PureConnect Quality of Service

Technical Reference

Abstract

This document presents information regarding tested configurations that are proven to provide Quality of Service (QoS) with the PureConnect line of products. When QoS is applied correctly, it can resolve problems with audio and delays in Voice-over-Internet-Protocol (VoIP) communications.

For the latest version of this document, see the PureConnect Documentation Library at: http://help.genesys.com/cic.

For copyright and trademark information, see https://help.genesys.com/cic/desktop/copyright_and_trademark_information.htm.
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Introduction

The purpose of this document is to provide tested configurations proven to provide Quality of Service with the PureConnect line of products. When you apply QoS correctly, it can resolve problems with “choppy” audio, voice quality, and delays that make interaction difficult. This document assumes that the reader is familiar with Basic QoS principles and network characteristics such as bandwidth, delay, jitter, and packet loss.

Genesys provides the examples on the following pages as a courtesy so don’t take them as exact representations of any customer environment. It’s possible that the configurations presented aren’t the best solution for larger, more complex networks. The intent of this document is to demonstrate some best practices and to provide a resource for network engineers before implementation.
# PureConnect Quality of Service driver

## Supported products for the PureConnect QoS driver

The PureConnect product suite installs and uses its own QoS driver for its servers and client applications. This driver operates independently of the Microsoft QoS Packet Scheduler, which was the recommended QoS driver in CIC version 3.0. The PureConnect QoS driver is available in 64-bit and 32-bit versions and installs based on the supporting operating system.

The following product installation packages install the PureConnect QoS driver:

<table>
<thead>
<tr>
<th>Product</th>
<th>.ISO image directory</th>
<th>Installation file</th>
<th>Supported operating system</th>
</tr>
</thead>
<tbody>
<tr>
<td>Interaction Media Server</td>
<td>\Off-ServerComponents</td>
<td>MediaServer_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>Interaction Recorder Extreme Query</td>
<td>\Off-ServerComponents</td>
<td>IRExtremeQueryServer_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>Interaction Recorder Remote Content Service</td>
<td>\Off-ServerComponents</td>
<td>IRRemoteContentService_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>IC Session Manager (Off-host)</td>
<td>\Off-ServerComponents</td>
<td>SessionManager_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>IC Status Aggregator</td>
<td>\Off-ServerComponents</td>
<td>StatusAggregator_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>Loquendo Automatic Speech Recognition</td>
<td>\Off-ServerComponents</td>
<td>LoquendoASR_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
<tr>
<td>Nuance Recognizer</td>
<td>\Off-ServerComponents</td>
<td>ASRServerNuanceRecognizer_20nn_Rn.msi</td>
<td>64-bit</td>
</tr>
</tbody>
</table>
| CIC User Applications (32-bit)               | \ServerComponents      | ICUserApps_32bit_20nn_Rn.msi with at least one of the following applications selected:  
  - SIP Soft Phone  
  - Interaction Screen Recorder Capture Client | 32-bit                     |
| CIC User Applications (64-bit)               | \ServerComponents      | ICUserApps_64bit_20nn_Rn.msi with at least one of the following applications selected:  
  - SIP Soft Phone  
  - Interaction Screen Recorder Capture Client | 64-bit                     |

**Note:** It’s possible for future releases of PureConnect products to include the QoS driver in their installations.
PureConnect QoS driver installations

By default, the products that use the PureConnect QoS driver install that driver silently. The installation programs also install the required driver certificate to the Trusted Publishers list. The only exceptions are the CIC user application installation packages.

Following are situations where you do not want to install the PureConnect QoS driver:
- You use another form of QoS in your network environment.
- You want to use the PureConnect QoS driver, but do not want to add the certificate to the Trusted Publishers list.

For these situations, do not install the .msi file through Windows Explorer or a command line interface without parameters. PureConnect installation packages support the following alternative methods, which allow you to determine how to install the PureConnect QoS driver:
- Group Policy.
- Startup or logon script.
- Command line interface with parameters.

PureConnect QoS driver installation properties

With properties, you can specify certain aspects of the installation of the PureConnect QoS driver. The following list provides these properties and the associated functionality:
- DONOTINSTALL_ININQOSDRIVER - This property, when set to 1, prohibits installation of the QoS driver and the driver certificate.
- DONOTADDCERTOTRUSTEDPUBLIST - This property, when set to 1, allows installation of the QoS driver but does not install the driver certificate to the Trusted Publishers list. When you set this property, the installation program prompts you to confirm installation of the PureConnect QoS driver.

Note: If you do not select the Custom installation type, the PureConnect QoS driver and driver certificate install silently.
**PureConnect QoS driver (Recommended):** If selected, both the driver certificate and the QoS driver install. If the current permissions on the workstation do not allow modification of the Trusted Publishers list, a message indicating such appears.

**Other:** If selected, the program doesn’t add the driver certificate to the Trusted Publishers list and the doesn’t install the QoS driver. Use this option only when you have another QoS solution in place on the workstation.

**Note:** If you install SIP Soft Phone and click the **Other** option because another form of QoS is present on the workstation already, you must set the **QoS Enabled** option in the **Interaction Administrator Managed IP Phone Configuration** to **No**. Otherwise, the user cannot receive calls.
Note: During installation of the PureConnect QoS driver, including upgrades, the network connections of the personal computer interrupt briefly.

Other options for PureConnect QoS

The following list provides more information for working with the PureConnect QoS driver and driver certificate:

- If you uninstall PureConnect products that included the installation of the QoS driver and the driver certificate, the uninstall process removes the driver certificate from the Trusted Publishers list.
- You can manually add or remove the QoS driver certificate using the `certmgr.msc` utility or through a Group Policy.

Driver_status.exe QoS utility

Each installation of the PureConnect QoS driver includes the `driver_status.exe` file. Run this utility, without switches or parameters, to check the status of the QoS driver.
Use SAPI TTS without the PureConnect QoS driver

If you do not install the PureConnect QoS driver, you can still configure CIC to play SAPI TTS through VoIP calls.

To configure CIC to play SAPI TTS without the QoS driver installed:
1. On the CIC server or from a remote computer connected to the CIC server, start Interaction Administrator.
2. In the expanded list under the name of your CIC server, click the Server Parameters object. The right pane of the Interaction Administrator window displays all current server parameters and their associated values.
3. In the right pane, right-click and then click New from the resulting shortcut menu. The Entry Name dialog box appears.
4. In the Enter Parameter Name box, type SAPI DSCP Value and then click OK. The Parameter Configuration dialog box appears.
5. In the Parameter Value box, type 0x0 and then click OK. The new server parameter and its value appear at the bottom of the table in the right pane of the Interaction Administrator window.

The 0x0 value represents the Differentiated Services Code Point (DSCP) value that inserts in RTP packets of VoIP interactions. You can also set the DSCP value to any value between 0x00 and 0x3F.
RTP port usage

VoIP devices often use different Realtime Transfer Protocol (RTP) ports. PureConnect products use RTP ports depending on the intrinsic architecture used. Most IP telephones and other VoIP devices use varying RTP port ranges. Consult the documentation for your VoIP devices to determine the available RTP port ranges.

For the full list of supported IP telephone devices, gateways, and other VoIP devices, see http://testlab.genesys.com.

For common network port ranges used with RTP, see the CIC Port Maps and Data Flow Diagrams Technical Reference.
Marking

PureConnect products use Differentiated Services (DiffServ), and Class Selector PHB in some cases, to mark packets on an IP Network. RFC 2474 “Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers” describe these standards.

When possible, always mark packets at the source. PureConnect products provide the ability to mark packets as they leave Interaction Media Server, Interaction Gateway, telephones, and other PureConnect products. Configure all switches and routers that handle data and call flows to guarantee that the marked IP packets honor the QoS as they propagate through the network.

The endpoints QoS mark the RTP Packets, which use UDP, and SIP Packets, which through configuration can use UDP, TCP, or TLS. The QoS markings are in the 6-bit DSCP field of the ToS Byte of the IP header. The recommended (Default) value for RTP (voice) is 101110, which is EF / Expedited Forwarding. The recommended (Default) value for SIP Signaling is 011000, which is CS3 / Class Selector.

Source Endpoint Device Recommended Markings for RTP (voice) Recommended Markings for SIP (signaling) Where are the Configurations

| Device Type                  | RTP Markings                                      | SIP Markings                                      | Instructions                                                                 
|------------------------------|---------------------------------------------------|---------------------------------------------------|-------------------------------------------------------------------------------
| SIP in CIC server            | NA                                                | SIP packets marked with 011000 (which is CS3 / Class Selector) |
| IC Notifier                  | Notifier packets marked with 101110 (EF / Expedited Forwarding) | NA                                                | This DSCP marking value is not currently configurable. If you need different values, contact PureConnect Customer Care. |
| Interaction Media Server     | RTP packets marked with 101110 (which is EF / Expedited Forwarding) | NA                                                | Interaction Media Server RTP is marked with EF. Interaction Media Server does not use SIP signaling. For more information about marking on Interaction Media Server, see Interaction Media Server Technical Reference. |
| Interaction Gateway          | RTP packets marked with 101110 (which is EF / Expedited Forwarding) | SIP packets marked with 011000 (which is CS3 / Class Selector) |
| AudioCodes Mediant Gateway   | RTP packets marked with 101110 (which is EF / Expedited Forwarding) | SIP packets marked with 011000 (which is CS3 / Class Selector) |
| Polycom Phones              | RTP packets marked with 101110 (which is EF / Expedited Forwarding) | SIP packets marked with 011000 (which is CS3 / Class Selector) |

To alter this setting in Interaction Administrator, do the following steps:
1. In the left pane, select the Lines object.
2. In the right pane, double-click the Configuration item.
3. In the Line Configuration dialog box, select the SIP Line Configuration tab.
4. In the list box, click the Transport item.
5. Scroll down to the SIP DSCP Value list box and select the value that you want to assign.

Interaction Gateway exposes these values with the entire TOS Byte. The default value is 2E (46, 101110) EF.

The Mediant gateway defaults to DSCP value 101100 and marks packets by default. You can change this setting on the following page under Full Configuration\Network Settings\QoS Settings. Change the Default setting of 44 decimal (101000) for Control Premium QoS to 24 decimal (011000) for SIP signaling to use CS3.

Legacy Polycom telephones that cannot be provisioned by the CIC server must be configured manually with a sip.cfg file. Provisioned Phone markings can be controlled in Interaction Administrator.
To alter this setting in Interaction Administrator, do the following steps:
1. In the left pane, expand the server object.
2. Expand the Stations object.
3. Highlight the Default Station object.
4. In the right pane, double-click the Configuration item.
5. In the Default Station Configuration dialog box, select the Global SIP Station tab.
6. In the list box, select the Transport item.
7. In the SIP DSCP Value list box, select the value that you want to assign to SIP packets.

See CIC Managed IP Phones Administrator’s Guide.

The following image displays an example of setting the SIP QoS in the Line Configuration that is available through Interaction Administrator. Ensure that you are consistent with any changes that you make and consider the SIP lines and stations that you use. You can enter 18 in the SIP Line Configuration SIP DSCP Value field to have it display 011000 (which is CS3) correctly if you wish to use DSCP. No restart of Telephony Services is required. The next call from the server will use the new value.

Genesys highly recommends that you ensure the IP packets are getting marked correctly with the WireShark application, formerly known as Ethereal (http://www.wireshark.org). The following example displays the RTP header from a Polycom phone and the proper marking that is applied:
Differentiated Services Field: 0xb8 (DSCP 0x2e: Expedited Forwarding; ECN: 0x00)

1011 10.. = Differentiated Services Codepoint: Expedited Forwarding (0x2e)
.... ..0. = ECN-Capable Transport (ECT): 0
.... ...0 = ECN-CE: 0
Total Length: 200

It is not possible to mark the SIP Signaling coming from the CIC server. This can be accomplished at the switch level with a service-policy if necessary. Let’s assume that the CIC server is using port fast0/1 in the following example. This example can extend to other application flows that need to be marked. Marking can be CPU intensive for routers and switches. Always ensure that you have properly sized your equipment for Marking.

1. Enable QoS on the Switch
   Start in the privileged exec mode:
   ```
   switch(config)#mls qos
   switch(config)#policy-map mark-signal
   switch(config-pmap)#class voice-signal
   switch(config-pmap-c)#set dscp cs3
   switch(config-pmap-c)#exit
   switch(config-pmap)#exit
   ```

2. Create access lists to identify the SIP signaling traffic as it leaves the CIC server
   Start in the privileged exec mode:
   ```
   switch(config)#access-list 100 permit tcp any any eq 5060
   switch(config)#access-list 100 permit tcp any any eq 8060
   ```

3. Create a Class-Map that uses the Access List
   Start in the privileged exec mode:
   ```
   switch (config)#class-map match-all voice-signal
   switch(config-cmap)#match access-group 100
   switch(config-cmap)#exit
   ```

4. Apply the Class-Map to a Policy
   Start in the privileged exec mode:
   ```
   switch(config)#policy-map mark-signal
   switch(config-pmap)#class voice-signal
   switch(config-pmap-c)#set dscp cs3
   switch(config-pmap-c)#exit
   switch(config-pmap)#exit
   ```

5. Apply the Service-Policy to the CIC server interface
   Start in the privileged exec mode:
   ```
   switch(config)#int fast 0/12
   switch(config-if)#service-policy input mark-signal
   switch(config-if)#exit
   ```
Virtual Local Area Networks (VLANs) and QoS

Basic LAN QoS (Cisco) and VLANs in a single switch network

When deploying IP telephones, Genesys recommends that you place IP telephones in a voice virtual local area network (VLAN). A voice VLAN can protect the telephones from broadcast traffic as well as provide security. You should place all voice endpoints including the Interaction Media Servers and other servers in the CIC network environment in the voice VLAN. CIC clients should remain in a Data VLAN or the Default VLAN. The SIP Soft Phone can also be placed on the Default VLAN. Currently, it requires dual Network Interface Cards (NICs) if splitting the traffic was a requirement. This configuration is only possible on switches that support Layer 3 routing. Layer 3 routing is required to let the different networks communicate with each other. This is all done in the routing table on the switch.

<table>
<thead>
<tr>
<th>VLAN</th>
<th>Purpose</th>
<th>Recommended Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Data</td>
<td>The Default for Clients, Data Traffic</td>
<td>100</td>
</tr>
<tr>
<td>Voice</td>
<td>IP Phones, CIC server, Interaction Media Server, Interaction SIP Proxy, etc.</td>
<td>200</td>
</tr>
</tbody>
</table>

This example configuration is based on the recommended setup and the following network model using a Cisco Catalyst 3560 switch.
1. Create the VLAN for Voice.
   Start in the privileged exec mode:
   ```
   configure terminal
   vlan 200
   name "Voice"
   exit
   ```

2. Configure the VLAN interfaces with IP addresses.
   From the global configuration mode:
   ```
   interface vlan 100
   ip address 10.70.0.1 255.255.0.0
   
   interface vlan 200
   ip address 10.60.0.1 255.255.0.0
   ```

3. Assign the switch ports to the respective VLANs using static port assignment.
   From the global configuration mode:
   ```
   interface range FastEthernet 0/1-12
   switchport access vlan 200
   exit
   ```

4. If using CDP (alternative to step 3)
   Some environments may only have one data drop to the user. In this case, it is possible to make use of the PC port on the back of most of the Polycom telephone models.
   The Polycom telephones support CDP to configure the correct VLANs for voice and data. The client personal computer can be plugged into the PC port of the Polycom telephone and still appear on VLAN 100 while the voice traffic from the telephone appears on VLAN 200.
   Other endpoints that do not support CDP need to have static VLAN assignments on the switch. In the following example, ports 1-4 are statically assigned, and ports 5-12 are assigned using CDP.
   From the global configuration mode:
   ```
   interface range FastEthernet 0/1-4
   switchport access vlan 200
   mls qos trust dscp
   
   interface range FastEthernet 0/5-12
   mls qos trust dscp
   switchport voice vlan 200
   ```

5. Enable Layer 3 routing between the VLANs on the switch.
   From the global configuration mode:
   ```
   router rip
   version 2
   network 10.0.0.0
   
   ip routing
   ip route 10.60.0.0 255.255.0.0 Vlan200
   ip route 10.70.0.0 255.255.0.0 Vlan100
   ip route 0.0.0.0 0.0.0.0 Vlan100 10.0.0.1 //default route
   ```

   By design, DHCP broadcasts are not forwarded across VLANs. The following configuration enables the relay of DHCP broadcasts from VLAN 100 to VLAN 200. This is assuming the DHCP server resides on the default VLAN 100 and has an IP address of 10.70.0.5
   From the global configuration mode:
Advanced Security Design in VLANs with a single switch network

The following diagram displays how you could use VLANs to design a security environment where the voice network cannot contact or be contacted by the data network. Only the "ININ" VLAN is routed to the require networks. This may be needed for extra security-conscious deployments but note that it does increase the level of complexity.

![Diagram of VLANs]

When deploying IP telephones, it is recommended to place telephones in a voice VLAN. A voice VLAN can protect the telephones from broadcast traffic as well as provide security. CIC clients should remain in a data VLAN or the default VLAN. The SIP Softphone can also be placed on the Default VLAN. Currently, it requires dual Network Interface Cards (NICs) if splitting the traffic was a requirement. This configuration is only possible on switches that support Layer 3 routing. The Layer 3 routing is required to let the different networks communicate with each other. This is all done in the routing table on the switch.

<table>
<thead>
<tr>
<th>VLAN</th>
<th>Purpose</th>
<th>Recommended Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Data</td>
<td>The Default for Data Traffic</td>
<td>100</td>
</tr>
<tr>
<td>Voice</td>
<td>IP Phones only</td>
<td>200</td>
</tr>
<tr>
<td>ININ</td>
<td>CIC server, Interaction Media Server, Interaction SIP Proxy, and others</td>
<td>300</td>
</tr>
</tbody>
</table>

The VLANs are configured in such a way that the data VLAN 100 is not able to communicate with the voice VLAN 200. This creates a higher level of security by preventing RTP sniffing. Interaction Media Servers, Interaction Gateways, and the CIC servers that are on VLAN 3 are able to communicate with the default VLAN 100. This is to enable client connections and also to enable Web-based configuration from the client machines. If desired, you can place the Interaction Gateways and the Interaction Media Servers in the voice VLAN 200 for added security. However, in that case, Web-based configuration for the Interaction Gateways and other systems may only be performed from the CIC server.

This example configuration is based on the recommended setup and the following network model using a Cisco Catalyst 3560 switch:
1. Create the VLANs for voice and CIC devices.  
   Start in the privileged exec mode.

   ```
   configure terminal
   vlan 200
   name "Voice"

   vlan 300
   name "ININ"

   exit
   ```

2. Configure the VLAN interfaces with IP addresses.  
   From the global configuration mode:

   ```
   interface vlan 100
   ip address 10.70.0.1 255.255.0.0

   interface vlan 200
   ip address 10.60.0.1 255.255.0.0

   interface vlan 300
   ip address 10.50.0.1 255.255.0.0
   ```
3. Assign the switch ports to the respective VLANs using static port assignment.
   From the global configuration mode:
   ```
   interface range FastEthernet 0/1-4
   switchport access vlan 300
   
   interface range FastEthernet 0/21-24
   switchport access vlan 200
   
   exit
   ```

4. If using CDP: (alternative to Step 3)
   Some environments may only have one data drop to the user. In this case, it is possible to make use of the PC port on the back of most of the Polycom telephone models.
   The Polycom telephones support CDP to configure the correct VLANs for voice and data. Therefore, in this case, the client personal computer can be plugged into the PC port of the Polycom telephone and still appear on VLAN 100 while the voice traffic from the telephone appears on VLAN 200.
   Other endpoints that do not support CDP need to have static VLAN assignments on the switch. In the following example, ports 1-4 are statically assigned, and ports 5-12 are assigned using CDP.
   From the global configuration mode:
   ```
   interface range FastEthernet 0/1-4
   switchport access vlan 300
   mls qos trust dscp
   
   interface range FastEthernet 0/21-24
   mls qos trust dscp
   switchport voice vlan 200
   
   exit
   ```

5. Enable Layer 3 routing between the VLANs on the switch.
   From the global configuration mode:
   ```
   ip routing
   ip route 10.60.0.0 0.0.255.255 Vlan200
   ip route 10.65.0.0 0.0.255.255 Vlan300
   ip route 10.70.0.0 0.0.255.255 Vlan100
   ip route 0.0.0.0 0.0.0.0 Vlan100
   ```

   By design, DHCP broadcasts are not forwarded across VLANs. The following set of commands enables the relay of DHCP broadcasts to VLAN 200 only. This is assuming that the DHCP server resides on the default VLAN 100 and has an IP address of 10.70.0.5.
   ```
   Important!
   IP addresses should be statically assigned on the devices in VLAN 300. Or, if required, the following can also be applied to VLAN 300.
   ```
   From the global configuration mode:
   ```
   interface vlan 200
   ip helper-address 10.70.0.5
   exit
   ```

7. Implement security.
   Extended access-lists are used to restrict access to the voice VLAN from the data VLAN. The access-list should be applied to the egress of the VLAN 200 interface (10.60.x.x is the voice VLAN and 10.70.x.x is the data VLAN).
   From the global configuration mode:
   ```
   access-list 101 deny ip 10.70.0.0 0.0.255.255 10.60.0.0 0.0.255.255
   access-list 101 permit any any
   
   interface vlan 200
   ip access-group 101 out
   ```
VLAN trunking

If the need arises to use more than one switch in the network, you can create VLAN trunks between two switches using VLAN Trunk Protocol (VTP). The configuration can be propagated through the network by administering it at a single switch.

Warning!
Some networking professionals have strong opinions on using VTP because of security concerns.

This example configuration is based on the following network model that consists of two Cisco Catalyst switches and two VLANs.

In this case, Switch A is a Cisco Catalyst 3560 switch and is the server in the VTP domain. Switch B is a Catalyst 3500 XL, a client in the VTP domain. For the purposes of this test, the VTP domain is named qostest and password authentication is disabled. Enabling password authentication for the VTP domain is strongly encouraged for a production environment. Switch A and B are connected to each other through ports fa0/12 on each switch. That port is configured as a trunk port on both switches.

1. Set up the VLAN trunks on both of the switches. These commands must be run on both switches.
   From the privileged exec mode:
   ```
   Configure terminal
   interface fa0/12
   switchport trunk encapsulation dot1q
   switchport mode trunk
   ```

2. Set up VTP on switch A / Catalyst 3560.
   From the privileged exec mode:
   ```
   configure terminal
   ```
3. Set up VTP on switch B / Catalyst 3500 XL

   From the privileged exec mode:

   ```
   configure terminal
   vlan database
   vtp client
   vtp domain qostest
   ```

Of course, since both the voice and data VLAN information is set down a single trunk port, network congestion can occur in the trunk links. The next section deals with enabling QoS on the switches.
Larger networks with multiple switches

Using VLANs is useful for separating the voice traffic from the data traffic. This creates some freedom for the voice traffic from network congestion. In a single switch network, since voice traffic does not leave a switch to reach its destination, QoS issues are not experienced. But, in the case of larger LAN configurations, the voice VLAN could span across multiple switches and the bottlenecks could occur at the trunk or uplink ports. Along with the recommended VLAN configuration, the specific uplink interfaces on the switches should have QoS enabled for the voice traffic.

In the scenario pictured in the following diagram, clients from switch A are generating traffic to the clients on switch B. Both switches A and B are up linked to a Layer 3 switch which does the routing. The uplinks between switch A and B are saturated with data traffic which can cause audio problems for calls going from network A to network B.

Note:
Enabling QoS on the physical ports that connect directly to the voice devices and telephones improves the quality of audio as the switch gives priority to the IP packets marked with DSCP when switch queuing is configured.
**The required method for Layer 3 switches**

This method works and is tested on the Cisco Catalyst 3560. Assume an IP telephone is connected to FastEthernet 0/4. To enable QoS on interface fa0/4 of a switch, you must enter the following commands. Always use DSCP when possible as it can traverse Layer 2. The marking is maintained as the packet moves from Layer 2 to Layer 3.

From the privileged exec mode:

```bash
configure terminal
mls qos
interface fa0/4
mls qos trust dscp
end
```

**Alternative method for older switches (not recommended)**

The Cisco Catalyst 3500 XL series and other older switches are Layer 2 switches. They cannot see Layer 3 DSCP markings and, as a result, perform QoS with the old Layer 2 Class of Service (CoS) markings. You can configure these switches to mark all incoming packets from the physical ports where the telephones are connected with a Layer 2 default CoS value. The switch queues the higher priority IP packets with preference over the default untagged data. The following configuration has been tested on a Cisco Catalyst 3500 XL switch with 12 ports. You must make sure that you have a strategy in place to remark Layer 2 CoS to Layer 3 DSCP if you must use a Layer 2 switch.

From the privileged exec mode:

```bash
configure terminal
interface fa0/4
switchport priority default 5
```

**Ingress DSCP mapping (Cisco)**

As mentioned previously, Cisco Catalyst switches support trusting using DSCP (DiffServ), IP Precedence, or CoS values on ingress frames. Internally, the switch maps the IP precedence or CoS value to a DSCP value. The following tables illustrate the default mapping tables for CoS-to-DSCP and IP Precedence-to-DSCP, respectively. Based on these mappings, the packets are placed in the appropriate queue.

<table>
<thead>
<tr>
<th>CoS</th>
<th>0</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP</td>
<td>0</td>
<td>8</td>
<td>16</td>
<td>24</td>
<td>32</td>
<td>40</td>
<td>48</td>
<td>56</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>IP Precedence</th>
<th>0</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP</td>
<td>0</td>
<td>8</td>
<td>16</td>
<td>24</td>
<td>32</td>
<td>40</td>
<td>48</td>
<td>56</td>
</tr>
</tbody>
</table>

The above mappings are configurable. For example, the following command changes the mappings from the default CoS-to-DSCP map to this custom table. This command is automatically run during Cisco’s AutoQoS. AutoQoS should not be used. AutoQoS is designed for Cisco products.

From the privileged exec mode:

```bash
configure terminal
mls qos map cos-dscp 0 8 16 26 32 40 48 56
exit
```
AutoQoS (Cisco)

AutoQoS is a Cisco proprietary method for simplifying QoS configuration. It is not designed for PureConnect products nor any other non-Cisco Voice-over-Internet-Protocol (VoIP) deployment. Depending on how AutoQoS is implemented, it is very likely to result in an incorrect network configuration for a CIC system and is not recommended.

AutoQoS sometimes relies on Network Based Application Recognition (NBAR) to find VoIP traffic, and then apply an appropriate static template. If this is applied prior to deployment of the VoIP system, then the resulting template can be incorrect. Likewise, in cases where the live system is changed post deployment—for example when an IP phone is plugged in to a new switch port—then AutoQoS has no built-in mechanism for compensating. AutoQOS may have already configured the port for a "workstation".

Important!
Currently, the configurations generated by AutoQoS are static. If parameters are modified in these cases, Cisco AutoQoS needs to be disabled and re-enabled to take the modifications into consideration for enabling the appropriate QoS parameters. Disable Cisco AutoQoS before changing these configuration parameters and then re-apply it again.

**WAN QoS (Cisco)**

When working with a remote link, bandwidth can be limited and expensive. QoS becomes a requirement in these situations. The following configuration was verified on a Cisco Catalyst 3640 and a Cisco Catalyst 2811 using a serial link configured at 1.5Mbs. In our test, the link was saturated with data traffic and we were unable to have clear telephone conversations. All calls experienced “choppy” audio.

After applying the QoS commands, the excessive data was delay or dropped from the queue on the gateways and the audio became clear. Below is the tested configuration with Cisco IOS version QoS configuration best practices. This example is based on a Cisco Catalyst 2811 and Cisco Catalyst 3640 in the following network scenario:

1. Make class maps to match voice packets and SIP signaling:
   Start in privileged exec mode:

   ```
   configure terminal
   access-list 101 permit tcp any eq 8060 any
   access-list 101 permit tcp any any eq 8060
   class-map match-any voice
   match ip dscp ef
   exit
   class-map match-any voip-control
   match protocol sip
   match access-group 101
   exit
   ```

   **Note:**
   Using keywords such as `sip` match only standard IANA port numbers.

2. Make policy-maps and the bandwidth assignments.

   **Note:**
   Recommended values are used for bandwidth assignment. Fair-queuing is used for non-voice traffic.

   From the global configuration mode:

   ```
   policy-map voip-qos
   class voip-control
   bandwidth percent 2
   class voice
   bandwidth percent 70
   class class-default
   fair-queue
   exit
   ```

3. Associate the policy to the WAN interface. In this example the serial interface is used.

   From the global configuration mode:

   ```
   interface serial 1/0
   service-policy output voip-qos
   ```

Many more tweaks are possible and there are many other ways this could be accomplished. This example gives you general guidelines. Shaping and policing may also be required on a WAN connection.

**Class Based Weighted Fair Queuing / Low Latency Queue CBWFQ/LLQ**
CBWFQ/LLQ is considered the best practice for implementing QoS on WAN networks at the core edge routers. The configuration is based on the assumption that trust boundaries are established on the network infrastructure and traffic is marked at the access layer according to the following template:

<table>
<thead>
<tr>
<th>Application/Protocol</th>
<th>DSCP Marking</th>
<th>CoS Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice traffic (RTP)</td>
<td>EF</td>
<td>5</td>
</tr>
<tr>
<td>PureConnect Notifier</td>
<td>AF22</td>
<td>1</td>
</tr>
<tr>
<td>Recordings</td>
<td>AF11</td>
<td>1</td>
</tr>
<tr>
<td>Bulk transfer traffic</td>
<td>AF11</td>
<td>1</td>
</tr>
<tr>
<td>(Web/FTP/etc)</td>
<td>AF12</td>
<td></td>
</tr>
<tr>
<td></td>
<td>AF13</td>
<td></td>
</tr>
<tr>
<td>Transactional applications</td>
<td>AF21</td>
<td>2</td>
</tr>
<tr>
<td>(Databases etc)</td>
<td>AF22</td>
<td></td>
</tr>
<tr>
<td></td>
<td>AF23</td>
<td></td>
</tr>
<tr>
<td>Mission critical applications</td>
<td>AF31</td>
<td>3</td>
</tr>
<tr>
<td>(Core business traffic)</td>
<td>AF32</td>
<td></td>
</tr>
<tr>
<td></td>
<td>AF33</td>
<td></td>
</tr>
<tr>
<td>Interactive video traffic</td>
<td>AF41</td>
<td>4</td>
</tr>
<tr>
<td></td>
<td>AF42</td>
<td></td>
</tr>
<tr>
<td></td>
<td>AF43</td>
<td></td>
</tr>
<tr>
<td>IP routing</td>
<td>CS6</td>
<td>6</td>
</tr>
<tr>
<td>(RIP, BGP, OSPF etc)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Streaming video</td>
<td>CS4</td>
<td>4</td>
</tr>
<tr>
<td>Voice/Video signaling</td>
<td>CS3</td>
<td>3</td>
</tr>
<tr>
<td>(SIP, etc)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Network management traffic</td>
<td>CS2</td>
<td>2</td>
</tr>
<tr>
<td>(SNMP etc)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Scavenger traffic</td>
<td>CS1</td>
<td>1</td>
</tr>
<tr>
<td>Unclassified traffic</td>
<td>CS0 (BE)</td>
<td>0</td>
</tr>
</tbody>
</table>

This document concentrates on voice traffic only. As mentioned previously, all PureConnect voice endpoints are capable of marking their packets. Assuming there are no QoS problems on the LAN side, the issue arises on the edge devices where bandwidth is limited when connecting to a service provider or any other edge device of another network.

Reclassification, queuing, and congestion avoidance using CBWFQ/LLQ with a WRED tail drop policy is the best recommended configuration for such a scenario. PureConnect products do not use WRED extensions such as ECN.

The following table outlines the best practices for reclassification, queuing and congestion avoidance:
<table>
<thead>
<tr>
<th>Application/Protocol</th>
<th>Original DSCP Marking</th>
<th>New DSCP Marking</th>
<th>Drop Policy</th>
<th>Queue used</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice traffic (RTP)</td>
<td>EF</td>
<td>EF</td>
<td>Not applied</td>
<td>LLQ</td>
</tr>
<tr>
<td>PureConnect Notifier Traffic</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Recordings</td>
<td>AF11</td>
<td>AF22</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>Bulk transfer traffic</td>
<td>AF11, AF12, AF13</td>
<td>AF22</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(Web/FTP/etc)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transactional applications</td>
<td>AF21, AF22, AF23</td>
<td>AF32</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(Databases etc)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Mission critical applications</td>
<td>AF31, AF32, AF33</td>
<td>AF31</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(Core business traffic)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Interactive video traffic</td>
<td>AF41, AF42, AF43</td>
<td>CS4</td>
<td>DSCP-based</td>
<td>LLQ</td>
</tr>
<tr>
<td>IP routing</td>
<td>CS6</td>
<td>CS6</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(RIP, BGP, OSPF etc)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Streaming video</td>
<td>CS4</td>
<td>CS4</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>Voice/Video signaling</td>
<td>CS3</td>
<td>CS3</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(SIP, etc)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Network management traffic</td>
<td>CS2</td>
<td>AF21</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>(SNMP etc)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Scavenger traffic</td>
<td>CS1</td>
<td>CS0</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
<tr>
<td>Unclassified traffic</td>
<td>CS0 (BE)</td>
<td>CS0</td>
<td>DSCP-based</td>
<td>CBWFQ</td>
</tr>
</tbody>
</table>

The following example configuration outlines an implementation on the edge router of the network. The voice traffic is given a LLQ of 25% of the available bandwidth.

1. Make class maps to match network traffic on the edge router.
   Start in privileged exec mode:

```bash
configure terminal
class-map match-any voice
match ip dscp ef
class-map match-any bulk-data
match ip dscp af11
match ip dscp af12
match ip dscp af13
class-map match-any transactional
match ip dscp af21
match ip dscp af22
match ip dscp af23
class-map match-any mission-critical
match ip dscp af31
match ip dscp af32
match ip dscp af33
class-map match-any routing
match ip dscp cs6
class-map match-any voice-signaling
```
match ip dscp cs3
match protocol sip
class-map match-any net-management
match ip dscp cs2
exit

2. Make policy-maps that will handle the re-classification, queuing, and dropping.

   **Note:**
   The class voice-media and voice-signaling together get 25% priority (21+4).

   From the global configuration mode:

   ```
   policy-map qos-policy
   class voice-priority percent 21
   class bulk-data
   bandwidth percent remaining 20
   set ip dscp af22
   random-detect dscp-based
   class transactional
   bandwidth percent remaining 15
   set ip dscp af32
   class mission-critical
   bandwidth percent remaining 30
   set ip dscp af32
   class routing
   bandwidth percent remaining 5
   set ip dscp cs6
   class voice-signaling
   priority percent 4
   set ip dscp cs3
   class net-management
   bandwidth percent remaining 5
   set ip dscp af21
   class class-default
   set ip dscp 0
   random-detect dscp-based
   ```

3. Associate the policy to the WAN interface.
   In this example, a serial link running at T1 speed is used.
   From the global configuration mode:

   ```
   interface serial 1/0
   bandwidth 1544
   service-policy output qos-policy
   ```
Call Admission Control (CAC)

CIC has built-in Call Admission Control on the SIP Line. It can be used in situations where you must limit the amount of calls that the system should place. Adding just one more call can degrade the voice quality of all the other active calls when there is not sufficient bandwidth to support it. In the example below, you can see that we have limited the CIC server to have a maximum of 10 calls associated with this SIP line. This setting does not account for bandwidth. The person doing the implementation must plan for the codec type and layer 2 overheard required.

Note:
CIC cannot guarantee the bandwidth needed for other activities on the WAN in this configuration.
Basic LAN QoS (Dell)

Multiple VLAN with trunks

The following example deals with configuring the two VLAN model using Dell hardware. The Dell PowerConnect 6024 has layer 3 routing capability and is used to route between the VLANS and the Dell PowerConnect 3424P was used for end point connectivity to the clients and the telephones. The 3424P switch was selected because it supports IEEE 802.3AF inline power. The network model used in this following example:

1. Configure VLANs on the 6024 and the 3424P.
   On the 6024 – From the privileged exec mode:
   ```
   configure
   vlan database
   vlan 2
   exit
   ```

   On the 3424P – From the privileged exec mode:
   ```
   configure
   vlan database
   vlan 2
   exit
   interface range ethernet e1-10
   switchport mode access
   switchport access vlan 1
   exit
   interface range ethernet e10-20
   switchport mode access
   switchport access vlan 2
   ```
2. Configure VLAN trunks on both switches.
   Run these commands on both devices. The assumption is that port 24 is the uplink/trunk port on both switches.
   From the privileged exec mode:

   ```
   configure
   interface ethernet g24 //e24 for the 3424P
   switchport mode trunk
   switch port trunk allow vlan add 1,2
   exit
   ```

3. Configure routing and DHCP forwarding on the 6024.
   The assumption is that the DHCP server has an IP address of 10.70.0.5 and exists on VLAN1. On the Dell 6024 switch, VLAN routing is enabled by default and is not necessary to create routes for VLANs that appear directly connected to the switch.
   From the privileged exec mode:

   ```
   configure
   ip dhcp relay enable
   ip dhcp address 10.70.0.5
   router rip enable
   exit
   ```

4. Configure QoS on the 3424P.
   Unlike the Cisco switches, the QoS trust commands on the Dell switches are applied globally instead of on the individual interfaces.
   From the privileged exec mode:

   ```
   configure
   qos trust dscp
   exit
   ```
CIC switchover and QoS

CIC enables you to have the switchover feature, which uses one CIC server as a backup should the first CIC server become unresponsive or have a system hardware failure. In this situation, the secondary CIC server can use the PureConnect QoS driver to enable QoS for the ping and the auxiliary connection. To enable this feature, you must set the **Switchover Use Qos For Ping** server parameter to **enabled**. You can also set the following CIC server parameters for more QoS functionality:

- **Switchover Qos DSCP** – When this parameter is enabled, set it to the value in the QoS byte. Differentiated Service Code Point (DSCP) is the six most significant bits of a packet. You can use DSCP to prioritize QoS traffic on switchover.

- **Switchover Ping on Aux Connection** – When this parameter is enabled, set it to **Yes** or **1** to move the TS ping from the main data connection to the auxiliary connection.

**Note:**
To enable QoS on the ping on the auxiliary connection (not the main connection), you must enable both the **Switchover Ping on Aux Connection** and the **Switchover Use Qos For Ping** parameters. QoS will not be used on ping if only the **Switchover Use Qos For Ping** parameter is enabled.
Telephones

Polycom

Polycom telephones can be provisioned by the CIC server. This feature greatly reduces setup, configuration, and maintenance time. Polycom has chosen (101100) as the default DSCP value for SIP signaling. A future enhancement may allow for this to be modified. They are using the recommended value (101110) EF for RTP. You can modify this in the Advanced Options section in the Managed Phone configuration.

Legacy Polycom telephones will mark RTP and SIP. The version of the firmware load will determine the default values, but this can be adjusted in the Sip.cfg file.

```xml
<QOS>
  <Ethernet>
    <RTP qos.ethernet.rtp.user_priority="5"/>
    <CallControl qos.ethernet.callControl.user_priority="5"/>
      <Other qos.ethernet.other.user_priority="2"/>
  </Ethernet>
  <IP>
    <RTP qos.ip.rtp.min_delay="1" qos.ip.rtp.max_throughput="1" qos.ip.rtp.max_reliability="0"
      qos.ip.rtp.min_cost="0" qos.ip.rtp.precedence="5"/>
    <CallControl qos.ip.callControl.min_delay="1" qos.ip.callControl.max_throughput="0"
      qos.ip.callControl.max_reliability="0" qos.ip.callControl.min_cost="0" qos.ip.callControl.precedence="5"/>
  </IP>
</QOS>
```

Soft phones

See CIC Managed IP Phones Administrator's Guide for details on setting the DSCP for soft phones. The PureConnect QoS driver, by default, marks voice traffic (RTP) with an EF DSCP value. The associated SIP signaling is marked with a CS3 DSCP value. These values are set once the softphone is provisioned through Interaction Administrator.
Fax

The T.30 and T.38 fax protocols are sensitive to network conditions and should always have QoS set. The priority is based on the SIP line setting or station (ToS byte) in Interaction Administrator. Genesys recommends that you change this to EF (See Marking). Fax stations do not use this setting with TX T.38 packets. When the voice call switches to fax, the time constraints are handled within the fax protocol itself. The fax image is not voice. The fax tones (RTP) are considered to be voice. Telephony Services should not mark the DSF of the packet.

| ITU T.38 | Procedures for real-time Group 3 facsimile communication over IP networks | http://www.itu.int/rec/T-REC-T.38/e |

In the case of WAN deployments with business critical applications, it might be necessary to provide higher priority to the T.30 or T.38 traffic. The recommended solution is to classify the fax traffic using the Cisco MQC and assign higher priority as explained in the WAN QoS section of this document. Using the information below, you can change the default ports for T.30 or T.38 and use an access list or a class map to mark the T.30 or T.38 traffic.
Interaction Gateway

By default, Interaction Gateway sets all voice packets with a DSCP value of (101110) EF. You can also configure the SIP Marking from the default of CS3.

On the Settings > Media page of the Interaction Gateway web user interface, select Set RTP Marking on the Media > Type of Service and Signaling on Spans > Type of Service. Interaction Gateway exposes these values with the entire TOS Byte, depending on the release level. B8 is the same as EF and 60 is the same as CS3.
### TDM Spans

<table>
<thead>
<tr>
<th>Span</th>
<th>State</th>
<th>Protocol</th>
<th>Variant</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Tdm1</td>
<td>Ready</td>
<td>ISDN PRI</td>
<td>SESS</td>
<td></td>
</tr>
<tr>
<td>Tdm2</td>
<td>Not Ready</td>
<td>ISDN PRI</td>
<td>SESS</td>
<td></td>
</tr>
<tr>
<td>Tdm3</td>
<td>Ready</td>
<td>ISDN PRI</td>
<td>NI</td>
<td></td>
</tr>
<tr>
<td>Tdm4</td>
<td>Ready</td>
<td>ISDN PRI</td>
<td>NI</td>
<td></td>
</tr>
</tbody>
</table>

### SIP Span

<table>
<thead>
<tr>
<th>Span</th>
<th>State</th>
<th>Protocol</th>
<th>Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sip1</td>
<td>Ready</td>
<td>UDP</td>
<td>5080</td>
</tr>
<tr>
<td></td>
<td>Ready</td>
<td>TCP</td>
<td>5080</td>
</tr>
<tr>
<td></td>
<td>Ready</td>
<td>TLS</td>
<td>5081</td>
</tr>
</tbody>
</table>
Opening client ports

CIC client applications also need a way to contact the CIC server from the accounting side of the PIX. You can map the ports that are needed through the firewall. Those ports depend on the applications that are deployed.

<table>
<thead>
<tr>
<th>CIC application</th>
<th>Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>CIC clients</td>
<td>TCP 3952</td>
</tr>
<tr>
<td>Interaction Fax</td>
<td>TCP 3952</td>
</tr>
<tr>
<td>Interaction Voicemail</td>
<td>TCP 3952</td>
</tr>
<tr>
<td>Interaction Recorder Client</td>
<td>TCP 5597</td>
</tr>
<tr>
<td>Interaction Supervisor</td>
<td>TCP 5597</td>
</tr>
<tr>
<td>Interaction Designer</td>
<td>TCP 5597</td>
</tr>
<tr>
<td>Interaction Attendant</td>
<td>TCP 5597</td>
</tr>
<tr>
<td>Interaction Recorder Client</td>
<td>TCP 3952</td>
</tr>
<tr>
<td>SIP Soft Phone</td>
<td>TCP 3952</td>
</tr>
</tbody>
</table>
Interaction Media Server

For more information about Interaction Media Server and QoS, see Interaction Media Server Technical Reference.
## References and further reading

<table>
<thead>
<tr>
<th>Title</th>
<th>Author(s)</th>
<th>Publisher</th>
<th>URL</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco QOS Exam Certification Guide</strong></td>
<td>Odom, Cavanaugh</td>
<td>Cisco Press</td>
<td></td>
</tr>
<tr>
<td><strong>CCNP Switching Study Guide</strong></td>
<td>Lammle, Quinn</td>
<td>Sybex</td>
<td></td>
</tr>
<tr>
<td>Dell PowerConnect support and white pages</td>
<td></td>
<td></td>
<td><a href="http://www.dell.com">http://www.dell.com</a></td>
</tr>
</tbody>
</table>
Appendix A: OpenSSL Copyright

NOTICE
This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/).

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## Change log

<table>
<thead>
<tr>
<th>Date</th>
<th>Changes</th>
</tr>
</thead>
<tbody>
<tr>
<td>24-October-2011</td>
<td>Initial release for Interaction Center 4.0</td>
</tr>
<tr>
<td>6-March-2012</td>
<td>IC-93044 - QoS functionality for SAPI TTS should be optional</td>
</tr>
<tr>
<td>30-November-2012</td>
<td>Remove erroneous content regarding VLAN tagging capability</td>
</tr>
<tr>
<td>6-August-2013</td>
<td>- Removed references to obsolete products</td>
</tr>
<tr>
<td></td>
<td>- Updated screen shots</td>
</tr>
<tr>
<td></td>
<td>- Updated references to external standards and resources</td>
</tr>
<tr>
<td></td>
<td>- IC-107632 - Warning should prompt when network driver gets reset during an install</td>
</tr>
<tr>
<td>5-September-2014</td>
<td>Updated documentation to reflect changes required in the transition from version 4.0 SU# to CIC 2015 R1, such as updates to product version numbers, system requirements, installation procedures, references to Interactive Intelligence Product Information site URLs, and copyright and trademark information.</td>
</tr>
<tr>
<td>13-April-2015</td>
<td>- Update &quot;Copyright and Trademark Information&quot; page</td>
</tr>
<tr>
<td></td>
<td>- Modify formatting to new corporate standards</td>
</tr>
<tr>
<td>13-October-2015</td>
<td>- Updated cover page with new corporate branding</td>
</tr>
<tr>
<td></td>
<td>- Updated &quot;Copyright and Trademark Information&quot; page</td>
</tr>
<tr>
<td></td>
<td>- Replace rebranded graphics displaying example interfaces</td>
</tr>
<tr>
<td></td>
<td>- Minor edits</td>
</tr>
<tr>
<td>2-January-2017</td>
<td>- Updated &quot;Copyright and trademark information&quot; page</td>
</tr>
<tr>
<td></td>
<td>- IC-141374 - Modify QoS document to include Notifier QoS traffic with media server - Updated instances of Notifier DSCP marking from AF41 to EF</td>
</tr>
<tr>
<td></td>
<td>- IC-141949 - Tables need port ranges - Removed instances referencing Cisco IP telephones that are no longer supported. Updated RTP port usage topic. Added cross-reference to CIC Port Maps and Data Flow Diagrams Technical Reference.</td>
</tr>
<tr>
<td>21-September-2017</td>
<td>Rebranded to Genesys.</td>
</tr>
<tr>
<td>6-April-2018</td>
<td>Updated document format.</td>
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