Improving the Performance of Mobiling Communication Using Convolutional Coding with Pilot Bits

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Abstract:

The major impairments of the mobile wireless communication signal are Fading and Doppler effect. To reduce these effects, channel coding is required which provides for forward error correction. This enables the receiver to correct the errors which have occurred in the transmission path. Also, a control information consists of known pilot bits may be added to support channel estimation for coherent detection.

In this paper the Bit Error Rate (BER) performances is shown by means of simulation for Rayleigh fading channels with Doppler shift and Additive White Gaussian Noise (AWGN) is added. Here the convolutional code with ½ code rate is used as a channel coding technique, and sequence of pilot bits is mapped and inserted with known modulation scheme Quadrature Amplitude Modulation (QAM), these pilot bits are know to the receiver. At the receiving end coherent detection is done by channel estimating using the splited pilot bits, and then Viterbi decoding is investigated. Obviously, the results show that the performance in terms of BER of coded signal is much better when inserting pilot bits during signal modulation. To compensate the loss in spectrum efficiency and data rate, the use of QAM modulations with larger size constellation is suggested, with small loss of performance associated with an extended constellation. The whole work is done using MATLAB.
In recent years, the demand for multimedia applications over wireless mobile communication systems has grown very rapidly. Wireless mobile communication systems present several design challenges resulting from the mobility of users throughout the system and the time-varying channel. Also, there has been an increasing demand for efficient and reliable digital communication systems. The major concern of design engineers is to minimize the received error probability by making efficient use of power and bandwidth resources, while keeping system complexity reasonable in order to reduce cost and processing time [1].

Communication engineering resulted from the necessity that the communication systems must be robust in order to transmit the information from the source to the destination via a communication channel with high reliability. This in fact is a difficult task since in the physical world there exist many factors to cause errors. Detecting and correcting errors is very important while transmitting information. If errors occur and it cannot be detected and corrected, the received information at the remote site will be of little use, as it will differ considerably from the original source information. These considerable imperfections in the information sequence, caused by the channel are what the communication engineers try to minimize [1].

In digital mobile communications, fast fading introduces an error floor, degrades Bit Error Rate (BER) and inhibits the use of coherent detection. To mitigate these effects, the use of error control coding has been suggested in this paper. However, to realise the full potential benefits of coding, coherent detection is essential. To this end, the use of a tone to provide amplitude and phase reference has been proposed by several authors. Recently, an alternative to the pilot tone technique has emerged, where the transmitter periodically inserts known symbols. The insertion of pilot samples causes a lowering of the data rate and some power loss, which is the price paid for quasicoherent detection. The some suggestions of References [2] and [3] have been supplanted by the general analysis in Reference [4]. Just as with pilot tone schemes, it has been demonstrated that such a system improves the receiver performance over noncoherent detection and removes error floors [4].

Previous studies of Pilot Symbol Assist Modulation PSAM, were based on simulation and experimental implementations. Although they demonstrated feasibility, they did not provide the performance analysis needed before their results can be generalized [5,6].

The present paper supplies the missing analysis. It includes the BER for different Quadrature Amplitude Modulation (QAM), as well as expressions for the optimum interpolator at the receiver. Also an extension of this scheme to fully convolutional coding is presented and analysed. Exact error probability are derived. Although, for simplicity, Rayleigh channel is used, the analysis can be easily extended to the Rician fading model.

Convolutional Code:
Convolutional codes are the most important types of channel coding. They are widely used in digital communication systems. Convolutional coding (CC) were introduced in 1955 by Elias [7]. He showed that redundancy can be introduced into a data stream through the use of a Linear Shift Register (LSR) and a network of Exclusive OR gates (XOR) [8]. The initial development of convolutional code was dated back to the early 1970s and becomes practical and widely accepted after the invention of Viterbi algorithm for iterative decoding of convolutional code [9]. Convolutional encoders are linear and time-invariant system given by the convolution of a binary data stream with generator sequences. They can be implemented by shift registers. It can be argued that convolutional coding can achieve a larger coding gain than can be achieved using a block coding with the same complexity [10,11].

The most important reason for using convolutional coding in this paper is that the channel is a fading channel [12].

1- Encoding Process

Using the relatively simple convolutional encoder, with one input, two outputs, and two memories, shown in Fig(1) for each individual bit fed into the shift register, two bits are generated as output symbols [13].

![Convolutional Encoder Diagram](image)

A convolutional encoder is generally characterized in \((n, k, K)\) format, where:
- \(n\) is number of output line of the encoder.
- \(k\) is number of input line to the encoder.
- \(K\) is constraint length, \(K = (s + 1)k\), where \(s\) is the number of memory elements (Flip-Flops).

The Convolutional Code Rate (CCR) of a \((n, k, K)\) encoder is \(CR = k/n\), and possible state = \(2^{sK}\). The encoder shown in Fig(1) is a \((2, 1, 3)\), 1/2 code rate, and 4 possible states [9]. The coding rate is the major factor influencing error correction, for 1/3 system, one more XOR block produces one more output for every input bit.

Although any coding rate is possible, rate 1/n systems are most widely used due to efficiency of the decoding process. The output bit combination is described by a polynomial. It is characterized by the number and positioning of the taps at the shift register, which are indicated by, generator polynomials \(g\), the coefficients of which are 0 or 1, according to whether there was a tap at the respective position or not. It is common practice to combine the coefficients as octal numbers. Polynomials or octal numbers must be indicated individually for each output branch. The example of Fig(1), uses the polynomials:

\[
g_1(x) = 1 + x^2 \quad (5_{octal})
\]
\[
g_2(x) = 1 + x + x^2 \quad (7_{octal})
\]
which can formed as a matrix \((k \times n)\)
\[
g_n(x) = [1 + x^2, 1 + x + x^2]
\] 
............... (3)

For example, a convolutional encoder shown in Fig(1), it is conventional to assume that the memory elements are initialized with all zeros at the beginning of transmission.

The input stream can be represented as \(m_s(x) = 1 + x + x^4 + x^6 \in GF(2)\). The outputs are [8] :
\[
c_1(x) = m_s(x)g_1(x) = (1 + x + x^4 + x^6)(1 + x^2) = 1 + x + x^2 + x^3 + x^4 + x^8
\]
\[
c_2(x) = m_s(x)g_2(x) = (1 + x + x^4 + x^6)(1 + x + x^3) = 1 + x^3 + x^4 + x^5 + x^7 + x^8
\]

which corresponding to:
- \(m_s = [1, 1, 0, 0, 1, 0, 1]\), the outputs are:
- \(c_1 = [1, 1, 1, 1, 1, 0, 0, 0, 1]\), \(c_2 = [1, 0, 0, 1, 1, 1, 0, 1, 1]\),

and the interleaved stream is:
\(c = [11, 10, 11, 11, 01, 00, 01, 11]\)

2- State and Trellis Diagrams of the Model Encoder

A convolutional encoder is a state machine. For both encoding and decoding purposes. This can usually be represented by a state diagram as shown in Fig(2), considering again the convolutional encoder of Fig(1) [13].

![Fig(2) State diagram of (2, 1, 3) convolutional encoder](image)

Another way to document the same combination is the trellis diagram. Here the states are plotted below one another and time-sequentially next to each other as shown in Fig(3).
The sequence of states through such trellis for the encoder of Fig(1), with input sequence [1, 1, 0, 0, 1, 0, 1], and [11, 10, 11, 11, 01, 00, 11] output (for the example of item 1), is shown in Fig(4), the solid line shows the state sequence, input and output is labeled on each branch [13].

![Fig(4) The sequence of states through encoder trellis.](image)

### 3- Decoding Process

To minimize the probability of word error for convolutional codes the Viterbi algorithm is used [12]. It was discovered and analyzed by Andrew J. Viterbi in 1967 (hence the name "Viterbi"). It is a maximum-likelihood decoding procedure based on finding the trellis path with the smallest distance between its digit sequence and the received sequence [14].

The advantage of Viterbi decoding is that the complexity is not a function of the number of symbols in the codeword sequence. The algorithm involves calculating a measure of similarity, or distance, between the received signal, at a time \( t' \), and all the trellis path entering each state at a time \( t' \). The Viterbi algorithm removes from consideration of those trellis paths that could not possibly be candidate for maximum likelihood choice. When two paths enter the same state, the one having the best metric is chosen, this path is called the surviving path. This selection of surviving paths is performed for all the states. The decoder continues in this way to advance deeper into the trellis, making decisions by eliminating the least likely paths. The early rejection of the unlikely paths reduces the decoding complexity [15].

### The Doppler effect:

When there is relative movement between the transmitter and receiver, the carrier frequency, as perceived by the receiver, gets changed by some amount; this is known as the Doppler effect. The amount of frequency shift depends on the relative speed, the direction of movement and the frequency of the carrier. In mathematical terms [16]

\[
f_{\Delta} = \frac{v}{\lambda} \times \cos \theta
\]

where:
- \( v \) : the relative speed between the transmitter and receiver,
- \( \theta \) : the angle between the direction of motion and the wave propagation,
- \( \lambda \) : the carrier wave-length.
A Doppler shift can be negative as well as positive, meaning an apparent decrease or increase in frequency, respectively. However, most often the maximum absolute value is considered and normalized with respect to the symbol rate and denoted by $F_d T$:

$$F_d T = \frac{|Fr_d|}{f_{symbol}} \quad \text{............. (9)}$$

where $f_{symbol}$ is the symbol rate. A typical office environment has $F_d T$ value of about $2^{-3}$. Another parameter, often used to characterize the time varying nature of the channels is the coherence time which is related to the Doppler shift by [10]:

$$T_c \approx \frac{9}{16 m f_m} \quad \text{............. (10)}$$

where $f_m$ is the maximum Doppler shift given by $f_m = v / \lambda$.

**Fading Channel:**

Every wireless system has to combat transmission and propagation effects that are substantially more hostile than wired systems. In mobile radio communication, the signal offered to the receiver contains not only a direct line-of-sight radio wave, but also a large number of reflected radio waves. Even worse in urban areas, the line-of-sight is often blocked by obstacles, and collections of differently delayed waves together are received by a mobile antenna.

The result of multipath and the Doppler shift is fading. Fading is the rapid variation in signal strength over a short distance or time interval where the large scale attenuation is constant. Fading can be flat or frequency selective depending on the multipath structure of the channel, and slow or fast depending on the Doppler effect. Flat fading occurs when the bandwidth of the signal is less than the coherence bandwidth. This type of fading is common, and some communication systems are designed specifically to operate in very narrow bandwidth mode. If the signal bandwidth is wider than the coherence bandwidth then different frequencies suffer independent fading and the result is Inter-Symbol-Interference (ISI) [14]. In this paper flat fading is used for simplicity.

**QAM Modulation:**

Digital modulation involves converting a set of bits into a complex number (or a data symbol) representing a constellation point of the corresponding mapping scheme[17]. QAM is one of widely used modulation techniques because of its efficiency in power and bandwidth. In QAM system, two amplitude-modulated (AM) signals are combined into a single channel, thereby doubling the effective bandwidth. However, it must also be noted that when using a modulation technique such as 64-QAM, better signal-to-noise ratios (SNRs) are needed to overcome any interference and maintain a certain bit error ratio (BER).

In this paper the bits are mapped into symbols. The order of the modulation will depend on the subcarrier. A subcarrier with high SNR will be assigned more bits than a subchannel with low SNR. The bits will be mapped using MQAM. In Fig(5) a QAM, 4-QAM, and a 16-QAM constellation is sketched. One or more of 16 symbols may additionally be used as a pilot symbol, and/or may be used as a synchronization symbol [3].
**Simulation and Results:**

In the proposed system which is shown in Fig(6), the source data is first coded using convolutional Encoder with code rate $\frac{1}{2}$, constraint length is 3 and generator polynomials are (5,7), then the coded data modulated using QAM constellation modulator and the transmitter inserts a known pilot bits at the beginning of each coded symbol frame. These pilot bits must be known by the receiver.

This data transmitted through Rayleigh fading channels with Doppler shift for mobility and then Additive White Gaussian Noise (AWGN) is added to the signal that passes through the channel.

At the receiving end the received signal is split into pilot and data signals using pilot extractor. The received pilot signal from pilot extractor is used with the known pilot for channel estimation to qualify the received data signal using data compensator, then this signal is demodulated and finally a hard-decision is used to detect the data using Viterbi decoder. In this simulation, the data sink simply counts the number of errors that occurred to gather statistics used for investigating the performance of the system (Bit error probability).
To calculate Bit Error Rate BER performance of the proposed system, the number of data used in this simulation is 5000 bits, and the number of decode iterations was set to 5. The average BER are obtained by varying the values of SNR in the range of 0 to 30 dB.

In this paper three cases of study are obtained:

**Case 1:** In the first case, the effect of convolutional code and pilot bits insertion on the proposed system is studied when the system is stationary, this is shown in Fig(7). This figure illustrates the BER performance of four different transmission and reception techniques which are shown in figure legend. It can be noticed that the case when using both CC and pilot bits has the best BER performance in comparison with the other cases.

**Case 2:** In the second case, the effect of using convolution coding with and without pilot bits insertion on the proposed system is studied when the system is in moving state, as shown in Fig(8). The figure shows clearly that the BER curve is more sharper when using pilot bits with coding than curve with pilot but without coding, with SNR gain about 3.6 dB at BER $10^{-4}$.

While in case of [18] a turbo code is used, (a turbo code is formed from the parallel concatenation of two codes separated by an interleaver), the results of the simulation test shows that the improvement in error performance at very low error rates is about 4dB, which means that this result is very near to the result in this paper.
Also, the figure shows that the effect when pilot bits are not used with mobiling system, where the two curves in this case shows high BER, which makes the system unusable.

Fig(8) Comparison of BER versus SNR for different simulations over Rayleigh flat fading channel with AWGN for Mobiling case.

**Case 3** : In order to compensat the loss in spectrum which stem from high redundancy of $\frac{1}{2}$ code rate of convolutional coding in the previous case. High level of constellation are used here, and Fig(9) illustrates the performace of this system for PAM, QAM, and 16QAM.
It is clear that the use of 16QAM constellation can compensate the loss in data rate when using convolutional code with code rate of $\frac{1}{2}$ and and pilot insertion, with small reduce in BER performance, which indicate that this construction is useful for modern mobile devices with high data rate.

**Conclusion:**

This paper shows, and through the three cases of study, the benefit of using pilot bits insertion during signal modulation step besides convolutional coding for improving the BER performance of mobile communication by making use of these pilot bits to estimate the channel variations to compensate the received signal. The SNR gain about 3.6 dB at BER $10^{-4}$. Also, it is clear that this system is useless when pilot is not used for fading conditions. Finally, the effect of using higher QAM constellation is shown to recover the lack in BER when using pilot bits.

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