the channel matrix, $\mathbf{n}_j(k)$ is the noise vector, $\mathbf{x}_j(k)$ accounts for the noisy received signal vector, and the multiple-channel equalizer used to recover user i is represented by vector $\mathbf{w}_{ji}$. Each row of $\mathbf{H}_{ij}$ is formed by the impulse response of a sub-channel from transmitter i to receiver j. Each sub-channel is formed by using multiple antennas, oversampling or both. Then, if we assume that there are P_j nodes that transmit to node j, we have that

$$\mathbf{x}_j(k)=\sum_{i=1}^{P_j}\mathbf{H}_{ij}*s_i(k)+\mathbf{n}_j(k) \tag{1}$$

$$y_i(k)=\mathbf{w}_{ji}^{\mathrm{T}}\mathbf{x}_j(k) \tag{2}$$

where $\mathbf{H}_{ij}*s_i(k)$ represents the convolution of the signal $s_i(k)$ with each row (sub-channel) of matrix $\mathbf{H}_{ij}$. In what follows, we will drop the j index, as we will refer to a particular receiver.

3. RANDOM ACCESS MECHANISM

We want to achieve the separation of the $P_j = P$ transmitted signals given N snapshots of the received signal vector $\mathbf{x}(k), (k = 1, \cdots, N)$, and the knowledge of the values f_i, α_i. Notice that the method in [1] assumes the receiver knows the color code of the desired transmission. In our case this is equivalent to knowing the value of the pair f_i, α_i, for all colliding signals. For ad hoc networks this knowledge is unrealistic since there is no central control that could be in charge of the task of assigning color codes for different users. We propose that the color codes α_i be taken from a common codebook of finite length D. This way, each node has D possible virtual channels for transmission and D virtual channels for reception. So, without the knowledge of the f_i, α_i pair of values used by the transmitters in the actual collision, the receiver must search for possible transmissions in all the virtual channels of the codebook. The proposed method uses $f_i = f$ for all transmitters, with f a pre-specified value (known to all nodes) that depends on the network characteristics, as will be seen later. So, using a codebook containing the α_i values, the assumption of the knowledge of the color codes is dispensed at the expense of an increase in the complexity of the processing.

3.1. Throughput Analysis

The collision channel can be characterized by a reception matrix $\mathbf{E}$ whose elements $e_{i,j}$ represent the probability that j packets are correctly received given that i packets collide at the MPR node [8]. We consider that transmitters use one antenna and the receiver uses K antennas forming an antenna array. Then, for the approach proposed, we make the assumption that if $i > K$ packets coincide in the observation window at the MPR node, none of them will be correctly received. If the number of packets coinciding in the observation window is $i \le K$, the signal separation algorithm will be able to demodulate all those packets whose color codes α_i are not shared by others. We assume that packets select color codes independently and with equal probability from the codebook of size D. Therefore if $i > K$, $e_{i,j} = 0, \forall j$. On the other hand, if $i \le K$, $e_{i,j}$ will be equal to the probability of having exactly j unique codes out of i selected among D different codes, thus

$$e_{i,j}(D) = \frac{f_{i,j}(D)}{D^i}; \quad (i = 1,2,\ldots,K); \quad \left(j = 0,\ldots,i\right) \tag{3}$$

where $f_{i,j}(D)$ is the number of permutations with repetition of D codes taken i at a time in which exactly j codes are different. It can be shown that

$$f_{i,0}(D) = \sum_{m=0}^{i} (-1)^m C\binom{i}{m} P\binom{D}{m} (D-m)^{i-m} \quad ; \quad (i = 0,1,\ldots K) \tag{4}$$

$$f_{i,j}(D) = C\binom{D}{j} P\binom{i}{j} f_{i-j,0}(D-j) \quad ; \quad (i = 1,2,\ldots K) \quad ; \quad \left(j = 1,\ldots,i\right). \tag{5}$$

For the analysis of the throughput, we consider the simplified scenario of exponentially distributed packet inter-arrival times with arrival rate equal to λ. We assume that all the packets have the same length, JT seconds, where T is the symbol duration. Thus, the offered load will be $G = \lambda JT$. If the observation window equals V times the packet length (we consider here that $1 \le V \le 2$), it can be shown that the normalized throughput S is

$$S = \frac{e^{-(V+1)G}}{V+1} \sum_{i=1}^{K} \frac{\left((V+1)G\right)^i}{i!} \sum_{j=1}^{i} j\, e_{i,j}(D) \ . \tag{6}$$

Figures 3 and 4 show the normalized throughput S for different values of the parameters K, D and V. Figure 3 shows the impact of the observation window length for $K = 4$ and $D = 32$. Curves are shown for values of $V = [1, 1.5, 2]$. It is clear that the throughput is reduced when the observation window is increased. This happens because the larger the window, the more packets are included on average and the greater the probability that the MPR node will fail to decode the arriving packets. However, the shorter the observation window the more processing it is required. This is so because, in the absence of synchronization with the beginning of the packet, observation windows should overlap at least one packet length, such that a specific packet will be completely observed either in the actual, previous or next window. Figure 4 shows the impact of the codebook size on the normalized throughput S for $V = 2$, $K = 4$ and values of $D = [4, 8, 16, 32, 64, \infty]$. It is observed that the throughput increases as D increases, being bounded by the curve of the MPR with $D = \infty$, which is the case when every packet is assured to have a different color code. Note however that a finite value of D on the order of eight times the number of antennas K gives a reduction on the maximum throughput with respect to $D = \infty$ of only less than 8%. This means that the use of a codebook of codes as proposed here can be considered a good practical approach to the problem of code assignment among users. Note that due to the simplified assumptions used, the impact of the PPS algorithm on the preceding analysis of throughput was not taken into account. If the performance of the PPS algorithm were to be included in the analysis, the reception matrix would change somewhat. However, the conditions to be presented in the following section for the proper performance of the PPS algorithm do consider our proposal of a color code codebook.

4. INTERFERENCE CANCELLATION ANALYSIS

The transmitted data symbols $b_i(k)$ are real valued as stated in section 2. Let us suppose for the moment that both random access interference (RAI) and inter symbol interference (ISI) have been removed from user i. Thus the equalizer output in the noiseless case can be expressed by

$$y_i(k) = g_i(m) b_i(k-m) c_i(k-m) \tag{7}$$

where $g_i(m)$ is a real gain factor and m some delay. Now, if $y_i(k)$ is multiplied by the complex conjugate of the PPS sequence as shown in figure 2, the resulting signal $z_i(k)$ will be $z_i(k) = y_i(k) c_i^*(k-m) = g_i(m) b_i(k-m)$. This shows that if the equalizer output is given as in (7), $z_i(k)$ is real valued. We propose then to select the equalizer coefficients in such a way that the imaginary part of $z_i(k)$ be equal to zero, so that

$$y_{iI}(k)c_{iR}(k-m) = y_{iR}(k)c_{iI}(k-m) \quad . \tag{8}$$

Where I and R stand for the real and imaginary parts. This relation is the base for our proposed interference mitigation method. Because our method starts from (8) with the aim of achieving (7), it is very important to find necessary and sufficient conditions for (8) implying (7). These conditions are as follows. Once the lower bound on the maximum channel dispersion length M is found (this depends on the expected propagation conditions when deploying the system), the equalizer length in symbol periods, Q, can be selected. It can be shown that in order to achieve both RAI and ISI cancellation, $T_f = 1/f$ must be selected as the smallest prime number greater than $M+Q-2$. The size of the codebook D is then determined as not being a multiple of T_f. Finally the whole set of α_i 's are computed according to

$$\alpha_i = \frac{i-1}{2D} \quad (i = 1,\ldots,D) \ . \tag{9}$$

The derivation of these conditions is not given here due to lack of space, but it will be presented elsewhere.

5. PROPOSED ALGORITHM

This section focuses on the derivation of a closed form solution for the equalizer coefficients based on relation (8). Define

$$\begin{aligned}\mathbf{u}_i(k) = \Big[&-c_{iI}(k-m)\mathbf{x}_R^T(k) + c_{iR}(k-m)\mathbf{x}_i^T(k),\\ &\big(c_{iI}(k-m)\mathbf{x}_I^T(k) + c_{iR}(k-m)\mathbf{x}_R^T(k)\big)\Big]\end{aligned} \tag{10}$$

and

$$\mathbf{v}_i = \begin{bmatrix} \mathbf{w}_{iR} \\ \mathbf{w}_{iI} \end{bmatrix} \tag{11}$$

then from (8)

$$\mathbf{u}_i(k)\mathbf{v}_i = 0 \tag{12}$$

where $\mathbf{w}_{iR}$ and $\mathbf{w}_{iI}$ stand for the real and imaginary parts of vector $\mathbf{w}_i$ while $\mathbf{x}_R(k)$ and $\mathbf{x}_I(k)$ serve the same purpose but for the received data vector $\mathbf{x}(k)$. Let the number of received symbols be N (the full observation window), and define

$$\mathbf{U}_i = \left[\mathbf{u}_i^{\mathrm{T}}(1), \mathbf{u}_i^{\mathrm{T}}(2), \ldots, \mathbf{u}_i^{\mathrm{T}}(N)\right]^{\mathrm{T}} \tag{13}$$

then (12) is extended to form an homogeneous system as

$$\mathbf{U}_i\mathbf{v}_i = \mathbf{0} \tag{14}$$

where $\mathbf{0}$ is the $N \times 1$ zero vector. To account for noisy conditions and also to avoid the equalizer vector to converge to the trivial solution, we consider a constrained least squares problem. Therefore the objective is

$$\text{Minimize: } \|\mathbf{U}_i\mathbf{v}_i\|_2^2 = \mathbf{v}_i^{\mathrm{T}}\mathbf{R}_{\mathbf{U}_i}\mathbf{v}_i \quad ; \quad \mathbf{R}_{\mathbf{U}_i} = \mathbf{U}_i^{\mathrm{T}}\mathbf{U}_i \tag{15}$$

subject to $E\left\{y_{iR}^2(k)\right\} = 1$, where $y_{iR}(k) = \mathrm{Re}\left[y_i(k)\right]$. The solution of (15) under this constraint is given by the right generalized eigenvector associated to the smallest generalized eigenvalue of matrices $\mathbf{R}_{\mathbf{U}_i}$ and $\mathbf{R}_{\mathbf{x}}$, with $\mathbf{R}_{\mathbf{x}} = E\left\{\mathbf{x}\mathbf{x}^{\mathrm{T}}\right\}$, and

$$\begin{aligned}\mathbf{x} = \Big[&c_{iI}(k-m)\mathbf{x}_I^T(k) + c_{iR}(k-m)\mathbf{x}_R^T(k),\\ &\big(c_{iI}(k-m)\mathbf{x}_R^T(k) - c_{iR}(k-m)\mathbf{x}_I^T(k)\big)\Big]\end{aligned} \tag{16}$$

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6. SIMULATION RESULTS

Some computer simulations will now be carried out to assess the performance of the proposed method. The performance measure we have considered is the average signal to interference plus noise ratio (SINR) at the equalizer output as a function of the received SNR at node j for transmitter i that is defined from (1) as $\mathrm{SNR}_i \equiv E\left[\left\|\mathbf{H}_{ij} * s_i(k)\right\|^2\right] \Big/ E\left[\left\|\mathbf{n}_j(k)\right\|^2\right]$.

In this simulation we consider that the receiver node under test possesses a circular antenna array with 8 elements. The radius of the structure is equal to one wavelength of the transmitted carrier frequency $f_c = 900$ MHz. Six transmitters using BPSK data and modulated by polynomial phase sequences with $f = \frac{1}{11}$ and $\alpha_i = \frac{i-1}{2D}$, $(i = 1,\ldots,D;\, D = 6)$ are transmitted through a dispersive channel. The symbol period for all the sources is $T = 1\ \mu s$. Each transmitted signal arrives to the receiver via 4 different paths with different delays and angles of arrival. The receiver takes samples two times per symbol period ($L = 2$) and is equipped with a space-time equalizer with a temporal length of $Q = 4$ symbol periods (The total number of coefficients in the equalizer is $K \times L \times Q = 64$). The dispersion of the channel impulse response is about 3 to 4 symbol periods for all transmitters. Raised cosine pulse shaping filters with roll-off factors equal to 25% are used by all the transmitters to limit the bandwidth of the emitted signals. The packet size is $J = 200$ and the observation window width is $N = 400$. All the packets arrive asynchronously at the receiver with starting times [50, 120, 75, 240, 190 and 100 symbol periods] referenced to the beginning of the observation window. Transmitters 1,2 and 3 are strong transmitters being received with the same power. Transmitter 4 is a very weak transmitter being 30 dB weaker than 1,2 and 3. Transmitter 5 is just 10 dB stronger than 4 and transmitter 6 is 10 dB stronger that 5. As can be realized, this is a very hard situation for user 4. The multipath field is dispersive in time and space and the situation is clearly near-far. Figure 5 shows the simulation results together with the performance of the optimum MMSE solution as a benchmark. It is observed that, despite the strongly adverse conditions (the desired signal from user 4 is orders of magnitude weaker than the interference), our PPS algorithm is able to recover user 4 almost optimally as soon as its received signal to noise ratio reaches 0 dB. Similar curves are obtained for the other users. We have performed extensive simulations using widely different channel and interference conditions and they all show similar behavior to the one presented here.

7. CONCLUSIONS

An algorithm for the blind recovery of multiple packets in asynchronous random access wireless ad hoc networks has been proposed. The algorithm exploits the received signal structure imposed at the transmitters via modulation using polynomial phase sequences. This operation does not increase the transmitted signal bandwidth. Necessary and sufficient conditions on the parameters of the polynomial phase sequences for the removal of both RAI and ISI have been presented. Simulation results show that the algorithm gives useful results even under heavy near-far scenarios. A practical method for the selection of the PPS parameters has also been proposed. This is based on the use of a codebook known to all network nodes. The impact of the codebook size on throughput performance has also been analyzed. Due to the asynchronous nature of the traffic, we assumed block processing at the receiver using an observation window. The impact of the observation window length on throughput was also studied.

8. REFERENCES

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9. FIGURES

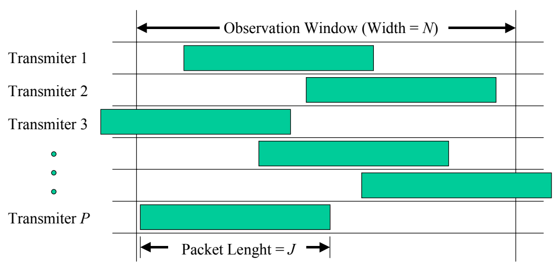

Figure 1: Packet reception model

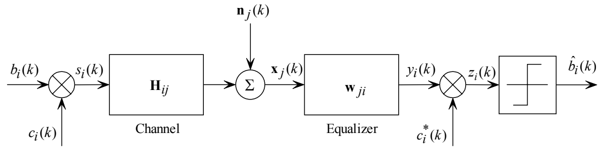

Figure 2: Discrete-time model for the communication between transmitter i and receiver j

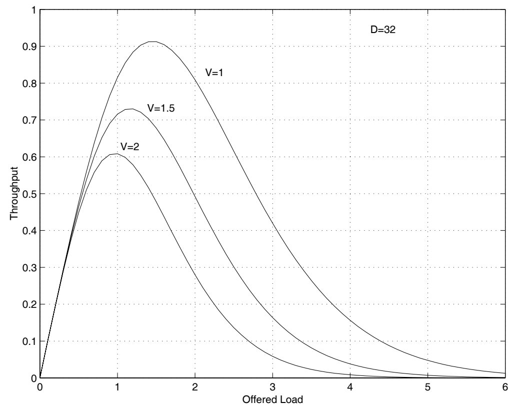

Figure 3: Impact on throughput of the observation window width $N = VJ$ ($K = 4$, $D = 32$)

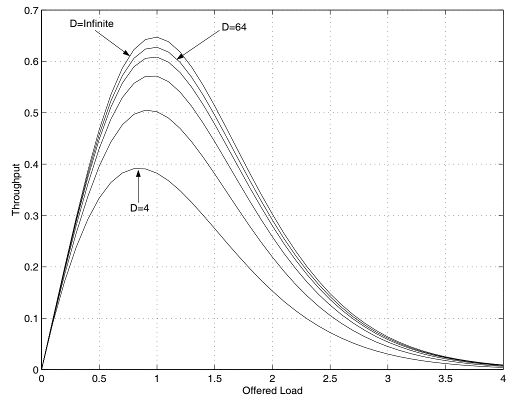

Figure 4: Impact on throughput of the codebook size D ($K = 4$, $V = 2$).

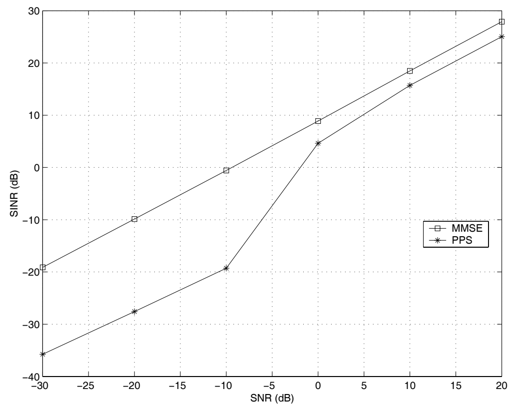

Figure 5: SINR performance for the weakest user