11th International Conference on Researches in Science, Technology and Management
Osmania University Campus, Hyderabad (India) RSTM-18
11th March 2018 www.conferenceworld.in ISBN : 978-93-87793-05-7

Mitigationof Security Risk in Satellite Communication by using Turbo coded DSSS

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ABSTRACT

When Turbo coding is employed with BPSK signaling, it execute very fine. Also its performance is very close with Shannon's limit. Although BPSK is straightforward to put into operation and we know all about its performance. Although BPSK is not efficient alone for in terms of bandwidth. Basic BPSK broadcasts one bit/signaling interval. Also when Turbo coding is employed, with BPSK as our theoretical result shows the spectral efficiency improved even further.

Key Words:- Turbo Code ,DSSS ,RSC Encoder, Decoder

I.INTRODUCTION

The Turbo code employed in this section having the transmission rate of 1/3 bits per signaling interval. Puncturing a Turbo coded bit stream and then using BPSK signaling for transmission improves the spectral efficiency efficiently. If we want to increase transmission period, the puncturing matrix is must to use in coding like turbo code which increase spectral efficiency by 1/3 per period. Although if we used BPSK alone its rate always remain smaller than one bit per period but it is not sufficient in terms of satellite communication. So if we want rate smaller than one bit per period there is need of turbo coding over which we directly use puncturing matrix, which is responsible for variation in rate. The main drawback of puncturing is that we get deteriorate the interpretation of the Turbo code, means increasing the rate degradation in the Turbo code performance .For the purpose to improve the interpretation of turbo code ,symbol-based (non-binary) turbo coding is an alternative, and if there are more transitions between different states and by using large interleaver size the decoding turn out to be more complex and also takes long time for decoding, but we get good interpretation of Turbo codes. Consider the of BPSK, the size of interleaver can vary in any size, let it would be N, although large quantity of interleavers are typically used to attain better performances.

The proposed work comprises of two identical Recursive systematic convolution encoders i.e. RSC encoders ,then the signal is passed through Direct sequence spread spectrum ,If the encoding initiates and finished at a identified state, the decoder for each code performs better at every interleaver.

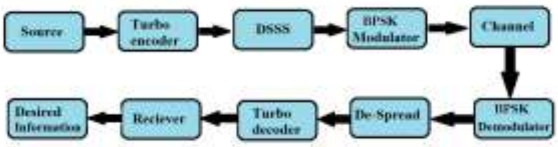

Fig 1.1 Proposed block diagram for performance improvement

The S_c represents circulation state and given by

$C_S = (I+G^n)^{-1}F_e$ (1.1)

Where

C_s = Circulation state of encoder

F_e = Final state of encoder

N = number of couples of data bits

$$G = \begin{bmatrix} 1 & 0 & 1 \\ 1 & 0 & 0 \\ 0 & 1 & 0 \end{bmatrix}$$

I = Identity matrix

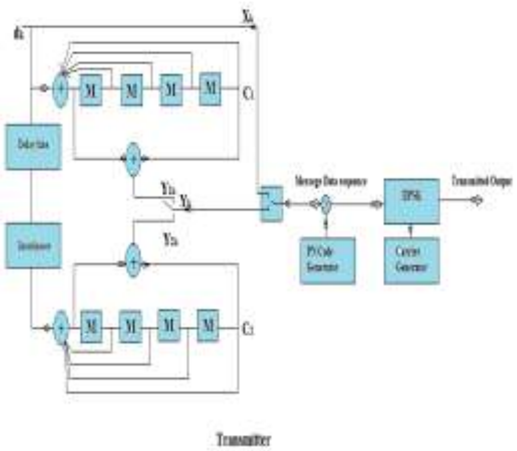

Fig. 1.2 Proposed Turbo coding with DSSS

ISBN : 978-93-87793-05-7

II.TURBO DECODING ALGORITHMS

In a distinctive turbo decoding, two decoders function iteratively and go by their conclusions and all iteration responsible for results. The soft outputs are produced by decoders for the improvement of the decoding interpretation and such decoder is called SISO ,imagine that the encoded information is broadcast at additive white Gaussian noise (AWGN) channel, and we obtained a corrupted sequence, let broadcast sequence X_k and corrupted sequence as Y_k, consider binary case ,in which each decoder based on the calculation of log-likelihood ratio (LLR) for the t^{th} data bit d_k, as follows .

$$\boldsymbol{R(d_k) = log\left[\frac{T(d_k=1|Y)}{T(d_k=0|Y)}\right]} \qquad (1.2)$$

$R(d_k)$ = log-likelihood ratio (LLR)

d_k=k^{th} data bit

Y = received sequence

The LLR can be decomposed into three independent terms as:

$R(d_k) = P_{apri}(d_k) + C(d_k) + E(d_k)$ (1.3)

$R(d_k)$ = log-likelihood ratio (LLR)

$P_{apri(dk)}$ = earlier sequence of d_k

$C(d_k)$ = medium dimension

$E(d_k)$ = exchanged outer message among decoders

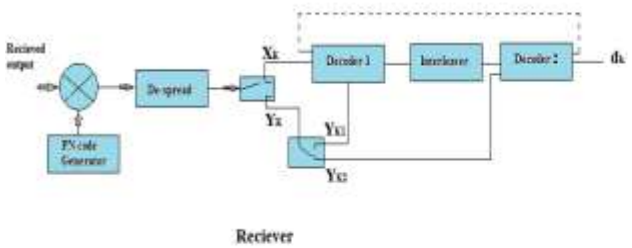

Fig. 1.3 Proposed Turbo coding with de spread DSSS

III.MAP ALGORITHM

MAP is the optimal but computationally complex algorithm. According to this algorithm, LLR values for each information bit can be calculated as:

$$L(d_k) = \ln\left[\frac{\sum_{Z_k}\sum_{Z_{k-1}}\gamma_1(Z_{k-1},Z_k)\alpha(Z_{k-1})\beta(Z_k)}{\sum_{Z_k}\sum_{Z_{k-1}}\gamma_0(Z_{k-1},Z_k)\alpha(Z_{k-1})\beta(Z_k)}\right] \quad (1.4)$$

α = forward state matric

β = backward state matric

γ = branch matric

Z_k= trellis state at time instant k

Where α is the forward state metric, β_k the backward state metric,$\gamma_i(Z_{k-1}, Z_k)$ the branch metric, andZ_k is the trellis state at time instant k. At state k, the forward state metric,$a_k(S_k)$ is given by:

$$\alpha_k(Z_k) = \sum_{j=0}^{1} \propto_{k-1}(Z_{k-1})\gamma_j(Z_{k-1}, Z_k) \quad (1.5)$$

The backward state metric, $\beta_k(Z_k)$ is given by:

$$\beta_k(S_k) = \sum_{j=0}^{1} \beta_{k+1}(Z_{k+1})\gamma_j(Z_k, Z_{k+1}) \quad (1.6)$$

For each possible transition the branch metric can be calculated as:

$$\gamma_i(Z_{k-1},Z_k) = C_k T(Z_k|Z_{k-1}) exp\left[\frac{2}{N_o}(y_k^Z x_k^Z(i) + y_k^T x_k^T(i, Z_{k-1},Z_k))\right]$$
(1.7)

where

$\gamma_i(Z_{k-1}, Z_k)$ = branch matric

C_k= constant

x_k^T = transmitted systematic data at transmitter side

x_k^T= parity bits at transmitter side

y_k^Z, y_k^T = received noisy bits at receiver side

Log-MAP algorithm

Log-MAP performs the calculations in the logarithmic domain by replacing the logarithm by the map* operator as follows

$$\text{map}^*(x,y) = \text{mn}(e^x + e^y) = \text{map}(x,y) + \log(1 + e^{-|y-x|}) \qquad (1.8)$$

Where the correction function having term log $(1 + e^{-|y-x|})$ can be measured using look-up table.

Max-Log-MAP algorithm

It approximates the computation of map* operator in Log-MAP algorithm by omitting the correction term, log$(1 + e^{-|y-x|})$ to become as follows:

$$\ln(e^x + e^y) \approx \text{map}(x,y) \qquad (1.9)$$

RCS Decoder Structure

Two types of symbols are possible, either 0 or 1 in binary case. The decoding process involves duo binary values and symbols for turbo encoding and transmitted as, (00, 01, 10, or 11) as a sequence. The equivalent probability is given as

$$\frac{T(d_k=00/y)}{T(d_k=00/y)} = \frac{T(d_k=00/y).\,T(d_k=00)}{T(d_k=00/y).\,T(d_k=00)} \qquad (1.10)$$

$$\frac{T(d_k=01/y)}{T(d_k=00/y)} = \frac{T(d_k=01/y).\,T(d_k=01)}{T(d_k=00/y).\,T(d_k=00)} \qquad (1.11)$$

$$\frac{T(d_k=10/y)}{T(d_k=00/y)} = \frac{T(d_k=10/y).\,T(d_k=10)}{T(d_k=00/y).\,T(d_k=00)} \qquad (1.12)$$

$$\frac{T(d_k=11/y)}{T(d_k=00/y)} = \frac{T(d_k=11/y).\,T(d_k=11)}{T(d_k=00/y).\,T(d_k=00)} \qquad (1.13)$$

d_k= transmitted symbol at time instant k

y= received continuous valued noisy symbol

Single log-likelihood ratio requires duo binary code for decoders which further require three log- likelihood ratios. The likelihood ratios for information couple (C_kG_k) can be symbolize in the form

$\Lambda_{c,g}(C_k,G_k) = \log\frac{T(C_k=c,G_k=g)}{T(C_k=0,G_k=0)}$ (1.14)

$C_{c,g}(C_k,G_k)$ = likelihood ratios of information couple (C_k, G_k)

where *(C, G)* can be (0, 1), (1, 0), or (1, 1).Now let y_k **(Z,** ^**Z**$_j$) denote the branch metric corresponding to state transition **Z,** ^**Z**$_j$ at time k. The branch metric depends on the information and parity couples that label the branch all beside the channel observation and extrinsic information at the decoder input. In particular, if transition **Z,** ^**Z**$_j$ is labelled by $(C_k,G_k,W_k,Y_k) = (c,g,w, y)$ then the branch metric y_k **(Z,** ^**Z**$_j$) is given by:

$\gamma_k(Z_i \rightarrow Z_j) = \Lambda^{(i)}_{c,g}(C_k, G_k) + w\Lambda(W_k) + y\Lambda(Y_k)$ (1.15)

$\gamma_k(Z_i \rightarrow Z_j)$ = branch matric

$\Lambda(C) = \log[P(C = 1)/P(C = 0)]$

The forward recursion is represented by

$\alpha_{k+1}(Z_j) = \text{map*}\{ \alpha_k(Z_i) + \gamma_k(Z_i \rightarrow Z_j)\}$ (1.16)

The forward metrics are normalized regarding metrix stored in state zero after computing the forward recursion for all Zj at time k+1as follows

$\alpha_{k+1}(Z_j) = \alpha_{k+1}(Z_j) - \alpha_{k+1}(Z_0)$ (1.17)

Again, let f_{K+1}**(Z**j) indicates the normalized backward metric at trellis state k+1 and state**Z**j andf_k**(Z,)** denote the backward metric at trellis state *k* and state **Z,** prior to normalization. The backward recursion is given by:

$\beta_k(Z_i) = \text{map*}\{ \beta_{k+1}(Z_j) + \gamma_k(Z_i \rightarrow Z_j)\}$(1.18)

The backward metrics are normalized regarding metrix stored in state zero after computing the backward recursion for all Z, at time k as follows

$\beta_k(Z_i) = \beta_k(Z_i) - \beta_k(Z_0)$ (1.19)

The sets of forward and backward metrics are then stored and employ to get the LLR values accordingly. For each branch the likelihood ration be capable of calculation as follows

$H_k(Z_i \rightarrow Z_j) = \alpha_k(Z_i) + \gamma_k(Z_i \rightarrow Z_j) + \beta_{k+1}(Z_j)$ (1.20)

For information pair (C_k,G_k) = (c, g) the likelihood is calculated as:

$t_k(c,g) = \text{map}^* \{H_k\}$ (1.21)

$Z_i \rightarrow Z_j$: (c,g) And the possible values for *(c,g)* are 01, 10, or 11. At the decoder output, the LLR value is given by:

$\Lambda_{c,g}^{(o)}(C_k, G_k) = t_k(c,g) - t_k(0,0)$ (1.22)

$\Lambda_{c,g}^{(o)}(C_k, G_k)$= LLR at decoder output

After the iteration process is completed, either by fixed amount of iterations or stand on several convergence criterion, the LLR of each bit in the couple (C_k, G_k) is computed to take the final decision by comparing them to:

$\Lambda(C_k) = \text{map}^* \{\Lambda_{1,0}^{(0)}(C_k, G_k), \Lambda_{1,1}^{(0)}(C_k, G_k)\} - \text{map}^* \{\Lambda_{0,0}^{(0)}(C_k, G_k), \Lambda_{0,1}^{(0)}(C_k, G_k)\}$
(1.23)

$\Lambda(G_k) = \text{map}^* \{\Lambda_{0,1}^{(0)}(C_k, G_k), \Lambda_{1,1}^{(0)}(C_k, G_k)\} - \text{map}^* \{\Lambda_{0,0}^{(0)}(C_k, G_k), \Lambda_{1,0}^{(0)}(C_k, G_k)\}$
(1.24)

where$\Lambda_{0,0}^{(0)}$(C_k, G_k) = 0

when we choose larger interleaver, the code word sequence which is having information posses weight of two, so to avoid that type of scenario in coding like Turbo the role of interleaver is important because interleaver is responsible for bit error probability . This property of turbo codes capable of use to find the approximate value of the bit error probability at upper bound, which is accurate when long interleavers are employed. The bound estimation utilizes only those terms that have a foremost effect on the overall interpretation. Based on this a hasty technique to calculate the significant expressions of a turbo encoder transfer function are extended. The proposed process is applied with the concenated DSSS with punctured codes so it is more suitably called pseudo sequence punctured code. By this result become more accurate result. The systematic weight is given as:

$u(k,m) = u(m) = 2t_{1,m}$ (1.25)

and the parity check weight is gives as:

$h(k,m) = kh_{core}^{m+1} + 2t_{2,m}$ (1.26)

We get $h_{core}^{m+jL} = h_{core}^{m}$and $t_{2,m+kM} = t_{2,m}$,because puncturing pattern has periodic. Computation of*h(k,m)* and, consequently, *G(w* = 2, *U*, *H*), requires numerical calculation of the **L** values of h_{core}^{m+1}. However, the assumption of pseudo-random puncturing can further simplify the computation of h_{core}^{m+1}.

In order to express h_{core}^{m+1} in a close-form, we first need to consider the autocorrelation function 0(i) of a sequence of length L, which is defined as

$$\theta(i) = \begin{cases} 2^v - 1, & if\ i = 0 \\ -1, & if\ 1 \leq i < L \end{cases} \quad (1.27)$$

On expansion is becomes

$$\theta(i) = \sum_{j=1}^{L} (4x_j x_{j+i} - 2x_j - 2x_{j+i} + 1)$$

$$= 4\sum_{j=1}^{L} x_j x_{j+i} - 2\sum_{j=1}^{L} x_j - 2\sum_{j<=1}^{L} x_{j+i} + L \backslash$$

$$= 4\sum_{j=1}^{L} x_j x_{j+i} - 2(2^{v-1}) - 2(2^{v-1}) + (2^v - 1)$$

$$\backslash= 4\sum_{j=1}^{L} x_j x_{j+i} - 2^v - 1$$

If we update h_{core}^{m+1} accordingly, it becomes

$$h(k,m) = \begin{cases} k2^{v-1}, & if\ m = 1 \\ k2^{v-2} + 2t_{2,m}, & if\ 2 \leq m \leq 2^v - 1 \end{cases} \quad (1.28)$$

We observe that h*(k,m)* can assume three distinct values, namely $k2^{v-2}$, $k2^{v-2} + 2$, and $k2^{v-1}$, in order of magnitude, but only when *v*> 2. Invoking the properties of PN sequences, and also only 2^{V-1} — 1 out of the2^V — 1 elements $t_{2,m}$ of the puncturing vector T_z are equal to 0. If the memory size is *v* =2, we find that $t_{2,m}$assumes the zero value only once, and this happens when *m* = 1. Consequently, the value of*h(k, m)* for a memory size of 2 can either be 2*k* if *m* = 1, or *k* + 2 if *m* > 1.

The rate of a punctured RSC encoder depends upon the number of codeword bits, both systematic and parity check, transmitted during the puncturing period M. We know that in the case of pseudo-randomly punctured RSC codes, vector Tz = [$t_{2)i}$ ... $t_{2,M}$] contains 2^{v-i} non-zero elements, hence 2^{v-i} parity check bits evade puncturing and, consequently, at least 2^{V-i}codeword bits are transmitted for every $M = 2^V$ — 1 input information bits. The rate of the code can be reduced by increasing the numeral of non-zero elements in the vector T^ = [$t_{i\,;i}$ ... $t_{i\,,M}$], which determines which systematic bits are eliminated during the puncturing period M. The dominant conditional weight enumerating function. The dominant conditional weight enumerating function$B(w = 2,U,H)$ can be obtained as follows

$$G(w=2,U,H)=\sum_{k=1}^{\lfloor (N-1)/(2^v-1)\rfloor}\sum_{m=1}^{2^v-1} G_{k,m}U^{u(m)}H^{z(k,m)}$$

where$G_{k,m}$assumes the form

$$G_{k,m}=\begin{Bmatrix} \left\lfloor\frac{N}{2^v-1}\right\rfloor-k, & if\ rem(N,2^v-1)<m \\ \left\lfloor\frac{N}{2^v-1}\right\rfloor-k+1, & otherwise \end{Bmatrix}$$

(1.29)

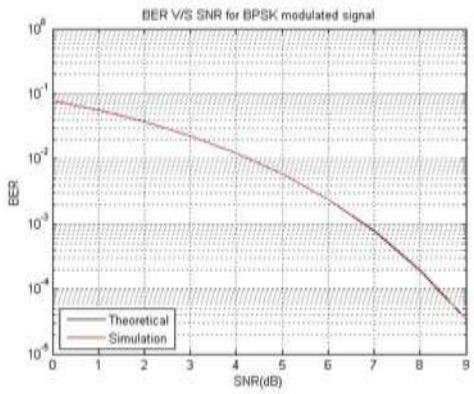

Fig 1.4 BPSK modulation in Channel

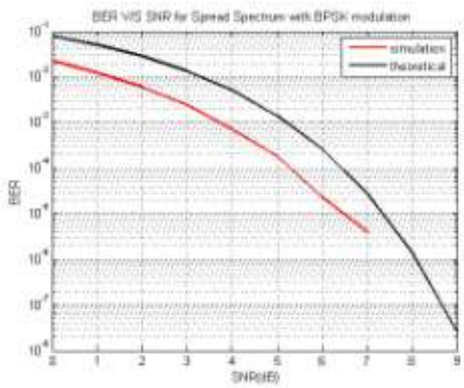

Fig. 1.5 DSSS performance with BPSK

Fig 1.4 shows BPSK modulation in AWGN channel and result is compared in terms of BER and SNR performance. Using this technique we can achieve only up to 10^{-1} dB and SNR has gone beyond the limit. Fig 1.5 shows that direct sequence spread spectrum is applied with binary phase shift keying as the result shows there is improvement in the result in terms of Bit error rate and Signal to noise ratio and there is much better improvement in SNR and it is approximate 7 dB.

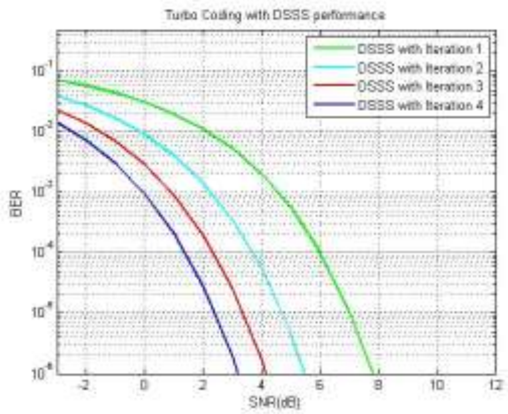

Fig 1.6 Turbo Coding Applied with BPSK

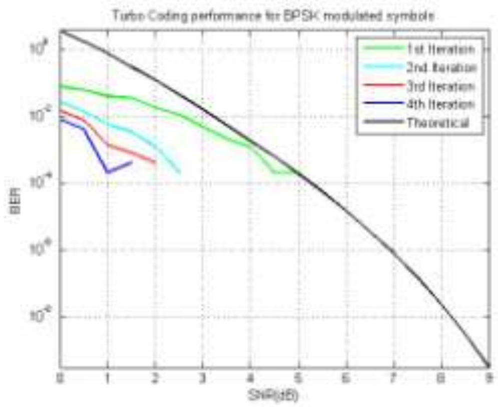

Fig 1.7 Combined performancewith Turbo Coded DSSS

Fig 1.6 shows turbo coding along with iteration is applied with Binary phase shift keying ,after each and every iterations there is an improvement in BER as well as SNR .Fig 1.7 shows that using the same modulation technique i.e. BPSK both the turbo coding and DSSS is applied, this scheme is shown in the form of graph at fig 1.7

IV. RESULT AND CONCLUSION

Analysis of different figures in this work the result is provided in table 1.1

Table 1.1

Proposed Technique with 4 iterations with code rate 1/3

SN	Proposed Technique	Modulation	Desired BER	Desired SNR(dB)	Channel
1	Turbo iteration 1with DSSS	BPSK Modulation	10^{-1}	8	AWGN
2	Turbo iteration 2 with DSSS	BPSK Modulation	10^{-2}	6	AWGN
3	Turbo iteration 3with DSSS	BPSK Modulation	10^{-2}	4	AWGN
4	Turbo iteration 4 with DSSS	BPSK Modulation	10^{-3}	3	AWGN

V.CONCLUSION

A new approach of turbo code along with DSSS is presented which overall provide good and secured communication among two stations. The modulation scheme is remain same for the above technique by using BPSK modulation scheme we reduce the complexity of system. USSS scheme is used instead of Frequency hopping scheme which result in enhancement of counter-jamming capabilities. The scenario given for this dissertation is that there is advanced knowledge of the communicating parties in the form of secret keys and after proper establishment of between two parties the further communication is possible. In this work we concentrated over satellite link between earth station and transponder. The main threats like Jamming, interference , spoofing and intrusion considered only and approach used here is focused to reduce MIJI attacks jointly and no single and individual method to reduce these threats are used in this dissertation. The counter measure technique at GNSS system requires new technology because threats are become advance according to present technique which more adversely affect the system. This dissertation has discussed many aspects of threat their analysis and mitigation technique separately. For this we analyze these threats from their origin and their originators. Although our result is only on analytical based and practically some of the parameters may vary.

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