Evaluation criteria for ADSL analog front end

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Introduction

The latest ADSL standard proposed by the International Telecommunication Union (ITU), with the draft recommendation G.992.3-ADSL2, is the next evolution for second-generation ADSL. A summary of the first-generation ADSL evaluation and design criteria is helpful in understanding first-generation ADSL in the mass market and in designing the specifications for second-generation ADSL.

The AFE1302 customer premise equipment (CPE) modem is one practical ADSL analog front end (AFE) design and evaluation example. In this design, an AFE1302 chip and AFE reference design printed circuit board are used for the ADSL CPE modem and CPE PCI card. The maximum reach of the CPE modem is an 18,000-ft (5.5-km) loop with a downstream net bit rate of 928 Kbps and an upstream rate of 416 Kbps.

The maximum transmitted power is defined by ITU G.992 power spectral density (PSD) mask requirements. The crosstalk from coexisting ADSL and ISDN lines can be minimized by limiting the maximum strength of the transmitted ADSL signal power over the twisted-pair long wires. In addition, Electromagnetic Compatibility Society legislation requires that the ADSL transmission system not interfere with AM/FM radio reception. These two requirements place a PSD limit on the strength of the transmitted ADSL signal power.

This article discusses some evaluation and design considerations for addressing transmitted data speed, telephone twisted-pair loop attenuation, the hybrid circuit, and the multitone power ratio (MTPR).

Transmitted data speed and the total signal-to-noise ratio

Transmitted data speed is the first consideration for ADSL CPE modem design and evaluation. The transmitted data speed is a function of the total signal-to-noise ratio (SNR). The total SNR is the telephone loop SNR plus the modem receive path SNR.

The discrete multitone (DMT) modulation was standardized for the ADSL system by the telecommunication standardization sector of the ITU. For DMT-based ADSL, each subcarrier or tone is spaced at 4.312-kHz intervals. The subcarrier assignment is defined by ITU standard G.992. For example, the subcarrier assignment in G.992.1 annex A can be used for frequency division multiplexing to separate upstream and downstream signals. The subcarriers 31–255 (ITU G.992.1), or subcarriers 31–127 (ITU G.992.2), are reserved for downstream. The subcarriers 0–30—e.g., a maximum total of 31 subcarriers—may be assigned for upstream. The lowest-frequency subcarriers may be set to zero to allow for voice on the same line, such as in plain old telephone service (POTS).

The lowest-frequency subcarrier used for upstream is determined by the POTS/ADSL splitting filter. The number of upstream and downstream subcarriers is determined by the receive and transmit filters. The actual number of subcarriers employed to modulate data may be less than the maximum and is determined during the initialization sequence. The transmitter designates a subset of the maximum available subcarriers for a connection during the channel analysis phase.

The process of ADSL DMT modulation actually modulates each subcarrier as $2^{b(i)}$ quadrature amplitude modulation (QAM). The superscript b(i) denotes the number of bits in the i*th* subcarrier. Subcarriers with a lower SNR will be assigned fewer bits to do a small-number QAM constellation. Subcarriers with a higher SNR will be assigned more bits to create a large-number QAM constellation. Suppose that the i*th* subcarrier is assigned 8 bits; then the size of the carrier QAM constellation is $2^8 = 256$ QAM.

The bits are determined by the SNR measured during the channel analysis initialization procedure. The initialization of the CPE modem is performed in five steps: Handshake procedures, channel discovery, transceiver training, channel analysis, and exchange.

During the exchange phase, each receiver communicates to its far-end transmitter the number of bits and relative power levels used on each DMT tone or subcarrier, as well as the final data rate information. After the successful initialization sequence, the transceivers can start communication with actual data.

The channel analysis phase is used to measure the channel characteristics for both directions of transmission. In other words, it measures the channel transfer function versus the frequency response characteristic. The downstream channel characteristics are measured at the CPE side, and the upstream channel characteristics are measured at the central office (CO) side.

During the channel analysis phase, the receiver estimates the transmitted channel gain of each subcarrier in preparation for computing the total SNR for each subcarrier. Then each subcarrier is assigned the number of bits it will carry. The sum of all the bits assigned to all of the subcarriers within the transmitting period (per DMT symbol) determines the transmitted data speed.

The number of bits assigned per subcarrier can be calculated by

$$b(i) = \log_2[1 + snr(i) \times g/\gamma \times m]. \quad (1)$$

Further, $2^{b(i)} - 1 = snr(i) \times g/\gamma \times m$, and

$10\log_{10}[2^{b(i)} - 1]$
$= 10\log_{10}[snr(i)] + 10\log_{10}(g) - 10\log_{10}(\gamma) - 10\log_{10}(m)$.

Only if $2^{b(i)} >> 1$ is the following equation true:

$10\log_{10}[2^{b(i)}] = SNR\ (dB) + G\ (dB) - \Gamma\ (dB) - M\ (dB)$.

Therefore,

$$b(i) = [SNR\ (dB) + G\ (dB) - \Gamma\ (dB) - M\ (dB)]/3\ dB\ (bits). \quad \textbf{(2)}$$

Variable definitions for the preceding equations are as follows:

- b(i) is the number of bits in the *ith* subcarrier.
- i is the subcarrier index from 0 to N – 1; N is the total usable subcarrier number; and the maximum N = 256.
- snr(i) is the SNR per subcarrier; it is a real value that represents the ratio between the received signal power and the received noise power for that subcarrier. SNR (dB) = $10\log_{10}[snr(i)]$.
- γ or Γ is a constant determined by the required bit error rate (BER). For example, $\gamma = 9.55$ and $\Gamma = 10\log_{10}(\gamma) = 9.8$ (dB) for BER $\leq 10^{-7}$.
- g or G = $10\log_{10}(g)$ (dB) is a gain provided by Reed-Solomon error correction coding to make the system robust against impulsive noise bursts.
- m or M = $10\log_{10}(m)$ (dB) is the margin to represent the amount of increased noise relative to the noise power that the system is designed to tolerate and still meet the target BER of $10 \times e^{-7}$, accounting for all coding (trellis coding and Reed-Solomon forward error correction) gains included in the design. This margin can prevent too many bits from swapping if the SNR changes. Normally M = 6 dB is used to prevent online swapping.

G (dB), Γ (dB), and M (dB) are constants. One bit is added to the assigned b(i) if SNR (dB) increases 3 dB in the subcarrier channel; this is the “one-bit-per-3-dB” rule.

Assuming that G (dB) = 2 dB, Γ (dB) = 9.8 dB, and M (dB) = 6 dB, a typical b(i) and SNR (dB) versus the tone number is shown in Figure 1.

The transmitted data speed can be calculated by

$$C = [\Sigma b(i)]/t \text{ for } i = 0 \text{ to } N - 1,$$

where C is the transmitted data speed and t is the transmission period.

Example 1: If the gain of trellis and Reed-Solomon coding is not included, the attainable net rate or the maximum data rate is

$$C = \{\Sigma[SNR\ (dB) - \Gamma\ (dB) - M\ (dB)]/3\ dB\}/t\ (bps)$$

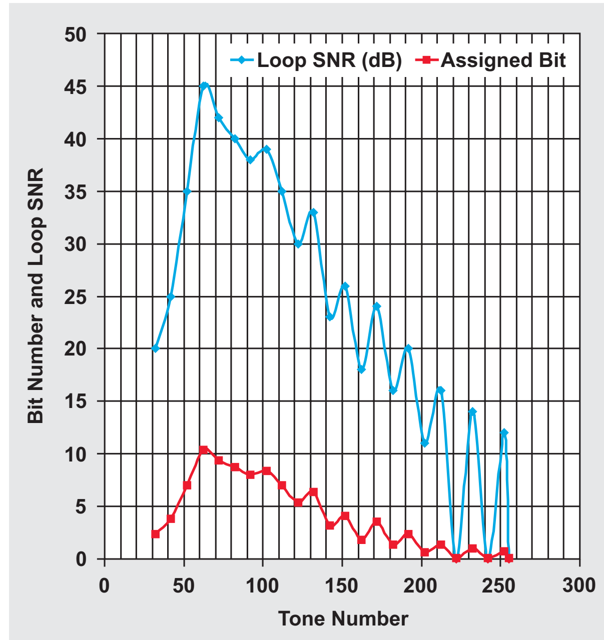
for i = 0 to NSC – 1, where NSC is the number of subcarriers.

Example 2: Additionally, if M = 6 dB and G = 6 dB, the attainable net rate can be simplified as follows:

$$C = \{\Sigma[SNR(i) - \Gamma]/3\ dB\}/t\ (bps)$$

for i = 0 to NSC – 1. For the duration of a DMT symbol, t = 250 µs. The data symbol rate (4000 data symbols/second) is the net average rate at which symbols carrying data frames are transmitted. However, in order to insert

Figure 1. Bit assignment and loop SNR relationship



the synchronization symbol, the symbol rate is defined as the rate at which all symbols, including the synchronization symbol, are transmitted; that is, (69/68) × 4000 = 4058.8 symbols/second.

From the equation given earlier, the transmitted data speed is determined by the total SNR. The modem RX path noise is not covered in this article. The transmitted DMT signal power and loop noise are specified in the recommended ITU G.992 as follows:

- For G.992.1, the downstream PSD is –40 dBm/Hz, from 25.875 to 1104 kHz, for a total transmission power not greater than 20.4 dBm if all subcarriers are used.
- For G.992.2, the downstream PSD is –40 dBm/Hz, from 25.875 to 552 kHz, for a total transmission power not greater than 16.2 dBm if all subcarriers are used.
- The upstream normal transmit PSD, for the channel analysis signal (R-REVERB1) and all subsequent upstream signals, is –38 dBm/Hz, which is equivalent to –1.65-dBm total transmit power in any 4.3125-kHz subcarrier. The maximum transmit PSD should be no higher than –37 dBM/Hz, for an aggregate transmit power not greater than 12.5 dBm if all subcarriers are used.
- The telephone loop noise is assumed to be 31.62 nV/$\sqrt{\text{Hz}}$, or –140 dBm/Hz. It is specified as the quiet line noise PSD N(f) for a particular subcarrier in G.992.3-ADSL2; and it is the rms level of the noise present on the telephone line when no ADSL signals are present on the line.

A practical evaluation result of the transmission channel capacity is the net data speed for the AFE1302 CPE modem. The evaluation is based on using the subcarriers 36–127 for downstream with a total transmission power of less than 15.99 dBm during the training phase and less than 6.06 dBm during the SHOWTIME state. The SHOWTIME state is the normal operation state after all initialization and training are completed. Taking into account all noise contributions, the loop noise test results referred to the telephone line are less than –148 dBm/Hz over receiver bandwidth for the AFE1302 CPE modem. The tested downstream net bit rate can reach 928 Kbps with an 18,000-ft (5.5-km) loop. For the test environments, the loop simulator NSA400 is set as a 26-AWG twisted-pair telephone wire without a bridged tap, the noise margin is 6 dB, and the interleave depth is 16 with no trellis coding.

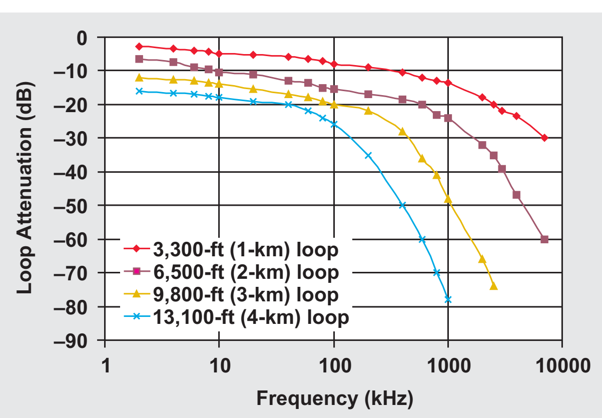
Twisted-pair telephone-loop attenuation

The reach of an ADSL system is the distance over telephone lines that it can transmit and receive information. Reach is the key performance parameter for ADSL and is limited by twisted-pair telephone-loop attenuation.

Telephone-loop attenuation is defined as the difference between the total maximum transmitted power at one end of the telephone line and the total received power at the other end of the line. Loop attenuation is mainly dictated by wire diameters, loop length, and the transmitted signal frequency. The most frequently used telephone twisted-wire diameters are 0.4 mm for 26-AWG wire and 0.5 mm for 24-AWG wire. The typical loop length varies from 6,000 to 18,000 ft (1.8 to 5.5 km) in North America, Europe, and Asia. ADSL signals cover the bandwidth of 25.875 kHz to 1104 kHz as specified in ITU G.992.

According to POTS application statistics, the typical voice signal loss through a telephone loop is 3 to 6 dB per mile (1.6 km) within the voice bandwidth of 300 Hz to 4 kHz. Because ADSL signal frequency is much higher than the POTS signal, and because line attenuation increases as signal frequency increases, the loop attenuation is greater within the ADSL signal bandwidth.

Figure 2. A 24-AWG twisted-pair telephone-loop attenuation vs. frequency



The transfer function of a twisted-pair telephone loop is dominated by the skin effect: High-frequency currents tend to flow only in the outer portion, or skin, of the conductor, resulting in an increase of the attenuation at higher frequencies. In the case of a twisted-pair loop, the attenuation is a function approximately proportional to $f^{1/4}$ for frequencies below 350 kHz, and approximately proportional to $f^{1/2}$ for frequencies above 350 kHz.

The experimental data in Figure 2 shows that the loop attenuation increases from around 5 dB per 3300 ft (1 km) at 10 kHz to over 15 dB per 3300 ft (1 km) at 1 MHz. This means that a 13,100-ft (4-km), 24-AWG twisted-pair telephone loop has an attenuation of over 20 dB at 100 kHz, and over 70 dB at 1 MHz. Over the ADSL signal bandwidth, the twisted-pair loop attenuation varies by more than 50 dB.

During channel analysis, the CO-side receiver calculates the average loop attenuation. Based on the channel analysis signal (R-REVERB1) using subcarriers 7–18, the average upstream loop attenuation is calculated as the difference between the total transmitted power at the CPE side and the total received power measured at the CO side. In G.992.3-ADSL2, for a given loop length, the loop attenuation is defined as

$$\text{LATN (dB)} = 10 \times \log[\Sigma|H(f)|^2/\text{NSC}]$$

for i = 0 to NSC – 1. Variables for this equation are defined as follows:

- NSC is the number of subcarriers.
- H(f) is the channel characteristic function per subcarrier.
- LATN is the difference in decibels between the power received at the near end and that transmitted from the far end over all subcarriers; its dynamic range is from 0 to 102.2 dB.

For example, assume that LATN = 58.2 dB, N = –140 dBm/Hz, G = M = 6 dB, the transmitted signal power is –30 dBm/Hz, and the received signal power is S = –88.2 dBm/Hz for a particular subcarrier. With Equation 2, b(i) can be calculated as 14 bits, which means that 14 bits of data can be assigned on that subcarrier. But if LATN increases to 88.2 dB for a longer loop, then b(i) is only 4 bits because of the one-bit-per-3-dB rule.

A longer-reach loop and a higher subcarrier frequency result in greater loop attenuation, which means that fewer data bits can be assigned to subcarriers, and lower data speed can be achieved.

Hybrid circuit

The ADSL hybrid circuit is a three-port network used to pass the transmit signal from the AFE transmission port to the telephone-loop port and to pass the receive signal from the telephone loop port to the AFE receive port. The ADSL hybrid circuit is also used to eliminate the transmit signal from the receive signal in the receive path. It allows full-duplex transceiver operation in a twisted-pair telephone loop. ADSL hybrid echo return loss is the amount of attenuation between the transmitted signal power and the reflected echo power in the receive path, normally expressed in decibels. It is a key target of evaluation and design for an ADSL AFE. The higher the echo return loss, the less of the transmitted signal power gets into the receive path.

$$H_{echo} = 10 \times |\log(P_{echo}/P_{tx})|,$$

where H_{echo} is the hybrid echo return loss, P_{tx} is the transmitted signal power, and P_{echo} is the portion of transmitted power that enters the receive channel (also called the reflected echo power in the receive path).

The actual twisted-pair loop impedance Z(f) is the frequency function of the telephone line. The magnitude of the twisted-pair impedance is about 600 Ω for the POTS bandwidth; but for the ADSL signal bandwidth, it is about 100 Ω without bridged taps and varies from 60 to 100 Ω with bridged taps.

The hybrid circuit used in the AFE1302 CPE modem is purely resistive and basically an electrical bridge circuit. When the differential drive resistive network becomes balanced, the amount of hybrid echo is minimized.

In a DMT-based ADSL system, the ADSL hybrid circuit and the receive path filter are the critical factors for receiver performance. The transmit path signal and noise power need to be sharply attenuated by the hybrid echo path and receive filter. They can significantly improve CPE modem reach and downstream data speed.

For the typical value of 100-Ω equivalent twisted-pair loop impedance, a total 40-dB attenuation of a hybrid circuit and receive high-pass filter is the minimum evaluation and design requirement. A 20-dB return loss can be easily achieved through the hybrid circuit. The hybrid plus the third-order, LC RX filter on-board can provide a total echo loss of more than 40 dB. That is, the hybrid circuit can attenuate the transmitted power and noise by 40 dB (see Figure 3).

The multitone power ratio

The multitone power ratio (MTPR) is an important feature in the evaluation and design of DMT-based ADSL systems. Better MTPR performance in both the transmission and receive paths results higher data rates in the ADSL system.

The MTPR is the ratio of the power in one subcarrier to the noise power in another selected empty subcarrier. There are no data bits assigned to the selected empty subcarrier in a DMT modulation. The MTPR indicates the degree to which a subcarrier QAM signal is corrupted by distortion from all other subcarrier QAM signals.

The measurement and evaluation for this kind of corruption are different from that for traditional single-tone distortion. Signal-to-noise and distortion ratio (SINAD), spurious-free dynamic range (SFDR), single-tone harmonic distortion, two-tone intermodulation distortion (IMD), third-order intercept point (IP_3), and total harmonic distortion (THD) are used only for expressing single- or two-tone signal integrity and spectral properties.

The MTPR, on the other hand, indicates how the tested device—such as line driver, receiver, line transformer, TX filter, RX filter, or DAC/ADC—responds to the discrete multitone signal, which is not 1 or 2 tones but may be up to 255 or more.

One of the design tasks for the ADSL CPE modem is to maintain the fidelity of the discrete multitone signal. All the analog components used on the modem—such as line transformer, common-mode choke, hybrid circuit, TX filter, RX filter, and line driver and receiver—must be designed so that they cause minimal corruption of the discrete multitone signal.

For any subcarrier, the minimum transmitter MTPR shall be at least 38 dB as specified in G.992, and at least 44 dB as recommended by G.992.3.

Figure 3. Hybrid echo return loss plus RX filter attenuation vs. frequency

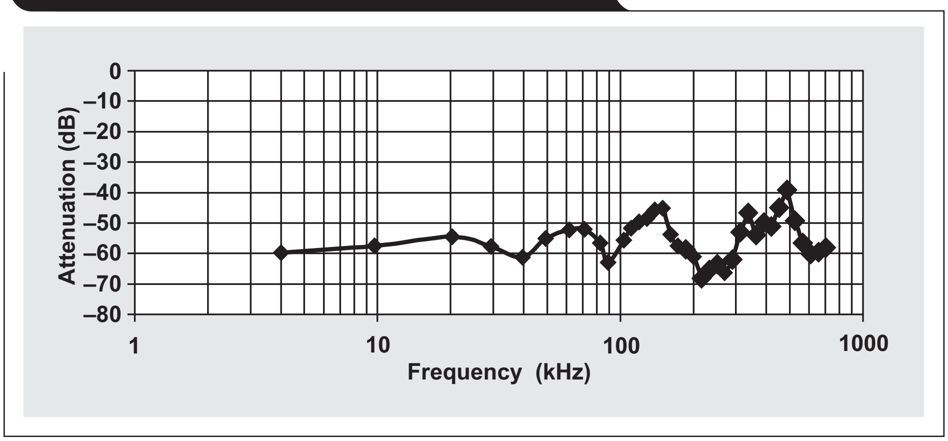
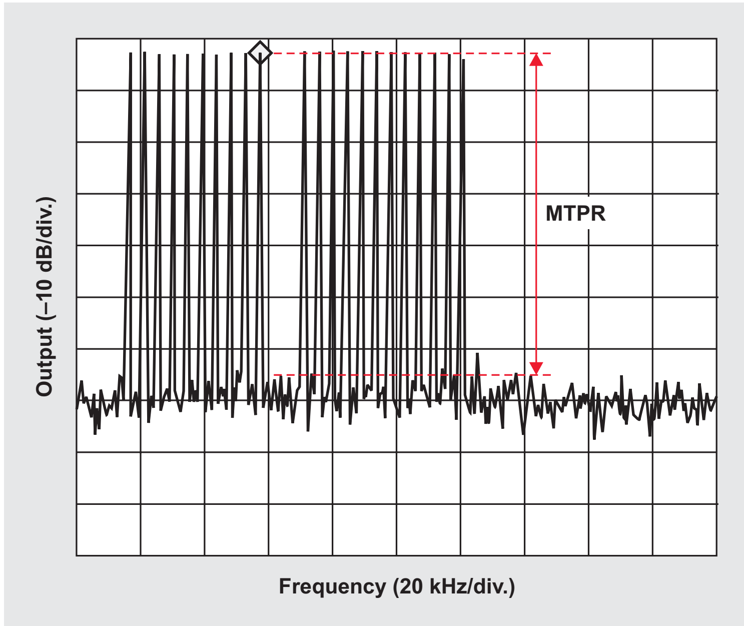


Figure 4. Transmitter path MTPR performance of AFE1302



The MTPR of a tested device can be measured as the dynamic range from peak power in a subcarrier to the peak distortion in an empty tone. For example, a test result shows that an MTPR of 65.82 dB can be achieved in the AFE1302 CPE modem transmission path. The test tool is National Instruments PXI-1002 arbitrary waveform generator 5411.

The transmitter path is defined as the path from the DSP interface TX output to the twisted-pair telephone line interface with 100-Ω impedance (see Figure 4). The transmitted multitone signal is from tones 6–29 and with notch tones 16 and 17. The internal AFE1302's TX programmable gain amplifier gain is 0 dB, and the external TX line driver gain is 15 dB. The tested result of the MTPR is 65.82 dB.

Summary

This article has discussed some evaluation and design considerations for an ADSL AFE. The most critical evaluation criterion is an ADSL transceiver data speed that is limited by the total SNR. The second issue is frequency variation of telephone-loop attenuation and its impact on reach. This article further pointed out that the improvement on hybrid echo return loss significantly increases the received data speed and the reach. The MTPR, a kind of corruption that differs from the traditional single-tone distortion, is another important criterion for designing and evaluating an ADSL AFE.

References

For more information related to this article, you can download an Acrobat Reader file at www-s.ti.com/sc/techlit/*litnumber* and replace *"litnumber"* with the **TI Lit. #** for the materials listed below.

Document Title	TI Lit. #
1. "ITU-T Draft Recommendation G.992.3-ADSL2," International Telecommunication Union, Geneva, Switzerland, April 29–May 10, 2002.	—
2. "ADSL Analog Front-End," Data Sheet	sbws014

Related Web sites

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