Introduction to Digital Subscriber Lines

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1. Introduction

- Telephone service was first provided by A. G. Bell in 1877. Now, about 740 million subscriber lines exist in the world (according to ITU) and 73% of them are for residential use. In U.S., there are 160 million lines.
- Originally, the public switched telephone network (PSTN) is developed for analog speech signals. Thus, only low-speed digital transmission is allowed.
- However, due to the advance of silicon and digital signal processing technology, high-speed transmission now is possible.

- Growth of the transmission rate:
- Motivation for higher bit rates:
  - Growth of PC

- Internet access

The growth of Internet hosts (courtesy of Network Wizards, http://www.nw.com/).
– Small office/Home office (SOHO)

The increasing number of telecommuters in the United States will continue into the year 2000.

- Why try to use a new technology on old lines?
  - Because the copper is already there. It is an installed base.
- Installing a new base of any kind is costly and time-consuming.
- The Market demand:

U.S. xDSL Modem Market - 1996 - 2002
There are three ways to provide digital access through the telephone subscriber line.

- **Analog (modem):**

- **Digital (ISDN):**
2. Subscribe Loop Environment

- Telephone network:
- A *subscriber loop* is the twisted pair telephone loop connecting a subscriber to the central office (CO).
- Resistance design rule: a maximum loop resistance above 1500 ohms will meet powering, signaling, and transmission requirements.
- To extend the service area and reduce loop deployment cost, the digital loop carrier (DLC) was introduced as an electronic multiplexing device.

![Diagram](image)

- **American wire gauge (AWG):**

<table>
<thead>
<tr>
<th>AWG</th>
<th>Metric size (mm)</th>
<th>Loop resistance (Ohms/mile)</th>
</tr>
</thead>
<tbody>
<tr>
<td>28</td>
<td>0.32</td>
<td>685</td>
</tr>
<tr>
<td>26</td>
<td>0.4</td>
<td>441</td>
</tr>
<tr>
<td>24</td>
<td>0.5</td>
<td>277</td>
</tr>
<tr>
<td>22</td>
<td>0.63</td>
<td>174</td>
</tr>
</tbody>
</table>

- **Category:**

<table>
<thead>
<tr>
<th>Category</th>
<th>Bit rates</th>
<th>Applications</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>unspecified</td>
<td>low-speed data circuit (DDS)</td>
</tr>
<tr>
<td>2</td>
<td>1 Mbps</td>
<td>16 Mbps 10BaseT and 4 Mbps token ring</td>
</tr>
<tr>
<td>3</td>
<td>16 Mbps</td>
<td>10BaseT and 16 Mbps token ring</td>
</tr>
<tr>
<td>4</td>
<td>20 Mbps</td>
<td>10/100BaseT, high-speed copper technology</td>
</tr>
</tbody>
</table>
- The carrier serving area (CSA) design rule (for DLC loops):
  - The maximum CSA loop length is 12 Kft for 24 AWG.
  - The maximum CSA loop length is 9 Kft for 26 AWG.
- Speech signal is digitized at the CO with a rate of 64 Kb/s (8 bits x 8 K).
- Due to the quantization noise, the analog modem speed cannot exceed 56 Kb/s.
- Digital data between COs are usually communicated using optic fibers.

- The loop length distributions:
- The feeder and distribution cables are bundled into binder groups 25, 50, and 100 pairs.
- It is common practice to connect a twist-pair from a feeder cable with more than one cables (bridged tap).

For loops beyond 5.5 Km, the signal loss at frequencies above 1 KHz is unacceptable. Series inductors placed at 1.8 Km result in a flatter spectrum at the voice band (loading coil)
There are many impairments in the subscriber loops

- Channel distortion
- Channel attenuation
- Crosstalk noise
- Impulsive noise
- Background/thermal noise
- Radio frequency interference
- Echoes

The channel distortion (ISI):

Inter-symbol interference:
- Channel impulse responses:

- Channel attenuation:

<table>
<thead>
<tr>
<th>DSL type</th>
<th>Transmission peak (Volts)</th>
<th>Maximum power loss (dB)</th>
<th>Minimum received peak (Volts)</th>
</tr>
</thead>
<tbody>
<tr>
<td>ISDN (144kb/s)</td>
<td>2.5</td>
<td>42</td>
<td>0.02</td>
</tr>
<tr>
<td>HDSL (1.5Mb/s)</td>
<td>2.5</td>
<td>35</td>
<td>0.045</td>
</tr>
<tr>
<td>ADSL (1.5Mb/s)</td>
<td>15</td>
<td>45</td>
<td>0.085</td>
</tr>
<tr>
<td>VDSL (26 Mb/s)</td>
<td>3-4</td>
<td>30</td>
<td>0.09-0.12</td>
</tr>
</tbody>
</table>
Since telephone subscriber loops are organized in binder group, there is crosstalk between each twisted pair.

Near end crosstalk (NEXT):

The crosstalk effect is frequency dependent. It is more apparent for the high frequency signal components.

It is generally accepted that NEXT is increased with \( f^{1.5} \).

Far end crosstalk (FEXT):

FEXT is increased with \( f^2 \). However, since the far-end signal is weak, FEXT is usually less concerned.

The transmission rate is essentially limited by the NEXT crosstalk noise.
- Rates vs. loop length:

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- Impulsive noise can come through connections of telephone lines or from the influence of an electromagnetic field.
- Impulsive noise is characterized as a random pulse waveform whose amplitude is much higher than the background noise.
- The frequency of impulses is between 1 and 5 per minute and is somewhat related to daily activities.
- Impulse noise #1 used in test:

![Graph showing impulse noise #1](image1)

- Impulse noise #2 used in test:

![Graph showing impulse noise #2](image2)
- If the receiver is properly designed and implemented, the thermal noise level can be smaller than the background noise.
- The background noise level for the twist-pair telephone loop plant is around -140 dBm/Hz (dBm=10 x log10(watts x 1000)).
- The radio frequency interference comes from AM MV, AM SM, and HAM. This problem is more serious for very high speed DSL.

- Echoes arises in a full-duplex transmission in a telephone loop.

- Due to the impedance mismatch problem, echoes thus arise.
- Note that the echo power is usually much stronger than that of the receiver signal.
3. Transmission and Signal Processing

- A typical digital transmission system:

![Diagram of a digital transmission system]

- Modulation is the process converting each data symbol vector into a continuous-time signal (for transmission).
- The 2B1Q line code (ISDN and HDSL):
  - “2 bits per one quartenary” symbol

\[ f(t) = \sin[\pi t(1-\alpha) / T] + 4\alpha t \cos[\pi t(1+\alpha)] / T \]
\[ \pi[1-(4\alpha t / T)^2] / T \]

*\( \alpha \): the roll-off factor

This is called the square-root raised cosine (SRRC) filter

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- Quadrature amplitude modulation (QAM):

\[ \varphi_1(t) = \sqrt{2/T} \psi(t) \cos(2\omega_c t) \]
\[ \varphi_2(t) = \sqrt{2/T} \psi(t) \sin(2\omega_c t) \]

where \( \psi(t) \) is a baseband function such as the square-root raised cosine one.

- A successive QAM signal:

\[ x(t) = \sqrt{2/T} \sum_k x_{1,k} \psi(t-kT) \cos(\omega_c t) - x_{2,k} \psi(t-kT) \sin(\omega_c t) \]

- Note that the sinusoidal signals are not shifted by \( kT \) on the \( k \)-th symbol. The basis functions are not periodic.
In the complex plane:

Carrierless amplitude/phase modulation (CAP):
- Using the complex representation, we can have QAM as

\[
x(t) = \text{Re}\left\{ \sum_k x_k \psi(t - kT) e^{j\omega_k t} \right\}
\]

\[
= \text{Re}\left\{ \sum_k x_k e^{j\omega_k T} \psi(t - kT) e^{j\omega_k (t-kT)} \right\}
\]


Original QAM:

An alternative QAM

\[
\gamma(t) = \psi(t - kT) e^{j\omega (t-kT)}
\]
The CAP system is the result of removing the phase rotator and the phase de-rotator in a QAM system.

\[ x(t) = \sqrt{\frac{2}{T}} \sum_k x_{1,k} \psi(t - kT) \cos(\omega_c(t - kT)) - x_{2,k} \psi(t - kT) \sin(\omega_c(t - kT)) \]

Thus, there is no carrier and \( \omega_c \) is simply a parameter that indicates the center of the transmission passband.

CAP and QAM are fundamentally equivalent in performance - only implementation is different.

The probability of bit error for QAM/CAP

\[ P_b \approx \frac{2(1 - L^{-1})}{\log_2 L} Q \left[ \sqrt{\frac{2 \log_2 L}{L^2 - 1} \text{SNR}} \right] \]

\( L \): the number of amplitude level (1 - D)

The CAP basis functions:

---

*Impulse Response \( g(t) \) of a Square-root Raised Cosine Shaping Filter.*
- The basic multichannel (muticarrier) transmission:

\[
\begin{align*}
X_1 & \xrightarrow{\Phi_1(t)} Y_1 \\
X_2 & \xrightarrow{\Phi_2(t)} + \xrightarrow{\text{channel}} \xrightarrow{\Phi_2(T-t)} Y_2 \\
& \vdots \\
X_N & \xrightarrow{\Phi_N(t)} \xrightarrow{\Phi_N(T-t)} Y_N
\end{align*}
\]

- Note that if \( \Phi_i(t) \)'s occupy different frequency bands, the orthogonal conditions can be satisfied.

- The simplest \( \Phi_i(t) \) function is the sinusoid having a single frequency. Consider the sampled \( \Phi_i(t) \).

\[ \Phi_k(n) = e^{j\Omega_0 n}, \quad \Omega_0 = \frac{2\pi}{N}, \quad k = 0, 1, \ldots, N-1 \]

- Property:

\[ e^{j\Omega_0 n} \xrightarrow{\text{DFT}} \frac{1}{N} \sum_{n=0}^{N-1} e^{jk(k-m)\Omega_0 n} \begin{cases} 1, & k = m \\ 0, & k \neq m \end{cases} \]

\[ e^{-j\Omega_0 n} \xrightarrow{\text{DFT}} \]
Thus, we can use the property to perform modulation.

\[ Ae^{j(\ell \Omega_0 n + \theta)} \]

Thus, we can define symbols in the frequency domain.

\[ Ae^{j\theta} = a + bj \]

\[ Ae^{-j(\ell \Omega_0 n + \theta)} \]

\[ Ae^{-j\theta} = a - bj \]

Thus, we can define symbols in the frequency domain.

For example:

\[ A \cos(\ell \Omega_0 n + \theta) \]

For multitones (multichannels/multicarriers), we can have

\[ \sum_{\ell} A_\ell \cos(\ell \Omega_0 n + \theta_\ell) \]
In the receiver, we then have

\[ \sum_{t} A_{t} \cos(\Omega t n + \theta_{t}) \]

**ISI:**

To compensate the channel effect in frequency domain (circular convolution), the guard interval is

If the channel response is known (shorter than CP), then

\[ y^{m}(n) = x^{m}(n) \otimes h(n) \]

\[ \Rightarrow X^{m}(e^{j\omega}) = \frac{Y^{m}(e^{j\omega})}{H(e^{j\omega})} \]
The discrete multitone (DMT) system (ITU g.dmt, ANSI T1.413):

- The DMT downstream parameters:
  - Symbol rate: 4 kHz
  - FFT size: 512 samples
  - Cyclic prefix: 32 samples
  - Sampling rate: 2.208 MHz
  - Frequency spacing: 4.3125 KHz

- The DMT upstream parameters:
  - Symbol rate: 4 kHz
  - FFT size: 64 samples
  - Cyclic prefix: 4 samples
  - Sampling rate: 276 KHz
  - Frequency spacing: 4.3125 KHz
• An equalizer is a commonly used device to compensate for the channel effect.

\[
\begin{align*}
 y(n) &= \hat{a}(n - \Delta) \\
(\hat{H}(z)) &\text{ (If no noise, } E(z) = \frac{1}{\hat{H}(z)} z^{-\Delta})
\end{align*}
\]

• Adaptive equalizer:

• Adaptive echo canceller:

• Timing recovery:
  - The purpose of timing recovery is to recover a symbol rate clock, which is used in the far-end transmitter D/A device, from the received waveform.
4. Standards

- ISDN: integrated services digital network. This can be seen as the first DSL service (since 1986).

  B channel ($64 \times 2 = 128 \text{K}$) : for data transmission
  D channel ($16 \text{K}$) : for signaling and user data packet
  Framing and line control ($16\text{K}$)
  Total $160 \text{Kb/s}$

- The low data rate ultimately become a major drawback of this DSL technology.
- Nevertheless, many ISDN lines still deployed (1.7 million in 1994 and 6 million in 1996)
- HDSL (high-bit rate digital subscriber loop)
- HSDL was essentially up-scaled the ISDN designs. The first HDSL service was placed in 1992.
- It is now used to replace the T1 and E1 services for cost reduction. This benefit is mainly due to the elimination of mid-space repeaters.

![Diagram showing HDSL setup]

2 pairs wires: 1.544 Mb/s
3 pairs wires: 2.048 Mb/s

- 2B1Q and CAP are the de-facto standards in North America.
- 2B1Q and CAP have the same performance.
- 2B1Q is the selected line code for ETSI standard.
- ADSL (Asymmetric DSL)
- The ADSL concept evolves during the early 1990. At first, ADSL was considered at a fixed rate 1.5 Mb/s downstream and 16 Kb/s upstream for MPEG-1.
- Recently, ADSL finds wide applications with the internet access.
  - Downstream: up to 9 Mb/s
  - Upstream: up to 1 Mb/s
  - Simultaneously provides POTS service
  - One pair of wires
- The conceptual definition of ADSL began in 1989 and the ITU gave preliminary approval to a set of ADSL recommendations in 1998.
- ADSL spectrum:

![ADSL Spectrum Diagram]

- DMT was selected as the ANSI standard in 1993.

- VDSL (Very-high-bit rate DSL):
  - VDSL is an extension of ADSL technology to higher rates up to 52 Mb/s downstream (26 Mb/s two ways)
  - Discussion of VDSL in standard committee began in 1994. VDSL is intended to support all applications, voice, data, video (even HDTV).
  - VDSL will be primarily used for loops fed from an optical network unit (ONU) which is typical located less than a KM from the customer.
  - Both QAM and DMT have been proposed for the standard candidates.
**VDSL service area:**

Coverage area of VDSL when deployed from remote ONUs.

**VDSL service range:**

Range and reach of VDSL from a central location.