What is ADSL?

Asymmetric Digital Subscriber Line (ADSL) is:

- A Telephone Loop Technology that uses existing phone lines
- Provides high-speed data and analog voice (Data over Voice)
- Dedicated digital line for an IP connection
- Data rates (North America) combinations of:
  - Upstream/downstream
    - 256 kbps/256 kbps
    - 384 kbps/128 kbps
    - 384 kbps/384 kbps
    - 384 kbps/1.5 Mbps
    - and many others
- Wide range of CPE options, including Ethernet 10baseT Interfaces.
- Dedicated ISP connection (static or dynamic addresses)
- Can support an IP subnet (from 1 to 254 IP addresses, depending on ISP)
- Priced lower than dedicated private line (T1) connections

The idea to distribute media through telephone wires are _NOT_ new, it has existed since early 1953, to distribute analog radio over telephone wires.

ADSL Applications

ADSL was designed to provide a dedicated, high-speed data connection for Internet/Intranet Access, using existing copper phone lines. This allows ADSL to work on over 60-80% of the phone lines existing in the U.S. without modification. Additionally, ADSL provides speeds approaching T1 (1.5Mbps), which are much greater than analog modems (56kbps) or ISDN (128kbps) services provided over the same type of line. ADSL is usually priced to be much less other dedicated digital services, and is expected to priced somewhere between T1 and ISDN services (including the ISDN usage charges). The Telcos see ADSL as a competitive offering to the Cable Company's Cable Modems, and as such, are expected to provide competitive pricing/configuration offerings. Although Cable Modems are advertised as having 10-30Mbps bandwidth, they use a shared transmission medium with many other users on the same line, and therefore performance varies, perhaps greatly, with the amount of traffic and other users. ADSL is positioned for Home and Small Office (SOHO) applications that require high-speed Internet Access. Since it also provides dedicated access, It can be used for interconnecting low-bandwidth servers to the Internet, and would provide a great access solution for 5-20 PCs in an Office location. It is also a great solution for those Linux power users that just want high speed access from home:-).
What is xDSL/DSL?
Digital Subscriber Line (DSL) provides a dedicated digital circuit from your home to the Telcos central office, using analog telephone line. DSL also provides a separate channel for voice phone conversations, which means analog calls (voice, fax, etc.) can be carried at the same time high-speed data is flowing across the line. DSL uses the frequency spectrum between 0kHz-4kHz for Analog Voice, and 4kHz-2.2MHz for data. xDSL is a generic acronym for a family of dedicated services, where the "x" stands for:

- **ADSL** Asymmetric Digital Subscriber Line: 1.5 Mbps-384kbps/384-128kbps
- **HDSL** High-bit-rate Digital Subscriber Line: 1.5 Mbps/1.5 Mbps (4Wire)
- **SDSL** Single-line Digital Subscriber Line: 1.5 Mbps/1.5 Mbps (2Wire)
- **VDSL** Very high Digital Subscriber Line: 13 Mbps-52 Mbps/1.5 Mbps-2.3 Mbps.
- **RADSL** Rate Adaptive Digital Subscriber Line: 384kbps/128kbps
- **UDSL** Universal Digital Subscriber Line: 1.0Mbps-384kbps/384kbps-128kbps

where Xbps/Ybps is X=Downstream Bit rate, Y=Upstream Bit rate

What are the various types of xDSL?

There are several forms of xDSL, each designed around specific goals and needs of the marketplace. Some forms of xDSL are proprietary, some are simply theoretical models and some are widely used standards. They may best be categorized within the modulation methods used to encode data. Below is a brief summary of some of the known types of xDSL technologies.

**ADSL**
Asymmetric Digital Subscriber Line (ADSL) is the most popular form of xDSL technology. The key to ADSL is that the upstream and downstream bandwidth is asymmetric, or uneven. In practice, the bandwidth from the provider to the user (downstream) will be the higher speed path. This is in part due to the limitation of the telephone cabling system and the desire to accommodate the typical Internet usage pattern where the majority of data is being sent to the user (programs, graphics, sounds and video) with minimal upload capacity required (keystrokes and mouse clicks). Downstream speeds typically range from 768 Kb/s to 9 Mb/s Upstream speeds typically range from 64Kb/s to 1.5Mb/s.

**ADSL Lite (see G.lite)**

**CDSL**
Consumer Digital Subscriber Line (CDSL) is a proprietary technology trademarked by Rockwell International.

**CiDSL**
Globespan's proprietary, splitterless Consumer-installable Digital Subscriber Line (CiDSL).

**EtherLoop**
EtherLoop is currently a proprietary technology from Nortel, short
for Ethernet Local Loop. EtherLoop uses the advanced signal modulation techniques of DSL and combines them with the half-duplex "burst" packet nature of Ethernet. EtherLoop modems will only generate hi-frequency signals when there is something to send. The rest of the time, they will use only a low-frequency (ISDN-speed) management signal. EtherLoop can measure the ambient noise between packets. This will allow the ability to avoid interference on a packet-by-packet basis by shifting frequencies as necessary. Since EtherLoop will be half-duplex, it is capable of generating the same bandwidth rate in either the upstream or downstream direction, but not simultaneously. Nortel is initially planning for speeds ranging between 1.5Mb/s and 10Mb/s depending on line quality and distance limitations.

**G.lite**
A lower data rate version of Asymmetric Digital Subscriber Line (ADSL) was been proposed as an extension to ANSI standard T1.413 by the UAWG (Universal ADSL Working Group) led by Microsoft, Intel, and Compaq. This is known as G.992.2 in the ITU standards committee. It uses the same modulation scheme as ADSL (DMT), but eliminates the POTS splitter at the customer premises. As a result, the ADSL signal is carried over all of the house wiring which results in lower available bandwidth due to greater noise impairments. Often a misnomer, this technology is not splitterless per se. Instead of requiring a splitter at customer premises, the splitting of the signal is done at the local CO. ADSL lite or g.lite is a sub-rated ADSL solution, with reduced digital signal processing requirements than full-rate ADSL systems. Under the name g.lite, the ADSL Lite has a downstream data rate of 1.5 Mbit/s or less. It has a similar reach to full rate ADSL systems. ADSL Lite is seen as the key to mass deployment of ADSL services, because the adaptation of this technology into cheap off the shelf modems is ideal for consumer use.

**G.shdsl**
G.shdsl is a ITU standard which offers a rich set of features (e.g. rate adaptive) and offers greater reach than many current standards. G.shdsl also allows for the negotiation of a number of framing protocols including ATM, T1, E1, ISDN and IP. G.shdsl is touted as being able to replace T1, E1, HDSL, SDSL HDSL2, ISDN and IDS technologies.

**HDSL**
High Bit-rate Digital Subscriber Line (HDSL) is generally used as a substitute for T1/E1. HDSL is becoming popular as a way to provide full-duplex symmetric data communication at rates up to 1.544 Mb/s (2.048 Mb/s in Europe) over moderate distances via conventional telephone twisted-pair wires. Traditional T1 (E1 in Europe) requires repeaters every 6000 ft. to boost the signal strength. HDSL has a longer range than T1/E1 without the use of repeaters to allow transmission over distances up to 12,000 feet. It uses pulse amplitude modulation (PAM) on a 4-wire loop. See also SDSL

**HDSL2**
High Bit-rate Digital Subscriber Line 2 was designed to transport T1 signaling at 1.544 Mb/s over a single copper pair. HDSL2 uses overlapped phase Trellis-code interlocked spectrum (OPTIS).
IDSL
ISDN based DSL developed originally by Ascend Communications. IDSL uses 2B1Q line coding and typically supports data transfer rates of 128 Kb/s. Many end users have had to suffice with IDSL service when full speed ADSL was not available in their area. This technology is similar to ISDN, but uses the full bandwidth of two 64 Kb/s bearer channels plus one 16 Kb/s delta channel. The differences between IDSL and ISDN are:

- ISDN passes through the phone company's central office voice network; IDSL bypasses it by plugging into a special router at the phone company end
- ISDN requires call setup; IDSL is a dedicated service
- ISDN may involve per-call fees; IDSL may be billed at a flat rate with no usage charges

MDSL
Usually this stands for multi-rate Digital Subscriber Line (MDSL). It depends on the context of the acronym as to its meaning. It is either a proprietary scheme for SDSL or simply a generic alternative to the more common ADSL name. In the former case, you may see the acronym MSDSL. There is also another proprietary scheme which stands for medium-bit-rate DSL. Confused yet?

RADSL
Rate Adaptive Digital Subscriber Line (RADSL) is any rate adaptive xDSL modem, but may specifically refer to a proprietary modulation standard designed by Globespan Semiconductor. It uses carrierless amplitude and phase modulation (CAP). T1.413 standard DMT modems are also technically RADSL, but generally not referred to as such. The uplink rate depends on the downlink rate, which is a function of line conditions and signal to noise ratio (SNR).

SDSL
Symmetric Digital Subscriber Line (SDSL) is a 2-wire implementation of HDSL. Supports T1/E1 on a single pair to a distance of 11,000 ft. The name has become more generic over time to refer to symmetric service at a variety of rates over a single loop. HDSL systems are now available that use 1 single copper pair at 2.048 Mbit/s. These are termed SDSL, Single Rate Digital Subscriber Line. Typical applications for this technology is providing T1 (1.5 Mbit/s) and E1 (2.048 Mbit/s) PAM lines.
The initial specifications for HDSL transmission in the United States called for dual duplex operation with a 2B1Q line code. The choice of line code is based on speed of implementation, since 2B1Q is the U.S. standard for basic digital transmission. In Europe, 2B1Q was also agreed on.

**UDSL**
Universal DSL. See G.1lite.

**VDSL**
Very High Bit-rate Digital Subscriber Line (VDSL) is proposed for shorter local loops, perhaps up to 3000 ft. Data rates exceed 10Mb/s. VDSL transmission can be used at the end of an optical fibre link for the final drop to the customer over a copper pair. In fibre-to-the-curb (FTTC) systems, the VDSL tail may be up to 500 meters long, and rates of 25 to 51 Mbit/s are proposed. In fibre-to-the-cabinet (FTTCab) systems, the tail may be over a kilometer, and rates of 25 Mbit/s are being considered.

VDSL uses DMT, especially because of its adoption for ADSL by ANSI. As for ADSL, the performance of the DMT for VDSL can be improved by bit interleaving and forward error correction.

The spectrum for VDSL transmission extends to 10 MHz for practical systems, as compared to about 1 MHz for ADSL transmission. However, it starts at a higher frequency of about 1 MHz, to reduce the interaction with other transmission systems at lower frequencies and to simplify the filter specification. Power levels for VDSL need to be lower than for ADSL because copper pairs radiate more at higher frequencies, generating greater electromagnetic interference.

**x2/DSL**

x2/DSL is a modem from 3Com that supports 56 Kbps modem communication but is upgradeable through new software installation to ADSL when it becomes available in the user's area. 3Com calls it "the last modem you will ever need."

<table>
<thead>
<tr>
<th>HDSL rate</th>
<th>No of pairs</th>
<th>Pair Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.544 Mbit/s</td>
<td>2</td>
<td>784 kbit/s</td>
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<tr>
<td>2.048 Mbit/s</td>
<td>3</td>
<td>784 kbit/s</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>1176 kbit/s</td>
</tr>
<tr>
<td></td>
<td>1</td>
<td>2048 kbit/s</td>
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<tr>
<td>DSL Type</td>
<td>Description</td>
<td>Data Rate Downstream; Upstream</td>
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<td>---------------</td>
<td>--------------------------------------------------</td>
<td>---------------------------------</td>
</tr>
<tr>
<td>IDSL</td>
<td>ISDN Digital Subscriber Line</td>
<td>128 Kbps</td>
</tr>
<tr>
<td>CDSL</td>
<td>Consumer DSL from Rockwell</td>
<td>1 Mbps downstream; less upstream</td>
</tr>
<tr>
<td>DSL Lite</td>
<td>&quot;Splitterless&quot; DSL without the &quot;truck roll&quot;</td>
<td>From 1.544 Mbps to 6 Mbps</td>
</tr>
<tr>
<td>G.Lite</td>
<td>&quot;Splitterless&quot; DSL without the &quot;truck roll&quot;</td>
<td>From 1.544 Mbps to 6 Mbps</td>
</tr>
<tr>
<td>HDSL</td>
<td>High bit-rate Digital Subscriber Line</td>
<td>1.544 Mbps duplex on two</td>
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<td>twisted-pair lines;</td>
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<td></td>
<td></td>
<td>2.048 Mbps duplex on three</td>
</tr>
<tr>
<td></td>
<td></td>
<td>twisted-pair lines</td>
</tr>
<tr>
<td>SDSL</td>
<td>Symmetric DSL</td>
<td>1.544 Mbps duplex (U.S. and</td>
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<tr>
<td></td>
<td></td>
<td>Canada); 2.048 Mbps (Europe) on</td>
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<tr>
<td></td>
<td></td>
<td>a single duplex line downstream</td>
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<td></td>
<td></td>
<td>and upstream</td>
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<tr>
<td>ADSL</td>
<td>Asymmetric Digital Subscriber Line</td>
<td>1.544 to 6.1 Mbps downstream;</td>
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<tr>
<td></td>
<td></td>
<td>16 to 640 Kbps upstream</td>
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<td></td>
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<tr>
<td>RADSL</td>
<td>Rate-Adaptive DSL from Westell</td>
<td>Adapted to the line, 640 Kbps</td>
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<tr>
<td></td>
<td></td>
<td>to 2.2 Mbps downstream; 272 Kbps</td>
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<tr>
<td></td>
<td></td>
<td>to 1.088 Mbps upstream</td>
</tr>
<tr>
<td>UDSL</td>
<td>Unidirectional DSL proposed by a company in Europe</td>
<td>Not known</td>
</tr>
<tr>
<td>VDSL</td>
<td>Very high Digital Subscriber Line</td>
<td>12.9 to 52.8 Mbps downstream;</td>
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<tr>
<td></td>
<td></td>
<td>1.5 to 2.3 Mbps upstream</td>
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<tr>
<td></td>
<td></td>
<td>1.6 Mbps to 2.3 Mbps downstream</td>
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</table>
ANSI T1.413 defines the requirements for the single asymmetric digital subscriber line (ADSL) for the interface between the telecommunications network and the customer installation in terms of their interaction and electrical characteristics. ADSL allows the provision of voiceband services (including POTS and data services up to 56 kbit/s) and a variety of digital channels. In the direction from the network to the customer premises, the digital bearer channels may consist of full duplex low-speed bearer channels and simpler high-speed bearer channels; in the other direction, only low-speed bearer channels are provided.

This transmission requires a normal, single twisted pair of telephone wires.

This standard includes descriptions for:

- echo cancellation
- trellis code modulation
- loop timing at either end
- dual latency
- transport of network timing reference
- transport of STM or ATM or DTM
- reduced overhead framing mode

To be able controlling and service user equipment as well as switches and routers in the network cloud the EOC was invented. EOC is compatible with ATM similar mechanisms.

Embedded Operations Channel (EOC) - Uses a DS0 (64kbps, clear channel) on the primary DS1 facility (time slot # 12). And another DS0 is used for EOC protection on a separate DS1 facility. The EOC channel is primarily used for carrying Path Protection Switching (PPS) and OAM&P-related messaging. The higher layer protocols used with this channel include LAPD subset for layer 2, Convergence sub-layer & Common Management Information Service Element (CMISE)/Remote Operations Services Element (ROSE)/ASN.1 sub-layer for layer 3.
Why so many speeds?
ADSL has to work over existing phone lines, which were designed 100 years ago, and were never designed for digital services (See the FAQ answers for more information). Also, ADSL is a new service, and all the providers are trying to find the right price/feature combinations that will make it in the market.
For the average user, the basic way of thinking about it is to segment the options into three categories:

- **Low End Residential**
  Speed ranges from 384kbps-128kbps, Asymmetric

- **High End Residential or Business End User**
  Speed ranges from 1.5Mbps-384kbps, Asymmetric

- **High End Server**
  Speed ranges from +2.0Mbps-1.1kbps, Symmetric

What is crosstalk?
Crosstalk refers to the interference between channels. In the xDSL world, the interference between nearby cables can have a negative impact on the performance of the affected cable(s). Have you ever been on the phone and heard some other conversation, not yours, in the background? If so, you have experienced the effect of crosstalk.

Near-end crosstalk (NEXT) occurs when the transmitter sends a signal and a nearby transceiver at the same end of link, through capacitive and inductive coupling, "hears" the signal.

Far-end crosstalk (FEXT) occurs when the transmitter sends a signal and a transceiver at the far end of the link, through capacitive and inductive coupling, "hears" the signal. FEXT will be of more concern in an asymmetrical system such as ADSL than symmetrical systems like HDSL. This is because strong signals originating from the near end, can interfere with the weaker signals originating at the far end.
What is the effect of noise?

Noise may be defined as the combination of unwanted interfering signal sources whether it comes from crosstalk, radio frequency interference, distortion, or random signals created by thermal energy. Noise impairs the detection of the smallest analog levels which may be resolved within the demodulator. The noise level along with the maximum clip level of an analog signal path set the available amplitude dynamic range.

The maximum data rate of a modem is limited by the available frequency range (bandwidth) and signal-to-noise ratio (SNR) which is amplitude dynamic range. If more of either is available, more bits may be transferred per second. The information carrying limit was discussed theoretically by Claude Shannon and is known as Shannon's limit, or information theory.

Because modems run close to Shannon's limit today, no further advances will be made to traditional telephone line modems other than incremental improvement of V.90. The frequency range of the audio channel is very limited at about 4 kHz. V.34+ modems are limited to a maximum data rate of 33.6Kb/s by an SNR of about 36 dB caused mostly by network PCM quantization noise. While V.90 improves the SNR by utilizing the network PCM levels directly, it is still subject to Shannon's limit.

xDSL modems take advantage of the spectrum above the telephone audio channel. While operating with somewhat less amplitude dynamic range they increase data rates by greatly increasing the frequency range of the communication signal (from about 10 kHz to over 1.0mHz). To do this they require the installation of special equipment at the central office and customer premise.

Shannon’s formula Relationship between bandwidth, frequency, and data rate.

Bit rate for a periodic signal with frequency f with two levels: bit rate (capacity) = 2f since each period can encode 2 bits.

\[ \text{Nyquists theorem: capacity} = 2(W)\log_2(M) \]

\[ \text{Shannons formula: capacity} = (W)\log_2(1 + S/N) \]

- \( M \) = number of signal levels
- \( W \) = bandwidth in Hz.
- \( S/N \) = signal to noise ratio in decibles. dB = 10log10(Signal Power/Noise Power)
- Bandwidth = (highest frequency - lowest frequency) 
(Spectrum = range of frequencies.)

All transmission media have limited bandwidth.

**Limited bandwidth will distort the signal.** Thus, the bandwidth determines the quality of a signal. To analyze the effect of limited bandwidth on the quality of a signal, Fourier analysis can be used. Depending on modulation different S/N ratios is necessary to satisfy BER (Bit Error Ratio) demands in relation to quality. Practical values is in the range of at least in the range 6dB to 12dB. Observe! Shannon did not predict data compression. The table show practical speeds for standard low quality 0,4mm CU wire used worldwide for POTS.
What are echo suppressors and echo cancellers?

These are active devices used by the phone company to suppress the reflection of an analog signal or positive feedback (singing) on the phone network. The effect of the echo on a voice connection is undesirable. Imagine that as you spoke into the phone's microphone, there was a short delay and you hear your own voice back over the earpiece. A soft echo that comes back fast enough is not bothersome to the average person. A more delayed echo is annoying.

A echo suppressor works by allowing only one direction to transmit at a time so as to entirely eliminate the effect of an echo. An echo suppressor is able to switch between each end very rapidly, typically within 5msec. Network echo suppressors make full-duplex communication impossible. However, modems can deactivate these devices by sending the 2100 Hz answer tone at the beginning of the connection.

An echo canceller subtracts a locally generated replica of the predicted echo based on the signal propagating in the forward direction. Echo cancellers do allow full-duplex operation and are generally preferred over echo suppressors in voice calls. But when network echo cancellers compete with echo cancellers within the modem they are problematic. Typically they reduce data rates to 9.6Kb/s or lower. Network echo cancellers are deactivated by placing 180 degree phase reversals every 450msec on answer tone. As long as carrier is maintained, they are supposed to remain deactivated.

xDSL is not affected by network echo suppressors/cancellers because they are part of the CODEC signal processing.

What is a CODEC?

CODEC is an abbreviation for coder/decoder. Specifically it converts a voice grade analog signal to u-law or A-law encoded samples at an 8 kHz sampling rate. xDSL bypasses the CODECs at the central office by separating the xDSL signal and voice frequencies in a POTS splitter. The voice signal is passed to a CODEC while the xDSL signal terminates in a DSLAM, the xDSL equivalent of a CODEC. All analog modems will be limited by CODEC’S. Codecs can be seen as low pass filters.
Modulation methods

There are three different modulating techniques presently used to support xDSL. These are carrierless amplitude phase modulation (CAP), 2B1Q and discrete multi-tone modulation (DMT). But to understand them, we need to look at the classic modulations technologies used by xDSL within those three.

What is DPSK/QPSK?

Dual Phase Shift Key and Quadruple Phase Shift Key, used for baseband modulation/encoding of digital data trains on a single carrier. The method uses different phases to represent digital values. DPSK uses two and QPSK uses four. This is the elementary methods of most digital modulations. Often people refers to bit spectral density, this is the number of bit per Hz bandwidth. 2Bit per HZ for QPSK.

What is QAM?

Quadrature amplitude modulation (QAM) is a method for encoding data on a single carrier frequency. The modulation encodes data (or bits) as discrete phase plus amplitude changes of a carrier tone. The phase vectors are arranged in a pattern of points called a constellation from which the transmitted point is selected based on the data to be sent.

The modem sends the symbols as abrupt changes in phase and amplitude, but only as what emerges from a sharp cutoff filter which carefully limits the bandwidth. The transmitted signal occupies slightly more than ±1/2 the modulation rate either side of the carrier frequency. The excess bandwidth, perhaps as much as 10%, is required for recovering symbol timing within the remote receiver.

The receiver has to pick which point was transmitted with great reliability. It may employ adaptive equalization or other methods to reduce intersymbol interference to levels which are acceptable for discriminating the received point. The background noise level of the receiver limits the number of distinct constellation points which may be reliably determined, and hence limits the data rate for a given symbol rate.
QAM has become the dominate modulation for high speed voice band modems. Examples are V.22bis, V.27, V.29, V.32bis, V.34. About every 2/3 of a carrier cycle the phase or amplitude is changed to a new value. This signaling rate is known as the baud (or symbol) rate. The highest QAM baud rate in use today for telephone line modems is 10/7 of 2400 Hz or about 3429 baud on a 1920 Hz carrier in V.34. By encoding something between 9 to 10 bits per baud a final data rate of 33.6Kb/s is developed. To encode this number of bits, over 1000 different phase/amplitude values must be resolved by the receiver. This is a nontrivial process involving adaptive equalizers, trellis/viterbi coding, and other highly sophisticated signal processing.

Transmit path:
scrambler -> symbol generator -> 3x upsample (S1,0,0,S2,0,0,S3,...) ->
complex transmit baseband FIR filter -> e^jwt carrier modulation ->
scale real signal output -> DAC converter

The baseband filter is about 3 dB down at ±1/2 symbol rate, so for 3429 baud the signal out of the filter extends from -1715 Hz to +1715Hz. This is shifted by the positive 1920 Hz carrier to +205Hz to +3635Hz. One can see that this just fits in the frequency spectrum of the voice band telephone network. This filter, the analog electronics and the phone channel smear any given symbol over a 10msec period of the signal (about 32 symbols).

The scrambler is very important. It randomizes the signal so an adaptive equalizer in the remote modem can build the inverse channel response (including the transmit filter). The smearing (or intersymbol
interference) is largely eliminated by dynamically adjusting adaptive equalizer coefficients with the goal of minimizing least square error in the received points. The major adaptation is done during the training phase, although the feedback loops remain active throughout the connection. Other impairments to be solved are gain normalization, timing recovery, carrier offset frequency, phase jitter removal, and echo cancellation.

### Simplified QAM phase diagrams

**What is PCM?**

Pulse code modulation (PCM) is used in the phone network to reduce the data rate required for voice grade audio to less than 64Kb/s. It uses either u-law (North America) or A-law (Europe) as the compression method. Any given 8 kHz analog audio sample is converted to 4 bits of mantissa, 3 bits of exponent, and a sign bit. This code has a characteristic that quantization noise is proportional to signal amplitude and does not become objectionable to the average telephone user. For a conventional modem this noise floor limits the available dynamic range to about 36 dB which sets the maximum data rate. The least significant bit of the mantissa may be periodically stolen for signaling within the phone network (called robbed-bit signaling) further increasing the noise.

The 8-bit codes are processed through the telephone switching network in fixed time slots. There exists an ever increasing hierarchy of data rates to support this. A DS0 is a 64Kb/s time slot. 24 DS0s become a DS1. 4 DS1s become a DS2 (now obsolete). 7 DS2s become a DS3, etc.

The physical layer of a DS1 (T1) may be remodulated as alternate mark inversion for passing over a wire pair as a method to concentrate local loops. Repeaters regenerate the signal every 6000-9000'. These signals may coexist with xDSL in the same wire bundle.
What is PAM?

Pulse amplitude modulation (PAM) is the physical layer of an ISDN or HDSL connection. The modulation consists of sending discrete amplitude levels (symmetric about 0 volts) at a regular rate. Both use the two binary quaternary (2B1Q) line code. Four analog voltages (called quaternary symbols) are used to represent the four possible combinations of two bits. These symbols are assigned the names +3, +1, -1, and -3. So each amplitude level being held for one symbol time communicates two bits.

The following diagram is typical of the 2B1Q waveform at the transmitter:

```
+3 =  2.64V + .--.        .--.                    .--.
+1 =  0.88V +   |  |        |  |                    |
-1 = -0.88V + --'     |  |     |  .-----'  `--.     |
-3 = -2.64V +         |  |     |  |           |     |
```

One might assume this is a digital signal relative to the definition in, but by the time the signal has reached the receiver these discrete levels have diffused into each other because of phone line induced amplitude and phase distortion. This is called intersymbol interference. Therefore an adaptive equalizer must be used to restore the levels to values which may be discriminated for recovering the data. The symbol timing is recovered by examining the squared signal energy for a tone at the modulation rate. Transitions between levels cause the instantaneous power to dip on average provided there is adequate excess bandwidth.

PAM differs from the other modulations in that it is baseband modulation and does not use a carrier. Some versions of HDSL increase the number of levels to 16 which communicates four bits per symbol in the same bandwidth.

2B1Q

2B1Q represents a straightforward signal type that has two bits per baud arranged as a quaternary or four level pulse amplitude modulation scheme. It essentially transmits data at twice the frequency of the signal. Used by ISDN. See PAM.

4B3T Used by ISDN. See PAM.
What is V.90?

V.90 is actually a variant of PAM. It has 256 PCM levels from which to choose a more limited set. The spacing between levels is set by the u-law or A-law characteristic described in [6.2]. The inner levels become more closely spaced so some of these must be excluded for reasons of limited signal-to-noise ratio. In addition, outer codes are excluded to keep transmit power on the local loop below -12dBm, a formal limit established by the FCC. V.90 includes a spectral shaping algorithm to prevent sending signal at DC.

V.90 bypasses the problems associated with a conventional modem. It recognizes that with enough signal processing the original PCM samples sent by the phone company may be resolved as individual levels using a 16-bit A/D converter on the receiving end. Audio is sent through the digital network as 8-bit u-law or A-law samples. Of course, the telco D/A converter, reconstruction filter, and phone network blur the levels into one continuous signal, so it's up to the receiver to reconstruct what was sent. An additional problem is recovering symbol (i.e. PCM sample) timing information which must be inferred from the residue of modulation at a frequency around 4 kHz. By just selecting a limited set of codes with say 64 levels, 6 bits per 8 kHz symbol may be sent for a data rate of 48Kb/s. More levels, more data, but a maximum of about 53.3Kb/s is a practical limit.

What is CAP?

Carrierless amplitude and phase (CAP) modulation is a proprietary standard implemented by Globespan Semiconductor. While the name specifies that the modulation is "carrierless" an actual carrier is imposed by the transmit band shaping filter through which the outbound symbols are filtered. Hence CAP is algorithmically identical to QAM. The upstream symbol rate is 136K baud on a 113.2KHz carrier, while the downstream symbol rate is 340K baud on a 435.5KHz carrier, 680K baud on a 631KHz carrier, or 952K baud on a 787.5KHz carrier. This allows the modem to be symbol rate adaptive to varying line conditions (see RADSL). The QAM modulation is also rate adaptive by varying the number of bits per symbol.

One advantage CAP claims to have is a lower peak-to-average signal power ratio relative to DMT. This means that the drivers and receivers may operate at lower power than DMT because they are not required to have the peak signal capacity that is required in the DMT circuitry. This is mitigated by the infrequency of the really high signal peaks in DMT which may be just considered to be another form of noise if they happen to clip.

CAP's principle advantage is its installed base of modems. It is actively being deployed in many trial markets and is available from several manufacturers. CAP ADSL offers 7.5 Mbit/s downstream with only 1 Mbit/s upstream. Compared to DMT it is slightly inferior and DMT is now the official ANSI, ETSI and ITU-T standard for ADSL. One twisted copper pair supports POTS on the 0-4 KHz range. CAP based DSL technology uses frequencies sufficiently above the POTS "voice band" to provide bandwidth for low-speed upstream and high speed downstream channels.
What is DMT?

Discrete multitone (DMT) modulation is a method by which the usable frequency range is separated into 256 frequency bands (or channels) of 4.3125KHz each. These are intimately connected to the FFT (fast Fourier transform) algorithm which DMT uses as its modulator and demodulator. The FFT is not perfect in separating the frequencies into individual bands, but it does well enough, and it generates spectra which are fully separable on the receiving end. By dividing the frequency spectrum into multiple channels DMT is thought to perform better in the presence of interference sources such as AM radio transmitters. It is also better able to focus its transmit power on those portions of the spectrum in which it is profitable to send data.

The assignment of channels is less flexible, but typical settings might be channels 6-31 for upstream (24KHz-136KHz), 32-250 for downstream (136KHz-1.1MHz). The modulation used on any given frequency channel is QAM. Channels 16 and 64 are reserved for pilot tones which are used to recover timing. The number of bits per symbol within each channel may be independently selected allowing the modem to be rate adaptive.

The use of the FFT is considered to be somewhat substandard to other orthogonal transformations such as the discrete wavelet transform which do a better job of isolating the individual frequency spectra. The FFT is chosen for its computational efficiency.

While DMT is off to a slow start in the marketplace, it is expected to dominate for two reasons: it is thought to perform better for technical reasons and there is an ANSI standard behind it (not to mention Intel/Microsoft support).

xDSL members is based on OFDM
Orthogonal Frequency Division Multiplexing, this means that we have several carriers in a broadband used for transmitting (or receiving) one data train in parallel. This is absolutely necessary to get as close as possible to the Shannon limit. This technique is also belonging to the Spread Spectrum family, a former military standard for making it hard to listen on communications.

The xDSL members uses different numbers of carriers, in xDSL, named tones. Some uses more
tones for traffic downstream to the customers and less upstreams, other uses the same numbers in both directions when others uses all in one direction and then reverses direction and uses all upstreams. Some of the carriers are used for synchronization and service channels. It is also possible to select dynamic numbers of carriers, thus for adaptation to the local wire between CO and customer device.
**ANSI T1.413** specifies DMT modem for ASDL applications

**UpStream:** 275 kHz sampling rate, 32 tones 0 … 138 kHz

**DownStream:** 2.208 MHz sampling rate, 256 tones 0 … 1.104 MHz

Symbol rate 4000 symbols/s. Each sub-channel is 4.3 kHz wide

max rate 32 kb/s per channel (compare to V.90 modem)

Half or Full duplex

ANSI T1.413 divides the frequency range between 0 MHz and 1.1 MHz in 288 subchannels or tones, 32 subchannels for upstream traffic and 256 subchannels for downstream traffic. The tone spacing is approximately 4.3 kHz. The lower part (0-4 kHz) of the frequency range is used for POTS, and from 26 kHz up to 1.1 MHz there are 249 channels used for the ATM traffic.

Quadrature Amplitude Modulation (QAM) is used as the modulation scheme in each of the subchannels. Therefore each subchannel can bear a moderate amount of data.

The video and data information is transmitted as ATM cells over the copper wires. POTS uses baseband signalling (Baseband on twisted pair wires. Baseband is a one-channel signal, a flow of ones and zeros). The telephony is separated from the ATM cell flow at the customers premises and the DSLAM (Digital Subscriber Line Access Module) by means of a filter or splitter.

The ANSI (American National Standards Institute) standard specifies two options, one with Frequency Division Multiplexing where the lower subchannels are used in the upstream direction and the rest is used in the downstream direction. The other option uses echo cancellation where the same subchannels are used for upstream and downstream traffic. These are also referred to as category 1 and category 2 modems, respectively. Category 1 modems provide a bandwidth of 6 Mbit/s in the downstream direction and up to 640 Kbit/s in the upstream direction.

Category 2 modems provide a bandwidth of up to 8 Mbit/s in the downstream direction and up to 1 Mbit/s in the upstream direction.

External disturbances and attenuation can impact on the transmission quality of the subchannels. One of the advantages with DMT is that it adjusts the bandwidth on each channel individually, according to a signal/noise ratio. This is termed Rate Adaptive. In other words, there will be a higher bit rate (bandwidth) where the noise (disturbances) is low, and a lower bit rate, or none, where the noise is high. This adjustment occurs when the ADSL line is being initialised before it is taken into service.

**ANSI T1.413a** also defines a method for generating a 16-bit vendor ID based on the 4-letter vendor code assigned according to T1.220.
The lower part in frequency spectra, 0-4KHz, is for the analog POTS. The interleave up to 26KHz is necessary to avoid noise.

Sub-carrier spacing is 4.3125 kHz - 256 total sub-carriers

<table>
<thead>
<tr>
<th>Carrier</th>
<th>Frequency</th>
<th>Meaning</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0 Hz</td>
<td>DC - not used for data</td>
</tr>
<tr>
<td>5</td>
<td>25 kHz</td>
<td>lower limit for upstream data</td>
</tr>
<tr>
<td>18</td>
<td>80 kHz</td>
<td>Approx. limit for 2B1Q ISDN</td>
</tr>
<tr>
<td>28</td>
<td>120 kHz</td>
<td>Approx. Limit for 4B3T ISDN</td>
</tr>
<tr>
<td>32</td>
<td>138 kHz</td>
<td>upper limit for upstream data</td>
</tr>
<tr>
<td>64</td>
<td>276 kHz</td>
<td>Pilot - not used for data</td>
</tr>
<tr>
<td>256</td>
<td>1104 kHz</td>
<td>Nyqvist - not used for data</td>
</tr>
</tbody>
</table>
**ADSL over ISDN or for short AOI**

ISDN provides the user with two 64-Kbps channels (referred to as B or bearer channels) as well as a lower-speed signaling channel (D channel) that is often used for X.25 data packet services. Each of the two 64-kbps channels can support the simultaneous transport of voice and data. On the voice side, ISDN offers enhanced calling features such as digital voice quality, speed dialing, call return, caller ID, call forwarding, and local number portability. On the data side, both bearer channels can be bonded, yielding a 128-Kbps bidirectional data connection. ADSL over ISDN (AOI) promises even higher speeds over the same link while preserving the features of ISDN.

We have now seen that it is possible to implement xDSL on copper wire supposed for POTS, and with various speeds and qualities, depending on distance from CO. We have also seen that there is a form of ADSL over ISDN called IDSL, not to mix up with ADSLoISDN. IDSL was only the digital service of ISDN.

However customers with ISDN must also be able to implement xDSL without scrapping their existing ISND. As you know, there are different forms of ISDN. N-ISDN with basic rate 2B+D (BRI) is the most common end-user, the other form is primary rate 30B +1D. B-ISDN Broadband ISDN with up to 2^24 B channels of 64Kbit/s each.

There are several ways running ADSL over ISDN. xDSL uses only the higher frequency spectra of the copper wire, ISDN is told to use all the bandwidth of the copper wire since it is a base band signal see modulations 2B1Q and 4B3T. It is perfectly possible to use the upper part here to, but ISDN is using more bandwidth than analogue POTS. We just increase the interleave to 80 for 2B1Q with 160kBit/s and 120kHz for 4B3T. But there are some small spectral problems with ISDN and ADSL, they overlap eachother in the lower part of spectra.

AOI’s main hurdle is the overlap of the ISDN and ADSL frequency spectrums (see Figure below). ISDN occupies the frequency spectrum of the twisted pair up to 80 kHz (2B1Q ISDN line coding) or 120 kHz (4B3T ISDN line coding). ADSL occupies the frequency spectrum from 26 kHz to 1.1 MHz (256 bins, each 4 kHz wide). A guard band from 4 to 26 kHz separating POTS and ADSL services is left to allow for splitter (filter) rolloff.

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**Echo-cancellation (EC) or Frequency Division Multiplexing (FDM) Modes**
To eliminate the spectrum overlap, one approach would be for ADSL to occupy its normal frequency spectrum and transmit ISDN voice and data bit streams inband (within the ADSL payload). Both G.Lite and G.DMT would be free to activate and use the twisted pair unencumbered. A splitter would no longer be necessary for either G.DMT or G.Lite and this approach would remain ITU compliant. However, the routing of voice-ISDN over ADSL means that all traffic must pass through the ADSL modem, creating two major issues. The first problem is that the power requirements of an ADSL modem make it unlikely that the modem can be line powered. A local power failure could interrupt lifeline voice services. Another issue is that the ADSL processing of the ISDN information would introduce latency exceeding 2 ms, violating the ISDN standard.

(This is also a suggested solution to transport regular ISDN traffic using xDSL as transport, in best case with short distances is can be possible to achieve 1 E1 channel over traditional copper wire.)

An alternative to in-band transmission would be to move the entire ADSL signal above the frequency spectrum utilized by ISDN. This is called up-banding. The ADSL activation procedure would no longer be able to use the spectrum normally allotted to it for startup, as this spectrum would fall directly into the ISDN spectrum. Thus, both ADSL modems would need to be up-banded. This approach is under consideration by the G.DMT specification as Annex B.

G.DMT Annex C addresses the issues of AOI in Japan. Japanese loops carry more wires, use different insulation techniques, utilize half-duplex ISDN, and often carry power within the same conduit.

Up-banding will allow ADSL and ISDN services to coexist, preserving lifeline communications and adhering to the ISDN standard.

For B-ISDN there are yet no solutions with ADSL, B-ISDN is not yet fully standard either.
**ATM over ADSL**

Finally there is also suggestions how to transport ATM traffic with ADSL. It is also fully under T1.314.

**Definition of functional blocks.**

**Access Node.**

The Access Node performs the adaptation between the ATM Core Network and the Access Network. In the downstream direction it may perform routing/demultiplexing, while in the upstream direction it may perform multiplexing/concentration.

B-NT1, B-NT, B-NT+TA or B-NT+TE.

This functional block performs the functions of terminating the ADSL signal entering the users premises via the twisted pair cable and providing either the T, S or R interface towards the terminal equipment. Such an interface may be absent in the case of integration of the functional block with the Terminal Equipment. Its functions are: terminating/originating the transmission line, handling the transmission interfacing and OAM functions. In addition, it may optionally include routing/multiplexing of the ÔFastÔ and ÔInterleavedÔ flows.

**ATM layer functions.**

In the Access Node this function performs in the downstream direction routing/demultiplexing on a VPI and/or VCI basis, while in the upstream direction multiplexing/concentration, again on a VPI and/or VCI basis. If implemented in the B-NT1, B-NT, B-NT+TA or B-NT+TE, this function performs cell routing/(de)multiplexing to the ÔFastÔ or ÔInterleavedÔ channel in the upstream direction and routing/(de)multiplexing of the two flows into a single ATM stream per T interface instantiation in the downstream direction.
TC.
ATM Transmission Convergence sublayer functional block.

ATU-C.
ADSL Transceiver Unit at the Central Office end.

ATU-R.
ADSL transceiver Unit at the remote terminal end.

Channelisation.
For ATM systems the channelisation of different payloads is embedded within the ATM data stream using different Virtual Paths and/or Virtual Channels. Hence, the basic requirements for ATM are for at least one ADSL channel downstream and at least one ADSL upstream channel.

The ANSI T1.413 standard [2] gives the possibility to use both the Interleaved and Fast paths for services with requirements for either high error performance or low latency respectively. The real need for this dual nature for ATM services depends on the service/application profile, and is yet to be confirmed. Consequently, different configurations of the ADSL access could be considered.

More specifically, possibly three latency classes could be envisaged:
- Single latency, not necessarily the same for each direction of transmission.
- Dual latency downstream, single latency upstream.
- Dual latency both upstream and downstream.

For the transport of only ATM over ADSL, all modems shall use the AS0 channel downstream and the LS0 channel upstream for the single latency class. Channels AS1 and LS1 are reserved for dual latency. The dual latency option will be specified in a future issue of this document.

A hybrid implementation of one or more Bit Synchronous (Plesiochronous) channels together with the ATM channels is not precluded by the above. The bandwidth occupied by the Bit Synchronous channel must first be reserved before allocating the remaining bandwidth to the ATM channel.

Data Rates
Modems compliant to ANSI T1.413 standard can be programmed to provide bearer channel data rates which are multiples of 32 kbit/s. This facility may be exploited for the transport of ATM. Channel data rates can be set on a semi-permanent basis depending upon the loop characteristics for the particular user. Complete flexibility is therefore given to the Network Operator.

BER
ANSI T1.413 standard [2] specifies a BER of 10^-7 with a 6dB margin. The Network Operator may decide on a BER/ Latency/ Range combination that meets the required service quality for the network. The effect of ADSL performance impairments on ATM performance is for further study.
How to wrap it together.
ADSL is made up of several parts (shown by Figures 1 and 2):
- ADSL Network Termination (ANT) and Network Interface Card (NIC)
- Splitter or Splitterless Design
- DSLAM and Telco Loop
- ISP connection

Figure 1: ADSL Block Diagram (POTS Splitter)

Using POTS Splitter

Using digital switch
CPE: ADSL ANT and NIC

The Customer Premises Equipment (CPE) for ADSL consists of the ANT and/or NIC card. The ADSL Network Termination (ANT), shown in Figure 1, is located at your home or office, provides an IP connection. ANTs come in several types:

- Router ANT with 10/100baseT Interface
- Bridge ANT with 10/100baseT Interface
- ANT with ATMF Interface
- ANT with USB Interface
- Integrated ANT/NIC Card

In each case, the ANT/NIC provides the a router address to an ISP. Each Telco will specify the configurations that they will allow. The most desirable configuration for the Linux user is the ANT with a 10baseT Interface, since the cost/setup is the easiest. The other options require special drivers, which have, to date, not been made available for Linux. The bad news is that some providers allow only integrated ANT/NIC PCI cards that do not have Linux Drivers.

Warning! Make sure any third party ANT/NIC you may purchase are compatible with your Telco provider. There are two major line encodings for ADSL (CAP, DMT), and several options for IP encapsulation. Your Telco should provide you a list of allowable options. The ANT is connected to your house's inside wire (2 wire phone line). This inside wire is connected to the data side of a POTS splitter, or, in the case of the splitterless version, directly connected to the local loop. Figure 1 shows the POTS splitter wiring, and Figure 2 shows a splitterless type. In my case, I was provided with an Alcatel ANT, which supports a 10baseT (wired as a Crossover) RJ45 jack. I understand that a NIC card that fits directly in the PC will be available sometime in the future.

Figure 2: ADSL Block Diagram (Splitterless Design)
**Splitter or Splitterless Design**

Somehow, the digital and analog signals need to be separated for all of this to work. Thus, a filter needs to be placed in the signal path at some point. There are two methods for doing this: Using a POTS Splitter or using RJ11 phone jack filters.

First, in the POTS splitter method, device is located on the "side of the house" where the Telco line is connected. The splitter provides two functions. First, it is the "demarcation point" that separates the Telco wiring from the inside wiring. Second, it "splits" the DSL signal from Telco into a separated data channel and a voice channel. The voice channel is a normal analog phone line (2 wire), and the data channel is sent to the ANT. The splitter is a passive, non-powered device, which will allow the voice channel to operate even if the power fails at the home location. The Telco signal is sent to the splitter using an existing 2 wire line to the home. The Splitter is housed in the Network Interface Device (NID) on the outside of your house.

Second, in the splitterless design, shown in Figure 2, the outside local loop is connected directly to the inside wire at the Subscriber Network Interface (SNI), the same box that is used today at your house. At each extension jack where you wish to plug in an analog phone, you place a special jack that contains a filter that removes the digital signal. This is called an RJ11 filter (RJ11 is the official Telco term used for your 4/6 pin phone jack). The extension used for your ANT does NOT use a filter (otherwise it won't work). That's all there is to it! It should also be noted that some low speed ADSLs will not require RJ-11 filters.

The splitterless design is very desirable from the Telco point of view, as they won't have to roll any trucks to do the install work, and allows them to offer ADSL at a lower price. For most users, it doesn't really matter, in fact, the analog phones will still work without the RJ11 filter in place. The only thing is that you will hear a bit of a high pitched whine when you use the phone. However, this is not recommended, as later version may damage the phone or have some other nasty effect.
DSLAM

A Digital Subscriber Line Access Multiplexer (DSLAM) is a network device, usually at a telephone company central office, that receives signals from multiple customer Digital Subscriber Line (DSL) connections and puts the signals on a high-speed backbone line using multiplexing techniques. Depending on the product, DSLAM multiplexers connect DSL lines with some combination of asynchronous transfer mode (ATM), frame relay, or Internet Protocol networks. DSLAM enables a phone company to offer business or homes users the fastest phone line technology (DSL) with the fastest backbone network technology (ATM).

The DSLAM
Using DSLAM as distributor
Using xDSL for transport

An ISP connects to the DSLAM via a high-speed data connection, usually ATM over a T3 (45Mbps) or OC-3 (155Mbps). The important thing here is that an ISP must "subscribe" with your Telco to provide this connection.
This is the Frequently Asked Questions (FAQ) section for xDSL.

1. Q: Are there ADSL Standards.
   A: Sort of. The U.S. Bell Operating Companies have standardized on Discrete Multi-Tone (DMT) ANTs (ANSI T1.413) in their current rollout. Most others should follow their lead in the states. There are other types of ANTs, most notably Carrier-less Amplitude Phase Modulation (CAP), which, of course, are incompatible with each other.
   A biased comparison from an DMT-based vendor on this subject can be found at the Aware. Still, it provides the best detail on this issue I have seen so far.
   A rather expensive copy of the ANSI standard can be ordered at: American National Standards Institute ANSI Home Page
   Asymmetric Digital Subscriber Line (ADSL) Metallic Interface
   ANSI T1.413-1995
   Note: ANSI T1.413 Issue 2 was released September 26, 1997

2. Q: Can I use ATM to connect to ADSL ANT?
   A: Yes, you can! Some ADSL ANT (at least the Alcatel version) has a ATM Forum 25Mbps interface, which connects to a PCI NIC card. However, I have not yet heard of any Linux drivers for such cards.

3. Q: Why the heck does ADSL have all these bit rates (384/1.5/8M/20M/etc) options?
   A: The basic problem is the 100 year old design of the copper loop. It works great for analog phone, but it presents a real challenge for a digital signal. Remember that the distance of a loop is inversely proportional to the data rate that it can carry. Rate-Adaptive technologies are great for making a digital signal work in many situations, but it can't provide a consistent bandwidth for all applications, especially for very long (over 18 kilofeet) loops. The different bandwidth that you see advertised reflect various marketing wars of vendors equipment, and the Telco struggle to finalize on a "standard" set of data rates. I think that the 384k/1.5Mbps will become the standard for now. The high bit rates will only be available for special application and/or situations, since they can only be provided on a small percentage of the available loops.
   Also, check out the next question on the loop impairments that cause this to happen.

4. Q: What are all these loop impairments (bridge taps, loading coils) that could disqualify my line from using ADSL? (thanks to Bruce Ediger)
   Load coils: in-line inductances that improve voice-frequency transmission characteristics of a telephone circuit. Essentially, a "load" steals energy from high frequencies and gives it to lower frequencies. Typically only used in very long (>9,000 ft) phone lines.
   By "bridges" I assume you mean "bridged taps". In older neighborhoods, the phone wiring will have been used by more than one customer. Perhaps these customers lived at different (though near-by) addresses. The unconnected "spur" of wiring is a "bridged tab" on the currently connected circuit.
   Digital loop carriers: there's a bunch of systems for carrying more than one voice transmission on a single pair of wires. You can shift the frequencies up or down, or you can digitize the voice transmissions and divide the telephone circuit by time or code or something. The more general term is "pair gain".
   These things cause different problems for high-frequency communication.
   Loads will completely mess up things by filtering high frequencies and passing low frequencies. They probably also change the "delay envelope", allowing some frequencies to arrive before others. One byte's tones will interfere with the next byte's.
   Bridged taps act as shunt capacitances if they're long in relation to the signals wavelength, and they'll actually act as band pass filters if they're about 1/4 wavelength of the signal. That is, they'll pass particular frequencies freely. Particular tones of a DMT modem might get shunted back, rather than passed along to the receiving modem, reducing bandwidth for that telephone line.
   Pair gain, digital or analog, limit the bandwidth available to one transmission in order to multiplex several on one wire. High and low tones of a DMT transmission get filtered out by the apparatus.
bridged tap, etc on the transmission characteristics of a telephone line. It's pretty expensive, however.

5. Q: Do you have examples of ADSL ANTs?
A: Short Answer: Yes. Real Answer: The evolution of this technology is moving too rapidly for anyone to keep up to date in a HOWTO. A good source of ADSL ANTs is the ADSL Forum Home Page. Go to the Vendors pages to see what's happening. However, I will provide a list of some of the current technology as of June 1998.

- **Router ANT with 10/100baseT Interface**
  - Examples: Flowpoint 2000 DSL (CAP), Netspeed Speedrunner 202 (CAP), Speedrunner 204 (CAP), 3COM Viper-DSL (CAP), StarNet Ezlink 500/100 (DMT), Westell ATU-R-Flexcap (CAP), Aware x200

- **Bridge ANT with 10/100baseT Interface**
  - Examples: Alcatel A1000 (DMT), Westell ATU-R-Flexcap2 (CAP)

- **ANT with ATMIF Interface**
  - Examples: Alcatel A1000 (DMT), Netspeed Speedrunner 203 (CAP), Ariel Horizon II

- **Bridge ANT with V.35 Serial Interface (T1, Serial Router)**
  - Examples: Westell ATU-R

- **ANT with USB Interface**
  - Rumored to being pushed by Intel.

- **Integrated ANT/NIC Card**
  - Examples: Netspeed PCI Runner (CAP), Efficient Networks Speedstream 3020 (DMT)
  - These are NOT endorsements of the products listed, just provided for illustration.;-)
Glossary

2 wire Copper Loop

The two wire twisted pair from the Telco Central Office that terminates at a customer location.

ADSL

Asymmetric Digital Subscriber Line

ANT

ADSL Network Termination (a.k.a. the ADSL modem)

ATM

Asynchronous Transfer Mode - provides high-speed packet switching from 155 Mbps to (currently) 2 Gbps. Used to provide backbone switching for the Internet.

ATMF-25Mbps

ATM Forum Interface - 25 Mbps speed, provided by a PCI NIC card. One of the interfaces used between the ANT and PC.

Central Office

Usually refers to one of two meanings - 1) The Telco Building that houses Telephone equipment 2) The Telco Voice Switch that provides dial tone.

CPE

Customer Premises Equipment - The Telco term for customer equipment (i.e. the stuff you are responsible for fixing). Examples are CSU/DSU, modems, ANTs, and your phone.

DHCP

Dynamic Host Configuration Protocol - The IP protocol used to set up dynamically assigned IP addresses.

DS0

The basic digital circuit for Telcos - offered at 56 kbps or 64 kbps. Can support one analog voice channel.

DSLAM

Digital Subscriber Line Access Multiplexer - The Telco equipment that concentrates and multiplexes the DSL lines.

xDSL

Digital Subscriber Line - A term describing a family of DSL services, including ADSL, SDSL, VDSL, etc.

HDC

See Section 2

ISDN

Innovations Subscribers Don't Need; I Still Don't kNow or maybe Integrated Services Digital Network, a digital phone service that uses a single copper pair to run 2B (64k) + 1D (16k) channels that can be used for switched
voice or data.

**ISP**

Internet Service Provider

**NID**

Network Interface Device - The housing used to protect the ADSL splitter from the elements.

**NIC**

Network Interface Card - A PC card (PCI/ISA) that supports the required network interface. Usually an Ethernet 10baseT or an ATMF-25Mbps Card..

**POTS**

Plain Old Telephone Service - The service that provides a single analog voice line. (i.e. your phone line)

**Recursion**

See "Recursion"

**SNI**

Subscriber Network Interface - The Telco term for the phone wiring housing on the side of your house. It designates the point between the Telco side and the Inside Wire. This is also called the Demarcation Point.

**Splitter**

The passive device (low-bandpass filter) at the SNI that splits the ADSL signal into separate voice and data channels.

**Splitterless**

An ADSL installation that does not require the Splitter. For higher speeds, a RJ11 filter is placed on every extension phone jack where an analog phone is used, thus providing the filtering at the jack, rather than at the NID. For lower speeds, no filter is required.

**SOHO**

Small Office HOme

**T1**

a.k.a DS1 - A digital dedicated line at 1.544 Mbps, used for both Voice (24 DS0s) or Data.

**T3**

a.k.a DS3 - A digital dedicated line at 44.736 Mbps, provides for both Voice (672 DS0s or 28 DS1s) or Data