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Introduction to Voice and Telephone Technology
Session 401
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Voice Is Not A Network

- Voice is an <u>Application</u>

- 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

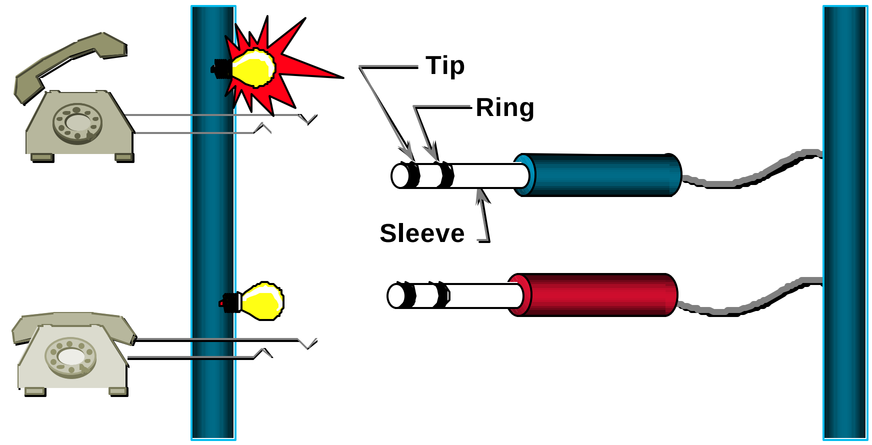
Agenda

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

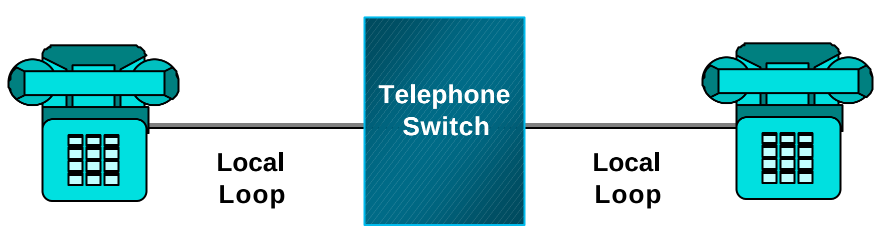
Telephony Equipment

- 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

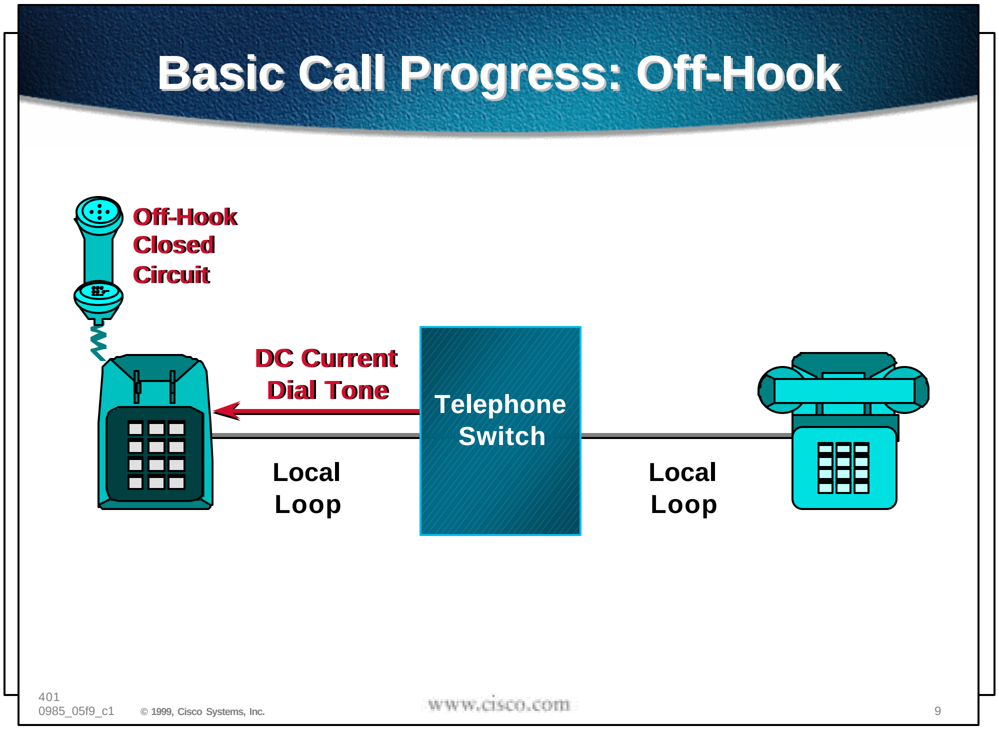
Analog Telephony—Connection Basics



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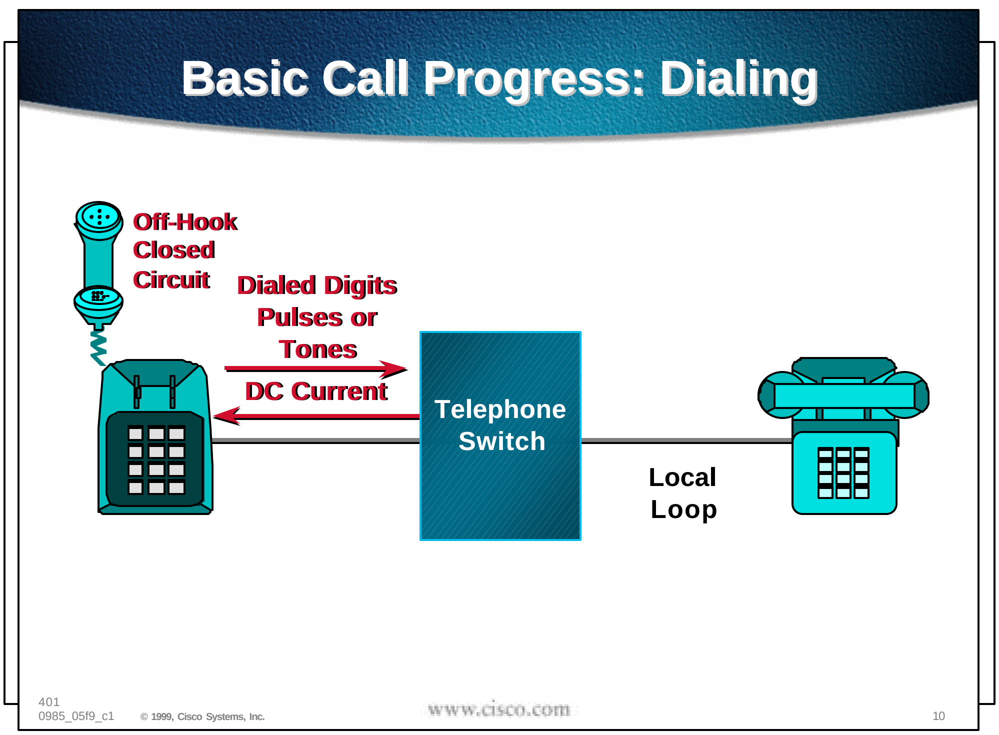
Basic Call Progress: Dialing
Off-Hook
Closed
Circuit
Dialed Digits
Pulses or
Tones
DC Current
Telephone
Switch
Local
Loop
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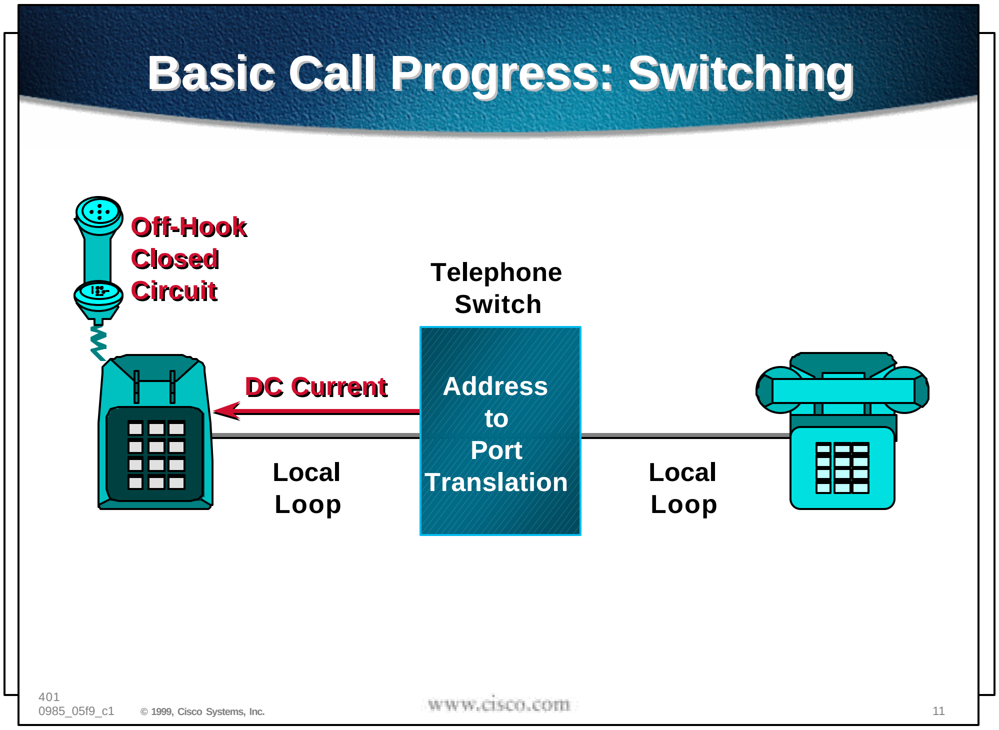

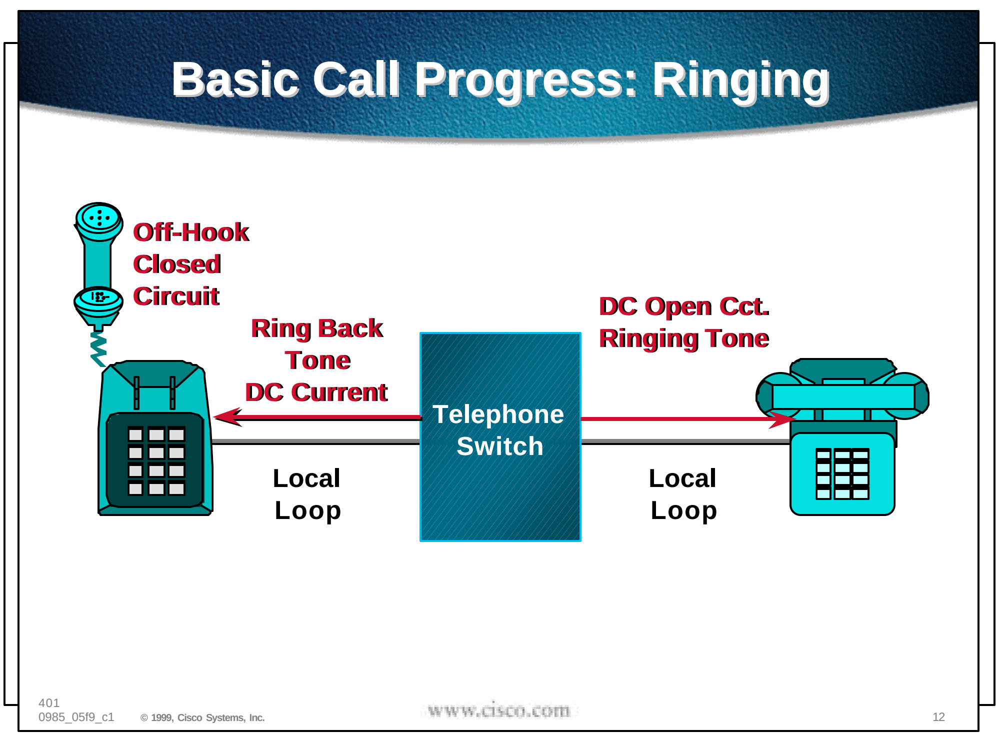

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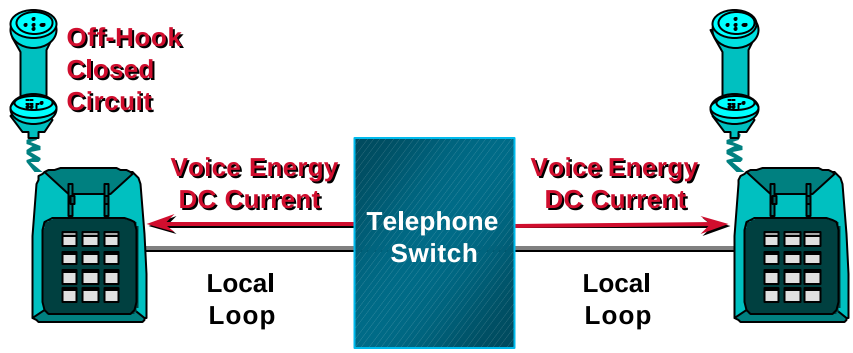
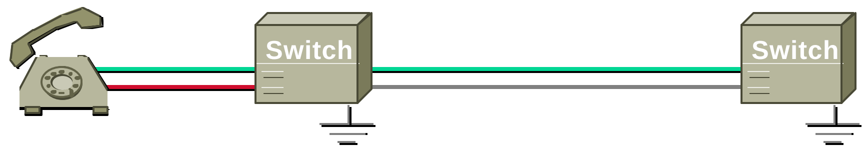
Basic Call Progress: Talking



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

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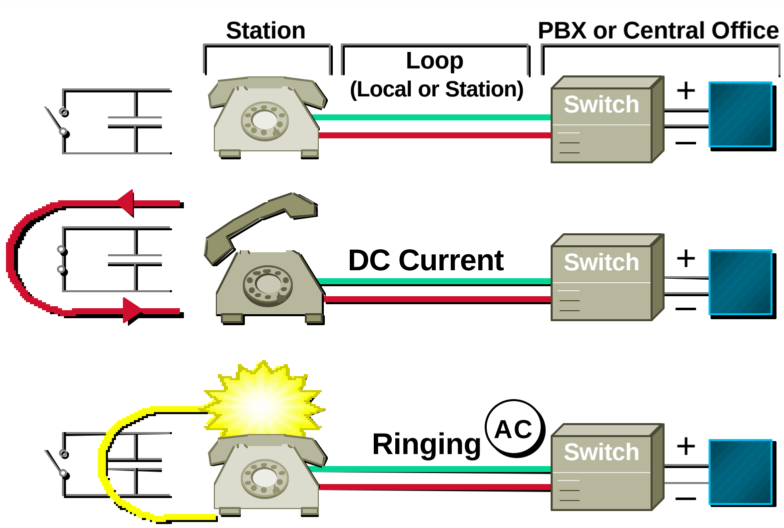
Loop Start



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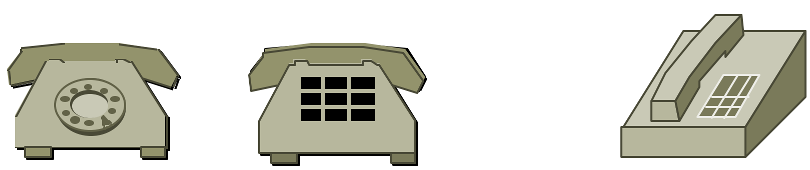
E&M Signaling

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

State	E-Lead	M-Lead
On-Hook	Open	Ground
Off-Hook	Ground	Battery Voltage

Signaling and Addressing



Dial Pulse **DTMF** **ISDN**

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

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

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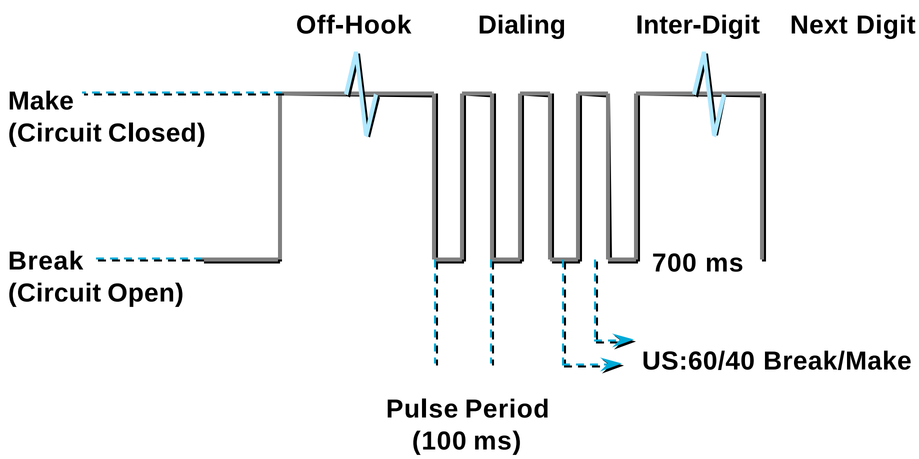
Pulse Dialing

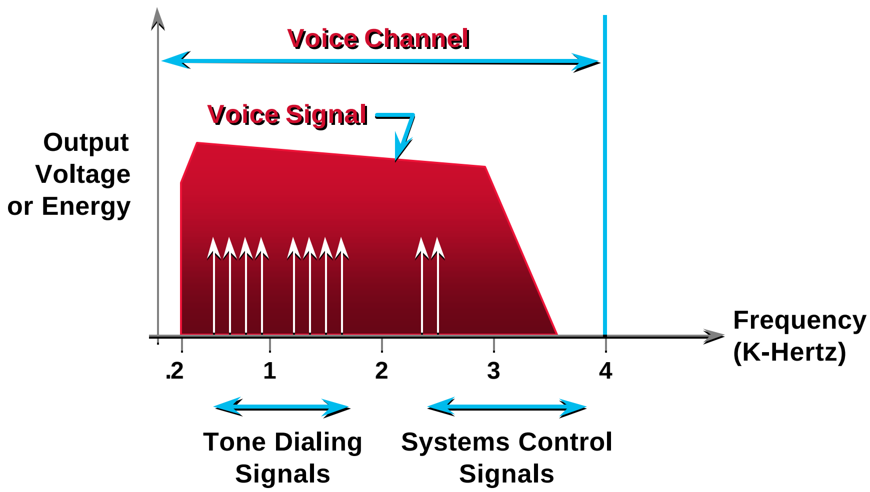


Tone Dialing

Dual Tone Multifrequency (DTMF)

	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Voice Channel Bandwidth



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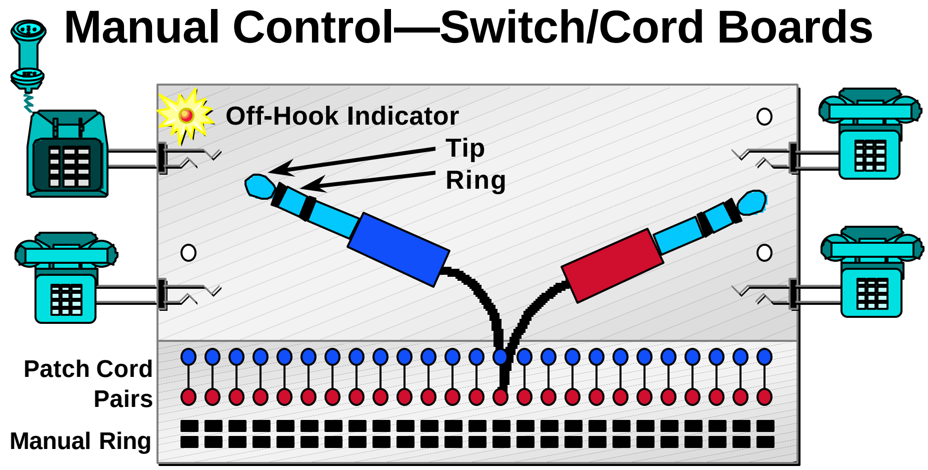
Switching Systems

Manual Control—Switch/Cord Boards



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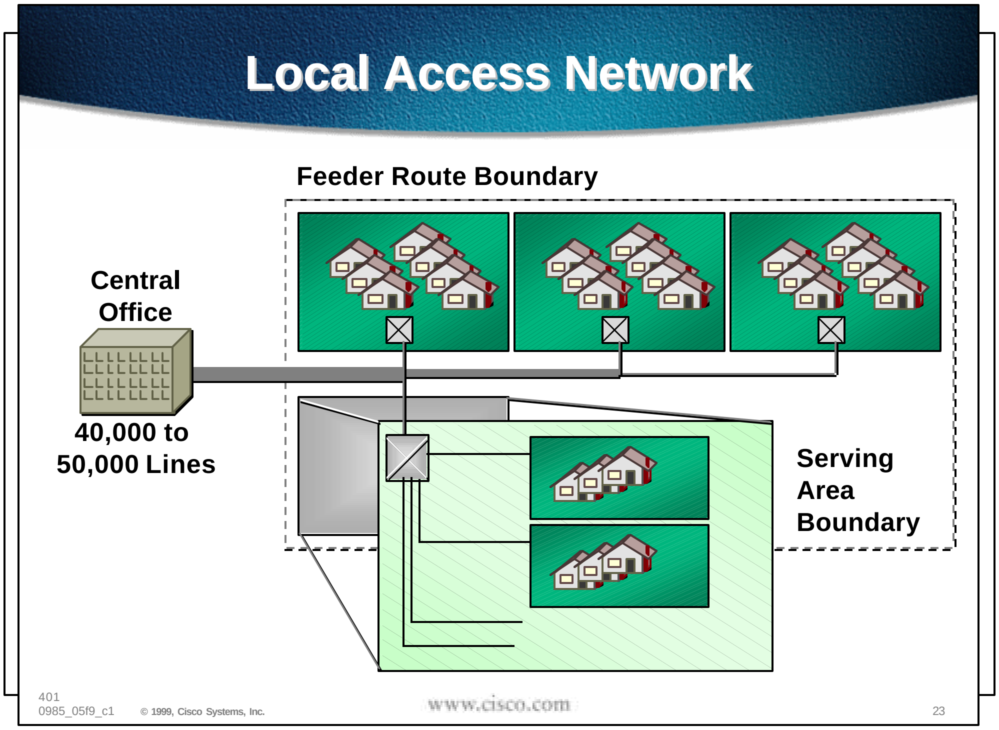
Local Access Network
Feeder Route Boundary
Central Office
40,000 to 50,000 Lines
Serving Area Boundary
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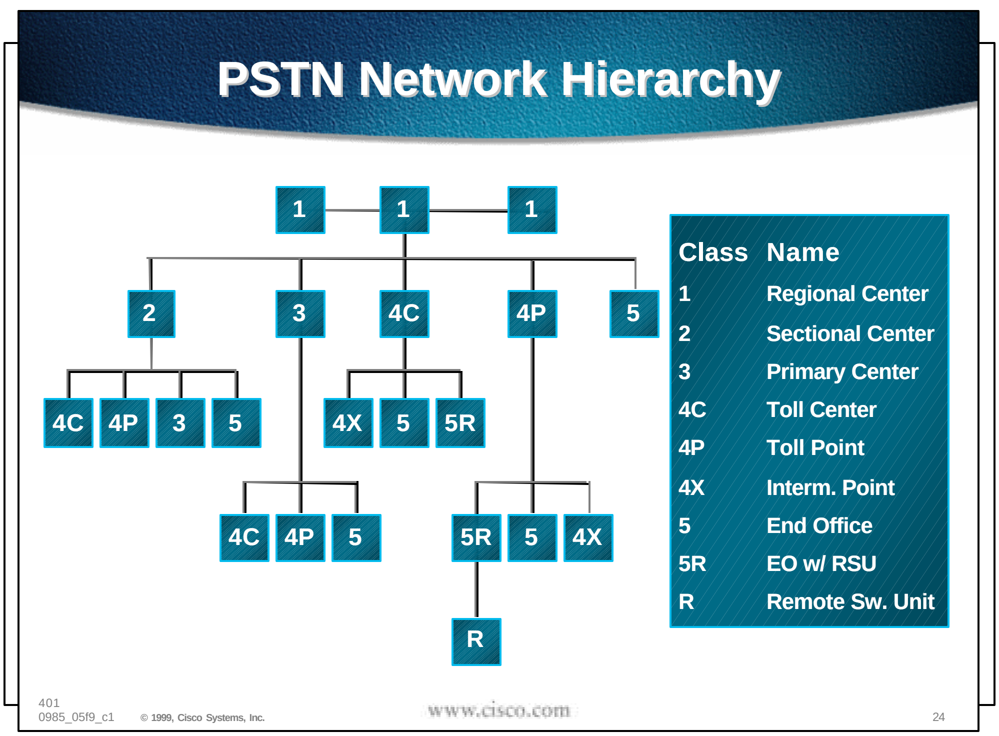
PSTN Network Hierarchy
1
1
1
2
3
4C
4P
5
4C
4P
3
5
4X
5
5R
4C
4P
5
5R
5
4X
R
Class Name
1 Regional Center
2 Sectional Center
3 Primary Center
4C Toll Center
4P Toll Point
4X Interm. Point
5 End Office
5R EO w/ RSU
R Remote Sw. Unit
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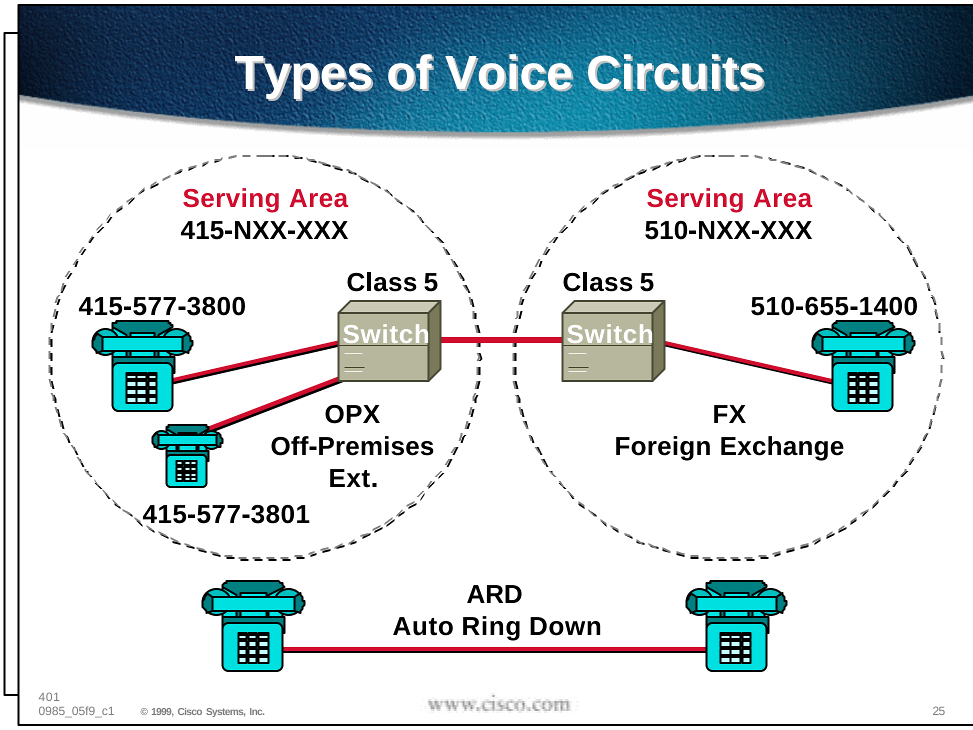
Types of Voice Circuits
Serving Area
415-NXX-XXX
Serving Area
510-NXX-XXX
415-577-3800
Class 5
Switch
Class 5
Switch
510-655-1400
OPX
Off-Premises
Ext.
FX
Foreign Exchange
415-577-3801
ARD
Auto Ring Down
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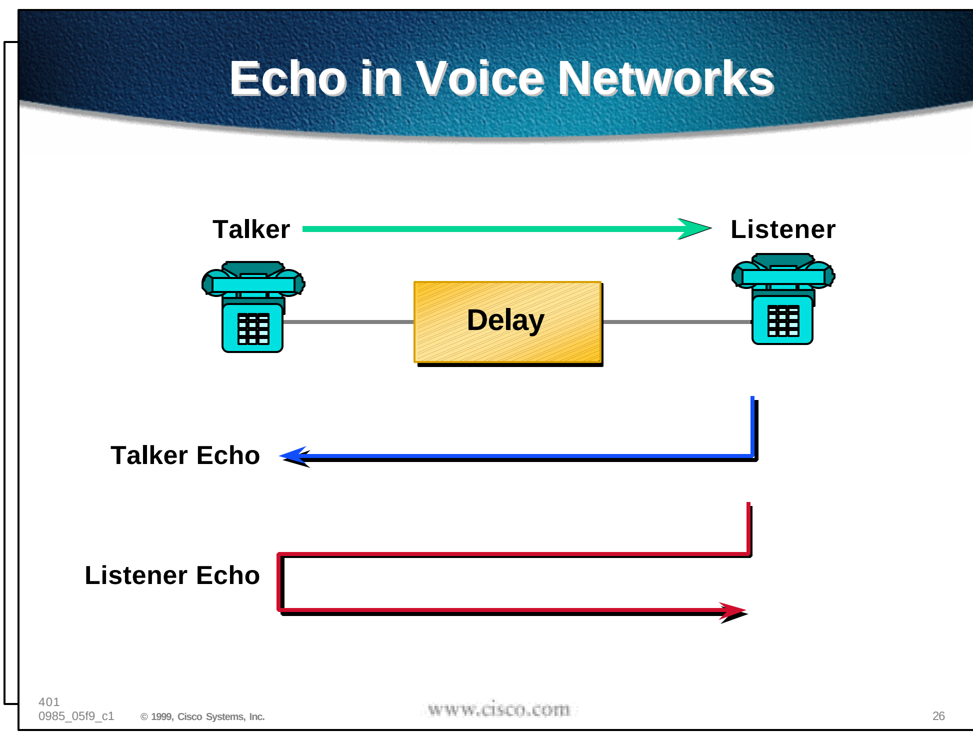
Echo in Voice Networks
Talker
Listener
Delay
Talker Echo
Listener Echo
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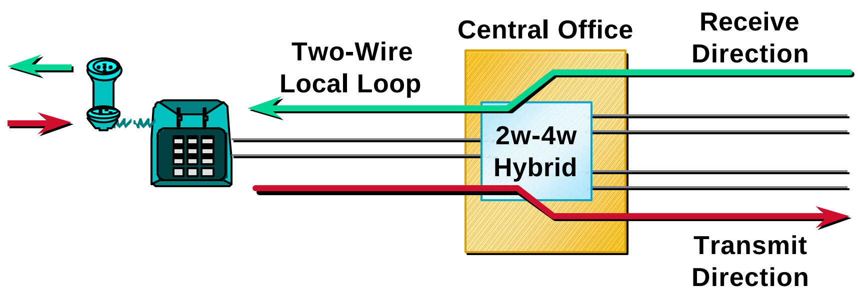
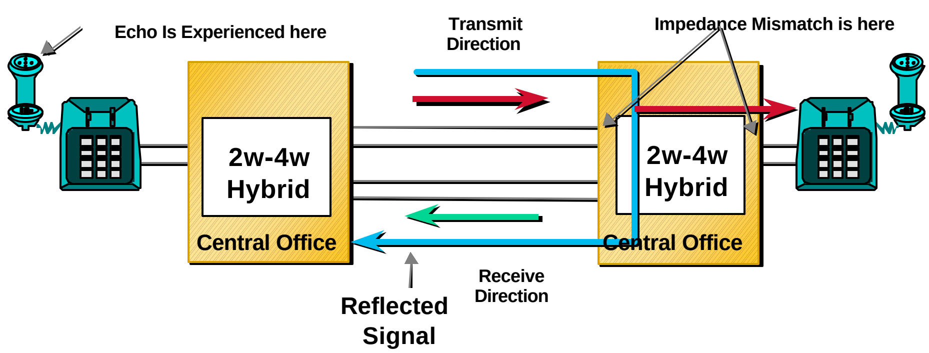
Normal Signal Flow



- 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?

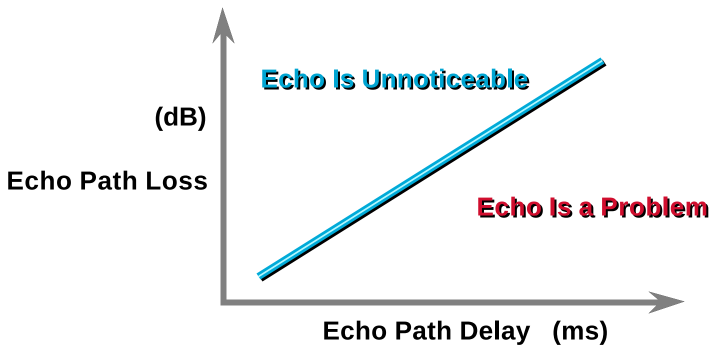
Echo Is Due to a Reflection



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



Ways to Defeat Echo

- **Increase the loss in the echo path**

  Can often be the solution

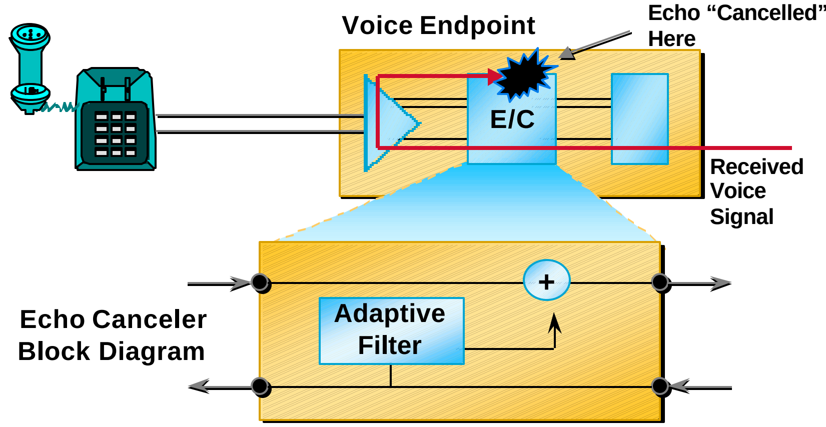
  Disadvantage: static setting and reduces the signal strength of the speaker

- **Echo suppresser**

  Acts like a noise gate, effectively making communications half-duplex

Echo Canceler

Most Effective Means for Removing Echo



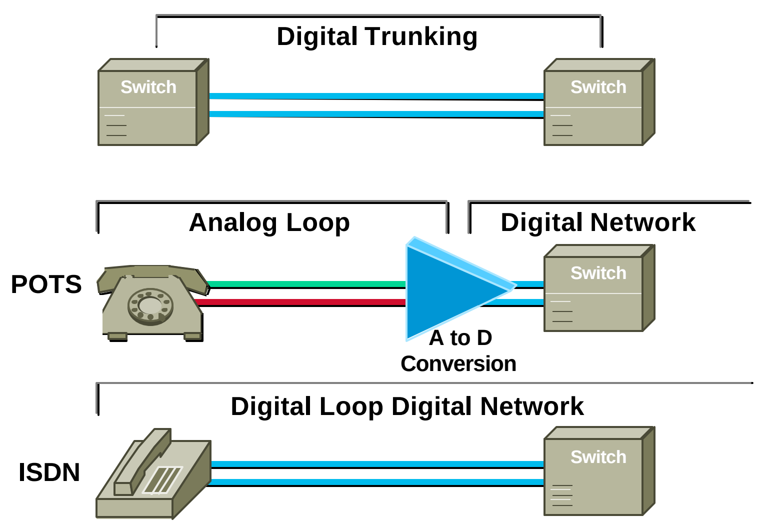
Summary

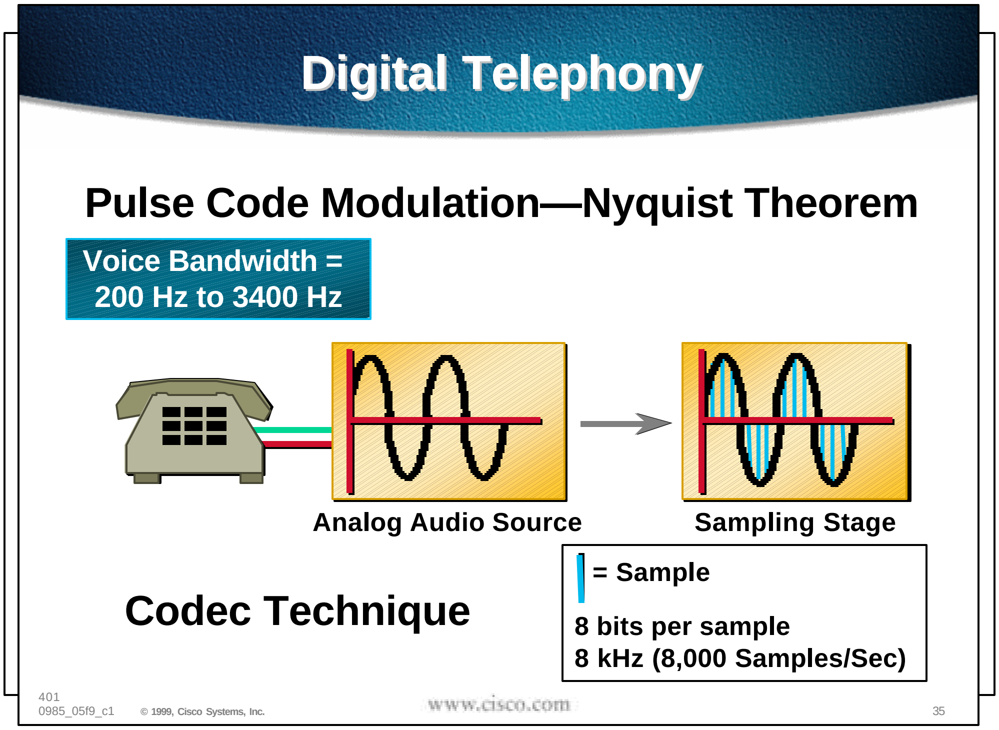
- Information exchange based on voltage, current flow, grounding, and so on
- Analog voice technology dates back to the late 1800s

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 Telephony
Pulse Code Modulation—Nyquist Theorem
Voice Bandwidth = 200 Hz to 3400 Hz
Analog Audio Source
Sampling Stage
= Sample
8 bits per sample
8 kHz (8,000 Samples/Sec)
Codec Technique
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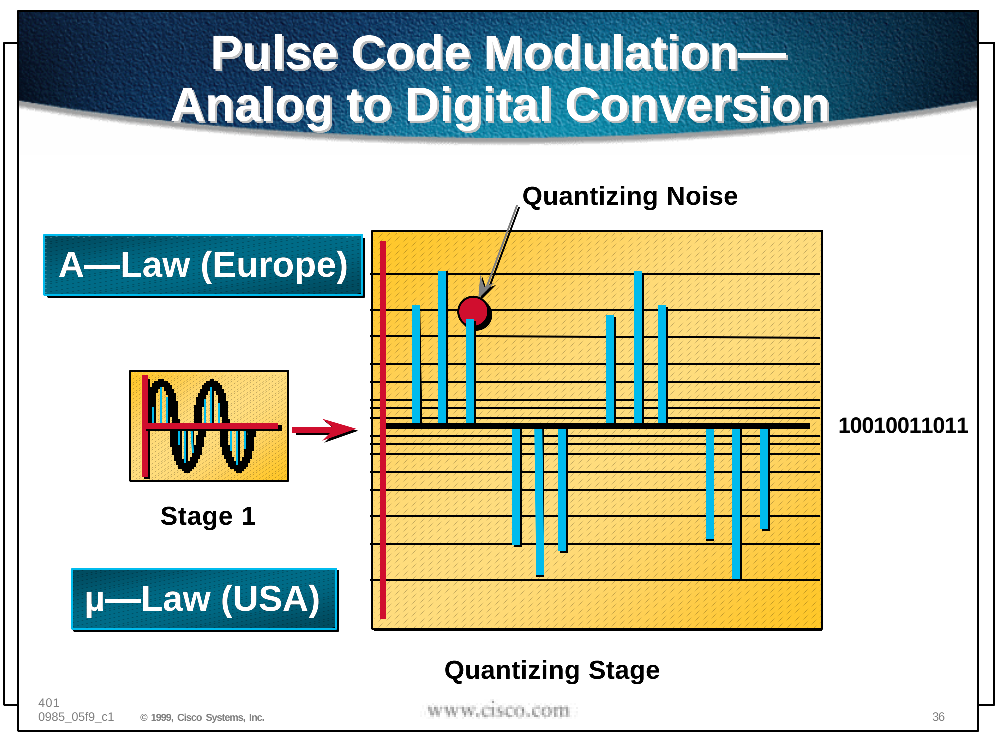
Pulse Code Modulation—Analog to Digital Conversion
Quantizing Noise
A—Law (Europe)
10010011011
Stage 1
μ—Law (USA)
Quantizing Stage
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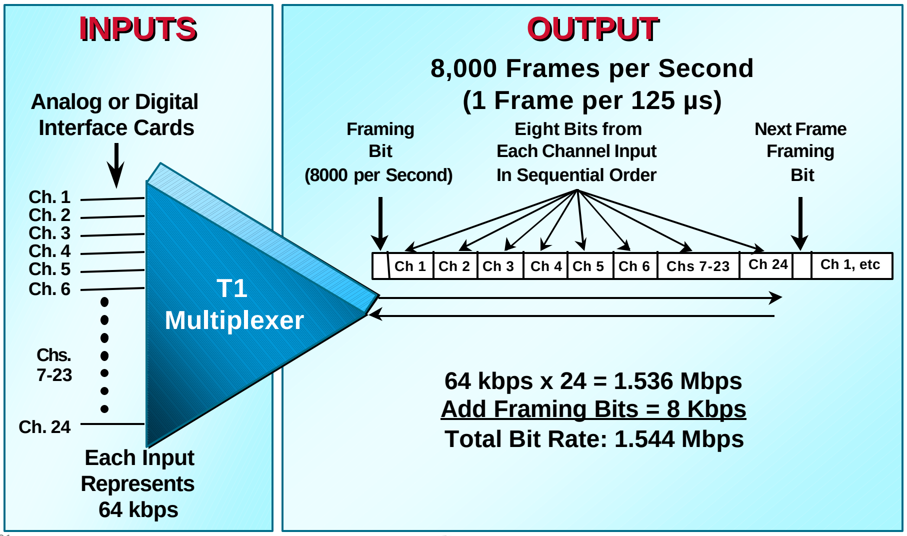
Time Division Multiplexer Example: T1 Channel Bank



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

	Framing Bits	Bit Use in Each Channel Time Slot		Signaling—Bit Use Options		
Frame Number	Framing Bit Value	Traffic	Signaling	T	2	4
1	1					
2	0					
3	0					
4	0					
5	1					
6	1	Bits 1–7	Bit 8	*	A	A
7	0					
8	1					
9	1					
10	1					
11	0					
12	0	Bits 1–7	Bit 8	*	A	B

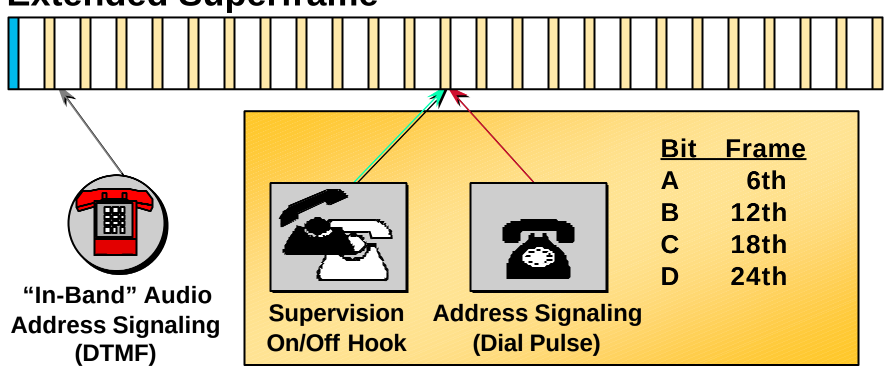
Extended Superframe (ESF)

Frame Number	S Bits: Fe	S Bits: DL	S Bits: BC	Bit Use in Each Channel Time Slot: Traffic	Bit Use in Each Channel Time Slot: Signaling	Signaling—Bit Use Options: T	2	4	16
1	–	m	–						
2	–	–	C1						
3	–	m	–						
4	0	–	–						
5	–	m	–						
6	–	–	C2	Bits 1–7	Bit 8	*	A	A	A
7	–	m	–						
8	0	–	–						
9	–	m	–						
10	–	–	C3						
11	–	m	–						
12	1	–	–	Bits 1–7	Bit 8	*	A	B	B
13	–	m	–						
14	–	–	C4						
15	–	m	–						
16	0	–	–						
17	–	m	–						
18	–	–	C5	Bits 1–7	Bit 8	*	A	A	C
19	–	m	–						
20	1	–	–						
21	–	m	–						
22	–	–	C6						
23	–	m	–						
24	1	–	–	Bits 1–7	Bit 8	*	A	B	D

Digital Signaling Schemes

Channel Associated Signaling

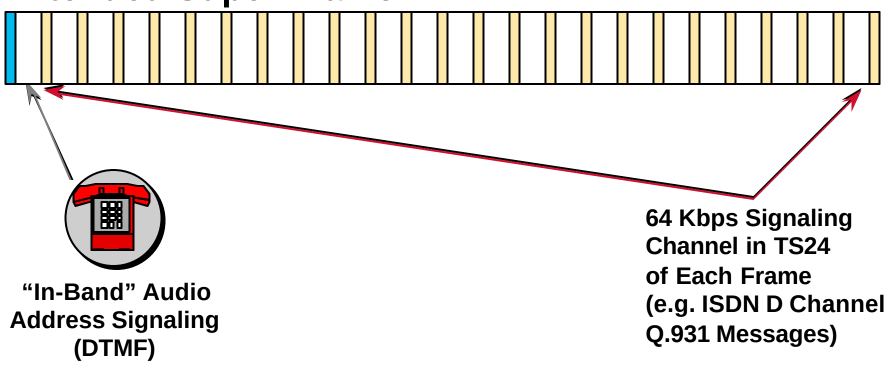
Extended Superframe



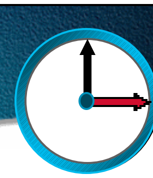
Digital Signaling Schemes

Common Channel Signaling

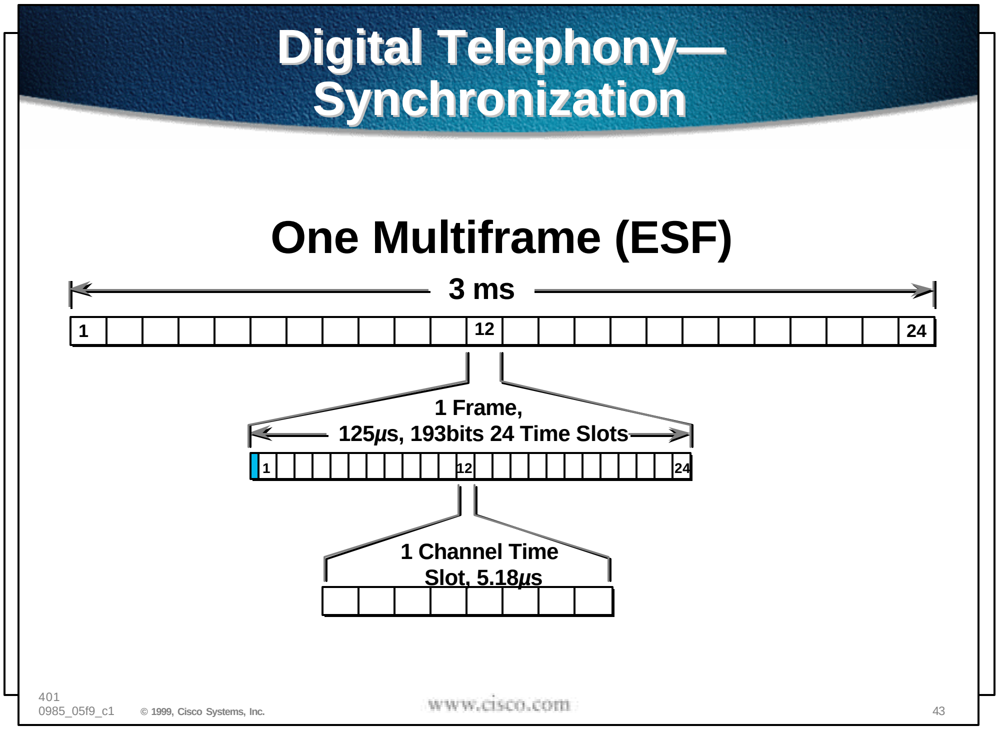
Extended Super Frame

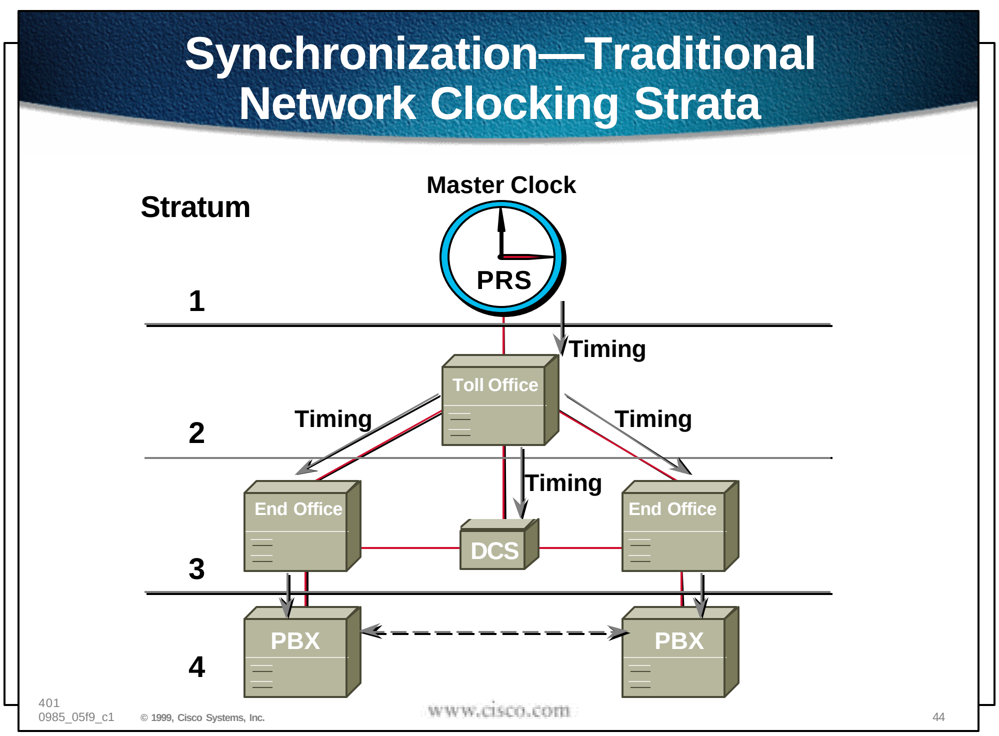


Digital Telephony—Synchronization



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





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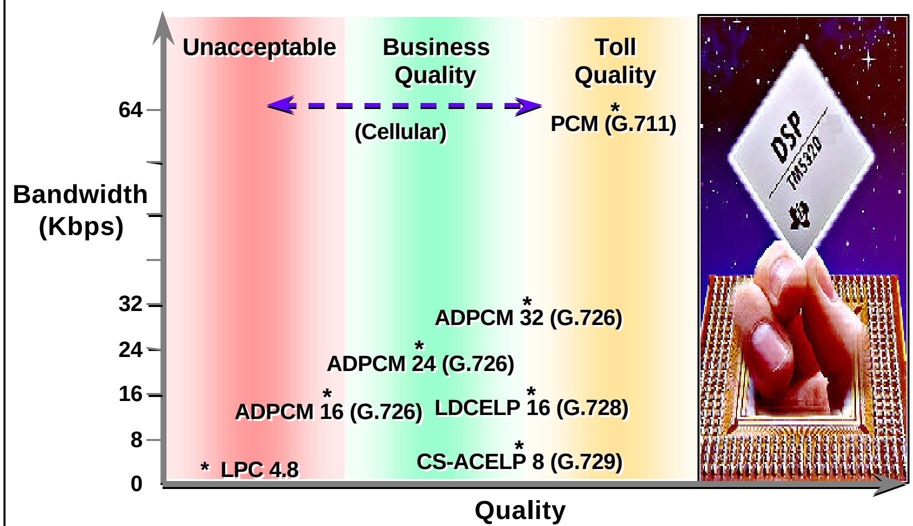
Agenda

- 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



Speech-Coding Schemes

- **Waveform coders**

  Non-linear approximation of the actual waveform

  Examples: PCM, ADPCM

- **Vocoders**

  Synthesized voice

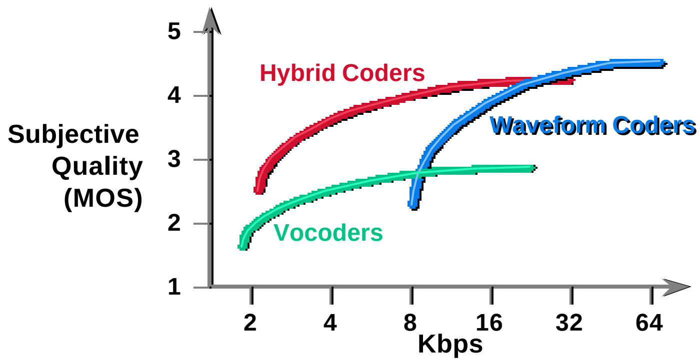
  Example: LPC

- **Hybrid coders**

  Linear waveform approximation with synthesized voice

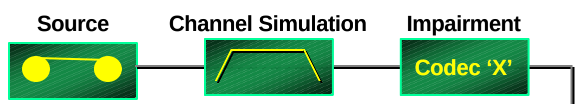
  Example: CELP

Subjective Impairment Analysis: Mean Opinion Scores



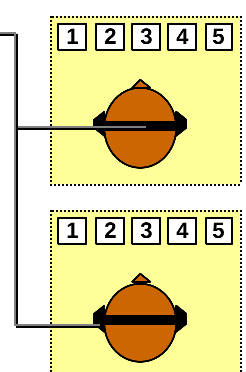
Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just Perceptible, not Annoying
3	Fair	Perceptible and Slightly Annoying
2	Poor	Annoying but not Objectionable
1	Bad	Very Annoying and Objectionable

Measuring Mean Opinion Scores: ITU P.800 Series



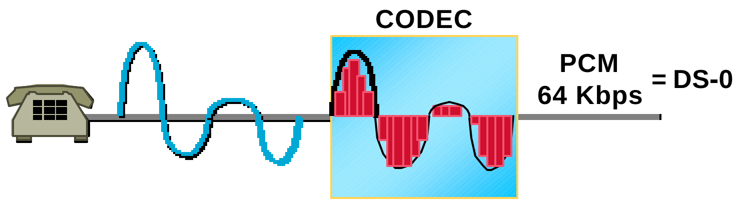
"Nowadays, a chicken leg is a rare dish"

Rating	Level of Speech Quality	Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying and objectionable

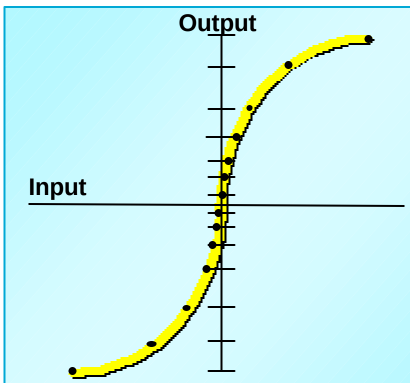


Digitizing Voice: PCM Waveform Encoding Review

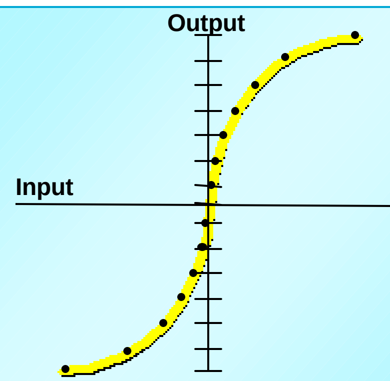
- **Nyquist Theorem: sample at twice the highest frequency**
  - Voice frequency range: 200-3400 Hz
  - Sampling frequency = 8000/sec (every 125µs)
  - Bit rate: (2 x 4 kHz) x 8 bits per sample = 64,000 bits per second (DS-0)
- **By far the most commonly used method**



Nonlinear vs. Linear Encoding

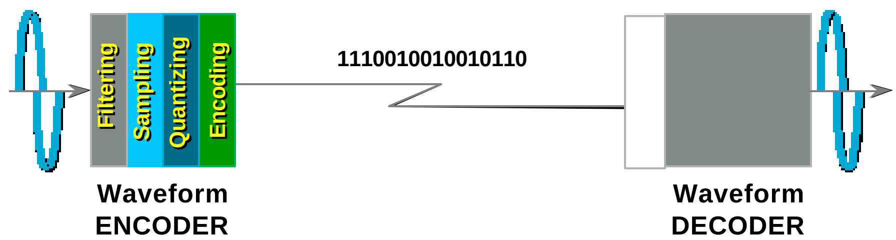


Nonlinear Encoding
Closely Follows Human Voice Characteristics. High Amplitude Signals have More Quantization Distortion.



Linear Encoding
Relatively Easy to Analyze, Synthesize and Regenerate. All Amplitudes Have Roughly Equal Quantization Distortion.

Voice CODECs: Waveform Coders



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

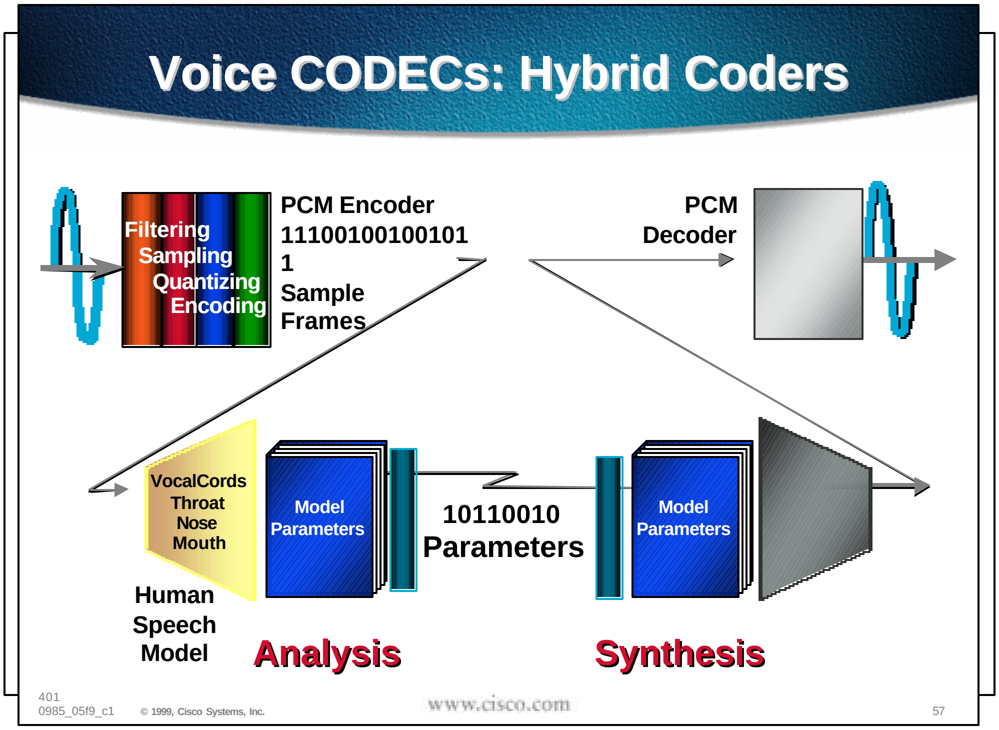
    24 Kbps = 8 Kbps x 3 bits/sample

    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
Filtering
Sampling
Quantizing
Encoding
PCM Encoder
111001001001011
Sample
Frames
PCM
Decoder
VocalCords
Throat
Nose
Mouth
Human
Speech
Model
Model
Parameters
10110010
Parameters
Model
Parameters
Analysis
Synthesis
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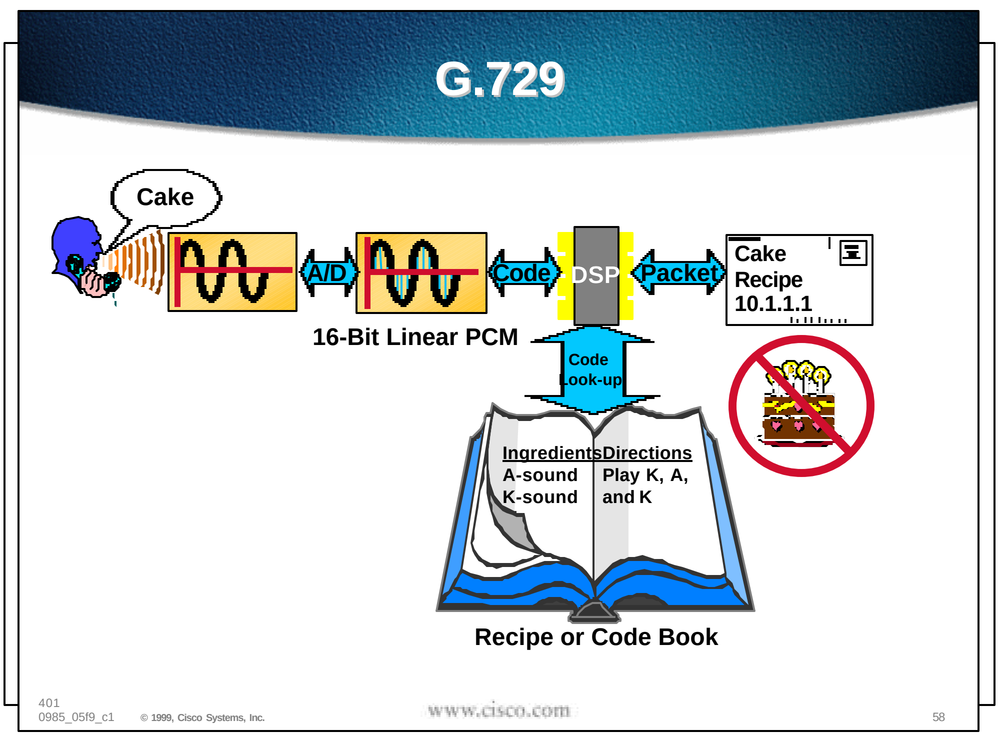
G.729
Cake
A/D
Code
DSP
Packet
Cake
Recipe
10.1.1.1
16-Bit Linear PCM
Code
Look-up
Ingredients
Directions
A-sound
K-sound
Play K, A,
and K
Recipe or Code Book
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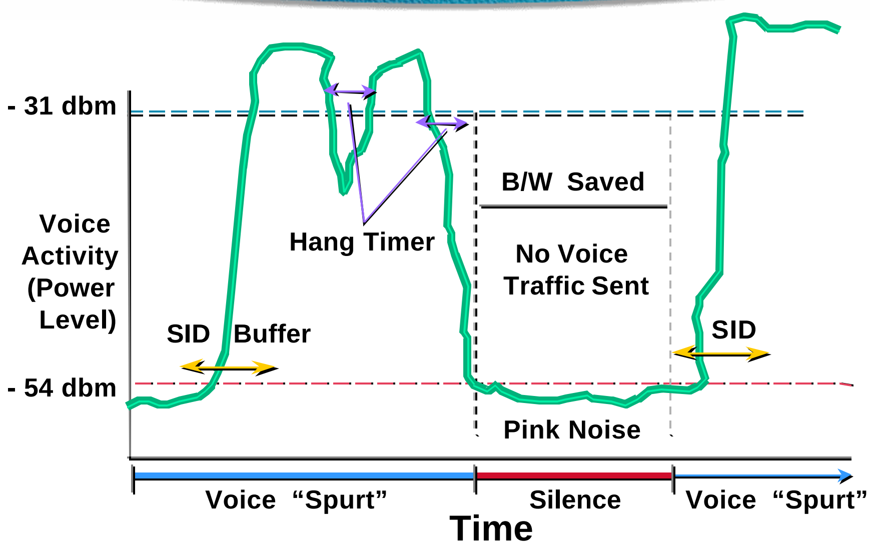
Digital Speech Interpolation (DSI)

- 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



Bandwidth Requirements

Voice Band Traffic

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

Agenda

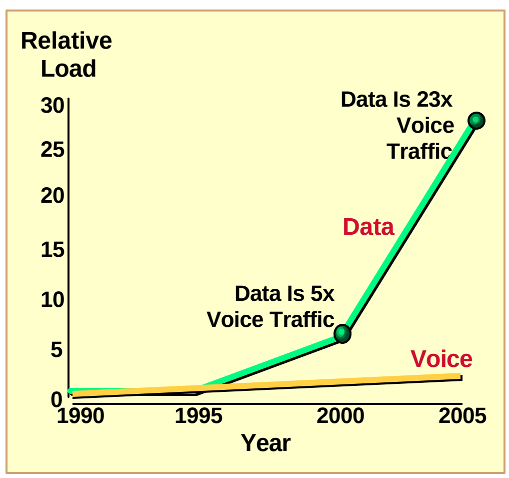
- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques

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

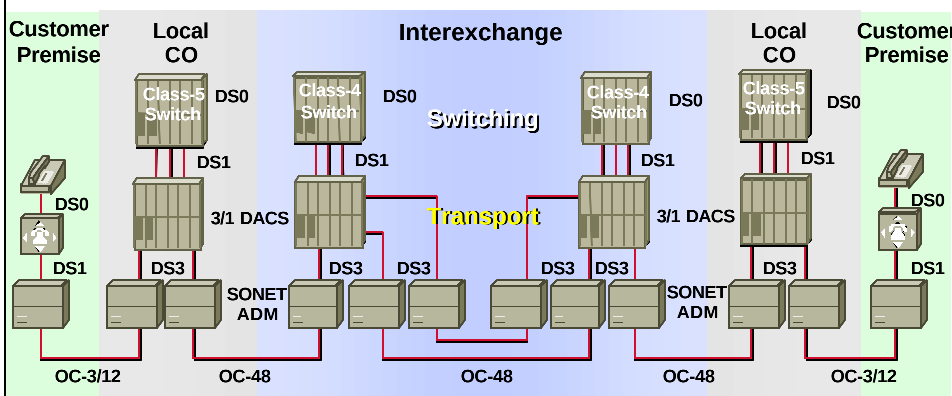
Evolution from TDM-based transport to packets/cells or a combination



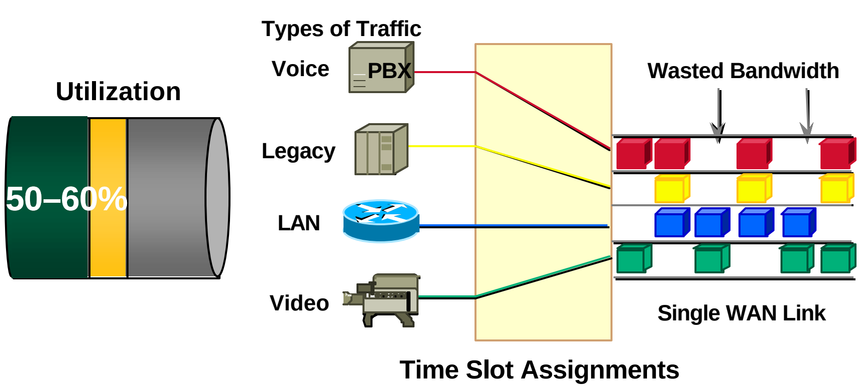
Source: Electronicast

The Tyranny of the DS0

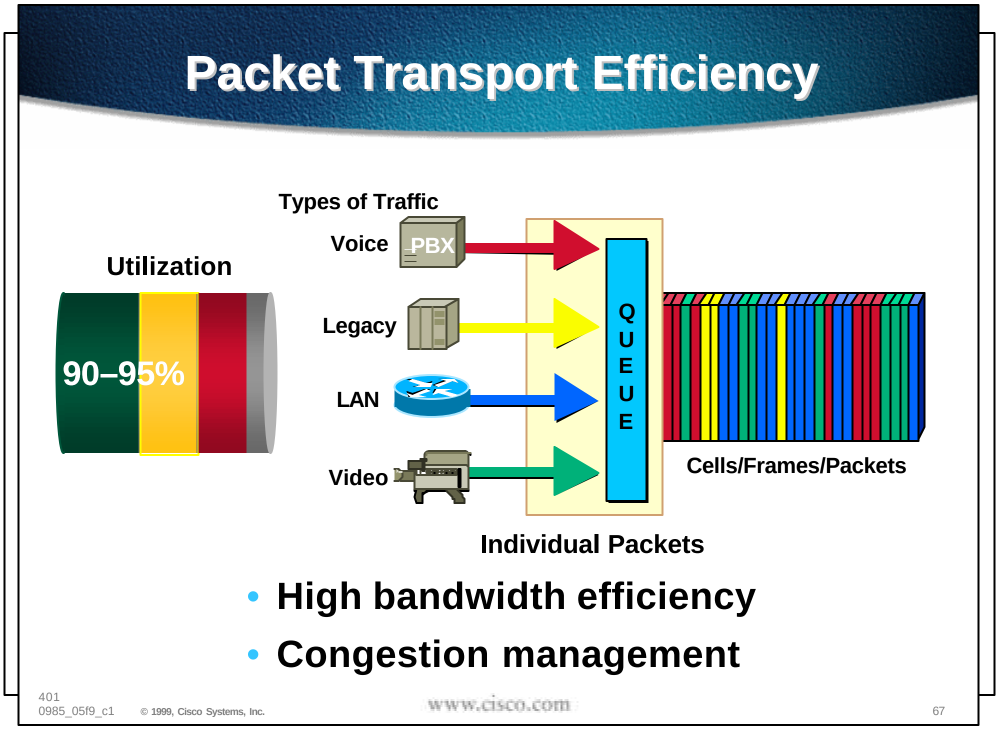
- Switching and transport based on circuits
- Rigid structure yields high cost for packet



TDM Transport Efficiency



- **Wasted bandwidth**
- **No congestion**


Packet Transport Efficiency
Types of Traffic
Voice
PBX
Legacy
LAN
Video
Utilization
90–95%
QUEUE
Cells/Frames/Packets
Individual Packets
• High bandwidth efficiency
• Congestion management
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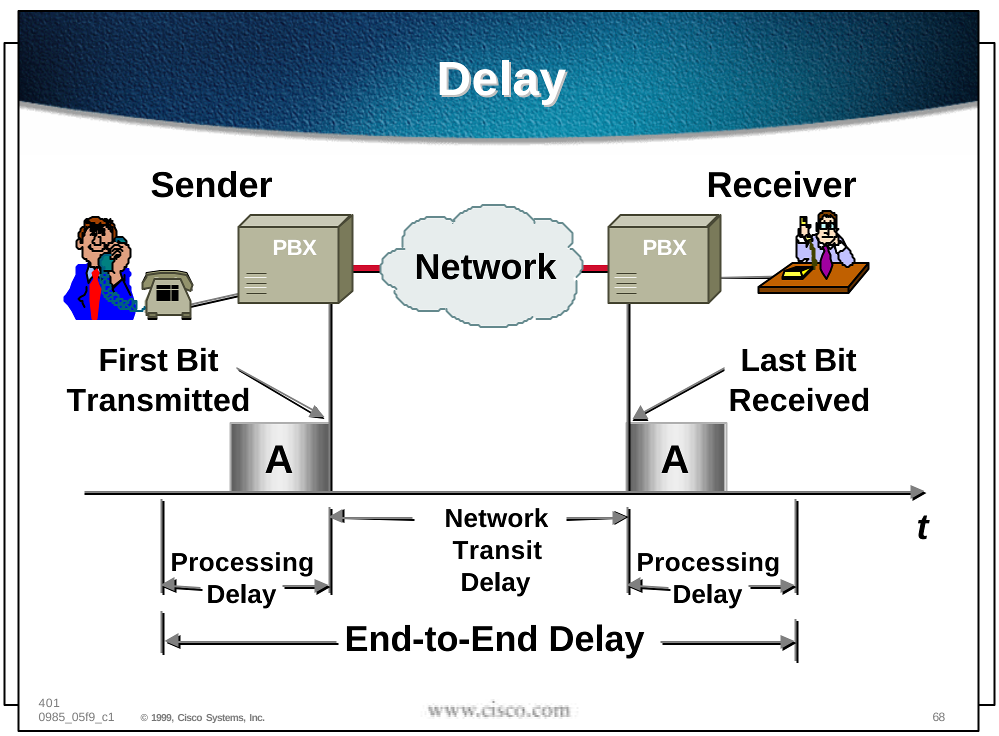
Delay
Sender
Receiver
PBX
Network
PBX
First Bit
Transmitted
Last Bit
Received
A
A
t
Network
Transit
Delay
Processing
Delay
Processing
Delay
End-to-End Delay
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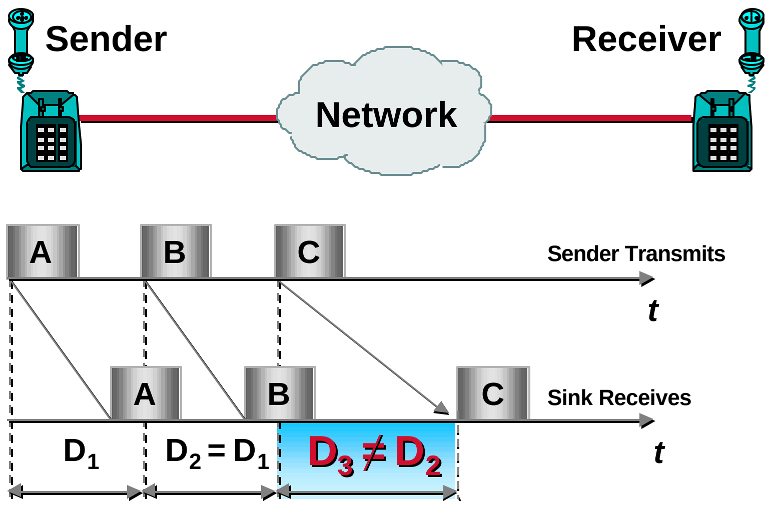
Delay Variation—"Jitter"



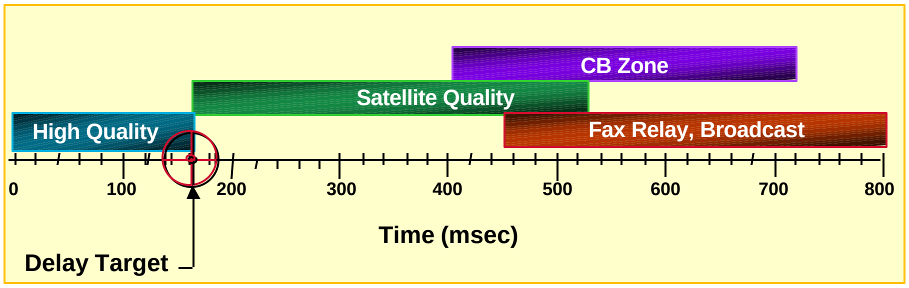
Voice Delay Guidelines

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

ITU's G.114 Recommendation

Delay in Perspective

Cumulative Transmission Path Delay



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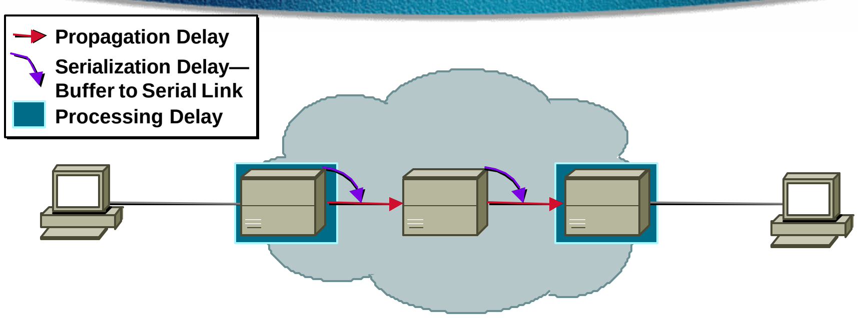
Fixed Delay Components



- **Propagation—Six microseconds per kilometer**
- **Serialization**
- **Processing**

  Coding/compression/decompression/decoding

  Packetization

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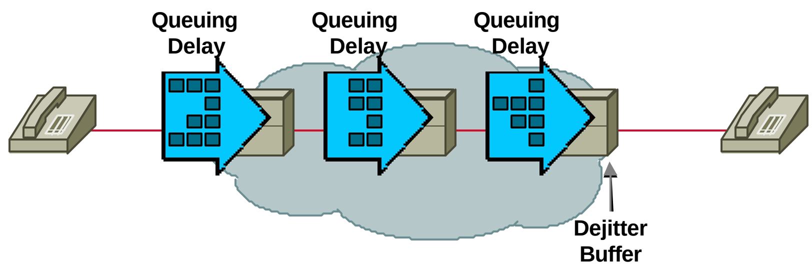
Variable Delay Components



- Queuing delay
- Dejitter buffers
- Variable packet sizes

An Example

- Assumptions:

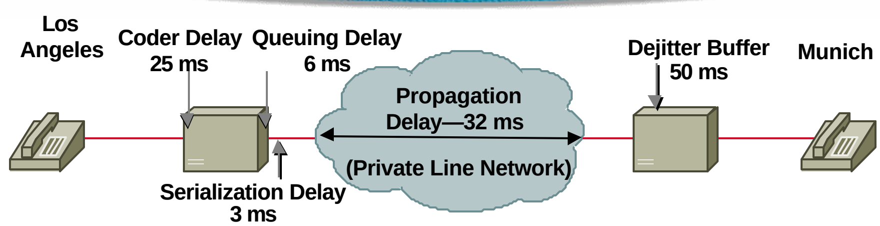
  We have eight trunks

  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



	Fixed Delay	Variable 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	110 msec	

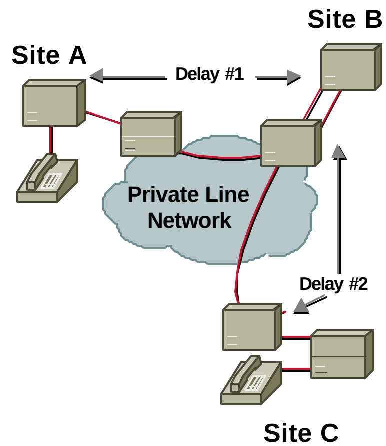
Variable Delay Component

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

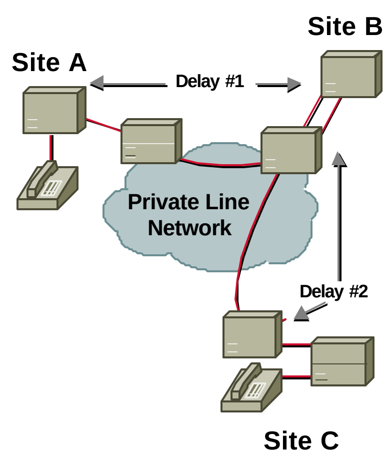
Delay Calculation

	Fixed Delay	Variable Delay
DELAY #1 Coder Delay G.729	25 msec	
Packetization Delay (Included in Coder Delay)		
Max Queuing Delay 64 kbps Trunk		21msec
Serialization Delay 64 kbps Trunk	3 msec	
Propagation Delay (Private Lines)	32 msec	
Dejitter Buffer	50 msec	
Tandem Switch	—	
Delay #1 Total	110 msec	



Delay Calculation

	Fixed Delay	Variable Delay
DELAY #1 Total	110 msec	
DELAY #2 Coder Delay G.729	25 msec	
Packetization Delay (Included in Coder Delay)		
Max Queuing Delay 2 Mbps Trunk		.7 msec
Serialization Delay 2 Mbps Trunk	0.1 msec	
Propagation Delay (Private Lines)	5 msec	
Dejitter Buffer	50 msec	
Delay #2 Total	80 msec	
Total Delay	190 msec	



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 <u>Application</u>
- 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

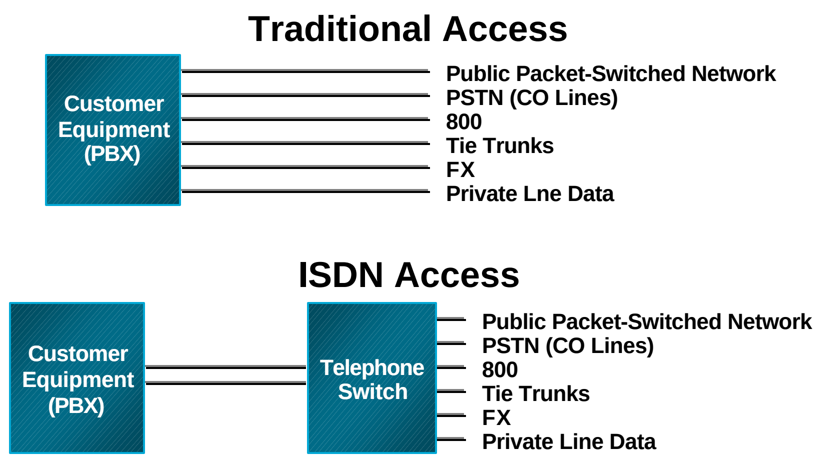
Digital Voice Signaling Techniques

- 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



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Terminology

- B channel “bearer channel”

  64 kbps

  Carries information (voice, data, video, etc.)

  DS-0

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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

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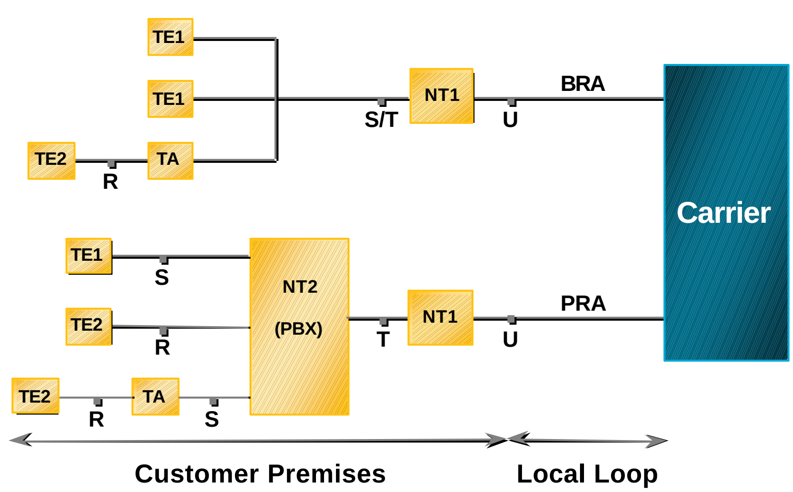
ISDN Reference Points



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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**

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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**

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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

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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

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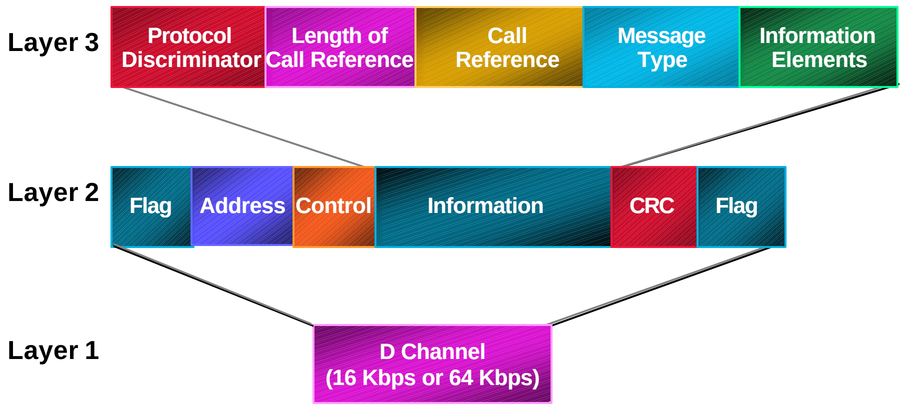
D-Channel Encapsulation



ISDN CCS (Q.930/931) Messages

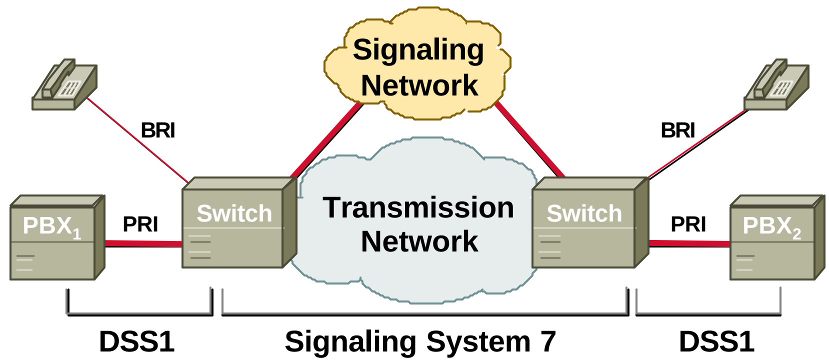
Call Establishment
- Alerting
- Call proceeding
- Connect
- Connect ack
- Progress
- Setup
- Setup ack

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

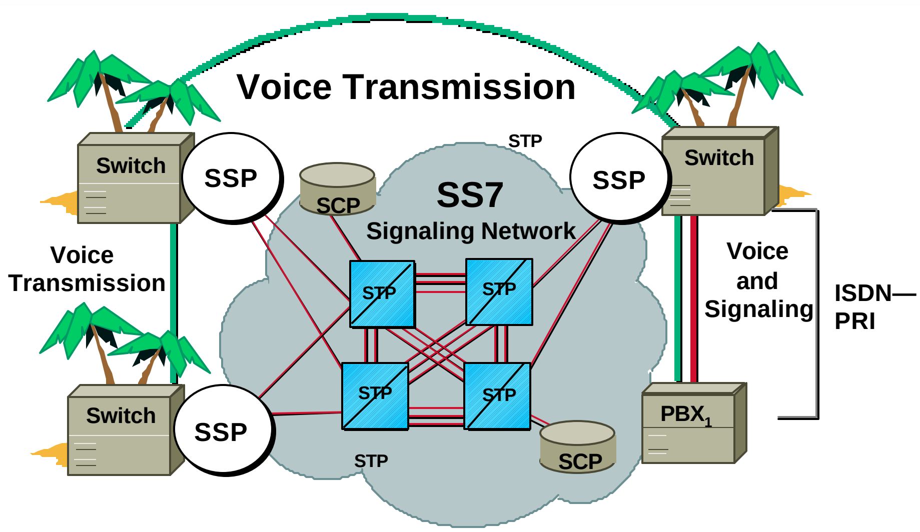
Miscellaneous
- Congestion control
- Facility
- Information
- Notify
- Register
- Status
- Status inquiry

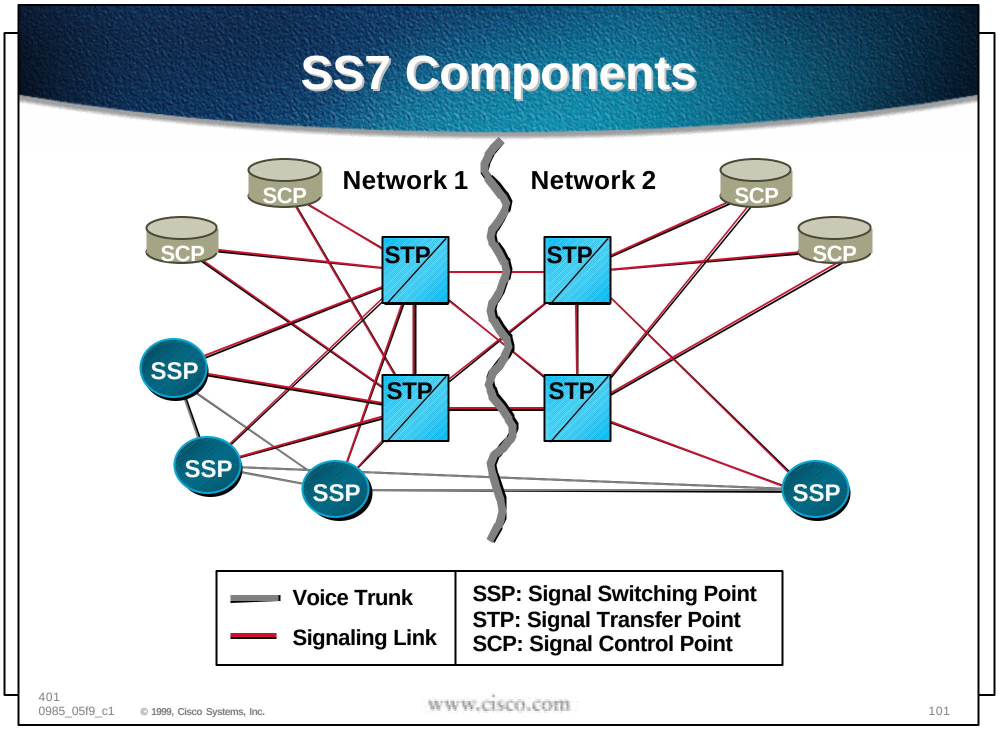
Public ISDN and Signaling System 7



DSS1 Is a Public ISDN Protocol

ISDN and SS7 "The Bridge Between the Islands"




SS7 Components
Network 1
Network 2
SCP
SCP
SCP
SCP
STP
STP
STP
STP
SSP
SSP
SSP
SSP
Voice Trunk
Signaling Link
SSP: Signal Switching Point
STP: Signal Transfer Point
SCP: Signal Control Point
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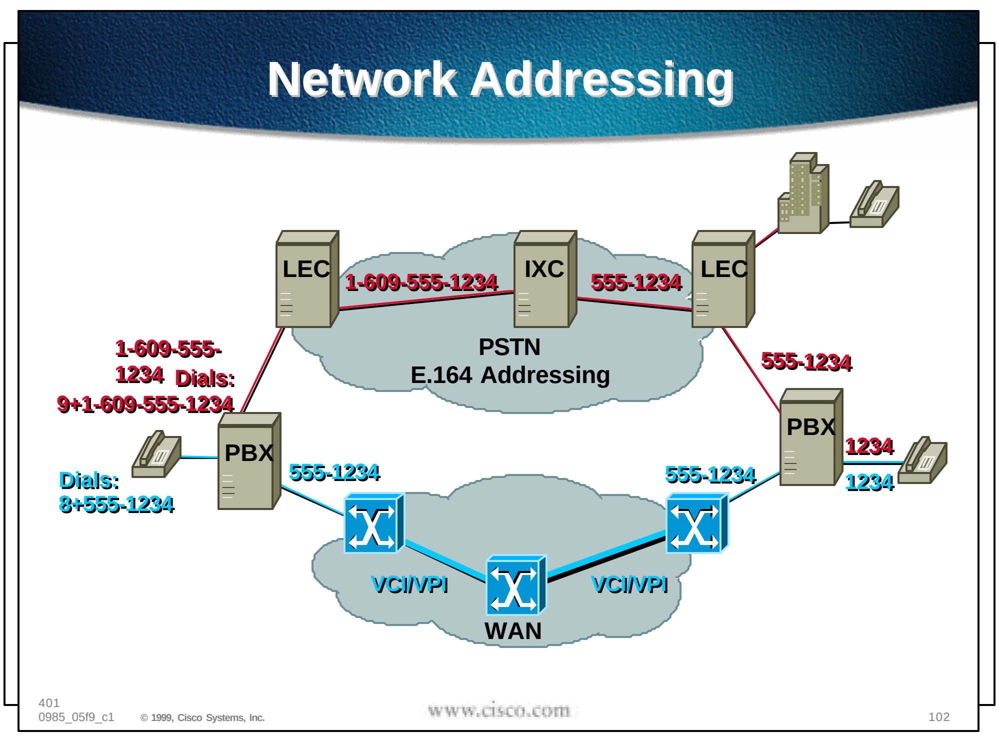
Network Addressing
LEC
IXC
LEC
1-609-555-1234
555-1234
PSTN
E.164 Addressing
1-609-555-1234 Dials:
9+1-609-555-1234
555-1234
PBX
PBX
Dials:
8+555-1234
555-1234
555-1234
1234
1234
VCI/VPI
VCI/VPI
WAN
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Agenda

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Thank You!

- Q & A
- Please Fill Out Evaluation Forms
- THANK YOU!

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