

*TOMORROW starts here.*



Cisco *live!*

# Call Admission Control and Quality of Service for Collaboration

BRKUCC-2667

Glen Lavers

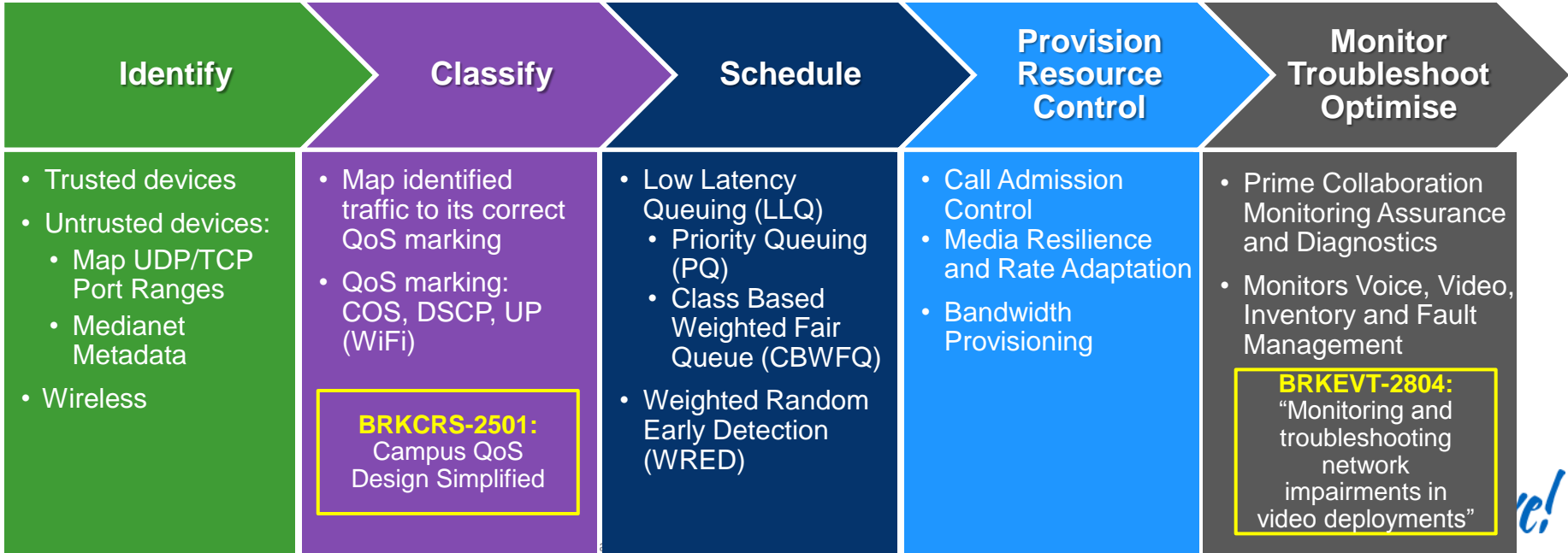
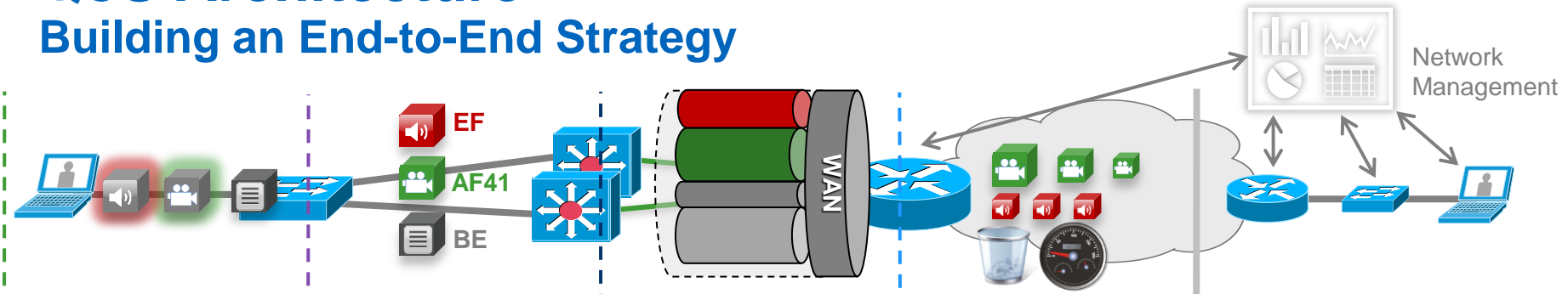
Senior Technical Marketing Engineer

# Agenda

- Introduction – Session Overview
- Media Resilience and Adaptation
  - Resilience techniques
  - Dynamic adaptation and advantages
- Enhanced Locations CAC Architecture
  - Network Modelling
  - Locations Bandwidth Manager (LBM)
  - Inter-Cluster E-LCAC with LBM
- QoS Architecture
  - Approach Overview
  - Identification & Classification
  - Queuing / Scheduling

# QoS Architecture

## Building an End-to-End Strategy

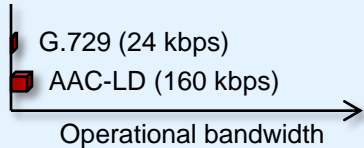




# Media Resilience and Adaptation

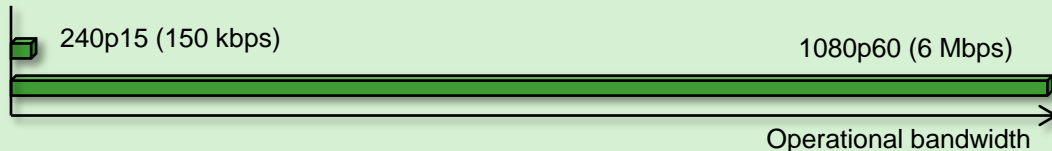
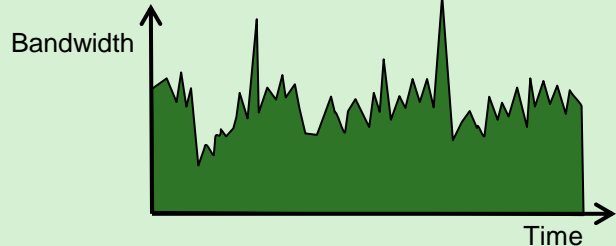
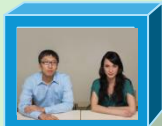
# Video Traffic: Requirements and Profiles

## AUDIO



- **Bandwidth:**
  - Constant bitrate (smooth)
  - Small footprint
  - Narrow operational range (1:6)
- **Loss-sensitive**
- **Delay-sensitive**

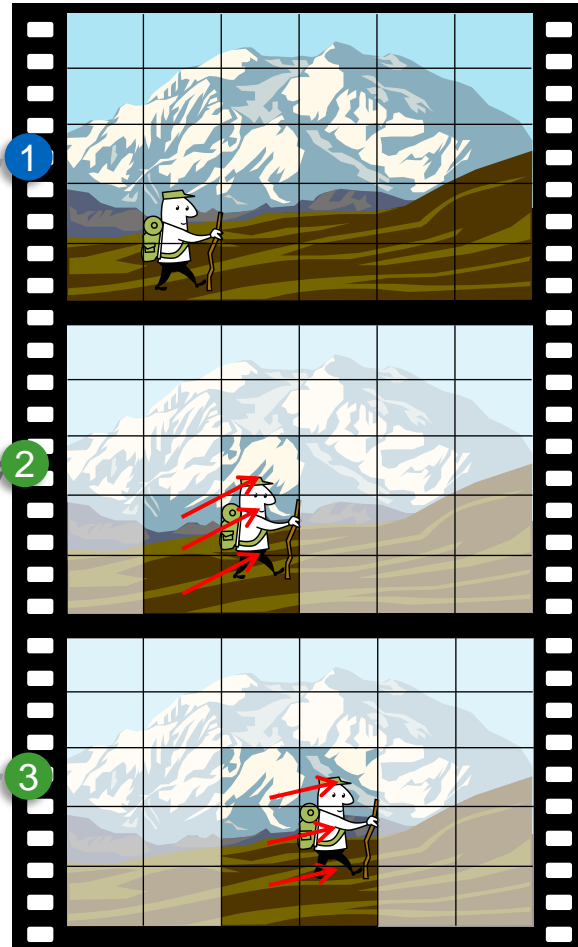
## VIDEO



- **Bandwidth:**
  - Variable bitrate (bursty)
  - Medium/large footprint
  - Wide operational range (1:40)
- **Loss-sensitive**
- **Delay-sensitive**

# Video Traffic

## Video Encoding Basics



### 1 I-Frame

*"Intra-coded" picture*

- Entire picture encoded as a static image
- No reference to other frames

### 2 P-Frame

*"Predicted" picture*

- Based on a previously encoded frame (1)
- Only the differences from that frame are encoded

### 3 P-Frame

*"Predicted" picture*

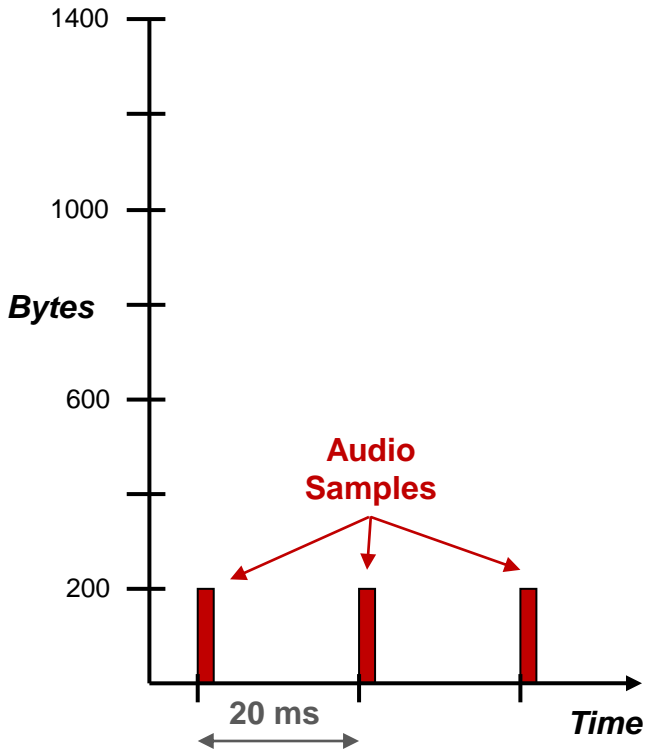
- Reference for prediction can be another P-Frame (2)

For more, see **BRKCOL-2777**:  
"Emerging Video Technologies:  
H.265, SVC, WebRTC / HTML5"

# Video Traffic

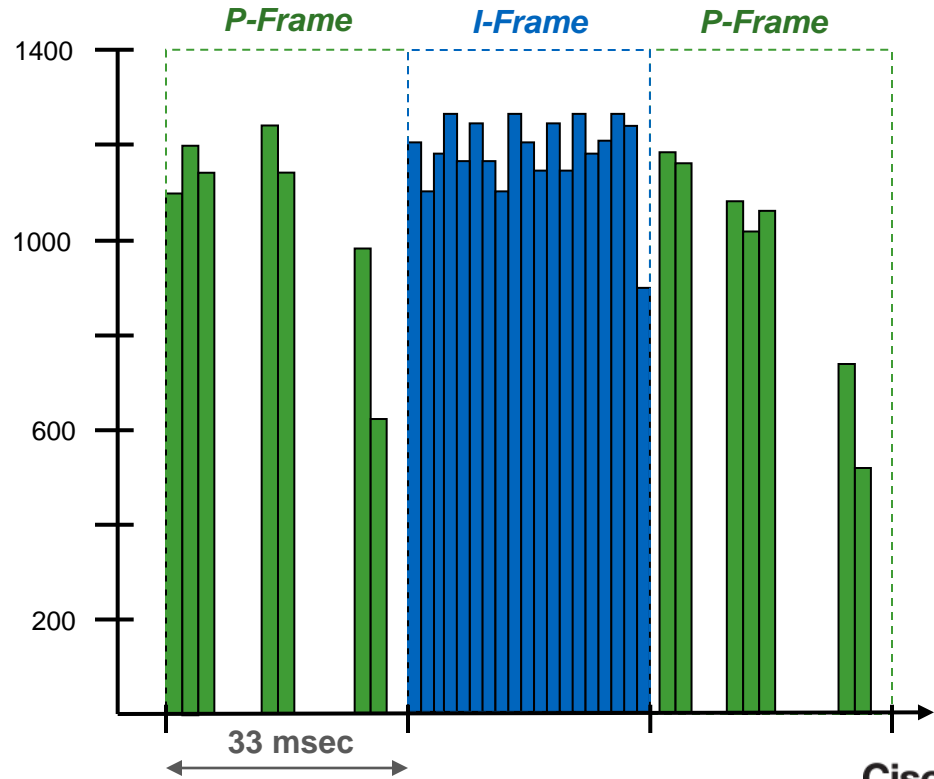
## Audio vs. Video Packet Distribution

### Audio Packets



BRKUC-2667

### Video Packets



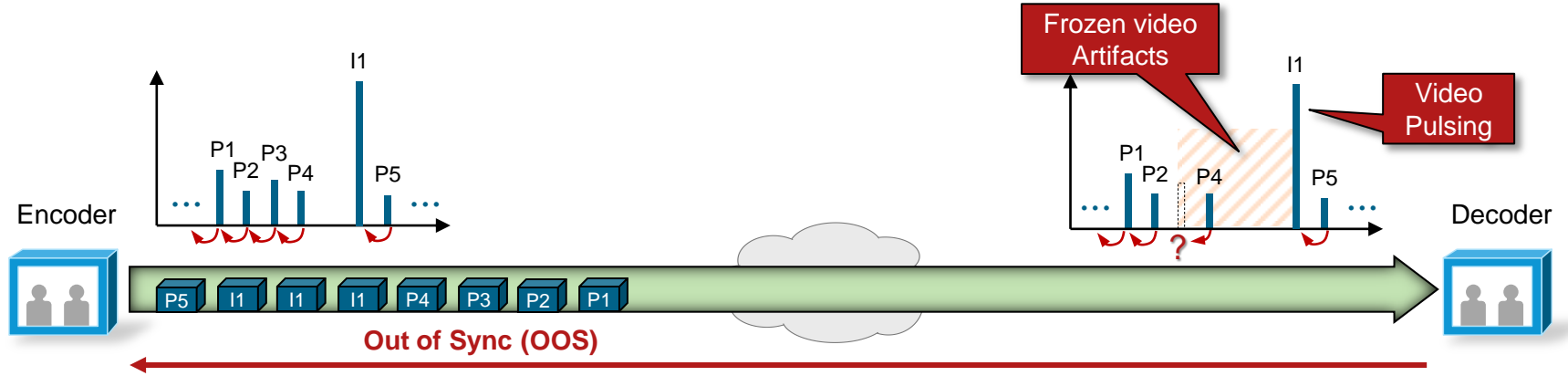
© 2014 Cisco and/or its affiliates. All rights reserved.

Cisco Public



# Video Traffic

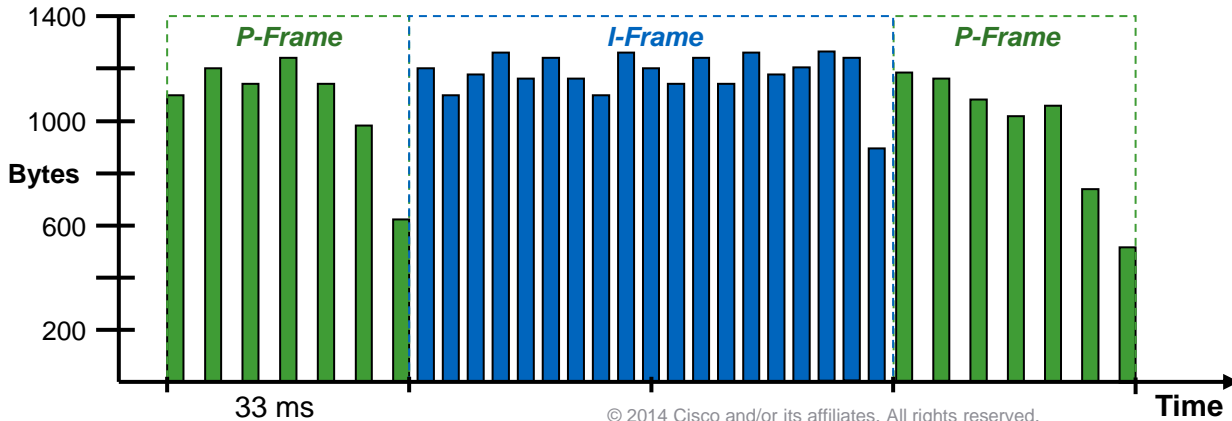
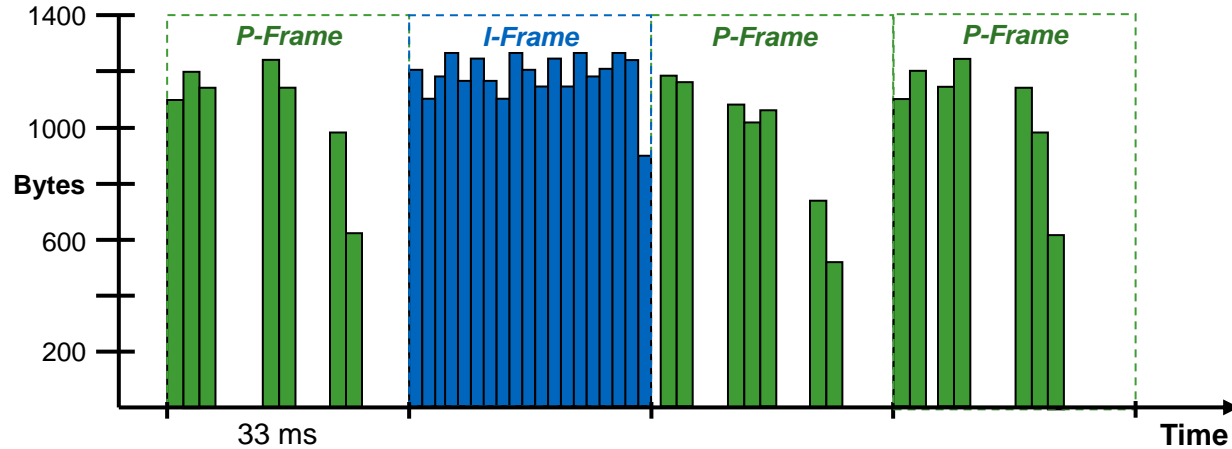
## Impact of Packet Loss on a Video Stream



- Loss of a P-frame triggers request for a new I-frame
  - Encoding and transmitting large I-frame takes time
  - If any of the I-frame packets get lost, the process needs to restart
  - I-frame creates burst that risks exacerbating network congestion (more packet loss!)
- Flickering/pulsing of video when new I-frame arrives
  - Video freeze or artifacts when multiple packets are lost

# Media Resilience

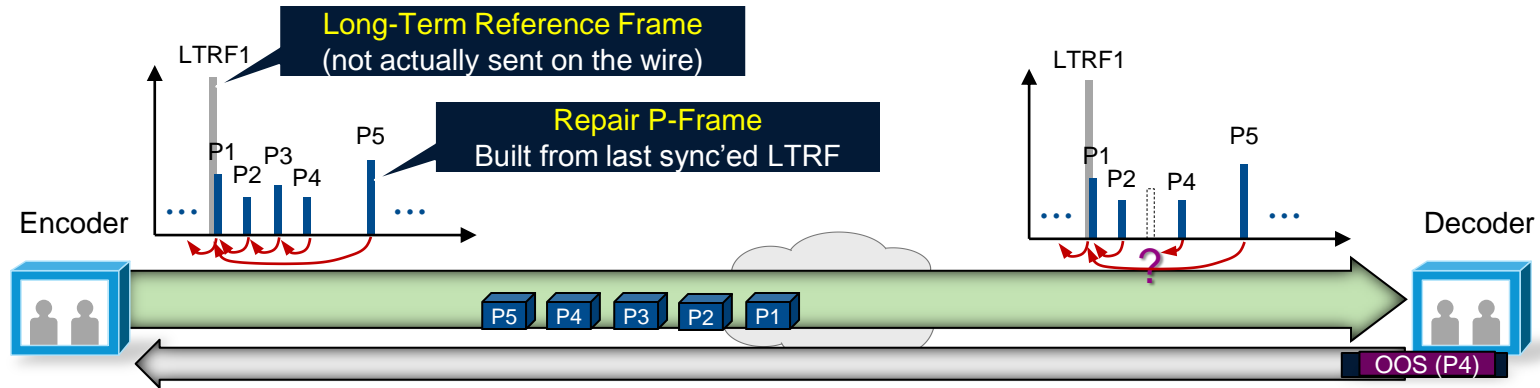
## Encoder Pacing



- Each frame must be packetised onto the wire in 33 ms
- Endpoint packet scheduler disperses packets as evenly as possible
- Large I-frames may need to be “spread” over 2 or 3 frame intervals  
Encoder may then ‘skip’ 1-2 frames to stay within bitrate budget

# Media Resilience

## Long Term Reference Frame (LTRF) with Repair

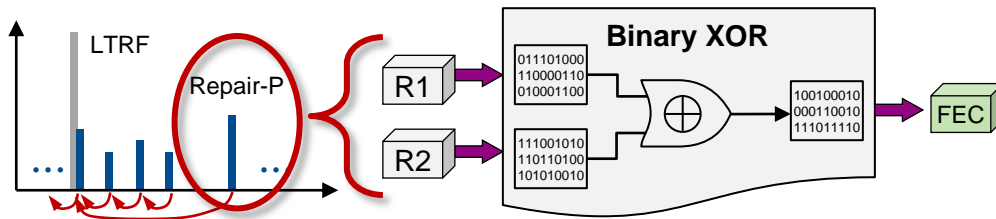


- Keep encoder and decoder in sync with active feedback messages
  - Encoder instructs decoder to store raw frames at specific sync points as Long-Term Reference Frames (part of H.264 standard)
  - Decoder uses “back channel” (i.e. RTCP) to acknowledge LTRF’s
- When a frame is lost, encoder creates a “Repair” P-frame based on the last synchronised LTRF instead of generating a new I-frame

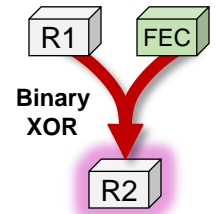
# Media Resilience

## Forward Error Correction (FEC)

Encoder



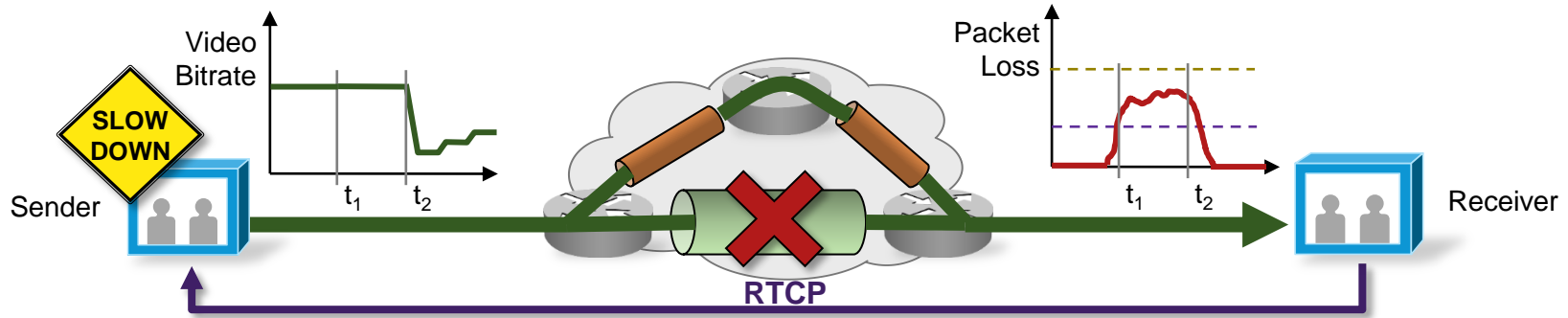
Decoder



- Allows decoder to recover from limited amount of packet loss without losing synchronisation
- Can be applied at different levels ( $x$  FEC packets every  $N$  data packets) to protect “important” frames in lossy environments
- Correction code can be basic (binary XOR) or more advanced (Reed-Solomon)
- Trade-off is bandwidth increase—best suited for non-bursty loss

# Rate Adaptation

## Key Idea



- Receiver observes delay and packet loss over periods of time and signals back using RTCP Receiver Reports (RR)
- Reports cause the sender to adjust bitrate so as to adapt to network conditions (*downspeeding, upspeeding*)
- Two approaches possible:
  - **Sender-initiated** adjustment based on RTCP Receiver Reports
  - **Receiver-initiated** adjustment via call signalling (H.323 flow control, TMBRR, SIP Re-invite) or explicit request in RTCP message

# “Smart” Media Techniques

Support in Cisco Collaboration Devices

remove encoder pacing column if removing encoder pacing slide

Endpoint / Bridge	Encoder Pacing	Rate Adaptation	FEC	LTRF Repair
89xx, 99xx	✓	future	future	--
DX	✓	✓	future	future
WebEx	✓	✓	✓	future
TX	✓	✓	future	✓
Jabber	✓	✓	✓	✓
C/EX/MX/SX/Profile	✓	✓	✓	✓
TS	✓	✓	✓ (3.1)	✓ (3.1)
MCU	✓	✓	✓ (4.5)	✓ (4.5)

ClearPath

Cisco *live!*

# “Smart” Media Techniques

## Key Takeaways

- Burstiness of traffic and mobility of the endpoints make deterministic provisioning for interactive video difficult for network administrators
- **Media resilience** mechanisms help mitigate impact of video traffic on the network and impact of network impairments on video
- Dynamic **rate adaptation** creates an opportunity for more flexible provisioning models for interactive video in Enterprise networks
- Media resilience and rate adaptation also help preserve user experience when video traffic traverses the Internet or non-QoS-enabled networks

# Provisioning and Resource Control Agenda

- **Enhanced Locations CAC Architecture**
  - Network Modelling
  - Locations Bandwidth Manager (LBM)
  - Inter-Cluster E-LCAC with LBM

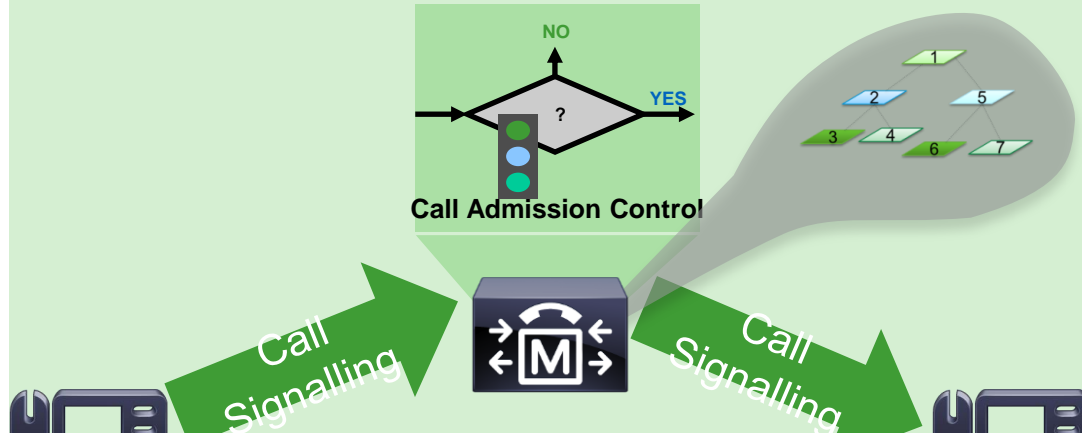




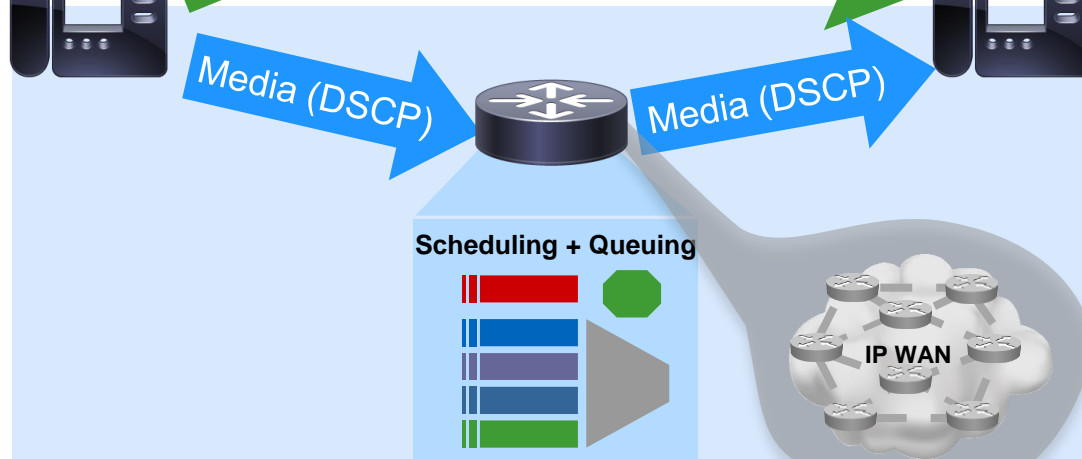
# What's New for Enhanced Locations CAC in 10.0

- Full support for Cross Cluster Extension Mobility
  - Nothing special to configure
  - Requires ELCAC, Inter-cluster ELCAC and Cross Cluster Extension Mobility to all be functioning individually.
- Video Promotion and Flexible DSCP
  - Supported on DX650
  - Planned support for 99xx/98xx video phones, CTS, TX and C series
- Support for Encrypted signalling between inter-cluster LBM Hubs (only related to inter-cluster ELCAC)

# Control Plane



E-LCAC



QoS

# Data Plane



# LCAC Limitations

## Limited WAN Topology Support:

- Hub and Spoke WAN topology support
- Large gap between RSVP and Locations CAC

## Multi Cluster Support:

- Multiple Clusters that managed endpoints in same branch sites could only inefficiently subdivide inter-branch bandwidth to avoid quality degradation (Ships in the night CAC)

## TelePresence:

- Did not Support CAC (overlay design)
- TelePresence and UC or 3rd party video on a single cluster
- Limited CAC support for TelePresence video interoperability (P2P calls without an MCU)

# E-LCAC Solutions

## Network Modelling:

- Convert UCM locations to a model capable of supporting real network topologies

## Inter-Cluster (Inter-cluster) CAC:

- Implement a bandwidth-accounting scheme that works between multiple Unified CM clusters and dynamically learns the topology from one another

## Immersive Bandwidth Allocations:

- Implement an immersive BW pool in locations CAC
- Provide better CAC interop support between TelePresence Video and Desktop Video

# Network Modelling



Enhanced Locations CAC Architecture

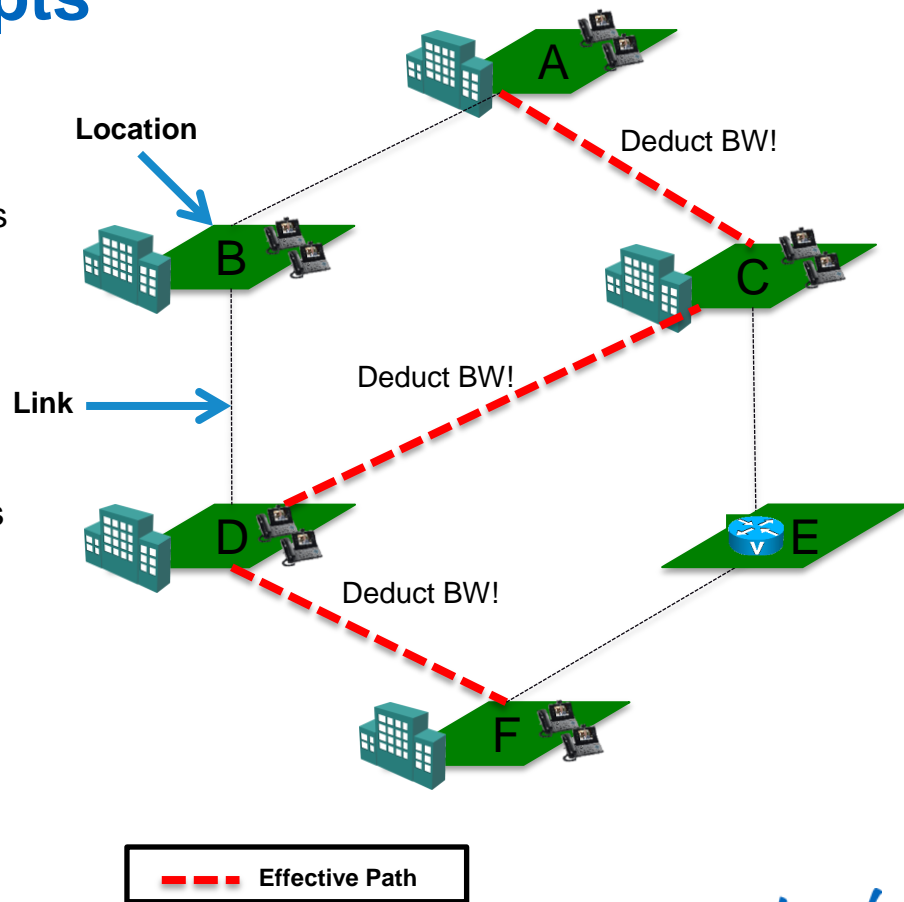
**Network Modelling**

Locations Bandwidth Manager (LBM)

Inter-Cluster E-LCAC with LBM

# Network Modelling - Concepts

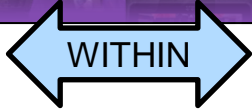
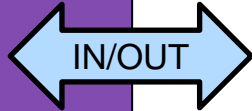
- Administrator builds a Network Model using locations and links
- A **Location** represents a LAN. It could contain endpoints or simply serve as a transit location between links for WAN network modelling
- **Links** interconnect locations and are used to define bandwidth available **between** locations. Links logically represent the WAN link
- **Weights** are used on links to provide a “cost” to the “effective path”. Weights are pertinent only when there is more than 1 path between any 2 locations
- UCM calculates shortest paths (least cost) from all locations to all locations and builds the effective paths
- The **Effective paths** are the paths with the “least cumulative weight”
- UCM tracks bandwidth across any link that the network model indicates from originating Location to terminating location.



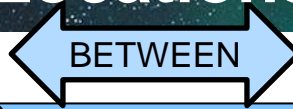
# Network Modelling – Locations and Links

## Location A

BW	Allocated
Audio	Unlimited
Video	100MB
Immersive	250MB

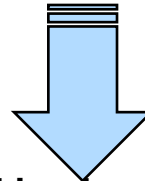


Locations Provide Bandwidth Accounting **WITHIN** the Location as well as **IN** or **OUT** of the Location



Link A < > B	
BW	Allocated
Audio	1500k
Video	3000k
Immersive	5000k

Links Provide Bandwidth Accounting Between Locations And Interconnect Locations



## Linking Locations



## Location B

BW	Allocated
Audio	Unlimited
Video	Unlimited
Immersive	Unlimited



TelePresence



TelePresence and UC Video Endpoints Can Reside in the Same Location\*

# Network Modelling – Locations and Links

## Intra-Location Bandwidth Allocation – TelePresence Immersive

- Links Interconnect Locations to Build the Topology. Bandwidth Values and Weight are Assigned to Links
- Intra-location Bandwidth Limits are Assigned to a Location to CAC ALL calls made **TO/FROM/WITHIN** the Location. Intra-location Bandwidth Values are Unlimited by Default.

**Location Information**

Name\* PDX

**Links - Bandwidth Between PDX and Adjacent Locations**

Locations (1 - 4 of 4) Rows per Page 50

Find Locations where name begins with Find Clear Filter

<input type="checkbox"/>	Location ^	Weight	Audio Bandwidth	Video Bandwidth	Immersive Bandwidth
<input type="checkbox"/>	BLD	50	80	384	UNLIMITED
<input type="checkbox"/>	EUG	50	80	384	UNLIMITED
<input type="checkbox"/>	SEA	50	80	384	UNLIMITED
<input type="checkbox"/>	YVR	50	80	384	UNLIMITED

Add Select All Clear All Delete Selected

**Links - Bandwidth Between PDX and Adjacent Locations**

Location SEA

Weight\* 50

Audio Bandwidth  Unlimited  80 kbps

Video Bandwidth  None  384 kbps  Unlimited

Immersive Video Bandwidth  None  kbps  Unlimited

If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Save Close

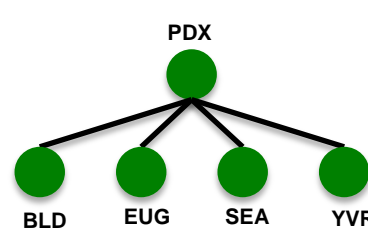
**Intra-location - Bandwidth for Devices WITHIN This Location**

Audio Bandwidth  Unlimited  kbps

Video Bandwidth  Unlimited  kbps  None

Immersive Video Bandwidth  Unlimited  kbps  None

If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.



As Viewed From  
The Perspective of  
The PDX Location.  
Serviceability  
Provides More  
Tools for Topo  
Visibility

Cisco live!

# Network Modelling – Locations and Links

## The Location Admin Page Has Been Updated To Configure Location Links

- By default when a new location is created a link to Hub. None will be added with unlimited audio bandwidth, 384 kb of both video and immersive bandwidth
- RECOMMENDATION: DELETE the link when it's not needed

The screenshot displays the Cisco Location Admin interface. On the left, the 'Location Information' section shows the 'Name\*' field set to 'NewLocation'. Below it, the 'Links - Bandwidth Between This Location and Adjacent Locations' section features a dropdown menu with 'Hub\_None' selected. The main panel, titled 'Links - Bandwidth Between NewLocation and Adjacent Locations', shows a table with one entry: 'Hub\_None' with a weight of 50, unlimited audio bandwidth, and 384 kbps for video and immersive bandwidth. A 'Delete Selected' button is highlighted with a black box. A green arrow points from the bottom right towards the 'Delete Selected' button.

**Location Information**

Name\* NewLocation

**Links - Bandwidth Between This Location and Adjacent Locations**

Location: BLD, EOS, **Hub\_None**, NYC, PDX

Weight\*: 50

Audio Bandwidth:  Unlimited  [ ] kbps

Video Bandwidth:  None  384 kbps  Unlimited

Immersive Video Bandwidth:  None  384 kbps  Unlimited

If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

**Links - Bandwidth Between NewLocation and Adjacent Locations**

Locations (1 - 1 of 1) Rows per Page 50

Find Locations where name begins with [ ] Find Clear Filter [+] [-]

<input type="checkbox"/>	Location ^	Weight	Audio Bandwidth	Video Bandwidth	Immersive Bandwidth
<input checked="" type="checkbox"/>	Hub_None	50	UNLIMITED	384	384

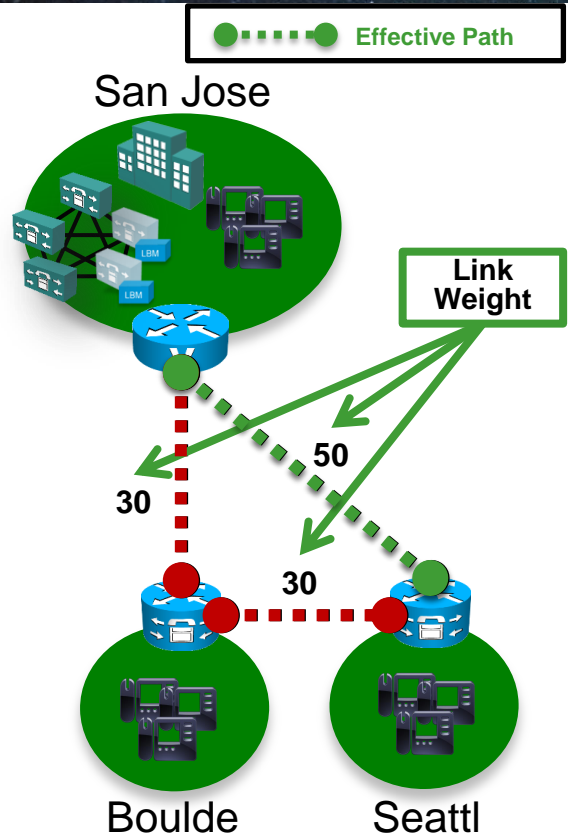
Add Select All Clear All **Delete Selected**

[Show Advanced](#)



# Network Modelling in Locations CAC

## Links, Weights and “Effective Path”



- Weight provides the ability to force a specific path choice when multiple paths between 2 locations are available
- When Multiple Paths are configured yet only 1 will be selected based on “Weight”. This path is the “**Effective Path**”
- Weight is used to determine path cost, lowest weight path from source to destination is selected
- Weight is static and does not change with regards to the “effective path” from one location to another

### EFFECTIVE PATH

Path 1:

San Jose > Seattle (Weight = 50 = 50)

Path 2:

San Jose > Boulder > Seattle (Weight = 30 + 30 = 60)

# Network Modelling in Locations CAC

## Links, Weights and “Effective Path”

- The Locations Bandwidth Manager (LBM) service computes the effective path from source location to destination location:
  - Sum weight of links across each possible path from source to destination
  - The least cost value of the path’s weight determines the “Effective Path”
  - A tie break of equally weighted paths is determined by LBM based on location name
  - Once the effective path is determined, all subsequent calls that have the same source and destination locations will use the same “Effective Path”

***Serviceability > Tools > Locations > Effective Path: Provides the Ability to Ascertain the “Effective Paths” Configured in the Topology***

Detailed Path View

Location Name	Weight (1-100)	Audio Bandwidth (kbps)	Video Bandwidth (kbps)	Immersive Bandwidth (kbps)
PDX		Configured: Unlimited Available: Unlimited	Configured: Unlimited Available: Unlimited	Configured: <b>Unlimited</b> Available: <b>Unlimited</b>
▼	50	Configured: <b>160</b> Available: <b>160</b>	Configured: <b>2048</b> Available: <b>2048</b>	Configured: <b>Unlimited</b> Available: <b>Unlimited</b>
BLD		Configured: Unlimited Available: Unlimited	Configured: Unlimited Available: Unlimited	Configured: <b>Unlimited</b> Available: <b>Unlimited</b>
▼	50	Configured: <b>160</b> Available: <b>160</b>	Configured: <b>2048</b> Available: <b>2048</b>	Configured: <b>Unlimited</b> Available: <b>Unlimited</b>
NYC		Configured: Unlimited Available: Unlimited	Configured: Unlimited Available: Unlimited	Configured: <b>Unlimited</b> Available: <b>Unlimited</b>

Bf

# Network Modelling

## Key Takeaways

- Enhanced Locations CAC is a Static Model-Based CAC Mechanism
- E-LCAC is a Model of the “Routed Network” Attempting to Represent How The WAN Network Topology Routes Media
- Network Modelling is NOT Dynamic like RSVP
- The Model Needs to be Updated When the Network Topology Changes
- E-LAC is Call-Based (No Asymmetric or Unidirectional Bandwidth Deductions)
- Intra-location bandwidth assignment and deduction. The default is set to unlimited.

# Locations Bandwidth Manager (LBM)



Enhanced Locations CAC Architecture

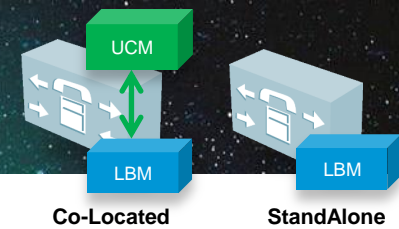
Network Modelling

**Locations Bandwidth Manager (LBM)**

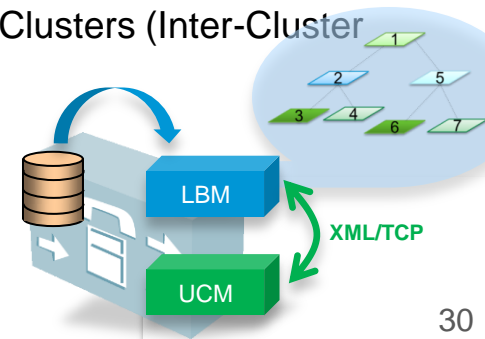
Inter-Cluster E-LCAC with LBM

# Location Bandwidth Manager (LBM)

## Service Overview



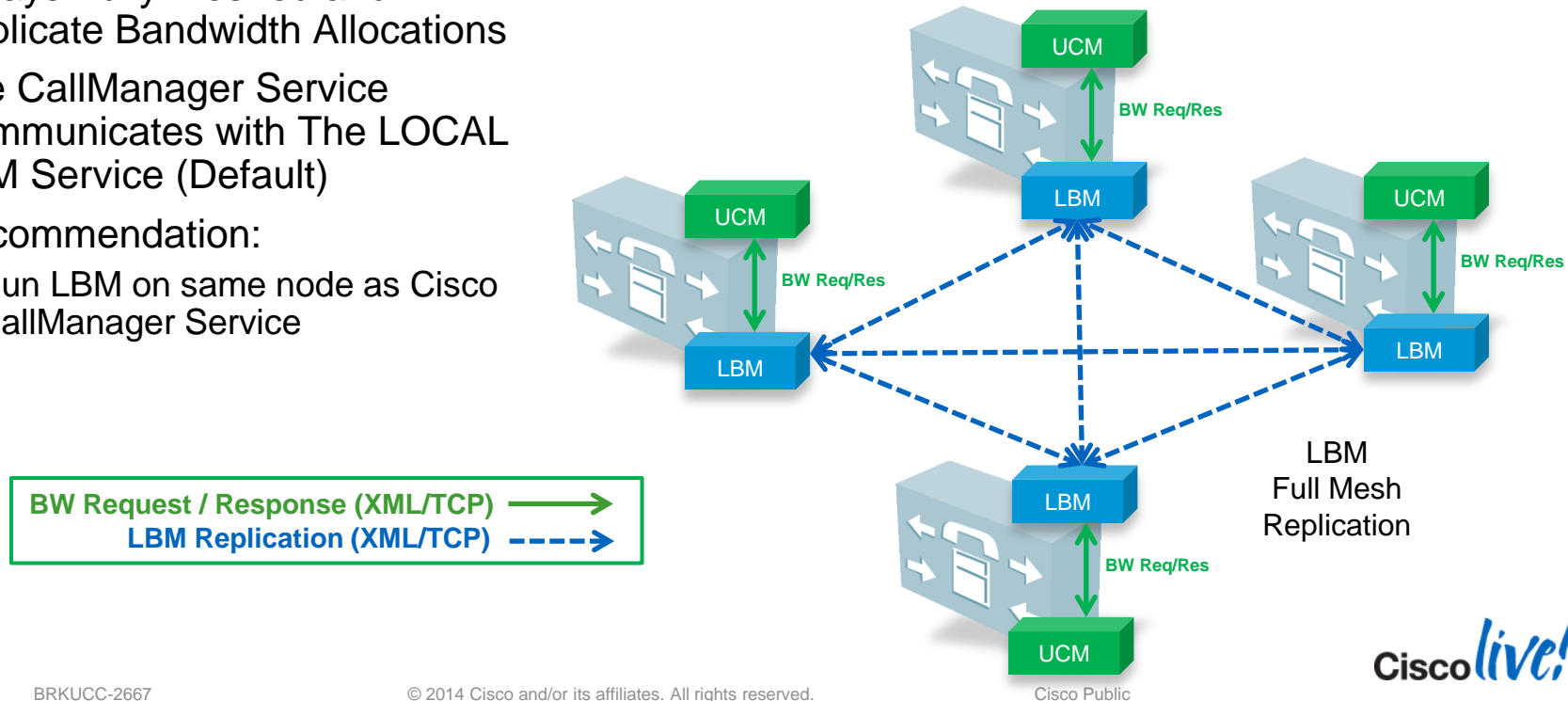
- LBM is a New Unified CM Feature Service
- LBM Service is Enabled by Default When Upgraded from a Pre-9.0 Installation
- For Fresh Installs The LBM Service Needs to be Manually Activated (like CCM service)
- LBM Can Run on Any UCM Subscriber or Standalone
- For E-LCAC to Function LBM Must Be Enabled
- Functions of LBM:
  - Location Path Assembly and Calculation
  - Servicing Bandwidth Requests from Unified CM Call Control (XML/TCP)
  - Replication of Bandwidth Information to Other LBMs Within and Between Clusters (Inter-Cluster Locations CAC)
  - Provides Configured and Dynamic information to Serviceability
  - Updates Location RTMT counters



# Location Bandwidth Manager Communication and LBM Replication

- LBM Services Within a Cluster Are Always Fully Meshed and Replicate Bandwidth Allocations
- The CallManager Service Communicates with The LOCAL LBM Service (Default)
- Recommendation:
  - Run LBM on same node as Cisco CallManager Service

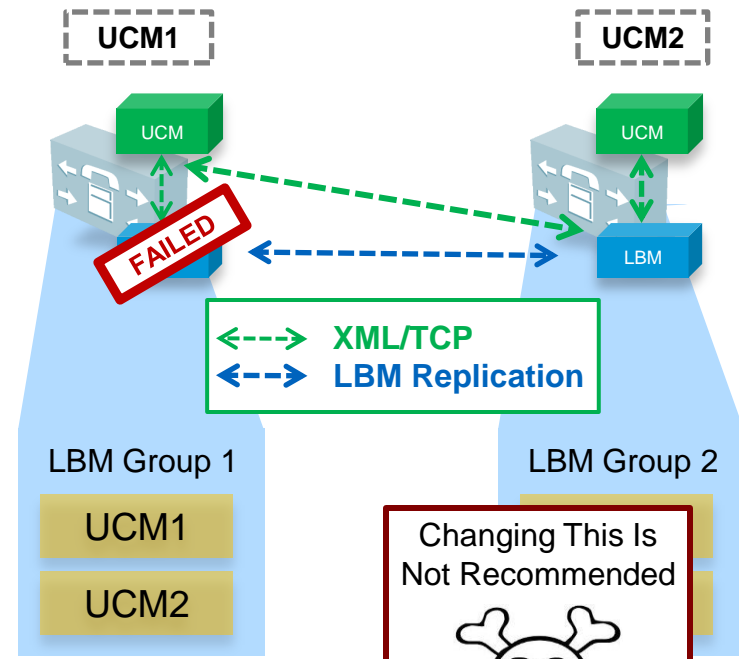
## 4 node Cluster



# LBM Redundancy

## Service and Recommendations

- LBM Group allows control of Active and Standby LBMs
- Provides redundancy of LBM for the UCM service during an LBM “service outage”
- **Unified CM LBM Usage Order:**
  1. LBM Group Designation (1, 2)
  2. Local LBM
  3. Service Param: “Call Treatment when no LBM available” (allow calls = Default)
- **LBM Recommendations:**
  - Run LBM on same node as Cisco CallManager Service
  - LBM Group: Co-located LBM first, LBM from local subscriber second, otherwise no redundancy.



Clusterwide Parameters (Call Admission Control)	
<a href="#">Call Counting CAC Enabled</a> *	False
<a href="#">Audio Bandwidth For Call Counting CAC</a> *	152
<a href="#">Video Bandwidth For Call Counting CAC</a> *	500
<a href="#">UCM to LBM Periodic Reservation Refresh Timer</a> *	5
<a href="#">Maximum Bandwidth Deduction Duration</a> *	720
<a href="#">Call Treatment When No LBM Available</a> *	Allow Calls
<a href="#">Locations Media Resource Audio Bit Rate Policy</a> *	Lowest Bit Rate

# Location Bandwidth Manager (LBM)

## LBM Group Configuration

### LBM Group Config

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾

Server

- Cisco Unified CM
- Cisco Unified CM Group
- Phone NTP Reference
- Date/Time Group
- BLF Presence Group
- Region Information
- Device Pool
- Device Mobility
- DHCP
- LDAP
- Location Info
  - Location
  - Location Bandwidth Manager Group
  - Location Bandwidth Manager Hub Group
- SRST
- MLPP

Manager Groups

Group (1 - 1 of 1)

Name
SEA_LBM

### Location Bandwidth Manager Group Configuration

Save

**Status**

Status: Ready

**Location Bandwidth Manager Group Setting**

Name\* SEA\_LBM

Description

**Location Bandwidth Manager Group Members**

Active Member\* 10.10.30.41

Standby Member < None >

Save

**Cisco Unified Communications Manager Information**

Cisco Unified Communications Manager: CM\_SEAUCM (used by 4 devices)

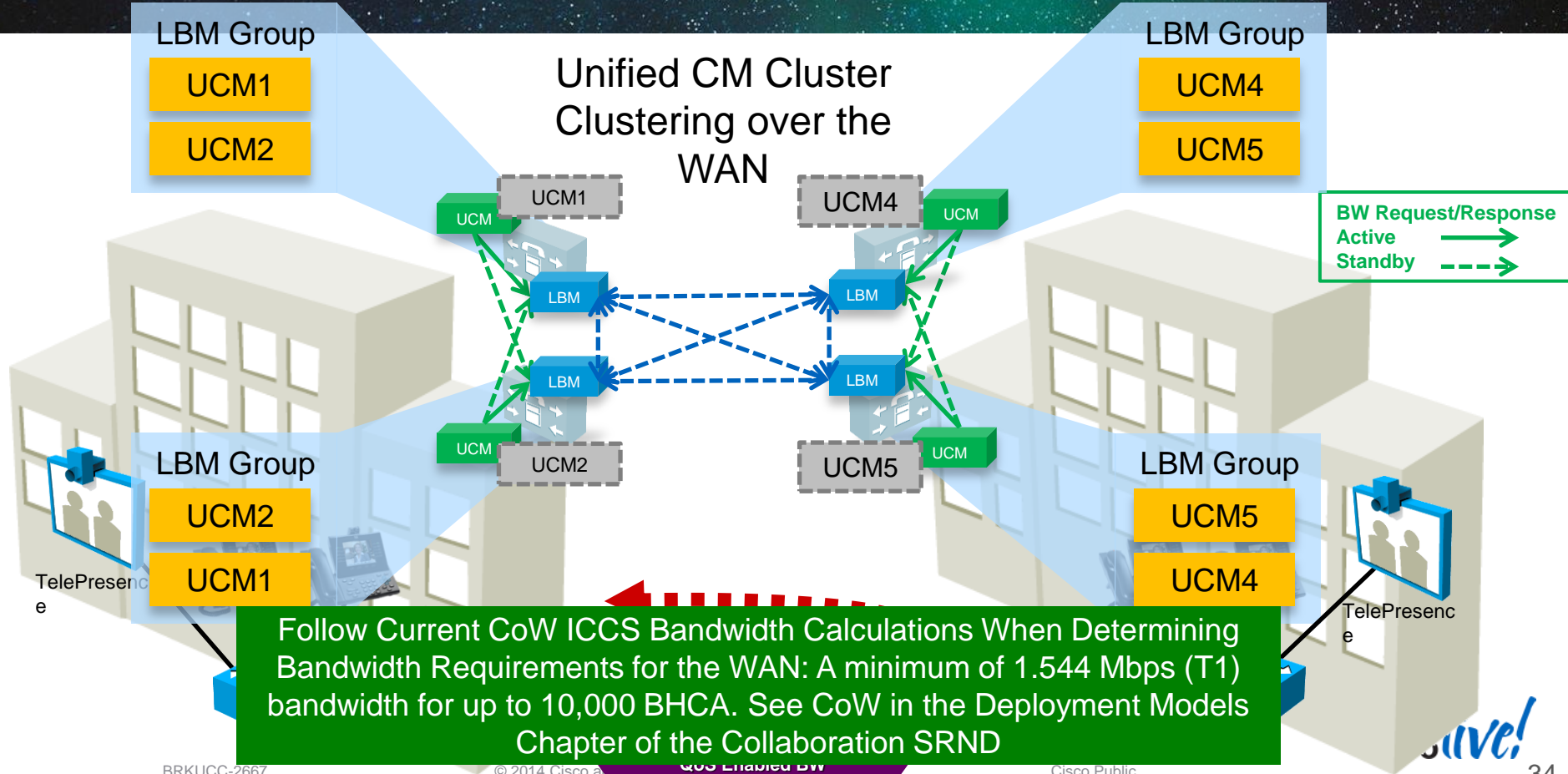
**Server Information**

CTI ID	1
Cisco Unified Communications Manager Server*	10.10.30.41
Cisco Unified Communications Manager Name*	CM_SEAUCM
Description	SEAUCM
Location Bandwidth Manager Group	SEA_LBM

An LBM group association determines which LBMs a UCM service communicates with. In absence of an LBM group UCM communicates with the local LBM ONLY.



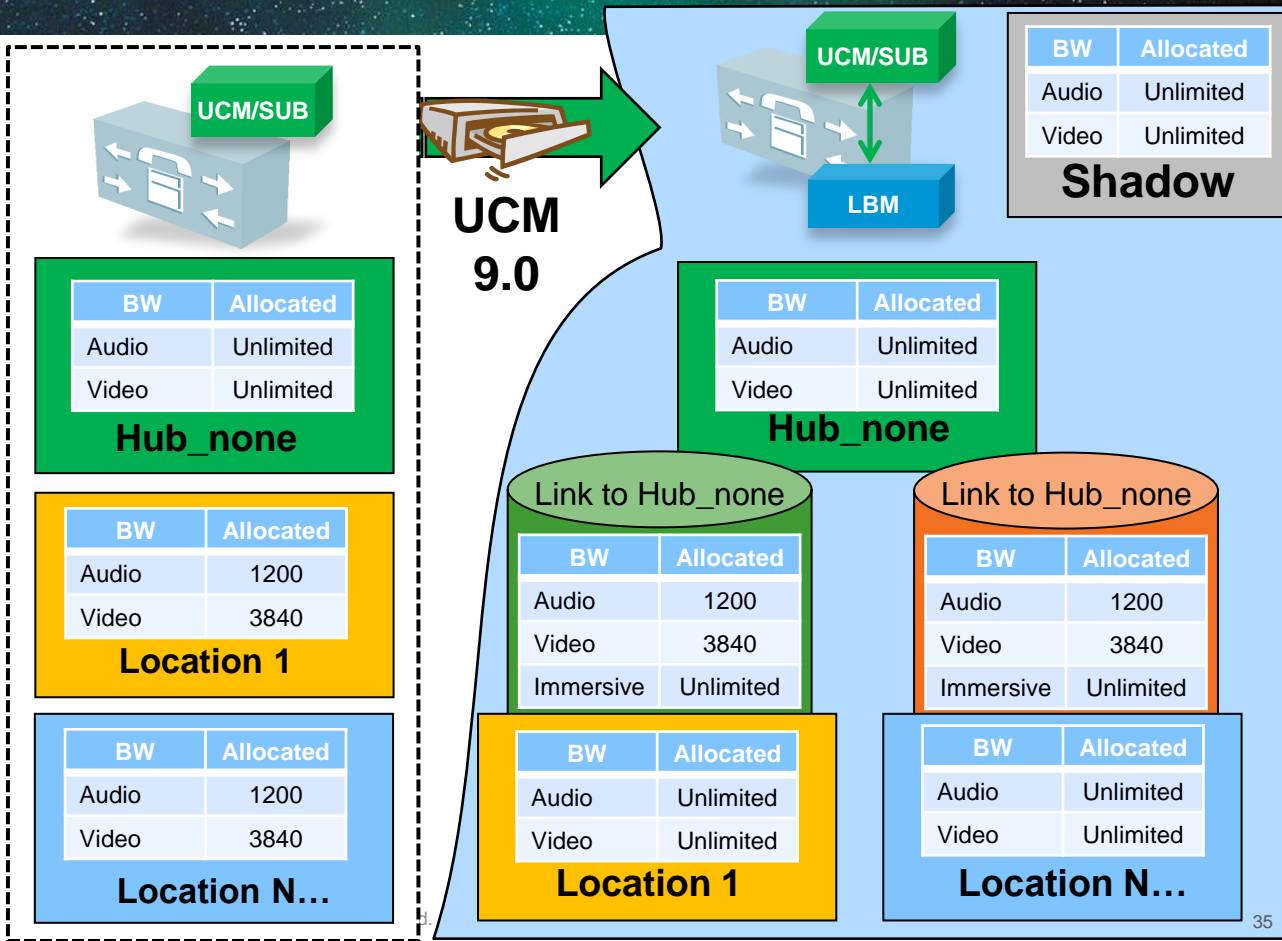
# Clustering over the WAN (CoW) - LBM



# Migration to Enhanced Locations CAC

## Settings After An Upgrade To 9.0

- LBM is activated on each UCM subscriber
- No LBM groups or LBM hub groups
- UCM service communicates with local (on node) LBM
- Fully meshed LBM services
- No inter-cluster E-LCAC
- Intra-location bandwidth values are set to unlimited
- Location bandwidth values are migrated to a link inter-connecting the migrated location and Hub\_None
- Phantom and Shadow locations have no links



# Sizing and Performance

- 2000 Max Locally Configured Locations
- 8000 Max Total Replicated Locations (Globally distinct locations)
- Sizing with LBM Co-located (Cisco Sizing Tool Will Assume LBM Impact)

# Key Takeaways

## Summary

- LBM is a New Feature Service
- LBM is Fully Meshed Within The Cluster
- LBM is Responsible For Modeled Topology and Servicing UCM Requests
- Recommendations for LBM Group Usage
  - Run LBM on Each Unified CM Subscriber
  - Use LBM Group to create a backup LBM for the CallManager service

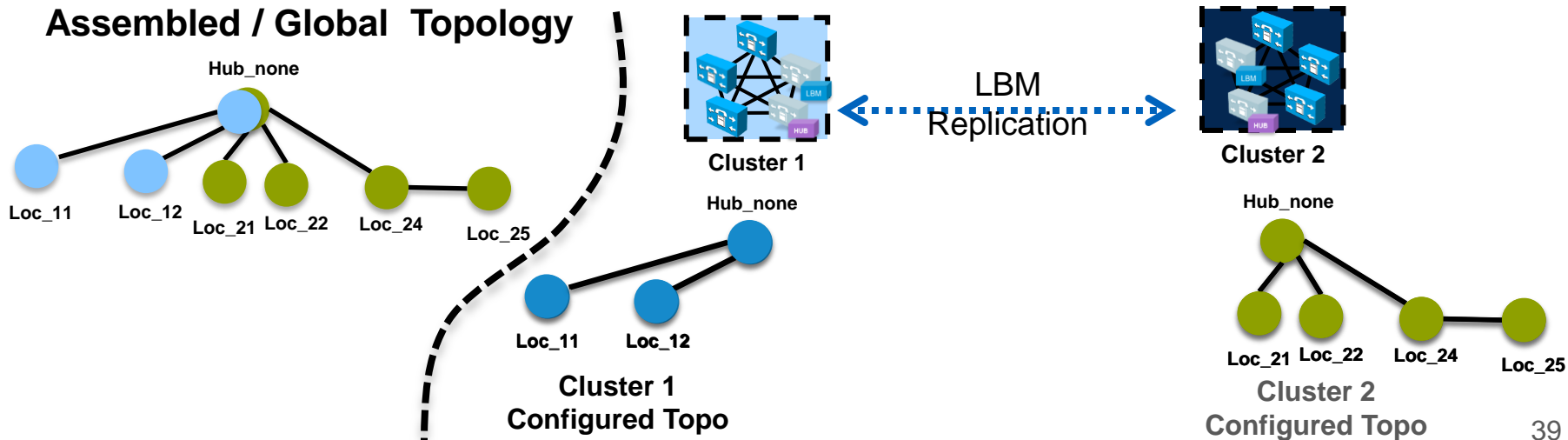
# Inter-Cluster E-LCAC with LBM



Enhanced Locations CAC Architecture  
Network Modelling  
Locations Bandwidth Manager (LBM)  
**Inter-Cluster E-LCAC with LBM**

# Inter-Cluster Enhanced Locations CAC

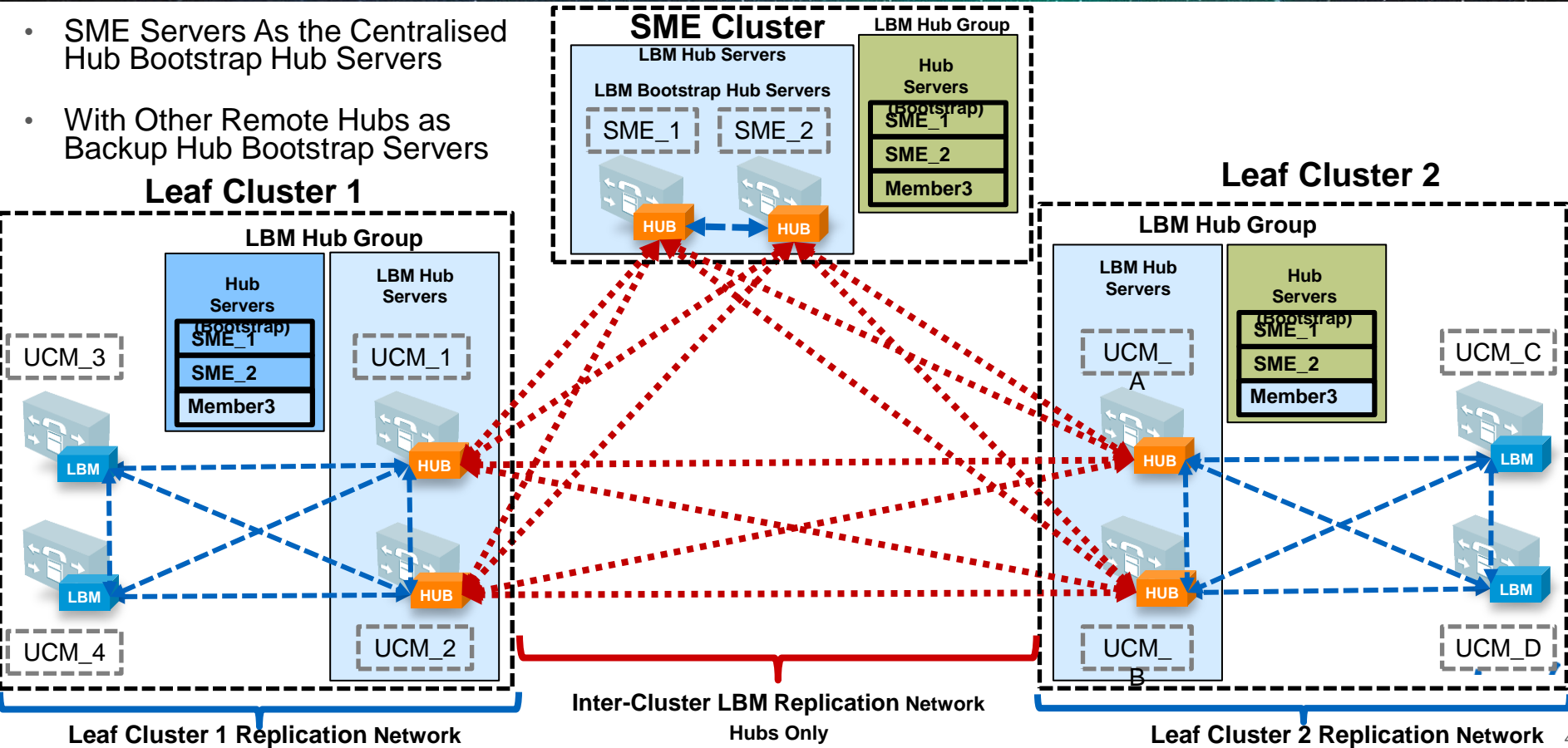
- Extends Enhanced Locations CAC Network Modelling Across Multiple Clusters
- Each Cluster Manages Its Own Topology
- Each Cluster Then Propagates Its Topology to Other Clusters Configured In the LBM Inter-Cluster Replication Network
- Each Cluster Then Creates a Global Topology (“Assembled Topology”) Piecing Together Each Clusters Replicated Topology



# LBM Network – Hubs, Spokes and Hub Bootstrap

Centralised Hub Bootstrap server for the LBM replication network

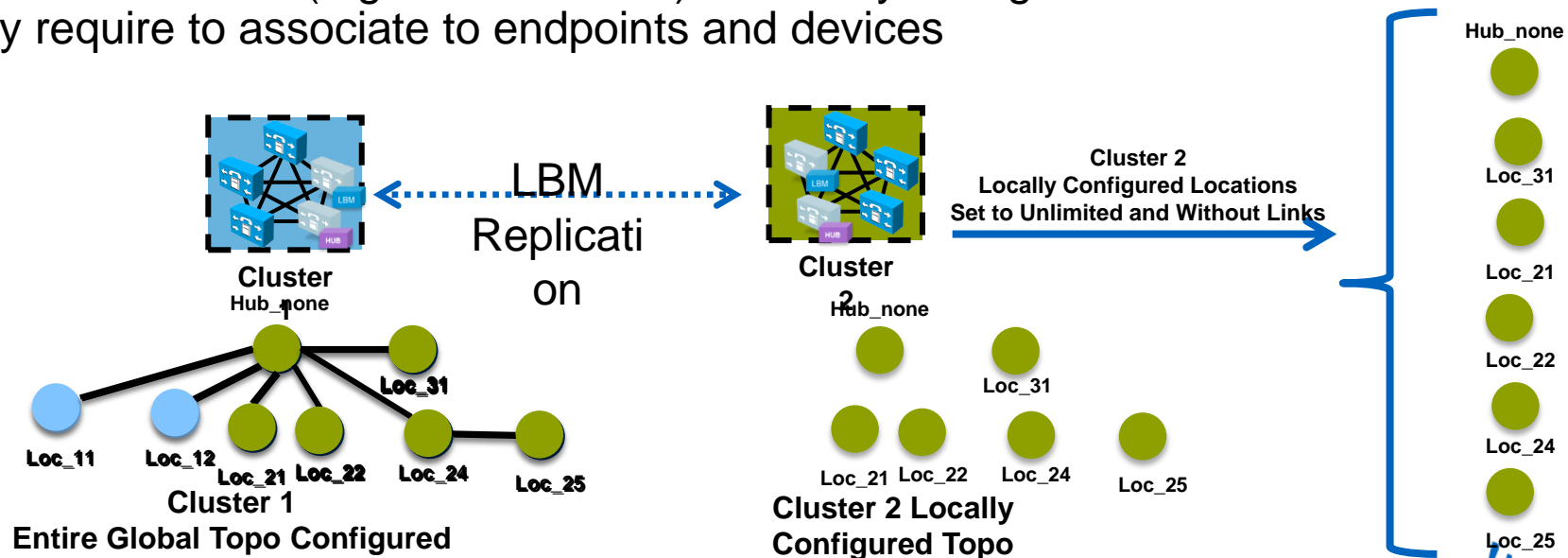
- SME Servers As the Centralised Hub Bootstrap Hub Servers
- With Other Remote Hubs as Backup Hub Bootstrap Servers



# Inter-Cluster Enhanced Locations CAC

## Locations and Links Management Cluster

- Single Cluster manages ALL Locations and Links for the entire Locations Replication Network
- All Other Clusters (e.g. Leaf Clusters) need only configure the Locations that they require to associate to endpoints and devices

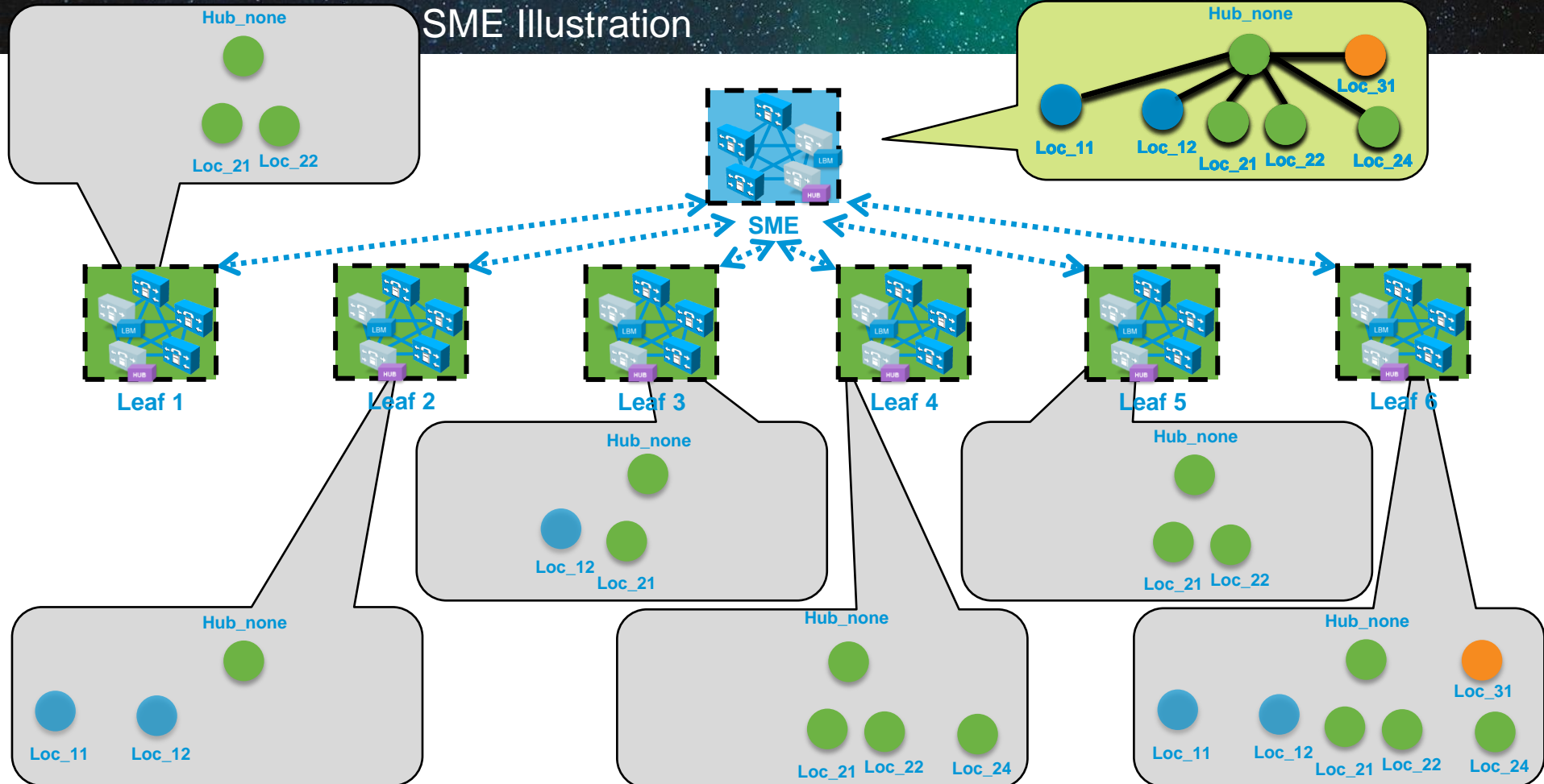




# Location and Link Mgmt Cluster

←.....→ LBM Replication

SME Illustration



# Inter-cluster Enhanced Locations CAC

For Further Information

See the addendum for more information on the following:

- Inter-cluster CAC Operation and Configuration
- Inter-cluster CAC design: Location and Link Management Cluster
- TelePresence and UC Video Differentiation in Admission Control

For a complete overview of E-LCAC see Orlando Presentation and VoD

[www.ciscolive365.com](http://www.ciscolive365.com) > BRKUCC-2343

[https://www.ciscolive.com/online/connect/sessionDetail.wv?SESSION\\_ID=7978&backBtn=true](https://www.ciscolive.com/online/connect/sessionDetail.wv?SESSION_ID=7978&backBtn=true)

# Audio and Video Admission Control

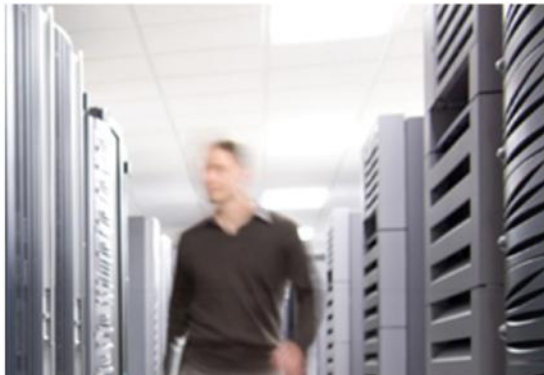
## Considerations

### No Admission Control

- Over-provision queues
- Rely on video rate adaptation and media resilience capabilities
- Audio is much easier to over-provision in pervasive video deployments
- QoS critical and rate adaptation is highly beneficial for both managed/unmanaged networks
- Benefits: Simplicity

### Admission Control

- Strict provisioning (Mapping CAC to Queuing)
- Mobility? Device Mobility feature (Add OPEX)
- Jabber? Medianet Metadata is recommended to align QoS / CAC
- Benefits:
  - Manage lower bandwidth links, use AAR for PSTN redirect
  - Ensure quality audio during the busy hour by avoiding oversubscription and packet loss
  - Safe when over-provisioning is not an option

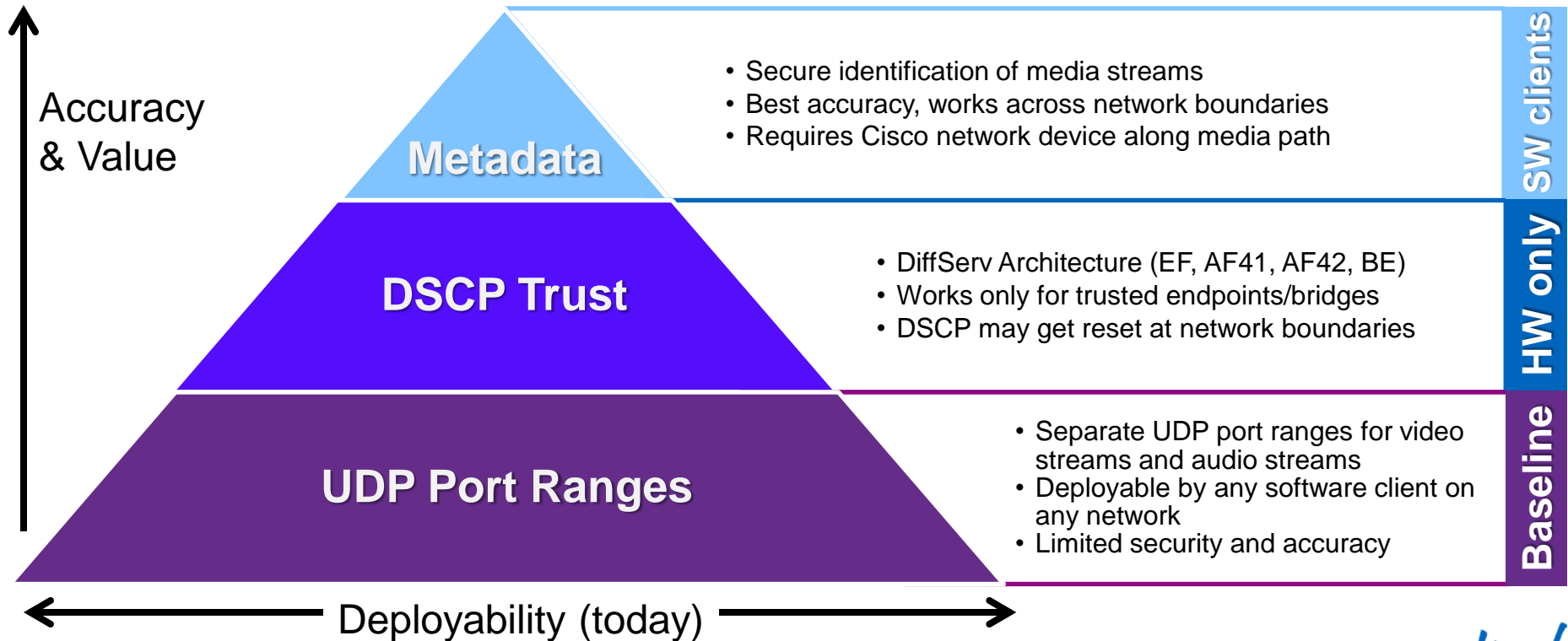


## QoS Architecture:

**APPROACH OVERVIEW**  
IDENTIFICATION & CLASSIFICATION  
QUEUING & SCHEDULING

# QoS Strategy

## Collaboration Traffic Identification

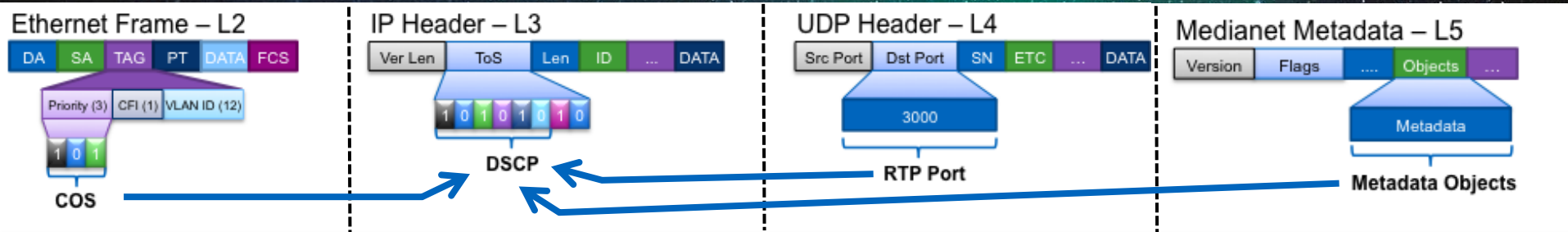


# QoS Recommendations (RFC 4594-Based)

Application Class	Per-Hop Behaviour	DSCP	Queuing & Dropping	Application Examples
VoIP Telephony	EF	46	Priority Queue (PQ)	Cisco IP Phones (G.711, G.729)
Broadcast Video	CS5	40	BW Queue	Cisco IP Video Surveillance / Cisco Enterprise TV
Realtime Interactive	CS4	32	BW Queue + DSCP WRED	Cisco TelePresence
Multimedia Conferencing	AF4	34/36/38	BW Queue + DSCP WRED	Cisco Unified Personal Communicator, WebEx
Multimedia Streaming	AF3	26/28/30	BW Queue + DSCP WRED	Cisco Digital Media System (VoDs)
Network Control	CS6	48	BW Queue	EIGRP, OSPF, BGP, HSRP, IKE
Call-Signalling	CS3	24	BW Queue	SCCP, SIP, H.323
Ops / Admin / Mgmt (OAM)	CS2	16	BW Queue	SNMP, SSH, Syslog
Transactional Data	AF2	18/20/22	BW Queue + DSCP WRED	ERP Apps, CRM Apps, Database Apps
Bulk Data	AF1	10/12/14	BW Queue + DSCP WRED	E-mail, FTP, Backup Apps, Content Distribution
Best Effort	DF	0	Default Queue + RED	Default Class
Scavenger	CS1	8	Min BW Queue (Deferential)	YouTube, iTunes, BitTorrent, Xbox Live

# Trust and Enforcement:

## QoS Marking in the LAN and WAN

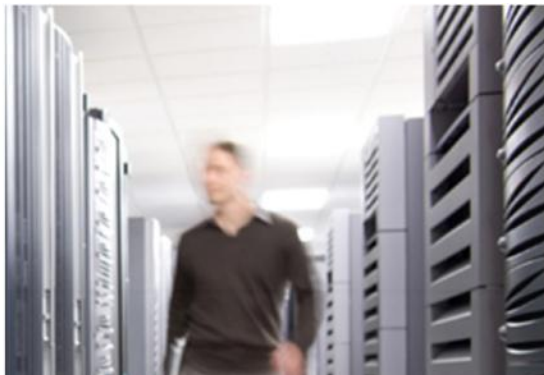


- Values 0-7
- Limited values to differentiate multiple traffic beyond 7 classes
- Goal: Use when needed in L2 environments to map to DSCP. Use DSCP when possible.

- 0-64: EF, CS5, CS4, AF4, AF3, CS6, CS3, CS2, AF2, AF1, DF, CS1, ....
- Many more values to differentiate traffic
- Goal: Get all traffic to a DSCP value

- Protocol Destination Port range.
- Difficult to differentiate Audio and Video
- Goal: Use to apply DSCP to traffic when no COS or DSCP is available.

- Large quantity of matching criteria to differentiate traffic based on signalling, media type, device models, etc....
- Requires network support
- Goal: Use to apply DSCP to traffic when no COS or DSCP is available.



## QoS Architecture:

APPROACH OVERVIEW  
**IDENTIFICATION & CLASSIFICATION**  
QUEUING & SCHEDULING



# Identification and Classification Agenda

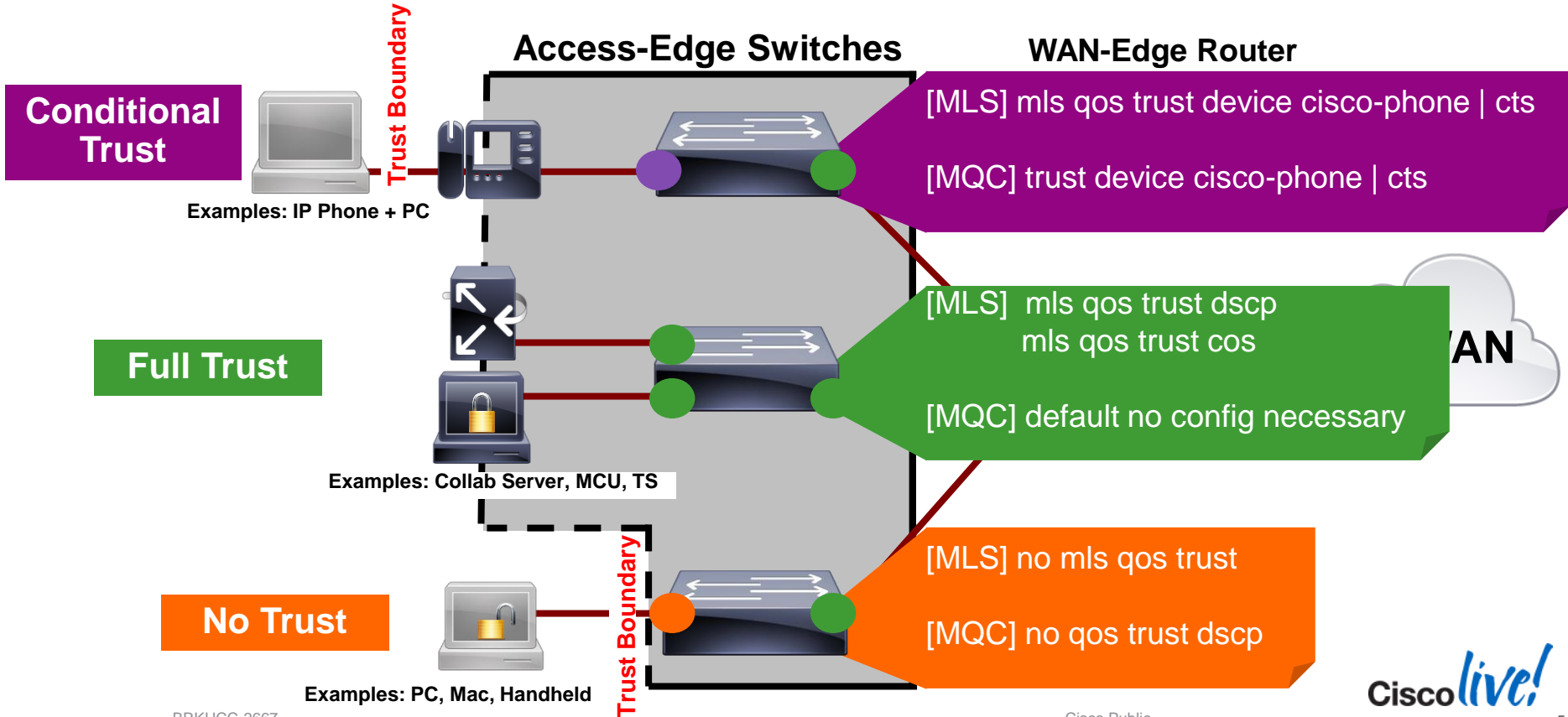
- **Trusted Devices**
- Untrusted Devices:
  - Mapping UDP/TCP Port Ranges
  - Medianet Metadata



# Campus QoS Design Considerations

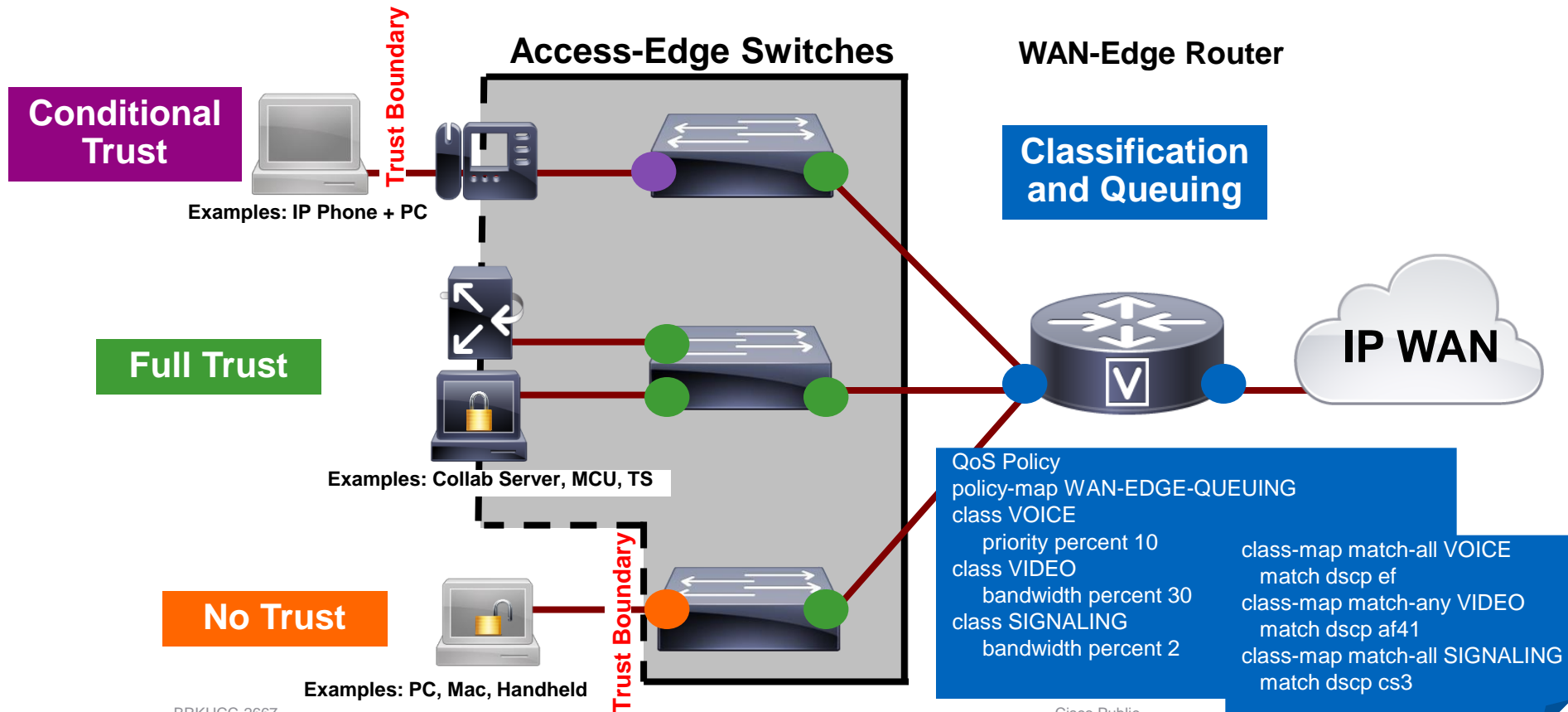
## Trust Boundaries

BRKCRS-2501 Campus QoS  
Design Simplified

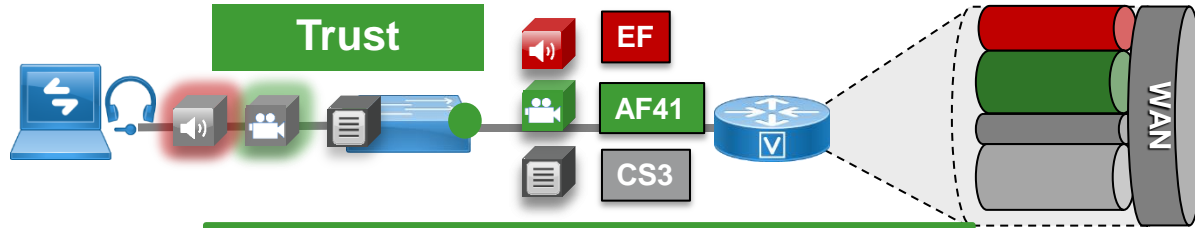


# Campus QoS Design Considerations

## Trust Boundaries



# Egress Classification and Queuing



1

**! This section applies the policy-map to the Interface**  
Router (config-if)# service-policy output EGRESS-QUEUING  
**! Attaches service policy to interface**

**! This section configures the bandwidth for all collab traffic**  
policy-map EGRESS-QUEUING  
class VOICE  
priority percent 10  
**! Provisions 10% LLQ to VOICE class**  
class VIDEO  
bandwidth percent 30  
**! Provisions 30% CBWFQ to VIDEO class**  
class SIGNALING  
bandwidth percent 2  
**! Provisions 2% CBWFQ to SIGNALING class**  
...

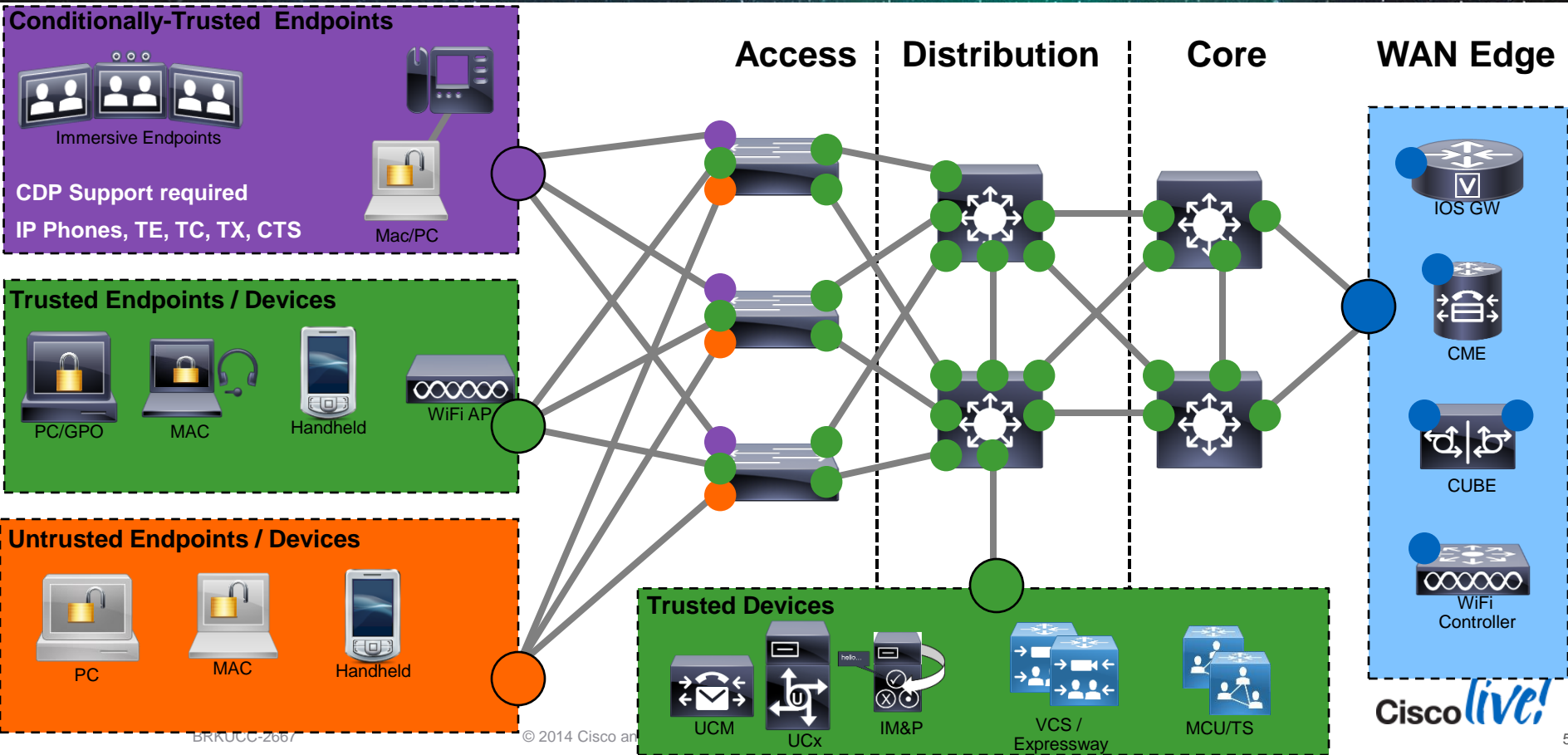
2

**! This section applies the policy-map**  
class-map match-all VOICE  
match dscp ef  
class-map match-any VIDEO  
match dscp af41  
match dscp af42  
class-map match-all SIGNALING  
match dscp cs3

3

# Trust and Enforcement:

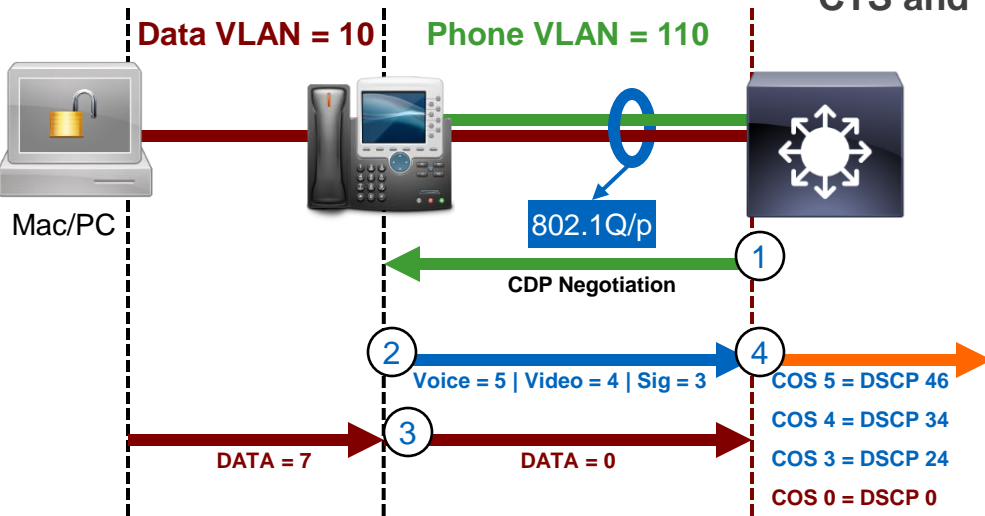
## Devices and Places in the Network



# Quality of Service

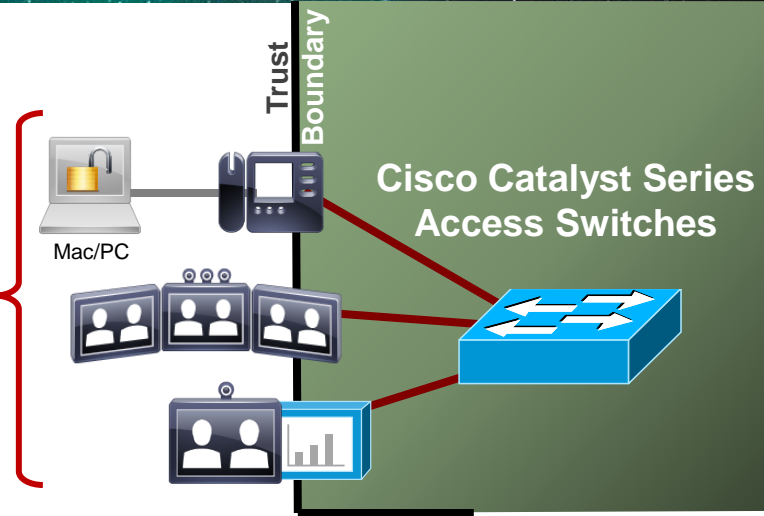
## CDP to Establish QoS Trust Boundary – “Conditional Trust”

1. CDP exchange, trust boundary extended to endpoint
2. Phone sets COS (audio 5, video 4, signalling 3)
3. Phone re-writes COS of Computer port traffic to 0
4. Switch trusts COS from endpoint and maps COS → DSCP



**CDP:**

Cisco IP Phones  
TE and TC Series  
CTS and TX Series



### Commands:

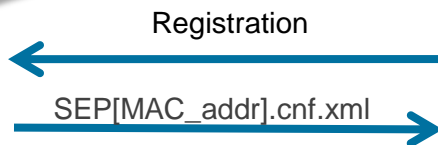
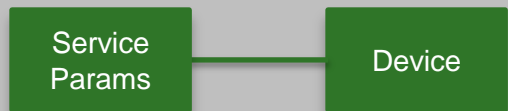
```
[MLS] mls qos trust device cisco-phone | cts  
[MQC] trust device cisco-phone | cts
```

# Quality of Service

## DiffServ Configuration on Cisco Endpoints

UCM Provides DSCP values via the service parameters to endpoints for media marking:

Clusterwide Parameters (System - QoS)		
<a href="#">Priority Class</a> *	Normal Priority	Normal Priority
<a href="#">DSCP for Audio Calls</a> *	46 (101110)	46 (101110)
<a href="#">DSCP for Video Calls</a> *	34 (100010)	34 (100010)
<a href="#">DSCP for Audio Portion of Video Calls</a> *	34 (100010)	34 (100010)
<a href="#">DSCP for TelePresence Calls</a> *	32 (100000)	32 (100000)
<a href="#">DSCP for Audio Portion of TelePresence Calls</a> *	32 (100000)	32 (100000)



TelePresence

**DSCP for TelePresence Calls**  
**DSCP for Audio Portion of TelePresence Calls**



UC Video

**DSCP for Audio Calls**  
**DSCP for Video Calls**  
**DSCP for Audio Portion of Video Calls**

# TelePresence Endpoints in Unified CM

**Add a New Phone**

Next

**Status**

Status: Ready

**Select the type of phone you would like to create**

Phone Type\* -- Not Selected --

Next

\*.- indicat  
\*\*.- Crea

- Cisco TelePresence
- Cisco TelePresence 1000
- Cisco TelePresence 1100
- Cisco TelePresence 1300-47
- Cisco TelePresence 1300-65
- Cisco TelePresence 1310-65
- Cisco TelePresence 3000
- Cisco TelePresence 3200
- Cisco TelePresence 500-32
- Cisco TelePresence 500-37
- Cisco TelePresence Codec C40
- Cisco TelePresence Codec C60
- Cisco TelePresence Codec C90
- Cisco TelePresence EX60
- Cisco TelePresence EX90
- Cisco TelePresence MX200
- Cisco TelePresence MX300
- Cisco TelePresence Profile 42 (C40)
- Cisco TelePresence Quick Set C20
- Cisco TelePresence SX20

- TelePresence endpoints are identified as immersive video endpoints fixed setting (Not Configurable)
- Check Devices for Capability:
- Cisco Unified Reporting Tool > “Immersive Video Support for TelePresence Devices”

- Cisco TelePresence TX9000
- Cisco TelePresence TX9200
- Cisco Telepresence Profile 42 (C20)
- Cisco Telepresence Profile 42 (C60)
- Cisco Telepresence Profile 52 (C40)
- Cisco Telepresence Profile 52 (C60)
- Cisco Telepresence Profile 52 Dual (C60)
- Cisco Telepresence Profile 65 (C60)
- Cisco Telepresence Profile 65 Dual (C90)
- Cisco Unified Client Services Framework
- Cisco Unified Communications for RTX
- Cisco Unified Mobile Communicator
- Cisco Unified Personal Communicator
- Cisco Virtualization Experience Client (VXC 6215)
- Generic Desktop Video Endpoint
- Generic Multiple Screen Room System
- Generic Single Screen Room System



# Unified CM QoS Classification

**\* New in 10.0**

Unified CM System QoS Values and CAC Pool Associations

Service Parameter Name	Media Stream Type	DSCP Value	PHB Value	CAC Pool
DSCP for Audio Calls	Audio Only	46	EF	Voice
*DSCP for Audio Portion of Video Calls	Audio of Video	34	AF41	Video
DSCP for Video Calls	Video of Video	34	AF41	Video
*DSCP for Audio Portion of TelePresence Calls	Audio of TP	32	CS4	Immersive
DSCP for TelePresence Calls	Video of TP	32	CS4	Immersive

# Trusted Devices

## Summary



What we've covered:

- **Conditional trust:** Endpoints using CDP
- **Full trust:** Collab Servers, MCUs, Fully Trusted Desktop/Devices
- **Untrusted:** Desktops, Laptops, Mobile Handhelds

What's next

- **Untrusted:** Jabber Devices – Traffic identification and remarking

See Addendum for Server  
QoS Configurations

For more information online:

Search in [www.cisco.com](http://www.cisco.com) > Medianet Campus QoS Design Guide

[http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN\\_and\\_MAN/QoS\\_SRND\\_40/QoS\\_Campus\\_40.html](http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND_40/QoS_Campus_40.html)

Search in [www.cisco.com](http://www.cisco.com) > Medianet WAN Aggregation QoS Design Guide

[http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN\\_and\\_MAN/QoS\\_SRND\\_40/QoS\\_SWAN\\_40.html](http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND_40/QoS_SWAN_40.html)

Breakout Sessions for more information:

**BRKCRT-2501 Campus QoS Design—Simplified: Friday, Jan 31, 9:00 AM - 11:00 AM**

# Identification and Classification Agenda

- Trusted Devices
- **Untrusted Devices:**
  - **Mapping UDP/TCP Port Ranges**
  - Medianet Metadata



# Mapping Identifiable Media and Signalling Streams

1. Identifying media and signalling streams from the client based on Layer 4 (Transport) port ranges (Protocol TCP/UDP and Ports).
  - The protocol port ranges are configured in Unified CM and are passed to the endpoint during device registration. The endpoint then uses these port ranges for signalling and media negotiation.
2. Classify the signalling and media streams and remark with a corresponding DSCP
  - Network Access Control Lists (ACL): Method consists of configuring ACLs to classify and mark DSCP based on protocol and port ranges
  - Windows Global Policy Objects (GPO): Method consists of configuring GPO's to classify and mark DSCP based on protocol, port ranges and application and relies on DSCP trust to pass through the network

# Mobile Portfolio: Jabber Clients



iOS



## Win, Mac



## Tablet



## Smartphone



## Web SDK



# QoS Classification and Marking in Jabber Products

## ■ **Classification in Windows 7 and 8**

- Global Policy Objects (GPOs) which specifies Protocol, Port and Application as means of identification of traffic by which to mark QoS

## ■ **Classification in Windows 2000 and XP**

- Windows 2000 and XP have a different model for allowing the application to mark QoS, which is called Generic Quality of Service (GQoS). Jabber for Windows has implemented GQoS allowing the application to inform the OS to mark the desired DSCP. Turning this function on in Windows 2000 and XP is explained in the following Microsoft Knowledge Base article: <http://support.microsoft.com/kb/248611>

## ■ **Classification in Mac OSX**

- Natively marks DSCP

## ■ **Classification in iOS (iPhone and iPad)**

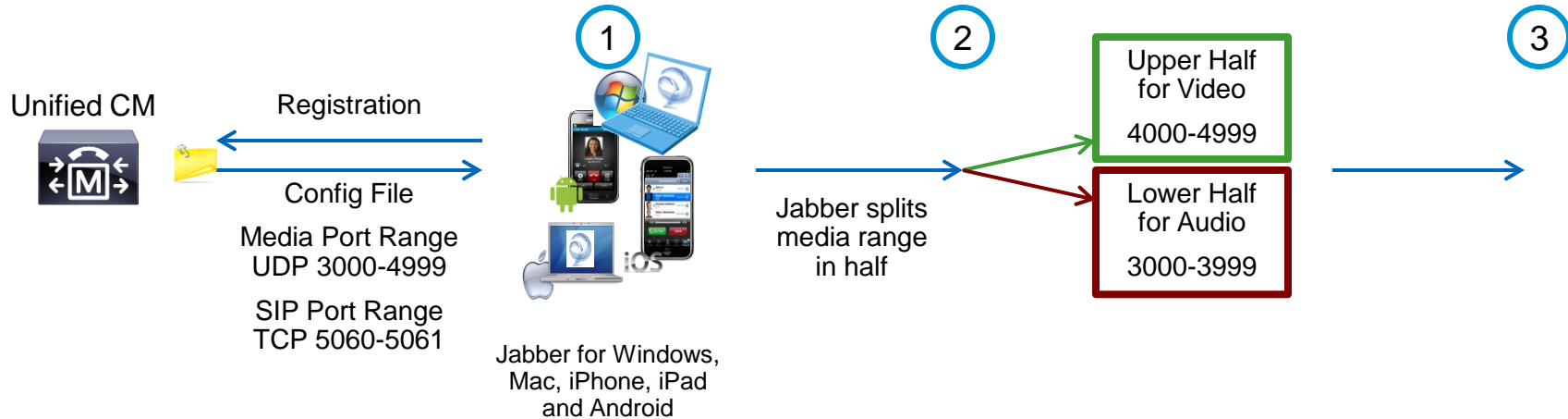
- Natively marks DSCP

## ■ **Classification in Android**

- Natively marks DSCP

Latest Jabber Releases

# Jabber's Use of UDP Port Ranges



1. Client registration, download configuration file
2. Split media port range in half, upper half for video and lower half for audio

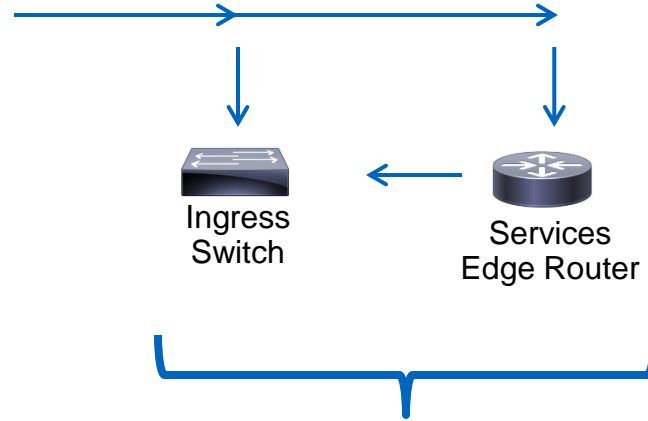
# Jabber's Use of UDP Port Ranges

3 →

## 3. Configure Network with ACL's:

Media is identified on UDP port ranges 3xxx and 4xxx and classified and remarked to EF and AF41 respectively.

SIP Signalling is identified on TCP Port range 5060-5061 to and classified and remarked to CS3.



## Example QoS Policy for Jabber Clients:

- UDP Port Range 3xxx Mark DSCP EF
- UDP Port Range 4xxx Mark DSCP AF41
- TCP Port 5060-5061 Mark DSCP CS3



# Configuring Jabber Client

## Managing Media Ports: SIP Profile

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name\* **Jabber for Windows SIP Profile**

Description SIP Jabber Profile

Default MTP Telephony Event Payload Type\* 101

Early Offer for G.Clear Calls\* Disabled

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\* TIAS and AS

User-Agent and Server header information\* Send Unified CM Version Information as

Timer T2 (msec)	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	3000
Stop Media Port*	4999
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off

**Device Information**

Registration Registered with Cisco Unified

IP Address 10.10.10.79

Active Load ID image\_a

Download Status Unknown

Device is Active

Device is trusted

Device Name\* CSFPCGLEN

Description Jabber for Windows

Device Pool\* SJC [View Details](#)

Common Device Configuration < None > [View Details](#)

Phone Button Template\* Standard Client Services Framework

Common Phone Profile\* Standard Common Phone Profile

Calling Search Space PDX

**Protocol Specific Information**

Packet Capture Mode\* None

Packet Capture Duration 0

BLF Presence Group\* Standard Presence group

SIP Dial Rules < None >

MTP Preferred Originating Codec\* 711ulaw

Device Security Profile\* Cisco Unified Client Services Framework - Standard SI

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Jabber for Windows SIP Profile

Digest User < None >

Media Termination Point Required

Unattended Port

Require DTMF Reception

# Configuring Jabber Client

## Managing Media Ports: SIP Profile / Voice Only

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name\* Jabber Voice Only SIP Profile

Description SIP Jabber Profile

Default MTP Telephony Event Payload Type\* 101

Early Offer for G.Clear Calls\* Disabled

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\* TIAS and AS

User-Agent and Server header information\* Send Unified CM Version Information

Timer Register Expires (seconds)\* 3600

Timer T1 (msec)\* 500

Timer T2 (msec)\* 4000

Retry INVITE\* 6

Retry Non-INVITE\* 10

Start Media Port\* 3000

Stop Media Port\* 3999

Call Pickup URI\* x-cisco-serviceuri-pickup

Call Pickup Group Other URI\* x-cisco-serviceuri-opickup

Call Pickup Group URI\* x-cisco-serviceuri-gpickup

Meet Me Service URI\* x-cisco-serviceuri-meetme

User Info\* None

DTMF DB Level\* Nominal

**Protocol Specific Information**

Packet Capture Mode\* None

Packet Capture Duration 0

BLF Presence Group\* Standard Presence group

MTP Preferred Originating Codec\* 711ulaw

Device Security Profile\* Cisco Dual Mode for iPhone - Standard SIP Non-Secur

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Jabber Voice Only SIP Profile [View Details](#)

Digest User < None >

**Device Information**

Registration Unknown

IP Address Unknown

Device is Active

Device is trusted

Device Name\* TCTGLEN

Description iPhone

Device Pool\* PDX [View Details](#)

**Product Specific Configuration Layout**

Allow End User Configuration Editing Disabled

iPhone Country Code

Cisco Usage and Error Tracking Enabled

Disallow Shared Preset Wi-fi Networks

Default Ringtone Normal

Video Capabilities Disabled

# Configuring Jabber for Windows Client

## Managing Signalling Port: IP Security Profile

**Phone Type**

**Product Type:** Cisco Dual Mode for iPhone  
**Device Protocol:** SIP

---

**Real-time Device Status**

**Registration:** Unknown  
**IPv4 Address:** None

---

**Device Information**

Device is Active  
 Device is trusted

Device Name\*   
Description   
Device Pool\*  [View Details](#)  
Common Device Configuration  [View Details](#)  
Phone Button Template\*

**Protocol Specific Information**

Packet Capture Mode\*   
Packet Capture Duration   
BLF Presence Group\*   
MTP Preferred Originating Codec\*   
**Device Security Profile\***   
Rerouting Calling Search Space   
SUBSCRIBE Calling Search Space   
SIP Profile\*  [View Details](#)  
Digest User

**Phone Security Profile Information**

**Product Type:** Cisco Dual Mode for iPhone  
**Device Protocol:** SIP

Name\*   
Description   
Nonce Validity Time\*   
Device Security Mode   
Transport Type\*

Enable Digest Authentication  
 Exclude Digest Credentials in Configuration File

---

**Phone Security Profile CAPF Information**

Authentication Mode\*   
Key Size (Bits)\*   
Note: These fields are related to the CAPF Information settings on the Phone Configuration page.

---

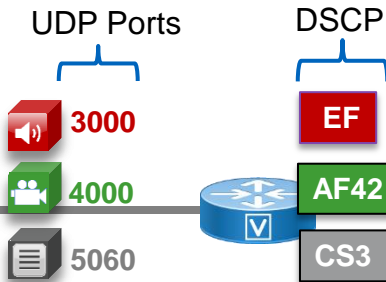
**Parameters used in Phone**

SIP Phone Port\*

# Jabber Client Summary / Best Practices

- Use the SIP Profile to configure media port range (default 16384-32766)
- Use the Sip Security Profile to configure the signalling port range (default 5060 or 5061 for secure signalling)
- If you have SCCP devices in the same network use a port range outside of 16384-32766 to avoid overlap and incorrect remarking
  - **Unified CM 9.1 expands SIP media port range to 2048-65535**
- Video Enablement:
  - Disable video if you do not want device to send or receive video
  - Video capable devices ALWAYS divide the port ranges (even if video is disabled)
  - Devices that do NOT support video (version dependent) use the entire port range for audio-only.

# Ingress Classification



**! This section applies the policy-map to the Interface Router**  
`Router (config-if)# service-policy input INGRESS-MARKING`  
**! Attaches service policy to interface**

**! This section configures the ACL's**

```
access-list 100 permit udp any any range 3000 3999
access-list 101 permit udp any any range 4000 4999
access-list 102 permit tcp any any range 5060 5061
```

**! This section configures the classes**

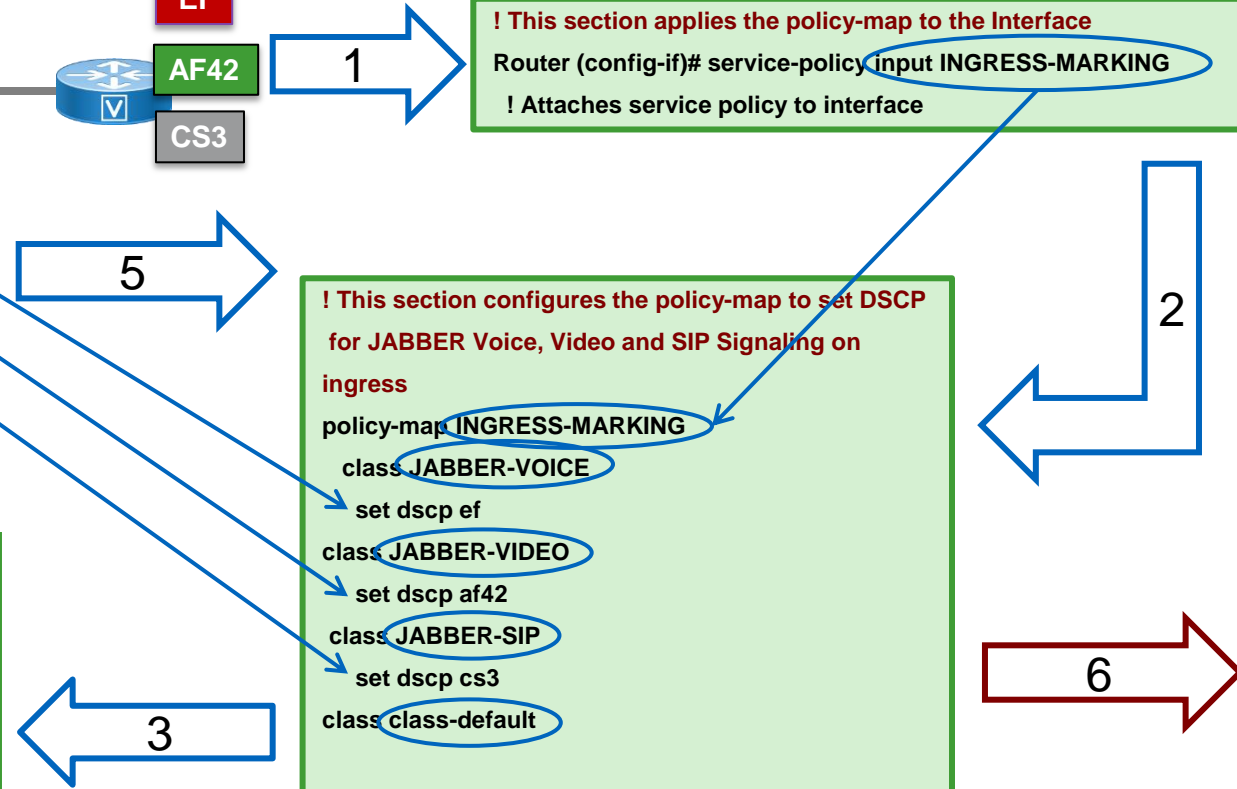
```
class-map match-all JABBER-VOICE
match access-group 100

class-map match-all JABBER-VIDEO
match access-group 101

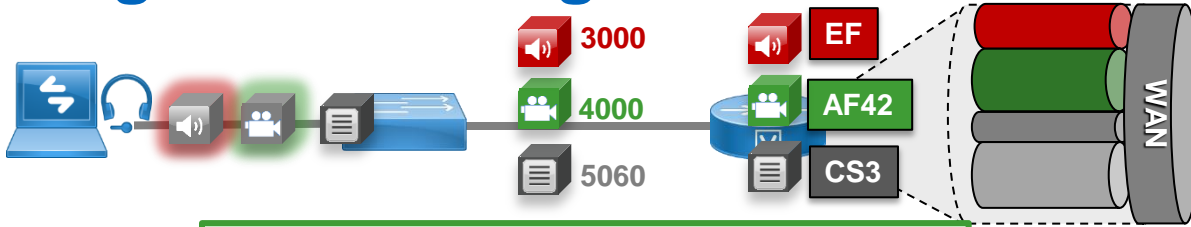
class-map match-all JABBER-SIP
match access-group 102
```

**! This section configures the policy-map to set DSCP for JABBER Voice, Video and SIP Signaling on ingress**

```
policy-map INGRESS-MARKING
class JABBER-VOICE
set dscp ef
class JABBER-VIDEO
set dscp af42
class JABBER-SIP
set dscp cs3
class class-default
```



# Egress Queuing



6

**! This section applies the policy-map to the Interface**  
Router (config-if)# service-policy output EGRESS-QUEUEING  
**! Attaches service policy to interface**

**! This section configures the bandwidth for all collab traffic**

policy-map EGRESS-QUEUEING

class VOICE

priority percent 10

**! Provisions 10% LLQ to VOICE class**

class VIDEO

bandwidth percent 30

**! Provisions 30% CBWFQ to VIDEO class**

class SIGNALING

bandwidth percent 2

**! Provisions 2% CBWFQ to SIGNALING class**

...

7

**! This section applies the policy-map**

class-map match-all VOICE

match dscp ef

class-map match-any VIDEO

match dscp af41

match dscp af42

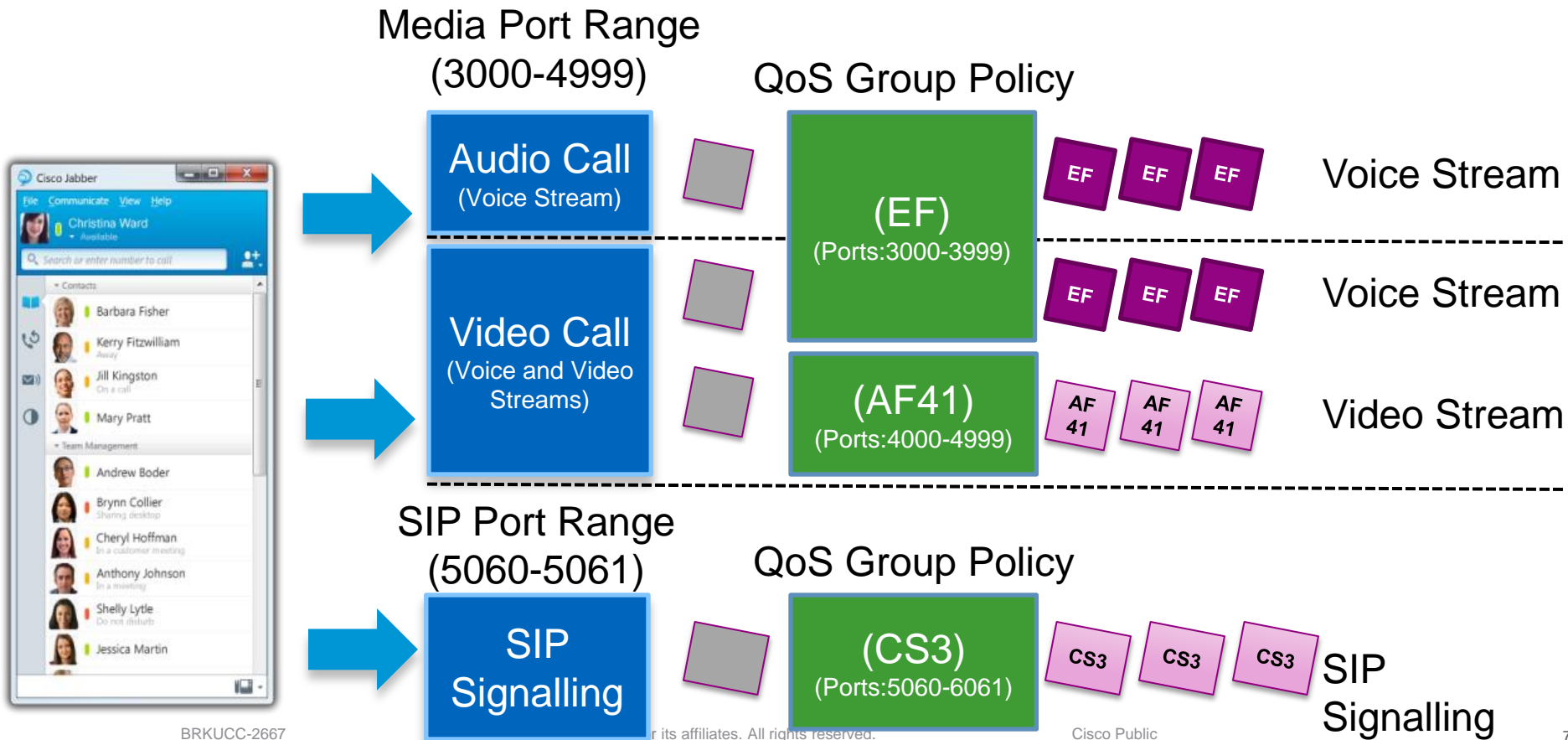
class-map match-all SIGNALING

match dscp cs3

8

# Jabber For Windows “To Trust or Not to Trust”

## Group Policy Objects



# Summary / Best Practice Recommendations

## Trusted / Native Marking

- Windows 7 and 8 require GPOs set QoS
  - GPOs use TCP and UDP port ranges to set QoS
  - Cannot differentiate audio-only from audio of a video call
- Previous Windows versions allow the application to set QoS natively
- All other Jabber clients (latest versions) mark EF for audio-only, AF41 for audio and video of a video call.

## Mapping Identifiable Media and Signalling Streams (Network ACLs)

- QoS Strategies
  - Mark audio EF and Video AF41
  - Mark audio AF41 and Video AF42
  - Mark audio and video AF42
- Recommended Remarking Policy
  - Remark using UDP/TCP Port ranges OR Use MSI and Medianet Metadata (next section)

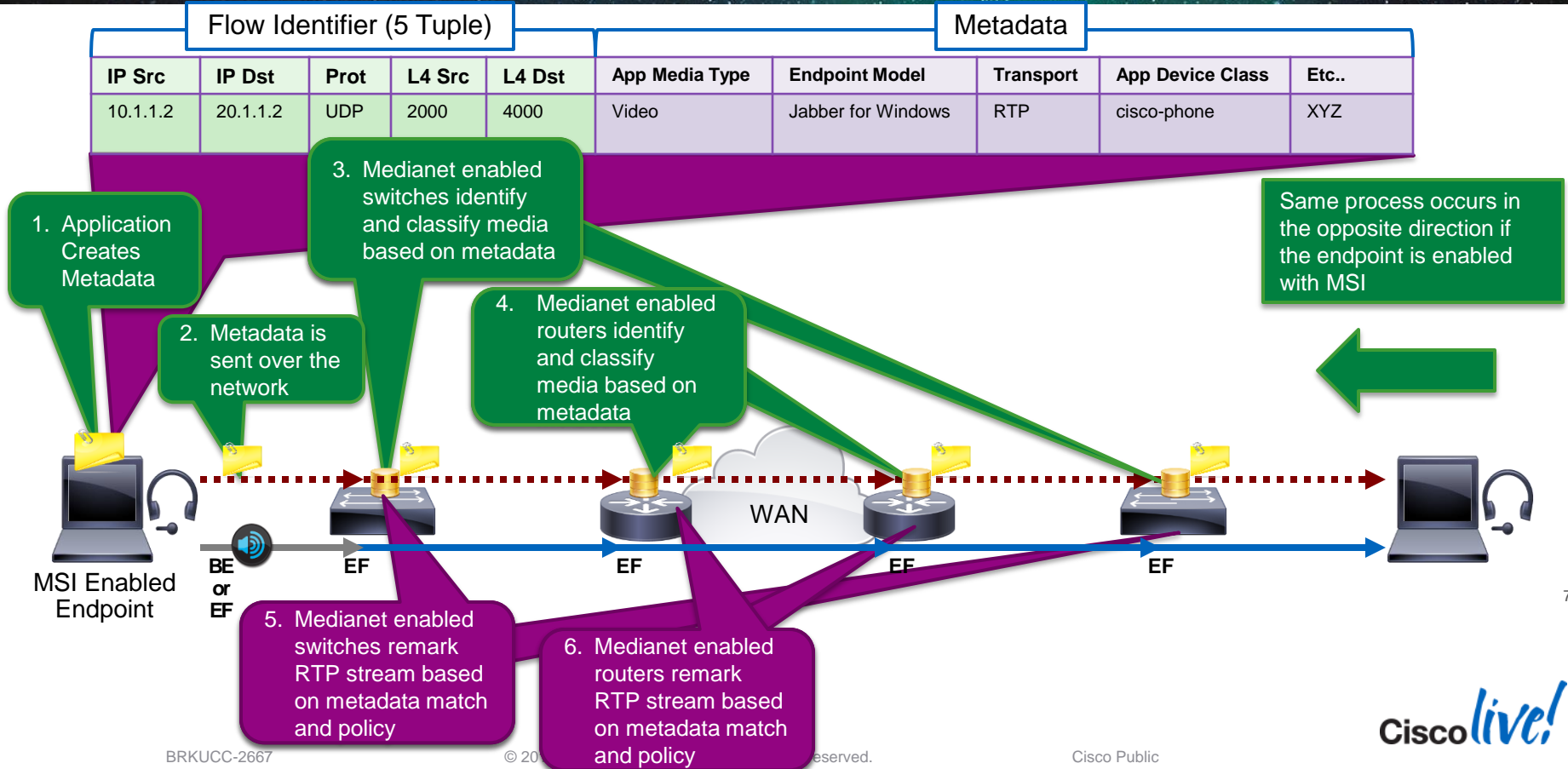


# Classification and Identification Agenda

- Trusted Devices
- **Untrusted Devices:**
  - Mapping UDP/TCP Port Ranges
  - **Medianet Metadata**



# Medianet Flow Metadata



# Medianet Flow Metadata

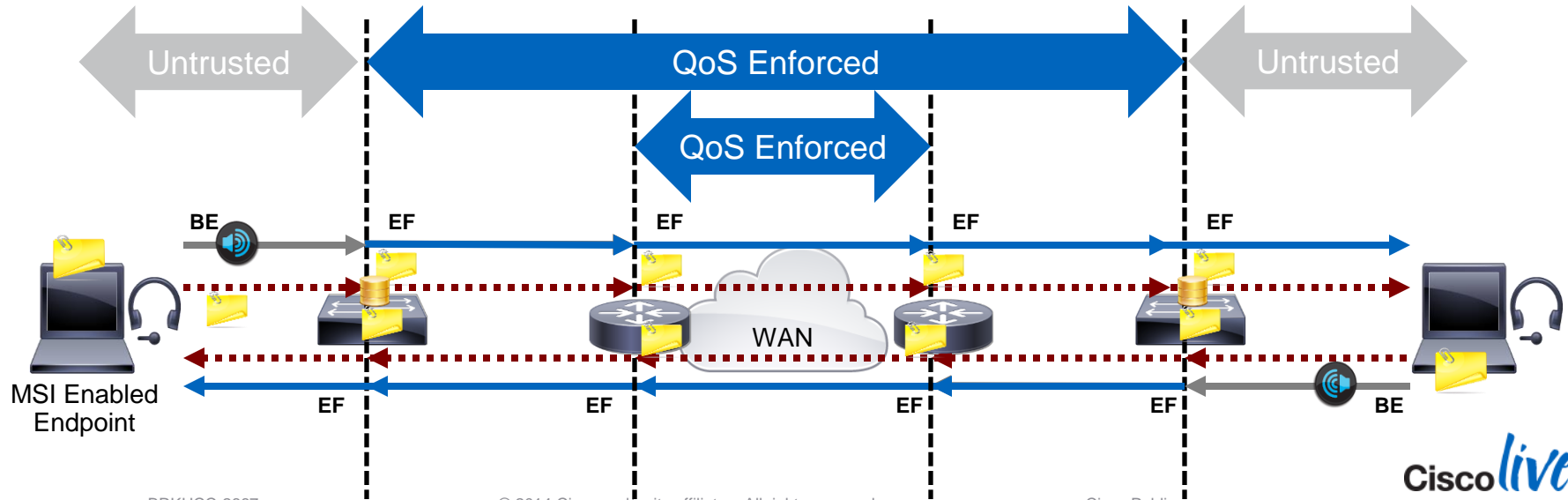
## What's the Benefit?

```
(config)#metadata flow
```

```
(config)#interface g0/1  
(config-interface)#metadata flow
```

- Enable metadata flow globally, or per interface
- RSVP snooping** required on L2 switch

- Mark on switches and trust on routers enforcing QoS end-to-end across the network
- Mark on routers to enforce QoS on the WAN (good starting point).



# Network Devices Supporting Medianet Metadata

Platform	IOS / Switch Image	Package	Medianet Feature
Cisco Catalyst 4500E Supervisor Engine 7-E and 7L-E, Cisco Catalyst 4500E Supervisor Engine 6-E, Cisco Catalyst 4500 Supervisor Engine 6L-E, Cisco Catalyst 4500X Series	XE 3.3.0SG or later 15.1.(1)SG or later	IP Base or higher	Media Awareness: <ul style="list-style-type: none"> <li>• Flow Metadata</li> <li>• Media Services Proxy</li> </ul>
Cisco Catalyst 4900M, Catalyst 4948E, and Catalyst 4948E-F Switches	15.1(1)SG or later	IP Base or higher	Media Awareness: <ul style="list-style-type: none"> <li>• Flow Metadata</li> <li>• Media Services Proxy</li> </ul>
Cisco Catalyst 6500-E Series Switches Supervisor Engine 2T Cisco Catalyst 6500-E Series Switches Supervisor Engine 720	15.0(1)SY or later 15.1(2)SY or later	IP Base or higher	Media Awareness: <ul style="list-style-type: none"> <li>• Flow Metadata</li> <li>• Media Services Proxy</li> </ul>
Cisco 880*, 990*, 1900, 2900 and 3900 Series Integrated Services Routers	15.2(1) T	Data	Media Awareness: <ul style="list-style-type: none"> <li>• Flow Metadata</li> </ul>
	15.2(3) T, *15.2(4) M2	Data	Media Awareness: <ul style="list-style-type: none"> <li>• Media Services Proxy</li> </ul>
	15.4(1)T 15.4(1)S	Data	Reverse Flow Metadata Support
Cisco ASR 1000 Series Aggregation Services Routers	Cisco IOS XE 3.7 or later	Advanced Enterprise	Media Awareness: <ul style="list-style-type: none"> <li>• Flow Metadata</li> </ul>



# Collaboration Endpoints Supporting Medianet Metadata

Collaboration Endpoints	Version
Cisco Jabber for Windows	UC 9.0(1) or later
Cisco Jabber for Mac	9.2.1 or later
Cisco Jabber for iOS and Android	Planned*
Cisco TelePresence (EX, C, MX, SX, TX, CTS 500-32, TX1300 and TX9000 ). MSI Included in software install	TC 6.0 or later
WebEx	WebEx (WBS28)

Current defect in TX6.3 and TC7.0! To be fixed in 7.0.2 and 6.4?

\* MSI for iOS and Android support planned. Come talk to me if this is important to you!



# QoS: Metadata Based Classifications

## Application Identifier

- match application telepresence-media
- match application cisco-phone

## Dynamic Attribute

- match application attribute media-type audio
  - match application attribute media-type video
  - match application attribute media-type audio-video
- } Audio-Only Call
- } Video Call

## Application Group

- match application application-group webex-group
- match application application-group telepresence-group

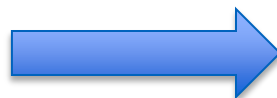
## Category

- match application attribute category voice-video
- match application attribute category business-productivity-tools

# Class-map Migration for Existing Egress QoS Policy

Class-map match-all Jabber

Match access-group 110



Class-map match-**any** Jabber

Match application **cisco-phone**

Match access-group 110

Match ....



Class-map: Jabber (match-any)

100198 packets, 10599254 bytes

30 second offered rate 75000 bps

**Match: application cisco-phone**

99535 packets, 10512016 bytes

30 second rate 75000 bps

**Match: access-group 110**

663 packets, 87238 bytes

30 second rate 0 bps

# Class-map Migration for Existing Ingress QoS Policy

```
class-map match-all VOICE
  match access-group 100
class-map match-any VIDEO
  match access-group 101
class-map match-all SIGNALING
  match access-group 102
```



```
class-map match-any VOICE
  match application attribute media-type audio
  match access-group 100
class-map match-any VIDEO
  match application attribute media-type audio-video
  match application attribute media-type video
  match access-group 101
class-map match-any SIGNALING
  match application attribute sub-category control-and-signaling
  match access-group 102
```

```
policy-map INGRESS-MARKING
  class VOICE
    set dscp ef
  class VIDEO
    set dscp af41
  class SIGNALING
    set dscp cs3
  class class-default
```

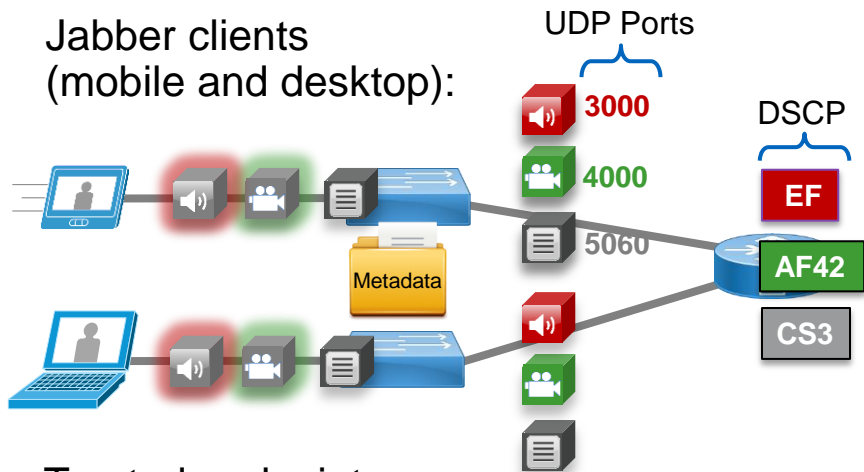
Note: No Device Type Differentiation.  
This policy applies to ALL metadata capable endpoints that match these attributes.



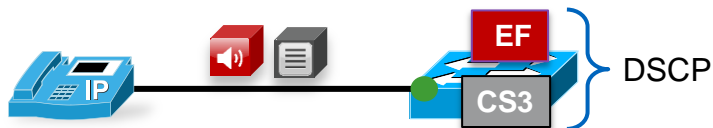
# Identification and Classification

## Bringing it all Together

Jabber clients  
(mobile and desktop):



Trusted endpoints:



- Jabber classification based on UDP port ranges and ACL's (mobile clients) and/or metadata (desktop clients):
  - Audio of **all** Jabber calls (voice-only and video) is marked EF
  - Video of Jabber calls is marked AF42
- Video endpoint and IP phone classification based on conditional trust and CDP:
  - Audio and video streams of video calls are marked AF41
  - Voice-only calls are marked EF

# Ingress Classification

## Port Ranges and Metadata

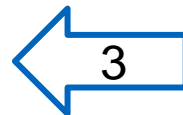
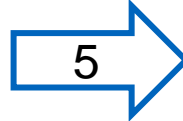
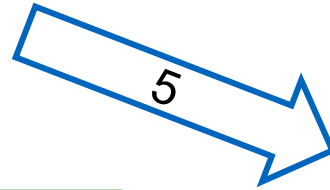
**! This section configures the ACL's**

```
access-list 100 permit udp any any range 3000 3999
access-list 101 permit udp any any range 4000 4999
access-list 102 permit tcp any any range 5060 5061
```



**! This section configures the classes**

```
class-map match-any JABBER-VOICE
  match application attribute media-type audio
  match access-group 100
class-map match-any JABBER-VIDEO
  match application attribute media-type audio-video
  match application attribute media-type video
  match access-group 101
class-map match-any JABBER-SIP
  match application attribute sub-category control-and-signaling
  match access-group 102
```

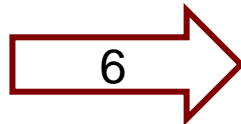
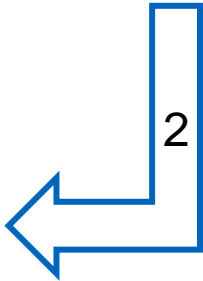


**! This section applies the policy-map to the Interface**

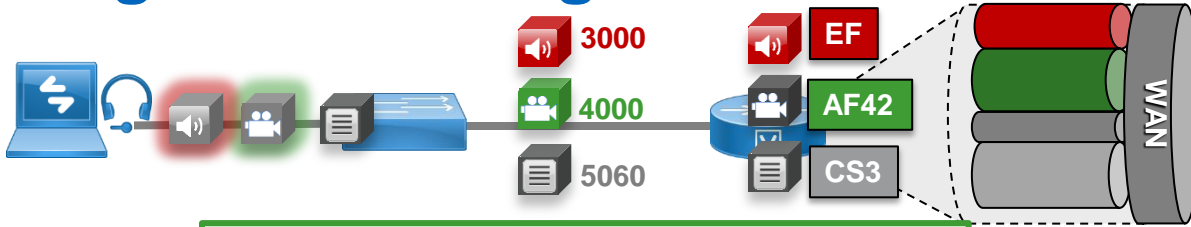
```
Router (config-if)# service-policy input INGRESS-MARKING
! Attaches service policy to interface
```

**! This section configures the policy-map to set DSCP for JABBER Voice, Video and SIP Signaling on ingress**

```
policy-map INGRESS-MARKING
  class JABBER-VOICE
    set dscp ef
  class JABBER-VIDEO
    set dscp af41
  class JABBER-SIP
    set dscp cs3
  class class-default
```



# Egress Queuing



6

**! This section applies the policy-map to the Interface**  
Router (config-if)# service-policy output WAN-EDGE-QUEUING  
**! Attaches service policy to interface**

**! This section configures the bandwidth for all collab traffic**  
policy-map WAN-EDGE-QUEUING  
class VOICE  
priority percent 10  
**! Provisions 10% LLQ to VOICE class**  
class VIDEO  
bandwidth percent 30  
**! Provisions 30% CBWFQ to VIDEO class**  
class SIGNALING  
bandwidth percent 2  
**! Provisions 2% CBWFQ to SIGNALING class**  
...

7

**! This section applies the policy-map**  
class-map match-all VOICE  
match dscp ef  
class-map match-any VIDEO  
match dscp af41  
match dscp af42  
class-map match-all SIGNALING  
match dscp cs3

8

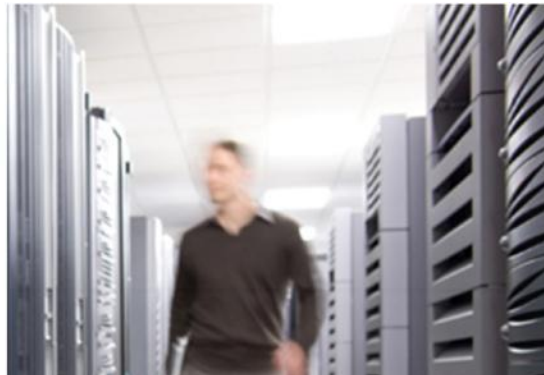
Queuing on EGRESS  
All marked traffic:  
1. Trusted  
2. Metadata  
3. Port Ranges

# Medianet Metadata

## Benefits and Best Practices Recommendations

### Benefits

- More secure!
  - Don't have to trust the whole OS, just the application!
  - Less worries about rogue endpoints usurping TCP or UDP port range to hijack your QoS.
- Easy and safe migration from ACL port range mechanism!
  - Simply add the metadata match criteria into your already configured class-maps
  - Ensure they are the first criteria



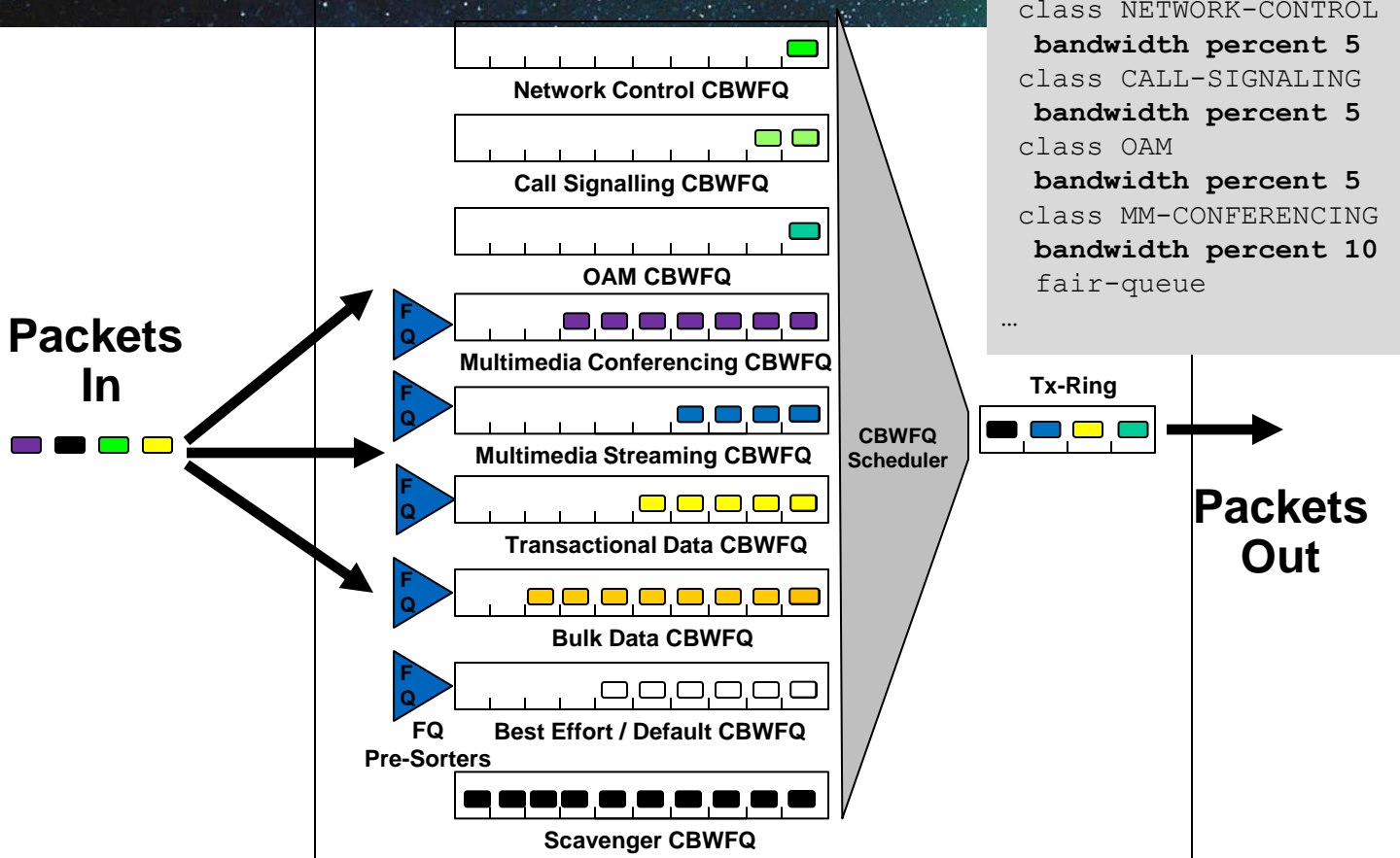
## QoS Architecture:

APPROACH OVERVIEW  
IDENTIFICATION & CLASSIFICATION  
**QUEUING & SCHEDULING**

# IOS QoS Mechanisms and Operation

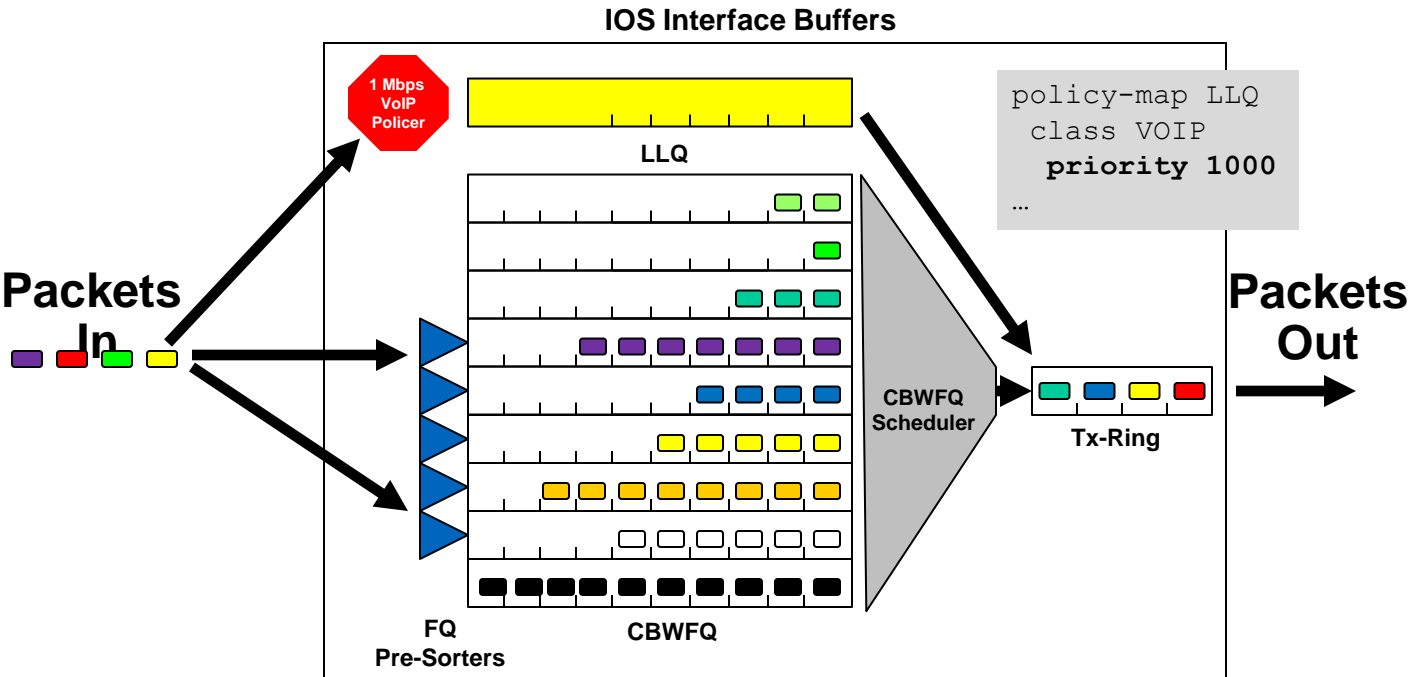
## CBWFQ Operation

### IOS Interface Buffers



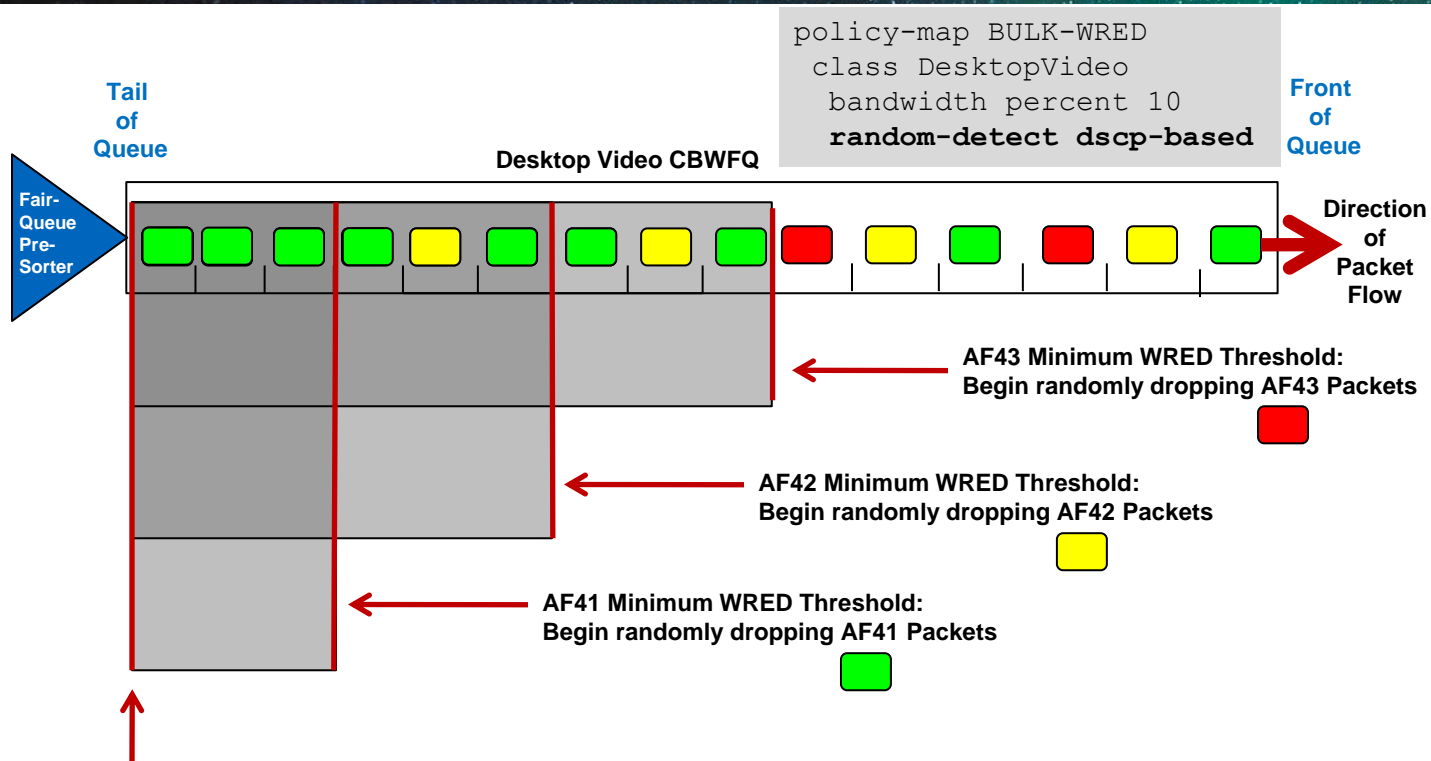
# IOS QoS Mechanisms and Operation

(Single) LLQ Operation



# IOS QoS Mechanisms and Operation

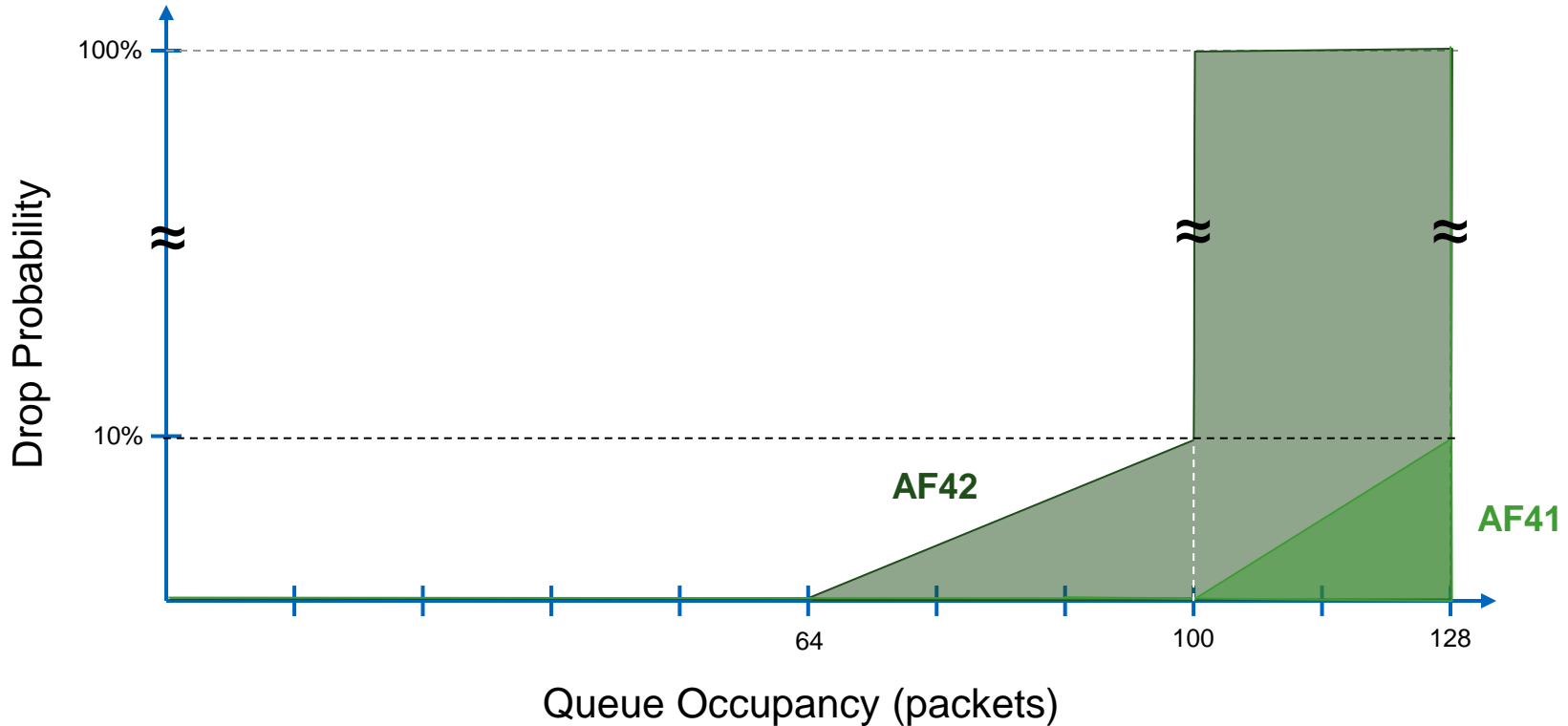
## DSCP-Based WRED Operation



Maximum WRED Thresholds for AF41, AF42 and AF43 are set to the tail of the queue in this example

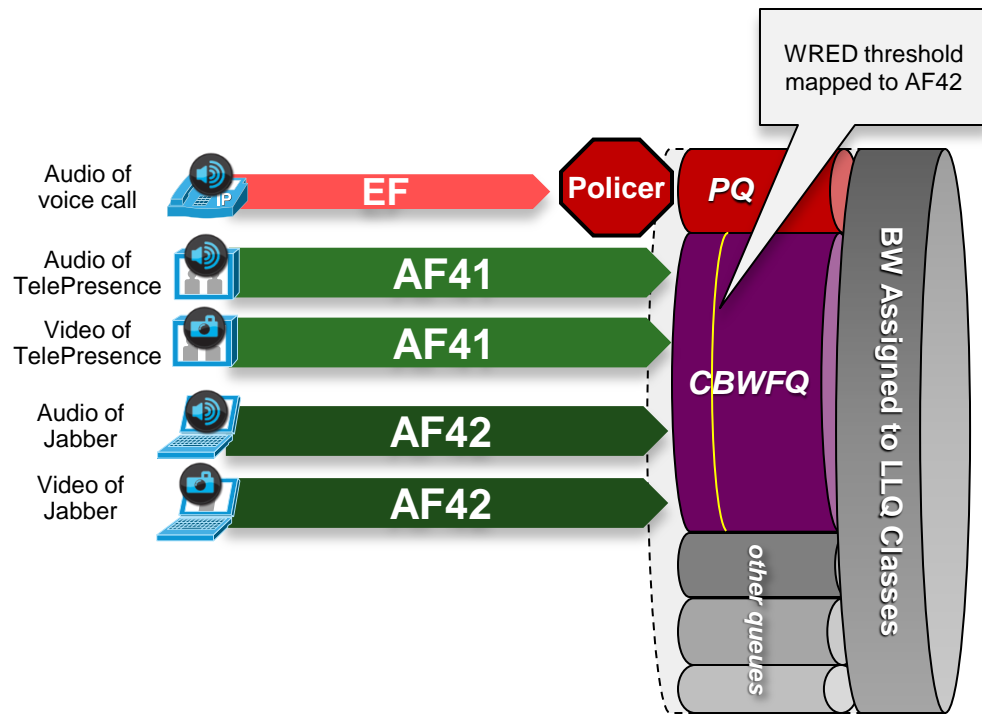


# WRED Thresholds



# Queuing and Scheduling

## Summary / Example



- Map all audio (EF) traffic to PQ
- Map all video (AF41 and AF42) to a class-based queue, in which:
  - AF41 is tail dropped
  - AF42 is mapped to a WRED threshold



Q & A

# Complete Your Online Session Evaluation

## Give us your feedback and receive a Cisco Live 2014 Polo Shirt!

Complete your Overall Event Survey and 5 Session Evaluations.

- Directly from your mobile device on the Cisco Live Mobile App
- By visiting the Cisco Live Mobile Site [www.ciscoliveaustralia.com/mobile](http://www.ciscoliveaustralia.com/mobile)
- Visit any Cisco Live Internet Station located throughout the venue

Polo Shirts can be collected in the World of Solutions on Friday 21 March 12:00pm - 2:00pm



## Learn online with Cisco Live!

Visit us online after the conference for full access to session videos and presentations.

[www.CiscoLiveAPAC.com](http://www.CiscoLiveAPAC.com)



**CISCO** <sup>TM</sup>