2013

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Abstract

Forward Error Correction (FEC) codes used by receivers to correct transmission errors without retransmission add a considerable amount of redundant bits to data bits. The addition of redundant bits lowers the overall network throughput, thus increasing the demand for more required bandwidth. In this paper we investigate and discuss various techniques used in FEC and show their effects to data communication in terms of bandwidth utilization. Additionally we propose improvement of (2, 1, 2) Convolutional encoder to (3, 2, 3) encoder. The proposed improvements increase the code rate from 1/2 to 2/3 hence reducing error control information and increasing bit rate. The received codeword can be decoded by Soft-Output Viterbi Algorithm.

Keywords: FEC, Bandwidth, Convolutional Codes, Code Rate, Soft Output Viterbi Algorithm

1. Introduction

In the recent years wireless communication industry has experienced a drastic growth of traffic volumes on its networks. According to Cisco Visual Networking traffic update, the Global mobile data traffic grew by 70% in 2012(Cisco-Visual-Networking-Index, 2013). Global mobile data traffic reached 885 petabytes per month at the end of 2012, up from 520 petabytes per month at the end of 2011(Cisco-Visual-Networking-Index, 2013). The rapid traffic growth lies on large increase in the number of mobile devices users; the emerging of popular bandwidth-intensive applications such as rich media gaming, streaming media, video conferencing, mapping and navigation applications, telemedicine and virtue education applications as new services (Bazelon, Jackson, & McHenry, 2011). The invention of new devices such as smart phones, tablets and notebooks that use the aforementioned services contributes much to the situation (Hanzo et al., 2012). All of the above mentioned reasons require the availability of wireless networks that are capable to offer more bandwidth or higher data rate.

Bandwidth allocation in wireless communication involves the process of assigning radio frequencies to different applications. The radio frequency spectrum is a finite and limited natural resource which is increasingly in demand due to a large growing number of wireless telecommunication services such as mobile phone, radio and TV broadcasting, space research, environmental monitoring and other communication services that ensure wellbeing of life in the world.

Mobile-cellular penetration rates stand at 96% globally; 128% in developed countries; and 89% in developing Countries(ITU-ICT-Facts-and-Figures, 2013). In developing countries such as Tanzania 75% of people live in rural areas which are characterized by inadequate resources such as teachers and books in schools, doctors in health centers and other necessary facilities. The emergence of e-services such as e-education, virtual education, e-health, telemedicine and other e-business services brings up new hope to reach remote and marginalized communities in rural areas. However, most of the aforementioned services are bandwidth intensive and require high data rate networks to operate in wireless communication systems. Contrary to that, Tanzanian rural areas are characterized by limited ICT infrastructure that are dominated by low data rate such as GPRS, EDGE and VSAT network services which are also too expensive for the rural population.

Despite the high demand for bandwidth a considerable portion of it is used to transfer control information rather than actual or intended information. Communication systems are congested by various control information such as error control, security and application format which are system use information. Any attempt to reduce these control information while maintaining quality of service in data transmission favors the communication system in two ways. First, more bandwidth is offered for the intended data (End user data) and therefore increasing the capacity of the transmission network. Secondly, communication costs are reduced as end users are charged based on the amount of bandwidth used during transmission of data.

This work proposes a new FEC Convolutional code model that reduces system error control information in wireless communication systems. Thus, offering more room for actual user information to be delivered at the same cost.

Section 2 of this paper presents the state of art of selected FEC algorithms with their bandwidth requirements. Section 3 presents the proposed model of improved FEC Convolutional code and section 4 concludes this work.

2. State Of The Art

There are many types of FEC codes in use today; however, FEC codes are basically divided into two main categories which are Block codes and Convolutional codes (Liumeng, 2011; Sijia & Zexi, 2011). Block codes work on fixed-size blocks (packets) of bits or symbols of predetermined size. The output of block codes only depends on the immediate inputs to the same block; while Convolutional codes work for bit or symbol streams transmission with outputs that depends not only on the immediate input but also the previous inputs.

Convolution codes (CC) are one of the popular FEC codes in use today. This type of code was first introduced by Elias in 1955(Morelos-Zaragoza, 2006). Through the age, FEC Convolutional code has found numerous applications in wireless communication including digital terrestrial, satellite communication and broadcasting systems and space communication systems (IMT-2000, GSM, IS-95,WiMAX (802.16), WCDMA, GPRS, UMTS, CDMA 2000, DVB-T and DVB-S) (Morelos-Zaragoza, 2006). The most popular decoding mechanism of CC is the Viterbi algorithm introduced by Viterbi in 1967(Viterbi, 1967). CCs have memory and are defined as *(n, k, m)* codes, where *n* is the number of output bits, *k* is the number of input bits at a time, and *m* is the memory length of encoder(Liumeng, 2011). If an encoder receives *k* bits of information as input at a time and gives out *n* bits as output (codeword) at a time, then *k/n* is a code rate of that particular encoder. The common code rate for this codes are 1/2,1/3, 1/4 (Johannesson & Stahl, 1999) and 4/8(Johannesson, Stahl, & Wittenmark, 2000).

To reduce the number of bits transmitted, various puncturing schemes are used which discard selected bits from the encoder output. Typical puncture rates are 2/3, 3/4, 5/6, and 7/8. For example, a 3/4 puncture rate means that for every 3 input bits, 4 output bits are transmitted from the encoder output rather than the 6 bits that are actually generated. Puncturing can be implemented using external logic to the Convolutional encoder(Francis & Green, 2007). This allows the freedom to change between the various puncture rates but it increases data processing load. Higher gains of CCs are also achieved by Convolutional Turbo Code (CTC) and concatenation of CCs with other schemes (Benedetto, Garello, & Montorsi, 1998). A good example of this is DVB-T where Convolutional code is concatenated with Reed Solomon as inner and outer code(Francis & Green, 2007).

According to ViaSat (ViaSat, 2013), CTCs perform extremely well at lower code rates. Communication systems with less bandwidth and power constraints can use low rate concatenated code to achieve high coding gains or very low error rates. However, tight bandwidth situations and power constraints communication system drive this solution in opposite direction. Thus, demanding for an interleaver to be integrated into FEC systems to enhance burst error dispersion as used in most of the contemporary communication systems (Francis & Green, 2007; ViaSat, 2013). More efforts are still needed to improve FEC codes to support communication systems to meet modern applications bandwidth requirements.

Below are few selected FEC block codes that are used in serial or parallel concatenation with Convolutional Codes. This selection includes Reed-Solomon (R-S) codes, Bose-Chaudhuri-Hochquenghem (BCH), Low Density Parity Codes (LDPC) and Turbo Product Codes (TPC). In communication systems the aforementioned codes are sometimes concatenated in either serial or parallel form to give a desired exceptional performance. In serial arrangement there is inner and outer channel code. This fact makes the situation even worse as the extra bandwidth required for the communication system is a total of all redundant bits in inner and outer code.

2.1 Reed Solomon (R-S)

R-S is a class of non-binary cyclic block codes defined by *(n, k)* code; where *n* is the output block size and *k* is the number of information symbols and *(n-k)* gives a number of redundant information. This type of code is used in communication systems as component code for building more powerful codes through concatenation such as Reed Solomon Turbo Product Code (RS-TPC). R-S codes type has its application in WiMAX (802.16), DVB-T, DVB-S, DVB-C, ISDB, DMB, DOCSIS and GPON.

2.2 Bose Chaudhuri Hacquenghem (BCH)

BCH is a class of cyclic block codes; it is used as a component for building more powerful codes. BCH encodes

k data bits into n code bits by adding n-k parity checking bits for the purpose of detecting and checking the errors. Given the length of the codes $n=2m-1$ for any integer $m \geqslant 3$, we have t (where $t<2m-1$), as the bound of the error correction. The number of parity checking bits is n-$k \leqslant mt$. Its applications are found in Video Broadcasting, DVB-S2 and memory controllers.

2.3 Low Density Parity Codes (LDPC)

LDPC is a FEC linear block code defined by a very sparse parity check matrices which allow for parallel iterative decoder. LDPC works with a code rate from 0.67 to 0.92 (Gho & Kahn, 2012), Applications of LDPC are also in China Multimedia Mobile Broadcasting (CMMB), IEEE 802.11n-2009 (Wi-Fi standard), WiMAX, VSAT, DVB-T2, DVB-C and DVB-S2

2.4 Turbo Product Codes (TPC)

TPC is a class of high-performance FEC codes developed in 1993, TPC are finding use in 3G mobile communications and (deep space) satellite communications as well as other applications where designers seek to achieve reliable information transfer over bandwidth or latency-constrained communication links in the presence of data-corrupting noise(ViaSat, 2013). Code rates for selected FECs are presented in table 1.

Table 1: Selected FEC Codes with Their Corresponding Code Rates

S/N	Code Type	Supported Code Rate
1	LPDC	$\leqslant$ 1
2	TPC	$\leqslant$ 0.93
3	CTC	$\leqslant$ 0.83333
4	R-S	$\leqslant$ 0.9375
5	BCH	t<=12 ; t=1,2, (3)
6	CC	$\leqslant$ 0.5

In the next section, we present the proposed model of FEC convolution code to be used in application layer for encoding and decoding data streams.

3. System Model

The model improves the *(2, 1, 2)* Convolutional encoder from four states (i.e. S_0, S_1 S_2, and S_3) to eight states (i.e. S_0, S_1, S_2, S_3, S_4, S_5, S_6, and S_7). This improvement results into a two level encoder. The inner level states which include the first four states (i.e. S_0, S_1 S_2, and S_3) and the outer level states that include the rest of the states (i.e. S_4, S_5 S_6, and S_7). We assume that:

- The encoding process starts and ends at state S_0.
- The implementation of the proposed code is in the application layer to support specific real time applications.

At each state, there are four possible ways to move to other states, where two ways allow the encoder to move to other two states of the same level and the other two ways to another level. All the same there are four possible ways to enter a state from other states, where two ways come from states of the same level and another two ways enter a state from states of another level as shown in the fig. 1.

3.1 Encoding

The encoding process in the fig. 1 starts and terminates at state S_0. By following the arrows leaving a state, the *input/output* labels (e.g. *01/011* representing *01* as input bits and *011* as the codeword associated to it) gives guidance through the encoding process. We easily follow the arrows from state to another while taking care of the input and codeword generated. Suppose the bits *{10-11-10-00-01}* are supplied as a string of input to the encoder (starting with *10*), then the encoder will produce the following string as codeword at the output *{100-111-110-011-011}* and the encoder will change states from S_0, S_4, S_5, S_6, S_0, *to* S_1.

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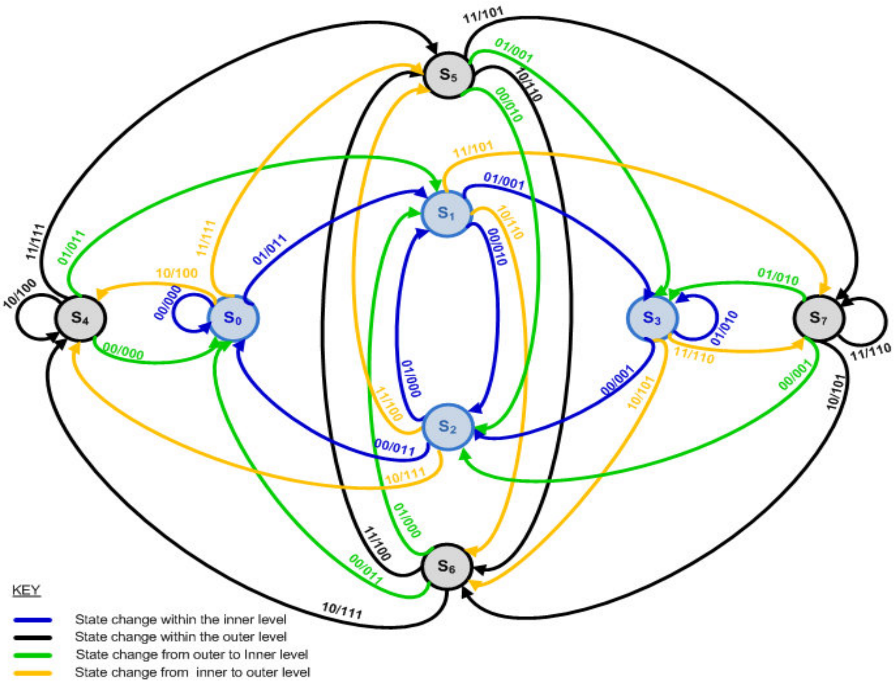

Figure 1: Proposed model of (3, 2, 3) convolution encoder

3.2 Decoding

The Soft-Decision Output Viterbi algorithm (SOVA) proposed by Viterbi (Viterbi, 1967) and improved by Joachim Hagenauer and Peter Hoeher (Hagenauer & Hoeher, 1989) is used to decode the received bits. SOVA decodes received data against random errors by comparing the received code sequence with every possible code sequence of a node or state. A bit by bit comparison yield a branch metric also known as hamming distance which is used to determine the most likely path of correct data.

Trellis diagram presented in fig. 2 contains information of states and uses time as a horizontal axis to show the possible paths through the states. The decoding process starts at S_0 where four possible branches with their possible codes can be compared with the received codeword at the top. The path with minimum comparison metrics is the most likely path with correct data. The comparison metrics are calculated cumulatively as you move along from time zero onwards. Viterbi realized that at any stage when different paths converge into one state, then only those paths with smaller hamming metric need to be remembered and not all paths. Therefore, the surviving path with minimum hamming distance metrics is the winner path.

For example, the trellis diagram in fig. 2 shows how eight states are used to decode the received codewords stream *{100-111-110-011-011}* to give back the original data stream *{*i.e. *10-11-10-00-01}*.

Through the red colored path in the trellis diagram which shows how the decoding process moves through S_0, S_4, S_5, S_6, S_0, to S_1.

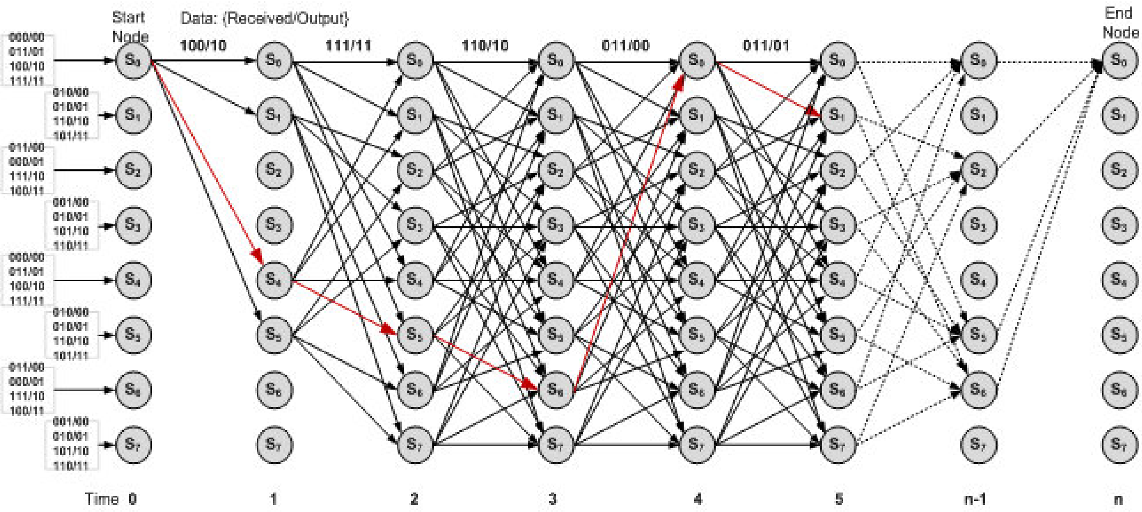

Figure 2: Trellis diagram to show the path

4. Conclusion

We have discussed the selected FECs, their applications and respective code rates. The improved FEC Convolutional Code model from (2, 1, 2) to (3, 2, 3) is also presented and enables the algorithm to improve the code rate by 0.17 from 1/2 to 2/3 and hence reducing error control information in transmission media. The paper shows clearly that the improved FEC Convolutional Code can significantly improve performance i.e data throughput of wireless communication systems in comparison with the rate of 1/2 convolutional coding case. Hence, the reduction of operational costs. The decoding process that uses the normal SOVA has also been presented.

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