Frequently Asked Questions

This appendix contains frequently asked questions about Cisco IPICS, and provides answers to these questions:

Q. Does Cisco IPICS allow multiple Cisco IPICS servers to use the same RMS?

A. No, Cisco IPICS does not support the use of multiple Cisco IPICS servers for the same RMS. Each server must have the use of resources on a corresponding RMS to ensure proper functionality.

Q. Does Cisco IPICS support more than one RMS in the same location.

A. Yes.

Q. What makes a channel remote?

A. A channel is remote when it is in a different multicast domain than the user who is accessing it.

Q. What does the designation “REMOTE” mean for a PMC location?

A. The REMOTE location is available only to PMC users. When a PMC user chooses the REMOTE location from the Location drop-down list box, connectivity is established with the appropriate RMS via a SIP-based unicast connection for each channel or VTG that has been assigned to the user. For more detailed information about locations, see the “Managing Locations” section on page 2-38.

Q. If I only have one router in a location and my channel is defined as ALL, will the channel be accessible to a user?

A. Yes. However, if a router location is defined as ALL, a channel that is not also configured as ALL will not be accessible to users or VTGs that the router supports.

The ALL location defines the scope or reachability of a multicast address. For this reason, the ALL location is applicable to channels and VTGs, which are associated with multicast addresses, but not applicable to IP phones or RMS components, which are not associated with multicast addresses. For more detailed information about locations, see the "Managing Locations" section on page 2-38.

Q. How many resources (voice ports, multicast addresses) do I need?

A. The following guidelines apply to the use of resources:

- Every channel that is active in a VTG uses one DS0 pair (also called a loopback)
- Every sub-VTG in a VTG uses one DS0 pair
- Every SIP connection uses one DS0 pair per channel or VTG per user, per location
- Local channels do not use any DS0 pairs
- G.729, which is used for a SIP connection, requires DSP resources
- A dial connection uses two DS0 pairs (for two multicast addresses) for the first dial user, and then one DS0 per subsequent dial user

The following items do not use voice resources:

- A user with an associated channel (the system only uses resources when the user logs in from a remote location)
- A VTG that includes only users
- User Groups
- Channel Groups

Q. Why would a VTG suddenly become active or inactive unexpectedly?

A. If a VTG unexpectedly becomes active or inactive, it could be because the VTG can be associated to a policy and the policy can execute, which could change the status of the VTG.

Q. What is the difference between the *Belongs To* attribute and the *Accessible To* attribute for an ops view?

A. The Belongs To attribute determines the ops view to which the resource belongs, or that the ops view owns. After a new ops view is created, the system administrator can associate resources, such as channels or users, to the ops view. The operator creates an operator user who belongs to that ops view and who can manage the ops view resources that are visible within the specific ops view.

The Accessible To attribute specifies that the resource is accessible to, or visible to, the ops view(s). Users only have access to the resources that are accessible to the ops view to which they belong. See "Configuring and Managing Cisco IPICS Operational Views" for more information about these attributes.

Q. What codecs does the dial engine support?

A. The dial engine supports only G.711 u-law. This codec is not configurable.

Q. How are license ports and DS0 loopback port resources counted in Cisco IPICS release 2.0(1)?

A. A single LMR (LMR Port) license is used when a channel is enabled.

A single PMC license is used each time that a PMC user logs in to the system. If a PMC user logs in multiple times, a license is used for each login occurrence.

A single IP phone license is used each time that an IP phone user (PMC xml client) logs in to the system.

A single Multicast (Multicast Port) license is used when a VTG is activated.

A single PSTN (Dial User) license is used in each of the following scenarios:

- One license is used for an active inbound call
- One license is used for an active outbound call

A single DS0 loopback pair is used in the following scenarios:

- For each remote channel on a PMC
- For each channel in an active VTG

- For each instance of an active VTG that is accessed by a dial-in or dial-out user, regardless of the number of users who are connected to the VTG

A single Ops View license is used for each configured ops view.

Q. Will an IP phone keep working if it loses connectivity to the Cisco IPICS server while the phone user is logged in to the Cisco IPICS service?

A. If a phone loses connectivity to the Cisco IPICS server while the phone user is logged in to the Cisco IPICS service, the service retains its current state and the user can continue to use the PTT functionality for the channel or VTG that is currently selected. However, the phone cannot connect to other channels or VTGs until connectivity to the server is re-established.

Q. In the case of a notification action that is in the form of an e-mail, sms, or page and a dial notification to a large number of users (for example, 100 users), what is the sequence of notification events?

A. The dial engine uses a scalable, multi-threaded dial-pool implementation for dialing out to users. Ports from the available dial pools are used by the currently executing policy notification/invite actions. If there are fewer dial ports available than what is needed, the other policy actions are put in a waiting state until more ports become available.

A call is considered successful when the call recipient authenticates. If there is no authentication, the system moves to the next dial preference that is listed in the Dial Preferences for the user in the user profile until either the call is successful or every number has been tried by the system. For detailed information, see the "Allocating Dial Ports for the Dial-In/Invite and Notification Features" section on page 6-29 and the "Managing Communications Preferences for a User" section on page 3-23.

Q. During a dial-out to users, does a dispatcher get notified about numbers that have not yet been reached and is there any way to determine how long it should take to reach all the participants in a VTG?

A. Dialed numbers display in the **Policy Execution Status > Executed/Executing Policy** window, showing which numbers have been reached and which are still in progress.

For each available port, the user must authenticate by entering a digit ID/PIN and then the notification message is played. Whenever errors occur, such as the user entering an incorrect digit ID or PIN and/or a timeout occurring when the user is not reached, the dial-out notification takes longer to complete. The total time for dial-out notification depends on these factors. For more information, see the "Viewing Information about Executing or Executed Policies" section on page 7-22.

Q. How do you integrate the dial engine into an existing network that runs an earlier version of Cisco Unified CallManager and does not have native SIP trunk support?

A. This integration can be accomplished by using a Cisco IOS router that runs Cisco Unified CallManager Express as the SIP provider and configuring an H.323 Intercluster Trunk (ICT) between the Cisco Unified CallManager and the SIP provider. For detailed information, refer to the *Solution Reference Network Design (SRND) for Cisco IPICS Release 2.0(1)*.