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Description	

A MIMO Turbo Equalizer for Frequency-Selective Channels With Unknown Interference

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***Abstract*—A new space-time turbo equalization algorithm is derived for frequency-selective multiple-input–multiple-output (MIMO) channels with unknown interference. The algorithm is an extension of our proposed MIMO equalization algorithm [12], which performs joint channel estimation, multiple user's signal detection, and decoding, all in an iterative manner. This paper's proposed algorithm uses estimates of the correlation matrix of composite unknown interference-plus-noise components to suppress the unknown interference while effectively separating multiple users' signals to be detected (referred to as "known user" later). The correlation matrix of the composite unknown interference-plus-noise components can be estimated by time averaging the instantaneous emperical correlation matrix over the training period. Since the iterative channel estimation yields better channel estimates as more iterations are performed, thereby the estimate of the correlation matrix of the unknown interference-plus-noise components also becomes more accurate. This results in better signal detection performances, even in the presence of unknown interferers. A series of computer simulations show that this paper's proposed algorithm can properly separate known users' signals while suppressing unknown interference.**

***Index Terms*—Multiple-input–multiple-output (MIMO) system, space-time signal processing, turbo equalization.**

I. INTRODUCTION

RECENTLY, multiple-input–multiple-output (MIMO) mobile communication systems have attracted great attention because of their vast channel capacity [1]. The logical structure of a system uplink has a MIMO radio-network topology in which, as shown in Fig. 1, each user uses a single antenna and the receiver uses multiple antennas to detect the simultaneous users' transmitted signals. Another MIMO communication scenario is a downlink in which both transmitter and receiver use multiple antennas and the transmitter transmits different information sequences via the multiple antennas. Obviously, the MIMO uplink aims to increase the cellular system's user capacity while the MIMO downlink aims to enhance radio channel's bit rate. This paper assumes MIMO uplink in which all users transmit their signals sharing the same time- and frequency-slots without spreading the signals in the frequency domain. Since broadband signal transmission renders the MIMO channel to be severely frequency selective, the receiver has to adequately detect the multiple users' signals in the presence of multiple access interference (MAI) as well as intersymbol interference (ISI).

Several types of detectors for MIMO channel signal transmission have been proposed. Obviously, signal detectors that perform the maximum likelihood sequence estimation (MLSE) for multiple users achieve optimal performance [2] and [3]. Their computational complexity, however, grows exponentially with the sum of the channel memory length of each simultaneous user. The BLAST and its family detectors[4] and [5] can reduce the computational complexity to a practical level by using adaptive linear filtering and successive signal-detection techniques. However, the BLAST detectors are not designed to fully exploit the path diversity in frequency-selective channels.

The discovery of turbo codes [6] and [7] has brought about the creation of iterative (turbo) signal-processing concepts in adaptive equalization and multiuser signal-detection techniques. In [8], turbo processing is effectively employed to enhance the performances of the BLAST system. [9] has derived a computationally efficient iterative multiuser detector for code-division multiple-access (CDMA) systems. The conceptual basis of [9]'s detector is the soft interference cancellation followed by minimum mean square-error filtering (SC/MMSE). New equalization algorithms for single-user frequency-selective channels have been derived based on SC/MMSE [10] and [11]. In [12], we applied the SC/MMSE concept to MIMO signal detection in frequency-selective fading channels and derived a new computationally efficient space–time (ST) turbo-equalization algorithm. Reference [12]'s proposed ST-turbo equalizer performs joint channel estimation, multiple user's signal detection, and decoding, all in an iterative manner. The iterative channel estimator achieves high accuracy in estimating channel parameters, even if only relatively short unique word sequences are transmitted from the multiple users as signal references. This is because in addition to the unique word sequences, the estimator can also use as signal references the reliable portions of multiple users' estimated information symbols fed back from each user's channel decoder. The channel parameters are re-estimated using the signal references iteration-by-iteration; the more iterations, the higher the channel-estimation accuracy. Despite the simple structure of [12]'s proposed detector, it can take full advantage of the path diversity inherent within the multipath channels without requiring prohibitively large computational effort.

This paper assumes the presence of unknown users' signals in MIMO channels. Since, in cellular environments, interference from users outside the reference cell degrades the receiver's signal-detection performance, the detector also has to be robust against unknown interference as well as the MAI and ISI components from the known users. Reference [13] uses a subspace

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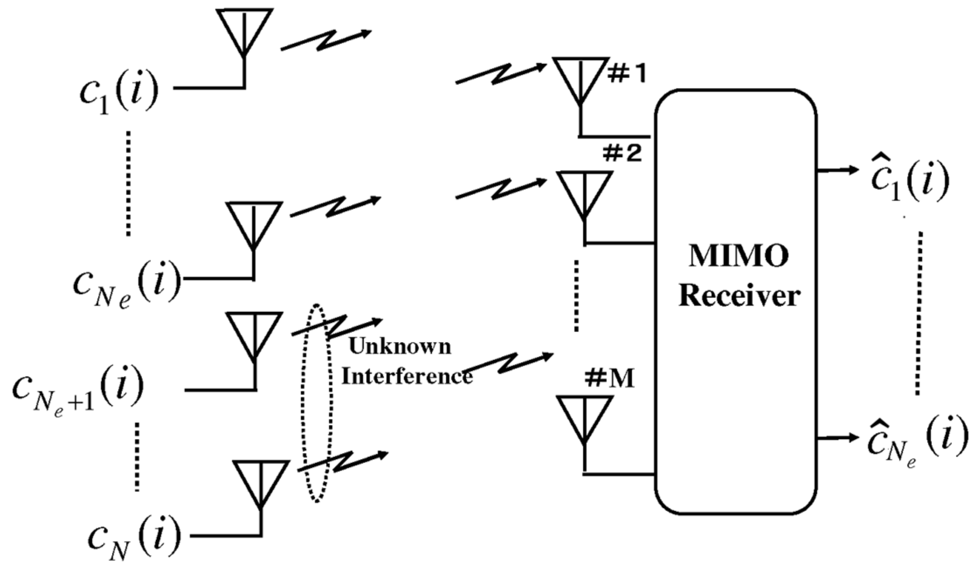

Fig. 1. MIMO channel model.

method to blindly estimate and suppress unknown interference in CDMA communication channels. Reference [14] proposes a blind Bayesian turbo multiuser detector algorithm for CDMA systems that uses long spreading codes, taking into account the presence of unknown interference. In [14], the unknown interference signals are regarded as colored Gaussian noise and the statistical knowledge related to the colored noise is used in signal detection. A goal of this paper is to propose a new ST-turbo equalization algorithm in frequency-selective MIMO channels based on the SC/MMSE concept, in which account is taken of the presence of unknown interference. This paper assumes that unique word sequences are transmitted from the known users' transmitters as signal references. Reference [12]'s proposed MIMO signal detector is modified so that the MMSE filter can effectively suppress the unknown interference. Empirical instantaneous-correlation matrix of the composite unknown interference-plus-noise vector is calculated by using the signal references used in the channel re-estimation process. The estimated correlation matrix is then used to derive the MMSE filter taps so that the unknown interference can be suppressed. Even with this simple method, a more accurate estimate of the correlation matrix of the composite unknown interference-plus-noise vector can be obtained as more iterations are performed. This is because with [12]'s proposed iterative channel-estimation method, the more iterations, the higher the accuracy of the channel estimates, resulting in better estimates of the correlation matrix of the composite unknown interference-plus-noise vector.

This paper is organized as follows. Section II describes the frequency-selective MIMO channel model assumed in this paper. A mathematical ST representation of the MIMO channel is given. Section III proposes a new MIMO turbo equalization algorithm that can suppress unknown interference. Section IV describes a technique for iteratively estimating known users' channels and the correlation matrix of a composite unknown interference-plus-noise vector. In Section V, results of computer simulations are shown to verify the effectiveness of the proposed MIMO turbo equalization algorithm. Concluding remarks are given in Section VI.

II. MIMO Channel Model

Fig. 1 shows a MIMO channel model with unknown interference: there are N users in total and the receiver has M antennas. We assume that the receiver aims to detect N_e users $(N_e < N)$ and the rest $(= N - N_e)$ of users are unknown to the receiver.

This paper assumes a coded system as shown in Fig. 2. The information symbols $c_n(i)$s are first encoded by each user's channel encoder, where i and n denotes the symbol and user indices, respectively. The coded symbols are then interleaved and modulated according to the modulation format used. The modulated symbols $b_n(k)s$ are then transmitted over frequency-selective channels. k denotes the symbol index of modulated symbols. At the receiver, discrete time measurement at the mth antenna yields the sampled value series $r_m(k)$ of the antenna output as

$$r_m(k) = \sum_{n=1}^{N_e} \sum_{l=0}^{L-1} h_{mn}(l) b_n(k-l) + \sum_{n=N_e+1}^{N} \sum_{l=0}^{L-1} h_{mn}(l) b_n(k-l) + v_m(k) \quad (1)$$

where L is the channel memory length. Without loss of generality, the channel memory length is assumed to be identical for all the N users. $h_{mn}(l)$ is a discrete time representation of the channel impulse responses between the nth user and the mth receiver antenna. $v_m(k)$ is additive white Gaussian noise (AWGN).[1] Stacking those measurements into a vector form, which is equivalent to sampling in the space domain, results in

$$\boldsymbol{r}(k) \equiv [r_1(k), r_2(k) \ldots r_M(k)]^T \quad (2)$$

[1]Equation (1) holds regardless of the synchronism in symbol timing among the users. In fact, impulse responses of the channel and the filters in transmitters and receivers can be folded into $h_{mn}(l)$ (see [15]).

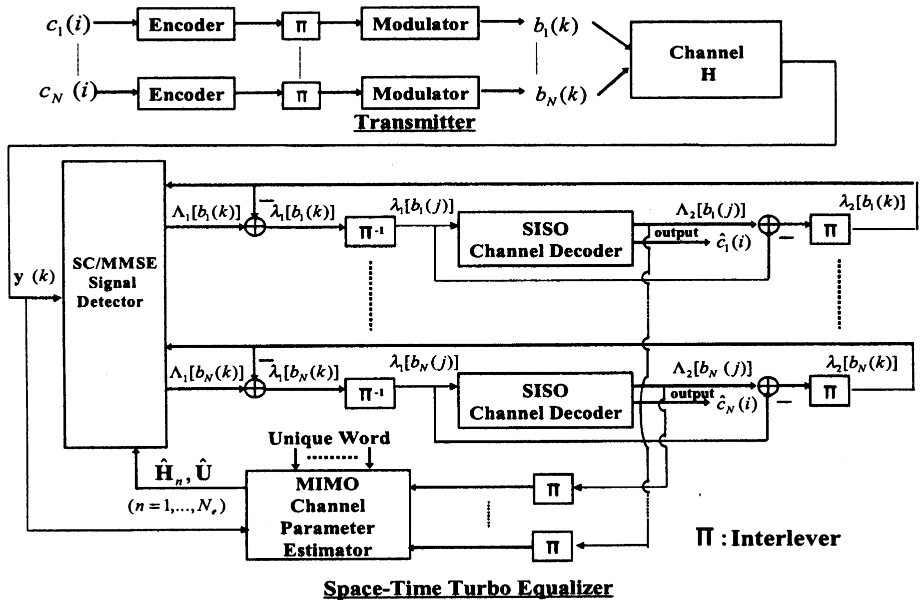

Fig. 2. System model.

$$= \sum_{n=1}^{N}\sum_{l=0}^{L-1} \boldsymbol{h}_n(l) b_n(k-1) + \sum_{n=N_e+1}^{N}\sum_{l=0}^{L-1} \boldsymbol{h}_n(l) b_n(k-l) + \boldsymbol{v}(k) \tag{3}$$

where

$$\boldsymbol{h}_n(l) = [h_{1n}(l), h_{2n}(l) \ldots h_{Mn}(l)]^T \tag{4}$$

$$\boldsymbol{v}(k) = [v_1(k), v_2(k) \ldots v_M(k)]^T. \tag{5}$$

Temporal sampling then takes place to capture the multipath signals, which yields the following ST representation of the received signal $\boldsymbol{y}(k)$:

$$\boldsymbol{y}(k) \equiv \left[\boldsymbol{r}^T(k+L-1), \boldsymbol{r}^T(k+L-2) \ldots \boldsymbol{r}^T(k)\right]^T \tag{6}$$

$$= \sum_{n=1}^{N_e} \boldsymbol{H}_n \boldsymbol{b}_n(k) + \sum_{n=N_e+1}^{N} \boldsymbol{H}_n \boldsymbol{b}_n(k) + \boldsymbol{n}(k) \tag{7}$$

where

$$\boldsymbol{H}_n = \begin{bmatrix} \boldsymbol{h}_n(0) & \ldots & \boldsymbol{h}_n(L-1) & & \boldsymbol{O} \\ & \ddots & & \ddots & \\ \boldsymbol{O} & & \boldsymbol{h}_n(0) & \ldots & \boldsymbol{h}_n(L-1) \end{bmatrix} \tag{8}$$

is the ST channel matrix of the nth user with $\boldsymbol{b}_n(k)$ and $\boldsymbol{n}(k)$ being

$$\boldsymbol{b}_n(k) = [b_n(k+L-1) \ldots b_n(k) \ldots b_n(k-L+1)]^T \tag{9}$$

and

$$\boldsymbol{n}(k) = \left[\boldsymbol{v}^T(k+L-1) \ldots \boldsymbol{v}^T(k)\right]^T \tag{10}$$

respectively.

III. ST Turbo Equalization

A. System Model

Fig. 2 shows a block diagram of the soft canceller followed by MMSE (SC/MMSE) ST turbo equalizer for frequency-selective MIMO channels. The receiver is comprised of a MIMO channel estimator, an SC/MMSE signal detector, and single-input–single-output (SISO) channel decoders for the N_e known users.

Binary phase-shift keying (BPSK) is assumed as a modulation scheme used. The detector produces the log likelihood ratio (LLR) for each coded bit as

$$\Lambda_1\left[b_n(k)\right] = \log \frac{Pr\left[b_n(k)=+1|\boldsymbol{y}(k)\right]}{Pr\left[b_n(k)=-1|\boldsymbol{y}(k)\right]} \tag{11}$$

$$\equiv \lambda_1\left[b_n(k)\right] + \lambda_2^p\left[b_n(k)\right] \tag{12}$$

where $\lambda_1[b_n(k)]$ is the extrinsic information fed to the n-th user's SISO channel decoder following the SC/MMSE detector, and $\lambda_2^p[b_n(k)]$ is the a priori information provided by the nth user's channel decoder.

The channel decoders derive the LLR for each coded bit as

$$\Lambda_2\left[b_n(j)\right] = \log \frac{Pr\left[b_n(j)=+1|\lambda_1\left[b_n(j)\right],\ j=0,\ldots,B-1\right]}{Pr\left[b_n(j)=-1|\lambda_1\left[b_n(j)\right],\ j=0,\ldots,B-1\right]} \tag{13}$$

$$\equiv \lambda_2\left[b_n(j)\right] + \lambda_1^p\left[b_n(j)\right], \tag{14}$$

where, with j being symbol-timing index after deinterleaving, $\lambda_2[b_n(j)]$ is the extrinsic information fed back to the detector and $\lambda_1^p[b_n(j)]$ is the a priori information provided by the detector. B is the burst length. Finally, the information symbol of each user is decoded as

$$\hat{c}_n(i) = \text{sign}\left(\Lambda_2\left[c_n(i)\right]\right), \quad (n=1,\ldots,N_e) \tag{15}$$

where

$$\Lambda_2[c_n(i)] = \log\frac{Pr[c_n(i)=+1|\lambda_1[b_n(j)], j=0,\ldots,B-1]}{Pr[c_n(i)=-1|\lambda_1[b_n(j)], j=0,\ldots,B-1]}. \tag{16}$$

B. SC/MMSE Signal Detector

A block diagram of the SC/MMSE detector is shown in Fig. 3. The detector consists of N_e independent detectors. In the fol-

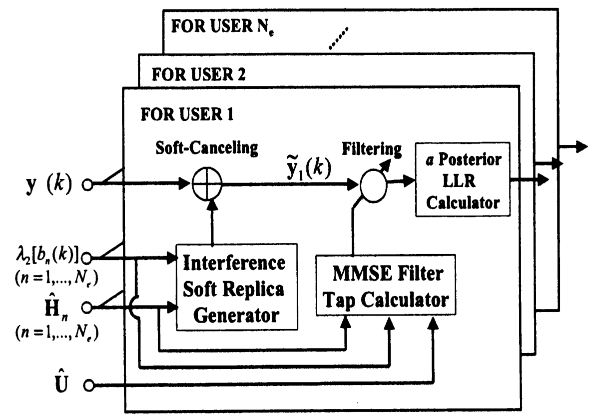

Fig. 3. SC/MMSE signal detector.

lowing, we assume the first user is the user of interest. The same algorithm should apply to signal detection of the other $N_e - 1$ users.

Utilizing the extrinsic information provided by the N_e users' channel decoders, the detector first forms soft estimates of the N_e users' kth symbols as

$$\tilde{b}_n(k) = \tanh\left[\frac{\lambda_2\left[b_n(k)\right]}{2}\right], (n = 1, \ldots N_e) \tag{17}$$

which are used to form *soft replica* $\sum_{n=1}^{N_e} \boldsymbol{H}_n \tilde{\boldsymbol{b}}_n(k)$ of the MAI and ISI components. The soft replica is then subtracted from the received signal vector $\boldsymbol{y}(k)$ to produce the first user's signal estimate vector as

$$\tilde{\boldsymbol{y}}_1(k) = \boldsymbol{y}(k) - \sum_{n=1}^{N_e} \boldsymbol{H}_n \tilde{\boldsymbol{b}}_n(k) \tag{18}$$

with

$$\tilde{\boldsymbol{b}}_n(k) = \left[\tilde{b}_n(k+L-1)\ldots\tilde{b}_n(k)\ldots\tilde{b}_n(k-L+1)\right]. \tag{19}$$

For $n = 1$,

$$\tilde{\boldsymbol{b}}_1(k) = \left[b_1(k+L-1)\ldots 0 \ldots \tilde{b}_1(k-L+1)\right]. \tag{20}$$

with its Lth element being zero. Equation (18)'s process is referred to as *soft interference cancellation*.

The objective of the rest of the algorithm is to suppress the residuals of ISI and MAI from the known users left after the soft interference cancellation and also to suppress MAI from unknown interferers. An adaptive linear filter is used for this purpose: the $M \times L$-vector $\boldsymbol{w}_n(k)$ of the filter taps is determined so that the MSE between the filter output and the signal point corresponding to the first user's detected symbol is minimized as

$$\boldsymbol{w}_1(k) = \arg\min_{\boldsymbol{w}_1(k)} \left\|\boldsymbol{w}_1^H(k)\tilde{\boldsymbol{y}}_n(k) - b_1(k)\right\|^2. \tag{21}$$

Obviously, the Wiener solution $\boldsymbol{w}_1(k)$ to this minimization problem is given as

$$\boldsymbol{w}_1(k) = \left[\sum_{n=1}^{N_e} \boldsymbol{H}_n \boldsymbol{\Lambda}_n(k) \boldsymbol{H}_n^H + \sum_{n=N_e+1}^{N} P_n \boldsymbol{H}_n \boldsymbol{H}_n^H + \sigma^2 \boldsymbol{I}\right]^{-1} \boldsymbol{h}_1 \tag{22}$$

$$\equiv \left[\sum_{n=1}^{N_e} \boldsymbol{H}_n \boldsymbol{\Lambda}_n(k) \boldsymbol{H}_n^H + U\right]^{-1} \boldsymbol{h}_1 \tag{23}$$

where

$$\boldsymbol{h}_1 \equiv \left[\boldsymbol{h}_1^T(L-1), \ldots, \boldsymbol{h}_1^T(0)\right]^T \tag{24}$$

$$U \equiv \left[\sum_{n=N_e+1}^{N} P_n \boldsymbol{H}_n \boldsymbol{H}_n^H + \sigma^2 \boldsymbol{I}\right] \tag{25}$$

and

$$\boldsymbol{\Lambda}_n(k) = \mathrm{diag}\left[P_n - \tilde{b}_n^2(k+L-1) \ldots P_n - \tilde{b}_n^2(k) \ldots P_n - \tilde{b}_n^2(k-L+1)\right]. \tag{26}$$

For $n = 1$

$$\boldsymbol{\Lambda}_1(k) = \mathrm{diag}\left[P_1 - \tilde{b}_n^2(k+L-1) \ldots 0 \ldots P_1 - \tilde{b}_1^2(k-L+1)\right] \tag{27}$$

with its (L, L)-element being zero.

P_n is nth user's signal power ($= E[|b_n(k)|^2]$). U is the correlation matrix of the composite unknown interference-plus-noise vector.

By approximating error at the MMSE filter output by a Gaussian process [10], the extrinsic information to be delivered to the channel decoder can be derived as

$$\lambda_1\left[b_1(k)\right] = \log \frac{Pr\left[\boldsymbol{y}(k)|b_1(k) = +1\right]}{Pr\left[\boldsymbol{y}(k)|b_1(k) = -1\right]} \tag{28}$$

$$= \frac{4Re\left[z_1(k)\right]}{1 - \mu_1(k)} \tag{29}$$

where $z_1(k)$ is the filter output

$$z_1(k) = \boldsymbol{w}_1^H(k)\tilde{\boldsymbol{y}}_1(k) \tag{30}$$

and

$$\mu_1(k) = \boldsymbol{h}_1^H \left[\sum_{n=1}^{N_e} \boldsymbol{H}_n \boldsymbol{\Lambda}_n(k) \boldsymbol{H}_n^H + U\right]^{-1} \boldsymbol{h}_1. \tag{31}$$

IV. Iterative Estimation for Channel Parameters

Obtaining the MMSE filter taps requires knowledge about channel impulse responses of the N_e known users and also the correlation matrix U defined in the previous section. In [12], we assumed no unknown interference exists on the channel. Hence, the correlation matrix U is reduced to $\sigma^2\boldsymbol{I}$. In [12], we proposed a channel-estimation technique that iteratively estimates the channel impulse responses of the N_e known users and also σ^2 as an minimum mean square error of the estimator. In this paper, however, since we assume unknown interference exists on the channel, the matrix U has to be estimated as correlation matrix of composite unknown interference-plus-noise vector for the MMSE filter to suppress the unknown interference. Thus, in the following, we extend the method in [12] and show a procedure to iteratively estimate matrix U of the composite unknown interference-plus-noise vector as well as the channel impulse responses of the N_e known users, which can jointly work with the iterative equalization algorithm.

A. Iterative Estimation for Channel Impulse Response

It is assumed that each of the N_e users' information sequence is headed by a unique sequence whose waveform and timing are known to the receiver. Prior to the first iteration for the MIMO equalization, the N_e known users' channel impulse responses are estimated by using the unique word sequences as signal references. Assuming that the N_e known users' unique word sequences are uncorrelated with the composite unknown interference-plus-noise component, the recursive least square (RLS) algorithm may be used to estimate the N_e users' channel impulse responses. Initial estimates of the N_e users' channel impulse responses are obtained at the end of the unique word period. The detector then runs the first iteration of the MIMO equalization algorithm described in Section III. The first iteration produces initial soft estimates of the N_e known users' transmitted symbols given by

$$\tilde{b}_n(k) = \tanh\left[\frac{\Lambda_2\left[b_n(k)\right]}{2}\right]. \tag{32}$$

Obviously, the larger the $|\tilde{b}_n(k)|$'s value, the more reliable it is, which suggests that the hard decisions of $\tilde{b}_n(k)$s' having relatively large $|\tilde{b}_n(k)|$ values can be used as additional signal references for channel estimation. Thresholding may properly identify the reliable soft estimates. Additional signal references are then given as signal points corresponding to the hard decision results for the symbols identified as being reliable.

Prior to the second iteration, the RLS parameter estimation algorithm is run again using both the unique word waveform and the information symbols identified as being reliable. The estimates of the channel impulse responses are then updated. The detector runs the second iteration for the MIMO equalization using the updated channel estimates. This process is repeated. Because of the turbo principle, the $|\tilde{b}_n(k)|$ values increase with the iteration number, thereby yielding more additional reference signals. This results in better estimates of the channel impulse responses.

When unknown interferers exist in the channel, the RLS channel estimator needs a longer training period to estimate the channels of the known users, since the RLS channel estimator has to decorrelate interference components from unknown interferers. Since the iterative channel estimation algorithm provides additional signal references, it is quite effective in the presence of unknown interferers.

B. Estimation for Correlation Matrix of Unknown Interference Plus Noise

The correlation matrix U of the composite unknown interference-plus-noise vector can also be iteratively estimated by simply time-averaging the instantaneous emperical correlation matrix as

$$\begin{aligned}\hat{\boldsymbol{U}} = &\frac{1}{Tu}\sum_{k=1}^{Tu}\left(\boldsymbol{y}(k) - \sum_{n=1}^{N_e}\hat{\boldsymbol{H}}_n\boldsymbol{b}_n^u(k)\right)\left(\boldsymbol{y}(k) - \sum_{n=1}^{N_e}\hat{\boldsymbol{H}}_n\boldsymbol{b}_n^u(k)\right)^H \\ &+ \frac{1}{Ta}\sum_{k=1}^{Ta}\left(\boldsymbol{y}(k) - \sum_{n=1}^{N_e}\hat{\boldsymbol{H}}_n\tilde{\boldsymbol{b}}_n(k)\right)\left(y(k) - \sum_{n=1}^{N_e}\hat{\boldsymbol{H}}_n\tilde{\boldsymbol{b}}_n(k)\right)^H\end{aligned} \tag{33}$$

where T_u and T_a are the numbers of unique word symbols and the additional reference symbols, respectively. $\boldsymbol{b}_n^u(k) = [b_n^u(k+L-1), \ldots, b_n^u(k), \ldots, b_n^u(k-L+1)]^T (n = 1, \ldots, N_e)$ is the reference signal vector comprised of unique word symbols

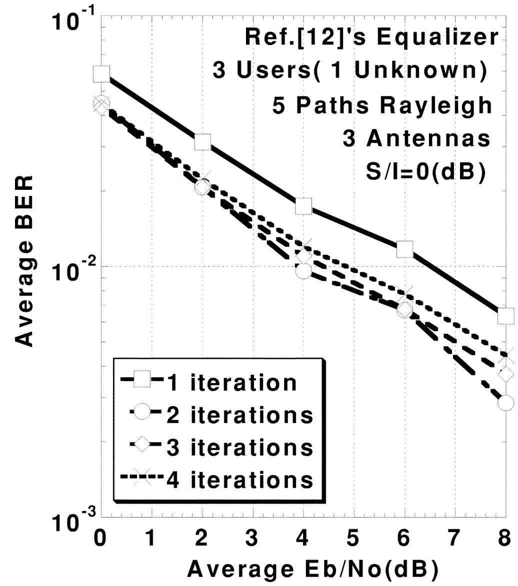

Fig. 4. BER performance of [12]'s equalizer: $\mathrm{S/I} = 0$ (dB).

and $\hat{\boldsymbol{b}}_n(k) = [\hat{b}_n(k+L-1), \ldots, \hat{b}_n(k), \ldots, \hat{b}_n(k-L+1)]^T (n = 1, \ldots, N_e)$ is the additional reference signal vector comprised of the selected information symbols identified as being reliable. $\hat{\boldsymbol{H}}_n s (n = 1, \cdots, N_e)$ are the estimated channel matrix of the N_e known users. The estimate of the correlation matrix is updated at each iteration. Since with more iterations $\hat{\boldsymbol{H}}_n$s become more accurate and T_a becomes larger, a more accurate estimate of the correlation matrix U can be obtained.

V. Simulation Results

This section presents results of computer simulations conducted to evaluate performances of the proposed MIMO turbo equalizer. All simulations assumed that channel frequency selectivity is due to an L-path propagation scenario with each path experiencing frequency-flat Rayleigh fading and each of L path components has identical average power. It is also assumed that the received signal power of N_e known users are identical with each other. The ratio of the each known user's signal power-to-the-total of $N - N_e$ unknown users' signal powers is denoted as S/I. A rate 1/2 nonsystematic convolutional code with the constraint length of 3 and generators $[G_1, G_2] = [5, 7]_{oct}$ was used. One burst has 900 coded symbols, headed by a 100-symbol unique word sequence for channel estimation. The RLS algorithm was used for channel estimation. The threshold value for iterative channel estimation was set at 0.5. N users are assumed to be symbol- and frame-synchronized for simplicity. A random interleaver was assumed. The Max-Log-MAP algorithm was used in the SISO channel decoders. More than 5000 simulation runs were conducted to obtain each plot in the figures shown below.

Figs. 4 and 5 show for $N = 3$, $N_e = 2$, $L = 5$, $M = 3$, and $S/I = 0\ (dB)$ bit error rate (BER) performances of [12]'s and the proposed algorithms, respectively. The normalized maximum Doppler frequency f_dT_s, normalized by the symbol duration T_s, was set to 1/20 000. E_b is defined as the average per-information-bit energy of each known user's signal received by

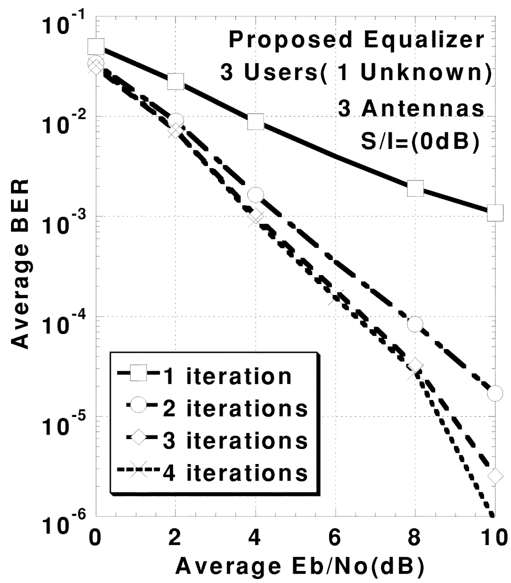

Fig. 5. BER performance of proposed equalizer: $\mathrm{S/I} = 0$ (dB).

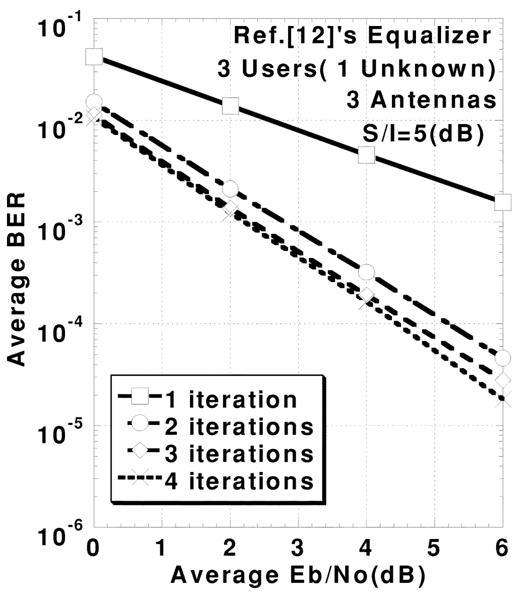

Fig. 6. BER performance of [12]'s equalizer: $\mathrm{S/I} = 5$ (dB).

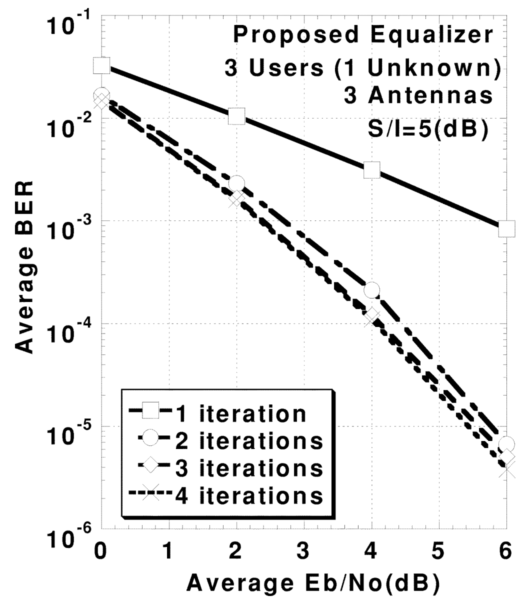

Fig. 7. BER performance of proposed equalizer: $\mathrm{S/I} = 5$ (dB).

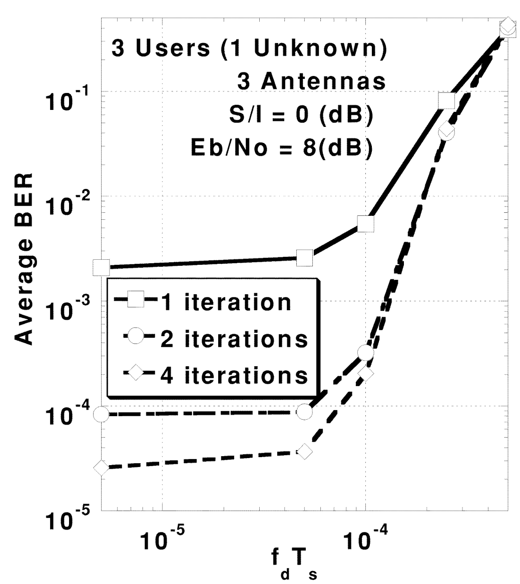

Fig. 8. BER performance of proposed equalizer versus $fdTs$.

one antenna element. The BER curves were obtained by averaging over all users' BERs. Fig. 4 shows that since [12]'s equalizer does not take unknown interferes into account in the received signal, the equalizer fails to achieve reasonable iteration gain.

On the other hand, as shown in Fig. 5, the proposed equalizer achieves significant iteration gain. This is because the MMSE filter suppresses composite unknown interference well by using estimated correlation matrix of the unknown interference-plus-noise vector.

Figs. 6 and 7 show for $N = 3$, $N_e = 2$, $L = 5$, $M = 3$, and $S/I = 5\,(dB)$ BER performance of [12]'s and the proposed algorithms, respectively. The normalized maximum Doppler frequency $f_d T_s$, normalized by the symbol duration T_s, was again set to 1/20 000. It is found that the BER performances are almost same. This is because unknown interference is weak, which does not cause serious model-mismatch problems even in [12]'s detector.

Fig. 8 shows BER performance versus maximum Doppler frequency $f_d T_s$, normalized by the symbol duration T_s, for $N = 3$, $N_e = 2$, $L = 5$, $M = 3$, and $S/I = 0\,(dB)$ with the iteration number as a parameter. It is found that reasonable BER can be maintained if $f_d T_s$ is smaller than 10^{-4} but the performance degrades if $f_d T_s$ is larger than 10^{-3}. This is because the proposed detector does not track the channel variation due to fading.

VI. Conclusion

We have proposed an iterative equalization algorithm for frequency-selective MIMO channels with unknown interference by extending our proposed turbo MIMO equalization algorithm [12]. The proposed algorithm works jointly with the estimations of known users' channel impulse responses as well as that of the correlation matrix of composite unknown interference-plus-noise vector, all in an iterative manner. It has been found that this paper's proposed MIMO turbo equalization algorithm is effective in achieving reasonable iteration gain when signal power of unknown interference is as large as those of known users. Future study will include issues regarding the numerical stability with the proposed algorithm under certain computational environments such as round-off and dynamic range due to the fixed points. Performance comparison with other unknown interference cancellation techniques such as [14] is also left as future study.

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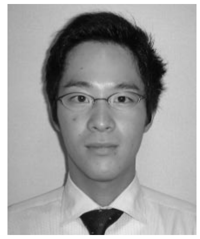

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