PERFORMANCE OF ASYMMETRICAL DIGITAL SUBSCRIBER LINE (ADSL)SYSTEM IN PRESENCE OF IMPULSE NOISE [+]

أداء منظومة خط المشتركين الرقمي اللاتناظري بوجود الضوضاء النبضية

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Abstract

A digital subscriber line (DSL) technology provides transport of high-bit rate digital information over telephone lines. This high-speed digital transmission requires advanced signal processing to overcome transmission impairments resulting from: crosstalk noise from the signals present on the other wires in the same binder, radio noise and impulse noise. Impulse noise (IN) is nonstationary crosstalk from temporary electromagnetic events in the vicinity of phone lines. The effects are temporary and typically at much lower frequencies with higher power level and reduces the system performance if cannot be overcome to it.

In this paper we focus on this effect on ADSL system for two channels (fast channel and interleaved channel).After that we study the performance of the system by calculating the probability of error (Pe) with and without impulse noise for two channels and comparing between them.

The system was simulated by using MATLAB simulation tools, from the simulation results the Pe was found 6.4449×10^{-6} and 7.3472×10^{-5} for fast channel with the effect of AWGN disturbance and IN disturbance respectively at 26dB energy signal to noise ratio (Es/No) whereas the Pe was found 6.1242×10^{-6} for interleaved channel with the effect of AWGN disturbance and IN disturbance at 26dB energy signal to noise ratio (Es/No).Within the paper an overview is provided about the major building block of the simulator and some analysis results are present by depending on MATLAB simulink tools.

المستخلص:

أن تقنية خط المشترك الرقمي(DSL) تمكننا من نقل معلومات رقمية بسرعة عالية عبر خطوط الهاتف السلكية المعتادة . وأن هذه السرعة العالية تتطلب معالج أشارة متقدم للتغلب على معوقات ألارسال الناتج من: ضوضاء الحديث المتبادل بين المشتركين لخطوط أخرى والموجود ضمن نفس حزمة ألاسلاك ، الضوضاء ألاذاعية وكذلك الضوضاء النبضية (IN) وان هذه الضوضاء غير ثابتة لأنها ناتجة من المجالات الكهرومغناطسية المؤقتة والتي تكون على مقربة من خطوط الهاتف وأن هذا التأثير وقتي ذو تردد قليل وقدرة عالية بحيث يقلل من أداء المنظومة أذا لم يتم التغلب عليه.في هذا البحث سوف نركز على تاثيرالضوضاء النبضية (IN) على منظومة خط المشتركين الرقمي اللاتناظري (ADSL) ولقناتين [قناة سريعة (Fast) وقناة مضفورة (Interleaved)] وندرس أداء هذه المنظومة بواسطة أحتساب معدل الخطاء(Pe) للبيانات المرسلة تحت تاثيرهذه الضوضاء مرة وبدون هذه الضوضاء مرة اخرى ومن ثم مقارنتها معا .تم عمل محاكاة للنظام باستعمال ادوات برنامج MATLAB حيث وجد

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بان معدل الخطاء يساوي $6.4449x10^{-6}$ و $7.3472x10^{-5}$ للقناة السريعة بتاثير (ضوضاء قناة كـــاوس AWGN وبتاثير الضوضاء النبضية) على التوالي ولنسبة طاقة الاشارة الى الضوضاء بمقدار 26dB بينما وجد معدل الخطاء يساوي $6.1242x10^{-6}$ للقناة المضفورة بتاثير(ضوضاء قناة كاوس AWGN وبتاثير الضوضاء النبـــضية) علــــى التوالي ولنسبة طاقة الاشارة الى الضوضاء بمقدار 26dB . وايضا نعطي نظرة عامة حول بنــــاء ألاجزاءالرئيـــسية لنظام المحاكاة وسوف نقدم بعض التحليلات والنتائج بالأعتماد على أدوات برنامجMATLAB.

Introduction:

Digital Subscriber Line (DSL) is an always-on internet connection that uses your existing telephone line and gives you an internet connection several times faster than a traditional dial-up account .There are two general categories of DSL: symmetric and asymmetric. Symmetric DSL provides the same service bit-rate in both directions upstream (from the user to the network) and downstream (from the network to the user). Asymmetric DSL provides downstream bit-rate more than upstream bit-rate. DSL service has a fixed cost and allows you to talk on the telephone and be on the internet at the same time, using the same telephone line. Typical telephone cabling is capable of supporting a greater range of frequencies (around 1MHz).

With DSL modems, the digital signal is not limited to 4 kHz of voice frequencies, as it does not need to travel through the telephone switching system.DSL modems enable up to 1MHz of bandwidth to be used for transmitting digital (data) along side analogue (voice) signals on the same wire by separating the signals, thereby preventing the signals from interfering with each other. Although ADSL (Asymmetric Digital Subscriber Line) transmission systems are widely used today in many countries as the primary high-speed service access solution for residential and small office customers there remain open questions in terms of quality of service and transmission performance. One of them is that most ADSL performance models [1] cover stationary error sources like crosstalk, radio interference and background noise but don't take into account impulsive noise as a typical non-stationary impairment. The reason might be that "the causes of impulses are so diverse that any distillation for engineering and measurement purposes necessarily has some bias"[2]. Nevertheless impulsive noise is injected into copper lines often with a very high magnitude compared to background noise level and contains spectral components which overlap with the used ADSL frequency bands. Therefore this electromagnetic disturbance cannot be neglected and needs to be investigated in order to determine the resulting impairments on the bit-error statistics of the transmission system and to search for specific optimizations which cope with this phenomenon.

Available research results on this topic were published in [3,4]. An approach presented in [3] doesn't consider the spectral properties of impulsive noise, but simply models it as a white noise process.Hence correlation between errors in different tones (e.g. harmonic oscillations, disturbances in neighboring tones) cannot be covered by this model. In [4] a simulation tool is described which during impulse duration superimposes the QAM symbol constellation of every DMT tone with an impulsive noise component of a certain magnitude and phase, which must be known. However it is difficult to extract these parameters from spectral analysis because they may considerably vary during impulse duration [5]. A possibility to overcome these problems is to use time-domain impulse waveforms for the analysis, which requires IDFT/DFT modulation algorithms to be included into the simulator.

In this paper the effect of impulse noise was taken on the **256-ADSL** system, this system will be described details in Section two witch provides an overview about the architecture of the real ADSL system, Section three provides an overview about the modulation types used in

ADSL system and section four provides the major building blocks of the ADSL simulation. After that in the fifth section some analysis results are presented which contain bit–error rat calculations for fast and interleaved channel of an ADSL system under the effect of impulse noise disturbance and AWGN disturbance.

ADSL System:

Asymmetric Digital Subscriber Line technology is a transmission technology that can support high digital data rates over a pair of wires known as the twisted pairs. While the plain old telephone service such as telephones and fax machines are being used, the ADSL signals can still be transmitted at higher frequencies (not used by POTS devices) in the frequency spectrum [6]. This technology is a full duplex transmission system meaning that data can be sent in both directions simultaneously. Most of the ADSL standards focus on the DMT modulation [7] which is the modulation that we are using for this paper. Generally, ADSL system shown in figure (1), the DMT modulation consists of QAM data transmitted on 256 equally 4.3125 KHz spaced channels using a bandwidth of 1.104 MHz Since the downstream and upstream data rates are not the same (8 Mb/s downstream and 800 kb/s upstream), there are only 32 channels used when transmitting upstream.

There is more impairments effect on ADSL channel like: crosstalk, line attenuation, frequency interference and impulse noise, this paper focus on impulse noise impairment. Impulse noise is a short burst of relative high energy noise that can potentially cause errors [8]. The examples of impulse noise could be as common as ringing of the telephone, turning switch on and off etc. The voltage of these impulse noises is small when compared to normal signal voltage levels. However, when there is high signal attenuation at high frequencies, these noises can be quite significant at the receiver hence causing reception errors. The system can be protected from impulse noise by interleaving.

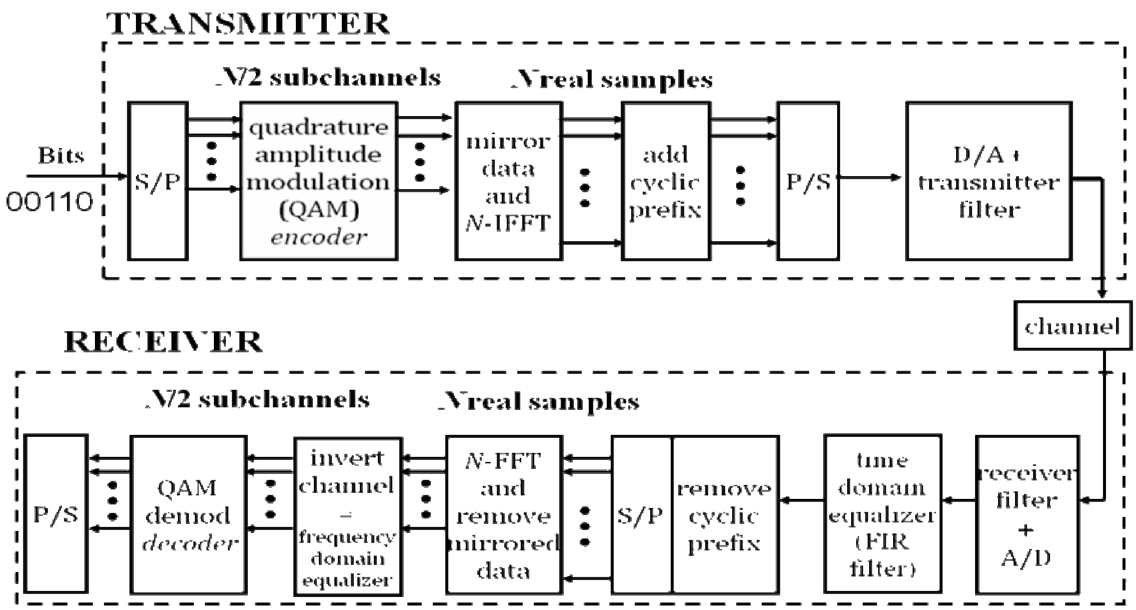

Figure 1: Block diagram of ADSL system.

Modulation Techniques:

In the present day, there are two main competing line modulation techniques to conserve bandwidth in ADSL implementations, which are Carrierless Amplitude/Phase (CAP) modulation [9,10] and Discrete Multi-tone (DMT) modulation which its relatives to QAM[11].

DMT modulation is commonly used in ADSL implementations, which has been ratified by ANSI as a standard coding for ADSL. Following two sections discussed the details of QAM and DMT modulation.

1. **QAM (Quadrature Amplitude Modulation)**

Figure (2) below shows an example of a QAM modulation system that can send two binary data bits per QAM symbol. The two binary data bits allow 4 possible points for any combination of bits and those 4 points are then constructed a QAM constellation; this is commonly called 4-QAM. In general, other numbers of bits can be transmitted per QAM symbol. For example, if 4 bit per symbol were transmitted as shown in Figure (3), a 16 points constellation is formed, which is called 16-QAM [12].

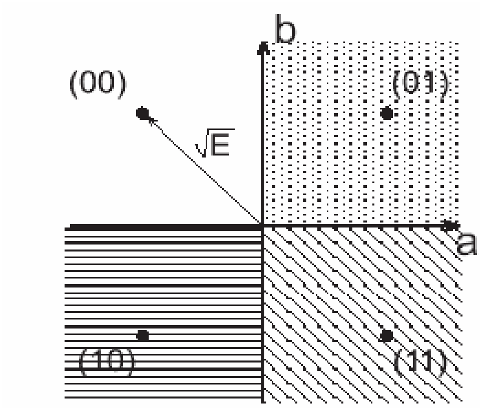

Figure2: An example of a 4-QAM system sending two bits per symbol per channel.

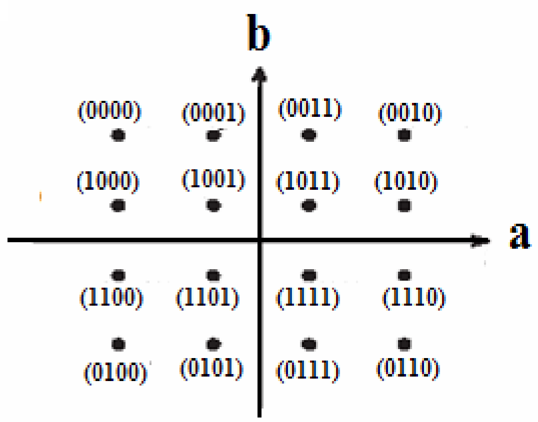

Figure 3: An example of a 16-QAM system sending four bits per symbol per channel.

2. DMT Modulation

The modulation technique that has become standard for ADSL is called the discrete multitone technique (DMT) as shown in figuer (4) which combines QAM and FDM [13], the ANSI selected DMT modulation over QAM and CAP for 3 reasons:

1. Better transmission performance than QAM.
2. Easier implementation than the CAP methods
3. Bandwidth-on-demand flexibility.

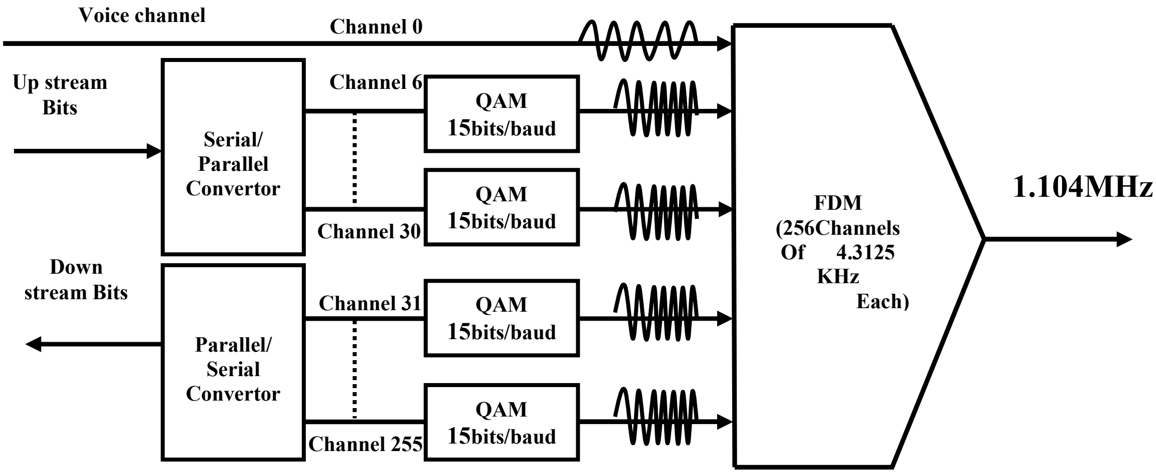

Figure 4: DMT modulation splits the channels into 256 channels.

Figure (5) below shows how the bandwidth can be divided into the following [13]:-
1-Voice Channel 0 is reserved for voice communication.
2- Idle Channels 1 to 5 are not used, to allow a gap between voice and data communication.
3- Upstream data and control channels 6 to 30 (25 channels) are used for upstream data transfer
and control .One channel is for control , an 24 channels are for data transfer .If there are 24 channels ,each using 4KHz(out of 4.3125KHz available) with QAM modulation , we have 24x4000x15, or a 1.44-Mbps bandwidth ,in the upstream direction.
4- Downstream data and control Channels 31 to 255 (225 channels) are used for downstream data transfer and control. One channel is for control, and 224 channels for data .If there are 224 channel, we can achieve up to 224x4000x15 or , 13.44 Mbps bandwidth in downstream direction.

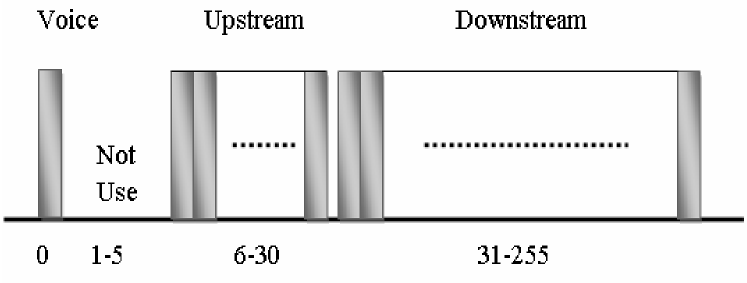

Figure 5: bandwidth division.

The bit error rate for ADSL system:

In conventional (one tone) QAM the signal occupies the channel bandwidth W and M points in the signal constellation are used; there are $\sqrt{M}$ points on each one-dimensional edge of the two-dimensional MQAM constellation. This is shown in Figure (3) in section 3.1.
For this one-tone MQAM system, there is a maximum bit rate which can be supported at a given error rate for a specified SNR. That is, there is a maximum value of M because the error

probability increases as *M* is increased. It is known t hat the symbol error probability for MQAM with received power P is given by [14].

$$\ldots\ldots(1) P_r(e) = 4\left(1-\frac{1}{\sqrt{M}}\right) Q\left[\sqrt{\frac{3}{M-1}\,\frac{P}{WN_o}}\right]$$

Where *Q(.)* is the Gaussian Q function defined as[15]:

$$\ldots\ldots(2) Q(x) = \int_x^\infty \frac{e^{-y^2/2}}{\sqrt{2\pi}}\, dy$$

The ADSL Simulation:

The ADSL simulation tool comprises the modulation/demodulation and coding/decoding functions of an ITU-T G.992.1, compliant transceiver together with a transmission channel model with some modification as illustrated in Figure (6), the ADSL simulation based on MATLAB simulik, contains the following parts:-

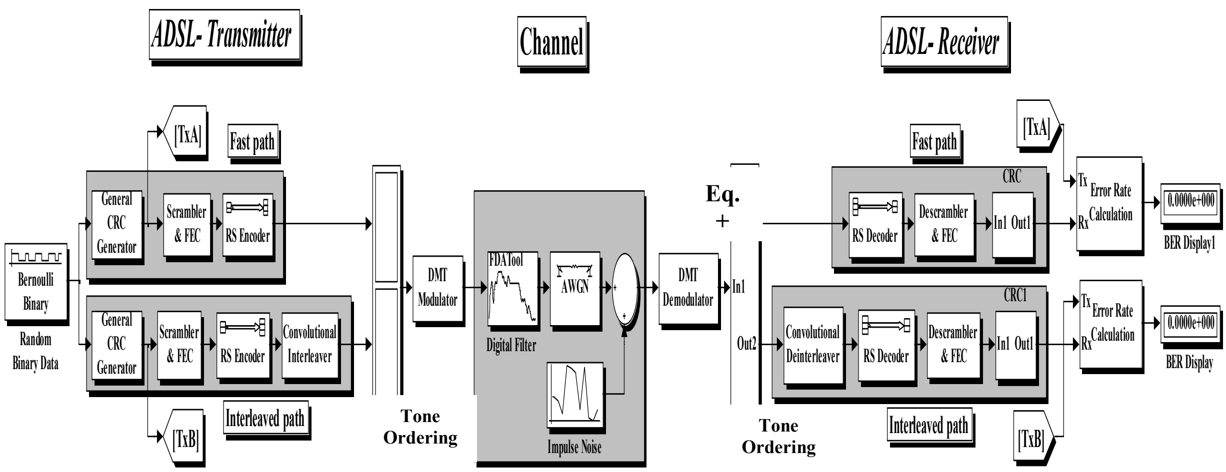

Figure 6: Simulation diagram of ADSL system.

1. The Transmitter

Two parallel data paths have been implemented in the model, the first path **is a fast channel (non-interleaved)** providing low delay for real-time applications this path processes the first 766-bits of each 1552-bit data frame. The fast channel is composed of a CRC generator, a self-synchronizing scrambler and a shortened [N, K] Reed-Solomon (**RS**) encoder with codeword length N and message length K. The CRC generator serves for calculation of a checksum over the entire frame which is used for frame error detection at the receiver. The scrambler randomizes the data stream e.g. in order to warrant a time-invariant transmission power. The byte-oriented RS encoder has the capability to correct up to (N-K)/2 bytes of a codeword, at the expense of lengthening the frame by N-K bytes and thus of requiring a higher transmission rate.

While the second path the interleaved channel improving transmission quality for error-sensitive applications it processes the last 776-bit frame.

Data transmitted through using this interleaved path will have much better immunity to impulse noises. However, the downside is that the data signals will be delayed accordingly. Hence the data signals requiring low latency transfer can be assigned to the fast data path without having the interleaving process. The interleaved channel contains additionally a convolutional interleaver which reorders the bytes of one codeword over D codeword's in the transmission stream (D is the interleaver depth).

Both downstream (high data rate of some Mbit/s) and upstream (low data rate of some hundred Kbit/s), transmission scenarios can be modeled by loading the DMT tones at corresponding frequencies with a bit number between 2 and 15 according to the selected data rate.

Each path appends eight cyclic redundancy check (CRC) bits to its 776-bit frame, scrambles the bits, and encodes them using a shortened Reed –Solomon (**RS**) code. The scrambling and encoding operations interpret the bits as integers between 0 and 127. In the second path but not the first, a convolutional interleaver block interleaves the encoded data. This interleaving operation increases the second path's resistance to burst errors but also its latency .The data from the two routes is concatenated and modulated.
Data from the fast buffer is modulated to the low frequency subcarriers, while data from the interleaved buffer is modulated to the high frequency subcarriers, according to the bit allocations vector as shown in figure (7). The DMT contains 16 Modulator Bank represents a set of 16 Rectangular QAM Modulator Baseband blocks. The DMT technique allocates different numbers of bits to different subchannels. Each copy of the modulator block acts as a distinct subchannel, and uses the 256-element vector. The power spectral density of transmitted signal shown in figure (8).

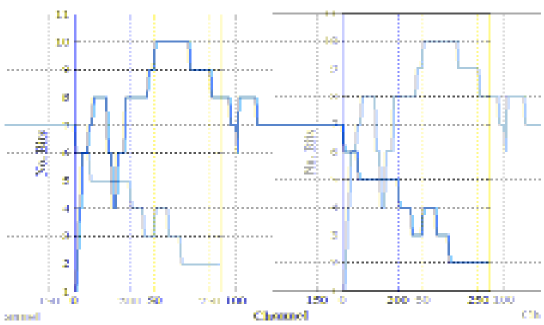

Figure 7: Bit allocations vector (256 subchannels transmitted).

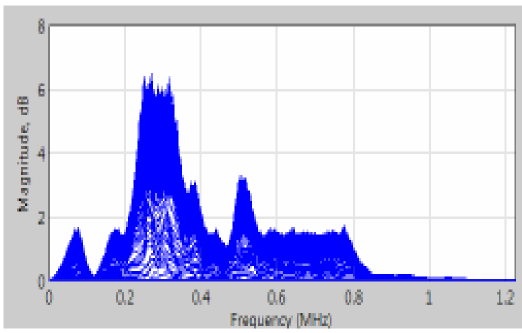

Figure 8: Power spectral density of transmitted signal.

2. <u>The Channel</u>

The ADSL transmitter is transmitted the signal over twisted -pair cable has AWGN with E_s/N_o=26d B and adding impulse noise (IN) to channel.
For impulse-noise (IN) modeling, we propose to use a statistical impulse noise generator as shown in block diagram in figure (9) which is especially useful for fast equipment tests, or a stochastic generator, which should be preferred for performance tests.

The beginning of an impulse event is triggered by the inter-arrival time generator and the length of an impulse event is determined by the length generator. Only during the impulse event, the voltage generator is active and supplies random voltage samples. The voltage sequence representing the impulse event may be windowed by a non-rectangular window of the length provided by the length generator.

The inter-arrival times are exponentially distributed [16, 17]:

$$\ldots\ldots (3)\, f(t) = \lambda\, e^{-\lambda t}$$

Where λ is the mean impulse rate. The mean impulse rate is proposed to be a variable parameter and should be less than 20 kHz.

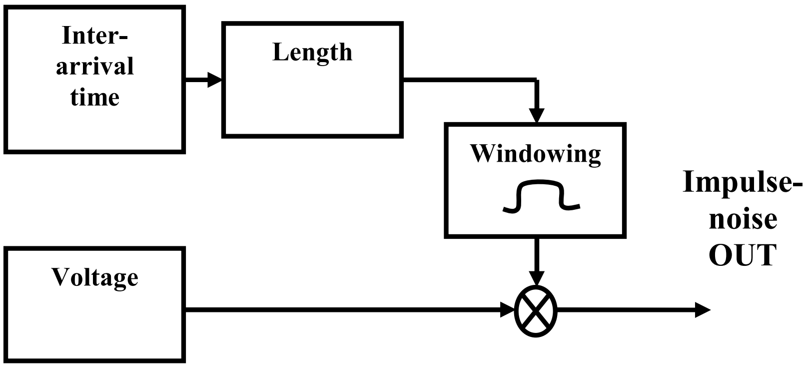

Figure 9: Block diagram of a statistical IN generator.

The impulse-lengths are distributed according to the superposition of log-normal densities in the following equation [16]:

$$f_1(t) = \frac{1}{\sqrt{2\pi}} \left[\frac{B}{s_1 t}\, e^{\frac{-1}{2S_1^2} ln^2 \left(t/_{t_1} \right)} + \frac{(1-B)}{s_2 t}\, e^{\frac{-1}{2S_2^2} ln^2 \left(t/_{t_2} \right)} \right] \quad \ldots\ldots(4)$$

Where B, S_1, t_1 and S_2 ,t_2 represented measured parameters for customer side and central office side respectively. The density of the voltage generator is given by the following equation [17]:

$$f_i\ (u) = \frac{1}{240\, u_0}\ e^{-|u/u_0|^{1/5}} \quad \ldots\ldots (5)$$

Where u_o typical value between 3nV to 18nV.

In the current approach the generation of impulse noise is modeled by white noise with lower frequency and a higher power level as shown in figure (10) and figure(11) which are represent power spectral density (PSD) and timing diagram of IN respectively. Impulse noise which is still a rather rough approximation because it does not incorporate the amplitude and the spectral characteristics of impulsive noise as explored in reference [18].

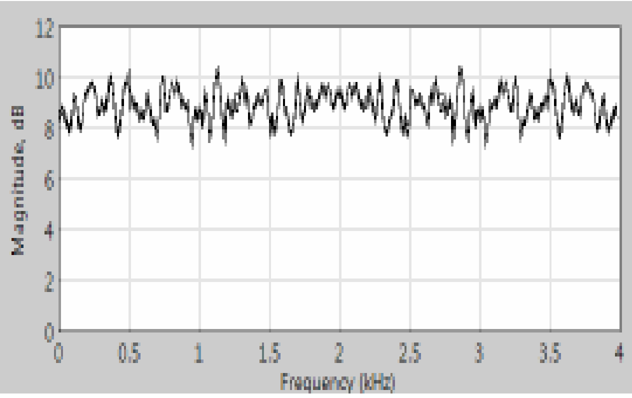

Figure 10: Power spectral density of IN.

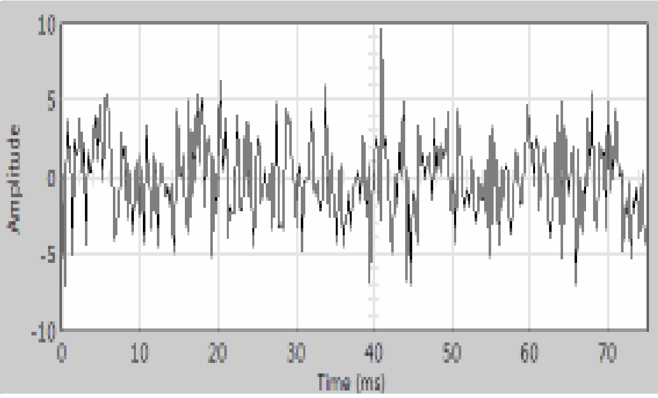

Figure 11: Timing diagram of IN.

3 The Receiver

At the receiver side at first , remove the cyclic prefix (CP) then Discrete Fourier Transform (DFT) block converts the received signal back to frequency domain .Subsequently linear equalizer increases the gain of subcarriers reciprocally to their attenuation within the transmission channel as shown in figure (1) in section2.Now the complex symbols for each subcarrier can be extracted .The QAM demodulator provides the received bits for the corresponding tone. The resulting bit stream of all subcarriers is then demultiplexed into the fast channel and the interleaved channel where de-interleaver, RS decoder and descrambler do the decoding functions .This procedure has the advantage of distributing burst errors which occur during transmission of one codeword over several codeword's which increases the probability to correct these errors by FEC (Forward Error Correction) in the RS decoder. However some time is required for byte reordering, thus the interleaving function generates a significant delay of up to 16 ms.

The receiver attempts to undo each operation that the transmitter performs. Much of the receiver’s design is straight forward; for example, to undo the actions of the convolutional interleaver block, use a convolutional de-interleaver block with the same mask parameters. The frequency domain equalizer in the DMT demodulator subsystem mitigates the channel distortion. The power spectral density of received signal shown in figure (12) below, it’s clear that the effect of impulse noise at low frequency with high power level when compared with transmitted signal in figure (8).

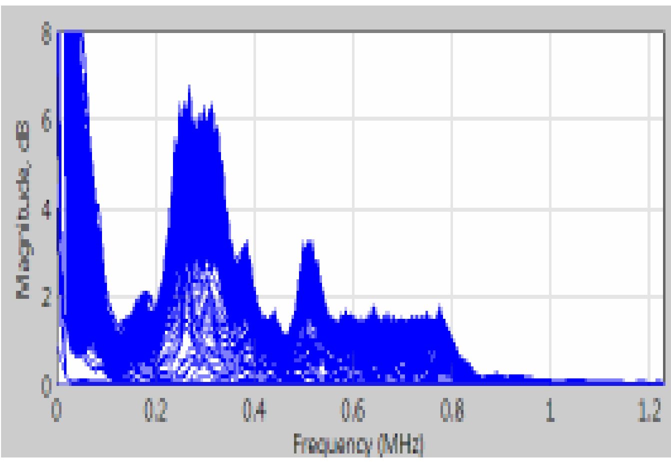

Figure 12: Power spectral density of received signal.

Simulation results

When the simulation system shown in figure (6) started simulation process, after (1000 ms) simulation time, the number of bits approximately was 3 million bits for each E_s/N_o reading .The bit error rate BER was calculated by using error rate calculation block from (communication block set for MATLAB-simulink) for different cases of energy signal to noise ratio E_s/N_o from AWGN block in channel part as shown in simulation diagram. The BER calculated for two cases: The first case with impulse noise disturbance and the second case without impulse noise disturbance (AWGN disturbance only).The results of these calculations for two cases are listed in table(1).

Table (1) the results of Es/N ratio with respect to BER for Fast and Interleaved channel with and without impulse noise (IN).

E_S/N (dB)	AWGN Disturbance		Impulse Noise (IN) Disturbance	
	Fast channel BER	Interleaved channel BER	Fast channel BER	Interleaved channel BER
0	4.9389×10^{-1}	4.9707×10^{-1}	4.9386×10^{-1}	4.9710×10^{-1}
2	4.9000×10^{-1}	4.9450×10^{-1}	4.8997×10^{-1}	4.9450×10^{-1}
4	4.8326×10^{-1}	4.8983×10^{-1}	4.8319×10^{-1}	4.8989×10^{-1}
6	4.7321×10^{-1}	4.8266×10^{-1}	4.7318×10^{-1}	4.8267×10^{-1}
8	4.5686×10^{-1}	4.6974×10^{-1}	4.5687×10^{-1}	4.6975×10^{-1}
10	4.3151×10^{-1}	4.4863×10^{-1}	4.3153×10^{-1}	4.4864×10^{-1}
12	3.9489×10^{-1}	4.1615×10^{-1}	3.9509×10^{-1}	4.1615×10^{-1}
14	3.4601×10^{-1}	3.7100×10^{-1}	3.4639×10^{-1}	3.7100×10^{-1}
16	2.8440×10^{-1}	3.1136×10^{-1}	2.8493×10^{-1}	3.1137×10^{-1}
18	2.0914×10^{-1}	2.3579×10^{-1}	2.1003×10^{-1}	2.3578×10^{-1}
20	1.2877×10^{-1}	1.4869×10^{-1}	1.3008×10^{-1}	1.4869×10^{-1}
22	6.0322×10^{-2}	6.9835×10^{-2}	6.2045×10^{-2}	6.9835×10^{-2}
24	7.3137×10^{-3}	9.6311×10^{-3}	9.9861×10^{-3}	9.6311×10^{-3}
26	6.4449×10^{-6}	6.1242×10^{-6}	7.3472×10^{-5}	6.1242×10^{-6}

From the above table can notice the probability of error convergent at lower values of E_s/N_o but the probability of error divergent at higher values of E_s/N_o.

Figure (13) shows the relationship between E_s/N_o and BER for fast and interleaved channel for AWGN disturbance, the interleaved channel had better probability of error than fast channel (non-interleaved) especially at greater than 26dB energy signal to noise ratio (E_s/N_o), as result the BER for interleaved channel equal 6.1242×10^{-6} and for fast channel equal 6.4449×10^{-6} at 26 dB for E_s/N_o .

When the effect of impulse noise (IN) disturbance was taken on the system, the probability of error for interleaved channel was better than the fast channel as shown in figure (14) which shows the probability of error for interleaved channel equal 6.1242×10^{-6} and for fast channel equal 7.3472×10^{-5} at 26 dB for E_s/N_o.

Figure (15) shows the comparison between the effect of (AWGN and IN) disturbance on the fast channel .The effect of IN disturbance increased the probability of error from 6.4449×10^{-6} to 7.3472×10^{-5} at 26 dB for E_s/N_o.

Figure (16) shows the comparison between the effect of (AWGN and IN) disturbance on the interleaved channel. It's found that the Pe was extremely closer for two cases.

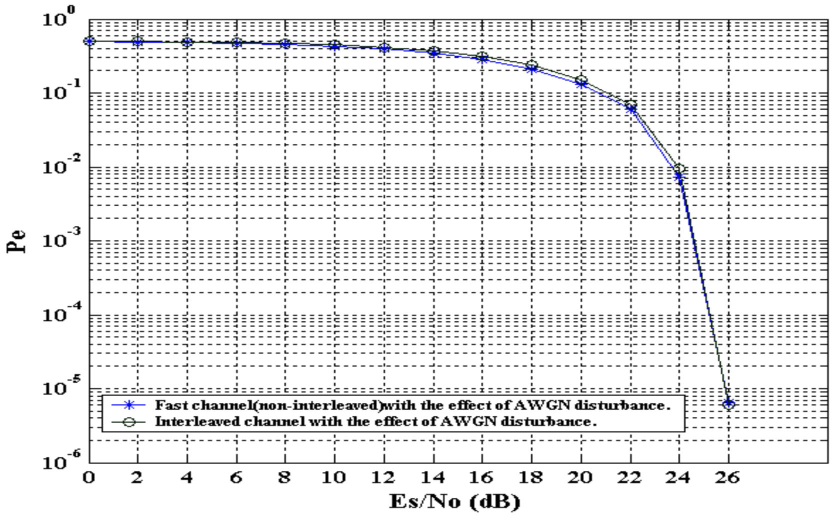

Figure 13: The probability of error for (Fast and interleaved) channel with the effect of AWGN.

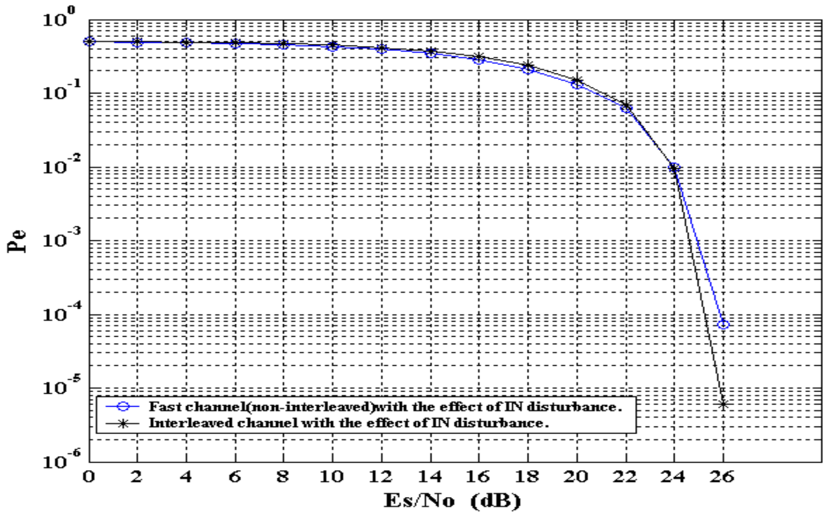

Figure 14: The probability of error for (fast and interleaved) channels with the effect of IN.

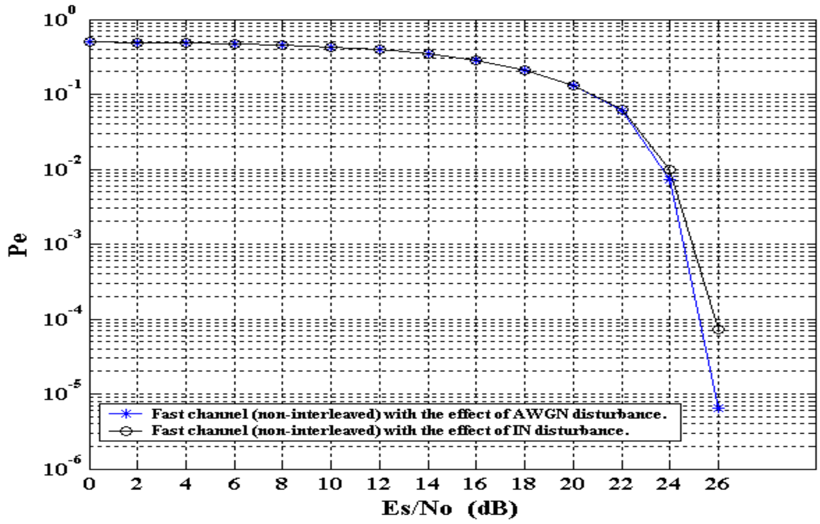

Figure 15: The probability of error for fast Channel with and without effect of IN.

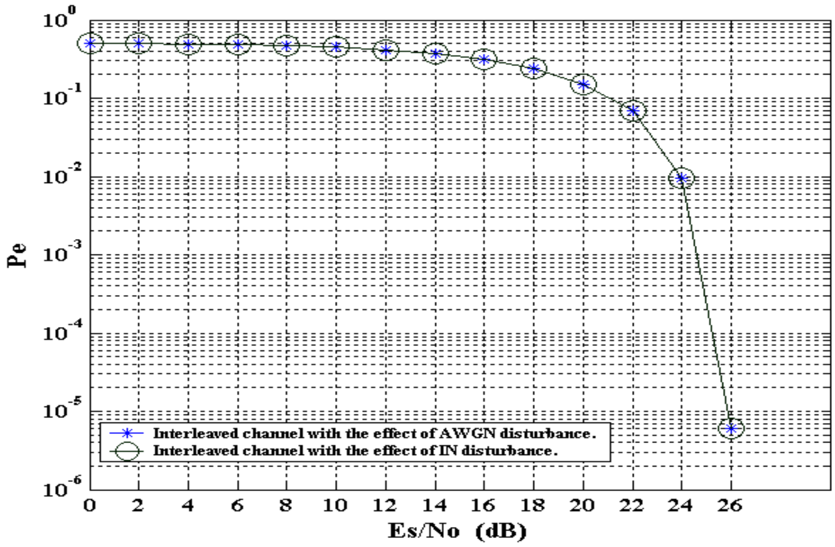

Figure 16: The probability of error for interleaved channel with and without effect of IN.

<u>Conclusions</u>

1-The simulation results shows that the performance of the system was affected by the impulse noise disturbance.

2-The fast channel had high probability of error 7.3472 x10^{-5} than interleaved channel which had low probability of error 6.1242x10^{-6} in presence of impulse noise disturbance.

3-The impulse noise disturbance had less effect on the interleaved channel and the interleaved channel provides lower error in two cases, these results lead us to recommend the use of interleaved channel in the ADSL system for solving the problem of impulse noise disturbance.

4-For comparison the result in this paper with another paper like the paper introduced in reference [20] which the Pe is 1.5x$10^{-5}$and 1.4x10^{-6} with the effect of AWGN disturbance for fast and interleaved channel respectively whereas the Pe is 6.4x10^{-5} and 1.9x10^{-6} with the effect of IN disturbance for fast and interleaved channel respectively at 26dB energy signal to noise ratio E_s/N_o. The results within our paper are closer than with the result introduced in Ref. [19].

7-MATLAB software is very good for implementing ADSL system design for monitoring the flow of signals and deals with blocks set in flexibility in spite of it is limitations in some blocks.

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