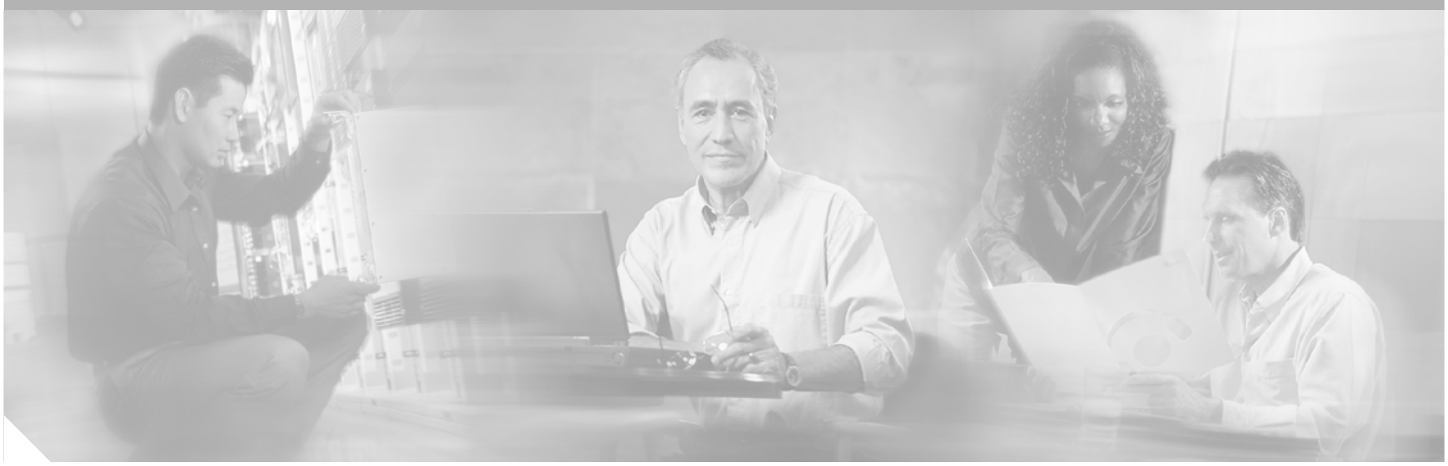

Cisco BTS 10200 Softswitch System Description

Software Release 900-03.01.00

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Preface

Preface

This preface explains the audience, purpose, and organization of the *Cisco BTS 10200 Softswitch System Description*. It also defines the conventions that are used to present specifications and information.

Audience

The *Cisco BTS 10200 Softswitch System Description* is intended for use by service provider management, system administration, and engineering personnel who are responsible for designing, installing, configuring, and maintaining networks that use the Cisco BTS 10200 Softswitch system. A familiarity with telco products and networking systems is recommended.

Purpose

The *Cisco BTS 10200 Softswitch System Description* describes the features and functions of the Cisco BTS 10200 Softswitch. This document includes descriptions of the new Release 3.1 features, including support for the following functions:

- Dial offload
- Support for network access server (NAS)
- Geographically distributed redundancy and survivability (GDRS) configuration
- Call control functions for SIP-enabled networks and devices
- Call control functions for H.323-based gateways and media endpoints

Note The H323 feature flag is set to *off* in this release.

For information on provisioning and operating the Cisco BTS 10200 Softswitch, see the "Related Documentation" section on page xiii.

Organization

The *Cisco BTS 10200 Softswitch System Description* is organized as follows:

- Chapter 1, "Network Overview," describes the historical use of PSTN systems for voice and packet networks for data, as well as the alternative for integrated voice/data services. It also describes the role of the Cisco BTS 10200 Softswitch in the integrated voice/data network.
- Chapter 2, "Product Description," describes the components of the Cisco BTS 10200 Softswitch, and provides data on system capacity and regulatory compliance. This chapter also includes a drawing and description of the equipment.
- Chapter 3, "Network Applications With the Cisco BTS 10200 Softswitch," describes the network applications supported by the Cisco BTS 10200 Softswitch, including, local, long distance, trunking types, and PSTN gateway for H.323 and SIP networks.
- Chapter 4, "External Interfaces and Services," describes the external interfaces of the Cisco BTS 10200 Softswitch, and details of the services provided.
- Chapter 5, "Site Requirements," lists the site requirements for the installation of the Cisco BTS 10200 Softswitch, including equipment requirements and network parameters.
- Chapter 6, "Reliability and Availability," describes the redundant systems and connections that the Cisco BTS 10200 Softswitch uses to ensure reliability and availability.
- Chapter 7, "System Maintenance," describes the management of network resources, such as gateways, trunks, and terminations.
- Chapter 8, "Events and Alarms," summarizes the event and alarm functions supported by the Cisco BTS 10200 Softswitch.
- Chapter 9, "Call Agent Component Description," provides details about the call processing and signaling functions of the call agent component of the Cisco BTS 10200 Softswitch.
- Chapter 10, "Feature Server Component Description," describes the functions of the feature server components of the Cisco BTS 10200 Softswitch.
- Chapter 11, "EMS/BDMS Component Description," describes the functions of the element management system and bulk data management server components of the Cisco BTS 10200 Softswitch.
- Chapter 12, "Routing Subsystem," explains how the Cisco BTS 10200 Softswitch performs call routing functions.
- Chapter 13, "Network Features," describes the network features supported by the Cisco BTS 10200 Softswitch, including casual dialing, directory and operator services, 911 and other *n*11 functions, and tandem services.
- Chapter 14, "Subscriber Features," describes the subscriber features supported by the Cisco BTS 10200 Softswitch, including POTS and Centrex services.
- Chapter 15, "Feature Interactions," describes the interactions among the various features offered by the Cisco BTS 10200 Softswitch.
- Chapter 16, "Announcements," lists the release cause codes supported by the Cisco BTS 10200 Softswitch, and the audio announcements that can be played by specified gateways.
- Chapter 17, "Feature Tones" lists the conditions that trigger the system to play specific alerting tones, and the auditory patterns for those tones.
- Appendix A, "Defined Cause Codes" provides industry standards information regarding release cause codes.

- Glossary.

Conventions

This publication uses the document conventions listed in this section.

Table 1 Font Conventions

Convention	Definition	Sample
Times bold	Text body font used for any argument, command, keyword, or punctuation that is part of a command that the user enters in text and command environments.	This is similar to the UNIX **route** command.
Times italic	Text body font used for publication names and for emphasis.	See the *Cisco BTS 10200 Softswitch Operations Manual* for further details.
`Courier`	Font used for screen displays, prompts, and scripts.	`Are you ready to continue?  [Y]`
`Courier bold`	Font used to indicate what the user enters in examples of command environments.	Login: **`root`** Password: **`<password>`**

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the manual.

Tip Means *the following information will help you solve a problem*. The tips information might not be troubleshooting or even an action, but could be useful information or information that might save time.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Warning **Means *danger*. You are in a situation that could cause bodily injury. Before you work on any equipment, you must be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. To see translated versions of the warning, refer to the *Regulatory Compliance and Safety* document that accompanied the device.**

Related Documentation

The *Cisco BTS 10200 Softswitch Command Line Interface (CLI) Guide* and *Cisco BTS 10200 Softswitch Operations Manual* are supplied with the shipment of the Cisco BTS 10200 Softswitch system. For additional technical details about your system, please contact Cisco TAC.

Obtaining Documentation

The following sections provide sources for obtaining documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at the following sites:

- http://www.cisco.com
- http://www-china.cisco.com
- http://www-europe.cisco.com

Documentation CD-ROM

Cisco documentation and additional literature are available in a CD-ROM package, which ships with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or as an annual subscription.

Ordering Documentation

Cisco documentation is available in the following ways:

- Registered Cisco Direct Customers can order Cisco Product documentation from the Networking Products MarketPlace:

  http://www.cisco.com/cgi-bin/order/order_root.pl

- Registered Cisco.com users can order the Documentation CD-ROM through the online Subscription Store:

  http://www.cisco.com/go/subscription

- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco corporate headquarters (California, USA) at 408 526-7208 or, in North America, by calling 800 553-NETS(6387).

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We appreciate your comments.

Obtaining Technical Assistance

Cisco provides Cisco.com as a starting point for all technical assistance. Customers and partners can obtain documentation, troubleshooting tips, and sample configurations from online tools. For Cisco.com registered users, additional troubleshooting tools are available from the TAC website.

Cisco.com

Cisco.com is the foundation of a suite of interactive, networked services that provides immediate, open access to Cisco information and resources at anytime, from anywhere in the world. This highly integrated Internet application is a powerful, easy-to-use tool for doing business with Cisco.

Cisco.com provides a broad range of features and services to help customers and partners streamline business processes and improve productivity. Through Cisco.com, you can find information about Cisco and our networking solutions, services, and programs. In addition, you can resolve technical issues with online technical support, download and test software packages, and order Cisco learning materials and merchandise. Valuable online skill assessment, training, and certification programs are also available.

Customers and partners can self-register on Cisco.com to obtain additional personalized information and services. Registered users can order products, check on the status of an order, access technical support, and view benefits specific to their relationships with Cisco.

To access Cisco.com, go to the following website:

http://www.cisco.com

Technical Assistance Center

The Cisco TAC website is available to all customers who need technical assistance with a Cisco product or technology that is under warranty or covered by a maintenance contract.

Contacting TAC by Using the Cisco TAC Website

If you have a priority level 3 (P3) or priority level 4 (P4) problem, contact TAC by going to the TAC website:

http://www.cisco.com/tac

P3 and P4 level problems are defined as follows:

- P3—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- P4—You need information or assistance on Cisco product capabilities, product installation, or basic product configuration.

In each of the above cases, use the Cisco TAC website to quickly find answers to your questions.

To register for Cisco.com, go to the following website:

http://www.cisco.com/register/

If you cannot resolve your technical issue by using the TAC online resources, Cisco.com registered users can open a case online by using the TAC Case Open tool at the following website:

http://www.cisco.com/tac/caseopen

Contacting TAC by Telephone

If you have a priority level 1 (P1) or priority level 2 (P2) problem, contact TAC by telephone and immediately open a case. To obtain a directory of toll-free numbers for your country, go to the following website:

http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

P1 and P2 level problems are defined as follows:

- P1—Your production network is down, causing a critical impact to business operations if service is not restored quickly. No workaround is available.
- P2—Your production network is severely degraded, affecting significant aspects of your business operations. No workaround is available.

CHAPTER 1

Network Overview

Background—Traditional PSTN Equipment

Traditional telephone services are engineered and offered over the public switched telephone network (PSTN) via plain old telephone service (POTS) equipment at customer premises. The PSTN is managed and owned by telephony service providers using Class 5 central office (CO) switches and Class 4 tandem switches to provide basic subscriber services, as well as some enhanced service features. The overall network consists of the following switches:

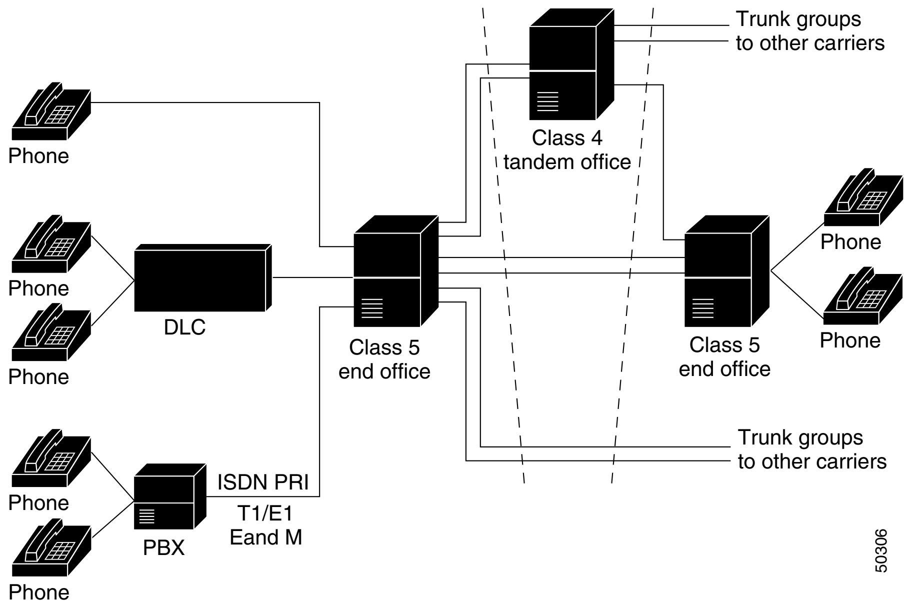
- Class 5 switches, providing individual subscriber interfaces
- Class 4 access tandem switches, providing interoffice trunk interfaces
- Class 4 end office switches for long distance services

Long distance service providers connect to local service providers via specific equal-access interfaces. Signaling between offices is based on one of the following methods:

- In-band multifrequency (MF) signaling
- Signaling System 7 (SS7), common channel signaling (CCS) standard

The majority of PSTN equipment today transports SS7 signaling. An example of a traditional local carrier PSTN network is shown in Figure 1-1.

Figure 1-1 Example of Traditional PSTN



Class 5 CO switching equipment is designed around a PSTN switch fabric, and equipped with subscriber interfaces such as

- Analog copper loops
- ISDN interfaces
- Digital loop carrier (DLC)

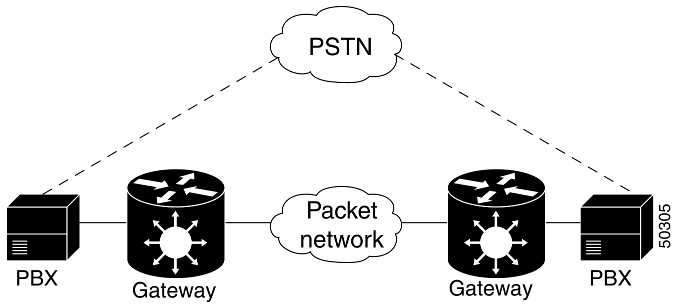
A Class 5 end office provides call processing as well as enhanced voice service features. A Class 4 tandem office provides basic call control and interoffice signaling, typically SS7.

Voice Over Packet Gateways

Traditional data services are offered over dedicated packet networks, separate from the PSTN-based voice networks. Packet networks are based on IP routers, ATM switches and frame relay switches. These data networks have grown quickly to support the increasing demand for data transmission. Systems have been put in place to transport voice and associated call signaling over the data network using tunneling methods.

A large number of gateway types—for example, H.323 VoIP gateways and gatekeepers—have been developed and deployed over packet networks such as IP/ATM networks. (See Figure 1-2.)

Figure 1-2 Example of Gateways Used with Packet Network as a PSTN Bypass



Packet/PSTN Network Convergence

As an alternative to maintaining two different types of networks (PSTN for voice, and packet networks for data), the solution is to bring PSTN-parity voice functionality to the data networks. These next-generation networks are based on the following requirements:

- Provide PSTN-parity voice services
- Seamlessly integrate with Web content
- Permit the end user to configure and customize services in a flexible manner via Web access

Neither traditional PSTN equipment nor the current VoIP gateway systems offer these integrated voice/data services.

Cisco BTS 10200 Softswitch In the Packet Network

Telecommunications innovations have resulted in a migration from voice telephony to packet-based networks. Cisco Open Packet Telephony (OPT) helps service providers support the requirements of both the traditional and new markets, as well as deliver new services. The Cisco BTS 10200 supports the use of OPT technology to deploy PSTN-parity voice functionality on the network. The Cisco BTS 10200 provides next-generation integrated voice and data switching solutions for packet networks. It is a standards-based solution that provides advanced call control and service intelligence to both telephony and packet networks.

The Cisco BTS 10200 provides the major functions performed by traditional Class 4 and Class 5 switching systems. Class 4 and 5 switches are large systems whose functionality can be partitioned into two main areas, each with several subsystems:

- Bearer path for ingress (entry into) and egress (exit from) the network
  - Ingress I/O subsystem
  - Egress I/O subsystem
- Switching control
  - TDM switching fabric and call processing engine
  - Signaling system interface
  - Operations Support System (OSS) interface

In the Class 4 and 5 systems, the call processing engine interfaces with the other subsystems as follows:

- Interfaces with the I/O subsystems for in-band signaling control
- Interfaces with the media path control
- Interfaces with the signaling system interface to provide SS7 protocol processing and ISDN D-Channel signaling
- The Cisco BTS 10200 solution distributes this switching control and bearer path infrastructure onto packet networks. The key functions of the legacy switching control systems are mapped into, and implemented by, the Cisco BTS 10200. The bearer path infrastructure is provided via the media gateways, which interface TDM facilities and analog phone lines into the packet network. These media gateways provide media encoding/decoding and packetization/depacketization functions.

Cisco BTS 10200 Functions

The Cisco BTS 10200 provides the following functions:

- Call control intelligence for establishing, maintaining, routing and terminating voice calls through the IP or ATM network via media gateways, while seamlessly operating with circuit-switched networks
- PSTN-parity routing mechanisms for voice calls including local access and transport area (LATA), interLATA, international, operator services, and emergency services routing
- Call processing, subscriber services and features, billing support and carrier class availability/reliability for subscribers and trunks connected to media gateways
- Key voice handling features, such as call waiting, call holding, call transferring, multiline hunting and caller identification
- Traffic measurements, such as call-completion counters, resource status and congestion information
- Event reports, including user provisioning of report filters
- Redundant hardware and software fail-safes to provide reliable and outage-free operation
- PSTN and intelligent network features for IP or ATM network subscribers
- Control of announcement servers
- Communication with existing OSS and network management systems (NMS) to support fault, configuration, accounting, performance, and security (FCAPS) functions
- SS7 signaling, and interoperability with legacy PSTN equipment
- Interoperability with PBX equipment via ISDN-PRI and CAS protocols
- Generation of triggers allows service providers to offer enhanced services using external service platforms (consistent with the ITU CS-2 call model)
- Enhanced Centrex services (virtual office) for business subscribers, including telecommuters and mobile workers
- Dial offload—Dial offload involves intercepting Internet traffic at inbound Class 5 locations and carrying this traffic over the packet network (instead of the PSTN) to the Internet service providers (ISP).
- Call control functions for the H.323-based gateways and end points

Note The H323 feature flag is set to *off* in this release.

- Call control functions for tandem applications
- Call control functions for SIP-enabled networks

Call Types Supported

The Cisco BTS 10200 supports the following types of calls:

- PSTN-to-PSTN Network calls—The Cisco BTS 10200 supports PSTN-to-PSTN network calls in conjunction with PSTN media gateways. A PSTN-to-PSTN call originates and terminates within a PSTN network without converting to a packet network. These are called PSTN-on-net calls.
- PSTN-to-Packet Network calls—The Cisco BTS 10200 supports PSTN-to-packet calls, which are calls that originate on a PSTN network and terminate on a packet network. These are often called off-net calls.
- Packet-to-PSTN Network calls—The Cisco BTS 10200 supports packet-to-PSTN calls. A packet-to-PSTN call is one that originates on a packet network and terminates on a PSTN network. These are called off-net calls.
- Packet-to-Packet calls—The Cisco BTS 10200 supports packet-to-packet calls. An IP-to-IP call is one that originates on an IP Network and terminates on an IP network. These are called IP-on-net calls. An ATM-to-ATM call (ATM-on-net call) originates and terminates on an ATM network.
- PSTN to Packet to PSTN calls—The Cisco BTS 10200 supports PSTN to packet to PSTN calls. This call originates on an ingress PSTN circuit and travels over a packet network to terminate on an egress PSTN port.
- MGCP-H.323 bridge—Long distance traffic is bridged directly between MGCP and H.323 networks. There is no need for H.323 call signaling data to be terminated at adjacent media gateways before converting them to MGCP messages or other signaling protocols.

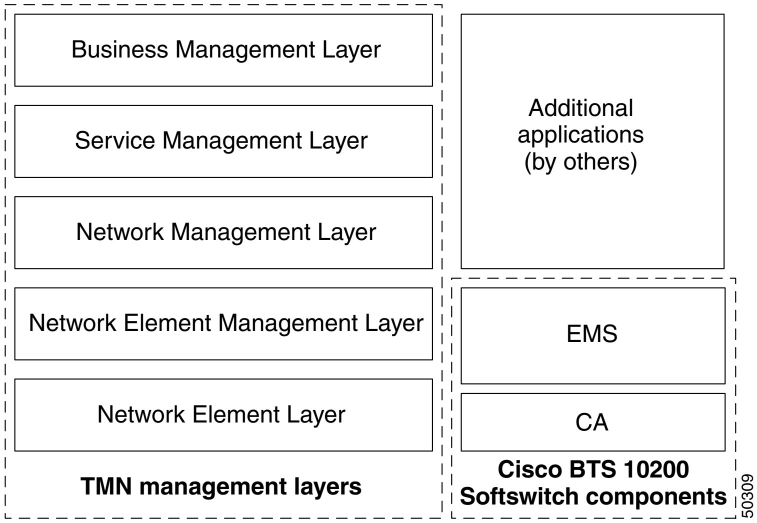
Note The H323 feature flag is set to *off* in this release.

- MGCP-SIP bridge—Traffic is bridged directly between MGCP and SIP networks. There is no need for SIP call signaling data to be terminated at adjacent media gateways before converting them to MGCP messages or other signaling protocols.

Cisco BTS 10200 Role in the TMN Model

This section describes the role of the Cisco BTS 10200 in the Telecommunications Management Network (TMN) model. Based on the basic ITU-T TMN model, shown in Figure 1-3, the Cisco BTS 10200 is involved in the NE Layer and NE Management Layer.

Figure 1-3 Cisco BTS 10200 Softswitch Components In the TMN Model



The role of each TMN layer is described below.

The Business Management Layer (BML) contains the following elements:

- Network planning
- Intercarrier agreements
- Strategic planning
- Enterprise-level management

The Service Management Layer (SML) contains the following elements:

- Customer interface
- Service provisioning
- Account management
- Customer-complaint management
- Integrated faults, billing, and QoS

The Network Management Layer (NML) contains the following elements:

- End-to-end network view
- All data aggregated to the network view
- Physical entity awareness

The Network Element Management Layer (NEML) contains the following elements:

- Subnet management
- Element management
- Reduced workload on NML
- Common NEs aggregated in a network

The Network Element Layer (NEL) contains the following elements:

- Performance data generation
- Self-diagnostics

- Alarm monitoring and generation
- Protocol conversions
- Billing generation

CHAPTER 2

Product Description

This chapter describes the components of the Cisco BTS 10200 Softswitch, and provides data on system capacity and regulatory compliance.

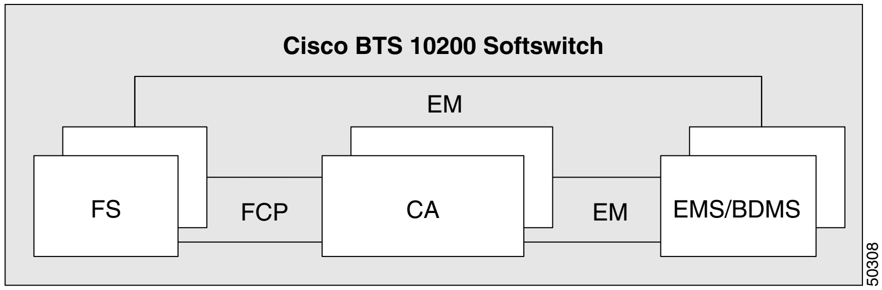
Cisco BTS 10200 Softswitch Components

The Cisco BTS 10200 network element (NE) consists of four logical components (see Figure 2-1) in a distributed architecture:

- CA—The call agent (CA) component serves as a call management system and media gateway controller. It handles the establishment, processing, and tear-down of telephony calls.
- FS—The feature servers (FS) provide POTS, Centrex, Tandem, and Advanced Intelligent Network (AIN) services to the calls controlled by the CAs and also provides processing for service features such as call forwarding, call waiting, local number portability, and so forth. The FS is an independent process from the CA.
- EMS—The element management system (EMS) controls the entire Cisco BTS 10200, and acts as a mediation device between an NMS and one or more CAs. It is also the interface for the provisioning, administration, and reporting features of the Cisco BTS 10200.
- BDMS—The bulk data management system (BDMS) coordinates the collection of billing data from the CA, and the forwarding of billing records to the service provider billing server.

The EMS and BDMS are colocated on one host machine and the CA and FS are colocated on another host machine. Interworking of CA and FS with media gateways (MGW) provides PSTN-parity voice service and a reliable migration platform for next-generation services.

Figure 2-1 Cisco BTS 10200 Components and Internal Interfaces



EM – Element management link
FCP – Feature control protocol
CA – Call agent
EMS/BDMS – Element management system/bulk data management system

Cisco BTS 10200 Database Features

Key data structures for the Cisco BTS 10200 are stored in shared memory and are accessible to any process in the system.

- A library of read-write locks controls access to shared memory
- The data structures are implemented using
  - Oracle data structure in the EMS
  - IDX data structure in CA/FS

The Cisco BTS 10200 provides the following user OAM&P capabilities:

- Perform system administration and security functions
- Show, add, change, or delete the database information through a local or remote interface
- Display reports of events, alarms, and failures
- Display and control the status of a component

The Cisco BTS 10200 database contains data for basic call processing and for special call features. The tables and parameters of the database are described in the *Cisco BTS 10200 Softswitch CLI Guide*.

Cisco BTS 10200 Equipment

The Cisco BTS 10200 consists of the following equipment:

- Call agent/feature server (CA/FS)—Two application servers, each with four CPUs.
- Element management system/bulk data management system (EMS/BDMS) server—Two administration processors, each with one CPU.
- Two switch routers.
- Alarm panel.
- Power distribution unit (PDU) for DC systems.
- Cabinet, 93 in. (236 cm) high x 24 in. (61 cm) wide x 36 in. (92 cm) deep, with 84 in. (213 cm) available vertical rack space.

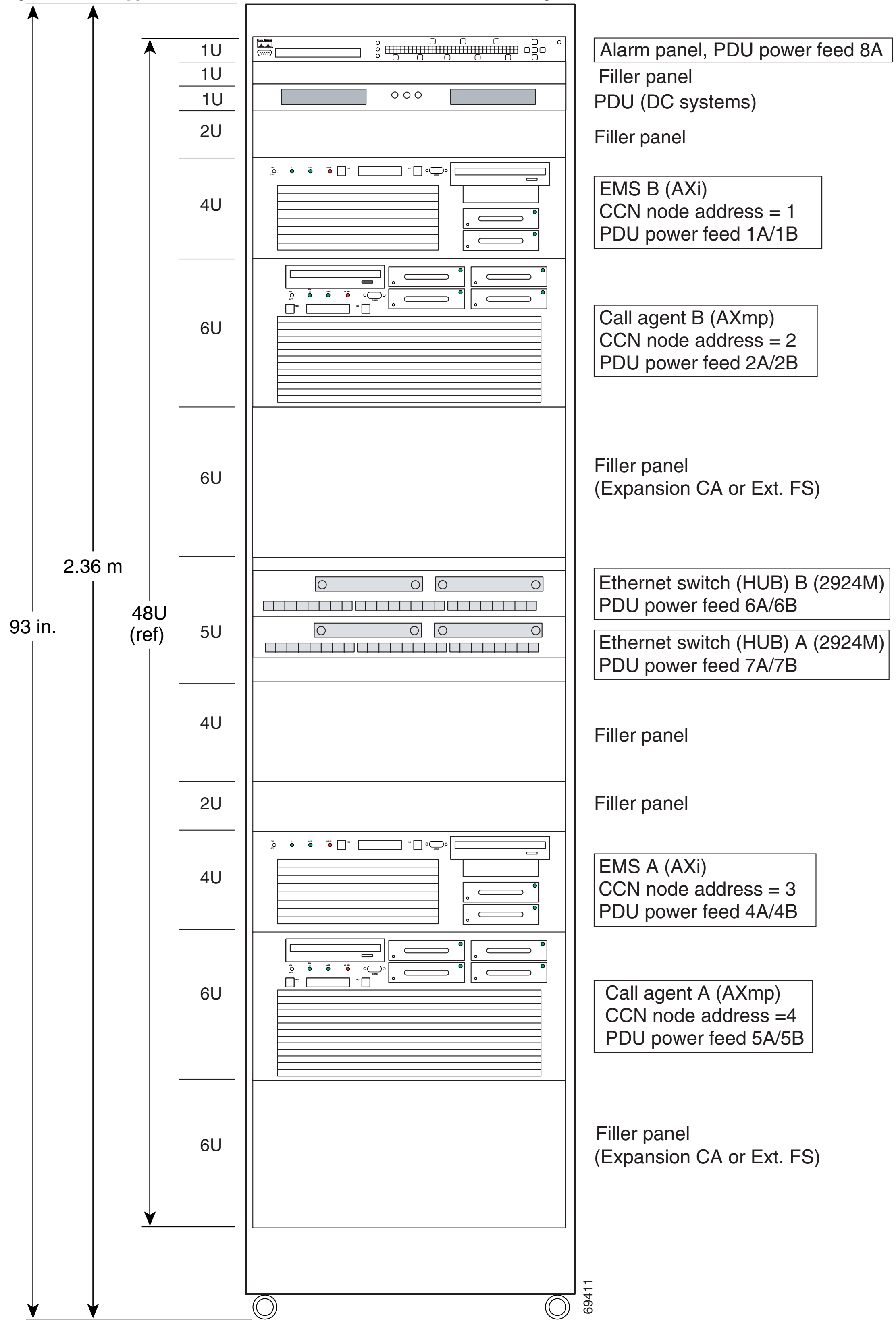
Note A rack-mounted system is also available.

The hard disks in the EMS/BDMS and CA/FS components store the provisioned database, and maintain a minimum of 48 hours of data in the areas of billing, traffic measurements, alarms, events, security logs, and activity logs.

The complete cabinet-mounted Cisco BTS 10200 weighs 1050 lbs. (477 kg), including the cabinet and all equipment as shown in Figure 2-2. A rack-mounted system weighs 570 lbs. (257 kg).

DC-powered systems require two (redundant) feeds of 40A at -48 VDC. AC-powered systems require two (redundant) circuits of 20A at 120 VAC.

Figure 2-2 Typical Cisco BTS 10200 Softswitch Rack Configuration



Geographically Distributed Redundancy and Survivability for Dial Offload Application

Note The GDRS configuration is currently used only with the dial offload application. See the "Dial Offload Application" section on page 3-5 for a description of the application.

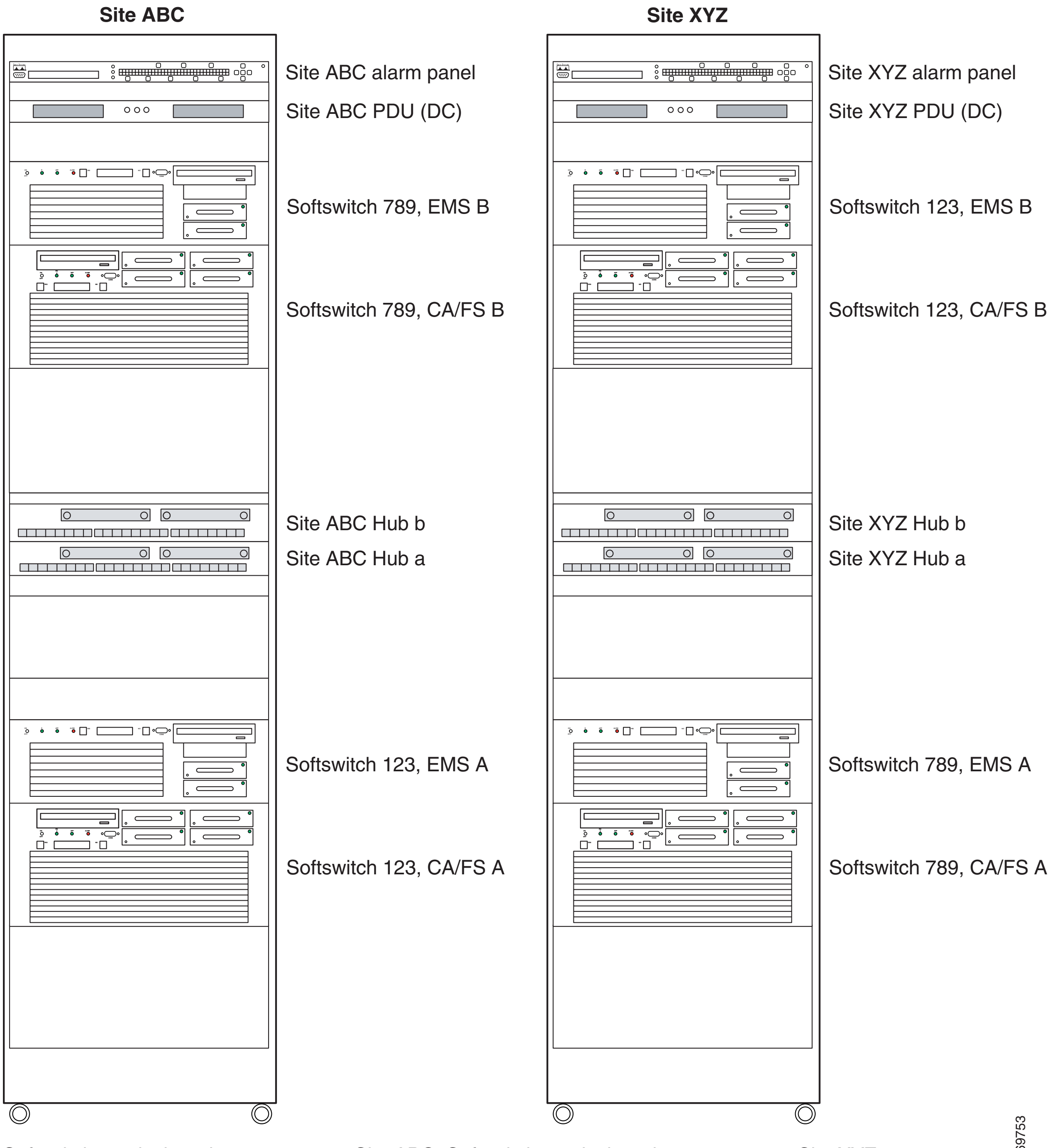
The Cisco BTS 10200 can be configured for geographically distributed redundancy and survivability (GDRS). Geographical separation is intended to improve system survivability in case of a catastrophic event. In this configuration, the primary side (Side A) of each component (EMS, BDMS, CA, FSPTC and FSAIN) is installed at one location, and standby side (Side B) of each component is installed at another location. Figure 2-3 shows an example with two independent Cisco BTS 10200 Softswitches in two locations:

- Side A of the Cisco BTS 10200 designated "Softswitch 123" is located at Site "ABC"
- Side B of the Cisco BTS 10200 designated "Softswitch 123" is located at Site "XYZ"
- Side A of the Cisco BTS 10200 designated "Softswitch 789" is located at Site "XYZ"
- Side B of the Cisco BTS 10200 designated "Softswitch 789" is located at Site "ABC"

The Cisco BTS 10200 is configured as a GDRS system (or non-GDRS system) at the time of the initial software installation.

Note If it is ever necessary to change the system configuration (GDRS or non-GDRS), this can only be accomplished by changing the system's physical configuration and IP addresses, and then reinstalling the software. During reinstallation, the system will display a prompt for setting the GDRS configuration. Cisco recommends that you contact the TAC for more detailed instructions prior to initiating such a change.

Figure 2-3 GDRS Example



Softswitch 123 is the primary system at Site ABC, Softswitch 789 is the primary system at Site XYZ.

Design Limits

The design limits for the Cisco BTS 10200 are listed in Table 2-1. The values listed in the table are the maximum that a fully equipped system can support. The achievable values for a particular system depend upon the levels for which it has been licensed.

Table 2-1 Cisco BTS 10200 Softswitch Design Limits

Category	Parameter	Value
Calls with 3-minute holding time (for basic calls)	Maximum number of call setups per second [1]	200
	Maximum number of simultaneous calls	36,000
Calls with 20-minute holding time (for trunk-trunk calls)	Maximum number of call setups per second [1]	50
	Maximum number of simultaneous calls	60,000
Announcements	Maximum number of announcements	1,000
	Maximum number of announcement trunks	6,120
Ports, Trunks, and Gateways	Maximum number of subscriber terminations	50,000
	Maximum number of trunk endpoints	50,000
	Maximum number of trunk groups	10,000
	Maximum number of gateways	10,000
Routes	Maximum number of routes	2,000
	Maximum number of route guides	1,000
SS7 and PRI	Maximum number of SS7 OPCs [2]	1
	Maximum number of SS7 routesets (DPCs) [3]	256 to 3,500
	Maximum number of SS7 linksets	127
	Maximum number of SS7 links	16
	Maximum number of SS7 trunks	50,000
	Maximum number of CICs per point code [4]	16,384
	Maximum number of PRI spans (groups) [5]	2,000
Billing	Size of billing record sent from Cisco BTS 10200 to external billing server for a basic call	400 bytes
	Maximum billing record capacity in SQL[6] database	10 million [7]

1. Maximum number of calls per second is less with use of the geographically distributed redundancy and survivability (GDRS) feature.
2. OPCs = Originating point codes
3. DPCs = Destination point codes
4. CICs = Circuit identification codes
5. PRI = Primary rate interface
6. SQL = Structured Query Language
7. 10 million billing records are sufficient to hold 48 hours of billing data on a typical Cisco BTS 10200 for maximum traffic load under normal circumstances.

Regulatory Compliance

The Cisco BTS 10200 complies with the standards listed in Table 2-2.

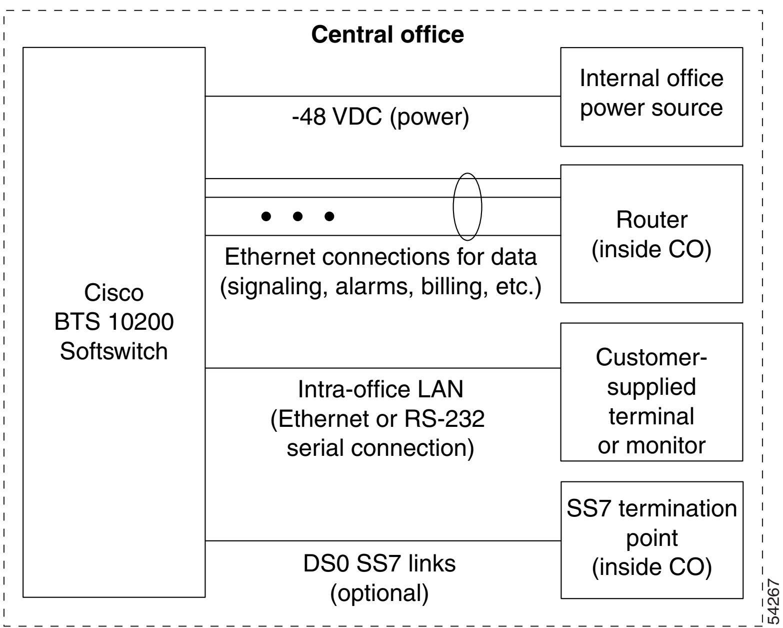
Table 2-2 Standards Compliance

Specification	
Information Technology Equipment	UL[1] 1950
EMI[2]	FCC[3] Class A (47CFR, Part 15)
Environmental compatibility	NEBS[4] Level 1 and Level 3 Requirements (SR-3580)

1. UL = Underwriters Laboratories
2. EMI = electromagnetic interference
3. FCC = Federal Communications Commission
4. NEBS = Network Equipment-Building System

The Code of Federal Regulations Title 47 (CFR 47) Part 68, *Connection of terminal equipment to the telephone network*, does not apply to this product. All of the points of demarcation are located at the first physical connection external to the Cisco BTS 10200 frame, which are at customer-provided equipment internal to the CO. See Figure 2-4 for a block diagram.

Figure 2-4 Block Diagram of Cisco BTS 10200 Connection to Internal CO Equipment



CHAPTER 3

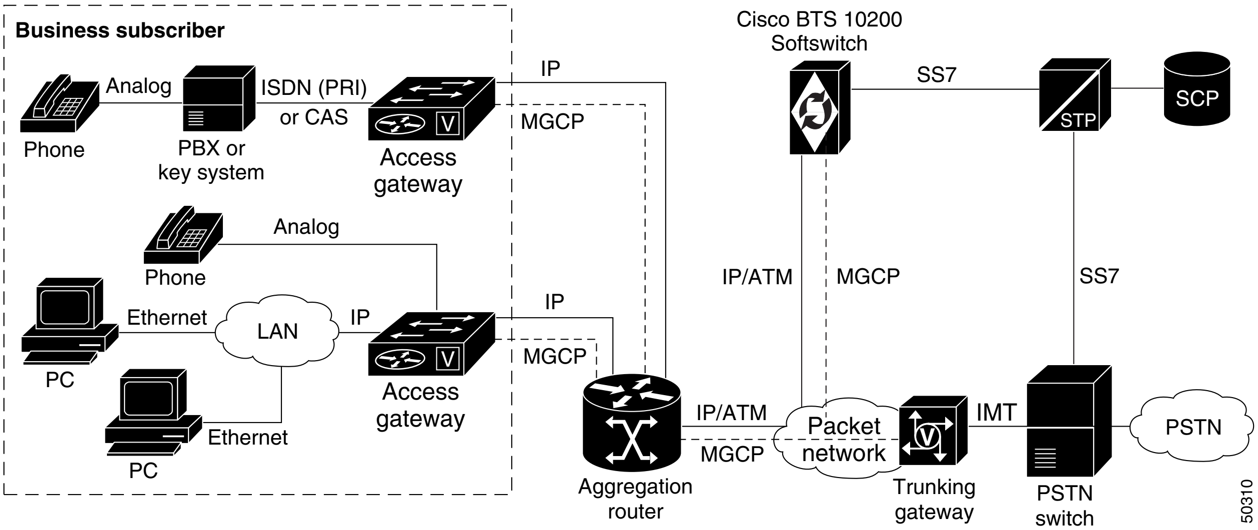
Network Applications With the Cisco BTS 10200 Softswitch

Carriers can migrate from existing PSTNs to next-generation networks by deploying the Cisco BTS 10200 Softswitch and MGWs. The Cisco BTS 10200 and MGWs perform the major switching functions previously performed by Class 5 CO and Class 4 Tandem switches. This allows carriers to offer additional crossover (packet, voice, and Internet) services. Cisco BTS 10200 applications include

Business Subscriber Application

A small or medium business subscriber might have 8 to 16 phone lines and a T1 connection to the Internet. The subscriber might want to allocate as much bandwidth to data services as possible, with voice services allowed to preempt the data bandwidth. This application is shown in Figure 3-1.

Figure 3-1 Example of Business Subscriber Application with Cisco BTS 10200 Softswitch



These small business group applications can be provided over existing wiring, allowing connectivity with WANs, LANs, Internet service providers (ISPs) and ATM networks.

The business subscriber application provides the following features:

- PSTN-parity voice and data transmission
- Increased bandwidth and speed to enable and enhance multimedia communication

The Cisco BTS 10200 supports Centrex services for business subscribers, including telecommuters and mobile workers. These enhancements are made possible by the separation of call-control processing and call-feature processing in the Cisco BTS 10200. This architecture allows carriers to bundle small business applications together for specific customers. This architecture enables a virtual office—allowing use of the full capabilities of a corporate network from a remote locations.

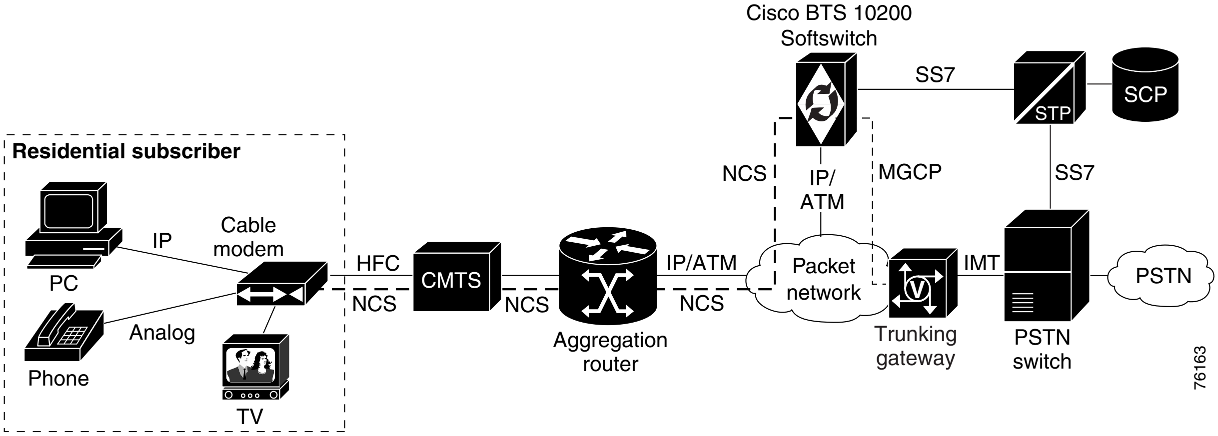
The small business application service can be a message-rate business service with a 2-line minimum and a 99-line maximum when direct inward dialing (DID) and direct outward dialing (DOD) features are activated on all lines.

Residential Subscriber with Cable Modem or Access Router

A typical residential subscriber is accessed via one of the following methods:

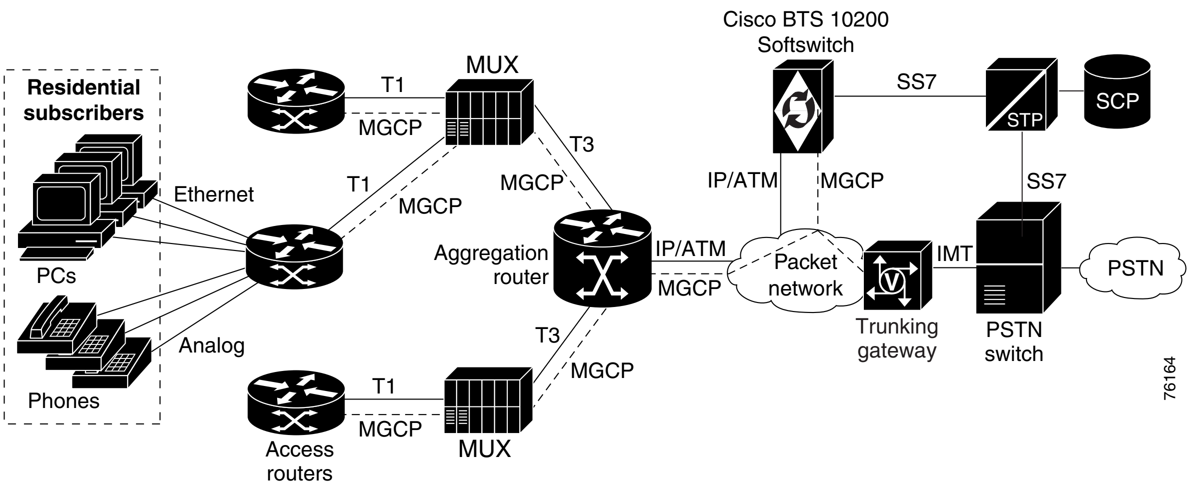
- Cable modem (customer premises equipment [CPE]) as shown in Figure 3-2
- Directly from an access router on the packet network as shown in Figure 3-3

Figure 3-2 Example of Local Cable Application



Note In Figure 3-2:
HFC = Hybrid fiber coax
CMTS = Cable modem termination system
NCS = Network-based call signaling

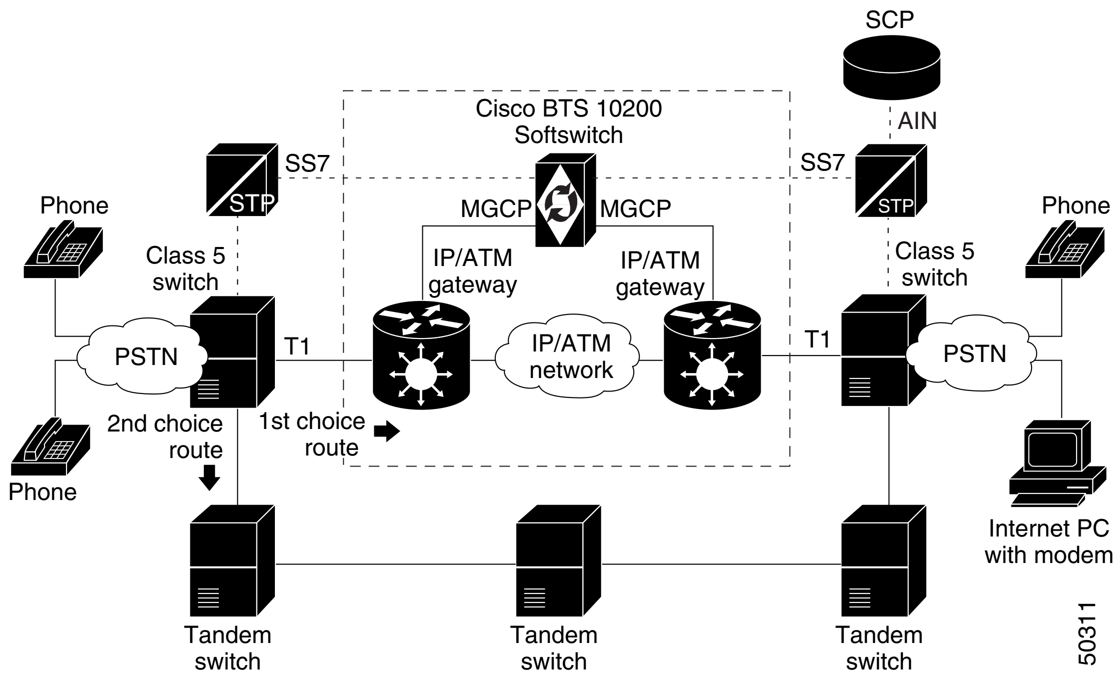
Figure 3-3 Example of Residential Application with Access Routers



Long Distance Network Application

Carriers can offload long distance Tandem traffic to a packet network for cost savings. The path through the Tandem switch can be used as a backup in this case. The Cisco BTS 10200 accesses a service control point (SCP) for Advanced Intelligent Network (AIN) features (local number portability [LNP] and toll-free calls). This application is shown in Figure 3-4.

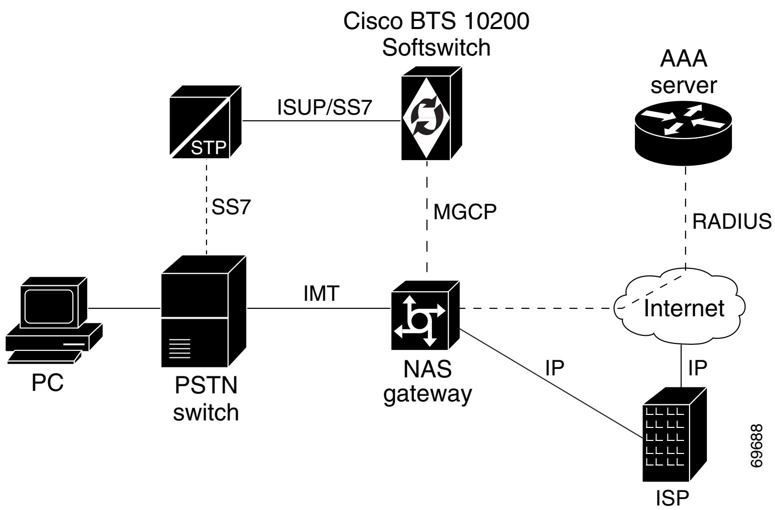
Figure 3-4 Example of Long Distance Network Application with Cisco BTS 10200 Softswitch



Dial Offload Application

When an end user logs on to the Internet, the PSTN switch frequently carries the call over a Primary Rate Interface (PRI) to the end user's ISP. The rapid growth in Internet traffic has caused increased demand and resulted in network congestion. Dial offload involves intercepting Internet traffic at inbound Class 5 locations and carrying this traffic over the packet network (instead of the PSTN) to the Internet service providers (ISP). The Cisco BTS 10200, along with network access server (NAS) gateways, provides an alternate route to the ISP, thus reducing the usage on the PRI circuits. In this scenario (see Figure 3-5), the local switch routes the call to the NAS gateway over an intermachine trunk (IMT). The gateway sends the bearer traffic (usually data to and from a PC) over a controlled-access packet network to the ISP, which maintains a node on the Internet.

Figure 3-5 Example of Dial Offload

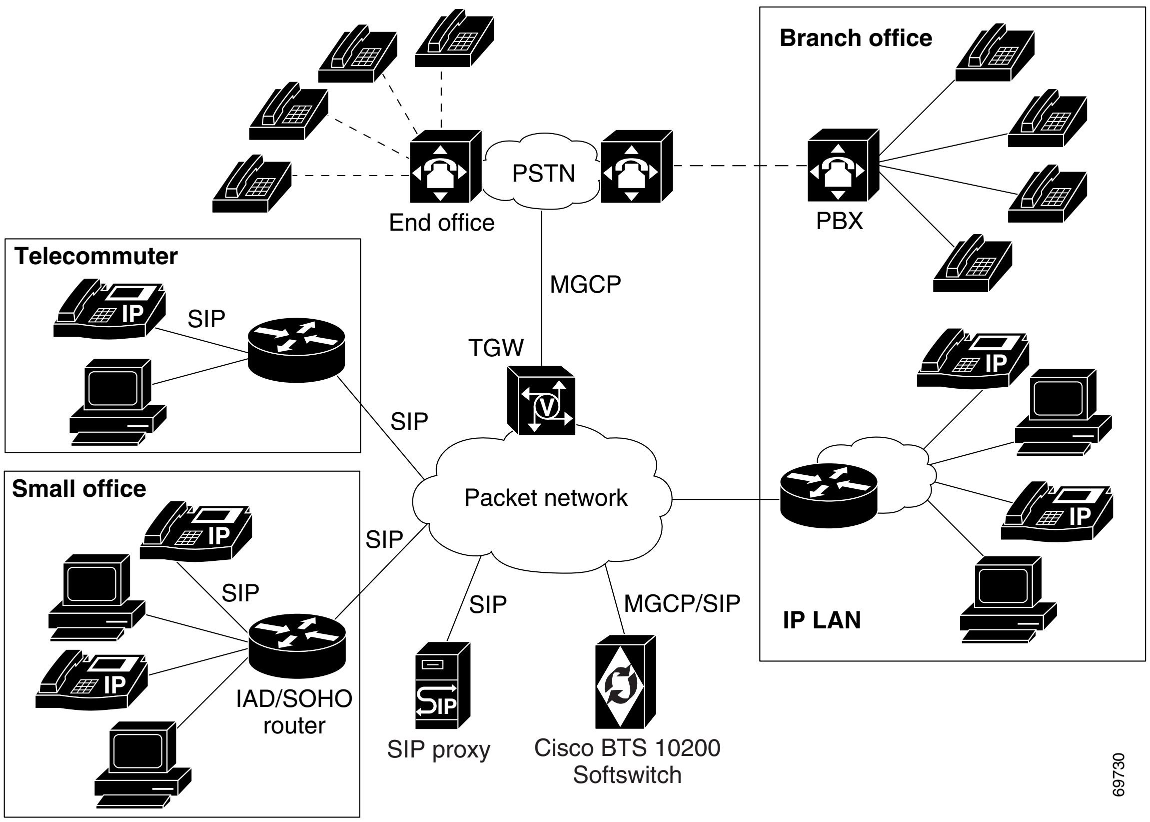


Note In Figure 3-5,
AAA = authentication, authorization, and accounting
NAS = Network access server
RADIUS = Remote authentication dial-in user service.

PSTN Gateway for SIP-Based Networks

The Cisco BTS 10200 routes traffic directly between MGCP-based networks and SIP-based networks. There is no need for SIP call signaling data to be terminated at adjacent media gateways before converting them to MGCP messages or other signaling protocols. This application is shown in Figure 3-6.

Figure 3-6 Example of Cisco BTS 10200 Softswitch Support for SIP-based Network



Applications for H.323-Based Networks

Note The H323 feature flag is set to *off* in this release.

The Cisco BTS 10200 provides H.323 signaling and interworking with various protocols of IP networks and the PSTN. The Cisco BTS 10200 and MGW together behave as an H.323 gateway, and interface to a gatekeeper (GK). The Cisco BTS 10200 can support up to four H.323 gateway instances. H.323 is a set of standards of the International Telecommunications Union (ITU-T) for the real-time transmission of multimedia traffic (audio, video and data) over packet networks.

PSTN Gateway for H.323-Based Networks

Figure 3-7 shows an example of a network using a Cisco BTS 10200 Softswitch to interconnect H.323-based networks with PSTN switches and MGCP gateways. H.323 and MGCP signaling protocols are both native to the Cisco BTS 10200.

Figure 3-7 Sample Network with Cisco BTS 10200 Softswitch Connecting H.323, PSTN and MGCP Infrastructures

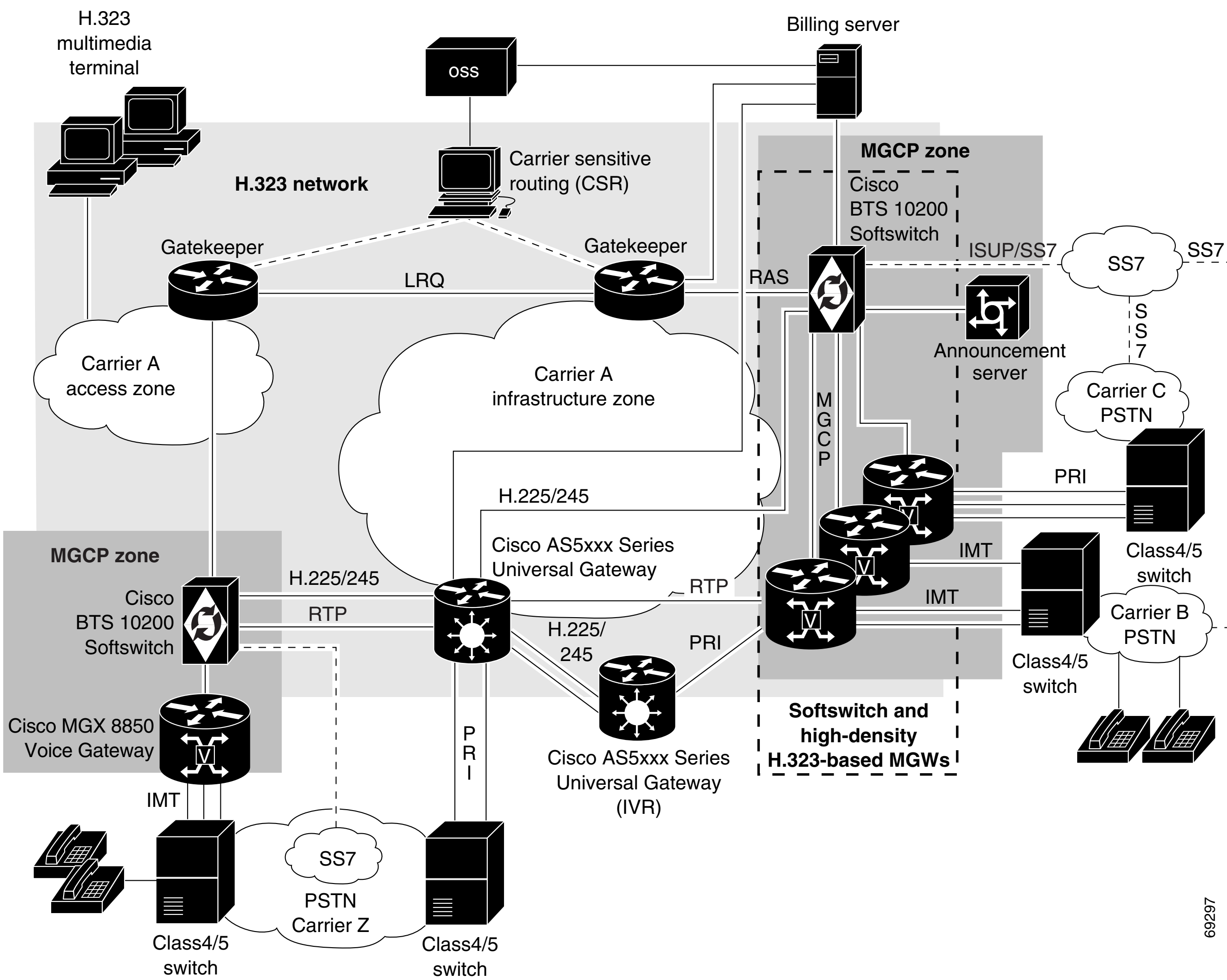


Table 3-1 Acronyms Used in Figure 3-7

Acronym	Definition
IMT	Intermachine trunk
IVR	Interactive voice response
LRQ	Location request
MGCP	Media gateway control protocol

Table 3-1 Acronyms Used in Figure 3-7

Acronym	Definition
PRI	Primary rate interface
PSTN	Public switched telephone network
RAS	Registration, admissions and status
RTP	Real-time transfer protocol
SS7	Signaling System 7

Support for High-Density H.323-Based MGW

Note The H323 feature flag is set to *off* in this release.

Figure 3-8 shows an example of connections between the PSTN and an existing H.323-based network using multiple voice gateways. Connections must be provisioned between individual gateways on each side of the H.323-based network. Bearer traffic and H.323 signaling traffic are both carried through the H.323-based network. Figure 3-9 shows a Cisco BTS 10200 and high-density MGW achieving the same capacity as the multiple voice gateways (Figure 3-8). This system saves POP space and simplifies provisioning for the service provider. It also separates the bearer layer from the call control layer.

Figure 3-8 Example of PSTN Connection to H.323-Based Network without a Softswitch

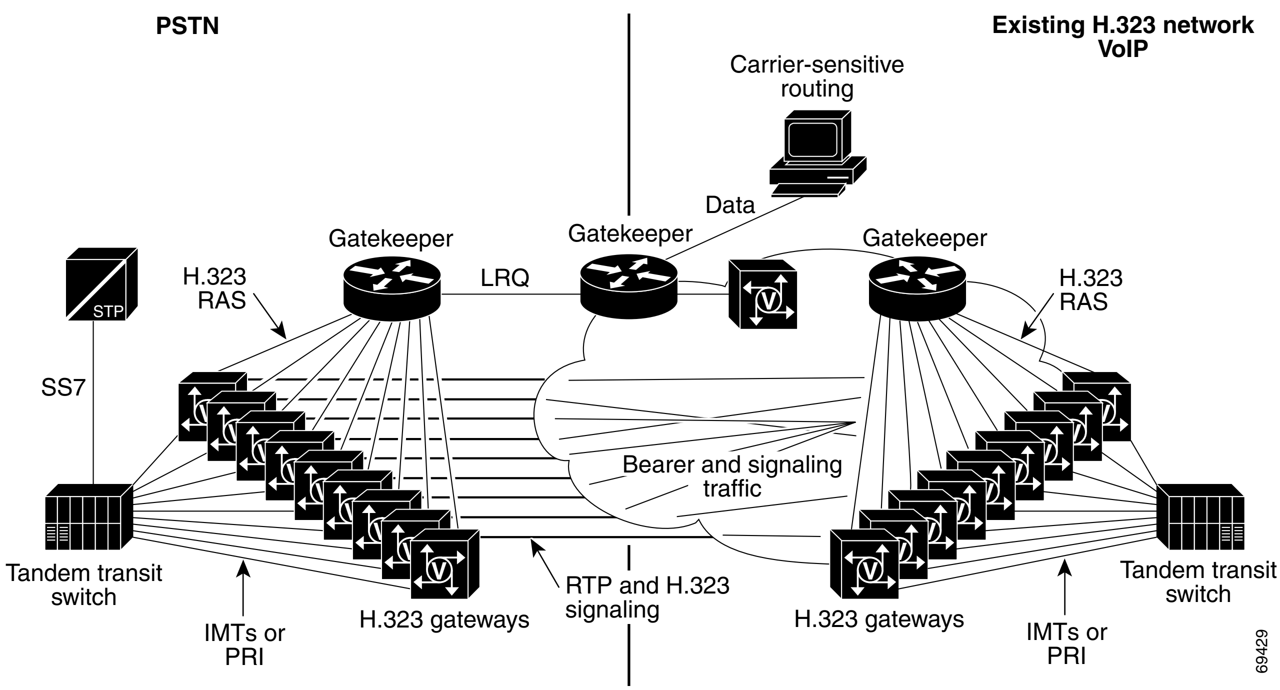
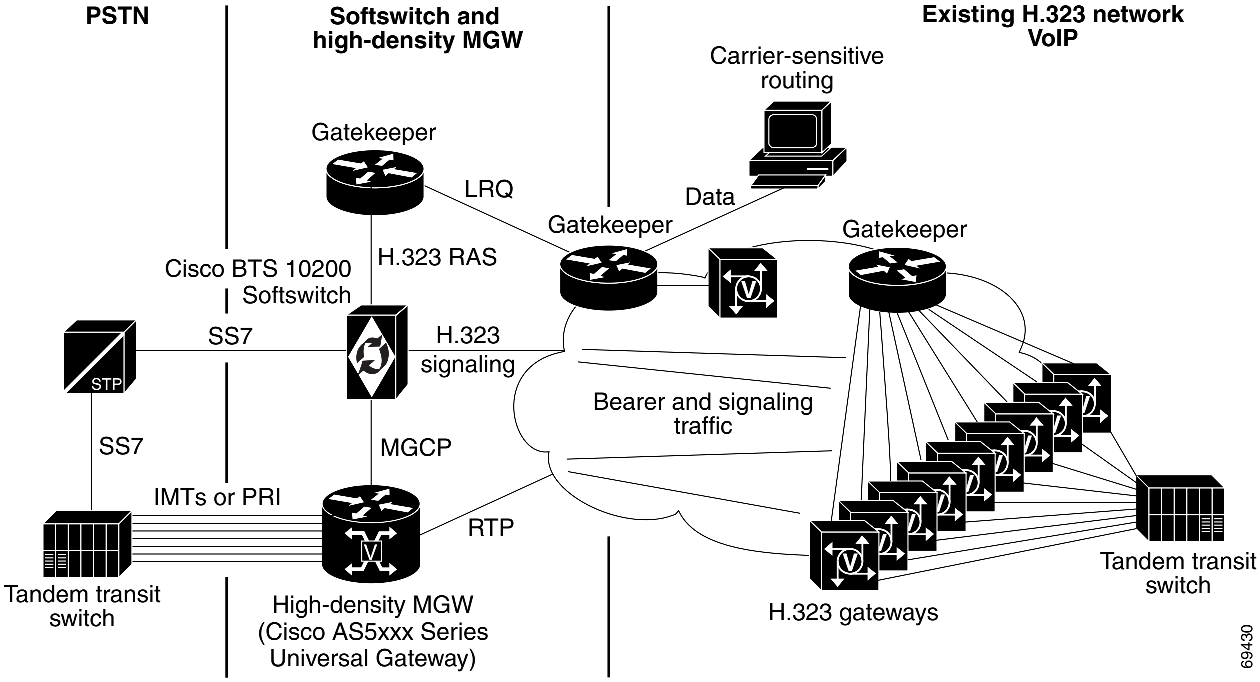


Figure 3-9 Example of PSTN Connection to H.323-Based Network with Cisco BTS 10200 Softswitch and High-density MGW



The Cisco BTS 10200-controlled high-density MGCP gateways interwork with existing H.323 VoIP ports (for example, the Cisco AS5xxx Series Universal Gateway). This allows the following capabilities on an MGCP-based network:

- Integrated access (voice/data trunks)
- Third party H.323 gateway or H.323-based server support
- Internet telephony service providers interfaces

Note Please contact Cisco for the appropriate type of Cisco AS5xxx Series Universal Gateway to use with your specific application.

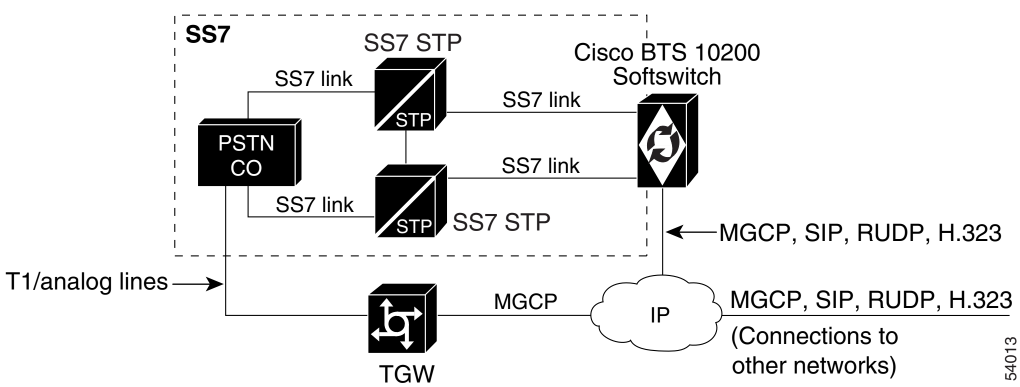
External Interfaces and Services

This section describes the external interfaces of the Cisco BTS 10200 Softswitch, and the services provided over these interfaces:

SS7 Interface

The Cisco BTS 10200 provides an SS7 signaling link interface to connect with peer PSTN switches and Service Control Points (SCPs). This interface is used to carry ISDN user part/transaction capabilities applications part (ISUP/TCAP) messages. It conforms to the ANSI SS7 specification. The basic interface of the Cisco BTS 10200 to the SS7 network is shown in Figure 4-1.

Figure 4-1 Cisco BTS 10200 Softswitch Interface to SS7 Network



H.323 Interface and Services

> **Note** The H323 feature flag is set to *off* in this release.

The Cisco BTS 10200 provides native H.323 signaling, which allows the Cisco BTS 10200 to communicate directly with H.323 gatekeepers (GK) and gateways, without the need for intermediate gateways. This allows H.323 Internet VoIP traffic to be carried seamlessly into the PSTN networks. The Cisco BTS 10200 and MGW together behave as an H.323 gateway, and interface to a gatekeeper (GK). The Cisco BTS 10200 can support up to four H.323 gateway instances.

The following H.323 services are supported:

- Hairpin (Call redirection)—On an incoming PSTN call, if the GK cannot route the incoming call, or if the GK determines that the call termination is on PSTN, the Cisco BTS 10200 can send the call back out to the PSTN or to another H.323 call leg.
- Auto Reattempt (Route Advance)—When a call does not complete, the Cisco BTS 10200 can reattempt the call based on the release cause code. The Cisco BTS 10200 action for each received cause code is provisionable in the Cause Code Map table. The call can be reattempted to the same trunk group or to another trunk group. In the H.323-based networks, the trunk group is mapped to an H.323 gateway. The choice of H.323 gateway depends on the cause code mapping analysis, and on carrier-dependent routing rules.
- Modem/Fax passthrough—The Cisco BTS 10200 allows MGCP gateways and H.323 gateways to transmit and receive faxes using in-band T.38 signaling. The CA does not perform the T.38 signaling.
- Information received response (IRR)—When a GK sends an IRQ message to an H.323 gateway, the gateway responds with an IRR message. IRRs contain per call information. Cisco BTS 10200 gateways have the capability to pack multiple call block information messages inside a single IRR message, thereby reducing network traffic. The service provider can provision the CA_CONFIG table for the number of call blocks to pack into a single IRR message.
- Resource availability indicator (RAI)—The RAI is an indication sent by a gateway to a GK to indicate a change in resource availability. The service provider can provision three fields in the H323_GW table to control the RAI behavior of the Cisco BTS 10200:
  - MAX_VOIP_CALLS—The total number of calls a Cisco BTS 10200 H.323 gateway will support.
  - HIGH_WATER_MARK—A percentage of the max_voip_calls. Once this level is reached, the Cisco BTS 10200 sends an RAI with outOfResources=TRUE.
  - LOW_WATER_MARK—A percentage of the max_voip_calls. Once this level is reached, the Cisco BTS 10200 sends an RAI with outOfResources=FALSE.
- Calling number information delivery (Octet 3A)—The Cisco BTS 10200 delivers calling number information (such as calling number, presentation restrictions, and so forth) to the terminating H.323 gateway via a Cisco proprietary octet 3A field in the H.323/H.225 setup message.

SIP and SIP-T Interfaces and Capabilities

Session initiation protocol (SIP) is a request-response protocol for initiating, maintaining, and terminating multimedia sessions. The Cisco BTS 10200 Softswitch uses SIP and the SIP-T MIME attachment to allow the PSTN and IP networks to exchange signaling information. SIP and SIP-T protocols are described in this section.

SIP Communications

A softswitch provides control intelligence for remote operation of telephony MGWs and their resources using MGCP signaling. When a telephony service provider deploys a softswitch into the public domain, it confronts the massive density of users that require services. A single softswitch cannot, by itself, address such high densities, involving hundreds of MGWs and spanning several geographical regions. Thus, there needs to be more than one softswitch. In fact there could be hundreds of softswitches, from varying vendors, belonging to various service providers, with various scales and roles within the network architecture. IP-based carrier voice communication networks must interwork between their respective domains. Therefore, the distributed softswitches must work together (versus a centralized softswitch that can work in isolation). In addition, distributed softswitches are required to interwork with the PSTN, in which case PSTN call signaling needs to be communicated between softswitches. SIP allows distributed softswitches to communicate with each other to provide end-users the required service. SIP is defined in the IETF document *RFC 2543, SIP: Session Initiation Protocol*. Additional reference information is available on the IETF web site:

- http://www.ietf.org/internet-drafts/draft-ietf-sip-rfc2543bis-02.txt (used in conjunction with *RFC 2543*)
- http://www.ietf.org/internet-drafts/draft-ietf-sip-call-flows-05.txt
- http://www.ietf.org/internet-drafts/draft-ietf-sip-srv-02.txt
- http://www.ietf.org/internet-drafts/draft-ietf-sip-100rel-04.txt
- http://www.ietf.org/internet-drafts/draft-rosenberg-sip-3pcc-02.txt
- http://www.ietf.org/internet-drafts/draft-levy-sip-diversion-02.txt
- http://www.ietf.org/internet-drafts/draft-mahy-sip-message-waiting-02.txt
- IETF document *RFC 2976, SIP INFO method*
- http://www.ietf.org/internet-drafts/draft-mahy-sip-signaled-digits-00.txt
- http://www.ietf.org/internet-drafts/draft-vemuri-sip-t-context-00.txt

Connecting SIP-Based Networks and PSTN

The Cisco BTS 10200 Softswitch provides interfaces to both SS7-based and SIP-based network entities, allowing the PSTN and SIP-based networks to operate as a unified network. The Cisco BTS 10200 enables calls to be routed between the two networks.

The Cisco BTS 10200 communicates with the following SIP servers:

- SIP proxy—The Cisco BTS 10200 can communicate with a SIP proxy/registrar, enabling SIP user agents in the domain of that proxy to route calls to the PSTN and receive calls from the PSTN.
- SIP redirect—The Cisco BTS 10200 can communicate with a SIP redirect server/registrar, enabling mobile SIP user agents registered with that server to receive calls from the PSTN.

- SIP user agent—The Cisco BTS 10200 can communicate with a SIP user agent for peer-to-peer SIP communications.

The Cisco BTS 10200 provides call data for billing on SIP calls. The following fields are supported in the call detail data records for calls that originate or terminate on a SIP trunk:

- SIP user name—The user portion of the FROM header, or the user portion of the DIVERSION header if a DIVERSION header is present.
- SIP call ID—Call ID header used in the SIP communication
- SIP adjacent hop address—IP address of the next upstream or downstream hop, depending on the direction of the call.

The Cisco BTS 10200 supports the following SIP protocol capabilities:

- Reliable provisional response—Each SIP request results in one or more provisional responses, which provide information on call progress, such as trying (100), alerting (180), queuing (182), and session progress (183). However, when run over UDP, SIP does not guarantee that these messages are delivered reliably, or in order. Compliance with the 100rel extension provides reliability of provisional responses.
- 3XX redirect response—The Cisco BTS 10200 honors the redirection response to a SIP INVITE request, and redirects the call accordingly. 3XX responses may contain one or more contacts. The Cisco BTS 10200 will attempt to place the call to the first contact specified in a 3XX response.
- SIP hairpin—The Cisco BTS 10200 supports hairpin functions for SIP calls. SIP requests go out from the Cisco BTS 10200 to a proxy and are routed back to the Cisco BTS 10200.
- Third party call control (3PCC)— The Cisco BTS 10200 can interoperate with a SIP back-to-back user agent in order to set up calls using the SIP 3PCC model. 3PCC refers to the ability of one entity (the controller) to set up and manage a call between two or more other parties. Call control signaling passes between the controller and each of the other parties. However, the media (call content) passes between the other parties.
- ANI-based routing for SIP calls—The Cisco BTS 10200 can receive SIP calls from individual lines and from remote IP-CENTREX systems. Typically, IP-CENTREX capability is offered to an enterprise or business group. The Cisco BTS 10200 receives these calls over a SIP-based trunk group and performs ANI-based routing onto the network. For additional details on ANI-based routing, see Chapter 12, “Routing Subsystem”.
- OPTIONS method—The Cisco BTS 10200 responds to an OPTIONS request with a 200 response containing the following headers:
  - Allow header—identifies the supported methods
  - Contact header—provides a URL where the user can be reached for further communications
  - Supported header—lists the option tags for supported extensions
- DTMF relay
  - SUBSCRIBE/NOTIFY—This is a method for passing DTMF digits via out-band channel.
  - INFO method—In this method, defined in IETF *RFC 2976*, the INFO request allows for the carrying of session related control information (DTMF digits) generated during a SIP session.
- Support for local ringback—The Cisco BTS 10200 will start a local ringback when a 180 message indicates that remote ringback is not being applied. This can occur when the remote UAS does not have the capability to play ringback. The absence of remote ringback is identified by the lack of session description protocol (SDP) attachment in an incoming 180 message.

- Support for voicemail—The Cisco BTS 10200 provides a SIP interface to compliant voicemail systems. The voicemail system can use the SIP NOTIFY message to turn on the visual message waiting indicator (VMWI) on a Cisco BTS 10200 subscriber phone.
- Support for diversion header—The SIP diversion header allows support for a number of enhanced features. In the legacy PSTN, redirection information is carried via ISDN/ISUP signaling. In SIP, the diversion header serves this purpose. The Cisco BTS 10200 maps the original called number, redirecting number, and redirection number from PRI to SIP. The methodology is based on the IETF document draft-levy-sip-diversion-02.txt.
- Support for user agent client (UAC) and user agent server (UAS) forking—The Cisco BTS 10200 supports UAC and UAS forking as follows:
  - UAC forking—After the Cisco BTS 10200 initiates a call, it may receive provisional as well as final responses from multiple user agent UAS. The Cisco BTS 10200 accepts the first final response, and sends a BYE to all subsequent successful final responses.
  - UAS forking—A SIP request from a UAC may arrive at the Cisco BTS 10200 as two separate requests via different paths (due to forking at proxies en route). Each of these requests will be routed independently by the Cisco BTS 10200. If the Cisco BTS 10200 is acting as a PSTN gateway, and receives forked INVITEs with differing RequestURIs, but the same call-Id, it will generate two call legs and add a unique To-tag in its response on each call leg. This allows the UAC to distinguish responses from the two call legs.
- Support for using INFO without a message body—During or after a call is set up, a remote user agent (UA) can send an INFO without body to the Cisco BTS 10200 to ensure that the call is still active. When receiving an INFO without body, the Cisco BTS 10200 simply replies with a 200 OK if the call exists, or a 481 Call Leg/Transaction Does Not Exist, if the call does not exist.
- Cancel initiation support—The Cisco BTS 10200 sends a CANCEL request to the applicable UAS or proxy server to cancel a pending INVITE.
- Support on type of service (TOS) for SIP signaling—SIP messages travel over the same network as voice traffic, so if this network becomes congested, the voice data may delay the SIP signaling packets. This could cause delays in call setup. The Cisco BTS 10200 allows the service provider to raise the priority of SIP packets in relation to other traffic, thus reducing this delay. TOS values are provisionable (via CLI commands) on a per trunk-group basis in the SOFTSW-TG-PROFILE table.
- Support for DNS SRV lookup for initiating SIP calls—The DNS SRV host name resolution feature allows the Cisco BTS 10200 to interoperate with proxy farms and find proxies and redirect servers. The SRV resource record (RR) allows the use of several servers for a single domain, service and protocol. This enables operators to move services from host to host with little impact, and to designate some hosts as primary servers for a service and others as backup. This feature is provisionable (via CLI) using the DNS-SRV token in the SOFTSW-TG-PROFILE table.
- Mapping URI to CIC—The Cisco BTS 10200 maps the Request-URI parameter to the TNS parameter used in the outgoing SS7 setup message. The receiving switch uses this data to select its outgoing trunk. The Cisco BTS 10200 can process either of the following Request-URI formats:
  - CIC parameter (cic=xxxx) is provided in the Request URI
  - Called party number in the Request-URI is prefixed with (101+CIC)

SIP-T Communications

SIP-T is a MIME attachment to SIP that enables PSTN-MIME encoding to carry PSTN signaling information elements through the packet network. SIP-T encapsulates the SS7 ISUP information between the IP and PSTN networks, thus allowing the two networks to work together. Only the network elements that use the tunneled PSTN signaling data will access this data.

MGCP Interfaces

Media Gateways (MGW) provide bearer paths between voice and packet networks, as well as connection control, endpoint control, auditing, and status functions. These gateways are equipped with voice coders that convert voice into packets, and voice decoders that convert packets into voice. Connections are grouped in calls, which means that a call can have one or more connections. One or more CA sets up the connections and calls.

The Cisco BTS 10200 connects to a variety of MGWs using Media Gateway Control Protocol (MGCP). It is based upon the evolving industry standards for MGCP, including the following MGCP variants:

- Network-based call signaling (NCS) protocol specification (PacketCable 1.0 variation)—supports cable access for voice application, including the Cisco UBR 7246 and Cisco UBR 924 universal broadband routers.
- MGCP (IETF Version 0.1, Draft 5, Feb. 1999).
- MGCP (IETF RFC-2705, Version 1.0, dated October 1999).

The MGCP-VARIANT parameter in the MGW-PROFILE table is used to identify the MGCP variant that a MGW supports.

The MGCP interface performs the following functions:

- Handles MGW initialization
- Provides endpoint auditing
- Provides MGW fault management
- Provides maintenance and administration of each termination, MGW operational states, and so forth.
- Carries call-control signaling
- Carries media-path control signaling

The Cisco BTS 10200 supports several special-purpose MGCP-based functions:

- Codec selection service is the process a CA uses to find a common codec (coder/decoder) type between originating and terminating call leg so a call can go through. The preferred codec type for originating and terminating calls is provisioned by the service provider using the QoS table in the Cisco BTS 10200 database. The QoS can be configured for subscriber or TG. The CA makes a decision on actual codec type based on a combination of the following conditions:
  - The codec types available on the MGW—the MGW dynamic profile (list of supported codecs reported by MGW) or MGW static codec list (list of supported codecs configured in the Cisco BTS 10200)
  - The codec type provisioned in the QoS table in the Cisco BTS 10200 database
    If a certain codec type is provisioned in QoS table but not available in MGW dynamic profile or TG profile, that type cannot be used. When no matching code is found, default pulse code modulation mu-law (PCMU) codec is used.

The following codec types are supported:

- G.711 mu-law (PCMU)
- G.711 A-law (PCMA)
- G.723.1 high rate
- G.723.1 Annex A High rate
- G.723.1 Low rate
- G.723.1 Annex A Low rate
- G.729
- G.729 Annex B
- G.726 32K rate (Default)
- G.726 24K rate
- G.726 16K rate
- G.728

- Resource Reservation Protocol (RSVP) is an IETF protocol for providing integrated services and reserving resources on the IP network. The service provider provisions the preferred reservation profile (guaranteed, controlled load, or best effort) in the Cisco BTS 10200 QoS table. When a reservation is needed on a connection, the Cisco BTS 10200 specifies the preferred reservation profile to the gateway. Whether or not RSVP will be done depends on the configuration of the gateway as well as the preferred reservation profile specified by the Cisco BTS 10200. If the best effort RSVP profile is specified, RSVP is not performed.
- Announcement server—An announcement server is a media server that stores network-based announcements and plays them to a caller upon request from the Cisco BTS 10200. The announcement server interfaces with the Cisco BTS 10200 using MGCP protocol. Every Cisco BTS 10200 in the network requires its own announcement server.
- Dual tone multifrequency (DTMF) signaling is transported across the IP network under MGCP control.
- Channel associated signaling (CAS) is used with the MGCP interworking function.
- ISDN primary rate interface (PRI) signaling is transported using backhaul RUDP protocol between the AGW and the CA.

Note RUDP stands for Reliable User Datagram Protocol, a Cisco Systems proprietary signaling backhaul protocol.

T.38 Fax Relay Interface

The Cisco BTS 10200 communicates with MGWs to permit faxes to be transported across the IP network in an MGCP control environment. The process is based on the following standards documents:

- IETF document *RFC 2833, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
- ITU-T document *Recommendation T.38 (06/98) - Procedures for real-time Group 3 facsimile communication over IP networks*

The process uses the MGW-controlled mode for fax relay. In this mode, the CA passes the session description protocol (SDP) information between the MGWs as in a normal voice call, and the CA is unaware that SDP extensions are being used by the MGWs for T.38 fax relay. After the SDP is exchanged, and if the MGW detects the fax call, it negotiates the codec to invoke the T.38 fax procedure automatically without CA involvement. The MGWs communicate with each other by using the realtime transport protocol (RTP) name signaling event (NSE) mechanism to exchange information.

PSAP Support for Enhanced 911 Emergency Services

The Cisco BTS 10200 supports call routing to a designated emergency service bureau (ESB), normally called the public safety answering point (PSAP) in the United States. Depending on municipal requirements and procedures, an ESB attendant can transfer calls to the proper agency, collect and relay emergency information to the agency, or dispatch emergency aid directly for one or more participating agencies. 911 calls are location dependent and must be selectively routed to the appropriate PSAP depending on where the call originates. Once the caller is connected to the PSAP attendant, the PSAP system typically displays the caller’s directory number to the PSAP attendant. Additional data (such as the subscriber’s name, address, and closest emergency response units) may also be retrieved from the local carrier automatic location identification (ALI) database and displayed to the PSAP attendant.

For additional description of support for 911 services, see the “Emergency Services (911)” section on page 13-3.

CALEA Interface

General Description of CALEA Implementation

The Cisco BTS 10200 supports the Communications Assistance for Law Enforcement Act (CALEA) in two phases. The first phase addresses the call data function and the second phase addresses call content.

Note The current Cisco BTS 10200 software supports the call data function. The call content function will be supported in a later release.

- For the call data phase, the Cisco BTS 10200 provides the PacketCable EMS/Radius interface for the transmission of call identifying information to the CALEA delivery function (DF) server. The Cisco BTS 10200 implementation is independent of the network application, such as cable or MGCP.
- The call content phase (later release) provides for capturing voice in the form of a replicated realtime transport protocol (RTP) stream. The replicated RTP stream is sent to the CALEA DF server upon request from the CA. The CA request is addressed to the appropriate intercept access point, such as:
  - An aggregation router—Sent via control commands based on the Common Open Policy Service (COPS) protocol
  - A trunking gateway—Sent via MGCP commands

Note These CALEA call data and call content features are currently based on industry-developed standards, including the PacketCable Electronic Surveillance Specification. Cisco will post information regarding the impact on its current approach to CALEA of any future determinations as to the technical requirements necessary to permit telecommunications carriers to comply with CALEA's assistance capability requirements.

CALEA Call Data Function

There are two call data functions supported by the Cisco BTS 10200.

- The Cisco BTS 10200 supports a secure provisioning interface to process intercept and wiretap requests from law enforcement agencies. (The service provider organization can limit viewing and provisioning of these parameters to selected authorized personnel.) The applicable parameters (entered via CLI) include the subscriber ID and DN, trap type, and call data channel for data transmission. The trap type specifies whether the tap order is a pen register (outgoing call information), a trap and trace (incoming call information), a pen and trace (incoming and outgoing call information), or an intercept (bidirectional plus the call content).
- The Cisco BTS 10200 supports the requirements of the enhanced PacketCable Event Messages Specification (EMS) PKT-SP-EM-I02-001128 (November 28, 2000) tailored for CALEA as specified by the PacketCable Electronic Surveillance Specifications PKT-SP-ESP-I01-991229 (December 1, 1999) and PKT-SP-ESP-D02-991207 (December 7, 1999). Full call-identifying information (call data) is shipped to a CALEA DF server from the Cisco BTS 10200 for the subject under surveillance for various call types (basic call, call forwarding, etc.). A CALEA DF server compliant with the PacketCable Electronic Surveillance Specifications, and per Cisco wiretap interface specification (for example, Comverse Infosys STAR-GATE), must be deployed in the network.

FCP Interface

Feature control protocol (FCP) is a multipurpose Internet mail extension (MIME) application on top of the session initiation protocol (SIP). FCP uses SIP for transport, and is the external protocol between the FS and the CA. It carries call state control and status information that is needed for external feature control, and provides an interface for both external and internal feature servers.

OSS Interface

The Cisco BTS 10200 OSS interface consists of the following parts:

- Command Line Interface (CLI) over secure shell (SSH) and telnet
- Common Object Request Broker Architecture (CORBA)
- SNMP traps, status, control, and measurement functions, and provisionable community strings
- Billing interface (FTP)
- Bulk provisioning (FTP)

Site Requirements

This section presents a summary of the installation site requirements for the Cisco BTS 10200 Softswitch. Installation procedures are provided separately.

Required Facilities

The Cisco BTS 10200 interfaces with a variety of NEs using various protocols. The facilities connecting the Cisco BTS 10200 to these NEs are customer supplied.

Operator Access to the Cisco BTS 10200 Softswitch NE

Operator access to the Cisco BTS 10200 NE is by a telnet or secure shell (SSH) session to the EMS over Ethernet. Communications can be interactive or can be in batch mode using FTP. See the "EMS OAM&P Interfaces" section on page 11-1 for additional user interface details.

Equipment Requirements

This section describes the Cisco BTS 10200 equipment requirements.

Note See Chapter 2, "Product Description," for a summary of the equipment supplied with the Cisco BTS 10200.

Site Physical Requirements

The physical requirements for installation of the Cisco BTS 10200 should be documented in the *Cisco BTS 10200 Softswitch Building Environment and Power Site Survey* document, available from Cisco TAC.

Electrical Power Requirements

DC-powered systems require two (redundant) feeds of 40A at -48 VDC. AC-powered systems require two (redundant) circuits of 20A at 120 VAC.

Shelf Interconnect Cables

Your Cisco BTS 10200 shipment includes all of the cables necessary for connection among the various Cisco BTS 10200 components.

Cables and Addresses for Connection to Other NEs

T1 cables are not included with the Cisco BTS 10200 order, and should be supplied by the service provider if used.

Network Data Definition

Certain network data need to be provided to Cisco so that each Cisco BTS 10200 Softswitch node can be given the appropriate initial software configuration. This configuration will ensure that the Cisco BTS 10200 will be able to communicate with the service provider network. The data requirements for a typical system are outlined below. Please contact Cisco to receive Network Site Survey sheets applicable to your specific system when preparing this information.

The Cisco BTS 10200 relies on ICMP Router Discovery Protocol (IRDP) for dynamic updating of router tables. The routers connected directly to the Cisco BTS 10200 are assumed to be IRDP capable, and the service provider network is assumed to be IRDP capable. (If this is not the case, contact Cisco for a review of Cisco BTS 10200 configuration options.) During installation, the service provider should turn on IRDP in the routers connected to the Cisco BTS 10200.

Note IRDP is an extension to Internet Control Message Protocol (ICMP) that provides a mechanism for routers to advertise useful default routes.

General Network Configuration Data

The following general network configuration data are required:

- Site ID—unique CORBA identifier, for example, dallas77
- Allowed operator access modes—Secure shell (SSH) login only, or SSH and Telnet login modes.
- NTP timehost—IP address of timehost for synchronization of all Cisco BTS 10200 components
- Subnet mask used for the networks that interface with the Cisco BTS 10200
- Hostnames and IP addresses of primary and secondary DNS servers
- Alarm Panel—hostname and IP address to use for alarm panel (alarm panel is supplied and installed by Cisco in the Cisco BTS 10200 rack)

External Network Configuration Data for EMS, BDMS, CA, and FS

The following external network configuration data are required for Side A and Side B EMS/BDMS, and Side A and Side B CA/FS:

- Hostnames and domain names
- External (public) IP addresses of hosts and gateways for external access via telnet, FTP, and so forth.
- External (public) IP addresses of hosts and gateways for dual MGCP and SIP traffic links:
  - Used for billing functions on EMS/BDMS
  - Used for MGCP and SIP functions on CA/FS

Internal Network Configuration Data

Six unused private nonroutable subnets are required for internal communications between the CA/FS and EMS/BDMS. Please contact Cisco for recommended subnet values.

Reliability and Availability

Availability Goals

The goals of the Cisco BTS 10200 Softswitch are to achieve

- 99.999% availability
- Robust switchover of redundant systems
- Retention of stable calls during switchover
- Fast and accurate data replication among redundant systems

To achieve these goals, the Cisco BTS 10200 operates with redundancy of both hardware and software for all major system components: CA, FS, EMS, and BDMS. In addition, the communication lines (LAN) among the components are also redundant.

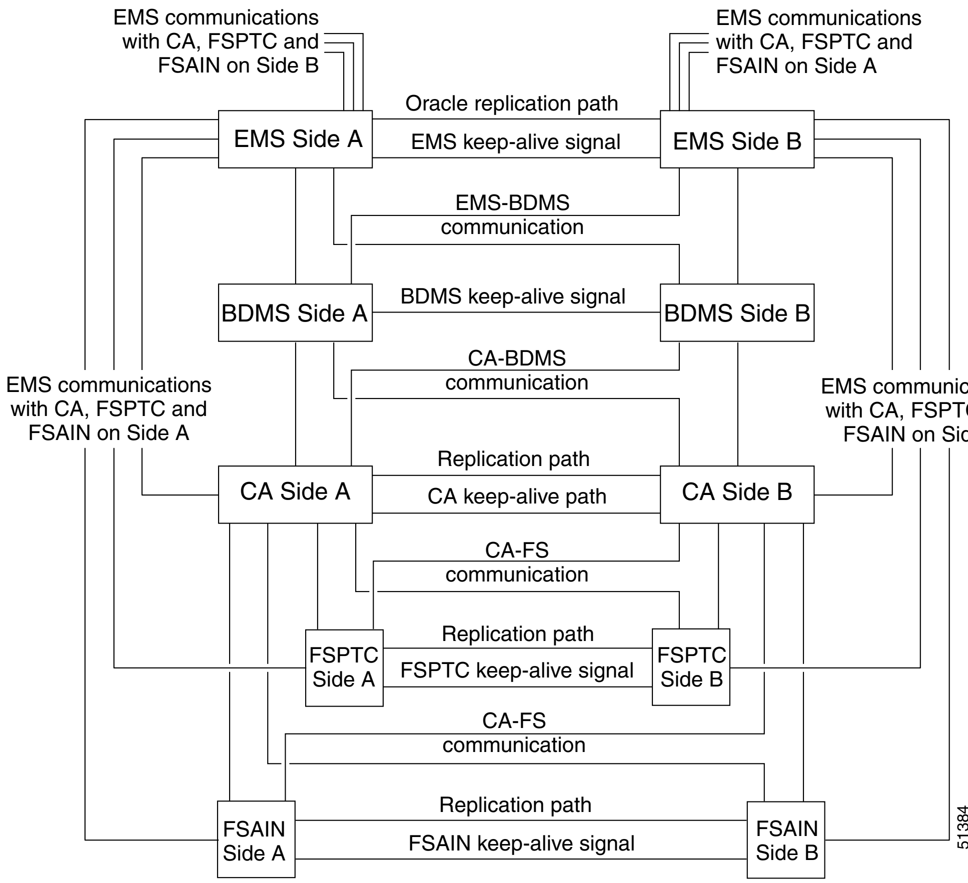
Active and Standby Configurations

Cisco BTS 10200 components are always deployed in a dual active/standby configuration, with the two sides running on separate computers (hosts). The active side of each component is backed up by a standby side on the other host. There is no traffic load sharing between the active and standby sides; the active side performs all of the call processing, and the standby does none. Each system has two types of hosts:

- One host with an EMS and bulk data management system (BDMS) together—The EMS and BDMS run as independent applications. The active side of each application is configured and running on one host system, and standby sides are configured and running on a mated host system. Each side maintains a keep-alive channel with the corresponding mate side. The keep-alive process on each side determines if the mate is faulty. If so, the standby side takes over and runs as active. An Oracle replication process runs between the two sides of the EMS to allow automatic database updates.
- One host with a CA, POTS/Centrex/Tandem FS, and AIN FS together—The CA, POTS/Centrex/Tandem FS and AIN FS run as independent applications. The active side of each application is configured and running on one host system, and standby sides are configured and running on a mated host system. Each side maintains a keep-alive/replication channel with the corresponding mate side. The keep-alive process on each side determines if the mate is faulty. If so, the standby side takes over and runs as active. A replication process runs between the two sides of the each component to allow automatic database updates.

Figure 6-1 shows the redundancy operations—communications, replication, and keep-alive signal—that occur between Side A and Side B of the Cisco BTS 10200.

Figure 6-1 Cisco BTS 10200 Redundancy Operations Between Side A and Side B Components



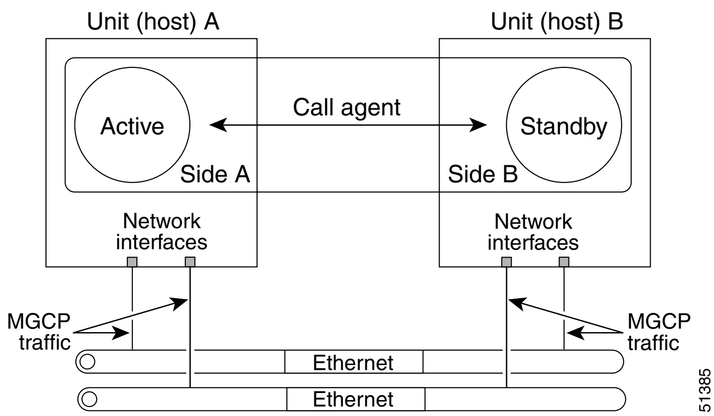
At installation, Side A is designated as primary, and Side B as secondary. In normal operation, the side designated primary runs in active mode, and the side designated secondary runs in standby mode. If there is a failure on the active side (or if the operator intentionally brings down the active side), the other side becomes active and takes over the traffic load. All stable calls continue to be processed without any loss of calls. There is no service outage, but during a switchover, transient calls can be impacted. When the side that failed is brought back in service, it remains in standby mode, and the system runs in normal duplex mode. A manual switchback (done during a maintenance window) may be needed to make Side A active again.

The operator can manually switch (force) either the primary or the secondary side to go active (FORCED_ACTIVE mode), which automatically forces the other side into FORCED_STANDBY mode.

Redundant LAN Connectivity

IP traffic between components of the Cisco BTS 10200 is divided into a number of paths that use separate network interfaces connected to different LANs. For each type of traffic, there are two separate network interfaces and two links connected to two different LANs. This ensures that no single-point failure in the LAN brings down the system. If one link fails, the other link can be used to carry the same traffic. This principle illustrated by the example shown in Figure 6-2.

Figure 6-2 Redundant LAN Connectivity Example



The complete set of virtual LANs (VLANs) is shown in Figure 6-3. When Side A and Side B of a Cisco BTS 10200 are located at separate sites to provide geographically distributed redundancy and survivability (GDRS), the VLANs are set up as shown in Figure 6-4 and Figure 6-5.

Figure 6-3 Cisco_BTS_10200 VLAN Links

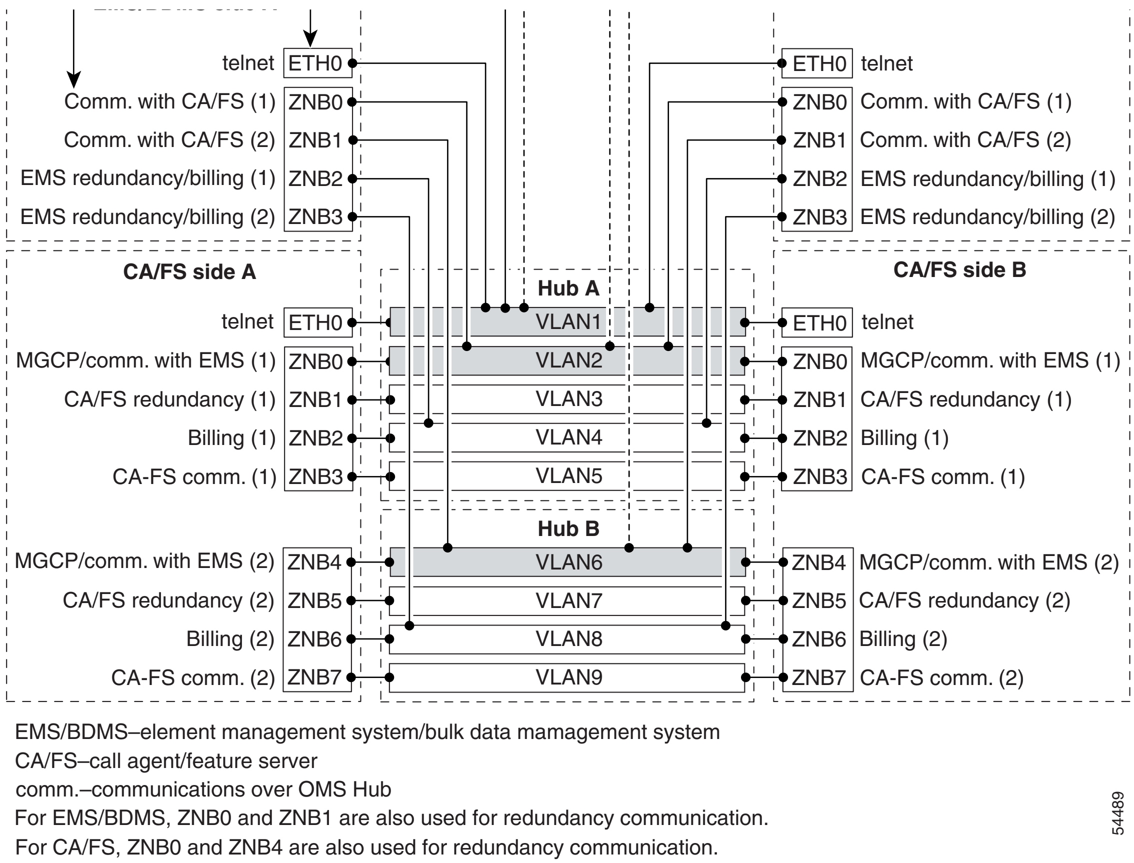


Figure 6-4 Cisco_BTS_10200 VLAN Links for GDRS (Site with Softswitch 123 Primary)

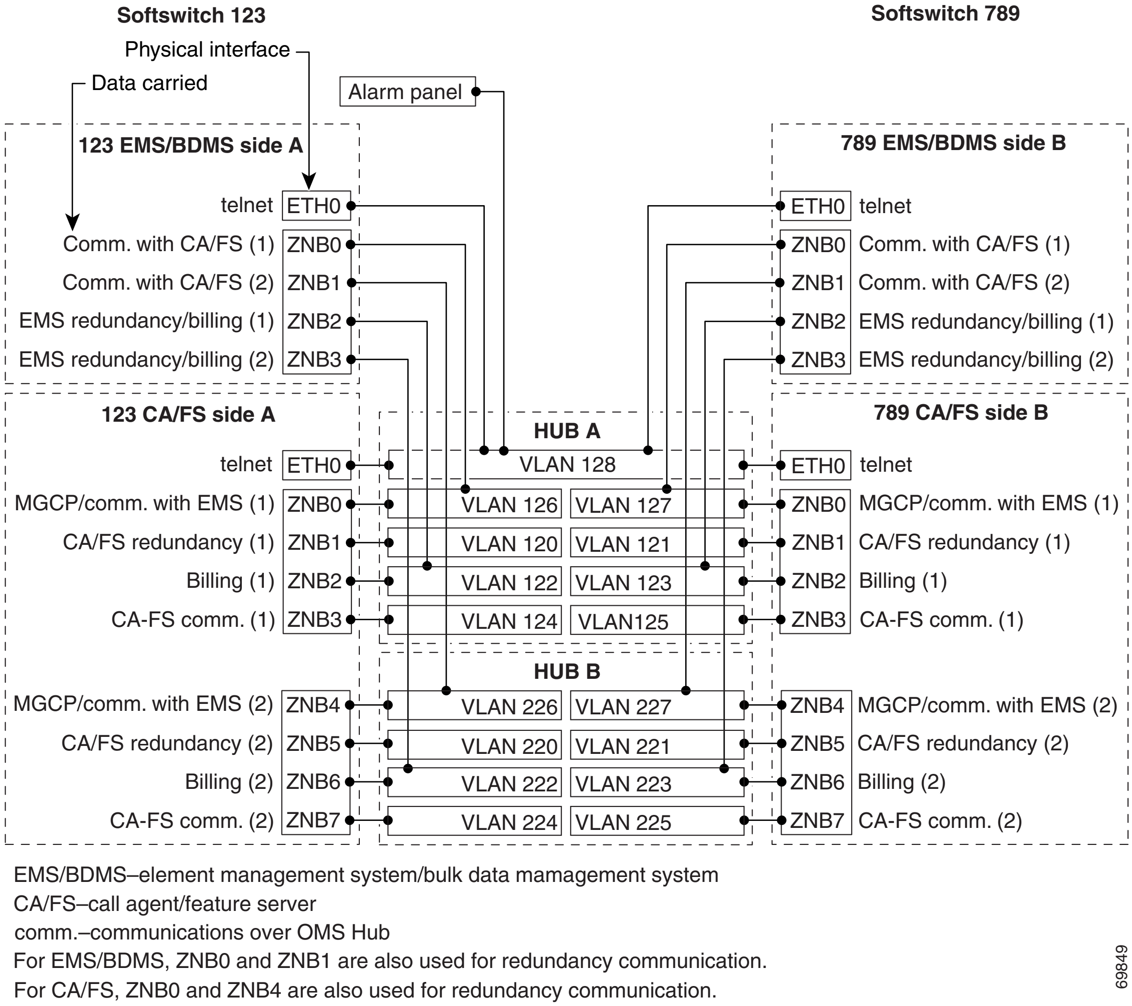
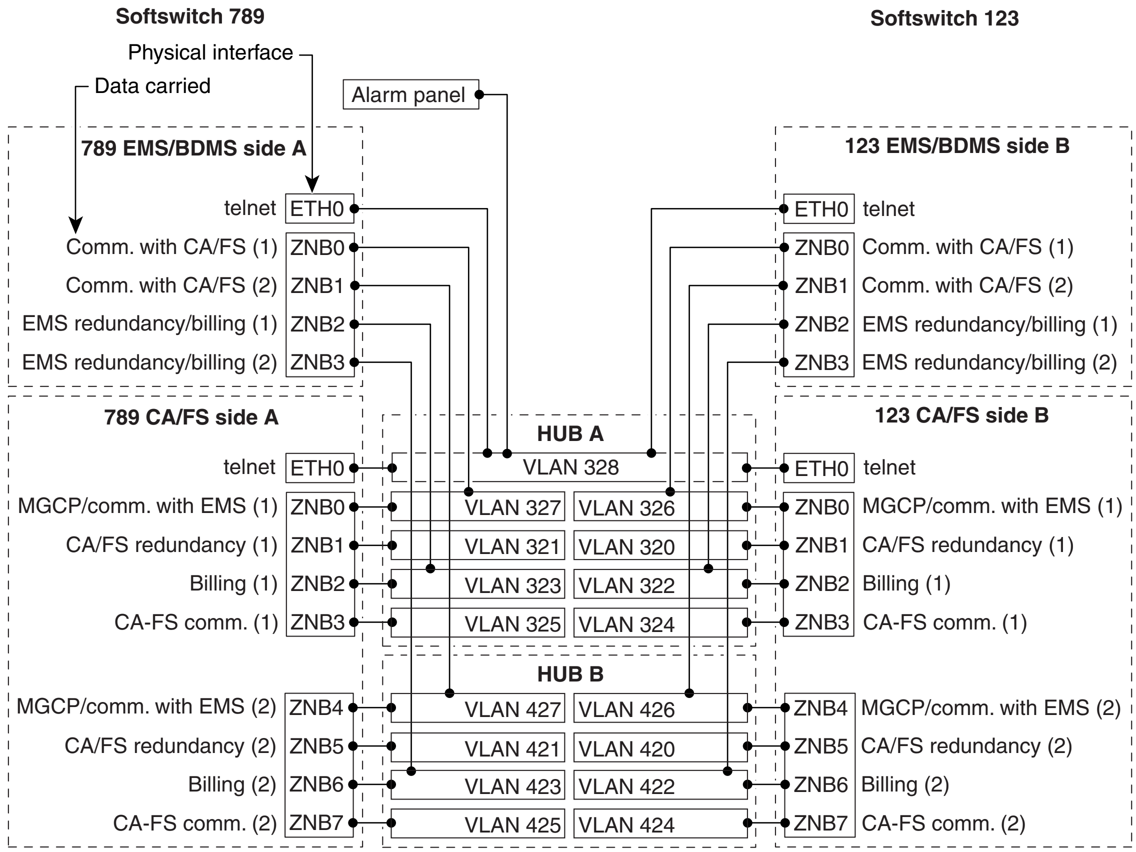


Figure 6-5 Cisco_BTS_10200 VLAN Links for GDRS (Site with Softswitch 789 Primary)



EMS/BDMS–element management system/bulk data mamagement system
CA/FS–call agent/feature server
comm.–communications over OMS Hub
For EMS/BDMS, ZNB0 and ZNB1 are also used for redundancy communication.
For CA/FS, ZNB0 and ZNB4 are also used for redundancy communication.

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Network Architecture for Data Replication

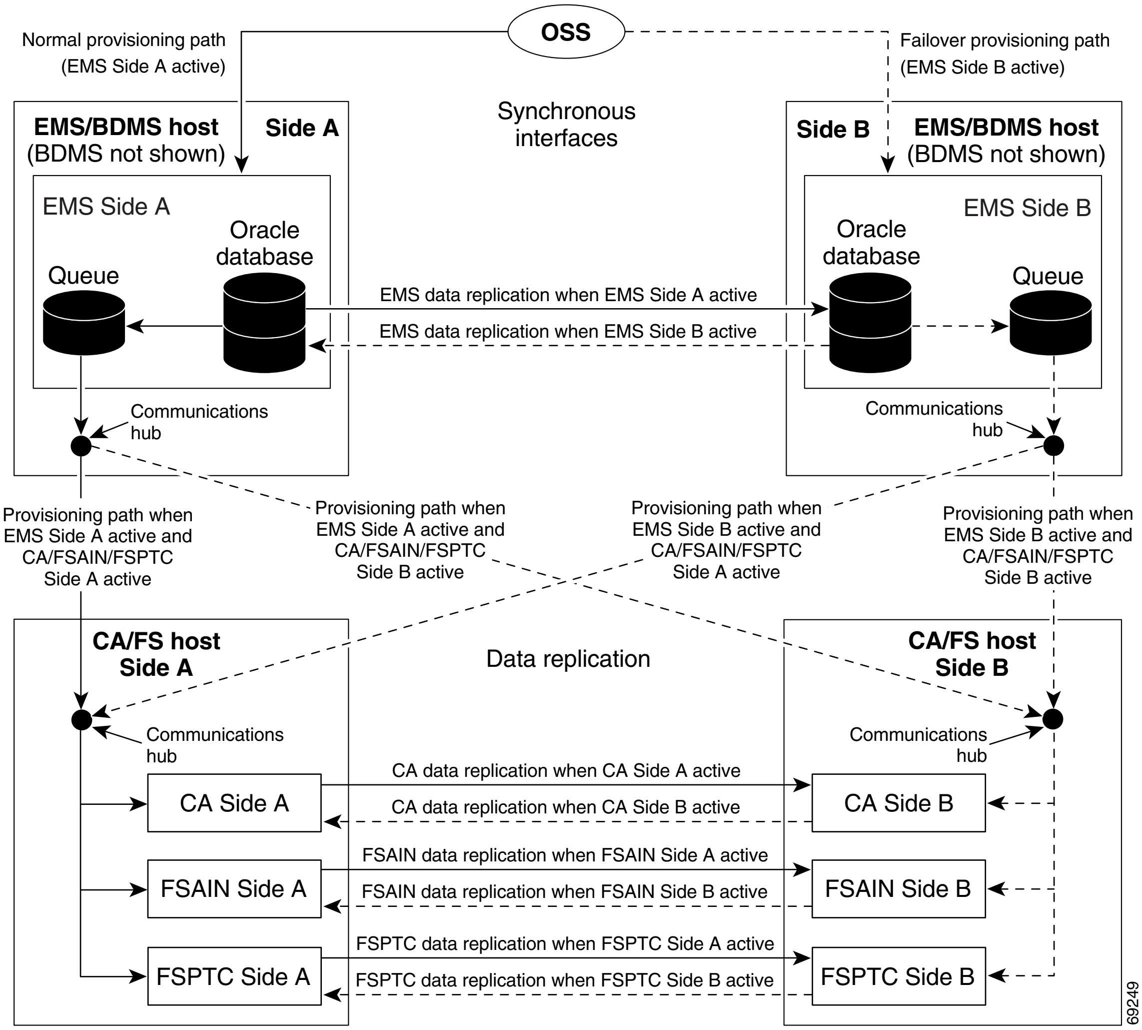
Call and feature data from the active side are replicated to the standby side at specific checkpoints of a call (when a call is answered, released, and so forth).

The following replications are carried out automatically by the system:

- Provisioned data is replicated from active EMS to standby EMS using ORACLE Multi-Master replication.
- Stable call data is replicated between active side and standby side of the CA.
- Stable feature data is replicated between active FS and standby FS.
- Billing data is replicated between active CA and standby CA.
- Provisioning data is replicated from active CA/FS to standby CA/FS.

The network architecture (communications and replication paths) for the EMS are shown in Figure 6-6. Similar information for the BDMS is shown in Figure 6-7.

Figure 6-6 Network Architecture for EMS Replication

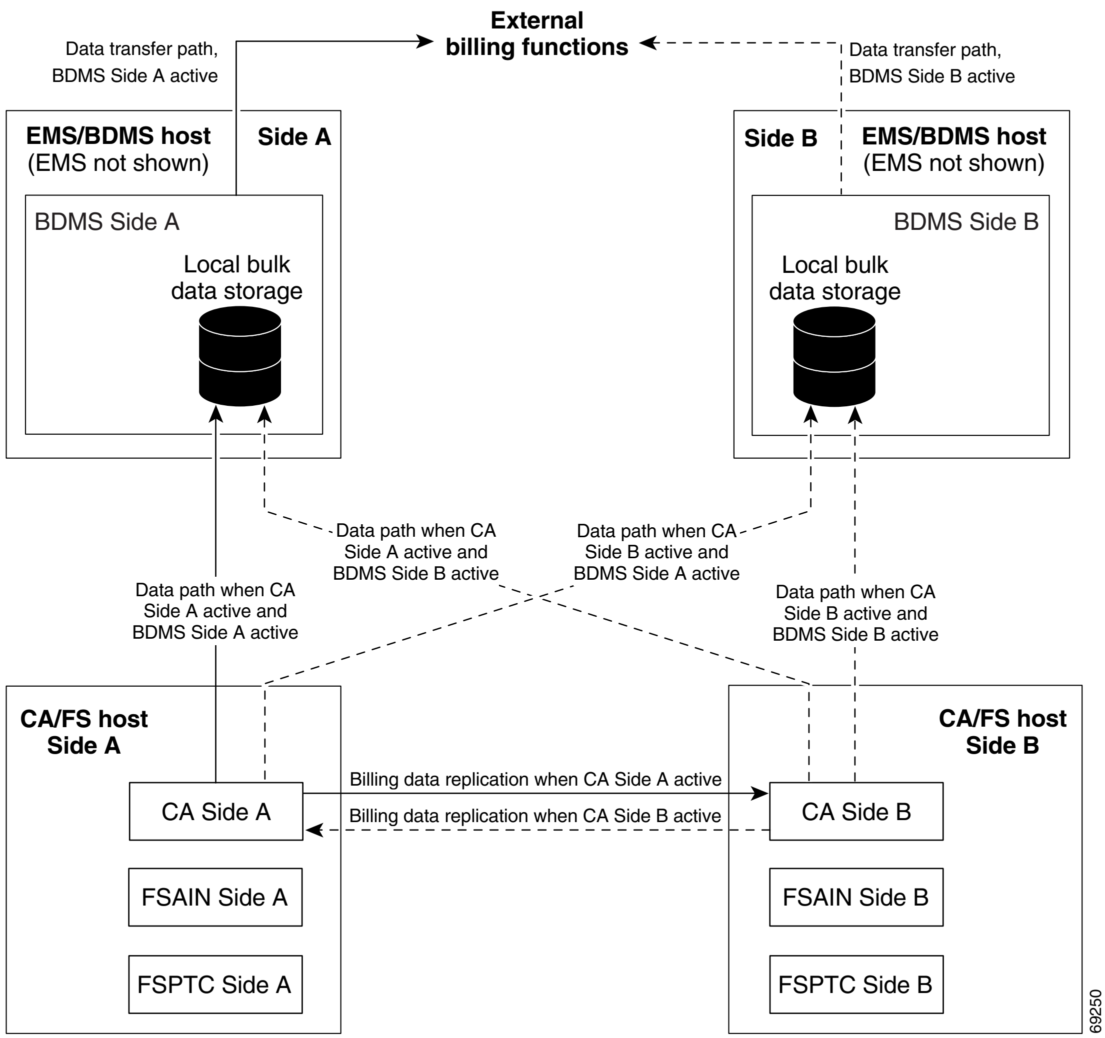


Note The EMS and BDMS components have two sides (Side A and Side B) running on different host machines. Each component pair operates independently. For example, EMS Side A might be standby while BDMS Side A is active. In that case, EMS Side B would be active and BDMS Side B would be standby.

The CA, FSAIN and FSPTC components have two sides (Side A and Side B) running on different machines. Each component pair operates independently from the other two component pairs. For example, CA Side A might be active, while FSAIN Side A is standby and FSPTC Side A is active. In that case, CA Side B would be standby, FSAIN Side B active, and FSPTC Side B standby.

The BDMS is not shown in Figure 6-6.

Figure 6-7 Network Architecture for BDMS Replication



Note The EMS and BDMS components have two sides (Side A and Side B) running on different host machines. Each component pair operates independently. For example, EMS Side A might be standby while BDMS Side A is active. In that case, EMS Side B would be active and BDMS Side B would be standby.

The EMS is not shown in Figure 6-7.

Fault Scenarios

Under normal conditions, the Cisco BTS 10200 performs internal communications and replication activities as shown in Figure 6-8. Alternative communications and replication paths are used if one side of an application becomes faulty. Examples of communications and replication paths under various fault scenarios are presented in Figure 6-9 through Figure 6-11.

Figure 6-8 Communications and Replication Paths under Normal Conditions

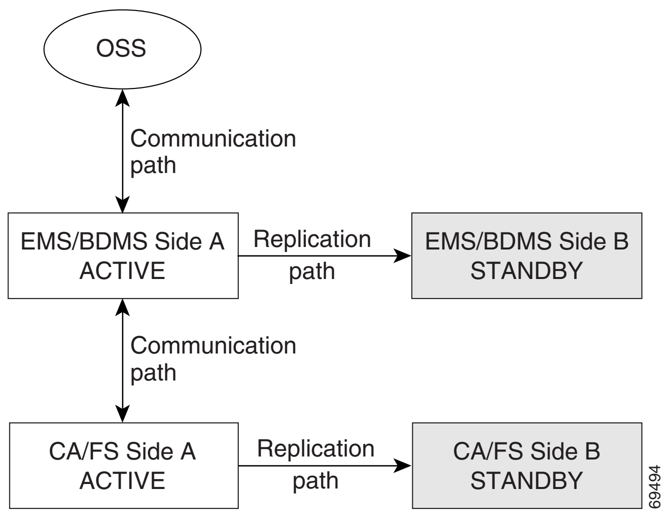


Figure 6-9 Fault Scenario—Side A EMS/BDMS Faulty

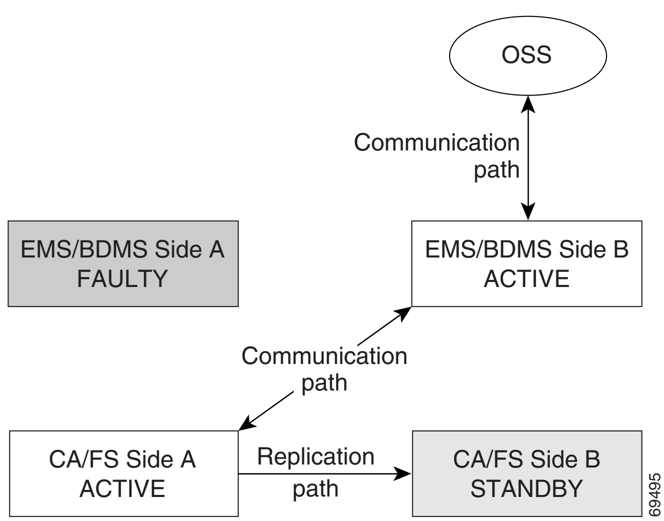


Figure 6-10 Fault Scenario—Side A CA/FS Faulty

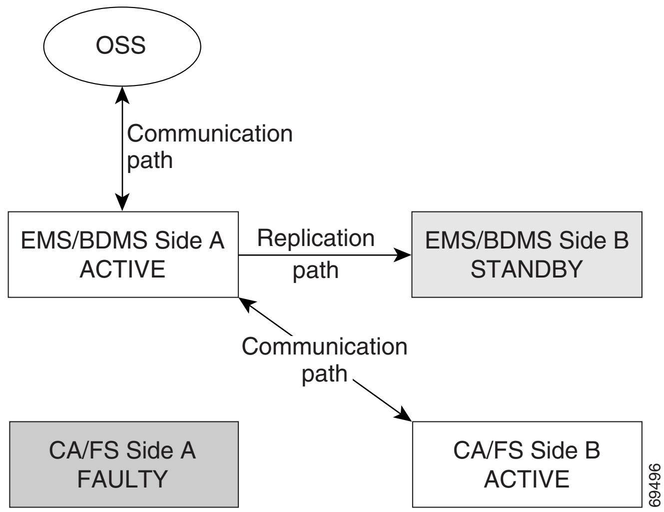
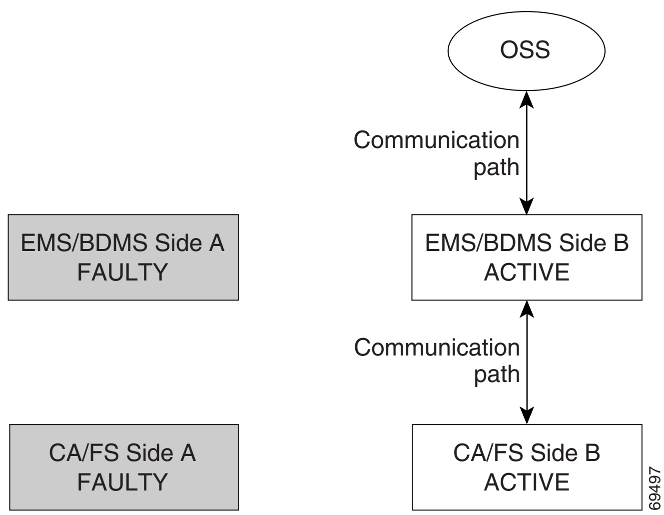


Figure 6-11 Fault Scenario—Side A EMS/BDMS and Side A CA/FS Faulty



DNS Lookup

The Cisco BTS 10200 does not store or use absolute IP addresses. Instead, it locates network connections by looking up domain names on the service provider domain name server (DNS). The service provider DNS translates the domain names into IP addresses. During software installation, the

Cisco BTS 10200 is configured with the data it needs to communicate with the service provider DNS. This configuration data is stored in the opticall.cfg file, and some critical domain names are also populated in the "etc/hosts" file in each host machine.

Caution The service provider DNS must be redundant. Otherwise, failure of one server could bring down the entire system.

Caution If your system is configured for geographically distributed redundancy and survivability (GDRS), the location of the IP connectivity broker (arbiter) is important. To give the best chance of avoiding dual failures, Cisco strongly recommends that you use a host outside the POPs of the two Cisco BTS 10200 locations for your IP connectivity broker. Otherwise a catastrophic outage, such as a fiber cut at the POP, may cause a dual outage. Only by selecting a separate point can POP failures be properly discriminated from internal failures.

Integrity of Provisioning Data and Billing Data

The Cisco BTS 10200 provisioning interfaces are:

- OSS-EMS interface
- EMS-CA interface
- EMS-FS interface

The Cisco BTS 10200 bulk data interfaces are:

- External billing systems-BDMS interface
- BDMS-CA interface

These interfaces require synchronization, redundancy, audits, backup, and recovery mechanisms to ensure data integrity. These mechanisms function as follows:

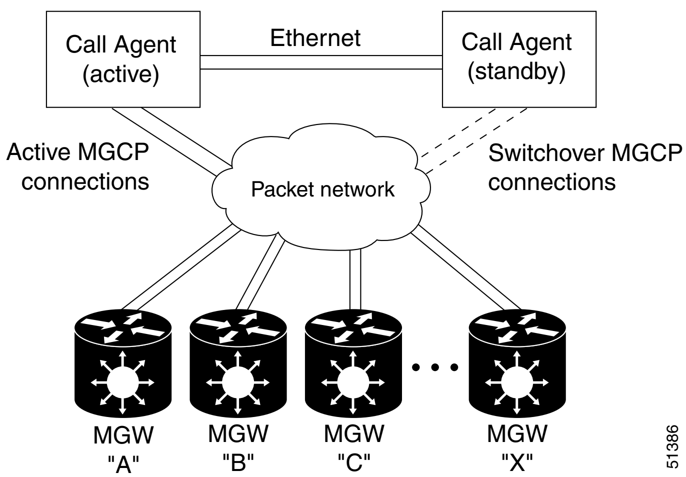
- Synchronization—The OSS-EMS interface normally operates in synchronous mode. The OSS sends transactions to the EMS for immediate execution, and normally does not queue failed transactions. However, the EMS-CA/FS interface operates in asynchronous mode. The EMS queues transactions that cannot be executed immediately in the CA/FS and delivers them later.
- Redundancy—The active and standby application sides of the EMS, CA, and FS operate in redundant mode. EMS redundancy uses the Oracle Multimaster tool for data replication. Oracle also provides server-enforced objects to maintain data integrity in the EMS, and performs database transaction control. For the CA and FS interfaces, an internal data replication function ensures consistency between the active and standby database contents.
- Audits—The audit mechanism checks the consistency between the active EMS database and the active CA/FS shared memory database. The audit detects loss of data integrity from communication failures, dual-system outages, software errors, or corruption of the shared memory.
- Database backup and recovery mechanisms—Backup and recovery mechanisms provide additional data integrity protection. The procedures for performing the backup and recovery are provided in the *Cisco BTS 10200 Softswitch Operations Manual*. The system administrator should refer to these procedures as needed:

- The system administrator should enable routine backup processes via Oracle crontab to archive the database contents. The backup files should also be transferred to a remote ftp site periodically.
- If the current database becomes corrupted or lost on both Side A and Side B EMS, the system administrator can recover and restore the data from the archives.

MGCP Connections Between CA and MGW

The Media Gateway Control Protocol (MGCP) connections between CA and media gateway (MGW) are shown in Figure 6-12. The active and standby sides of the CA operate as a pair, with the active side processing calls. Each MGW performs DNS lookup of the Side A and Side B CA for IP address resolution, so that it can communicate with the other side. Each MGW stores internally two IP addresses for each side of the CA (total of four IP addresses). Under normal conditions, the MGW communicates only with the active CA, using MGCP. If the active CA fails, the standby CA becomes active and establishes communication with the MGW.

Figure 6-12 MGCP Connections Between CA and MGW



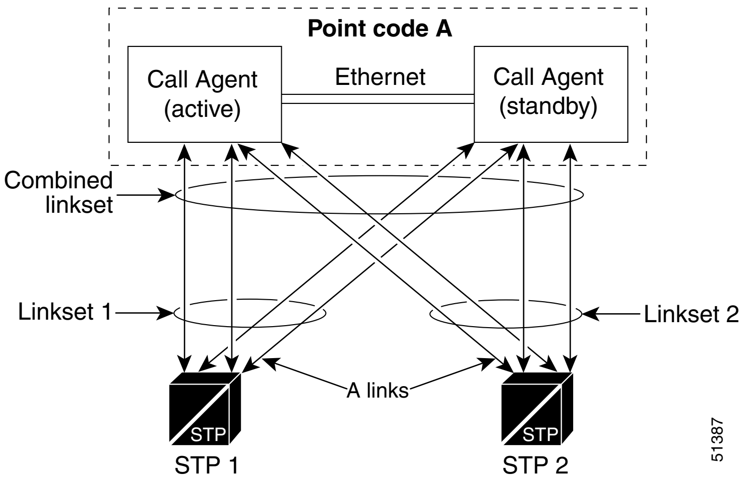
SS7 Connections Between CA and STP

The CA connects via SS7 A-links to a signal transfer point (STP) mated pair (shown as STP1 and STP2 in Figure 6-13). To the STP pair, the two sides of the CA appear as a single network element with a single point code. Therefore, all of the A-links are typically assigned to one combined linkset.

In the typical configuration, each CA terminates up to 8 A-links per linkset to an STP. The example in Figure 6-13 shows four links per linkset. The actual number of links should be based on capacity.

The Cisco BTS 10200 ensures SS7 redundancy via the Ulticom SS7 stack dual computing element (CE) configuration. Each side of the CA runs its own SS7 stack and the two sides communicate over redundant Ethernet links. SS7 messages received on links attached to the standby side are passed to the active side for call processing. If there is a complete failure of one side of the CA, the other side continues processing SS7 traffic from both STPs without service interruption.

Figure 6-13 SS7 Connection Redundancy Example



Note The SS7 A-links operate differently in the geographically distributed redundancy and survivability (GDRS) configuration of the Cisco BTS 10200. The Cisco BTS 10200 deactivates the A-links at the MTP2 level on the standby side of the CA and activates the A-links on the active side of the CA. The active side of the CA handles all of the SS7 signaling. If there is an automatic switchover (the active side of the CA goes down and the previously-standby CA becomes active), the Cisco BTS 10200 automatically activates all of the A-links on the newly active side and deactivates all of the A-links on the other CA side. When the CA side that was down is brought back in service, it remains in standby mode, and the A-links on the active CA continue to handle all SS7 signaling.

For additional information on the GDRS configuration, see Chapter 2, "Product Description."

CHAPTER 7

System Maintenance

Introduction

The Cisco BTS 10200 Softswitch provides a user interface for administering and monitoring resource states of the following resources:

- Subscribers
- Trunks
- Trunk groups (TG)
- Trunking gateways (TGW), which provide PSTN trunk interfaces (ISDN, CAS, and SS7)
- Devices that provide subscriber line interfaces:
  - Residential gateways (RGW)
  - Integrated access devices (IAD)
  - Cable modems

The resources are hierarchical:

- Allowed subscriber states depend upon the current MGW state.
- Allowed trunk states depend upon the current trunk group state, which in turn depends on the current TGW state.

There are two types of service states:

- Administrative—The state that the Cisco BTS 10200 operator has provisioned for the resource.
- Operational—The physical condition of the resource.

These two types of service states are independent of each other. This is illustrated with the following example.

> A Cisco BTS 10200 operator wants to place an MGW connection in service, and enters the command to do so. The administrative state is now In Service (INS). However, the link between the Cisco BTS 10200 and the MGW might be inoperable (cut, damaged, or placed out of service by the owner/operator of the MGW), or the MGW itself might be physically removed or placed out of service. Thus, the operational state of the MGW link is Out of Service (OOS). A status report of the MGW lists both the administrative state (INS) and operational state (OOS) of the link to the MGW.

Administrative Service States

The operator can enter commands to query and change the administrative service state of network resources. Table 7-1 lists the administrative (operator-controlled) service states. For a detailed description of the specific status and control commands to use, see the *Cisco BTS 10200 Softswitch Operations Manual*.

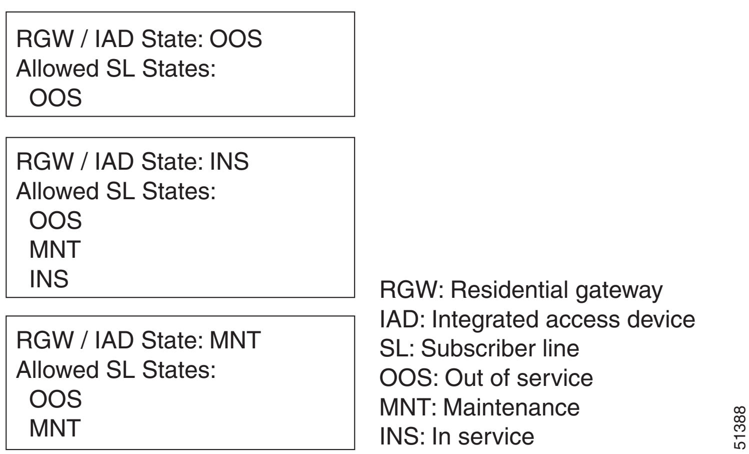
Table 7-1 Administrative Service States

Service State	Description
In Service (INS)	The resource is deployed and services can be provided to network customers.
Out of Service (OOS)	The resource is not deployed and services cannot be provided to network customers.
Maintenance (MNT)	The resource is deployed and placed in this state for testing, diagnostic tests, or periodic maintenance. Services cannot be provided to network customers.

Subscriber Line Maintenance

Individual subscriber lines can be placed into any of the three administrative service states, INS, OOS and MNT. The relationship between subscriber line state and the RGW state is provided in Figure 7-1.

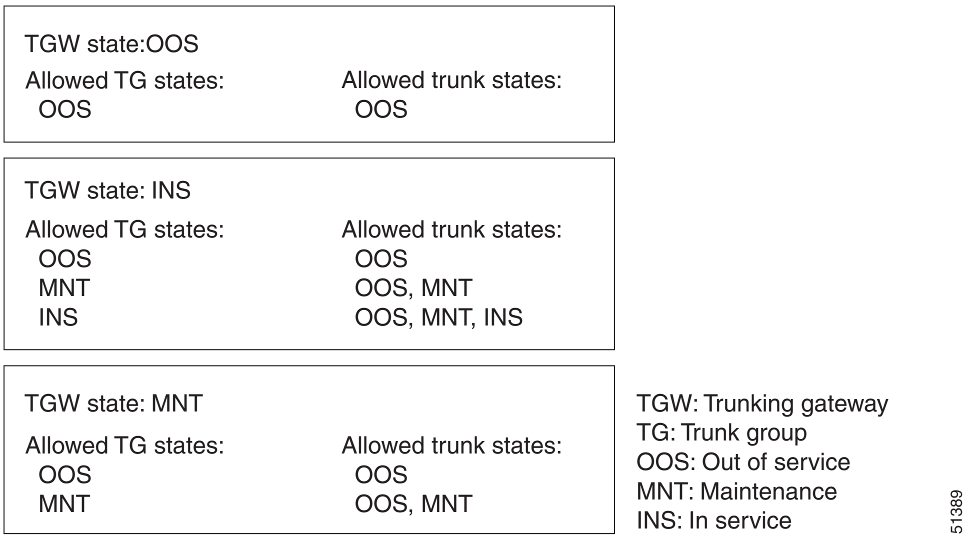
Figure 7-1 Subscriber Line Service States



Trunk and Trunk Group Maintenance

Individual trunks and trunk groups can be placed into any of the three administrative service states, INS, OOS, and MNT. The relationship between trunk/trunk group state and the TGW state is provided in Figure 7-2.

Figure 7-2 Trunk and Trunking Gateway Service States



CHAPTER 8

Events and Alarms

Event Levels

The Cisco BTS 10200 Softswitch sends system status and trouble information to the operator interface. Each event is tagged with one of the following designations:

- Critical—Service (call processing) is severely affected
- Major—Significant loss of service
- Minor—Minimal or momentary loss of service
- Warning—Information providing cautionary advice about a potential service impact
- Info—Information only, indicating various stages of system operation

Alarm reports are the subset of event reports that include critical, major and minor.

Note See the *Cisco BTS 10200 Softswitch Operations Manual* for a detailed list of events, alarms, and threshold values.

Alarm Functions

Critical, major, and minor events are all alarmed. Cisco BTS 10200 alarm features include:

- Alarm log
- Display of current active alarm status
- Operator alarm-acknowledgment capability
- Alarm-inhibit capabilities that allow the operator to mask certain noncritical alarms
- Operator-created alarm filters categorize the alarms logged into the file

Alarms can be generated in any of the Cisco BTS 10200 components—EMS, BDMS, CA, FSAIN, or FSPTC. The alarm information is forwarded from the affected component to the event management subsystem in the EMS.

Event Management

The event management subsystem in the EMS supports the system operator by providing these services:

- Formats event information
- Supports report types: call processing, configuration, database, signaling, maintenance, OSS, billing, statistics, and security
- Provides the ability to provision report parameters
- Provides the ability to view summary reports and execute queries
- Provides the ability to view reports in near real-time on demand
- Provides report queries based on time, type, or severity level
- Manages and distributes event indications on demand to subscribed adapters (CLI, SNMP, and so forth)
- Formats system reports, including faults and informational messages

CHAPTER 9

Call Agent Component Description

The call agent (CA) provides signaling and call processing (call setup and teardown) for the Cisco BTS 10200 Softswitch. The call state machine is based upon the ITU-T standard capability set (CS-2). Figure 9-1 and Figure 9-2 show the originating and terminating state machines implemented in the CA.

Figure 9-1 Originating Basic Call State Machine—Based on ITU-T CS-2

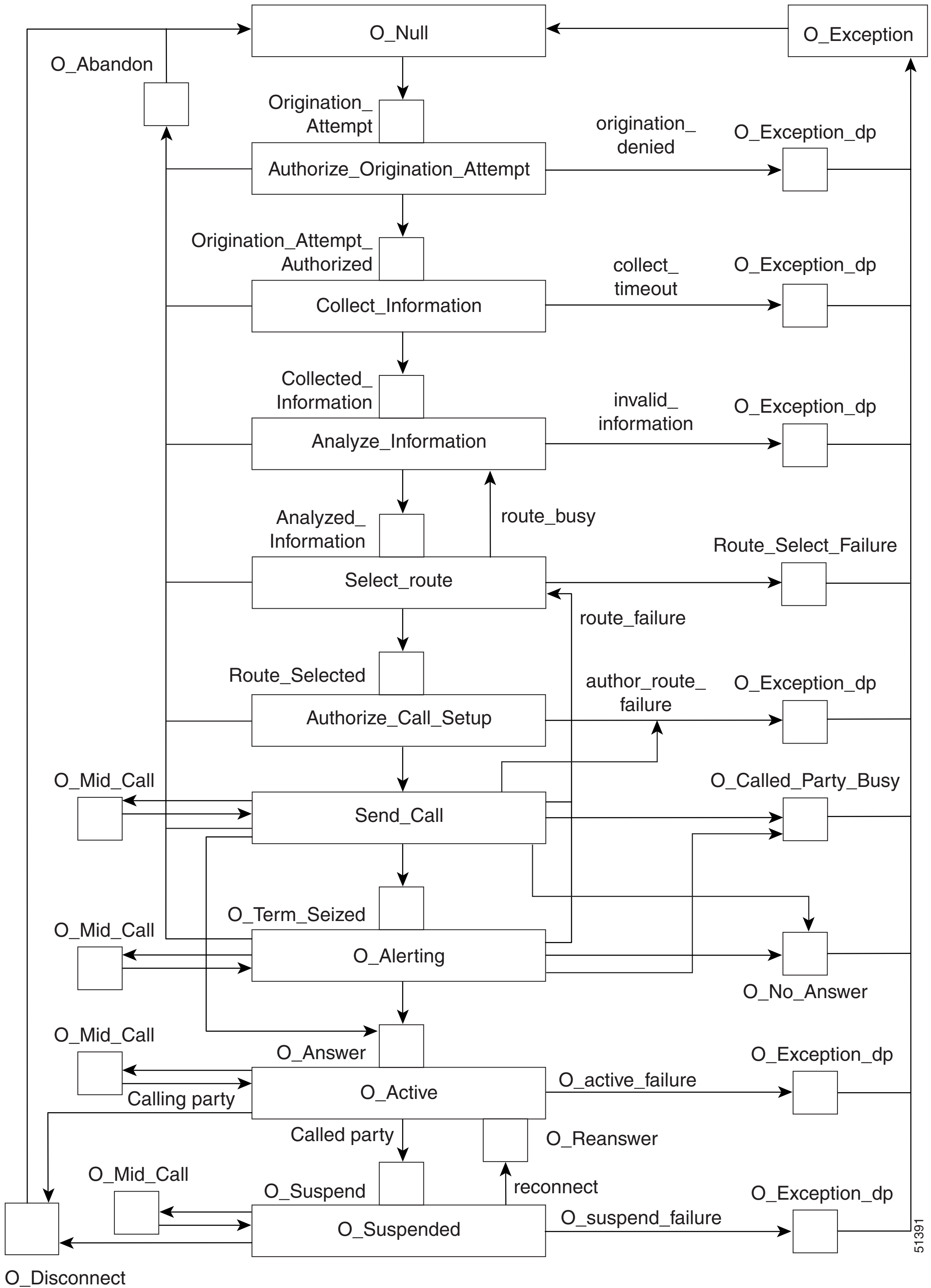
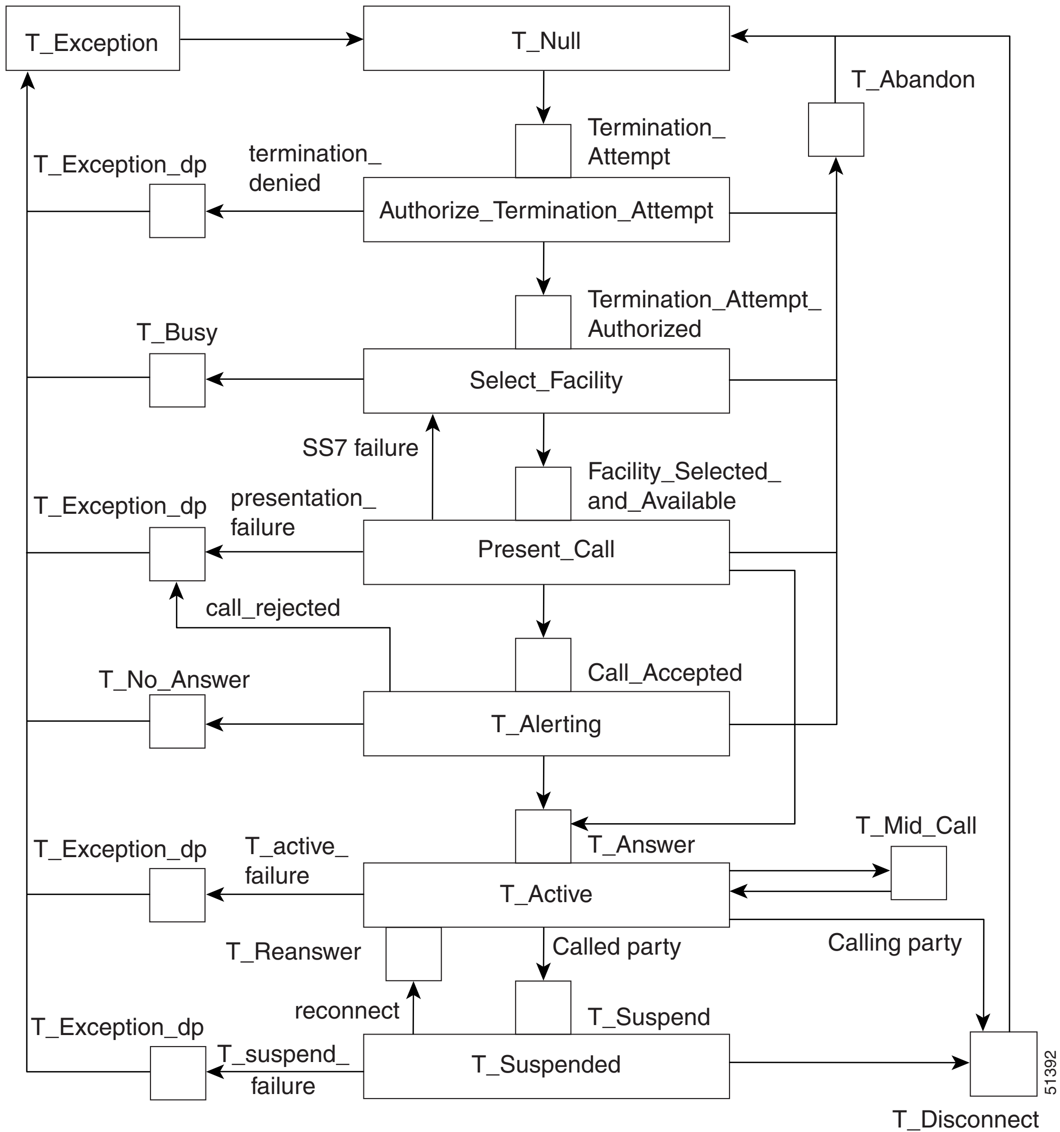


Figure 9-2 Terminating Basic Call State Machine—Based on ITU-T CS-2



Signaling

Signaling features are discussed in the following sections.

Overview

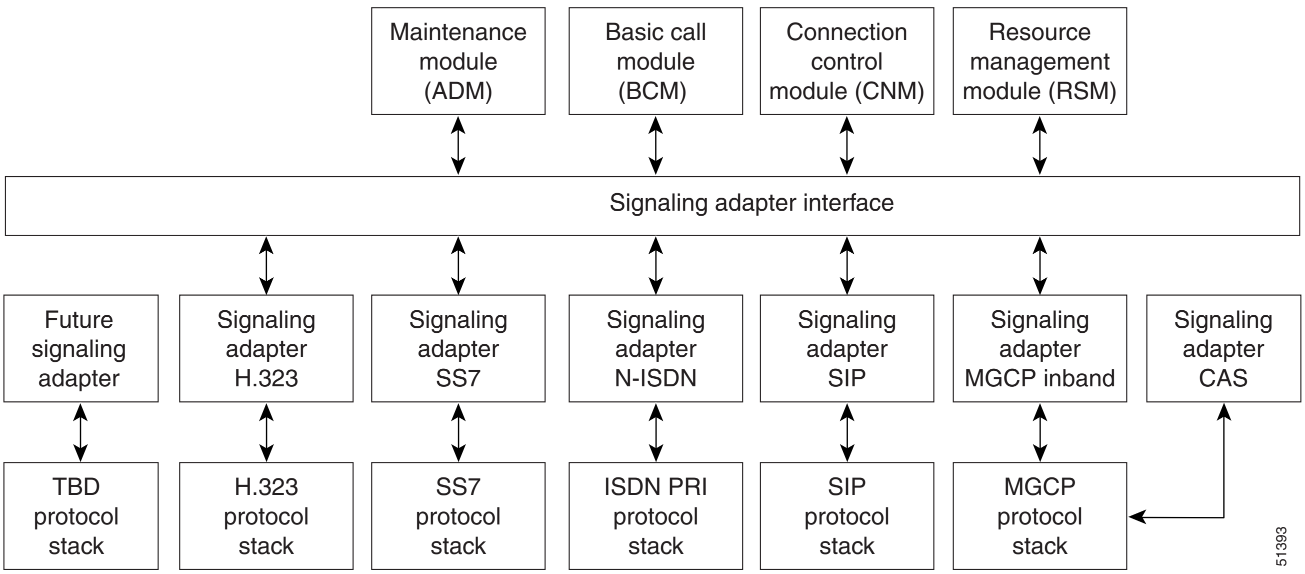
The CA provides monitoring and control of external NEs. It connects to multiple networks—using several types of signaling systems—via the signaling adapter interface (SAI). See Figure 9-3. The SAI converts incoming and outgoing signaling to and from the standard internal format of the CA. This interface allows the CA to connect to multiple networks and exchange signaling messages to set up, tear down and transfer calls.

The types of external signaling supported are

- CAS
- SS7
- ISDN (PRI)
- SIP and SIP-T
- MGCP
- H.323

Note The H323 feature flag is set to *off* in this release.

Figure 9-3 Signaling Adapter Architecture



Signaling Adapter Functions

The signaling adapters perform the following functions:

- Provide uniform primitives (signaling indications) for all interactions between different protocol stacks and the CA modules.
- Provide uniform data structures containing common information elements from different signaling protocols.
- Provide call control primitives for exchanging all call signaling messages between CA and the signaling network.
- Provide maintenance primitives for signaling link hardware maintenance and signaling protocol stack provisioning.

The individual signaling subsystems are described in the sections that follow.

SS7 Subsystem

The following acronyms are used in this section:

- ISUP—ISDN user part
- MTP—Message transfer part
- TCAP—Transaction capabilities application part
- AIN—Advanced intelligent network
- CNAM—Calling name delivery
- DPC—Destination point code
- LIDB—Line information database
- OPC—Originating point code
- SCP—Service control point
- SSP—Service switching point

The SS7 subsystem in the standard Cisco BTS 10200 Softswitch has the following maximum capabilities. Note that the actual values for some parameters depend upon the levels for which the particular Cisco BTS 10200 Softswitch is licensed.

- Four channelized T1 physical connections
- SS7 MTP
  - One OPC
  - 127 linksets
  - 16 links
  - 3,500 DPCs
  - 8 possible routesets per DPC
  - 8 combined linksets
  - 16 links per linkset
  - 16 unique cluster tables
  - 32 unique member tables
- SS7 ISUP
  - 100,000 SS7 trunks
  - 16,384 CICs per point code
- ANSI TCAP messages to interface CNAM LIDBs
- ANSI TCAP messages to interface toll-free (800) databases in the SCP
- AIN 0.1 messages to interface LNP and toll-free (800) databases

Note that the user can choose the desired toll-free (800) application—AIN 0.1 or TCAP—in the Cisco BTS 10200 Softswitch by editing configuration parameters in the opticall.cfg file at time of Cisco BTS 10200 Softswitch software installation. The available choices are

- AIN 0.1 procedures for SSP/SCP interaction based on Telcordia GR-1284/1285. Info_Analyzed and Analyze_Route messages are exchanged (in that order) between SSP and SCP.
- TCAP procedures for services based on Telcordia GR-533 procedures.

The following subsystem numbers and translation types are selectable at the time of Cisco BTS 10200 Softswitch software installation. They appear as parameters in the opticall.cfg file (see Table 9-1).

Note These parameters can be modified using the CA-CONFIG table in the Cisco BTS 10200 Softswitch database. See the *Cisco BTS 10200 Softswitch CLI Reference Guide* for provisioning details.

Table 9-1 Subsystem Number and Translation Type Parameters for opticall.cfg

Parameter	Description	Default Value
ssntoll[1]	Subsystem number for toll-free (800) service (248 is typically used for AIN 0.1, and 254 for TCAP)	254
ssnlnp	Subsystem number for LNP (local number portability) service	247
ssncnam	Subsystem number for CNAM service	232
tttoll[1]	Translation type for toll-free (800) service (8 is typically used for AIN 0.1, and 254 for TCAP)	254
ttlnp	Translation type for LNP service	11
ttcnam	Translation type for CNAM service	5
msgtype	Type of message used for sending queries to the SCP and receiving responses from the SCP. (Allowed msgtype values are AIN01 and TCAP)	TCAP

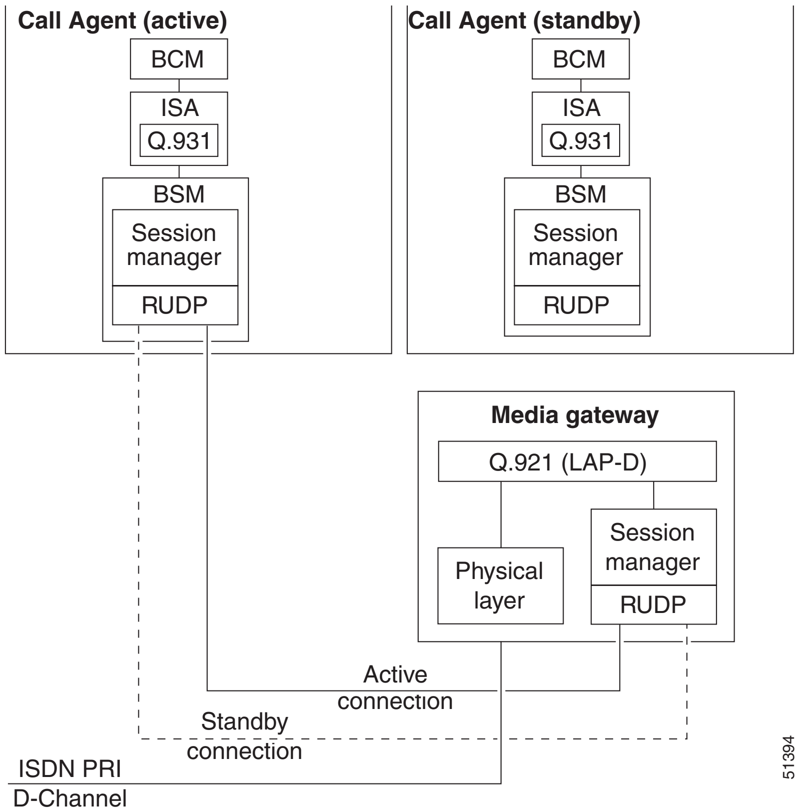
1. During installation of Cisco BTS 10200 Softswitch software, the appropriate ssntoll and tttoll parameter values can be entered to support either AIN01 or TCAP Toll (800) services.

ISDN Subsystem

The ISDN Adapter subsystem includes:

- Q.931 network side only (no user side)
- ISDN PRI - NI2
- ISDN backhaul is supported with Session Manager/R-UDP Protocols (see Figure 9-4).

Figure 9-4 R-UDP Based Backhaul Design



Note In Figure 9-4:
R-UDP = Reliable User Datagram Protocol, a Cisco Systems proprietary signaling backhaul protocol.
BCM = Basic call module in the CA
ISA = ISDN signaling adapter in the CA

The Q.931 Network Layer interfaces with the session manager. The session manager provides two basic functions: the ability to manage sessions in a group, and the ability to manage a session set in a fault-tolerant configuration. The session manager is based on the client server model. The MGW implements the client side and the Cisco BTS 10200 Softswitch implements the server side. The session manager client and server both send messages: start, stop, active, and standby.

ISDN primary rate interface (PRI) signaling is transported using backhaul R-UDP protocol between the access gateway and the CA. Each ISDN trunk group has up to 32 trunks, each with 24 channels. Each trunk is identified by its gateway, slot, port, and channel designations.

SIP Subsystem

The SIP subsystem includes:

- SIP-T for CA to CA communications
- SIP user agent (UA) functionality

MGCP and CAS Subsystem

The MGCP subsystem includes:

- MGCP
- VoIP bearer path control
- VoATM bearer path control
- VoATM extensions for MGCP/ATM
- VoATM bearer path control supports ATM extensions for MGCP for AAL1 SVC circuits.
- VoATM bearer path control supports ATM extensions for MGCP for AAL2 SVC circuits.
- Support for Resource Reservation Protocol (RSVP) on connection.

The following applications of CAS/MGCP are supported:

- Operator services interface and a PSAP (911) systems interface.
- Legacy operator services system interface via MF/T1 trunks.
- PBX interfaces

H.323 Subsystem

The H.323 adapter subsystem includes:

- H.225/RAS signaling control
- H.245 connection control

Note The H323 feature flag is set to *off* in this release.

CHAPTER 10

Feature Server Component Description

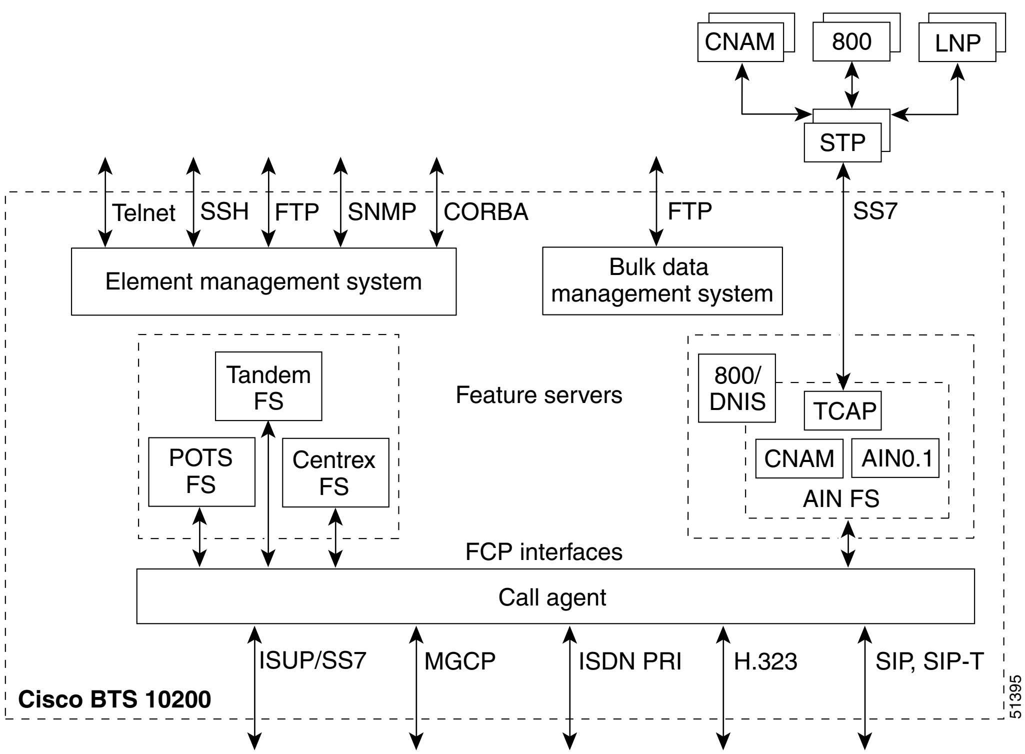
Feature Server Architecture

There are two different types of feature servers (FSs) in the Cisco BTS 10200 Softswitch.

- POTS/Centrex/Tandem Feature Server (FSPTC)
- Advanced Intelligent Network Feature Server (FSAIN)

The purpose of the FSs is to provide access to features through a well defined interface. The Cisco BTS 10200 architecture logically separates the FSs (which provide feature control) from the CA (which provides call control) with a clear interface—Feature Control Protocol (FCP)—defined between them. The FSs provide support for POTS, Centrex, AIN, 800 service, and other enhanced services. The FSs are colocated on the same machine as the CA. The basic architecture is shown in Figure 10-1.

Figure 10-1 Feature Servers in the Cisco BTS 10200 Softswitch Architecture



Features and Services

Table 10-1 and Table 10-2 show the types of services and features provided by the Cisco BTS 10200 and identify the FS where each service/feature data set is stored.

Table 10-1 Service Types and Features on POTS/Centrex/Tandem FS

Service Type	Feature Name
Class of Service Restrictions	Number Blocking Restrictions based on category of service: • 900 Blocking • Directory Assistance Blocking • International Blocking • 976 Blocking • National Black/White List • International Black/White List • Casual Black/White List • Account Codes • Authorization Codes
Screening features	Selective Call Forwarding Selective Call Acceptance Selective Call Rejection, Call Block
	Distinctive Ringing/Call Waiting
POTS features	Analog DID for PBX (FXO) DOD for PBX
	Multiple Directory Numbers (Teen Service)

Table 10-1 Service Types and Features on POTS/Centrex/Tandem FS (continued)

Service Type	Feature Name
Common features (POTS and Centrex)	Call Forwarding Unconditional Remote Activation of Call Forwarding Remote Call Forwarding
	Call Forwarding On Busy
	Call Forwarding No Answer
	Calling Number Delivery Blocking Calling Name Delivery Blocking Calling Identity Delivery and Suppression
	Calling Number Delivery
	Calling Name Delivery (No External Query)
	Calling Identity Delivery on Call Waiting
	Anonymous Call Rejection
	Automatic Callback (Repeat Dialing)
	Automatic Recall (Call Return)
	Call Block (Reject Caller)
	Call Waiting
	Cancel Call Waiting
	Customer-Originated Trace
	Do Not Disturb
	Hotline Service Warmline Service
	Interactive Voice Response Functions
	Multiline Hunt Group (MLHG)
	Speed Call (1-digit and 2-digit)
	Three-way Calling
	Usage-Sensitive Three-Way Calling
	Visual Message Waiting Indicator

Table 10-1 Service Types and Features on POTS/Centrex/Tandem FS (continued)

Service Type	Feature Name
Basic Centrex features	Customized Dialing Plan
	Intercom Dialing
	Semi/Fully Restricted Lines
	Direct Inward Dialing (DID) Distinctive Alerting/Call Waiting Indication on DID Direct Outward Dialing (DOD)
	Incoming/Outgoing Simulated Facility Group
	Call Transfer
	Call Hold
	Call Park and Call Retrieve Directed Call Pickup (With and Without Barge-in
	Group Speed Call
Tandem features	ANI Screening

Table 10-2 Service Types and Features on AIN FS

Service Type	Feature Name
Database services (AIN 0.1)	Local Number Portability (LNP)
	External 800 Query[1]
	CNAM[2] Query
Database services (Local)	Local 800 Database
	DNIS[3]

1. An external 800 query can be based on either AIN0.1 or GR-533 (National TCAP).
2. CNAM = Calling name delivery
3. DNIS = Dialed number identification service

CA to FS Process

An FS is invoked from a detection point (DP) in the CA (see Table 9-1 for a list of DPs). In each DP, the CA checks if any triggers are armed. If a trigger is armed, the CA checks if the trigger applies to the subscriber, group, or office (in that order). If the trigger is applicable, the CA invokes the FS associated with that trigger. Examples of these triggers are illustrated in Figure 9-1 and Figure 9-2. The Cisco BTS 10200 call processing mechanisms are based on the ITU-T CS-2 call model.

Automatic Call Gap Function

The automatic call gap (ACG) function is supported for an AIN FS connected to a service control point (SCP). When an SCP sends a message to the AIN FS regarding the allowed query rate, the Cisco BTS 10200 adjusts its query rate accordingly.

EMS/BDMS Component Description

Introduction

The element management system (EMS) and the bulk data management system (BDMS) are separate applications colocated on a single host. The EMS manages the Cisco BTS 10200 Softswitch components and provides operations, administration, management, and provisioning (OAM&P) interfaces for monitoring and control. The management functions of the EMS are partitioned along the guidelines described by the Telecommunications Management Network (TMN) standards. This facilitates future migration to these standards with minimal impact to the existing architecture. The BDMS manages the billing functions of the Cisco BTS 10200.

The EMS is described in the following sections:

- "EMS OAM&P Interfaces" section on page 11-1
- "Hardware Monitoring Subsystem" section on page 11-3
- "Traffic Management Subsystem" section on page 11-4
- "Fault Management Subsystem" section on page 11-5
- "Event Management Subsystem" section on page 11-5
- "Queuing and Audit Management Subsystem" section on page 11-6
- "Configuration Management Subsystem (Status and Control)" section on page 11-6
- "Security Management Subsystem" section on page 11-7

The BDMS system is described in the "Billing Data and Bulk Data Management System" section on page 11-7.

EMS OAM&P Interfaces

The EMS provides a flexible mechanism to transport information over any protocol to any external device. The EMS interface design takes into account that each carrier has its own unique set of Operations Support Systems (OSS). The EMS provides a decoupling layer between the external protocols used within the service provider network and the internal protocols of the FS and CA. The core Cisco BTS 10200 system does not need to interpret the specific data formats used by the other carrier network elements.

Local operators on a workstation or PC use a command line interface (CLI) or a menu assisted command (MAC) interface to communicate with the EMS. Sessions can be either interactive or batch mode:

- Interactive sessions can use either of the following login methods:
  - Secure shell (SSH)
  - Telnet
- Batch mode (bulk provisioning) using file transfer protocol (FTP).

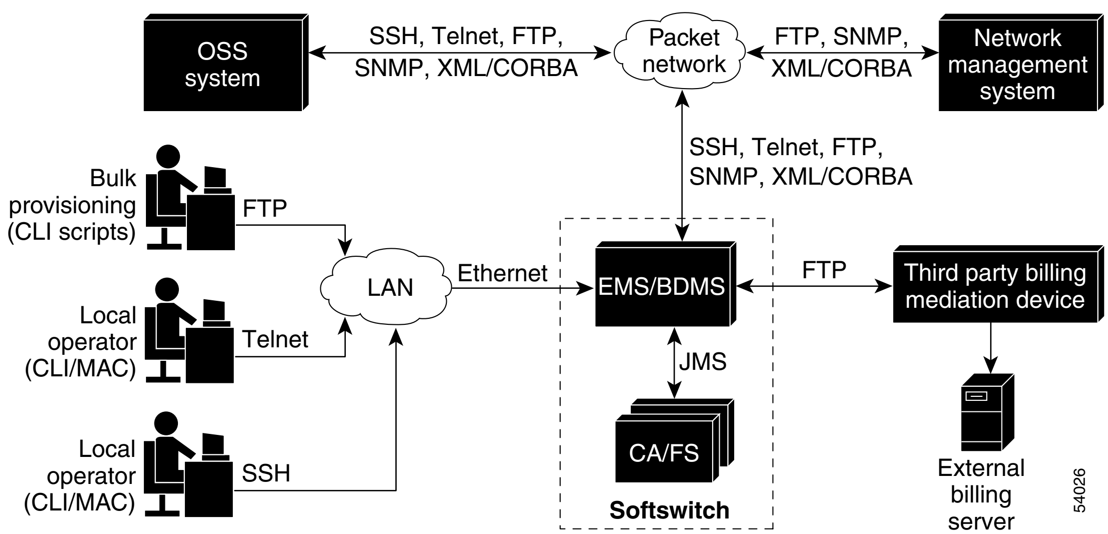
Note During software installation at the factory, a user-access parameter is set according to the preference indicated by the service provider. The options are SSH only, or SSH and Telnet.

Note For additional information on security, see the "Security Management Subsystem" section on page 11-7

The database holds up to 100 unique user accounts, and up to 16 user sessions can be active at any time.

Figure 11-1 shows how the Cisco BTS 10200 fits into a carrier network and coordinates communication with service provider operators and billing centers. The service provider operations personnel and systems communicate with the EMS. The service provider billing systems communicate with the BDMS.

Figure 11-1 Service Provider Communications with EMS and BDMS



Hardware Monitoring Subsystem

The hardware monitoring subsystem monitors the CPUs, memory consumption, disk, and disk control utilization, and returns information and alarms as appropriate. The monitored items are as follows

- Individual CPU(s) on a physical node
  - Percentage of CPU idle
  - Percentage of CPU in system mode
  - Percentage of CPU in user mode
  - Percentage of CPU in blocked I/O
- Memory consumption
  - Total real memory
  - Free memory available
  - Total swap space
  - Free swap space available
- Disk and disk control utilization on a physical disk and controller level. This measurement is dependent on the driver support supplied for the SCSI controller from the vendor. The hardware monitoring subsystem reports the following information for disk and disk control utilization.

  Note The device names follow those of a Solaris kernel. These vary from device to device. A physically understandable mapping of these devices may be required.

  - Utilization of disk devices by disk partition
  - Transfer rates, transactions per second (TPS) and hard or soft errors.

The hardware monitoring subsystem tracks the five most heavily used processes in the system and generates alarm reports on devices and facilities that exceed their threshold settings. Alarms are generated under the following conditions.

Note Threshold settings in the list below are examples only.

- A process exceeds 70 percent of the CPU.
- The CPU is over 90 percent busy (10 percent idle).
- The load average exceeds 3 for at least a five-minute sample.
- Memory is 95 percent exhausted and swap is over 50 percent consumed. This indicates that the system is spending excessive time in paging virtual memory.
- Any disk reports utilization by partition greater than 70 percent.

Traffic Management Subsystem

This section describes the general traffic and performance measurements processed by the Cisco BTS 10200. This information is used by service providers to integrate their measurement collection systems with the EMS.

The Traffic Management Subsystem provides the following functions:

- Collection of all statistics in 15-minute intervals
- Saves 48 hours of statistical data in 15-minute increments in persistent store
- Provides summary report display for the past 24 hour period
- Permits on-demand report queries keyed by collection interval
- Issues events as appropriate

Types of Counters

The following types of counters are collected on the EMS, BDMS, CA, and FS:

- CA and POTS FS measurements on SIP messages and subscriber features
- AIN FS measurements
- Service interaction manager (SIM) measurements
- MGCP adapter (MGA) measurements
- ISUP measurements
- Call processing measurements
- SIP adapter (SIA) measurements
- ISDN measurements
- Billing measurements
- SNMP measurements
- Trunk group usage measurements
- Announcement measurements
- Network Access Server (NAS) measurements
- H.323 measurements

Note The H323 feature flag is set to *off* in this release.

A detailed list of all counters is provided in the *Cisco BTS 10200 Softswitch Operations Manual*.

Fault Management Subsystem

The fault management subsystem provides the following functions:

- Provides interface from event management for fault reporting
- Issues events for all abnormalities found by constant monitor scans
- Issues events as appropriate

Event Management Subsystem

The event management subsystem provides the following functions:

- Handles all event notifications from the CA, FS, or fault management subsystem
- Formats events and writes into event log. Event information includes
  - Level
  - Category
  - Description
  - Instance
  - Applicable data (additional event information)
  - Event origin
  - Event date and time
- Delivers event information to any subscribed entities over Telnet or SNMP
- Generates alarms when necessary
- Reports potential faults to the fault management subsystem
- Maintains active alarm status
- Provides reports of event log keyed by event date and time, category, or event level
- Provides capability for user to subscribe to events based on interest level or category
- Provides an on-demand alarm summary report
- Provides capability to provision event throttle levels
- Provides for provisioning the severity level range to alarm
- Provides a display of all events available on the system by category and instance
- Provides for setting and clearing alarms

The current and previous calendar day's events (up to 48 hours of event logs) are maintained in the system. After 48 hours, the oldest events are sequentially overwritten by new events. Events can be throttled by provisioning a throttling level for any event. The events are halted when the number of events issued exceeds the provisioned level in a 30-minute period.

Queuing and Audit Management Subsystem

The queuing and audit management subsystem performs the following functions:

- Queuing function—The transaction queue is maintained in the EMS Oracle database. The queuing management process checks the transaction queue several times per minute for new entries. If transaction entry(ies) are found, they are sent to the appropriate CA and/or FS. If the content and format of the data in a transaction are consistent with the database in the applicable CA or FS, a commit statement is issued, and the completed transaction is removed from the queue. If the content and format of the data are not consistent, an event is issued specifying that the data is inconsistent and administrator intervention is required.
- Audit function—The audit functions are provisionable using the CLI. There are two types of audits---scheduled audits and operator-initiated audits:
  - Scheduled audits—Audit schedules are provisioned using the "**add scheduled-command**" command. At the scheduled times, the audit manager scans and compares the EMS tables to the CA and/or FS tables. If any inconsistency is found, the system generates a report.
  - Operator-initiated audits—The system operator can initiate either a general audit (**audit database** command) or a table-specific audit (**audit <table name>** command).

Note To access audit reports via a Web browser, use the primary EMS DNS or IP address with port 10200 (for example, http://priems17:10200 or http: //190.101.100.201:10200).

Configuration Management Subsystem (Status and Control)

The configuration management subsystem performs the following functions:

- Provisions all Cisco BTS 10200 components
- Reports the administrative and operational status, and changes the administrative status of components. Components include:
  - FS
  - CA
  - EMS
  - BDMS
  - Trunk groups (SS7, CAS, ISDN, Announcement [MGCP], Softswitch [SIP], and H.323)

  Note The H323 feature flag is set to *off* in this release.

  - Trunk terminations
  - Subscriber terminations
  - MGW

Security Management Subsystem

The security management subsystem is responsible for all Cisco BTS 10200 access security concerns. The security management subsystem provides the following functions:

- Performs user authentication
- Manages the level of access on a per user basis
- Provides an interface for the system administrator to provision users and security classes
- Provides session-oriented security measures
- Provides transaction oriented security measures
- Logs all access activity to a log
- Maintains security log for 48 hours
- Provides user view of the security log reports
- Provides support for an activity log for data retention, stored in a flat file for efficiency. The number of entries stored is based on a factory preset value. Once the maximum number of entries is reached, the oldest files are deleted to create room for the new entries.
- Provides support for password aging—This is the number of days that a password is valid. The time period is provisionable (default 30 days).
- Provides support for password aging notification—The EMS warns the user that the current password is set to expire. The warning notification period is provisionable (default 4 days).
- Provides access via SSH—SSH implementation supports a secure connection between two nonsecure sites. This means that all traffic on that socket will be encrypted.

Note During software installation at the factory, a user-access parameter is set according to the preference indicated by the service provider. The options are SSH only, or SSH and Telnet.

Billing Data and Bulk Data Management System

The Cisco BTS 10200 provides the following billing functions:

- Provides FTP transfer of call data records to a remote billing server or third-party billing mediation device
- Supports batch record transmission via FTP
- Issues events as appropriate including potential billing data overwrites
- Saves up to 10 million billing data records in persistent store (10 million records are adequate to hold 48 hours of billing data on a typical Cisco BTS 10200 for maximum traffic load under normal circumstances)
- Supports user-provisionable billing subsystem parameters
- Supports on-demand call detail block (CDB) queries based on ranges of timestamps, ranges of sequence numbers, a calling number, a called number, or last record written

The Bulk Data Management System (BDMS) application in the Cisco BTS 10200 gathers all billing-related call events from call processing, formats them into a standard format, and transmits them to an external billing collection and mediation device (see Figure 11-1). The interface to the billing mediation device can vary from carrier to carrier, thus the BDMS supports a flexible profiling system. This profiling system allows the Cisco BTS 10200 to adapt quickly to any variation of the interface to the billing mediation device, or to a variation in the record keeping system.

The BDMS transmits billing records via FTP to a remote server that is part of the service provider billing system. The FTP transfer occurs automatically every *n* minutes, where *n* is a number from 1 to 60 that can be provisioned in the Cisco BTS 10200 by the service provider. The default value is 15.

See the *Cisco BTS 10200 Softswitch Operations Manual* for billing procedures, basic call billing data and feature billing data. See the *Cisco BTS 10200 Softswitch CLI Guide* for individual command options.

CHAPTER 12

Routing Subsystem

The routing subsystem includes:

- Digit analysis
- Subscriber termination
- Carrier routing
- Trunk group termination
  - SS7 routing
  - SIP routing
  - CAS routing
  - ISDN routing (if connected to PBX)
  - H.323 routing

Note The H323 feature flag is set to *off* in this release.

Routing selections include:

- Least-cost routing (LCR)
- Call routing based on dialed number or ANI
- Call routing based on TOD, DOW, DOY
- Call routing based on originator
- Call routing to carriers for calls that are casual dialed
- Call routing to intraLATA PIC, interLATA PIC, and international PIC
- Call screening based on destination, origination, service request, and type of call

Numbering Plans include:

- North American numbering plan (NANP)
- Support for service codes (N11)
- Vertical services access codes (*XX), including the following capabilities:
  - Service provider can activate or deactivate service codes
  - Service provider can reassign the routing for each service code
  - Support for 411 directory service
  - ANI delivery on 911 calls

The MGW and networks that connect to the Cisco BTS 10200 Softswitch are controlled by direct connections. The external interfaces connecting other switching systems or CAs are represented by trunk groups (TGs). A TG can be an IP interface (or a range of Real-time Transport Protocol [RTP] ports on an IP interface) or a group of DS0 channels connected to a Class 4 or 5 switch.

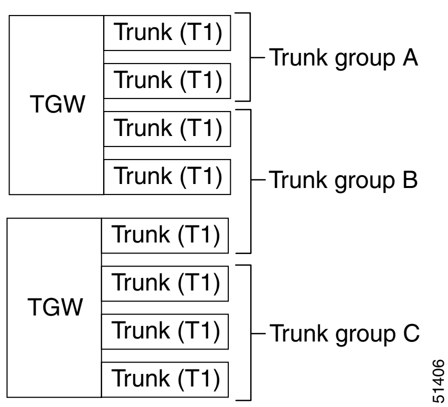
Signaling links (SLs) define the signaling interface properties between the Cisco BTS 10200 and other network elements. For example, a signaling link connecting the Cisco BTS 10200 and a Class 4 or 5 switch defines SS7 properties. Signaling connecting the Cisco BTS 10200 and another call agent defines CA to CA signaling properties (SIP-T), and the signaling link connecting the Cisco BTS 10200 and PBX or Class 5 defines ISDN PRI properties.

The TG in association with SLs provides call signaling and bearer paths. One or more TGs can share the same signaling links.

The following conditions are supported by the Cisco BTS 10200 routing subsystem (see Figure 12-1):

- A TG can span across multiple trunking gateways (TGWs).
- A TGW can be associated with more than one trunk group connecting to different nodes.

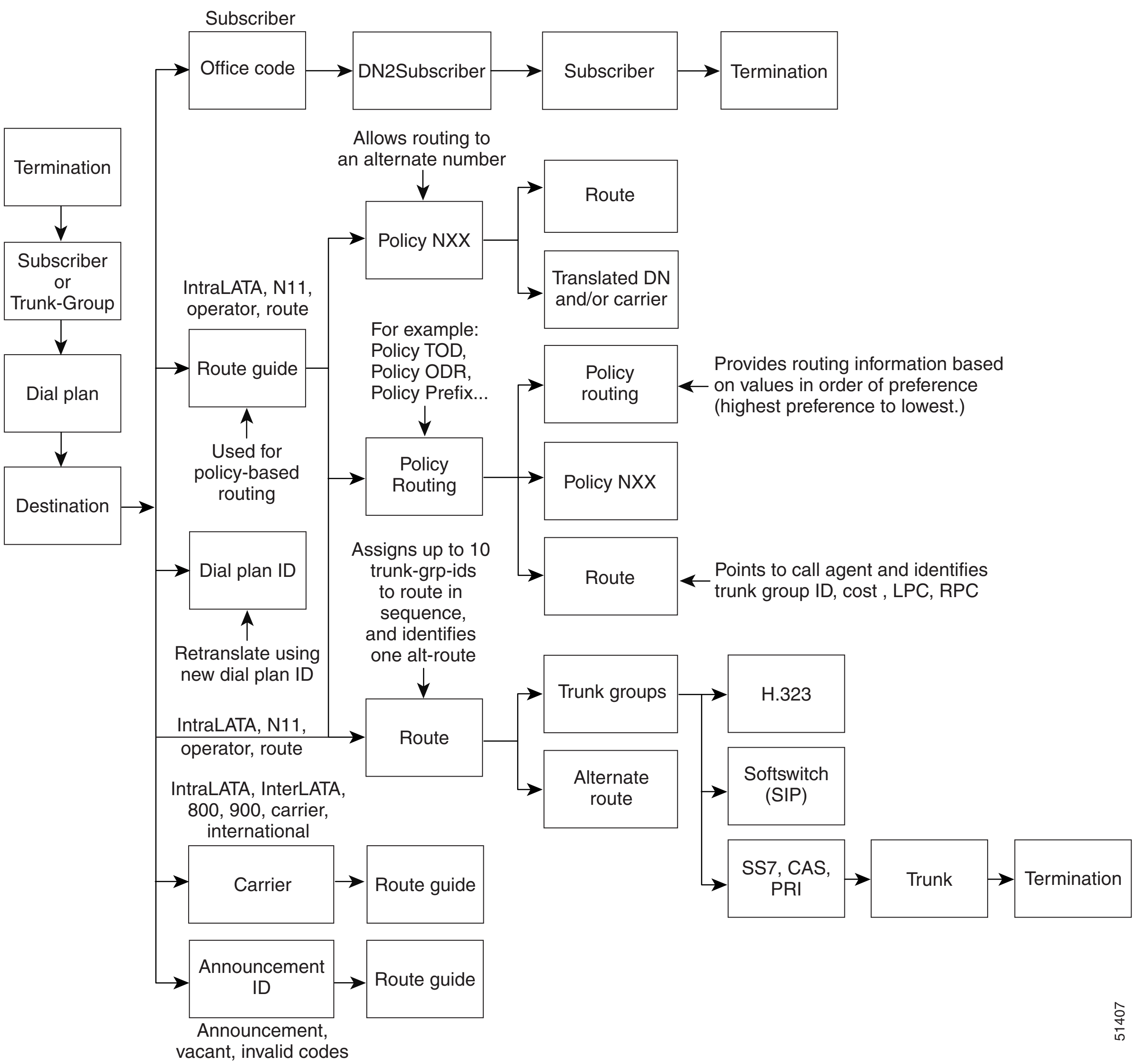
Figure 12-1 Relationships of TGW, TG, and Trunks



Route Selection

As shown in Figure 12-2, when a request for routing is received by the routing subsystem, the subsystem first searches through the termination table to determine whether the called number is a subscriber. If the called number is a subscriber, the dial plan table processes the call through the dn2subscriber and subscriber tables. If the called number is not a subscriber, the call is routed to the appropriate trunk, carrier, or announcement.

Figure 12-2 Route Selection Flow



Policy-Based Routing

If policy-based routing is required, the dial plan entry for the corresponding called number (NPA/NXX) is associated with the route group ID. The Cisco BTS 10200 supports the following routing policies:

- Least-Cost Routing, page 12-4
- Prefix-Based Routing, page 12-4
- Line-Based Routing, page 12-4
- ANI-Based Routing, page 12-5
- Region-Based Routing, page 12-5
- Time of Day (TOD) Routing, page 12-5
- Percentage-Based Routing, page 12-6
- Route ID, page 12-6
- Trunk Group Selection Policies, page 12-6

Least-Cost Routing

With least-cost routing (LCR), the least expensive TG in a route is chosen. The TGs in a list may be provisioned in any order. The call processing function will find the relative cost of each TG in the route from the TG table and order them from least to most expensive. The least expensive idle trunk will be selected from this newly ordered TG list.

Prefix-Based Routing

Prefix-based routing is used to route calls to an interexchange carrier. A service provider can choose to route calls over different TGs based on the prefix dialed. The supported call types and their associated prefixes are

- National (1+)
- International (011+)
- Operator (0-, 00)
- National operator (0+)
- International operator (01+)
- Toll free (8xx)
- Cut-through calls (101xxxx+#)
- Directory assistance (555-1212, NPA-555-1212)

Line-Based Routing

Routing of calls can be based on the originating subscriber line class (such as coin, coinless, hotel/motel or multiparty). If an incoming call is received over a feature group D SS7 TG, the originating line information parameter is used to determine the route.

ANI-Based Routing

Automatic number identification (ANI) routing is based on the subscriber ID of the calling party. The service provider can enable ANI-based routing by setting the ANI-BASED-ROUTING flag in the TRUNK-GRP table. When a call is received by the Cisco BTS 10200, the Cisco BTS 10200 performs a subscriber lookup based on the ANI (DN), and uses the subscriber properties to route the call.

Note For a remote CENTREX or PBX application, each enterprise (or business group) is entered as a subscriber on the Cisco BTS 10200. Each ANI (DN) within the business group is linked to this subscriber ID.

The following are examples of ANI-based routing:

- For outgoing interLATA calls, the Cisco BTS 10200 needs to route the call to the nearest Access Tandem (AT) switch. The Cisco BTS 10200 first analyzes the calling party's DN and determines the subscriber ID. The Cisco BTS 10200 uses the subscriber information (dial plan ID or POP) to determine which AT switch is the nearest, and routes the call. The Cisco BTS 10200 provides data for transit network selection (TNS) in an SS7 outgoing setup message.
- For 911 calls, the Cisco BTS 10200 analyzes the calling party's DN, determines the subscriber ID, and routes the call to the PSAP designated for this subscriber.
- If Class of Service (COS) screening is required, the Cisco BTS 10200 analyzes the calling party’s DN, determines the subscriber ID, and allows or denies the outgoing call according to the parameters provisioned for this subscriber.

Region-Based Routing

If region-based routing is required, the Cisco BTS 10200 uses a look-up table to convert the calling party’s dialing number (DN) to a region. The Cisco BTS 10200 then routes the call according to the information provisioned for this region. If ANI is not available, the Cisco BTS 10200 routes the call according to the region assigned to the originating TG.

Time of Day (TOD) Routing

The TOD feature has the following subfeatures:

- Routing based on day of year (DOY)—Up to 10 DOY entries are supported. If a matching DOY entry is found, the call is routed on the basis of the DOY. Otherwise (if no matching DOY entry is found), the call is routed based on the DOW/TOD table.
- Routing based on day of week (DOW) and time of day (TOD)—Up to 10 DOW/TOD entries are supported. Ranges for both DOW and TOD can be provisioned. If an entry corresponding to the DOW/TOD is not found, a default entry should be provided in the database for routing.

Percentage-Based Routing

Percentage-based routing allows call distribution of a specified percentage of calls to different route groups. Up to 5 different percentage ranges (each directed to a specific route group) can be entered. The percentage ranges can be specified to a granularity of 1 percent, and the ranges should cover a total of 100 percent. A random number generator is used to sort calls into the specified percentage ranges, and the percentage range is used to route each call.

Route ID

The route table is used when the routing policy is specified as route ID. Up to 10 TGs can be provisioned per route ID. A TG is selected based on the trunk group selection policy specified in the route table.

Trunk Group Selection Policies

The TG selection policies are applied once a route has been selected. Route selection is based on the route selection policy, and each route can have up to 10 TGs. If all the TGs in a route are busy, the route can point to an alternate route. The following assumptions are made regarding TGs:

- TG is controlled by one CA
- TG can be split across multiple MGWs
- All Trunks in a TG have the same physical characteristics back to the IP network

TG selection policies are also applied based on the user requirements. The TG selection policies are not applied if the user has not requested any capabilities. The selection policy supported is selection of TG by cost.

Trunk Selection Policies in the TG

Trunk selection policies are applied to each TG, and allow the service provider to control the assignment of trunks. The CA supports the following trunk selection policies, provisionable in the TG table:

- Ascending (default)—Select trunks in ascending order
- Odd—Select only odd numbered trunks
- Descending—Select trunks in descending order
- Least recently used—Select least recently used trunk
- Even—Select only even numbered trunks

Glare Resolution Parameter in the TG

The glare parameter in the TG table is provisioned for bothway trunks only (those that carry both incoming and outgoing calls). Glare is a condition in which the stations at both ends of the trunks seize the trunk simultaneously. The glare parameter defines which station will be given priority to seize the trunk:

- Odd—This trunk group is master of odd numbered trunks.

- Even—This trunk group is master of even numbered trunks.
- All—This trunk group is master of all trunks.
- Slave—This trunk group yields any trunk in glare condition.

Note When provisioning ISDN trunk groups, glare must be set to All. Setting glare to Slave can cause CIC/trunk instability

Trunk Group Types

The following TG types are supported in the TRUNK-GRP table in the Cisco BTS 10200 database:

- ANNC
- SOFTSW
- CAS
- ISDN
- SS7
- H323

Note The H323 feature flag is set to *off* in this release.

The required fields (tokens) for each of these TGs in the TRUNK-GRP table are listed in the following sections. In addition to the tokens listed here, there are several optional tokens that can be provisioned by the user as desired. See the *Cisco BTS 10200 Softswitch CLI Reference Guide* for a complete list of all token values.

Announcement Trunk Group

The announcement TG (ANNC TG) is used to define announcement trunks. The announcements are played on an MGCP-controlled MGW. In an ATM environment, an ATM MGW is connected to an announcement server via TDM trunks. The ANNC TG is also used to define the TDM trunk group. The fields for the ANNC TG are listed in Table 12-1.

Table 12-1 ANNC TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (ANNC)	Identifies the TG type as Announcement/MGCP

Softswitch Trunk Group

The fields in the softswitch TG are listed in Table 12-2.

Table 12-2 Softswitch TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (SOFTSW)	Identifies the TG type as Softswitch (CA to CA trunks)
SOFTSW-TSAP-ADDR	TSAP address of the Cisco BTS 10200
TG-PROFILE-ID	TG profile ID to be used
DIAL-PLAN-ID	Dial plan ID to be used

CAS Trunk Group

The fields in the CAS TG are listed in Table 12-3.

Table 12-3 CAS TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (CAS)	Identifies the TG type as CAS
TG-PROFILE-ID	TG profile ID to be used
DIAL-PLAN-ID	Dial plan ID to be used

ISDN Trunk Group

The fields in the ISDN TG are listed in Table 12-4.

Table 12-4 ISDN TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (ISDN)	Identifies the TG type as ISDN
MAIN-SUB-ID	ID of main subscriber number on PBX
TG-PROFILE-ID	TG profile ID to be used
DIAL-PLAN-ID	Dial plan ID to be used
MGW-ID	MGW ID to be used

Table 12-4 ISDN TG Required Fields and Descriptions (continued)

Field (Token)	Description
DCHAN-SLOT	D-Channel slot number on MGW
DCHAN-PORT	D-Channel port number on MGW

SS7 Trunk Group

The fields in the SS7 TG are listed in Table 12-5.

Table 12-5 SS7 TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (SS7)	Identifies the TG type as SS7
DPC	Destination Point Code
TG-PROFILE-ID	TG profile ID to be used
DIAL-PLAN-ID	Dial plan ID to be used

H.323 Trunk Group

Note The H323 feature flag is set to *off* in this release.

The fields in the H.323 TG are listed in Table 12-6.

Table 12-6 H.323 TG Required Fields and Descriptions

Field (Token)	Description
ID	Unique identifier for this TG
CALL-AGENT-ID	ID of CA for this TG
TG-TYPE (H323)	Identifies the TG type as H.323
H323-GW-ID	Specifies the gateway ID for this trunk group
TG-PROFILE-ID	TG profile ID to be used

Carrier-Based Routing and Service Provider-Based Routing

This section describes the carrier-based and service provider (SP) based routing features. In a wholesale network environment, the wholesale network operator (NO) owns and operates the facility. The NO provides transport facilities to carry voice calls on behalf of smaller SPs and for SPs that are not facility based. Some SPs may own facilities terminating directly on NO equipment. The Cisco BTS 10200 supports two types of routing options for the NO:

- Route calls over the SP facilities if a route exists on the SP facilities
- Route calls over NO facilities if SP based facilities do not exist

Note When Cisco BTS 10200 database tables are named in this section, refer to the *Cisco BTS 10200 Softswitch CLI Reference Guide* for detailed provisioning options.

Carrier Selection for Outgoing Calls

For outgoing interLATA calls, the Cisco BTS 10200 selects the carrier based on one of the following:

- The subscriber's presubscribed interexchange carrier (PIC)
- The dialed carrier ID if the call was a casual call

Note The CARRIER table can be provisioned to route calls for the carrier according to information provided by the SP. If a specific SP ID is entered, the routing options in the SP table are used, rather than routing options in the CARRIER table. This is referred to as SP-based routing.

For outgoing toll-free calls, the Cisco BTS 10200 receives the carrier ID based on a toll-free query to a service control point (SCP).

For outgoing 500 (PCS rate) and 900 (premium rate) calls, the DESTINATION table can be provisioned with unique carrier IDs for each of these call types. The Cisco BTS 10200 routes the calls via these assigned carriers.

Carrier Selection for TG-Originated Calls

In a Class 4 environment, the Cisco BTS 10200 is connected to several PSTN switches. For trunk groups (TGs) between the Cisco BTS 10200 and a Tandem switch, the TG table can be provisioned to route the call based on either the carrier ID or the SP ID.

The TGs can be provisioned as either of the following:

- A shared TG that carries traffic for multiple carriers—The Cisco BTS 10200 uses carrier identification parameter (CIP) or transit network selection (TNS) parameter to determine the carrier ID.
- A dedicated TG that carries traffic only for one particular carrier—The Cisco BTS 10200 uses CIP, TNS, or the carrier ID assigned to the TG.

If a SP ID is assigned to the TG, the SP ID is used for routing. The Cisco BTS 10200 identifies the carrier for the TG-originated call, and uses one of the following (provisionable) methods to route the call:

- The dial plan provisioned for a SP specified in the TG table
- The dial plan provisioned for a SP specified in the CARRIER table

Note To enable SP ID selection in the TG table (for SP-based routing), the TRAFFIC TYPE token in the TG table must be provisioned as TANDEM.

Technical Prefix Group Routing for H.323 Applications

Note The H323 feature flag is set to *off* in this release.

The Cisco BTS 10200 can perform SP-based routing on H.323-controlled calls using technical prefix digits. The Cisco BTS 10200 maps technical prefix to SP ID according to provisioning in the TECHNICAL PREFIX GROUP table.

Network Features

The Cisco BTS 10200 Softswitch supports the network features described in Table 13-1. Some of these features are defined in Telcordia LSSGR documents. In general, Cisco BTS 10200 Softswitch features delivered via gateway clients behave identically to their PSTN counterparts.

Note For subscriber features, see Chapter 14, "Subscriber Features."

Table 13-1 Network Features

Feature Description	Reference	LSSGR Volume
Numbering Plan and Dialing Procedures		
Casual Dialing (Dial Around)		
Directory Services (411, 555-1212, 0+ Listing Services)	FSD-30-17-0000 TR-TSY-000532	10
Easily Recognizable Codes	GR-2892-CORE SR-2275, Sec. 3.3	
Emergency Services (911)	FSD-15-01-0000 TR-TSY-000529 TR-TSY-000350 SR-4163	9
Information Service Calls (900 and 976)		
n11 support (311, 411, 611, 711, 811)	FSD-30-16-0000 TR-TSY-000532	10
Operator Services • Subscriber Access To Operator (0, 00, 0+, CAC+0+, 01+, CAC+01+) • Busy Line Verification and Operator Interrupt Services	FR-271 (OSSGR)	
Vertical Service Codes		
Tandem Service		
ANI Screening		
Class Of Service (COS) Restrictions		

Table 13-1 Network Features (continued)

Feature Description	Reference	LSSGR Volume
Other Network Features		
Dialing Parity (IntraLATA Toll Presubscription)	FSD-20-24-0040 TR-TSY-000693	10
Local Number Portability (LNP)	ANSI T1S1.6	
Split-NPA	INC97-0404-016	
Toll-Free Database Service	SR-2275, Sec. 14.6	
Trunk Testing		
Default Office Service ID		

Numbering Plan and Dialing Procedures

Numbering plan and dialing procedures include the following sections:

Casual Dialing (Dial Around)

Casual dialing, also known as dial around, specifies whether the carrier supports 101XXXX calls. The digit map CLI command tokens provide the digit pattern. The digit pattern specifies all possible acceptable patterns. An example of a casual digit pattern is 1010321 or 1010220. The digit map table tells the MGW how to collect and report dialed digits to the CA. Subscribers can prefix their interLATA or international calls with 101XXXX. Dial around supports the following casual calls:

- 101XXXX + 0/1 + NPA + NXX-XXXX
- 101XXXX + 0/00
- 101XXXX + 011/01 + CC + NN

Directory Services (411, 555-1212, 0+ Listing Services)

The Cisco BTS 10200 Softswitch supports the directory services access feature as specified in LSSGR module TR-TSY-000352 (FSD-30-17-0000), *Interface To Directory Assistance Systems*.

Directory services allows a subscriber to obtain the listed telephone number for a given name and address. The caller dials a specific service number to reach directory services, also referred to as directory assistance (DA). When a subscriber dials one of the following digit patterns, the Cisco BTS 10200 Softswitch routes the call to the applicable directory services in the PSTN:

- 411 or 555-1212 (DA)
- 1+411, 1+555-1212 (toll DA)
- 1-NPA-555-1212 (mostly for out-of-town/state numbers)
- 1-800/888-555-1212 (toll free numbers)
- 0+ listing services

The service to the caller can be provided manually by a live operator, automated via a voice or DTMF recognition system, or by a combination of these. The volume level from an automated voice-response unit, however, should be comparable to that of a live operator. Different network operators can employ different systems in providing directory services.

A typical directory services request requires that the caller first give the name of the town and city. The caller then provides the name of the person or business that the caller wants to call, including the spelling of unusual names. Finally, the caller states if the request is for residence or business. Additional services include handling multiple requests made during the same call and automatic connection to the person (or business) the caller wants to call.

Easily Recognizable Codes

The Cisco BTS 10200 Softswitch supports selected easily recognizable codes (ERCs) as described in Telcordia SR-2275, *Bellcore Notes On the Network*, Section 3.3. The supported ERCs are

- 500 personal communications services (PCS)—See INC-95-0407-009 for PCS description
- 700 service access calls (SAC)—range of codes used by IXCs to provide services on the network
- Toll free service call features (800, 888, 877, 866, 855, 844)—See the "Toll Free Services" section on page 13-7 for description
- 900/976 information service calls—See the "Information Service Calls (900 and 976)" section on page 13-4 for description)

References:

SR-2275, *Bellcore Notes On the Network*

GR-2892-CORE, *Switching And Signaling Generic Requirements For Toll-Free Service Using AIN*

Emergency Services (911)

The Cisco BTS 10200 Softswitch supports emergency services (911) as specified in LSSGR module TR-TSY-000529, Public Safety.

Other reference documents include

- LSSGR module TR-TSY-000529, *Public Safety*
- Telcordia SR-4163—*E9-1-1 Service Description*
- Telcordia TR-TSY-000350—*E911 Public Safety Answering Point: Interface Between a 1/1A ESS Switch And Customer Premises Equipment*

Note Refer to the referenced standards for a complete description of 911 services and features.

Emergency services is a public safety feature providing emergency call routing to a designated emergency service bureau (ESB), normally called the public safety answering point (PSAP) in the United States. The 3-digit 911 number is assigned for public use in many areas of the United States and Canada for reporting an emergency and requesting emergency assistance. Depending on municipal requirements and procedures, an ESB attendant can transfer the call to the proper agency, collect and relay emergency information to the agency, or dispatch emergency aid directly for one or more participating agencies.

911 calls are location dependent and must be selectively routed to the appropriate PSAP depending on where the call originates. The routing process is

- In the PSTN, the local serving end office routes the call to the designated E-911 tandem for that serving area.
- The E-911 then routes the call to the proper PSAP.

Once the caller is connected to the PSAP attendant, the PSAP system typically displays the caller's directory number to the PSAP attendant. Additional data (such as the subscriber's name, address and closest emergency response units) may also be retrieved from the local carrier automatic location identification (ALI) database and displayed to the PSAP attendant.

Examples of special emergency functions include:

- Operator callback—Allows the PSAP to automatically ring back the caller.
- End-to-end called-party hold—The Cisco BTS 10200 Softswitch keeps the connection active even if the caller goes on hook.
- Operator disconnect—Allows the PSAP to terminate the call even though the caller has not gone on hook.

Note These special emergency functions can be provided via a CAS TGW that supports ESB trunks or emergency service line (ESL) trunks with MF signaling.

Information Service Calls (900 and 976)

Information service calls (ISC) provide a variety of announcement-related services on a national or local basis. There are two general categories of this service:

- Public announcement services (PAS)—Weather, sports, horoscope, and so forth
- Media stimulated calling (MSC)—Telephone voting, radio station call-ins, and so forth

National calls are dialed as 1-900-xxx-xxxx and local calls are dialed as NPA-976-xxxx.

n11 support (311, 411, 611, 711, 811)

The following reference material is for further information about n11 calling:

LSSGR module FSD-30-16-0000 (TR-TSY-000532), Service Codes N11

The Cisco BTS 10200 Softswitch supports n11 services as described in the following sections.

Nonemergency Services (311)

Some city governments offer 311 service to provide nonemergency information to the community. The caller dials 311 and the call agent translates this to the closest nonemergency access office.

The Cisco BTS 10200 Softswitch supports nonemergency services (311) for routing calls to a specified route type and identification. Routes for all nonemergencies (311) are allocated through the destination table by defining the call type ("call-type=non-emg") and the routing information for the dialed digits.

Directory Assistance (411)

The 411 service provides directory assistance. See the "Directory Services (411, 555-1212, 0+ Listing Services)" section on page 13-2.

Repair Service (611)

The 611 service connects to the local telephone repair service.

Telecommunications Relay Services (711)

The 711 service provides access to telecommunications relay services (TRS).

Local Billing Services (811)

The 811 service connects to the local telephone billing office.

Other n11 Codes

The following codes are not currently supported in the Cisco BTS 10200 Softswitch:

- 211 service (caller access to information from government service agencies and certain public charity groups)
- 511 (caller access to information about local traffic conditions)

Operator Services

This section describes the services available to operators.

Subscriber Access To Operator (0, 00, 0+, CAC+0+, 01+, CAC+01+)

The Cisco BTS 10200 Softswitch supports operator services as specified in Telcordia Requirement FR-271, *Operator Services Systems Generic Requirements (OSSGR)*.

Operator services is a call-processing function whereby callers can access either a live operator or an automated function to complete calls or gain access to information. The service provider can provide this feature or outsource it to a third-party vendor. Some additional functions accomplished by operator services include automatic call distribution, billing detail recording, and information retrieval. The following numbers are commonly used to access operator services:

- 0—Local operator support

- 00—Operator support outside the “local” calling area, using a presubscribed interexchange carrier (PIC)
- 0+ area code and number—Operator support when the destination number is known (that is, for collect calls, calling card calls, person-to-person calls, and so forth, using PIC
- CAC+0+—Operator services, using a dialed carrier access code (CAC)
- 01+CC+NN—international operator services, using PIC
- CAC+01+CC+NN—International operator services, using a dialed CAC

Operator services provided to callers include:

- Assistance
- General information
- Directory assistance
- Dialing instructions
- Rate information
- Credit recording
- Trouble reporting
- Call completion
- Alternate billing services (ABS)
- Calling card calls
- Collect calls
- Third-number calls
- Handling options
- Person-to-person
- Conference calls
- Call transfer
- Real-time rating
- Rate quotes
- Time and charges
- Notify

Busy Line Verification and Operator Interrupt Services

Busy line verification (BLV) service permits the user to obtain operator assistance to determine if a called line is in use. The user dials 0, waits for the operator to pick up the line, and requests BLV service.

Operator interrupt service works as follows:

- The user calls the operator and requests BLV service regarding a specific called line.
- The operator provides the BLV service.
- If not prohibited in advance by the party on the called line, the operator interrupts the conversation in progress and relays a message.

- If the interrupted party at the called line is willing to hang up, they do so.
- The user can originate a new call to the called DN.

At the user's request, the operator has the option to directly connect the user to the called line.

BLV and operator interrupt services are based on the Telcordia document GR-1176, (FSD 80-01-0300), *Custom Call-handling Features*, part of Telcordia family of requirements OSSGR (FR-271).

Toll Free Services

The purpose of toll free services is to have the called party, rather than the calling party, charged for the call. These calls are prefixed with the 1+800 or 1-8nn service access codes. The seven digits following the 800/8nn codes are used for routing the call. The Cisco BTS 10200 Softswitch communicates with a database called the toll-free database service, which contains information for routing the call. The database service provides information about the network service provider selected to complete the call, and information for translating the toll-free number to a specific 10-digit directory number (DN). The routing of the call can vary depending on the arrangements made between the toll-free subscriber and the network service provider. These arrangements can include selective routing based on the time of day, day of week, and location from which the call originates.

All aspects of toll-free calling are transparent to the caller. A caller expects to dial 1-800/8nn-NXX-XXXX to reach the desired destination. The company that translates the number to a specific DN, and the company that routes the call, must appear transparent to callers. Most callers are not aware that their dialed 800/8nn number is changed into a specific DN. What matters to the callers is that they reach what they perceive to be the called number, and they are not billed for the call.

Vertical Service Codes

Vertical service codes (VSC) allow subscribers to activate and deactivate services from their own station. Table 13-2 lists the VSCs (applicable to POTS systems) that are preprovisioned in the Cisco BTS 10200 Softswitch. These can be changed in the VSC table via CLI commands. For Centrex systems, star codes (codes consisting of an asterisk plus a two digit number) can be entered in the custom dial plan table to perform these same functions. These features are described in the "Subscriber Features" section on page 14-1.

Table 13-2 Vertical Service Code Table—Features and Preprovisioned Codes

Features	Codes
Call Forwarding Unconditional (CFU)	
Call Forwarding Unconditional Activation (CFUA) and Call Forwarding Unconditional Deactivation (CFUD)	*72 Activation *73 Deactivation
Call Forwarding Busy (CFB)	
Call Forwarding Busy Variable Activation (CFBVA) and Call Forwarding Busy Variable Deactivation (CFBVD)	*90 Activation *91 Deactivation

Table 13-2 Vertical Service Code Table—Features and Preprovisioned Codes (continued)

Features	Codes
Call Forwarding No Answer (CFNA)	
Call Forwarding No Answer Variable Activation (CFNAVA) and Call Forwarding No Answer Variable Deactivation (CFNAVD)	*92 Activation *93 Deactivation
Selective Screening of Incoming Calls	
Selective Call Forwarding (SCF) Activation and Selective Call Forwarding (SCF) Deactivation	*63 Activation *83 Deactivation
Selective Call Acceptance (SCA) Activation and Selective Call Acceptance (SCA) Deactivation	*64 Activation *84 Deactivation
Selective Call Rejection (SCR) Activation and Selective Call Rejection (SCR) Deactivation	*60 Activation *80 Deactivation
Distinctive Ringing on Call Waiting (DRCW) Activation and Distinctive Ringing on Call Waiting (DRCW) Deactivation	*61 Activation *81 Deactivation
Caller ID Features	
Calling Identity Delivery and Suppression Per Call (CIDS)	*82 for delivery function *xx for suppression function (provisionable by service provider)
Calling Name Delivery Blocking (CNAB)	*95 Blocks calling name delivery for the call
Calling Number Delivery Blocking (CNDB)	*67 Toggle
Anonymous Call Rejection	
Anonymous Call Rejection (ACR) Activation and Anonymous Call Rejection (ACR) Deactivation	*77 Activation *87 Deactivation
Cancel Call Waiting	
Cancel Call Waiting (CCW)	*70 Cancel call waiting

Table 13-2 Vertical Service Code Table—Features and Preprovisioned Codes (continued)

Features	Codes
Automatic Callback	
Automatic Callback (AC) Activation and Automatic Callback (AC) Deactivation **Note** AC is also referred to as Repeat Dialing.	*66 Activation *86 Deactivation
Automatic Recall	
Automatic Recall (AR) Activation and Automatic Recall (AR) Deactivation **Note** AR is also referred to as Call Return.	*69 Activation *89 Deactivation
Speed Call	
Access the speed call feature	*74 (for 8-number speed call) *75 (for 30-number speed call)
Do Not Disturb	
Do not disturb activation and Do not disturb deactivation	*78 Activation *79 Deactivation
Customer-Originated Trace	
Access the Customer-Originated Trace (CT) Function	*57

Tandem Service

This section describes the following tandem services:

- ANI Screening, page 13-9
- Class Of Service (COS) Restrictions, page 13-10

ANI Screening

Automatic number identification (ANI) is used for long distance access service. The ANI screening feature validates the ANI on incoming feature group D calls from the PSTN before routing. All ANIs supported by the Cisco BTS 10200 Softswitch are stored in the feature server database. If an ANI is not available, or does not appear in the feature server ANI table, the TG default data is checked to see if casual calls are allowed. If casual calls are not allowed, the call is denied and routed to an announcement. If the ANI exists in the table, the ANI status is checked next. The ANI status can either be allowed or blocked. If the status is blocked, the call is blocked and routed to an announcement. The ANI table also includes information on class of service (COS), and authorization/account codes.

Class Of Service (COS) Restrictions

This service allows blocking calls based on their originating line type, for example, regular subscriber, hotel/motel, coin- phone, prison lines, and so forth. When the CA receives the called and calling party numbers, the following screening checks are performed (in this order):

- II white list and black list
- Called number white list and black list (domestic or international)
- Directory assistance calling restriction
- Operator-assisted calling restriction
- Type of call restrictions (local calling or international calling toll restriction, for example)
- Calling party is allowed to call the called party (check if restricted by COS)

COS restrictions can be assigned to any ANI, authorization code, trunk group, or location. When multiple COS restrictions are assigned to a call, then the most restrictive COS for that call is used. The order of precedence is as follows:

1. Use the COS assigned to ANI if found in the ANI screening table
2. If not found in ANI screening table, use the COS assigned to the TG
3. If an authorization code is required, then use the COS assigned to Auth Code

When a call is blocked due to COS screening, the call event shows which type of screening blocked the call. The service provider can provision the treatment of blocked calls, and can include, for example, playing an announcement or sending a cause code to the originator.

Dialing Parity (IntraLATA Toll Presubscription)

The Cisco BTS 10200 Softswitch supports this feature in accordance with LSSGR module FSD-20-24-0040 (TR-TSY-000693), Presubscription Indication.

Dialing parity—also known as intraLATA toll presubscription—allows subscribers to select a telecommunications company for intraLATA calls (local toll calls) in the same way they select a long distance provider. With dialing parity, subscribers are able to dial the number they want and have a preselected carrier—a CLEC, ILEC, or a long distance carrier—automatically handle the call if it is a local (intraLATA) toll call. Preselecting a local toll carrier eliminates the need for dial-around service for local toll calls (101XXXX numbers). Prior to implementation of dialing parity, long distance carriers provided intraLATA service by dialing around an ILEC or CLEC via 101XXXX numbers (carrier access codes—CACs).

Local access and transport areas (LATA) were created after the breakup of the AT&T system. LATAs are also known as called service areas or local toll calling areas. CLECs and ILECs provide two types of calls to their subscribers within the LATA:

- Local calls
- Local toll calls

Local toll calls are typically calls to places more than 16 miles from the subscriber location in urban areas and more than 13 miles in rural areas. Local toll calls do not qualify as either local or long distance—they are between the two and are subject to different rates.

Local Number Portability (LNP)

The Cisco BTS 10200 Softswitch supports the local number portability (LNP) feature. See the following documentation for further reference:

Technical Requirements for Number Portability—Switching Systems provides unofficial agreement within T1S1.6. T1S1 is the ANSI accredited body for signaling. This document is available at http://www.atis.org.

Number portability (NP) permits subscribers who change their local phone company to keep their existing telephone number. An FCC order requires this feature in the 100 top metropolitan service areas in the United States. NP permits calls to be routed to the subscriber's new local switch without any particular per-call action required of either the calling or called party.

The Cisco BTS 10200 Softswitch supports the LNP function as follows:

- Ranges/blocks of ported numbers are provisionable in the Cisco BTS 10200 Softswitch, with block size granularity from 100 to 10,000 DNs per block.
- During the call processing if the dialed digits/destined digits match 3 to 10 contiguous digits of a ported NPA-NXX-XXXX at the Info_Analyzed/ Collected_Info TDP, a query is initiated to an external database using AIN message Info_Analyzed. This LNP trigger is also known as the public office dialing plan (PODP) trigger.
- The Cisco BTS 10200 Softswitch processes the received response (Analyze_Route) from the TCAP query and determines if the dialed digits have been translated to a location routing number (LRN) or not:
  - If the CalledPartyID received from the Analyze_Route differs from the dialed digits (that is, the LRN comes back), the call is routed based on the received CalledPartyID as the ISUP IAM CalledPartyNumber and sets the ForwardCallIndicator parameter to "Number translated". But it further includes the ISUP GenericAddress Parameter (GAP) set to the dialed digits.
  - If the CalledPartyID received from the Analyze_Route is the same as the dialed digits (that is, no LRN comes back), the call is routed based on the received CalledPartyID as the ISUP IAM CalledPartyNumber and sets the ISUP ForwardCallIndicator (FCI) parameter to "Number translated".
  - When the LNP query results in an error, the call is routed based on the dialed digits/destination digits, and does not include the ISUP GAP, and the FCI is set to "Number not translated".

Split-NPA

For additional information on split-NPA, see the ATIS document INC97-0404-016, *NPA Code Relief Planning & Notification Guidelines*.

When DNs are exhausted within an NPA, an additional NPA is assigned to the region. The new NPA may be allocated as an overlay over the existing NPA, in which case there is no major impact to the Cisco BTS 10200 Softswitch. However, when the new NPA is assigned based on a geographical split of the region, there are significant impacts. The assignment of the new NPA based on a geographical split is referred to as split-NPA.

The split-NPA feature impacts both provisioning (EMS) and call processing subsystems in the Cisco BTS 10200 Softswitch. Several provisioning tasks must be performed to introduce a new NPA into a region, including:

- Duplicate records (tasks to be performed before permissive period)

- Update ANI records (tasks to be performed during permissive period)
- Cleanup (tasks to be performed after permissive period)

Note Permissive period is the time frame where both old NPA and the new NPA can be dialed to reach the subscribers affected due to the split-NPA feature.

Before the permissive period begins, subscribers affected due to the Split-NPA can be reached only via the old NPA. Duplicate records for both old and new NPA are created before the permissive period begins.

During the permissive period, both old and new NPA can be dialed (10-digit dialing is required to reach a subscriber in the affected NPA). The subscriber (ANI) and subscriber feature data records are updated to the new NPA during the permissive period.

Once the permissive period ends, subscribers affected due to the split-NPA can be reached only via the new NPA. It is also referred as the mandatory dialing period for the new NPA. The duplicate records created before the permissive period are cleaned up after the mandatory period begins.

Toll-Free Database Service

See SR-2275, *Bellcore Notes on the Network*, Section 14.6 for additional information.

The Cisco BTS 10200 Softswitch provides the ability to translate inbound/outbound 800 numbers at the FS using a local 800 database. The 800 service supports the following features:

- Origin dependent routing
- Time of day routing
- Percentage based routing
- Information digit-based screening
- Black/white list screening

The Cisco BTS 10200 Softswitch also supports optional DNIS service. In an 800 DNIS service, when a call is terminated to a PBX (call center), 4 digits are outpulsed to the PBX to identify the originally dialed 800 number. In case of custom DNIS, up to 22 digits can be outpulsed with additional information such as:

- Original 800 number dialed
- Automatic number identification (ANI)
- Originating line information of the calling party

When a translated number (for an original 800 call) is received, the Analyzed Info DP triggers the FS. The Cisco BTS 10200 Softswitch looks up the DNIS and TG information for the call. The DNIS information is then outpulsed to the PBX. If an overflow condition is encountered, the call is routed to the overflow trunk. The overflow trunk can be a PSTN trunk.

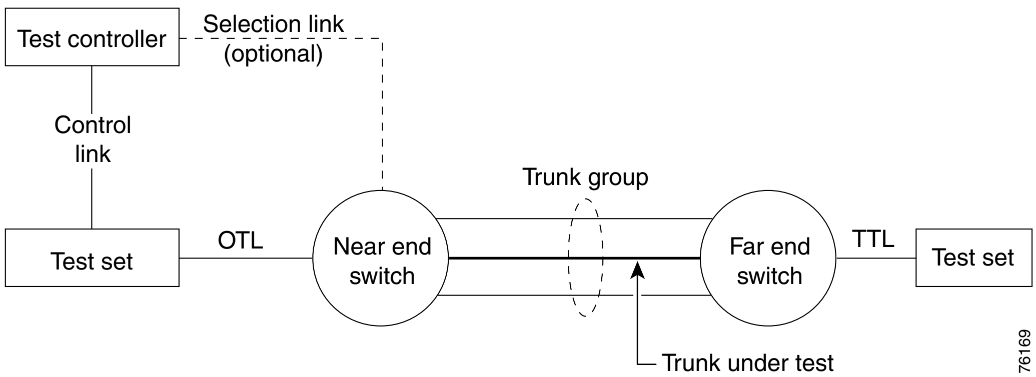
Trunk Testing

Trunk testing is used to determine the transmission quality of the shared trunks that interconnect switching systems. Trunk testing is extremely important in monitoring system health, because it is the only practical way to objectively determine the performance of individual trunks.

Trunk testing requires the equipment and test lines listed below. (Some additional types of equipment and lines may also be used.) A basic system setup is shown in Figure 13-1.

- Test controller
- Test set(s)
- Originating test line (OTL)
- Terminating test line (TTL)

Figure 13-1 Trunk Testing Setup



Near End Test Origination Test Calls

The Cisco BTS 10200 Softswitch supports calls used to test individual trunks that connect a local gateway with a gateway or PSTN switch at a remote office. The Cisco BTS 10200 Softswitch supports OTL and TTL capability. User-provided test equipment and, optionally, test controllers may be connected to the test lines. Proper selection of test equipment and test functions helps to ensure interoperability between different carriers.

Note The processes described in this section are applicable to the Cisco BTS 10200 Softswitch. The processes may work differently on other switches.

The process for testing a Cisco BTS 10200 Softswitch OTL is as follows:

1. The user should verify that the remote CO has the desired 1xx test line available.
2. The user sets up a test device on a CAS TGW that is connected to the local Cisco BTS 10200 Softswitch.
3. The user provisions the CAS-TG-PROFILE table, setting TEST-LINE = YES. (Provisioning commands are described in the *Cisco BTS 10200 Softswitch CLI Reference Guide*.)
4. On the test device at the CAS TGW side, the user enters digits representing the circuit to be tested and the test to be performed:

- TG, for example 0003
- Trunk number, for example 0018

  The complete trunk address in this example is 00030018.
- Test type (10x), for example 104

  The technician dials KP-00030018-104-ST.

5. The Cisco BTS 10200 Softswitch automatically inserts either 9581 or 9591 in front of the test type digits to create a dialing string.

   The complete test string in this example is ***PREFIX | 00030018 | 9581104 | END***.

> **Note** Alternatively, with the Cisco BTS 10200 Softswitch, the user can dial the test type with the 9581 or 9591 included:
> *KP-00030018-9581104-ST.*

6. The Cisco BTS 10200 Softswitch selects the trunk to be tested based on the user-defined trunk address.
7. The TGW outpulses the digits to the remote switch over the designated trunk.

1xx Test Line

When the Cisco BTS 10200 Softswitch is the near end switch, the following process takes place at the remote switch:

- The remote switch recognizes the trunk test prefix (9581 or 9591) on the incoming signal, and the test type is used to route the test to the appropriate test line.
- The appropriate tests are performed on the test set.
- Additional test processes may occur, depending on the specific test configuration.

When the Cisco BTS 10200 Softswitch is supporting the TTL capability (test call originated at another switch), the process is as follows. The Cisco BTS 10200 Softswitch receives the 958 or 959 call, recognizes the 958 or 959 type, and routes the test to the appropriate test line.

Default Office Service ID

One service ID (the default office service ID, typically ID=999) is reserved for provisioning of switch-based features. These switch-based features can include certain network features and certain usage-sensitive features, as described below. The service provider must provision this service ID in the service table, and define these features in the feature table.

> **Note** The default value of the office service ID is 999. This value is provisionable in the CA-CONFIG table.

Caution The system does not validate or restrict the provisioning of features on this office service ID. However, entries other than the ones listed below will have undefined results. Do not enter features other than the ones listed below.

> **Note** Refer to the *Cisco BTS 10200 Softswitch CLI Guide* for specific feature provisioning requirements.

- Network features—When provisioned, the system makes these features available for all subscriber lines:
  - Local network portability (LNP)
  - Toll-free services (8XX)
  - Emergency services (911)
  - Busy line verification (BLV)
- Usage-sensitive features—When provisioned, the system makes these features available to all subscribers without the need to actually subscribe these features to individual lines:

  > **Note** These features can also be assigned to individual subscribers using other service IDs.

  > **Note** See Chapter 14, “Subscriber Features” for details of these subscriber features.

  - Usage-sensitive three-way calling (USTWC)
  - Customer originated trace (COT)
  - Automatic callback activation and deactivation—AC_ACT and AC_DEACT (or AC, if the AC umbrella feature was created)
  - Automatic recall activation and deactivation—AR_ACT and AR_DEACT) (or AR, if the AR umbrella feature was created)

Subscriber Features

The Cisco BTS 10200 Softswitch supports subscriber features, including selected custom local area signaling service (CLASS) features, as listed in Table 14-1. Most of these features are defined in Bellcore LSSGR documents. In general, Cisco BTS 10200 Softswitch features delivered via gateway clients behave identically to their PSTN counterparts.

Note Some features can be accessed and controlled by the subscriber using a handset and vertical service codes (VSCs). VSCs are provisionable by the service provider. See Chapter 13, "Network Features" for a list of the VSCs that are preprovisioned in the Cisco BTS 10200 Softswitch.

Table 14-1 Subscriber Features

Feature Description	LSSGR FSD No.	LSSGR Vol.	Subscriber Category
Call Forwarding Features			
Call Forwarding Unconditional: • Activation/Deactivation Modes From Service Provider and Local Phone • Remote Activation of Call Forwarding • Remote Call Forwarding	FSD-01-02-1401 TR-TSY-000580 FSD-01-02-1402 GR-581	7	Centrex POTS
Call Forwarding on Busy	FSD-01-02-1450 TR-TSY-000586	7	Centrex POTS
Call Forwarding on No Answer	FSD-01-02-1450 TR-TSY-000586 FSD-01-02-2200 TR-TSY-001520	7	Centrex POTS
Calling Identity Features			
Calling Identity Delivery Blocking • Calling Number Delivery Blocking • Calling Name Delivery Blocking • Calling Identity Delivery and Suppression	FSD-01-02-1053 TR-NWT-000391	5	Centrex POTS

Table 14-1 Subscriber Features (continued)

Feature Description	LSSGR FSD No.	LSSGR Vol.	Subscriber Category
Calling Number Delivery	FSD-01-02-1051 TR-NWT-000031	5	Centrex POTS
Calling Name Delivery	FSD-01-02-1070 TR-NWT-001188	5	Centrex POTS
Calling Identity Delivery on Call Waiting	FSD-01-02-1090 TR-NWT-000575	5	Centrex POTS
Class of Service Restrictions			
Authorization Codes and Precedence			Centrex POTS
Restriction Details for Various Categories of Service • Casual Call Restrictions (101XXXX) • NANP Call Restrictions (Toll Restrictions) • International Call Restrictions • NANP Black and White Lists (Number Blocking) • Block 900 Calls • Block 976 Calls • Block DA Calls • Block Info Calls • Block Time/Weather Calls • Block NANP Operator Assistance Calls • Block International Operator Assistance Calls • Combination of Call Categories and Associated Restrictions			Centrex POTS
Account Codes and Authorization Codes			Centrex POTS
Number Blocking			Centrex POTS
Direct Inward/Outward Dialing for PBX			
Analog DID for PBX	TIA/EIA-464B	n/a	POTS only
DOD For PBX	FSD-04-02-0000 TR-TSY-000524	8	POTS only

Table 14-1 Subscriber Features (continued)

Feature Description	LSSGR FSD No.	LSSGR Vol.	Subscriber Category
Direct Inward/Outward Dialing for Centrex			
DID for Centrex	FSD-01-01-1000 TR-TSY-000520	5	Centrex only
DOD for Centrex	FSD-01-01-1000 TR-TSY-000520	5	Centrex only
Call Handling Features for Centrex Subscribers Only			
Call Hold	FSD-01-02-1305 TR-TSY-000579	7	Centrex only
Call Park and Call Retrieve	FSD-01-02-2400 GR-2913-CORE	8	Centrex only
Directed Call Pickup (With and Without Barge-In)	FSD-01-02-2800 TR-TSY-000590	7	Centrex only
Distinctive Alerting/Call Waiting Indication	FSD-01-01-1110 GR-520-CORE	5	Centrex only
General Features Applicable to Centrex and POTS Subscribers			
Anonymous Call Rejection	FSD-01-02-1060 TR-TSY-000567	6	Centrex POTS
Automatic Callback (Repeat Dialing)			Centrex POTS
Automatic Recall (AR)—Call Return			Centrex POTS
Call Block (Reject Caller)			Centrex POTS
Call Waiting	FSD-01-02-1201 TR-NWT-000571	5	Centrex POTS
Cancel Call Waiting	FSD-01-02-1204 TR-TSY-000572	6	Centrex POTS
Call Transfer	FSD-01-02-1305 TR-TSY-000579	7	Centrex POTS
Customer-Originated Trace	FSD-01-02-1052 GR-216-CORE	5	Centrex POTS
Do Not Disturb	FSD-01-02-750 SR-504	2	Centrex POTS
Hotline Service (See also "Warmline Service")			Centrex POTS

Table 14-1 Subscriber Features (continued)

Feature Description	LSSGR FSD No.	LSSGR Vol.	Subscriber Category
Interactive Voice Response Functions			Centrex POTS
Multiline Hunt Group (MLHG)	FSD-01-02-0802 TR-TSY-000569	5	Centrex POTS
Multiple Directory Numbers			POTS only
Speed Call • Speed Call for Individual Subscribers • Group Speed Call (Centrex and MLHG only)	FSD-01-02-1101 TR-TSY-000570	5	Centrex POTS
Subscriber-Controlled Services and Screening List Editing	FSD-30-28-1000 GR-220-CORE	11	Centrex POTS
• Selective Call Forwarding	FSD-01-02-1410 TR-TSY-000217	7	
• Selective Call Acceptance			
• Selective Call Rejection	FSD-01-02-0760 TR-TSY-000218	4	
• Distinctive Ringing/Call Waiting [1]	FSD-01-01-1110 TR-TSY-000219	5	
Three-Way Calling	FSD-01-02-1301 TR-TSY-000577	6	Centrex POTS
Usage-Sensitive Three-Way Calling	FSD-01-02-1304 GR-578-CORE	6	Centrex POTS
Visual Message Waiting Indicator	GR-2942-CORE		Centrex POTS
Warmline Service (See also "Hotline Service")			Centrex POTS
Default Office Service ID			

1. Distinctive ringing/call waiting (DRCW) is applicable only to individuals within a multiline hunt group (MLHG).

Call Forwarding Features

Call forwarding is a group of features allowing incoming calls to a subscriber line to be forwarded to another telephone number, including a cellular phone number, under various circumstances. Call forwarding features allow a subscriber line to be forwarded to a number that itself can be forwarded. This chaining of call forwards is allowed to a maximum of five different stations as long as none of the station numbers appears twice in the forwarding list (in order to prevent loops). Before forwarding a call outside of a zone or off net, the system must determine if the forwarding station already has an active call that has been forwarded to the same destination. If so, forwarding is denied to the second call and a station busy signal is returned to the caller.

The following types of call forwarding features are provided by the Cisco BTS 10200 Softswitch:

- Call forwarding unconditional (CFU)
  - Activation from a local phone
  - Remote activation of call forwarding (RACF)
- Call forwarding on busy (CFB)
- Call forwarding on no answer (CFNA)

Call Forwarding Unconditional

The Cisco BTS 10200 Softswitch implements the call forwarding unconditional (CFU) feature as specified in LSSGR module FSD-01-02-1401 (TR-TSY-000580), *Call Forwarding Variable*.

CFU allows the user to forward all calls regardless of the status of the user's line. A typical forwarding address is voicemail, a remote telephone, or an attendant. A user whose station is idle, and that has CFU activated, receives a reminder ring when incoming calls are forwarded, but is not able to answer the call. A reminder ring is a half second burst of ringing. The reminder ring is not applied when the forwarding station is off hook.

Activation/Deactivation Modes From Service Provider and Local Phone

CFU has multiple activation options as follows:

- Activated permanently at subscription time by the service provider—The service provider provisions the forward-to number at subscription time to the subscriber-specified DN. All calls made to the subscriber's line will be forwarded to the single forward-to number that was provisioned.
- Activated by user:
  - The user lifts the handset, and listens for dial tone.
  - The user presses ***72**. If CFU can be activated, the system returns 3 short beeps and then the dial tone.
  - The user enters the number (local, long distance, or international) where calls are to be forwarded.
  - The user receives an appropriate error if the forward-to number is invalid.
  - If the feature can be activated to the forward-to number entered, the system returns a confirmation tone and attempts to place a courtesy call to the forward-to number.
  - If the forwarded-to party answers the courtesy call, the CFU feature is activated.
  - If the forwarded-to line is busy or does not answer, the CFU feature is not activated. The user can still activate CFU by repeating the activation procedure within two minutes of the first attempt. No courtesy call is set up during the second attempt. The user hears a confirmation tone. If more than two minutes elapse before the second attempt, the second attempt is treated as a first attempt.
  - CFU is now activated, and will stay active until it is deactivated using *73.

CFU deactivation options are as follows:

- Service provider deactivation at subscriber request.

- Code-deactivated by user (service code *73)—The user deactivates the CFU feature by lifting the receiver handset, receiving a dial tone, and dialing the CFU deactivation code—***73**. If the feature can be deactivated, the system responds with a confirmation tone followed by the dial tone.

Remote Activation of Call Forwarding

Remote activation of call forwarding (RACF) permits the user to control their CFU functions when they are away from their phone. The service provider sets up this function for the user, and designates a DN the user should call to access IVR functions that control the RACF feature. Once set up for RACF, the user can take the following actions from a remote station:

- Activate CFU
- Deactivate CFU
- Change the target DN of CFU

The procedure is similar to making call-forwarding changes at a home or local business phone, but requires the additional step of dialing the remote location:

- The user dials a remote-access DN and is prompted to enter the directory number of their home phone and then the RACF authorization code (a personal identification code, PIN). The PIN can be shared by a group, or can be unique to the individual subscriber.
- Upon successful validation of the PIN, the user's current CFU activation status is checked.
  - If the CFU feature is currently inactive (calls are not being forwarded) the user is prompted to enter a DN to which calls should be forwarded.
  - If the CFU feature is currently active (calls are being forwarded) the user is given the option of deactivating CFU or changing the DN to which call should be forwarded.
- A subscriber with a unique PIN can change their PIN using the VSC function (*xx code is assigned and provisioned by the service provider). The PIN can only be changed from the base phone.

Note A shared (nonunique) PIN is usually assigned to the subscriber group by the service provider. It can be changed only by the service provider, and not through handset provisioning.

Remote Call Forwarding

The Cisco BTS 10200 Softswitch implements the remote call forwarding (RCF) feature as specified in LSSGR module FSD-01-02-1402 (GR-581), *Remote Call Forwarding*.

RCF allows incoming 7-digit or 10-digit calls to be routed automatically to a remote DN, which can be in another NANP area. RCF is activated by the service provider at customer request. With the RCF feature, all calls to the specified DN are always forwarded to a remote address. This service is similar to CFU service with these exceptions:

- Forwarding is always activated and not controlled by the customer. (The forwarded-to number cannot be changed by direct customer action.)
- No local office terminal (physical telephone) is associated with the dialed number from which forwarding occurs.
- Simultaneous calls can be active between the base switching office and the remote RCF terminal.

The billing data produced by the Cisco BTS 10200 Softswitch identifies the invoked feature as CFU and not RCF. The calling party is charged for the call to the RCF DN. The called party (RCF DN) can be charged for the CFU feature usage. The service provider can also charge the called party (RCF DN) for the call from the RCF base DN to the remote DN.

Call Forwarding on Busy

The Cisco BTS 10200 Softswitch implements the call forwarding on busy (CFB) feature as specified in LSSGR module FSD-01-02-1450 (TR-TSY-000586), *Call Forwarding Subfeatures.*

CFB allows a user (the called party) to instruct the network to forward calls when their line is busy. A typical forwarding number is voicemail. Since the forwarding station is off hook when the CFB feature is executed, no reminder ring is generated. CFB is usually set up by the service provider at the subscriber's request.

Activation/Deactivation Modes

The service provider can offer either of the following options on CFB:

- CFB is activated permanently at subscription time by service provider—The service provider provisions the forward-to number at subscription time to the subscriber-specified DN.
- CFB is set up so that an individual user can activate and deactivate CFB.

When the option is made available, user activation and deactivation of CFB works as follows:

- To activate CFB:
  - The user lifts the handset, and listens for dial tone.
  - The user presses ***90**. If CFB can be activated, the system returns 3 short beeps and then the dial tone.
  - The user enters the number (local, long distance, or international) where calls are to be forwarded.
  - If the feature can be activated to the number entered, the system activates the feature and returns a confirmation tone (2 beeps), followed by the dial tone. The user receives an appropriate error if the forward-to number is invalid.
  - CFB is now activated, and will stay active until it is deactivated using *91.

> **Note** Centrex subscribers can specify a second forwarding number for in-group calls, but they cannot program this forwarding number via handset.

- To deactivate CFB:
  - The user lifts the handset, and listens for a dial tone.
  - The user presses ***91**. If CFB can be deactivated, the system deactivates the feature and returns 3 short beeps and then the dial tone.

CFB Interactions

CFB interacts with the call waiting (CW) feature as follows:

- If a user with CW is already involved in a call, the next incoming call is not forwarded. However, any additional simultaneous incoming calls will be forwarded.
- If a user with CW has gone off hook but has not yet completed a call or the call is in a ringing state, and there is an incoming call, the call will be forwarded.

Call Forwarding on No Answer

The Cisco BTS 10200 Softswitch supports the call forwarding on no answer (CFNA) feature as specified in the following references:

- LSSGR module FSD-01-02-1450 (TR-TSY-000586), *Call Forwarding Subfeatures*
- LSSGR module FSD-01-02-2200 (GR-1520), *Ring Control*

CFNA allows a user (the called party) to instruct the network to forward calls when their own station does not answer the incoming call within the first 5 rings (default setting, but number of rings is configurable). A typical forwarding number is voicemail. Because the forwarding station is ringing when the CFNA feature is executed, no reminder ring is generated. CFNA is usually set up by the service provider at the subscriber's request.

Note Ring control allows a user to change the number of rings before forwarding the call. This service can be used with either rotary or dual tone multifrequency (DTMF) equipped customer premises equipment (CPE). The service provider sets this up at the subscriber's request.

CFNA Activation/Deactivation Modes

The service provider can offer either of the following options on CFNA:

- CFNA is activated permanently at subscription time by the service provider—The service provider provisions the forward-to number at subscription time to the subscriber-specified number.
- CFNA is set up so that the individual user can activate and deactivate CFNA.

Subscriber Activation/Deactivation Procedures

When the option is made available, user activation and deactivation of CFNA works as follows:

- To activate CFNA:
  - The user lifts the handset, and listens for a dial tone.
  - The user presses ***92**. If CFNA can be activated, the system returns a dial tone.
  - The user enters the number (local, long distance, or international) where calls are to be forwarded.
  - If the feature can be activated to the number entered, the system activates the feature and returns a confirmation tone (3 beeps), followed by the dial tone. The user receives an appropriate error if the forward-to number is invalid.
  - CFNA is now activated, and will stay active until it is deactivated using *93.

Note Centrex subscribers can specify a second forwarding number for in-group calls, but they cannot program this forwarding number via handset. The service provider sets this up at the Centrex subscriber's request.

- To deactivate CFNA:
  - The user lifts the handset, and listen for a dial tone.
  - The user presses ***93**. If CFNA can be deactivated, the system deactivates the feature and returns 3 short beeps and then the dial tone.

Calling Identity Features

Calling identity features include:

- Calling Identity Delivery Blocking, page 14-9
  - Calling Number Delivery Blocking, page 14-10
  - Calling Name Delivery Blocking, page 14-10
  - Calling Identity Delivery and Suppression, page 14-11
- Calling Number Delivery, page 14-11
- Calling Name Delivery, page 14-12
- Calling Identity Delivery on Call Waiting, page 14-12

Calling Identity Delivery Blocking

The Cisco BTS 10200 Softswitch supports calling identity delivery blocking (CIDB) features as specified in LSSGR module FSD-01-02-1053 (TR-NWT-000391), *Calling Identity Delivery Blocking Features*

CIDB allows caller to control whether or not their calling identity information is delivered with outgoing calls. Identity includes directory number (DN) and/or name of the caller. CIDB does not affect the presentation of caller's information when making 911 calls.

The CIDB feature affects the presentation status (PS) of the calling identity information. The PS is a flag that lets the network know if it is permissible to deliver the information to the called party. Both the calling number and calling name have PS information associated with them. There are two types of PS flags.

- A permanent PS (PPS) is a stored attribute about the caller's line (privacy status provisioned by the service provider). At subscription time, the subscriber requests the service provider to provision default values for the PPS flags. The possible values are:
  - Public—Calling identity information (calling name and/or calling number) is delivered with outgoing calls.
  - Private (anonymous)—Calling identity information (calling name and/or calling number) is not delivered with outgoing calls.

- Per-call PS (PCPS) has significance only to the current outgoing call. On a per-call basis, a caller with the CIDB feature enabled can override the default values for the PS flags. The per-call features are listed in Table 14-2 and described in the "Calling Number Delivery Blocking" section on page 14-10 through the "Calling Identity Delivery and Suppression" section on page 14-11.

Table 14-2 Per-Call CIDB Feature Summary

Identity Item	Permanent Privacy Status		Effect Of CNDB (*67)		Effect Of CNAB (*95)		Effect Of CIDS Delivery (*82)		Effect Of CIDS Suppression (*xx [1])	
	Number	Name	Number PS	Name PS	Number PS	Name PS	Number PS	Name PS	Number PS	Name PS
Identity:	Public	Public	Private	Private[2]	Public	Private	Public	Public	Private	Private
Number [CND]	Public	Private	Private	Private	Public	Public	Public	Public	Private	Private
	Private	Public[2]	Public	Public	Private	Private	Public	Public	Private	Private
+ Name [CNAM]	Private	Private	Public	Private	Private	Private[2]	Public	Public	Private	Private
Number:	Public	n/a	Private	n/a	n/a	n/a	Public	n/a	Private	n/a
[CND] only	Private	n/a	Public	n/a	n/a	n/a	Public	n/a	Private	n/a

1. The service provider provisions the two-digit code "xx" for the suppression feature. The value of xx is not preprogrammed in the Cisco BTS 10200 Softswitch.
2. When the number is private, no name query is performed.

Calling Number Delivery Blocking

Calling number delivery blocking (CNDB) allows the caller to control the status of their caller number privacy on a per-call basis. For all new calls, the privacy status reverts back to the PPS.

To use the CNDB feature, the caller does the following:

- The caller goes off hook and receives a dial tone.
- The caller enters ***67**. The system responds with 3 beeps and a dial tone.
- The caller enters the desired phone number for the remote station. The CNDB function toggles the PPS of the caller's number for this call:
  - If the PPS is private, CNDB makes the PS public for this call
  - If the PPS is public, CNDB makes the PS private for this call

Note When the number is private, no name query is performed.

Calling Name Delivery Blocking

Calling name delivery blocking (CNAB) allows the caller to control the status of their caller name privacy on a per-call basis. For all new calls, the privacy status reverts back to the PPS.

To use the CNAB feature, the caller does the following:

- The caller goes off hook and receives a dial tone.
- The caller enters ***95**. The system responds with 3 beeps and a dial tone.

- The caller enters the desired phone number for the remote station. The CNAB function toggles the PPS of the caller's name for this call:
  - If the PPS is private, CNAB makes the PS public for this call
  - If the PPS is public, CNAB makes the PS private for this call

Note When the number is private, no name query is performed.

Calling Identity Delivery and Suppression

The Cisco BTS 10200 Softswitch supports the delivery function (VSC *82) and the suppression function (VSC *xx) of calling identity delivery and suppression (CIDS). CIDS is a per-call feature.

Note The service provider provisions the two-digit code "xx" for the suppression feature. The value of xx is not preprogrammed in the Cisco BTS 10200 Softswitch.

CIDS allows a caller to explicitly indicate on a per-call basis whether both the calling name and calling number will be treated as private or public. The following conditions apply:

- CIDS Delivery—If the CIDS Delivery code (*82) is input, the current call is treated as public regardless of the default privacy status permanently associated with the calling name and number.
- CIDS Suppression—If the CIDS Suppression code (*xx) is input, the current call is treated as private regardless of the default privacy status permanently associated with the calling name and number.

For all new calls, the privacy status reverts back to the PPS.

To use the CIDS feature, the caller does the following:

- The caller goes off hook and receives a dial tone.
- The caller enters ***82** or ***xx** and receives a dial tone again.
- The caller enters the desired phone number.
- The caller's ID will be displayed or blocked as follows:
  - For *82, the caller's ID will be delivered (that is, it will not be blocked) at the remote station, assuming the remote station has the caller ID function.
  - For *xx, the caller's ID will be blocked at the remote station, assuming the remote station has the caller ID function.

The next time this caller makes a phone call, the default caller ID settings will apply, unless the caller again enters ***82** or ***xx**.

Calling Number Delivery

The Cisco BTS 10200 Softswitch supports the calling number delivery (CND) feature as specified in the Telcordia LSSGR module FSD-01-02-1051 (TR-NWT-000031), *Calling Number Delivery, and GR-30-CORE, Voiceband Data Transmission Interface.*

The CND feature provides CPE with the date, time, and DN of an incoming call. When the called subscriber line is on hook, the calling party information is delivered during the long silent interval of the first ringing cycle. Telcordia document GR-30-CORE specifies the generic requirements for transmitting asynchronous voice band data to the CPE.

Calling Name Delivery

The Cisco BTS 10200 Softswitch supports the calling name delivery (CNAM) feature as specified in LSSGR module FSD-01-02-1070 (TR-TSY-001188), *Calling Name Delivery Generic Requirements.*

CNAM is a terminating user feature allowing CPE connected to a switching system to receive a calling party's name, number, and the date and time of the call during the first silent interval. If a private status is assigned with the name, the name will not be delivered and a private indicator code P is sent to the CPE. If the name is not available for delivery, the switch sends an out-of-area/unavailable code O to the CPE.

When transferring or forwarding calls internally, the private calling name can be obtained from the Cisco BTS 10200 Softswitch databases and passed on to the internal called user.

Calling Identity Delivery on Call Waiting

The Cisco BTS 10200 Softswitch supports the calling identity delivery on call waiting (CIDCW) feature as specified in LSSGR module FSD-01-02-1090 (TR-TSY-000575), *Calling Identity Delivery on Call Waiting.*

CIDCW is a service that enables a called party to receive information about a calling party on a waiting call while off hook on an existing call. CIDCW provides the capability of calling identity delivery (CID) information to the called party on waiting calls. CIDCW is considered an enhancement of the CW feature, and requires the basic CW feature, along with the CND or CNAM feature.

When a called party has been alerted of an incoming call, the called party places the current far-end party on hold by pressing the **Flash** button or **hookswitch** to retrieve the waiting call. The Flash button or hookswitch can be used to toggle between the current far-end party and the held party. The details of these functions are as follows:

- A called party currently on a call receives notification, via a short beep repeated 3 times, that another call is coming in.
- The incoming name and/or number is displayed after the first beep.
- The called party can either ignore the new call or can accept it while putting the existing call on hold.
- To ignore the new call, the called party continues uninterrupted on the existing call, and the beep indication will time out after 3 repetitions.
- To accept the new call and put the existing call on hold, the called party presses and releases the **Flash** button or **hookswitch**.
- To alternate between the two calls, the called party can press and release the **Flash** button or **hookswitch**.
- If either one of the remote stations goes on hook, the remaining remote station continues as a normal session with the called party.
- The called party can end either session by going on hook during the currently active session. This ends the session. The phone will ring to indicate that the other party is still holding. The called party can pick up the phone and the session to that calling party resumes as a normal call.

If the calling party's caller ID is not available (for example, if the caller has blocked caller ID) then the called party's caller ID display will indicate an anonymous call or other unidentified caller message as in the caller ID feature.

Class of Service Restrictions

The class of service (COS) restrictions are defined for a for a subscriber or a location. Restrictions can be any of the following types:

- Call category restrictions:
  - Local—Only calls within the local service area are allowed
  - Toll—Only intraLATA toll calls are allowed
  - Intrastate—Only calls within the state are allowed
  - National—Only calls within the country are allowed
  - All—All NANP calls are allowed (default)
  - International Select—Calls to countries included in the select list
  - International—Allows international calls subject to the blocked country code list
- Black list and white list restrictions:
  - Black list—List of numbers that are not allowed
  - White list—List of numbers that are allowed when restriction type is white list
  - List of allowed country codes—Contains list of allowed country codes for the international select list
  - List of blocked country codes—Contains list of blocked country codes

Note Black/white list restrictions take precedence over call category restrictions.

The details of these restrictions are discussed further in the sections that follow.

Authorization Codes and Precedence

Authorization codes can be assigned to call category restrictions. If a user is restricted from making a certain category of call, and has a subscription to an authorization code, the call can be allowed if the user enters the correct authorization call.

Note The user will be prompted by an authorization tone (2 beeps) after which they must enter the correct authorization code.

Authorizations codes cannot be provided for black/white list restrictions.

Restriction Details for Various Categories of Service

Restriction details are provided in the sections that follow.

Casual Call Restrictions (101XXXX)

The casual call restrictions are used to allow/restrict calls dialed with a casual code prefix (101XXX). The following restrictions can be provisioned:

- No casual calls allowed—User cannot make 101XXXX calls
- All casual calls allowed—User can make 101XXXX calls
- 101XXXX white list—Only a predefined set of XXXX codes can be dialed
- 101XXXX black list—All XXXX codes can be dialed except for a predefined set

For NANP operator calls (0+NPA-NXX-XXXX) and international operator calls (01+CC+NN), casual-call screening is not performed, even if the casual-call restriction is provisioned in the cos-restrict table for the calling party.

NANP Call Restrictions (Toll Restrictions)

The NANP call restrictions are used to allow/restrict calls to destinations based on a predefined grouping of LATA, state, country, or group of countries. Customers can subscribe to one of the following:

- All NANP calls—There are no restrictions on calls to NANP areas
- National Only—Calls are restricted to within the country
- IntraState Only—Calls are restricted to within the state
- IntraLATA Only—Calls are restricted to within the area defined as the local service area
- Local Only—Calls are restricted to local only

For NANP operator calls (0+NPA-NXX-XXXX), NANP call restriction screening is not performed, even if the NANP call restriction is provisioned in the cos-restrict table for the calling party.

International Call Restrictions

The international call restrictions are used to restrict/block calls made outside the country, and to certain countries within the NANP. The following restrictions can be applied:

- No international calls allowed—Does not allow any international calls outside NANP
- International white list—Allows only those calls that have a prefix noted in the white list
- International black list—Does not allow any calls that have a prefix noted in the black list
- All international calls allowed—No restrictions are applied on any international calls

For international operator calls (01+CC+NN), international call restriction screening is not performed, even if the international call restriction is provisioned in the cos-restrict table for the calling party.

NANP Black and White Lists (Number Blocking)

This restriction allows/blocks NANP category calls to a predefined list. Customers can subscribe to one of the following:

- No restrictions (default)
- NANP white list—Only calls within a predefined prefix list can be called. The list could consist of NPA or NPANXX codes. Three to ten digits can be specified for this restriction.
- NANP black list—All calls within a predefined prefix list are blocked. The list could consist of NPA or NPANXX codes. Three to ten digits can be specified for this restriction.

Block 900 Calls

This service allows all calls of the form 1-900-XXX-XXXX to be blocked. The service provider provisions blocking or unblocking of the 900 service based on customer request.

Block 976 Calls

This service allows all calls of the form 976-XXXX or NPA-976-XXXX to be allowed/blocked The service provider provisions blocking or unblocking of this service based on customer request.

Block DA Calls

This service allows all calls of the form 411, 1+411 or NPA-976-XXXX to be allowed/blocked. The service provider provisions blocking or unblocking of this service based on customer request.

Block Info Calls

This service allows all calls to information services to be allowed/blocked. The service provider provisions blocking or unblocking of this service based on customer request.

Block Time/Weather Calls

This service allows all calls to time and weather services to be allowed/blocked. The service provider provisions blocking or unblocking of this service based on customer request.

Block NANP Operator Assistance Calls

This service allows 0+ calls (0+NPA-NXX-XXXX) to be allowed/blocked. The service provider provisions blocking or unblocking of this service based on customer request.

Block International Operator Assistance Calls

This service allows 01+ calls (01+CC+NN) to be allowed/blocked. The service provider provisions blocking or unblocking of this service based on customer request.

Combination of Call Categories and Associated Restrictions

For any call, it is possible that a combination of call categories are applicable—for example, a casual operator-assisted national call. Under these conditions the restrictions are applied based on the following priority.

- Highest priority—InterLATA, IntraLATA, International, 900, 976, DA call restrictions

- Lower priority—0-, 0+ and 01+ calls
- Lowest priority—Casual call restrictions

If a customer is subscribed to an authorization code, the system checks for all restrictions before prompting the subscriber to enter the code. When the proper code is entered, the call will be placed.

Account Codes and Authorization Codes

The Cisco BTS 10200 Softswitch supports account codes and authorization codes as described in this section.

Account Code Description

Account codes provide collection of 1 to 12 digits to allow call charging to user projects, departments or special accounts. The user activates account codes by dialing a number (usually a long distance call) that requires an account code for call completion. A prompt tone is issued after the digits are dialed. The user then enters the assigned account code. The Cisco BTS 10200 Softswitch does not verify the account code. The account code is provided in the call detail records (CDRs) associated with the call.

Account codes are not applicable, and are not collected, for any call that requires an authorization code. Account codes are not collected for any of the following call types, even if an account code requirement is provisioned in the cos-restrict table for the calling party:

- 0+, NANP Operator calls
- 01+, INTL Operator calls
- Local calls

Authorization Code Description

Authorization codes, also referred to as validated account codes, can be used by an intended user or group to override selected COS calling restrictions. An example of the authorization code would be one in which a user may be restricted from making long distance calls. The user can override the restriction by dialing an authorization code that has enough privileges to make long distance calls. The user takes the following action when an authorization code is required:

- The user goes off hook and receives a dial tone.
- The user dials a DN. The system determines that an authorization code is required and returns a confirmation tone (2 beeps) to the user.
- If the user enters the correct authorization code, the call is completed. If the code is incorrect, or the associated account is invalid, the call is diverted to a preselected announcement.

Number Blocking

The Cisco BTS 10200 Softswitch supports number blocking on certain types of calls on a per subscriber basis. Number blocking prevents certain types of calls from being completed from a particular line or station. When the caller encounters a call that is blocked, the caller can receive any of several blocking treatments such as reorder tone, announcement or routing to an attendant. Number blocking is activated/deactivated and administered by the service provider on a per line or per group of lines basis.

The service provider can prohibit calls based on different dialing plans and formats (for example, NPA, NPA-NXX-XXXX, 011+ for international calls, and so forth) combined with other restriction criteria such as type of service (for example, flat rate versus usage rate).

Direct Inward/Outward Dialing for PBX

The Cisco BTS 10200 Softswitch supports the direct inward dialing (DID) and direct outward dialing (DOD) features for PBX.

Analog DID for PBX

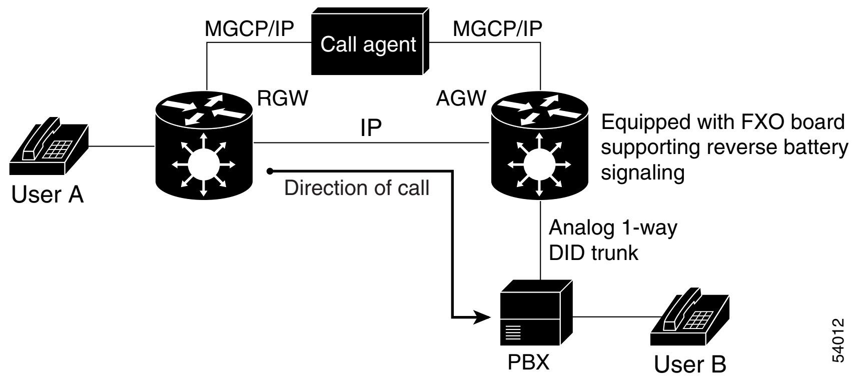
The Cisco BTS 10200 Softswitch supports analog DID for PBX as specified in TIA/EIA-464B, *Requirements for Private Branch Exchange (PBX) Switching Equipment*, April 1, 1996.

The analog DID one-way feature allows incoming calls to a local PBX network to complete to a specific station without attendant assistance. The station address is provided by the CA that controls an access gateway (AGW) connecting to the PBX. The number of digits to be outpulsed by the AGW to the PBX is configurable in the CA.

Note For guidance in provisioning the CA to support this feature, see the Cisco *BTS 10200 Softswitch Operations Manual.*

Figure 14-1 shows a typical application, with a residential user (UserA) attempting to call a PBX user station (UserB). UserB is identified by a specific set of extension digits in the PBX. The Cisco BTS 10200 Softswitch uses MGCP signaling to communicate with the AGW, and controls the outpulsing of digits from the AGW to the PBX. A foreign exchange office (FXO) board in the AGW uses reverse battery signaling (per TIA/EIA-464B) to communicate with a DID trunk board in the PBX over an analog DID one-way trunk. When completing a call termination to the PBX, the extension digits for UserB are outpulsed from the AGW toward the PBX. The PBX receives the extension digits and then completes the call to UserB.

Figure 14-1 Example of PBX Analog DID One-Way Application



DOD For PBX

Reference: LSSGR module FSD 04-02-0000 (TR-TSY-000524), *Direct Inward Dialing*.

The DOD feature allows outgoing calls from a specific station to be completed through the local PBX network without attendant assistance. The CA serving the PBX recognizes the station address and routes the call to the PBX.

Direct Inward/Outward Dialing for Centrex

The Cisco BTS 10200 Softswitch supports the following direct inward/outward dialing features for Centrex systems as specified in LSSGR module FSD 01-01-1000 (TR-TSY-520), *Basic Business Group*:

- DID
- Distinctive ringing for DID
- DOD

DID for Centrex

DID provides a Centrex group with the ability to receive a call from the PSTN without attendant intervention. The receiving Centrex station appears as a serving line to the CA.

The Cisco BTS 10200 Softswitch supports distinctive ringing for DID. The service provider can provision the distinctive ringing feature for Centrex customers. When enabled (da-cwi=yes in the centrex-grp table), the customers within the Centrex group will be able to receive different ringing patterns (distinctive ringing) and CW alerting as follows:

- Calls originating within the same Centrex (also referred to as inside calls or extension dialing)
  - Ringing pattern: 2 seconds of ringing followed by 4 seconds of silence
  - CW pattern: 0.3-second beep
- Incoming calls originating outside the Centrex (outside calls)
  - Ringing pattern: 800 ms of ringing, 400 ms of silence, 800 ms of ringing, 4 seconds of silence
  - CW pattern: 0.1 seconds beep, 0.1 seconds silence, 0.1 seconds beep

Note Incoming calls originating from a *different* Centrex are treated as outside calls.

DOD for Centrex

The Cisco BTS 10200 Softswitch supports the DOD feature. DOD provides a Centrex group with the ability to make a call to the PSTN without attendant intervention. The sending Centrex station appears as a serving line to the CA.

Call Handling Features for Centrex Subscribers Only

The Cisco BTS 10200 Softswitch provides Centrex-group functionality. A Centrex group is an emulation of a PBX by a Class 5 switch, and is typically assigned to a business group. The service provider can provision the values for the main subscriber of the Centrex group, and those properties are applied to the entire Centrex group. The service provider can also provision the parameters for simulated facility group (SFG) control, if SFG is desired. Both the incoming SFG (ISFG) and outgoing SFG (OSFG) are provisionable.

The following features are available to Centrex subscribers only:

- Call Hold, page 14-19
- Call Park and Call Retrieve, page 14-20
- Directed Call Pickup (With and Without Barge-In), page 14-20
- Distinctive Alerting/Call Waiting Indication, page 14-21

Call Hold

The Cisco BTS 10200 Softswitch supports the call hold (CHD) feature as specified in LSSGR module FSD-01-02-1305 (TR-TSY-000579), *Add On Transfer And Conference Calling Features*.

Note This feature is available only to Centrex subscribers.

The CHD feature allows the user to temporarily put an active call on hold and then make another call. The user can then return to the original call, and alternate between the two calls.

The user uses the CHD feature as follows:

- To perform a CHD during an active call:
  - The user (the activating party) presses the **Flash** button or **hookswitch** and then presses ***52**.
  - The network responds by putting the remote station on hold, providing silent termination. The system also returns 3 beeps, followed by a dial tone, to the activating party.
  - If the activating party does nothing, the network waits 4 seconds, then removes the dial tone. In this case, the activating party can recall the held party by using the **Flash** button or **hookswitch**.
  - If the activating party dials another remote station, then the system rings that station, and a new call is initiated if the remote station goes off hook.

- The CHD activation procedures (**Flash** button or **hookswitch** followed by *52) can be used to toggle between the two calls. If the activating party disconnects while a party is on hold, the network responds by ringing the activating party's line. If the line is not answered within 6 ring cycles, the held party is disconnected. The held party does not hear an audible ringback during this ringing cycle.

Call Park and Call Retrieve

Call park (CPRK) and call retrieve (CPRK-RET) are defined for a call park subscriber group (CPSG), which is a subset of the Centrex subscriber group who have privileges to park and retrieve calls. Members of the CPSG can park and retrieve calls on a DN within their own CPSG. If desired, this feature can be used to transfer calls from one CPSG member to another.

CPRK allows a user in a business group to park an active call on a designated parking DN, leaving the user free to make other calls. The parked caller hears either music-on-hold or silence. The parking party is periodically reoffered the parked call. If the parking party accepts the reoffer attempt, or if another authorized user in the CPSG retrieves the call, then the call is connected. Otherwise, after three reoffer attempts, the call is released or forwarded as provisioned.

To park an active call:

- The parking party uses the **Flash** button or **hookswitch**, receives a recall dial tone, and dials the CPRK Access Code
- The parking party dials the DN of the desired CPSG member (or just hangs up or dials # to park the call against their own DN)
- A confirmation tone is provided to the parking party to confirm that the call is parked

To retrieve a parked call:

- The retrieving party dials the CPRK-RET access code and gets a dial tone
- The retrieving party dials the DN on which the call is parked
- The call is now connected between the calling party and the retrieving party

There is no deactivation procedure for this feature. The parked call is either connected or forwarded as described above.

Directed Call Pickup (With and Without Barge-In)

Directed call pickup allows a user in a basic business group (BBG) to answer a call to a telephone from another telephone in the BBG. There are two types of directed call pickup, with and without barge-in, each with its own activation access code. These codes are assigned by the administrator of the BBG, and can range from 2 to 65535.

The procedure for directed call pickup without barge-in (DPN) is as follows:

- The process begins when a telephone rings in the BBG, and a member of the BBG at a remote phone would like to pick up the call from the ringing telephone line
- At the remote telephone line, the user lifts the handset, and listens for a dial tone
- The remote user dials the DPN activation access code ***xx**
- The system returns a recall dial tone
- The remote user dials the extension associated with the ringing line

- The remote line is connected to the incoming call that actually terminated at the ringing line
- The original called line is now idle and available to originate and to receive calls
- If the incoming call has already been picked up by another member of the BBG, the additional DPN requests are routed to a rejection announcement or tone.

The procedure for directed call pickup with barge-in (DPU) is as follows:

- The process begins when a telephone rings in the BBG, is answered by the party, and another member of the BBG at a remote line wants to join the conversation
- At the remote telephone line, the user lifts the handset, and listens for a dial tone
- The remote user dials the DPU activation access code ***xx** (where xx represents the digits assigned for DPU activation in the BBG)
- The system returns a recall dial tone
- The remote user dials the extension on which the active call is taking place
- The system plays a confirming tone and establishes a three-way call (TWC) between the remote line and the original two parties
- The remote BBG user can press the **Flash** button or **hookswitch** to drop the other BBG party related to the original call
- If the remote BBG user goes on hook, the two-way connection will be reestablished between the calling party and the original BBG party

Distinctive Alerting/Call Waiting Indication

The distinctive alerting/call waiting indication (DA/CWI) feature is based on the Telcordia document *GR-520-CORE, Features Common to Residence and Business Customers I (FSDs 00 to 01-01-1110)*. DA/CWI provides Centrex users special ringing and CW tones on DID calls. The Centrex administrator can activate this feature for some or all of the business group lines (BGLs) in the basic business group (BBG). Any call terminating at a designated BGL will receive the appropriate distinctive ringing or CW tone.

General Features Applicable to Centrex and POTS Subscribers

The following features are available to both Centrex and POTS subscribers:

- Anonymous Call Rejection
- Automatic Callback (Repeat Dialing)
- Automatic Recall (AR)—Call Return
- Call Block (Reject Caller)
- Call Waiting
- Cancel Call Waiting
- Call Transfer
- Customer-Originated Trace
- Do Not Disturb
- Hotline Service

- Interactive Voice Response Functions
- Multiline Hunt Group (MLHG)
- Multiple Directory Numbers
- Speed Call
- Subscriber-Controlled Services and Screening List Editing
- Three-Way Calling
- Usage-Sensitive Three-Way Calling
- Visual Message Waiting Indicator
- Warmline Service

Anonymous Call Rejection

The Cisco BTS 10200 Softswitch supports the anonymous call rejection (ACR) feature as specified in LSSGR module FSD-01-02-1060 (TR-TSY-000567), *Anonymous Call Rejection.*

The ACR feature allows users to reject calls from parties that have set their privacy feature to prevent calling number delivery. When ACR is active the called party receives no alerting of incoming calls that are rejected. The incoming call is rerouted to a denial announcement indicating that private numbers are not accepted by the called party. To complete a call to the party with ACR, the calling party must enter the vertical service code (*82) to activate calling identity delivery and then place a call to the party with ACR. Incoming calls to the called party with ACR are checked even if the called party is offhook.

ACR has multiple activation options as follows:

- Activated permanently at subscription time by service provider.
- Activated by user:
  - The user lifts the handset, and listens for a dial tone.
  - The user presses ***77**. If ACR can be activated, the system returns a confirmation tone.
  - ACR is now activated, and will stay active until it is deactivated using *87.

ACR deactivation options are as follows:

- Service provider deactivation at user request.
- Deactivated by user:
  - The user lifts the handset and listens for a dial tone.
  - The user presses ***87**. The system responds with a confirmation tone.
  - ACR is now deactivated, and will stay inactive until it is activated using *77.

Automatic Callback (Repeat Dialing)

Automatic callback (AC), also called repeat dialing, allows the user to request the system to automatically call the most recently dialed number. The system will keep attempting to call the number for up to 30 minutes, and the remote station will be rung automatically when the called party becomes idle. The system alerts the user with distinctive ringing. Up to 20 AC requests can be active at any time. The service provider can set up this service for the user, or the user can access it on a usage-sensitive basis.

AC is activated as follows:

- The user calls a remote station, receives a busy signal or no answer, and hangs up.
- The user lifts the handset again, and listens for dial tone.
- The user presses ***66**. One of the following scenarios will occur:
  - Audible ring—Indicates that the call setup is being attempted immediately.
  - The delayed processing announcement—This announcement is given to indicate that the line the customer is calling is busy and that the system will attempt to complete the call when the called line is idle.
  - A short term denial announcement. “We are sorry. Your AC request cannot be processed at this time. Please try again later or dial directly.”
  - A long term denial announcement. “The number you are trying to reach cannot be handled by AC. Please dial directly.”
  - A denial announcement indicating that the called party has a call rejection feature active and is not accepting calls from you.

AC is deactivated as follows:

- The user goes off hook, receives a dial tone, and dials the deactivation code, ***86**.
- Once the deactivation code is dialed the user hears an announcement stating that all outstanding AC requests have been deactivated.

Automatic Recall (AR)—Call Return

Automatic Recall (AR), also called Call Return, allows the user to request the system to automatically redial of last incoming call (that is, the station that called the user) when that station becomes idle. The system will keep attempting to call the number for up to 30 minutes, and the remote station will be rung automatically when the called party becomes idle. The system alerts the user with distinctive ringing. Up to 20 AR requests can be active at any time. The service provider can set up this service for the user, or the user can access it on a usage-sensitive basis.

AR is activated as follows:

- The user receives a call (ringing) from a remote station, but does not pick up.
- The user lifts the handset and listens for dial tone.
- The user presses ***69**. One of the following scenarios will occur:
  - Audible ring—Indicates that the call setup is being attempted immediately.
  - The delayed processing announcement—This announcement is given to indicate that the line the customer is calling is busy and that the system will attempt to complete the call once the called line is idle.
  - A short term denial announcement. “We are sorry. Your AR request cannot be processed at this time. Please try again later or dial directly.”
  - A long term denial announcement. “The number you are trying to reach cannot be handled by AR. Please dial directly.”
  - A denial announcement indicating that the called party has a call rejection feature active and is not accepting calls from you.

AR is deactivated as follows:

- The user goes off hook, receives a dial tone, and dials the deactivation code, ***89**.
- Once the deactivation code is dialed the user hears an announcement stating that all outstanding AR requests have been deactivated.

Call Block (Reject Caller)

The call block (reject caller) feature allows the user to block incoming calls from the DN of the last received call. For the call block feature to work, the user must already be subscribed to the selective call rejection (SCR) feature. Once call block is activated against a specified DN, that DN remains in the SCR list of the subscriber. A subscriber who wishes to block callers (like sales calls, etc.) but does not know the caller's DN, can use this feature. Call block can be provided to POTS, CENTREX and MLHG subscribers.

Provisioning call block includes the following CLI operations:

- Configuring the feature table for call block
- Provisioning a two-digit star code (*xx) for call block activation:
  - Adding a star code entry in the vertical service code (VSC) table (for POTS subscribers)
  - Adding a star code entry in the custom dial plan table (for CENTREX subscribers)
- Creating a service with call block and SCR
- Assigning the service to the subscriber via the subscriber service profile table

An idle user (if subscribed to SCR) can dial the call block activation code indicating that the last incoming caller's DN is to be added to their SCR list.

A confirmation tone is given to indicate successful activation. In cases of error or the user not subscribed to call block, a reorder tone is given. If the user is trying to activate call block while on an active call, the user is reconnected to the original call.

The user can deactivate call block for this DN by removing the DN from their SCR list. This is done by using the screen list editing (SLE) function of the SCR feature.

Note For details of the SCR feature, see the "Selective Call Rejection" section on page 14-31.

Call Waiting

The Cisco BTS 10200 Softswitch supports the call waiting (CW) feature as specified in LSSGR module FSD-01-02-1201 (TR-NWT-000571), *Call Waiting*.

CW informs a busy station that another call is waiting through the application of a 300 ms, 440 Hz tone. Ten seconds after the initial tone, a second tone is applied if the waiting call has not been answered. To answer the waiting call and place the original call on hold, the user presses the **Flash** button or **hookswitch**. A subsequent flash returns the user to the original call. Additional flashes can be used to toggle between the two calls as long as they are both still connected. The waiting call hears ringing until it is answered.

When a waiting call is accepted, there are two active sessions. To end the currently active session, the user goes on hook. The user's phone will then ring to indicate that the other caller is still holding. The user can pick up the phone to resume that session.

If a media gateway-connected handset is off hook, but no active call yet exists (that is, receiving a dial tone), then an incoming call receives a station busy tone and CW is not activated.

Only one instance of CW can be active for a given subscriber line at any given time. Thus, if a subscriber line were involved in both an active call and a waiting call, then an additional incoming call attempt results in the caller receiving a busy tone or being forwarded (CFB). The user involved in the CW call is not aware of the additional incoming call attempt.

Cancel Call Waiting

The Cisco BTS 10200 Softswitch supports cancel call waiting (CCW) feature activation via service code *70, as specified in LSSGR module FSD 01-02-1204 (TR-TSY-000572), *Cancel Call Waiting.*

CCW allows a user to disable CW, which also disables the calling identity delivery/call waiting (CIDCW) feature for the duration of a call. CCW is normally included as an integral part of a service package containing the CW and CIDCW features. CCW is useful when the user does not want to be interrupted during an important call or during an outgoing data/fax call.

The user activates and deactivates the CCW feature as follows:

- To make an uninterrupted new call:
  - The user lifts the handset, and listens for a dial tone.
  - The user presses ***70**. The system responds by disabling the CW/CIDCW features and returning three short beeps and then the dial tone.
  - Now CCW is activated for the duration of the call, until the user goes on hook again.
  - After the user goes on hook, the CW/CIDCW service will be back in effect automatically.

Note There are exceptions. If a user was involved in a multiparty call, and received a ringback after going on hook, CCW remains active for the call.

- To make a currently active call go uninterrupted:
  - The user presses **Flash** button or **hookswitch**
  - The user presses ***70**. The system responds by disabling the CW/CIDCW features and returning three short beeps and then the dial tone.
  - Now CCW is activated for the remainder of the current call, until the user goes on hook again.
  - The user presses **Flash** button or **hookswitch** to return to the original call.

After the current call is completely released, the CW service will be back in effect automatically.

Call Transfer

The Cisco BTS 10200 Softswitch supports the call transfer (CT) feature as specified in LSSGR module FSD-01-02-1305 (TR-TSY-000579), *Add On Transfer And Conference Calling Features*.

Note This feature is available only to Centrex subscribers.

CT allows a user to add a third party or second call to an existing two-party call. CT also allows the user to hang up while involved in the two calls and connect the remaining two parties in a two-way connection.

To activate a CT call, a user involved in a stable two-way call takes the following steps:

- The user (the initiating party) presses the **Flash** button or **hookswitch**. This places the remote end on hold and returns a recall dial tone.
- The initiating party dials the DN of the third party.

Note If the initiating party presses the **Flash** button or **hookswitch** before completing dialing, the original two-way connection is reestablished.

- When the third party answers, only the initiating party and the third party can hear and talk. This allows the initiating party to speak privately with the third party before sending the second flash.
- If the initiating party presses the **Flash** button or **hookswitch** after successfully dialing the third party number, a three-way conference is established, regardless of whether the third party answers the call. If the initiating party hangs up after successfully dialing the third-party number, a two-way call is established between the held party and the third party, regardless of whether the third party answers the call.

Customer-Originated Trace

Customer-originated trace (COT) allows users who have been receiving harassing or prank calls to activate an immediate trace of the last incoming call, without requiring prior approval or manual intervention by telephone company personnel.

After an harassing or prank call is terminated, a user who wishes to trace the call goes off hook, receives a dial tone, and dials the COT activation code (*57). When the trace has been completed, the user receives a COT success tone or announcement, such as, "You have successfully traced your last incoming call. Please contact your telephone company for further assistance." (Information about a traced call is made available to the telephone company or to a telephone company-designated agency, usually law enforcement, but not to the user who initiated the trace). Because COT is activated on a per-call basis, the service is deactivated when the user goes on hook.

If the trace cannot be performed, an appropriate tone or announcement is played.

Note For an incoming call to be traced, the incoming call must have been answered by the called party, or a ringing tone or call waiting tone must have started on the called line.

All COT trace records are stored in the EMS of the Cisco BTS 10200 Softswitch for retrieval purposes. A maximum of 10,000 traces are stored in a circular file format (oldest record overwritten).

Do Not Disturb

The do not disturb feature is activated and deactivated by the user. It routes calls destined to the user's DN either to a special do not disturb announcement or to a special tone. A user can dial the activation code (*78) to enable this service, and dial the deactivation code (*79) to disable the service.

Hotline Service

Hotline service is a dedicated private line between a subscriber phone and a predetermined DN. The service is activated by the service provider at the request of the subscriber. When the hotline user picks up the phone, the Cisco BTS 10200 Softswitch rings the predetermined DN instantly.

An exclusive telephone DN is required for the hotline feature, and certain limitations apply to its use:

- None of the VSC star (*) features are available on this line
- The user cannot originate additional outgoing calls by pressing the Flash button or hookswitch. Therefore, the user will not be able to originate multiparty or conference calls from the hotline phone.

Only the service provider can deactivate hotline service.

Note See also the "Warmline Service" section on page 14-32. Warmline service is a combination of hotline service and regular phone service.

Interactive Voice Response Functions

The Cisco BTS 10200 Softswitch supports interactive voice response (IVR) functions for activation of remote call forwarding (RACF) and screening list editing (SLE) features. To use the RACF feature, the user dials a specified DN (assigned by the service provider) and is connected to the appropriate IVR media server. The user enters a personal ID number (PIN) to access the IVR functions, and follows the voice prompts of the IVR server to activate/deactivate or edit their RACF options. To use the SLE feature, the user dials one of several VSC (*xx) numbers and is connected to the appropriate IVR media server. The user follows the voice prompts to edit their screening lists.

For additional details of these features, refer to the following sections:

- Remote Activation of Call Forwarding, page 14-6
- Subscriber-Controlled Services and Screening List Editing, page 14-29

Multiline Hunt Group (MLHG)

The Cisco BTS 10200 Softswitch supports multiline hunt service as specified in LSSGR module FSD-01-02-0802 (TR-TSY-000569), *Multiline Hunt Service.*

A multiline hunt group (MLHG) is a telecommunications channel between two points, such as a telephone company CO/switching center and a call center, PBX or key system. Typically, a business has more stations (telephones) than lines, and hunting features allow sharing of a group of lines by many individual stations for both incoming and outgoing calls. Hunting refers to the process of a call reaching a group of lines. The call tries the first line of the group. If that line is busy it tries the second line, then it hunts to the third, etc. A hunt group is simply a series of lines organized in such a way that if the first line is busy the next line is hunted and so on until a free line is found. Often this arrangement is used on a group of incoming lines. Each line in a MLHG has a terminal number that identifies its position in the group.

There are three types of hunting patterns:

- Regular—Starts from the top of the group list every time, then goes to the first idle line it finds
- UCD (uniform call distribution)—Assigns a call to the most idle line

- Circular—Stops/Starts from the previous line

The originating features are assigned *only on a group basis*. There are no individual features for origination. The terminating features can be assigned as a group, individual, or both.

Originating calls are billed to the individual DN if assigned, otherwise it is billed to the group DN.

If the terminal associated with the Dial DN is busy and it has preferential hunt list, the preferential hunt list is checked for an idle terminal. If all terminals in the preferential hunt list are busy, the system starts regular or circular hunting from the last terminal in the preferential hunt list.

There can be up to 64 preferential hunt lists per MLHG, and a maximum of 18 terminals are allowed in a preferential hunt list. There can be a maximum of 512 terminals in an MLHG.

Multiple Directory Numbers

Multiple directory numbers (MDN) service is also known as teen service. It enables one primary DN and one or two secondary DNs to be assigned to a single line termination. A specific unique ringing pattern is assigned to each DN, so that each incoming call can be individually identified. A distinctive CW tone is also assigned to each DN so that each incoming call can be individually identified when the line is busy.

Billing for this service is charged to the primary telephone number (or to the number designated as the billing DN). If DRCW is activated, MDN is inhibited. For calls originating from a MDN line, the primary DN is used as caller-ID, if an ID is offered to the called party.

Note The MDN feature is available to POTS users only.

Speed Call

The speed call feature is based on LSSGR module FSD-01-02-1101 (TR-TSY-000570), *Speed Calling*.

Speed Call for Individual Subscribers

The speed call feature allows a user to program the phone line so that they can dial selected or frequently called numbers using just one or two digits. After programming the line from their handset, the user can enter the one- or two-digit number, followed by the # symbol or a four-second delay, and the system automatically dials the applicable DN. The programming data is stored in the SC1D (one-digit) or SC2D (two-digit) table of the Cisco BTS 10200 Softswitch. These tables can also be programmed by the service provider via CLI commands.

To program the line, the user listens for a dial tone, then enters a service code. There are two service codes available:

- *74 is used for one-digit speed call, which accommodates up to 8 numbers (2 through 9)
- *75 is used for two-digit speed call, which accommodates up to 30 numbers (20 through 49)

After entering the service code, the user can either enter the end-of-dialing signal (#) or wait four seconds to receive the recall dial tone (stutter tone). After receiving recall dial tone, the user enters the appropriate one-digit or two-digit speed code followed by the complete phone number (including any prefixes such as 1 and the area code). A confirmation tone is then returned to the user, followed by a delay of one to two seconds, and then regular dial tone. Changes to existing programmed speed codes are also made in the manner described above.

After the speed code is programmed, the user can speed call as follows:

1. Go offhook and enter the one- or two-digit speed code instead of the phone number.
2. Press the # symbol or wait for four seconds.
3. The system automatically places the call to the DN associated with the speed code.

Group Speed Call

The group speed call feature allows members of a Centrex group or multiline hunt group (MLHG) to program a list so that they can select and dial frequently called numbers using one or two digits. The programming follows the same process as described for individual subscribers, using the same service codes. (See the "Speed Call for Individual Subscribers" section on page 14-28.) A customer is allowed both one- and two-digit speed calling. In the case of shared lists for group speed calling, only one of the customers sharing the list may have the customer-changeable option. The switch is able to provide a given line with both a shared list and an individual list with the requirement that one must be a one-digit list and the other a two-digit list. If speed calling is assigned to a multiline hunt group, all members of that group have access to the shared group speed call list. If, however, a line in the group also has individual speed calling, then the individual speed calling will take precedence over the group speed calling.

Subscriber-Controlled Services and Screening List Editing

Subscriber-controlled services allow individual users to screen and manage their incoming calls. The user can specify lists of DNs for which incoming calls are to be screened and given any of the following treatments:

- Selective Call Forwarding
- Selective Call Acceptance
- Selective Call Rejection
- Distinctive Ringing/Call Waiting

The user can create screening lists, add DNs to the lists, and edit the lists, via the screening list editing (SLE) function as described in the Telcordia document *GR-220-CORE, Screening List Editing*. The user performs the SLE functions, and activates/deactivates the services, via VSCs. Each VSC connects the user to the appropriate IVR media server functions. The VSCs are preprovisioned in the Cisco BTS 10200 Softswitch as listed below.

Note For each feature, a pair of preprovisioned VSCs is listed. Either VSC in the pair can be used to access the IVR server to perform all review, edit, activation, and deactivation functions. The service provider has the option of reprovisioning VSCs as desired.

- Selective Call Forwarding—*63/*83
- Selective Call Acceptance—*64/*84
- Selective Call Rejection—*60/*80
- Distinctive Ringing/Call Waiting—*61/*81

The individual features are described in the following sections.

The Cisco BTS 10200 Softswitch provides the necessary CORBA interface for service providers interested in building web-based applications that permit users to perform these SLE functions via the web. A web CORBA software development kit (SDK) is provided as part of the Cisco BTS 10200 Softswitch product.

Selective Call Forwarding

The selective call forwarding (SCF) feature screens each incoming call to determine whether the DN is on a list of DNs, provisioned by the user (called party), to receive automatic forwarding treatment. The user also sets the forward-to number. Any incoming calls from DNs that are on the SCF screening list are forwarded to the designated number. Any incoming calls from DNs not on the SCF screening list receive regular treatment (they are not forwarded).

The user accesses and controls the SCF properties from their handset via a VSC and IVR interaction. The user can add or delete DNs on the screening list, change the forward-to number, review the screening list, and activate or deactivate SCF. As a convenience, the system allows the user to add or delete the last caller’s number to the screening list by entering **01** at the prompt. The system recognizes the “01” command and translates it into the last-received DN.

The following conditions apply to the use of the SCF feature:

- TWC and CW are disabled while the user is editing the list or activating/deactivating the SCF feature.
- If SCF is active, it takes precedence over all other call forwarding features including CW and DRCW. It does not take precedence over SCR.
- The forward-to number defined in SCF can be the same number used by other call forwarding features, or it can be different.
- Once the SCF feature is activated, it remains active until it is deactivated.

Selective Call Acceptance

The selective call acceptance (SCA) feature screens each incoming call to determine whether the DN is on a list of DNs, provisioned by the user (called party), to be accepted. Any incoming calls from DNs on the SCA screening list are accepted, but any incoming calls from DNs not on the SCA screening list are blocked (receive terminating treatment).

The user accesses and controls the SCA properties from their handset via a VSC and IVR interaction. The user can add or delete DNs on the screening list, review the screening list, and activate or deactivate SCA. As a convenience, the system allows the user to add or delete the last caller’s number to the screening list by entering **01** at the prompt. The system recognizes the 01 command and translates it into the last-received DN.

The following conditions apply to the use of the SCA feature:

- TWC and CW are disabled while the user is editing the list or activating/deactivating the SCA feature.
- Once the SCA feature is activated, it remains active until it is deactivated.

Selective Call Rejection

The selective call rejection (SCR) feature screens each incoming call to determine whether the DN is on a list of DNs, provisioned by the user (called party), to be blocked. The blocked caller is connected to an announcement stating that their call is not presently being accepted by the called party. Any incoming calls from DNs not on the SCR screening list receive regular treatment (they are not blocked).

The user accesses and controls the SCR properties from their handset via a VSC and IVR interaction. The user can add or delete DNs on the screening list, review the screening list, and activate or deactivate SCR. As a convenience, the system allows the user to add or delete the last caller's number to the screening list by entering **01** at the prompt. The system recognizes the 01 command and translates it into the last-received DN.

The following conditions apply to the use of the SCR feature:

- TWC and CW are disabled while the user is editing the list or activating/deactivating the SCR feature.
- If SCR is active, it takes precedence over all call forwarding features including CW and DRCW.
- Once the SCR feature is activated, it remains active until it is deactivated.

Note The call block/reject caller feature (see "Call Block (Reject Caller)" section on page 14-24) provides another way for the user to selectively reject calls from the last caller.

Distinctive Ringing/Call Waiting

The distinctive ringing/call waiting (DRCW) feature screens each incoming call to determine whether the DN is on a list of DNs, provisioned by the user (called party), to receive special ringing or CW alerting treatment. If the incoming DN is on the DRCW screening list, the system alerts the user with a special ring or a special CW tone. Any incoming calls from DNs not on the SCR screening list receive regular treatment (regular ringing and CW alerting tones).

The user accesses and controls the DRCW properties from their handset via a VSC and IVR interaction. The user can add or delete DNs on the screening list, review the screening list, and activate or deactivate DRCW. As a convenience, the system allows the user to add or delete the last caller's number to the screening list by entering **01** at the prompt. The system recognizes the 01 command and translates it into the last-received DN.

Once the DRCW feature is activated, it remains active until it is deactivated.

Three-Way Calling

The Cisco BTS 10200 Softswitch supports the three-way calling (TWC) feature as specified in LSSGR module FSD-01-02-1301 (TR-TSY-000577), *Three-Way Calling*.

TWC is a feature provisioned by the service provider in response to a request from the subscriber. TWC allows a subscriber to add a third party to an existing two party conversation.

To activate a TWC, a user involved in a stable two-way call takes the following steps:

- The user presses the **Flash** button or **hookswitch**. This places the remote end on hold.
- The user hears the recall dial tone (three tones and then a dial tone), indicating the system is ready to receive the DN for the third party.
- The user dials the DN of the third party.

Note If the user presses the **Flash** button or **hookswitch** before completing dialing, the original two-way connection is reestablished.

- When the third party answers, only the user (who initiated the TWC) and the third party can hear and talk. This allows the user to speak privately with the third party before sending the second flash.
- If the user presses the **Flash** button or **hookswitch** after successfully dialing the third party number, a three-way conference is established.
- If either of the called parties (the two stations remote from the initiating party) hangs up, the call continues as a single-call session.
- If the initiating party hangs up during a TWC, all parties are disconnected.

Note During a TWC, the CW feature does not work for the party that initiated the TWC, but does work for the two called parties.

When in a TWC, the last party added can be disconnected by using the **Flash** button or **hookswitch**.

Usage-Sensitive Three-Way Calling

The Cisco BTS 10200 Softswitch supports usage-sensitive three-way calling (USTWC) feature as specified in LSSGR module FSD-01-02-1304 (TR-TSY-000578), *Usage-Sensitive Three-Way Calling*.

USTWC allows a user to add a third party to an existing two party conversation. It provides all the functionality of TWC (see the "Three-Way Calling" section on page 14-31) without requiring the user to subscribe to the service. The service provider may charge differently for the use of this service. The usage-sensitive features can be enabled/inhibited per user by turning on/off the usage-sensitive option for the user.

The user activates and uses this service in the same manner as TWC.

Visual Message Waiting Indicator

The visual message waiting indicator (VMWI) is associated with the voice mail service. When a call is forwarded to a voice mail system, and the caller leaves a message, the voice mail system sends the Cisco BTS 10200 Softswitch an MWI signal via SIP. The Cisco BTS 10200 Softswitch forwards a VMWI signal to the called party, and the called party's telephone indicator light turns on. When the called party retrieves the message, the voice mail system signals the Cisco BTS 10200 Softswitch to clear the VMWI indicator, and the light on the telephone turns off.

Warmline Service

Warmline service is a combination of hotline service (see the "Hotline Service" section on page 14-27) and regular phone service on the same line. The service is activated by the service provider at the request of the subscriber. The service provider provisions a timeout parameter in the FEATURE table (default is four seconds), and the warmline service uses that timeout value as follows:

- Use of warmline for regular phone service—The user takes the handset off hook, receives dial tone, and starts dialing a regular call before the timeout expires.
- Use of warmline as a hotline—The user takes the handset off hook, receives dial tone, but does not dial any digits. After the timeout expires, the system automatically calls the predetermined DN.

Note This timeout is a switch-level timeout common to all subscribers, and normally not changed on a per-subscriber basis.

Note This variable timeout feature operates with MGWs that are compliant with MGCP1.0 (per IETF document *RFC 2705*) or higher. For MGWs compliant with MGCP0.1 only, the timeout is not variable.

An exclusive telephone DN is required for the warmline feature, and certain limitations apply to its use:

- None of the VSC star (*) features are available on this line
- The user cannot originate additional outgoing calls by pressing the Flash button or hookswitch. Therefore, the user will not be able to originate multiparty or conference calls from the warmline phone.

Only the service provider can deactivate warmline service

Default Office Service ID

One service ID (the default office service ID, typically ID=999) is reserved for provisioning of switch-based features. These switch-based features can include certain network features and certain usage-sensitive features, as described below. The service provider must provision this service ID in the service table, and define these features in the feature table.

Note The default value of the office service ID is 999. This value is provisionable in the CA-CONFIG table.

Note Refer to the *Cisco BTS 10200 Softswitch CLI Guide* for specific feature provisioning requirements.

Caution The system does not validate or restrict the provisioning of features on this office service ID. However, entries other than the ones listed below will have undefined results. Do not enter features other than the ones listed below.

- Network features—When provisioned, the system makes these features available for all subscriber lines:

  Note See Chapter 13, "Network Features" for details of these network features.

  - Local network portability (LNP)
  - Toll-free services (8XX)
  - Emergency services (911)
  - Busy line verification (BLV)
- Usage-sensitive features—When provisioned, the system makes these features available to all subscribers without the need to actually subscribe these features to individual lines:

  Note These features can also be assigned to individual subscribers using other service IDs.

  - Usage-sensitive three-way calling (USTWC)
  - Customer originated trace (COT)

- Automatic callback activation and deactivation—AC_ACT and AC_DEACT (or AC, if the AC umbrella feature was created)
- Automatic recall activation and deactivation—AR_ACT and AR_DEACT) (or AR, if the AR umbrella feature was created)

CHAPTER 15

Feature Interactions

This section describes the interactions among the various features offered by the Cisco BTS 10200 Softswitch. It includes the following topics:

The service provider defines the features and services for their system, and assigns these services to subscribers. A service is a collection of features. Each feature has static information, stored in the feature table, regarding triggers, feature defaults, associated features, and vertical service codes. When a service is created, the system automatically maps the service with the triggers. The system uses internal information about triggers and trigger detection points (TDP), based on the ITU-T CS-2 call model, to process features during a call. The system has internal information to handle features that interact with other features at specific detection points. The system also handles features that are inhibited when certain other features are already invoked on the subscriber line.

Note See Chapter 13, "Network Features" and Chapter 14, "Subscriber Features" for detailed descriptions of individual features.

Overview of Features and Services

The service provider uses CLI commands to provision the features and services for their system. The feature table contains all the static information for a feature, such as:

- Trigger detection point (TDP)
- Trigger ID (TID)
- Trigger type
- Vertical service code, if any
- Feature server
- Feature defaults

- Associated features, if any (for example, CFU_ACT and CFU_DEACT can be associated with CFU)

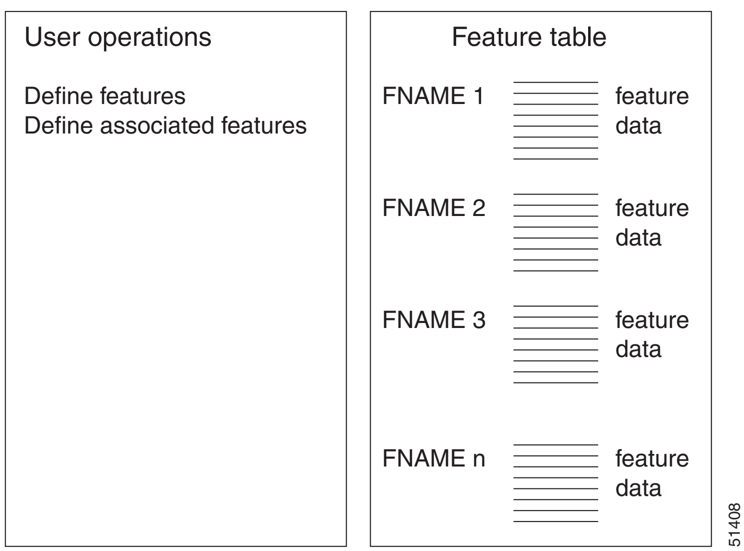
A service is a collection of one or more features (up to 10 features per service). Each service is identified by a unique service ID numeric value. Each feature within a service may have one or more triggers. When a service is created, the system automatically registers the triggers. During call processing, the services are triggered based on TDP and TID. The Cisco BTS 10200 Softswitch supports provisioning of up to 50 services per subscriber.

Note Limitation—If services are defined (by the service provider) such that they share the same TDP-TID pair, the Cisco BTS 10200 Softswitch will support a maximum of 10 services for that TDP-TID pair.

Creation of Features and Services

Figure 15-1 through Figure 15-3 show the process of creating features, assembling features into services, and assigning services to individual subscribers (or subscriber groups). The provisioning operations listed in these figures are performed using CLI commands. Feature provisioning steps are provided in the *Cisco BTS 10200 Softswitch Operations Manual*. Detailed reference information on commands and parameters (tokens) is provided in the *Cisco BTS 10200 Softswitch CLI Reference Guide*.

Figure 15-1 Defining Features and Associated Features



Note Associated features, such as CFU_ACT and CFU_DEACT, must be defined first, and then they can be linked (associated) with the main feature (CFU in this case).

Figure 15-2 Assigning Features to Services

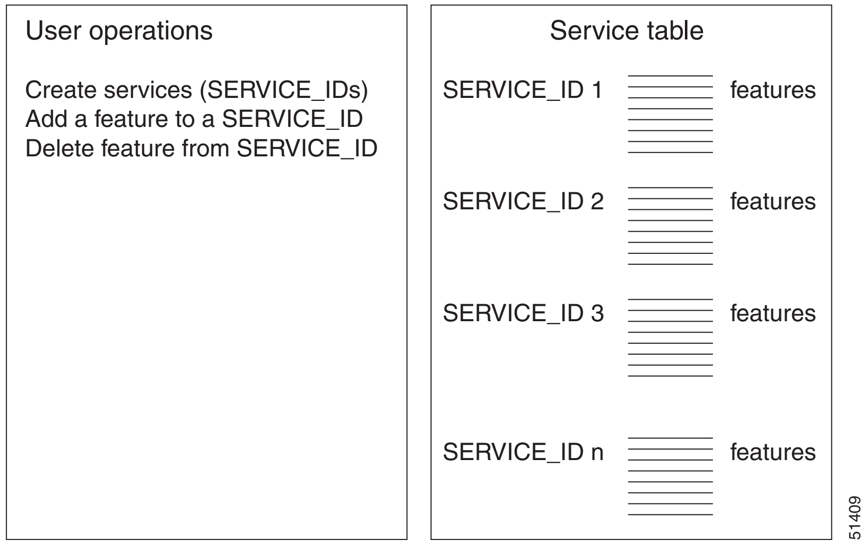
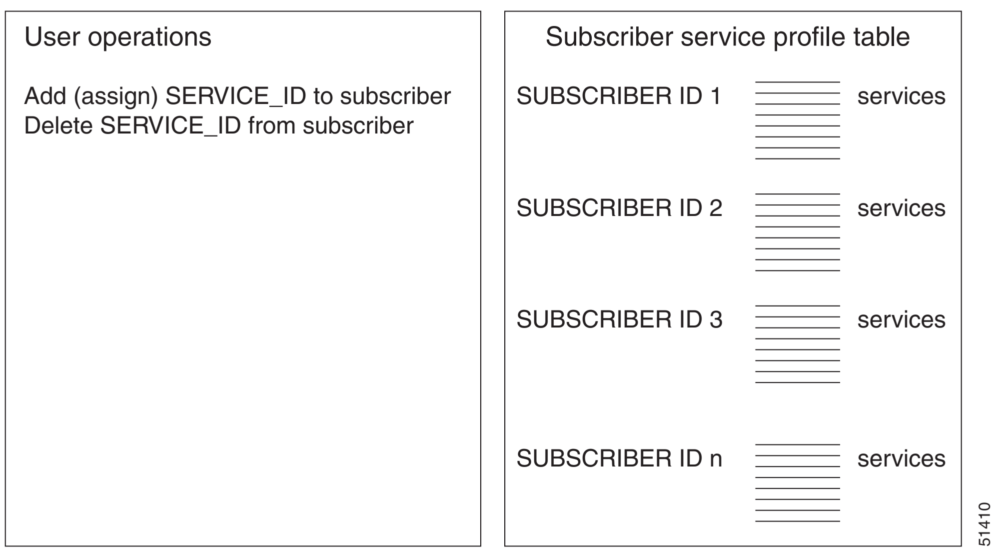


Figure 15-3 Assigning Services to Subscribers



Trigger Detection Points and Trigger IDs

The TDPs for the Cisco BTS 10200 Softswitch are illustrated in Figure 15-4 and Figure 15-5.

Note The basic call module of the Cisco BTS 10200 Softswitch contains the triggers specified in the standard ITU-T CS-2 call model, as well as several additional triggers.

Figure 15-4 Cisco BTS 10200 Softswitch Originating Call States and Triggers

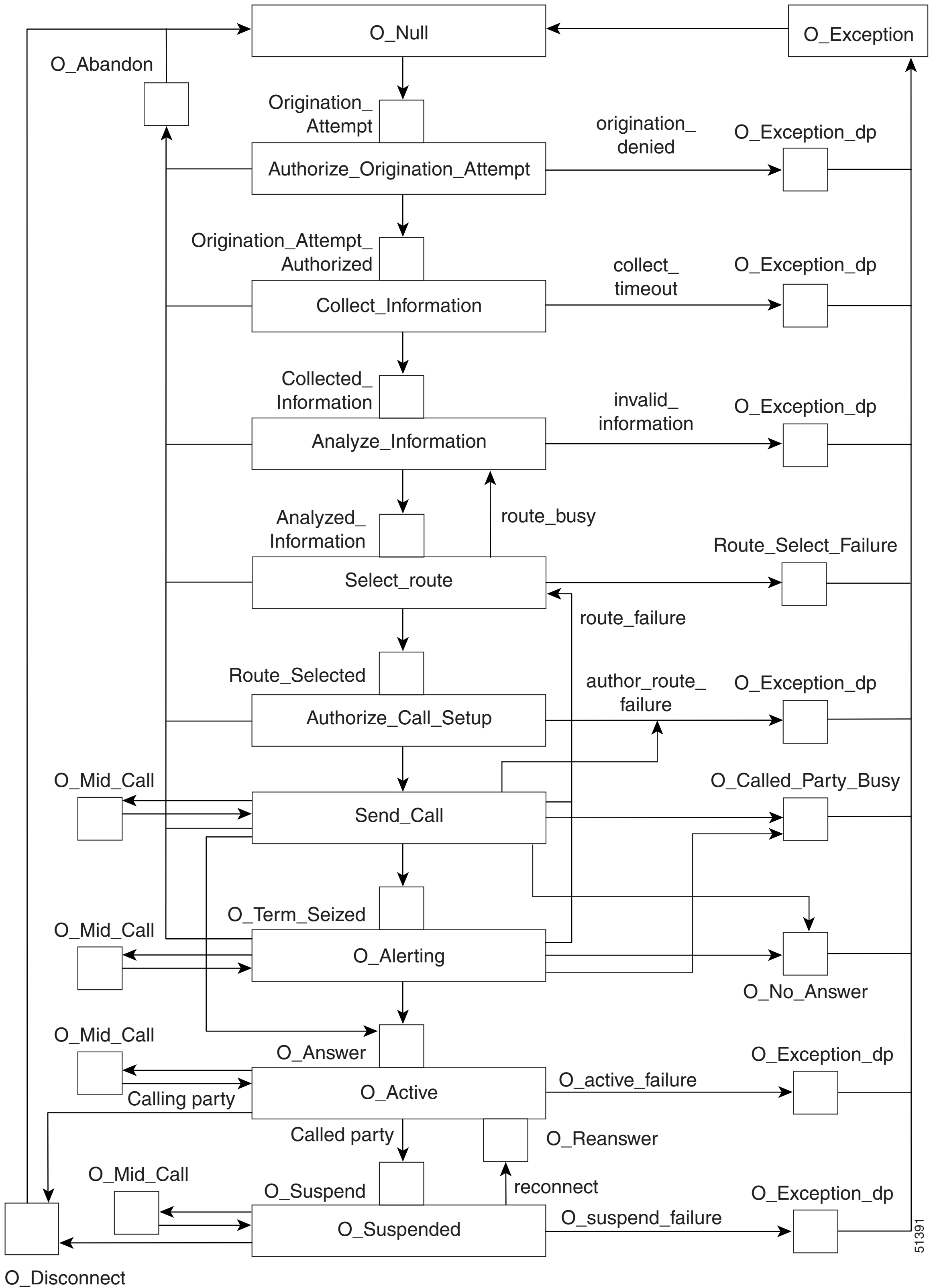


Figure 15-5 Cisco BTS 10200 Softswitch Terminating Call States and Triggers

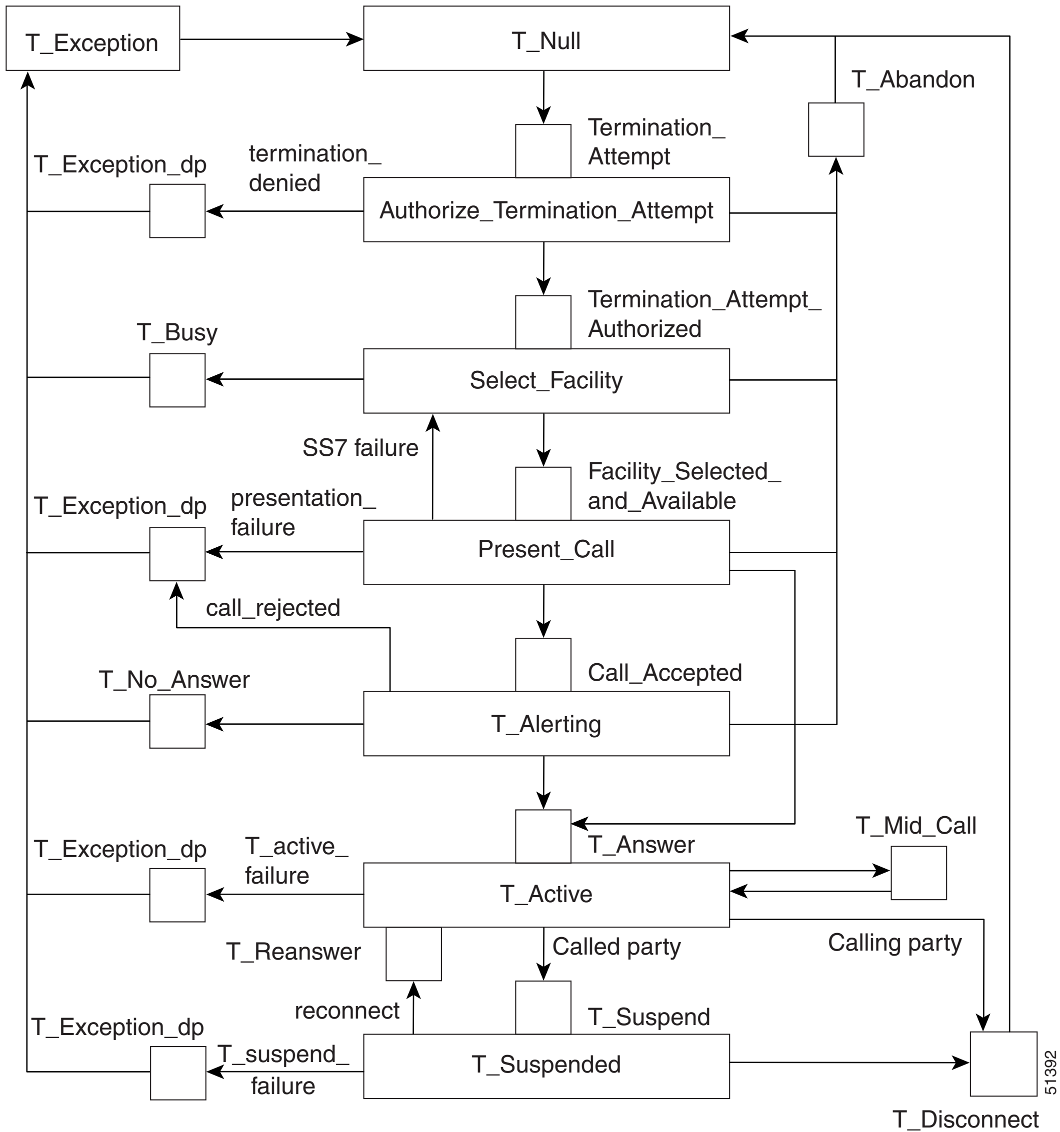


Table 15-1 lists the trigger detection point(s) and trigger ID(s) for each feature.

Tip The feature acronyms used in this section are defined in the "Acronyms for Features" section on page 15-19.

Note Vertical service codes (VSC) are assigned to some of these features. See the "Vertical Service Codes" section in Chapter 13, "Network Features," for VSC information.

Table 15-1 Features and Service Triggers

Feature	Description	Trigger Detection Point	Trigger ID
8XX	Toll free number	COLLECTED_INFORMATION	SPECIFIC_DIGIT_STRING
911	Emergency service	COLLECTED_INFORMATION	911_TRIGGER
AC	Automatic callback (includes AC_ACT and AC_DEACT)		
AC_ACT	Automatic callback activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
AC_DEACT	Automatic callback deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
ACR	Anonymous call rejection	TERMINATION_ATTEMPT_ AUTHORIZED	TERMINATION_ATTEMPT_ AUTHORIZED
ACR_ACT	ACR activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
ACR_DEACT	ACR deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
ANI	Automatic number ID	no TDPs	no triggers
AR	Automatic recall (includes AR_ACT and AR_DEACT)		
AR_ACT	AR activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
AR_DEACT	AR deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
BLV	Busy line verification	TERMINATION_ATTEMPT	BLV
CBLK	Call block (reject caller)	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CCW	Cancel call waiting	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CDP	Customize dial plan	COLLECTED_INFORMATION	CUSTOMIZE_DIALING_PLAN
CFB[1]	Call forwarding busy	T_BUSY	T_BUSY
CFBVA	CFB variable activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CFBVD	CFB variable deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CFNA[1]	Call forwarding no answer	CALL_ACCEPTED	CALL_ACCEPTED
CFNAVA	CFNA variable activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CFNAVD	CFNA variable deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CFU[1]	Call forwarding unconditional	TERMINATION_ATTEMPT_ AUTHORIZED	TERMINATION_ATTEMPT_ AUTHORIZED
CFUA	CFU activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CFUD	CFU deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE

Table 15-1 Features and Service Triggers (continued)

Feature	Description	Trigger Detection Point	Trigger ID
CHD[2]	Call hold	O_MID_CALL	O_SWITCH_HOOK_FLASH_ IMMEDIATE
		T_MID_CALL	T_SWITCH_HOOK_FLASH_ IMMEDIATE
CIDCW	Caller ID with call waiting	T_BUSY	T_BUSY
CIDS	Delivery function of calling identity delivery and suppression	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CIDSS	Suppression function of calling identity delivery and suppression	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CNAB	Calling name delivery blocking	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CNAM	Calling name delivery	FACITLITY_SELECTED_AND_ AVAILABLE	TERMINATION_RESOURCE_ AVAILABLE
CND	Calling number delivery	FACITLITY_SELECTED_AND_ AVAILABLE	TERMINATION_RESOURCE_ AVAILABLE
CNDB	CND blocking (toggles the privacy indicator)	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
COS	Class of service screening	COLLECTED_INFORMATION	COS_TRIGGER
COT	Customer originated trace	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
CPRK	Call park access code	O_MID_CALL	O_SWITCH_HOOK_FLASH_ IMMEDIATE
		T_MID_CALL	T_SWITCH_HOOK_FLASH_ IMMEDIATE
CPRK_RET	CPRK retrieval access code	no TDPs	no triggers
CT	Call transfer	O_MID_CALL	O_SWITCH_HOOK_FLASH_ IMMEDIATE
		T_MID_CALL	T_SWITCH_HOOK_FLASH_ IMMEDIATE
CW	Call waiting	T_BUSY	T_BUSY
DACWI	DACWI on DID calls	TERMINATION_ATTEMPT_ AUTHORIZED	TERMINATION_ATTEMPT_ AUTHORIZED
DND	Do Not Disturb	TERMINATION_ATTEMPT_ AUTHORIZED	TERMINATION_ATTEMPT_ AUTHORIZED
DND_ACT	Do not disturb activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
DND_DEACT	Do not disturb deactivation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
DPN	Directed call pickup without barge-in	no TDPs	no triggers
DPU	Directed call pickup with barge-in	no TDPs	no triggers

Table 15-1 Features and Service Triggers (continued)

Feature	Description	Trigger Detection Point	Trigger ID
DRCW	Distinctive ringing call waiting	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
DRCW_ACT [3]	DRCW activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
		T_ANSWER	T_ANSWER
GSC1D	1-digit group speed call	COLLECTED_INFORMATION	SC1D_TRIGGER
GSC2D	2-digit group speed call	COLLECTED_INFORMATION	SC2D_TRIGGER
HOTLINE	Hotline feature	O_ATTEMPT_AUTHORIZED	O_ATTEMPT_AUTHD
ISFG	Incoming simulated facility group (SFG) for Centrex	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
LIDB	LIDB query for the CNAM feature	TERMINATION_ATTEMPT	CNAM
LNP	Local number portability	COLLECTED_INFORMATION	LNP_TRIGGER
MDN	Multiple directory numbers	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
OSFG	Outgoing SFG for Centrex	ROUTE_SELECTED	ROUTE_SELECTED
RACF [4]	Remote activation of call forwarding	T_ANSWER	T_ANSWER
RACF_PIN	RACF PIN change	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
		T_ANSWER	T_ANSWER
RCF	Remote call forwarding	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
SC1D	1-digit speed call	COLLECTED_INFORMATION	SC1D_TRIGGER
SC1D_ACT	1-digit speed call activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
SC2D	2-digit speed call	COLLECTED_INFORMATION	SC2D_TRIGGER
SC2D_ACT	2-digit speed call activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
SCA	Selective call acceptance	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
SCA_ACT [5]	SCA activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
		T_ANSWER	T_ANSWER
SCF	Selective call forwarding	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
SCF_ACT [6]	SCF activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
		T_ANSWER	T_ANSWER
SCR	Selective call rejection	TERMINATION_ATTEMPT_AUTHORIZED	TERMINATION_ATTEMPT_AUTHORIZED
SCR_ACT [7]	SCR activation	COLLECTED_INFORMATION	VERTICAL_SERVICE_CODE
		T_ANSWER	T_ANSWER

Table 15-1 Features and Service Triggers (continued)

Feature	Description	Trigger Detection Point	Trigger ID
TWC	Three way call	O_MID_CALL	O_SWITCH_HOOK_FLASH_ IMMEDIATE
		T_MID_CALL	T_SWITCH_HOOK_FLASH_ IMMEDIATE
USTWC	Usage sensitive three way call	O_MID_CALL	O_SWITCH_HOOK_FLASH_ IMMEDIATE
		T_MID_CALL	T_SWITCH_HOOK_FLASH_ IMMEDIATE
WARMLINE	Warmline feature	O_ATTEMPT_AUTHORIZED	O_ATTEMPT_AUTHD

1. Multiple call forwarding is enabled by default, but can be disabled.
2. For Centrex users, an access code (typically provisioned as *52) can be used in conjunction with the CHD feature.
3. DRCW_ACT provides access to interactive voice response (IVR) server for activation, screening list setup and editing, and deactivation of DRCW.
4. RACF is an IVR-based function that can be used with CFU.
5. SCA_ACT provides access to IVR server for activation, screening list setup and editing, and deactivation of SCA.
6. SCF_ACT provides access to IVR server for activation, screening list setup and editing, and deactivation of SCF.
7. SCR_ACT provides access to IVR server for activation, screening list setup and editing, and deactivation of SCR.

Note The features DRCW_ACT, SCA_ACT, SCF_ACT, and SCR_ACT are collectively referred to as screening list editing (SLE) functions.

Feature Precedence

If the call processing function in the CA detects a TDP, it sends a trigger, if applicable, to the appropriate FS. After receiving the trigger, the FS controls the call as needed. With multiple features assigned to a single service package, it is possible for more than one feature to trigger at the same TDP. When that occurs, the Cisco BTS 10200 Softswitch uses the feature precedence table (Table 15-2), along with the subscription information of the subscriber, to determine which feature to provide. If multiple features are included in a service package (as they often are), it is important for the service provider to be able to identify to their subscribers which feature takes precedence at a particular TDP. The TDPs for each feature, and the precedence conditions for specific feature pairs, are defined in the system and cannot be changed. The precedence functionality is implemented in accordance with the LSSGR specification.

Note See the *Cisco BTS 10200 Softswitch CLI Reference Guide*, service-trigger table, for additional information about triggers for multiple features that are grouped into a service package.

Tip As shown in Figure 15-4 and Figure 15-5, a call reaches TDPs in a specified sequence consistent with the CS-2 call model. A feature triggered at an earlier TDP is **not** said to have precedence over a feature triggered at a later TDP. Precedence refers to a scenario in which two features occur **at the same TDP**, and the Cisco BTS 10200 Softswitch uses internally programmed rules to determine which feature takes precedence at that TDP.

Table 15-2 Feature Precedence

No.	Trigger Detection Points (TDP)	Precedence
1	TERMINATION_ ATTEMPT_AUTHORIZED	• SCR has priority over: SCF DRCW SCA ACR CFU DND MDN DACWI • ISFG has priority over CFU • SCF has priority over: DRCW SCA ACR CFU DND MDN DACWI • DRCW has priority over: SCA ACR [1] CFU [1] DND MDN DACWI • SCA has priority over: ACR CFU DND MDN DACWI • ACR has priority over: CFU [1] DND MDN DACWI • CFU has priority over: DND MDN DACWI • DND has priority over: MDN DACWI • MDN has priority over: DACWI
2	FACILITY_SELECTED _AND_AVAILABLE	• CNAM has priority over CND (CNAM includes CND. If a subscriber has both features, CNAM is provided.)
3	T_BUSY	• CIDCW has priority over: CW CFB • CW has priority over CFB
4	O_ATTEMPT_AUTHORIZED	• HOTLINE has priority over WARMLINE

Table 15-2 Feature Precedence (continued)

No.	Trigger Detection Points (TDP)	Precedence
5	COLLECTED_INFORMATION	• CDP and VSC are independent features, with different triggers • CDP has priority over COS • Call agent does not report COS trigger for VSC dialed
6	O_MIDCALL and T_MIDCALL	• CT has priority over TWC

1. If all three features (DRCW, ACR, and CFU) are assigned to a subscriber, CFU takes precedence over ACR and DRCW.

If a called party (subscriber) is assigned both the DND and CFB features, and has activated them, an incoming call will be forwarded to the CFB forward-to DN whether the called party is busy or not.

Feature Inhibition

Feature inhibition is defined as an interaction where the subscriber's current feature status inhibits other features from being provided. The inhibition functionality is implemented in accordance with the LSSGR specification. This table is preset in the system and cannot be modified. Table 15-3 shows how features are inhibited by various other features.

Tip If a call is released at a particular TDP, the later TDPs will not be reached, and the features associated with those later TDPs will not occur. This is a direct result of the TDP sequencing, and is not defined as inhibition. Feature inhibition occurs when a trigger is reached, but one of the features associated with the TDP has been inhibited by a feature that occurred at an earlier TDP.

Note MDC refers to midcall, which is a function activated when the user presses the Flash button or hookswitch during a call. In Table 15-3, MDC is treated as an internal feature, with the following meaning:

Certain features inhibit MDC. This means that when one of those features is invoked, the Cisco BTS 10200 Softswitch ignores the Flash/hookswitch function.

MDC inhibits several features. This means that those features cannot be supplied to the user after the user presses the Flash/hookswitch.

Table 15-3 Feature Inhibition

Feature	Feature State	Inhibited Features	Remarks
911	Invoked	CIDCW COS CW MDC	
ACR	Deactivated	ACR	
BLV	Invoked	ACR CFB CFNA CFU CIDCW CNAM CND COS CT CW DND DRCW ISFG MDC MDN OSFG RACF SCA SCF SCR TWC USTWC	
CCW	Invoked	CIDCW, COT, CW	
CFU	Invoked	CFU	Applicable only if MCF flag for CFU in the feature table is set to no (N).
	Deactivated	CFU	
CFB	Invoked	CFB	Applicable only if MCF flag for CFB in the feature table is set to no (N).
	Deactivated	CFB	
CFNA	Invoked	CFNA	Applicable only if MCF flag for CFNA in the feature table is set to no (N).
	Deactivated	CFNA	

Table 15-3 Feature Inhibition (continued)

Feature	Feature State	Inhibited Features	Remarks
CHD	Invoked	911 AC, AC_ACT, AC_DEACT AR, AR_ACT, AR_DEACT CBLK CFBVA, CFBVD CFNAVA, CFNAVD CFUA, CFUD CIDCW CNDB COT CPRK, CPRK_RET CW DPN DPU	
CIDCW	Invoked	CIDCW CNAM CND CW MDC TWC	
CNDB	Invoked	CNDB	
COT	Invoked	CIDCW CT CW MDC TWC USTWC	
CT	Invoked	AC, AC_ACT, AC_DEACT AR, AR_ACT, AR_DEACT CBLK CIDCW COT CPRK_RET CT CW MDC TWC	
CW	Invoked	CIDCW CNAM CND CW MDC TWC	
DND	Deactivated	DND	
DRCW	Deactivated	DRCW	

Table 15-3 Feature Inhibition (continued)

Feature	Feature State	Inhibited Features	Remarks
DRCW_ACT	Invoked	CHD CIDCW CT CW MDC TWC	
HOTLINE	Assigned (provisioned by service provider)	CT MDC TWC USTWC VSC based features	
MDC	Invoked	AC, AC_ACT, AC_DEACT ACR_ACT, ACR_DEACT AR, AR_ACT, AR_DEACT CBLK CFBVA, CFBVD CFUA, CFUD CIDCW CNDB COT CPRK_RET CW DND_ACT, DND_DEACT DPN DPU MDC SC1D_ACT, SC2D_ACT	
SC1D_ACT	Invoked	CFUA	
SC2D_ACT	Invoked	CFUA	
SCA	Activated	ACR, DND	
	Deactivated	SCA	
SCA_ACT	Invoked	CHD CIDCW CT CW MDC TWC	
SCF	Deactivated	SCF	
SCF_ACT	Invoked	CHD CIDCW CT CW MDC TWC	
SCR	Deactivated	SCR	

Table 15-3 Feature Inhibition (continued)

Feature	Feature State	Inhibited Features	Remarks
SCR_ACT	Invoked	CHD CIDCW CT CW MDC TWC	
TWC	Invoked	AC, AC_ACT, AC_DEACT AR, AR_ACT, AR_DEACT CBLK CIDCW COT CPRK_RET CT CW MDC TWC	
WARMLINE	Assigned (provisioned by service provider)	CT MDC TWC USTWC VSC based features	

Examples of Interactions

Feature interaction examples are presented in this section for the following scenarios:

- Three-way calling
- Call waiting
- Calling number delivery

Three-Way Call Interaction

The following interactions pertain to three-way calling (TWC):

- TWC can interact with itself. Given three parties involved in a call, any party with the TWC feature who has not already added can flash and add on another party. In other words TWC can be recursively used to join more than 3 parties.
- A customer who has initiated TWC cannot initiate TWC again while in a TWC conference call.
- The use of TWC does not restrict the call waiting capabilities of the customers who did not initiate TWC.
- The initiator of TWC does not receive CW calls or the CW tone while in a TWC mode or while a party is on hold.
- When a line that is not the initiator of TWC receives a CW call, a flash is not interpreted as a request for TWC (that is, CW takes precedence over TWC in this case).
- TWC can be used to disable CW during an existing conversation.
- When CW is in effect, it takes precedence over TWC. When CW is disabled, TWC treatment is given when the customer flashes.
- If a customer activates cancel call waiting and then originates TWC, CW remains disabled until all connections are torn down. If either of the noncontrolling parties of TWC disconnect (or are disconnected by the controller), CW remains disabled for the remaining two-way connection.
- If the initiator of TWC hangs up with a party on hold, the initiator will be rung back and connected to the held party on answer. If the initiator's CW was disabled prior to hanging up on the held party, it remains disabled after the customer answers the ringback.
- Flashes are ignored after a two-way call has been set up to a 911 attendant. This means that for the duration of the 911 call, the TWC feature cannot be used.
- A customer involved in a two-way call can flash and use TWC to add-on a 911 attendant. All subsequent flashes will be ignored.

Call Waiting Interaction

The following interactions pertain to call waiting (CW):

- If a line has call forwarding on busy (CFB) and CW, the CW service normally takes precedence over CFB.
- Given a line that has both CFB and CW and is in a talk state, the first call attempting to terminate is treated as a CW call. Subsequent termination attempts will be call forwarded (.that is, CFB is invoked only if a call is already waiting).

- If CW treatment cannot be given (for example, because the line is dialing or ringing), then CFB takes effect.
- CW and cancel call waiting (CCW) cannot be invoked simultaneously.
- When CW is disabled via CCW, it only applies to calls terminating at the subscriber line. It does not affect calls terminating at other subscriber lines.
- During a call to a 911 attendant, the CW service is inhibited (that is, no CW tone).

Calling Number Delivery Interaction

The following interactions pertain to the calling number delivery (CND) feature:

- No CND Data is sent during or after a CW tone.
- CND data is sent for held and waited parties during the first silent interval of ringback that results from the customer going on hook in response to a CW tone.

Acronyms for Features

Table 15-4 lists the abbreviations and acronyms used in this section.

Table 15-4 Acronyms for Features

Abbreviation	Definition
8XX	Toll free service
911	Emergency services
AC	Automatic callback
AC_ACT	Automatic callback activation
AC_DEACT	Automatic callback deactivation
ACR	Anonymous call rejection
ACR_ACT	Anonymous call rejection activation
ACR_DEACT	Anonymous call rejection deactivation
AR	Automatic recall
AR_ACT	Automatic recall activation
AR_DEACT	Automatic recall deactivation
BLV	Busy line verification
CBLK	Call block (reject caller)
CCW	Cancel call waiting
CDP	Custom dial plan
CFB	Call forward on busy
CFBVA	Call forward busy variable activation
CFBVD	Call forward busy variable deactivation
CFNA	Call forward on no answer

Table 15-4 Acronyms for Features (continued)

Abbreviation	Definition
CFNAVA	CFNA variable activation
CFNAVD	CFNA variable deactivation
CFU	Call forward unconditional
CFUA	Call forward unconditional activation
CFUD	Call forward unconditional deactivation
CHD	Call hold
CIDB	Calling identity delivery blocking
CIDCW	Calling identity delivery on call waiting
CIDS	Delivery function of calling identity delivery and suppression
CIDSS	Suppression function of calling identity delivery and suppression
CNAB	Calling name delivery blocking
CNAM	Calling name delivery
CND	Calling number delivery
CNDB	Calling number delivery blocking
COS	Class of service screening
COT	Customer originated trace
CPRK	Call park
CPRK_RET	Call park retrieve
CT	Call transfer
CW	Call waiting
DACWI	Distinctive alerting/call waiting indication
DID	Direct inward dial
DND	Do not disturb
DOD	Direct outward dial
DPN_O and DPN_T	Call pickup without barge-in (originate/terminate)
DPN	Call pickup without barge-in
DPU	Call pickup with barge-in
DPU_O and DPU_T	Call pickup with barge-in (originate/terminate)
DRCW	Distinctive ringing/call waiting
DRCW_ACT	Distinctive ringing/call waiting activation
FIM	Feature interaction manager
ISFG	Incoming simulated facility group
LIDB	Line information database

Table 15-4 Acronyms for Features (continued)

Abbreviation	Definition
LNP	Local number portability
MCF	Multiple call forwarding
MDC	Mid-call
MDN	Multiple directory numbers
MLHG	Multiline hunt group
OSFG	Outgoing simulated facility group
RACF	Remote activation of call forwarding
RACF-PIN	Remote activation of call forwarding personal ID number
RCF	Remote call forwarding
SCA	Selective call acceptance
SC1D_ACT	1-digit speed call activation
SC2D_ACT	2-digit speed call activation
SCA_ACT	Selective call acceptance
SCF	Selective call forwarding
SCF_ACT	Selective call rejection
SCR	Selective call forwarding
SCR_ACT	Selective call rejection
SLE	Screen list editing
TF	Toll free
TWC	Three-way calling
USTWC	Usage sensitive three-way calling
VSC	Vertical service code

CHAPTER 16

Announcements

The Cisco BTS 10200 Softswitch supports announcement features by sending announcement requests to a customer-supplied announcement server. An announcement request can be generated by the Cisco BTS 10200 in response to either of the following conditions:

- A call was released (did not go through), and an accompanying release cause code is activated on the Cisco BTS 10200. The Cisco BTS 10200, in turn, signals the announcement server to play a designated audio file.
- The service provider has provisioned all calls to the target DN to be routed automatically to a designated announcement.

The announcement server accesses prerecorded audio files that can be played to the caller. The audio files are provided by one of the following servers:

- Cisco AS53xx series Announcement Server
- Cognitronics CX500 Media Resource Server

Note Contact Cisco for details about these servers. The announcement server is customer supplied.

Release Cause Codes

Industry standard release cause code specifications are available in the following documents:

- ANSI document *T1.650-1995, Integrated Services Digital Network (ISDN) – Usage of the Cause Information Element in Digital Subscriber Signaling System Number 1 (DSS1)*
- ITU-T Recommendation Q.850, *Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part*

The service provider can link any of the supported cause codes to any announcement ID, and each announcement ID can be linked to a specific audio file. The system will trigger the appropriate recording to play when a cause code is activated in the system. Table 16-1 shows the default mapping of cause codes to announcement IDs and announcement files. The service provider can use CLI commands to provision the following changes to the default mapping:

- Use the **change release-cause** command to change the mapping of release cause codes to announcement IDs.
- Use the **change announcement** command to change the mapping of announcement IDs to audio files.

Note If no announcement is available for a specific cause code, a reorder tone is played to the calling party.

Tip With the Cisco AS53xx series Announcement Server, the service provider can enter new announcement file names and use their own audio announcement files. The announcement files must be in 8-bit mu-Law encoded, Next/Sun AU format (.au extension).

DNs Provisioned for Routing to Announcement

The service provider can provision all calls to a specific DN to be routed automatically to a designated announcement. This is provisioned using CLI commands in the DN2SUBSCRIBER table, as follows:

- Change the administrative status of the announcement service by setting the **status** token to **annc.**
- Designate the announcement to be played by setting the **annc-id** token to the appropriate announcement id. The announcement ID must be one that is listed in Table 16-1.

Mapping of Cause Codes and Announcement Files

Table 16-1 shows the default mapping of cause codes to announcement IDs and announcement files.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
152—Receiver off hook 170—Collected digits timeout	10	ann_id_10.au	If you'd like to make a call, please hang up and try again. If you need help, hang up and then dial your operator.	10	If you'd like to make a call, please hang up and try again. If you need help, hang up and then dial your operator.
34—No circuit available 38—Network out of order 153—TG overflow 155—DID TG overflow	11	ann_id_11.au	We're sorry, all circuits are busy now. Please try your call again later.	11	We're sorry, all circuits are busy now. Please try your call again later.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
6—Channel unacceptable 29—Facility rejected 31—Normal, unspecified 57—Bearer capability not authorized 58—Bearer capability not presently available 65—Bearer capability not implemented 66—Channel type not implemented 81—Invalid call reference value 82—Identified channel does not exist 83—A suspended call exists, but this call identity does not 84—Call identity in use 85—No call suspended 86—Call having the requested call identity has been cleared 87—User not member of Centrex user group 88—Incompatible destination 89—Invalid endpoint reference 90—Nonexistent Centrex user group 91—Invalid transit network selection 92—Too many pending add-party requests 95—Invalid message, unspecified	12	ann_id_12.au	We're sorry, your call did not go through. Please try your call later.	12	We're sorry, your call did not go through. Please try your call later.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
96—Mandatory IE[1] missing 97—Message type nonexistent or not implemented 98—Message not compatible with call state, or message type nonexistent or not implemented 99—IE nonexistent or not implemented 100—Invalid IE contents 101—Message not compatible with call state or threshold exceeded 102—Recovery on timer expiry 103—Parameter nonexistent or not implemented, passed on 104—Incorrect message length 110—Message, with unrecognized parameter, discarded 111—Protocol error, unspecified 127—Interworking, unspecified	12				
42—Switching equipment congestion 173—TG congestion	13	ann_id_13.au	We're sorry, all circuits are busy now. Please try your call again later.	60	We're sorry. Due to heavy calling, we cannot complete your call at this time. Will you please hang up and try your call later. If your call is urgent, please try again now.
8—Prefix 0 dialed but not allowed 9—Prefix 1 dialed but not allowed 157—Prefix 1 or 0 present	14	ann_id_14.au	We're sorry, it is not necessary to dial a 1 or 0 when calling this number. Please hang up and try your call again.	14	We're sorry, it is not necessary to dial a 1 or 0 when calling this number. Please hang up and try your call again.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
10—Prefix 1 not dialed but required 156—Prefix 1 or 0 absent	15	ann_id_15.au	We're sorry, you must first dial a 1 or 0 when calling this number. Please hang up and try your call again.	15	We're sorry, you must first dial a 1 or 0 when calling this number. Please hang up and try your call again.
159—HNPA area code absent	16	ann_id_16.au	We're sorry. When placing a local call it is now necessary to dial an area code followed by the 7-digit local number. Please hang up and redial using the complete 10-digit number.	16	We're sorry. When placing a local call it is now necessary to dial an area code followed by the 7-digit local number. Please hang up and redial using the complete 10-digit number.
3—No route to destination 28—Invalid number format (address incomplete) 158—Prefix error 175—New NPA before start date 176—Old NPA after end date	17	ann_id_17.au	We're sorry, your call cannot be completed as dialed. Please check the number and dial again.	17	We're sorry, your call cannot be completed as dialed. Please check the number and dial again.
1—Unallocated (unassigned) number 4—Vacant area code or CO code 23—Redirection to new destination	18	ann_id_18.au	The number you dialed is not in service. Please check the number and dial again.	18	The number you dialed is not in service. Please check the number and dial again.
20—Subscriber absent 169—Centrex station OOS	19	ann_id_19.au	We're sorry. You have reached a number that has been disconnected or is no longer in service. If you feel you have reached this recording in error, please check the number and try your call again.	59	We're sorry. You have reached a number that has been disconnected or is no longer in service. If you feel you have reached this recording in error, please check the number and try your call again.
151—Temporarily disconnected	20	ann_id_20.au	The party you are calling has temporarily disconnected their service.	20	The party you are calling has temporarily disconnected their service.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announce-ment ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
163—Feature restricted or not subscribed (CA)	24	ann_id_24.au	We're sorry, your call cannot be completed as dialed. Please check your instruction manual or call the business office for assistance.	315	We're sorry, but the service you are trying to use is not available on this line.
164—Authorization code invalid (CA)	25	ann_id_25.au	The authorization code you have dialed is not valid. Please hang up, check the authorization code, and try again.	837	The authorization code you have dialed is not valid. Please hang up, check the number, and try again.
1101—Do not disturb (DND) activated on local station	30	ann_id_30.au	Do not disturb has been successfully activated for you. All calls to you will be blocked until you deactivate it.	89	The feature is now activated.
1102—DND not activated on local station	31	ann_id_31.au	We're sorry. Do not disturb could not be activated successfully.	671	Cannot be activated at this time.
1103—DND deactivated on local station	32	ann_id_32.au	Do not disturb has been successfully deactivated for you. Calls to you will not be blocked.	88	The feature is now deactivated.
1104—DND not deactivated on local station	33	ann_id_33.au	We're sorry. Do not disturb could not be deactivated successfully.	673	Cannot be processed at this time.
1105—DND currently active on remote station	34	ann_id_34.au	The party you are trying to call has the do not disturb feature currently activated. Please try your call again later.	3	We're sorry, the customer you are trying to reach cannot be disturbed. Will you please hang up and try your call again later.
2—No route to specified transit network 167—Carrier access code (CAC) OOS 168—Cut-through not supported	57	ann_id_57.au	We're sorry, the long-distance company you have selected is unable to complete your call at this time. Please contact your long-distance company for assistance.	69	We're sorry, the long-distance company you have selected is unable to complete your call at this time. Please contact your long-distance company for assistance.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announce-ment ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
1053—Anonymous call reject	61	ann_id_61.au	Your call has been properly delivered, but the party you are trying to reach is not accepting calls from callers who do not allow delivery of their telephone number. Please hang up, do not block the delivery of your number, and call again.	299	We're sorry. The party you are calling does not wish to talk to callers who block their numbers. If you wish to reach this party, please hang up and place your call again without blocking your number.
1062—Customer originated trace successful	70	ann_id_70.au	The last call has been traced. If you wish to investigate further, contact your telephone company annoyance call center with the date and time of the trace.	293	Your call has been traced. If you wish to investigate further, contact your telephone company annoyance call center with the date and time of the trace.
1063—Customer originated trace denied	71	ann_id_71.au	We're sorry. The customer originated trace cannot be performed from this line.	315	We're sorry, but the service you are trying to use is not available on this line.
1064—Customer originated trace facility trouble	72	ann_id_72.au	We're sorry. Due to the telephone company facility trouble, your trace cannot be completed at this time. Please try again later.	312	We're sorry. Due to telephone company facility trouble your call cannot be completed at this time. Will you try your call again later?
1065—Trace buffer empty	73	ann_id_73.au	We're sorry. The trace cannot be performed. The number of the last person who called you is not available.	247	We're sorry. The number of the last person who called you is not available.
1066—Trace already done	74	ann_id_74.au	We're sorry, but the last call has already been successfully traced.	329	The feature is already successfully activated.
1061—Trace not available for calling number out of service area	75	ann_id_75.au	We're sorry. The trace cannot be performed. The calling number is out of our service area.	291	This is the customer originated trace feature. A trace cannot be performed because the calling number is out of our service area.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announce-ment ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
1050—Invalid deactivation, already deactivated	80	ann_id_80.au	The feature is already successfully deactivated.	330	The feature is already successfully deactivated.
1071—Rejected by selective call acceptance feature	81	ann_id_81.au	The party you called has its selective call acceptance feature activated. Your number is not on the call acceptance list.	210	The party you called has its selective call acceptance feature activated. Your number is not on the call acceptance list.
1072—Rejected by selective call rejection feature	82	ann_id_82.au	The party you are calling is not currently accepting calls. Please call back at another time.	878	The party you are calling is not currently accepting calls. Please call back at another time.
1081—All automatic recall requests deactivated	83	ann_id_83.au	All outstanding automatic recall requests have been deactivated.	287	All outstanding automatic recall requests have been deactivated.
1082—Short-term denial of automatic recall [2]	84	ann_id_84.au	Your automatic recall request cannot be processed at this time. Please try again or dial directly.	201	Your automatic recall request cannot be processed at this time. Please try again or dial directly.
1083—Long-term denial of automatic recall	85	ann_id_85.au	The number you are trying to reach cannot be handled by the automatic recall features. Please dial directly.	285	The number you are trying to reach cannot be handled by the automatic recall and callback features. Please dial directly.
1084—Denial of automatic recall for anonymous target DN	86	ann_id_86.au	We're sorry. The number you are trying to reach is private and cannot be called using the automatic recall feature.	283	We're sorry. The number you are trying to reach is private and cannot be called using the automatic recall feature.
1085—Resume automatic scanning	87	ann_id_87.au	We're sorry. Your called party has just become busy. The system will continue checking the line. You will be notified by ringing when the party is free.	286	We're sorry. Your called party has just become busy. The system will continue checking the line. You will be notified by special ringing when the party is free.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announce-ment ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
1086—Activate automatic scanning	88	ann_id_88.au	The number you are calling is busy. The system will check the line for the next 30 minutes. You will be notified by ringing when the party is free.	288	The number you are calling is busy. The system will check the line for the next 30 minutes. You will be notified by special ringing when the party is free.
1091—All automatic callback requests deactivated	89	ann_id_89.au	All outstanding automatic callback requests have been deactivated.	284	All outstanding automatic callback requests have been deactivated.
1092—Short-term denial of automatic callback [2]	90	ann_id_90.au	We're sorry. Your automatic callback request cannot be processed at this time. Please try again or dial directly.	201	We're sorry. Your automatic callback request cannot be processed at this time. Please try again or dial directly.
1093—Long-term denial of automatic callback	91	ann_id_91.au	The number you are trying to reach cannot be handled by the automatic callback features. Please dial directly.	285	The number you are trying to reach cannot be handled by the automatic recall and callback features. Please dial directly.
1073—Activate reject caller	92	ann_id_92.au	Reject caller feature has been activated successfully. Your last calling number has been added to your selective call rejection list.	89	The feature is now activated.
165—CAC temporarily OOS	112	ann_id_12.au	We’re sorry, the long distance company you have dialed is experiencing a temporary service problem. Please try your call later.	51	We're sorry, the long-distance company you have dialed is experiencing a temporary service problem. Please try your call again later.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	**Contents**	**Number**	**Contents**
22—Number changed	118	ann_id_18.au	The number you dialed is not in service. Please check the number and dial again.	59 300 301	<old number> has been changed. The new number is <new number>. Please make a note of it. <old number> has been changed. The new number is <new number>.
27—Destination out of order	119	ann_id_19.au	We're sorry. You have reached a number that has been disconnected or is no longer in service. If you feel you have reached this recording in error, please check the number and try your call again.	12	We're sorry, your call did not go through. Please try your call later.
1052—Invalid entry	124	ann_id_24.au	We're sorry, your call cannot be completed as dialed. Please check your instruction manual or call the business office for assistance.	353	We're sorry. The service or option requested is not available. Please check your instruction manual or call the business office for assistance.
162—Invalid CAC received	157	ann_id_57.au	We're sorry, the long-distance company you have selected is unable to complete your call at this time. Please contact your long-distance company for assistance.	50	We're sorry, your call cannot be completed with the access code you dialed. Please check the code and try again, or call your long-distance company for assistance.
21—Call rejected 41—Temporary failure 44—Requested circuit or channel not available 47—Resource unavailable, unspecified	212	ann_id_12.au	We're sorry, your call did not go through. Please try your call later.	65	Due to network difficulties, your call cannot be completed at this time. Please try your call again later.

Table 16-1 Default Mapping of Cause Codes to Prerecorded Announcements (continued)

Cause Code Number and Name	Announcement ID	Announcements Provided on Cisco AS53xx Announcement Server		Announcements Provided on Cognitronics Media Resource Server	
		File Name	Contents	Number	Contents
160—CAC present but not required	217	ann_id_17.au	We’re sorry, your call cannot be completed as dialed. Please check the number and dial again.	23	We're sorry, it is not necessary to dial a long-distance company access code for the number you have dialed. Please hang up and try your call again.
1051—Resources unavailable	312	ann_id_12.au	We’re sorry, your call did not go through. Please try your call later.	346	We're sorry. We are unable to complete your call at this time. Please try your call again later.
161—CAC absent but required	317	ann_id_17.au	We’re sorry, your call cannot be completed as dialed. Please check the number and dial again.	52	We're sorry, a long-distance company access code is required for the number you have dialed. Please dial your call again with the access code.
(no number)—Call park activated	901	ann_id_901.au	We're sorry that you are still waiting. Please continue to hold.	323	The person you are waiting for will answer your call soon. Please stay on the line.
(no number)—Caller not reconnected after call park	902	ann_id_902.au	We're sorry. Your call has not been reconnected within the allotted time. Please try your call again later.	320	The person you are calling is busy. Please try again later.

1. IE = information element
2. Cause Code numbers 1082 and 1092 both invoke Cognitronics announcement number 201. However, the Cisco BTS 10200 passes additional information to the Cognitronics system indicating whether the 1082/recall or the 1092/callback announcement is required.

CHAPTER 17

Feature Tones

Introduction and References

The Cisco BTS 10200 Softswitch supports special tones on various subscriber and operator features by sending MGCP messages to the gateways. These tones are based on the information in the following documents:

- Telcordia document *GR-506-CORE, Signaling for Analog Interfaces*
- Telcordia document *TR-NWT-506, Issue 3, Signaling*
- Telcordia document *GR-590-CORE, Call Pickup Features (FSD 01-02-2800)*
- Telcordia document *GR-317-CORE, Switching System Generic Requirements for Call Control Using the Integrated Services Digital Network User Part (ISDNUP)*
- Telcordia document *GR-219-CORE, Distinctive Ringing/Call Waiting (FSD 01-01-1110).*
- IETF document *RFC 2705, Media Gateway Control Protocol (MGCP) Version 1.0*

List of Tones Applicable to Specific Features

Table 17-1 lists the tones used with each feature, and the condition that triggers the sending of each tone.

Note See the Glossary for the meaning of acronyms used in this table.

Table 17-1 Feature Tones

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
AC	ALERTING PATTERN 3	
ACR	No special tone delivered	
ACRA ACRD	CONFIRMATION TONE	Anonymous call rejection (ACR) was successfully activated or deactivated by user actions.
	REORDER TONE	ACR was not successfully activated or deactivated by user actions.
AR	ALERTING PATTERN 3	

Table 17-1 Feature Tones (continued)

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
BLV/OI	REORDER TONE	Normal access is not available. There is a local office problem. The line is momentarily unavailable. No-test access is not available.
	BUSY VERIFICATION	CFU is activated on the terminating line. Terminating line is a data-only line or a denied line.
	PERMANENT SIGNAL TONE	Line up to receiver off-hook tone. Terminating line receiving a permanent signal announcement. Terminating line is high and wet (battery and ground shorted) or high and dry (off hook for an extended period).
CW CIDCW	CW TONE	If called party has MDN feature: primary DN matched. If called party is in Centrex system with DACWI: there is no extension for the number dialed. If called party has DRCW feature: calling party is not on the DRCW screening list.[2]
	CW TYPE 2	If called party has MDN feature: second DN matched. If called party is in Centrex system with DACWI: extension exists for the number dialed.
	CW TYPE 3	If called party has MDN feature: third DN matched.
	CW TYPE 4	If called party has DRCW feature: calling party is on the DRCW screening list.[2]
	STUTTER TONE	For Centrex subscriber with CHD feature and currently on an active call: A third party calls in, and the called party hears a call-waiting tone. The called party presses Flash button or switchhook to place the current remote station on hold, and hears the stutter tone. The called party has the following options: • Press Flash button or switchhook again to return to the original call. • Dial a designated vertical service code (VSC)—typically *52—to be connected to the new calling party; the first calling party is kept on hold.
	TONES OFF	No tones will be played for CW or CIDCW. (Tones are turned off under certain special circumstances.)
	ALERTING PATTERN 1	Alerting pattern (ringing) is provided to the calling party and called party, as applicable, for all reconnect, re-ring, callback and recall scenarios.

Table 17-1 Feature Tones (continued)

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
CCW	CONFIRMATION TONE	User in two-way call cancels call waiting.
	DIAL TONE	POTS or Centrex user picks up phone to cancel call waiting.
	STUTTER TONE	User places the other party on hold (CHD) and then activates CCW while call is still on hold.
	ALERTING PATTERN 1	User goes on hook with the other party still on hold; the system provides alerting pattern (ringing).
CDP	DIAL TONE	User is granted access to an outside (public) line, typically after dialing 9.
	ALERTING PATTERN 3	Member of a Centrex group receives an incoming call from the group attendant.
CFU	REMINDER RING TONE	Alerting pattern (ringing) is provided on the called station to indicate that a call has been received and automatically forwarded.
CFU-ACT	STUTTER TONE	User has successful activated CFU from the handset.
	DIAL TONE	User has dialed the CFU-ACT star code, and the system is ready to receive digits for the forward-to DN. 1-second timer elapses following the confirmation tone.
	CONFIRMATION TONE	Centrex user successfully activates extension forwarding. If the user has multiple call forwarding (MCF), the user has successfully activated a chain call forwarding scenario POTS user receives ROUTE SELECTED DIALING PLAN.
	REORDER TONE	The CFU-ACT attempt was not successful due to • Attempt to activate CFU when it was already activated • Attempt to forward calls to a DN that could not be reached • Attempt to forward call from a DN to itself.
CFU-DEACT	CONFIRMATION TONE	User successfully deactivates CFU.
	DIAL TONE	1-second timer elapses following the confirmation tone.
	REORDER TONE	User attempts to deactivate CFU when it was already deactivated
CFB-ACT and CFNA-ACT	DIAL TONE	User has dialed the CFB-ACT or CFNA-ACT star code, and the system is ready to receive digits for the forward-to DN.
	CONFIRMATION TONE	User successfully activates CFB or CFNA.
	DIAL TONE	1-second timer elapses following the confirmation tone.
CFB-DEACT and CFNA-DEACT	CONFIRMATION TONE	User has dialed the CFB-DEACT or CFNA-DEACT star code, and CFB or CFNA has been deactivated.
	DIAL TONE	Issued after a 1-second timer elapses following the confirmation tone.

Table 17-1 Feature Tones (continued)

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
CHD	STUTTER TONE	For Centrex subscriber (controlling party) currently on an active call: The controlling party places the other party on hold by pressing the Flash button or switchhook, and hears the stutter tone. Controlling party has the following options: • Press Flash button or switchhook again to return to the original call. • Dial a designated vertical service code (VSC)—typically *52—hear the stutter tone again, then dial the number of a third party. The first calling party is kept on hold.
	ALERTING PATTERN 1	Alerting pattern (ringing) is provided to the calling party and called party, as applicable, for all reconnect, re-ring, callback and recall scenarios.
CNAM CND	No special tone delivered	
CNDB CNAB CIDB CIDS	DIAL TONE	User has dialed the star code for the identity blocking feature, and the system is ready to receive digits for the DN to be called.
COS: Account Codes	CONFIRMATION TONE	The system prompts the user to enter the account code.
COS: Authorization Codes	CONFIRMATION TONE	The system prompts the user to enter the authorization code.
CPRK	REORDER TONE	User has dialed the call park (CPRK) access code, but is not subscribed to the CPRK feature. The user is subscribed to the CPRK feature, and has dialed the CPRK access code, but the CPRK attempt was not successful. **Note** In this case (CPRK attempt was not successful), the reorder tone is played for two seconds, and then the user is reconnected to the original call.
	STUTTER TONE	User presses Flash button or switchhook to park the call.
CPRK_RET	REORDER TONE	The user is subscribed to the CPRK feature, and has dialed the CPRK access code, but is unable to retrieve the call.
	STUTTER TONE	The user enters the CPRK_RET access code, and the system is waiting for the user to dial the extension against which the parked call should be retrieved.
CT/TWC	ALERTING PATTERN 1	User hangs up with one party on hold.
DACWI	ALERTING PATTERN 3	Distinctive ring pattern.

Table 17-1 Feature Tones (continued)

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
DPN	STUTTER TONE	User has dialed DPN access code, and DPN access has been granted.
	REORDER TONE	Reorder tone is returned to the user who initiated a DPN request when any of the following occurs: • The DPN feature has not been assigned to the requesting line. • The dialed extension is not assigned in the business group dialing plan. • The line associated with the dialed extension is not being rung. (Note that "being rung" should not include being given call-waiting treatment.) • The call has been answered, picked up, or abandoned. • The requesting line is not allowed to pick up the particular call because of being assigned the fully restricted terminating or the denied termination feature.
DPU	STUTTER TONE	User has dialed DPU access code, and DPU access has been granted.
	REORDER TONE	Reorder tone is returned to the user who initiated a DPU request when any of the following occurs: • The dialed extension is not assigned in the business group dialing plan. • The line associated with the dialed extension is not assigned the DPU feature. • The line associated with the dialed extension is neither being rung nor involved in a stable two-way call. (Note that "being rung" should not include being given call-waiting treatment. Note also that DPU should not allow a user to barge-in on the controller of a multiway connection, that is, a call-waiting configuration, a call-hold configuration, or a conference call.) • The call is abandoned by the caller before the DPU request is recognized or has been picked up by a line without DPU assigned. • The requesting line is not allowed to pick up the particular call because of being assigned the fully restricted terminating or the denied termination feature.
	CONFIRMATION TONE	Barge-in connection is being processed and connection will occur within one second. **Note** Confirmation tone is repeated twice.
DRCW	ALERTING PATTERN 1	DN of incoming call is *not* on the DRCW screening list.
	ALERTING PATTERN 6	DN of incoming call is on the DRCW screening list.
	CW TONE	DN of incoming call is *not* on the DRCW screening list.
	CW TYPE 4	DN of incoming call is on the DRCW screening list.
Emergency—911	ALERTING PATTERN 1	After a normal two-party call, the user presses the Flash button or hookswitch, dials 911, and then hangs up before the 911 operator answers.

Table 17-1 Feature Tones (continued)

Feature	Tone	Condition(s) That Cause the Specified Tone To Be Played [1]
MDN	ALERTING PATTERN 1	Station is on hook and there is an incoming call to primary DN.
	ALERTING PATTERN 4	Station is on hook and there is an incoming call to secondary DN.
	ALERTING PATTERN 5	Station is on hook and there is an incoming call to the third DN.
	CW TONE	Station is off hook and there is an incoming call to primary DN.
	CW TYPE 2	Station is off hook and there is an incoming call to secondary DN.
	CW TYPE 3	Station is off hook and there is an incoming call to the third DN.
MWI [3]	MWI TONE	User, who is subscribed to MWI service, has a message waiting.
MIDCALL	STUTTER TONE	After pressing Flash button or hookswitch and the system acknowledges it is as a valid midcall action.
	ALERTING PATTERN 1	User goes on hook with the other party still on hold; the system provides alerting pattern (ringing).
SC1D-ACT SC2D-ACT	STUTTER TONE	Stutter tone is used once after the user enters the *74 (SC1D activation) or *75 (SC2D activation) to begin the process of collecting the information required to provision one of the speed call slots. After a speed call slot has been successfully provisioned, the user will again receive the stutter tone to signify that the speed call slot was successfully provisioned.
VMWI [3]	STUTTER TONE	User, who is subscribed to VMWI service, has a message waiting, but the serving MGW does not have a visual indicator.

1. When more than one condition is listed for a single tone, any one of the conditions can cause the tone to be played.
2. For more information on the screening list, see the "Subscriber-Controlled Services and Screening List Editing" section on page 14-29.
3. MWI = message waiting indicator; VMWI = visual message waiting indicator.

Tone Frequencies and Cadences

Table 17-2 lists the frequency and cadence for tones applicable to subscriber and operator features. Tones are requested by the Cisco BTS 10200 Softswitch and delivered to the subscriber or operator by the MGW. Some MGWs can be provisioned to play tone cadences different than the ones described in this table.

Table 17-2 Subscriber and Operator Tone Descriptions

Tone	Frequency (Hz)	Cadence Played by MGW
Alerting pattern (ringing) 1	440 + 480	2 sec on, 4 sec off, repeating
Alerting pattern (ringing) 2	440 + 480	0.8 sec on, 0.4 sec off, 0.8 sec on, 4.0 sec off, repeating
Alerting pattern (ringing) 3	440 + 480	0.4 sec on, 0.2 sec off, 0.4 sec on, 0.2 sec off, 0.8 sec on, 4 seconds off, repeating
Alerting pattern (ringing) 4	440 + 480	0.3 sec on, 0.2 sec off, 1 sec on, 0.2 sec off, 0.3 sec on, 4 sec off, repeating
Alerting pattern (ringing) 5	440 + 480	0.5 sec on once
Alerting pattern (ringing) 6	440 + 480	1 sec on, 3sec off, repeating

Table 17-2 Subscriber and Operator Tone Descriptions (continued)

Tone	Frequency (Hz)	Cadence Played by MGW
Busy verification (used for operator BLV [1])	440	2 sec burst, followed by 0.5 sec burst every 10 sec
CW tone	440	0.3 sec on once
CW Type 1	440	0.3 sec on once
CW Type 2	440	0.1 sec on, 0.1 sec off, 2 times
CW Type 3	440	0.1 sec on, 0.1 sec off, 3 times
CW Type 4	440	0.1 sec on, 0.1 sec off, 0.3 sec on, 0.1 sec off, 0.1 sec on
Confirmation tone	350 + 440	0.1 sec on, 0.1 sec off, 3 times
Dial tone	350 + 440	steady on
Line busy tone	480 + 620	0.5 sec on, 0.5 sec off, repeating
Message waiting indicator tone	350 + 440	10 bursts (0.1 sec on, 0.1 sec off), then steady on
Off-hook warning tone (receiver off-hook tone)	1400 + 2060 + 2450 + 2600	0.1 sec on, 0.1 sec off, repeating
Permanent signal (used for operator BLV [1])	480	Steady on
Reminder ring tone (ring splash)	440 + 480	0.5 sec ring
Reorder tone	480 + 620	0.25 sec on, 0.25 sec off, repeating
Ringback tone (audible ringing)	440 + 480	2 sec on, 4 sec off (repeated)
Stutter (recall) dial tone	350 + 440	3 bursts (0.1 sec on, 0.1 sec off), then steady on

1. BLV = busy line verification

Table 17-3 lists the maintenance tones used for continuity testing. See the Telcordia document GR-317-CORE for additional details.

Table 17-3 Maintenance Tone Descriptions

Tone	Frequency (Hz)	Description
2010-Hz continuity tone	2010	Used for single-tone test under either of the following conditions: • The circuit is a 4-wire circuit at both the transceiver end and the distant end • The circuit is a 2-wire circuit at the transceiver end
1780-Hz continuity tone	1780	Used for dual-tone test with a 4-wire circuit at the transceiver end and a 2-wire circuit at the distant end

Defined Cause Codes

For reference purposes, defined cause codes are listed in Table A-1 through Table A-7.

Table A-1 Cisco BTS 10200 Softswitch Defined Cause Codes—Normal Events

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Normal Event			
1. [Note 1]	1. Unallocated number	1. Unallocated (Unassigned) number	N/A
2.	2. No route to specified transit network	2. No route to specified transit network	N/A
3. [Note 1]	3. No route to destination	3. No route to destination	N/A
4. [Note 2]	N/A	6. Channel unacceptable	
5. [Note 3]	N/A	7. Call awarded and being delivered in an established channel	
6.	16. Normal clearing	16. Normal call clearing	N/A
7.	17. User busy	17. User busy	401. Phone is already off hook
	—	—	402. Phone is already on hook
8.	18. No user responding	18. No user responding	Notified event
9.	19. User alerting, no answer	19. User alerting, no answer	Notified event
10. [Usr->Nw]	21. Call rejected	21. Call rejected	
11. [Note 1]	22. Number changed	22. Number changed	
12.[1]	26. Misrouted call to a ported number	N/A	

Table A-1 Cisco BTS 10200 Softswitch Defined Cause Codes—Normal Events (continued)

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
13. [Note 3]	N/A	26. Nonselected user clearing	
14.[2]	27. Query on Release (QoR) number not found (No procedures for US networks		
15.	27. Destination out of order	27. Destination out of order	520
16. [Note 1]	28. Address incomplete	28. Invalid number format (incomplete number)	N/A
17. [nw->usr]		29. Facility rejected	N/A
18. [Note 4]		30. Response to STATUS ENQUIRY	N/A
			500. The transaction could not be executed because the endpoint is unknown
19. [Note 5]	31. Unspecified (default)	31. Unspecified (default)	

1. ANSI Standard cause only
2. ANSI Standard cause only

Table A-2 Cisco BTS 10200 Softswitch Defined Cause Codes—Network Resource Unavailable

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Network resource unavailable			
31.	34. No Circuit available	34. No circuit/channel available	
32.		38. Network out of order	Can not reach MGW, network interface down
33.	41. Temporary failure	41. Temporary failure	400. Transaction not completed due to transient error
34.	42. Switching congestion	42. Switching equipment congestion	N/A
35. [Note 6]	43. Access information discarded[1]	43. Access information discarded	

Table A-2 Cisco BTS 10200 Softswitch Defined Cause Codes—Network Resource Unavailable

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
36.	Circuit Reservation Rejected (CMJ)	44. Requested circuit/channel not available	501.Endpoint not ready
37.	47. Resource unavailable (default)	47. Resource unavailable, unspecified	502. Endpoint does not have sufficient resource

1. ANSI standard cause value

Table A-3 Cisco BTS 10200 Softswitch Defined Cause Codes—Service Or Option Not Available

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Service or option not available			
41.	57. Bearer capability not authorized	57. Bearer capability not authorized	N/A
42.	58. Bearer capability not presently available	58. Bearer capability not presently available	N/A
43.	63. Service/Option not available (default)	63. Service or option not available, unspecified	N/A

Table A-4 Cisco BTS 10200 Softswitch Defined Cause Codes—Service/Option Not Implemented

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Service/option not implemented			
51.	65. Bearer capability not implemented	65. Bearer capability not implemented	N/A
52.		66. Channel type not implemented	N/A
53.		69. Requested facility not implemented	N/A
54.		70. Only restricted digital information bearer capability is available	N/A
55.	79. Service/Option not implemented (default)	79. Service or option not implemented, unspecified	N/A

Table A-6 Cisco BTS 10200 Softswitch Defined Cause Codes—Protocol Error

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Protocol error (local significance only)			
81.	97. Message type nonexistent or not implemented	98. Message not compatible with call state or message type nonexistent or not implemented or 41 Temporary failure	

Table A-6 Cisco BTS 10200 Softswitch Defined Cause Codes—Protocol Error (continued)

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
82.	99. Parameter not existent or implemented—discarded	99. Information element nonexistent or not implemented or 41 Temporary failure	
83.		100. Invalid information element contents	
84.	101. Parameter not existent or implemented—passed on	101. Message not compatible with call state or 41 Temporary failure	
85.	102. Recovery on timer expiry	102. Recovery on time expiry	
86.	111. Protocol error (default)	111. Protocol error, unspecified or 41 Temporary failure	510. Protocol Error was detected

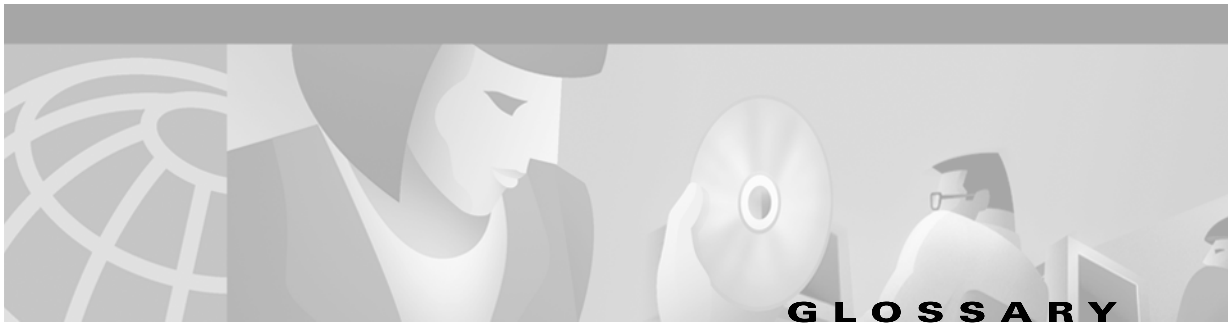
Table A-7 Cisco BTS 10200 Softswitch Defined Cause Codes—Interworking

Generic Cause Code	SS7 A.2.4.3/GR-905-CORE	ISDN TABLE I.2/Q.931	MGCP
Interworking			
91.	127. interworking—unspecified (default)	127. Interworking, unspecified	

The numbered notes are listed below:

1. If, following the receipt of a SETUP message or during overlap sending, the network determines that the call information received from the user is invalid, (for example, an invalid number), then the network follows the procedures for a cause such as one of the following:
   #1—Unassigned (unallocated) number
   #3—No route to destination
   #22—Number changed
   #28—Invalid number format (incomplete number)
2. The SETUP message can specify the channel identification information element (IE) for B-channel selection preference indicating preferred channel and alternative channel. If there is no B-channel available for the given criteria, this cause code is sent.
3. In the case of call offering, where multiple terminating exchanges are served with SETUP messages and they in turn respond back with CONNECT messages, the network sends a RELEASE message to the selected users with the cause code call awarded and being delivered in an established channel (#7), and with cause code nonselected user clearing (#26).
4. Upon receipt of a STATUS ENQUIRY message, the receiver responds with a STATUS message, reporting the current call state (the current state of an active call or a call in progress, or the null state if the call reference does not relate to an active call or to a call in progress) and cause #30, as a response to the STATUS ENQUIRY.

5. The terminating exchange can interpret an unspecified cause code as normal clearing (#16), and the originating exchange during the call setup procedure can interpret it as call rejected (#21).
6. IEs with a length exceeding the maximum length are treated as an IE with content error. But for access IEs (for example, a user to user information element or a called-party subaddress IE), cause #43, access information discarded, is used instead of cause No. 100, invalid IE contents.



GLOSSARY

A

AAA	Authentication, Authorization, and Accounting
AC	Automatic callback
AC_ACT	Automatic callback activation
AC_DEACT	Automatic callback deactivation
ACR	Anonymous call rejection
ACRA	Anonymous call rejection activation
ACRD	Anonymous call rejection deactivation
ADSL	Asymmetric digital subscriber line
AGW	Access gateway
AIN	Advanced Intelligent Network
AIOD	Automatic identified outward dialing
ALI	Automatic location identification
AMA	Automated message accounting
ANC	Announcements module
ANI	Automatic number identification
ANS	Announcement server
ANSI	American National Standards Institute
API	Application program interface
AR	Automatic recall
AR_ACT	Automatic recall activation
AR_DEACT	Automatic recall deactivation
AT	Access Tandem
ATA	Address translation adapter

ATIS	Alliance for Telecommunications Industry Solutions
ATM	Asynchronous transfer mode

B

BAF	Bellcore AMA format
BBG	Basic business group
BCM	Basic call module
BDMS	Bulk data management system
BEM	Billing event message
BGDP	Basic group dialing plan
BGL	Business group line
BLA	Billing adapter
BLV	Busy line verification
BP	Block pair
BRIDS	Bellcore rating input database system
BS	Billing server
BTA	Basic trading area

C

CA	Call agent
CAC	Carrier access code
CALEA	Communications Assistance for Law Enforcement Act
CAMA	Centralized automatic message accounting
CAS	Channel-associated signaling
CAT	Customer access treatment
CBR	Constant bit rate
CCS	Common-channel signaling
CCW	Cancel call waiting

CDB	Call data block
CDP	Customized dial plan
CDR	Call detail record
CE	Computing element
CFB	Call forwarding on busy
CFB-ACT	Call forwarding on busy activation
CFB-DEACT	Call forwarding on busy deactivation
CFB-VA	Call forwarding on busy variable activation
CFB-VD	Call forwarding on busy variable deactivation
CFM	CLASS feature module
CFNA	Call forwarding on no answer
CFNA-ACT	Call forwarding on no answer activation
CFNA-DEACT	Call forwarding on no answer deactivation
CFU	Call forwarding unconditional
CFU-ACT	Call forwarding unconditional activation
CFU-DEACT	Call forwarding unconditional deactivation
CHD	Call hold
CIC	Circuit identification code, carrier identification code
CID	Calling identity delivery, also Caller ID (see also CND)
CIDB	Calling identity delivery blocking
CIDCW	Calling identity delivery on call waiting
CIDS	Calling identity delivery and suppression (per call)—delivery part
CIDSS	Calling identity delivery and suppression (per call)—suppression part
CIP	Carrier identification parameter
CLASS	Custom local area signaling services
CLC	Carrier liaison committee
CLEC	Competitive local exchange carrier
CLEI	Common language equipment identifier
CLI	Command line interface

CLLI	Common language location identifier
CMIP	Common management information protocol
CMS	Call management system
CMTS	Cable modem termination system
CNAB	Calling name delivery blocking
CNAM	Calling name delivery
CND	Calling number delivery, Calling number display
CNDB	Calling number delivery blocking
CNM	Connection module, Customer network management
CO	Central office
COCUS	Central office code utilization survey
CODEC	Coder/decoder, Compression/decompression
COPS	Common open policy service protocol
CORBA	Common object request broker architecture
COS	Class of service (screening)
COT	Customer-originated trace, Continuity testing, Central office termination
CPCN	Certificate of public convenience and necessity
CPE	Customer premises equipment
CPRK	Call park
CPRK_RET	Call park retrieve
CPSG	Call park subscriber group
CPU	Call pickup
	Central processing unit
CS	Capability set (for example, CS-2)
CSA	Callpath services architecture
CSN	Circuit switched network
CSR	Carrier sensitive routing
CT	Call transfer, Call type
CW	Call waiting

CWI	Call waiting indication

D

DA	Directory assistance, Distinctive alerting
DACWI	Distinctive alerting/call waiting indication
DCPU	Directed call pickup (See also DCN and DCU)
DF	Delivery function (CALEA)
DID	Direct inward dialing
DLEC	Data local exchange carrier
DN	Directory number
DND	Do not disturb
DNS	Domain name server
DNIS	Dialed number identification service
DOD	Direct outward dialing
DOW	Day of week
DOY	Day of year
DP	Dial plan, Dial pulse, Demarcation point
DPN	Directed call pickup without barge-in
DPN_O	Call pickup without barge-in (originate)
DPN_T	Call pickup without barge-in (terminate)
DPU	Directed call pick-up with barge-in
DPU_O	Call pickup with barge-in (originate)
DPU_T	Call pickup with barge-in (terminate)
DRCW	Distinctive ringing/call waiting
DPC	Destination point code
DQoS	Distributed quality of service
DSL	Digital subscriber line
DSP	Digital signal processing

DSX	Digital system cross-connect frame
DTMF	Dual tone multifrequency

E

E-1	European equivalent of T1
E-911	Enhanced 911
E & M	"Ear and Mouth" switch-to-switch signaling on PSTN
EA	Equal access
EC	Echo cancellation
ECSA	Exchange Carriers Standards Association
EDP	Event detection point
EMS	Element management system, Event Message Specification (Packet Cable)
ERC	Easily recognizable codes
ERQNT	Embedded request for notification
ESB	Emergency service bureau
ESL	Emergency service line
ETSI	European Telecommunications Standards Institute

F

FCAPS	Fault, configuration, accounting, performance, security
FCP	Feature control protocol
FGB	Feature group B
FGD	Feature group D
FIM	Feature interaction manager
FS	Feature server
FSAIN	Feature server for Advanced Intelligent Network services
FSPTC	Feature server for POTS, Tandem, and Centrex services
FTP	File transfer protocol

FXO Foreign exchange office

FXS Foreign exchange station

G

GAP Generic address parameter

GSM Global system for mobile communications

GUI Graphical user interface

H

HFC Hybrid fiber coax

HLR Home location register

HNPA Home numbering plan area

HTML Hypertext markup language

HTTP Hypertext transfer protocol

I

IAD Integrated access device

IANA Internet Assigned Numbers Authority

ICAP Inter-call agent protocol

ICMP Internet Control Message Protocol

IDDD International direct distance dialing

IE Information element

IETF Internet Engineering Task Force

ILEC Incumbent local exchange carrier

IMT Intermachine trunk

IN Intelligent network

INC Industry Numbering Committee

IP Internet protocol

IPM	Impulses per minute
IRDP	ICMP Router Discovery Protocol
ISA	ISDN adapter
ISDN	Integrated Services Digital Network
ISFG	Incoming simulated facility group
ISO	International Standards Organization
ISP	Internet service provider; independent service provider (for VMWI feature)
ISS	ISDN stack
ISUP	ISDN user part
ITU	International Telecommunications Union
IVR	Interactive voice response
IXC	Interexchange carrier

J

JCA	Java cryptography architecture, Java console agent
JCM	Java console module
JDBC	Java dataBase connectivity
JMS	Java message service

K

KAM	Keepalive module
Kbps	Kilobits per second

L

LAN	Local area network
LATA	Local access and transport area
LCR	Least cost routing
LDAP	Lightweight directory access protocol

LEC	Local exchange carrier
LERG	Local exchange routing guide
LIDB	Line information database
LNP	Local number portability
LPC	Local point code
LRN	Local routing number
LRQ	Location request (H.323 signaling)
LRU	Least recently used
LSA	Local serving area
LSSGR	LATA Switching Systems Generic Requirements

M

Mbps	Megabits per second
MCF	Multiple call forwarding
MCS	Media gateway control stack
MDC	Mid-call
MDN	Multiple directory numbers
MF	Multifrequency
MG (MGW)	Media gateway
MGA	Media gateway adapter
MGC	Media gateway controller
MGCP	Media gateway control protocol
MGW (MG)	Media gateway
MIB	Management information base
MIME	Multipurpose Internet mail extensions
MLHG	Multiline hunt group
MNM	Maintenance module
ms	millisecond

MSA	Metropolitan statistical area
MSU	Message signal units
MTA	Metropolitan trading area
MTP	Message transport part
MTU	Maximum transmission unit
MWI	Message waiting indicator

N

NANP	North American Numbering Plan
NAS	Network access server
NCS	Network-based call signaling
NE	Network element
NEBS	Network Equipment Building Standards
NFAS	Nonfacility-associated signaling
NIS	Network information service
NMS	Network management system
NO	Network operator
NOC	Network operations center
NPA	Numbering Plan Area
NSE	Name signaling event
NTP	Network time protocol
NU	Network unit
nxx	NANP digits: n=2, 3, …9 and x=0, 1, …9

O

OAM	Operations, administration and maintenance, Operations administration module
OAM&P	Operations, administration, maintenance and provisioning
OCN	Operating company number

OBCSM	Originating basic call state machine
OI	Operator interrupt
OLI	Originating line information
OMS	OptiCall™ Messaging System
OPC	Originating Point Code
OPT	Open Packet Telephony
OS	Operating system
OSA	Open service adapter
OSFG	Outgoing simulated facility group
OSI	Open Systems Interconnection
OSS	Operations support system
OSSGR	Operator Services Systems Generic Requirements

P

PBX	Private branch exchange
PCM	Pulse code modulation
PCMA	Pulse code modulation A law
PCMU	Pulse code modulation mu law
PCS	Personal communications services
PCSNDB	Personal communications services numbering database
PDU	Power distribution unit
PIC	Presubscribed interexchange carrier, Point in call
PLT	Platform
POI	Point of interface, Point of interconnection
POP	Point of presence
POPD	Public office dialing plan
POSIX	Portable operating system interface UNIX
POTS	Plain old telephone service

PPP	Point to point protocol
PPQ	Point to point queuing
PPS	Permanent presentation status
PRI	Primary rate interface
PS	Presentation status
PSAP	Public safety answering point
PSTN	Public switched telephone network
PVC	Permanent virtual circuit

Q

QoS	Quality of service

R

RACF	Remote activation of call forwarding
RACF-PIN	Remote activation of call forwarding personal ID number
RADIUS	Remote authentication dial-in user service
RAID	Redundant array of inexpensive disks
RAS	Remote access server
	Registration, Admissions, and Status (signaling function in H.323 for communications to gatekeeper),
RCF	Remote call forwarding
RDBS	Routing database system
RDM	Redundancy module
RDT	Recall dial tone
RFC	Request for comment (IETF)
RGW	Residential gateway
RIP	Routing information protocol
ROH	Receiver off hook
RPC	Remote point code, Remote procedure call
RQNT	Request for notification

RR	Resource record
RSA	Rural service area
RSIP	Restart in progress
RSM	Resource module
RSVP	Resource reservation protocol
RTM	Routing module
RTP	Realtime transport protocol
R-UDP	Reliable User Datagram Protocol (Cisco Systems proprietary signaling backhaul protocol)

S

S7A	SS7 adapter
S7S	SS7 stack (DGM&S)
SAC	Service access code
SAI	Signaling adapter interface
SC1D	Speed call 1-digit
SC2D	Speed call 2-digit
SCA	Selective call acceptance
SCF	Selective call forwarding
SCP	Service control point, Signal control point
SCR	Selective call rejection
SDK	Software development kit
SDP	Session description protocol
SFG	Simulated facility group
SGCP	Simple gateway control protocol
SIA	SIP adapter
SID	System identification number
SIM	Service interaction manager
SIP	Session initiation protocol

SLE	Screening list editing
SMA	SNMP adapter
SMDS	Switched multimegabit data service
SMS	Service management system
SNMP	Signaling network management protocol
SOHO	Small office home office
SP	Service Provider
SPCS	Stored program control system
SQL	Structured Query Language
SS7	Signaling System 7
SSF	Service switching function
SSP	Service switching point, Signal switching point
STP	Signal transfer point
SVC	Switched virtual circuit

T

T1	Trunk level 1
T3	Trunk level 3
TAP	Telocator alphanumeric paging protocol
TBCSM	Terminating basic call state machine
TCAP	Transaction capabilities application part
TCP	Transmission control protocol
TCP/IP	Transmission control protocol/Internet protocol
TDD	Telecommunications device for the deaf
TDM	Time division multiplexing
TDP	Trigger detection point
TF	Toll free
TG	Trunk group

TGW	Trunking gateway
TMN	Telecommunications management network
TNS	Transit network selection
TOD	Time of day
TOPS	Traffic operator position system
TOS	Type of service
TPM	Terminating point master
TRS	Telecommunications relay services
TSAP	Transport service access point
TTY	Text typewriter
TWC	Three way calling

U

UAA	User authentication adapter
UAC	User agent client
UAS	User agent server
UBR	Universal broadband router (Cisco)
UCD	Uniform call distribution
UDP	User datagram protocol
URI	Uniform resource identifier
URL	Universal resource locator
USTWC	Usage-sensitive three way calling

V

VBR	Variable bit rate
VLAN	Virtual local area network
VMWI	Visual message waiting indicator
VoATM	Voice over ATM

VoIP Voice over IP

VSC Vertical service code

W

WAN Wide area network

WFI Waiting for instruction

X

***x*DSL** (generic) digital subscriber line

Y

Z