Introduction to Voice and Telephone Technology

Session 401
Voice Is Not A Network

- Voice is an Application
- Complete understanding of Voice Application fundamentals helps us to design and build better Networks

Objective

To Prepare the Data Communications Professional for Voice and Data Network Integration by Providing Voice Technology Fundamentals
### Agenda

- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques

### Telephony Equipment

- Telephone set
- Key system
  - Optimizes use of telephone sets to lines
  - Mechanical to electronic
  - Two to ten telephone handsets is typical
- PBX (Private Branch Exchange)
  - Advanced features and call routing
  - Tens to hundreds of telephone handsets
- Central office switch
Analog Telephony—Connection Basics

Basic Call Progress: On-Hook

DC Voltage
Open Circuit
No Current Flow
Basic Call Progress: Off-Hook

- Off-Hook Closed Circuit
- DC Current Dial Tone
- Local Loop
- Telephone Switch
- Local Loop

Basic Call Progress: Dialing

- Off-Hook Closed Circuit
- Dialed Digits Pulses or Tones
- DC Current
- Local Loop
Basic Call Progress: Switching

- Off-Hook Closed Circuit
- DC Current
- Local Loop
- Telephone Switch
- Address to Port Translation
- Local Loop

Basic Call Progress: Ringing

- Off-Hook Closed Circuit
- Ring Back Tone DC Current
- Local Loop
- Telephone Switch
- DC Open Cct. Ringing Tone
- Local Loop
Basic Call Progress: Talking

- Off-Hook Closed Circuit
- Voice Energy DC Current
- Local Loop
- Telephone Switch
- Voice Energy DC Current
- Local Loop

Analog Telephony—Signaling

- Supervisory
- Addressing
- Call progress
Analog Telephony—Supervisory Signaling

- **Loop start**
  - Almost all telephones
  - Current flow sensed

- **Ground start**
  - Switch Trunk Lines
  - Momentary ground ring lead

### Loop Start

Station | Loop (Local or Station) | PBX or Central Office
---|---|---

1. **DC Current**
2. **Ringing**

Cisco Systems Confidential
E&M Signaling

• PBXs, switches
  Separate signaling leads for each direction
  E-Lead (inbound direction)
  M-Lead (outbound direction)
  Allows independent signaling

<table>
<thead>
<tr>
<th>State</th>
<th>E-Lead</th>
<th>M-Lead</th>
</tr>
</thead>
<tbody>
<tr>
<td>On-Hook</td>
<td>Open</td>
<td>Ground</td>
</tr>
<tr>
<td>Off-Hook</td>
<td>Ground</td>
<td>Battery Voltage</td>
</tr>
</tbody>
</table>

Signaling and Addressing

Analog Transmission
“In-Band” Signaling
0–9, *, # (12 Digits)

Digital Transmission
“Out-of-Band” Message-Based Signaling

Dial Pulse
DTMF
ISDN
Pulse Dialing

Make (Circuit Closed)

Break (Circuit Open)

Off-Hook  |  Dialing  |  Inter-Digit  |  Next Digit

Pulse Period (100 ms)

US: 60/40 Break/Make

700 ms

Tone Dialing

Dual Tone Multifrequency (DTMF)

<table>
<thead>
<tr>
<th>1209</th>
<th>1336</th>
<th>1477</th>
<th>1633</th>
</tr>
</thead>
<tbody>
<tr>
<td>697</td>
<td>1</td>
<td>2</td>
<td>3</td>
</tr>
<tr>
<td>770</td>
<td>4</td>
<td>5</td>
<td>6</td>
</tr>
<tr>
<td>852</td>
<td>7</td>
<td>8</td>
<td>9</td>
</tr>
<tr>
<td>941</td>
<td>*</td>
<td>0</td>
<td>#</td>
</tr>
</tbody>
</table>

Cisco Systems Confidential
Voice Channel Bandwidth

Output Voltage or Energy

Frequency (K-Hertz)

Voice Channel

Voice Signal

.2 1 2 3 4

Tone Dialing Signals

Systems Control Signals

Switching Systems

Manual Control—Switch/Cord Boards

Off-Hook Indicator

Tip Ring

Manual Ring

Patch Cord Pairs
Local Access Network

Feeder Route Boundary

Central Office

40,000 to 50,000 Lines

Serving Area Boundary

PSTN Network Hierarchy

Class | Name
--- | ---
1 | Regional Center
2 | Sectional Center
3 | Primary Center
4C | Toll Center
4P | Toll Point
4X | Interm. Point
5 | End Office
5R | EO w/ RSU
R | Remote Sw. Unit

Cisco Systems Confidential
Types of Voice Circuits

- Serving Area 415-NXX-XXX
  - 415-577-3800
  - Class 5 Switch
  - 415-577-3801
- Serving Area 510-NXX-XXX
  - 510-655-1400
  - Class 5 Switch
  - FX Foreign Exchange

ARD
Auto Ring Down

Echo in Voice Networks

- Talker
- Delay
- Listener

- Talker Echo
- Listener Echo
Normal Signal Flow

- Two- to four-wire hybrid combines receive-and transmit-signals over the same pair
- Two-wire impedance must match four-wire impedance

How Does Echo Happen?

Echo Is Due to a Reflection

Impedance Mismatch at the 2w-4w Hybrid Is the Most Common Reason for Echo
**Echo Is Always Present**

Echo as a Problem is a Function of the Echo Delay, and the Magnitude of the Echo

![Graph](image)

- **Echo Is Unnoticeable**
- **Echo Is a Problem**
- **Echo Is Always Present**

**Ways to Defeat Echo**

- **Increase the loss in the echo path**
  - Can often be the solution
  - Disadvantage: static setting and reduces the signal strength of the speaker

- **Echo suppresser**
  - Acts like a noise gate, effectively making communications half-duplex
Echo Canceler

Most Effective Means for Removing Echo

Echo Canceler Block Diagram

Summary

- Information exchange based on voltage, current flow, grounding, and so on
- Analog voice technology dates back to the late 1800s
Agenda

- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques

Digital Telephony

Digital Trunking

Switch

Analog Loop

POTS

A to D Conversion

Digital Network

Switch

ISDN

Digital Loop Digital Network

Switch

Cisco Systems Confidential
Digital Telephony

Pulse Code Modulation—Nyquist Theorem

Voice Bandwidth = 200 Hz to 3400 Hz

Analog Audio Source → Sampling Stage

Codec Technique

Sampling Stage

= Sample
8 bits per sample
8 kHz (8,000 Samples/Sec)

Pulse Code Modulation—Analog to Digital Conversion

Quantizing Stage

A—Law (Europe)

Quantizing Noise

µ—Law (USA)

10010011011

Cisco Systems Confidential
**Time Division Multiplexer**  
**Example: T1 Channel Bank**

### INPUTS
- Analog or Digital Interface Cards
- Ch. 1, Ch. 2, Ch. 3, Ch. 4, Ch. 5, Ch. 6, Ch. 7-23, Ch. 24

**OUTPUT**
- 8,000 Frames per Second (1 Frame per 125 µs)
- 8 Bits from Each Channel Input in Sequential Order

**T1 Multiplexer**

- Each input represents 64 kbps

- 64 kbps x 24 = 1.536 Mbps
- Add Framing Bits = 8 Kbps
- Total Bit Rate: 1.544 Mbps

---

**DS1 Superframe (D4) Format**

- 193rd bit of each frame used for frame synchronization
- D4 framing is 12 frames
- D4 framing pattern is: 100011011100
- Channel Associated Signaling (CAS) robs the LSB of every byte in frames 6 and 12 for AB bits
- Common Channel Signaling (ISDN) uses TS 24

### Table: Framing Bits

<table>
<thead>
<tr>
<th>Frame Number</th>
<th>Framing Bit Value</th>
<th>Traffic</th>
<th>Signaling</th>
<th>T</th>
<th>2</th>
<th>4</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>1</td>
<td></td>
<td>Bits 1-7</td>
<td>Bit 8</td>
<td>*</td>
<td>A</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>9</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>11</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>12</td>
<td>0</td>
<td></td>
<td>Bits 1-7</td>
<td>Bit 8</td>
<td>*</td>
<td>A</td>
</tr>
</tbody>
</table>

Cisco Systems Confidential
### Extended Superframe (ESF)

<table>
<thead>
<tr>
<th>Frame Number</th>
<th>S Bits</th>
<th>Bit Use in Each Channel Time Slot</th>
<th>Signaling—Bit Use Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fe DL BC</td>
<td>Traffic</td>
<td>Signaling</td>
<td>T</td>
</tr>
<tr>
<td>1 2 0</td>
<td>m m m</td>
<td>-</td>
<td>C1</td>
</tr>
<tr>
<td>3 4 5</td>
<td>m m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>6</td>
<td>- C2</td>
<td>Bits 1–7</td>
<td>Bit 8</td>
</tr>
<tr>
<td>7 8 9 10</td>
<td>0 m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>11</td>
<td>- C3</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>12 1</td>
<td>- -</td>
<td>-</td>
<td>Bits 1–7</td>
</tr>
<tr>
<td>13 14 15</td>
<td>m m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>16 17</td>
<td>0 m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>18</td>
<td>- C4</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>19 20 21</td>
<td>m m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>22 23</td>
<td>0 m m</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>24 1</td>
<td>- -</td>
<td>-</td>
<td>Bits 1–7</td>
</tr>
</tbody>
</table>

### Digital Signaling Schemes

#### Channel Associated Signaling

**Extended Superframe**

- **Supervision On/Off Hook**
- **Address Signaling (Dial Pulse)**

**Bit Frame**
- A 6th
- B 12th
- C 18th
- D 24th
Digital Signaling Schemes

Common Channel Signaling

Extended Super Frame

“In-Band” Audio Address Signaling (DTMF)

64 Kbps Signaling Channel in TS24 of Each Frame (e.g. ISDN D Channel Q.931 Messages)

Digital Telephony—Synchronization

- Bit synchronization
  Primary reference source
  Ones density
- Time-slot synchronization
  Bits/bytes/channels
- Frame alignment
  193rd Bit Pattern
Digital Telephony—Synchronization

One Multiframe (ESF)

3 ms

1 Frame, 125µs, 193 bits 24 Time Slots

1 Channel Time Slot, 5.18µs

Synchronization—Traditional Network Clocking Strata

Stratum

1 2 3 4

Master Clock

PRS

Timing

Toll Office

End Office

DCS

PBX

Cisco Systems Confidential
Agenda

- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques

Voice Coding and Compression

- Speech-coding schemes
- Subjective impairment analysis: mean opinion scores
- Digitizing voice
- Voice compression
  ADPCM
  CELP (LD-CELP and CSA-CELP)
  Silence removal techniques (DSI using VAD)
Voice Compression Technologies

Bandwidth (Kbps)

Quality

Unacceptable
Business Quality
Toll Quality

(Commercial)

PCM (G.711)
ADPCM 32 (G.726)
ADPCM 24 (G.726)
ADPCM 16 (G.726)
LDCELP 16 (G.728)
CS-ACELP 8 (G.729)

Speech-Coding Schemes

• Waveform coders
  Non-linear approximation of the actual waveform
  Examples: PCM, ADPCM

• Vocoders
  Synthesized voice
  Example: LPC

• Hybrid coders
  Linear waveform approximation with synthesized voice
  Example: CELP
Subjective Impairment Analysis: Mean Opinion Scores

<table>
<thead>
<tr>
<th>Score</th>
<th>Quality</th>
<th>Description of Impairment</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>Just Perceptible, not Annoying</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>Perceptible and Slightly Annoying</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>Annoying but not Objectionable</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
<td>Very Annoying and Objectionable</td>
</tr>
</tbody>
</table>

Subjective Quality (MOS) vs. Kbps

Hybrid Coders vs. Waveform Coders vs. Vcoders

Measuring Mean Opinion Scores: ITU P.800 Series

Source → Channel Simulation → Impairment → Codec 'X'

“Nowadays, a chicken leg is a rare dish”

<table>
<thead>
<tr>
<th>Rating</th>
<th>Level of Speech Quality</th>
<th>Distortion</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>Just perceptible but not annoying</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>Perceptible and slightly annoying</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>Annoying but not objectionable</td>
</tr>
<tr>
<td>1</td>
<td>Unsatisfactory</td>
<td>Very annoying and objectionable</td>
</tr>
</tbody>
</table>
Digitizing Voice: PCM Waveform Encoding Review

- Nyquist Theorem: sample at twice the highest frequency
  Voice frequency range: 200-3400 Hz
  Sampling frequency = 8000/sec (every 125µs)
  Bit rate: \((2 \times 4 \text{ kHz}) \times 8\) bits per sample
  \(= 64,000\) bits per second (DS-0)

- By far the most commonly used method

Nonlinear vs. Linear Encoding

- Nonlinear Encoding: Closely follows human voice characteristics. High amplitude signals have more quantization distortion.
- Linear Encoding: Relatively easy to analyze, synthesize and regenerate. All amplitudes have roughly equal quantization distortion.
Voice CODECs: Waveform Coders

- Filter
- Sampling
- Quantizing
- Encoding

Waveform ENCODER

1110010010010110

Waveform DECODER

Voice Compression

- Objective: reduce bandwidth consumption
  Compression algorithms are optimized for voice
  Unlike data compression: these are “loose”

- Drawbacks/tradeoffs
  Quantization distortion
  Tandem switching degradation
  Delay (echo)
Voice Compression—ADPCM

- Adaptive Differential Pulse Code Modulation
  - Waveform coding scheme
  - Adaptive: automatic companding
  - Differential: encode the changes between samples only
  - Rates and bits per sample:
    - 32 Kbps = 8 Kbps x 4 bits/sample
    - 24 Kbps = 8 Kbps x 3 bits/sample
    - 16 Kbps = 8 Kbps x 2 bits/sample

Voice Compression—CELP

- Code excited linear predictive
- Very high voice quality at low-bit rates, processor intensive, use of DSPs
- G.728: LD-CELP—16 Kbps
- G.729: CSA-CELP—8 Kbps
  - G.729a variant—“stripped down” 8 kbps (with a noticeable quality difference) to reduce processing load, allows two voice channels encoded per DSP
Voice CODECs: Hybrid Coders

PCM Encoder
11100100100101
Sample Frames

PCM Decoder

Analysis
Model Parameters

Synthesis
10110010
Model Parameters

Vocal Cords
Throat
Nose
Mouth

Human Speech Model

G.729

16-Bit Linear PCM

Recipe or Code Book

A-sound
K-sound

Code

Recipe

10.1.1.1

Cake

Recipe

Play K, A, and K

Invalid

Cisco Systems Confidential
Digital Speech Interpolation (DSI)

- Voice Activity Detection (VAD)
- Removal of voice silence
- Examines voice for power, change of power, frequency and change of frequency
- All factors must indicate voice “fits into the window” before cells are constructed
- Automatically disabled for fax/modem

Voice Activity Detection

<table>
<thead>
<tr>
<th>Voice Activity (Power Level)</th>
<th>Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>- 54 dbm</td>
<td>Voice “Spurt”</td>
</tr>
<tr>
<td>- 31 dbm</td>
<td>Hang Timer</td>
</tr>
<tr>
<td>B/W Saved</td>
<td>No Voice Traffic Sent</td>
</tr>
<tr>
<td>Pink Noise</td>
<td>Voice “Spurt”</td>
</tr>
<tr>
<td>SID Buffer</td>
<td>Silence</td>
</tr>
<tr>
<td>SID</td>
<td>B/W Saved</td>
</tr>
</tbody>
</table>

Cisco Systems Confidential
Bandwidth Requirements

Voice Band Traffic

<table>
<thead>
<tr>
<th>Encoding/Compression</th>
<th>Result Bit Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 PCM A-Law/µ-Law</td>
<td>64 kbps (DS0)</td>
</tr>
<tr>
<td>G.726 ADPCM</td>
<td>16, 24, 32, 40 kbps</td>
</tr>
<tr>
<td>G.729 CS-ACELP</td>
<td>8 kbps</td>
</tr>
<tr>
<td>G.728 LD-CELP</td>
<td>16 kbps</td>
</tr>
<tr>
<td>G.723.1 CELP</td>
<td>6.3/5.3 kbps (Variable)</td>
</tr>
</tbody>
</table>

Agenda

- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques
Voice Network Transport

- Voice Network Transport is typically TDM circuit-based:
  - T1/E1
  - DS3/E3
  - SONET (OC-3, OC-12, etc.)
- But can also be packet-based:
  - ATM
  - Frame Relay
  - IP

Data Is Overtaking Voice

Evolution from TDM-based transport to packets/cells or a combination

Source: Electronicast
The Tyranny of the DS0

- Switching and transport based on circuits
- Rigid structure yields high cost for packet

TDM Transport Efficiency

Utilization

- Wasted bandwidth
- No congestion

Types of Traffic
- Voice
- Legacy
- LAN
- Video
Packet Transport Efficiency

Types of Traffic
- Voice
- Legacy
- LAN
- Video

Utilization
- 90–95%

Individual Packets
- High bandwidth efficiency
- Congestion management

Delay

Sender
- First Bit Transmitted
- PBX

Receiver
- Last Bit Received
- PBX

Network Transit Delay

Processing Delay

End-to-End Delay

Cisco Systems Confidential
Delay Variation—“Jitter”

Sender Transmits

A

B

C

Network

Sink Receives

D1

D2 = D1

D3 ≠ D2

Voice Delay Guidelines

<table>
<thead>
<tr>
<th>One Way Delay (msec)</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0–150</td>
<td>Acceptable for Most User Applications</td>
</tr>
<tr>
<td>150–400</td>
<td>Acceptable Provided That Administrations Are Aware of the Transmission Time Impact on the Transmission Quality of User Applications</td>
</tr>
<tr>
<td>400+</td>
<td>Unacceptable for General Network Planning Purposes; However, It Is Recognized That in Some Exceptional Cases This Limit Will Be Exceeded</td>
</tr>
</tbody>
</table>

ITU’s G.114 Recommendation
Delay in Perspective

Cumulative Transmission Path Delay

Time (msec)

Delay Target

Fixed Delay Components

- Propagation—Six microseconds per kilometer
- Serialization
- Processing
  Coding/compression/decompression/decoding
  Packetization
Variable Delay Components

- Queuing delay
- Dejitter buffers
- Variable packet sizes

An Example

- Assumptions:

  We have eight trunks
  
  We are going to use CS-ACELP that uses 8 Kbps per voice channel
  
  Our uplink is 64 Kbps
  
  Voice is using a high priority queue and no other traffic is being used
Delay Calculation

- Propagation Delay—32 ms
- Coder Delay—25 ms
- Serialization Delay—3 ms
- Dejitter Buffer—50 ms
- Queuing Delay—6 ms

Total Delay: 110 msec

### Fixed Delay
- Coder Delay G.729 (5 msec Look Ahead): 5 msec
- Coder Delay G.729 (10 msec per Frame): 20 msec
- Packetization Delay—Included in Coder Delay
- Max Queuing Delay 64 kbps Trunk: 21 msec
- Serialization Delay 64 kbps Trunk: 3 msec
- Propagation Delay (Private Lines): 32 msec
- Network Delay (e.g., Public Frame Relay Svc)
- Dejitter Buffer: 50 msec

### Total Delay: 110 msec

Variable Delay Calculation

- We have eight trunks, so in the worst case we will have to wait for seven voice calls prior to ours
- To put one voice frame out on a 64Kbps link takes 3msec
- 1 byte over a 64Kbps link takes 125 microseconds. We have a 20 byte frame relay frame with 4 bytes of overhead. 125 * 24 = 3000 usecs or 3 msec
- Does not factor in waiting for a possible data packet or the impact of variable sized frames
- Assumes voice prioritization of frames
Delay Calculation

<table>
<thead>
<tr>
<th></th>
<th>Fixed Delay</th>
<th>Variable Delay</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>DELY #1</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Coder Delay G.729</td>
<td>25 msec</td>
<td></td>
</tr>
<tr>
<td>Packetization Delay</td>
<td></td>
<td></td>
</tr>
<tr>
<td>(Included in Coder Delay)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Max Queuing Delay 4 Mbps Trunk</td>
<td>21 msec</td>
<td></td>
</tr>
<tr>
<td>Serialization Delay 4 Mbps Trunk</td>
<td>3 msec</td>
<td></td>
</tr>
<tr>
<td>Propagation Delay (Private Lines)</td>
<td>32 msec</td>
<td></td>
</tr>
<tr>
<td>Dejitter Buffer</td>
<td>50 msec</td>
<td></td>
</tr>
<tr>
<td>Tandem Switch</td>
<td>—</td>
<td></td>
</tr>
<tr>
<td><strong>Delay #1 Total</strong></td>
<td>110 msec</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th></th>
<th>Fixed Delay</th>
<th>Variable Delay</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>DELY #2</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Coder Delay G.729</td>
<td>25 msec</td>
<td></td>
</tr>
<tr>
<td>Packetization Delay</td>
<td></td>
<td></td>
</tr>
<tr>
<td>(Included in Coder Delay)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Max Queuing Delay 2 Mbps Trunk</td>
<td>.7 msec</td>
<td></td>
</tr>
<tr>
<td>Serialization Delay 2 Mbps Trunk</td>
<td>0.1 msec</td>
<td></td>
</tr>
<tr>
<td>Propagation Delay (Private Lines)</td>
<td>5 msec</td>
<td></td>
</tr>
<tr>
<td>Dejitter Buffer</td>
<td>50 msec</td>
<td></td>
</tr>
<tr>
<td><strong>Delay #2 Total</strong></td>
<td>80 msec</td>
<td></td>
</tr>
<tr>
<td><strong>Total Delay</strong></td>
<td>190 msec</td>
<td></td>
</tr>
</tbody>
</table>
Other Useful Voice QoS Schemes in IP

- Custom Queuing, Priority Queuing and Weighted Fair Queuing (WFQ)
- Resource Reservation Protocol (RSVP)
- IP Precedence Bit setting in the ToS Field of the IP Header
- Compressed Real Time Protocol (CRTP)

Summary

- Voice traffic engineering principles still apply
- Packet-based voice trunks can provide efficiency with high quality if properly engineered
- The biggest impact on voice quality over a data network will be as a result of the delay and delay variation
Repeat: Voice Is Not A Network

- Voice is an Application
- Complete understanding of Voice Application fundamentals helps us to design and build better Networks

Agenda

- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques
Digital Voice Signaling Techniques

- ISDN
- Q.930/Q.931
- Signaling System 7
- Voice addressing

ISDN

- Integrated Services Digital Network
  - Part of a network architecture
  - Definition for the access to the network
  - Allows access to multiple services through a single access

- Standards-based
  - ITU recommendations
  - Proprietary implementations
### Network Access

#### Traditional Access
- Customer Equipment (PBX)
- PSTN (CO Lines)
- 800
- Tie Trunks
- FX
- Private Line Data

#### ISDN Access
- Customer Equipment (PBX)
- Telephone Switch
- Public Packet-Switched Network
- PSTN (CO Lines)
- 800
- Tie Trunks
- FX
- Private Line Data

### Terminology

- **B channel** “bearer channel”
  - 64 kbps
  - Carries information (voice, data, video, etc.)
  - DS-0
Terminology (Cont.)

- D channel “signaling channel”
  - 16 Kbps or 64 Kbps
  - Carries instructions between customer equipment and network
  - Carries information
  - Can also carry packet switch data (X.25) for the public packet switched network

Terminology (Cont.)

- BRA/BRI (Basic Rate Access/Basic Rate Interface)
  - 2 B + D
  - 2 x 64 Kbps + 16 Kbps = 144 Kbps (not including overhead)
  - Designed to operate using the average local copper pair
Terminology (Cont.)

- **PRA/PRI** (Primary Rate Access/Primary Rate Interface)
  
  23 B + D
  
  23 x 64 Kbps + 64 Kbps (D Channel) + 8 Kbps (Frame Alignment bit) = 1.544 Mbps
  
  Designed to operate using T1/E1
  
  In E1 environments: 30 B + D

---

**ISDN Reference Points**

- **TE1**
- **TE2**
- **NT1**
- **NT2** (PBX)
- **BRA**
- **S/T**
- **U**
- **PRA**
- **Local Loop**
- **Customer Premises**
- **Carrier**

---

Cisco Systems Confidential
ISDN Reference Points

- **NT1**
  - Terminates local loop
  - Coding and transmission conversion
  - Maintenance and performance monitoring
  - Functions as a CSU

ISDN Reference Points (Cont.)

- **TE1**
  - ISDN compatible equipment

- **TE2**
  - Non-ISDN compatible equipment
  - Requires TA

- **TA**
  - Interfaces available for different TE2
  - E.g. RS-232, X.21, V.35, PC-Bus, video, etc.
ISDN Reference Points (Cont.)

• NT2
  Typically a PBX
  Provides switching functions
  Handles Layer 2 and Layer 3 protocols

Access to ISDN

• At the S-reference point:
  RJ-45 (receive and transmit pair)
  Optional power can be provided for TE devices
  Distance:
    1 Km (1 x TE only),
    200 m (8 x TE), 500 m (4 x TE)
  When more than one TE, wires act as a bus
  CSMA/CD
  Limitation: cannot have an extension phone
### Access to ISDN

- **At the U-Reference point (BRA)**
  - Standards differ NA, France, UK vs. Germany vs. Japan
  - In North America, designed to use as much of existing copper plant available
    - Two wire, unloaded local loops are 99% of total
  - Up to 5.5 Km loop length

- **At the U-Reference point (PRA)**
  - T1/E1 standard

### D Channel

- **ISDN Access Protocols are carried in the D channel**
- **Layer 2 and Layer 3 protocol specifications**
  - Protocol specifications are identical for BRA and PRA
- **Layer 2, Q.920/921, LAP-D**
  - Supports the communications for Layer 3
  - Maintains the connections between devices
- **Layer 3, Q.930/931**
  - Call setup, call supervision, call tear down, and supplementary services
  - Uses standard set of messages to communicate
D-Channel Encapsulation

Layer 3
- Protocol Discriminator
- Length of Call Reference
- Call Reference
- Message Type
- Information Elements

Layer 2
- Flag
- Address
- Control
- Information
- CRC
- Flag

Layer 1
- D Channel (16 Kbps or 64 Kbps)

ISDN CCS (Q.930/931) Messages

Call Establishment
- Alerting
- Call proceeding
- Connect
- Connect ack
- Progress
- Setup
- Setup ack

Call Information
- Hold
- Hold ack
- Hold reject
- Resume
- Resume ack
- Resume reject
- Retrieve
- Retrieve ack
- Retrieve reject
- Suspend
- Suspend ack
- Suspend reject
- User information

Call Clearing
- Disconnect
- Release
- Release complete
- Restart
- Restart ack

Miscellaneous
- Congestion control
- Facility
- Information
- Notify
- Register
- Status
- Status inquiry
Public ISDN and Signaling System 7

DSS1 Is a Public ISDN Protocol

ISDN and SS7 “The Bridge Between the Islands”
Agenda

• Basic Analog Telephony
• Basic Digital Telephony
• Voice Coding and Compression Techniques
• Voice Transport and Delay
• Supplemental Slides: Digital Voice Signaling Techniques

Thank You!

• Q & A
• Please Fill Out Evaluation Forms
• THANK YOU!
Please Complete Your Evaluation Form

Session 401