



CHAPTER 7

Overview of Cisco Unified Communications Call Routing

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Once the network infrastructure has been put in place for your Cisco Unified Communications System, call routing applications, components, and services can be layered on top of this infrastructure. There are numerous applications and features that can, and in some cases must, be deployed on the network infrastructure. In general, you should deploy the following call routing components, features, and services:

- Call processing agent — Provides telephony services and call routing capabilities.
- Dial plan — Provides endpoint numbering, dialed digits analysis, and classes of restriction to limit types of calls that a user can make.
- Call admission control — Provides mechanisms for preventing oversubscription of network bandwidth by limiting the number of calls that are allowed on the network at a given time based on overall call capacity of the call processing components and network bandwidth.
- Video telephony services — Provide the ability to provision and register video endpoints as well as to set up, route, and maintain video calls on the network.
- PSTN gateways and provider voice and data services — Provide access to voice and data networks outside the enterprise, including the PSTN, Internet, and service provider IP-based trunks.
- Remote site survivability — Provides continuation of basic telephony services at remote sites when the central-site telephony services are unavailable due to failed or flapping network connectivity.

The chapters in this part of the SRND cover the features, components, and services mentioned above. Each chapter provides an introduction to the component or service, followed by discussions surrounding architecture, high availability, capacity planning, and design considerations. The chapters focus on design-related aspects of the applications and services rather than product-specific support and configuration information, which is covered in the related product documentation.

This part of the SRND includes the following chapters:

- [Call Processing, page 8-1](#)

This chapter examines the various types of call processing applications and platforms that facilitates IP telephony call routing. The chapter examines the call processing architecture, including hardware options, Unified CM clustering capabilities, high availability considerations for call processing, and capacity planning.

- [Dial Plan, page 9-1](#)

This chapter explores dial plan features and functions that enable the call processing application to route calls to appropriate numbers. The chapter considers various aspects of dial plan services, including dial plan constructs, dial plan numbering options and design considerations, classes of restriction, inbound and outbound calling features, and dial plan and call routing redundancy mechanisms.

- [Emergency Services, page 10-1](#)

This chapter discusses accessing emergency services through Public Safety Answering Points (PSAPs) on the PSTN from within the enterprise IP telephony environments, an important aspect of most deployments due to possible critical needs for medical, fire, and other emergency response services. The chapter provides an overview of the various emergency service components both inside and outside the enterprise. It also discusses planning, 911 network service providers, gateway interfaces, and number-to-location mapping.

- [Call Admission Control, page 11-1](#)

This chapter examines the potential for oversubscribing IP links, which causes the voice quality for phone calls to become unacceptable. It also examines the use of call admission control to allow only a certain number of simultaneous calls on the network at a given time to prevent oversubscription. This chapter covers call admission control types, including location-based call admission control and RSVP, as well as design and deployment guidelines for successfully deploying admission control services.

- [IP Video Telephony, page 12-1](#)

This chapter covers video telephony, an important and integral part of collaborative communication. The chapter discusses video telephony components, protocols and codecs, multipoint conferencing, and gatekeeper aspects for video call routing.

- [Gateways, page 13-1](#)

This chapter explores voice and IP gateways, which are critical components of Unified Communications deployments because they provide the path for connecting to phones on the public telephone network. This chapter looks at gateway traffic types and patterns, protocols, capacity planning, and platform selection, as well as fax and modem support.

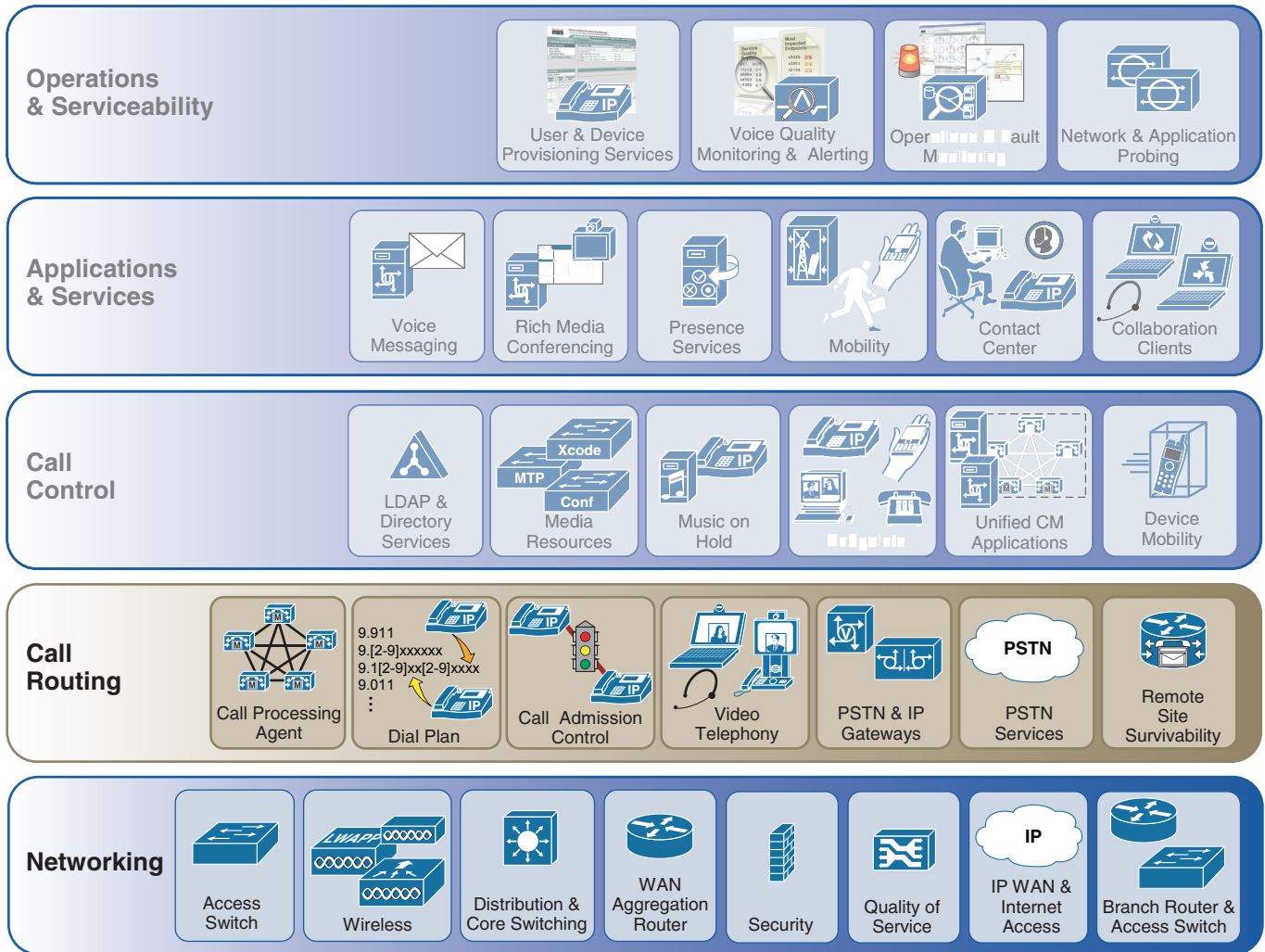
- [Cisco Unified CM Trunks, page 14-1](#)

This chapter covers both intercluster and provider trunks, which provide the ability to route voice calls over IP and leverage various Unified Communications features and functions. This chapter discusses H.323 and SIP trunks, codecs, and supplementary services over these trunks, as well as sizing of trunks to accommodate network call load.

Architecture

Just as with other network and application technology systems, Unified Communications call routing components and services must be layered on top of the underlying network infrastructures. Figure 7-1 shows the logical location of call routing applications and services in the overall Cisco Unified Communications System architecture.

Figure 7-1 Cisco Unified Communications Call Routing Architecture



Unified Communications call routing components and services such as call processing agents and IP and PSTN gateways rely on the underlying network infrastructure for network connectivity and access. By connecting to the underlying network infrastructure, call routing components and features are able to leverage end-to-end network connectivity and quality of service to access both the enterprise and public telephone networks. In turn, call routing applications and services provide basic Unified Communications functions such as call control, dial plan, call admission control, and gateway services to other applications and services in the deployment. For example, a Unified CM cluster connects to the IP network through a switch in order to communicate with other devices and applications within the

network as well as to access other devices and services in other locations. At the same time, the Unified CM cluster provides services such as phone registration and media resource provisioning and allocation to call control components and services such as IP phones.

Further, just as call routing components rely on the network infrastructure for network connectivity, call routing components and services are also often dependent upon each other for full functionality. For example, while Unified CM provides registration and call routing services to various IP endpoints within the network, it is completely dependent upon gateways and gateway services to route calls beyond the enterprise.

High Availability

As with the network infrastructure, critical Unified Communications call routing services should be made highly available to ensure that required features and functionality remain available if failures occur in the network or with individual call routing components. It is important to understand the various types of failures that can occur and the design considerations around those failures. In some cases, the failure of a single server or component (for example, a subscriber node in a Unified CM cluster) might have little or no impact due to the redundant nature of the Unified CM clustering mechanism. However, in other cases a single failure can impact multiple components or services. For example, the failure of a PSTN or IP gateway could result in loss of access to the public telephone network, and even though a call processing agent such as Unified CM is still available and able to provide most features and services, it cannot route calls to the PSTN because there is no path available if the gateway fails. To avoid these types of situations, you should deploy multiple PSTN gateways to provide redundant gateway services, and you should configure the call processing agent to handle call routing to both gateways as needed.

For features and services such as dial plan and call admission control, high availability considerations include temporary loss of functionality due to network connectivity or call processing agent application server failures, resulting in the inability of the call agent to route calls and therefore the inability for callers to make calls. Oversubscription of the network could also occur if call admission control services are not available to the endpoints initiating a call. For example, if RSVP call admission control is in use and an RSVP agent fails or loses connectivity to the network, the call may still go through but without the call admission control service being aware of the call, thus potentially resulting in poor quality. To avoid these types of scenarios, provide call admission control resiliency by deploying multiple RSVP agents so that a failed RSVP agent will not prevent another RSVP agent from providing the call admission control service.

High availability considerations are also a concern for components and services such as video endpoints and remote site survivability. For deployments with network-attached remote sites where devices are leveraging call processing services from an agent in a central site, remote site survivability using SRST, for example, can ensure that local phones within the remote site will still receive call processing services in the event of a connectivity failure to the central site. Likewise, to ensure that video endpoints are highly available, you can deploy more than one multipoint control unit (MCU) in case one fails.

Capacity Planning

The network infrastructures must be designed and deployed with consideration for the capacity and scalability of the individual components and the overall system. Similarly, deployments of call routing components and services must also be designed with attention to capacity and scalability considerations. When deploying various call routing applications and services, not only is it important to consider the scalability of the applications and services themselves, but you must also consider the scalability of the underlying network infrastructure. Certainly the network infrastructure must have available bandwidth and be capable of handling the additional traffic load that the call routing components will create. Similarly, the call routing infrastructure and its components must be capable of handling all the required device configurations and registrations as well as the call load or busy hour call attempts (BHCA),

For example, with call processing agents such as Unified CM, it is critical to assess the size of the deployment in terms of number of users, endpoints, and calls per user per hour, and to deploy sufficient resources to handle the required load. If a call processing agent is undersized and does not have sufficient resources, features and services will begin to fail as the load increases. Two of the chief considerations when attempting to size a call processing deployment are the call processing type and the call processing hardware. Both of these are critical for sizing the system appropriately given the number of users, locations, devices, and so on. As an example, Cisco Unified Communications Manager has a much higher capacity than Cisco Unified Communications Manager Express and should therefore be used for larger deployments. In addition, the server platform selected to run the call processing agent will, in many cases, determine the maximum load.

Capacity planning for remote site survivability is much the same in that it relies on backup call processing hardware. Selecting the appropriate Cisco IOS platform to provide backup or survivable call processing services typically begins with determining the number of devices or users that must be supported at that site in the event that connectivity to the central site is disrupted. Equally critical in this sizing exercise is the local PSTN gateway services. In the event of a central site connection failure, will the local PSTN gateway have sufficient circuits to be able to route all calls without blocking during the busiest hour? If the answer is no, adding additional gateways or trunks will be necessary to appropriately size the remote site for backup call processing.

PSTN and IP gateways must also be sized appropriately for a deployment, so that sufficient capacity is available to handle all calls in the busiest hour. In some cases, you might have to deploy multiple PSTN or IP gateways to provide enough resources.

When sizing call admission control, ensure that sufficient bandwidth is available over network connections to support the required number of calls. If sufficient bandwidth is not available, additional network capacity, gateways, and IP or telephony trunks may be required.

Sizing dial plan services is also important. However, in most cases dial plan capacity in terms of the number of endpoints or phone numbers, route patterns, or other dial plan constructs, is completely dependent upon the type of call processing agent and platform used.

For components and services such as video telephony, appropriate sizing is just as critical. Capacity planning considerations for video telephony center mainly on network bandwidth, available video ports, and MCU sessions. In most cases additional capacity can be added by increasing the number of application servers and MCUs or by upgrading server or MCU hardware with higher-scale models, assuming the underlying network infrastructure is capable of handling the additional load.

For a complete discussion of system sizing, capacity planning, and deployment considerations related to sizing, refer to the chapter on [Unified Communications Design and Deployment Sizing Considerations, page 29-1](#).

