

Experimental Evaluation of Trellis Coded Modulation (TCM) System Performance

H. N. OGBOKÉ¹, V. C. OGBOH², C. A. OBIORA-OKEKE³, N. A. NWOYE⁴
^{1, 2, 3, 4} Department of Electrical Engineering, Nnamdi Azikiwe University, Awka

Abstract- *Uncoded signals has constituted unbearable interferences in communication systems. The Limited bandwidth, a scarce resource does not encourage more traffic that constitute high bit error rate. The experimental based evaluation of Trellis-Coded Modulation (TCM) performance was carried to address the interferences observed and prove high rate data communication at a very high speed while maintaining the same bandwidth. This experiment was executed using Telecommunication Instructional Manual System (TIMS). The experimental result shows a great margin of coding gain against the theoretical M-ary shift keying (4-ASK) with an appreciable bandwidth efficiency. The observation is the possibility that information transmission will increase dramatically at low power for better system performance. Furthermore, digital communication engineers have the opportunity to access different modulation for effective transmission and necessary trade-offs.*

Indexed Terms- Coding, Trellis-Coded Modulation, Bandwidth, Communication, Transmission

I. INTRODUCTION

In evaluating the performance of Trellis-Coded Modulation (TCM), one of the methods used was experimental based evaluation of 4-ASK TCM performance using Telecommunication Instructional Modelling System (TIMS) equipment. Both analogue and digital systems can be experimented through this equipment. The block diagrams and mathematical equations used to represent electrical circuits modelled in Eurocard sized modules. It is a dual rack system with plug-ins for the upper modules and lower rack for the fixed modules. The output of a module automatically becomes the input for another module and the combinations yield the desired results. The model consists of signal generation by oscillation, Signal processing by multipliers and filters, signal

measurement by a frequency counter and a digital signal processing; TMS320C50. The system noise is typical act 40dB below the TIMS analogue reference level. Switching and tuning enables preset and/or reset for appropriate setting to achieve the desired result. Furthermore, it was modelled to use an oscilloscope to monitor the signal waveform at any stage in the modelling process. The experimental set-up consists of the transmitter known as the TCM generator and the noise channel to introduce Additive White Gaussian Noise (AWGN) noise. The BER instrumentation model records the bit errors between the transmitted and received signals.

In evaluating the performance of Trellis-Coded Modulation (TCM), one of the methods used was experimental based evaluation of 4-ASK TCM performance using Telecommunication Instructional Modelling System (TIMS) equipment. It is a model used to perform and obtain results of different dimensions as it relates to telecommunication systems. Both analogue and digital systems can be experimented through this piece of equipment. The block diagrams and mathematical equations used to represent electrical circuits are modelled in Eurocard sized modules. Its physical feature is a dual rack system with the upper rack as plug-ins for the modules and the lower rack is for the fixed modules. The modules deliver outputs that eventually become the input to other modules and the combinations of necessary modules yield the desired result. Each module, fixed or plug-in, has a specific function that could be categorised into any of the following:

- Signal Generation – Oscillators etc.
- Signal Processing – Multipliers, filters, etc.
- Signal Measurements – Frequency counter, and
- Digital signal processing – TMS320C50

TIMS modules have certain conventions that must be adhered to. The front panel of each module have sockets that serve to interconnect them through the

provision of interconnecting leads. The left side and the right side are designated for input and output respectively. The yellow colour sockets are for analog signals while the red colour sockets are for digital (TTL) signals. The analogue signals are adjusted to the TIMS analogue reference level of 4 Volt peak-to-peak. The system noise is typically at 40 dB below the TIMS analogue reference level. The modules are unique and switching and tuning are provided on modules as required to pre-set and/or reset them to appropriate parameter setting to achieve the desired result. Furthermore, it has been modelled to use oscilloscope to monitor the signal waveform at any stage in the modelling process [1].

A. Experimental evaluation of 4-ASK TCM performance using TIMS equipment

In other to effectively carryout this experiment two models were used. The TCM model as shown in figures 4. 1a and b, and the reference model as shown in figure 2.

The TCM model comprises

B. Module Functions and Basic Properties

- The Sequence Generator

The sequence generator is a device that uses a common external clock signal to generate two independent pseudorandom sequences. It has a SYNC output provided to start the sequences. The SG is powered by the clock signal through the TTL clock input and a pseudo random binary waveform is obtained from the TTL output. Four independent sequence pairs are available from lengths of 2^5 , 2^8 , 2^8 and 2^{11} . Input Clock Range TTL 1Hz to 1MHz, Analog < 500Hz to > 10kHz.

- The Convolutional Encoder Module

A continuous sequence of data bits is mapped into a continuous sequence of convolutionally encoded bits. A systematic convolutional encoder code 2 with rate, $R = 1/2$, and constraint length, $n = 4$ was used. During encoder operation, an extra bit is added to the serial input message (data) stream which results to an output bit rate being twice that of the input bit rate which depends configuration of the shift registers and the modulo-2 adders.

Output coded bits are presented in both serial and parallel TTL-level format, as well as 2-level and 4-

level bipolar format. Required bit clock signals are generated within the encoder module, derived from a single master clock input

The master clock, M.CLK, is a TTL-level signal, from TIMS MASTER SIGNALS

module's 8.33kHz SAMPLING CLOCK output. Since it is the transmitted bit rate, then the clock will run at half the rate of common bit rate, 2.083KHz i.e. 1.042KHz is required to run the sequence generator which is used to represent the message. This sampling clock is provided by the convolutional encoder

NORMAL mode was used, the encoder module maps and outputs the input data sequence into the selected convolutional code, CODE 2.

The frequency relationship between the input and output clock signals is $S.CLK = M.CLK / 8$, to provide a sampling clock of 1.042KHz as the bit rate with master clock frequency of 8.33KHz. the convolutional encoder produces a 4-level output.

- NOISE GENERATOR MODULE

This is a broadband noise source, with a 12 step (0dB to +22dB) output amplitude attenuator. The output noise level can be varied in discrete steps of 2dB. This is used to introduce AWGN noise to the channel to enable BER measurements be made.

- ADDER

This has two analog input signals A(t) and B(t) where the noise and signal may be added together, in adjustable proportions G and g. The resulting sum is presented at the output for further processing.

- TUNEABLE LPF

This is a frequency selective module that allows a certain frequency from low to cut-off frequency to pass and is varied using the TUNE control. The GAIN control is used to vary the amplitude of the output signal. It has a bandwidth (frequency range) of 2.0 kHz < WIDE < 12 kHz over which the signal is transmitted without expansion. The module is used for the band-limiting of the ASK signal as required by the TCM.

• INTEGRATE AND DUMP

This module consists of two independent functional blocks namely, a variable digital delay for TTL level clock signals which is used for aligning the phase of a bit clock to a data stream and a dual channel sampling block, integrate & dump and holding functions that can be switched in three combinations, Sample & Hold; Integrate & Dump; Integrate & Hold.

The DELAY control knob enables a digital phase delay function by varying the time between the positive edge of the signal at the B.CLK input, with respect to the positive edge of the output signal at CLK.OUT.

The output of this module is the input to the Viterbi decoder (TIMS320 DSP-HS) module.

• VITERBI DECODER (TIMS320 DSP-HS)

This is a convolutional decoder with Viterbi algorithm designed to generated continuous sequence of data bits from a continuous sequence of convolutionally encoded bits. The output is fed to the error correcting utilities for comparison with the transmitted signal.

• ERROR CORRECTING UTILITIES

This module compares the decoded message with the transmitted message by an X-OR logic gate and errors are recorded on the frequency counter for BER. It has a pulse count setting to ascertain the number of bit counts to compute the BER measurement [1].

II. EXPERIMENTAL PROCEDURE

A. Experimental Set-up

The Transmitter (TCM generator) made up of the sequence generator module and the convolutional encoder module

The Channel (noise channel) which consists of the noise generator module, the adder module, and the tuneable low pass filter module

To access the performance of TCM, experimental evaluation of trellis-Coded Modulation (TCM) was executed using Telecommunication Instructional Manual System (TIMS) equipment and the modules were modelled as shown in the figures in this chapter.

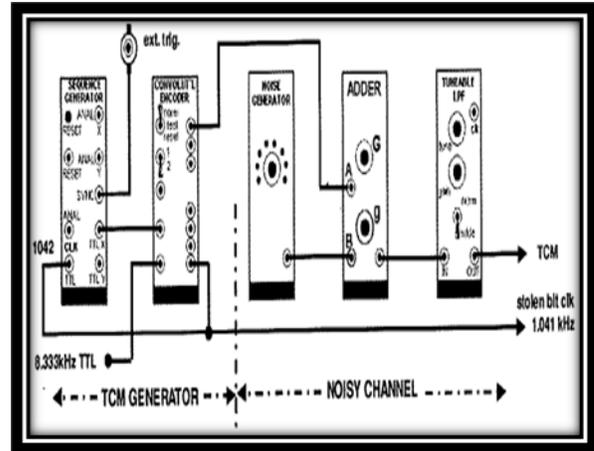


Figure 1.0: TCM generator and channel model [1]

• The Transmitter

To model this experiment, some modules require being pre-set to their operating conditions. This is to enable the smooth operation of the equipment and for result oriented purposes.

To set up the transmitter depicted in the sequence generator (SG) has to be set to the required sequence. The short sequence, 2^5 (32 bits) was selected by switching both toggles of the on-board switch SW2 up. Then the SG was patched up in the TIMS panel to be powered. The SYNC output provided is powered by the external trigger from the oscilloscope which starts the sequences. Next the SG was clocked at the frequency of 1.042KHz which was supplied by the Master clock signal at 8.33KHz through the convolutional encoder that splits the frequency by eight (8). Hence, a bit clock of 1.042 KHz of waveform (green) was fed to the SG. The TCM transmitter operates at the message bit rate of 1.042 KHz to generate the pseudorandom output message of figure 27a(yellow waveform) that was fed to the convolutional encoder serial data input. The oscilloscope was used to confirm the clock frequency. The convolutional encoder was set to normal mode while the code select switch was switched to 2 to enable the encoder operate as a systematic convolutional encoder suitable for TCM to avoid catastrophic codes that adversely affect the coding gain. The convolutional encoder, as a default setting, produced A 4-level output (00 01 10 11), since we are experimenting on 4-ASK modulation. Using the oscilloscope a 4-level output was affirmed as depicted in figure 27b.

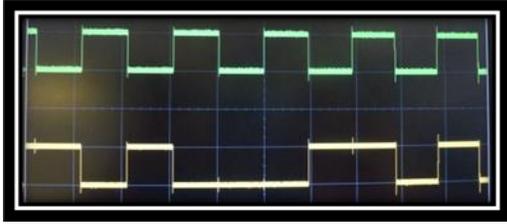
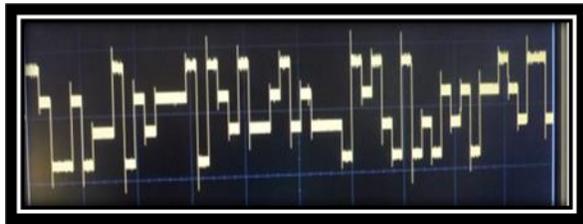


Figure 2.0: a) Input and output of sequence generator;



- The Channel

b) 4 level output of convolutional encoder

The channel consists of three modules, the noise generator (NG), the adder and the tuneable low pass filter. The system was patched up with the NG disconnected to facilitate alignment of the two sequences as will be discovered later. The front GAIN control of the Tuneable LPF was set to mid-way position with the toggle switched to its widest bandwidth-*the bandwidth of operation of the TCM system*. This expands the filter's range to above 10 kHz. This was to allow low to cut-off frequency to pass. Above this frequency the message will be lost (cut off). The Gain control of the ADDER was used to set the 4-level output from the LPF to 0.2volt peak-to-peak observed at the display unit. When noise at 14 dB was introduced to the signal there was a change in the output waveform as shown in figure 1.0.



Figure 3.0: 4 level output of convolutional encoder with 14 dB noise

B. The Detector & Viterbi Decoder

The TCM detector/decoder unit employed in this work to perform the retrieval of the transmitted data

comprises the Integrate & dump and the Viterbi decoder (TIMS 320 DSP-HS) modules as shown in figure 4.0. These modules also require parameter setting for efficient operation.

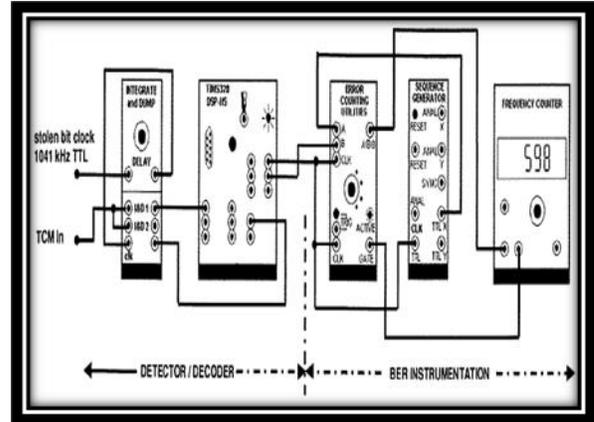


Figure 4.0: TCM detector/decoder model with BER Instrumentation model

For the Integrate and Dump, the following parameter settings were implemented:

- The on-board switch SW1 was set to Integrate and Hold (I&H1) to make Integrate & Dump (I&D1) subsystem to perform an Integrate and Dump operation
- The on-board switch SW2 was set to Integrate and Dump (I&D2) to make Integrate & Dump (I&D2) subsystem to perform an Integrate and Hold operation.
- The toggles on-board switch SW3 was set as follows; upper button to the LEFT, and lower button to the RIGHT. These govern the range of delay introduced by the DELAY from the front panel control.

The detector/decoder was patched as shown in the figure 4.0 noting that EPROM has been installed in TIMS320 DSP-HS to guarantee soft decision detector/decoding operation. Through the oscilloscope, the Integrate & Hold operation from I&D1 output as well as the Integrate & Dump operation from I&D2 output were observed. Its delay knob was used to maintain the 4-level signal from the Integrate & Hold output which had 8-level signal output.

For correct operation of the Viterbi decoder, the Gain of the Tuneable LPF was returned so that the input to ADC1 of TMS320 DSP-HS module is 3 volt peak-to-peak (the 4-levels should be ± 1.5 and ± 0.5 volts). This was monitored from the display unit.

C. Viterbi Decoder

With the aid of the oscilloscope, the BIO input and the TTL output #1 of the decoders was confirmed at 1.042KHz. This is to ensure that the reference sequence generator in the BER instrumentation unit operates (triggered) at the same rate with SG at the transmitter to generate the same message for comparison with received message for error check.

The waveform monitored through the oscilloscope showed that the two waveforms (transmitted and received) were not the same until after alignment. This means there is a considerable off-set (i.e. processing and other delay) between the two waveforms as a result of the system. Figure 5.0a is the Txd and Rxd waveforms while figure 5.0b is the BER recorded.

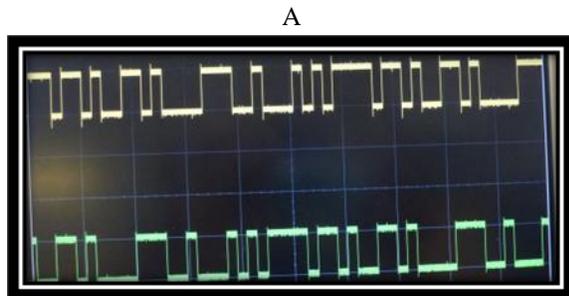
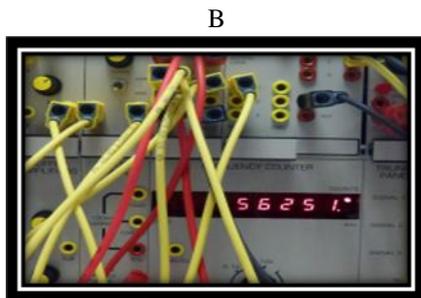


Figure 5.0: a) Unaligned waveform of transmitted and received msg.

5.0 (b) BER of unaligned transmitted & received message



- BER Instrumentation

The Bit error ratio (BER) measurement was instrumentation that encompasses:

- I. Error counting utilities module
- II. A sequence generator module and
- III. The frequency (error) counter module

BER instrumentation model is made up of three modules; the error counting utilities, the reference sequence generator and the frequency counter as illustrated in figure 4.0. This model was used to compare and record the number of bit disagreements (errors) between the transmitted and received signals. It was initially used to align and synchronise the transmitted and received data before the introduction of noise to ensure any bit error was due to noise interference.

In setting up this model, only the sequence generator was adjusted to short (bit) sequence ($2^5=32$) through the on-board switch for the two SGs to generate the same random data for the alignment of both SGs. The SG was later switched to long sequence i.e. $2^{11}=2048$ bits for a better result that gave rise to the readings in the table 1.

The instrumentation was then patched up with the TCM model as shown in figure 8. Having aligned the two sequence generators signal output waveforms in time and data bits through the oscilloscope and confirming it with a confidence count of 1 from the error counter. See figure 6.0.

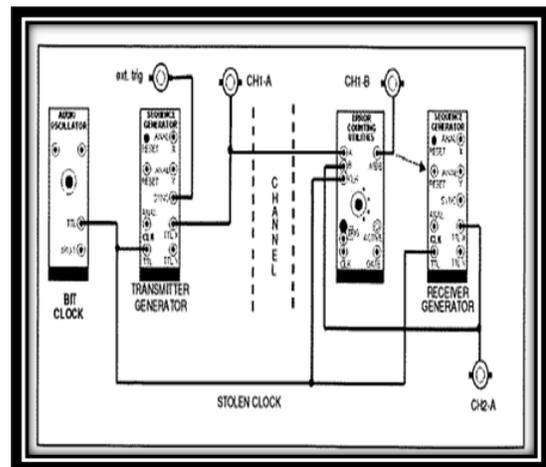


Figure 6.0: Two sequence alignment using sliding window correlator model [19]

Through the use of X-OR gate of error counting utilities, the received and locally generated sequences were aligned in data bits by establishing a connection between it and the reset button. The transmitter bit clock was “stolen” to clock (power) the error counting module and the ref SG to operate at the same rate with the transmitter SG. The misalignment and alignment were observed by the oscilloscope with errors and no error respectively as shown in figures 5.0a and 7.0a. Before carrying out the BER measurement, the principle of adding noise was observed to obtain the SNR as explained in figure 8.0.

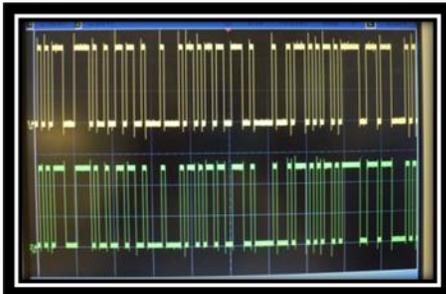


Figure 7.0: a) Aligned Transmitted & Received message (Long seq);



B

b) Confidence count obtained

C. Noise Addition – Principle

The Wideband True RMS Meter was used to measure the signal and noise voltage of the system. The oscilloscope and Wideband True RMS Meter were connected to the model to observe the signal at the output of the channel.

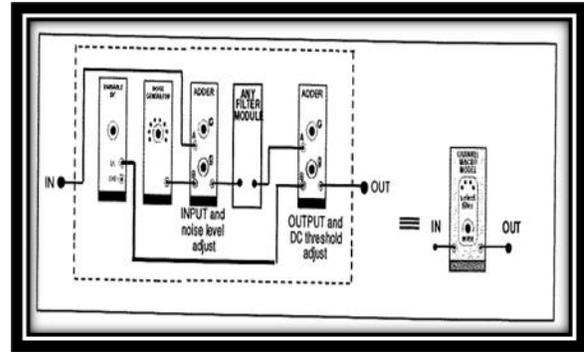


Figure 8.0: TCM Noise addition principal model

Reducing the signal amplitude to zero using the ‘G’ gain control of the input adder, the Noise generator attenuator was set to maximum output while the noise introduction into the channel with the adder ‘g’ gain was tuned to maximum. A reading on the RMS meter was recorded (as N volt rms amplitude). Now, unplugging the noise cord the signal was introduced to obtain the same reading (as S volt rms amplitude). This sets the SNR to zero from the expression SNR set at a reference value

$$10 \times \log_{10} \left(\frac{S^2}{N^2} \right) = 10 \times \log_{10} (S/N)^2$$

Which implies

$$20 \times \log_{10} \left(\frac{S}{N} \right)$$

Where $S = N = 2.21 \text{ V}$

SNR = 0 dB

Noise voltage readings were taken at different noise setting (range 0 dB to 22 dB) as shown in the table 1.

Now the signal was introduced and errors were recorded at different SNRs. It was observed that as the noise level is reduced in steps of 2dB from the 22dB to 0dB, the number of errors decreased exponentially to zero. When the SNR was above 11.04 dB, there was no error which means high SNR. Errors were recorded at pulse count of 10^6 .

Table 1.0: Experimental Result

Noise generator setting, (dB)	Signal voltage, S (volt)	RMS meter Noise N (volt)	SNR = $20\log(S/N)$ (in dB)	E_b/N_0 (dB) =	Error count @($\times 10^6$)	BER = (Error count-1)/ 10^6
0	2.21	2.21	0	-3.0103	169103	0.1691
2	"	1.72	2.18	-0.8303	98309	0.0983
4	"	1.35	4.28	1.2697	26539	0.0255
6	"	1.05	6.46	3.4497	2560	0.0026
8	"	0.85	8.62	5.6097	216	0.00023
10	"	0.62	11.04	8.0297	3	0
12	"	0.46	13.63	10.6197	1	0
14	"	0.35	16.0	12.9897	1	0
16	"	0.26	18.59	15.5797	1	0
18	"	0.19	21.31	18.2997	1	0
20	"	0.13	24.61	21.5997	1	0
22	"	0.09	27.8	24.7897	1	0

D. The Reference System

The reference system of figure 9.0 is devoid of a coding/decoding scheme such as the convolutional encoder as used in figure 26, the TCM experiment. This was used to compare with the later to obtain the coding gain as a result of the introduction of TCM.

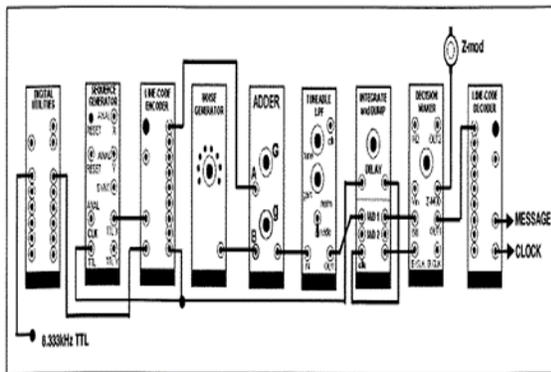


Figure 9.0: TCM Reference system model

The model consists of a digital utility, a sequence generator and a line code encoder which serves as a transmitter as well as a noise generator, an adder and a tuneable low pass filter that serve as noisy channel unit. While the integrate and dump, the decision maker and the line code decoder forms the detector/decoder unit. The BER instrumentation model discussed earlier

will be used to measure and record the bit disagreements for bit error rate calculation. The processes taken except the inclusion and settings of the convolutional encoder and the Viterbi decoder (TIMS320 DSP-HS) is as applicable to TCM model.

Unfortunately, this experiment could not be carried out due to unavailability of digital utilities module. This module is used to set the bit clock (bit rate) at the same rate as the TCM model in order to effectively compare results and obtain the required coding gain.

3. TCM Matlab Implementation

A. Generator Polynomial

This is the basis on which a measurable coding gain is achieved. It takes into consideration the number of inputs k and the expected outputs at the respective outputs n.

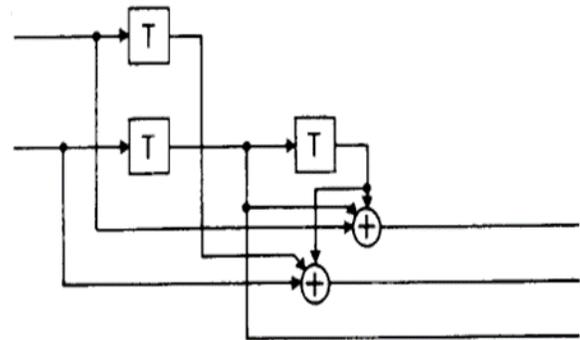


Figure 10: Rate 2/3 Systematic encoder convolutional coder with feedforward connection [20]

Using figure 35, the Constraint length (CL) = K = 2 & 3; this is a vector of 1- by - k matrix, where k is the number of inputs to the encoder k=2.

Code generator = $[h_0, h_1, \dots, h_n] = n = 3$; this is a vector of k-by-n matrix in octal representation, where n is the number of outputs from the encoder. Below are the impulse responses and their generator polynomials.

$$h_0 = 0 = 0; h_1 = 10_2 = D = 02_8; h_2 = 01_2 = 1 = 1_8 \text{ (CL=2)}$$

$$h_0 = 010_2 = D = 02_8; h_1 = 001_2 = 1_8; h_2 = 0100_2 = D^2 = 04_8 \text{ (CL=3)}$$

Consider the connections of figure 35 as in the equivalent scheme from Fig. 9 (21), then the following generator matrix was obtain for poly2trellis: $G = [1\ 2\ 0; 4\ 1\ 2]$. The constraint lengths for these 2 inputs are given by the vector $CL = [2\ 3]$.

Finally, your trellis is generated as:
`trellis = poly2trellis(constraint length(CL), [code generator(CG)])`
`= poly2trellis (K,`
`[h0,h1,...hn])`
`= poly2trellis([2 3]; [1 2`
`0; 4 1 2])`

B. TCM Standard Codes

The table 2 gives standard codes for 8 & 16-PSK , 16-QAM and above for optimum coding gains.

Table 2a: Codes for phase and quadrature amplitude modulation

No. of states 2^v	m	Parity-check coefficients			d_{free}/Δ_i^2	Asympt. coding gain [dB]	
		h^2	h^1	h^0		$G_{PSK-PSK}$ (m=2)	$G_{PSK-PSK}$ (m=3)
4	1	—	2	5	4.000*	3.01	—
8	2	04	02	11	4.586	3.60	—
16	2	16	04	23	5.172	4.13	—
32	2	34	16	45	5.758	4.59	—
64	2	066	030	103	6.343	5.01	—
128	2	122	054	277	6.586	5.17	—
256	2	130	072	435	7.515	5.75	—
4	1	—	2	5	1.324	—	3.54
8	1	—	04	13	1.476	—	4.01
16	1	—	04	23	1.628	—	4.44
32	1	—	10	45	1.910	—	5.13
64	1	—	024	103	2.000*	—	5.33
128	1	—	024	203	2.000*	—	5.33
256	2	374	176	427	2.085	—	5.51

Table 2b: Codes for phase and quadrature amplitude modulation

No. of states 2^v	m	Parity-check coefficients			d_{free}/Δ_i^2	Asympt. coding gain [dB]			
		h^2	h^1	h^0		$G_{QAM-PSK}$ (m=3)	$G_{PSK-PSK}$ (m=4)	$G_{QAM-PSK}$ (m=5)	$G_{QAM-PSK}$ (m=6)
4	1	—	2	5	4.0*	4.36	3.01	2.80	3.01
8	2	04	02	11	5.0	5.33	3.98	3.77	3.98
16	2	16	04	23	6.0	6.12	4.77	4.56	4.77
32	2	10	06	41	6.0	6.12	4.77	4.56	4.77
64	2	061	016	101	7.0	6.79	5.44	5.23	5.44
128	2	012	014	203	8.0	7.37	6.02	5.81	6.02
256	2	304	056	401	8.0	7.37	6.02	5.81	6.02
512	2	0510	0316	1001	8.0*	7.37	6.02	5.81	6.02

The No of states, 2^v is the number of states of the trellis diagram and v is the constraint length. The m is the number of inputs to the encoder while m is the

total inputs to the signal mapper. The parity check coefficients are the code generators.

B. TCM MATLAB Codes Implementation

- FOR TCM PSK (this is also applicable to TCM QAM except for modulation and demodulation objects and the parameters as is applicable to each)

```
%DEFINE PARAMETERS (Refer to the appendix for the main code)
M = 8;% Alphabet size
n = 300000;% number of bits to process per iteration
k = log2(M); % Number of bits per symbol
codeRate = 2/3; % Ration of input to output of encoder
constlen = [2 3]; % Number of impulse responses that resets the register to zero state
nsamp = 1; % Oversampling rate
codeGen = [1 2 0; 4 1 2]; % Generator matrix in octal representation
EbNodB = 0:1:24; %Eb/No values (in dB) to simulate
MinNoErr = 100; %Do repeated simulation rounds till at least this no of symbol errors received
MaxNoSymbols = 1e9; % Maximum number of symbols in total simulation
```

The parameters used for the MATLAB implementation of the codes below are given in the defined parameters.

C. TRELIS STRUCTURE

`t = poly2trellis([2 3];[1 2 0; 4 1 2]);%`

The poly2trellis structure gives information about the block diagram regarding: 1. The constraint length [2 3] – the number of impulse response that will reset the shift registers to all zero. It also determines the coding gain; as the higher it is in number the higher (or better) the coding gain. Expressed in MATLAB as a vector of 1-by- k matrix, where k is the number of inputs to the convolutional encoder.

2. The code generator [1 2 0; 4 1 2] is the resultant effect of the convolution principle carried out by the encoder which gives the parity check coefficients in binary numbers and expressed in octal representation, obtained from the generator polynomials. This determines the number of connections to the modulo-

2 adder that results to an output, the coefficients. The number of elements determines the number of outputs, n and represented in MATLAB as a vector of k -by- n matrix. This also contributes to the coding gain to be obtained.

E. MODULATION SYSTEM OBJECT

```
hMod = comm.PSKTCMModulator(t,'ModulationOrder', 8);
```

The system object, `comm.PSKTCMModulator` implements trellis-coded M-ary phase shift keying (PSK TCM) modulation by convolutionally encoding the binary input signal and then mapping the result to a PSK constellation with specified properties set to the specified values as stated below. This object carries out the convolution of the inputs, k and generates an output, n with a parity bit, $p=1$. Hence, output $n=k+1$. It also doubles the input by a factor of 2 i.e. 2^{k+1} and gives a signal constellation of double the input by the function of a signal mapper through the principle of set partitioning. Furthermore, it is mapped in a waveform (nonbinary) number in the trellis diagram of the number of states desired to ensure the best Euclidean distance (ED). For instance, in this work;

- The trellis structure = t ; the result of the `poly2trellis` structure
- The TerminationMethod is set to “Continuous” the encoder retains the encoder state for use with the next input.
- The ResetInputPort is set to False because the step method does not require additional input.
- The Modulation Order is set to the number of points to the signal constellation which is equal to 2^n for a rate k/n convolutional code.

F. DATA GENERATION AND MODULATION

```
data = randi([0 1], n,1); % Generates binary data
modData = step(hMod, data);%Modulate data by the
object handle for transmission
tr_modData = modData; % transmitted message
```

Here binary data is generated as specified in the parameter section and the step function processes the input data according to the object algorithm as stated

above and returns the encoded and modulated data to `modData` for transmission. It is transmitted in real and imaginary numbers that maintains the ED as specified by Ungerboeck’s set-partitioning.

G. NOISE INTRODUCTION TO THE MESSAGE

```
%% ADDITION OF NOISE
ttxtlb = ' Computing BER at Eb/No = '; PSKtype =
'8-PSK ';
for i = 1:length(EbNodB);
    snr(i)=EbNodB(i)+10*log10(k);
    disp(['***** ',PSKtype,ttxtlb,num2str(EbNodB(i)),
dB *****]);
    NoErr=0; NoSymbols=0; FramCount=1;%
Initialise to zero
    while NoErr < MinNoErr && NoSymbols <
MaxNoSymbols
        disp([' FramCount = ',num2str(FramCount),'']);
        hAWGN =
comm.AWGNChannel('NoiseMethod', ...
'Signal to noise ratio
(SNR)', 'SNR',snr(i),'SignalPower',3);
        nos_msg =step(hAWGN,tr_modData);
        recv_msg = nos_msg;
```

The introduction of noise is to model a real-time scenario of energy per bit E_b to noise spectral density N_o (E_b/N_o) between 0 to 16 in dB to calculate the signal to noise ratio (SNR) that is introduced to the message through the system object `hAWGN`. A loop statement “for” is introduced to add noise level in steps of 1 dB to 16 dB to the message per iteration. The “disp” function displays the modulation type (8-PSK) and E_b/N_o that is being computed. The while loop sets condition that if `NoErr` is not up to `MinNoErr` and `NoSymbols < MaxNoSymbols` continue the iteration. The `hAWGN` system object has several properties set to model the system as desired. The choice of setting is given below.

- The noise method is set at SNR as calculated above
- The signal power is set at 3 watt.

The step function processes the data for transmission according to the noise object to cause interference to the message such that BER measurement will be calculated.

H. DEMODULATION/ERROR SYSTEM OBJECT FUNCTION

```
%% DEMODULATION
hDemod = comm.PSKTCMDemodulator(t,
    'ModulationOrder', 8, ...
    'TracebackDepth', 16);
msg_rx = step(hDemod, recv_msg);
% Error calculation
hError = ...
comm.ErrorRate('ReceiveDelay', hDemod.TracebackDepth*log2...
(t.numInputSymbols));
errorStats = step(hError, data, msg_rx);
Num_er = errorStats(2);
NoErr = NoErr + Num_er; NoSymbols = NoSymbols + length(data);
FramCount = FramCount + 1;
```

The hDemod system object performs a reverse action as the PSKTCM modulator using Viterbi algorithm. It detects, decodes and demodulates received message. It performs the function of a convolutional decoder that traces the path the message sequence has travelled through the encoding trellis while comparing it with the transmitted message. This is achieved through a decision-making process known as maximum-likelihood criterion.

The coding gain is obtained through a process as it performs the function of choosing on the trellis code structure the minimum distance of a signal state of a subset among other signal waveforms as they change state. Each transition from a particular subset (e.g. 0 4 2 6) has a minimum ED. It traces the path from divergence and emergence from the all zero path. The sum of the squared MED of the sequences gives the free distance of the coded signal. It has the following properties to set to achieve the desired result.

- The output of the poly2trellis structure designated as “t”.
- Termination Method is set at 'Continuous' because the object saves the internal metric at the end of each frame to be used by the next frame.
- The Modulation Order is set as the number of elements in constellation=8
- The Reset Input Port is set to “false”
- Traceback Depth is generally 3 or 4 times the constraint length and is the number of bits the

decoder will run through before it makes choice of which path is correct. The longer the traceback length the more accurate the output will be.

The step function processes the received message according to the demodulation object to enable the message be recovered.

The Error Rate system object compares the transmitted data with the received data and calculates the bit disagreements as the error rate (BER). The properties are set as:

- Samples to consider: all samples, hence is set at “Entire frame”
- ResetInput Port is set at “False” since it does not need to reset the error statistics via an input to the step method.

The last two line of codes that ends the while loop gives the number of errors and symbols for each Eb/No being computed.

```
NumErr(i) = NoErr;
BER(i) = NoErr/(NoSymbols*k);
disp([' Num of Err = ', num2str(NumErr(i))]);
disp([' BER = ', num2str(BER(i))]);
```

These lines of codes bring the for loop to the end by displaying the number of errors calculated and BER by dividing NoErr obtained by total number of bits processed within the errors were obtained.

```
EbNodB = 0:1:18;
semilogy(EbNodB, BER, 'r--', 'LineWidth', 2)
xlabel('Eb/No(dB)', 'LineWidth', 2);
ylabel('BER', 'LineWidth', 2); grid on; hold on
% parameters for BER Theoretical
v = 2*pi/M; z = 1-cos(v);
A = 10.^(EbNodB./10);
BER3 = (1/k)*erfc(sqrt(1/2*A.*z*k)); % Theoretical BER for PSK(M>2)
semilogy(EbNodB, BER3, 'bo-', 'LineWidth', 2)
% title('CODING GAIN OF TCM FOR 8-PSK', 'fontSize', 16, 'fontWeight', 'b');
legend('8-PSK TCM Simulation', '8-PSK Theoretical')
hold off
```

The codes above plots Eb/NodB against BER graph of the simulated and theoretical 8-PSK using semilog on the ordinate.

IV. INCORPORATION OF RAYLEIGH CHANNEL

% Create Rayleigh fading channel object for reflected paths where ts is the sample time in secs and fd the maximum Doppler shift in hertz..

```
chan = rayleighchan(ts, fd);
pskSig = step(hMod, tx); % PSK modulate the signal
fadedSig = filter (chan, pskSig); 5 % Apply the channel effects
rxSig = step(hAWGN,fadedSig); % Add Gaussian noise
rx = step (hDemod, rxSig); % Demodulate
```

A. Rayleigh Channel

In a wireless mobile communication system, a signal can travel from transmitter to receiver over multiple reflective paths; this phenomenon is referred to as multipath propagation. The effect can cause fluctuations in the received signal’s amplitude, phase, and angle of arrival, giving rise to the terminology multipath fading.

Rayleigh channel have randomly time-variant impulse responses that are a consequence of the constantly changing physical characteristics of the media. Scintillation is used to describe the multipath fading caused by physical changes in the propagating medium, such as variations in the density of ions in the ionospheric layers that reflect transmitted high-frequency (HF) radio signals. Consequently, if the same signal is transmitted at HF in two widely separated time intervals, the two received signals will be different.

As a result of this a severe penalty in SNR must be paid as a consequence of the fading characteristics of the received signal. This penalty can be reduced dramatically by means of efficient modulation/coding and demodulation/demodulating techniques.

One characteristics of a multipath medium is the time spread introduced in the signal that is transmitted through the channel.

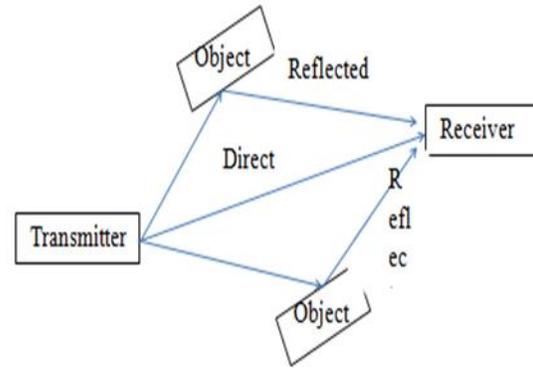


Figure 11: Multipath Propagation environment

Coding for fading channel: The structure of errors in a fading channel is different from that in an AWGN channel. The presence of error bursts is noticeable: when the channel shows (instantaneous) high attenuation due to the destructive interference of multipath components, then the error probability is much larger than in constructive interference. Correction of these error bursts can be achieved either by using codes that are especially suited for bursty errors RS codes. Alternatively, interleaving “breaks up” the bursty structure of errors[13]

V. RESULTS AND ANALYSIS

A. Experimental Result

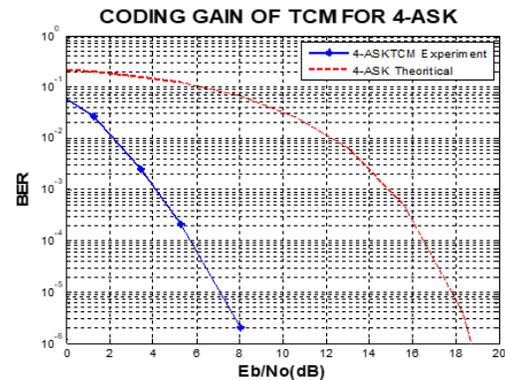


Figure 12: a) Experiment coding gain; b) Literature graph [17]

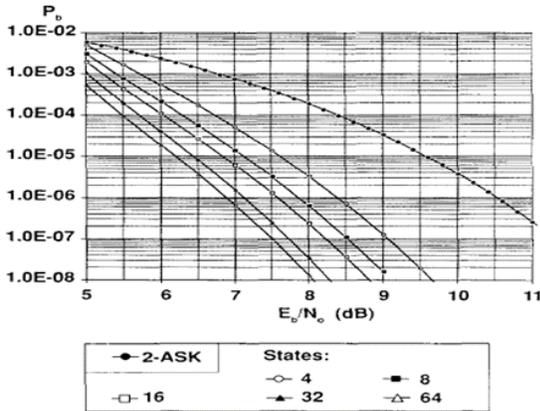


Fig. 4. BER for best Ungerboeck codes for 4-ASK.

This graph of figure 37a compares the theoretical and TCM experiment of 4-ASK of E_b/N_o against BER to obtain the coding gain. The trend shows a sharp decreasing linear graph of BER as E_b/N_o increases. At the BER of 10^{-5} the coded and theoretical ASK have E_b/N_o of 7.2 dB and 18 dB respectively offering a coding gain of about 11 dB as a result of TCM. The bandwidth efficiency is $\log_2(4) = 2$ bits/s/Hz.

B. Literature Result

The trend is the same as in the case of the experiment of figure 37b. At the BER of 10^{-5} , the power of transmission is 7.6 dB which compares favourably with the experimental result although a difference of 0.4 dB.

It is worthy to note that the original equipment manufacturer (OEM), the manufacturer of TIMS equipment instructed that the signal be set at 4-volt peak-to-peak, the reference level with a second adder module which is unavailable to effectively measure the noise to be added to the system. Hence, the difference in the result is probably the unavailability of extra module to measure noise added as specified.

C. SIMULATION RESULTS

I. PSK Simulation Results

This section analyses the performance of TCM in two different modulation schemes, PSK and QAM using computer simulations. Their performances are compared to other results to ascertain their performance.

II. Simulation Result

Figure 12 compares the trends of the simulated TCM, uncoded and theoretical 8-PSK modulation with respect to the coding gain at a particular BER. The TCM simulation and uncoded modulation showed a coding gain of about 3.5 dB (3.6 dB) at BER of 10^{-5} which is almost the same value with the theoretical PSK. Furthermore, for 16 PSK 16 states as shown in figure 39, at the same BER, there was E_b/N_o of about 13 dB and 17.8 dB for coded and uncoded system respectively of which coding gain of 4.8 dB was achieved compared to 3.5 dB in the case of 8 PSK.

III. PSK Literature Results

As seen in Figure 38a the BER decreases as E_b/N_o increases in a near-exponential manner in all the cases studied. Also for a given E_b/N_o , the uncoded modulation appeared to have the highest values for BER while the TCM simulation had the least. This can be attributed to the gain in power as a result of the integration of the coding and modulation functions. Furthermore, this trend agrees quite well with published results[17] as shown in figure 38b except for a few discrepancies; the very obvious being the E_b/N_o values at a given BER. These discrepancies are most likely due to variations in configuration, signal power and trellis structure. However, at BER of 10^{-5} there is E_b/N_o of 9.7 dB compared to 7.0 dB of the literature with a difference of 2.7 dB.

These differences are probably because of parameter differences used in the literature varies with the simulation parameters such as the trellis structure and the signal power.

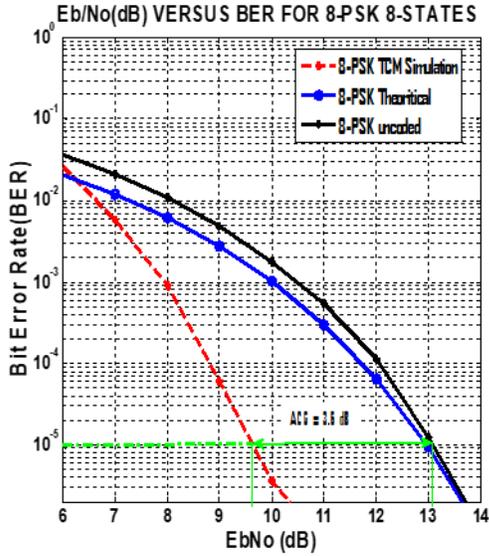


Figure 12: 8-PSK a) Simulated result, b) Literature result

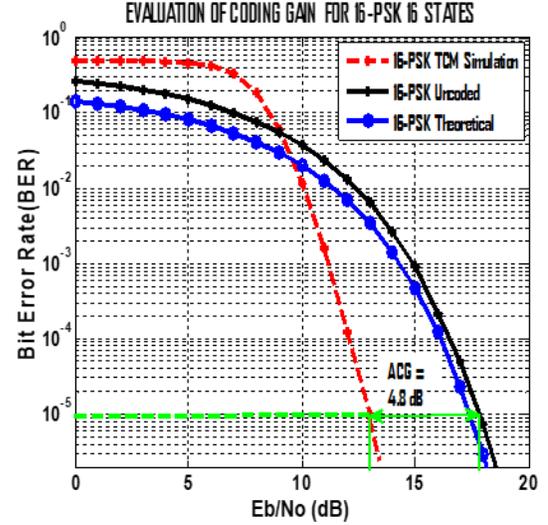


Figure 13: 16 PSK simulated result

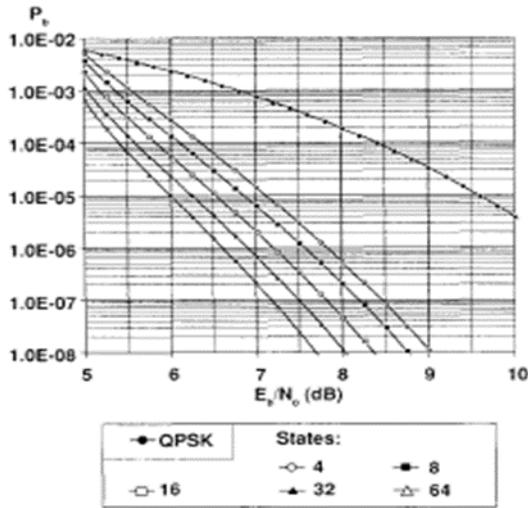


Fig. 2. BER for best Ungerboeck codes for 8-PSK.

IV PSK Bandwidth Efficiency

The figure 14 shows the performance of 8-PSK with 8 states and 16 PSK 16 states. The trend shows that as the bandwidth efficiency increases, the power efficiency also increases giving a rising exponential curve known as Shannon's limit. With this modulation scheme, the bandwidth efficiency is $k = \log_2(M) = 3$ and 4 bits/s/Hz respectively corresponding to 9.7 dB and 13 dB respectively in figure 38. Evaluating with MATLAB reveals that coded 8 and 16-PSK are 6.02 dB and 9.32 dB less power efficient than Shannon's specification respectively. Hence, 8 and 16-PSK are more power efficient and ensures better system performance than uncoded 8 and 16-PSK modulation.

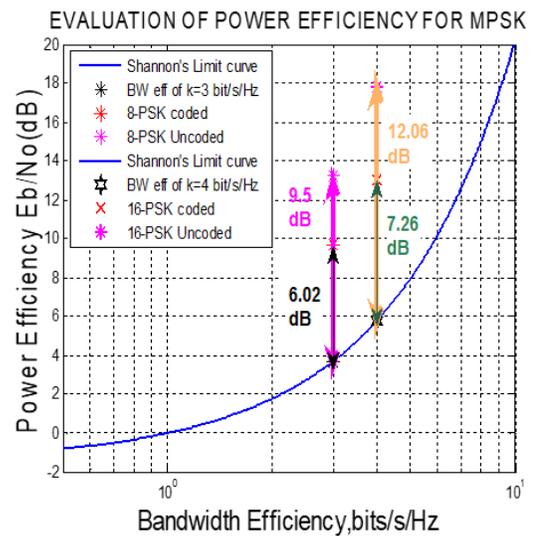


Figure 14: MPSK Power efficiency graph

V. QAM Simulation Results

In a similar development, in 16-QAM TCM simulation of figure 15, the trend of a declining BER with increasing E_b/N_0 dB as observed in the previous PSK modulation. Again three cases were compared, coded, uncoded modulation and the theoretical illustrates what the power difference should be at a particular BER. For instance, using BER of 10^{-5} as a reference, a signal of good quality can be transmitted at 7.8 dB as against 12.8 dB of uncoded 16-QAM resulting to a savings of 5 dB. This is as a result of applying TCM.

Also, in 4-QAM TCM simulation of figure 42, with the same trend still maintained at the same BER of 10^{-5} with the uncoded and theoretical agreeing, the coded scheme is at 5.8 dB as against 9.6 dB of uncoded 4-QAM resulting to a coding gain of 3.7 dB.

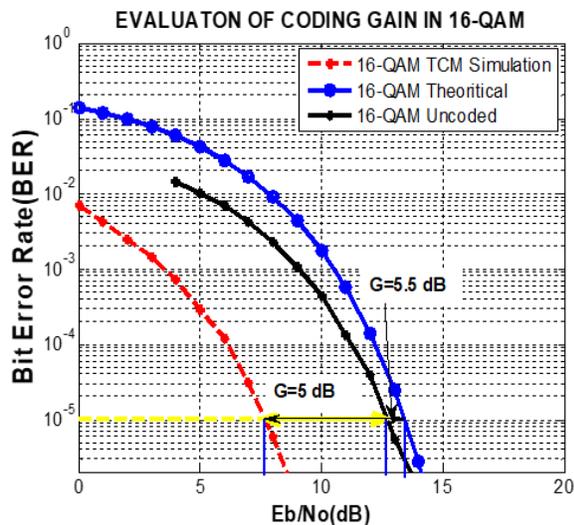


Figure 15: 16-QAM Simulation of E_b/N_0 versus BER

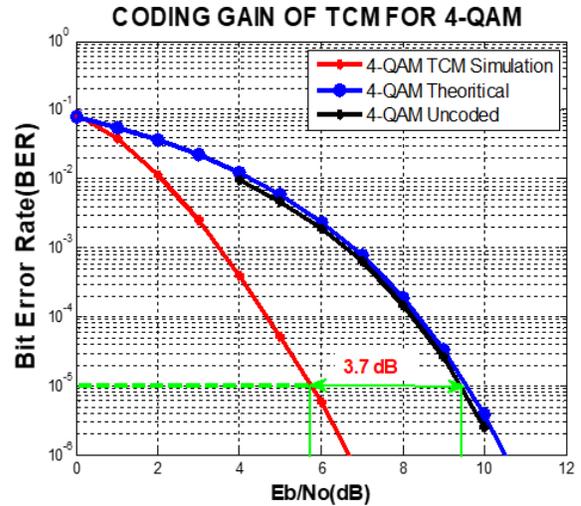


Figure 16: 4-QAM Simulation of E_b/N_0 versus BER

VI. QAM Bandwidth Efficiency

The spectral efficiency of QAM TCM modulation is depicted in figure 16. The graph reveals that TCM and uncoded 16 QAM having 4 bits/s/Hz ($k = \log_2(M)$) is 2.2 dB and 7.2 dB less power efficient than Shannon power specification respectively. Hence, 16 QAM TCM the system performance is more power efficient. While for 4-QAM TCM with 2 bits/s/Hz is 4.04 dB less power efficient than Shannon specification against 7.74 dB of uncoded modulation scheme. Consequently, 16-QAM has a higher coding gain and better power efficiency than 4 QAM as it is closer to Shannon's limit.

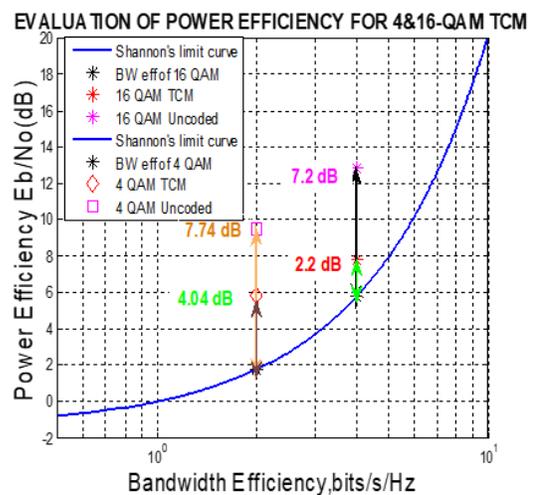


Figure 17: Power Efficiency plane

VII. COMPARISON OF TCM WITH OTHER CODED MODULATION SCHEMES

The performance of TCM could be ascertained by comparing it with other coded modulation techniques. In context, comparison is made of Turbo Trellis Coded Modulation, Bit- interleaved coded modulation and BICM-ID with TCM.

From figure 45, TCM performs better than BICM in AWGN channel but BICM-ID performs better than TCM and BICM over the same channel. Above all, TTCM in figure B shows superior performance over the modulation schemes under consideration.

Figure 18 & 19 shows that at BER of 10^{-4} the E_b/N_0 of TTCM is about 3.8 dB compared to 5.3 dB of TCM and 6.4 dB of BICM.

VIII. ANALYSIS OF RESULTS

Table 3: Overall Analysis Result

S/N	MODULATION	ASK	PSK	QAM	Trend		
1	Alphabet size(M)	4	8	16	4	16	Increasing trend, ASK to QAM
2	Coding Gain (dB)	11	3.5	4.8	3.7	5.0	Increasing trend (ASK to QAM)
3	E_b/N_0 (dB)@ 10^{-5} BER	7.2	9.7	13.0	5.8	7.5	Increasing per scheme
4	Power Efficiency(dB)	5.4	6.2	7.26	4.04	2.2	Decreasing per scheme

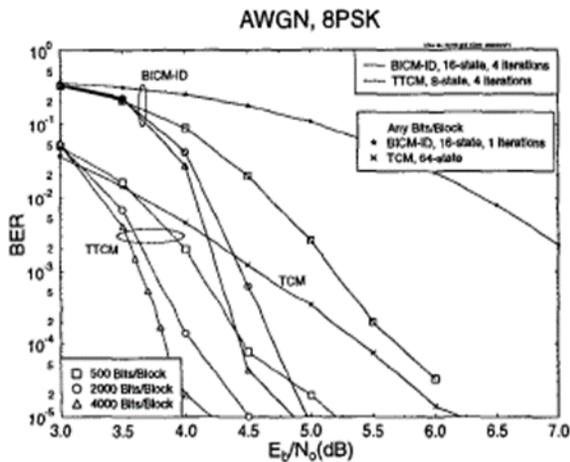


Figure 18: Graph of 8-PSK in TTCM, BICM & BICM-ID in AWGN at different bits/block[23]

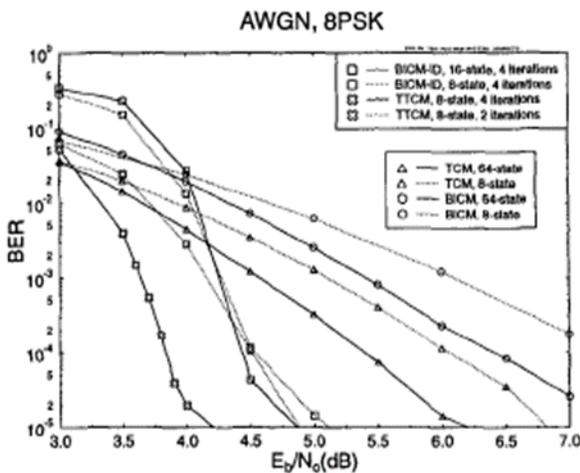


Figure 19: : Graph of 8-PSK in TTCM, BICM & BICM-ID in AWGN [23]

Table 8 gives the overall breakdown of the performance of TCM for three modulation schemes at four different parameters namely; the alphabet size (M), the coding gain, the E_b/N_0 (dB) at 10^{-5} BER and the Power Efficiency(dB).

The table reveals that as M increases for QAM TCM the coding gain and the E_b/N_0 (dB) increases while the power efficiency decreased approaching Shannon’s limit. Similarly, for PSK TCM as M increases all the parameters under consideration also increase. The increase in M shows an increase in number of bits/symbol and leads to conservation of bandwidth (i.e. trading power for bandwidth efficiency). Furthermore, the signal power required is in the order of $QAM < PSK < ASK$.

The increase in coding gain as M increases at a given BER conserves power (E_b/N_0) while transmitting signal or data intelligibly. This ensures that at low power of transmission the number of errors recorded is low and ensures good quality of service. This is evident in the power efficiency graph. It is noticeable that at higher values of M, coding gain is improves further and the power efficiency improves as it approaches Shannon’s specification limit.

The practical implication of coding gain is that high power efficiency is achieved for better system performance while minimising waste of power. Also, it enable Digital Communication design Engineers

choose among different modulation schemes for efficient communication and trade-offs.

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