CISCO SYSTEMS

White Paper

# H.323 and SIP Integration

## Introduction

Service providers are deploying packet telephony technologies as an alternative to traditional circuit-switched telephone networks in order to shift traditional voice services to packet- based networks and to create new services that combine data, voice, and video information. The lower cost associated with converged data and voice networks is a prime driver for deployment of packet telephony.

Cisco offers packet voice solutions that offer architectural and protocol flexibility, including the choice of H.323 [International Telecommunications Union (ITU)—T Recommendation H.323], Media Gateway Control Protocol (MGCP), and Session Initiation Protocol (SIP) for signalling and call control. In particular; H.323 and SIP are often compared and contrasted with each other. MGCP is not being covered because the focus of this document is how to integrate SIP into existing H.323 networks. The premise is that most existing packet telephony customers are currently running H.323, and we are explaining a means of incorporating SIP, to enhance their networks.

Each protocol provides its own set of advantages and disadvantages within a packet voice network. It is possible to use both protocols within the same network, and it is definitely necessary to interconnect networks using one or the other. The H.323 protocol has been available for several years, and carriers have made a significant investment to build out many large, H.323-based networks.

SIP is growing in popularity due to its ability to easily combine voice and Internet- based services. SIP interoperability and coexistence with H.323 is very important to maximize the return on current investments and to support new deployments that might use SIP as an alternative packet telephony signaling protocol.

This white paper will address current techniques available to address issues regarding the interoperability and coexistence of H.323 and SIP in packet telephony networks.

## Overview

While SIP is growing in popularity due to its use of and similarity to Internet technologies such as HTTP, H.323 is more widely deployed and the standard is more mature. One of the key strategic advantages of Cisco packet telephony solutions is the ability to originate and terminate H.323 and SIP calls on the same Cisco voice gateway on a dial-peer basis.

The Cisco packet voice architecture, based on Cisco IOS® Software, is the foundation for Cisco packet telephony protocol development. The Cisco architecture allows for protocol flexibility and enables, on a call-by-call basis, use of a particular session protocol. This flexibility allows customers to deploy SIP networks on proven packet telephony infrastructures, while still maintaining core H.323 functionality within their networks. With the ability to support the connection of customers and carriers using either protocol, service providers can offer a variety of application hosting and sharing services, and be more aggressive in pursuing wholesale opportunities via new services.

Some principles for coexistence that are critical for successful multiprotocol deployments are transport capabilities across time-division multiplexing (TDM) interfaces, dual tone multifrequency (DTMF) processing capabilities and fax relay support. In deployments where both protocols are used, it is important that there are no performance limitations related to the call mix between SIP and H.323 calls, and that there is no significant deviation in calls-per-second measurements compared to a homogeneous SIP or H.323 network.

Cisco gateways provide support for coexistence of SIP and H.323 calls beginning with Cisco IOS Software Release 12.2(2)XB. Figure 1 illustrates packet voice architectures for wholesale call transport and Figure 2 illustrates termination services for application service providers (ASPs) where SIP and H.323 are used simultaneously for signaling.

Figure 1
SIP and H.323 Wholesale Call Transport

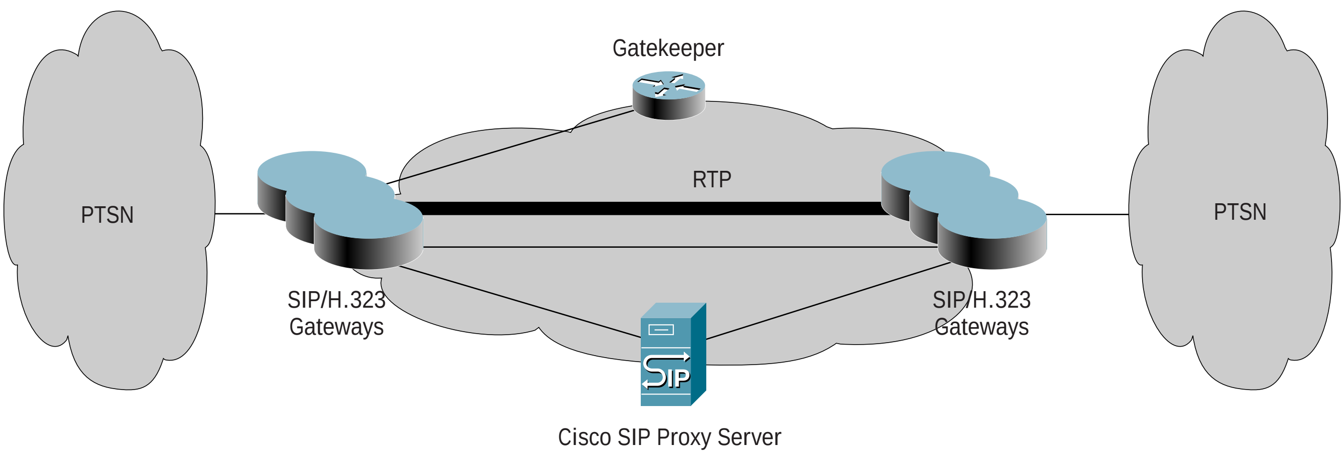
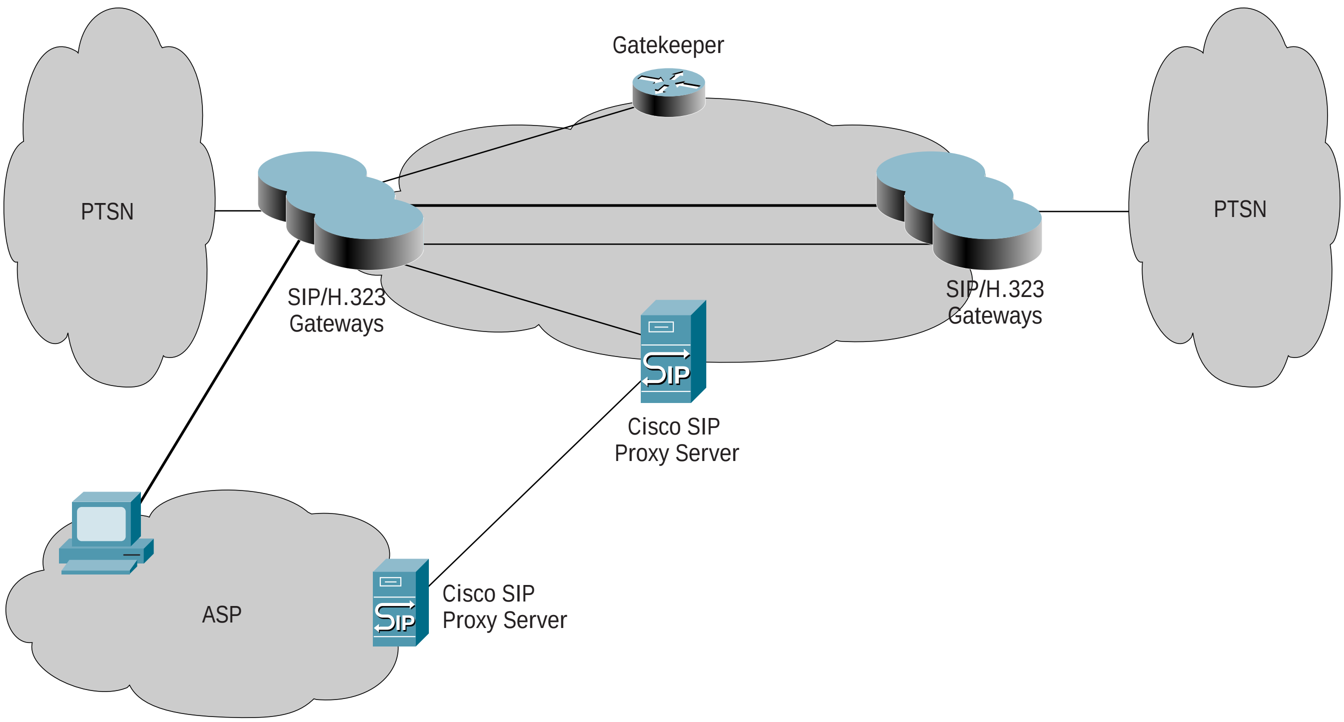


Figure 2

SIP-Based Termination Services for ASPs and H.323-Based Wholesale Call Transport




Calls from an H.323 gateway are routed through an H.323 network using an H.323 gatekeeper. Correspondingly, SIP gateways use a SIP redirect or Cisco SIP Proxy Server for call routing.

One of the key advantages of H.323-based networks is an ability to manage available resources for call routing via H.323 gatekeepers. The gatekeeper is the logical "switch" of the H.323 network, providing several basic services to all endpoints in its zone.

Services include address translation (alias name/number-to-network address), endpoint admission control (based on bandwidth availability, concurrent call limitations, or registration privileges), bandwidth management, and zone management (the routing of calls originating or terminating in the gatekeeper zone, including multiple path reroute). Gateways coordinate calls by communicating with gatekeepers using the Registration, Admission, and Status (RAS) protocol.

Within a SIP network, an "infrastructure" proxy server can control call routing. The proxy server performs functions such as registration, authentication, authorization, network access control, and network security. It also finds the next-hop routing information based on received or translated destination URLs or E.164 addresses.

While support of the two protocols on a single gateway is critical, another integral part of dual-protocol deployment is the ability for H.323 gatekeepers and SIP proxies to interwork and share routing capabilities. One method that was introduced to support time-to-market requirements uses routing interaction between a Cisco SIP Proxy Server and an H.323 gatekeeper.

The business model for some carriers using the Cisco Global Long Distance Solution is to provide origination and termination of voice-over-IP (VoIP) minutes for several other service providers. This business model has been very successful with deployment of H.323-based services, but these Cisco customers would also like to attract additional SIP-based service providers. Ideally, these customers would like to use their existing voice-gateway infrastructure to support additional SIP-based offerings.

Cisco has provided these carriers with a way to add new SIP services by adding capabilities to the Cisco SIP Proxy Server to allow it to "handshake" with an H.323 gatekeeper using the H.323 RAS protocol.

By enabling a SIP proxy server to communicate with an H.323 gatekeeper using RAS location request, location confirmation, and location reject messages and responses, a Cisco SIP Proxy Server can obtain optimized routing information from VoIP gateways that have been deployed in the service provider's network.

While the Cisco SIP Proxy Server could supply routing information to SIP gateways, this scheme allows a packet voice carrier to not only use its existing routing structure on its H.323 gatekeepers, but to also take advantage of the H.323 RAI functionality for a more efficient network. The Cisco SIP Proxy Server actually acts like another gatekeeper to the H.323 network.

This optimized routing structure provides shorter post-dial delay and more efficient usage of gateway resources. It must be stressed that the SIP-proxy-to-gatekeeper communication is used only for call routing and not for any type of protocol translation.

This type of communication between SIP- and H.323-based components is also only used for call signaling. Real-time Transport Protocol [RTP]-based media information (such as packetized voice) will still only be communicated directly between SIP endpoints.

Figure 3 illustrates a typical configuration where a Cisco SIP Proxy Server interfaces with an H.323 network, and Figure 4 demonstrates a generic call flow for calling from a SIP phone to a SIP/H.323 gateway using call signaling between a Cisco SIP Proxy Server and a Cisco H.323 gatekeeper.

Figure 3

A Cisco SIP Proxy Server Uses the H.225 RAS Protocol to Interface with a Cisco H.323 Gatekeeper to Provide SIP-Based Services

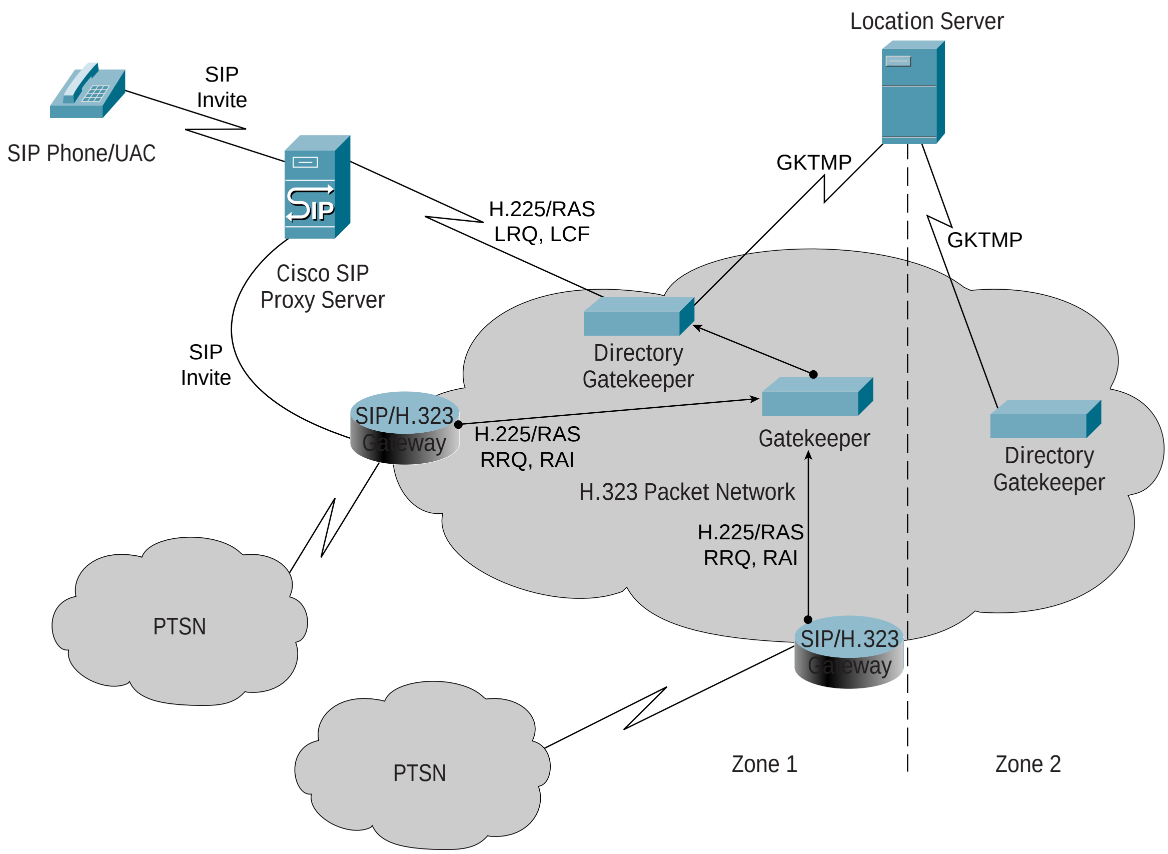
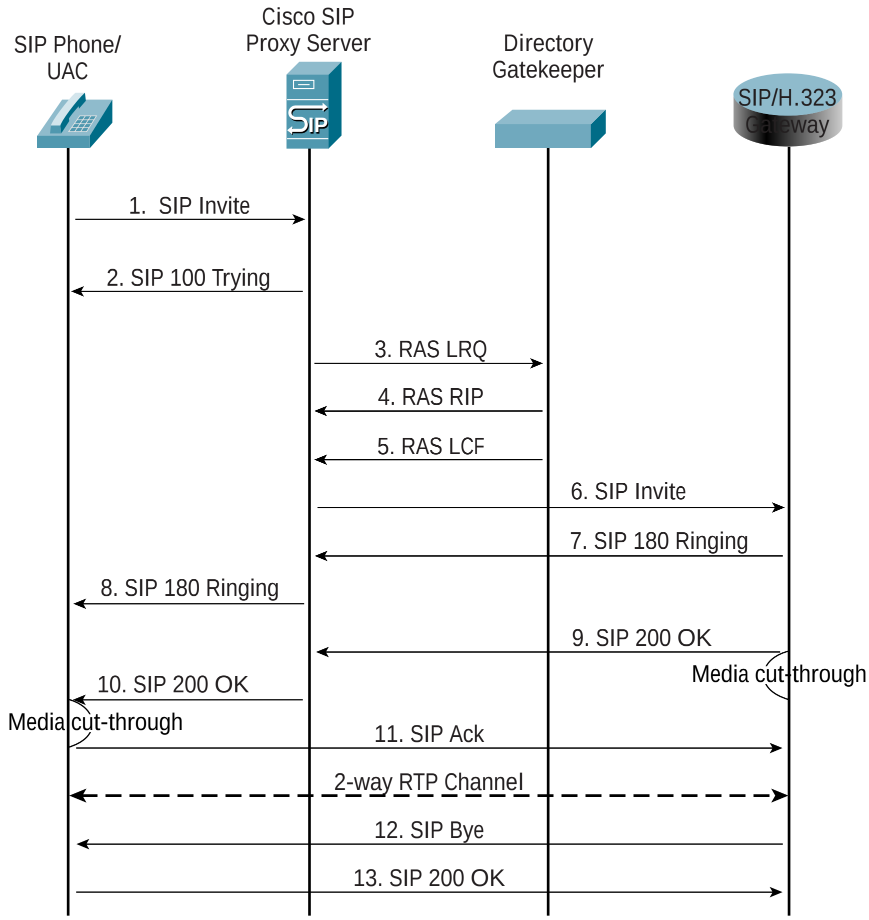


Figure 4

SIP Phone-to-SIP/H.323 Gateway-Call Via Cisco SIP Proxy Server




Provisions for communication between the Cisco SIP Proxy Server and H.323 gatekeepers, along with other features that allow SIP and H.323 to coexist on a common network, allow Cisco customers the ability to build out hybrid networks that include both SIP and H.323 traffic. The adoption of SIP will encourage the growth of new applications within these hybrid networks.

## Conclusion

Cisco offers packet voice solutions that offer architectural and protocol flexibility, with support for H.323, MGCP, and SIP. H.323 and SIP are both used today for call control and signaling service provider packet telephony network rollouts.

While each call control and signaling protocol offers advantages and disadvantages within different segments of a carrier network, Cisco solutions make it possible for service providers to use H.323 and SIP in the same network. Cisco has addressed coexistence and interoperability issues to enable service providers to optimize their networks and to have the flexibly to meet divergent customer needs.

Cisco is currently working closely with customers, partners, and standards bodies to address issues that will affect the deployment of mixed-protocol networks. The Cisco commitment to architectural and protocol flexibility is apparent in Cisco packet telephony solutions, which are based on industry standards and offer the highest level of interoperability as well as broad options for interconnection.

For more information on Cisco Packet Telephony Solutions, visit: www.cisco.com/go/telephony

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