
NETWORKERS
2000
Cisco Systems
2001
1192_05_2000_c1
© 2000, Cisco Systems, Inc.
1


Introduction to Voice and Telephone Technology
Session 2001
Cisco Systems
2001
Presentation_ID
© 2000, Cisco Systems, Inc.
2

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

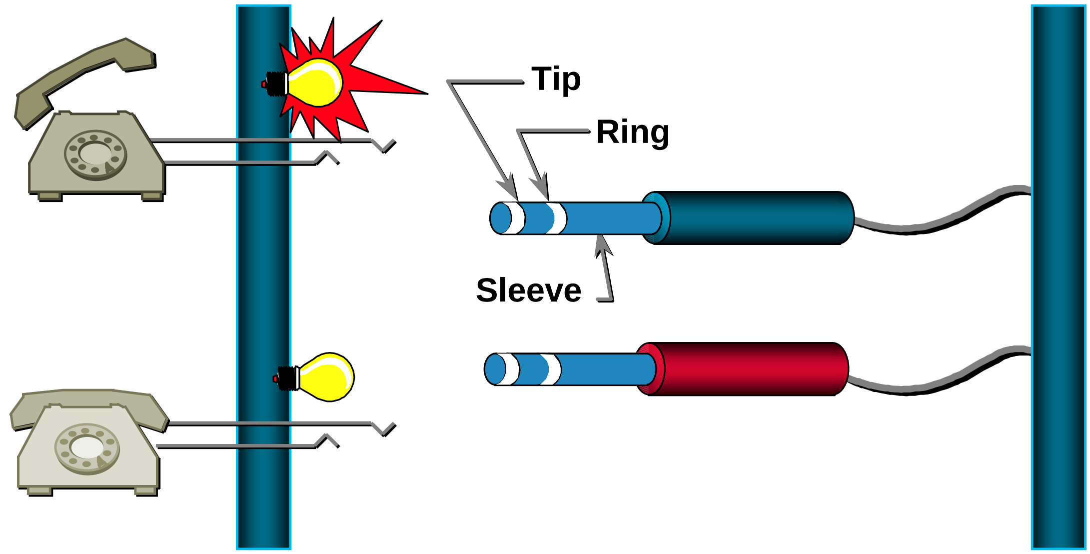
Agenda

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

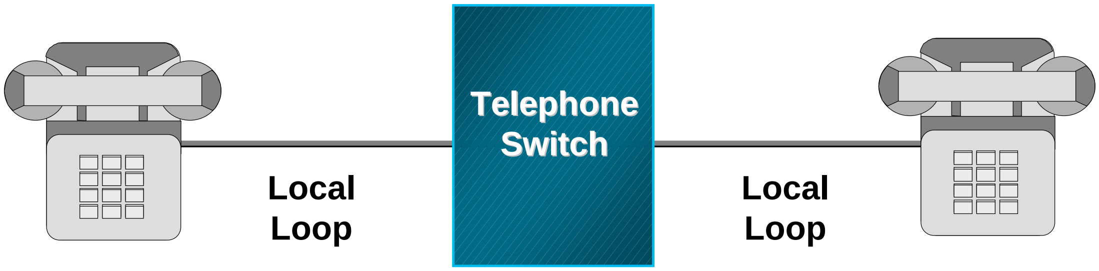
Telephony Equipment

- 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

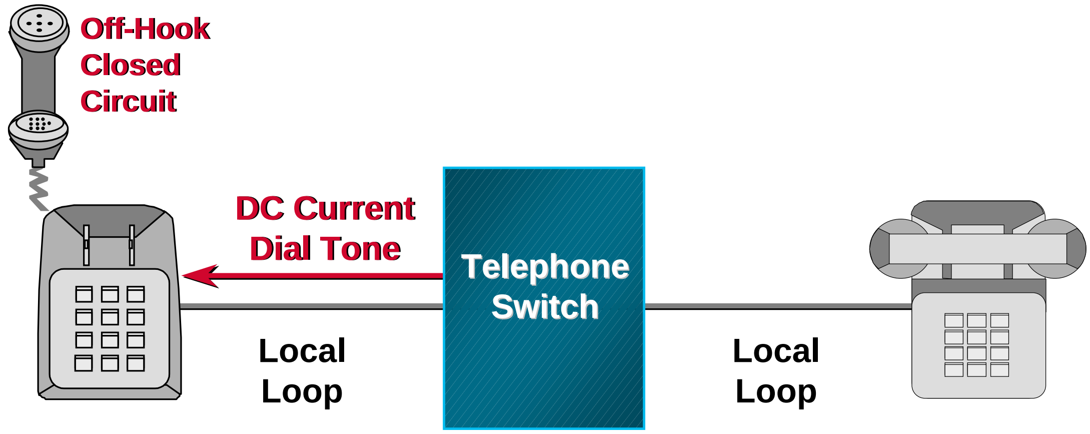


Basic Call Progress: On-Hook

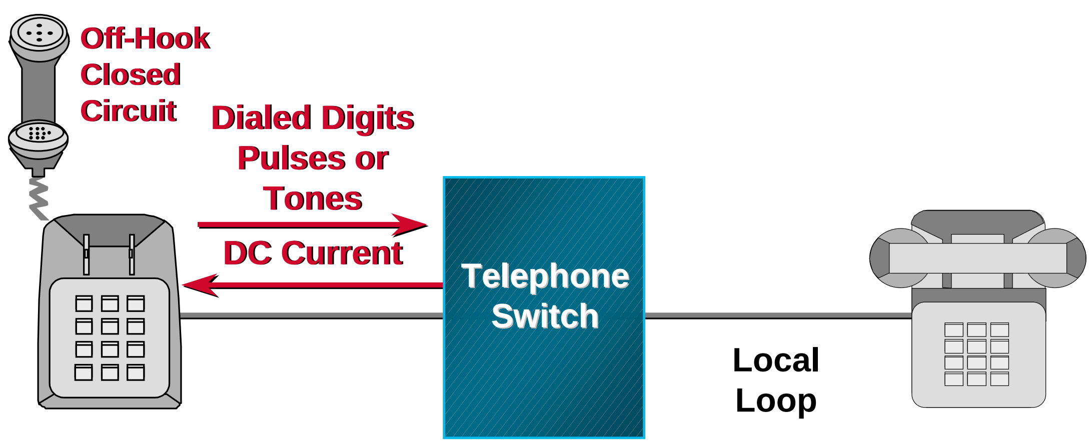


DC Voltage
Open Circuit
No Current Flow

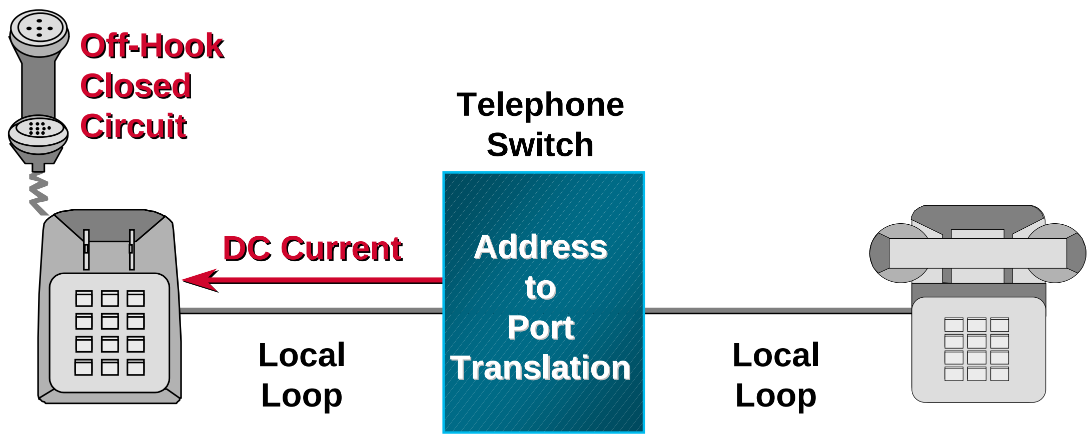
Basic Call Progress: Off-Hook



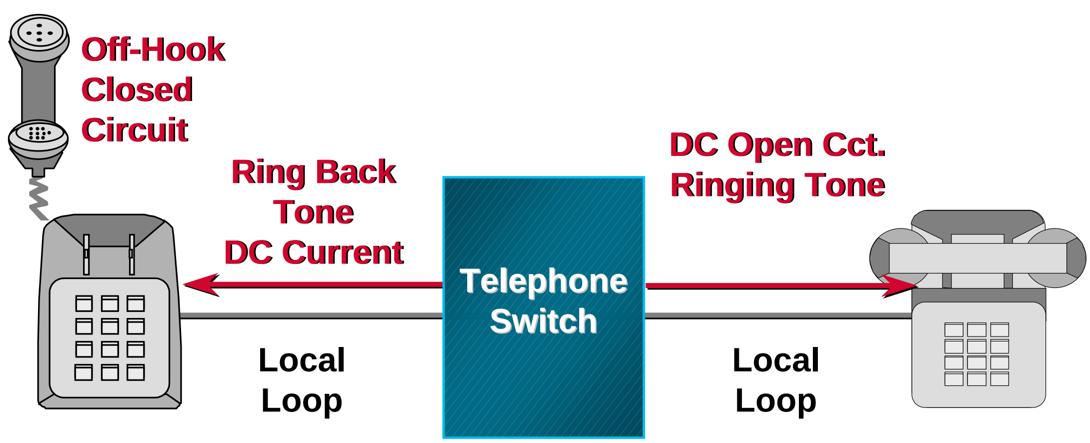
Basic Call Progress: Dialing



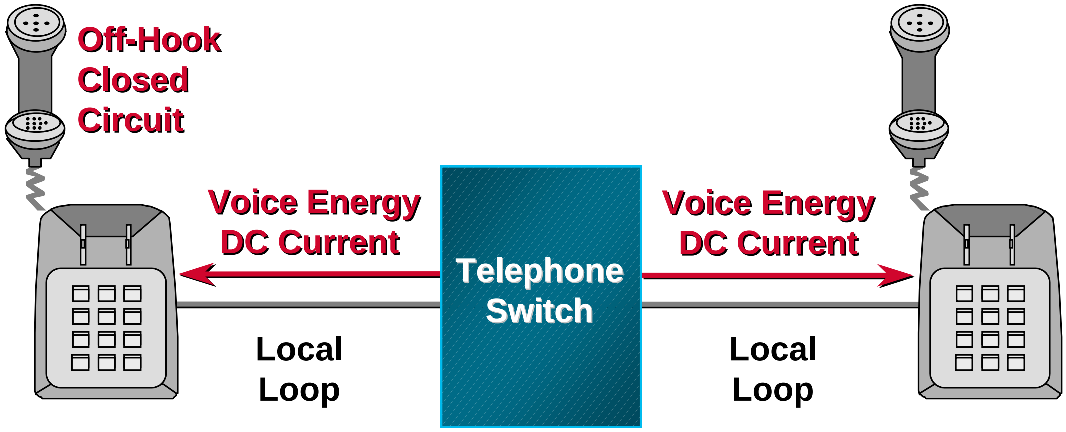
Basic Call Progress: Switching



Basic Call Progress: Ringing



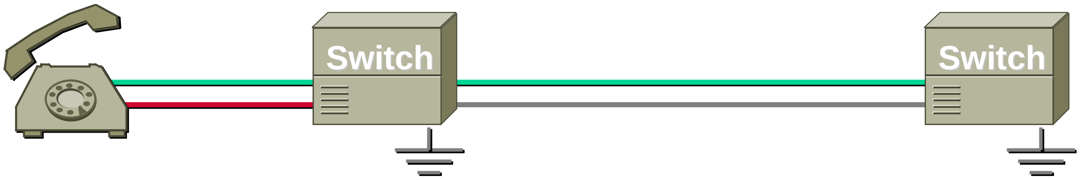
Basic Call Progress: Talking



Analog Telephony—Signaling

- Supervisory
- Addressing
- Call progress

Analog Telephony—Supervisory Signaling



- **Loop start**

  Almost all telephones

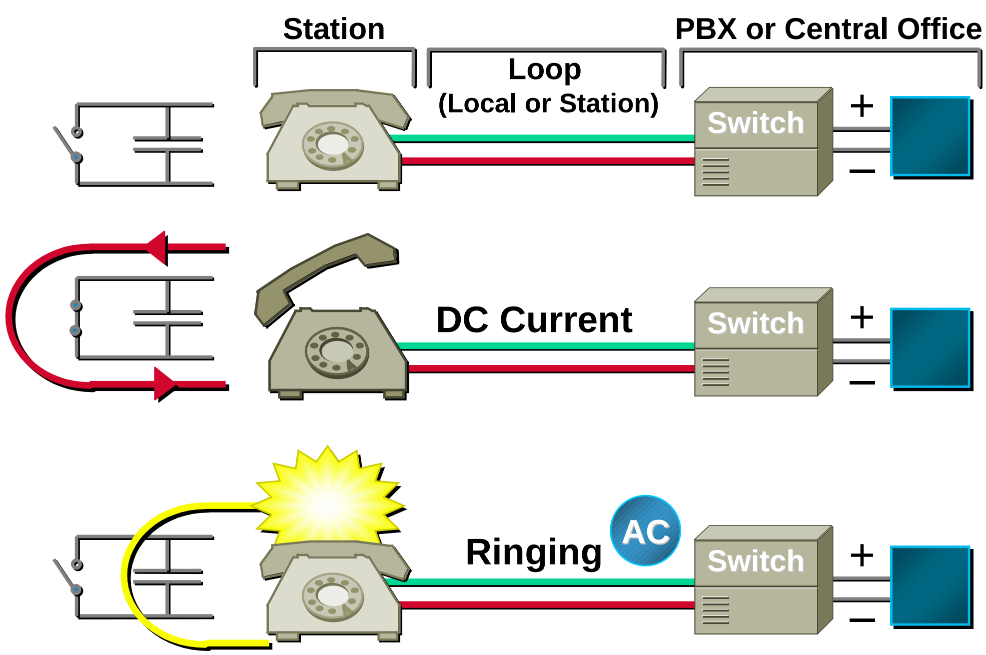
  Current flow sensed

- **Ground start**

  Switch Trunk Lines

  Momentary ground ring lead

Loop Start

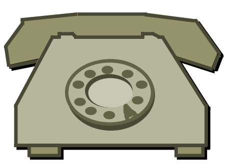


E&M Signaling

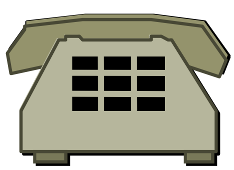
- PBXs, switches
  - Separate signaling leads for each direction
  - E-Lead (inbound direction)
  - M-Lead (outbound direction)
  - Allows independent signaling

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

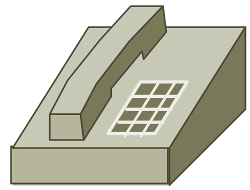
Signaling and Addressing



Dial Pulse



DTMF

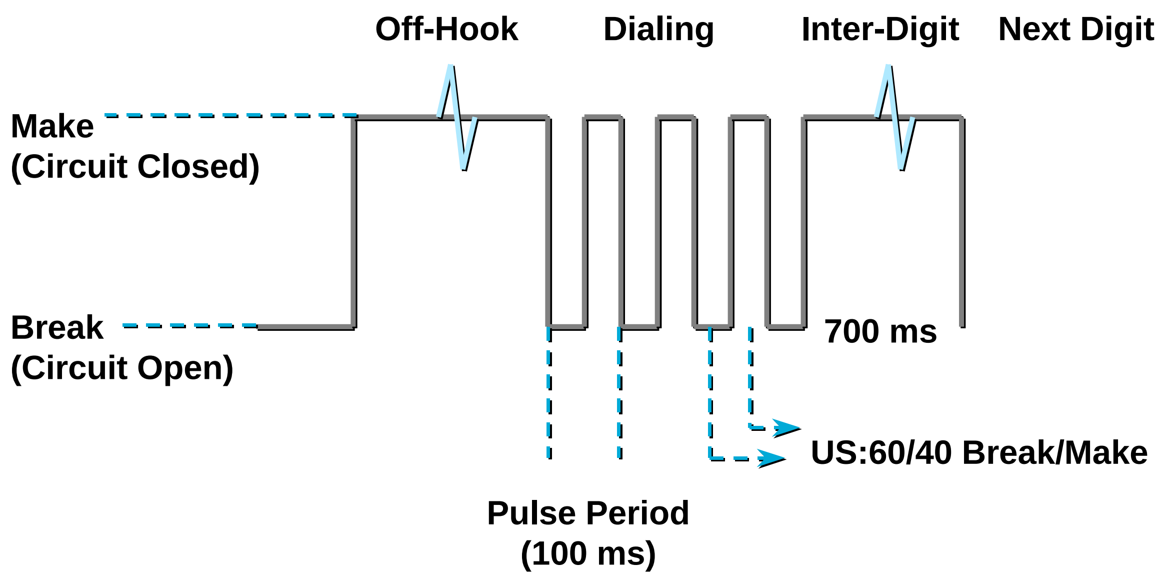


ISDN

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

Digital Transmission
"Out-of-Band"
Message-Based
Signaling

Pulse Dialing

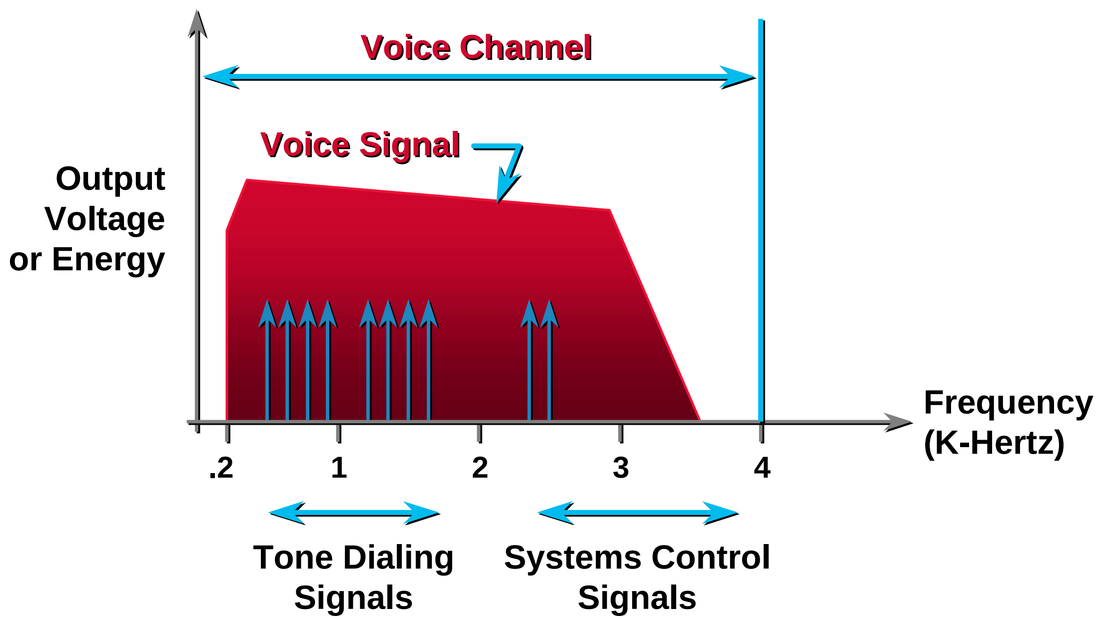


Tone Dialing

Dual Tone Multifrequency (DTMF)

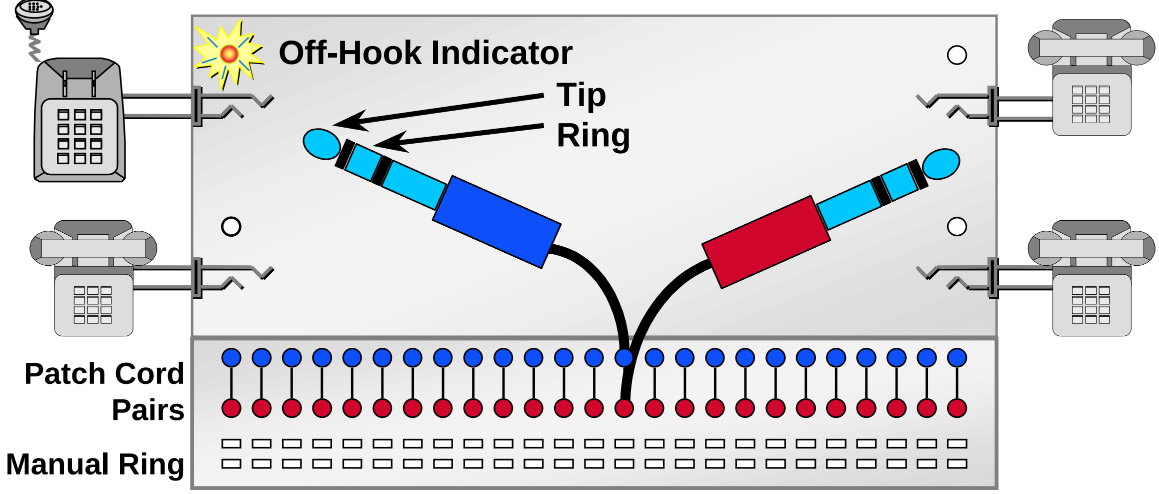
	1209	1336	1477	1633
697	1	2	3	A
770	4	5	6	B
852	7	8	9	C
941	*	0	#	D

Voice Channel Bandwidth

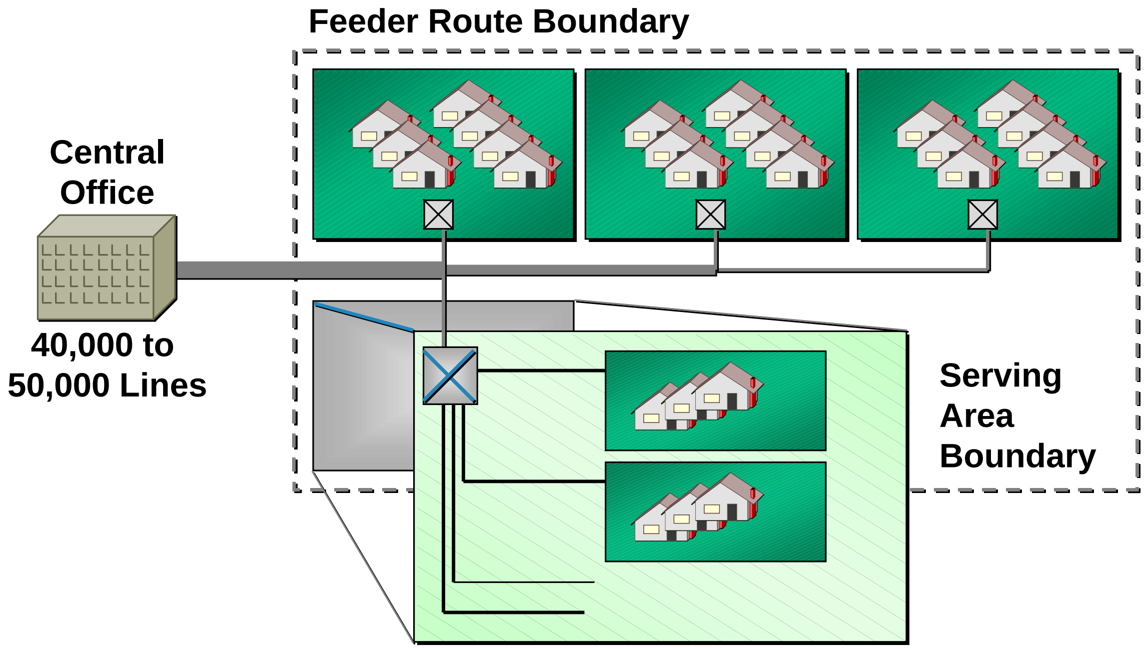


Switching Systems

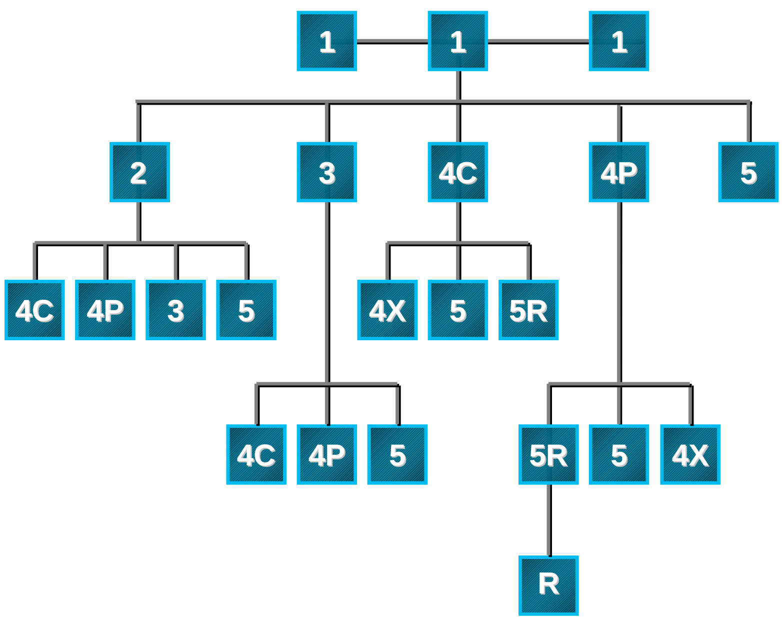
Manual Control—Switch/Cord Boards



Local Access Network

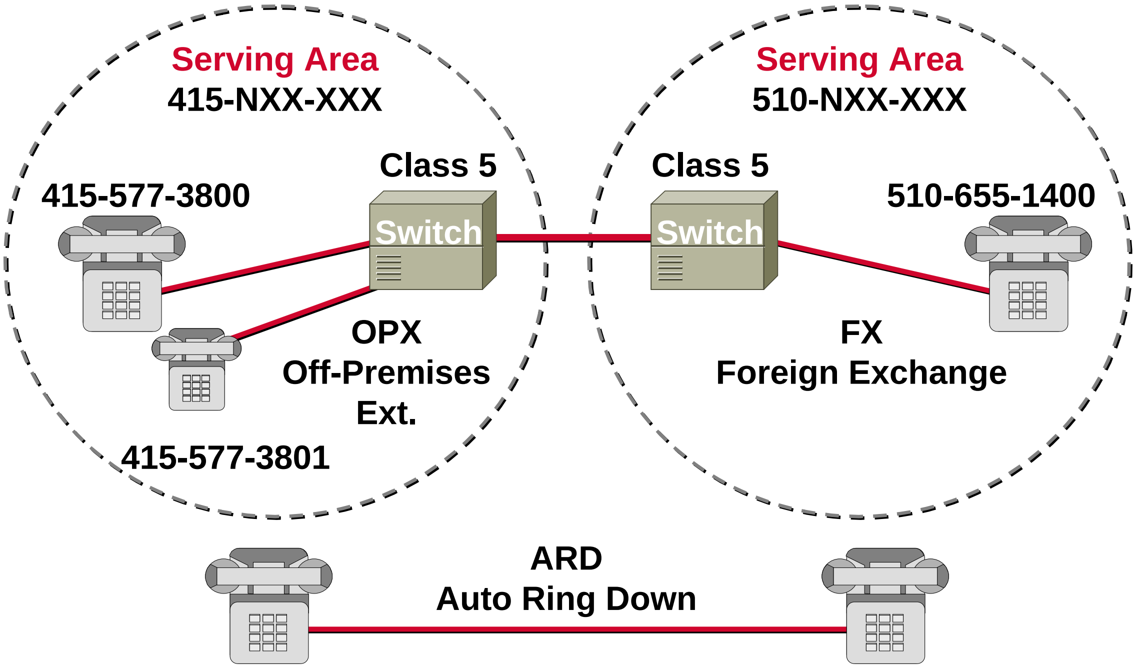


PSTN Network Hierarchy

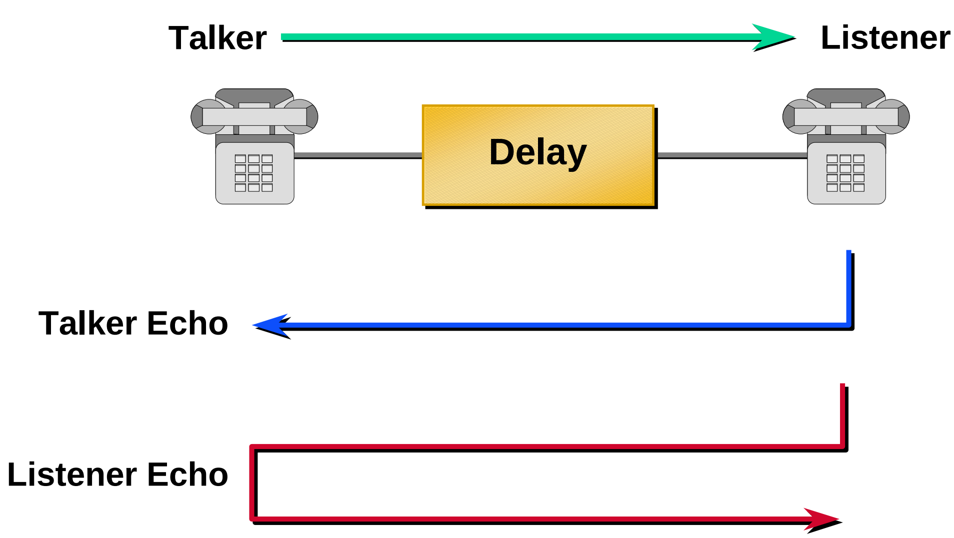


Class	Name
1	Regional Center
2	Sectional Center
3	Primary Center
4C	Toll Center
4P	Toll Point
4X	Interm. Point
5	End Office
5R	EO w/ RSU
R	Remote Sw. Unit

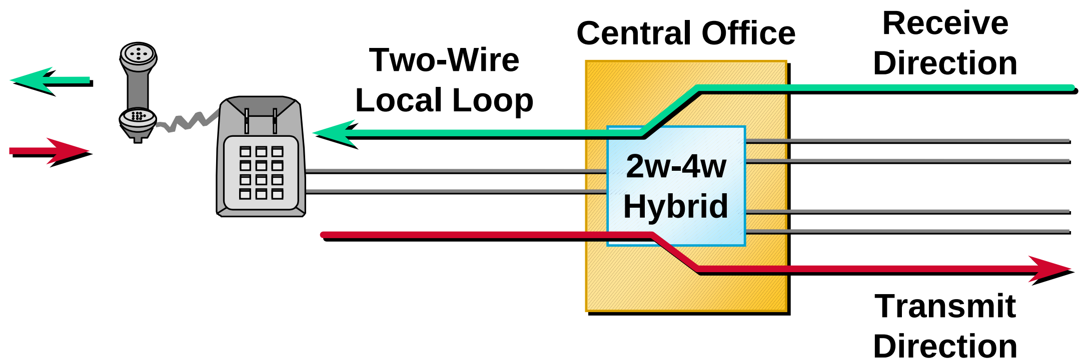
Types of Voice Circuits



Echo in Voice Networks



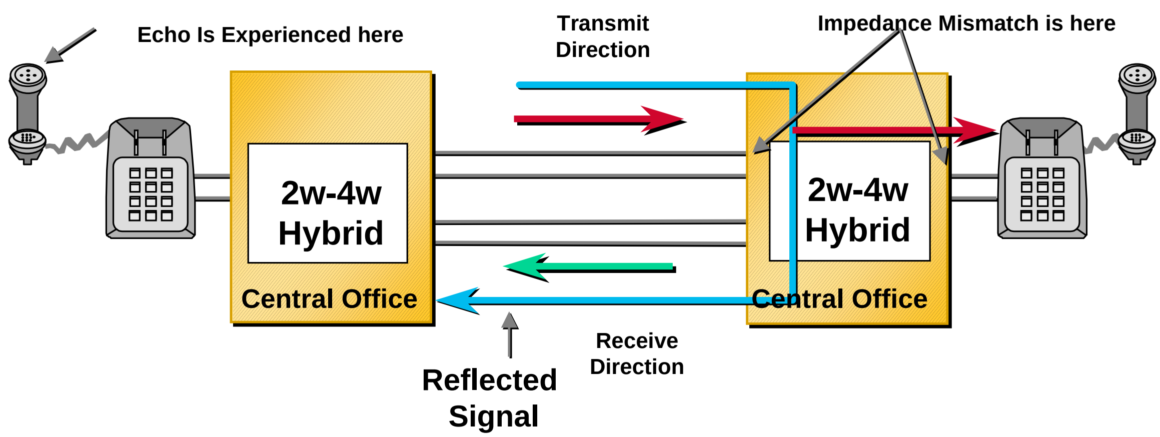
Normal Signal Flow



- 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?

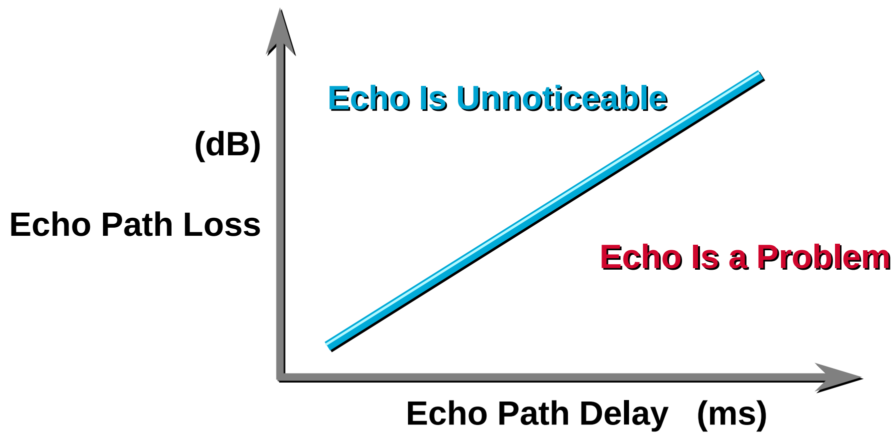
Echo Is Due to a Reflection



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

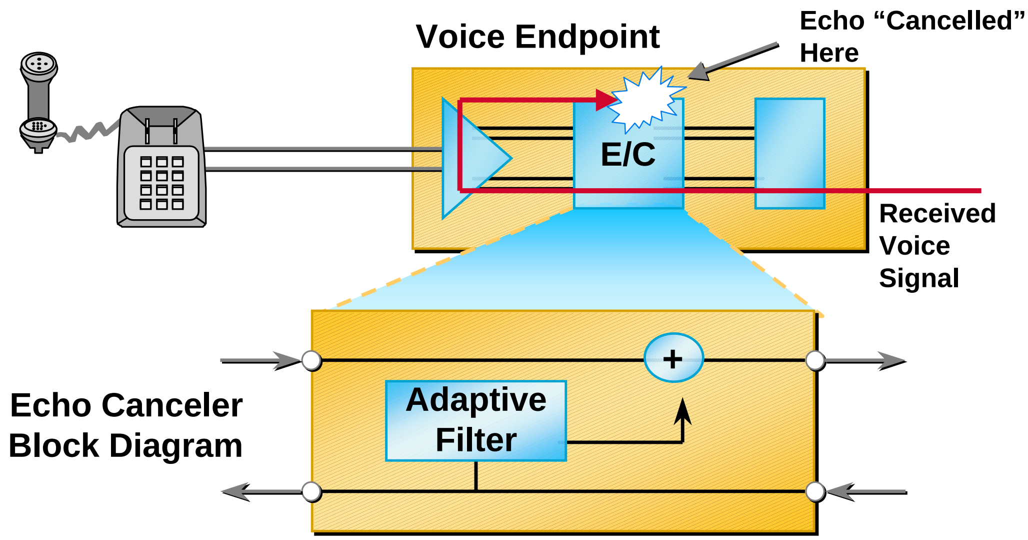
  Disadvantage: static setting and reduces the signal strength of the speaker

- **Echo suppresser**

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

Echo Canceler

Most Effective Means for Removing Echo



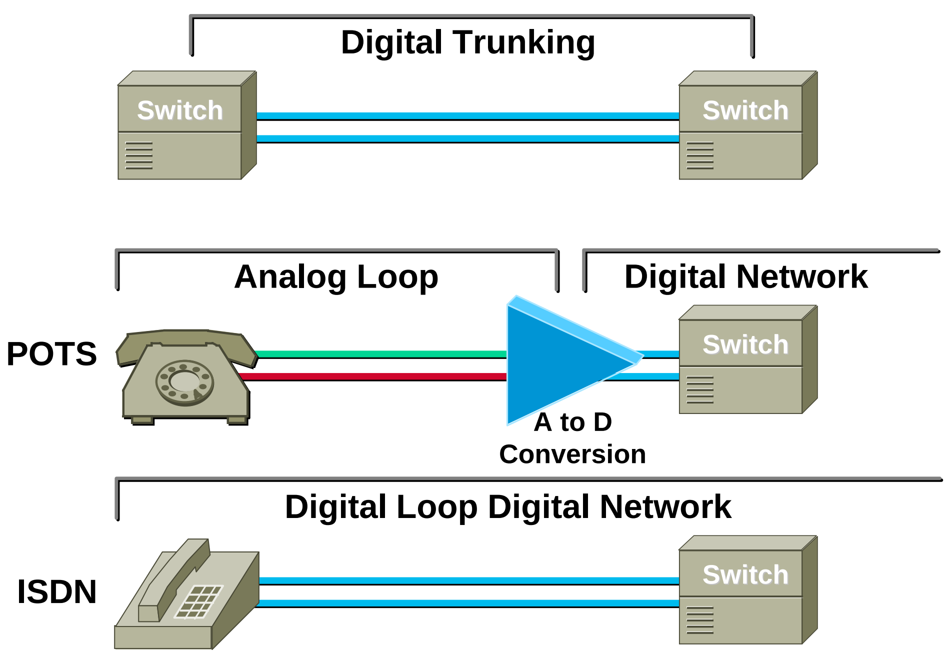
Summary

- 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

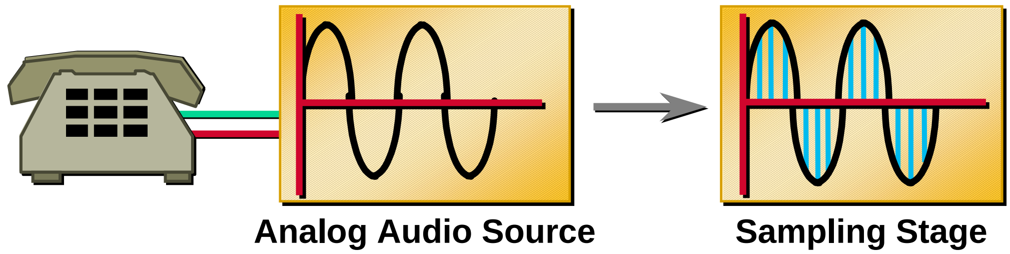
Digital Telephony



Digital Telephony

Pulse Code Modulation—Nyquist Theorem

Voice Bandwidth =
200 Hz to 3400 Hz



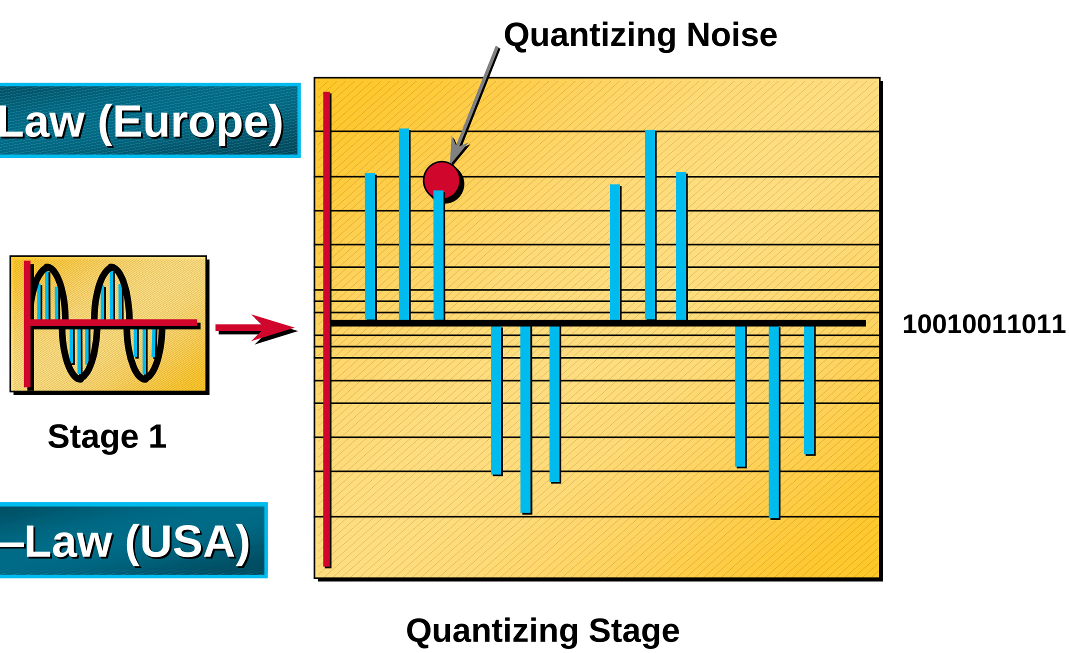
Codec Technique

= Sample

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

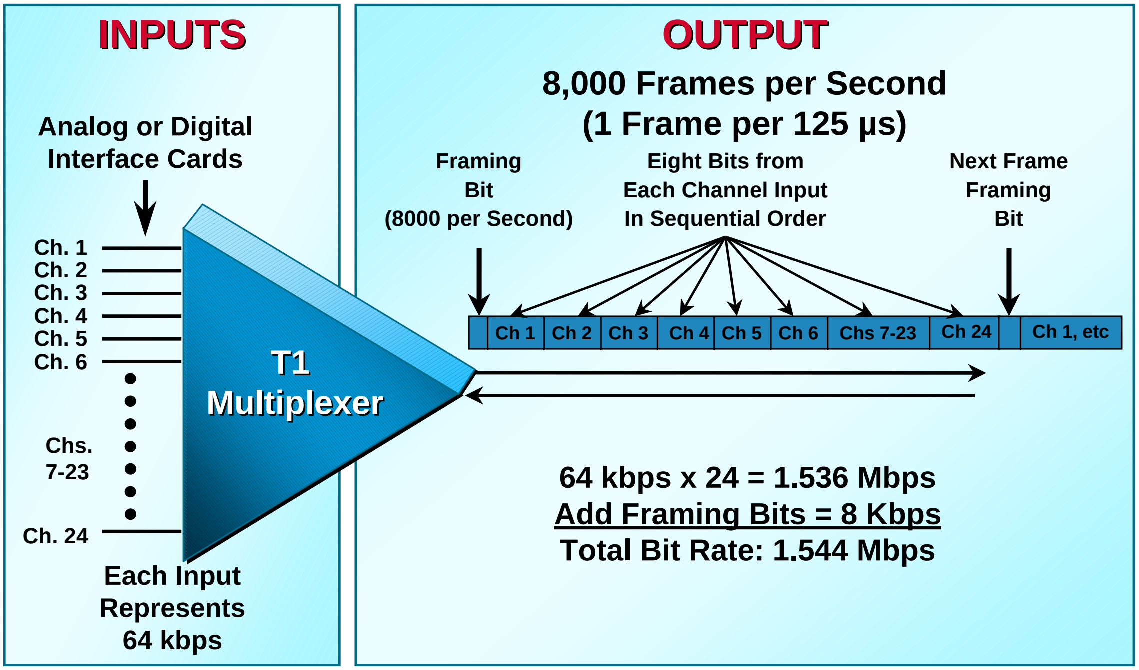
Pulse Code Modulation—
Analog to Digital Conversion

A—Law (Europe)



μ—Law (USA)

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

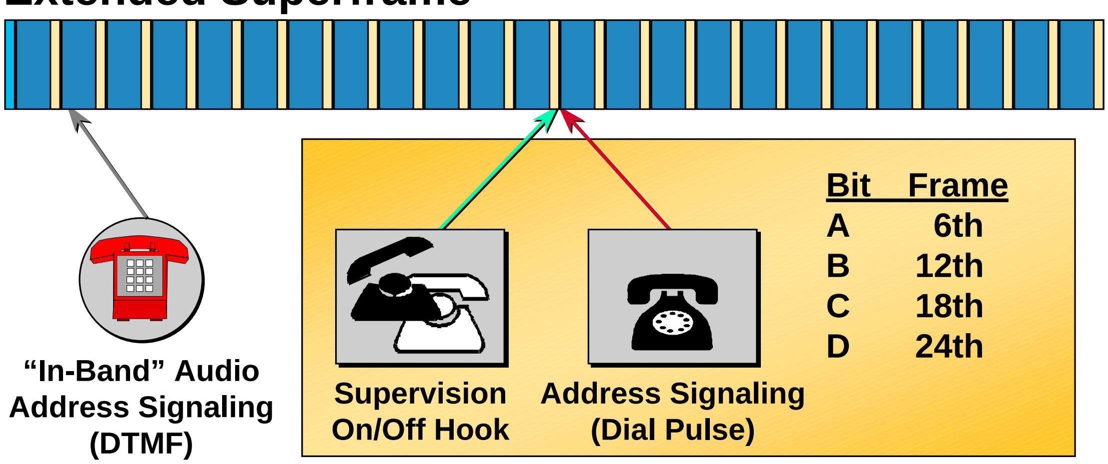
	Framing Bits	Bit Use in Each Channel Time Slot		Signaling—Bit Use Options		
Frame Number	Framing Bit Value	Traffic	Signaling	T	2	4
1	1					
2	0					
3	0					
4	0					
5	1					
6	1	Bits 1–7	Bit 8	*	A	A
7	0					
8	1					
9	1					
10	1					
11	0					
12	0	Bits 1–7	Bit 8	*	A	B

Extended Superframe (ESF)

	S Bits			Bit Use in Each Channel Time Slot		Signaling—Bit Use Options			
Frame Number	Fe	DL	BC	Traffic	Signaling	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

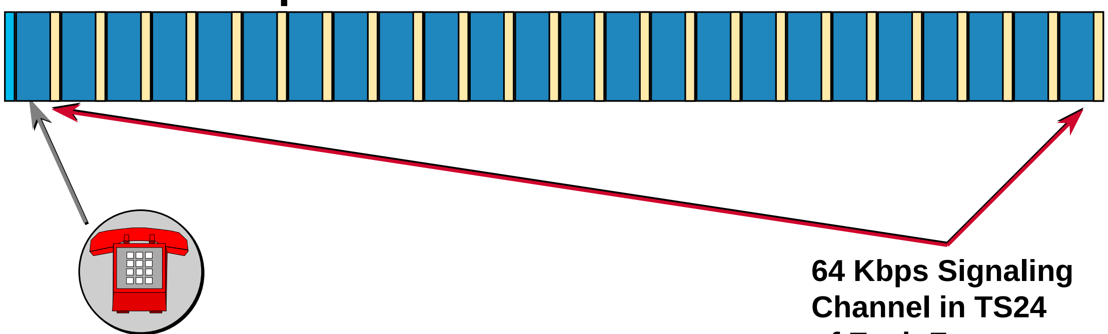
Digital Signaling Schemes

Extended Superframe



Digital Signaling Schemes

Extended Super Frame



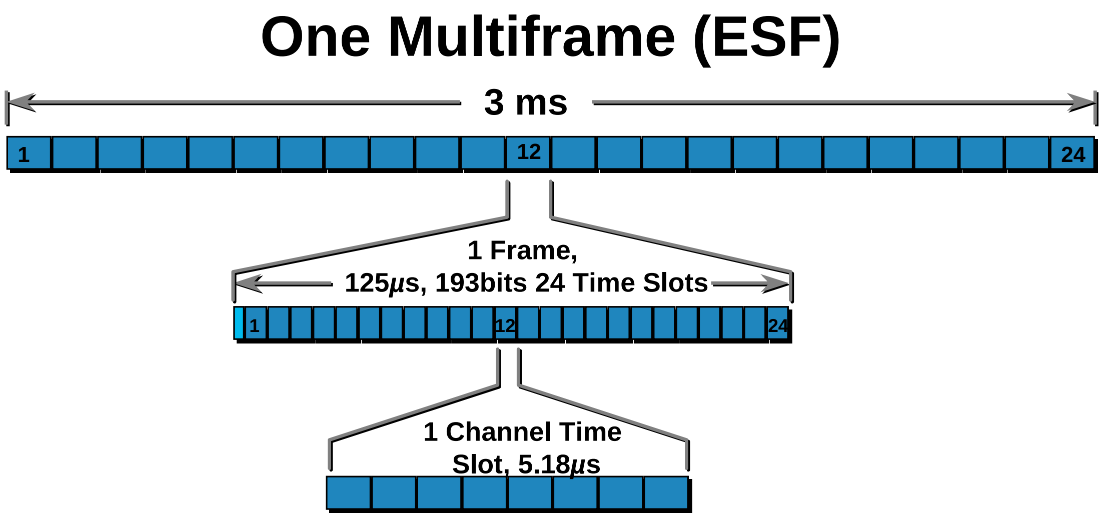
“In-Band” Audio
Address Signaling
(DTMF)

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

Digital Telephony—Synchronization

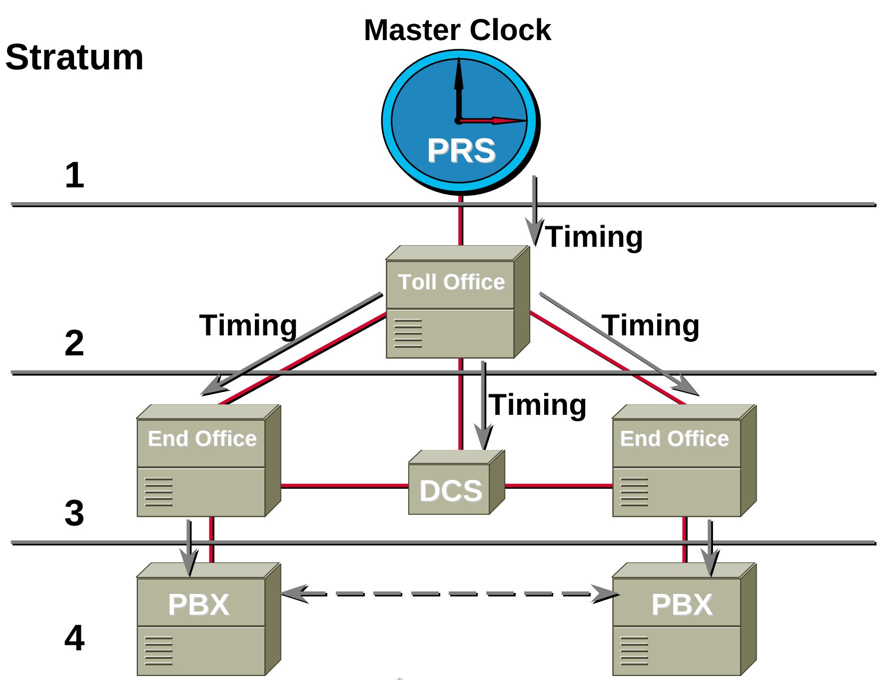
- **Bit synchronization**
  - Primary reference source
  - Ones density
- **Time-slot synchronization**
  - Bits/bytes/channels
- **Frame alignment**
  - 193rd bit pattern

Digital Telephony—Synchronization



  cisco.com

Synchronization—Traditional Network Clocking Strata



  cisco.com

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

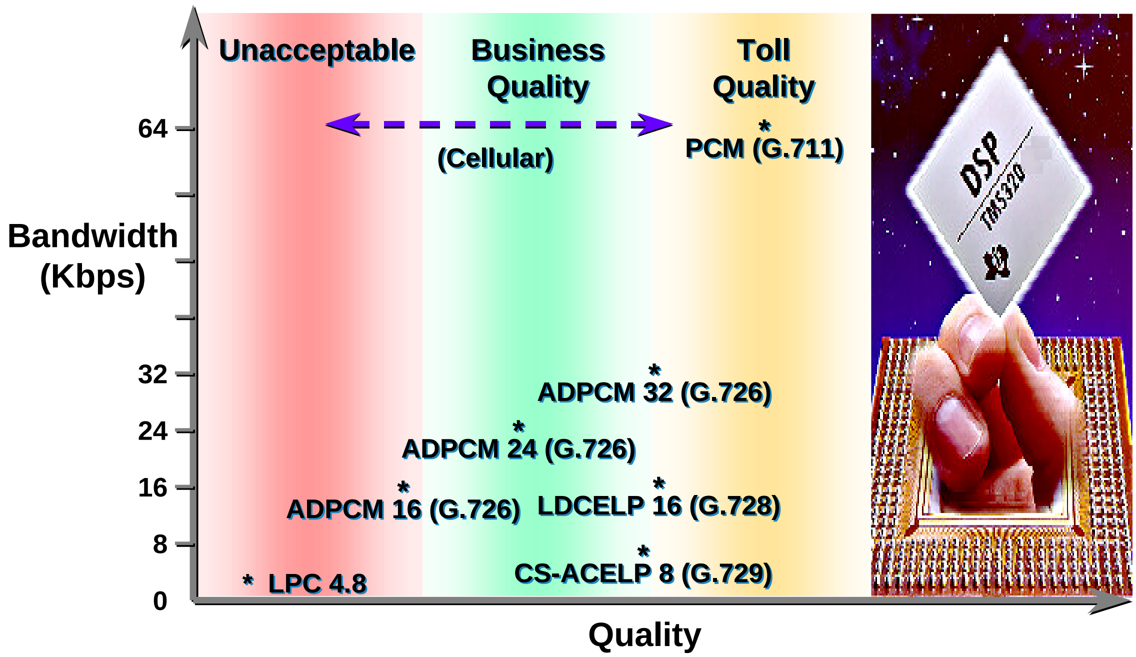
- Speech-coding schemes
- Subjective impairment analysis: mean opinion scores
- Digitizing voice
- Voice compression

    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

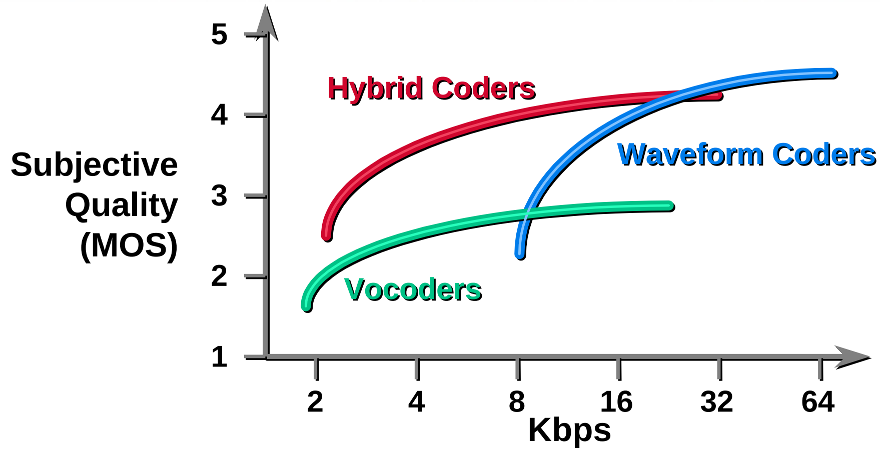
  Example: LPC

- **Hybrid coders**

  Linear waveform approximation with synthesized voice

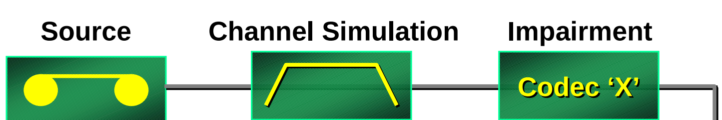
  Example: CELP

Subjective Impairment Analysis: Mean Opinion Scores



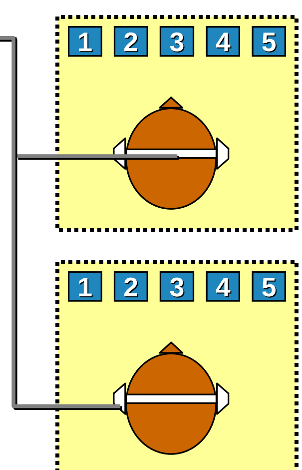
Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just Perceptible, not Annoying
3	Fair	Perceptible and Slightly Annoying
2	Poor	Annoying but not Objectionable
1	Bad	Very Annoying and Objectionable

Measuring Mean Opinion Scores: ITU P.800 Series



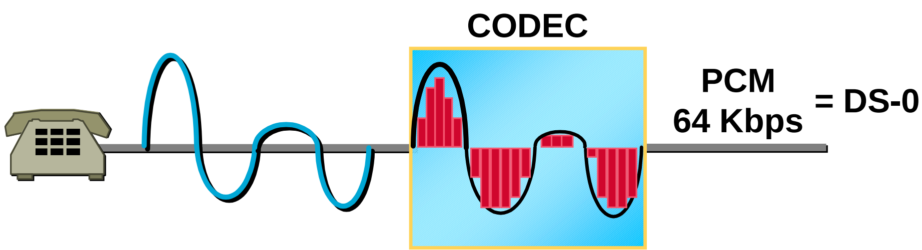
"Nowadays, a chicken leg is a rare dish"

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

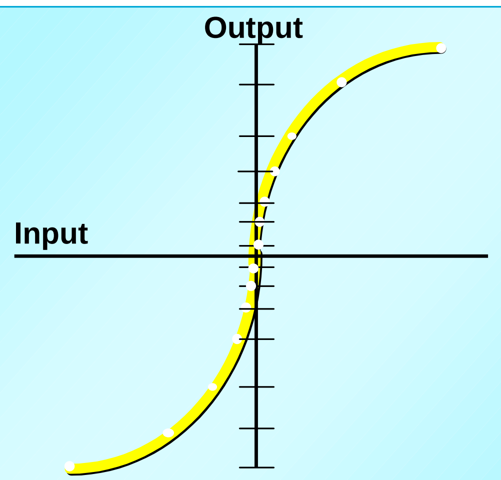


Digitizing Voice: PCM Waveform Encoding Review

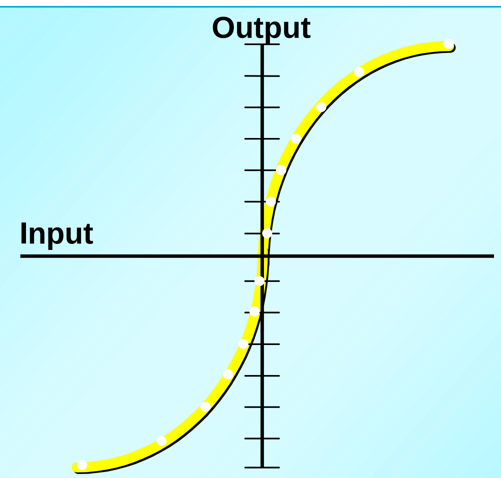
- **Nyquist Theorem: sample at twice the highest frequency**
  - Voice frequency range: 200-3400 Hz
  - Sampling frequency = 8000/sec (every 125µs)
  - Bit rate: (2 x 4 kHz) x 8 bits per sample = 64,000 bits per second (DS-0)
- **By far the most commonly used method**



Nonlinear vs. Linear Encoding

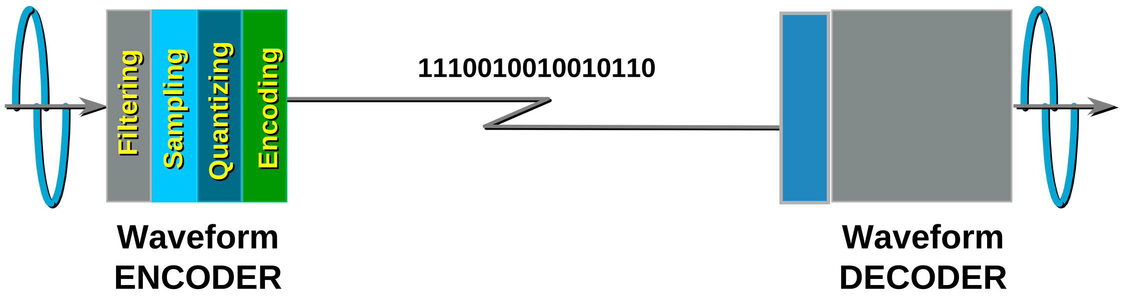


Nonlinear Encoding
Closely Follows Human Voice Characteristics. High Amplitude Signals have More Quantization Distortion.



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

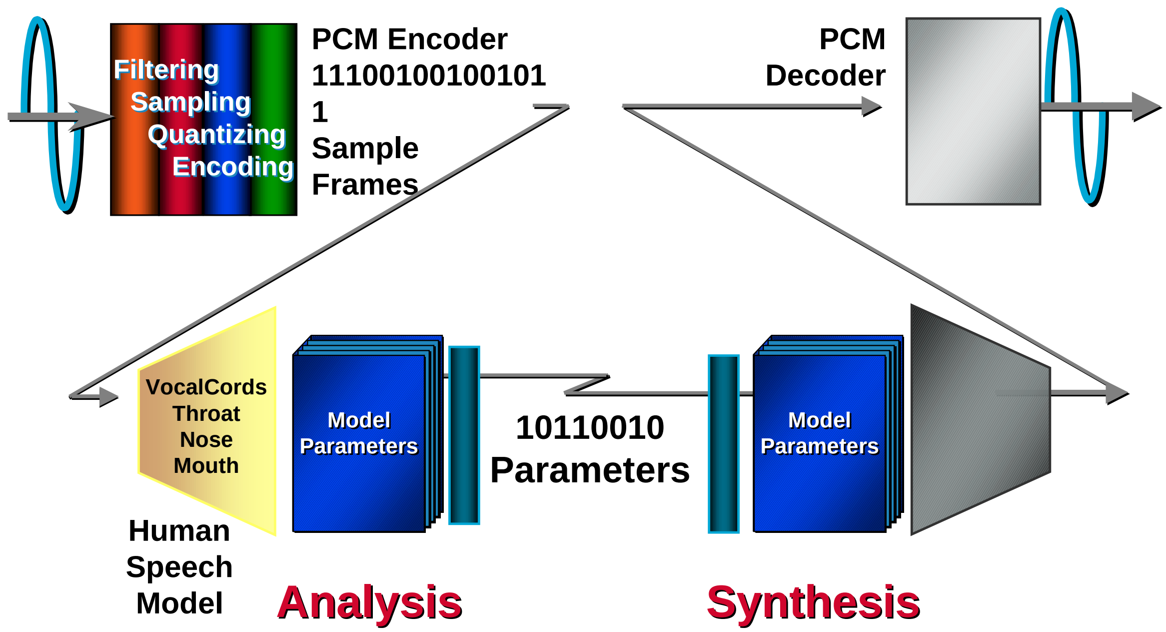
    16 Kbps = 8 Kbps x 2 bits/sample

Voice Compression—CELP

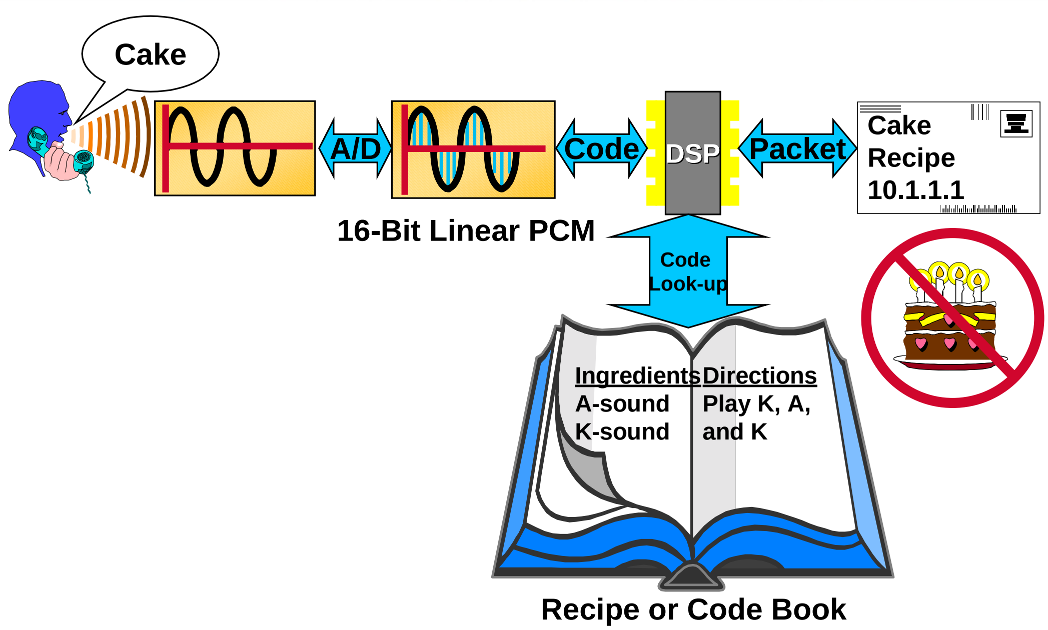
- Code excited linear predictive
- Very high voice quality at low-bit rates, processor intensive, use of DSPs
- G.728: LD-CELP—16 Kbps
- G.729: CSA-CELP—8 Kbps

  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



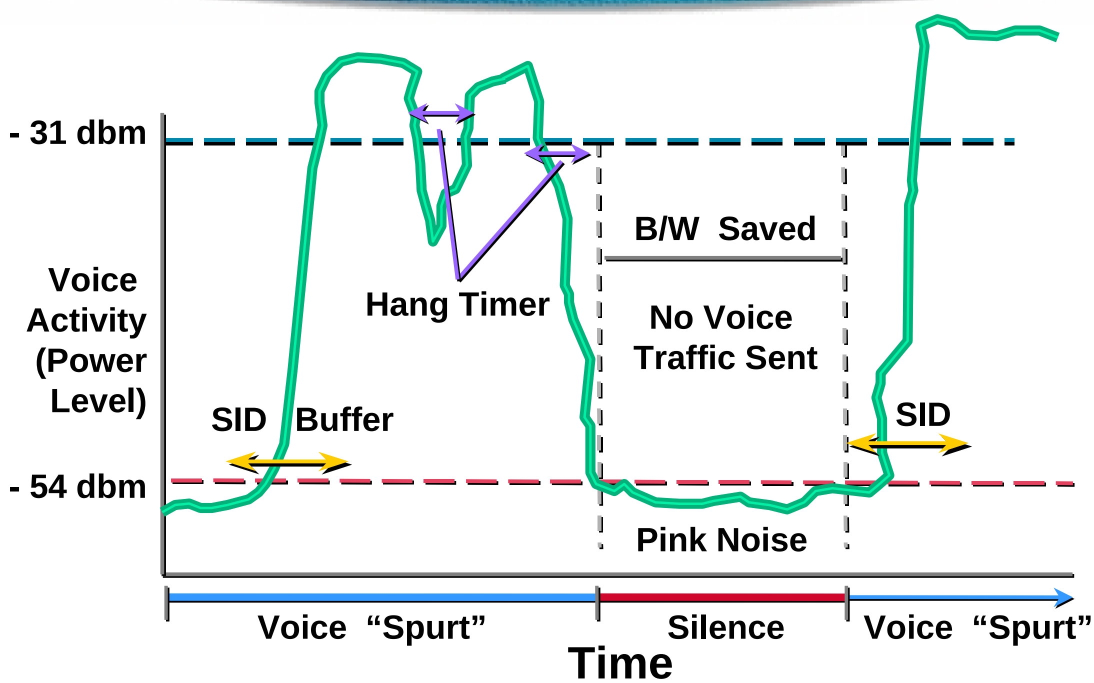
G.729



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

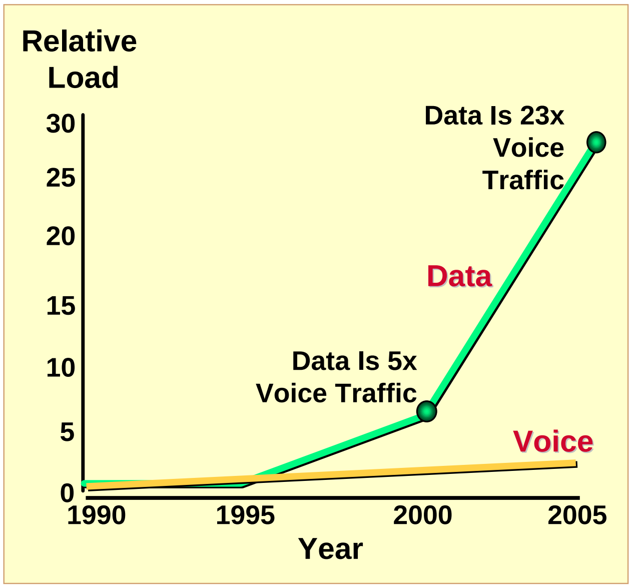
- Basic Analog Telephony
- Basic Digital Telephony
- Voice Coding and Compression Techniques
- Voice Transport and Delay
- Supplemental Slides: Digital Voice Signaling Techniques

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

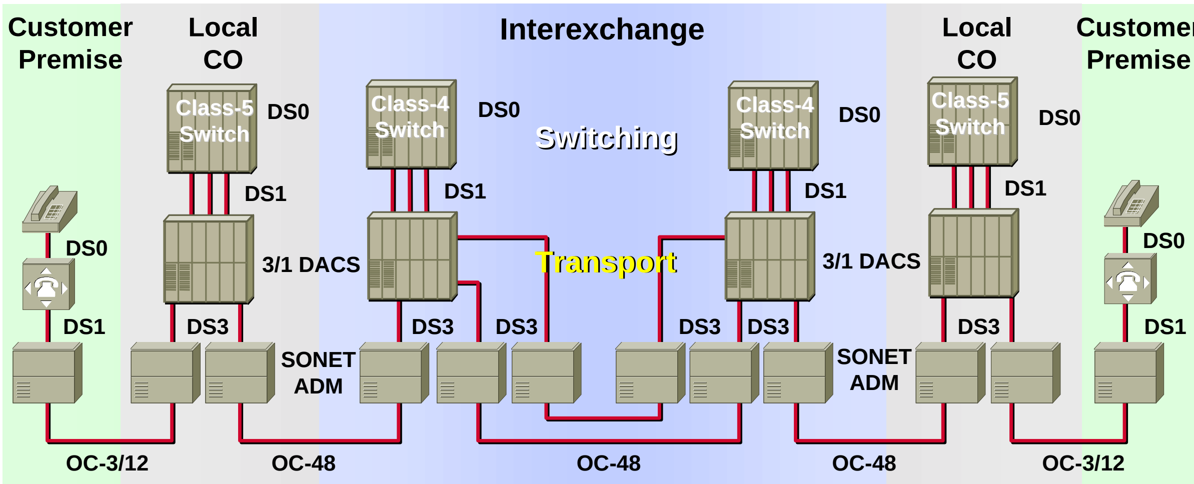
- Evolution from TDM-based transport to packets/cells or a combination



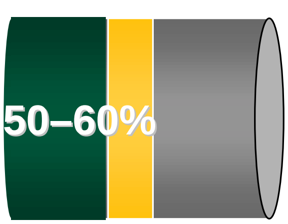
Source: Electronicast

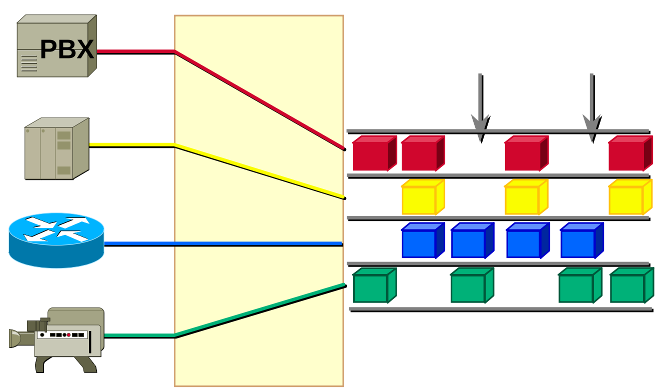
The Tyranny of the DS0

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



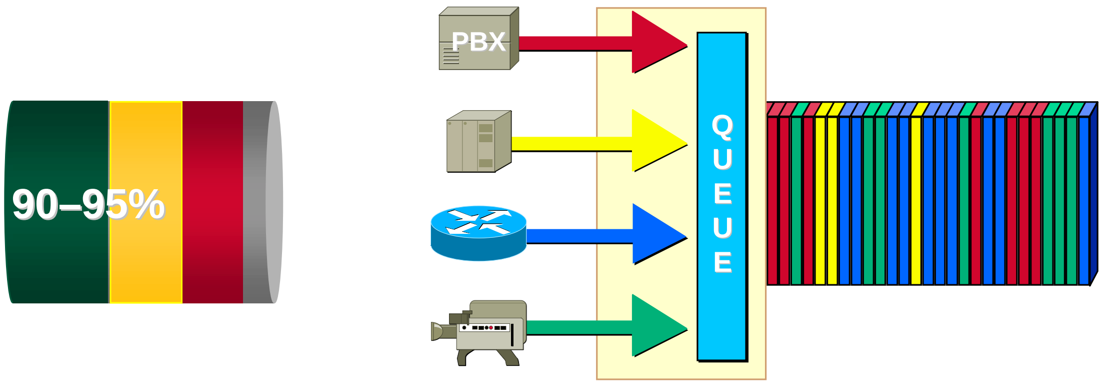
TDM Transport Efficiency





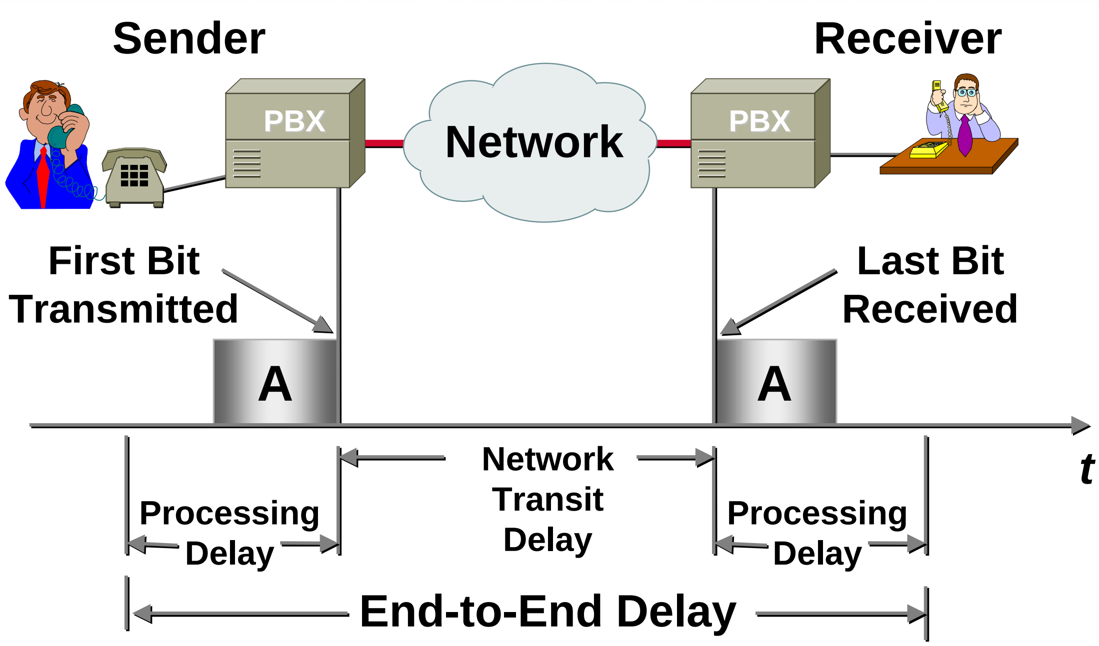
- Wasted bandwidth
- No congestion

Packet Transport Efficiency

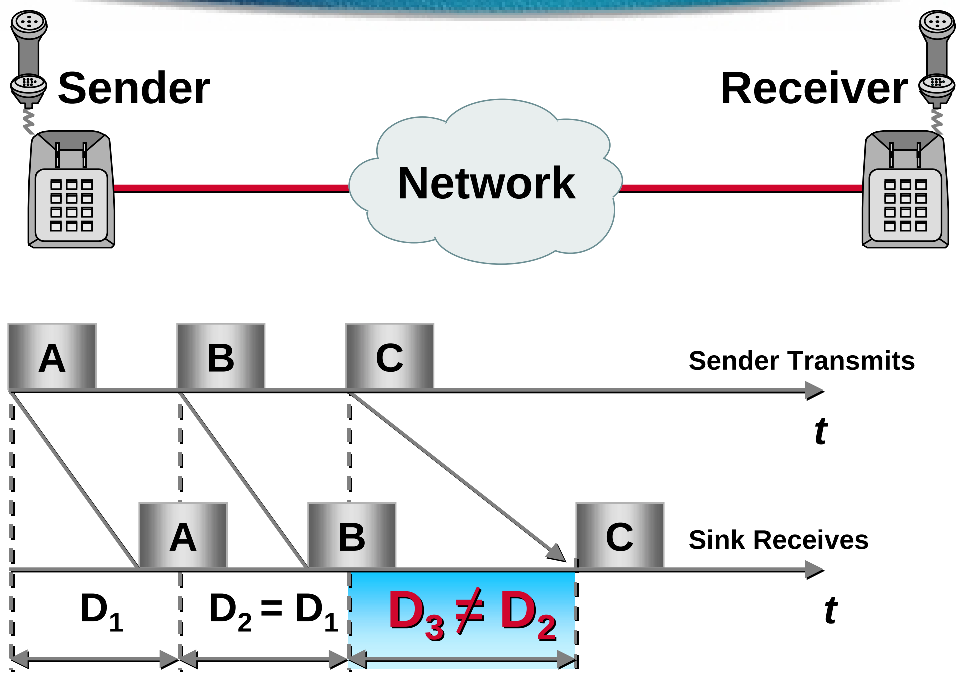


- High bandwidth efficiency
- Congestion management

Delay



Delay Variation—"Jitter"



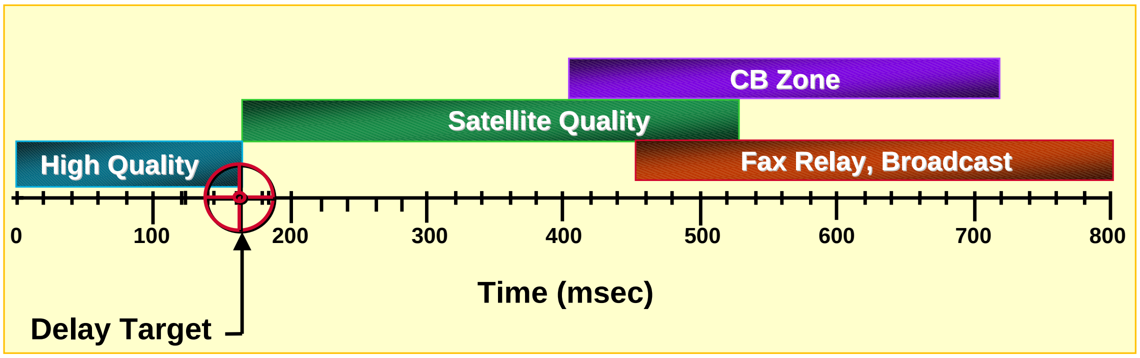
Voice Delay Guidelines

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

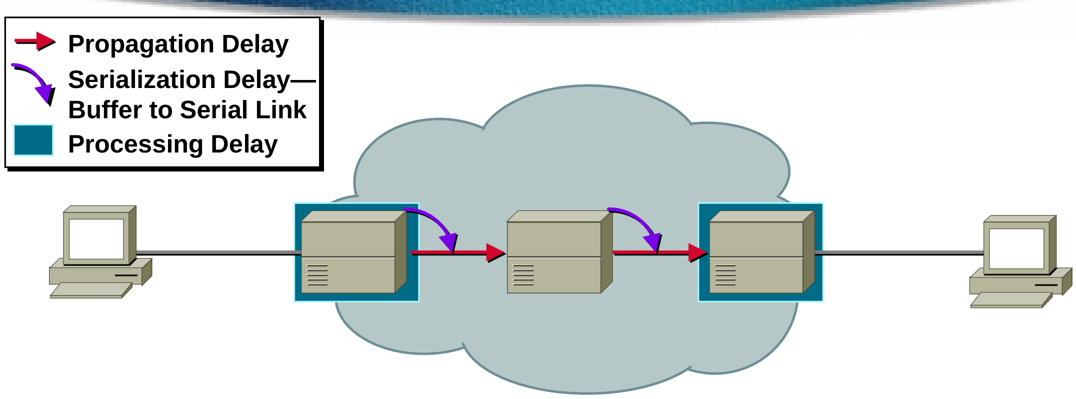
ITU's G.114 Recommendation

Delay in Perspective

Cumulative Transmission Path Delay



Fixed Delay Components

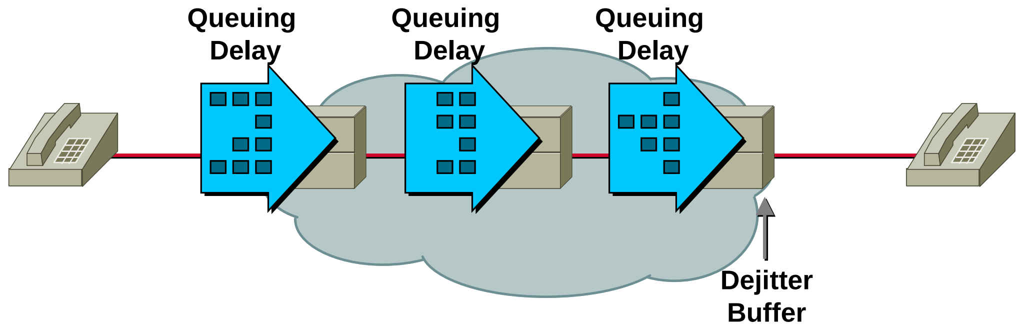


- Propagation—six microseconds per kilometer
- Serialization
- Processing

  Coding/compression/decompression/decoding

  Packetization

Variable Delay Components



- Queuing delay
- Dejitter buffers
- Variable packet sizes

An Example

- Assumptions:

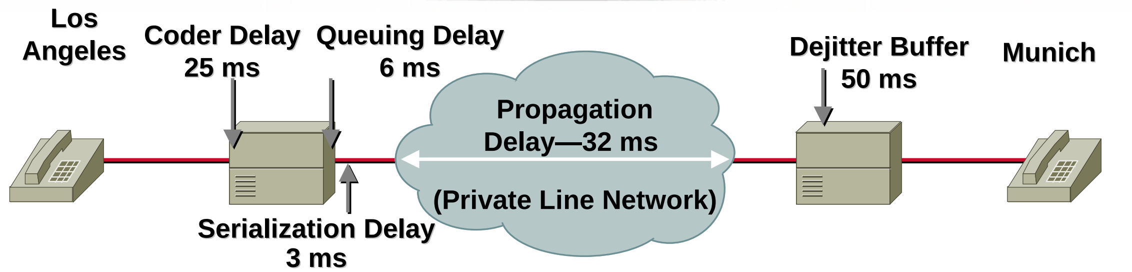
  We have eight trunks

  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



	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	

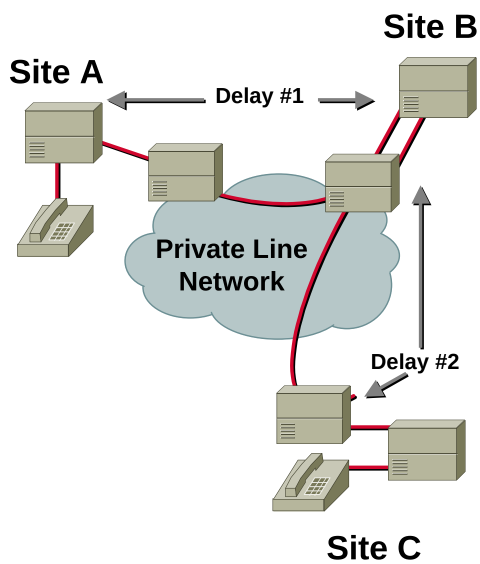
Variable Delay Component

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 64 Kbps link takes 3 msec
- 1 byte over a 64 Kbps 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

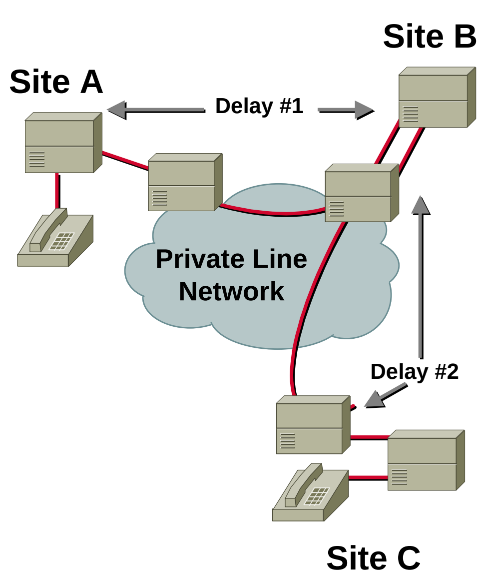
Delay Calculation

	Fixed Delay	Variable Delay
DELAY #1 Coder Delay G.729	25 msec	
Packetization Delay (Included in Coder Delay)		
Max Queuing Delay 64 kbps Trunk		21 msec
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 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

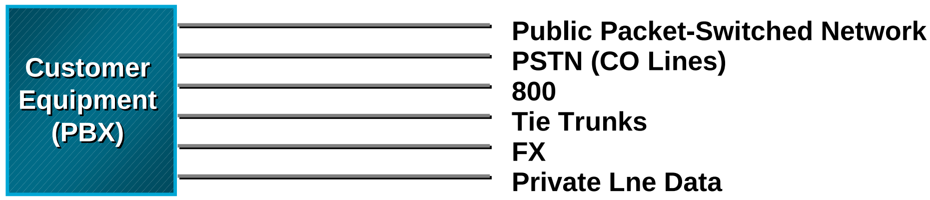
- ISDN
- Q.930/Q.931
- Signaling System 7
- Voice addressing

ISDN

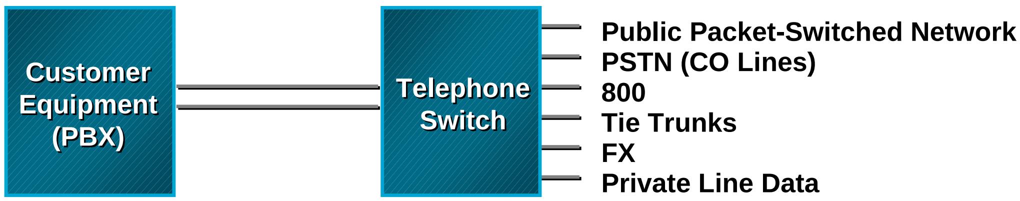
- Integrated Services Digital Network
  - Part of a network architecture
  - Definition for the access to the network
  - Allows access to multiple services through a single access
- Standards-based
  - ITU recommendations
  - Proprietary implementations

Network Access

Traditional Access



ISDN Access



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)**

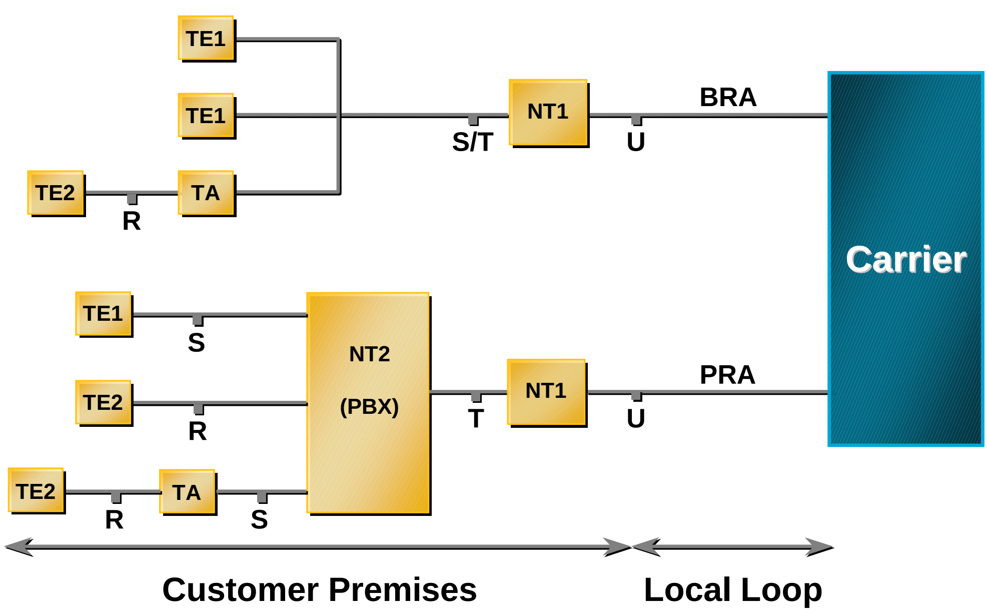
  23 B + D

  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



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**

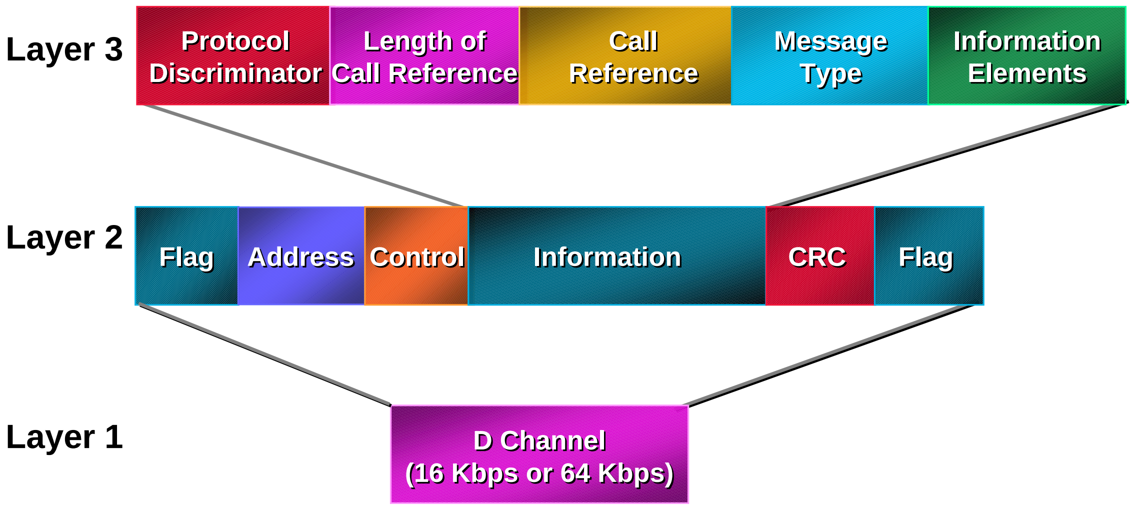
Access to ISDN

- 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



ISDN CCS (Q.930/931) Messages

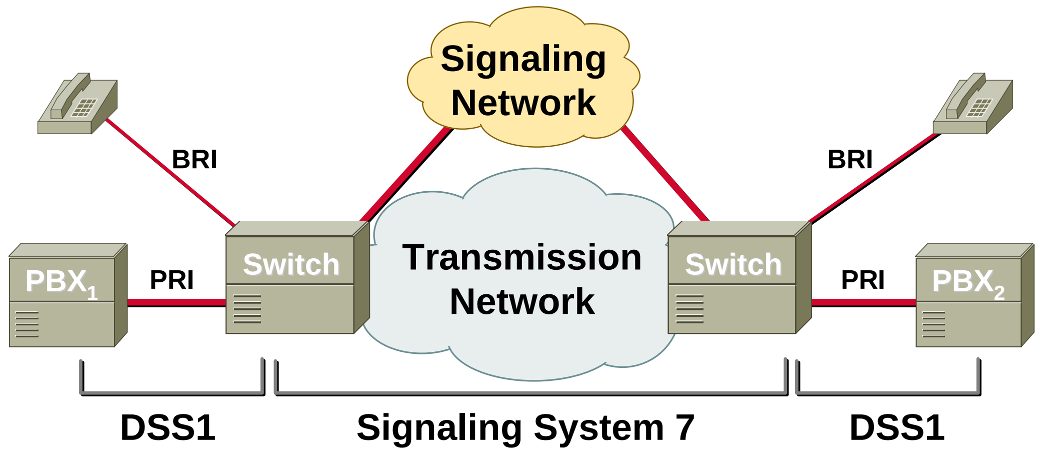
Call Establishment
- Alerting
- Call proceeding
- Connect
- Connect ack
- Progress
- Setup
- Setup ack

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

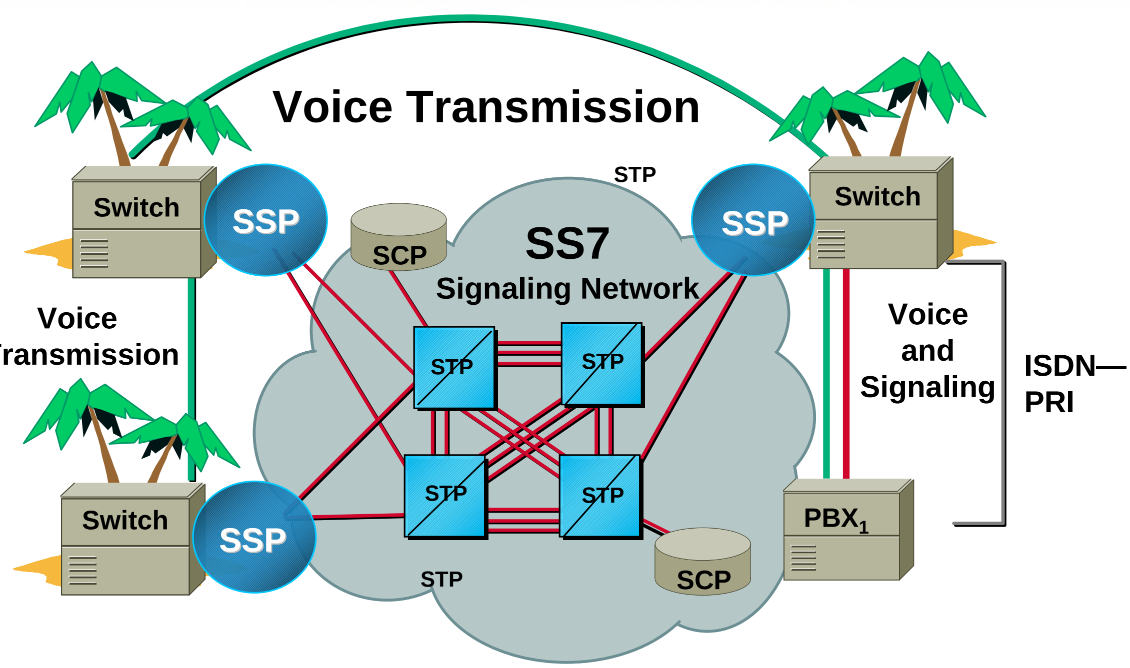
Miscellaneous
- Congestion control
- Facility
- Information
- Notify
- Register
- Status
- Status inquiry

Public ISDN and Signaling System 7

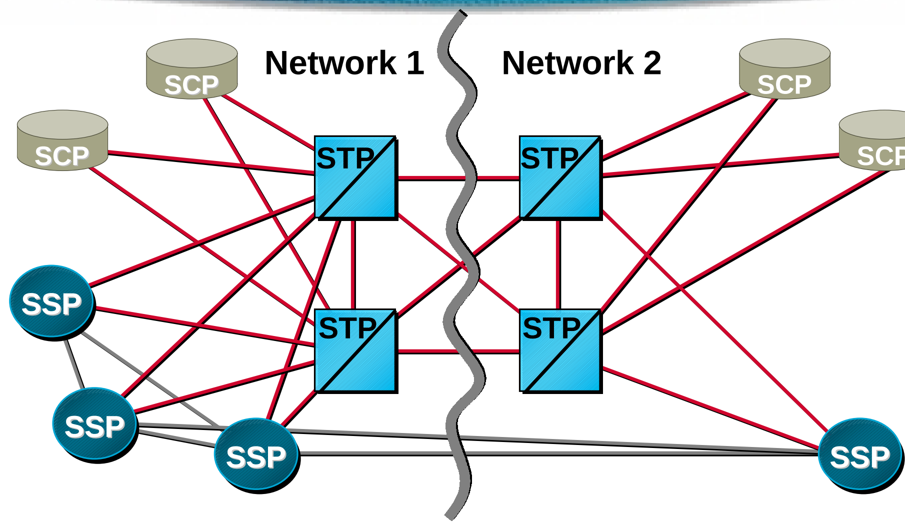


DSS1 Is a Public ISDN Protocol

ISDN and SS7 "The Bridge Between the Islands"

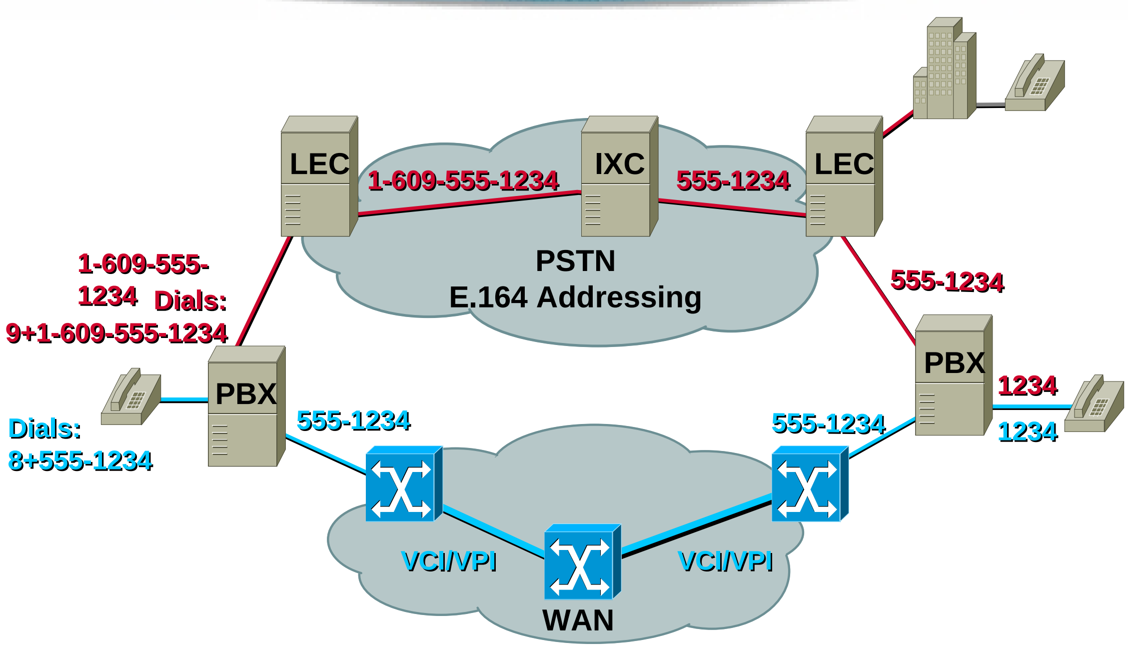


SS7 Components



Voice Trunk	SSP: Signal Switching Point
Signaling Link	STP: Signal Transfer Point
	SCP: Signal Control Point

Network Addressing



Agenda

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

Introduction to Voice and Telephone Technology

Q&A

Please Complete Your Evaluation Form

Session 2001

CISCO SYSTEMS

EMPOWERING THE INTERNET GENERATION℠