CHAPTER 1

Overview

This document describes the tools available to troubleshoot potential problems you may have with Cisco CallManager. Also, this document describes call traces and debug outputs and how to use them to diagnose problems with call flow and call events.

This chapter contains the following topics:

Understanding Cisco IP Telephony Topology

It is very important to have an accurate topology, access to all the network devices, and terminal services for management of the Cisco CallManager.

Adding IP telephony to a new or existing network requires significant planning to ensure success. Since real-time voice traffic has different requirements than data traffic, the network must be designed with low latency and Quality-of-Service (QoS) in mind. As with any network that carries mission-critical traffic, it is imperative that the network administrator maintains accurate, detailed diagrams of the network topology. In a crisis situation, it is important to know not just the broad overview of the network, but also which ports are connected to network components, such as routers, switches, Cisco CallManager servers, gateways, and other critical devices. Plan the network with redundancy and scalability in mind.

Caution Cisco does not support using hubs for shared connectivity to the switches as they can interfere with correct operation of the IP telephony system.

Topology Checklist

Use the following checklist to be sure you have properly documented your network topology. You will need:

- A topology diagram that shows all network devices and critical components with the port or interface numbers to which they are attached, and to which VLAN (if applicable) they belong. Special designations should be used for ports that are configured in trunking or channeling mode.
- The root of the spanning-tree should be configured and all normally blocking ports should be identified.
- Any WAN circuits should be identified with the amount of bandwidth (or CIR, in the case of frame-relay) they carry.

Any WAN interface will require special consideration, since this is a potential source of congestion. Cisco IP Phones and gateways set the RTP stream IP precedence field to five; however, this only tags the RTP packet. It is up to the network administrator to ensure that the network is configured for prioritization and call admission control so that the Voice over IP (VoIP) traffic can be serviced with minimal delay and contention for resources.

For additional information on this topic, see:

http://www.cisco.com/warp/public/793/voip/