Deploying QoS for Voice and Video in IP Networks
Session VVT-213

Session Objectives

- To be able to design and implement an AVVID infrastructure that can guarantee voice quality while enabling video conferencing, streaming video and mission critical data applications
- Presentation follows the IP telephony QoS design guide on CCO
- All designs based on:
  Cisco CallManager 3.1 and above
  CatOS 5.5(8) and Cat IOS 12.1(2)E and above
  IOS 12.1(2)T-12.1(5)T and above
The AVVID Design Model

The OSI Stack Revisited

Application Layer

Scalable Call Control Layer

Highly Available, QoS-Enabled Infrastructure Layer

3 Steps for CoS/QoS Implementation

- **Classification**—Marking the packet with a specific priority denoting a requirement for special service from the network
- **Scheduling**—Assigning packets to one of multiple queues (based on classification) for expedited treatment through the network
- **Provisioning**—Accurately calculating the required bandwidth for all applications plus element overhead
QoS is Needed to Minimize Packet Loss, Delay and Delay Variation

Where QoS Is Needed

Central Campus

QoS—Campus Access
- Speed and Duplex Settings
- Classification/trust on IP Phone and Access Switch
- Multiple Queues on IP Phone and Access Ports

QoS—Campus Distribution
- Layer 3 Policing
- Multiple Queues on All Ports; Priority Queuing for VoIP
- WRED Within Data Queue for Congestion Management

QoS—WAN
- Low-latency Queuing
- Link Fragmentation and Interleave
- Bandwidth Provisioning
- Admission Control

QoS—Branch
- Classification and Trust Boundaries on IP Phone, Access Layer Switch and Router
- Multiple Queues on IP Phone and All Access Ports

Remote Branch

Agenda

- Quality Concerns with IP Telephony and Multimedia Applications
- General AVVID QoS Design Guidelines
- Connecting the IP Phone
- Designing the Campus
- Enabling the WAN
- Managing the QoS Infrastructure
Factors which Degrade Voice Quality

Packet Loss

- Packet loss
  
  Current Cisco GW DSP CODEC algorithms can correct for 30 msec of lost voice—1 G.729A voice packet contains 20 msec of voice
  
  Lost packets induce “clipping” and temporarily expand the jitter buffer, which increases end-to-end latency
  
  One lost FAX over IP packet causes a MODEM retrain; 2 drops cause a call disconnect
  
  Causes of packet loss: Network quality, network congestion and delay variation (jitter buffer under-runs)

Factors which Degrade Voice Quality

Variable Delay—Jitter Buffer Under Runs

Cisco GW DSPs Uses an Adaptive Jitter Buffer Which
Only Has 10 msec of “Extra” Buffer
Packet Dropped If Instantaneous Jitter Is > 10 msec

50ms of possible Jitter Buffer

<=10ms
Calculated Jitter Buffer Based on Variable Network Delay in msec (packet RTP Timestamp)
Factors which Degrade Voice Quality

End-to-End Delay

ITU G.114 states one-way delay $\leq 150$ msec  ~200 msec is acceptable

- **CODEC**
  - G.729A = 25 msec (20msec+5msec look ahead)

- **Queuing**
  - Queuing delay = serialization delay as utilization approaches 100%

- **Serialization**

- **Propagation and network delay**
  - 6.3 usec/km + network delay (variable)

- **Jitter buffer**
  - 20-50 msec

Example of PCM (64Kbps)
IP Telephony Call

- Consistent, easily managed packet rate
- A G.711 call is really 80Kbps over a data network
- Layer 2 overhead not included
- VAD/silence suppression is not enabled in this example
Factors which Degrade Video Conferencing Quality

- Unlike voice, video has a very high, extremely variable packet rate
- Much higher average MTU
- Queuing
  - The LLQ will fill to capacity regularly
  - Queuing delay = serialization delay as utilization approaches 100%
- End-to-end delay
  - 200 msec target delay budget
- Jitter buffer
  - 20-70 msec

Example of 384 Kbps Video (30 fps) Conferencing Traffic (CIF)

- “I” frame is a full sample of the video
- “P” and “B” frames use quantization via motion vectors and prediction algorithms

“I” Frame

- 1024-1518 Bytes

“I” Frame

- 1024-1518 Bytes

“P” and “B” Frames

- 128–256 Bytes

600Kbps

32Kbps
Video Conferencing Traffic Packet Size Breakdown (CIF)

- 65–128 Bytes: 1%
- 1025–1518 Bytes: 37%
- 129–256 Bytes: 34%
- 513–1024 Bytes: 20%
- 257–512 Bytes: 8%
- 1025–1518 Bytes: 20%

Video Conferencing Traffic Packets per Second Breakdown (CIF)

- Low PPS
- High PPS
Factors which Degrade Streaming Video Quality

• Has a very high, extremely variable packet rate
• Much higher average MTU
• Queuing
  Because of the tolerance for e-2-e delay, streaming video should go into a bw-based queue
• End-to-end delay
  4–5 secs
• Jitter buffer
  1 MB (read long latency tolerance)

Streaming Video Traffic Packet Size Breakdown (MPEG-1)

1025–1518 Bytes 54%
65–128 Bytes 16%
129–256 Bytes 1%
257–512 Bytes 13%
513–1024 Bytes 16%
Agenda

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• General AVVID QoS Design Guidelines

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Layer 2 Class of Service

Three Bits Used for CoS
802.1p bits (802.1D User Priority)

Note: IP Phones Do Not Support ISL
Layer 3 Type of Service

Layer 3 Type of Service

Layer 3
IPV4

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<th>Version</th>
<th>Length</th>
<th>ToS 1 Byte</th>
<th>Len</th>
<th>ID</th>
<th>Offset</th>
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<th>Proto</th>
<th>FCS</th>
<th>IP-SA</th>
<th>IP-DA</th>
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<td>4</td>
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IP Precedence

DSCP

Unused Bits; Flow Control for DSCP

Standard IPV4: Three MSB Called IP Precedence (DiffServ May Use Six D.S. Bits Plus Two for Flow Control)

Differentiated Services Code Point (DSCP)

- Fundamentally, just a way to classify, or differentiate traffic
- A number placed in the IP header to assist in isolating a class of traffic
- Occupies 6 bits out of what used to be the TOS byte
- The other two bits are for Explicit Congestion Notification (ECN)
- Typically used in conjunction with a Per-Hop Behavior (PHB)
  E.g., RFC (EF PHB-2598 or AF PHB-2597)
Defined Per-Hop Behaviors

- **EF**
  EF PHB (Expedited Forwarding—RFC 2598) can be used to build a low loss, low-latency, low jitter, assured bandwidth, end-to-end service
  “Virtual leased-line”

- **AFxy**
  AF PHB (Assured Forwarding—RFC 2597) gives domains the ability to offer different levels of traffic forwarding assurance
  x = 4 AF classes are defined (AF1y-AF4y)
  y = 3 drop preferences/probabilities per class

- **Best effort**

Diff-Serv Behaviors

<table>
<thead>
<tr>
<th>Per-Hop Behaviours (PHB)</th>
<th>DiffServ Code Points (DSCP)</th>
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<td>Assured Forwarding</td>
<td>Low Drop Pref</td>
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<td>AF11</td>
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<td>Class 2</td>
<td>AF21</td>
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<tr>
<td>Class 3</td>
<td>AF31</td>
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<td>Class 4</td>
<td>AF41</td>
</tr>
<tr>
<td>Best Effort</td>
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</table>
Cisco AVVID Classification

Voice with the “Encore” CallManager Release

- Voice
  - VolP control channels
    - CoS = 3, IP Prec = 3, DSCP = AF31
    - H.323 = TCP 1720, 11xxx (RAS = TCP 1719) 12.2(1)T
    - Skinny = TCP 2000-200 CCM 3.0(5)
    - ICCP = TCP 8001-8002 CCM 3.0(8)
    - CTI (TAPI/JTAPI) = TCP 2748
    - MGCP = UDP 2427, TCP 2428 CCM 3.1
  - VolP RTP bearer channels
    - CoS = 5, IP Prec = 5, DSCP = EF
    - UDP 16384-32767 CCM 3.0+ and IOS 11.3+

Cisco AVVID Classification (Cont.)

Video Classification

- Video
  - Video conferencing
    - CoS = 4, IP Prec = 4, DSCP = AF41
    - RAS = TCP 1719, H.323 = TCP 1720 & 11xxx, UDP = Depends...
  - Streaming video (IP/TV)
    - CoS = 1, IP Prec = 1, DSCP = AF13—Recommended for enterprises
    - UDP = IP/TV 3.2 provides customer port configuration
  - 3rd party video partners
    - VCON—Can set ToS
      http://techsup.vcon.com/Docs/ToS-setting%20utility.doc
    - PictureTel—Can set ToS
    - Polycom—Can set ToS
    - RadVision—Can not set ToS
Cisco AVVID Classification, Cont.

Data Application Classification

• Data
  
  \[
  \text{CoS = 0-2, IP Prec = 0-2, DSCP = 0-AF23}
  \]
  
  Some data applications may need special handling from the network
  
  This can be for business, technical or Layer 8 reasons

• Recommendations
  
  Only classify when necessary
  
  Modifying WRED thresholds may be required to insure performance
  
  For a CoS/ToS = 2 applications, configure queue #1’s 2nd threshold (CoS/ToS = 2) to drop at 95% instead of 50%

Connecting the IP Phone

General Guidelines

• Use auto-negotiation on the wiring closet switch port and PC NIC
• Separate all voice traffic onto a voice specific subnet
• Use portfast to decrease IP phone boot time
• IP Phone VoIP RTP bearer traffic will use CoS/ToS=5/EF
• Classify all VoIP control traffic to CoS/ToS=3/AF31
• Extend and enforce trust boundary at IP phone (set port qos <mod/port> trust-ext ___); never allow PC applications to send traffic at CoS/ToS 5-7 except SoftPhone
Designing the Campus

General Guidelines

- A robust, modern switching design is a requirement
- Multiple queues are required on all interfaces to guarantee voice quality
  2900 XL (8 MB DRAM), 3500 XL, 4000, 6000
- Catalyst 5000 designs should use a separate path for voice traffic
- Voice RTP bearer traffic should always go into the highest priority queue; video and voice call control should go into queue #2 regardless of device
- Distribution layer switches must have the ability to map between CoS and ToS values

Building the Branch Office

General Guidelines

- The WAN branch router MUST support advanced Cisco QoS tools
- Use 12.2(1)T* in the router to map between layer 2 and layer 3 classification schemes
- Use a branch switch with multiple queues
- 802.1Q trunking between the router and switch for multiple VLAN support (separation of voice/data traffic) is preferred
Enabling the WAN

General Guidelines

- Use LLQ on all WAN interfaces in an AVVID network
  - Voice (DSCP=EF) → LLQ
  - Video conferencing (DSCP=AF41) → LLQ (conferencing)
  - Call control (DSCP=AF31) → CBWFQ (minimum 8Kbps)
  - Streaming video (DSCP=AF13) → CBWFQ (Kbps depends upon IP/TV policy)
- Use LFI on WAN connections below 768Kbps
  - Don’t use LFI on any video over IP solutions
- Traffic shaping is required for all frame-relay and ATM/FR networks
- Use cRTP carefully; pay attention to the IOS and interface caveats
- Call admission control is required when the number of calls can overwhelm the provisioned LLQ (PQ) bandwidth

Agenda

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- Managing the QoS Infrastructure
Cisco IP Phone Port Speed and Duplex

Single Cable

- Cisco IP phone ports auto-negotiate; not user configurable until CCM 3.1(x); best practice: set switch port to auto-negotiation
  - cat6k-access> (enable) set port speed 5/1-48 auto
  - cat3500(config)# interface FastEthernet0/1
cat3500(config-int)# speed auto
- If the PC NIC or switch is configured to full/100, the phone will negotiate to half/100
Connecting the IP Phone

Catalyst 4000 and 6000—Single Cable

Auxiliary VLAN = 110

PC VLAN = 10

802.1Q trunk with 802.1p
Layer 2 CoS

Native VLAN (PVID); no configuration changes needed on PC

Catalyst 6000

IP Phone 10.1.110.3

Desktop PC 171.1.10.3

- cat6k-access> (enable) set vlan 10 name 171.1.10.0_data
- cat6k-access> (enable) set vlan 110 name 10.1.110.0_voice
- cat6k-access> (enable) set vlan 10 5/1-48
- cat6k-access> (enable) set port auxiliaryvlan 5/1-48 110
- cat6k-access> (enable) set port speed 5/1-48 auto
- cat6k-access> (enable) set port host 5/1-48

Catalyst 3500—Single Cable

Auxiliary VLAN = 112

PC VLAN = 10

802.1Q trunk with 802.1p
Layer 2 CoS

Native VLAN (PVID); no configuration changes needed on PC

Catalyst 3500

IP Phone 10.1.112.3

Desktop PC 171.1.12.3

interface FastEthernet0/1
switchport trunk encapsulation dot1q
switchport trunk native vlan 12
switchport mode trunk
switchport voice vlan 112
speed auto
spanning-tree portfast
vlan database
vlan 112

interface FastEthernet0/1
switchport trunk encapsulation dot1q
switchport trunk native vlan 12
switchport mode trunk
switchport voice vlan 112
speed auto
spanning-tree portfast
vlan database
vlan 112
Queuing on the IP Phone

- PQ is for CoS=5 Flows and BPDUs
- Round-Robin Scheduling with a PQ Timer
- Access Layer switch RX queue can be: FIFO, 1Q4T or 1P1Q4T (Future)

PC CoS Settings Are Not Trusted

Default—Recommended

```
cat6k (enable)# set port qos 2/1 trust-ext untrusted

untrusted: Phone
ASIC Will Re-
Write CoS = 0
```
Port Trust on the Catalyst 6000

- **set port qos** `<mod/port>` **trust-ext ____**
  
  Only applies to port trust on the IP phone PC ethernet port
  
  Un-related to actual cat6k port trust

- **set port qos** `<mod/port>` **trust ____**
  
  Applies to the actual cat6k port trust rules
  
  untrusted (default), trust-cos, trust-ipprec, trust-dscp
  
  Current 10/100 cards require an additional ACL to actually enable port trust:
  
  ```
  cat6k-access> (enable) set qos enable
  cat6k-access> (enable) set port qos 5/1-48 trust-trust-cos
  cat6k-access> (enable) set port qos 5/1-48 vlan-based
  cat6k-access> (enable) set qos acl ip ACL_IP-PHONES
  trust-cos ip any any
  cat6k-access> (enable) commit qos acl all
  cat6k-access> (enable) set qos acl map ACL_IP-PHONES 110
  ```

Connecting the IP Phone

SoftPhone

- SoftPhone sets VoIP-RTP to DSCP = EF
- No CoS manipulation
- No VoIP control plane classification
- Trusting the SoftPhone DSCP settings requires trusting all DSCP tags from the PC
  
  ```
  cat6k-access> (enable) set vlan 10 6/1-24
  cat6k-access> (enable) set port host 6/1-24
  cat6k-access> (enable) set port qos 6/1-24 trust trust-dscp
  ```
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Is QoS Needed in the Campus?

“Just throw more bandwidth at it. That will solve the problem!”

Maybe, Maybe Not; Campus Congestion Is a Buffer Management Issue
Access, Distribution and Core Queuing

Area’s Where QoS Maybe a Concern

- Output buffers can reach 100% in campus networks
- When an output buffer congests, dropped packets occur at the ingress interfaces
- QoS required when there is a possibility of congestion in buffers
- Multiple queues are the only way to “guarantee” voice quality

Tx Buffer Congestion

Ethernet Switch

- Data Flows “Hog” Tx Buffer
- Additional Flows, including voice, can not get access to Tx Buffer
Guaranteed Voice Requires Multiple Queues

Queues Transmit Scheduled on RR/WRR Format

- Catalyst 6000
  - WRR or PQ/WRR
  - WRR is a 255:5 ratio of high/low queues
- Catalyst 4000
  - Round-Robin
- Catalyst 2900 XL and 3500 XL
  - Exhaustive PQ'ing scheme
- IP Phone
  - Round-Robin with a priority timer for PQ

Round Robin, Weighted Round Robin or Priority Queuing Used for Scheduling Between Queues
Campus QoS

Catalyst Switches which Support Multiple Queues

Queuing/Scheduling Capabilities Depend on Hardware:

- **Wiring Closet**
  - 3500—2Q1T TX (10/100 Mbps)
  - 8Q1T TX (1000 Mbps—Only 2 active)
  - 4000—2Q1T TX (10/100/1000 Mbps)
  - 6000—2Q2T TX (10/100/1000 Mbps)
  - 1P2Q2T TX (1000 Mbps)*
  - 1Q4T RX (10/100/1000 Mbps)
  - 1P1Q4T RX (1000 Mbps)*

- **Distribution/Core**
  - 6000—2Q2T TX (10/100/1000 Mbps)
  - 1P2Q2T TX (1000 Mbps)
  - 1Q4T RX (10/100/1000 Mbps)
  - 1P1Q4T RX (1000 Mbps)

* Next generation Cat6k 10/100 Linecards will be able to take advantage of the additional PQ

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**Access Switch 10/100 Port TX Queuing**

- **Cat6K**
  - Cat6K 10/100BaseT Interface has 2Q2T TX Queue
  - WRR Between Queues (255:5 Ratio of High/Normal)

- **Cat4K**
  - Cat4K 10/100BaseT Interface has 2Q1T TX Queue
  - RR Between Queues

- **Cat 3500 XL**
  - 2900 XL and 3500 XL 10/100BaseT Interfaces have 2Q1T TX Queue
  - Exhaustive PQ'ing Scheme (not configurable)
QoS In Catalyst 6000 Switches

Access Layer—Cat6K

• Port can trust DSCP, IP Prec or CoS
  Recommended: trust-cos
  10/100 cards require an additional step of configuring
  ACL to trust traffic
• Any traffic which “hits” the MSFC will receive a CoS
  of “0”...DSCP and CoS map required
• Current 10/100 Cards don’t have the additional TX and RX
  PQs
• Only switch which really can support SoftPhone QoS
• Output scheduling consists of:
  Assigning traffic to queues based on CoS
  Configuring threshold levels
  Modifying buffer sizes (expert mode)
  Assigning weights for WRR (expert mode)

Catalyst 6000 Example

Access Layer—Cat6K

```bash
cat6k-access> (enable) set qos enable
cat6k-access> (enable) set qos acl ip ACL_IP-PHONES
  trust-cos ip any any
cat6k-access> (enable) set qos acl ip ACL_VIDEO-CONF
  trust-ipprec ip host 10.70.247.200 any
  cat6k-access> (enable) set qos acl ip ACL_IPTV trust-
  ipprec ip any any
  cat6k-access> (enable) set port qos 5/1-48 trust trust-
  cos
  cat6k-access> (enable) set qos acl map ACL_IP-PHONES 110
  cat6k-access> (enable) set port acl map ACL_VIDEO-CONF 4/2
  cat6k-access> (enable) set qos acl map ACL_IPTV 4/4
  cat6k-access> (enable) set qos map 1p2q2t tx 2 1 cos 3
  cat6k-access> (enable) set qos map 2q2t tx 2 1 cos 3
  cat6k-distrib> (enable) set qos acl-map acl 0 14 16 26
  cat6k-distrib> (enable) set qos ipprec-dscp-map 0 14 16
  26 34 46 48 56
cat6k-access> (enable) set port qos 1/1-2 trust trust-cos
```
QoS in Catalyst 4000/2948G

Access Layer—Cat4K

- Input classification based on incoming CoS label (802.1p)
- If no CoS, packet gets assigned a CoS value which is “switch-wide”
- All ports are considered “trusted”
- The output ports have a 2Q1T capability
- CoS valuesmapped to output queues in pairs
- Queues are serviced in a round-robin fashion

Catalyst 4000/2948G Example

Access Layer—Cat4K

```bash
cat4k> (enable) set qos enable
cat4k> (enable) set qos map 2q1t 1 1 cos 0-1
cat4k> (enable) set qos map 2q1t 2 1 cos 2-3
cat4k> (enable) set qos map 2q1t 2 1 cos 4-5
cat4k> (enable) set qos map 2q1t 2 1 cos 6-7
cat4k> (enable) show qos info runtime
Run time setting of QoS:
QoS is enabled
All ports have 2 transmit queues with 1 drop thresholds (2q1t).
Default CoS = 0
Queue and Threshold Mapping:
Queue Threshold CoS
----- -------- ---------------
1 1 0 1
2 1 2 3 4 5 6 7
```

Queue CoS mapping occurs in groups of 2
QoS in Catalyst 3500/2900 XL

Access Layer—3500/2900 XL

- Input classification based on incoming CoS label (802.1p)
- If no CoS, packet can be assigned a port-based CoS value
- The ports have a 2Q1T TX capability
- CoS values mapped to default output queues—**not configurable**
  - 0-3 = Low Priority Queue
  - 4-7 = High Priority Queue
- Queues are serviced via priority scheduling
- GigaStack architecture is not supported for guaranteed voice quality because it’s a shared media
- No way to view queue configuration and statistics with the current HW
- 2900 XL requires 8 MB DRAM

Catalyst 3500/2900 XL Example

Access Layer—3500/2900 XL

```
interface FastEthernet0/1
  power inline auto
  speed auto
  switchport trunk encapsulation dot1q
  switchport trunk native vlan 12
  switchport mode trunk
  switchport voice vlan 112
  switchport priority extend cos 0
  spanning-tree portfast
interface GigabitEthernet0/1
  switchport trunk encapsulation dot1q
  switchport mode trunk

vlan database (EXEC Mode)
  vlan 112
```
QoS Is Catalyst 6000 Switches

Distribution Layer—Cat6K

- Typically Gig-E for all connections
  TX = 1P2Q2T
  RX = 1P1Q4T
- Use dual layer 3 distribution layer switches and load balance VLANs using HSRP; tweak STP, HSRP and routing protocols for fast convergence
- Distribution layer switch will perform all Layer 3 & Layer 2 classification mapping for layer 2 only access switches
- Any frames which “hit” the MSFC will receive a CoS value of “0”: mark DSCP and perform DSCP to CoS mappings
- “Trust” access layer switch CoS markings

Cat6K Access-Distribution Gig-E Uplink

- Cat6K Gig-E has 1P2Q2T TX Interface Queue
- Cat6K Gig-E Interface has a 1P1Q4T RX Queue
- PQ/WRR Queue Scheduler
- RX PQ gives frames priority access to backplane
- Must “trust” CoS for RX PQ Access
**Example AVVID QoS Network—Campus**

- Native IOS 6000
- Hybrid 6000
- Distribution
- Access

**Catalyst 6000 Example—Hybrid**

**Distribution Layer—Cat6K**

```
cat6k-access> (enable) set qos enable
cat6k-distrib> (enable) set qos map lp2q2t tx queue 2 1 cos 3
cat6k-distrib> (enable) set qos map 2q2t tx queue 2 1 cos 3
cat6k-distrib> (enable) set qos ipprec-dscp-map 0 14 16 26 34 46 48 56
cat6k-distrib> (enable) set qos cos-dscp-map 0 14 16 26 34 46 48 56
```

```
cat6k-distrib> (enable) set port qos 1/1-2,3/2 trust trust-cos
cat6k-distrib> (enable) set port qos 1/1-2,3/2 vlan-based
```

```
cat6k-distrib> (enable) set port qos 9/1 trust trust-dscp
```

```
cat6k-distrib> (enable) set port qos 9/1 port-based
```

```
cat6k-distrib> (enable) set qos acl acl ACL_TRUST-WAN trust-dscp ip any
```

```
cat6k-distrib> (enable) commit qos acl ACL_TRUST-WAN
```

```
cat6k-distrib> (enable) set qos acl map ACL_TRUST-WAN 9/1
```
Catalyst 6000 Example—Native

Distribution Layer—Cat6K

```
mls qos
mls qos map ip-prec-dscp 0 14 16 26 34 46 48 56
mls qos map cos-dscp 0 14 16 26 34 46 48 56
int range gigabitEthernet 1/1 - 2
  wrr-queue cos-map 2 1 3
  wrr-queue cos-map 2 2 4

! Trust CoS from the PFC enabled Access Switch
interface GigabitEthernet2/1
  description trunk port to PFC enabled cat6k-access
  no ip address
  wrr-queue cos-map 2 1 3
  wrr-queue cos-map 2 2 4
  mls qos vlan-based
  mls qos trust cos
  switchport
  switchport trunk encapsulation dot1q
  switchport mode trunk
```

```
Distribution Layer—Cat6K

! Trust CoS from the PFC enabled Access Switch
interface GigabitEthernet2/2
  description trunk port to PFC enabled cat6k-access
  no ip address
  wrr-queue cos-map 2 1 3
  wrr-queue cos-map 2 2 4
  mls qos vlan-based
  mls qos trust cos
  switchport
  switchport trunk encapsulation dot1q
  switchport mode trunk
```

```
Distribution Layer—Cat6K

! Trust CoS from the Layer 2 only Catalyst 4000 Access Switch
interface GigabitEthernet3/1
  description trunk port to layer 2-only cat4k
  no ip address
  wrr-queue cos-map 2 1 3
  wrr-queue cos-map 2 2 4
  mls qos vlan-based
  mls qos trust cos
  switchport
  switchport trunk encapsulation dot1q
  switchport mode trunk
```

```
Distribution Layer—Cat6K

! Trust CoS from the Layer 2 only 3500 Access Switch
interface GigabitEthernet3/1
  description trunk port to layer 2-only 3500
  no ip address
  wrr-queue cos-map 2 1 3
  wrr-queue cos-map 2 2 4
  mls qos vlan-based
  mls qos trust cos
  switchport
  switchport trunk encapsulation dot1q
  switchport mode trunk
```

Campus QoS—Access Layer

Control and Management Plane Traffic

- All VoIP Control Plane Traffic should be Classified as DSCP=AF31 in the VoIP Gateway or from the Cat6K PFC

  - Skinny Control: TCP 2000-2002
  - Skinny already classifies in CCM 3.0(5) and beyond
  - MGCP Control: UDP 2427 and TCP 2428
  - H.323 Control: TCP 1720
  - TCP 11000-11999

Skinny already classifies in CCM 3.0(5) and beyond
MGCP and H.323 Call Control traffic can be classified from
IOS 12.2(1)T

H.323 RTP and VoIP Control Channel Classification

ip qos dscp ef media
ip qos dscp af31 signaling

MGCP VoIP Control Channel Classification

cat6k-access> (enable) set qos acl ip ACL_VOIP_CONTROL dscp 26
udp any any eq 2427

H.323 Gateway

MGCP Gateway

CallManager

4/2 4/3 4/4
General Campus Recommendations

- Use switches that support multiple queues on both access and uplink ports and in-line power in the wiring closet
  - 2900 XL
  - 3500 XL (not GigaStacked; Daisy Chain OK)
  - 4000
  - 6000
- If 5000 is used in the wiring closet, use Saint 5 (10/100) linecards for uplinks or use 1 uplink for voice VLANs and one uplink for data VLANs
  - If 5000 is used in the wiring closet, use Saint 5 linecards

Agenda

- Quality Concerns with IP Telephony and Multimedia Applications
- General AVVID QoS Design Guidelines
- Connecting the IP Phone
- Designing the Campus
- Enabling the WAN
- Managing the QoS Infrastructure
QoS in the WAN

General Guidelines

- Use LLQ anytime VoIP over the WAN is involved
- Traffic shaping is a requirement for frame-relay/ATM environments
- Use LFI techniques for all links below 768Kbps
  Don’t use LFI for any video over IP applications
- TX-ring sizes may require modifications
- Properly provision the WAN bandwidth
- Call admission control is a requirement where VoIP calls can over-subscribe the provisioned BW
- Use cRTP carefully

IOS Recommendations

<table>
<thead>
<tr>
<th>Media</th>
<th>Minimum IOS</th>
<th>Prioritization</th>
<th>LFI</th>
<th>Traffic Shaping</th>
</tr>
</thead>
<tbody>
<tr>
<td>Leased Lines</td>
<td>12.1(2)T</td>
<td>LLQ</td>
<td>MLPPP</td>
<td>N/A</td>
</tr>
<tr>
<td>Frame-Relay ATM</td>
<td>12.2(1)T</td>
<td>LLQ</td>
<td>FRF.12</td>
<td>Shape to CIR</td>
</tr>
<tr>
<td>ATM</td>
<td>12.1(6)</td>
<td>Per VC LLQ</td>
<td>MLPPP over ATM</td>
<td>Shape to Guaranteed Portion to BW</td>
</tr>
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<td>12.1(5)T*</td>
<td>Per VC LLQ</td>
<td>MLPPP over ATM and Frame-Relay</td>
<td>Shape to Guaranteed Portion to BW</td>
</tr>
</tbody>
</table>
Sources of Trouble for VoIP

Provisioning

Voice Is Not Free—Especially on Low Speed Links—Engineer the Network for Data, Voice, and Video

Link Capacity = (Min BW for Voice + Min BW for Video + Min BW for Data) / 0.75

Voice

Video

Voice/Video

Data

Routing etc

LLQ = 33%

Sum of Traffic = 75%

Calculating VoIP Bandwidth Requirements

Provisioning

<table>
<thead>
<tr>
<th>CODEC</th>
<th>Sampling Rate</th>
<th>Voice Payload in Bytes</th>
<th>Packets per Second</th>
<th>Bandwidth per Conversion</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>20 msec</td>
<td>160</td>
<td>50</td>
<td>80 kbps</td>
</tr>
<tr>
<td>G.729A</td>
<td>20 msec</td>
<td>30</td>
<td>50</td>
<td>34 kbps</td>
</tr>
<tr>
<td>G.729A</td>
<td>30 msec</td>
<td>30</td>
<td>33</td>
<td>18 kbps</td>
</tr>
</tbody>
</table>

A More Accurate Method for Provisioning is to Include the Layer 2 Headers into the Bandwidth Calculations

<table>
<thead>
<tr>
<th>CODEC</th>
<th>Ethernet 14 Bytes of Header</th>
<th>PPP 6 Bytes of Header</th>
<th>ATM 53 Bytes Cells with a 48 Byte Payload</th>
<th>Frame-Relay 4 Bytes of Header</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 at 50 pps</td>
<td>95.6 kbps</td>
<td>82.4 kbps</td>
<td>106 kbps</td>
<td>81.6 kbps</td>
</tr>
<tr>
<td>G.729A at 50 pps</td>
<td>29.6 kbps</td>
<td>26.4 kbps</td>
<td>42.4 kbps</td>
<td>25.5 kbps</td>
</tr>
</tbody>
</table>
cRTP Update

Miscellaneous VoIP QoS Tools

- Fast switching enhancements in 12.1(1)T and 12.1(2)T are interface specific; also, each QoS feature might take the switching back to process switched; read release notes carefully
- 12.2.1T (target): cRTPovPPovATM-AAL5
  - PPP definitely
  - MLPPP under investigation
- cRTP over IETF FR VCs
  - Submitted a FR forum proposal for carrying cRTP in FR encap

cRTP VoIP Bandwidth Calculations

<table>
<thead>
<tr>
<th>CODEC</th>
<th>PPP 6 Bytes of Header</th>
<th>ATM 53 Bytes Cells with a 48 Byte Payload</th>
<th>Frame-Relay 4 Bytes of Header</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 at 50 pps</td>
<td>68 kbps</td>
<td>N/A</td>
<td>67 kbps</td>
</tr>
<tr>
<td>G.729 at 50 pps</td>
<td>62 kbps</td>
<td>N/A</td>
<td>63.5 kbps</td>
</tr>
<tr>
<td>G.729A at 50 pps</td>
<td>62 kbps</td>
<td>N/A</td>
<td>63.5 kbps</td>
</tr>
<tr>
<td>G.729A at 33 pps</td>
<td>10.5 kbps</td>
<td>N/A</td>
<td>10 kbps</td>
</tr>
</tbody>
</table>
Sources of Trouble for VoIP

Call Admission Control

“Protecting Voice from Voice”

Example:
WAN Bandwidth Can Only Support 2 Calls
What Happens when 3rd Call Attempted?

Call #1
Call #2
Call #3

VoIP Data Network

Call #3
Causes Poor Quality for ALL Calls

Need—To Prevent Third Call from Traversing IP WAN

PQ-CBWFQ (Low Latency Queuing)

Queuing

• Operation
  PQ is policed to BW to ensure other traffic is not starved
  Rate limit is per class, even if multiple classes point traffic to PQ
  Over-subscription of minimum possible BW is not allowed
  “Bandwidth” and “priority” mutually exclusive

• BW in the priority class
  Max allowable BW for “priority” classes is mincir (frame-relay)
  Recommended max BW for “priority” classes is 33%
  In order to take cRTP into account:
    7500—12.2(1) - Careful
    7200—12.2(1)T
    26/3600—12.2(1)
Low-Latency Queuing Logic Tree

Queuing

Layer 3 Queuing Subsystem

Layer 2 Queuing Subsystem

Link Fragmentation and Interleave

Packets In

Packets Out

VoD

CBWFQ

Layer 3 Queuing

Layer 2 Queuing

Fragment

Interleave

Low Latency Queuing

VoIP-Cntr

PQ Voice

PQ VC

PQ

VoD

SAP/Oracle

Default

WFQ

Low Latency Queuing Logic Tree

LLQ Example—WAN Router

VoIP—Queuing

class-map VoIP-RTP
   match access-group 100
class-map VoIP-Control
   match access-group 101

policy-map QoS-Policy
   class VoIP-RTP
      priority 100
   class VoIP-Control
      bandwidth 8
   class class-default
      fair-queue

!  Voice RTP Traffic
access-list 100 permit ip any any dscp cs5
access-list 100 permit ip any any dscp ef

!  Voice Control Traffic
access-list 101 permit ip any any dscp cs3
access-list 101 permit ip any any dscp af31

Leased Lines: 12.0.7T
interface Multilink 1
service-policy output QoS-Policy

ATM: 12.0.7T
interface ATM1/0.1 point-to-point
service-policy output QoS-Policy

VoIPovFR: 12.1.2T
map-class frame voipofr
frame cir 128000
frame mincir 64000
frame bc 1000
frame frag 160
service-policy output QoS-Policy
LLQ Example—WAN Router

Video—Queuing

class-map Video-Conf
match access-group 102
class-map Streaming-Video
match access-group 103
!
policy-map QoS-Policy

class Video-Conf
priority 450 30000
class Streaming-Video
bandwidth 150
class class-default
fair-queue
!

Leased Lines: 12.0.7T
interface Multilink 1
service-policy output QoS-Policy

ATM: 12.0.7T
interface ATM1/0.1 point-to-point
service-policy output QoS-Policy

FR: 12.1.2T
map-class frame vcofr
frame cir 128000
frame mincir 64000
frame bc 1000
frame frag 160
service-policy output QoS-Policy

Slow Link Efficiency Tools

Fragmentation and Interleave Not Needed on Links Greater than 768 kbps

Before

Real-Time MTU

Elastic Traffic MTU
214 ms Serialization Delay
for 1500 Byte Frame at 56 kbps

After

Elastic MTU
Elastic MTU
Real-Time MTU
Elastic MTU

Mechanisms

Pt to Pt Links: MLPPP with Fragmentation and Interleave
Frame Relay: FRF.12—12.1(5)T
ATM: MLPPP over ATM—12.1(5)T
ATM/Frame-Relay Interworking: MLPPP over ATM and Frame Relay—12.1(5)T

Fragment Size for MLPPP over ATM:
Fragment Size = (48 * Number_of_Cells)—10–8
Fragment Size Recommendations

LFI Fragment Information

Serialization Delay Matrix

<table>
<thead>
<tr>
<th>Link Speed</th>
<th>Frag Size</th>
<th>64 Bytes</th>
<th>128 Bytes</th>
<th>256 Bytes</th>
<th>512 Bytes</th>
<th>1024 Bytes</th>
<th>1500 Bytes</th>
</tr>
</thead>
<tbody>
<tr>
<td>56 kbps</td>
<td></td>
<td>9 ms</td>
<td>18 ms</td>
<td>36 ms</td>
<td>72 ms</td>
<td>144 ms</td>
<td>214 ms</td>
</tr>
<tr>
<td>64 kbps</td>
<td></td>
<td>8 ms</td>
<td>16 ms</td>
<td>32 ms</td>
<td>64 ms</td>
<td>128 ms</td>
<td>187 ms</td>
</tr>
<tr>
<td>128 kbps</td>
<td></td>
<td>4 ms</td>
<td>8 ms</td>
<td>16 ms</td>
<td>32 ms</td>
<td>64 ms</td>
<td>93 ms</td>
</tr>
<tr>
<td>256 kbps</td>
<td></td>
<td>2 ms</td>
<td>4 ms</td>
<td>8 ms</td>
<td>16 ms</td>
<td>32 ms</td>
<td>46 ms</td>
</tr>
<tr>
<td>512 kbps</td>
<td></td>
<td>1 ms</td>
<td>2 ms</td>
<td>4 ms</td>
<td>8 ms</td>
<td>16 ms</td>
<td>23 ms</td>
</tr>
<tr>
<td>768 kbps</td>
<td>640 used</td>
<td>1.2 ms</td>
<td>2.5 ms</td>
<td>5 ms</td>
<td>10 ms</td>
<td>15 ms</td>
<td></td>
</tr>
</tbody>
</table>

Fragmentation Size Matrix (Based on 10 msec Delay)

<table>
<thead>
<tr>
<th>Link Speed</th>
<th>Frag Size</th>
<th>56 kbps</th>
<th>70 Bytes</th>
</tr>
</thead>
<tbody>
<tr>
<td>64 kbps</td>
<td></td>
<td>64</td>
<td>32 Bytes</td>
</tr>
<tr>
<td>128 kbps</td>
<td></td>
<td>128</td>
<td>160 Bytes</td>
</tr>
<tr>
<td>256 kbps</td>
<td></td>
<td>256</td>
<td>640</td>
</tr>
<tr>
<td>512 kbps</td>
<td></td>
<td>512</td>
<td>640</td>
</tr>
<tr>
<td>768 kbps</td>
<td></td>
<td>768</td>
<td>1000</td>
</tr>
<tr>
<td>1536 kbps</td>
<td></td>
<td>1536</td>
<td>2000</td>
</tr>
</tbody>
</table>

TX-Ring Sizing

Misc. VoIP QoS Tools

- TX-Ring (TX-Queue on 7500 RSP) is an un-prioritized FIFO buffer which holds packets just before media transmission
- Used to make sure enough packets are queued in order to maximize available BW
- Will add to E-2-E delay numbers because serialization delay really equals:
  Serialization delay * number of packets in the TX-Ring buffer

<table>
<thead>
<tr>
<th>Media</th>
<th>Default TX-Ring Buffer Sizing (Packets)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PPP</td>
<td>6</td>
</tr>
<tr>
<td>MLPPP</td>
<td>2</td>
</tr>
<tr>
<td>ATM</td>
<td>8192—Must Be Changed For Low Speed VCs</td>
</tr>
<tr>
<td>Frame-Relay</td>
<td>64 (Per Main T1 Interface )</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Link Speed</th>
<th>Default TX-Ring Buffer Sizing (Packets)</th>
</tr>
</thead>
<tbody>
<tr>
<td>128 kbps</td>
<td>3</td>
</tr>
<tr>
<td>192 kbps</td>
<td>3</td>
</tr>
<tr>
<td>256 kbps</td>
<td>3</td>
</tr>
<tr>
<td>512 kbps</td>
<td>4</td>
</tr>
<tr>
<td>768 kbps</td>
<td>6</td>
</tr>
</tbody>
</table>
Traffic Shaping—Why?

1. Central to remote site speed mismatch
2. To avoid remote to central site over-subscription
3. To prohibit bursting above committed rate

What are you guaranteed above your committed rate?

Branch QoS

General Guidelines

All Incoming Voice from PSTN Classified as DSCP=EF within the Dial-Peer

Trust IP Prec from Proxy or Specific VC System

All Incoming VoIP from WAN Should Already Be Tagged as DSCP=EF/AF31 (Video=AF41) at the Central Site

Use LLQ/LFI on Branch Router WAN. Use mod-CLI for L3 to L2 Classification on the Ethernet Interface

Branch Ethernet switches with multiple queuing based on CoS on all interfaces

Classify all IP Phone VoIP traffic All Data traffic CoS=0
QoS In the Branch Office

- If any VoIP over the WAN is part of the design, advanced QoS tools are a requirement; specifically, LLQ and LFI
- Branch router will typically be 1750, 2600, 3600, Cat4k or Jobim
  - All of these support VoIP gateway interfaces: classify VoIP traffic
  - Only the 2600 and 3600 support 802.1Q/p...branch switch needs router to set 802.1p for queuing (10/100 interfaces)
- Catalyst scheduling capabilities depends on hardware:
  - Catalyst 2900 XL or 3500 XL
  - Catalyst 4000
  - Catalyst 6000

Branch Office Design

802.1Q Trunking

```conf
tarton (enable) set vlan 70 name data70
tarton (enable) set vlan 170 name voice170
tarton (enable) set vlan 70 2/1-48
ntarton (enable) set port host 2/1-48
ntarton (enable) set port auxiliarvlan 2/1-48
ntarton (enable) set port speed 2/1-49 auto
tarton (enable) set trunk 2/49 on dot1q 1-1005
```

Native VLAN=70

Aux VLAN=170
Layer 3 to Layer 2 Classification Mapping at the Branch

Requires the mod-cli Commands Available in IOS 12.1(5)T*

- class-map L3-to-L2-VoIP-RTP
  - match ip dscp EF
- class-map L3-to-L2-Video-Conf
  - match ip dscp AF41
- class-map L3-to-L2-VoIP-Control
  - match ip dscp AF31

! policy-map output-L3-to-L2
  - class L3-to-L2-VoIP-RTP
    - set cos 5
  - class L3-to-L2-Video-Conf
    - set cos 4
  - class L3-to-L2-VoIP-Control
    - set cos 3

! interface e0/0
  - service-policy output output-L3-to-L2

Branch Office Design

Single Subnet

- interface FastEthernet1/0
  - mac-address 0000.2600.0001
  - ip address 10.1.40.1 255.255.255.0
  - ip helper-address 10.1.10.10
  - service-policy output output-L3-to-L2
  - no ip mroute-cache
  - load-interval 30
  - speed 100
  - full-duplex

- interface FastEthernet0/2
  - description Port to IP Phone in single subnet
  - switchport trunk encapsulation dot1q
  - switchport trunk native vlan 40
  - switchport mode trunk
  - switchport voice vlan dot1p
  - spanning-tree portfast

! interface FastEthernet0/15
  - description Port to 1750 Router in single subnet
  - load-interval 30
  - duplex full
  - speed 100
  - switchport access vlan 40
Branch Office Video Conf Designs

Video Conf Zone Design Specific

**Single WAN Zone**

```
class-map Video-Conf
  match access-group 102
!policy-map QoS-Policy
  class Video-Conf
    priority 450 30000
  class class-default
    fair-queue
!
! Video-Conf Traffic
  access-list 102 permit ip host 10.70.249.200 any dscp cs4
  access-list 102 permit ip host 10.70.249.200 any dscp af41
```

**Multiple WAN Zones**

```
!H323 proxy
!interface FastEthernet/0.249
  descrip native subnet 10.70.249.0 data
  encapsulation dot1Q 70
  ip address 10.70.249.1 255.255.255.0
  service-policy output output-L3-to-L2
  no ip mroute-cache
  h323 interface
  h323 qos ip-precedence 4
  h323 h323-id vail-px@vail.com
  h323 gatekeeper ipaddr 10.70.249.254
```

WAN QoS—Leased Lines

VoIP over Leased-Line min IOS 12.1(2)T

**Queueing**

- **Low-Latency Queuing**
  - VoIP Bearer Plane PQ’d by IP Prec/DSCP (5/EF) Classification
  - VoIP Control Plane CBWFQ’ing by IP Prec/DSCP (3/24) Classification

- **T1 Clock or Above**
  - Video Conf PQ’d by IP Prec/DSCP (4/34) Classification
  - Streaming Video CBWFQ’ing by IP Prec/DSCP (1/14) Classification

- **MLPPP**
  - Link Speeds < 768kbps
  - Fragment Size = Max_Allowed_Jitter / (1 Byte / Line Speed in kbps)

**Supported**

- cRTP
AVVID over PPP QoS Example

```plaintext
interface Multilink1
  ip address 10.1.61.1 255.255.255.0
  ip tcp header-compression iphc-format
  no ip mroutecache
  load-interval 30
  service-policy output QoS-Policy
  ppp multilink
    ppp multilink fragment-delay 10
    ppp multilink interleave
    multilink-group 1
  ip rtp header-compression iphc-format

interface Serial0
  bandwidth 256
  no ip address
  encapsulation ppp
  no ip mroute-cache
  load-interval 30
  no fair-queue
  ppp multilink
  multilink-group 1
```

WAN QoS—Frame-Relay

Voice over Frame-Relay min IOS 12.1(2)T

<table>
<thead>
<tr>
<th>Queueing</th>
<th>Low-Latency Queuing per VC</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>VoIP Bearer Plane PQ'd by IP Prec/DSCP (5/EF) Classification</td>
</tr>
<tr>
<td>Video</td>
<td>VoIP Control Plane GFQ'ing by IP Prec/DSCP (9/24) Classification</td>
</tr>
</tbody>
</table>

Traffic Shaping: Frame-Relay Traffic Shaping

- Shape to CIR [CIR + Flags+CRC]
- Bc = CIR/100
- Be = 0
- MINCIR = Sum of all configured queues

LFI: MLPPP

- Link Speeds < 768kbps
- Fragment Size = Max_Allowed_Jitter / (1 Byte / Line Speed in kbps)

cRTP: Supported
AVVID over Frame-Relay QoS Example

interface Serial1
  no ip address
  encapsulation frame-relay
  load-interval 30
  frame-relay traffic-shaping

! interface Serial1.71 point-to-point
  bandwidth 256
  ip address 10.1.71.1 255.255.255.0
  frame-relay interface-dlci 71
  class VoIP

! map-class frame-relay VoIP
  frame-relay cir 250000
  frame-relay bc 1000
  frame-relay be 0
  frame-relay mincir 250000
  no frame-relay adaptive-shaping
  service-policy output QoS-Policy
  frame-relay fragment 320

256000 * 320/324
Rounded Down

WAN QoS—ATM

VoIP over ATM-Relay 12.1(5)T

<table>
<thead>
<tr>
<th>Queuing</th>
<th>Low-Latency Queuing per VC</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>VoIP Bearer Plane d by IP Prec/DSCP (5/EF) Classification</td>
</tr>
<tr>
<td></td>
<td>VoIP Control Plane CBWFQ'ing by IP Prec/DSCP (3/24) Classification</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Video</th>
<th>T1 Clock or Above</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Video Conf Po'd by IP Prec/DSCP (4/34) Classification</td>
</tr>
<tr>
<td></td>
<td>Streaming Video CBWFQ'ing by IP Prec/DSCP (1/14) Classification</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Traffic Shaping</th>
<th>Generic Traffic Shaping</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Shape to low VC</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>LF!</th>
<th>MLPPP over ATM in 12.1(5)T</th>
</tr>
</thead>
<tbody>
<tr>
<td>cRTP</td>
<td>Not Supported</td>
</tr>
</tbody>
</table>
AVVID over ATM QoS Example

interface ATM2/0
no ip address
no ip mroute-cache
no shutdown
atm pvc 1 0 16 ilmi
no atm ilmi-keepalive
!
interface ATM2/0.37 point-to-point
pvc cisco37 0/37
tx-ring-limit 7
abr 256 256
protocol ppp Virtual-Template2
!
interface Virtual-Template2
bandwidth 256
ip address 10.1.37.52 255.255.255.0
service-policy output QoS-Policy
ppp authentication chap
ppp chap hostname HQ_7200
ppp chap password 7 05080F1C2243
ppp multilink
ppp multilink fragment-delay 10
ppp multilink interleave

WAN QoS—ATM to Frame Relay

VoIP over Hybrid Networks 12.1(4)T

- Queuing: Low-Latency Queuing per VC
- Video: T1 Clock or Above
- Traffic Shaping: Generic Traffic Shaping
- L.166: MI. PPP over ATM and Frame Relay in 12.1(5)T
AVVID over ATM to Frame Relay Interworking QoS Example

Remote Frame Relay Configuration

interface Serial6/0
description T1 to Frame Relay switch
no ip address
encapsulation frame-relay
load-interval 30
no arp frame-relay
frame-relay traffic-shaping
!
interface Serial6/0.73 point-to-point
description 3640
no arp frame-relay
frame-relay interface-dlci 73 ppp
Virtual-Template2
class VoIP-256kbps
!
interface Virtual-Template2
bandwidth 254
ip address 10.1.37.51 255.255.255.0
service-policy output QoS-Policy
 PPP authentication chap
 PPP chap hostname R72HQ
 PPP chap password 7 05080F1C2243
 PPP multilink
 PPP multilink fragment-delay 10
 PPP multilink interleave
!
Central ATM Configuration

interface ATM2/0
no ip address
no ip mrouted-cache
no shutdown
atm pvc 1 0 16 ilmi
no atm ilmi-keepalive
!
interface ATM2/0.37 point-to-point
pvc cisco37 0/37
tx-ring-limit 7
abr 256 256
protocol ppp Virtual-Template2
!
interface Virtual-Template2
bandwidth 254
ip address 10.1.37.52 255.255.255.0
service-policy output QoS-Policy
PPP authentication chap
PPP chap hostname HQ_7200
PPP chap password 7 05080F1C2243
PPP multilink
PPP multilink fragment-delay 10
PPP multilink interleave

WAN QoS—ATM to Frame Relay

VoIP over Hybrid Networks—IGX Solution

End to End FRF.12 on single PVC

FRF.8 Service Inter-working
Occurs in the Carrier AND Gets Reversed at the IGX

Characteristics
1. Allows for L2 LFI (FRF.12) on a single PVC
2. Tested and works
3. Overcomes shortcomings of carriers not providing FRF.12 in cloud

Caveats
1. EXPENSIVE
Agenda

- Quality Concerns with IP Telephony and Multimedia Applications
- General AVVID QoS Design Guidelines
- Connecting the IP Phone
- Designing the Campus
- Enabling the WAN
- Managing the QoS Infrastructure

Campus Queue Configuration

- VLAN=10
- VLAN=11
- VLAN=12
- Catalyst 6500s
- 6000 with PFC
- 3000
- 3500
- 7200 WAN Router
- VVID=110
- VVID=111
- VVID=112
## Identifying Packets for Service Treatment

### IP Packet

<table>
<thead>
<tr>
<th>ToS Byte</th>
<th>Source IP Addr</th>
<th>Dest IP Addr</th>
<th>Src Port</th>
<th>Dest Port</th>
<th>IP RTP Header</th>
<th>Voice Payload</th>
</tr>
</thead>
</table>

### TCP/UDP Packet

- Source IP Addr
- Dest IP Addr
- Src Port
- Dest Port

### IP RTP Packet

- Source IP Addr
- Dest IP Addr
- Src Port
- Dest Port

---

## Enabling LLQ with QPM

### CBWFQ

- Priority

---

- CBWFQ
- Priority

---

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Presentation_ID.scr
### Configuring cRTP

<table>
<thead>
<tr>
<th>Version</th>
<th>IHL</th>
<th>Type Service</th>
<th>Total Length</th>
<th>Identification</th>
<th>Flags</th>
<th>Fragment Offset</th>
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<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Source Address</td>
<td></td>
<td>Destination Address</td>
</tr>
<tr>
<td></td>
<td>Source Port</td>
<td>Destination Port</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Length</td>
<td>Checksum</td>
<td>V</td>
<td>P</td>
<td>X</td>
<td>CC</td>
</tr>
<tr>
<td></td>
<td>Timestamp</td>
<td>Synchronization Source (SSRC) Identifier</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### What’s Coming…

- Cisco-wide consensus on voice control plane classification—completed
- Video QoS additions to QoS design guide—completed
- QPM Integration Testing and Appendix—completed
- Additional platform scalability testing—ongoing; check the ESE web site (9/01)
Deploying QoS for Voice and Video in IP Networks
Session VVT-213

Please Complete Your Evaluation Form
Session VVT-213