Application Report
SPAA005

DSP Solutions for Voiceband and ADSL Modems

Padmanand Warrier *World Wide Broadband Access Group*

This application report is a tutorial on voiceband and Asymmetric Digital Subscriber Line (ADSL) modems. It describes how ADSL technology builds upon the techniques developed for voiceband modems, and illustrates why Digital Signal Processing is important to transition from voiceband to ADSL modem technology. This application report is organized as follows:

- History of voiceband modems
- Key functions of voiceband modems
- Advanced features of V.34 modems
- Evolution of ADSL and Discrete Multi-Tone (DMT) line-coding
- Advantages of DMT for ADSL
- The need for Digital Signal Processing in modems
- ADSL implementation with the Texas Instruments (TI™) fifth generation chipset (TNETD2000™) using TMS320C6x core DSP technology
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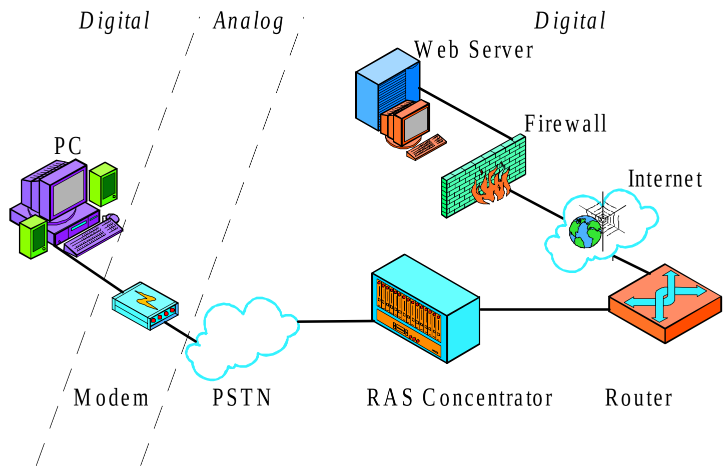
From Voiceband Modems to ADSL Modems

Most people are familiar today with the paradigm for connecting to the Internet via a dial-up connection as shown in Figure 1. In a dial-up connection, the user's PC is connected to a voiceband modem that dials up across the Public Switched Telephone Network (PSTN) to a Remote Access Service (RAS) Concentrator. The RAS Concentrator consists of several modems. The first available modem in the RAS Concentrator that is not in use serves to terminate the incoming phone call. The RAS concentrator is responsible for directing the upper layer traffic (such as Web browsing requests) to a router that in turn routes traffic to the appropriate destination.

Figure 1. Typical Dial-Up Connection



The device that makes this connection possible across a wide area is the voiceband modem. The modem at either end of the PSTN allows the user's PC and the Web Server to communicate to each other. Voiceband modems are used to carry digital data that originates in the user's PC (or Web Server) across the commonly available infrastructure of a telephone network. However, the telephone network was originally designed to carry voice signals in analog form. The modem transforms digital data to the necessary analog signals. At the other end, the receiving modem reverses the process, and converts the analog signals back into digital data that can be manipulated by the RAS Concentrator. Therefore, voiceband connections go through multiple analog to digital and digital to analog conversions[1]. Also, modem technology is subject to the bandwidth limitations imposed by the analog voice network between the subscriber and the Central Office. For these reasons, dial-up access is slow and ill suited to the megabit bandwidth requirements of rich, dynamic multimedia content.

ADSL technology can be used to overcome the issues faced by traditional voiceband modem technology. The copper loop used to carry voice traffic between a subscriber's premise and the Central Office at a Telco is inherently capable of sufficient bandwidth to carry megabits of data, depending on the distance of the loop. Voice traffic, however, utilizes only 4 KHz of bandwidth since 64 Kbps are sufficient to accurately reproduce human voice in telephone conversations. Hence, the copper loop has unused bandwidth that could be used to support high data rates depending on loop length.

Using ADSL, it is possible to achieve data rates that are hundred times the rates of today's 56 Kbps modems. ADSL technology utilizes the infrastructure already in place in the PSTN; the copper loops between the premises and a Central Office. It does not require a replacement of network equipment such as routers, switches, firewalls and Web servers used in today's paradigm for access. ADSL builds upon the techniques developed for modem technology for modulation, error detection and error correction. All of these factors make ADSL an attractive choice for high-speed Internet access.

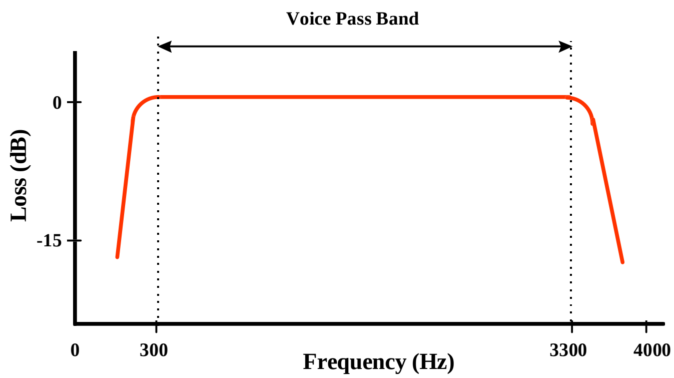
[1] Today's 56K voiceband modems are able to achieve higher speeds by eliminating one of these conversion steps. However, the maximum achievable speed is still limited to 56 Kbps. In the US, due to regulations, the actual maximum achievable speed is 53 Kbps.

Brief History of Voiceband Modems

The word *modem* stands for MOdulator/DEModulator. What is being *modulated* is an analog carrier signal residing in the voiceband spectrum. The modulated analog carrier signal is sent over the PSTN and represents the analog equivalent of the digital input that originates from a computer (the user's PC in the example of the Internet dial-up connection).

The voiceband is typically defined as the frequency band from 0 to 4 KHz, as shown in Figure 2. In actuality, the voice pass band is less than the full 4 KHz, typically being around 3 KHz. The remainder of the 4 KHz spectrum is used for out of band signaling and guard band considerations. The analog carrier is typically located near the center of the voiceband.

Figure 2. Voice Pass Band



The reasoning behind selecting a 4 KHz voiceband was primarily driven by economics. Even though the human audible range of frequencies is roughly 0 to 20 KHz, every day speech can be adequately transmitted using only a fifth of that bandwidth, i.e., 4 KHz. By designing the Public Switched Telephone Network (PSTN) to support 0-4 KHz, the cost and complexity was greatly reduced. However, this factor also limits the data rate. In 1948 Claude Shannon's research led to the famous Shannon's Law $C = Bw * Log_2(1+S/N)$, where C is the channel capacity (data rate), Bw is the available channel bandwidth, and S/N is the signal-to-noise ratio. According to Shannon's Law, the maximum data rate is ~ 35 Kbps.

The original modems in the 1950s used proprietary modulation protocols. Examples were the Bell 103 (300 bps U.S. standard) and the Bell 212A (1200 bps U.S. standard), based on Radio Frequency (RF) technology. In the 1960s, the International Telecommunications Union (ITU)[2] began to ratify standards based modem protocols beginning with V.21 (300 bps standard outside the U.S.). The preferred modulation for "V.dot" series of recommendations is based on a technique known as Quadrature Amplitude Modulation (QAM). For example, V.22bis (2400 bps standard) uses QAM.

[2] The ITU was formerly known as the CCITT.

From Shannon's law, it follows that the higher the S/N ratio, the greater the achievable data rate. This became the focus of research in modem technology. In the 1980s, the next major improvements came with the addition of echo cancellation and Trellis Coding. Echo cancellation allowed modem pairs to use the entire available bandwidth for both upstream and downstream. Trellis Coding made it possible to implement error correction in modems that resulted in the ability to extract information more reliably for a given S/N ratio. This enabled data rates to increase to 14.4 Kbps with V.32bis.

In early 1990, work began on raising the data rate beyond 14.4 Kbps to an initial goal of 19.2 Kbps. However, based on the adoption of several key technologies (Line Probing, Precoding, Multi-dimensional Trellis Coding, Shell Mapping, and Warping), a top speed of 28.8 Kbps was achieved in 1994. By 1996, V.34 standard based modems could achieve 33.6 Kbps, which is close to the theoretical limit. Recently, 56 Kbps modems have become available. These modems do not violate Shannon's law because communications is not between two modems, rather it occurs between a 56K modem and a digital system at the other end. This results in a reduced quantization noise because only one analog to digital conversion must take place.

Key Functions of Voiceband Modems

A modem implements protocols that cover the following three major functions:

- Modulation
- Error Control
- Data Compression

Protocols are necessary to implement these functions in a standard manner, so that a modem from one manufacturer can successfully interoperate with another. This section describes some of the modulation, error control and data compression protocols.

Modulation Protocols

The process of encoding a digital signal into an analog signal that represents and conveys the information over an analog channel is called line-coding. In the simplest case, a single signal element represents one bit. With complex techniques, a single signal element represents multiple bits, thereby increasing the effective data rate. Therefore, to achieve higher data rates, an important consideration is the number of bits/Hz that can be supported with an encoding scheme. Since the telephone network is both bandwidth limited (4 KHz) and power limited, line-coding schemes should not involve increasing either bandwidth or power. Consequently, line-coding techniques have focused on manipulating the analog carrier signal. An analog carrier signal has three attributes – amplitude, frequency, and phase. Modulation techniques involve manipulating one (or more) of these attributes. The common modulation techniques are:

- Amplitude Modulation (AM)

  Modulating only the carrier signal amplitude. Different amplitude levels can be employed to increase the number of bits/Hz.

- Frequency Modulation (FM)

  Modulating only the carrier signal frequency

- Phase Modulation (PM)

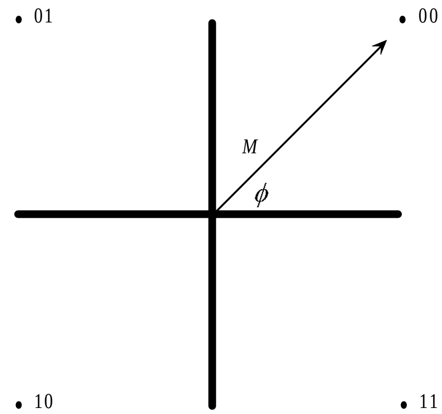
  Modulating only the carrier signal phase

- Quadrature Amplitude Modulation (QAM)

  Modulating both the carrier signal's phase and amplitude

Using appropriate modulation techniques, it is possible to increase the number of bits/Hz as illustrated in Figure 3. The digital values that can be encoded and the corresponding phase and amplitude are represented using a constellation diagram. In a constellation diagram, the length of the vector from the origin to the constellation point represents the amplitude of the carrier wave (M), and corresponding angle of the vector (ϕ) represents the phase.

Figure 3. 4 QAM Constellation Diagram



In Figure 3, each dot is known as a symbol. Table 1 shows the bit values for 4 QAM.

Table 1. Bit Values for 4 QAM

Bit Values	**Amplitude (M)**	**Phase (ϕ)**
00	1	45^0
01	1	135^0
11	1	225^0
10	1	315^0

By increasing the constellation size, and hence the bit density per symbol, higher data rates can be achieved. This technique is used in V.34 modems, as discussed later.

Modems use several modulation protocols for various data rates, as shown in Table 2.

Table 2. Evolution of Voiceband Modem Standards to ADSL

Name	Scope	Data Rate (Down/Upstream)	Duplex	Key Improvements
Bell 103	U.S.	300/300 bps	Half	
Bell 212A	U.S.	1200/1200 bps	Full	
ITU (CCITT) V.22	Outside U.S.	1200/1200 bps	Half	
ITU V.22 bis	International	2400/2400 bps	Full	
ITU V.32	International	9.6/9.6 Kbps	Full	Trellis Coding
ITU V.32bis	International	14.4/14.4 Kbps	Full	2-D Trellis, 2-D Shell Mapping
ITU V.34	International	33.6/33.6 Kbps	Full	4-D Trellis, 16-D Shell Mapping
ITU V.90[3]	International	56/33.6 Kbps	Full	Reduction of A/D conversions
ITU G.dmt[4]	International	6000/640 Kbps[5]	Full	DMT technology

Error Control Protocols

A telephone line is subject to several sources of errors called line impairments. The following are some common sources of errors:

- *Impulse noise.* This is caused by electrical interference. Examples are the effect of lightning storms, switching home appliances on/off, and so on.
- *White noise*. This is also called Gaussian noise. White noise is always present on the line and is caused by the movement of electrons in the line. In general, many noise sources add together, and the law of large numbers makes the cumulative interference look Gaussian. Shannon's Law is based on the assumption of white noise on the line.
- *Attenuation.* When a signal travels along the line, it looses some of its strength with distance. If the signal becomes too weak, it becomes more susceptible to errors.
- *Crosstalk.* Crosstalk occurs when signals from two lines interfere with each other.
- *Intermodulation Noise (IMD).* This is similar to crosstalk, except that the two signals combine to produce a frequency outside an allowable range of frequencies.

To improve data throughput, voiceband modems use techniques for both error detection and error correction. Error detection techniques are parity check and Cyclic Redundancy Check (CRC). Error correcting techniques are of two kinds (a) Forward Error Correction (FEC) and (b) Automatic Repeat Request (ARQ). FEC techniques are preferred in voiceband modems because errors can be corrected without requiring retransmission of the data.

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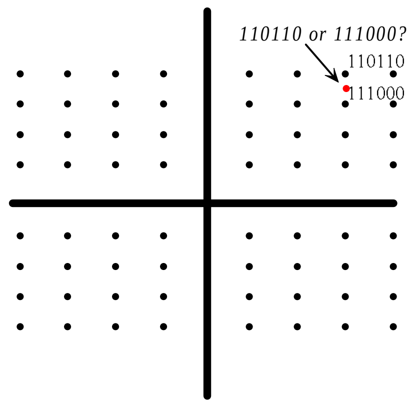
[3] V.90 is the 56 Kbps international standard. It combines the techniques based on X2 (a standard developed by US Robotics, now part of 3Com Corporation) and K56Flex (a standard developed by Lucent Technologies and Rockwell Semiconductor).
[4] G.dmt is the proposed ADSL International Standard. It is based on the ANSI standard T1.413, Issue 2.
[5] Actual data rates depend upon line conditions and loop lengths.

Figure 3Figure 3 illustrates a 4 QAM constellation. In this example, every symbol transmitted represents 2 bits of data (i.e., 2 bits/Hz). One way to increase data rate is to increase the constellation size and thus the bit density per symbol. As the constellation size increases, the granularity of the phase and amplitude differences between different constellation points diminishes. Therefore, it becomes increasingly difficult to accurately decipher the constellation points, especially in the presence of channel noise or interference, as shown in Figure 4.

Figure 4. 64 QAM Constellation



From Figure 4, it follows that increasing the distance between symbols (known as the Euclidean distance) increases detection probability. This is the goal of Trellis Coding[6], developed by Dr. G. Ungerboeck. Trellis coding introduces redundancy to a bit stream with a convolutional code. The output of the convolutional code is then mapped to an expanded constellation with a greater number of possible points. Thus, the technique is bandwidth-efficient because the symbol rate (and hence bandwidth) is not increased. The state evolution of the convolutional code couples with the particular constellation mapping of the Trellis Code design to restrict the possibilities for the sequence of symbols at the receiver. In this manner, the receiver is able to eliminate certain symbols from consideration during detection at the receiver. However, the transmitted *sequence* of symbols must now be detected. Normally this would be an overwhelmingly complex operation, but the Viterbi algorithm provides an efficient implementation of the sequence detection algorithm.

Modems use one or more of the following standard error control protocols:

- ❑ Microcom Networking Protocol (MNP) classes 2-4

  De facto standards developed by Microcom (now part of Compaq Computer Corporation). MNP framing defines several formats to be used during modem operation such as Link Request, Link Disconnect, Link Transfer, Link Acknowledgment, Link Attention, Link Attention Acknowledgment, Link Management, and Link Management Acknowledgment. Each of the frames perform a specific function, such as requesting link establishment, transmitting data across the link, acknowledging successful reception of data, requesting a change in the transmission data rate, requesting session termination, and so on.

[6] Prior to the discovery of Trellis Coding, Hamming Coding was popular. Today, most modems use Trellis Coding because it improves the performance of QAM based modems without requiring increased bandwidth.

- MNP class 10

  A de facto standard developed by Microcom for adverse line conditions.

- V.42

  The International standard for error correction, V.42 supports Link Access Procedure for Modems (LAPM) as the primary error control protocol, with fallback to MNP class 4.

Alternatively (or additionally), some modems may support a proprietary error control protocol.

Data Compression Protocols

Another method for improving data throughput is to compress the input data before it is sent. The receiving modem reverses the process and decompresses the data before passing it on. The compression and de-compression is done “on the fly”, i.e., data compression is performed as it is received from the sending computer and decompressed just before transmitting to the receiving computer (with some buffering to distinguish “redundancy” in the data stream). Common techniques used in data compression are Huffman coding, Run-Length coding and Lempel-Ziv coding.

Modems use one or more of the following standard data compression protocols:

- MNP class 5

  A de facto standard developed by Microcom. MNP Class 5 can achieve a maximum compression ratio of 2:1.

- V.42 bis

  The International standard for data compression protocols, V.42bis can achieve a maximum compression ratio of 4:1.

Alternatively (or additionally), some modems may support a proprietary data compression protocol to support higher compression ratios. Compression ratios, however, are highly dependent upon the type of input data. If there is no “redundancy” in the data stream (i.e., data has already been compressed), further compression will not yield any benefits.

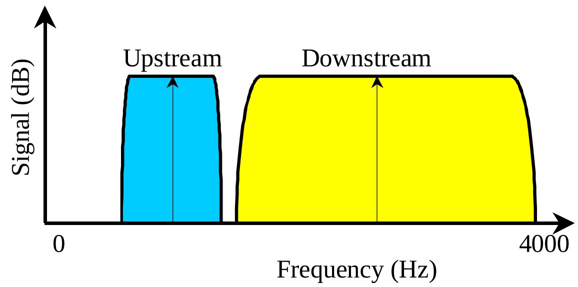
Advanced Features of V.34 Modems

The evolution to modern voiceband technology closely matched corresponding advances in silicon technology. The availability of specialized processors called Digital Signal Processors (DSPs) enabled modem engineers to economically implement algorithms to improve system performance. These advanced techniques allow V.34 to achieve data rates close to the theoretical limit. Many of these techniques are also applicable for ADSL. Some of the advanced features of V.34 modems are discussed below.

Echo cancellation

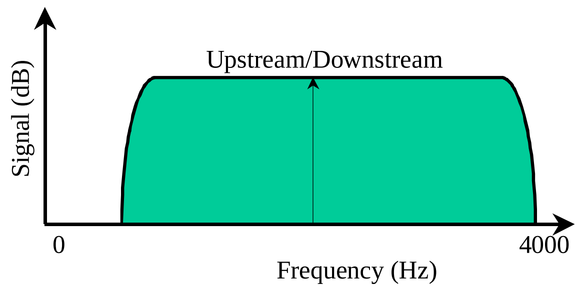
To achieve full duplex operation, voiceband modems split the pass band into two separate channels – one each for upstream and downstream. This is called Frequency Division Multiplexing (FDM). In FDM modems, there are two separate analog carrier signals, as shown in Figure 5.

Figure 5. FDM Technique for Full Duplex Operation



Modern voiceband modems employ echo cancellation techniques to achieve full duplex operation with a further increase in the data transmission rates. Echo cancellation based modems use a single channel for both transmission and reception, as shown in Figure 6. Echo cancellation involves removing the effects of the modem's transmission on the modem's reception. This allows a modem pair to use the entire available bandwidth for both directions.

Figure 6. Echo Cancellation Technique for Full Duplex Operation



Multi-Dimensional Trellis Coding

From Figure 3 and Figure 4, it can be seen that one way to increase data rate is to increase the constellation size, and hence, the bit density per symbol. However, as the constellation size increases, it becomes increasingly difficult to accurately decipher the constellation points, especially in the presence of channel noise or interference. Trellis Coding is used to solve this problem; it applies signal processing to extract information more reliably for a given S/N ratio.

Initially, 2D Trellis Coding was employed in V.32 bis modems. 2D Trellis Coding introduces controlled redundancy to a bit-stream using a convolutional code. With 2D Trellis Coding, introducing redundancy means adding one more bit/symbol. For example, for a 4 QAM constellation, the number of bits/symbol is 2; with redundancy, the number bits/symbol is 3. So, the constellation size now increases from 2^2 to 2^3, i.e., an 8 QAM constellation. In general, if the number of bits/symbol is N, 2D Trellis coding results in N+1 bits/symbol. Hence, the constellation size increases from 2^N to 2^{N+1}. In effect, 2D Trellis Coding doubles the constellation size. Doubling the constellation size is not a problem for smaller constellations. However, as constellation sizes get bigger, the problem of detecting constellation points due to increased density (shorter Euclidean distances between constellation points) is reintroduced. Therefore, it is desirable to introduce redundancy without doubling the constellation size. Multi-dimensional Trellis Coding provides the solution.

In 2D Trellis Coding, redundancy is introduced in a 2D space. However, if the redundant bits were to be distributed in multi-dimensions (M>2), the number of bits/symbol mapped to a 2D space is now less. Therefore, it is possible to introduce redundancy without doubling the constellation size. For example, with 4 QAM, it is possible to achieve redundancy with 5 bits for every two symbols, i.e., 2½ bits/symbol. So, the constellation size increases to $2^{2½}$, rather than 2^3. A practical limit on the number of dimensions for multi-dimensional TCM is 4 (i.e., M=4). This limit comes from striking a balance between increased complexity versus better performance.

Shell Mapping (Constellation Shaping)

Referring to the constellation diagrams shown previously, it can be seen that as constellations get larger, placing symbols in a rectangular 2D plane requires more energy at the corners rather than at the edges. This is not very power-efficient. Instead, if the outer boundary of the constellation were roughly circular (rather than rectangular), less energy would be required to place symbols at the edges. This results in power savings of the average energy required per symbol.

If we were to take a uniform, spherical distribution of symbols and project onto a 2D plane, we would end up with a non-uniform circular distribution of symbols, with the density of symbols highest at the center, and becoming less dense at the edges. This re-introduces the problem that the distances between symbols are small when they are located close to the center. To overcome this problem, additional algorithms are required to maintain a balance between power reduction and adequate symbol distance. These algorithms are often implemented in DSPs because they involve complex mathematical calculations.

Fast Training

In Figure 2, the voice pass band was depicted as linear. In actuality, the pass band response is non-linear because the higher frequencies get attenuated more than lower frequencies in the local loop. The equipment also introduces various non-linearities. To compensate for the non-linear behavior of the local loop, each modem's receiver must synchronize to the other's transmit signal during the initialization sequence. This is known as training. Training involves sending a known signal down the wire that the receiver can use to compare the received signal against the original non-distorted known signal. The receiver can then calculate the distortion effects of the channel and compensate accordingly.

Due to large error bursts on the line, it is possible for the modems to perform training after the connection has been up for a while. This is known as re-training. Training/re-training is a complex process that takes time. During the time period when modems are training/re-training, no data can be sent. Hence, it is desirable to reduce this period. Unfortunately, estimated training times became longer for higher data rates such as the 33.6 Kbps supported by V.34 modems. The necessary reduction of training times was achieved by (a) changing the training sequence to be based on receiver timing rather than transmitter timing and (b) better error recovery procedures to avoid complete re-initialization.

Fast training procedures are even more critical in ADSL since the data rates are much higher (in the range of Mbps rather than Kbps).

V.8 Handshake Negotiation

This is a handshake procedure at start-up that allows a V.34 modem to negotiate features and mode parameters with a remote modem. Using V.8 handshake, a V.34 modem can identify whether the remote modem is also V.34 capable, or whether it needs to fall back into any of the backward compatible modes for lower data rates. V.8 is used to determine the modulation, error control and data compression protocols to be used for the connection.

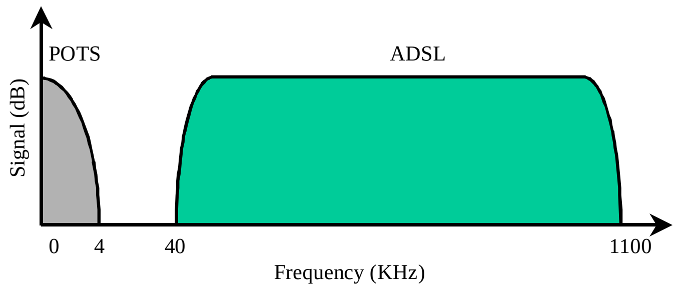
V.8 has a counterpart in ADSL, a handshake protocol called G.hs. G.hs has still not been defined by the ITU, however, like V.8 it is expected to be a mechanism to allow two ADSL modems to negotiate features and mode parameters.

Handshake procedures such as V.8 (and G.hs in the future) are also often implemented in programmable DSPs. DSPs offer the advantage of rapid prototyping and verification. Additionally, software modifications can be developed to address interoperability with multi-vendor implementations.

Evolution of ADSL and DMT Line-Coding

ADSL was developed in 1989 out of research work done by Bellcore. The fundamental idea is to reuse the copper loops that exist between premises and central offices to carry analog voice traffic. The reuse involves utilizing frequencies beyond the voice pass band. Copper is capable of carrying higher frequency signals (4 KHz to approximately 2 MHz), however, these signals attenuate more rapidly with distance than do signals at voiceband frequencies. Even so, for loop lengths of up to approximately 18,000 feet it is possible to get enough signal strength to carry information bits at the higher frequencies. ADSL uses a guard band to separate the voiceband POTS from ADSL frequencies, as shown in Figure 7. This allows POTS and ADSL to co-exist on the same wire.

Figure 7. Separation of POTS and ADSL Frequencies



Another characteristic of ADSL is the asymmetry – the ratio of upstream bandwidth to downstream bandwidth is approximately 1:10. Bellcore originally envisioned ADSL for Video on Demand (VOD) applications where downstream speeds of 1.5 Mbps would suffice for MPEG movies. Since then, the market for VOD has vanished due to lack of demand. Today, ADSL has found new life with Internet access. The inherent asymmetry makes it well suited for Web browsing applications, where the downstream content information is likely to require greater bandwidth than the upstream requests. The upstream and downstream rates have improved to 640 Kbps upstream and approximately 6 Mbps downstream (depending on loop length and loop conditions). As per the ADSL Forum, the following are the range of downstream speeds depending on the distance:

- Up to 18,000 feet 1.544 Mbps
- Up to 16,000 feet 2.048 Mbps
- Up to 12,000 feet 6.312 Mbps

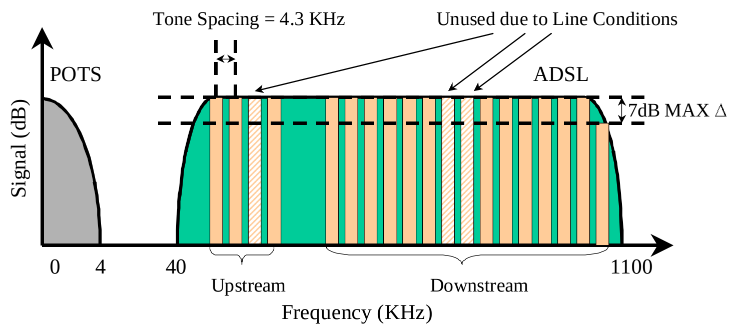
Upstream speeds range from 16 Kbps to 640 Kbps. Another improvement over the original idea was rate adaptation. Rate adaptation allows two DSL modems to adjust the upstream and downstream rates based on the loop conditions.

With the need to utilize the local loop for both analog voice and digital data, newer and more efficient line codes were required. DMT line-coding technique was developed around 1987 as a result of the research performed by Professor John M. Cioffi at Stanford University. John Cioffi then founded Amati Corporation in 1992, and developed the first ADSL modem product, called Prelude™, using off the shelf components. Prelude was tested by Telcos throughout the world to evaluate the basic technology. Amati incorporated lessons learned from the Prelude testing into its Overture™ series of ADSL transceivers and modems.

In the early 1990s, Bellcore evaluated various line code options for ADSL, and agreed upon DMT based on the successful demonstration of the technology by Amati. Today, standards bodies in the USA (ANSI) and Europe (ETSI) have adopted DMT as the line code of choice. The ITU Study Group 15 has also endorsed DMT.

DMT-based ADSL modems can be thought of as many (usually 256) “mini-modems”, 4 KHz each, that run simultaneously. DMT uses many carriers that create sub-channels, each sub-channel carries a fraction of the total information. The sub-channels are independently modulated with a carrier frequency corresponding to the center frequency of the sub-channel and processed in parallel. Each sub-channel is modulated using QAM and can carry between 0 to a maximum of 15 bits/symbol/Hz. The number of actual bits carried per sub-channel depends upon the line characteristics. Certain sub-channels can be left unused due to external interference. For example, an AM radio station causing radio frequency interference in a particular sub-channel can cause that sub-channel to be unused. DMT is illustrated in Figure 8.

Figure 8. Discrete Multi-Tone (DMT)



The theoretical maximum upstream bandwidth is

25 channels × 15 bits/symbol/Hz/channel × 4 KHz = 1.5 Mbps

The theoretical maximum downstream bandwidth is

249 channels × 15 bits/symbol/Hz/channel × 4 KHz = 14.9 Mbps

Advantages of DMT for ADSL

DMT offers several advantages as the line-coding technique for ADSL. Among them are:

- Evolution from V.34 modem technology
- Performance
- Robustness to line impairments
- Rate Adaptation

Evolution from V.34 Modem Technology

As discussed earlier, V.34 modems utilize several advanced techniques to maximize the data rates on noisy lines. DMT based ADSL modems represent a natural evolution from V.34 modem technology. DMT modems employ the following:

- QAM (each of sub-channels in DMT implements QAM)
- Echo cancellation[7]
- Multi-dimensional Trellis Coding
- Constellation Mapping

Additionally, DMT modems are expected to also implement the following technologies similar to V.34 modems

- V.8 negotiation handshake

  The ADSL equivalent is called "G.hs" (for handshake).

- Fast training

  Some modems already implement fast-training procedures today; however, additional procedures may be required to support operation of modems without a POTS splitter[8].

Performance

DMT increases modem performance because independent sub-channels can be manipulated individually with consideration to the line conditions. DMT measures the S/N ratio separately for each sub-channel, and accordingly assigns the number of bits carried by the sub-channel. Typically, the lower frequencies can carry more bits because they are attenuated to a lesser extent than higher frequencies. This procedure increases the overall throughput even under adverse conditions.

Robustness to Line Impairments

During initialization, DMT monitors the line conditions and computes the bit carrying capacity of each sub-channel based on its S/N ratio. If a sub-channel is experiencing external interference such as Radio Frequency Interference (RFI) and crosstalk, it may not be used at all in favor of other sub-channels.

Rate Adaptation

DMT can dynamically adapt the data rate to line conditions. Each sub-channel carries a certain number of bits depending on its S/N ratio. By adjusting the number of bits per channel, the DMT can automatically adjust the data rate.

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[7] Echo cancellation in a DMT modem is even more important than in a voiceband modem. Without echo cancellation, the upstream and downstream bandwidths are treated separately, with the lower frequencies used for upstream bandwidth. However, lower frequencies get attenuated less than higher frequencies, hence the bit carrying capacity of sub-channels that span these frequencies is greater. Therefore, echo canceling modems are able to provide greater downstream bandwidth by utilizing the lower frequencies – something that is not possible in FDM modems.

[8] Standard ADSL requires a POTS splitter device (either separate or built into the modem) to separate the POTS and ADSL frequencies. However, this requires the intervention of a telephone company technician to come and install the splitter at the customer premises and new wiring. The advantage of "splitterless" operation is that no new wiring is necessary, and a customer can simply utilize the DSL modem by plugging into any available Telco jack (similar to today's paradigm for voiceband modems). On the other hand, splitterless operation does imply interference between the POTS and ADSL frequencies. A new ITU standard called "G.lite" is expected to address splitterless operation. The "lite" comes from the fact that the speeds supported by this standard will be less than full rate ADSL. The lower speeds will provide approximately 1.5 Mbps downstream, which is still an order of magnitude better than the data rate of voiceband modems (56 Kbps).

The Need for Digital Signal Processing in Modems

DSPs and Voiceband Modems

The advanced features of V.34 modems make it possible to achieve data rates close to the theoretical limit defined by Shannon's law. It is Digital Signal Processing that makes it possible to implement these features in a cost-effective manner. Digital Signal Processing was a key factor in enabling the evolution of modems from previously slow data rates to V.34 speeds today. The evolution of Digital Signal Processing closely mirrors the advances in modem technology.

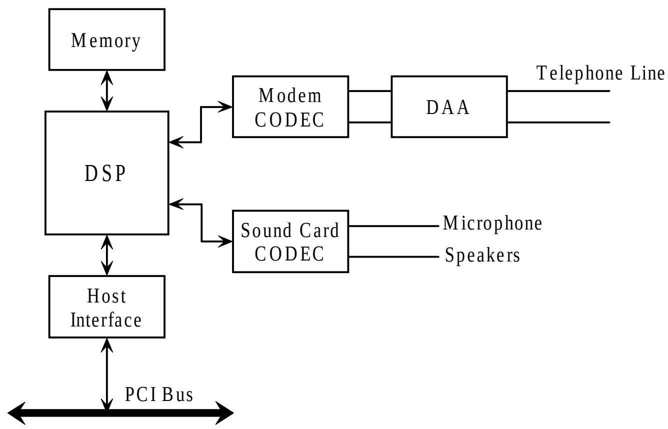
As discussed earlier, modems implement a variety of protocols to modulate the analog carrier signal in complex ways. In addition to protocol implementations, modems have to process the signal in the modem hardware and software. The actual transmission and reception of the signals are done in the hardware using a CODE/DECODE (CODEC) and a Data Access Arrangement (DAA) to provide isolation of the modem from the PSTN. However, the processing of the signal prior to transmission (and on the receiving modem, after reception) is often done in software. Therefore, modem software has to implement two different types of algorithms – (a) algorithms that implement the protocols in accordance to standards and (b) algorithms that process the signal according to certain mathematical functions (such as Fast Fourier Transforms). As modems became more complex, it was necessary to implement the mathematical functions in DSPs.

Digital Signal Processing consists of digitizing an analog signal and processing it in the digital domain, then changing it back into the analog domain. While this may seem inefficient at first, there are many advantages to processing the signal in the digital domain, since it simplifies mathematical complexity. Until the 1970s, Digital Signal Processing was considered mostly as a theoretical field because the available technology to implement the algorithms was too costly, slow, and bulky. With the PC revolution, however, advances in chip technology made it possible for cost-effective solutions in very small packages. In the early 1980s, Texas Instruments, the world leader in Digital Signal Processing, introduced a DSP capable of 5 million instructions per second (MIPs). The TI DSP had the ability to complete several functions (e.g., multiply-accumulate) in one cycle, thus making it a natural choice to implement the mathematical functions.

A trend began in the late 1980s and the early 1990s to start using DSPs to implement modem software. The programmability of DSPs allows the modem software developer to write flexible software that enables the modem to adapt to a variety of line conditions. Modern DSPs such as the TI C5x and C6x core DSP technology processors are capable of instruction speeds in the range of several hundred million instructions per second (MIPs) that allow modems to keep up the data rate even under noisy line conditions. Programmability also allowed modem manufacturers to ship a common hardware platform and provide upgrades in features and performance through software alone.

Simultaneously, there was a trend to take advantage of the DSP to provide multimedia capability for PCs – specifically for sound and computer telephony applications. This allows a PC to support features such as an answering machine, FAX machine, speakerphone and advanced capability such as surround sound and voice recognition. This combination of functions provides a really cost-effective solution, as shown in the block diagram in Figure 9. The DSP typically executes the modem software, sound card emulation software, and other features such as FAX and speakerphone.

Figure 9. Typical Block Diagram Combing Modem, Sound Card and Telephony Functions



The combination of these trends for DSPs – implementation of high-speed modem technology, programmability, and implementation of combined voice and modem functions enabled the DSP marketplace to grow. Today, Texas Instruments has shipped more than 40 million DSPs for modem solutions.

DSPs and DMT Modems

DSPs play an even more significant role in DMT modems since DMT works by dividing the available bandwidth into multiple sub-channels, and treating each sub-channel as a “mini-modem”. This requires an ADSL Transceiver to *simultaneously* modulate all sub-channels, a function that is performed efficiently and cost effectively on a DSP.

Additionally, DSPs allow modem designers to implement advanced error correcting techniques such as Reed-Solomon Coding. Reed-Solomon Coding is named after its inventors Irving S. Reed and Gustave Solomon who presented the basic idea in a paper in 1960. Reed-Solomon Coding is based on groups of bits, rather than individual bits. This enables Reed-Solomon Codes to deal with error bursts of several bits. For example, an electrical disturbance can clobber several bits at once – Reed-Solomon Coding enables error recovery under these conditions. Reed-Solomon Coding uses polynomials to create redundancy in a message (a group of bits manipulated together), such that the original message can be recovered even if multiple bits are in error. The use of polynomials makes implementation of the Reed-Solomon code efficient.

Programmable DSPs – a Future Proof Platform for Voiceband and DSL Modem Technology

The use of programmable DSPs for voiceband and ADSL modems is likely to remain the best choice as cost effective, future proof platforms for a number of reasons:

- ❑ Programmable DSPs offer several benefits
  - Rapid prototyping and verification, resulting in faster time to market
  - Tracking rapid evolution of new standards such as G.hs and G.lite

- Software upgrades that include code updates, feature set additions, interoperability code patches, and so on.

- DSPs are a natural fit for voice and video processing, hence the extra available MIPs can be utilized to develop value added applications that further enhance the Internet experience with multimedia features (e.g., streaming applications).

- Multitasking operating systems requires context switching between multiple tasks executing on the main processor – this keeps the processor busy for task management alone. As the PC becomes a device capable of multiple functions that must be constantly kept active (e.g., speakerphone, answering machine, and videophone), it is important to let the host CPU be a manager of tasks, rather than getting burdened with a single task.

- Several networking protocols require responses to “keep alive” messages, otherwise, the remote side may assume that the connection is inactive. These responses have to be made within specific timer values. A dedicated processor that is responsible for the connection is more efficient at responding to messages without causing unnecessary timer pops rather than a centralized host processor that has to perform many other functions.

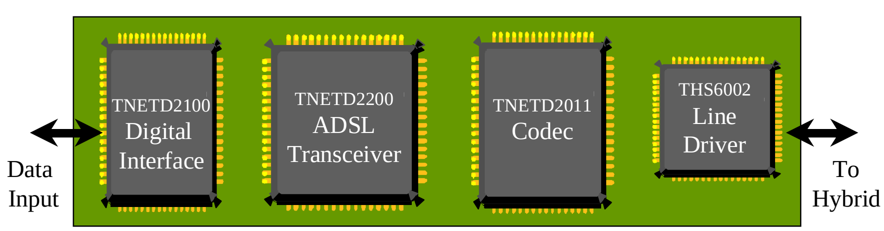
- Functions such as modem processing can be CPU intensive, and this trend is likely to continue with the even higher speeds and processing demanded by ADSL. Burdening the CPU with complex mathematical computation does not free it up for normal processing. Furthermore, CPU MIPs tend to be expensive. Therefore, these functions are more cost effectively performed on a dedicated processor.

- The price/performance of DSPs shall continue to provide great value as a solution. In addition to the PC market, total volumes in DSPs are also driven by other high volume areas such as the high-density disk drive industry, automotive industry, the cellular phone industry, video game industry, and so on.

Texas Instruments TNETD2000 ADSL Chipset

Texas Instruments and Amati Corporation (now a part of Texas Instruments) have leveraged their expertise in voiceband modems, ADSL and programmable DSP technology to develop the TNETD2000 ADSL chipset shown in Figure 10. This is a fifth generation chipset that incorporates many of the improvements from the field trials conducted by Amati Corporation.

Figure 10. Texas Instruments TNETD2000 ADSL Chipset



The following is a brief description of each of the devices in the chipset:

- The TNETD2100 Digital Interface has programmable serial interfaces to provide simple and clean standards-compliant data interfaces to the chipset. To support ATM implementations, each serial interface features a Byte Mark pin.
- The TNETD2200 ADSL Transceiver is based on the high-performance C6x core DSP technology. This fully programmable ADSL transceiver provides the necessary computationally intensive digital signal processing required for ADSL operation. It implements the following
  - Echo cancellation
  - Reed-Solomon Forward Error Correction
  - Multi-dimensional (4D) Trellis Coding
  - Rate adaptation
  - Full ADSL Data Rate of 800 Kbps upstream and 8 Mbps downstream at 12,000 feet
- The TNETD2011 Codec is a high precision mixed-signal device that provides the analog-to-digital (A/D) and digital-to-analog (D/A) conversions and associated filtering required for ANSI T1.413 Issue 2 modems.
- The THS6002 Line Driver provides the necessary high-speed line drivers and receives circuitry to drive the ADSL line. It is available in TI's patented PowerPad™ package, reducing the die size and greatly improving thermal dissipation characteristics.

This chipset offers the following benefits:

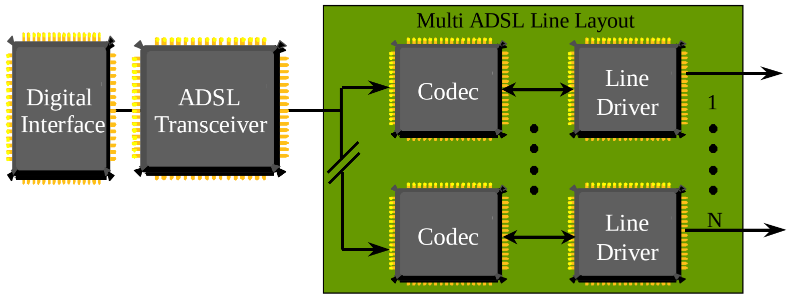
- ANSI T1.413 Issue 2 standard compliance via DSP programmability.
- All the devices have been optimized to work together to ensure maximum possible reach and data rate under varying line conditions.
- A programmable architecture facilitates quick and easy software upgrades, code updates and future standard implementations such as the proposed G.lite standard.
- An extensible architecture that can support a variety of interfaces – PCI, Ethernet and ATM (today); USB/IEEE 1394 and UTOPIA (future).
- A scalable architecture that supports a variety of data rates. This allows manufactures to build universal single board platforms that can be software upgraded.

For client applications, Texas Instruments provides the TNETD2000P™ chipset. This chipset incorporates *all* of the devices in the TNETD2000 chipset. Additionally, it includes the TNETD2300 Host Interface. This device handles all ATM, SAR, AAL5 (1024 VCI address range) and PCI bus[9] control functions for efficient host bus utilization. In addition, its advanced traffic shaping features ensures optimum network bandwidth utilization.

For Central Office applications, the TNETD2000C chipset supports a multi-line architecture, as shown in Figure 11.

[9] PCI 2.1 compliant interface and Bus Master capability with DMA bursting.

Figure 11. Multi-Line Architecture



The multi-line architecture offers the following benefits:

- *Cost reduction.* The transceiver was developed with multi-line communication applications in mind. This enables equipment manufactures to lower their cost per channel/line and reduce board space/power.
- *Scalability.* The current generation transceiver implements two full rate ADSL lines, but as the performance of the DSP core scales, so does the number of full rate ADSL lines it can support. Additionally, the current chip set can also support almost twice the number of lines carrying the lower rate proposed G.lite standard.

Conclusion

DSPs are necessary to evolve modem technology and to support the high data rates of ADSL. Many of the advanced features available today on voiceband and ADSL modems would not have been possible without the use of DSPs to implement complex mathematical algorithms in a cost-effective manner. Additionally, programmable DSPs offer the benefits of rapid prototyping and verification. This results in fast time to market, tracking of rapid evolution of new standards, introduction of software upgrades in a timely manner, and so on. DSPs are also useful in voice and video processing; hence, they are a natural fit as the platform of choice to develop cost effective multimedia solutions.

Texas Instruments, the world leader in DSP solutions, and Amati Corporation, the recognized leader in ADSL, have combined to develop the TNETD2000. This fifth generation chipset incorporates many of the ADSL features discussed in this paper. This chipset is based on a programmable DSP core to provide the highest performance and longest reach possible at any bit rate. The TNETD2000 chipset provides the customer with the benefits of programmability, extensibility and scalability.

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BBS	(281) 274-2323
email	dsph@ti.com

Americas

Phone	+1(972) 644-5580
Fax	+1(972) 480-7800
Email	sc-infomaster@ti.com

Europe, Middle East, and Africa

Phone	
Deutsch	+49-(0) 8161 80 3311
English	+44-(0) 1604 66 3399
Francais	+33-(0) 1-30 70 11 64
Italiano	+33-(0) 1-30 70 11 67
Fax	+33-(0) 1-30-70 10 32
Email	epic@ti.com

Japan

Phone	
International	+81-3-3457-0972
Domestic	+0120-81-0026
Fax	
International	+81-3-3457-1259
Domestic	+0120-81-0036
Email	pic-japan@ti.com

Asia

Phone	
International	+886-2-3786800
Domestic	
Australia	1-800-881-011

Asia (continued)

TI Number	-800-800-1450
China	10811
TI Number	-800-800-1450
Hong Kong	800-96-1111
TI Number	-800-800-1450
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