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Abstract

This paper gives a short introduction of Asymmetric Digital Subscriber Line (ADSL) transmission technology. ADSL is an interesting technology for the transport of VoD-services or other high bit-rate services over the ordinary phone-lines. ADSL uses a multi carrier modulation method called Discrete Multitone (DMT) which is also described in this paper.

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Initials, acronyms and symbols

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$\mathbf{1}$ **Introduction**

Asymmetric Digital Subscriber Lines (ADSL) are used to deliver high-rate digital data over existing ordinary phone-lines. A new modulation technology called Discrete Multitone (DMT) allows the transmission of high speed data.

ADSL facilitates the simultaneous use of normal telephone services, ISDN, and high speed data transmission, eg., video.

DMT-based ADSL can be seen as the transition from existing copper-lines to the future fiber-cables. This makes ADSL economically interesting for the local telephone companies. They can offer customers high speed data services even before switching to fiber-optics.

 $2¹$ $ADSL$

$\overline{2}$ **ADSL**

ADSL is a newly standardized transmission technology facilitating simultaneous use of normal telephone to services, data transmission of 6 Mbit/s in the downstream and Basic-rate Access (BRA).

ADSL can be seen as a FDM system in which the available bandwidth of a single copper-loop is divided into three parts. See figure 1. The baseband occupied by POTS is split from the data channels by using a method which guarantees POTS services in the case of ADSL-system failure (eg. passive filters).

Figure 1: Frequency Spectrum of ADSL

2.1 Application Architecture

A possible ADSL system is illustrated in figure 2 [1, p. 3]. A flexible way to connect various servers to corresponding application's device is to use ATM-switches. Local ATM-switch is connected to an access module in a telephone central office. The access module is used to connect the ATM network to phonelines. In the access module ATM data stream from server is decomposed and routed to the corresponding phone-lines.

There is a large number of different kind of servers that can be accessed by an ADSL system. Those servers shown in figure 2 are not only future but also today's technology.

An employee using a work-at-home-server can take full advantage of the high-speed capabilities of an ADSL-system in many ways, e.g., running licensed software, downloading CAD, documents etc.

Video-on-Demand-service is one of the most interesting aspect of ADSL. By using MPEG-coded video it is possible to deliver video-quality movies over existing copper-loops to customers. A video-quality can be achieved by only 1.5 Mbps data rate. Together with pure VoD-services there might exist combined movie/information/advertizer-services in which commercial and non-commercial information providers and advertizers can deliver their information.

¹ANSI T1E1.4/94-007R

Figure 2: ADSL System Architecture

System Architecture 2.2

ADSL System reference model, which is shown in gure 3, describes the basic blocks of an ADSL-system.

The decomposed and routed data from the access module, see figure 2, is connected to an ATU-C (ADSL Transceiver Unit - Central Office) in which the data will be converted into analog signals. The analog signals are then carried with POTS signals to remote end. ATU-C also receives and decodes data coming from customers premises send by ATU-R (remote).

Both ATU-C and ATU-R (ADSL Transceiver Unit) are described in more detail in figures 4 and 5.

The splitter either combines or separates the signals depending on the direction of the transmission. It protects MTS from voice-band interference generated by both ATU's and on the other hand it protects ATU's from MTS-related signals.

NOTES

1 The V interface is defined in terms of logical functions; not physical

2 The V interface may consist of interface(s) to one or more switching systems

3 Implementation of the V and Tsm interfaces is optional when interfacing elements are integrated into a common element

4 The splitter function may be integrated into the ATU

5 A digital carrier facility (e.g., SONET extension) may be interposed at the V interface when the ATU-C is located at a remote site

6 The nature of the CI distribution (e.g., bus or star, type of media) is for further study

7 More than one type of Tsm interface may be defined, and more than one type of T-sm interface may be provided from an ATU-R

8 Due to the asymmetry of the signals on the line, the transmitted signals shall be distinctly specified at the U-R and U-C reference points

9 A future issue of this standard may deal with CI distribution requirements

Figure 3: ADSL System Reference Model

2.3 ADSL Transport Capacity

The dierent additional transport classes for n-classes for n-maps with the 2M-2M-1, 2M-2 and 2M-2 and 2M-3. In 2M-1 corresponds the highest rate and shortest range.

ADSL downstream transport capacity is basically from 2.048 Mbps to 6.144 Mbps. At 6.144 Mbps it is possible to achieve the range of about 3 kilometers [1]. The lower the transmission rate is the longer the range will be. Upper limit is according to tests about 9 kilometers [2]. It is possible to achieve higher

Figure 4: ATU-C Transmitter reference model

Figure 5: ATU-R Transmitter reference model

data rates of 52 Mbps and 155 Mbps, corresponding range of one mile and a quarter mile, if the used transmission media is ber [1]. By using DMT ADSL it is also possible to use other data rates, the exact rate depends only on interface circuits. So the system is flexible enough to support, eg., T1. The downstream bit rates are summarized in table 3.

1. The 16 kbit/s duplex C channel is transported entirely within the overhead dedicated to synchronization capacity.

2. 544 kbit/s is required when a 160 kbit/s and a 384 kbit/s optional duplex bearer are both included.

3. The duplex C channel is not included in total bearer channel rates for transport class 2M-3; it's included in the overhead.

4. The overhead required for FEC in not shown in this table.

Table 3: Downstream bit rates

ADSL upstream transport capacity is $0 - 640$ kbit/s depending on transport class. The aggregate upstream bit rates are summarized in table 4.

ATM can be transported over ADSL and the components of the aggregate bit rates are summarized in

2.4 Framing

The downstream and upstream data channels are synchronized to the 4 kHz ADSL DMT (Discrete Multi Tone) symbol rate, and multiplexed into two separate data buffers (fast and interleaved).

ADSL uses the superframe structure shown in figure 6. Each superframe is composed of 68 ADSL data frames, which are encoded and modulated into DMT symbols. From the bit-level and user data perspective, the DMT symbol rate is 4000 baud (period $= 250 \mu s$). Because of the sync symbol inserted to the end of each superframe, the transmitted DMT symbol rate is 69/68 * 4000 baud.

Eight bits per ADSL superframe are reserved for the crc, and 24 indicator bits (ib0-ib23) are assigned for

bearer, then the maximum total bearer channel capacity and maximum aggregate rate will increase by 32 kbit/s.

2. For transport class 2M-3, the duplex C channel is 16 kbit/s; this is not included in the total bearer channel rates because it is transported entirely within the overhead dedicated to synchronization capacity.

3. 544 kbit/s obtained when both optional duplex bearers are included.

4. The overhead required for FEC is not shown in this table.

3. These values have been calculated by $\frac{7}{47} \times n \times 2.048$ Mbit/s and rounded up to the \parallel nearest 32 kbit/s multiple

(FEC output or constellation encoder input data frame, points (B),(C))

Figure 6: ADSL superframe structure

OAM functions. The "fast" byte of the fast data buffer carries either crc, eoc or synchronization bits.

Each user data stream is assigned to either the fast or the interleaved buffer during initialization.

2.5 **Scrambling**

The binary data stream outputs from the fast or interleaved buffers are scrambled separately using the following algorithm for both:

$$
d'_n=d_n\oplus d'_{n-18}\oplus d'_{n-23}
$$

where d_n is the n-th output from the fast or interleaved buffer, and d'_n is the n-th output from the corresponding scrambler. Scrambling can be performed independent of symbol synchronization.

Forward Error Correction (FEC) is used to assure optimal performance. It is based on Reed-Solomon coding and it must be implemented. The size of the Reed-Solomon codeword is $N = K + R$, in which the number of check bytes R and codeword size N vary depending on the number of bits assigned to either fast or interleaved buffer.

The Reed-Solomon codewords in the interleave buffer are convolutionally interleaved. The interleaving depth values are either 16, 32 or 64 (32 or 64 for 2.048 Mbit/s based systems).

2.7 Tone ordering

A DMT time-domain signal has a high peak-to-average ratio (its amplitude distribution is almost Gaussian), and large values may be clipped by the D/A-converter. The error signal caused by clipping can be considered as an additive negative impulse for the time sample that was clipped. The clipping error power is almost equally distributed across all tones in the symbol in which clipping occurs. Clipping is therefore most likely to cause errors on those tones that have been assigned the largest number of bits (and therefore have the densest constellation). These occasional errors can be reliably corrected by the FEC coding if the tones with the largest number of bits have been assigned to the inter leave buffer.

The number of bits and the relative gains to be used for every tone are calculated in the ATU-R receiver, and send back to the ATU-C. The pairs of numbers are typically stored, in ascending order of frequency or tone number i , in a bit and gain table.

The "tone-ordered" encoding assigns the first B_F bytes (8 B_F bits) from the symbol buffer to the tones with the smallest number of bits assigned to them, and the remaining B_I bytes (8 B_I bits) to the remaining

2.8 Constellation encoding

Constellation encoder can be implemented with or without trellis coding. The system performance can be improved by block processing of Wei's 16-state 4-dimensional trellis code. It is possible to achieve 2-3 dB better coding gain and the overall improvement in coding gain by well designed ADSL system can be about 5.5 dB [1].

For a given sub-channel, the encoder selects an odd point (X, Y) from the square-grid constellation based on the b bits $v_{b-1}, v_{b-2}, \ldots, v_1, v_0$. For convenience of description, these b bits are identified with an integer label whose binary representation is $(v_{b-1}, v_{b-2}, \ldots, v_1, v_0)$. For example, for $b = 2$, the four constellation points are labeled 0, 1, 2, 3 corresponding to $(v_1, v_0) = (0, 0), (0, 1), (1, 0), (1, 1)$, respectively.

Even values of b

Example constellation are shown in figure 7.

The 4-bit constellation can be obtained from the 2-bit constellation by replacing each label n by the $2x2$ block of labels:

$$
4n+1 \quad 4n+3
$$

$$
4n \quad 4n+2
$$

The same procedure can be used to construct the larger even-bit constellations recursively

Figure 7: Constellation labels for $b = 2$ and $b = 4$

Odd values of b

Example constellation for the case $b = 5$ is shown in figure 8.

Figure 8: Constellation labels for $b = 3$ and $b = 5$

if b is odd and greater than 3, the 2 MSBs of X and the 2 MSBs of Y are determined by the 5 MSBs of the ^b bits.

The 7-bit constellation can be obtained from the 5-bit constellation by replacing each label n by the 2x2 block of labels:

$$
4n+1 \quad 4n+3
$$

$$
4n \qquad 4n+2
$$

Transmitter

The transmitter includes all analog transmitter functions: the D/A -converter, the anti-aliasing filter, the hybrid circuitry and the MTS splitter.

2.9 Initialization

The task of the initialization process is to maximize the throughput and reliability of the link. This process is also transparent to the vendors choice of the method of separating upstream and downstream signals (either FDM or echo cancellation). See also section 3.2.

The channel attribute values determined by the initialization procedure include the number of bits and relative power levels to be used on each DMT sub-carrier, as well as any messages and final data rates information. Table 6 describes the main stages of the initialization procedure.

Table 6: Overview of initialization

2.10 High-level on-line adaptation $-$ bit swapping

Bit swapping enables an ADSL system to change the number of bits assigned to a subcarrier, or change the transmit energy of a subcarrier without interrupting data flow. The bit swap process uses the aoc channel.

3 **ADSL Modulation Methods**

ANSI standard [4] describes a basic ADSL system which uses DMT (Discrete Multitone) modulation. There is also at least one other ADSL system available. This system facilitates Carrierless AM/PM (CAP). In this chapter the DMT modulation method is described.

3.1 Discrete Multitone (DMT)

The basic idea of DMT is to split the available bandwidth into a large number of subchannels. DMT is able to allocate data so that the throughput of every single subchannel is maximized. If some subchannel can not carry any data, it can be turned off and the use of available bandwidth is optimized. The examples in figure 9 give an idea about of the functionality of DMT. [1]

Figure 9: Examples of DMT

First an equal number per tone is transmitted to measure the characteristics of the line. The processing of the signal takes place in ATU-R, and the optimized bit distribution information will be delivered for ATU-C by using the same phone-line at a secure low speed.

The first example describes a segment of 24-gauge twisted pair phone-line. Low frequencies are eliminated by the transformer coupling. The attenuation at the higher frequencies depends on the length of the phone-line.

The second example includes the notch in spectrum that is illustrative of bridge taps and also the

interference of an AM radio station.

A third example shows that DMT is also an interesting possibility for other transmission channels, such as coaxial cable-TV networks, as well.

3.2 ADSL DMT Modulation

In ADSL DMT-systems the downstream channels are divided into 256 4-kHz-wide tones. The upstream channels are divided into 32 subchannels. See also the frequency spectrum of the ADSL-channels in figure 1.

Some of the most important parameters for standardized ADSL DMT are described below. Note, that these values differ for both ATU-C and ATU-R.

Carrier 64 $(f = 276 \text{ kHz})$ is reserved for a pilot. The data modulated onto the pilot subcarrier shall be constant 0,0. Use of this pilot allows resolution of sample timing in a receiver modulo-8 samples.

Nyquist frequency

The carrier at the Nyquist frequency (256) may not be used for data.

Modulation by the inverse discrete Fourier transform (IDFT)

The modulating transform defines the relationship between 512 real values x_k and the Z_i

$$
x_k = \sum_{i=0}^{511} exp\left(\frac{j\pi ki}{256}\right) Z_i
$$

The encoder and scaler, see figure 4, generate only 255 complex values of Z_i (plus zero at dc, and one real value if the Nyquist frequency is used). In order to generate real values of x_k these values shall be augmented so that the vector Z' has Hermitian symmetry.

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Synchronization symbol

The synchronization symbol permits recovery of the frame boundary after micro-interruptions that might otherwise force retraining.

Cyclic prefix

The last 32 samples of the output of the IDFT $(x_k$ for $k = 480$ to 511) are prepended to the block of 512 samples and read out to the D/A converter in sequence. The cyclic prefix is used for data and synchronization symbols.

ATU-R

For ATU-R the maximum number of subcarriers is 31 and carrier 16 is reserved for a pilot.

The modulating transform defines the relationship between 64 real values x_k and the Z_i'

$$
x_k = \sum_{i=0}^{64} \exp\left(\frac{j\pi ki}{32}\right) Z_i
$$

for $k = 0$ to 63.

For the cyclic prefix the 4 last samples are used.

Figure 10: Multidrop extension example of ADSL

4 ADSL Progress

There are couple of interesting prospects in the future of ADSL. The development of High-Speed ADSL is on its way. There are also proposals for in-house extensions of ADSL. The interconnections of ADSLsystems and Broadband Services are also under study.

Below are listed a few points to take into account, when the progress of ADSL DMT-systems is consider.

- 1. Speed up
	- 6 Mbps is not enough
	-
	- by the state of the state of the state possible possible possible and the state of the state
- 2. Broadband services
	- ATM over ADSL or within ADSL ?
- 3. DIM (DMT Information Bus)
	- A possible in-house extension for ADSL

The DMT Information Bus (DIB), see figure 10, is based on the fact, that ADSL DMT modems can interoperate on the common subchannels. The central-office modem can "talk" to severeal remote modems simultaneously. Of course the remote modems need to be loop-timed in sample and symbol clock. This kind of approach has the advantage, that no in-house rewiring is needed. [1]

References

- [1] J. M. Cioffi. ADSL Stanford University, 1994.
- [2] W. Y. Chen, D. L. Waring. Applicability of ADSL to Support Video Dial Tone in the Copper Loop IEEE Communications Magazine, pp. 102-109, May 1994.
- [3] K. Sistanizadeh, P. S. Chow, J. M. Cioffi. Multi-Tone Transmission for Asymmetric Digital Subscriber Lines (ADSL) ICC93, pp. 756-760.
- [4] ANSI/T1E1.4/94-007, Asymmetric Digital Subscriber Line (ADSL) Metallic Interface.
- [5] P. F. Prunty Delivery of TV Over Existing Phone Lines SMPTE Journal, pp. 586-594, September 1994.
- [6] K. Salminen Simulation Model for Digital Television Broadcasting System using Orthogonal Frequency Division Multiplexing, M. Sc. Thesis, Tampere University of Technology, 1993.