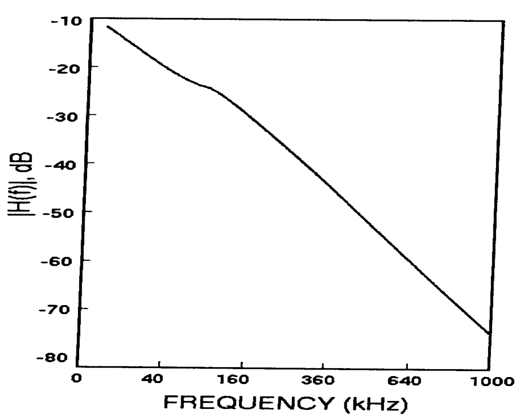
7. ADSL and High-Rate Digital Communication

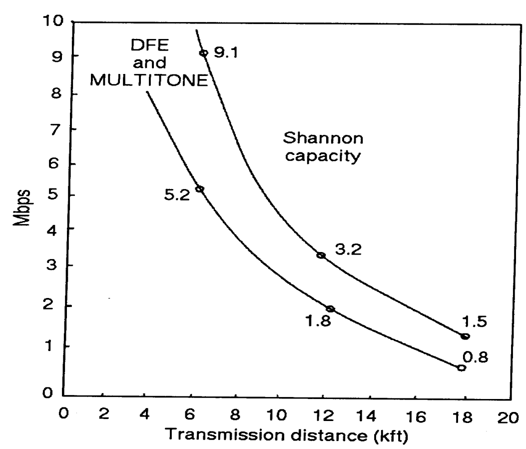
7. 1 Digital Subscriber Line Technologies

Digital Subscriber Line Technologies (xDSL) is a revitalized transmission technology facilitating simultaneous use of Plain Old Telephone Services (POTS) and data transmission of up to 6.0 Mbits/s over the existing infrastructure of copper wiring. In particular, Asymmetric DSL (ADSL) is a newly standardized technique (ANSI/T1.413). In the past year, xDSL have attracted a great deal of attention as the access solution of the future in both the home and business application environments. Originally, xDSL technologies were proposed as an intermediate access solution for the residential area before the extensive installation of hybrid fiber coax (HFC) or fiber to the home (FTTH). It has become apparent that these last two systems will not be widely accepted in decades so the intermediate solution of xDSL is now seen as a deployment for at least few decades.

1. What is behind xDSL technologies?

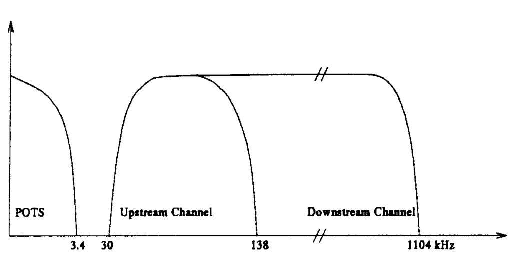
For decades, conventional wisdom has held that analog modems would reach (did reach) 56 kbits/s ceiling in terms of maximum possible bandwidth without compression. In actuality, the 56 kbits/s magic number refers to only to the amount of bandwidth that is theoretically possible over the audible spectra of frequencies, which is the bottom 4.0 kHz of total spectra available on a typical pair of telephone wire. Below, we present the attenuation characteristics for a typical twisted-pair (12 gauge, 12,000 ft) wire and more critically, we plot the capacity versus transmission distance under the new regimes used in xDSL technologies.





As it can be seen from figures that the digital transmission rates of up to 6.0 Mbits/s for distances 6,000 ft or less (1 Km or less) is possible over the majority of the existing twisted-pair copper wire installations throughout the world.

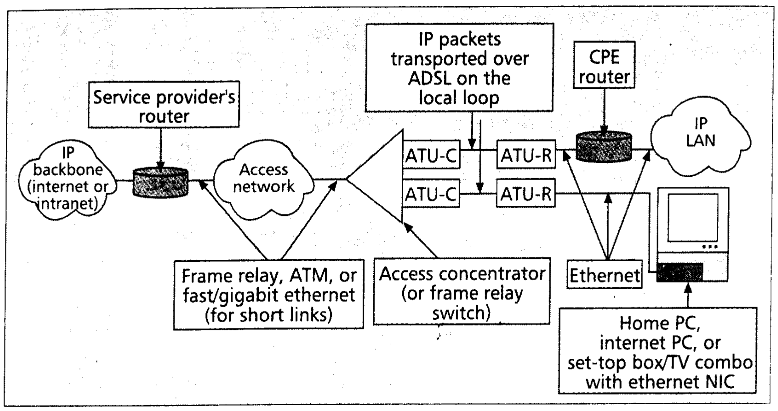
2. How xDSL systems utilize the copper subscriber loop?



ADSL and its technological cousins can be seen as an Orthogonal Frequency-Domain Multiplexing (OFDM) system in which the available bandwidth of a single copper-loop is divided into three parts. In particular, the ADSL spectrum shown here has a splitter to guarantee a ***simultaneous*** POTS channel for the lowest 3.4-4.0 kHz and a digital data channel on the remaining 30-

1,104 kHz. ADSL standard, as the adjective “asymmetric”, allocates 30-138 kHz for the upstream data and the rest to the downstream information sequences in an uneven fashion. Before we go into the system details let us discuss a few applications for the xDSL technologies.

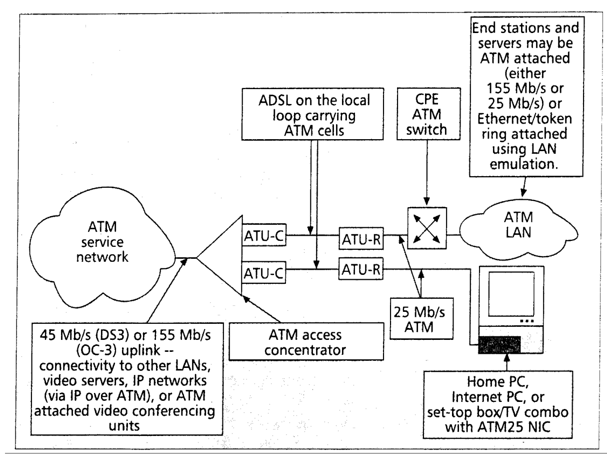
7. 2 Applications of Digital Subscriber Line Technologies



1. **Intranet Access.** An organization that has already implemented an Intranet will require the higher bandwidth afforded by xDSL to link their office/branch offices and telecommuters to the more demanding business oriented applications.

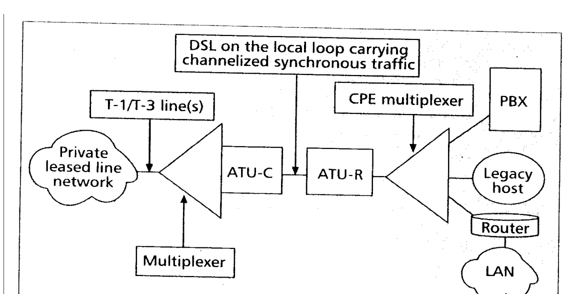
2. **Low-Cost and High-Throughput, LAN-to-LAN Connectivity**: These emerging xDSL technologies have the potential to prove far more effective in this role than ISDN or traditional leased lines.

3. **Frame Relay Access:** Since xDSL operates at the physical layer, it could emerge as the most cost-effective method of carrying frame relay traffic from the service subscriber to the frame relay network.



4. **ATM Network Access:** As with (3), xDSL can also be used to carry ATM cells to an ATM access device, where they are statistically multiplexed over an ATM backbone.

5. **Leased Line Provisioning:** xDSL can be used to greatly reduce the cost of provisioning T-1/E-1 lines from the central office (CO) to the customer’s site.



■ Figure 5. *Circuit switched model.*

Some more terminology:
IP : Internet Protocol.
ATU-C: ADSL Terminal Unit for CO.
ATU-R: ADSL Unit for the remote site.

7.3 Speeds and Feeds for ADSL Systems

ADSL as presently standardized is defined as having (7) transport classes: 4 classes based on multiples of T-1 (1.5 Mbits/s) downstream bandwidth and three classes based on E-1 (2.0 Mbits/s) bandwidth as tabulated below.

Transport Class	1	2	3	4	2M1	2M2	2M3
Downwards simplex ch.	6.144 Mb/s	4.608 Mb/s	3.072 Mb/s	1.536 Mb/s	6.144 Mb/s	4.096 Mb/s	2.048 Mb/s
Upstream duplex ch.	640 kb/s 576 kb/s usable BW	608 kb/s 544 kb/s usable BW	608 kb/s 544 k/s usable BW	176 kb/s 160 kb/s usable BW	640 kb/s	608 kb/s	176 kb/s
Control Ch.	64 kb/s	64 kb/s	64 kb/s	16 kb/s	64 kb/s	64 kb/s	16 kb/s
POTS Ch.	64 kb/s	64 kb/s	64 kb/s	64 kb/s	64 kb/s	64 kb/s	64 kb/s

7.4 Other xDSL Systems

There are several digital subscriber loop technologies related to ADSL.

- **Symmetric DSL (SDSL):** The same amount of BW is allocated to both upstream and downstream links. The price paid for maintaining BW symmetry is lower aggregate BW. Systems operating at 384 kb/s, 768 kb/s, 1.5 Mb/s (T-1) and 2 Mb/s (E-1) are available. Because of theses restrictions SDSL is not likely to be a serious contender in low-cost markets.

. **Very-High Rate DSL (VDSL):** Like the ADSL case, it is an asymmetrical transmission scheme operating in the range 30-51 Mb/s over extremely short distances (150-300 m). It is anticipated that VDSL can have market penetration in conjunction with fiber to the curb (FTTC) deployment in its last link between the curb and the user terminal equipment.

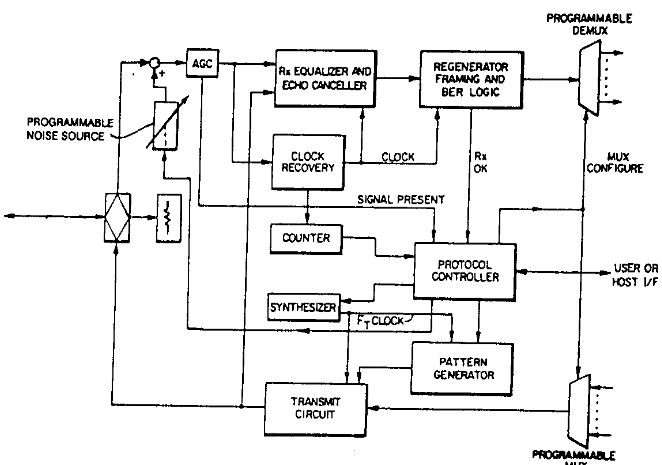


Fig. 1. RA-DSL system block diagram.

- **Rate-Adaptive DSL (RA-DSL):** Here the line speed is automatically adjusted based on a series of initial tests that determine the maximum speed possible on a given line. This is designed to take the guesswork out of the picture on the gauge of the wire, the variations in length, and the condition of the loop.

- **High-Bit-Rate DSL (HDSL):** It is the most widely deployed xDSL technology and it has been commercially available for sometime now. Unlike the other xDSL technologies, HDSL uses 2-pairs of copper cable rather than one and does not carry POTS. They provide either 1.5 or 2 Mb/s of symmetrical BW up to 4,000 meters from the CO. It is attractively used in T-1/E-1 provisioning since it eliminates the need for repeaters, loop conditioning, or pair selection. These advantages have been one of the reasons why the lease line prices have come down significantly.

7.5 xDSL Systems vs. Cable Modems

Cable modems designed to provide multi-megabit BW over existing CATV networks is xDSL systems' primary competitor in the residential access market. However, data services based on CATV's coaxial cable network infrastructure possess a number of shortcomings including (i) lack of penetration in commercial areas, (ii) lack of sizeable and organized CATV infrastructure outside North America and Japan, (iii) security, (iv) lack of experience in network management, and (v) most critically "Shared Bandwidth Access."

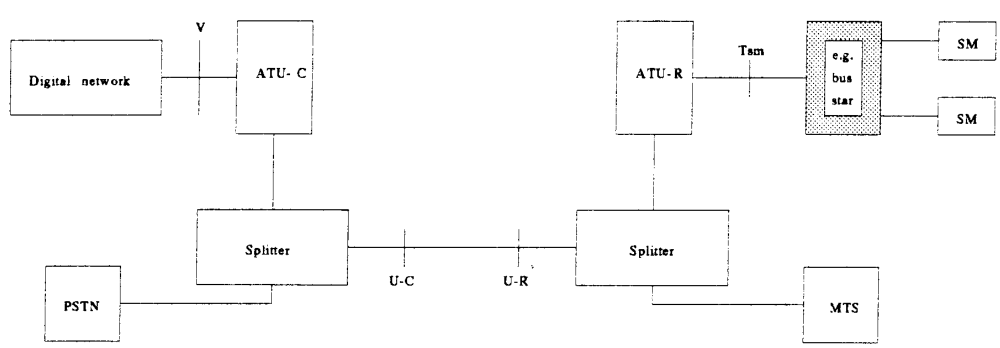
Perhaps the greatest shortcoming to broadband services deployed over CATV networks is the fact that all users in the service area must share the available bandwidth. Even though, cable modems can provide raw BW up to aggregate rate of 51 MB/s, each time a subscriber is added in a given service area, the BW available to each user is decreased. For instance, if 100 subscribers are in a service area, then effectively each user has 300 kb/s if the overall rate were 30 Mb/s. Below we present a cost/performance table for Internet access technologies measured by cost-of-service per unit bandwidth.

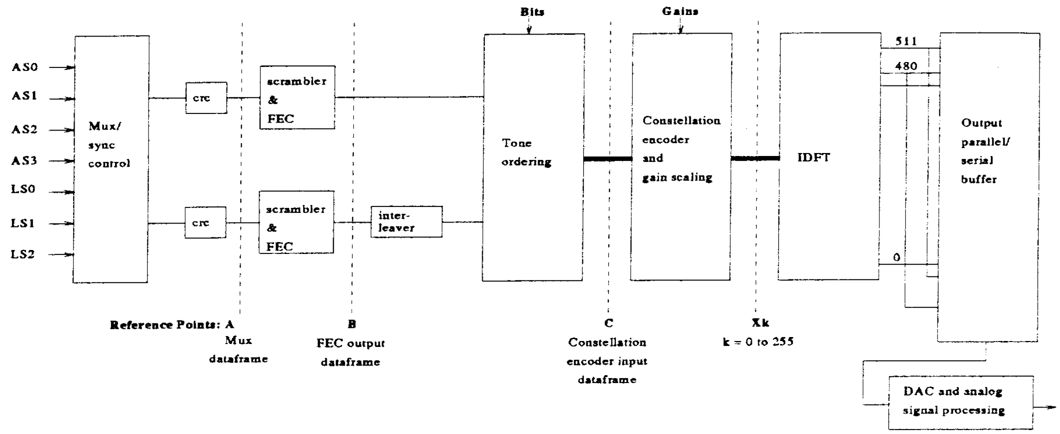
technology	Number of nodes connected to one CO/headend	Bandwidth per node, — downstream/ upstream	Monthly cost of Internet access	Cost of bandwidth provided ($/kb/s)
nodem	225	6 Mb/s/.74 Mb/s	$90	$.015/$1.22
b/s modem	225	0.0288 Mb/s	$11	$3.82
RI	225	0.128 Mb/s	$60	$4.69
sed line	225	1.5 Mb/s	$1,200	$7.74
nodem	1	10 Mb/s/ 1 Mb/s	$50	$.05/$.5
nodem	5	2 Mb/s/.2 Mb/s	$50	$.25/$2.50
nodem	25	.4 Mb/s/.04 Mb/s	$50	$1.25/$12.50
nodem	225	.044 Mb/s/ .004 Mb/s	$50	$11.36/$113.64

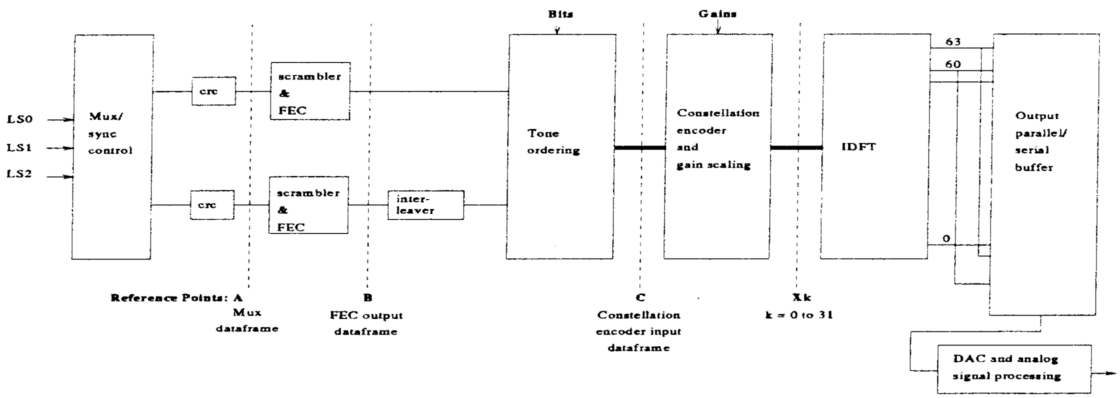
5. Internet access technologies measured by cost-of-service/unit bandwidth.

7. 6 Highlights of ADSL Standard Architecture

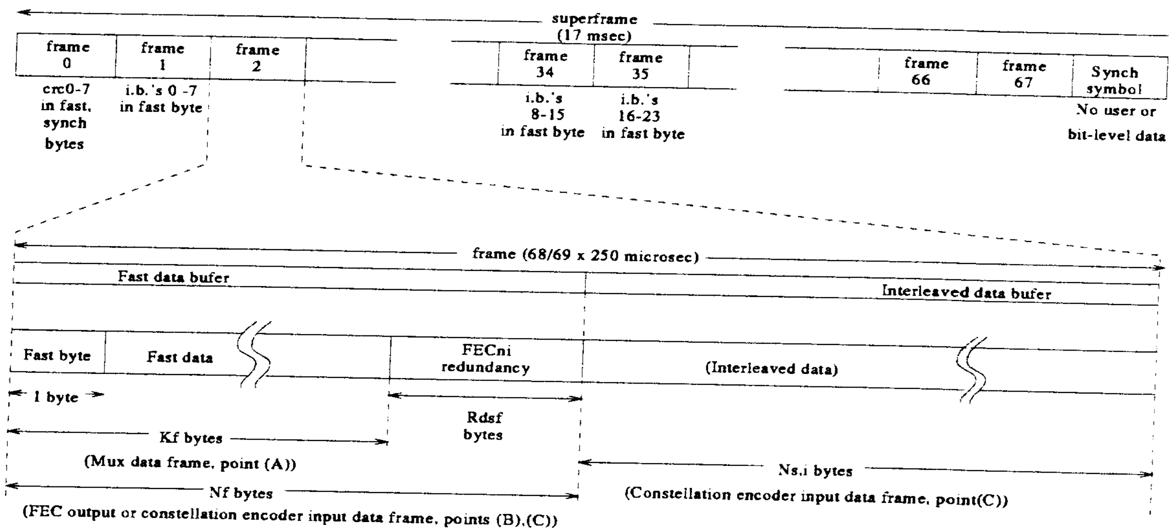
System reference model: ADSL System reference model is depicted below. Decomposed and routed data from the digital network is connected to an ATU-C (Transceiver Unit-Central Office) where the data will be converted into analog signals. The analog signals are then multiplexed and carried with POTS signals to the remote end. ATU-C also received and decodes data coming from customer premises sent by ATU-R (Remote.) In addition to combining or separating, the splitter also protects POTS from voice-band interference generated by both ATU's. Similarly, it protects ATU's from POTS-related signals as well. Next we present the ATU-C Transmitter and ATU-R Receiver Units.







Framing Structure: The downstream and upstream data channels are synchronized to the 4.0 kHz ADSL Discrete Multi Tone (DMT) symbol rate, and multiplexed into two separate data buffers called “fast” and “interleaved.” It uses a superframe structure shown below.



- Each ADSL 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 4,000 bauds with a symbol period of 250 μs.
- One sync symbol is inserted to the end of each superframe; this yields a transmitted symbol rate of $\frac{69}{68}.4000$ bauds.
- 8 bits per ADSL superframe are reserved for the "crc," and 24 indicator bits $(ib_0 - ib_{23})$ are assigned for OAM functions (Operation, administration, and maintenance).
- "Fast Byte" of the fast data buffer carries either "crc (cycling redundancy check code", "eoc (embedded operations channel)" or synchronization bits.
- Each user data stream is assigned to either fast or the interleaved buffer during initialization.

Scrambling and Forward Error Correction (FEC):

If the n^{th} output from the fast or interleaved buffer is d_n and d_n' is the n^{th} output from the corresponding scrambler, data streams from both the fast or interleaved buffers are scrambled separately according to:

$$d_n' = d_n \oplus d_{n-18}' \oplus d_{n-23}' \qquad (7.1)$$

Forward error correction is based on Reed-Solomon coding, which we have studied in Chap 5. In the ADSL terminology, the size of the RS codeword is defined by $N = K + R$, in which the number of check bytes are R and the codeword size is N depending on the number of bits assigned to either fast or interleaved buffer.

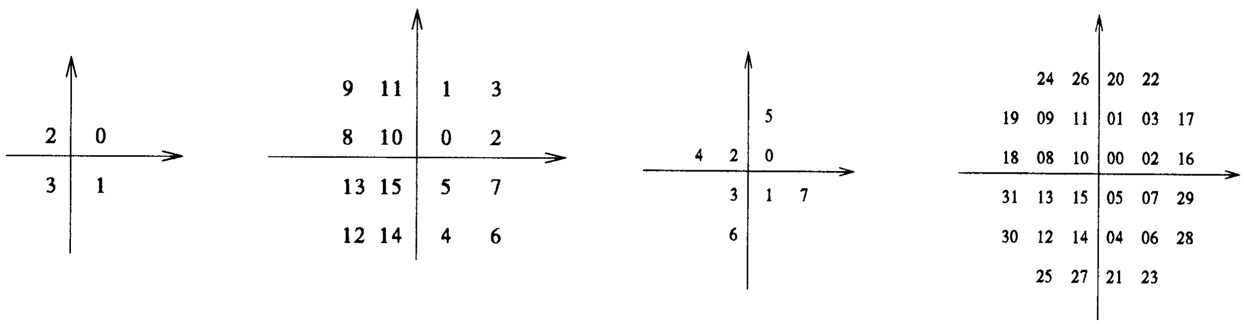
- RS codewords in the interleave buffer are convolutionally interleaved and the interleaving depth values are either 16, 32, or 64 for 1.5 Mb/s-based systems and they are 32 or 64 for 2 Mb/s-based systems.

Discrete Multi Tone (DMT) Modulation and Tone Ordering will be discussed separately.

Constellation Encoding:

Constellation encoding can be implemented with or without "trellis coding." However, for high rates it is unavoidable. It is based on an improved version of Ungerboeck Codes due to Wei. It is has 4-dimensional trellis coder with 16-states.

1. For a given sub-channel, the encoder selects an odd point (X,Y) from the square-grid based on b bits: $\{v_{b-1}, v_{b-2}, \cdots v_1, v_0\}$.
2. These b bits are identified with an integer label whose binary representation is given by: $(v_{b-1}, v_{b-2}, \cdots v_1, v_0)$. For instance: for b=2, the 4 constellations are labeled 0,1,2,3 as shown below.



3. Even values of b: Higher order constellations are obtained from the 2-bit ones above by replacing each label by the 2x2 block of labels: $\begin{Bmatrix} 4n+1 & 4n+3 \\ 4n & 4n+2 \end{Bmatrix}$
4. Odd values of b: Using the 3-bit labeling as basis, the two MSB of X and the two MSB of Y are determined by the five MSBs of the b bits. For instance, the 7-bit constellation is obtained by replacing each label n by the same 2x2 labeling scheme given above.

Transmitter:

It includes all analog transmitter functions, such as the D/A converter, the anti-aliasing filter, the hybrid circuitry and the POTS splitter.

Before the actual data transfer start, an initialization process is launched to maximize the throughput and reliability of the link. This process is transparent to the vendors' choice of method separating upstream and downstream signals. They include the following stages for both the ATU-C and ATU-R equipment:

Activation and ACK,
Transceiver Training,
Channel Analysis, and
Exchange.

Here the number of bits relative power level for each DMT sub-carrier, as well as any messages and final data rates, are passed between the two sides.

ADSL DMT Modulation Specs:

- Downstream channels are divided into 256 4kHz-wide tones.

- Upstream channels are divided into 32 sub-channels.
- Pilot: Carrier-64 $(f = 276\,kHz)$ is reserved for a pilot tone. The data modulated into the pilot sub-carrier is a constant 0,0. Use of this pilot allows resolution of sample timing in a receiver modulo-8 samples.
- The carrier at the Nyquist frequency (256) may not be used for data.

ATU-C Modulation by the Inverse Discrete Fourier Transform (IDFT):

If a particular data sequence is assigned the i^{th} point on the 2-D constellation with coordinates:

$$(X_i, Y_i) \quad \Rightarrow \quad Z_i = X_i + jY_i \tag{7.2}$$

and if the encoder output is multiplied by a fine gain adjuster:

$$Z_i^{'} = g_i . Z_i \tag{7.3}$$

Then modulating transform determines between these weighted complex values the 512 real values:

$$x_k = \sum_{i=0}^{511} Z_i^{'} . \exp(j \frac{pki}{256}) \quad for\ k = 0,1,\cdots,511 \tag{7.4}$$

In order to generate real values of x_k we must augment them to have Hermitian symmetry:

$$Z_i^{'} = conj[Z_{512-i}^{'}] \quad for\ i = 257,\cdots,511 \tag{7.5}$$

Synchronization Symbols:

Synchronization symbol permits recovery of the frame boundary after micro-interruptions that might otherwise necessitate retraining, which could be costly.

- Symbol rate: $f_{sym} = 4{,}000$; Sub-carrier separation : $\Delta f = 4{,}312.5\,Hz$
- IDFT Size: $N = 512$ then a cyclic prefix of (40) samples could be used:

$$(512 + 40)x4000 = (512x4312.5) = 2{,}208{,}000 \tag{7.6}$$

- The cyclic prefix, however, is shortened to (32) samples and a synchronization symbol with a nominal length 544 is inserted after every 68 data samples:

$$(512 + 32)x69 = (512 + 40)x68$$

- Data pattern used in the synchronization symbol is a pseudo-random sequence with a specific generation structure and a seed.
- The last (32) samples of the output of the IDFT is appended to the block of 512 samples and read out to the DAC in sequence: $x_{480}, x_{481}, x_{482}, \cdots, x_{511}, x_0, \cdots, x_{511}$.

ATU-R Modulation by the Inverse Discrete Fourier Transform (IDFT):

- Maximum number of sub-carriers is 31 and carrier_16 is reserved for pilot.
- Modulating transform is adjusted to reflect that:

$$x_k = \sum_{i=0}^{63} Z_i^{'} . \exp(j \frac{pki}{32}) \quad for\ k = 0,1,\cdots,63 \tag{7.7}$$

The encoder generates only 31 complex values of $Z_i^{'}$ plus zero at DC and one real value if Nyquist frequency is used. In order to generate real values from (7.7), this time the Hermitian symmetry condition is changed to:

$$Z_i^{'} = conj[Z_{64-i}^{'}] \quad for\ i = 33,\cdots,63 \tag{7.8}$$

- For synchronization and cyclic prefix there are similar modifications to reflect that.
- Symbol rate: $f_{sym} = 4{,}000$; Sub-carrier separation : $\Delta f = 4{,}312.5\,Hz$
- IDFT Size: $N = 64$ then a cyclic prefix of (5) samples could be used:

$$(64+5)x4000 = (64x4312.5) = 276{,}000$$

- The cyclic prefix, however, is shortened to (4) samples and a synchronization symbol with a nominal length 68 is inserted after every 68 data samples:

$$(64+4)x69 = (64+5)x68$$

- Data pattern used in the synchronization symbol is a pseudo-random sequence with a specific generation structure and a seed.
- The last (4) samples of the output of the IDFT is appended to the block of 64 samples and read out to the DAC in sequence: $x_{60}, x_{61}, x_{62}, x_{63}, x_0, \cdots, x_{63}$.

(Details of these and many other aspects of the ADSL can be found in the ANSI Standard T1.413.)

7.7Carrierless AM/PM (CAP) Modulation

There has been a real competitive debate on the choice of the digital modulation scheme to be used in xDSL technologies. In particular, two techniques have been fully deployed with success in each: Discrete Multi Tone (DMT) and Carrierless Amplitude and Phase (CAP) modulation techniques. These systems are generically *Orthogonal Transceiver-based Architectures*. Before we discuss the DMT system in somewhat more detail, it is proper to introduce CAP modulation since it has significant overlaps with the QAM and TCM schemes we have seen earlier.

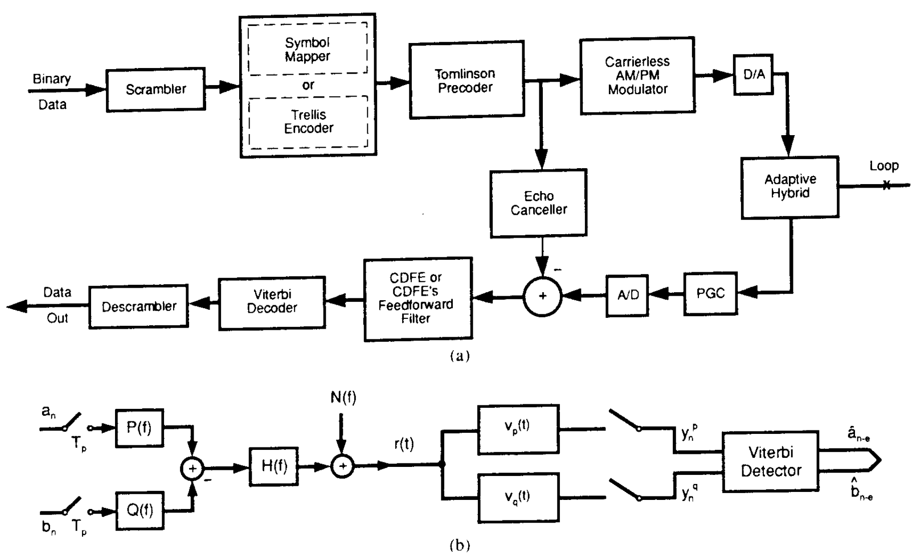


Fig. 1. (a) Generic transceiver architecture. (b) Carrierless AM/PM.

As it is shown in the generic transceiver architecture, the scrambled and convolutionally coded (TCM) bit stream is first converted to two independent symbol sequences $\{a_n\}$ *and* $\{b_n\}$ that are generated at a rate 1/T. The transmitted analog waveform is then formed by superimposing the outputs of two transversal bandpass filters,

$$s(t) \equiv \sum_n a_n.p(t-nT) - \sum_n b_n.q(t-nT) \tag{7.9}$$

To effectively decouple these two PAM dimensions at the output of the detector's feedforward filter; the impulse responses constitute a Hilbert transform pair (in-phase and quadratic terms).

It is worth noting that typical QAM does not have the same spectral capabilities as CAM. The latter provides the flexibility to generate asymmetric spectra as well, which is very critical for ADSL. The major difference between a CAM receiver of above figure and that of QAM that it does not incorporate any demodulation at any point in the received signal's path.

The optimum receiving front-end for carrierless maximum likelihood sequence estimation (CMLSE) is formed from two parallel passband filter that constitute a Hilbert transform pair as well. In general, the analog component of the optimum receiving filter for all carrierless equalization receivers can be seen to comprise a passband matched filter (PMF) defined in the frequency domain as:

$$H_{PMF}(f) \equiv \begin{cases} P^*(f).H^*(f)/n(f) & if\ \{f : N(f) > 0\} \\ 0 & otherwise \end{cases} \tag{7.10}$$

in tandem with an ideal phase splitter (v_p, v_q). *N(f)* is the power spectrum of the additive system noise (possibly colored.)

- Given a carrierless equalizer, a target error rate, and the channel impairment conditions, it is theoretically possible to maximize the system's throughput capability. For AWGN and NEXT-dominant environments, this optimizes system simulations have resulted in favorable performance.
- For the 4000-meter (12,000-ft) 24-gauge cable and a bit error rate of 10^{-6}, the maximum NEXT-limited throughput is $R = 1.8 Mb/s$. This information rate is achieved by mapping an average of 5.84 bits to each 2-D symbol, corresponding to a symbol rate of 308 Kbauds.
- If optimization is based on a transmission rate (1.544 Mb/s or half-rate). Then for T1 case, a symbol rate of 257.33 Kbaud on 64-point 2-D constellation is fairly optimal (HSDL choice!)
- At half the T1 rate, however, the optimal symbol rate is 193 kbaud on 16-point constellation.

7.8 Discrete Multi-Tone (DMT) Modulation

In general, multitone modulation methods used in transceivers use an optimized frequency division assignment of energy and bits to maximize the reliable communication over bandlimited channels. These systems are easily described because they use frequency-division-multiplexing (FDM) to transport a single-input data sequence on several carriers within the usable frequency range of a physical channel, such as twisted-wire pairs in POTS. The simplicity of MDT methods accounts for their use in some of the earliest data transmission modems of 1957. However, these techniques gained recognition with the introduction of voiceband and groupband modems in late eighties, such as Telebit's Trailblazer and NEC's multitone modems.

The excellent high-performance/cost tradeoff of DMT also makes it a strong candidate for the high-bit-rate xDSL technologies as it was accepted in the ADSL Standard T1.413.

The basic concept is to use a transform technique to divide the transmission system into a set of frequency-indexed sub-channels that appear to be modulated and demodulated independently. With a careful bit and transmit power allocation strategy, it has been demonstrated that such systems are capable of performing close to theoretical limits. There are a number of candidates to achieve frequency-indexing with proven and/or simulated results:

- Discrete Fourier Transform (DFT) based approach as it is standardized in ADSL.
- Multi-resolution orthogonal frequency division multiplexing (OFDM) based on Wavelets and its binary subsets:
- Complete-tree structured subband structures based on QMF or polyphase filtering and
- Octave-band structured (pruned-tree) subband structures.

Before we present the orthogonal frequency division transmultiplexers used in xDSL technologies, in particular, HDSL and ADSL, we would like to give more insight to the concept of multi-carrier systems.

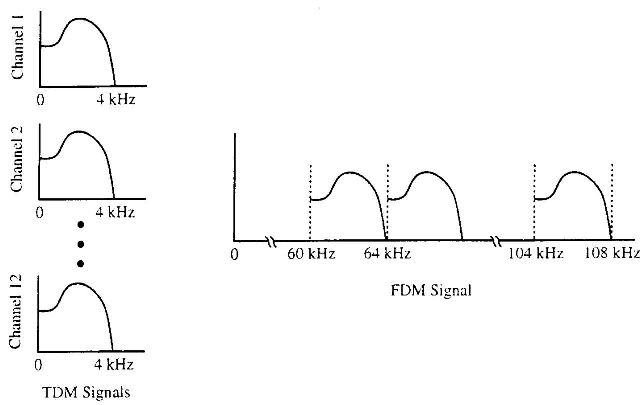


Figure 11.61 Spectrums of TDM signals and the FDM signal.

In these systems shown on the left a number of input signals (12 in the figure) are sampled at Nyquist rate or higher, or a sequence of input bits is divided into N subsymbols. They are interpolated by a factor of N (12 in our case) and modulated independently and summed to form a composite signal, which converted into an FDM analog signal, by a D/A converter. In our example, we have the band 60-108 kHz=12x4,000 Hz. At the receiver, the attenuated and corrupted signal is converted into a digital signal by A/D conversion followed by a bank of 12-demodulators whose outputs are then decimated, resulting the baseband signal set. In the case of binary data communication, the outputs of these decimators are passed through a parallel-to-serial converter to obtain the stream of data.

In the case of binary data signaling, a rectangular pulse train or a raised-cosine pulse sequence, we can modulate a stream of input data $a_r; r = 0,1,2,\cdots, N-1$ by a number of sub-carriers, which are harmonics of a sinusoidal signal: $Cos(2prt/T)$. Then modulated sub-carriers are summed and transmitted as one composite signal.

We can see from the following diagram that 6-pulses are to be transmitted and they will probably be received as shown. To alleviate the problems encountered in a bandlimited channel such the twisted-wire pair cable, each binary digit modulated by one of six harmonics of the fundamental sinusoid as shown above. We next add them up to form a composite signal. A the receiver, we have to push this composite signal through a bank of coherent demodulators to undo the process of modulation.

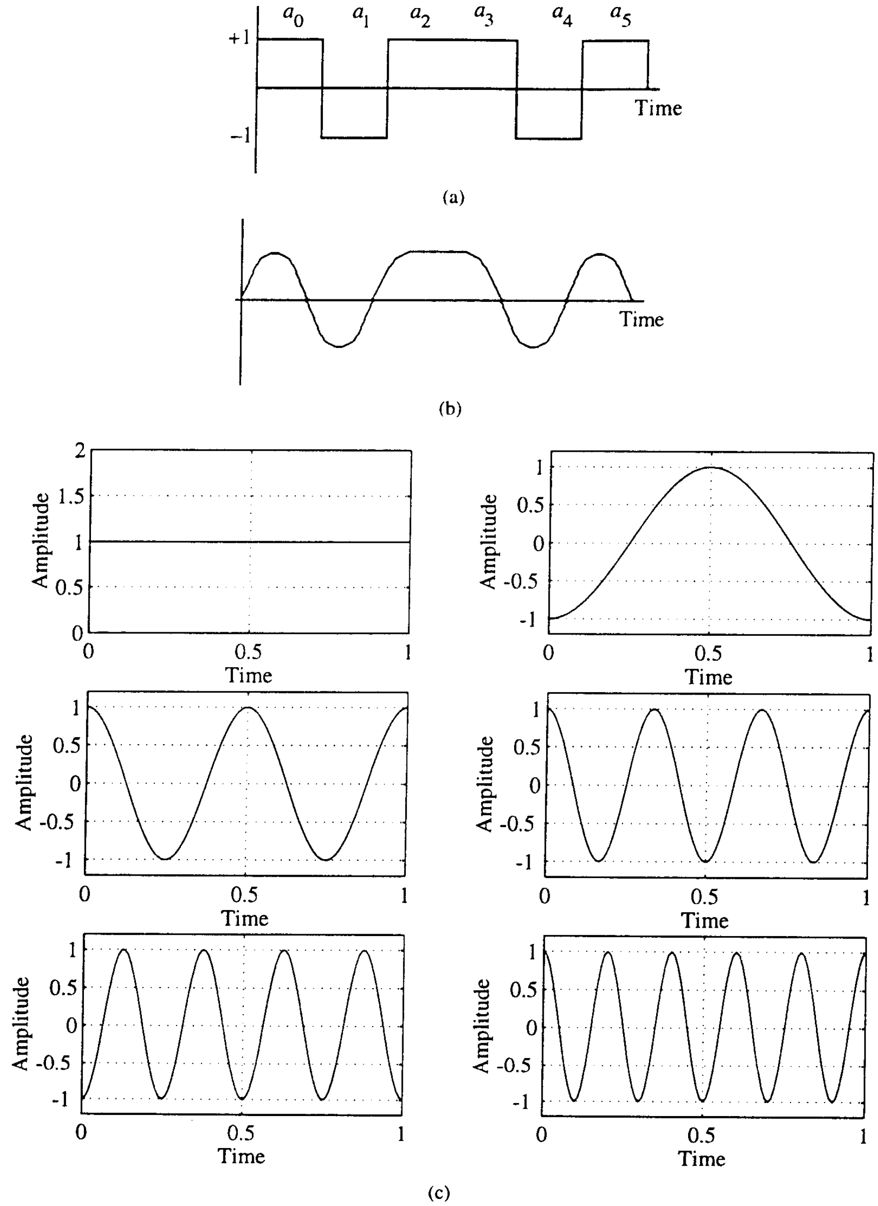


Figure 11.62 (a) Serial binary data stream, (b) baseband serially transmitted signal at the receiver, and (c) signals generated by modulating a set of subcarriers by the digits of the pulse train in (a).

What is the system structure DMT for xDSL systems, such as ADSL or HDSL?

Let $\{a_k[n]\}$ and $\{b_k[n]\}$ be two real-valued sequences of length M to be transmitted over a bandlimited channel. We assume that these are operating at a sampling rate F_S.

- Define a new set of complex sequence of length N=2M by:

$$a_k[n] \equiv \begin{cases} 0 & k = 0 \\ a_k[n] + jb_k[n] & 1 \le k \le (N/2) - 1 \\ 0 & k = N/2 \\ a_{N-k}[n] - jb_{N-k}[n] & (N/2) + 1 \le k \le N - 1 \end{cases} \tag{7.11}$$

It is worth noting that this definition guarantees a real-valued signal due to the Hermitian symmetry above.

- Let us apply an inverse DFT(IDFT) to for a new set of N signals:

$$u_l[n] = (1/N).\sum_{k=0}^{N-1} a_k[n].W_N^{-lk} \quad for\ l = 0,1,\cdots, N-1 \qquad (7.12)$$

where $W_N = e^{-j\frac{2p}{N}}$ is the N-point DFT kernel.

- Each of these N signals is then up-sampled by a factor of N and time-interleaved to generate a composite signal $\{x[n]\}$ operating at a rate: $N.F_S = 2.F_c$.
- The composite signal is converted into an analog signal $x_a(t)$ by passing it through a D/A followed by a synthesizing LP filter. The analog signal is then transmitted over the channel.

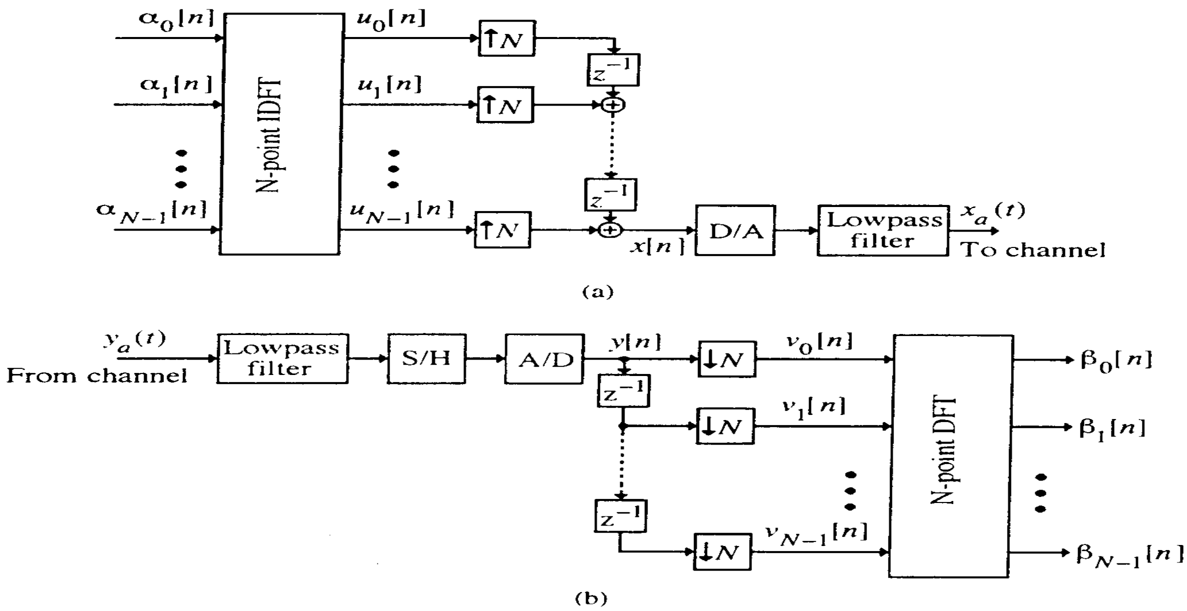


Figure 11.63 The DMT scheme. (a) Transmitter, and (b) receiver.

- At the receiver, the received possibly corrupted signal $y_a(t)$ pre-processed and digitized at a rate $NF_S = 2F_C$.
- They are de-interleaved by a delay chain of N-1 units whose outputs are then downsampled by a factor N to generate the signal set $\{v_l[n]\}$.
- Applying DFT to these N-signals will result in

$$b_k[n] = \sum_{l=0}^{N-1} v_k[n].W_N^{lk} \quad for\ l = 0,1,\cdots, N-1 \qquad (7.13)$$

- If we assume the frequency the frequency response of the channel is flat passband, and the processes between IDFT and DFT operations are lossless then we can show that

$$y[n] = x[n]$$

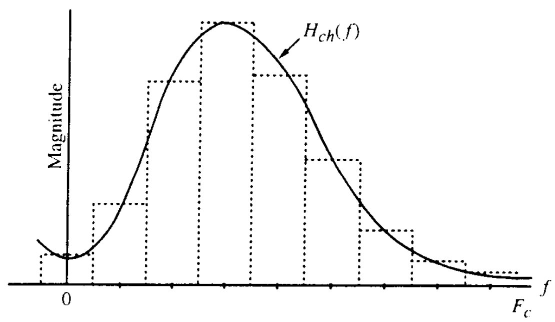


Figure 11.64 Frequency response of a typical bandlimited channel.

- Since the transmission channels have a bandpass frequency response $H_{ch}(f)$ with a magnitude response falling to zero at some frequency F_C, we need to use a channel equalizer for reliable transmission as discussed earlier.
- For large DFT length, as the case in ADSL systems, the channel can be treated as a composition of a series of contiguous narrow bandwidth bandpass sub-channels with flat top (dotted lines.)
- In this case, each sub-channel can be approximated by a single complex number given by the value of its frequency response at $w = 2pk / N$. These values can be determined by first transmitting a known signal to train the system as mentioned in ADSL structures above. The actual data samples are then divided by these complex numbers at the receiver to compensate for channel distortion.

7.5 HDSL/ADSL Systems using DMT Modulation

Finally, we will present an HDSL system due to Cioffi that uses the DMT modulation precisely the way it was discussed in the previous section.

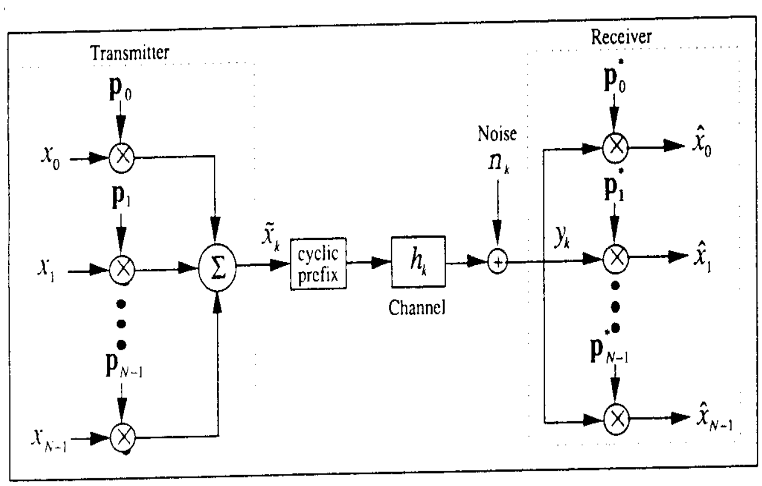


Fig. 3. Simplified illustration of DMT transceiver and channel.

Consider the simplified illustration of a DMT transmitter, receiver, and the channel for high-rate digital communication as shown in the left. As just presented, a block of input bits are divided into N-subsymbols and then independently modulated by N-Dim sampled sinusoid modulating vectors $\underline{P}_n$ and then summed to form a composite signal samples with Hermitian symmetry: $x_k; k = 0, \cdots, N-1$, where

$$x_n = x_{N-n}^* \quad for\ n = 1, \cdots, N-1. \tag{7.14}$$

In this notation, the n^{th} modulating vector is given by: $\underline{P}_n = [p_{n,0}\ p_{n,1} \cdots p_{n,N-1}]$, where each component is the inverse DFT kernel:

$$p_{n,k} = \frac{1}{\sqrt{N}} W_N^{2pkn/N} \qquad for\ k, n = 0, 1, \cdots, N-1 \tag{7.15}$$

- We can also think this as N/2 QAM channels. The transmitter of an HDSL system with an 8-tap Cyclic-Prefix, sampling rate 640 kHz and N=512 is depicted below.

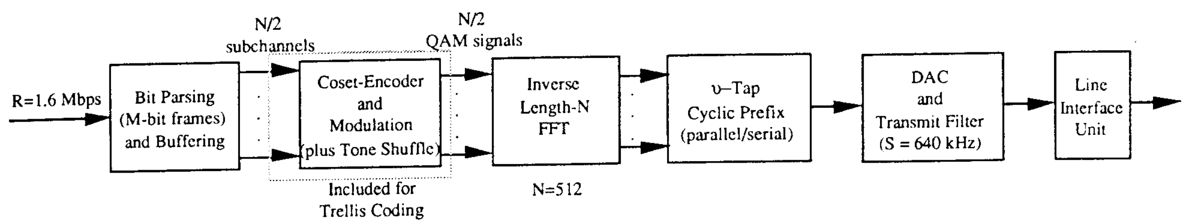


Fig. 4. Block diagram of a specific DMT HDSL transmitter.

- For a single twisted-pair cable to carry R=1.6 Mb/s single-duplex information, the input data stream is parsed into M-blocks:

$$M = \frac{N+v}{S}.R = \frac{512+8}{640{,}000}.1{,}600{,}000 = 1{,}300 \tag{7.16}$$

- These bits are then transformed into a maximum $N/2 = 256$ QAM subsymbols that are then applied to a trellis encoder. Trellis encoder sequentially processes the frequency-indexed subsymbols to avoid the large latency and memory requirement that would occur with multiple trellis encoders. This process includes shuffling on frequency-indexed subsymbols to avoid any minor correlation between the noise on adjacent channels under FEXT and/or NEXT regimes.
- The output of the trellis encoder is modulated using the inverse DFT kernels of (7.15).
- Finally, an 8-sample cyclic prefix as mentioned previously is inserted and sent to the line.
- The corresponding DMT HDSL Receiver block diagram is shown below.

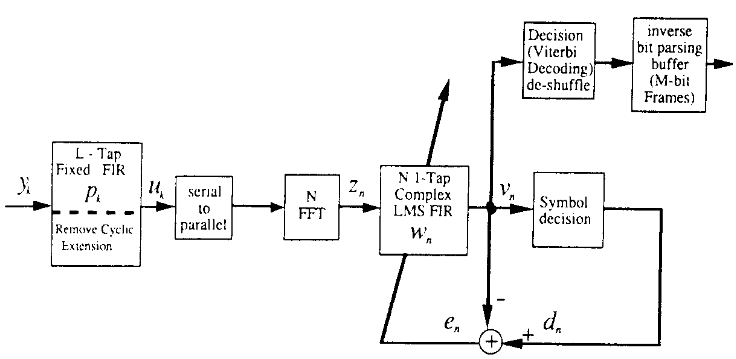


Fig. 6. Specific block diagram of a DMT HDSL receiver.

A short fixed L-tap feedforward equalizer is used to process the channel output y_k sequentially. This equalizer does not completely remove ISI, but rather confines the impulse response so that its length is approximately $v+1$ sample periods or less, which results in an approximately a minimum MSE AWGN channel. The last N samples of the $N+v$ samples that correspond to the transmit block are extracted from the equalizer output and a serial-to-parallel conversion is performed.

- An N-point FFT yields: $z_n ; n = 0,1,\cdots, N-1$ and they are multiplied by N-complex 1-tap adaptive filters, $w_n ; n = 0,1,\cdots, N-1$ so that a common decision device can be used to estimate the subsymbols on each subchannel.
- The initial tap settings are given by:

$$w_n = A_n^{-1} \quad n = 0,1,\cdots, N-1 \tag{7.17}$$

where:

$$A_n = \sum_{k=0}^{N-1} a_k . W_N^{-j2pkn/N} \quad for\ n = 0,1,\cdots, N-1 \tag{7.18}$$

And finally, a_k are the FFT coefficients of the channel impulse response or the samples of the channel frequency response as depicted with dotted lines in Figure 12.64 above.

- The resulting output data: $v_n = w_n . z_n$ is then decoded. If there is a Viterbi decoder in the system then it comes into operation.
- The symbol decisions are used only to derive an error signal, so that the LMS algorithm can be used for updating.

$$e_n = d_n - v_n \tag{7.20}$$

with the standard updating formulation:

$$w_{n+1} = w_n + 2.m_n.e_n.z_n^* \tag{7.21}$$

where m_n is the step-size for each sub-channel.

(The performance of this system and its power and computational requirements are discussed in detail in J.S. Chow, J.C. Tu, and J.M. Cioffi, “A Discrete Multitone Transceiver System for HDSL Applications,” IEEE Journal on Selected Areas in Communications, Vol. SAC-9, pp. 895-908, August 1991.)