Understanding Video Telephony

Cisco CallManager supports video telephony and thus unifies the world of voice and video calls. Video endpoints use Cisco CallManager call-handling features and access a unified voice and video solution for dialing and connecting video calls.

The Cisco CallManager video telephony solution offers these features:

- Supports video and video-related features, such as far-end camera control (FECC)
- Supports multiple logical channels that are needed to allow the transmission of video streams
- Transmits midcall, media-related messages that are needed for video (that is, transmits commands or indications that are needed for video calls)
- Supports H.323, Skinny Client Control Protocol (SCCP), and Session Initiation Protocol (SIP)
- Enhances locations and regions to provide bandwidth management
- Provides serviceability information, such as Call Detail Records (CDRs), about video calls

This section covers the following topics:

Introducing Video Telephony

The following topics discuss the details of video telephony in the Cisco CallManager environment:

- Additional Configuration for Video Calls, page 44-9
- Conference Control for Video Conferencing, page 44-9

Video Calls

The typical video call includes two or three Real-Time Protocol (RTP) streams in each direction (that is, four or six streams). The call can include the following stream types:

- Audio (same codecs as a normal call with additional codecs G.722 and G.728)
- Video (H.261, H.263, and Cisco VT Camera wideband video codecs) at a different port
- Far-end camera control (FECC) (optional)

SIP video supports the following video calls by using the SIP Signaling Interface (SSI):

- SIP to SIP
- SIP to H.323
- SIP to SCCP
- SIP intercluster trunk

SIP video calls also provide media control functions for video conferencing.

Call control for video calls operates the same way as the call control that governs all other calls. Refer to the "Call Control" section on page 22-2 in the Media Resource Management chapter.

Video Codecs

Common video codecs include H.261, an older video codec, H.263, a newer codec that gets used to provide internet protocol (IP) video, and H.264, a high-quality codec. The system supports H.264 for calls that use the Skinny Client Control Protocol (SCCP), H.323, and SIP protocols on originating and terminating endpoints only. The system also supports Regions and locations.

H.261 and H.263 codecs exhibit the following parameters and typical values:

- Bit rates range from 64 kbps to a few mbps. These bit rates can exist in any multiple of 100 bps.
- Resolution:
  - One-quarter Common Interchange Format (QCIF) (Resolution equals 176x144.)
  - Common Interchange Format (CIF) (Resolution equals 352x288.)
  - 4CIF (Resolution equals 704x576.)
  - Sub QCIF (SQCIF) (Resolution equals 128x96.)
  - 16CIF (Resolution equals 1408x1152.)
  - Custom Picture Format
- Frame Rate: 15 frames per second (fps), 30 fps
- Annexes: D.1, D.2, F, I, J, K, L.4, L.8, N, P.5, T, U, N, U, W

The Cisco VT Camera wideband video codec, which is a fixed-bit-rate codec, runs on a PC that is linked to a phone. This codec enables the PC to associate with a call that the phone receives. Cisco CallManager currently supports intracluster Cisco VT Camera wideband video codec calls but not intercluster Cisco VT Camera wideband video codec calls.

Cisco VT Advantage supports the Cisco VT Camera wideband video and H.263 codecs which can be used for intracluster and intercluster calls respectively. The support is based on correct configuration with related capabilities and regions. This support also applies to mid-call.

The bandwidth of video calls equals the sum of the audio bandwidth and the video bandwidth. The total bandwidth does not include overhead.

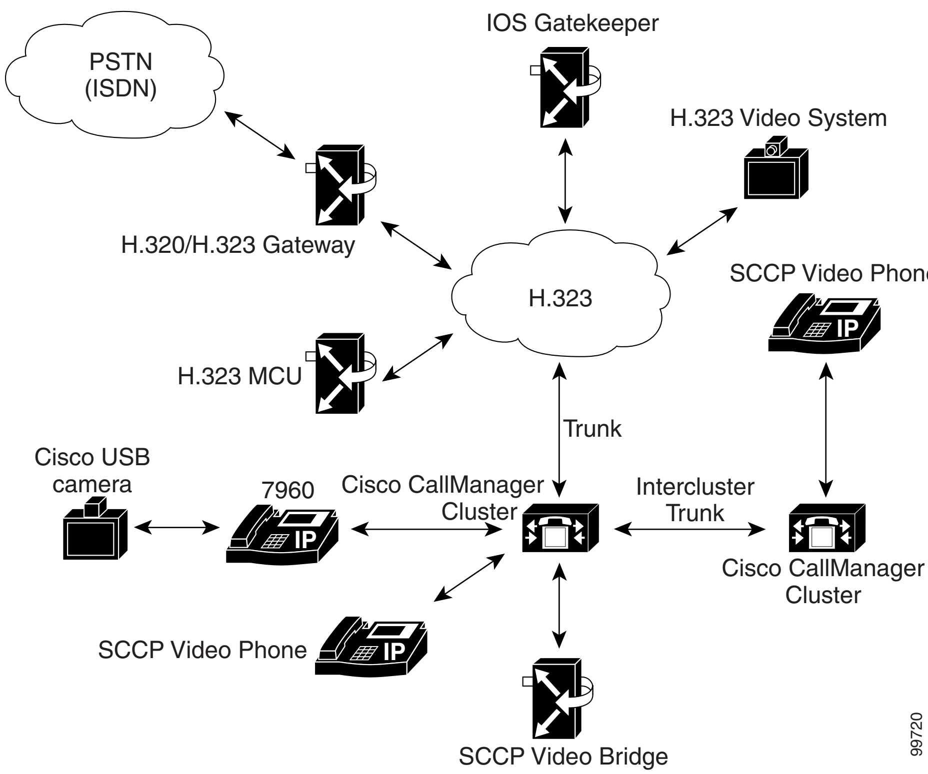
Example

A 384-kbps video call may be G.711 at 64 kbps (for audio) plus 320 kbps (for video). This sum does not include overhead. If the audio codec for a video call is G.729 (at 24 kbps), the video rate increases to maintain a total bandwidth of 384 kbps. If the call involves an H.323 endpoint, the H.323 endpoint may use less than the total video bandwidth that is available. Regardless of protocol, the endpoint may always choose to send at less than the max bit rate for the call.

Video Network

Figure 44-1 provides an example of a video network. In a successful video network, any endpoint can call any other endpoint. Video availability only exists if both endpoints are video enabled. Video capabilities extend across trunks.

Figure 44-1 Video Network Example



The Cisco video conference portfolio comprises the following H.323 devices:

- Cisco IP/VC 3511 (Video Bridge or Media Control Unit [MCU])
- Cisco IP/VC 3521 (BRI H.323/H.320 gateway)

- Cisco IP/VC 3526 (PRI H.323/H.320 gateway)
- Cisco IP/VC 3540 MCU (chassis-based bridge/gateway unit, which accepts multiple cards, and which supports H.323 and the Skinny Client Control Protocol. The IPVC Gateways only support H.323.)
- IOS H.323 Gatekeeper

Each of these devices supports the internet protocol (IP) network; the gateways support the Integrated Services Digital Network (ISDN).

Refer to the "Conference Bridge Configuration" section of the *Cisco CallManager Administration Guide* for details of configuring the Cisco IP/VC 3511 (MCU), 3540 (MCU) in Cisco CallManager Administration.

Enabling an Audio-Only Device with Video

You can enable an audio-only device with video by using a Cisco application, Cisco VT Advantage. You can associate the application with a Cisco IP Phone. This association can occur before a call is made or during a call (mid-call). Cisco IP Phones 7940/41, 7960/61, and 7970/71 support Cisco VT Advantage.

For example, a call occurs from a Cisco IP Phone 7960 to a video phone. The call is established as audio only. After Cisco VT Advantage is associated with the Cisco IP Phone 7960, the call gets reestablished as a video call.

During the association, Cisco CallManager receives updated capabilities for the phone via existing SCCP messages. After the updated capabilities are received, Cisco CallManager negotiates for video.

The media layer checks whether the regions allow video and whether both parties have video capabilities. If these conditions are met, the media layer establishes the video channels, and a video call gets established. Avoiding violation of administrative bandwidth constraints makes the region check necessary.

If the initial call involves an IP phone without a video, only audio location bandwidth gets reserved, and the media layer establishes an audio-only call.

H.323 Video

H.323 video exhibits the following characteristics:

- H.323 endpoints can be configured as H.323 phones, H.323 gateways, or H.323 trunks.
- Call forwarding, dial plan, and other call-routing-related features work with H.323 endpoints.
- H.323 video endpoints cannot initiate hold, resume, transfer, park, and other similar features.
- If an H.323 endpoint supports the empty capability set (ECS), the endpoint can be held, parked, and so forth.
- Some vendors implement call setup in such a way that they cannot increase the bandwidth of a call when the call gets transferred or redirected. In such cases, if the initial call is audio, users may not receive video when they are transferred to a video endpoint.
- No video media termination point (MTP) nor video transcoder currently exists. If an audio transcoder or MTP is inserted into a call, that call will be audio only. This is true when the IPVC audio transcoding capabilities is not being used. When the IPVC transcoders are used you can transcode the audio and send/receive video.
- For H.323 video calls, users must specify video call bandwidth.

Dynamic H.323 Addressing

You can configure a H.323 client with the E.164 address that is registered with the gatekeeper. E.164 addressing facilitates H.323 configuration and call routing by allowing the Cisco CallManager to route all calls in place of the gatekeeper. The gatekeeper that is to be configured requires the following characteristics:

- Forward all calls to the Cisco CallManager for routing.
- Calls that are routed from the Cisco CallManager must not be routed back to the Cisco CallManager.

Registering with the Gatekeeper

At boot time, Cisco CallManager loads static configuration information such as the E.164 address and the configured gatekeeper for each H.323 client. The H.323 clients in the same gatekeeper zone stay in one group. A registration with the gatekeeper gets initiated for the group. The process does not require individual registration for each member of the group.

H.323 clients that belong to the same gatekeeper but different zone remain part of a different group, and only one registration is initiated for this group. H.323 devices that belong to a different gatekeeper zone remain part of another group, and only one registration is initiated for this group. All members of the same group use the same technology prefix.

Call Processing

During an outbound call where the H.323 client is the called party, Cisco CallManager routes the call on the basis of DN to the H.323 device. Cisco CallManager uses the H.323 device configuration to determine whether the gatekeeper is configured and sends an Admission Request Message (ARQ) with the configured E.164 address. If the device is registered with the gatekeeper, the gatekeeper sends an Admission Confirm Message (ACF) with the device's current IP address. Cisco CallManager routes the call directly to this address.

During an inbound call where the H.323 device is the calling party, the gatekeeper routes the call to Cisco CallManager. Cisco CallManager uses the source E.164 address to determine whether the calling device is configured. Cisco CallManager uses the configuration to determine the configuration for that phone. The phone configuration includes regions, locations, MRGL, etc.

Note the following items:

- The system does not support E.164 addressing on H.323 trunks, intercluster trunks and H.323 gateways.
- Cisco CallManager does not resolve the device name when a gatekeeper-controlled H.323 client is configured. Cisco CallManager can access the gatekeeper field for the H.323 client to discover the device. This enables Cisco CallManager to bypass name resolution for the device name.
- Cisco CallManager supports a maximum of one E.164 number per gatekeeper-controlled H.323 client. If the gatekeeper field is populated, you cannot configure a second DN. If an H.323 client is configured for more than one DN, you cannot add the extra gatekeeper information to the database.
- The Gatekeeper routes call by using zone information when there is no zone prefix.

Configuration Notes

Note the following items for configuration purposes:

- You must ensure that gatekeeper is configured in Cisco CallManager before an H.323 client can specify that gatekeeper in its configuration. The Gatekeeper field stays empty by default.
- Ensure that the Gatekeeper field on H.323 client configuration is configured as it is for H.323 trunk.
- Be sure to add the gatekeeper name, technology prefix, zone, and E.164 fields to the H.323 client configuration. You do not need to add Terminal Type. Default specifies the gateway type. If the gatekeeper is not chosen for the gatekeeper field during configuration of each of these fields, these fields cannot populate.
- Gatekeeper, zone, technology prefix fields, and E.164 information display under H.323 Information group on H.323 Client configuration.
- When an H.323 client uses the same gatekeeper, zone and technology prefix as those of another client, consider both clients in the same group. This group represents a single endpoint to the gatekeeper.
- You cannot use the same zone name for the H.323 client and trunk. A zone that a H.323 client uses must differ from the one that an H.323 trunk or a gatekeeper-controlled intercluster trunk uses.
- Ensure the service parameter, Send Product Id and Version ID, is set to True.

If an H.323 client is configured with an E.164 address and a gatekeeper, the database stores this information when the configuration is updated. This information gets loaded at boot time or when the device is reset.

Skinny Client Control Protocol Video

Skinny Client Control Protocol video exhibits the following characteristics:

- If a Skinny Client Control Protocol phone reports video capabilities, Cisco CallManager automatically opens a video channel if the other end supports video.
- For Skinny Client Control Protocol video calls, system administration determines video call bandwidth by using regions. The system does not ask users for bit rate.

Skinny Client Control Protocol Video Bridging

Video conferencing requires a Skinny Client Control Protocol video bridge. Skinny Client Control Protocol video bridging exhibits the following characteristics:

- Skinny Client Control Protocol video bridging requires the same setup as an audio bridge.
- Skinny Client Control Protocol video bridging supports a mix of audio and video in a conference.
- Media resource group lists determine whether an endpoint receives an audio or video bridge. That is, the media resource group list configuration of the user who sets up the conference determines whether the conference is a video conference or an audio-only conference. Refer to the "Media Resource Group List Configuration" section for details of configuring a media resource group list.

SIP Video

Cisco CallManager video supports the SIP protocol, and both SIP trunks and lines support video signaling. SIP supports the H.261, H.263, and H.264 video codecs (it does not support the wideband video codec that is used by the VTA).

The following table lists the type of codes that SIP interfaces support.

Codec	RTP Payload Type
G.711 u-Law	0
GSM	3 (also referred to as GSM Full Rate)
G.723	4
G.711 a-Law	8
G.722	9
G.728	15
G.729	18 (support for combinations of AnnexA and AnnexB)

The Media Termination Point (MTP), which is used for RFC 2833, supports multiple logical channels within a session. A logical channel could be for audio or video. To support video channels, the MTP uses pass-through mode. Video pass-through is enabled if the MTP supports both pass-through and multiple logical channels. Not all MTP devices support multiple logical channels and pass-through mode.

Configuring a SIP Trunk for Video Calls

Perform the following steps to enable video calls on a SIP trunk:

- On the Trunk Configuration window in Cisco CallManager Administration, check the Retry Video Call as Audio check box if you want the call to use audio when the video connection is not available.
- Reset the trunk.

For more information, see the "Additional Configuration for Video Calls" section on page 44-9 and the "Trunk Interaction with H.323 Client" section on page 44-9.

Bandwidth Management

Bandwidth management for video calls gets managed through the call admission control that regions and locations provide in Cisco CallManager Administration.

Regions

Regions in Cisco CallManager allow the bandwidth of video calls to be set. Video call bandwidth, which is the sum of the video bandwidth and the audio bandwidth, does not include overhead.

Refer to the "Region Configuration" section of the *Cisco CallManager Administration Guide* for details of configuring regions in Cisco CallManager.

Locations

Locations in Cisco CallManager Administration include two pools, one pool for video calls and a separate pool for audio calls.

Refer to the "Location Configuration" section of the *Cisco CallManager Administration Guide* for details of configuring locations in Cisco CallManager.

RSVP

RSVP supports SCCP and SIP video calls. The RSVP policy for call admission control is configured by using the Location Configuration window in Cisco CallManager Administration. For more information on the RSVP functionality, see the "Resource Reservation Protocol" section on page 9-1.

Alternate Routing

If an endpoint cannot obtain the bandwidth that it needs for a video call, a video call retries as an audio call for the default behavior. To use route/hunt lists or Automated Alternate Routing (AAR) groups to try different paths for such video calls, uncheck the Retry Video Call as Audio setting in the configuration settings for applicable gateways, trunks, and phones. Refer to the "Route List Configuration" and "Automated Alternate Routing Group Configuration" sections of the *Cisco CallManager Administration Guide* for details.

DSCP Marking

Differentiated Services Code Point (DSCP) packet marking, which is used to specify the class of service for each packet, includes the following characteristics:

- Audio streams in audio-only calls default to EF.
- Video streams and associated audio streams in video calls default to AF41.
- You can change these defaults through the use of a service parameter. The following service parameter settings affect DSCP packet marking:
  - DSCP For Audio Calls (for media [RTP] streams)
  - DSCP For Video Calls (for media [RTP] streams)
  - DSCP for Audio Calls When RSVP Fails
  - DSCP for Video Calls when RSVP Fails
  - DSCP for ICCP Protocol Links

Phone Configuration for Video Calls

The following setting for video-enabled devices affects video calls:

- Retry Video Call as Audio—By default, this check box remains checked. Thus, if an endpoint (phone, gateway, trunk) cannot obtain the bandwidth that it needs for a video call, call control retries the call as an audio call. This setting applies to the destination devices of video calls.
- Video Capabilities Enabled/disabled—This drop-down list box turns video capabilities on and off.

Additional Configuration for Video Calls

The following configuration considerations also affect the ability to make video calls in Cisco CallManager:

- Trunk interaction with the H.323 client
- Call routing considerations
- Resetting gateway timer parameters

Trunk Interaction with H.323 Client

Trunk interaction with the H.323 Client for video calls functions identically to interaction functions for audio calls. Refer to the "Trunks and Gatekeepers in Cisco CallManager" section on page 42-2 in the Understanding Cisco CallManager Trunk Types chapter.

Call Routing for Video Calls

Call routing for video calls functions identically to call routing for audio calls.

Gateway Timer Parameter

For some bonding calls through the H.323/H.320 gateway, the gateway requires a longer time to exchange the H.323 TCS message. If the time required is greater than the timer setting for several Cisco CallManager service parameters, Cisco CallManager will drop the call.

If the default Cisco CallManager gateway timer values appears to be too short, Cisco CallManager drops the call before completion of the call connection. Cisco recommends increasing the following service parameter timers values to avoid call failure.

- H245TCSTimeout=25
- Media Exchange Interface CapabilityTimer=25
- Media Exchange Timer=25

Conference Control for Video Conferencing

Cisco CallManager supports the following conference controls capabilities:

- Roster/Attendee List
- Drop Participant
- Terminate Conference
- Show Conference Chairperson/Controller
- Continuous Presence

Cisco CallManager also supports the following video conference capabilities for Skinny Client Control Protocol phones:

- Display controls for video conferences. The Skinny Client Control Protocol phones can choose to use the continuous presence or voice-activated mode to view the video conference. When a mode is chosen, a message gets sent to the bridge to indicate which mode to use on the video channel. Switching between modes does not require renegotiation of media.

- Display participant information such as the user name in the video stream. The system can use the participant information for other conferencing features such as roster.

Video Telephony and Cisco Serviceability

Cisco Serviceability tracks video calls and conferences by updating performance monitoring counters, video bridge counters, and call detail records (CDRs).

Performance Monitoring Counters

Video telephony events cause updates to the following Cisco CallManager Serviceability performance monitoring counters:

- Cisco CallManager
  - VideoCallsActive
  - VideoCallsCompleted
  - VideoOutOfResources
- Cisco H.323
  - VideoCallsActive
  - VideoCallsCompleted
- Cisco Locations
  - VideoBandwidthAvailable
  - VideoBandwidthMaximum
  - VideoOutOfResources
  - VideoCurrentAvailableBandwidth
- Cisco Gatekeeper
  - VideoOutOfResources
- Cisco SIP
  - VideoCallsCompleted
  - VideoCallsActive

Refer to the *Cisco CallManager Serviceability System Guide* and *Cisco CallManager Serviceability Administration Guide* for details.

Video Bridge Counters

Video conference events cause updates to these Cisco video conference bridge performance monitoring counters:

- ConferencesActive
- ConferencesAvailable
- ConferencesCompleted
- ConferencesTotal

- OutOfConferences
- OutOfResources
- ResourceActive
- ResourceAvailable
- ResourceTotal

These counters also display in the Cisco CallManager object with the VCB prefix.

Refer to the *Cisco CallManager Serviceability System Guide* and *Cisco CallManager Serviceability Administration Guide* for details

Call Detail Records

Video telephony events cause updates to Call Detail Records (CDRs) in Cisco CallManager Serviceability. These CDRs include the following information:

- IP address and port for video channels
- Codec: H.261, H.263, H.264, Cisco VT Camera wideband video
- Call bandwidth
- Resolution: QCIF, CIF, SQCIF, 4CIF, 16CIF, or Custom Picture Format

Cisco CallManager also stores CDRs for mid-call video and supports the following call scenarios:

- Skinny Client Control Protocol to Skinny Client Control Protocol calls
- Skinny Client Control Protocol to Skinny Client Control Protocol calls across an intercluster trunk (ICT)

Note CDR is added when video is added mid-call, but CDR entry is not removed as part of mid-call video removal (for example, Cisco Video Telephony Advantage gets turned off).

Refer to the *Cisco CallManager Serviceability System Guide* and *Cisco CallManager Serviceability Administration Guide* for details.

Video Telephony Configuration Checklist

Table 44-1 provides a checklist to configure video telephony in Cisco CallManager Administration.

Table 44-1 Video Telephony Configuration Checklist

Configuration Steps		Related procedures and topics
Step 1	If you use regions for call admission control, configure regions for video call bandwidth. **Note** All devices have a default region, which defaults to 384 kbps for video.	Region Configuration, *Cisco CallManager Administration Guide* Call Admission Control, *Cisco CallManager System Guide*
Step 2	If you use locations for call admission control, configure locations for video call bandwidth.	Configuring a Location, *Cisco CallManager Administration Guide* Call Admission Control, *Cisco CallManager System Guide*
Step 3	If using RSVP for bandwidth management of SIP video calls, configure the RSVP service parameters, or set the RSVP policy in the Location Configuration window.	Configuring a Location, *Cisco CallManager Administration Guide* Configuring Service Parameters for a Service on a Server, *Cisco CallManager Administration Guide*
Step 4	To use a Cisco video conference bridge, configure the appropriate conference bridge for your network.	Conference Bridge Configuration, *Cisco CallManager Administration Guide*
Step 5	To configure a user to use the video conference bridge instead of using other conference bridges, configure the user’s media resource groups and media resource group lists accordingly.	Media Resource Group Configuration, *Cisco CallManager Administration Guide* Media Resource Group List Configuration, *Cisco CallManager Administration Guide*
Step 6	Configure the H.323 gateways in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect.	Gateway Configuration, *Cisco CallManager Administration Guide* Automated Alternate Routing Group Configuration, *Cisco CallManager Administration Guide* Route List Configuration, *Cisco CallManager Administration Guide*

Table 44-1 Video Telephony Configuration Checklist (continued)

Configuration Steps		Related procedures and topics
Step 7	Configure the H.323 phones in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect. Choose Enabled for Video Capabilities.	Cisco IP Phone Configuration, *Cisco CallManager Administration Guide* Automated Alternate Routing Group Configuration, *Cisco CallManager Administration Guide* Route List Configuration, *Cisco CallManager Administration Guide*
Step 8	Configure the H.323 trunks in your system to retry video calls as audio calls (default behavior) or configure AAR groups and route/hunt lists to use alternate routing for video calls that do not connect.	Trunk Configuration, *Cisco CallManager Administration Guide* Automated Alternate Routing Group Configuration, *Cisco CallManager Administration Guide* Route List Configuration, *Cisco CallManager Administration Guide*

Where to Find More Information

Related Topics

- Call Admission Control, *Cisco CallManager System Guide*
- Region Configuration, *Cisco CallManager Administration Guide*
- Location Configuration, *Cisco CallManager Administration Guide*
- Conference Bridge Configuration, *Cisco CallManager Administration Guide*
- Media Resource Group Configuration, *Cisco CallManager Administration Guide*
- Media Resource Group List Configuration, *Cisco CallManager Administration Guide*
- Automated Alternate Routing Group Configuration, *Cisco CallManager Administration Guide*
- Route List Configuration, *Cisco CallManager Administration Guide*
- Gateway Configuration, *Cisco CallManager Administration Guide*
- Cisco IP Phone Configuration, *Cisco CallManager Administration Guide*
- Trunk Configuration, *Cisco CallManager Administration Guide*

Additional Cisco Documentation

- Cisco IP Phone administration documentation and release notes (all models)
- Cisco IP Phone user documentation and release notes (all models)
- *Cisco CallManager Serviceability System Guide*
- *Cisco CallManager Serviceability Administration Guide*
- *Cisco IP/VC 3511 MCU and Cisco IP/VC 3540 MCU Module Administrator Guide*