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APSK Constellation Shaping and LDPC Codes Optimization for DVB-S2 Systems

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***Abstract* - Here an energy-efficient approach is presented for constellation shaping of a bit-interleaved low-density parity-check (LDPC) coded amplitude phase-shift keying (APSK) system. APSK is a modulation consisting of several concentric rings of signals, with each ring containing signals that are separated by a constant phase offset. APSK offers an attractive combination of spectral and energy efficiency, and is well suited for the nonlinear channels. 16 and 32 symbol APSK are studied. A subset of the interleaved bits output by a binary LDPC encoder are passed through a nonlinear shaping encoder. Here only short shaping codes are considered in order to limit complexity. An iterative decoder shares information among the LDPC decoder, APSK de-mapper, and shaping decoder. In this project, an adaptive equalizer is added at the receiver and its purpose is to reduce inter symbol interference to allow recovery of the transmit symbol. An adaptive equalizer one that automatically adapts to time-varying properties of the communication channel. Simulation results showed that the combination of shaping, use of LDPC codes, adaptive equalizer and iterative decoding can achieve a gain in excess of 1.5dB in AWGN compared with a system that does not use shaping and adaptive equalizer, uses an un optimized code from the DVB-S2 standard, and does not iterate between decoder and demodulator.**

***Index Terms*— APSK, BICM-ID, DVB-S2, LDPC.**

I. INTRODUCTION

Satellite communications are providing a key role in the worldwide digital information networks. Among other applications, satellite communications provide the platform for Direct-to-Home (DTH) digital TV broadcasting as well as interactive and subscription TV services, mobile services to ships, aircraft and land-based users, and data distribution within business networks. Satellite networks are essential part of the Internet backbone enabling both broadband and narrowband Internet access services from remote and rural areas where the satellite access provides a unique way to complement the terrestrial telecommunication infrastructure.

The exploitation of highly efficient coding techniques associated with power and spectrally efficient modulation schemes had been designed to operate over the satellite channel environment.

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DVB-S [1] is an abbreviation for "Digital Video Broadcasting – Satellite", is the original Digital Video Broadcasting Forward Error Correction and demodulation standard for Satellite Television and dates from 1995, while development lasted from 1993 to 1997. DVB-S is used in both Multiple Channel Per Carrier (MCPC) and Single Channel per carrier modes for Broadcast Network feeds as well as for Direct Broadcast Satellite. DVB-S is suitable for use on different satellite transponder bandwidths and is compatible with Moving Pictures Experts Group 2 (MPEG 2) coded TV services.

Digital Video Broadcasting - Satellite - Second Generation (DVB-S2) [2] is for broadband satellite applications that has been designed as a successor for the popular DVB-S system. It was developed in 2003. The standard is based on, and improves upon DVB-S and the electronic news-gathering (or Digital Satellite News Gathering) system, used by mobile units for sending sounds and images from remote locations world-wide back to their home television stations. It has been designed for:

- ❖ Broadcast Services for standard definition TV and HDTV.
- ❖ Interactive Services including Internet Access for consumer applications.
- ❖ Professional Applications, such as Digital TV contribution and News Gathering, TV distribution to terrestrial VHF/UHF transmitters, Data Content distribution and Internet trunking.

The DVB-S2 standard has been specified around three key concepts:

- ❖ best transmission performance,
- ❖ total flexibility,
- ❖ reasonable receiver complexity.

To achieve the best performance-complexity trade-off, DVB-S2 [3] benefits from more recent developments in channel coding (adoption of LDPC codes) and four modulation modes QPSK AND 8PSK are proposed for broadcast applications and 16APSK and 32APSK are used mainly for professional, semi-linear applications, but can also be used for broadcasting though they require a higher level of available carrier to noise ratio. The result is typically a 30% capacity increase over DVB-S under the same transmission conditions. In addition, for broadcast applications, DVB-S2 is not constrained to the use of QPSK and therefore it can deliver significantly higher bit rates over high power satellites, thus still increasing capacity gain with respect to DVB-S. DVB-S2 is so flexible that it can cope with any existing satellite transponder characteristics, with a large variety of spectrum efficiencies. Furthermore, it is not limited to MPEG-2 video and audio source coding, but it is designed to handle a variety of audio-video and data formats.

	DVB-S	DVB-S2
Input Interface	Single Transport Stream (TS)	Multiple Transport Stream and Generic Stream Encapsulation
Modes	Constant coding & modulation	Variable coding & modulation and adaptive coding & modulation
FEC	Reed Solomon (RS) 1/2, 2/3, 3/4, 5/6, 7/8	LDPC + BCH 1/4, 1/3, 2/5, 1/2, 3/5, 2/3, 3/4, 4/5, 5/6, 8/9, 9/10
Modulation	Single carrier QPSK	Single carrier QPSK with multiple streams.
Modulation schemes	BPSK, QPSK, 8PSK, 16QAM	BPSK, QPSK, 8PSK, 16APSK, 32APSK
Interleaving	Bit-interleaving	Bit-interleaving

Table 1: Modes and Features of DVB-S2 in Comparison to DVB-S.

This paper compares two systems, the existing and proposed systems. The existing system is one that has no adaptive equalizer in the receiver section. Here output from the AWGN channel goes directly to the APSK demodulator. The proposed system has an adaptive equalizer before the APSK demodulator, in this system output from AWGN channel goes to adaptive equalizer. The SNR and BER graphs shows the comparison of two systems.

II. AMPLITUDE PHASE SHIFT KEYING

Amplitude phase-shift keying (APSK) [4] is a modulation consisting of several concentric rings of signals, with each ring containing signals that are separated by a constant phase offset. APSK has recently become widely adopted, due primarily to its inclusion in the second generation of the Digital Video Broadcasting Satellite standard, DVB-S2, as well as some other standards such as DVB-SH, IPoS, GMR-2 3G, and ABSS. APSK is known to be both spectral and energy efficient, and is especially well suited for nonlinear channels. For a given modulation order *M*, an APSK constellation is characterized by the number of rings, the number of signals in each ring, the relative radii of the rings, and the phase offset of the rings relative to each other. The APSK modulation schemes exist in many variations (4+12 APSK, 5+11APSK, 6+10 APSK, 1+5+10 APSK, 4+12+16, etc.) having the different signal constellation. Among them, 4+12 APSK and 4+12+16 APSK, which are the modulation schemes achieving the most good performance in case of considering the nonlinear characteristics of an amplifier, were adopted in DVB-S2 and they are denoted as 16APSK and 32APSK techniques.

A. 16 symbol APSK shaping

The 16APSK [5] modulation constellation is composed of two concentric rings. The inner ring contains 4 symbols, while the outer ring contains 12 symbols, thus making a total of 16 symbols. The bit mapping is as indicated on the figure. This constellation is identical to the 16-APSK constellation in the DVB-S2 standard. The ratio of inner ring radius $R1$ and outer ring radius $R2$

$$\gamma = \frac{R2}{R1} \tag{1}$$

can be adapted to FEC channel coding method allowing performance optimization according to the channel characteristics. The ratio of the radius of the outer ring to the radius of the inner ring is denoted γ which according to the DVB-S2 standard may assume a value from the set [2.57, 2.60, 2.70, 2.75, 2.85, 3.15].

B. 32 symbol APSK shaping

The 32APSK [5] modulation constellation is constructed from three concentric rings. The inner ring has 4 symbols, the middle ring has 12 symbols and 16 symbols in the outer ring. This constellation is identical to the 32-APSK constellation in the DVB-S2 standard. The inner ring has radius R1, the intermediate ring has radius R2 and the outer ring has radius R3. The ratio of the radius of the middle ring to the radius of the inner ring is denoted γ_1, while the ratio of the radius of the outer ring to the radius of the inner ring is denoted γ_2 ,

$$\gamma 1 = \frac{R2}{R1} \tag{2}$$

and

$$\gamma 2 = \frac{R3}{R1} \tag{3}$$

According to the standard, the value of $\gamma = \{ \gamma_1, \gamma_2 \}$ must be one of the following: {2.53, 4.30}, {2.54, 4.33}, {2.64, 4.64}, {2.72, 4.87}, or {2.84, 5.27}.

III. LDPC CODES AND BICM-ID

A. LDPC codes

In information theory, a low density parity check (LDPC) code is a linear error correcting code, a method of transmitting a message over a noisy transmission channel, and is constructed using a sparse bipartite graph. LDPC [4] codes are capacity approaching codes, which means that practical constructions exist that allow the noise threshold to be set very close to the theoretical maximum (the Shannon limit) for a symmetric memoryless channel. The noise threshold defines an upper bound for the channel noise, up to which the probability of lost information can be made as small as desired. Using iterative belief propagation techniques, LDPC codes can be decoded in time linear to their block length. LDPC codes are finding increasing use in applications requiring reliable and highly efficient information transfer over bandwidth or return channel-constrained links in the presence of corrupting noise. LDPC codes are also known as Gallager codes, in honour of Robert G. Gallager, who developed the LDPC concept in 1960. The crucial innovation was Gallager's introduction of iterative decoding algorithms (or message-passing decoders) which he showed to be capable of achieving a significant fraction of channel capacity at low complexity.

B. BICM-ID

BICM [7] is the serial concatenation of a channel coder, interleaver and mapper and is used in most recent wireless standards due to its simplicity, flexibility and performance. At the receiver, the signal is consecutively demapped, deinterleaved and decoded. The performance of this standard BICM receiver can be greatly improved through iterative information exchange between the demapper and the decoder. BICM can be obtained by using a bit interleaver, π, between an encoder for a binary code and a memoryless modulator. In an iterating process, the feedback from the section which is less affected by the channel noise removes the ambiguity in the higher-order demodulation and enhances the decoding of the weak data sections. Iterative decoding of BICM not only increases the inter subset Euclidean distance, but also reduces the number of nearest neighbours. This leads to a significant improvement over both AWGN and fading channels.

The interleaver is critical to the high performance of BICM-ID [8] systems. The key idea in the design of a good interleaver for a BICM-ID system is to make the interleaved coded bits in the same channel symbol as far apart as possible. Two design objectives of a good interleaver are as follows:

- To increase the minimum Euclidean distance between any two coded sequences.
- To mitigate the error propagation during the iterative decoding.

A small block size might cause a substantial performance degradation to BICM-ID. It should be noted that increasing the block length leads to earlier convergence, in terms of both the SNR and the number of iterations. The reason for this is the reduction in the number of nearest neighbour codewords when the interleaver size is increased. However, increasing the interleaver size does not improve the performance of BICM-ID systems after a certain high SNR level.

IV. SYSTEM MODEL

A. Transmitter section

Figure 1 shows the transmitter section. The information bits from DVB-S2 had been encoded using a LDPC encoder. The encoded codewords are passed through a bit interleaver and the interleaved codeword is separated into two groups using a bit separator.

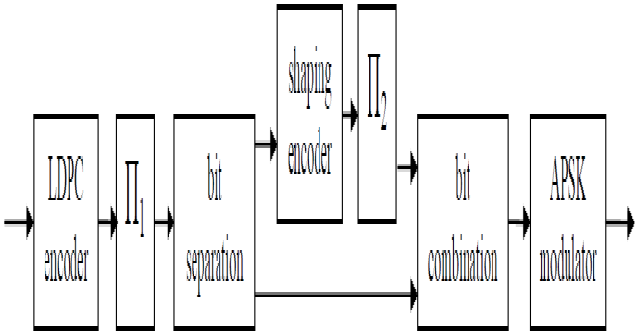

Figure 1: Transmitter section

One group is passed through a shaping encoder and the shaping used is 16 symbol and 32 symbol APSK. The shaped output is then interleaved by a second bit interleaver and passed to the bit combiner. The other group directly goes to the bit combiner and there the shaped bits and the unshaped bits are combined and it then goes to the APSK modulator. The modulated signal is transmitted through the AWGN channel. Noise is added when signal is transmitted through the channel.

LDPC Encoder

The LDPC encoder takes information from the DVB-S2 [9] system. Here, a syntax of Matlab is used. The syntax is H=dvbs2ldpc(r) returns the parity-check matrix of the LDPC code with code rate r from the DVB-S.2 standard. Possible values for r are 1/4, 1/3, 2/5, 1/2, 3/5, 2/3, 3/4, 4/5, 5/6, 8/9, and 9/10. The block length of the code is 64800. LDPC encoder constructs a binary symmetric LDPC code with a parity check matrix H. H must be a sparse zero-one matrix. N and N-K are the number of columns and rows in H. The last N-K columns in H must be an invertible matrix in GF(2). An LDPC encoder function in Matlab has the following properties. Only Parity Check Matrix is writable. All other properties are derived from it.
ParityCheckMatrix – H, stored as a sparse logical matrix.
Block length – N, total number of bits in a codeword.
NumInfoBits – K, number of information bits in codeword.
NumParityBits – N-K, number of parity bits in a codeword.

Bit Interleaver

Interleaving is frequently used in digital communication e systems to improve the performance of forward error correcting codes. If the number of errors within a codeword exceeds the error-correcting code's capability, it fails to recover the original code word. Interleaving ameliorates this problem by shuffling source symbols across several code words, thereby creating a more uniform distribution of errors. Therefore, interleaving is widely used for burst error-correction. The analysis of modern iterated codes, like turbo codes and LDPC codes, typically assumes an independent distribution of errors. Systems using LDPC codes therefore typically employ additional interleaving across the symbols within a code word.

Bit Separator

This block just separates the interleaved output into two sets. One set goes to shaping encoder and other set goes directly to bit combiner.

Shaping Encoder

The idea behind constellation shaping is that signals with large norm are used less frequently than signals with small norm, thus improving the overall gain by adding shaping gain to their original coding gain. The non-uniform signalling reduces the entropy of the transmitter output, and hence, the average bit rate. However, if points with small energy are chosen more often than points with large one, energy savings may compensate for this loss in bit rate. Theoretically, constellation points would be selected according to a continuous Gaussian distribution at every dimension, and thus achieve the maximum shaping gain. Here in this two shaping strategies are considered. They are 16 APSK shaping and 32 APSK shaping.

APSK modulator

Amplitude and phase-shift keying or asymmetric phase-shift keying (APSK), is a digital modulation scheme that conveys data by changing, or modulating, both the amplitude and the phase of a reference signal (the carrier wave). In other words, it combines both Amplitude-shift keying (ASK) and Phase-shift keying (PSK) to increase the symbol-set. It can be considered as a superclass of Quadrature Amplitude Modulation (QAM). The advantage over conventional QAM, for example 16-QAM, is lower number of possible amplitude levels, resulting in fewer problems with non-linear amplifiers. **T**he DVB-S2 [10] specification permits the use of 16APSK and 32APSK modes, allowing 16 and 32 different symbols respectively and are intended for mainly professional, semi-linear applications. The APSK modulator accepts the bit combiner output which consists of shaped and unshaped symbols and modulates it.

B. Receiver section

The receiver section is shown in figure 2.

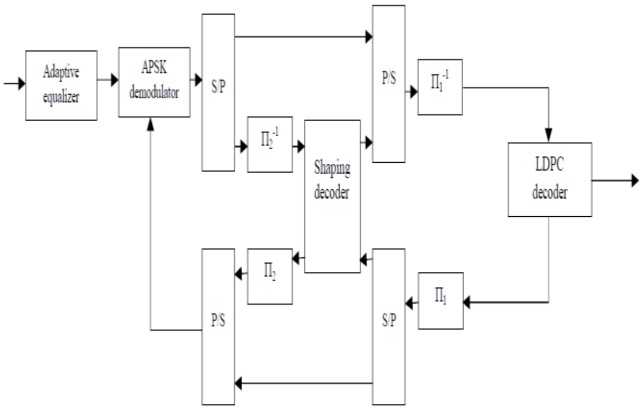

Figure 2: Receiver section

Adaptive Equalizer

The received signal goes to an adaptive equalizer. The purpose from using an adaptive equalizer is to compensate for the distortion and eliminate the inter-symbol interference [ISI] caused by the channel noise. Initially, a desired signal is defined for the adaptive equalizer. Then, the error estimation is calculated by making the difference between the equalizer output and the desired signal. The produced error is used to update the coefficients of the equalizer to reach its optimum values. The received signal is obtained by applying the equalizer output to a decision device. Here a BLMS (Block implementation of LMS) equalizer is used. It updates the equalizer parameters as it processes the data. A problem in DFE (Decision Feedback Equalizer) is the huge computation complexity which is due to the long feedback part of the DFE. In this solution the signal is divided into blocks. To get small delays, the length of the block can be much smaller than the filter's order. The LMS algorithm is convenient due to its computational simplicity and thus BLMS has better performance than DFE.

The output signal from the adaptive equalizer goes to an APSK demodulator and the output signal is passed through a serial to parallel converter and the output will be split into two sets. One set passes through a de-interleaver and then to a shaping decoder and its output to a parallel to serial converter. At the same time, other set of signal directly goes to the parallel to serial converter. The output from this converter is again de-interleaved and then it goes to LDPC decoder. The LDPC decoded output is then bit interleaved and passed through serial to parallel converter. The converted output is split into two sets. One set goes to shaping decoder then to a bit interleaver and then to parallel to serial converter, while the other set directly goes to this converter. Then its output goes to an APSK demodulator. This is the iterative process between decoding and demodulator.

V. SIMULATION RESULTS

16 APSK SIGNALS

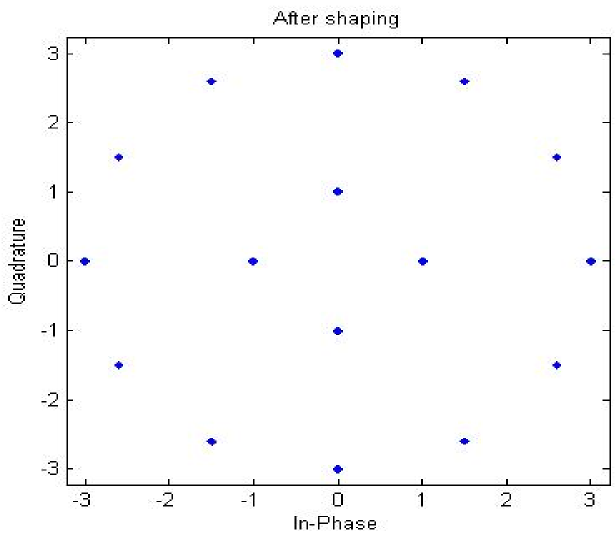

Figure 3: Constellation diagram after shaping.

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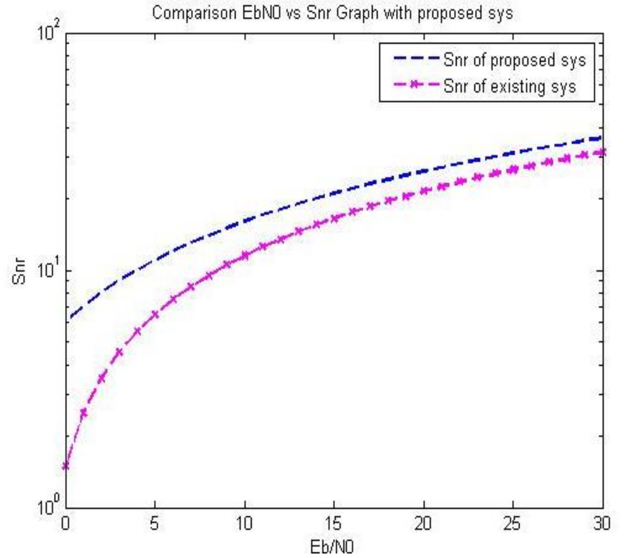

Figure 4: Snr comparison of existing and proposed systems.

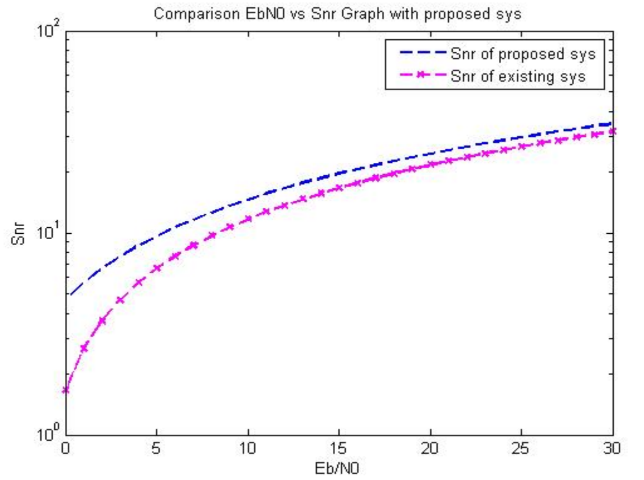

Figure 7:Snr comparison of existing and proposed systems.

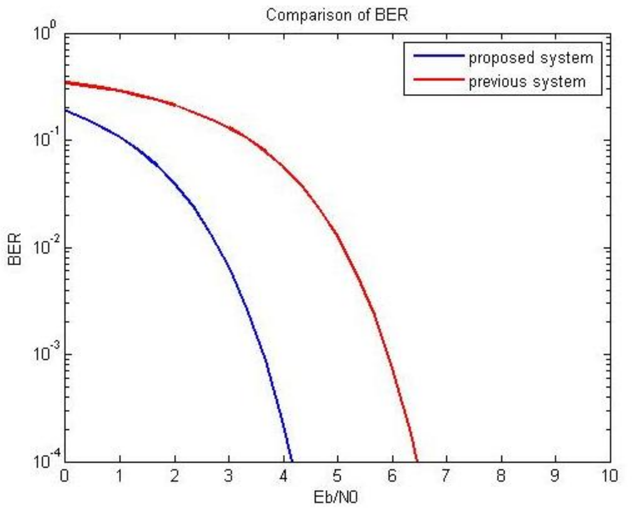

Figure 5: BER comparison of proposed and existing systems.

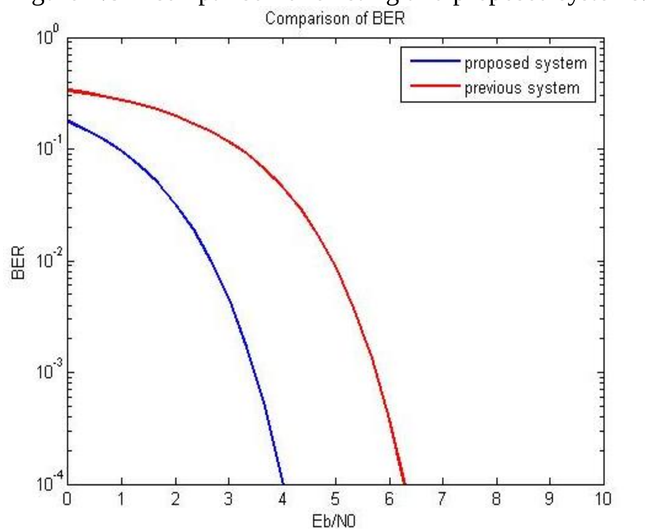

Figure 8: BER comparison of proposed and existing systems.

32 APSK SIGNALS

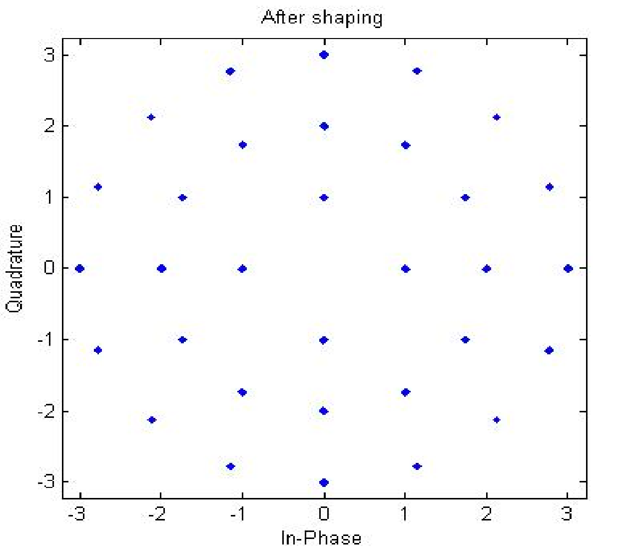

Figure 6: Constellation diagram after shaping

VI. CONCLUSION

The main objective of the project is to shape the APSK constellation diagram and to remove the ISI which had been added to the signal. This system can be used in DVB-S2 systems. The LDPC codes from DVB-S2 system is used here. Using LDPC encoder, bit interleavers, shaping encoder and APSK modulator the constellation shaping can be attained. At the receiver side, equalization, APSK demodulation, shaping decoding, LDPC decoding and an iterative feedback is done. The principle used here is BICM-ID and because of this iterative decoding the performance of this system can be improved over the BICM where the iterative process is not used. The use of the adaptive equalizer adds additional gain to this system. In case of 16 APSK and 32 APSK errors in proposed system decreased compared to existing system, thus the performance of the proposed system increased. 32 APSK has less bit error rate (BER) compared to 16 APSK. Simulation results proved that with the constellation shaping, BICM-ID, and the use of LDPC codes and adaptive equalizer a gain of more than 1.4 dB is achieved.

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