CISCO UNIFIED COMMUNICATIONS MANAGER v10 UPDATED TOPICS

LOCAL ROUTE GROUP ENHANCEMENT

Multiple Local Route Groups can be associated with Route Groups for Emergency Dialing. In releases 8 and 9, administrators could only have one Local Route Group per Device Pool.

Result:

- ✓ Multiple Local Route Groups can be associated with Route Groups.
- ✓ This simplifies the dial plan, reducing the number of Route Lists needed

SELF-PROVISIONING FOR END USER PHONES

- The Self-Provisioning feature allows an end user or administrator to add an unprovisioned phone to a Cisco Unified Communications Manager system with minimal administrative effort.
- In prior releases, TAPS allowed you to import Template Phones into the system and have the user provision any phone they receive as their own simply by plugging in the phone and having the user dial a special number and enter their extension. TAPS required a Contact Center Server environment in order to use TAPS, making its use much less common.

Result:

- ✓ In Cisco Unified Communications Manager 10.x, a new but similar tool called Self-Provisioning and Self-Care has been released to make the lives of the administrators easier.
- ✓ A phone can be added by plugging it into the network and following a few prompts to identify the user.

PRIME COLLABORATION PROVISIONING STANDARD

- Cisco Prime Collaboration Provisioning Standard is a tool that attempts to unify administration of UC environment to one interface. In version 10.x you can administer Cisco Unified Communications Manager, Unified Communications Manager Express, Unity, Unity Connection and Cisco Unity Express.
- The standard version is included with every install of Cisco Unified Communications Manager 10.x

Result:

- ✓ This solution provides a single interface for a single cluster for call control, messaging, presence, and video
- ✓ Administrator audit log and order tracking promote accountability, network security, and operational control, and they facilitate troubleshooting
- ✓ Lightweight Directory Access Protocol (LDAP) integration for user changes helps ensure consistency among systems
- ✓ Batch provisioning (single cluster only) increases operating efficiencies and reduces costs with scheduled scripts and templates that let an operator rapidly and consistently add or modify a large number of users or a large number of endpoints or device profiles (and the corresponding configuration settings)

INCOMING CALLED PARTY TRANSFORMS FOR SIP TRUNKS AND MGCP GATEWAYS CONNECTED PARTY TRANSFORMS FOR H.323 TRUNKS/GWS AND MGCP GWS

Incoming called party transformations enable normalization of incoming called party information prior to actually routing the call.

Result:

- ✓ Each numbering type can be configured with a calling search space used to apply called party transformation patterns to the calls. The calling search space should contain partitions containing called party transformation patterns exclusively.
- ✓ This allows the modifications applied to the called party number to be based on the structure of the called party number rather than strictly on its numbering type.
- ✓ The called party transformation patterns use regular expressions to match the called party number.
- ✓ The best-match process is used to choose between multiple matches, and the selected pattern's called party transformations are applied to the call.

URGENT PRIORITY FOR DN'S

- In prior Cisco Unified Communications Manager releases, the 15 second T302 interdigit timer would be used because there is a longer pattern that is “possibly” matched.
- Urgent Priority could only be used on Translation Patterns, Route Patterns, or Hunt Groups.

Result:

- ✓ With the 10.x release, the individual line extensions (DNs) can be marked as Urgent.
- ✓ This allows the DNs to ring immediately, even though there is a longer “possible” match available in the Route Plan Report.
- ✓ Urgent Priority can be marked on individual line extensions - DN’s.
- ✓ When using a global +164 dialing plan, users would have to wait 15 seconds when dialing another internal extension. With this this feature enabled, that wait is eliminated.

NETWORK BASED CALL RECORDING

Network based recording is available in Cisco Unified Communications Manager 10.x. Network based recording allows you to use the gateway to record calls.

Result:

- ✓ Allows Cisco Unified Communications Manager to route recording calls, regardless of device, location, or geography.
- ✓ Centralizes recording policy control.
- ✓ Captures calls that are extended off the enterprise network to mobile and home office phones.
- ✓ Cisco Unified Communications Manager selects dynamically the right media source based on the call flow and call participants.
- ✓ Enhanced SIP Header and CTI metadata enables applications to track recorded calls in both single and multi-cluster environments.
- ✓ Recording Serviceability counters and alarms help compliance officers ensure that calls are recorded by monitoring the real-time status and historical performance of the feature.

CISCO IM AND PRESENCE (FORMALLY CUPS) PART OF CISCO UNIFIED COMMUNICATIONS MANAGER 10.X CLUSTER

Cisco IM and Presence server is now managed as part of the cluster through Cisco Unified Communications Manager 10.x. It is still install as a separate server, however it must be configured at the Unified Communications Manager Publisher before install, just like all Unified Communications Manager Subscribers.

Result:

- ✓ Unified administration of IM and Presence and voice and video call-control users reduces time and effort to add, change, and verify user configuration.
- ✓ Common administration of IM and Presence and other nodes within a cluster simplifies installation, node configuration, and backup and restore.
- ✓ Integrated serviceability allows easy navigation and supports a single client for the Real-Time Monitoring Tool (RTMT).
- ✓ A common portal makes it easy for end users to manage their options and preferences for IM and Presence together with voice and video.

TRANSLATION PATTERN CAN NOW USE ORIGINATOR'S CALLING SEARCH SPACE

Cisco Unified Communications Manager always used the translation pattern’s calling search space to decide how to route the calls that used the pattern. In 10.x the administrator can chose to use originator's calling search space instead of the translations pattern CSS.

Result:

- ✓ This feature gives the admin more granularities when creating the route plan. Since the translation pattern is used for modifying the dialed digits, the originating devices CSS is lost and not taken into consideration after digit modification is performed at the translation pattern.
- ✓ This behavior can lead to users receiving elevated calling privileges. To prevent this, more complex configuration was needed.
- ✓ With this new feature, administrators can make sure that an original caller’s calling privileges are maintained after digit manipulation.

AUDIT LOG VIEWER

Cisco RTMT has the new setting Audit Log Viewer that displays historical record of configurations performed by users.

Result:

- ✓ Administrators can display the following messages in AuditLog Viewer:
- ✓ AuditApp Logs - To monitor actions of one or more users.
- ✓ Vos Logs - To monitor activities of a specific terminal, port, or network address of the system.

SELF-CARE PORTAL

Replaces the Cisco Unified CM User Options interface

Result:

- ✓ From the Cisco Unified Communications Self Care Portal, users can customize and control phone features and settings for their user account, their phone, and Jabber.

VMWARE ONLY SUPPORTED DEPLOYMENT

Cisco will no longer offer their Cisco Unified Communications Manager servers on MCU 7800 physical servers. Now all deployments must be in the virtual format.

Result:

- ✓ Administrators will install and maintain virtual Unified Communications servers using Cisco Unified Communications System (UCS).

SINGLE STEP PROCESS TO CHANGE BOTH HOSTNAME AND IP

- Historically, changing a server hostname involved a series of steps that required the use of the CLI and web graphical user interface (GUI). It also required the user to reboot single servers and whole clusters after the event.
- Cisco Unified Communications Manager has been updated with a simplified procedure for updating the IP address or hostname of a Unified Communications Manager server.

Result:

- ✓ The IP address and hostname of a Unified Communications Manager publisher or subscriber node can be updated from Cisco Unified Operating System Administration, or from the Command Line Interface
- ✓ All the existing steps in the procedure "Changing the IP Address and Hostname for Cisco Unified Communications Manager" are automated into a single step using the CLI or GUI.

CONFIGURABLE RTP AND SRTP PORT RANGES

The Configurable RTP and SRTP port ranges feature allows you to configure RTP and Secure RTP port ranges used by the software based conference bridges, Media Termination Points (MTP), Music on Hold (MoH), Annunciator media resources, and SIP endpoints.

Result:

- ✓ Two new service parameters—Start Media Port and Number of Ports have been added to configure RTP port ranges of IPVMS devices. The configurable port range is 9051-61000. The Start Media Port and Stop Media Port fields in the SIP profile are used to configure RTP port ranges for SIP end points. The configurable port range is 2048-65535.

Configurable Set of Nonpreemptable Numbers

Cisco Unified Communications Manager 10.x allows you to configure a list of destinations that cannot be pre-empted so that the calls on these destinations are not disconnected even if a higher precedence call is attempted.

Result:

- ✓ You can use this feature for calls to emergency services so that the emergency calls are not disconnected.
- ✓ An MLPP Preemption Disabled check box has been added to the Calling Party Transformation Pattern settings.
- ✓ You must check this check box to make the numbers in a transformation pattern nonpreemptable.

COMMERCIAL COST AVOIDANCE

If a caller within a Cisco Unified Communications Manager network calls a called party on an external number, Unified Communications Manager checks if an internal number exists for the called party in the LDAP database.

Result:

- ✓ If an internal number exists, the call is routed to the internal number. If the internal number is not found in the LDAP database, the call is routed to the original (external) number. To route the calls to the internal numbers, you must configure directory number alias for both the lookup and the sync servers. You must configure the LDAP server for Directory Number Alias Sync (sync server) to synchronize users from Unified Communications Manager database to the sync server. Finally, you must configure the LDAP server for Directory Number Alias Lookup (lookup server) to route the commercial calls to an alternate number.

MULTIPLE CODECS IN SDP ANSWER

This option applies when incoming SIP signals do not indicate support for multiple codec negotiation and Cisco Unified Communications Manager can finalize the negotiated codec.

Result:

- ✓ When this check box is checked, the endpoint behind the trunk is capable of handling multiple codecs in the answer SDP. For example, an endpoint that supports multiple codec negotiation calls the SIP trunk and Cisco Unified Communications Manager sends a Delay Offer request to a trunk. The endpoint behind the trunk returns all support codecs without the Contact header to indicate the support of multiple codec negotiation.
- ✓ In this case, Cisco Unified Communications Manager identifies the trunk as capable of multiple codec
- ✓ negotiation and sends SIP response messages back to both endpoints with multiple common codecs.

CISCO USER DATA SERVICES

Cisco User Data Services provides Cisco Unified IP Phones with the ability to access user data from the Cisco Unified Communications Manager database.

Result:

- ✓ With Release 10.x, Cisco Personal Directory (CCMPD) and Cisco CallManager Cisco IP Phone (CCMCIP) services are supported through Cisco User Data Services. If you disable Cisco User Data Services, the Directory button on Cisco Unified IP Phones will be disabled and your phones will not be able to use Cisco IP Phone services.

VIDEO ON HOLD

The Video on Hold feature is for video contact centers where customers that place a call are able to watch a specific video after initial consultation with the contact center agent. In this case, the agent selects the video stream that is played to the customer while the customer is on hold. In addition to the video contact center Video on Hold can be deployed within any enterprise if the deployment requires a generic video on hold capability.

Result:

- ✓ Cisco Unified Communications Manager now has a new configuration "Video on Hold Server" that allows a media content server to be provisioned under the existing "Media Resources" menu. The media content server can stream audio and video content when directed by Unified Communications Manager. The media content server is an external device that can store and stream audio and video content under Unified Communications Manager control using SIP as the signal protocol. The media content server is capable of providing hi-definition video content at 1080p, 720p, or lower resolutions such as 360p. In addition to the video contact center, Video on Hold can be deployed within any enterprise if the deployment requires a generic video on hold capability. Cisco MediaSense is used as the media content server.