The Softest Handoff Design Using Iterative Decoding (Turbo Coding)

Byung K. Yi, Sang G. Kim, and Raymond L. Pickholtz

Abstract: **Communication systems, including cell-based mobile communication systems, multiple satellite communication systems or multi-beam satellite systems, require reliable handoff methods between cell-to-cell, satellite-to-satellite or beam-to-beam, respectively. Recent measurement of a CDMA cellular system indicates that the system is in handoff at about 35% to 70% of an average call period. Therefore, system reliability during handoff is one of the major system performance parameters and eventually becomes a factor in the overall system capacity. This paper presents novel and improved techniques for handoff in cellular communications, multi-beam and multi-satellite systems that require handoff during a session. This new handoff system combines the soft handoff mechanism currently implemented in the IS-95 CDMA with code and packet diversity combining techniques and an iterative decoding algorithm (Turbo Coding). The Turbo code introduced by Berrou *et al.* has been demonstrated its remarkable performance achieving the near Shannon channel capacity [1]. Recently, Turbo codes have been adapted as the coding scheme for the data transmission of the third generation international cellular communication standards: UTRA and CDMA 2000. Our proposed encoder and decoder schemes modified from the original Turbo code is suitable for the code and packet diversity combining techniques. This proposed system provides not only an unprecedented coding gain from the Turbo code and its iterative decoding, but also gain induced by the code and packet diversity combining technique which is similar to the hybrid Type II ARQ. We demonstrate performance improvements in AWGN channel and Rayleigh fading channel with perfect channel state information (CSI) through simulations for at low signal to noise ratio and analyses using exact upper bounding techniques for medium to high signal to noise ratio.**

Index Terms: **CDMA, soft handoff, turbo code, iterative decoding, code combining, packet combining, UTRA, IMT-2000, CDMA 2000.**

I. INTRODUCTION

A cellular communication system, which considers broad representations of cell-based mobile communication systems, multiple satellite communication systems or multi-beam satellite systems, requires reliable handoff methods between cell-to-cell, satellite-to-satellite or beam-to-beam, respectively. Recent measurement, of an IS-95 CDMA cellular system indicates that the system is in handoff at about 35% to 70% of an average call period depending on traffic load and handoff parameter settings. Therefore, system reliability during handoff is one of the major system performance parameters and eventually becomes a function of the overall system capacity measurement. In this paper, we are proposing a novel and improved method for handoffs in cellular communication, in multi-beam and multi-satellite system that require handoff(s) during a session. This handoff method combines the soft handoff mechanism, currently implemented in the IS-95 CDMA [2], with the code diversity combining technique, the packet combining techniques and the iterative decoding algorithm (Turbo Coding). The Turbo code introduced by Berrou and *et al.* has demonstrated its remarkable performance achieving the near Shannon channel capacity [1].

Recently, Turbo codes have been suggested as the coding scheme for the data transmission of for the third generation international cellular communication standards: W-CDMA UTRA and CDMA-2000 [3], [4]. Our proposed encoder and decoder schemes modified from the original Turbo code is suitable for the code and packet diversity combining techniques, which is very similar to the hybrid Type II ARQ system. The proposed system provides not only an unprecedented coding gain from the combination of the new Turbo code encoder configuration and its iterative decoding, but also gain due to the code and packet diversity combining technique well suited for fading channels without bandwidth expansion. The new configuration of the Turbo encoder is structured to utilize the interleaved sequence for the transmission during handoff, which is normally not transmitted in the conventional Turbo code configuration [1].

We demonstrate performance improvements in AWGN channel and Rayleigh fading channel with perfect channel state information (CSI) through simulations and analyses using exact upper bounding techniques proposed by A. Viterbi *et al.* [5].

The paper is organized as follows: In Section II we present the modified encoder and decoder configurations suitable for the proposed handoff mechanism and their operations. In Section III we derive Bit Error Rate (BER) and word error rate (WER) performances for the given configuration. We will also develop performances of the configuration proposed for the third generation W-CDMA, which is the conventional Turbo code scheme and compare it with our proposed configuration through simulations. Simulation techniques have been used for BER and WER estimations at low to medium signal to noise ratios. In this paper, we are not considering the multi-path signal combining using RAKE receivers and CDMA multiple access performance for the sake of simple presentation and assuming that we have only multiple signal paths from/to multiple base stations and perfect CSI with fully interleaved flat Rayleigh fading. Also the

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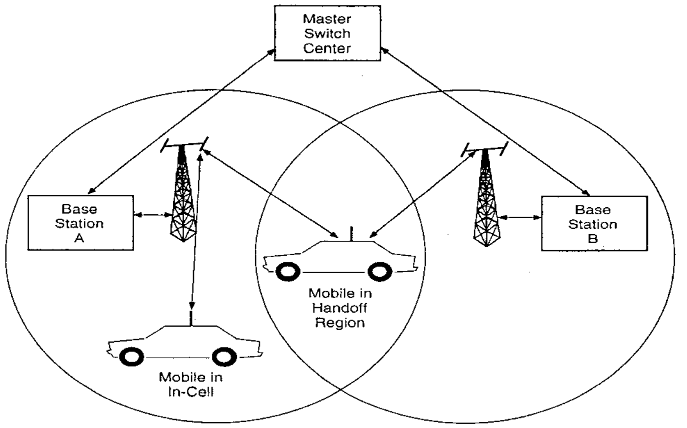

Fig. 1. Overview scheme of a cellular handoff procedure.

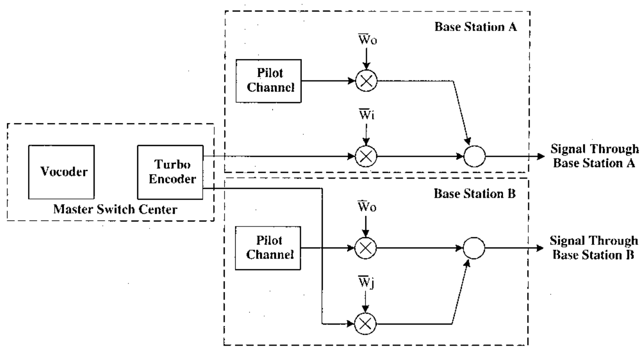

Fig. 2. The forward traffic signals to base station A and B engaging a handoff process.

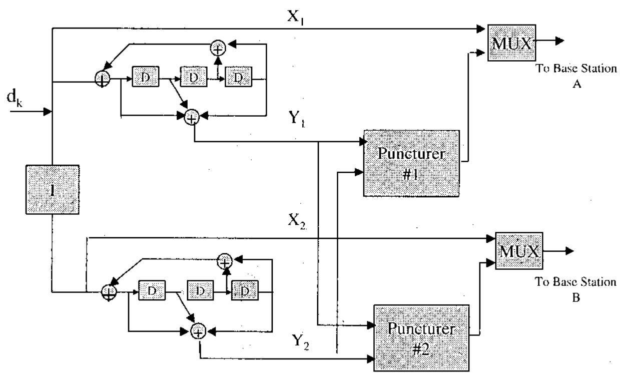

Fig. 3. Detailed proposed Turbo encoder configuration of the code rate 1/2 for the CCPC handoff operations.

performance derivations of the new handoff configuration and of the conventional one over AWGN and Rayleigh channels, using exact upper bounding technique for medium to high signal to noise ratio, are presented in Section IV. Finally Section V is consecrated to some conclusions and to opening further developments.

II. SYSTEM MODELS

A typical CDMA soft handoff is implemented by diversity combining in conjunction with the RAKE receiver, providing better call reliability and supporting the handoff process between cells in a manner that is transparent to the user. We propose a method in which the call reliability during handoff is enhanced compared to the diversity combining soft handoff method, currently proposed in UTRA and CDMA-2000 standards. The proposed handoff scheme is characterized by the combination of code combining and packet combining with the iterative decoding process which is similar to the scheme implemented in the Digital Audio Broadcasting (DAB) system [6]. The currently proposed systems utilize multiple receivers to detect multipath signals and/or signals from different base stations. These signals are time-delayed versions of the same signal and can be combined by a RAKE receiver and a diversity combiner. The diversity combiner uses the maximal ratio combiner on the same signals from different base stations, from different sectors or from different multipath signals. This scheme is similar to the ARQ with a site diversity instead of a time diversity. In the currently proposed novel handoff method, utilizing the interleaved signals with appropriate puncturing will take advantage of the code diversity and coding gain of the powerful iterative coding. Consider an overview of a cellular handoff process, which is illustrated in Fig. 1. If a mobile station travels from the coverage area of the base station A to the coverage area of the base station B, a handoff process must occur to maintain the call signal integrity. In the Code Combining and Packet Combining (CCPC) softest handoff scheme or, in short CCPC handoff, both base stations send differently Turbo encoded signals to the mobile station during handoff. As noted, concepts of the CCPC may apply to both forward channel and reverse channel communications.

A. *The Forward Channel Encoder of the CCPC*

Consider first the forward channel during handoff. The forward traffic signals to base stations A and B which are engaging a handoff process are illustrated in Fig. 2 and a detailed proposed Turbo encoder scheme is shown in Fig. 3. The forward data traffic sequence out of the source encoder is encoded by the Turbo-code instead of the convolutional code used in IS-95. Conventional IS-95 system transmits the same convolutionally encoded sequences through base stations, which are in the active set and the RAKE receiver combines those sequences using a maximal ratio combiner. Then the appropriate decoder decodes the combined sequence. As shown in Fig. 3, the CCPC encoder of the forward traffic during handoff to base stations A and B uses not only a systematic sequence and both parity sequences punctured appropriately, but also the interleaved sequence. For normal in-cell operations (not in handoff operation), the forward traffic transmits either one of code rate 1/2 Turbo encoded sequences. Whether it takes a systematic sequence with the representative punctured sequence or an interleaved sequence with the representative punctured parity sequence depends on the code assignment to the base stations. The Turbo encoder depicted in Fig. 3 includes an interleaver between two constituent recursive convolutional encoders to permute the information sequence in a random fashion. The two constituent encoders are not limited to only the Turbo code. Variety of codes, such as a non-recursive convolutional code serial concatenated Turbo code and etc., can be used.

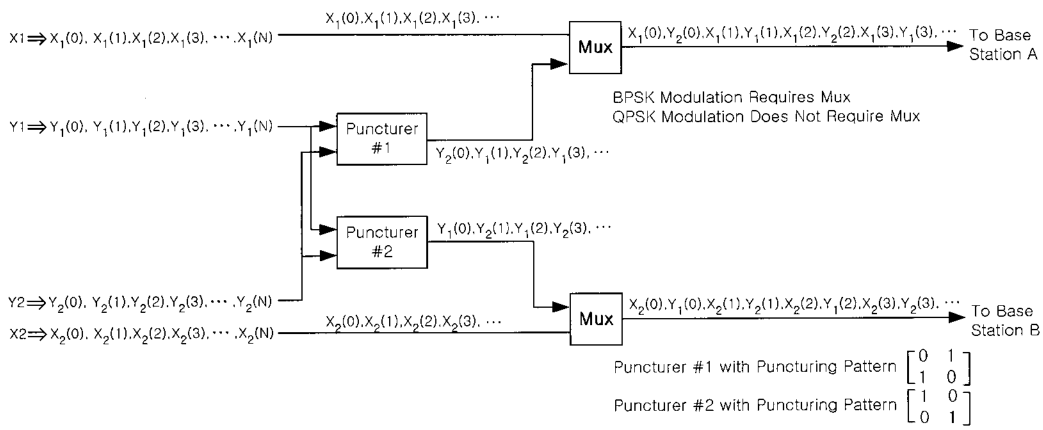

Fig. 4. Detailed parity puncturers and MUXs.

The size of the interleaver and the permutation algorithm are important parameters in Turbo coding performance [7]. Typically, as the interleaver size increases, the code performance is likewise enhanced. However, the interleaver size is limited by system constraints such as allowable voice latency and frame error rate. For example, voice communication cannot tolerate high latency; thus it can have an interleaver size up to 400 bits. Other data communications such as image, video and file transfer do not have such limitations on the interleaver size. In this paper, we address non-voice applications, which can afford the large interleaver size. We used the interleaver size of 2,048 bits for simulations and upper bounding analyses. The implementation of the proposed handoff scheme for the voice traffic with a short interleaver size will be addressed in a subsequent paper.

Permutation by the interleaver breaks the cross-correlation between two decoded sequences. The permutation method can be a purely random interleaver whose permutation map is generated randomly. It can be the pseudo random interleaver introduced by Berrou [8] or the analytical interleaver with the shortening algorithm proposed by Takeshita *et al.* and Yi [9], [10]. We used the random interleaver, generated whenever a frame of information was transmitted for simulations.

The operation of constituent encoders can be described by the following polynomial representation:

$$(1, g_2/g_1) = (1, 1 + D + D^3/1 + D^2 + D^3). \qquad (1)$$

Conventional octal representation for the illustrated code is $(1, g_2/g_1) = (1, 15/13)$. Here, g_1 represents the feedback connection and g_2 represents the feed forward connection. The Turbo encoder shown in Fig. 3 is one of the codes proposed by IMT-2000 third generation specifications.

The parity sequence outputs Y_1 and Y_2 of the two RSC encoders are inputs to the parity puncturers to generate two different punctured sequences described in greater detail in Fig. 4. The first puncturer punctures the parity outputs Y_2 and Y_1, generated by the systematic sequence X_1 and the interleaved sequence X_2, according to the puncturing pattern $[0, 1/1, 0]$ in an alternating Y_2 and Y_1 bit output sequences. The second puncturer punctures the parity output Y_1 and Y_2,according to the puncturing pattern $[1, 0/0, 1]$ in an alternating Y_1 and Y_2 bit output sequences.

In this manner, according to a proposed handoff method, the forward transmissions can provide a punctured code rate of $1/2$ for each signal path to one of the base stations A and B during handoff (or during an in-cell operation). By transmitting the uninterleaved systematic sequence $X1$ and the interleaved systematic sequence $X2$ in conjunction with the usage of the code diversity combining and the packet combining at the receiver in the mobile station, the system provides an overall code rate of $1/4$ during handoff and yields a better performance than the currently proposed soft handoff method. The forward channel of the conventional CDMA system is limited to a code rate of $1/2$. Therefore, transmitting the interleaved sequence is the key salient feature of the CCPC encoder. The result of the interleaved sequence with appropriate puncturing of the parity sequences and coding scheme, provides resistance to detrimental fading that may occur in cellular links is more effective.

B. The Forward Channel Decoder of the CCPC

However, the proposed approach requires multiple demodulator receivers to demodulate two or more differently encoded signal streams from different base stations. Each demodulator receiver contains a RAKE receiver to combine multipath signals. The demodulated signals are combined through the packet combiner and the code diversity combiner in conjunction with an iterative decoder. This new technique demonstrates that actual communication performance measured, by the BER (Bit Error Rate) and the WER (Word Error Rate) better than that of normal in-cell or in-beam operation and eventually this increases system capacity due to improved handoff reliability.

The combination of the handoff process with the iterative decoding process is inherently a robust and seamless transition, which is completely transparent to users. It provides the softest and the most reliable handoff scheme in cellular environments. In the meanwhile, it takes substantial coding gain provided by the iterative decoding [1] with the conventional code rate. At the receiver, the interleaved sequence and the uninterleaved original sequence are combined by the packet combiner and punctured parity check sequences are combined by the code combiner. The code combining and the packet combining techniques not only provide unprecedented coding gain, but also mitigate *detrimental* fading effects on cellular systems.

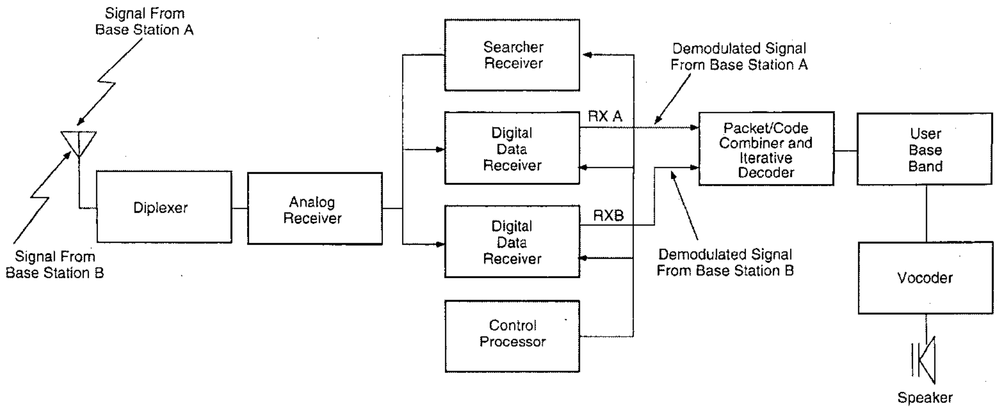

Fig. 5. The receiver configuration implemented for the CCPC handoff operations.

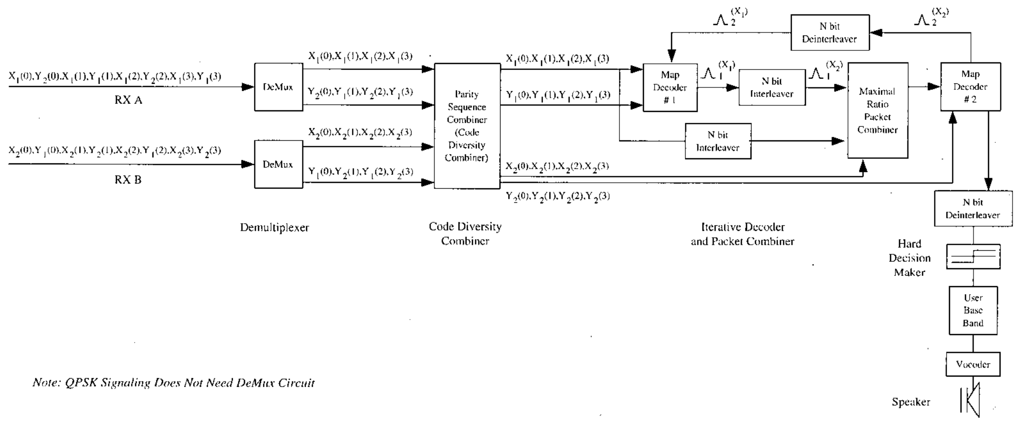

Fig. 6. The CCPC and iterative Turbo decoder.

The CDMA mobile receiver configuration modified to implement the newly proposed handoff, is illustrated in Fig. 5 and the detailed CCPC and iterative decoder is shown in Fig. 6. The antenna, diplexer, and analog receiver are standard elements of the CDMA cellular circuitry [11].

The receivers process two differently coded signal streams from two different base stations, instead of processing the same signals from both base stations as in a conventional IS-95 CDMA receiver. The outputs of the receivers are RxA and RxB, respectively. These output sequences are provided to the packet/code combiner and iterative decoder.

As illustrated in Fig. 6, the code combiner performs demultiplexing to systematic data sequences from the multiplexed signals and depunctures parity check sequences. Through this depuncturing and reshuffling process, code combining is achieved.

The maximal ratio packet combiner combines the interleaved version of the first MAP (Maximum a posteriori probability) decoder output with the interleaved sequences sent over the channel from a basestation and with interleaved sequence from another base station. The output signal $\Lambda_1^{(X_2)}$ (combined) from the maximal ratio combiner and the estimated parity sequences Y_2 are fed to the second MAP decoder. The output of the maximal ratio combiner is expressed by:

$$\Lambda_1^{(X_2)}(\text{combined}) = a_1\Lambda_1^{(X_2)} + a_2X_2 + a_3X_1(\text{interleaved}), \tag{2}$$

where a_1, a_2, and a_3 are measured relative fading parameters for the extrinsic information of the output of the first decoder, the demodulated interleaved information sequence and the uninterleaved sequence later interleaved at the decoder. The second MAP decoder also outputs the reliability data of the interleaved information sequence as an extrinsic feed back signal $\Lambda_2^{(X_2)}$.

The number of iterations performed depends on the system performance criteria, such as final bit error rate, latency tolerance and allowable processing power. Our performance demonstration uses 5 iterations for simulations.

C. The Soft Handoff Schemes for FDMA, TDMA, CDMA to CDMA, Generation to Generation

In FDMA, TDMA, CDMA to CDMA, Generation to Generation when the mobile unit moves out of a currently serviced cell area during a call, the received signal becomes weak and the present cell site requires a handoff. The system switches to a new channel while the call continues. We often called this process a hard handoff or a "break before connect," to highlight the advantages of the soft handoff's "connect before break." However, our newly proposed method allows even for FDMA, TDMA, CDMA to CDMA and Generation to Generation to have a type of soft handoff mechanism by the code and the packet diversity combining technique with an iterative decoding. It requires the receiver entity containing the capability of

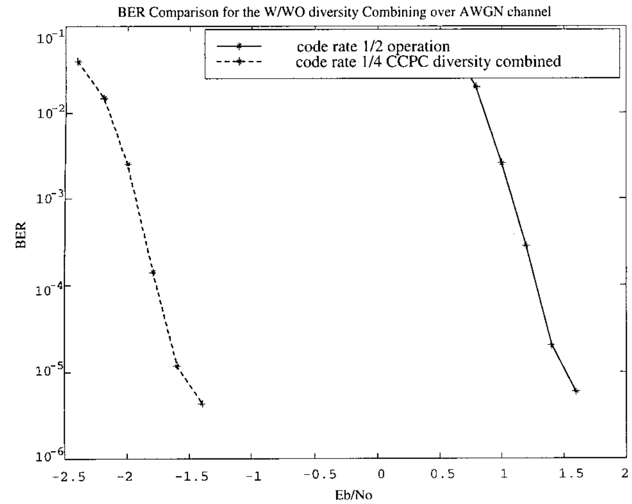

Fig. 7. Bit error rate comparison for with and without CCPC over AWGN channel.

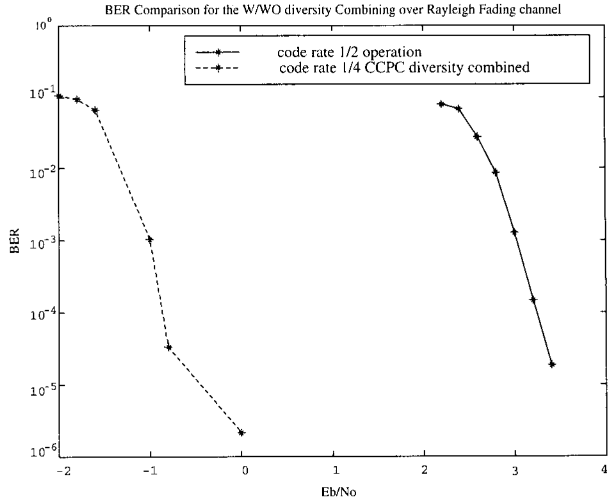

Fig. 9. Bit error rate comparison for with and without CCPC over Rayleigh channel.

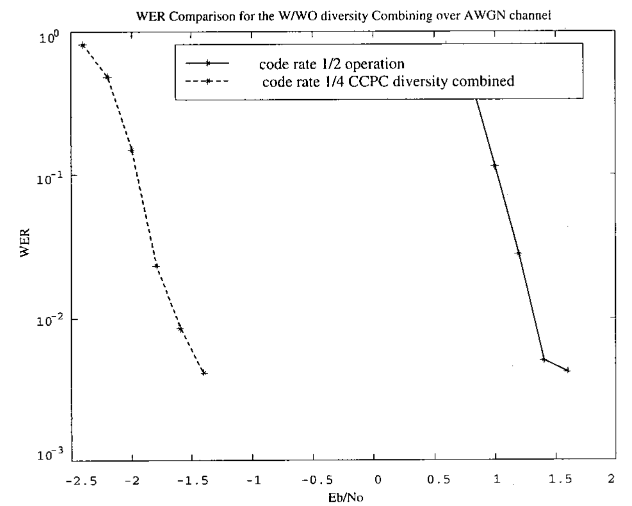

Fig. 8. Word error rate comparison for with and without CCPC over AWGN channel.

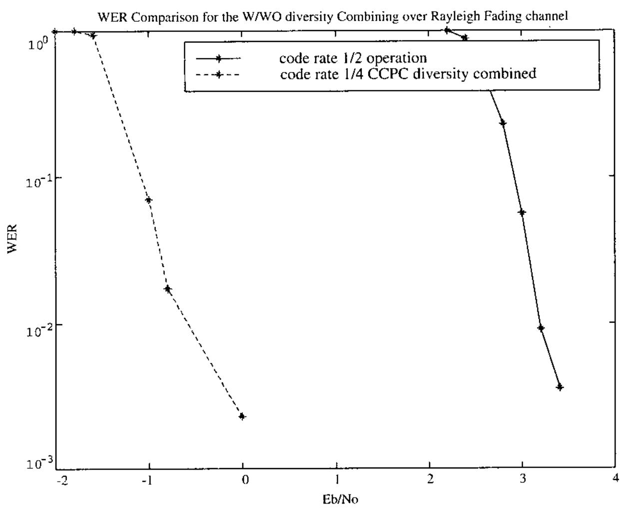

Fig. 10. Word error rate comparison for with and without CCPC over Rayleigh channel.

two channels—two frequency diversity receivers for FDMA and for multi-cell TDMA and two time slots in single cell TDMA, two CDMA receivers for the CDMA to CDMA handoff case or a dual phone receivers for the Generation to Generation handoff.

III. PERFORMANCE AND SIMULATION RESULTS

Computer simulations were used to evaluate BER and WER performances of the CCPC handoff scheme over AWGN and Rayleigh channels. Performances of the CCPC were compared with the scheme proposed for the third generation CDMA. We used the UTRA Turbo code with constraint length 4 with the generator polynomials of (1, 15/13) in octal shown in Fig. 3. The data frame length (word length) in all simulation results presented here was 2,048 bits, with the random interleaver algorithm.

Simulations were carried out for the optimal MAP decoding in both AWGN and Rayleigh channels. In the MAP decoding, the noise variances were assumed to be known by the decoder. For the Rayleigh channel, we assumed a perfect CSI with fully interleaved flat Rayleigh fading.

The BER and WER performances of the proposed handoff scheme have been compared with the conventional approach for AWGN channel shown in Figs. 7 and 8, respectively. The BER and WER performances for Rayleigh fading channel are shown in Figs. 9 and 10, respectively. It indicates that the CCPC handoff schemes demonstrate 3.4 dB and 4.1 dB diversity gains for AWGN and Rayleigh channels, respectively for the 10^{-4} BER at low to medium signal to noise ratio. As expected, diversity gains for AWGN channel is not dramatic. However, for Rayleigh channel the diversity gain is substantial. These gains can be also shown through the analytical upper bounding techniques for medium to high signal to noise ratio.

IV. PERFORMANCE AND UPPER BOUNDS

For an (n, k) linear block code, exact upper bounds to maximum likelihood WER and BER can be derived using the input-

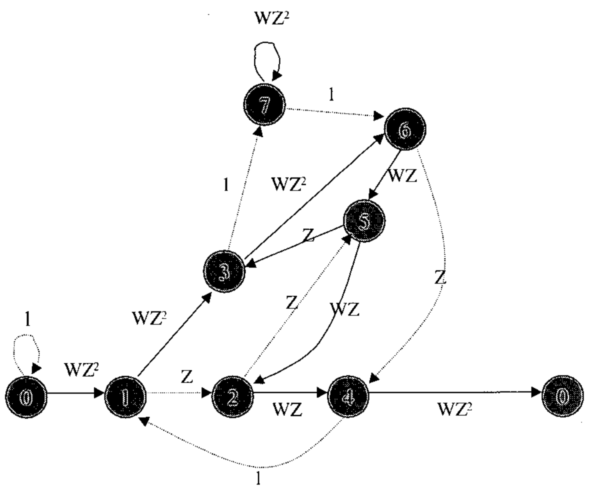

Fig. 11. The state diagram of the 3GPP Turbo constituent code.

output weight enumerating function (IOWEF) defined as,

$$A(W, Z) = \sum_{w=0}^{k} \sum_{z=0}^{n} A_{w,z} W^{w} Z^{z}, \tag{3}$$

where $A_{w,\,z}$ is the number of code words with Hamming weights associated with an input word of weight w and an output weight z .

Based on the code IOWEF, the application of the union bound yields an upper bound to WER and BER for an AWGN channel,

$$P_w(e) < \sum_{w=1}^{k} A(w, Z)|_{Z=e^{-R_c E_b/N_o}}, \tag{4}$$

and

$$P_b(e) < \sum_{w=1}^{k} \frac{w}{k} A(w, Z)|_{Z=e^{-R_c E_b/N_o}}, \tag{5}$$

where $A(w, Z) = \sum_{z=o}^{N} A_{w,\,z} Z^{z}$ is the conditional weight enumerating function (CWEF) which describes the weight distribution of the code words.

Since a terminated Turbo code can be interpreted as an (n, k) block code, close form upper bounds to WER and BER are derived using the CWEF of the Turbo code based on the knowledge of the CWEFs of the two constituent codes. Because the output code weight generated by the second encoder from the interleaved input word depends on the permutation of the interleaver, deriving the CWEF for the given interleaver is not an easy task. The introduction of the "uniform interleaver," which represents the ensemble average of all possible interleavers, allows the derivation of the CWEF of the Turbo code, considering the CWEFs of upper constituent code C_1 and lower constituent code C_2 independently [7]. The Turbo code closed form upper bound for the uniform interleaver is,

$$A^{C_P}(w, Z) = \frac{A^{C_1}(w, Z) A^{C_2}(w, Z)}{\binom{N}{w}}. \tag{6}$$

Due to the properties of the uniform interleaver, the obtained performance for each value of the signal to noise ratio is achievable by at least one deterministic interleaver. Remaining tasks are to calculate CWEFs of constituent codes.

There have been many proposals to derive the upper bound of the iterative coding schemes; Benedetto *et al.* obtained approximate expressions by using concatenated error event counts [12], Divsalar *et al.* used a recursive algorithm based on the transfer function method [13] and A. Viterbi demonstrated a recursion on the transfer function for the serial concatenated codes [5]. We used a recursion based on the transfer function on the log domain to avoid numerical problems within values assumed by $A^{C_1}(w, Z), A^{C_2}(w, Z)$ and $\binom{N}{w}$, which quickly exceed the numerical range supported by a standard computer. The CWEFs of the constituent codes can be derived from the state diagrams shown in Fig. 11 and its associated state transition matrix for the upper lower constituent code is shown below. The state transition matrix of the upper constituent code of the CCSDS Turbo code is described by the expression (7) shown at the bottom of this page, where W, Z, and Z^2 represent "one" input, "one" output, and two "one" outputs.

Note that for our proposed CCPC handoff scheme, the state transition matrix of the lower constituent code have the same transition matrix of the upper constituent code. Therefore the CWEF of the CCPC coding scheme can be derived by

$$A(w, Z) = \frac{A^{C_1}(w, Z) A^{C_2}(w, Z)}{\binom{N}{w}} = \frac{(A^{C_1}(w, Z))^2}{\binom{N}{w}}. \tag{8}$$

Also, the state transition matrix of the code rate 1/2 punctured code, which is used by the normal in-cell single base station, can be derived by modifying the state transition matrix (7).

$$C^1(W, Z) = \begin{bmatrix} 1 & 0 & 0 & & WZ^2 & 0 & 0 & 0 & \\ WZ^2 & 0 & 0 & & 1 & 0 & 0 & 0 & \\ 0 & Z & 0 & & 0 & WZ & 0 & 0 & \\ 0 & WZ^2 & 0 & & 0 & Z & 0 & 0 & \\ 0 & 0 & WZ & & 0 & 0 & Z & 0 & 0 \\ 0 & 0 & Z & & 0 & 0 & WZ & 0 & \\ 0 & 0 & 0 & WZ^2 & 0 & 0 & 0 & 1 & \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 & WZ^2 & \end{bmatrix}, \tag{7}$$

For the every even bit position, the unpunctured upper constituent code will have the same transition matrix the upper constituent code of the state transition matrix (7). The punctured lower constituent code will change the entries of the matrix as follows:

$$\begin{aligned} WZ^2 &\longrightarrow W \\ Z &\longrightarrow 1 \\ WZ &\longrightarrow W. \end{aligned} \tag{9}$$

Similarly, the every odd bit position, the transition matrix of the punctured upper constituent code may be derived by changing entries of the matrix in Eq. (7):

$$\begin{aligned} WZ^2 &\longrightarrow WZ \\ Z &\longrightarrow 1 \\ WZ &\longrightarrow WZ. \end{aligned} \tag{10}$$

And for the unpunctured lower constituent code the transition matrix will become:

$$\begin{aligned} WZ^2 &\longrightarrow WZ \\ Z &\longrightarrow Z \\ WZ &\longrightarrow WZ. \end{aligned} \tag{11}$$

Let us derive the recursive algorithm assuming the interleaver length of N bits and the number of S states (Note that the 3GPP Turbo code has 8 state constituent codes). CWEFs of the $A^{C_1}(w,Z)$ and $A^{C_2}(w,Z)$ for $w=0,1,2,\ldots,N$ can then be computed iteratively in N-steps, starting from state 0 by following all the edges in the trellis [14]. For each step,

- assign to each state s in the trellis, an array $A_w^{C_i}(s)$, $w=0,1,\ldots,N$;
- to force an initial starting state of "0," all values must be set to $A_w^{C_i}(0)=1$;
- also define temporary metric, $\tilde{A}_w^{C_i}$, $w=0,1,\ldots,N$ for each state S;
- then go N-steps through trellis doing the following at each step:
  - initialize $\tilde{A}_w^{C_i}(s)=0$, $w=0,1,\ldots,N$ for each state $s=0,1,\ldots,S-1$;
  - for each branch (trellis edge), e, find start state i and end state j. Let the input weight on that edge be $w(e)$ and output weight be $z(e)$. For all $k=0,1,\ldots,N$ such that $A_k(i)\neq 0$, compute $\tilde{A}_{k+w(e)}(j)=\tilde{A}_{k+w(e)}(j)=A_k(i)Z^{z(e)}$, where $Z=\exp(-R_cE_b/N_o)$ (Chernoff Parameter);
  - update $A_w^{C_i}(s)\leftarrow\tilde{A}_w^{C_i}(s)$, $s=0,1,\ldots,S-1$ and $w=0,1,\ldots,N$.
- after N-iterations, each state s will contain contributions of all paths starting 0 state and end ending 0 state. If terminating state is "0," then the terms of interest are $A_w^{C_i}(0)$, $w=0,1,\ldots,N$.

However, for each moderate interleaver size, the previous algorithmic approach leads to numerical problems. A much better approach is to operate the algorithm in log domain. Let us define the function $E-function$,

$$\begin{aligned} E-function(x,y) &\overset{Def}{=} \log(\exp(x)+\exp(y)) \\ &= \max(x,y)+\log(1+\exp(-|x-y|)), \end{aligned} \tag{12}$$

where the log terms can be approximated and implemented using a look-up table. Then we get,

$$P_w(e) < \sum_{w=1}^{N}\exp(U_w), \tag{13}$$

$$P_b(e) < \sum_{w=1}^{k}\frac{w}{N}\exp(U_w), \tag{14}$$

where $U_w=\alpha_w^{C_1}+\alpha_w^{C_2}-\gamma_w$ with $\alpha_w^{C_1}=\log(A_w^{C_1})$, $\alpha_w^{C_2}=\log(A_w^{C_2})$, and $\gamma_w=\log\begin{pmatrix}N\\w\end{pmatrix}$. The coefficient $\alpha_w^{C_1},\alpha_w^{C_2}$, and γ_w can be computed iteratively using $E-function$ as follows: for the $\alpha_w^{C_1}$ and $\alpha_w^{C_2}$, we now have for all $k=0,1,\ldots,N$ such that $\alpha_w^{C_1}(i)>-\infty$ (Initial conditions), $\tilde{\alpha}_{k+w(e)}(j)=E-function(\tilde{\alpha}_{k+w(e)}(j),\alpha_k(i)-z(e)R_cE_b/N_o)$ and for the γ_w, $\gamma_w=\gamma_{w-1}+\log(N-w+1)-\log(w)$, $w=1,2,\ldots,N$ and $\gamma_0=0$.

The BER and WER performances over AWGN channel derived through the algorithm described above are illustrated in Figs. 12 and 13 respectively. At medium to high signal to noise ratio beyond the cutoff rate, it still shows 3.6 dB gain. The BER and WER performance over Rayleigh channel are shown in Figs. 14 and 15. For the Rayleigh channel, diversity gains are greater than 4.5 dB.

V. CONCLUSIONS

We have proposed a new handoff scheme-CCPC and demonstrated performance improvements in AWGN channel and Rayleigh fading channel with perfect channel state information (CSI) through simulations and analysis using exact upper bounding techniques. It indicates that we can improve BER performances by 3.4 dB and 4.1 dB over AWGN and Rayleigh fading channels, respectively at low signal to noise ratio. These improvements will give rise to performance improvements during handoff operations and eventually increase the system capacity. We presented for the forward channel with code rate 1/2 case. We can implement this CCPC scheme for the reverse channel with code rate 1/3 in the same manner. Also, the proposed CCPC can be one of readily implementable Space-Time code schemes.

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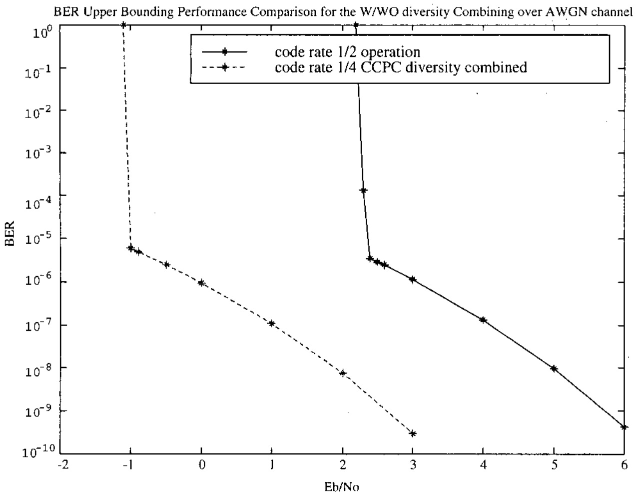

Fig. 12. Upper bound BER comparison for with and without CCPC over AWGN channel.

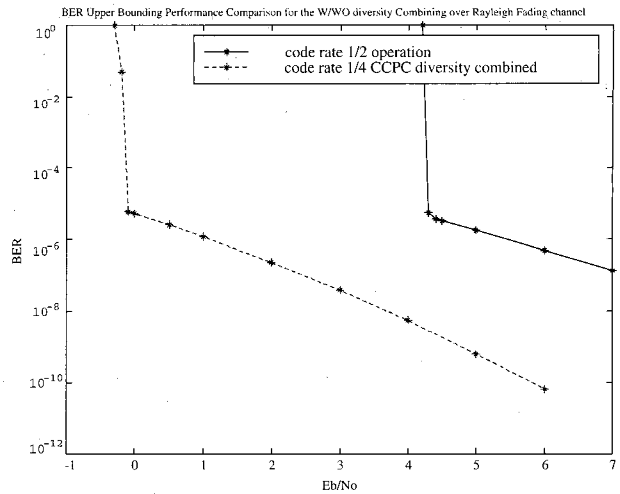

Fig. 14. Upper bound BER comparison for with and without CCPC over Rayleigh channel.

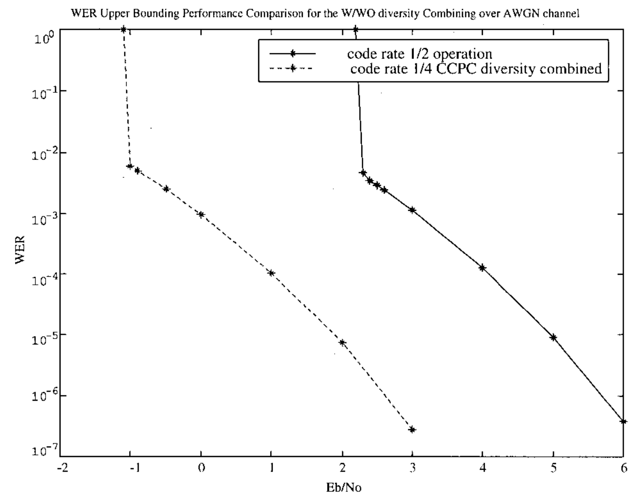

Fig. 13. Upper bound WER comparison for with and without CCPC over AWGN channel.

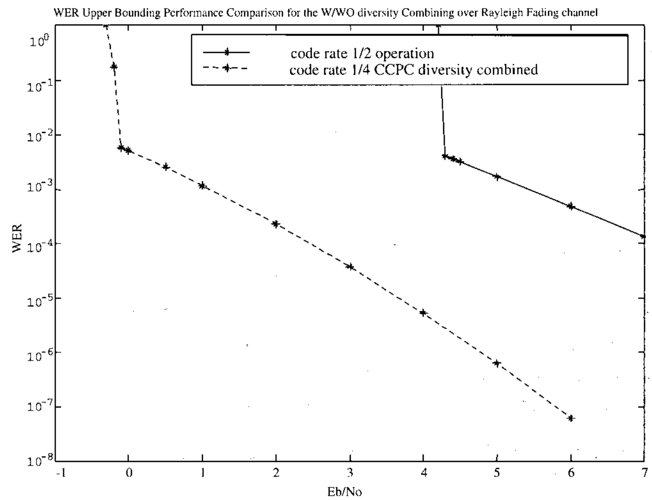

Fig. 15. Upper bound WER comparison for with and without CCPC over Rayleigh channel.

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