Performance Evaluation of Convolution Coded SCM with different Code Rates

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***Abstract* —A technique for signal code modulation (SCM) is given here with convolution coding with different code rates. Signal Code Modulation eliminates the inherent quantization noise component in digital communications, instead of conventionally making it minimal. In the SCM the primary analog signal is represented by a digital component (i.e. quantized) and an analog component consisting of quantization error. The combination of both these components is transmitted and used to reconstruct the original signal at the receiver. The SNR gain is achieved by digital component and almost an error free communication is possible by employing coding while system performance is improved by analog component when excess channel SNR is available. By increasing redundancy, BER is reduced appreciably or conversely lower E_b/N_o is required for an acceptable BER.**

***Index Terms* – Additive white Gaussian noise (AWGN), bit error rate (BER), quadrature amplitude modulation (QAM), signal code modulation (SCM), and symbol error rate (SER), demodulation-remodulation (Demod-Remod).**

I. INTRODUCTION

Traditionally the analog information is transmitted over a bandlimited AWGN channel using analog transmission, or sampling, quantizing, coding and using digital transmission. Analog modulation techniques (FM or PM) provide SNR improvement proportional to square root of modulation index, and hence are able to trade off bandwidth for SNR. Still, there is a significant gap between best achievable improvements as per Shannon's capacity theorem [4], which can be provided by the Digital modulation techniques using error correction codes. However, quantization error left while digitizing the analog signal can't be later recovered. This quantization distortion introduced at an early stage will be present regardless the transmission quality of the communication channel. In an alternative technique called SCM [1], the transmission of this quantization error and subsequent addition of it to digital component received, produces an exact representation of the original signal.

In this paper, we study the SCM which uses a combination of analog and digital modulation, and enjoys the advantages of both. This technique is based on the theme of representing the analog signal by a digital (i.e. quantized) component, and an analog component consisting of the quantization error. The combination of both the components provides an exact representation of the original signal. Both components are transmitted by the communication system and used to reconstruct the signal at receiver. The presence of analog residual improves system performance when excess channel SNR is available. The digital component makes it possible to employ coding to achieve near error free transmission as it provides an SNR gain.

The SCM technique may be employed in repeater application between two links with different SNRs, where first link has the high SNR characteristics, while the second one has lower SNR characteristics.

The SCM has the ability to vary the data rate i.e. order of QAM according to the available SNR, it can be used for retransmitting a digital communication signal over channels with different signal-to-noise ratios, without complete demodulation and remodulation (Demod-Remod) [2]. The SCM is also able to trade off bandwidth for SNR and gives a performance close to that of conventional Demod-Remod method.

II. SIGNAL CODE MODULATION

Signal Code Modulation is a mixed analog-digital technique for transmitting analog information over a noisy channel. SCM provides an analog pipe through which any bandlimited signal can pass, including truly analog information or the output of a digital modem. The operations that SCM performs on the input analog signal are simple, as illustrated in Figure 1. The waveform is sampled and quantized, just like a typical pulse code modulation (PCM) transmission, and the digital signal (i.e. quantized part $q_{i(n)}$) is then transmitted over the noisy channel using any digital technique, such as quadrature amplitude modulation (QAM). The digital signals are denoted by the symbol *D*. However, unlike PCM, SCM does not discard the quantization error i.e. $x_a(n)$. This error signal is extracted, amplified and then transmitted over the noisy channel as an analog symbol, *A*.

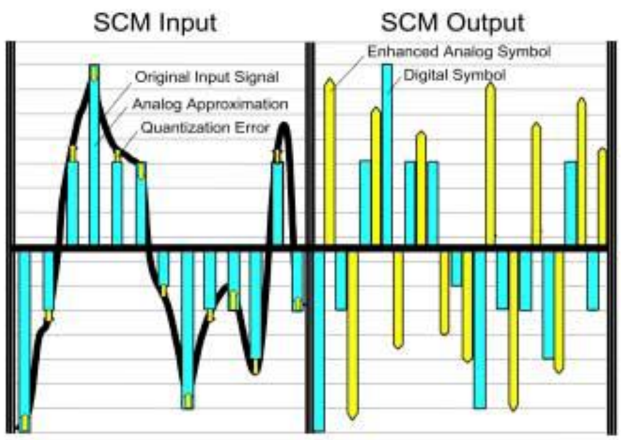

Figure 1: SCM operation on an input analog waveform.

$$x(n) = q_{i(n)} + x_a(n) \quad \ldots. (1)$$

where $x(n)$ is sampled analog signal appearing at the output of first link with higher order QAM signal, $q_{i(n)}$ is quantized part and $x_a(n)$is analog residual i.e. quantization error. The $\alpha\, q_{i(n)}$ represents D symbol, while $g_a^2\, x_a(n)$ is A symbol. Here α is scale factor, g_a^2 is gain factor which provides noise immunity.

The SCM transmission and reception processes are depicted in Figure 2. The transmission channel is divided into two time division multiplexed channels. Channel 1 is analog, and channel 2 is digital. In a process essentially identical to PCM, the original analog signal at the system input is sampled at the appropriate rate, based on the sampling theorem, and converted to digital values. The resulting *D* symbols are transmitted via channel 2 using a digital transmission technique optimized for the channel. Those *D* symbols represent *N* bits per analog input sample. To produce the quantization error *A*, the PCM data is converted back to analog and subtracted from the original input. This *A* symbol is amplified by a gain of 2^N or any gain that will optimize the voltage swing of the *A* symbol with that of channel 1.

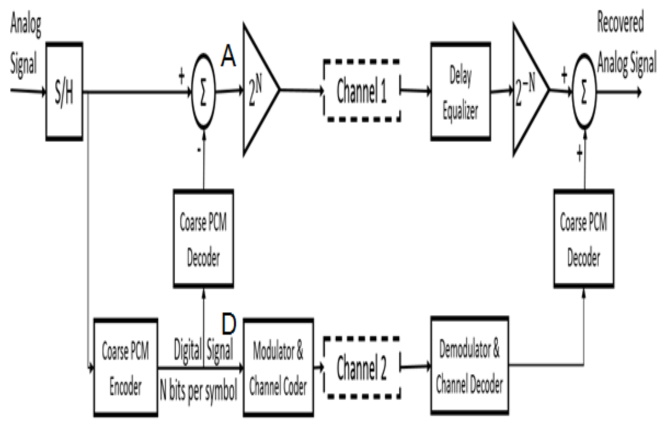

Figure 2: Conceptual block diagram of the SCM system.

The SCM receiver performs the opposite operation, combining the *A* and *D* symbols results into an exact representation i.e. analog stream replica of the original analog signal. This replica is not a precise copy of the original signal; because noise in the channels could vary the *A* symbols or cause bit errors in the *D* symbols. However, the 2^N amplitude gain in channel 1 has provided noise power immunity of 2^{2N}, as an attenuation is given to the received *A* symbols. This is one of the key benefits of SCM.

The modulation technique described above may be called as SCM-AD, where AD indicates that each input sample is converted into one analog and one digital symbol. There are other variations of the SCM system which use different combinations of A and D symbols e.g. SCM-ADD, SCM-ADDD, SCM-AAD etc. to balance the digital and analog gains. The SNR required to transmit 16-QAM is 12 dB lower than that required for transmitting 256-QAM (for a given BER 10^{-4}). This difference is called the digital gain of the SCM. The analog gain is provided by the amplification factor (i.e. gain factor g_a^2) by which the analog residual i.e. quantization error is amplified and it comes 10.9 dB for $g_a = 3.51$ [2]. Here both these gains are roughly same.

Whenever the signal is to be transmitted through two links where digital and analog gains differ significantly, the SCM-ADD/SCM-ADDD/SCM-AAD etc. an appropriate technique may be used to balance both the gains.

III. EVALUATION OF CODING SCM

Error performance analysis is performed by plotting the bit error-rate averaged over a large number of times versus E_b/N_o for AWGN channel. These graphs are drawn using Matlab simulation results. Simulations were run for direct transmission from output of first link to second link using similar order QAM as on first link, Demod-Remod approach, SCM applied at output of first link without any coding for digital symbols, convolution coding with code rate 3/4, with code rate 2/3 and also for Hamming coded SCM [6].

The following graphs indicates superiority of the corresponding coding scheme with respect to quality in terms of Bit Error Rate at fixed transmitted power i.e. E_b/N_o or in terms of required transmitted power i.e. E_b/N_o at a fixed BER.

a) BERs for SCM coded with code rate ¾ and others

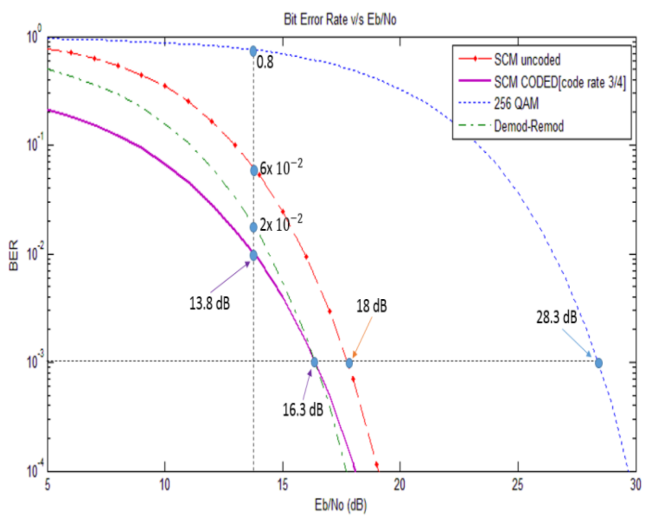

Figure 3: SCM: uncoded, convolution coded with rate 3/4, Demod-Remod and 256-QAM direct

Figure 3 shows the simulation result which reveals that at fixed E_b/N_o (13.8 dB) for conventional Demod-Remod approach the BER is 2×10^{-2} and for uncoded SCM it is 6×10^{-2} and for convolution coding with code rate 3/4, the BER is 1×10^{-2} .

Conversely, to achieve BER 0.001, the required transmitted power i.e. E_b/N_o is 18.0 dB (for uncoded SCM), and 16.3 dB (for coded with code rate ¾ and traditional Demod-Remod approach) which is 1.7 dB lower than uncoded SCM, a fair power saving.

b) BERs for SCM coded with code rate 2/3 and others

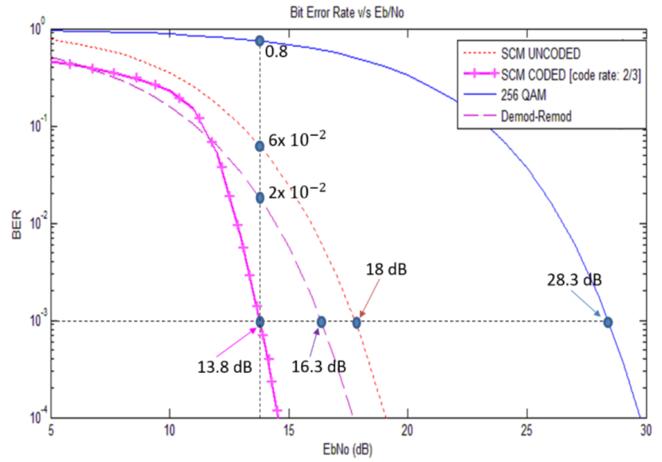

Figure 4: SCM: uncoded, convolution coded with rate 2/3, Demod-Remod and 256-QAM direct

Figure 4 shows the simulation result which reveals that at fixed E_b/N_o (13.8 dB) for conventional Demod-Remod approach the BER is 2×10^{-2} and for uncoded SCM it is 6×10^{-2} and for convolution coding with code rate 2/3, the BER is 1×10^{-3}, i.e. 20 times better than Demod-Remod approach and 60 times better than that of uncoded SCM.
Conversely, to achieve BER 0.001, the required E_b/N_o is 18 dB (for uncoded SCM), 16.3 dB (for Demod-Remod) and 13.8 dB (for coded with rate 2/3) which is 4.2 dB lower than uncoded SCM and 2.5 dB lower than for Demod-Remod.

c) BERs for SCM coded with code rate ¾ , with code rate 2/3 and others

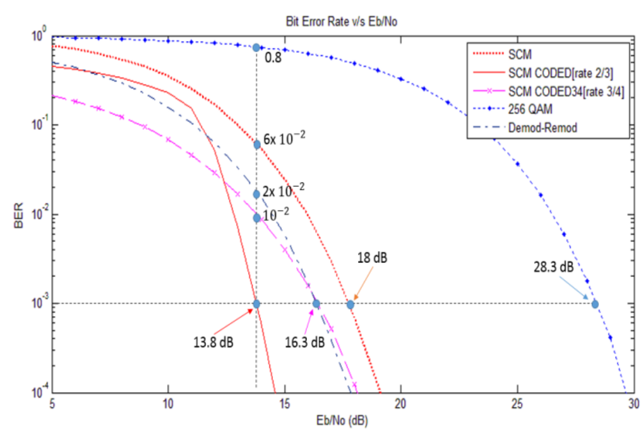

Figure 5: SCM (uncoded, convolution coded with rate 2/3 & 3/4), Demod/Remod and 256-QAM direct

Figure 5 shows the simulation result which reveals that at fixed E_b/N_o (13.8 dB) for conventional Demod-Remod approach the BER is 2×10^{-2}, for uncoded SCM it is 6×10^{-2} for convolution coding with code rate 3/4, the BER is 1×10^{-2} and for convolution coding with code rate 2/3, the BER is 1×10^{-3}, which is 20 times better than Demod-Remod approach, 60 times better than that of uncoded SCM and 10 times better than that of SCM coded with rate 3/4.

Conversely, to achieve BER 0.001, the required E_b/N_o is 18 dB (for uncoded SCM), 16.3 dB (for Demod-Remod and SCM coded with rate 3/4) and 13.8 dB (for coded with rate 2/3) which is 4.2 dB lower than uncoded SCM and 2.5 dB lower than for Demod-Remod and SCM coded with rate 3/4.

d) BERs for SCM coded with code rate ¾ , with code rate 2/3, with Hamming coded and others

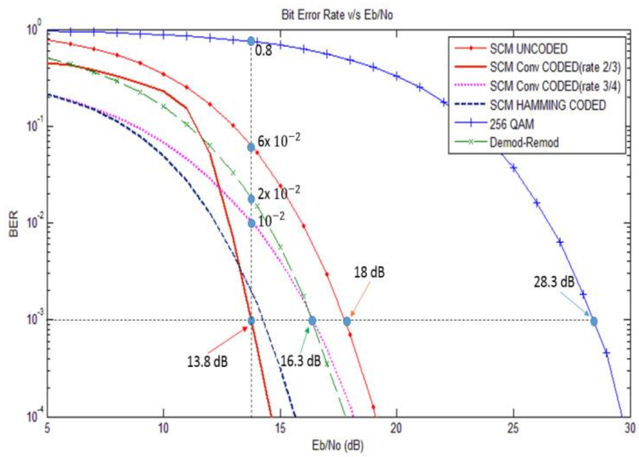

Figure 6: SCM (uncoded, convolution coded with code rate 2/3 & 3/4, hamming coded), Demod/Remod and 256-QAM

Figure 6 shows the simulation result which reveals that at fixed E_b/N_o (13.8 dB) for convolution coding with code rate 2/3, the BER is 1×10^{-3}, which is 20 times better than Demod-Remod approach, 60 times better than that of uncoded SCM, 2 times better than SCM with Hamming code (constraint length 7/4) and 10 times better than that of SCM coded with rate 3/4.

Conversely, to achieve BER 0.001, the required E_b/N_o is 18 dB (for uncoded SCM), 16.3 dB (for Demod-Remod and SCM coded with rate 3/4), 14.3 dB for SCM with Hamming code (constraint length 7/4) and 13.8 dB (for SCM coded with rate 2/3) which is 4.2 dB lower than uncoded SCM and 2.5 dB lower than for Demod-Remod and SCM coded with rate ¾, and 0.5 dB lower than that for SCM with Hamming coded.

IV. CONCLUSIONS

The BER graphs derived from SCM methods employing convolution coding with different code rates has been presented. These show that by increasing the redundancy, better performance is achieved i.e. the BER is reduced considerably at fixed E_b/N_o. In other words at fixed BER the required power level of signal is reduced appreciably. So whenever the bandwidth is not a prime factor, convolution coding with code rate 2/3 may be employed.

The SCM with coding with different code rates is able to trade off bandwidth for SNR and gives a performance far better than that of conventional Demod-Remod methods. Thus the SCM method effectively converts any low SNR communication link into a high SNR link at the cost of increased bandwidth.

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