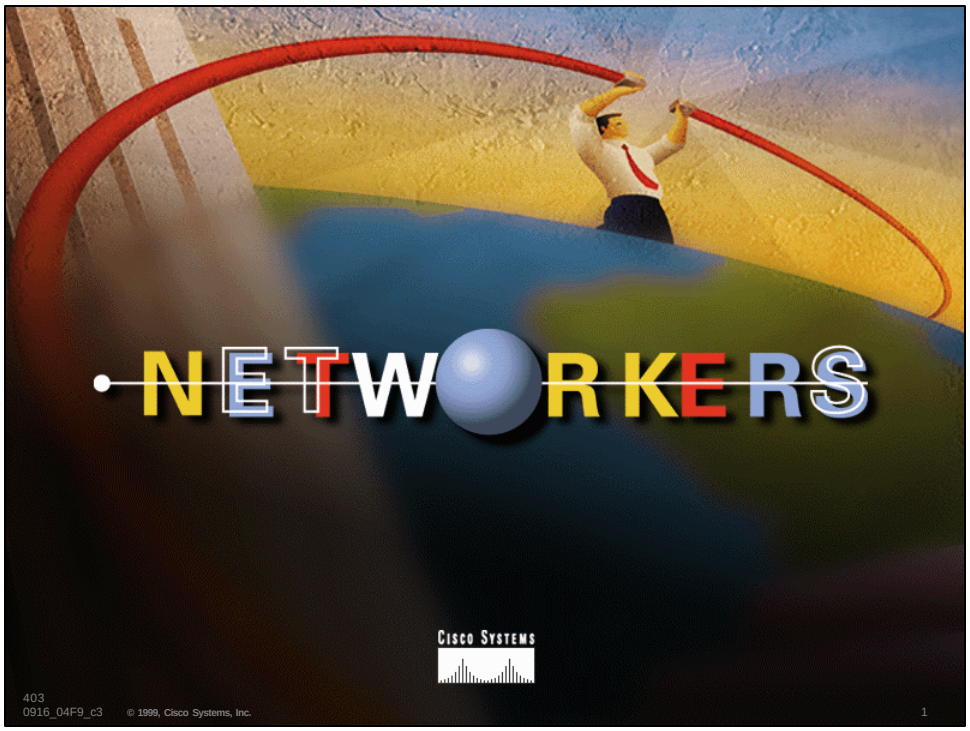


Objectives

- Understand the voice quality factors in an integrated data/voice network
- Describe integrated data/voice network transport mechanisms
- Understand the engineering requirements in data/voice networks to meet voice quality requirements
- Know where to look for the “Gremlins” in integrated data/voice networks

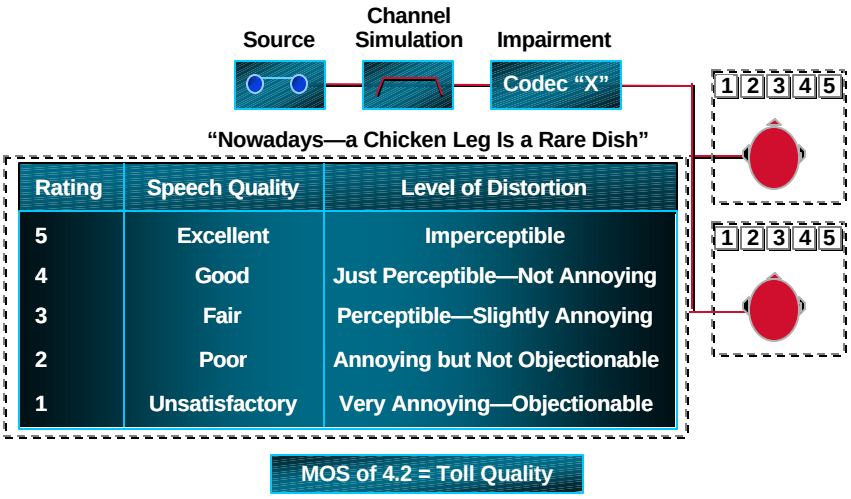
Agenda

- Establishing Target Parameters for Quality
- Network Technology Trade-Offs
- Voice Over Frame Relay Network Design
- Voice Over ATM Network Design

Voice Quality Factors

- Mean opinion score
- Coding
- Compression
- Delay

Mean Opinion Score



Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just Perceptible—Not Annoying
3	Fair	Perceptible—Slightly Annoying
2	Poor	Annoying but Not Objectionable
1	Unsatisfactory	Very Annoying—Objectionable

MOS of 4.2 = Toll Quality

Voice Quality Guidelines

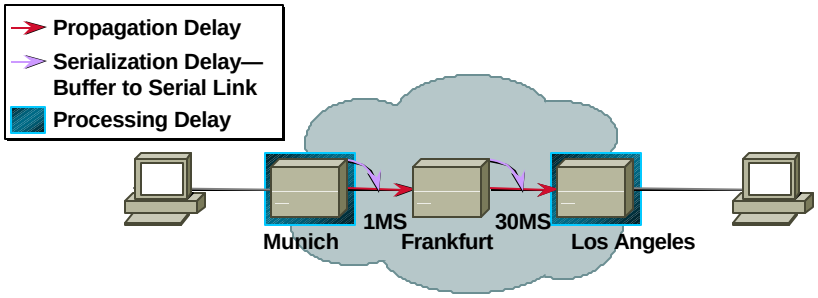
Compression Method	MOS Score	Delay (Msec)	Bit Rate (Kbps)
PCM (G.711)	4.4	0.75	64
ADPCM (G.726)	4.1	1	32–24–16
LD-CELP (G.728)	3.65	3–5	16
CS-ACELP (G.729)	3.9	15	8
CS-ACELP (G.729a)	3.65	15	8
MPMLQ or ACELP (G.723.1)	3.8	30	6.3–5.3

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

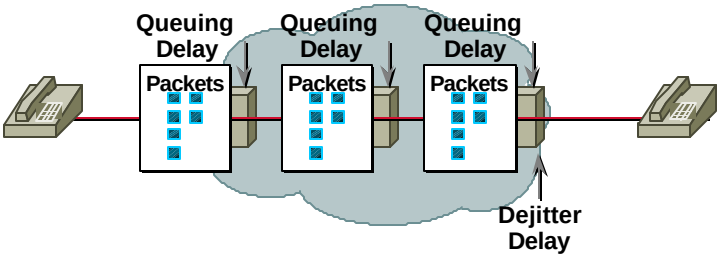
ITU's G.114 Recommendation

Fixed Delay Components



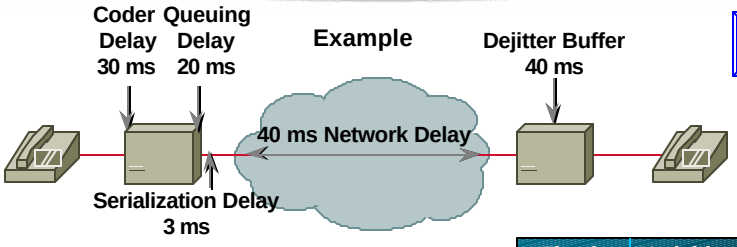
- Propagation—six microseconds per kilometer
- Serialization
- Processing
  - Coding/compression/decompression
  - Packetization

Variable Delay Components



- Dejitter buffers
- Queuing delay
- Variable packet sizes

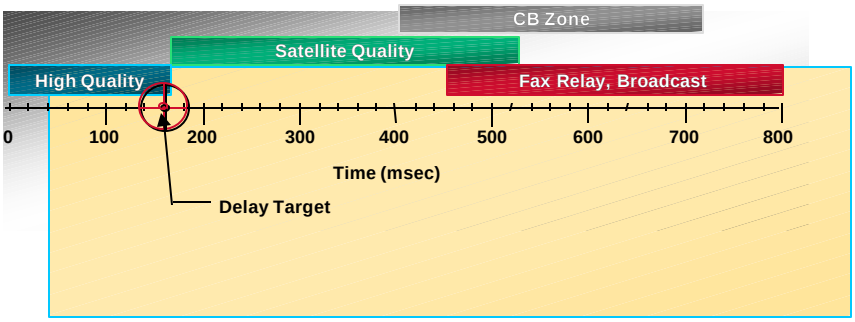
Calculate Delay Budget



	Fixed Delay	Variable Delay
Propagation Delay		
Coder Delay G.729 (15 Msec Per 10 Bytes)	30 Msec	
Packetization Delay—Included in Coder Delay		
Queuing Delay 64 Kbps Trunk		20 Msec
Serialization Delay 64 Kbps Trunk	3 Msec	
Network Delay (Public Frame Relay Network)	40 Msec	
Dejitter Buffer	40 Msec	
Total	113 Msec	20 Msec

Delay—How Much Is Too Much?

Cumulative Transmission Path Delay



ITU's G.114 Recommendation = 0 – 150msec 1-way delay

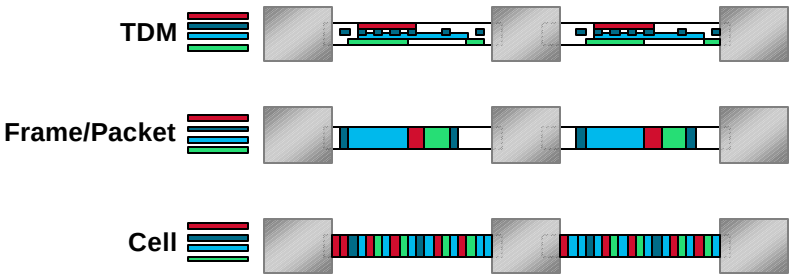
Target Parameters

- MOS—Business/toll quality
- Coding—G.729, G.711
- Compression—8k, 64k
- Delay—Calculate network

Technology Trade-Offs

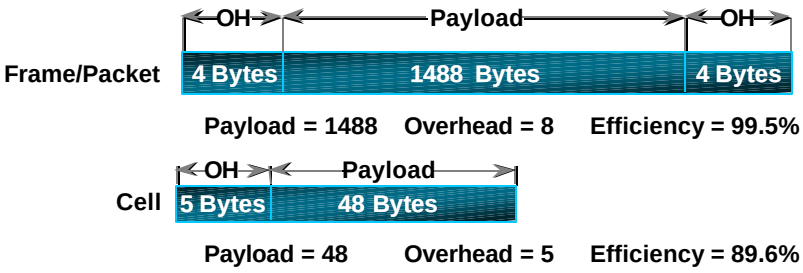
- Frame vs cell
- Mesh vs star
- Transport vs translate

TDM vs Frame vs Cell



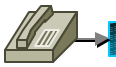
- TDM—Constant delay, wasted bandwidth
- Frame/packet—Variable delay, highly efficient
- Cell—Improved delay, less efficient

Packet Efficiency



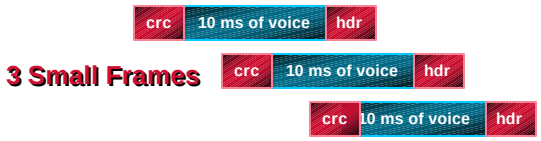
- Small vs large packet sizes
- Fixed vs variable sized packets

Voice Payload Options



Original Voice Information

- **Small Payload**
  - Low Delay
  - High Overhead
  - High PPS
  - High CPU Load

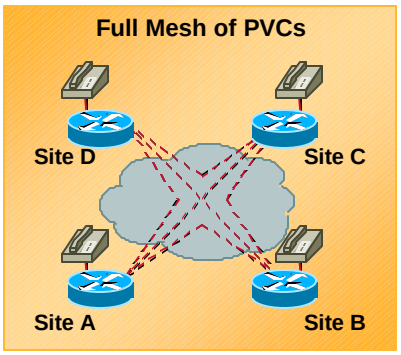


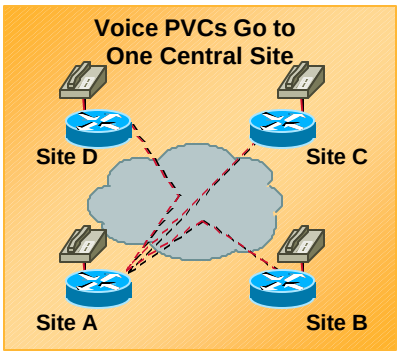
3 Small Frames

- **Large Payload**
  - High Delay
  - Low Overhead
  - Low PPS
  - Low CPU Load

1 Large Frame

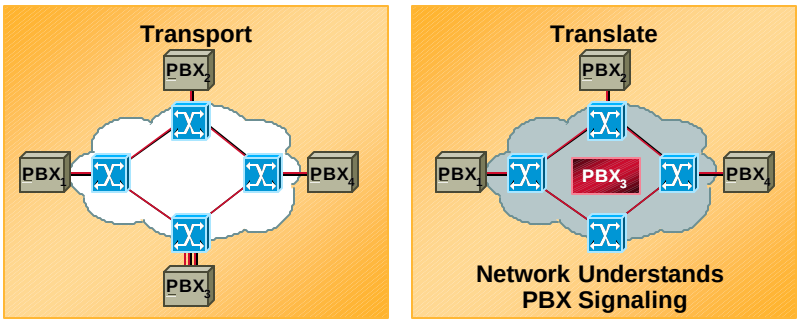
Network Design Options





- Separate voice and data PVCs—Maximizes quality of service
- Combine voice and data on one PVC—Minimizes recurring costs
- Or use some combination

Transport vs Translate Model



- ATM
- Frame Relay
- Router-based backbone—such as TCP/IP
- Or a mixture of the above

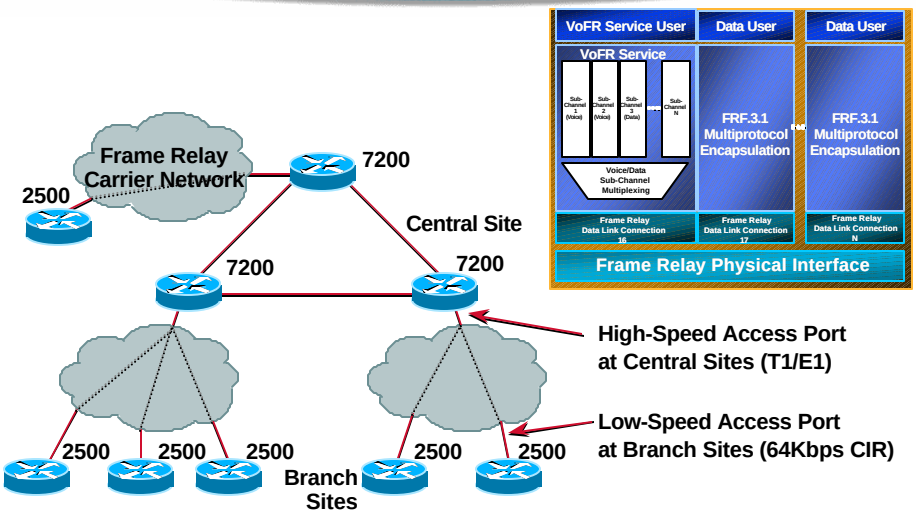
Design Trade-Offs

- TDM, frame, cell-efficiency, delay, cost
- Mesh vs star—Performance vs cost
- Transport vs translate—Signaling

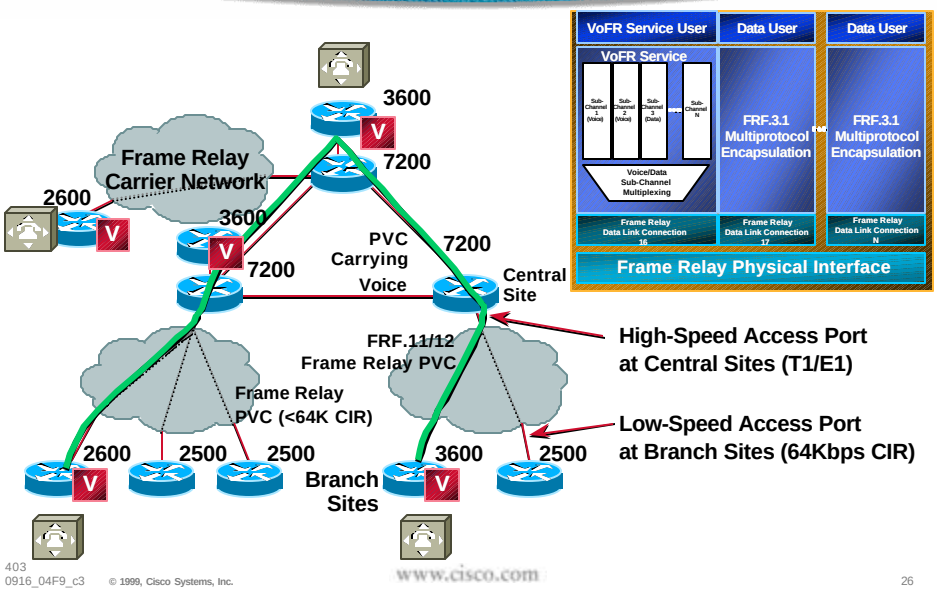
Voice Over Frame Relay Network Design

- A brief tour of FRF.11
- Why FRF.12?
- Network design considerations
- Bandwidth calculations

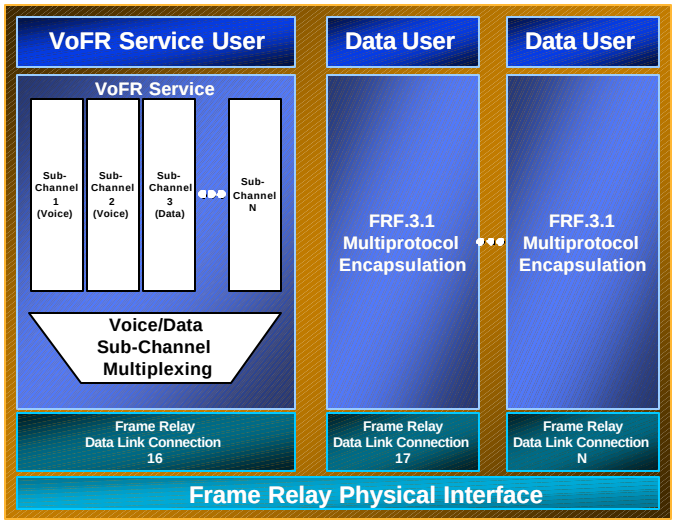
Data Over Frame Relay Network



Data/Voice Over Frame Relay

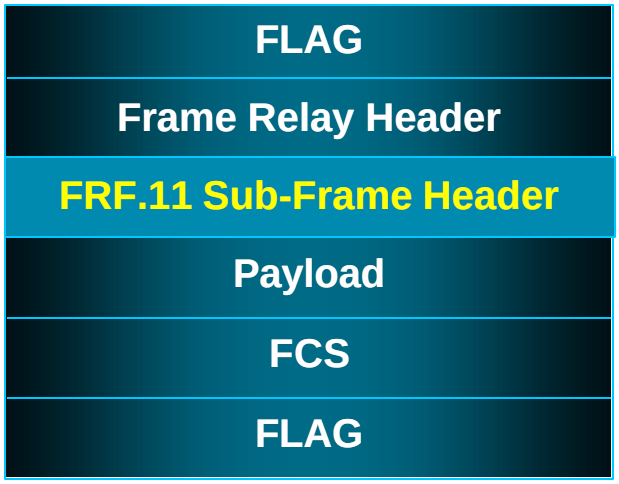


VoFR Multiplexing Model

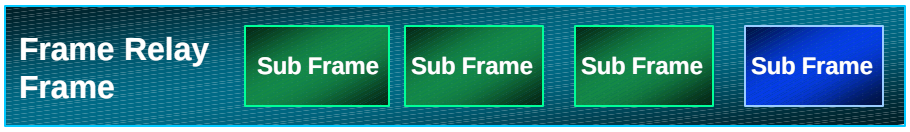


Source: Frame Relay Forum

FRF.11 Frame Format

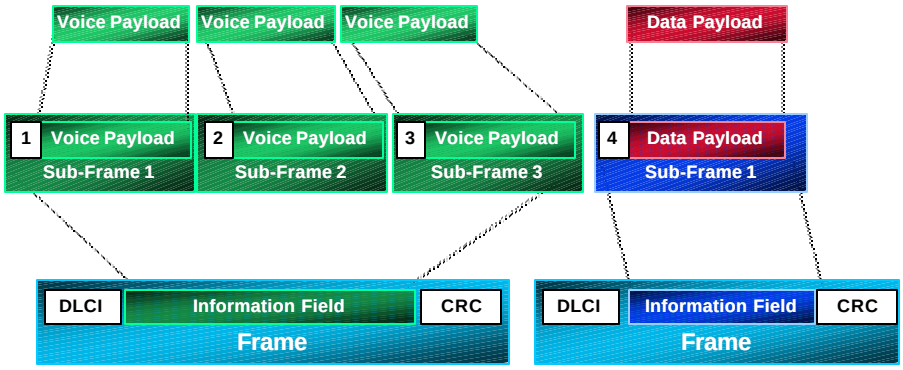


Voice and Data Encapsulation



- **Multi frames transport**

Multiple Sub-Channel Payloads in an FRF.11 Frame



Source: Frame Relay Forum

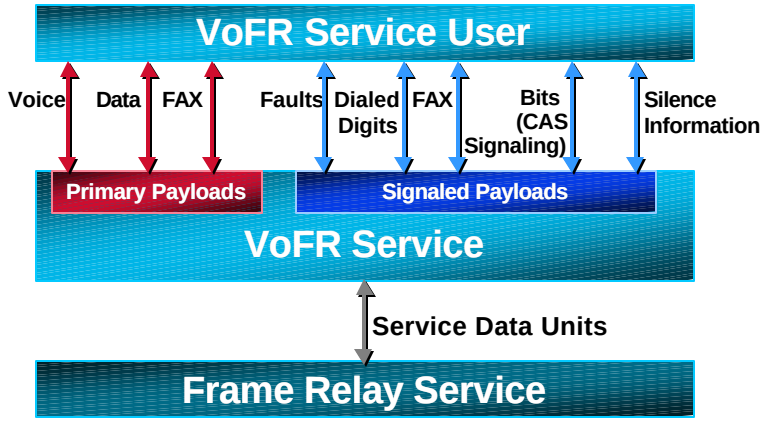
FRF.11 Concept

- Extension of frame relay application support for compressed voice
- Multiplexing of up to 255 sub-channels
- Support of multiple payloads
- Support of data sub-channel

FRF.11 Equipment Classes

- Class 1 equipment
  - Transmission equipment in high bandwidth environments
  - Requires G.727 EADPCM compliance
- Class 2 equipment
  - Optimizes performance over low bandwidth trunks

VoFR Services



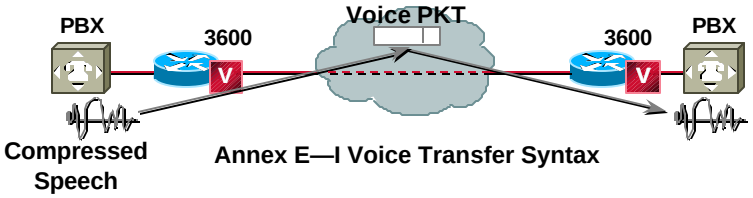
Source:
Frame Relay Forum

FRF.11 Class 2 Services

- Dialed digits transfer syntax (Annex A)
- Signaling bit transfer syntax (Annex B)
- Data transfer syntax (Annex C)
- FAX relay transfer syntax (Annex D)
- CS-ACELP transfer syntax (Annex E)
- PCM/ADPCM transfer syntax (Annex F)
- G.727 D/E EADPCM voice transfer syntax (Annex G)
- G.728 LD-CELP transfer syntax (Annex H)
- G.723.1 MP-MLQ dual rate speech coder (Annex I)

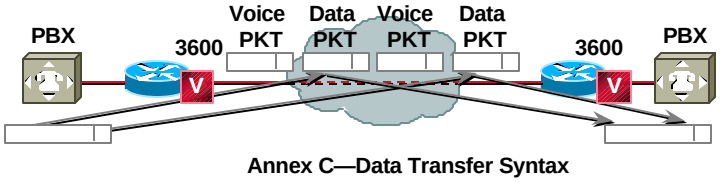
Note: Cisco Does Not Currently Support Annex G

FRF.11 Annex E, F, G, H, I: Voice Transfer Syntax



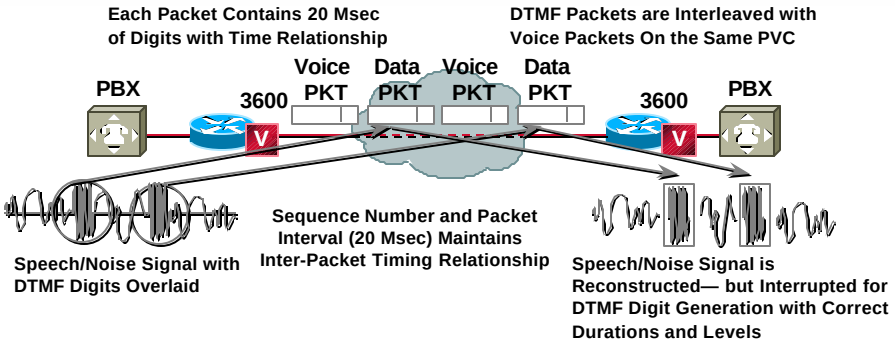
- Codec syntax specified per annex—ie; Annex E—CS-ACELP G.729
- Annex F—Generic PCM/ADPCM G.711, G.726, G.727
- Annex H—LD-CELP G.728, Annex I—G.723.1
- Sequence number and coding type are optional

FRF.11 Annex C Data Transfer Syntax



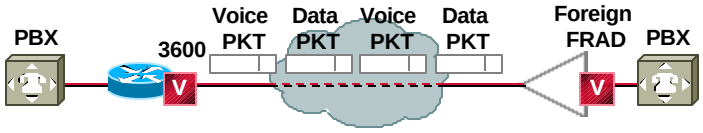
- FRF.11 sub-frame headers allow voice and data sub-channels within a PVC
- Each packet contains a whole or a fragment of an original data frame
- Original frames smaller than the fragmentation threshold are encapsulated with both the B and E bits set

FRF.11 Annex A Dialed Digit Transfer Syntax



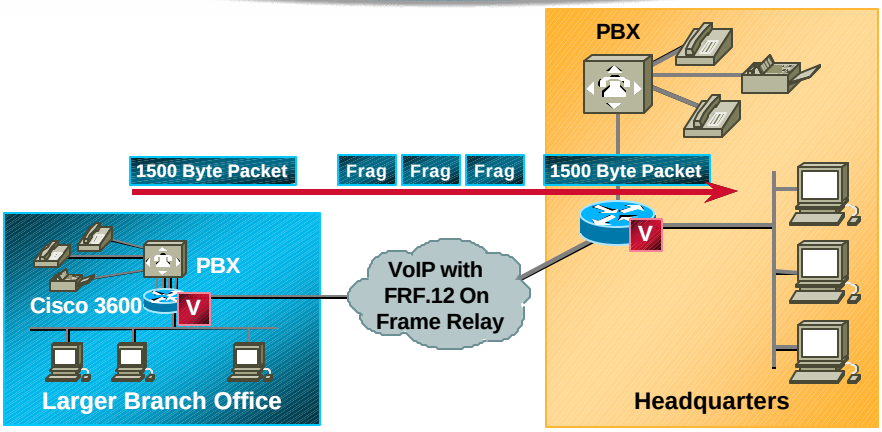
- Supports DTMF transmission for high compression Codecs
- Originating VFRAD detects DTMF and codes; destination VFRAD reproduces DTMF
- When there has been no DTMF activity for 60 msec, no more Annex A frames are generated until the next DTMF digit is detected

FRF.11 Interoperability



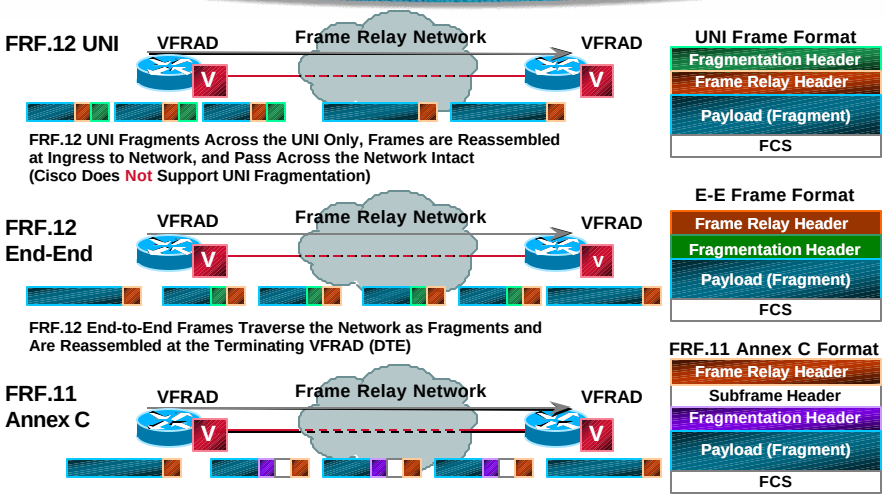
- In theory FRADs from different vendors can interoperate with FRF.11
- Your mileage may vary

Why FRF.12?



- Multiple PVCs—some with FRF3.1 data on same physical interface
- Can be used with VoIP
- Fragments and interleaves large data frames with voice packets

Comparison of FRF.11/12 Fragmentation Schemes



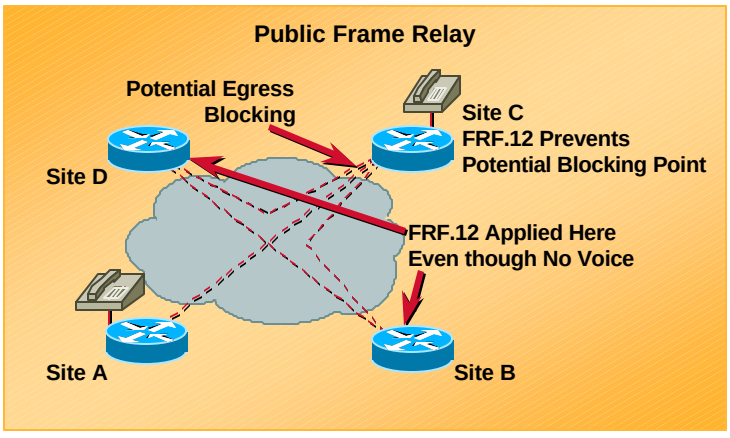
Fragmentation Frame Size Matrix

Real Time Packet Interval

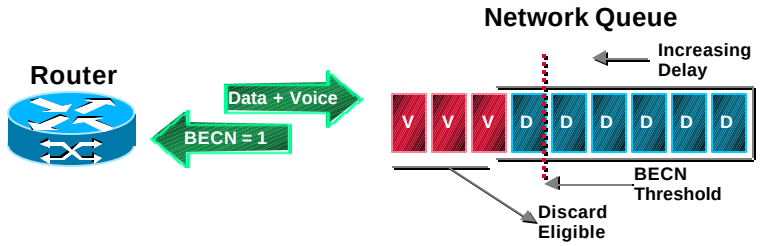
Link Speed	10ms	20ms	30ms	40ms	50ms	100ms	200ms
56kbps	70 Bytes	140 Bytes	210 Bytes	280 Bytes	350 Bytes	700 Bytes	1400 Bytes
64kbps	80 Bytes	160 Bytes	240 Bytes	320 Bytes	400 Bytes	800 Bytes	1600 Bytes X
128kbps	160 Bytes	320 Bytes	480 Bytes	640 Bytes	800 Bytes	1600 Bytes X	3200 Bytes X
256kbps	320 Bytes	640 Bytes	960 Bytes	1280 Bytes	1600 Bytes X	3200 Bytes X	6400 Bytes X
512kbps	640 Bytes	1280 Bytes	1920 Bytes X	2560 Bytes X	3200 Bytes X	6400 Bytes X	12800 Bytes X
768kbps	1000 Bytes	2000 Bytes X	3000 Bytes X	4000 Bytes X	5000 Bytes X	10000 Bytes X	20000 Bytes X
1536kbs	2000 Bytes X	4000 Bytes X	6000 Bytes X	8000 Bytes X	10000 Bytes X	20000 Bytes X	40000 Bytes X

X—Fragmentation not an issue due to BW + Interval Combination

Egress Blocking/Fragmentation

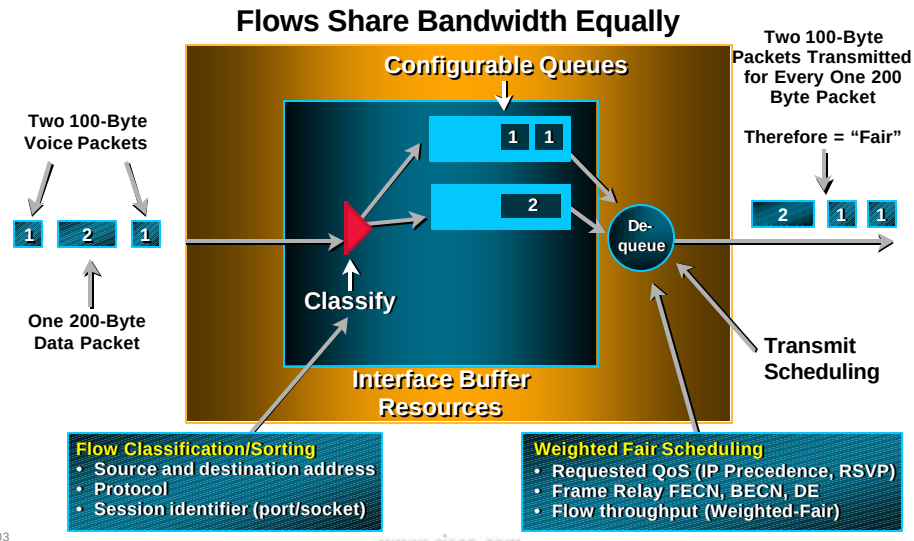


VoFR Network Design Considerations

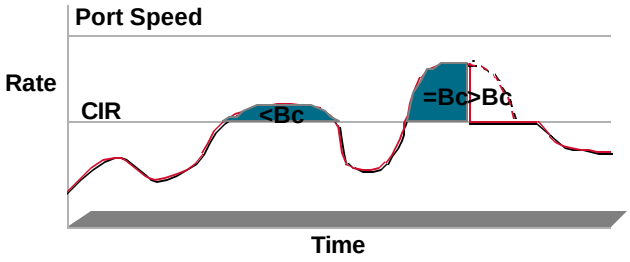


- If network queues are filling…
  - Voice may experience excessive delay
  - Voice may be discarded
  - Router throttles back on BECN—but it's too late—queues are full and voice packets are dropped
- Voice quality may be negatively affected

VoFR Design Considerations—Queuing (WFQ)

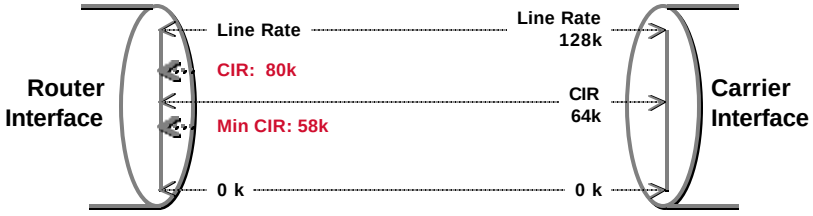


Frame Relay Traffic Shaping



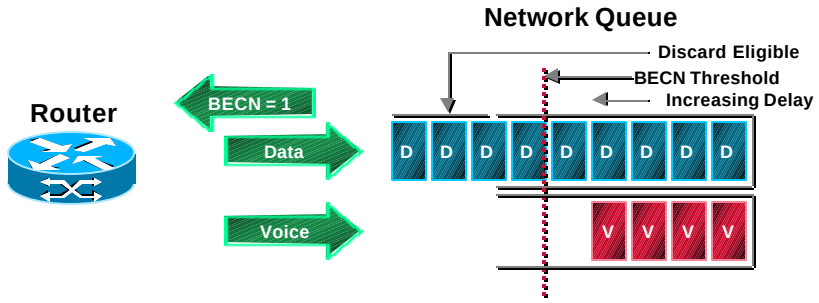
- Traffic shaping provides voice and data quality of service
- Allows better utilization of CIR by shaping data

Single PVC Design Considerations



- Goal is best voice quality, NOT data volume
- Contract CIR to meet calculated requirements
- Shape conservatively
- Have carrier set BECN threshold to a smaller setting
- Set traffic shaping parameters for finest granularity

Multiple PVC Design Considerations



- Separate queues maintain voice quality
  - More deterministic—voice won't burst
- Priority PVCs from Carrier for Voice

Calculating VoFR Bandwidth

- Assumptions
- G.729 Codec at 8Kbps
- 50 PPS (using 2–10ms samples)
- 2 bytes of DLCI header
- 2 bytes of FRF.11 header
- 1 byte of sequence number
- 2 byte CRC

Calculating VoFR Bandwidth

- Voice payload calculation
  - 20 Msec voice sample * 8 Kbps (for G.729)/ 8 bits/byte = 20 bytes
  - Note: to derive the payload for G.711, substitute 64 kbps = 160 bytes
- Packet size calculations
  - 20 byte payload + 7 byte Header = 27 bytes (Header = DLCI/FRF.11/seqn/CRC)
- Bandwidth calculations
  - 27 b/voice packet * 8 bits/byte * 50 pps = 10.8 Kbps per call

CIR Critical Factors

- PVC design
  - Full mesh vs star
  - Shared vs separate PVCs for voice and data
- Potential concurrent calls
  - Bandwidth per call
  - Switched through calls
- Pre-existing data environment
  - Utilization prior to adding voice

VoFR Summary

- FRF.11 standards-based voice and function syntax
- FRF.12 standards-based fragmentation for data, mitigates delay and delay variation
- Proper PVC design for network requirements
- Balance voice quality, delay, bandwidth, CIR

VoFR Summary

- Avoid Tandem (multiple) conversions
- Calculate delay for chosen design
- Check for Egress blocking
- Calculate CIR for total potential voice and/or data per PVC
- Configure queuing and traffic shaping

References

- [1] FRF.3.1, R. Cherukuri (ed), Multiprotocol Encapsulation Implementation Agreement, June 22–1995
- [2] FRF.9, D. Cantwell (ed), Data Compression Over Frame Relay Implementation Agreement, January 22–1996
- [3] FRF.11.1 K. Rehbehn, R. Kocen, T. Hatala (eds), Voice Over Frame Relay Implementation Agreement, December 1998
- [4] FRF.12, A. Malis (ed), Frame Relay Fragmentation Implementation Agreement, 1997
- [5] ITU Recommendation Q.922, ISDN Data Link Layer Specification for Frame Mode Bearer Services, 1992

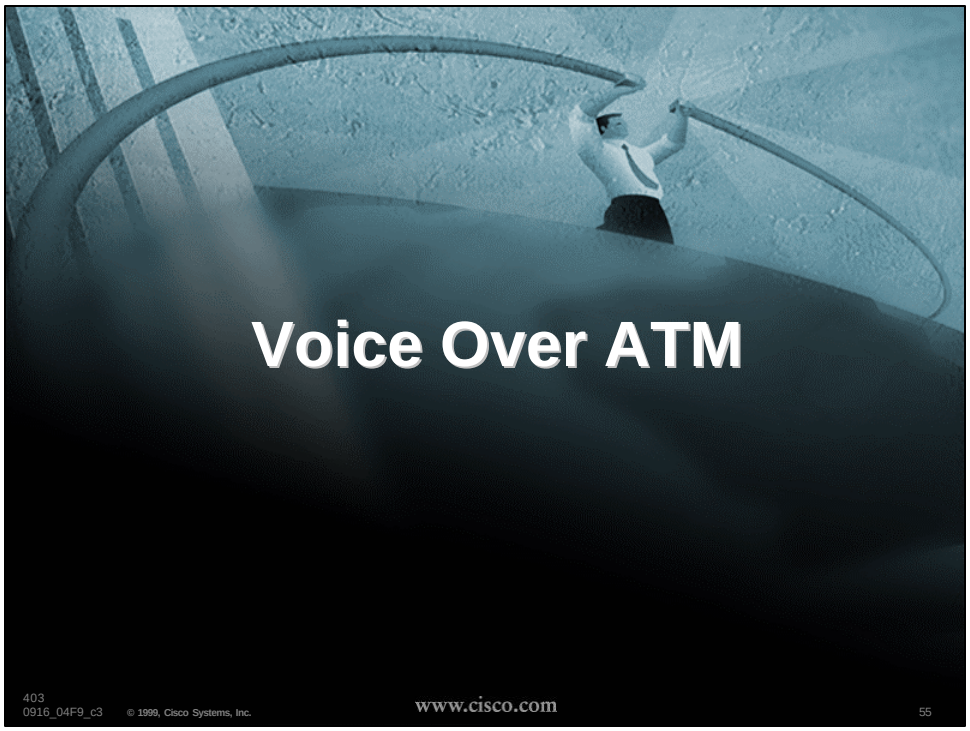
Web Sites

- **Cisco**

  http://www.cisco.com—search on VoFR

- **Frame Relay Forum**

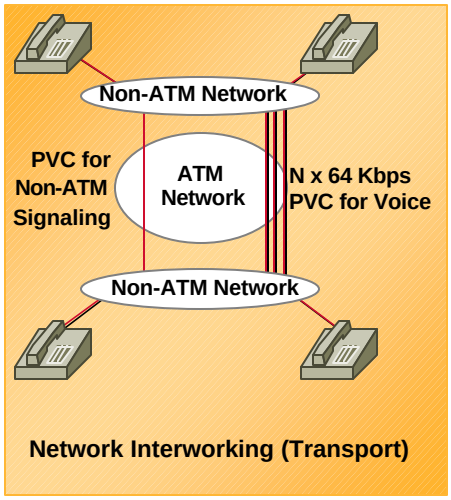
  http://www.frforum.com/

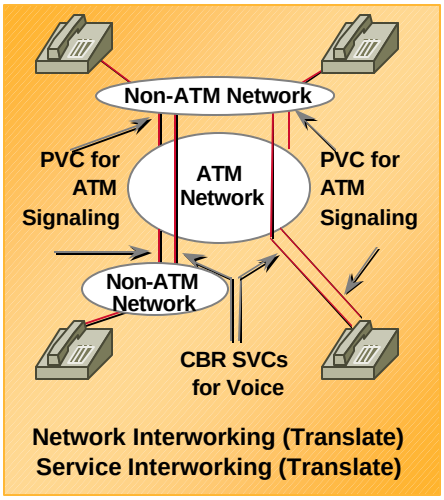


Voice Over ATM Network Design

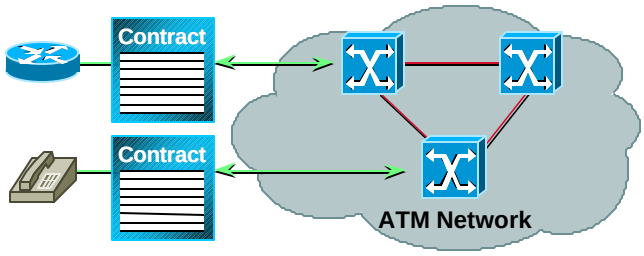
- ATM reference model
- ATM service categories
- Circuit emulation
- Packetized voice
- Network designs
- Bandwidth calculations

VoATM Signalling



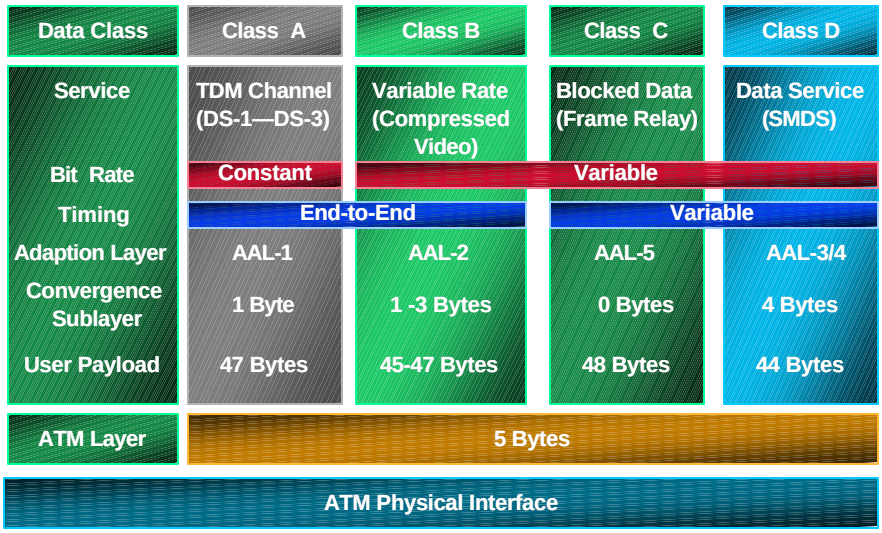


ATM Service Criteria

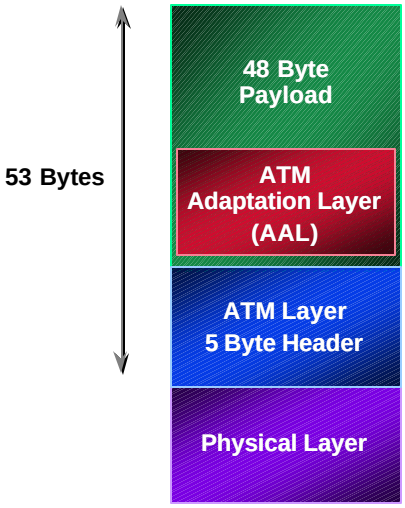


- **Contract**
  - Traffic descriptors
    - Peak cell rate
    - Sustainable cell rate
    - Maximum burst size
    - Minimum cell rate
  - Quality of service
    - Delay
    - Cell loss

ATM Protocol Model

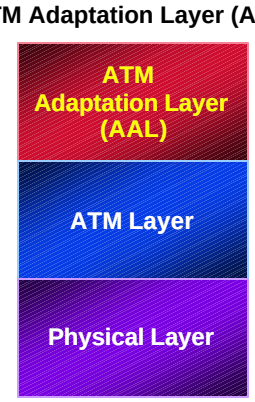


ATM Cell

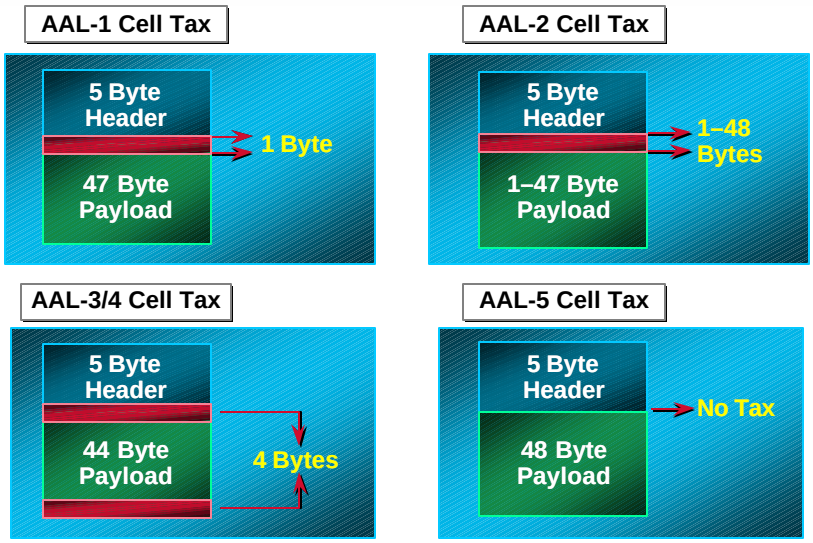


ATM Reference Model

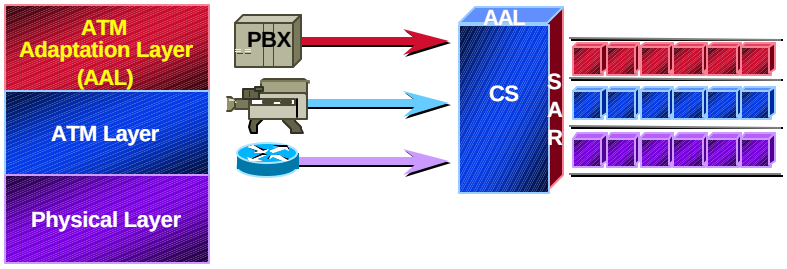
- **Two sublayers**
  - **Convergence Sublayer (CS)**
  - **Segmentation and Reassembly (SAR)**



AAL Cell Tax



ATM Adaptation Layer—AAL



- AAL = CS + SAR
- CS—cell tax
- SAR—cell <-> packet

ATM Classes of Service

- Service criteria
  - Traffic descriptors
  - QoS parameters
- Service categories
  - Constant Bit Rate (CBR)
  - Variable Bit Rate (VBR)
  - Unspecified Bit Rate (UBR)
  - Available Bit Rate (ABR)

ATM Service Criteria

Traffic Parameters

- **Peak Cell Rate—PCR—Maximum data rate a connection can handle without losing data**
- **Sustainable Cell Rate—SCR—Average ATM cell throughput the application is permitted**
- **Maximum Burst Size—MBS—Size of the maximum burst of contiguous cells that can be transmitted**
- **Minimum Cell Rate—MCR—Rate of an application's ability to handle latency**

ATM Service Criteria

QoS Parameters

- **Maximum Cell Transfer Delay—MCTD how long the network can take to transmit a cell from one endpoint to another**
- **Cell Delay Variation Tolerance—CDVT line distortion caused by change in interarrival times between cells aka jitter**

ATM Service Criteria

QoS Parameters

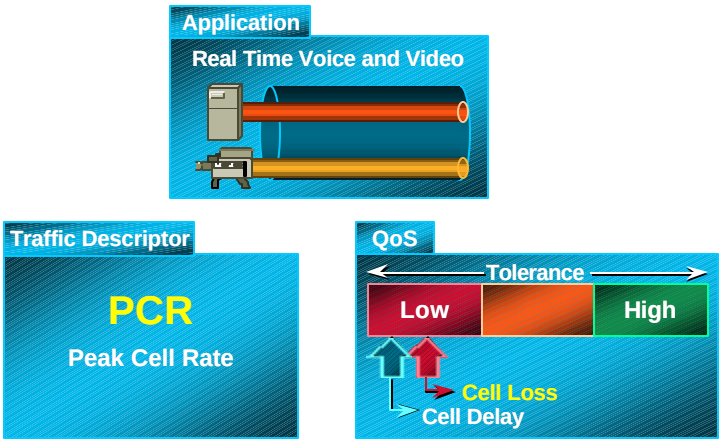
- Cell Loss Ratio—CLR acceptable percentage of cells that the network can discard due to congestion

ATM Classes of Service

- Service criteria
  - Traffic descriptors
  - QoS parameters
- Service categories
  - Constant Bit Rate (CBR)
  - Variable Bit Rate (VBR)
  - Unspecified Bit Rate (UBR)
  - Available Bit Rate (ABR)

ATM Service Categories

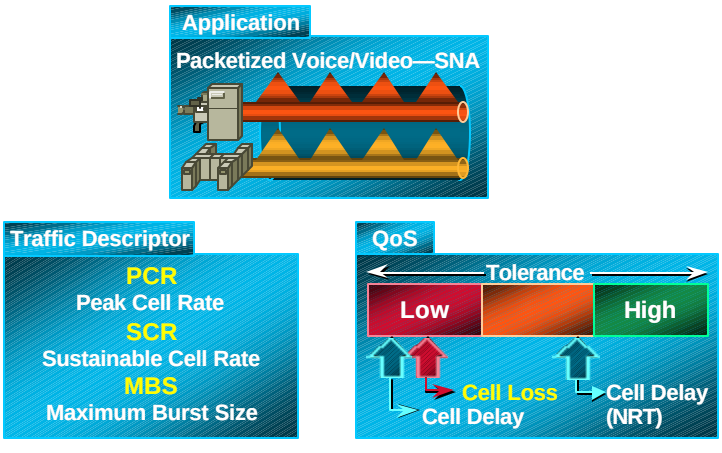
Constant Bit Rate (CBR)



403
0916_04F9_c3 © 1999, Cisco Systems, Inc. www.cisco.com 69

ATM Service Categories

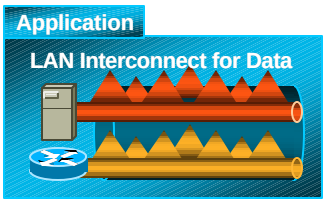
Variable Bit Rate (VBR-RT/VBR-NRT)

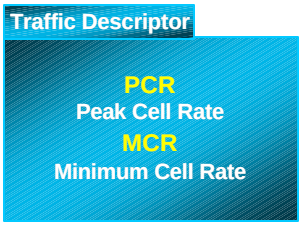


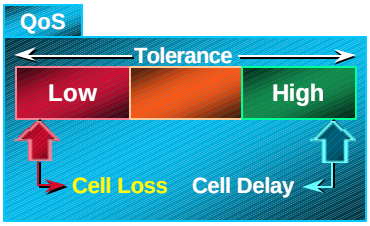
403
0916_04F9_c3 © 1999, Cisco Systems, Inc. www.cisco.com 70

ATM Service Categories

Available Bit Rate (ABR)



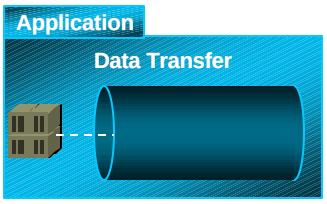


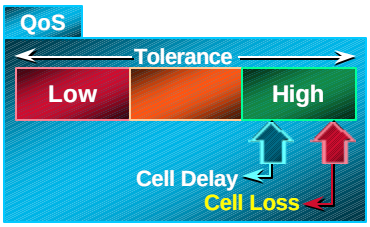


Also Uses Congestion Feedback Mechanisms

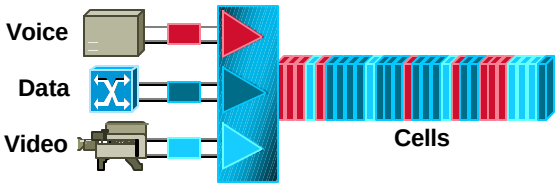
ATM Service Categories

Unspecified Bit Rate (UBR)



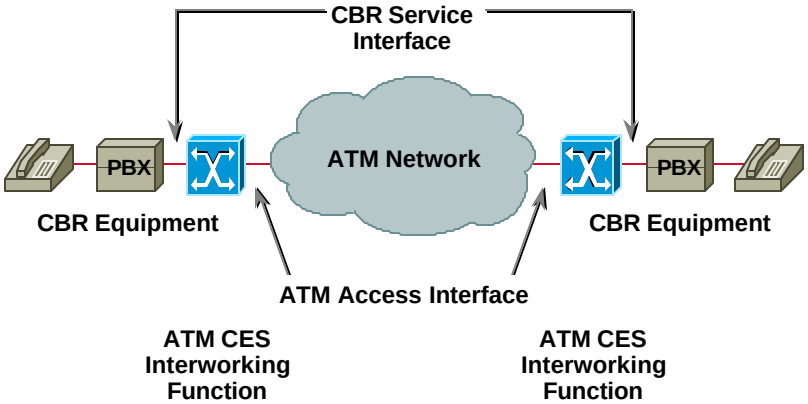


Characteristics of ATM

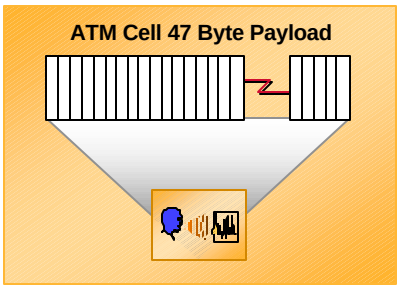


- Uses small—fixed-sized cells
- Connection-oriented
- Supports multiple service types
- Applicable to LAN and WAN

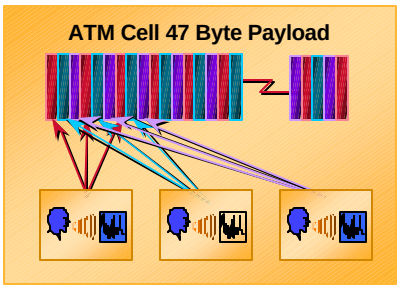
CES Reference Model



ATM Voice Transport Choices



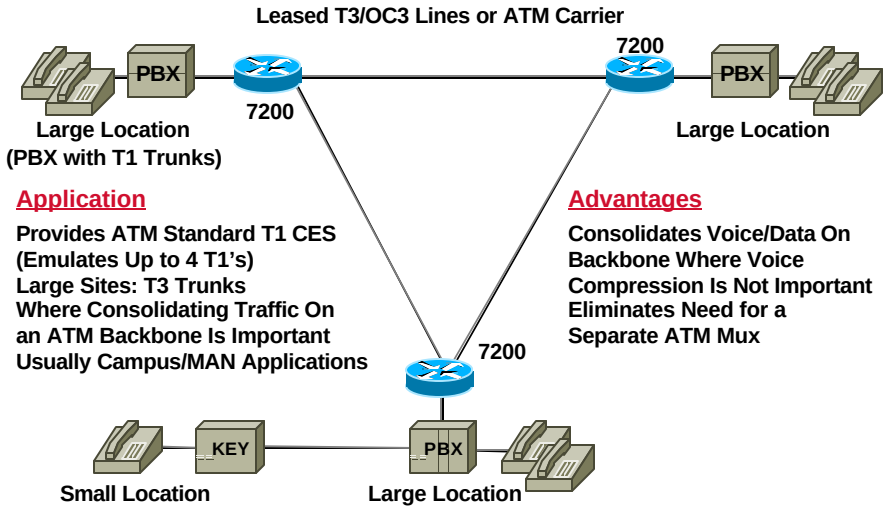
- Unstructured circuit emulation service
- Entire payload filled with one voice channel
- Or a continuous stream of bytes from an E1 circuit is placed into cells



- Structured circuit emulation service
- Entire payload filled with many voice channels
- DS0 structure information is maintained

Note: AAL1 Uses One Byte for Overhead

CES Voice Network



Application

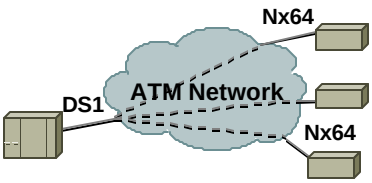
Provides ATM Standard T1 CES (Emulates Up to 4 T1's)
Large Sites: T3 Trunks Where Consolidating Traffic On an ATM Backbone Is Important
Usually Campus/MAN Applications

Advantages

Consolidates Voice/Data On Backbone Where Voice Compression Is Not Important
Eliminates Need for a Separate ATM Mux

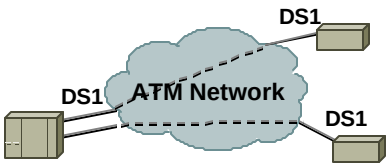
Structured vs Unstructured CES

Structured



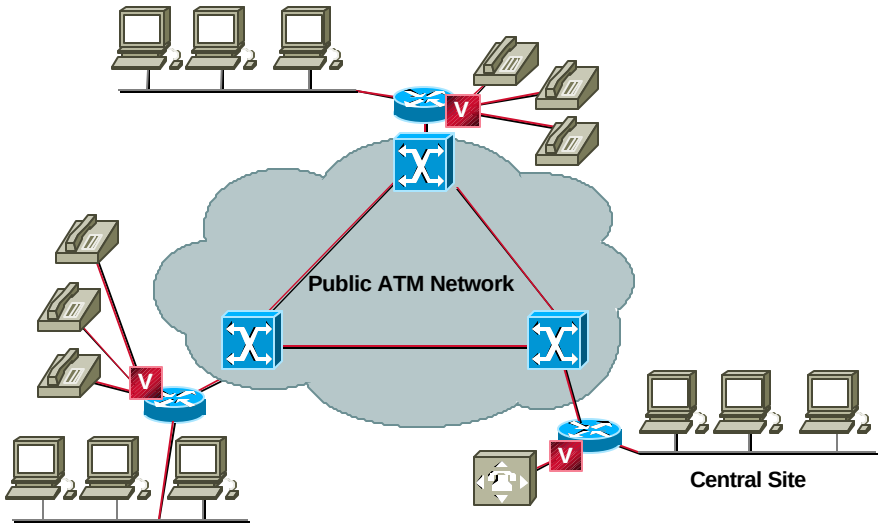
- Intended to emulate point-to-point fractional DS1 or E1 circuit
- Allows Nx64 Kbps independent emulated circuits to share one DS1
- Can be configured to minimize ATM bandwidth

Unstructured

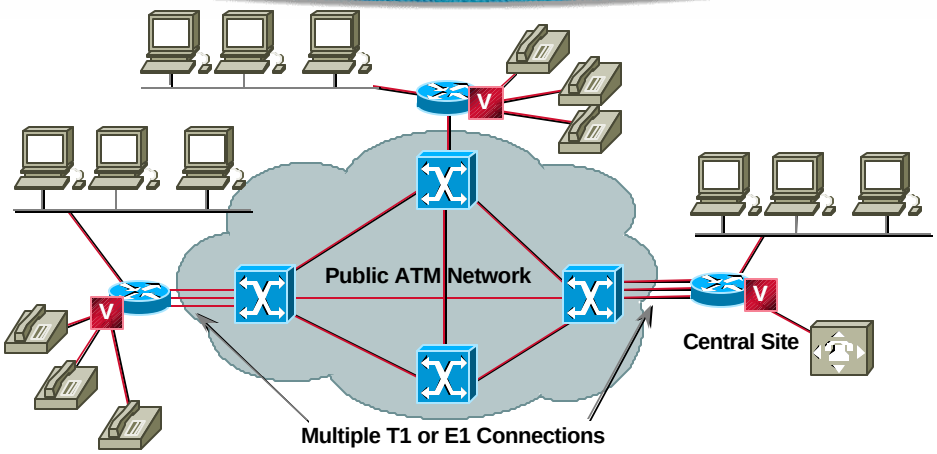


- Intended to emulate point-to-point DS1 or E1 circuit
- Allows one 1.54 or 2.04 Mbps emulated circuit per DS1
- Can be used with equipment with non-standard framing
- Allows simple configuration of service

Data/Voice Over ATM (AAL5)

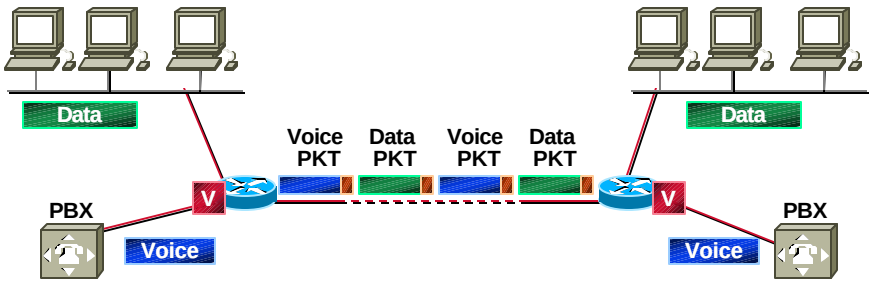


Inverse Multiplexing (IMA)



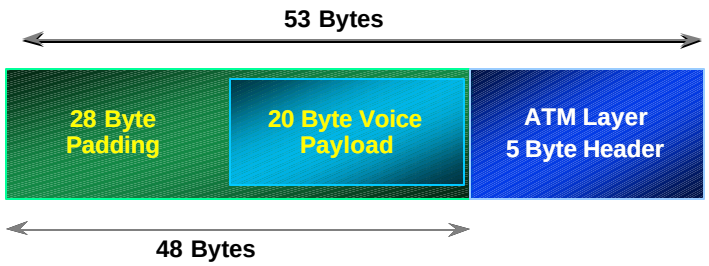
- Uses AAL5; 48 bytes for data
- Binds multiple PVCs to a single logical interface
- Prioritizes voice cells over data cells

ATM AAL5 Voice and Data Cells



- AAL 5 does not require convergence sub-layer
- 48 Byte payload available for voice/data
- Voice payload = voice sample + padding = 48 bytes
- 5 byte ATM header

ATM AAL5 Voice Cells

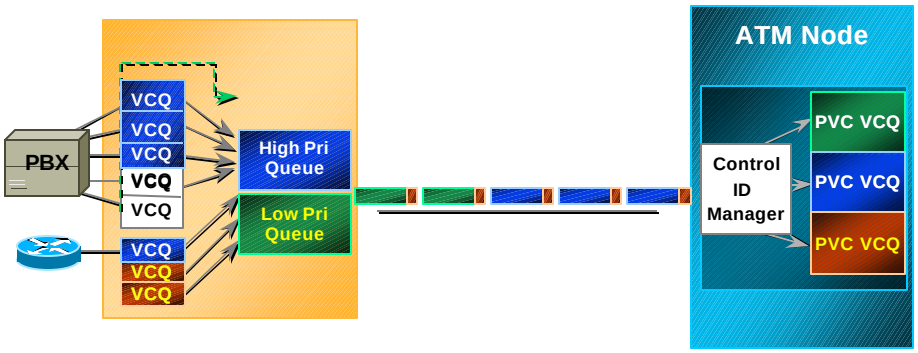


- G.729 compression with 20 ms voice sample
- No AAL5 CS “cell tax”
- 28 Bytes “overhead” due to padding

VoATM Bandwidth

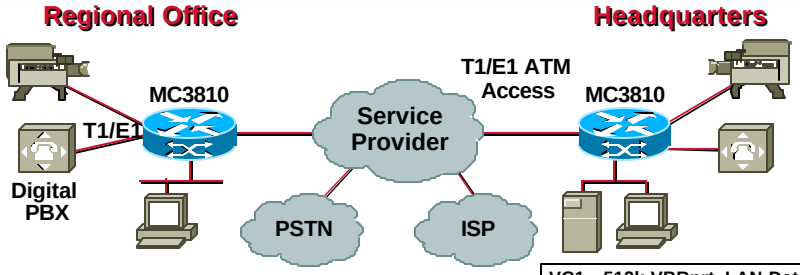
- Voice payload calculation
  - 20 msec voice sample * 8 Kbps (for G.729)/8 bits/byte = 20 bytes
  - Note: to derive the payload for G.711, substitute 64 Kbps = 160 bytes
- Packet size calculations
  - 20 byte payload + 28 byte pad +5 byte header = 53 bytes
- Bandwidth calculations
  - 53 b/voice packet * 8 bits/byte * 50 pps = 21.2 Kbps per call

Priority Queuing



Quality of Service (QoS)
Extends Across Entire Network

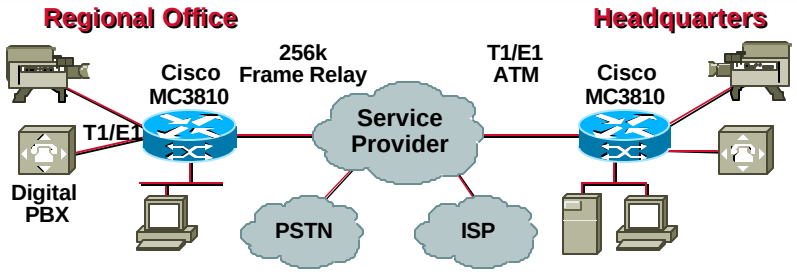
Regional Office—ATM Over T1/E1



VC1—512k VBRnrt, LAN Data
VC2—256k VBRrt, 24 Voice Calls
VC3—768k CBR, Video
VC4—VC8 80k CBR, Voice Circuits

- LAN data
- Internet connection
- Voice traffic
- Video application
- C.O. circuit via CES (future)

Frame Relay/ATM Interworking



- Network interworking
  - FRF.5
  - Frame Relay encapsulation
- Service interworking compatible
  - FRF.8
  - Carrier compatible

VoATM—Summary

- ATM reference model
- Fixed size cells—Delay
- Service category—CBR, VBR, ABR
- Service criteria for QoS, SCR, CDVT
- Chose service for requirements—Circuit emulation (AAL1) voice over AAL5
- Combined networks

References

ATM Forum af-phy-0016.000	DS1 Physical Layer Specification
ATM Forum af-phy-0064.000	E-1 Physical Layer Specification
ATM Forum af-saa-0032.000	Circuit Emulation
ATM Forum af-uni-0010.002	UNI Signaling 3.1
IETF RFC1483	Multiprotocol Over ATM
IETF RFC1577	IP Over ATM
IETF RFC1695	AToM MIB
ANSI T1.630—ITU I.363—I.363.	ATM AAL 1 (Constant Bit Rate)
ANSI T1.635—ITU I.363.5	ATM AAL 5 (Variable Bit Rate)

Web Sites

- **Cisco**

  http://www.cisco.com

- **ATM Forum**

  http://www.atmforum.com/