shortcut

Your Short Cut to Knowledge

CCVP GWGK
Quick Reference Sheets

Brion S. Washington

About the Author 2
About the Technical Editor 2
Icons Used in This Book 3
Introduction 4
Purpose of This Guide....5
Who Should Read This Guide?....5
Gatekeepers....6
H.323 Call Setup and Control 8
Gatekeeper Transaction Messages Protocol (GKTMP)....19
Hot Standby Router Protocol....22
Voice Gateways 24
Open Settlement Protocol 30
Getting a Gateway Up and Running with H.323 31
Advanced Gateway Options Media Resources....32
Transcoding and Conferencing 34
Drop and Insert Multiplexing 36
Analog Connectivity....37
Circuit Emulation 43
Faxing....44
Tcl Scripts....45
IP-to-IP Gateway (IPIPGW)....47
Advanced IPIPGW Commands....49
Advanced Gateways Options....51
Gateway Redundancy....51
Survivable Remote Site Telephony 52
SIP SRST 54
H.323 SRST Reminders....54
SRST on MGCP Gateways 54
Gateway Problem Solving....54
Dial Plans....55
Dial Peers 55
Call Legs....56
Hunt Groups....58
Digit Manipulation 58
Voice Translation Rules and Profiles 59
COR....61
Enterprise Versus Service Provider....62

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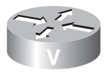
Icons Used in This Book



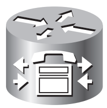
Voice-Enabled Router

IP Phone

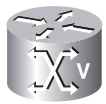
Phone

Network Cloud

SIP Server



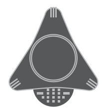
IP Telephony Router



Label Switch Router

PBX Switch

PBX



Phone Polycom

Introduction

For as long as anyone alive can remember, there has always been a way to pick up some sort of phone for chats with distant relatives or with companies that offer customer support. The protocols for this have been ever changing since the invention of the telephone. The old days of human operators gave way to central office switches, which in turn saw the uptake of private branch exchange (PBX) use in companies. Telecommunication methods are ever changing to meet customer expectations and to generate new services (and thus money).

In the early stages of voice communications, the rise of major telco companies brought numerous changes. For the most part, the changes proved to be beneficial to the growth of the industry. Among these benefits was the financial power to expand services to all parts of the world. Then, no matter where you lived, you probably had access to a phone. This almost universal access allowed providers to generate more revenue and reinvest in emerging technologies and research. The downside to this was that small companies could not compete with the giants of the field. When the smaller companies couldn't get a foothold into the market, larger companies continued their dominance. This market lockout started a selfish trend: Companies could do what they wanted with their equipment with little regard for client needs or interoperability with other companies. This monopoly created an environment in which the equipment was proprietary. Proprietary equipment, such as a PBX, from one company is probably not going to be interoperable with another company's PBX. The purchasing company is now tied to one brand of equipment.

Telecommunication monopolies have been declining for years. Everywhere you look, smaller companies are evolving, while creating new services and functionality for voice communications. A safe guess is that a few hundred billion phone calls are completed every year in the world. With the customer pool consisting of hundreds of millions, companies are competing to add services and growing their client base. Because smaller companies can now enter the market and challenge the established companies, proprietary networks of the past will be just that: of the past.

History shows that customer demand drives technology and new inventions. One demand has been for the intertwining of voice and data networks. But, most companies must answer to shareholders, and shareholders want a return on investment (ROI). The demand for the intertwining of voice and data networks, combined with the need to cut costs (and thus increase ROI), has helped spark the Voice over IP (VoIP) revolution.

Modern telecommunications use a variety of protocols and devices. VoIP was one of the technologies created to help combine voice and data networks. VoIP is a method of sending voice and video over data networks. VoIP has emerged as the de facto way for companies to send calls. Within VoIP, there are a few protocols to choose from, as follows:

- H.323
- SIP (Session Initiation Protocol)

- MGCP (Media Gateway Control Protocol)
- SGCP (Simple Gateway Control Protocol)
- H.248/MEGACO

Each of these protocols has its advantages and disadvantages. The protocols have different components to provide services. The components can be grouped by the function they serve. Examples of these components include gatekeepers (GK), gateways (GW), endpoints, circuits, call control, and different signaling types. Those who want to implement some type of VoIP need to understand which protocol best suits their voice needs.

Purpose of This Guide

This guide is intended to help readers pass the Gateway Gatekeeper (GWGK) exam required for the Cisco Certified Voice Professional (CCVP) and other voice-related certifications. The CCVP certification is valid for three years. After passing the CCVP, you may continue on to Cisco Certified Internetwork Export (CCIE) Voice certification. I hope readers find this guide useful as a preparation tool if they choose to pursue the CCIE Voice certification.

In this guide, you will find the necessary information to prepare for your exam, as outlined at http://www.cisco.com. The 642-452 Implementing Cisco Voice Gateways and Gatekeepers exam tests your ability to configure and set up various functions on these Cisco devices. As a tester, you must be able to decide which functions to configure based on the requirements. It is not this guide's intention to try to explain or even mention all the possible options that can be configured with Cisco GKs and GWs. Instead, I try to give you the information you *must have*, presented in such a way that you gain a full understanding of the subject matter.

Who Should Read This Guide?

Anyone who wants to augment his or her study material for the GWGK exam is the target audience for this guide. However, my intention also is to provide a deeper level of understanding for those who want to read a concise document to gain a fundamental understanding of GW and GK operation in Cisco environments. This guide does not delve into GK/GW minutiae. Instead, it provides a solid foundation on which you can build. For those readers who plan to prepare for additional certifications, this short cut could supplement your study material for other voice-related certifications (including CCIE Voice).

—Brion S. Washington

Gatekeepers

In the VoIP world of H.323, the GK plays a vital role if included in the network. The use of a GK is optional; if chosen, the GK must perform certain functions. Gatekeepers act like the brains in an H.323 network. They provide network access, bandwidth management, address translation, accounting, and dial plans. Gatekeepers allow the simple configurations of the majority of devices on your network, while requiring only a few devices to hold the majority of configurations. This architecture keeps call routing, security, and administration centralized to a few devices. Because the network only has a handful of devices that need to be configured when changes occur, the likelihood of errors is diminished.

NOTE

I use VoIP in this short cut to simplify the concept. Gatekeepers will work on any VoXX network using the H.323 protocol suite. Gatekeepers do not concern themselves with the medium or underlying technologies. Gatekeepers only need to make path selections and know whether resources are available for allocation.

Gatekeeper Communications

The GK operates on three different ports depending on whom it is communicating with. TCP port 1718 is used when communicating with other GKs. TCP port 1719 is used for H.323 devices to register with GK in their zones. The next port used is TCP port 1720 for call control.

When GKs are used in your H.323 network, GKs must/may provide certain mandatory and optional functions. Some of the functions are automatically configured when the GK service is started; others must be configured manually.

Mandatory and Optional Gatekeeper Functions

Mandatory Functions

Admission control: Controls endpoint admission into the H.323 network using H.225 Registration, Admission, and Status (RAS) messages, as follows:

- Admission Request (ARQ)
- Admission Confirm (ACF)
- Admission Reject (ARJ)

Admission control is automatically started when the GK is started.

Address translation: Translation of IP addresses to an E.164 phone number and vice versa.

Example 1: 44.44.1.31 = 402-555-5555

Example 2: 405-999-9978 = 192.168.1.109. Address translation must be configured manually.

Bandwidth control: Provides endpoint bandwidth management requirements using H.225 RAS messages. When an endpoint decides it needs a certain amount of bandwidth for the call, it sends a BRQ message to the GK asking for an amount of bandwidth to be set aside for the call. The GK responds with a BCF or BRJ message.

- Bandwidth Request (BRQ)
- Bandwidth Confirm (BCF)
- Bandwidth Reject (BRJ)

NOTE

Messages appear in the order in which they will happen on a GK-controlled network.

Zone management: Manages all registered endpoints in the zone. This provides centralized control for many devices within the zone.

Zone management is automatically started when the GK is started.

Optional Functions

Call authorization: The GK can restrict access for endpoints or GWs.

Bandwidth management: Reject admission when the required bandwidth is not available. The GK can reject calls if there is not enough bandwidth to support the call. All active calls are recorded by the GK to manage the bandwidth within the zone. The GK use the following formula to decide whether it can accept or reject the call request.

Available bandwidth = (Total allocated bandwidth) – (Bandwidth used locally) – (Bandwidth used by all alternates)

Call authorization and bandwidth management must be configured manually.

Call management: The GK maintains active call information. Used for resource allocation / route selection.

Call management is automatically started. Use **show** commands to display this information.

Call control signaling: The GK may route call signaling messages between endpoints using Gatekeeper Routed Call Signaling (GKRCS). Normally, the GK uses H.225 call signaling. The endpoints send messages directly to other endpoints without using GK signaling. Cisco gatekeepers do not support GKRCS.