Improved Method of Characteristics for Two way Subscriber Transmission Systems

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Abstract: The two way subscriber transmission systems have tendency to spread its carrier frequency bandwidth or information bit rate and average bit error rate according to popularization of high speed information through the digital communication system, transmission medium and the Internet. This fact is an important incentive to realize new systems. These two way subscriber transmission systems usually use same cable or same carrier frequency bandwidth for up stream channel and down stream channel. In the systems, the disturbances of noise, crosstalk or fading affect the characteristics. Specifically, these disturbances cause the decrease of information bit rate and degradation of transmission quality.

This paper proposes the improved method of their degradations using the particular feature of two way subscriber transmission systems and it makes clear proposed method is effective by theoretically and some numerical examples.

Keywords: two way subscriber transmission system, information bit rate, bit error rate, error correcting code, quadrature amplitude modulation, discrete multi-tone transmission.

1. INTRODUCTION

Recently, the most real digital communication systems which transmission medium typically includes cable, microwave, or high frequency radio links are required the high information bit rate efficiently and low noise. Because the Internet users rapidly increase and various high speed contents such as moving picture, precise video signal and etc. are treated through wide area network. Back born network is working almost enough high information bit rates, but subscriber line gives the limitation of information bit rate against these high speed signals. Therefore, two way subscriber transmission systems are requested to have broadband communication channels and simultaneously achieve high transmission quality [1]. But these cable and wireless two way subscriber transmission systems suffer disturbances such as internal and external noise, crosstalk or fading and so on. These disturbances cause the decrease of received information bit rate and degradation of transmission quality in communication systems.

This paper proposes new improvement method of these both undesirable characteristics utilizing the particular feature of two way subscriber transmission systems. Proposed system is constructed from up stream (low bit rate) channel, and down stream (high bit rate) channel which have plural different information bit rate determined by selection from M sets of modulation and demodulation functions. According to select suitable one by considering measured bit error rate in up stream channel, total average information bit rate and average bit error rate of down stream channel are simultaneously improved.

The feature of this system is that adopt variable transmission information bit rate in down stream channel.

In this paper, fundamental principle of this system is described and also evaluated the characteristics by theoretically and example numerically. It makes clear that proposed system is effective as actual subscriber transmission systems.

2. OUTLINE OF PROPOSED SYSTEM

The system is constructed using particular feature of two way transmission systems. In this chapter the feature and system principle are described.

2.1 Features of two way transmission systems

There are two types of transmission systems, one is a frequency division multiplex (FDM) system, in which the different frequency bands are used for up stream channel and down stream channel in same transmission medium. Other is time division multiplex (TDM) system that is used the different time intervals but same carrier frequency band to transmitting information bit in also same transmission medium. Fundamental principle of these transmission systems is shown in Fig.1.

The representative of former transmission system is an ADSL (Asymmetric Digital Subscriber Line) [2], and later is an ISDN (Integrated Services Digital Network) or PHS (Personal Handy Phone System) in Japan.

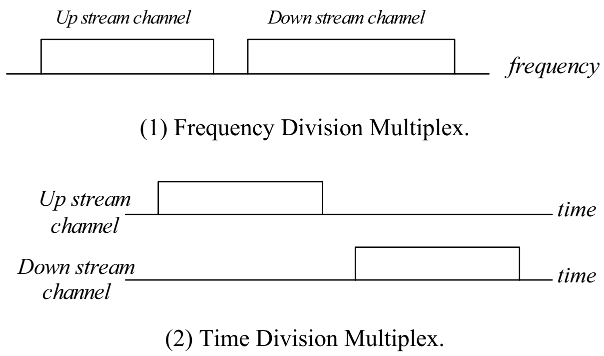


Fig.1 Using same medium in two way transmission systems.

General of two way subscriber transmission systems usually use same cable or same carrier frequency bandwidth or frequency radio link for up stream channel and down stream channel. In these cable and wireless two way subscriber transmission systems, carrier frequency band or time interval is used by dividing two parts for up stream channel and down stream channel, respectively. Therefore, transmission quality (for example: an average bit error rate; BER) of up stream and down stream channels is considered to be almost the same in short time duration.

This fact is an important feature of two way transmission systems and this feature is introduced to proposed system.

2.2 Fundamental principle

This proposed improvement method is devised by utilizing abovementioned particular characteristics. Proposed system consists of up stream channel for measurement of bit error rate and down stream channel which has M sets of different transmitting information bit rate. Operation principle of this system is as follows:

First, certain length and period of training sequence (for example: pseudo noise sequence or pseudo random noise sequence; PN or PRN sequence) is transmitted over up stream (low bit rate) channel on transmitting side and on receiving side the average information bit error rate is measured. According to this measured result, one of the M different information bit rates (set of modulation and demodulation function) in down stream (high bit rate) channel is selected and total information bit rate is increased rather than ordinary system that its modulation and demodulation function are fixed. Simultaneously, improvement of average bit error rate is achieved by error correcting code, BCH (Bose-Chadhuri-Hocquenghem) code and Reed-Solomon code which suitable for each information bit rate.

The system configuration of proposed improvement method is illustrated in Fig.2 and correlation of signals flow between up stream channel and down stream channel is shown in Fig.3.

3. DESIGN OF SYSTEM PARAMETERS

This system consists of combination of several techniques. The system parameters of these sub-techniques are as follows.

3.1 Measurement of average bit error rate in up stream channel

The training sequence of L bits length and certain period T_0 seconds generated from PN sequence generator is transmitted and average bit error rate of receiving signal is measured comparing with transmitting sequence. The PN sequence generator for comparison on receiving side is operated in synchronous with receiving signal.

Figure 4 shows configuration of training PN or PRN sequence generator that is set at both transmitting and receiving side. This PN sequence generator consists of n stages shift register and its operating function considered by linear feedback shift register and multiplexer and adder, its length of one period L is defined as $L = (2^n - 1)$ bits.

Block error probability P_r that an error occur any r bits in L bits length training sequence is represented as follows.

$$P_r = {}_L C_r p_e^r (1 - p_e)^{L-r} \tag{1}$$

p_e; average bit error rate in up stream channel.

$r = r_i$, $i = 1,2,\ldots, M$, $0 = r_0 < r_1 < \cdots < r_{M-1} < r_M = L$

These M difference bit error rate probability P_{r_i} is the threshold level for selecting an information bit rate (set of modulation and demodulation functions) over down stream channel.

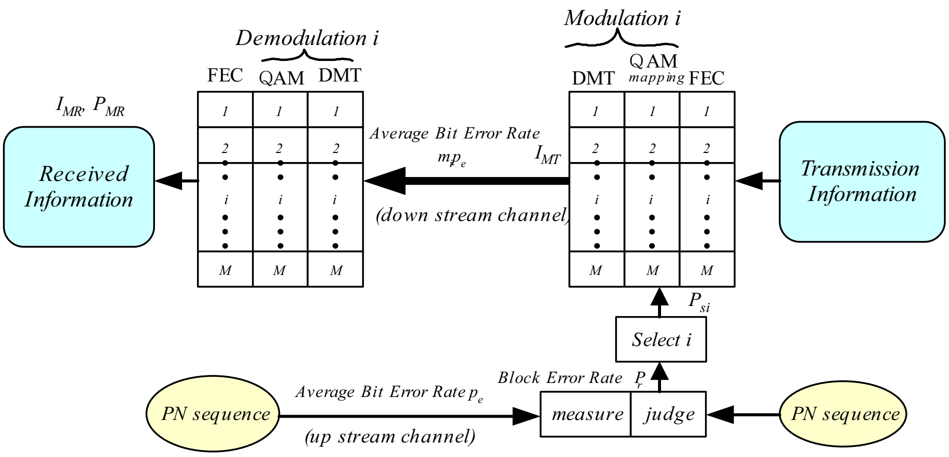
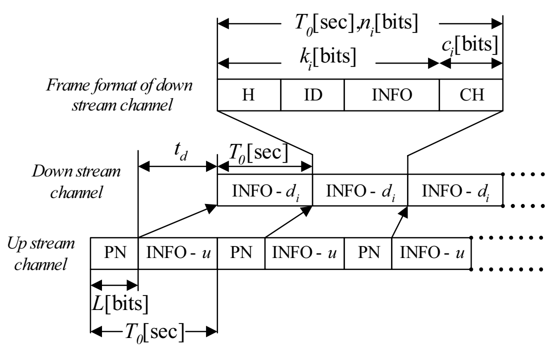


Fig.2 System configuration of proposed method.



PN; the training sequence.
INFO-u; information in up stream channel.
INFO-d_i; information in down stream channel ($i = 1,2,\ldots, M$)
H; header for synchronization, address and etc.
ID; identifier of modulation/demodulation function.
CH; check bit for error correcting (c_i [*bits*]).
t_d; propagation delay time.

Fig.3 Correlation of signal flow between up and down stream channels.

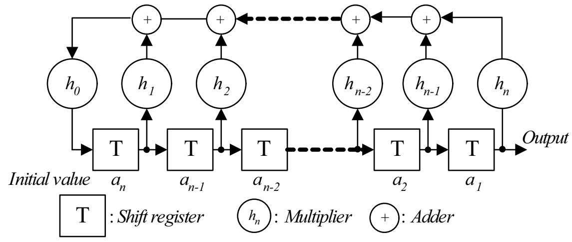


Fig.4 The circuit configuration of pseudo noise (Maximum length) sequence generator.

3.2 Modulation system on down stream high bit rate channel

Two kinds of modulation and demodulation techniques, that is, QAM (Quadrature Amplitude Modulation) and DMT (Discrete Multi-tone Transmission) are applied in this system [3]. The former of modulation and demodulation are used as a primary modulation and demodulation technique and other is a secondary one.

Base band information binary signals are mapping to each m_i ($i = 1,2,...,M$) bits of inphase component a_n, and quadrature component b_n by $2^{m_i} \times 2^{m_i}$ QAM technique. These signals are called a complex signal $X_n = a_n + jb_n$ and each a_n, b_n have m_i bits of information bit rate. The values X_n are the two-dimensional modulated inputs that are derived from standard QAM constellations. Example of QAM constellation is shown in Fig.5.

Other modulation and demodulation techniques, for example: multi-phase PSK (Phase Shift Keying) or APSK (Amplitude Phase Shift Keying) except QAM technique are also available. In the case of PSK, 2^{m_i} -ary PSK can be transmitted m_i bits in each time intervals.

The N units of these complex signals are modulated by different N carriers, which is DMT technique same as OFDM (Orthogonal Frequency Division Multiplex) technique, in which its modulation and demodulation function are using DFT (Discrete Fourier Transform) or IDFT (Inverse Discrete Fourier Transform) (secondary modulation).

This modulation process of proposed improvement method system can be used as next Inverse Discrete Fourier Transform (IDFT) [4].

$$x_k = \frac{1}{N}\sum_{k=0}^{N-1} X_n e^{j\frac{2\pi}{N}nk} \qquad k = 0,1,2,...,N-1 \tag{2}$$

$$X_n = a_n + jb_n$$

Each sample X_n is modulated every T seconds, that is, each m_i bits in inphase and quadrature components is transmitted on one DMT carrier in time interval T. Therefore, after this modulation following information bit rate is obtained.

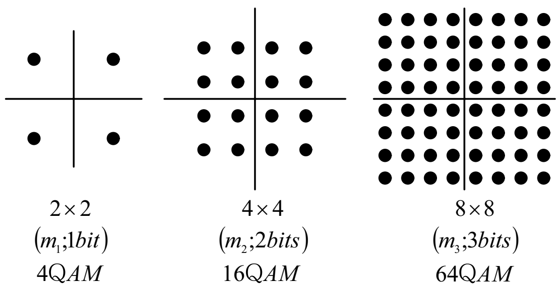


Fig. 5 $2^{m_i} \times 2^{m_i}$ QAM constellation.

$$I_i = \frac{2Nm_i}{T} \text{ [bits/sec]} \tag{3}$$

$i = 1,2,...,M$ and $I_1 < I_2 < \cdots < I_M$

The demodulation function is described following discrete Fourier transform (DFT).

$$X_n = \sum_{k=0}^{N-1} x_k e^{-j\frac{2\pi}{N}nk} \qquad n = 0,1,2,...,N-1 \tag{4}$$

The frequency allocation of DMT is shown in Fig.6 and Fig.7 shows QAM/DMT modulation and demodulation process.

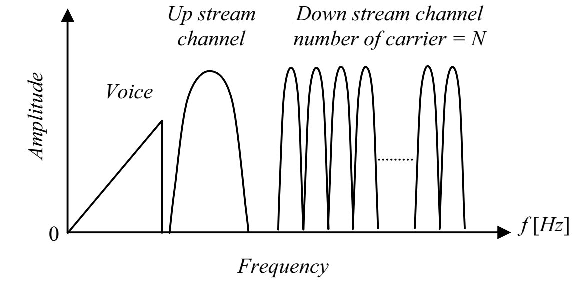


Fig.6 The frequency allocation of DMT.

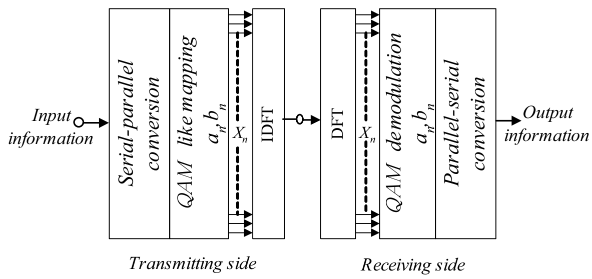


Fig.7 The QAM/DMT modulation and demodulation process.

3.3 Error correcting technique

Error correcting coding is introduced to improve an average bit error rate on down stream channel. The correcting code function used is a block systematic codes, that is (n_i,k_i) code and its correcting ability is t_i ($i = 1,2,...,M$) bits. The n_i is a total number of bit in one block (code word), k_i is a number of information bit and $(n_i$-$k_i)$ is denoted as a number of redundancy bit. The k_i/n_i represents transmission efficiency. The (n_i,k_i) and t_i are selected suitable value for each information bit rate. In general, it is desirable that higher error correcting ability is applied for high information bit rate channel, and in this case not to reduce transmission efficiency it is necessary to use longer block length.

BCH code is suitable for random error and RS code for burst error. BCH codes are important because the BCH codes outperform all other block codes with the same block length and code rate.

The average block error rate after correcting P_i is represented as next equation.

$$P_i = \sum_{j=t_i+1}^{n_i} {}_{n_i}C_j (m_i p_e)^j (1 - m_i p_e)^{n_i - j}$$

$$= \sum_{j=0}^{t_i} {}_{n_i}C_j \left(1 - m_i p_e\right)^j \left(m_i p_e\right)^{n_i - j} \quad (5)$$

p_e; average bit error rate in down stream channel.
m_i; information bit rate conveyed on one carrier in Discrete Multi-tone Transmission.

3.4 Total characteristics and evaluation criterion

Total characteristics are evaluated both transmitting and receiving information bit rate and transmission quality, that is average bit error rate.

Next equation shows the probability after select i information P_{s_i} that condition $P_{r_{i-1}} \le P_r < P_{r_i}$ is satisfied. There, the r_i ($i = 1,2,..., M$) is threshold error bit which changes the modulation and demodulation function.

$$P_{s_i} = \sum_{r=r_{i-1}}^{L} P_r - \sum_{r=r_i}^{L} P_r = \sum_{r=r_{i-1}}^{r} P_r$$

$$= \sum_{r=r_{i-1}}^{r_i} {}_L C_r p_e^r \left(1 - p_e\right)^{L-r} \quad (6)$$

$$\sum_{i=1}^{M} P_{s_i} = 1 \qquad i = 1,2,..., M$$

Under this condition, an information bit rate I_i is selected as down stream channel bit rate. The calculation process of probability P_{s_i} is represented in Fig.8.

Therefore, the total average transmission information bit rate I_{MT} on transmitted side is represented as following equation.

$$I_{MT} = \sum_{i=1}^{M} P_{s_i} I_i \frac{k_i}{n_i} \quad \text{[bits/sec]} \quad (7)$$

And actually (correctly received) average receiving information bit rate I_{MR} on received side is defined as next equation.

$$I_{MR} = \sum_{i=1}^{M} P_{s_i} I_i \frac{k_i}{n_i} \left(1 - P_i\right) \quad \text{[bits/sec]} \quad (8)$$

Furthermore, total average bit error rate P_{MR} is evaluated by following equation.

$$P_{MR} = \sum_{i=1}^{M} P_{s_i} P_i \frac{1}{n_i} \quad (9)$$

These I_{MR} and P_{MR} are an important evaluation criterion of this system (refer to Fig.2) and are compared with characteristics of ordinary system in which does not change modulation and demodulation function, that is $M = 1$.

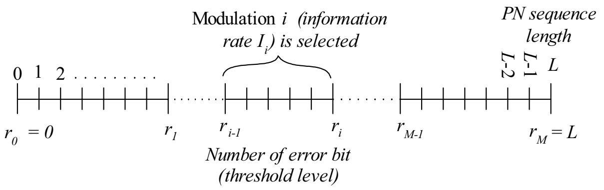


Fig.8 Calculation process of probability P_{si}

4. NUMERICAL EXAMPLES OF PROPOSED SYSTEM

To prove the usefulness of this system, abovementioned characteristics of two numerical examples are calculated.

4.1 Parameters used

Fundamental common parameters for two examples are as follows:

(1) Primary modulation (mapping)
$2^{m_i} \times 2^{m_i}$ QAM; 2 × 2 QAM (m_1 = 1[bit]), 4 × 4 QAM (m_2 = 2 [bits]), 8 × 8 QAM (m_3 = 3 [bits]).
(2) Secondary modulation (DMT)
$N = 256$; $T = 4000$ [baud]
(3) Error correcting code; (424,406) BCH code, and its correcting ability; 2 [bits].
(4) Average bit error rate of transmission medium
$p_e = 10^{-1} \sim 10^{-7}$ (variable).
(5) PN (Maximum length) sequence;
$L = 2^7 - 1 = 127$ [bits].
(6) The transmitting information block length; 424 [bits] (corresponding to ATM one cell length).
Individual parameter for each example is described in the next.

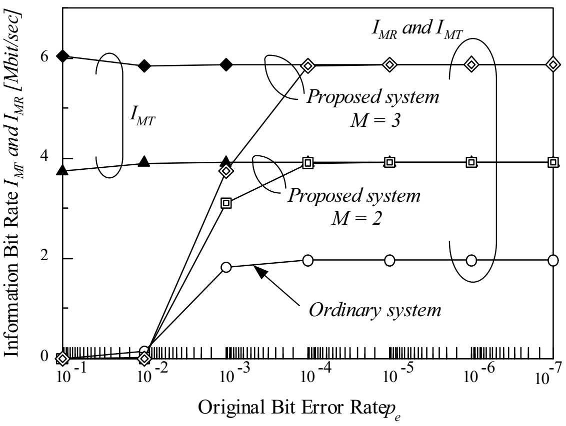
[Example 1]
A number of modulation and demodulation; $M = 2$, (m_1 = 1 [bit], m_2 = 2 [bits]).

[Example 2]
A number of modulation and demodulation; $M = 3$, (m_1 = 1[bit], m_2 = 2 [bits], m_3 = 3[bits]).

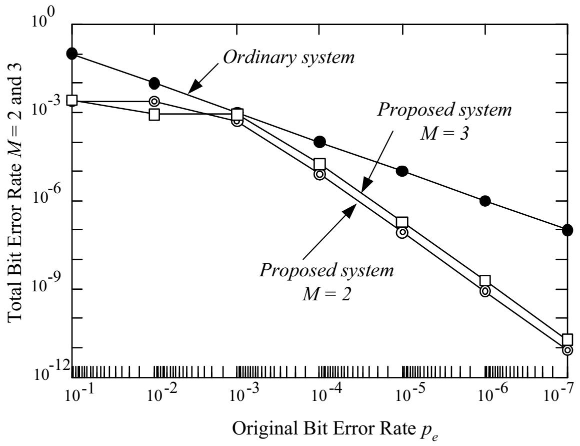
4.2 Result obtained and its evaluation

The numerical computation results of example 1 and 2 are shown in Fig.9, 9(a) represents transmitting and received information bit rate in down stream channel and 9(b) illustrated a total average bit error rate under condition $p_e = 10^{-1} \sim 10^{-7}$, $M = 2$ and 3.

Compared with an ordinary system, which does not change modulation and demodulation functions, the good results are obtained on both information bit rate and total average bit error rate. Almost twice (in the case of $M = 2$) or three times ($M = 3$) of correctly received average information bit rate and lower 2 ~ 4 figures of total average bit error rate are achieved in the region of $p_e < 10^{-4}$. In the region of higher original bit error rate p_e, an improvement effects are not so much. This fact is considered that error correcting function do not enough operate because many error bits occur in transmission line. The performance of an average information bit rate in the case of $M = 3$ is superior than the case of $M = 2$. It shows that precise control of modulation and demodulation function is more effective. The total average bit error rate for $M = 2$ and 3 is almost same although average information bit rate is increased.



(a) Average Information Bit Rate I_{MT} and I_{MR} vs Original Bit Error Rate.



(b) Total Bit Error Rate P_{MR} vs Original Bit Error Rate.

Fig.9 The results of example 1 and 2 under condition $p_e = 10^{-1} \sim 10^{-7}$, $M = 2$ and 3.

5. CONCLUSIONS

New method to be improving the information bit rate and bit error rate is proposed. And the analysis of characteristics is achieved theoretically and numerically. From these results, it makes clear that this method is very useful for two way transmission systems.

Obtained results through this study are summarized by next expressions.

(1) Transmitting PN sequence, the measuring method of bit error rate (number of error bits) variance in short time duration is shown.

(2) According to measured result, improved method of characteristics is proposed changing modulation and demodulation function. Also particular feature of two way transmission system is used in this system construction.

(3) Evaluation criterions of this system, both correctly received information bit rate and total average bit error rate are defined.

(4) Using above criterion, characteristics of numerical examples is calculated.

Near future problems are that the simulation of this system using another system parameters, for example both m_i and N are variable

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