MPS-PBX Configuration and Interconnectivity Manual

3.5
NN44100-504, 02.02
July 2009

Downloading Documentation

For the most current versions of Documentation, see the Avaya Support Web site: http://support.avaya.com.

Contact Avaya Support

Avaya provides a telephone number for you to use to report problems or to ask questions about your Product. The support telephone number is 1-800-242-2121 in the United States. For additional support telephone numbers, see the Avaya Web site: http://support.avaya.com.

Contents

Chapter 1: Preface

Scope

The MPS-PBX Configuration and Interconnectivity Manual describes product, configuration, and operations procedures that facilitate interconnectivity between the Avaya Media Processing Server (MPS) and the Meridian 1 (M1) or Communication Server 1000 (CS 1000) Private Branch Exchange (PBX). The manual explains hardware and software requirements, and crossover protocols between the MPS and the Avaya PBX.

Important:

All references to MPS 3.0 in this document (such as document titles, software versions, and illustrations) apply to all releases of MPS 3.X.

Note:

This document refers to the Avaya (MPS 500 and MPS 1000) systems as Media Processing Server (MPS) systems or Interactive Voice Response (IVR) systems.

Intended Audience

This manual is intended for MPS system users who purchase, configure, install, program and operate the system. This manual serves the needs of the user who:

Example

- has completed an installation maintenance training program.
- is familiar with site-specific operating procedures related to the MPS interface, as well as knowledge of application functions and other equipment to which the PS interface must be connected.
- has basic knowledge of the Solaris or the Windows operating systems.
- is familiar with the functions of the Avaya PBX.

In addition, the reader should be familiar with telecommunications and computer equipment, their functions, and associated terminology. The reader must also know the characteristics of the specific installation, including on-site power systems, computers, peripherals, and telephony components.

Customer service

Visit the Avaya Web site to access the complete range of services and support that Avaya provides. Go to www.avaya.com or go to one of the pages listed in the following sections.

Navigation

Getting technical documentation

To download and print selected technical publications and release notes directly from the Internet, go to www.avaya.com/support.

Getting product training

Ongoing product training is available. For more information or to register, you can access the Web site at www.avaya.com/support. From this Web site, you can locate the Training contacts link on the left-hand navigation pane.

Getting help from a distributor or reseller

If you purchased a service contract for your Avaya product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller for assistance.

Getting technical support from the Avaya Web site

The easiest and most effective way to get technical support for Avaya products is from the Avaya Technical Support Web site at www.avaya.com/support.

How to Use This Manual

This manual uses many standard terms relating to computer systems, software application functions, and the Internet. However, it contains some terminology that can be explained only in the context of the MPS Series. Refer to the Glossary of Avaya's Media Processing Server Series Terminology for definitions of MPS Series specific terms.

Read this manual from start to finish at least once. When you are familiar with the document, you can use the Table of Contents to locate topics of interest for reference and review.

If you are reading this document online, use the cross-reference links (shown in blue) to quickly locate related topics. Position your cursor over the cross-reference link and click once. Click any point in a Table of Contents entry to move to that topic. Click the page number of any Index entry to access that topic page.

Familiarize yourself with various specialized textual references within the manual, see Conventions Used in This Manual on page 10.

Note:

Periphonics is now part of Avaya. The name Periphonics, and variations thereof, appear in this manual only in reference to a product (for example, the PERImps package, the perirev command, and so on).

Organization of This Manual

The following briefly outlines the structure of this manual:

Example

- Chapter 1—Introduction Provides an overview of the MPS, the Avaya PBX, the package requirements, and the interconnectivity functions.
- Chapter 2—Installation Describes installation procedures for the Media Processing Server and the Avaya PBX. Outlines hardware and software requirements. Documents

cabling requirements between the two systems with details of pin structures and other connectors. Explores licensing requirements.

- Chapter 3—Configuration for CTI Functionality Documents the configuration processes for the MPS and the Avaya PBX for CTI functionality.
- Chapter 4—ISDN Configuration Documents the configuration processes for the MPS and the Avaya PBX for ISDN protocols.
- Chapter 5—SIP Configuration Documents the configuration processes for the MPS and the Avaya PBX for SIP protocols.
- Chapter 6—Protocols Provides information about protocols common to the MPS and the Avaya PBX.
- Chapter 7—Status Monitoring Describes monitoring status of the phone line and the Avaya PBX using MPS Manager.

Conventions Used in This Manual

This manual uses different fonts and symbols to differentiate between document elements and types of information. These conventions are summarized in the following table.

Table 1: Conventions Used in This Manual

Notation	Description
Normal text	Normal text font is used for most of the document.
important term	The Italics font introduces new terms, highlights meaningful words or phrases, or distinguishes specific terms from nearby text.
`system command`	This font indicates a system command or its arguments. Enter such keywords exactly as shown (that is, do not fill in your own values).
command, condition and alarm	Command, Condition and Alarm references appear on the screen in magenta text and reference the Command Reference Manual, the MPS Developer User's Guide, or the Alarm Reference Manual, respectively. Refer to these documents for detailed information about Commands, Conditions, and Alarms.
`file name / directory`	This font highlights the names of disk directories, files, and extensions for file names. It also shows what is displayed on a text-based screen (for example, to show the contents of a file).
on-screen field	This font indicates field labels, on-screen menu buttons, and action buttons.
<KEY NAME>	A term that appears within angled brackets denotes a terminal keyboard key, a telephone keypad button, or a system mouse button.
Book Reference	This font indicates the names of other publications referenced within the document.

Notation	Description
cross-reference	A cross-reference appears on the screen in blue. Click the cross-reference to access the referenced location. A cross-reference that refers to a section name accesses the first page of that section.
Note:	The italicized Note: identifies notes, important facts, and other keys to understanding.
	The Caution icon identifies procedures or events that require special attention. The icon indicates a warning that serious problems may arise if the stated instructions are not followed implicitly.
	The Flying Windows icon identifies procedures or events that apply to the Windows operating system only.[1]
	The Solaris icon identifies procedures or events that apply to the Solaris operating system only.[2]

1. Windows and the flying Window logo are either trademarks or registered trademarks of Microsoft Corporation.
2. Solaris® is a registered trademark of The Open Group in the U.S. and other countries.

Solaris and Windows Conventions

This manual depicts examples (command line syntax, configuration files, and screen shots) in Solaris format. Windows-specific commands, procedures, or screen shots are shown when required. The following table lists general operating system conventions used with either the Solaris or Windows operating system.

	Solaris	Windows
Environment	`$PPROHOME`	`%PPROHOME%`
Paths	`$PPROHOME/bin`	`%PPROHOME%\bin`
Command	`<command> &`	`start /b <command>`

Two-Button (Windows) vs. Three-Button (Solaris) Mouse

<SELECT>	Left button
<ADJUST>	Left and right together
<MENU>	Right button

<SELECT>	Left button
<ADJUST>	Middle button
<MENU>	Right button

Trademark Conventions

The following trademark information applies throughout for third party products discussed within this manual. Trademarking information is not repeated hereafter.

Solaris® and Motif® are registered trademarks of The Open Group in the U.S. and other countries.

Solaris, SunOS, OpenWindows, SPARC, and UltraSPARC are trademarks or registered trademarks of Sun Microsystems, Inc. in the United States and other countries.

Microsoft, MSSQL, Windows, Internet Explorer, and the Flying Windows logo are either trademarks or registered trademarks of Microsoft Corporation.

Oracle® is a registered trademark of Oracle Corporation.

Sybase™ and SYBASE™ are trademarks of Sybase, Inc. or its subsidiaries.

Informix® and INFORMIX® are registered trademarks of Informix Corporation or its affiliates.

Product Nomenclature Changes

The following product names changed with the latest Avaya MPS software release. All other references to the former name with respect to environment variables, directory paths, software package names, and so on remain the same. For example, the PeriProducer product is now referred to as the Media Processing Server Developer; however, its package name remains PERIppro.

Former Product Name	New Product Name
IVR Software	Media Processing Server Release x.x
PeriProducer	Media Processing Server Developer
PeriView	Media Processing Server Manager
PeriStudio	Media Processing Server Studio
PeriReporter	Media Processing Server Reporter
PeriSQL	Media Processing Server RDB
PeriVXML	Media Processing Server VXML Browser
CTI Suite	Communications Control Toolkit (CCT)
Open Signal Computing and Analysis Resource (OSCAR)	Speech Server

Japanese Denan Regulatory Compliance

Warning

Please be aware of the following while installing the equipment:

- Please use the connecting cables, power cord, and AC adaptors shipped with the equipment or specified by Avaya to be used with the equipment. If you use any other equipment, it may cause failures, malfunctioning or fire.
- Power cords shipped with this equipment must not be used with any other equipment. If the above guidelines are not followed, it may lead to death or severe injury.

警告

本製品を安全にご使用頂くため、以下のことにご注意ください。

- 接続ケーブル、電源コード、AC アダプタなどの部品は、必ず製品に同梱されております　添付品または指定品をご使用ください。添付品・指定品以外の部品をご使用になると故障や動作不良、火災の原因となることがあります。
- 同梱されております付属の電源コードを他の機器には使用しないでください。

上記注意事項を守らないと、死亡や大怪我など人身事故の原因となることがあります。

VCCI Class A Statement

Class A ITE

この装置は、情報処理装置等電波障害自主規制協議会 (VCCI) の規定に基づくクラスA 装置です。この装 置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を取るように要求されることがあります。

Translation: This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective action.

Chapter 2: Introduction

This chapter covers:

1. MPS Overview
2. Avaya M1 PBX Overview
3. Avaya CS 1000 PBX Overview
4. MPS-Avaya PBX Interface Diagram

MPS Overview

The Avaya Media Processing Server (MPS) Series product line consists of hardware and software for performing Interactive Voice Response (IVR) and media processing functions in call processing environments. The MPS specifically integrates call processing components with speech, telephony, data communications, and transaction processing functions.

The MPS 500 consists of one Telephony Media Server (TMS) that supports eight T1/E1 spans (up to 240 ports) or VoIP channels of comparable capacity, on Solaris platforms.

The MPS 1000 consists of one Telephony Media Server (TMS) that supports up to 64 T1/E1 spans (up to 1,536/1,920 ports). Multiple cabinets can be networked together to increase capacity to 384 T1/E1 spans (up to 9,216/11,520 ports) in a Managed Cluster. Multiple MPS 1000 Managed Clusters can be networked together for additional capacity.

The MPS enables writing self-service applications using the following options:

Example

- MPS Developer—graphical development tool
- VoiceXML—text-based markup language

Avaya M1 PBX Overview

The Avaya Meridian PBX portfolio consists of voice features, data connectivity, LAN communications, Computer Telephony Integration (CTI), and information services for communication applications for 20 to 16,000 users.

The Meridian suite has the following products:

Example

- IP Trunk and IP Line —These products allow you to add Communication Server IP Telephony applications to Meridian communication systems, or additional IP Telephony capacity to a Communication Server 1000 system.
- Integrated Conference Bridge (MICB) —A plug-and-play conference bridge card for the M1.
- Attendant Console M2250 —A fully digital console that provides high speed call processing.
- Meridian Automatic Call distribution —Automatic Call Distribution (ACD) that enables call distribution based on arrival.
- Attendant PC —CTI interface that brings the call processing functions of the M2250 Attendant Console to the PC.

In addition, the Meridian Portfolio has Avaya M1 PBXs in the following configurations:

Example

- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet
- Meridian 1 PBX 51C
- Meridian 1 PBX 61C
- Meridian 1 PBX 81
- Meridian 1 PBX 81C

Note:

When upgrading software, memory upgrades may be required on the Signaling Server, the Call Server, or both.

Avaya CS 1000 PBX Overview

Avaya Communication Server 1000 (CS 1000) is a full-featured IP-distributed communications system that delivers the benefits of network convergence and collaborative communications to today's increasingly virtual enterprise environments. The CS 1000 is a highly scalable communications platform with built-in reliability and survivability that can be distributed across IP LAN and WAN infrastructures. Core system components include three primary elements:

- CS 1000 Call Servers provide reliable call and connection management service. The Call Servers control the system software and are capable of supporting up to 15 000 IP clients per server, as well as supporting geographically redundant configurations to ensure business continuity.

- Signaling Servers perform important call control services such as registration of IP terminals, IP address translation and bandwidth control. They streamline the network dialing plan and simplify the scalability and management of CS 1000 networks.
- Enterprise Media Gateways support a complete range of analog and digital line and trunk interfaces across LAN or WAN infrastructures.

Example

In addition, the Communication Server 1000 Portfolio has Avaya CS 1000 PBXs in the following configurations:

- Communication Server 1000S (CS 1000S)
- Communication Server 1000M Chassis (CS 1000M Chassis)
- Communication Server 1000M Cabinet (CS 1000M Cabinet)
- Communication Server 1000M Half Group (CS 1000M HG)
- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Communication Server 1000E (CS 1000E)

System Migration from M1 to CS 1000

When particular Meridian 1 systems are upgraded to run CS 1000 Release 4.5 software and configured to include a Signaling Server, they become CS 1000M systems. The following table lists each Meridian 1 system that supports an upgrade path to a CS 1000M system.

This Meridian System...	Maps to this CS 1000M System
Meridian 1 PBX 11C Chassis	CS 1000M Chassis
Meridian 1 PBX 11C Cabinet	CS 1000M Cabinet
Meridian 1 PBX 51C	CS 1000M Half Group
Meridian 1 PBX 61C	CS 1000M Single Group
Meridian 1 PBX 81	CS 1000M Multi Group

Meridian 1 PBX 81C	CS 1000M Multi Group

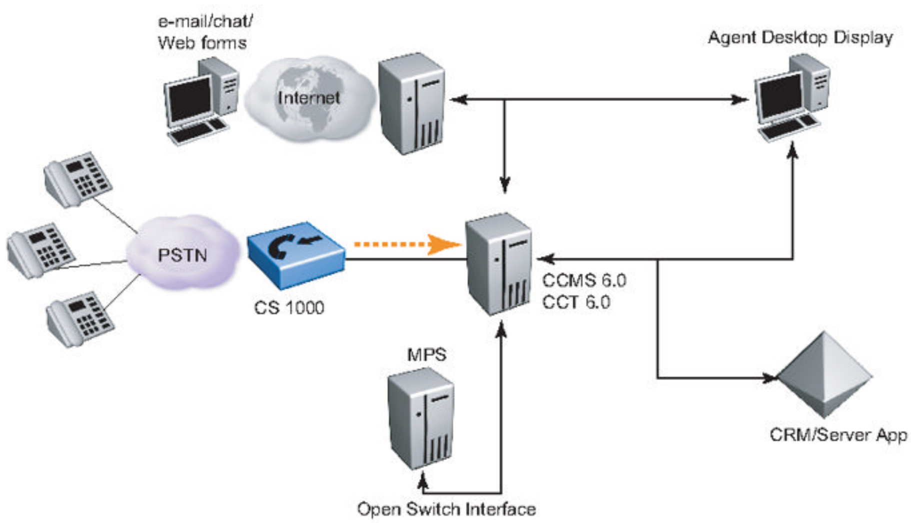
Example

For more information, see one or more of the following NTPs:

- Communication Server 1000M and Meridian 1: Small System Upgrade Procedures (553-3011-258)
- Communication Server 1000M and Meridian 1: Large System Upgrade Procedures (553-3021-258)
- Communication Server 1000S: Upgrade Procedures (553-3031-258)
- Communication Server 1000E: Upgrade Procedures (553-3041-258)

MPS-Avaya PBX Interface Diagram

The following diagram shows the network architecture, which in this example, includes the MPS and the Avaya CS 1000 PBX:



Chapter 3: Installation

This chapter covers:

1. Introduction
2. Installing the MPS
3. Installing the Avaya PBXs
4. The PERIplic License Server
5. Cabling Requirements
6. MPS-M1/CS 1000 Connection Methods
6. Lineside T1 Connection
7. Lineside E1 Connection

Introduction

The first step to interconnect the MPS 500 or MPS 1000 and the Avaya PBX is to install each of these entities individually.

The Avaya M1 PBX and CS 1000 Performance Enhancement Packages (PEPs) are available at the Avaya Enterprise Solutions PEP Library (ESPL). Go to https://app91.avaya.com/espl/ to access the PEP library. A login window appears. Enter the user name and password obtained during registration. Select options as appropriate.

Installing the MPS 500 and the MPS 1000

The MPS systems provide a wide range of programming and call handling capabilities and enable compatibility with existing network architectures and fresh e-commerce applications. This encourages wide scalability to meet increasing and varied call demands.

For information on installing and setting up the MPS 500, refer to the Media Processing Server 500 Hardware Installation and Maintenance Manual.

For information on installing and setting up the MPS 1000, refer to the Media Processing Server 1000 Hardware Installation and Maintenance Manual.

Installing the Avaya M1 and CS 1000 PBX

The Avaya M1 and CS 1000 systems are referred to as either Small Systems or Large Systems depending on the model of the PBX. The following two sections define small systems and large systems and reference the installation documentation for each system.

Installing the Small System

The following systems are referred to generically as Small Systems:

Example

- Communication Server 1000M Chassis (CS 1000M Chassis)
- Communication Server 1000M Cabinet (CS 1000M Cabinet)
- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet

For details on the installation process, refer to the following documents:

Example

- Communication Server 1000M and Meridian 1: Small System Planning and Engineering
- Communication Server 1000M and Meridian 1: Small System Installation and Configuration)

Installing the Large System

The following systems are referred to generically as "Large Systems":

Example

- Communication Server 1000M Half Group (CS 1000M HG)
- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Meridian 1 PBX 51C
- Meridian 1 PBX 61C

- Meridian 1 PBX 81
- Meridian 1 PBX 81C

Refer to the following manuals for information about the installation of the large system:

Example

- Communication Server 1000M and Meridian 1: Large System Overview
- Communication Server 1000M and Meridian 1: Large System Planning and Engineering
- Communication Server 1000M and Meridian 1: Large System Installation and Configuration

The PERIplic License Server

The PERIplic package has files and directories used for licensing relevant Avaya packages. Avaya licensing uses the client-server approach. A license server has a list of licenses that it can serve. A client (licensed software product) requests a license when it starts, and does not start without acquiring a license.

The server grants a license to the client if the requested release, the product number, and the identifying information the client supplies matches the information associated with the product license. A granted license remains in use until the client releases the license or the release interval of the license expires (each time the server receives a refresh message from the client holding a license, the interval is restarted).

For information on the PERIplic License Server, refer to Installing MPS Software on the Windows Platform Manual.

Note:

Individual licenses are needed for each of the MPS 500 or the MPS 1000 IVR ports that connect to the Avaya PBX.

Cabling Requirements

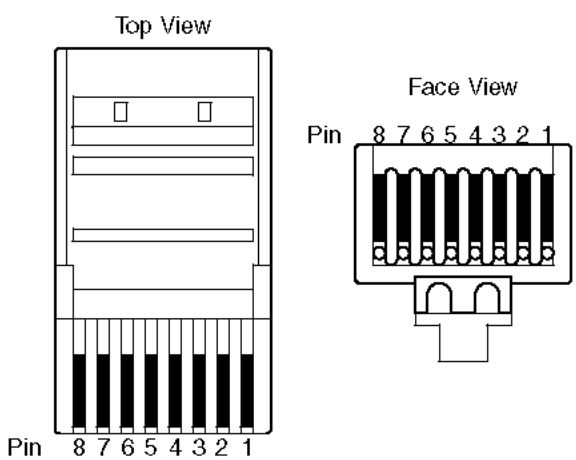
The MPS 500 or MPS 1000 and the Avaya PBX are connected using cables with different specifications.

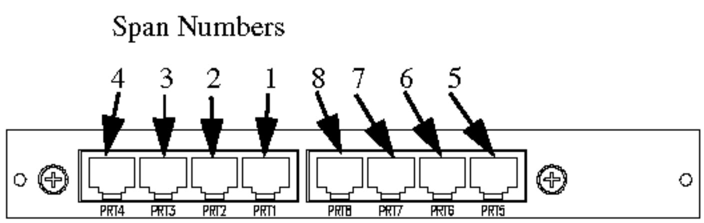
MPS 500 Connector Interfaces

The following information describes the MPS 500 Connector Interfaces. Information is provided for typical configurations so as to allow values to be determined for any specific configuration.

RJ-48C Digital Telephony Interface

Systems that are equipped with RJ48C connectors provide separate connectors for each PSTN span. The following illustration shows the RJ48C plug and pin-outs used to connect a single span (T1 or E1).

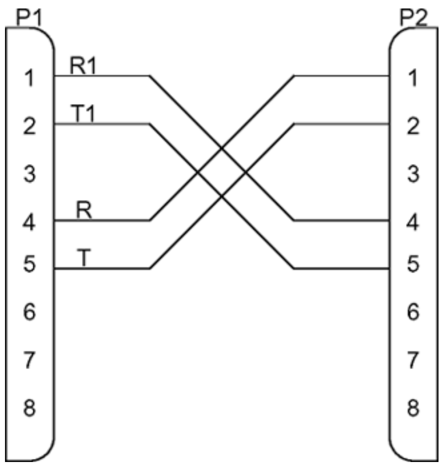




MPS 500 RJ-48C Rear Panel

RJ48C Crossover Cable

A special crossover cable can be fabricated for loopback or other offline tests where an outbound span is looped back directly to an inbound span. On a crossover cable, the transmit pins on one connector are connected to the receive pins on the other connector. Wire an RJ48C crossover cable as shown in the following illustration.

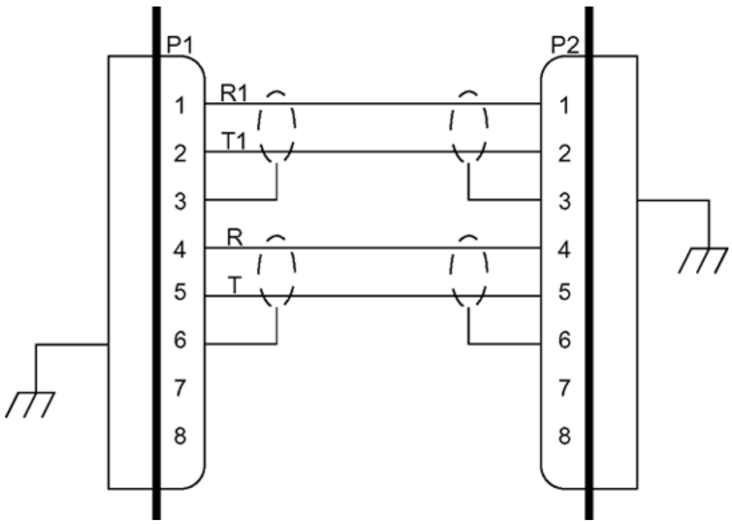


RJ48C Pin-outs

Pin	Signal
1	R1 (receive, ring)
2	T1 (receive, tip)
3	Not used
4	T (transmit, ring)
5	R (transmit, tip)
6	Not used
7	Not used
8	Not used

RJ48C Shielded Interface

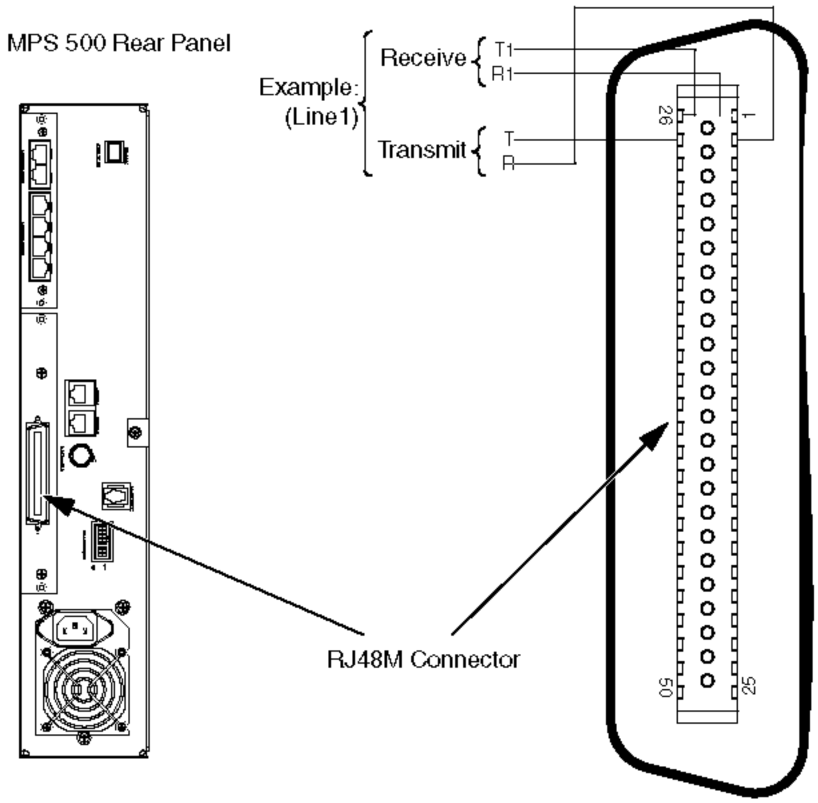
Some systems use RJ48C cables where the transmit and receive wire pairs are shielded. The shields for the transmit and receive wire pairs connect to connector pins 6 and 3, respectively. These shields connect to the chassis ground at the terminating equipment driving the signal. These connections are illustrated in the following interconnection diagram.



RJ48M

The TMS connects to the Public Switched Telephone Network (PSTN) through a USOC RJ48M 50-pin connector located on the front panel of each Digital Communications Controller (DCC). This interface provides capacity for up to eight T1 or E1 spans. Digital interface connections for span/line 1 are shown in the following illustration. Connections for all spans are provided in the following table.

Digital interface connections for span/line 1



RJ48M Direct Feed Pin-Outs

Line (Span)	Receive		Transmit	
	T1 (Tip)	R1 (Ring)	T (Tip)	R (Ring)
1	26	1	27	2
2	29	4	30	5
3	32	7	33	8
4	35	10	36	11
5	38	13	39	14
6	41	16	42	17
7	44	19	45	20
8	47	22	48	23

A standard direct feed cable for RJ48M connectors is connected at both ends in accordance with the preceding table. Each pin from both connectors is connected to the corresponding pin from the other.

The T1 interface is in accordance with ANSI-T1.403-1989. It can connect up to eight digital spans. Each span connection consists of up to four wires and pin connections — two for transmit and two for receive. Transmit channel connections are designated T (tip) and R (ring). The receive channel connections are identified as T1 and R1. For T1, each span handles 24 separate phone lines.

For E1, the connections and signal designations are exactly the same, except that each span handles 30 separate lines under the E1 transmission protocol. The DCC is dedicated either to a T1 or E1 interface, and the DCC installed in the system is dependant on the local PSTN standard.

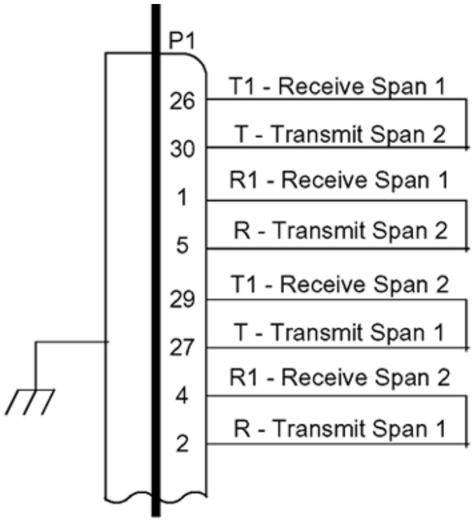
Note:

The terms tip and ring are telephone terms that originated when loop circuit connections were made manually with phone plugs (for example, the tip is positive and ring is the return).

RJ48M Loopback Connector

A special loopback connector is used for loopback and other offline tests. On the loopback connector (A0898687), the transmit pins of each span connect to the receive pins of another span. Span 1 is connected to span 2, span 3 to span 4, span 5 to span 6, and span 7 to span 8. The following table and illustration describe the connector configuration.

Line (Span)	Receive		Line (Span)	Transmit	
	T1 (Tip)	R1 (Ring)		T (Tip)	R (Ring)
1	26	1	2	30	5
2	29	4	1	27	2
3	32	7	4	36	11
4	35	10	3	33	8
5	38	13	6	42	17
6	41	16	5	39	14
7	44	19	8	48	23
8	47	22	7	45	20



Partial Loopback Connector Showing Span 1 connected to Span 2

The following information pertains to the cables and cards used to connect the MPS 500 to the Avaya M1/CS 1000 PBX. A Avaya cable kit includes all the parts required to connect to the MPS 500 IVR.

The adapter kit part number is A0394216, and contains the following parts.

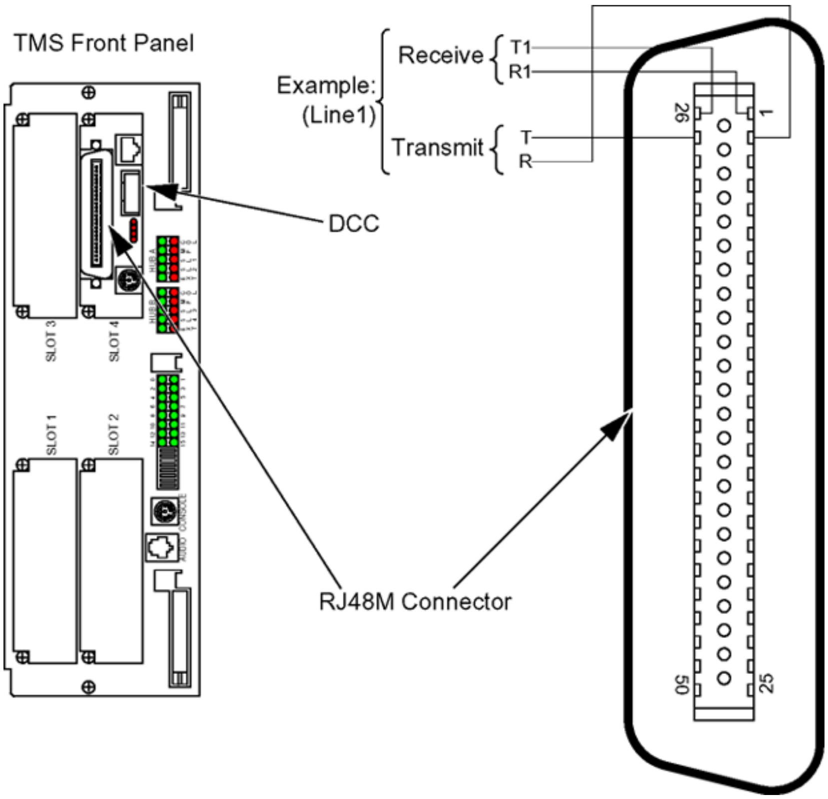
Adapter Kit Part No.	Cable	Adapter	Ethernet
A0394216	Male F50 Telco -> Male DB15 (NTBK04AA)	Female DB15 -> Female RJ48C	RJ48

MPS 1000 Connector Interfaces

The following information describes the MPS 1000 Connector Interfaces. Information is provided for typical configurations so as to allow values to be determined for any specific configuration.

RJ48M

The TMS connects to the public switched telephone network (PSTN) or the TelCo Connector Panel through a USOC RJ48M 50-pin connector located on the front panel of each DCC. This interface provides capacity for up to eight T1 or E1 spans. The digital interface connections for span/line 1 are shown in the following illustration. Connections for all spans are listed in the following table.



RJ48M Direct Feed Pin Outs

Line (Span)	Receive		Transmit	
	T1 (Tip)	R1 (Ring)	T (Tip)	R (Ring)
1	26	1	27	2
2	29	4	30	5
3	32	7	33	8
4	35	10	36	11
5	38	13	39	14
6	41	16	42	17
7	44	19	45	20
8	47	22	48	23

A standard direct-feed cable with RJ48M connectors is connected at both ends in accordance with the preceding table. Each pin of both connectors is connected to the corresponding pin of the other.

The T1 interface is in accordance with ANSI-T1.403-1989. The interface is capable of connecting up to eight digital spans. Each span connection comprises four wires/pin connections: two for transmit and two for receive. The transmit channel connections are designated T (tip) and R (ring). The receive channel connections are designated T1 and R1. For T1, each span handles 24 separate phone lines.

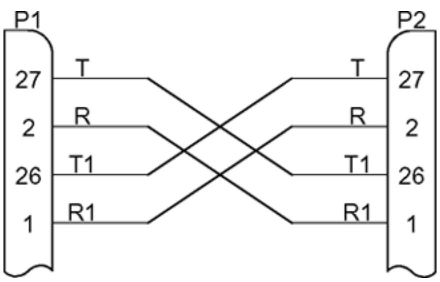
For E1, the connections and signal designations are exactly the same, except that each span handles 30 separate lines under the E1 transmission protocol. The DCC is dedicated to either a T1 or E1 interface, so the DCC installed in the system depends on the local PSTN standard.

Note:

The terms tip and ring are telephone terms that originated when loop circuit connections were manually made with phone plugs (that is, tip is positive and ring is the return).

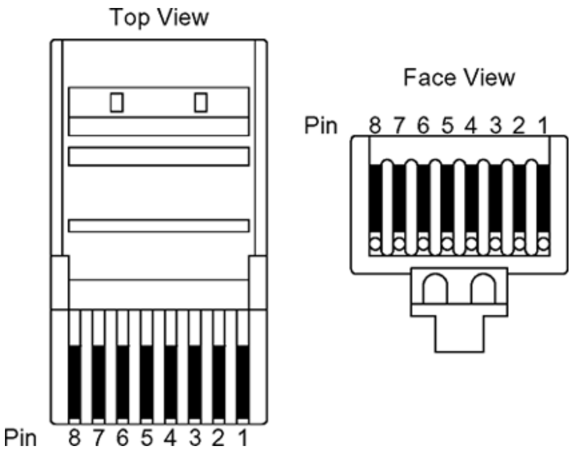
RJ48M Crossover Cable

For loop-back or other off-line tests where outbound spans are looped back directly to inbound spans, a special crossover cable can be fabricated. On a crossover cable, the transmit pins of each span on one connector are connected to the receive pins for that span on the other connector. For example, the connections for span 1, are made as shown in the following illustration. The same order applies to the remaining spans.



RJ48C

Systems that are equipped with a TCCP with RJ48C connectors provide separate connectors for each PSTN span. The following illustration and table show the RJ48C plug and pin-outs used to connect a single span (T1 or E1).



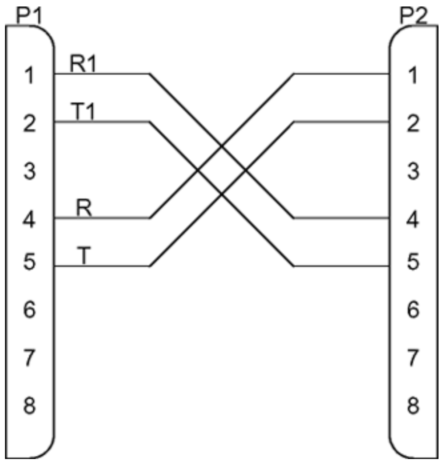
RJ48C Direct Feed Pin Outs

A standard direct feed cable with RJ48C connectors is connected at both ends in accordance with the following table. Each pin of either connector is connected to the corresponding pin of the other.

Pin	Signal
1	R1 (receive, ring)
2	T1 (receive, tip)
3	Not Used
4	T (transmit, ring)
5	R (transmit, tip)
6	Not Used
7	Not Used
8	Not Used

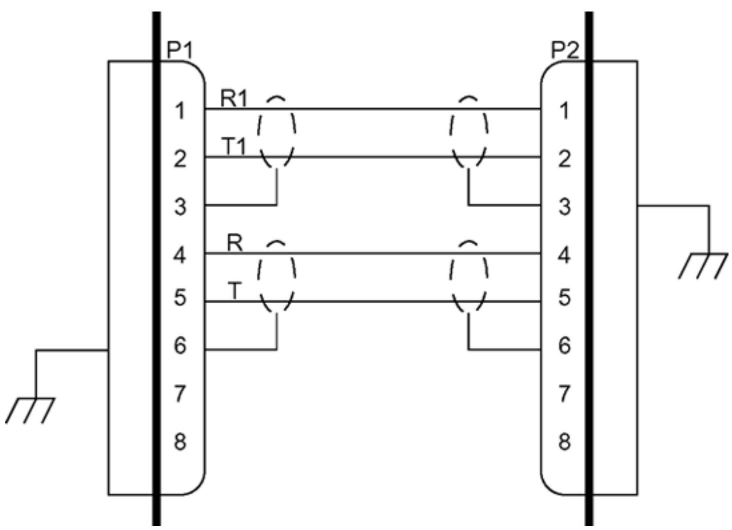
RJ48C Crossover Cable

For loop-back or other off-line tests where an outbound span is looped back directly to an inbound span, a special crossover cable can be fabricated. On a crossover cable, the transmit pins on one connector are connected to the receive pins on the other connector. An RJ48C crossover cable is wired as shown in the following illustration.



RJ48C Shielded Interface

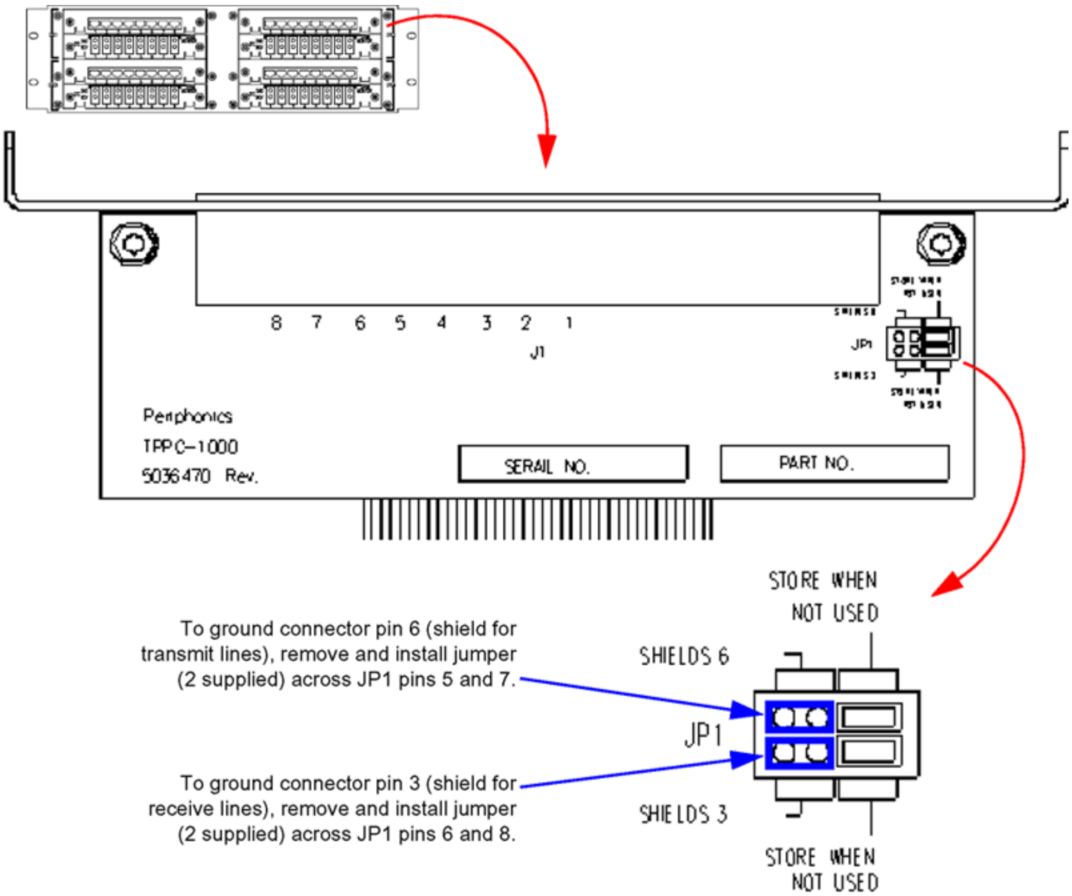
Some systems use RJ48C cables where the transmit and receive wire pairs are shielded. The shields for the transmit and receive wire pairs are connected to connector pins 6 and 3, respectively. These shields must be connected to the chassis ground at the terminating equipment that drives the signal. The connections are illustrated in the following interconnection diagram.



The TelCo Connector Panel (TCCP) provides a convenient way to connect the cable shields to chassis ground. Generally, this needs to be done only when the system is initially installed. If your system was installed by Avaya field support, grounding is already configured. However, if you previously connected to a non-shielded interface, and need to connect to equipment using shielded cables, jumpers must be installed in the TCCP to ground the shields.

Each RJ48C module is supplied with spare, uninstalled jumpers. Remove the two screws that secure the RJ48C module to the TCCP and unplug the module from the backplane. On JP1, remove one of the jumpers from its STORE WHEN NOT USED position and install it in the SHIELDS 6 position (across JP1, pins 5 and 7). This grounds the pin 6 shields (transmit lines) for all eight span connectors on the module.

Generally, the other spare jumper remains in the unused position. The shields for the receive lines are grounded at the connected equipment. Alternatively, ground the receive lines by moving the spare jumper to the SHIELDS 3 position (across JP1, pins 6 and 8). Each shield grounds at only one termination, as shown in the preceding interconnection diagram. Do not ground a shield at both ends.



MPS-M1/CS 1000 Connection Methods

The MPS can connect to the M1/CS 1000 PBX through two different methods:

- Through a Lineside T1 or Lineside E1 card, which permits CTI functionality.
- Through ISDN protocols, which does not permit CTI functionality.

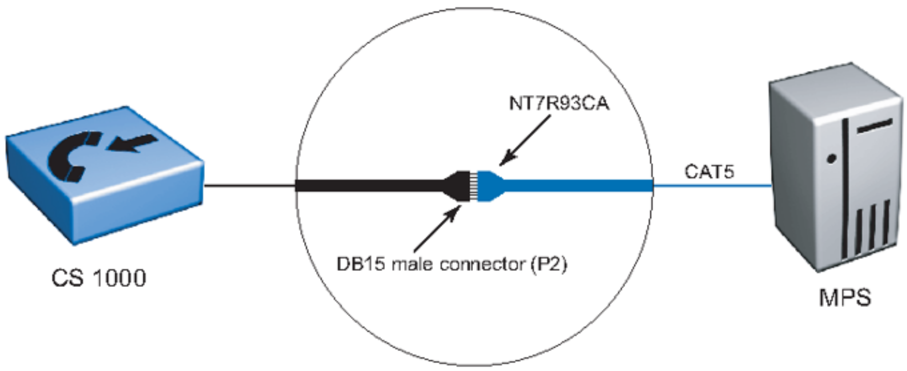
The following section documents how to connect the MPS with the M1/CS 1000 using a Lineside T1 or Lineside E1 connection. To connect the MPS to the M1/CS 1000 through ISDN protocols, see ISDN Configuration on page 57.

MPS Lineside T1 or E1 Connection to the M1/CS 1000

The following sections document how to connect the MPS to the M1/CS 1000 using a Lineside T1 or Lineside E1 connection.

Lineside T1 Connection

The following diagram shows the Lineside T1 cabling.



Lineside T1 Card NT5D11

The NT5D11 Lineside T1 interface card is an IPE line card that can be installed in the NT8D37 IPE module. A maximum of eight such cards can be installed. The Lineside T1 card interfaces one T1 line (24 channels) to the Avaya M1/CS 1000 PBX. This card occupies two card slots in the IPE shelf, utilizing 16 channels on slot 1 and eight channels on slot 2.

The Lineside T1 card emulates an analog line card to the Avaya M1/CS 1000 software. Each channel is independently configured by software control in the Analog (500/2500-type) Telephone Administration program LD 10. The Lineside T1 card comes with a Man Machine Interface (MMI) maintenance program that provides diagnostic information about the status of the T1 link.

Card Connections

The Lineside T1 card uses the NT8D81AA Tip and Ring cable to connect from the IPE backplane to the 25-pair amphenol connector on the IPE I/O (input/output) panel. The I/O panel connector then links directly to a T1 line, external alarm, and an MMI terminal or modem using the NT5D13AA Lineside T1 I/O that can be sourced from Avaya.

Cabling From the I/O Panel

The I/O panel is usually connected to the T1 link and other external devices through the NT5D13AA Lineside T1 I/O cable. This cable consists of:

Example

- A 25-pair amphenol connector (P1)
- A DB15 male connector (P2)
- A DB9 male connector (P3)
- A second DB9 male connector (P5)
- A DB9 female connector (P4)

To connect the I/O panel to the T1 link and other external devices using the NT5D13AA Lineside T1 I/O cable, perform the following steps:

1. Plug the 25-pair amphenol connector (P1) into the I/O panel.
2. Plug the DB15 male connector (P2) into the T1 line (use this plug with the NT7R93CA adapter) with the male-male CAT 5 cable. This, in turn, plugs into the MPS TMS.
3. Plug the DB9 male connector (P3) into the external alarm system.
4. Plug the second DB9 male connector (P5) into an MMI terminal or modem.
5. Plug the DB9 female connector (P4) into the next Lineside T1 card P4 connector for MMI daisy chaining.

The NT7R87BA (A0659794) adapter kit contains the NT7R93AA, BA, CA, and the NT7R89AA (45' RJ45 to RJ45).

NT7R93AA — CPE to Network Male DB15 (male) to RJ48

The following table pertains to settings prior to assembly cut wires from RJ48 pin 3 and 6:

RJ48 Connector (P1)	DB15 Connector (P2)	Wire Color	Signal Name
P1 - 1	P2 - 3	Blue	Tip In
P1 - 2	P2 - 11	Orange	Ring In
P1 - 3	P2 - 2	Black	No Connect
P1 - 4	P2 - 9	Red	Ring Out
P1 - 5	P2 - 1	Green	Tip Out
P1 - 6	P2 - 4	Yellow	No Connect
P1 - 7	P2 - 15	Brown	Signal 1
P1 - 8	P2 - 8	Grey	Signal 2

NT7R93BA — CPE to Network Female DB15 (female) to RJ48

RJ48 Connector (P1)	DB15 Connector (P2)	Wire Color	Signal Name
P1 - 1	P2 - 3	Blue	Tip In
P1 - 2	P2 - 11	Orange	Ring In
P1 - 3	P2 - 2	Black	GND
P1 - 4	P2 - 9	Red	Ring Out
P1 - 5	P2 - 1	Green	Tip Out
P1 - 6	P2 - 4	Yellow	GND
P1 - 7	P2 - 15	Brown	No Connect
P1 - 8	P2 - 8	Grey	No Connect

NT7R93CA — CPE to CPE Female DB 15 (female) to RJ48

The following table pertains to settings prior to assembly cut wires from RJ48 pin 3 and 6:

RJ48 Connector (P1)	DB15 Connector (P2)	Wire Color	Signal Name
P1 - 1	P2 - 1	Blue	Tip In
P1 - 2	P2 - 9	Orange	Ring In
P1 - 3	P2 - 2	Black	No Connect
P1 - 4	P2 - 11	Red	Ring Out
P1 - 5	P2 - 3	Green	Tip Out
P1 - 6	P2 - 4	Yellow	No Connect
P1 - 7	P2 - 15	Brown	Signal 1
P1 - 8	P2 - 8	Grey	Signal 2

Cable NT7R89AA — Cable Length 384 Inches (32ft)

RJ48 Connector (P1)	RJ48 Connector (P2)	Wire Color
P1 - 1	P2 - 1	Blue
P1 - 2	P2 - 2	Orange
P1 - 3	P2 - 3	Drain 1
P1 - 4	P2 - 4	Red
P1 - 5	P2 - 5	Green
P1 - 6	P2 - 6	Drain 2
P1 - 7	P2 - 7	NC
P1 - 8	P2 - 8	NC

Lineside E1 Connection

The Lineside E1 Interface Card (LEI) is an Intelligent Peripheral Equipment (IPE) line card. The LEI card provides an all-digital connection between E1 compatible terminal equipment, such as a voice mail system, and the Avaya M1/CS 1000 PBX.

Example

- The LEI interfaces one E1 line carrying 30 channels to the Avaya M1/CS 1000 PBX.
- The LEI emulates an analog line card to the Avaya M1/CS 1000 software.

  Each channel is independently configured by software control in the Analog (500/2500-type) Telephone Administration program LD 10.
- The LEI also comes equipped with an MMI interface program, which provides diagnostic information about the status of the E1 link.

Perform the following installation steps :

Example

- Install the NT5D33 version of the LEI in the NT8D37 IPE module.
- Install the NT5D34 version of the LEI in:
  - The PBX 11C NTAK11 Main Cabinet
  - The PBX 11C NTAK12 Expansion Cabinet
  - The NT1P70 Small Remote IPE Main Cabinet
  - The NTAK12 Small Remote IPE Expansion Cabinet

Before you order the LEI, you must know whether the installation uses twisted pair E1 (120 ohm) or coaxial E1 (120 ohm).

Example

- The NT5D35AA cable is ordered for twisted-pair E1 installations
- the NT5D36AA cable is ordered for customer sites that need 75 ohm E1 connections

Table 2: Lineside E1 Characteristics

Part Number	Description	Circuits	Line Type	Message Waiting	Architecture
NT5D33AB	LEI for IPE module	30	E1	None	Option 21-81 IPE

Part Number	Description	Circuits	Line Type	Message Waiting	Architecture
NT5D33AB	LEI for Option 11 and small remote cabinet	30	E1	None	Option 11 Small Remote Cabinet

Twisted-Pair Installations

For twisted-pair installations, E1 signaling for all 30 channels is transmitted over P2 connector pins 1, 3, 9 and 11, as shown in the following table. Plug the DB15 male connector labeled P2 into the E1 link. E1 transmit and receive pairs must be turned over between the LEI and CPE equipment that is hardwired without carrier facilities.

If the LEI is connected through E1 carrier facilities, wire the transmit and receive pairs straight through to the RJ48 and Telco demarc, the LTU, or other E1 carrier equipment. The E1 CPE equipment at the distant end similarly has transmit and receive wired straight from the RJ48 demarc at the distant end of the carrier facility.

75 ohm Coaxial Installations

For 75 ohm coaxial installations, E1 signaling for all 30 channels is transmitted over P2 connector pins 1, 3, 9 and 11 through an adapter and out of two coaxial connectors Tx (transmit) and Rx (receive). Tx is the LEI output, and Rx is the LEI input from the E1 stream.

E1 transmit and receive pairs must be turned over between the LEI and the CPE equipment that is hardwired without carrier facilities. If the LEI is connected through E1 carrier facilities, wire the transmit and receive pairs straight through to the RJ48 at the Telco demarc, the LTU, or other E1 carrier equipment. The E1 CPE equipment at the far end likewise has Tx and Rx wired straight from the RJ48 demarc at the far end of the carrier facility.

Table 3: Lineside E1 Backplane Pin-outs

I/O Panel Connector Pin	Lead Designations	LEI Connector Pin	LEI Cable Connector to External Equipment
1	E1 Tip Receive Data	11	DB15 Male to E1 (P2). LEI is CPI transmit and receive to network.
26	E1 Ring Receive Data	3	
2	E1 Tip Transmit Data	1	
27	E1 Ring Transmit Data	9	

Card Connections

1. Use the NT8D81AA Tip and Ring cable to connect the LEI from the IPE backplane to the 25-pair amphenol connector on the IPE I/O panel.
2. Use the NT5D35 or NT5D36 Lineside I/O cable to connect the I/O panel connector to an E1 line, external alarm and an MMI terminal or modem. This cable can be sourced from Avaya.

Cabling between the Lineside E1 and the MPS

The following table shows cabling between the MPS and the Lineside E1 card.

LSE 1 Card P2 Connector (male DB15)		MPS Connector (RJ48C)
E1 Tip Tx	--- 1 <--> 2 ---	E1 Tip Rx
E1 Ring Tx	--- 9 <--> 1 ---	E1 Ring Rx
E1 Tip Rx	--- 11 <--> 5 ---	E1 Tip Tx
E1 Ring Rx	--- 3 <--> 4 ---	E1 Ring Tx

Note:

Each LEI card has two part numbers (depending on the Avaya M1/CS 1000 PBX architecture) and two types of connectors (twisted-pair or 75 ohm coaxial cable), depending on the cabling used.

Chapter 4: Configuration For CTI Functionality

This chapter covers:

Configuring the Telephony Protocol

Once the MPS is set up, use the MPS Configurator to specify the telephone protocol the system uses. You must do this before you begin any processing. Once you start the MPS Configurator, it automatically detects:

Example

- the number of MPS units attached to this application processor
- the chassis number assigned to the MPS
- the type of cards the processor is using
- the Back Plane Slot (BPS) number of each slot
- the number of Digital Signal Processors (DSPs)

The MPS Configurator can later be used to identify and manage the Multimedia Format Files (MMFs) to load at boot time.

Note:

For more information on configuring the telephony protocol, refer to the Installing MPS Software on the Windows Platform Manual.

Configuring MPS for CTI Functionality

The MPS can be configured with the PBX using a Lineside T1 connection or through ISDN protocols. Of these two options, only a Lineside T1 connection enables CTI functionality. You cannot use ISDN protocols for CTI functionality. This chapter documents the procedures for MPS-PBX configuration for CTI functionality only.

Configuring the PBX

Connections can be made to the Avaya PBX in the following ways:

Example

- Using standard PBX hunt groups:
  - Traffic distribution may not be as even as ACD, so some ports may be used more extensively than others.
  - Graphical tools MPS Manager and MPS Reporter are used to provide statistical information on the traffic served by the IVR.
- Configuring IVR ports as ACD agents:
  - Provides automatic distribution of traffic across all IVR ports configured in the ACD-DN.
  - Provides integration into the contact center for enhanced routing treatments, real-time displays, and historical reporting, just like live agents in a contact center.
  - MPS Manager and MPS Reporter supplements the Contact Center Manager Server (CCMS) in providing statistical information on the traffic being served by the IVR.

Checklist for PBX Configuration

Perform the following processes to configure the Avaya PBX:

Example

- Configure Trunk Route Data in Load Program 16 (LD 16).
- Configure positive disconnect signals to the IVR configurations.
- Configure IVR ports in the ACD.

- Configure CTI parameters.
- Configure overlay setups.
- Configure 500/2500 sets (including IVR ports).
- Configure Digital Sets—this is not typically needed for IVR ports, but used as test sets or in the contact center environment with the CCT.
- Configure ACD DNs (agent queues).
- Configure controlled DNs.
- Configure DNIS notification.

Note:
Some Avaya PBX configuration items are dependant on customer requirements, such as CTI, DNIS, controlled DNs, or recorded announcements if all IVR ports are busy.

Configuration Interfaces

The MPS IVR can be configured with the PBX and the CCMS in several ways, depending on customer requirements. The following scenarios are possible:

IVR as the Front End

The advantages of having MPS at the front end are as follows:

Example

- Decreased call traffic on the CCMS because only callers requesting a live agent are transferred to the CCMS
- IVR ports do not count as active agents because they are set up as voice ports.

The disadvantages are as follows:

Example

- If the MPS IVR/ host has problems and the Avaya PBX still shows ACD agents (ports) logged in, you cannot easily send calls to the CCMS.
- The MPS at the front end cannot provide a single report with the number of calls offered to the IVR and the CCMS.

Contact Center Manager Server as the Front End

The advantages of having CCMS at the front end are as follows:

Example

- Front end CCMS can provide a single report with the number of calls offered to the MPS IVR and the CCMS.
- If the MPS IVR host has problems and the Avaya PBX still shows ACD agents/ports logged in, you can easily send calls to the CCMS.

The disadvantages are as follows:

Example

- Increase in call traffic to the CCMS.

  Each caller who wants to speak to a live agent must be transferred back to the CCMS.
- There is an increase in cost because IVR ports have to be set up as active agents.

Physical IVR Port Connection Information

Most IVR installations (about 90 percent) behind the Avaya PBX use the Lineside T1/ E1 interface, which provides 24 or 30 voice connections to the Avaya MPS.

Note:

For ISDN configurations, see ISDN Configuration on page 57. For SIP configurations, see SIP Configuration on page 79.

The following DIP switch settings are possible on the Lineside T1:

Example

- Framing D4
- Signaling Loop
- Coding AMI

Line supervision on T1 Failure

This setting determines the state in which all 24 ports of the Lineside T1 card appear to the Avaya PBX in case of T1 failure. Ports on the Avaya PBX can appear either in the on-hook or off-hook states on T1 failure.

Note:

All idle Lineside T1 lines go off-hook and seize a Digitone receiver when off-hook line processing is invoked on T1 failure. This may prevent DID trunks from receiving calls until the Lineside T1 lines time out and release the DTR.

IVR Port Configuration

IVR ports typically connect to the Avaya PBX on the Lineside T1 or E1 card. IVR ports can be configured on the Avaya PBX as:

Example

- Analog 500 DNs in a hunt chain
- Analog 500 agents in an ACD queue (for queue hold or IVR skillset functionality)

Contact Center Manager Server Call Flow

Example

- IVR ports from front-end call center

  Calls enter an IVR, and only when the caller chooses to talk to an agent does the IVR transfer the call to a CDN controlled by the CCMS.

- IVR ports used for Hold in Queue for IVR functionality

  Calls enter a CDN controlled by the CCMS. The call is controlled by the CCMS script, and the CCMS sends a message to the Avaya PBX to queue call IVR ports. The call secures place in the queue (skillset). Once the caller receives a message, the IVR port disconnects from the call and the CCMS continues script execution.

- IVR ports are agents in a skillset

  Calls enter a CCMS-controlled CDN. IVR ports are configured as live agents in a skillset. The CCMS script uses a queue to skillset command. The IVR port transfers calls back into different CDNs on the CCMS, based on caller selection.

IVR ports are set up as agents. When the CCMS receives the call, it uses one of two mechanisms to transfer the call to the IVR:

Example

- Queue to Skillset: Do the following:
  - Set up IVR ports with skills IVR in the CCMS and the script.
  - Do a Queue to SkillSet IVR.

    Contact Center Manager Server queues the call to the available IVR agent port.
  - If the caller wants to speak to a live agent, the IVR application must blind transfer the call to a different CDN that the CCMS controls.
- Give IVR with Treatment: Do the following:
  - Set up a queue for IVR ports.
  - Set up IVR ports as voice ports in the CCMS.
  - Direct the call to the CCMS CDN.

The CCMS script executes the Give IVR with Treatment command. If the IVR application needs to transfer the call back to the CCMS, it disconnects and CCMS takes the call back. The CCMS script utilizes HDX to request and receive IVR collected data.

Avaya PBX Configuration

This section explains how to configure the Avaya PBX with the CCT server to enable CTI messages to be generated, along with screen pops and other CTI functionalities.

Note:

For additional information and sample configurations, see Appendix D–M1/CS 1000 Sample Configurations in the MPS with CCTIVR Configuration and Interfaces Guide.

Configuration Checklist

1. Ensure all agent PCs are able to ping the CCT Server.
2. Develop a list of all agent names and ACD login IDs (for example, Jan Smith, 3458; Bill Jones, 3643). Specify the device numbers for every agent phone set to be monitored (this includes both the PBX DN and the position ID for each phone set).
3. Develop a list of all ACD-DNs (queues) and CDNs through which calls may flow.
4. Develop a list of all IVR ports with corresponding DID access numbers, if applicable, that run MPS Developer applications configured with CTI. The application attaches DNIS and some call data key/value pairs and then performs CTI mute transfers to one of the CDNs specified above. If the CTI mute transfer fails, the application issues a hookflash transfer to the same CDN.
5. Develop both a production call-flow diagram and a production CTI network diagram.
   a. Create a call-flow diagram that shows:
      - the DID CDN for incoming calls
      - the IVR hunt group
      - the IVR ACD-DN or queue
      - the IVR agent Position IDs
      - the CDNs to which the IVR application transfers calls, and any agent queues through which the call may pass
      - The desktop agents PBX DN, position ID, name, and ACD login ID
   b. Create a network diagram that shows all IP addresses, subnet masks, and default gateways for the following:
      - the CCT Windows 2003 server or MPS Windows 2003 Application Processor Server if CCT is co-resident
      - the CCT server and network cards in the IVR that are on the customer's LAN
      - the CCMS Server subnet (formerly named CLAN) port
      - any host computers, routers, or firewalls through which the data passes between the IVR and the CTI Agent desktops
6. Ensure all agent phone sets and IVR ports in the call center are CTI-enabled. This is typically done in the LD 10 and LD 11 programs. AST must be enabled.
7. Ensure all CDNs and ACD DNs within the production call flow are CTI-enabled. This is typically done in the LD 23 program. AST must be enabled.

Note:

AST in LD 23 is supported with only X11 Release 16 and earlier. X11 Release 17 and later use the prompt ISAP to enable associated set (AST) functionality to send CTI messages to the appropriate application servers that need the information.

Note:

VSID is configured for the VSID defined in LD 17 only when connecting to a Meridian Link Server. If you are using the Meridian Link Services Module on the CCMS system, then the VSID in LD 23 is left blank.

Configuring Trunk Route Data in Load Program 16

Requirements to attach data for CCT/CTI from a Avaya IVR are as follows:

1. Connect to the PBX using the 2500 station ports.
2. Configure ports according to status change message configuration requirements in the LD 15.
3. Configure the Lineside T1/E1 as the 2500 station port.
4. Use the VRU application to perform an attach data to the CCT server. This is optional but required for the DNIS, CLID and Treatment DN information to be passed from the switch or the CCMS to the IVR and back.
5. Agent queue and phone must be AST/ISAP enabled for CCT/CTI integration.
6. PBX software level must be 25.x or later.
7. Meridian Link software must be Release 4.x or later for the stand-alone Meridian Link Server.
8. Contact Center Release 6.0 or later with CCT Server

Configuring Positive Disconnect Signals to the IVR

In Load 10 at the FTR block in the SET DATA BLOCK, add ISP OSP.

The FTR configuration enables AB bit signaling (D4-AMI). ISP enables disconnect supervision for internal calls (transfers). OSP enables disconnect supervision for external calls (PBX to IVR). Check the Load printout for the FTR_DATA block. It is in LD 21, or just above the SPRE code.

Note:
Check the Load printout for the FTR_DATA block; it is in the LD 21, or just above the SPRE code.

Configuring IVR Ports in the ACD

Configuring IVR ports in the ACD requires Lineside T1 configuration. Configure the IVR ports to automatically log in to an ACD group. Because the IVR ports are not configured as agents, they do not need to explicitly log in to or out of the switch using SPRE codes.

Configuring CTI Parameters

Every DN, CDN, and ACD queue that requires monitoring (generating CTI messages) must be CTI-enabled.

Note:
The two most critical parameters are AST and IAPG:

- Set AST to on for analog sets 500/2500, which is how the IVR ports are configured.
- Set AST = key1 key2 (for example, 00 01) for digital sets, which is how most live agents are configured.
- Set IAPG to 1, which means all messages are broadcasted. Changing the IAPG value filters out certain types of messages, which is not desirable.
- Set ISAP to yes for ACD queues, and set the VSID parameter to the VAS ID value configured in the switch.

Configuring Overlay Setups

The following describes how to configure overlays:

1. Configure the LD 17 (Load Pgm 17) record.
2. Define an AML, using the Avaya default values.
3. Define the VAS server with a unique VAS ID assigned to the AML port.

4. Configure the VAS parameters. Configure Status Change Msgs in LD 15 by setting VSID to the VAS ID of the Meridian Link defined in LD 17. This is necessary only for non-ACD DNs, such as calls between agents.
5. Set SECU to yes. (Enabling this security parameter allows CTI transfers from IVR applications.)

Configuring 500/2500 Sets

The following describes how to configure the 500/2500 sets:

1. Configure the LD 10 (Load Pgm 10) record.
2. Set the AST prompt to yes.
3. Set the IAPG prompt to 1.
4. Set the AACD prompt to yes (for an ACD set).
5. Set the CLS (class of service) to have XFA (transfer allowed), AGTA (ACD services allowed) and (LDTA) Line Disconnect Tone Allowed.

   Note:
   For Option 11C systems, also set CLS to have the MBX A prompt turned on.
6. Set the FTR ACD prompt (queue and position ID in particular).

   Note:
   In a CCMS environment, you must deaquire any previously acquired DNs before making changes to the device in the switch. If you do not explicitly configure the CCMS to acquire the IVR ports, these ports may appear acquired when, in fact, they are not. Their bitmaps become visible, which makes it look like they have been acquired.
7. Set FTR OSP 1 - enable battery reversal answer and disconnect supervision for outgoing calls with absolute and assumed answer indication.
8. Set FTR ISP 75 - Enable hookflash disconnect supervision with flash timer in 10-millisecond units. If the numeric parameter is not entered and the saved value is null, it is defaulted to 75 (750 ms). Otherwise, it remains unchanged.

Configuring Digital Sets

The following describes how to configure agent or supervisor digital sets:

1. Configure the LD 11 (Load Pgm 11) record.
2. Set the AST prompt to indicate the keys that will be AST-enabled (maximum of two). For instance: AST = 00 01 (but keys can vary at each site). The first key is the agent incoming ACD position, and the second key is the PBX DN.
3. Set the IAPG prompt to 1. This means all CTI messages are sent without messaging filtering.

   The position ID is a separate setting also contained in the Load 11 program.
4. Program the Transfer key on each phone (if the transfer feature is to be used).
5. Program the Conference key on each phone (if the conference feature is to be used).

   Note:
   The set can either be an agent set or a supervisor set.
6. Document the position ID and PBX DN for every agent. These values are registered by CTI client applications to receive call presentation messages.

Configuring ACD DNs

This is the configuration for agent queues.

1. Configure the LD 23 (Load Pgm 23) record.
2. If a prompt labeled AST exists, set it to YES.
3. Set the ISAP prompt to YES.
4. Set the VSID prompt to the Meridian Link VAS ID assigned in LD 17.

After setting ISAP = YES, if the VSID parameter is not visible, exit from LD 23 and run it again. The VSID parameter becomes visible, and is most likely blank. Set VSID to the VAS ID configured in the switch. The VAS ID is typically between 0 and 15 for MLINK and between 16 and 31 for a CCMS environment.

Configuring Controlled DNs

The following describes how to configure controlled DNs:

1. Configure the LD 23 (Load Pgm 23) record.
2. Set the CNTL prompt to YES.
3. Verify that the ASID prompt is equal to the Meridian Link VAS ID assigned in LD 17.

If a CCMS has acquired the CDN, it will automatically set CNTL to YES and assign a different parameter (ASID) to the Meridian Link VAS ID. The VSID and HSID prompts appear blank and are not configurable.

Configuring DNIS Notification

The following describes how to configure DNIS notification:

1. Set the OPT prompt to DNIS in Customer Data Block in LD 15.
2. Set the DNIS prompt to YES to enable the route to pass DNIS in the Trunk Route.
3. Set the LENGTH prompt to the number of DNIS digits expected (usually 4, 7, or 10). You can specify any number up to 31.
4. Configure Incoming Digit Conversion (IDC) to direct DID or DNIS digits to ACD-DNs or CDNs. For the LD 16, set the IDC to YES and configure the IDC table number to be used for this route.
5. Configure IDC tables in LD 49.

IVR Information

The following information is required for MPS systems:

Example

- IP address
- the following lists:
    - list of all Position IDs for each phone port on the IVR
    - list of queues
    - list of CDNs through which a call may flow
    - list of modem lines for dial-up access

Note:

If a customer requires access to the IVR through a secure system, the customer must provide access for the programmer, field engineer, and technical support.

CCT Server Information

The following CCT Server information is required:

Example

- IP address of CCT Server
- IP address of the Avaya server subnet on the CCMS server
- host name of the CCT Server
- the following lists:
  - list of all queues, DNs, and CDNs though which calls may flow
  - list of all software applications to be installed on the CCT Server. Applications such as backup and security software are the responsibility of the customer to install, configure, and maintain.
  - phone lines for pcAnywhere to provide access to the CCT Server. If a customer requires access to the CCT Server through a secure system, the customer must provide access for the programmer, field engineer, and technical support.

CCTIVR Checklist

The following items are required:

- One or more test agent phone sets identical to the ones in the call center.
- One or more test computers identical to the ones in the call center alongside the test agent phone sets. These computers must be running the same operating system and have the same software applications installed as the production call center agents.
- One or more text PCs with pcAnywhere installed. Configure the these PCs as hosts waiting for a TCP/IP connection. Password protection is recommended.
- The test PCs must be able to ping the TAPI server.
- One or more test agent names and ACD login IDs (for example, CTI1, 1001 and CTI2, 1002). Know the PBX DNs and the position IDs for each phone set.
- The test agents must be set up to log into a single test queue automatically. Do not configure the phones to become Not Ready or Set Busy if a call is not answered.
- A single test CDN that transfers calls to the test agent queue specified above. This may require a writing a script and attaching it to the CDN.
- Two test DID access numbers that are connected directly to the CallPilot system. These test DID access numbers transfer calls to the IVR test lines (ports 95 and 96). On each

of these ports, run a test MPS Developer application configured for CTI. These applications attach DNIS, and some call data key value pairs, then perform a CTI mute transfer to the specified test CDN. If the CTI mute transfer fails, the application issues a hookflash transfer to the same CDN. Specific message logging must be in place to indicate the following:

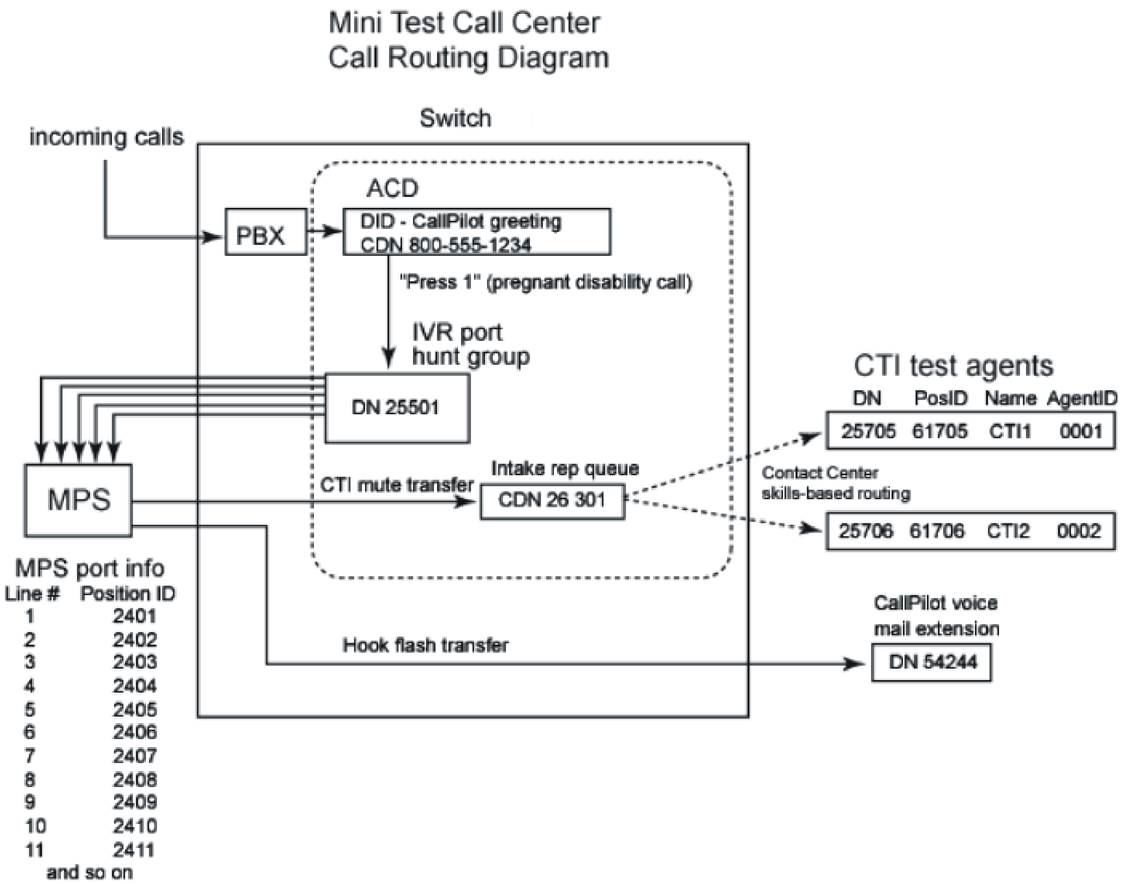
- whether the CTI resource was obtained
- if the resource was obtained, add messages after each add-data request indicating the result (success or failure)
- provide the request to issue a CTI mute transfer and the result
- if an error occurs, log it and provide information on the request to issue a hookflash transfer and the result

• The test agent phone sets must be CTI-enabled.

• The test agent PCs must have CCT server and Agent Desktop software installed.

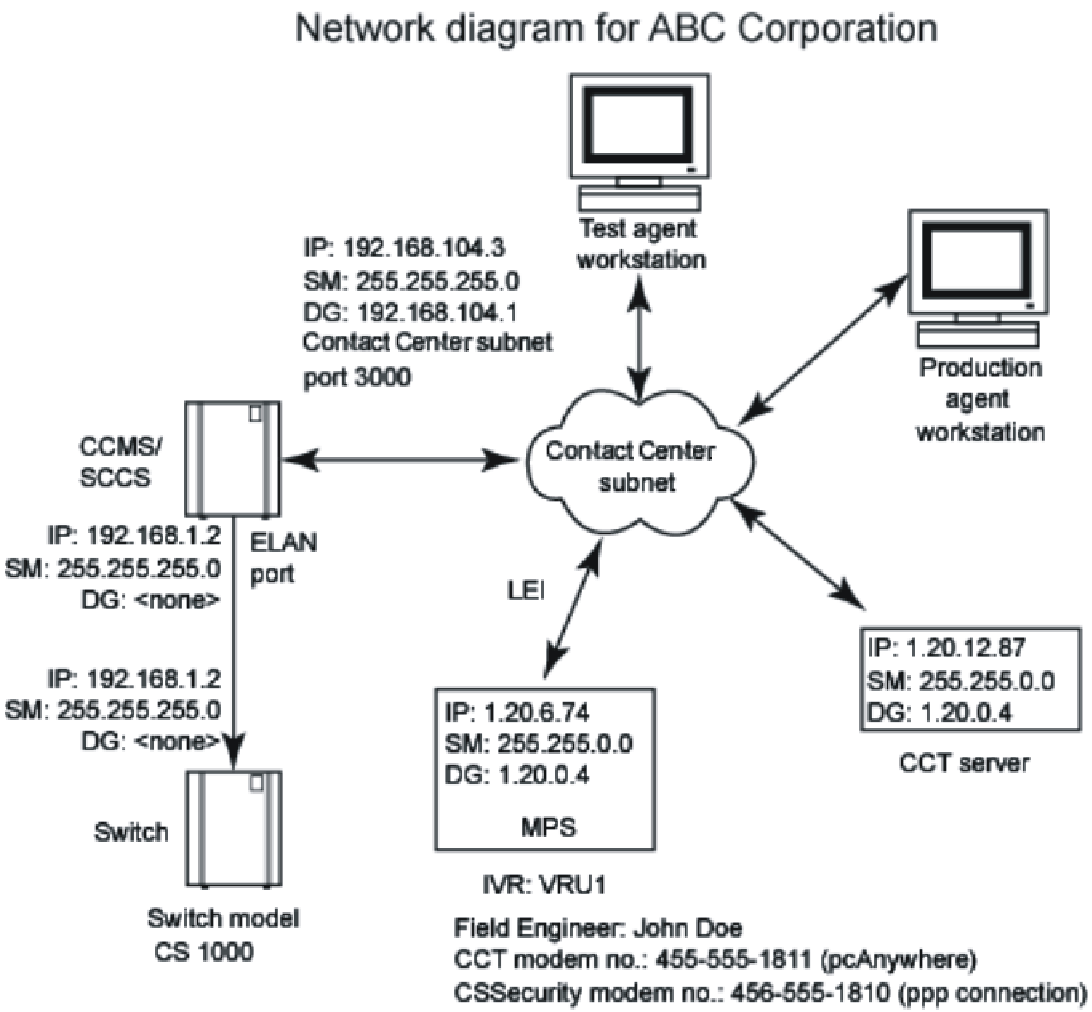
CTI Test Call Flow Diagram

Develop a test call-flow diagram similar to the one shown in the following diagram. It must contain all DNs, CDNs, and ACD DNs (queues, splits, route points, and so on) through which the call flows.



Sample CTI Network Diagram

Develop a CTI network diagram with all relevant IP information in a format similar to the one shown in the following diagram.



Avaya PBX-CCMS Sample Configuration

For details, refer to the MPS with CCTIVR Configuration and Interfaces Guide.

Chapter 5: ISDN Configuration

This chapter covers:

Configuring the Telephony Protocol

Once the MPS is set up, use the MPS Configurator to specify the telephone protocol the system uses. You must do this before you begin any processing. Once you start the MPS Configurator, it automatically detects:

Example

- the number of MPS units attached to this application processor
- the chassis number assigned to the MPS
- the type of cards the processor is using
- the Back Plane Slot (BPS) number of each slot
- the number of Digital Signal Processors (DSPs)

The MPS Configurator can later be used to identify and manage the Multimedia Format Files (MMFs) to load at boot time.

Note:

For more information on configuring the telephony protocol, refer to the Installing MPS Software on the Windows Platform Manual.

ISDN Protocol Communication

The MPS can communicate with the M1/CS 1000 PBX using an ISDN protocol. If you configure the MPS to communicate with the M1/CS 1000 switch using an ISDN protocol, CTI cannot be enabled. Because the CCT is not compatible with ISDN, this is a telephone connection only.

MPS Configuration

For details on configuring the MPS using an ISDN protocol, refer to the Avaya Media Processing Server Series Telephony Reference Manual.

Note:

Use the National ISDN option to configure the IVR to work with the M1/CS 1000.

```
ISDN SECTION
;
; This section is used only when using Unified ISDN. It is
; used to set the a D-Channel Map and to specify External
; Interface IDs for each span. The D-Channel Map is used
; to configure which span or spans provides D-Channels
; and for what spans the D-channels are provided and set
; the Variant of ISDN. The default is one D-Channel per
; span and a variant of Avaya. The values that can be
; used for Variants are:
;
; Avaya - NIS-A211-1
; National - SR-NNT-002120
; NTT - JT-Q931
;
; If this section is left commented out the default is
; Primary Rate National ISDN. The SPAN_EXT_ID line
; assigns an External Interface id to the spans. This is ;
; only used if the D_Channel Map is being set to something
; other then Default.
;
; Format of ISDN SECTION is:
;
; (The following example sets the D-Channel on span 1
; this D-Channel is used for all spans (191B+D)).
;
[ISDN]
;
;----------------------------------------------------------------------
```

; ;	DTC Num	PLI Slot	D-Chan Span	Spans	Switch Type
DCHANMAP	1	4	1	1,2,3,4,5,6,7,8	Avaya

```
; Example (1) - All spans configured as 23B+D. Each running a
; different variant of ISDN.
```

; ;	DTC Num	PLISlot	D-Chan Span	Spans	Switch Type

DCHANMAP	1	4	1	1	Avaya
DCHANMAP	1	4	3	3	National
DCHANMAP	1	4	4	4	Avaya
DCHANMAP	1	4	6	6	National
DCHANMAP	1	4	7	7	Avaya

```
; Example (2) - All spans configured 4 groups of 47B+D. ;
; Each running a;different variant of ISDN.
```

; ;	DTC Num	PLISlot	D-Chan Span	Spans	Switch Type
DCHANMAP	1	4	2	1,2	Avaya
DCHANMAP	1	4	6	5,6	National
DCHANMAP	1	4	8	7,8	Avaya

```
; Example (3) - All spans configured 2 groups of 95B+D. ;
; Each running a;different variant of ISDN.
```

; ;	DTC Num	PLISlot	D-Chan Span	Spans	Switch Type
DCHANMAP	1	4	4	1,2,3,4	Avaya

```
; Example (4) - All spans configured 1 group of 191B+D. ;
; Running;Avaya variant of ISDN.
```

; ;	DTC Num	PLISlot	D-Chan Span	Spans	Switch Type
DCHANMAP	1	4	8	1,2,3,4,5,6,7,8	Avaya

```
; Specify the SPAN_EXT_ID specification in the [ISDN]
; Section. For all configurations except 23B+D (PRI), ; ;
```

```
; external interface numbers for the spans can be defined.
; These are done in the SPAN_EXT_ID section as follows:
```

;	DTC Num	PLISlot	Span	Interface ID
SPAN_EXT_ID	1	4	1	10
SPAN_EXT_ID	1	4	2	11
SPAN_EXT_ID	1	4	3	12
SPAN_EXT_ID	1	4	4	13
SPAN_EXT_ID	1	4	5	14
SPAN_EXT_ID	1	4	6	15
SPAN_EXT_ID	1	4	7	16
SPAN_EXT_ID	1	4	8	17

MPS/CS 1000 Configuration

The LD 17 must be configured with the D-channel type set to NI2. Configure the switch for the network side of the interface as shown in the following example. The key configurations are followed with explanations in parenthesis.

Configure D-Channel in LD 17

```
ADAN DCH 37
CTYP MSDL
GRP 2
DNUM 8
PORT 1
DES IVR
USR PRI
DCHL 76
OTBF 32
PARM RS422 DTE
```

```
DRAT 64KC
CLOK EXT
```

IFC NI2 (interface type for D-channel is NI2)

```
ISDN_MCNT 300
CLID OPT0
CO_TYPE STD
```

SIDE NET (SIDE is NET – Network, the controlling switch)

```
CNEG 1
RLS ID **
RCAP COLP
T310 120
T200 3
T203 10
N200 3
N201 260
K 7
BSERV NO
```

Route Data Block Configured in LD 16

Software package 334 (NI2 call by call) is equipped, which forces the tie route to be programmed as call by call (CBC). You must turn off package 334 NI2 CBC before configuring the NI2 Tie route. After configuring the NI2 Tie route, perform a data dump and turn package 334 back on.

```
TYPE RDB
CUST 00
ROUT 28
DES IVR
```

TKTP TIE (Trunk type is Tie)

```
NPID_TBL_NUM 0
ESN NO
CNVT NO
SAT NO
RCLS EXT
DTRK YES
DGTP PRI
ISDN YES
MODE PRA
```

IFC NI2 (Interface type for D-channel is NI2)

```
CBCR NO
NCOS 5
SBN NO
PNI 00000
NCNA YES
NCRD NO
```

```
CHTY BCH
CTYP UKWN
INAC NO
CPFXS YES
CPUB OFF
DAPC NO
BCOT 0
INTC NO
DSEL VOD
PTYP PRI
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH RRB
TRMB YES
STEP
ACOD 60028
TCPP NO
PII NO
TARG 01
CLEN 1
BILN NO
OABS
INST
IDC NO
DCNO 0 *
NDNO 0
DEXT NO
ANTK
SIGO STD
ICIS YES
TIMR ICF 512
OGF 512
EOD 13952
NRD 10112
DDL 70
ODT 4096
RGV 640
GRD 896
SFB 3
NBS 2048
```

Trunks Configured in LD 14

```
DES IVR
TN 076 01
```

TYPE TIE (Trunk type is Tie)

```
CDEN SD
CUST 0
TRK PRI
PDCA 1
PCML MU
NCOS 5
RTMB 28 24
B-CHANNEL SIGNALING
```

```
TGAR 1
AST NO
IAPG 0
CLS CTD DTN CND WTA LPR APN THFD HKD
P10 VNL
TKID
DATE 13 JAN 2005
```

Confirm the Call Type in LD 86

```
REQ prt
CUST 0
FEAT dgt

DMI 28
DMI 28
DEL 0
CTYP LOC
```

CS 1000 ISDN Configuration

ISDN can be configured using the following two channels:

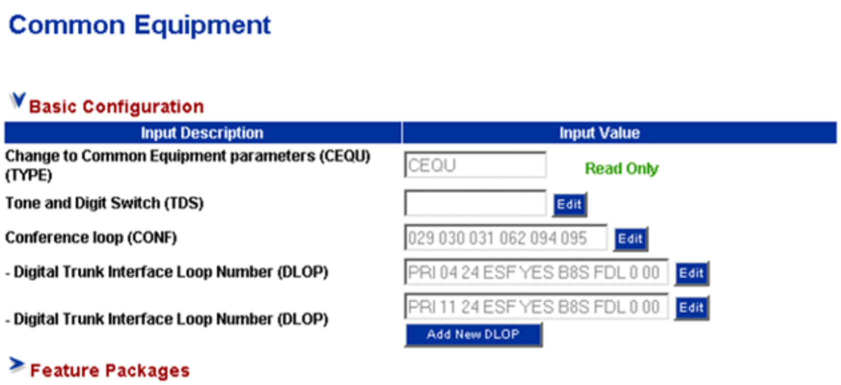
Example

- ISDN configuration using Element Manager
- ISDN configuration using overlay commands

ISDN Configuration Using Element Manager

Perform the following steps to configure ISDN using the Element Manager:

1. Move through Configuration > Call Server and select Common Equipment from the menu.
2. Create a PRI loop. If a loop already exists within the slot, remove it by disabling the corresponding span. The PRI loop is present in slots 2 and 4.



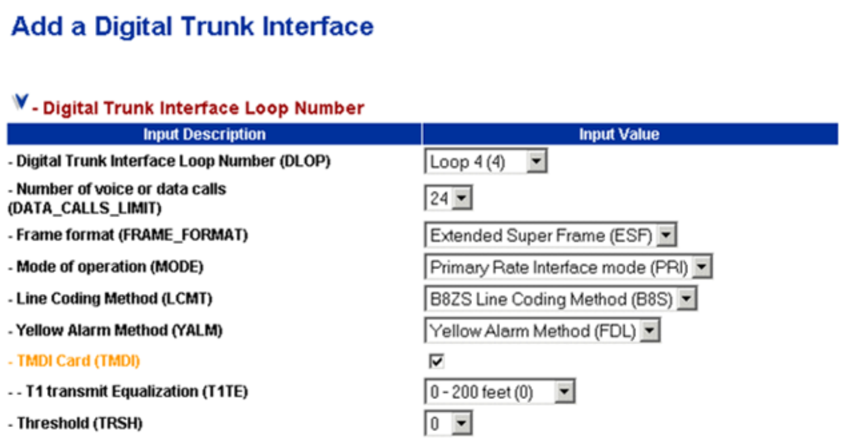
Common Equipment

Basic Configuration

Input Description	Input Value
Change to Common Equipment parameters (CEQU) (TYPE)	CEQU Read Only
Tone and Digit Switch (TDS)	Edit
Conference loop (CONF)	029 030 031 062 094 095 Edit
- Digital Trunk Interface Loop Number (DLOP)	PRI 04 24 ESF YES B8S FDL 0 00 Edit
- Digital Trunk Interface Loop Number (DLOP)	PRI 11 24 ESF YES B8S FDL 0 00 Edit Add New DLOP

Feature Packages

3. Click Add New DLOP to add a new Digital Trunk Interface Loop Number (DLOP).
    - Set Number of voice or data calls to 24 to specify the total number of channels.
    - Set Frame Format to Extended Super Frame (ESF).
    - Set Mode of Operation to Primary Rate interface mode (PRI).
    - Set Line Coding Method to B8ZS Line Coding Method (B8S).
    - Select the TMDI CARD (TMDI) check box to indicate TMDI is used.

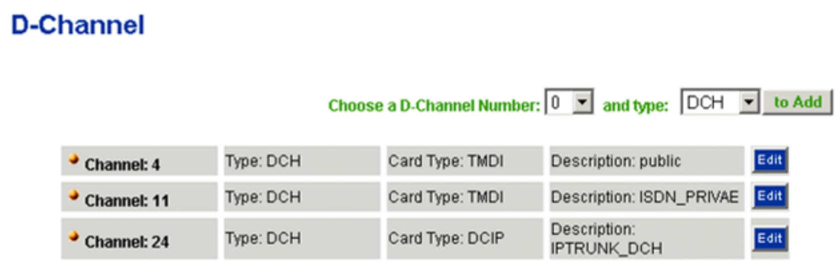


Add a Digital Trunk Interface

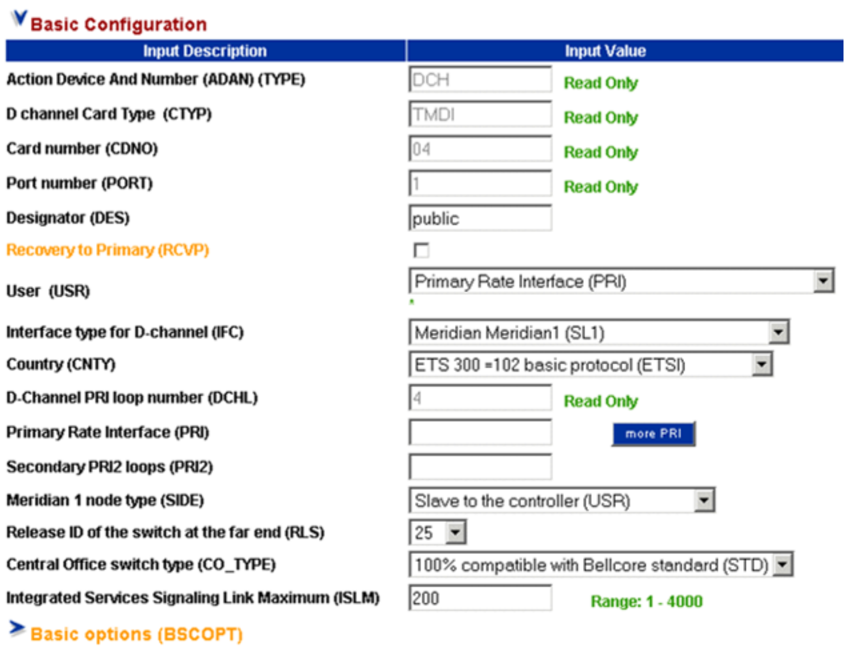
- Digital Trunk Interface Loop Number

Input Description	Input Value
- Digital Trunk Interface Loop Number (DLOP)	Loop 4 (4)
- Number of voice or data calls (DATA_CALLS_LIMIT)	24
- Frame format (FRAME_FORMAT)	Extended Super Frame (ESF)
- Mode of operation (MODE)	Primary Rate Interface mode (PRI)
- Line Coding Method (LCMT)	B8ZS Line Coding Method (B8S)
- Yellow Alarm Method (YALM)	Yellow Alarm Method (FDL)
- TMDI Card (TMDI)	☑
- - T1 transmit Equalization (T1TE)	0 - 200 feet (0)
- Threshold (TRSH)	0

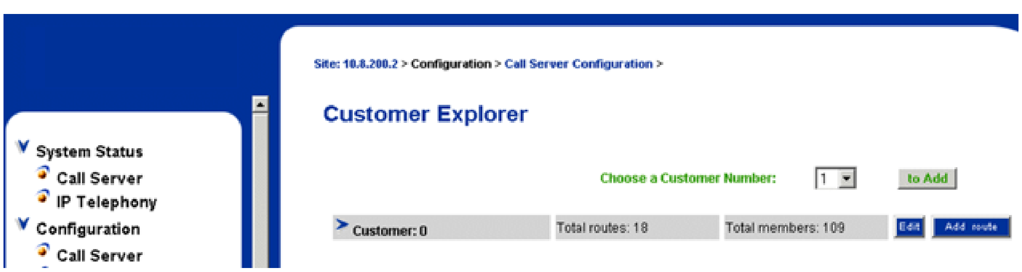
4. In the left panel, under Configuration select Call Server.
5. Select D-Channel from the list.
6. Create a D-Channel as shown in the following screen, where two PRI channels (3 and 4) exist, and channel 24 is a virtual D for IP routes.



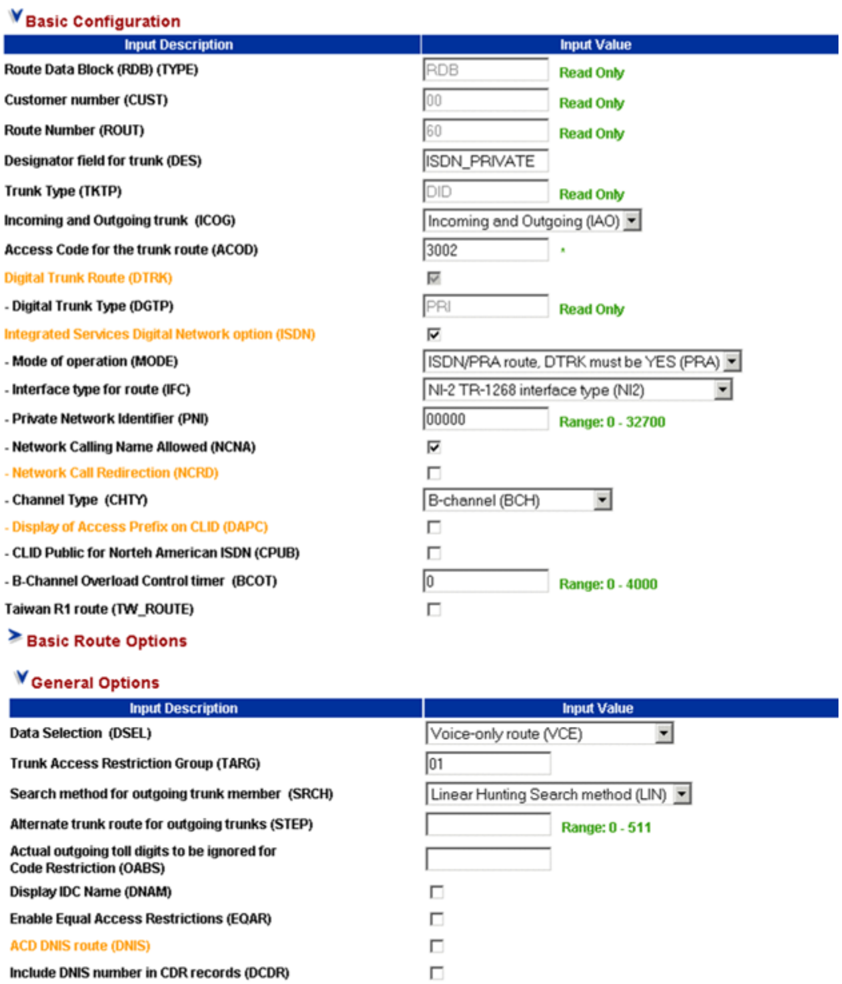
7. Click Edit next to any of the channels to specify the card (slot), protocol (NI2 - NET) towards the MPS and the 64kb baud rate. All these entries must match remote switch configuration for the ISDN DH.



8. Move through Configuration > Call Server > Customer Explorer. and create a route as shown, and specify trunks associated with the route.



In the following example, route 60 is DID and access code 3002. Select Voice only (VCE) and set TARG to 0, which allows full access with no dialing restrictions.



Basic Configuration

Input Description	Input Value	
Route Data Block (RDB) (TYPE)	RDB	Read Only
Customer number (CUST)	00	Read Only
Route Number (ROUT)	60	Read Only
Designator field for trunk (DES)	ISDN_PRIVATE	
Trunk Type (TKTP)	DID	Read Only
Incoming and Outgoing trunk (ICOG)	Incoming and Outgoing (IAO)	
Access Code for the trunk route (ACOD)	3002	*
Digital Trunk Route (DTRK)	☑	
- Digital Trunk Type (DGTP)	PRI	Read Only
Integrated Services Digital Network option (ISDN)	☑	
- Mode of operation (MODE)	ISDN/PRA route, DTRK must be YES (PRA)	
- Interface type for route (IFC)	NI-2 TR-1268 interface type (NI2)	
- Private Network Identifier (PNI)	00000	Range: 0 - 32700
- Network Calling Name Allowed (NCNA)	☑	
- Network Call Redirection (NCRD)	☐	
- Channel Type (CHTY)	B-channel (BCH)	
- Display of Access Prefix on CLID (DAPC)	☐	
- CLID Public for Norteh American ISDN (CPUB)	☐	
- B-Channel Overload Control timer (BCOT)	0	Range: 0 - 4000
Taiwan R1 route (TW_ROUTE)	☐	

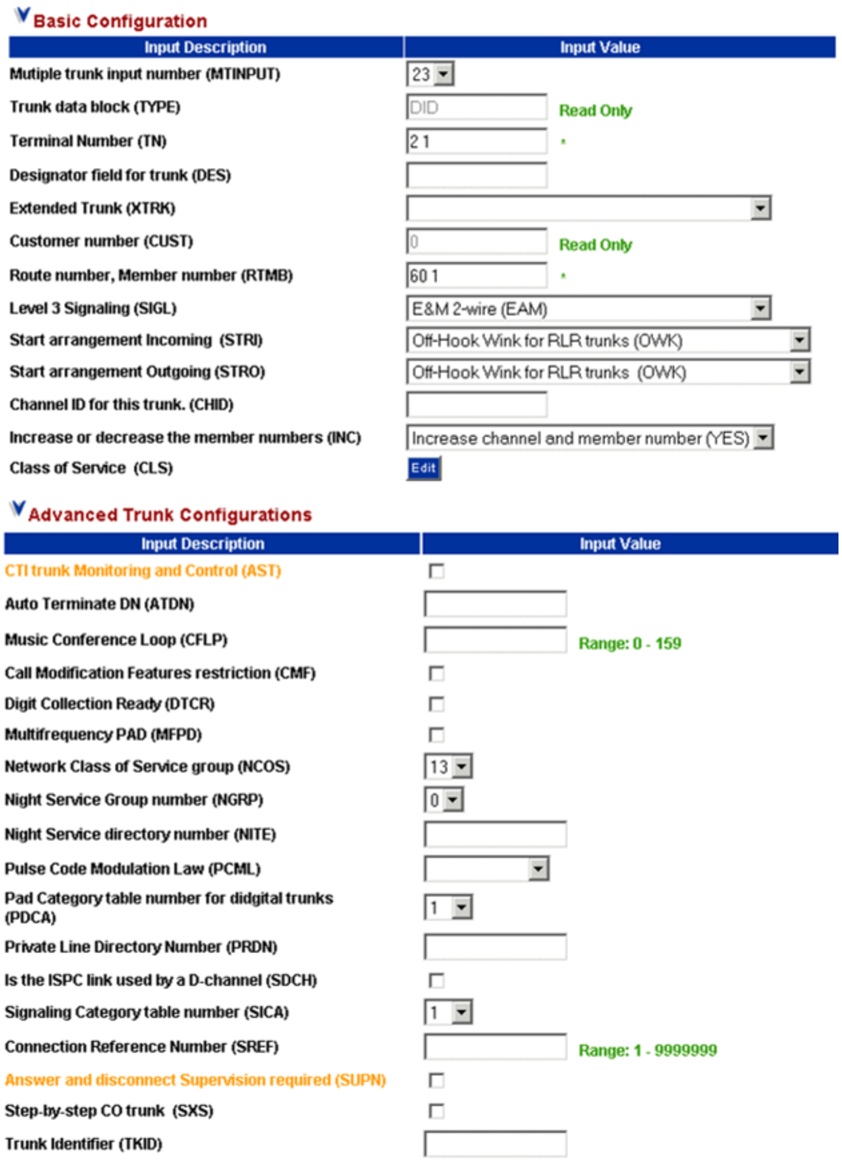
Basic Route Options

General Options

Input Description	Input Value	
Data Selection (DSEL)	Voice-only route (VCE)	
Trunk Access Restriction Group (TARG)	01	
Search method for outgoing trunk member (SRCH)	Linear Hunting Search method (LIN)	
Alternate trunk route for outgoing trunks (STEP)		Range: 0 - 511
Actual outgoing toll digits to be ignored for Code Restriction (OABS)		
Display IDC Name (DNAM)	☐	
Enable Equal Access Restrictions (EQAR)	☐	
ACD DNIS route (DNIS)	☐	
Include DNIS number in CDR records (DCDR)	☐	

9. On the Customer Explorer page, locate Route 60 and click Add Trunk. Create 23 trunk (channels), starting with slot 2 channel 1.

   For example, designate 21 from route 64 1. Ensure COS is high for outbound and low for inbound routes.



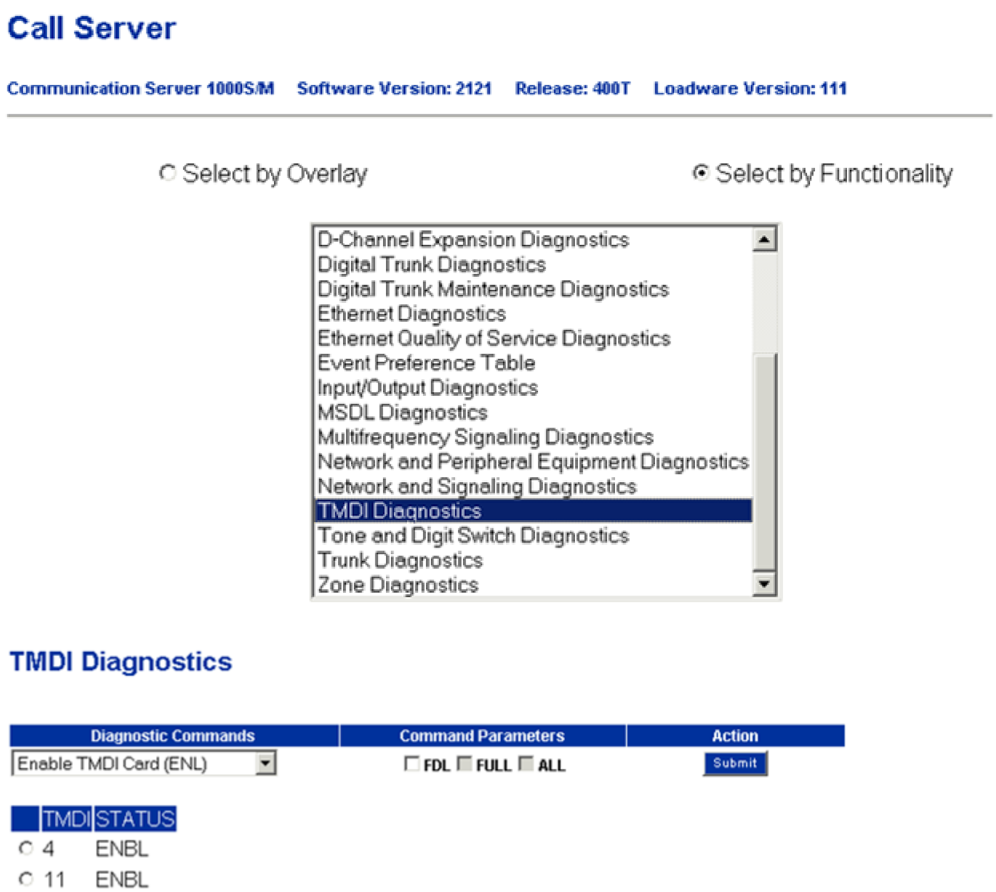
Basic Configuration

Input Description	Input Value
Mutiple trunk input number (MTINPUT)	23
Trunk data block (TYPE)	DID Read Only
Terminal Number (TN)	2 1 *
Designator field for trunk (DES)	
Extended Trunk (XTRK)	
Customer number (CUST)	0 Read Only
Route number, Member number (RTMB)	60 1 *
Level 3 Signaling (SIGL)	E&M 2-wire (EAM)
Start arrangement Incoming (STRI)	Off-Hook Wink for RLR trunks (OWK)
Start arrangement Outgoing (STRO)	Off-Hook Wink for RLR trunks (OWK)
Channel ID for this trunk. (CHID)	
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

Advanced Trunk Configurations

Input Description	Input Value
CTI trunk Monitoring and Control (AST)	☐
Auto Terminate DN (ATDN)	
Music Conference Loop (CFLP)	Range: 0 - 159
Call Modification Features restriction (CMF)	☐
Digit Collection Ready (DTCR)	☐
Multifrequency PAD (MFPD)	☐
Network Class of Service group (NCOS)	13
Night Service Group number (NGRP)	0
Night Service directory number (NITE)	
Pulse Code Modulation Law (PCML)	
Pad Category table number for didgital trunks (PDCA)	1
Private Line Directory Number (PRDN)	
Is the ISPC link used by a D-channel (SDCH)	☐
Signaling Category table number (SICA)	1
Connection Reference Number (SREF)	Range: 1 - 9999999
Answer and disconnect Supervision required (SUPN)	☐
Step-by-step CO trunk (SXS)	☐
Trunk Identifier (TKID)	

10. Move through System Status > Call Server in the left panel, and click Select by Functionality.
11. From the menu, select TDMI diagnostics. Enable the TMDI, which enables the D-Channel and brings up the span.
12. Confirm that the span and DCH are up and running.



ISDN Configuration Using Overlay Commands

Perform the following steps:

1. Disable the existing Span 4.

```
LOGI ADMIN1
PASS? <0000>
```

```
LD 96 .DIS TMDI 4 ALL
```

```
OK
.****
OVL000
```

2. Ensure Span 4 is disabled.

```
>LD 60 .STAT 4
```

```
PRI*  TRK  LOOP 4 - DSBL
FFMT/LCMT/YALMT:  ESF/B8Z/FDL
```

3. Check route number and trunk channels for loop 4. In the following example, route = 70, with 23-B channels.

```
>LD 21
```

```
REQ: LTM CUST 0
```

```
ROUT
ACOD
. . .
TYPE TLST
TKTP DID
```

```
ROUT 70
```

```
DES  ISDN_PUBLIC
TN  004 01 MBER  1  1_TO_23
TN  004 02 MBER  2  1_TO_23
TN  004 03 MBER  3  1_TO_23
TN  004 04 MBER  4  1_TO_23
TN  004 05 MBER  5  1_TO_23
TN  004 06 MBER  6  1_TO_23
TN  004 07 MBER  7  1_TO_23
TN  004 08 MBER  8  1_TO_23
TN  004 09 MBER  9  1_TO_23
TN  004 10 MBER  10  1_TO_23
TN  004 11 MBER  11  1_TO_23
TN  004 12 MBER  12  1_TO_23
TN  004 13 MBER  13  1_TO_23
TN  004 14 MBER  14  1_TO_23
TN  004 15 MBER  15  1_TO_23
TN  004 16 MBER  16  1_TO_23
TN  004 17 MBER  17  1_TO_23
TN  004 18 MBER  18  1_TO_23
TN  004 19 MBER  19  1_TO_23
TN  004 20 MBER  20  1_TO_23
TN  004 21 MBER  21  1_TO_23
TN  004 22 MBER  22  1_TO_23
TN  004 23 MBER  23  1_TO_23
```

4. Remove the 23 trunks associated with Route 70.

```
> LD 14
```

```
MEM AVAIL: (U/P): 2887986    USED U P: 187512 53845    TOT: 3129343
DISK RECS AVAIL: 447
ITG ISDN TRUNKS AVAIL:  2500    USED:     0    TOT:  2500
IP PEER H.323 TRUNKS AVAIL:     0    USED:    10
TOT:    10
AST     AVAIL:   100    USED:     0    TOT:   100
RAN CON AVAIL:    12    USED:     0    TOT:    12
MUS CON AVAIL:   100    USED:     0    TOT:   100
TNS     AVAIL:  2397    USED:   103    TOT:  2500
DATA PORTS AVAIL:  2500    USED:     0    TOT:  2500
TRADITIONAL TRUNKS AVAIL:  2454    USED:    46    TOT:  2500
```

```
REQ OUT 24 TYPE DID TN 4 1
```

```
OUT TRK    TN  004 01     RT  70     MB  1
MEM AVAIL: (U/P): 2888183    USED U P: 187370 53790    TOT: 3129343
DISK RECS AVAIL: 447
AST     AVAIL:   100    USED:     0    TOT:   100
ITG ISDN TRUNKS AVAIL:  2500    USED:     0    TOT:  2500
IP PEER H.323 TRUNKS AVAIL:     0    USED:    10    TOT:    10
RAN CON AVAIL:    12    USED:     0    TOT:    12
MUS CON AVAIL:   100    USED:     0    TOT:   100
TNS     AVAIL:  2398    USED:   102    TOT:  2500
DATA PORTS AVAIL:  2500    USED:     0    TOT:  2500
TRADITIONAL TRUNKS AVAIL:  2455    USED:    45    TOT:  2500
OUT TRK    TN  004 02     RT  70     MB  2
. . .
MEM AVAIL: (U/P): 2892320    USED U P: 184388 52635    TOT: 3129343
DISK RECS AVAIL: 449
AST     AVAIL:   100    USED:     0    TOT:   100
ITG ISDN TRUNKS AVAIL:  2500    USED:     0    TOT:  2500
IP PEER H.323 TRUNKS AVAIL:     0    USED:    10    TOT:    10
RAN CON AVAIL:    12    USED:     0    TOT:    12
MUS CON AVAIL:   100    USED:     0    TOT:   100
TNS     AVAIL:  2419    USED:    81    TOT:  2500
DATA PORTS AVAIL:  2500    USED:     0    TOT:  2500
TRADITIONAL TRUNKS AVAIL:  2476    USED:    24    TOT:  2500
OUT TRK    TN  004 23     RT  70     MB  23
>LD 21
PT1000
REQ: LTM
CUST 0
ROUT
ACOD ****
OVL000
```

5. Verify deletion of trunks, and that no trunk exists in Route 70.

```
>LD 21
```

```
REQ: LTM
CUST 0
ROUT 70
TYPE TLST
TKTP DID
ROUT 70
DES  ISDN_PUBLIC
```

6. Drop Route 70.

```
>LD 16 REQ OUT TYPE RDB CUST 0 DMOD ROUT 70
```

```
MEM AVAIL: (U/P): 2892950    USED U P: 184078 52315    TOT: 3129343
DISK RECS AVAIL: 450
RAN RTE AVAIL:  2500    USED:     0    TOT:  2500
REQ  ****
```

7. Verify route deletion.

```
>LD 21
PT1000
REQ: LTM
SCH0510
REQ: LTM
```

```
CUST 0
ROUT 70
SCH0858
ROUT ****
OVL000
>ERR SCH0858
SCH0858
Route number does not exist.
Severity:  Info
```

8. Drop the D-Channel 4 associated with this route.

```
>ld 17

REQ  CHG
TYPE CEQU
TDS
CONF
DLOP X4
DLOP
PRI2
DTI2
TIM000 21:51   7/3/2096 CPU 0
BUG845
BUG845 : 004FE201 0000010A 00080000 0036BFFF
BUG845 +   106C4A44 1083FBCC 1083F9CC 108F9872 108EDB48
BUG845 + 11510EA8 115111F0 114FFDDA 114FF6F4 109C91DA
BUG845 + 109C7FFA 109C78AC 109C5CBE 109C3172 10F00F08
BUG845 + 10F002D4 10EFFEA6 10EFFD88
MEM AVAIL: (U/P): 2895434    USED U P: 182208 51701    TOT: 3129343
DISK RECS AVAIL: 451
DCH  AVAIL:    79    USED:     1    TOT:    80
TMDI D-CHANNELS AVAIL:    99    USED:     1    TOT:   100
AML  AVAIL:    16    USED:     0    TOT:    16
REQ  ****
```

9. Verify that D-Channel 4 has been dropped or DLOP=4 does not exist.

   `>LD 22 REQ PRT TYPE CEQU`

```
CEQU
MPED 8D
SUPL  000  004  008  012
016  032  036  040
044  048  064  068
072 V096 V100 V104
V108 V112
XCT   000
CONF  029  030  031  062
094  095
DLOP  NUM DCH FRM TMDI LCMT YALM T1TE TRSH
PRI  02   24  ESF YES  B8S  FDL  0    00
MISP
REQ  ****
```

10. Create PRI loop 4 with essential parameters.

    `>LD 17 REQ CHG TYPE CEQU`

```


TDS
CONF
DLOP 4
```

```
MODE PRI
TMDI YES
LCMT
YALM FDL
T1TE
TRSH
```

11. Verify that D-Channel 4 has been added on.

```
>LD 22
REQ  PRT
TYPE CEQU
CEQU
MPED 8D
SUPL  000  004  008  012
016  032  036  040
044  048  064  068
072 V096 V100 V104
V108 V112
XCT   000
CONF  029  030  031  062
094  095
DLOP  NUM DCH FRM TMDI LCMT YALM T1TE TRSH
PRI  02   24  ESF YES  B8S  FDL  0    00
04   24  ESF YES  B8S  FDL  0    00
MISP
REQ  ****
```

12. Create new PRI DCH 4 for CARDNO=4, PROTO=NI2, SIDE=NETWORK 64KC.

```
>LD 17
CFN000
MEM AVAIL: (U/P): 2893705    USED U P: 183715 51923    TOT: 3129343
DISK RECS AVAIL: 450
DCH  AVAIL:    79    USED:     1    TOT:    80
TMDI D-CHANNELS AVAIL:    99    USED:     1    TOT:   100
AML  AVAIL:    16    USED:     0    TOT:    16
REQ  CHG
TYPE ADAN
ADAN NEW DCH 4
CTYP TMDI
```

CDNO 4

```
PORT 1
DES  1_TO_23
```

USR PRI IFC NI2

```
CO TYPE
ISDN_MCNT
CLID
DCHL 4
PRI
OTBF
```

DRAT 64KC SIDE NET

```
CNEG
RLS
RCAP
```

```
MBGA
TIMR
LAPD
BSRV
ADAN DATA SAVED
```

13. Verify that DCH is created.

    `>LD 22 REQ PRT TYPE ADAN DCH 4`

```
ADAN     DCH 4
CTYP TMDI
CARD 04
PORT 1
DES  1_TO_23
USR  PRI
DCHL 4
OTBF 32
PARM RS232  DTE
DRAT 64KC
CLOK EXT
IFC  NI2
ISDN_MCNT 300
CLID OPT0
CO_TYPE  STD
SIDE NET
CNEG 1
RLS  ID  **
RCAP COLP
MBGA NO
OVLR NO
OVLS NO
T310 120
T200 3
T203 10
N200 3
N201 260
K    7
BSRV  NO
```

14. Create new Route 70.

    `>LD 16 REQ NEW TYPE RDB`

```
CUST 0
DMOD
```

`ROUT 70`

```
DES  ISDN_PUBLIC
```

`TKTP DID`

```
SAT
RCLS
```

`DTRK YES DGTP PRI ISDN YES MODE PRA IFC NI2`

```
CBCR
PNI
NCNA
```

```
NCRD
CHTY
NCOS 7
CPFXS
CPUB
DAPC
BCOT
INTC
```

DSEL VCE! VOICE ONLY !

```
PTYP
AUTO
DNIS
DCDR
IANI
ICOG IAO
RANX
SRCH
TRMB
STEP
```

ACOD 3696

```
CLEN
TCPP
PII
```

15. Verify new Route 70.

>LD 21 REQ: PRT TYPE: RDB CUST 0 ROUT 70

```
TYPE RDB
CUST 00
DMOD
ROUT 70
DES  ISDN_PUBLIC
TKTP DID
NPID_TBL_NUM   0
SAT  NO
RCLS EXT
VTRK NO
DTRK YES
BRIP NO
DGTP PRI
ISDN YES
MODE PRA
IFC  NI2
CBCR NO
NCOS 7
SBN  NO
PNI  00000
NCNA YES
NCRD NO
CHTY BCH
CPFXS YES
CPUB OFF
DAPC NO
BCOT 0
INTC NO
DSEL VCE
PTYP PRI
```

```
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
RANX NO
SRCH LIN
TRMB YES
STEP
ACOD 3696
TCPP NO
PII NO
TARG
CLEN 1
BILN NO
OABS
INST
ICIS YES
TIMR ICF  512
OGF  512
EOD  13952
NRD  10112
DDL  70
ODT  4096
RGV  640
FLH  510
GRD  896
SFB  3
NBS  2048
NBL  4096
DRNG NO
CDR  NO
MUS  NO
EQAR NO
FRL  0 0
FRL  1 0
FRL  2 0
FRL  3 0
FRL  4 0
FRL  5 0
FRL  6 0
FRL  7 0
TTBL 0
ATAN NO
PLEV 2
MCTS NO
ALRM NO
ART  0
SGRP 0
AACR NO
```

16. Create 23 trunks for Route 70.

    `>LD 14 REQ NEW 23`

```
TYPE DID
TN  4 1
DES  1_TO_23
PDCA
PCML
CUST 0
NCOS 7
RTMB 70 1
B-CHANNEL SIGNALING
INC
```

```
NITE
AST
CLS  UNR
TKID
NEW TRK    TN  004 01     RT  70     MB  1
MEM AVAIL: (U/P): 2892277    USED U P: 184654 52412    TOT: 3129343
DISK RECS AVAIL: 449
AST     AVAIL:   100    USED:     0    TOT:   100
ITG ISDN TRUNKS AVAIL:  2500    USED:     0    TOT:  2500
IP PEER H.323 TRUNKS AVAIL:     0    USED:    10    TOT:    10
RAN CON AVAIL:    12    USED:     0    TOT:    12
MUS CON AVAIL:   100    USED:     0    TOT:   100
TNS     AVAIL:  2419    USED:    81    TOT:  2500
DATA PORTS AVAIL:  2500    USED:     0    TOT:  2500
TRADITIONAL TRUNKS AVAIL:  2476    USED:    24    TOT:  2500
NEW TRK    TN  004 02     RT  70     MB  2
. . .
MEM AVAIL: (U/P): 2887930    USED U P: 187636 53777    TOT: 3129343
DISK RECS AVAIL: 447
AST     AVAIL:   100    USED:     0    TOT:   100
ITG ISDN TRUNKS AVAIL:  2500    USED:     0    TOT:  2500
IP PEER H.323 TRUNKS AVAIL:     0    USED:    10    TOT:    10
RAN CON AVAIL:    12    USED:     0    TOT:    12
MUS CON AVAIL:   100    USED:     0    TOT:   100
TNS     AVAIL:  2398    USED:   102    TOT:  2500
DATA PORTS AVAIL:  2500    USED:     0    TOT:  2500
TRADITIONAL TRUNKS AVAIL:  2455    USED:    45    TOT:  2500
NEW TRK    TN  004 23     RT  70     MB  23
```

17. Verify trunk creation.

    >LD 11

    REQ: PRT TYPE: TNB TN 4 1

```
DATE
PAGE
DES
DES  1_TO_23
TN   004 01
TYPE DID
CDEN SD
CUST 0
TRK  PRI
PDCA 1
PCML MU
NCOS 7
RTMB 70 1
B-CHANNEL SIGNALING
NITE
STRI/STRO OWK OWK
AST  NO
IAPG 0
CLS  UNR DIP WTA LPR APN THFD HKD
P10 VNL
TKID
DATE  7 MAR 2096
NACT
```

    REQ: PRT TYPE: DID TN 4 23

```
DATE
```

```
PAGE
DES  1_TO_23
TN   004 23
TYPE DID
CDEN SD
CUST 0
TRK  PRI
PDCA 1
PCML MU
NCOS 7
RTMB 70 23
B-CHANNEL SIGNALING
NITE
STRI/STRO OWK OWK
AST  NO
IAPG 0
CLS  UNR DIP WTA LPR APN THFD HKD
P10 VNL
TKID
DATE  7 MAR 2096
NACT ****
```

18. Enable trunk and DCH.

>LD 60 .STAT 4

```
PRI*  TRK  LOOP 4 - DSBL
FFMT/LCMT/YALMT:  ESF/B8Z/FDL
.****
```

>LD 96 .STAT DCH

```
DCH 003 : OPER  EST   ACTV  AUTO         DES : ISDN_PRIVATE
DCH 004 : DSBL  RST         AUTO         DES : 1_TO_23
DCH 024 : OPER  EST   ACTV  AUTO         DES : IPTRUNK_DCH
.****
```

.ENL TMDI 4 ALL

```
DCH: 4  EST CONFIRM  TIME: 22:09:26  7/03/2096
DCH: 4  RLS CONFIRM  TIME: 22:09:26  7/03/2096
DCH1011 4 2
DCH: 4  EST CONFIRM  TIME: 22:09:26  7/03/2096
```

.STAT DCH

```
DCH 003 : OPER  EST   ACTV  AUTO         DES : ISDN_PRIVATE
```

DCH 004 : OPER EST ACTV AUTO DES : 1_TO_23

```
DCH 024 : OPER  EST   ACTV  AUTO         DES : IPTRUNK_DCH
.****
```

>LD 60 .STAT 4

```
PRI*  TRK  LOOP 4 - ENBL
FFMT/LCMT/YALMT:  ESF/B8Z/FDL
SERVICE RESTORE: YES
YEL ALM PROCESS: YES
```

```
ALARM STATUS : NO ALARM
CH 01 - IDLE DID  VCE  *   CH 02 - IDLE DID  VCE  *
CH 03 - IDLE DID  VCE  *   CH 04 - IDLE DID  VCE  *
CH 05 - IDLE DID  VCE  *   CH 06 - IDLE DID  VCE  *
CH 07 - IDLE DID  VCE  *   CH 08 - IDLE DID  VCE  *
CH 09 - IDLE DID  VCE  *   CH 10 - IDLE DID  VCE  *
CH 11 - IDLE DID  VCE  *   CH 12 - IDLE DID  VCE  *
CH 13 - IDLE DID  VCE  *   CH 14 - IDLE DID  VCE  *
CH 15 - IDLE DID  VCE  *   CH 16 - IDLE DID  VCE  *
CH 17 - IDLE DID  VCE  *   CH 18 - IDLE DID  VCE  *
CH 19 - IDLE DID  VCE  *   CH 20 - IDLE DID  VCE  *
CH 21 - IDLE DID  VCE  *   CH 22 - IDLE DID  VCE  *
CH 23 - IDLE DID  VCE  *   CH 24 - DCH 4
```

Chapter 6: SIP Configuration

This chapter covers:

Configuring the Telephony Protocol

Once the MPS is set up, use the MPS Configurator to specify the telephone protocol the system uses. You must do this before you begin any processing. Once you start the MPS Configurator, it automatically detects:

Example

- the number of MPS units attached to this application processor
- the chassis number assigned to the MPS
- the type of cards the processor is using
- the Back Plane Slot (BPS) number of each slot
- the number of Digital Signal Processors (DSPs)

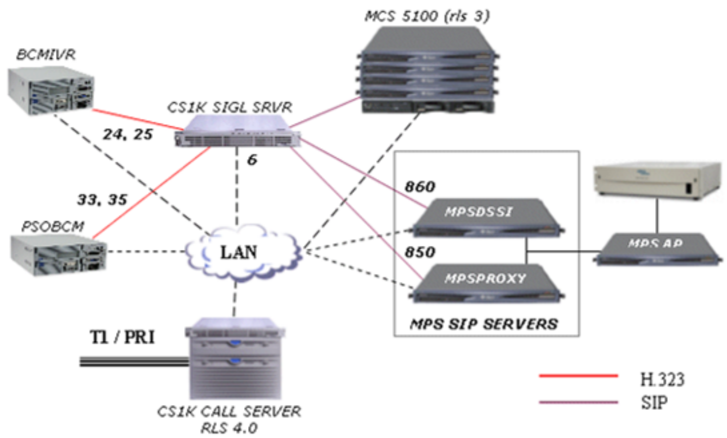
The MPS Configurator can later be used to identify and manage the Multimedia Format Files (MMFs) to load at boot time.

Note:

For more information on configuring the telephony protocol, refer to the Installing MPS Software on the Windows Platform Manual.

MPS-M1 SIP Configuration

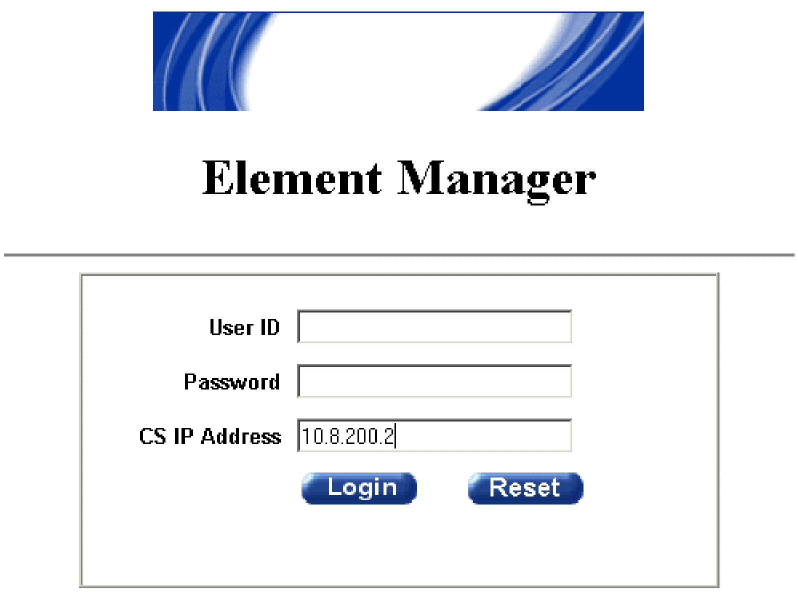
This procedure details the setup of SIP trunking between the CS 1000 Release 4 Signaling server and an MPS SIP server. The network diagram below illustrates a user calling into the CS 1000 Call Server through a T1/PR1.



Note:
Dial the 850X or 860X series of numbers to get SIP access to specific applications running on MPS lines.

Element Manager

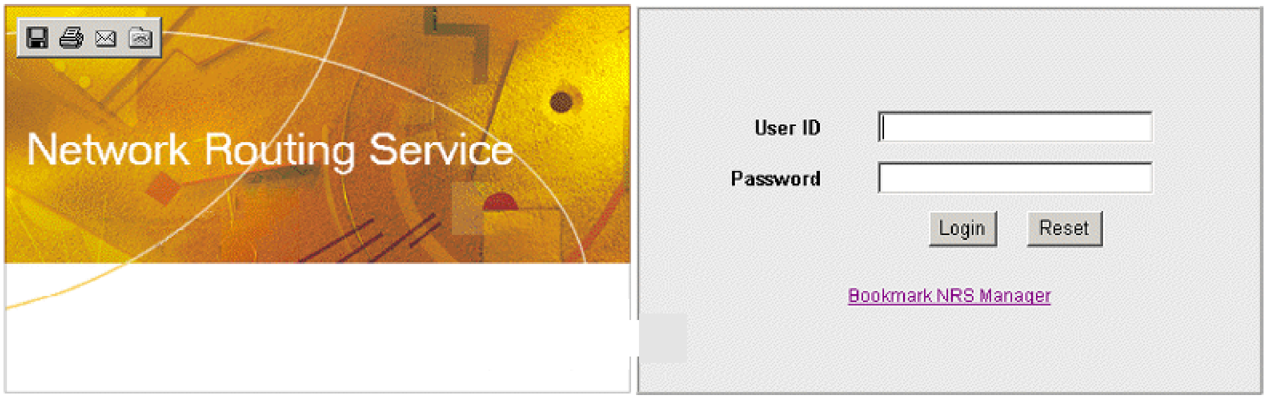
Use the Element Manager (http://<NODE IP>) to configure CS 1000 features relevant to call/ signaling servers, media gateway cards and CS 1000 numbering plans.



Network Routing Service

Use the Network Routing Service for VoIP numbering plans and SIP/H.323 GW/routing. Access the Network Routing Service through Element Manager > Network Numbering Plan > NRS.

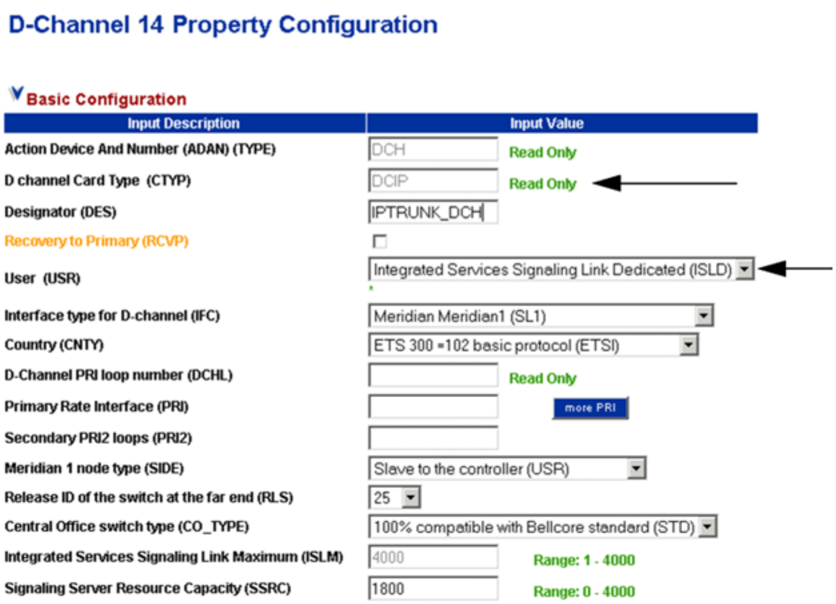
In the case of co-resident SIP GW and NRS, SIP GW uses the node IP address and NRS uses the TLAN IP address.



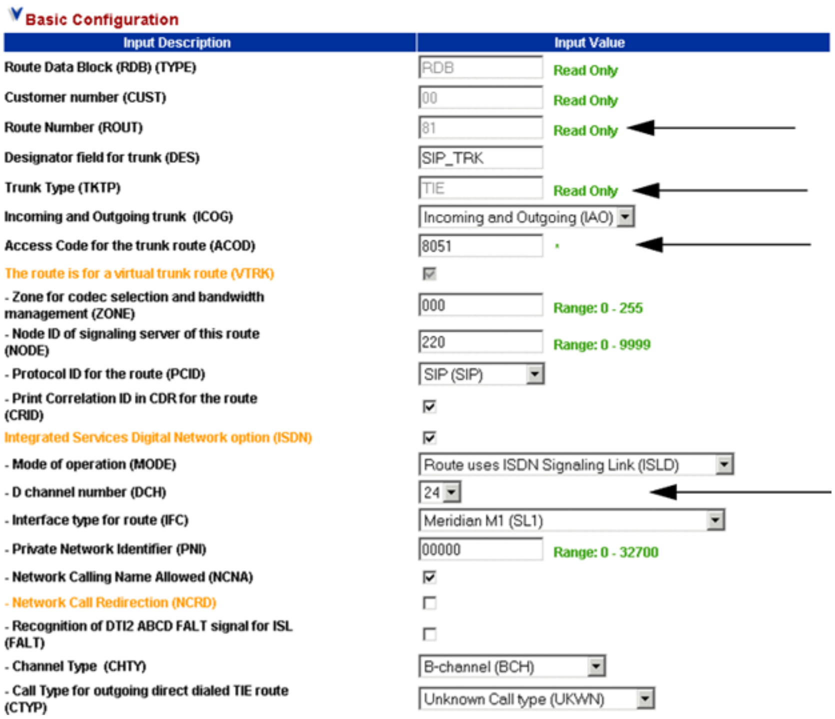
CS 1000 SIP Trunk Configuration

Follow the steps below for SIP trunk configuration:

1. At the Element Manager interface, go to Configuration > Call Server Configuration > D-Channel.
2. Create an IP DCH.



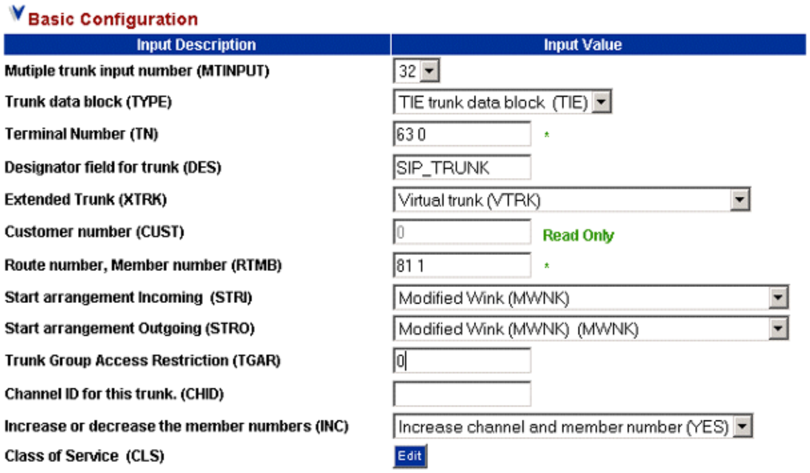
3. Move through Configuration > Call Server Configuration > Customer Explorer to create a SIP route associated with 32 SIP trunks, using the DCH 24 created earlier.



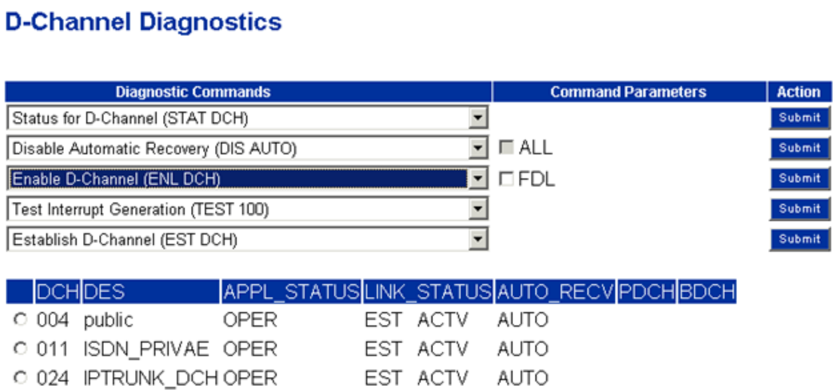
4. Create 32 trunks as shown, and check for available virtual cards using LD 97.

```
ld 97
REQ  prt
TYPE supl
SUPL
SUPL  SUPT SLOT XPEC0   XPEC1
. . .
096  VIRTUAL CARDS 61 - 64  81 - 84
100  VIRTUAL CARDS 65 - 68  85 - 88
104  VIRTUAL CARDS 69 - 72  89 - 92
108  VIRTUAL CARDS 73 - 76  93 - 96
112  VIRTUAL CARDS 77 - 80  97 - 99
```

Cards 61 and 62 were used for H.323 trunks, while 63 was free. The MTINPUT command indicates that 32 trunks are created from 63 0, RTMB 81 1.



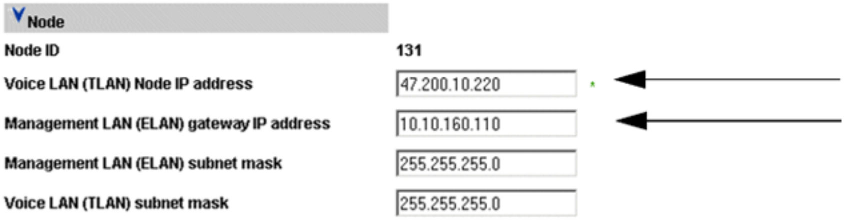
5. Go to System Status > Call Server > D-Channel Diagnostics and enable DCH. Under the Diagnostic Commands list, select Enable D-Channel (ENL DCH) from the third list.



CS 1000 SIP Server Configuration

Follow the steps below to configure the SIP server:

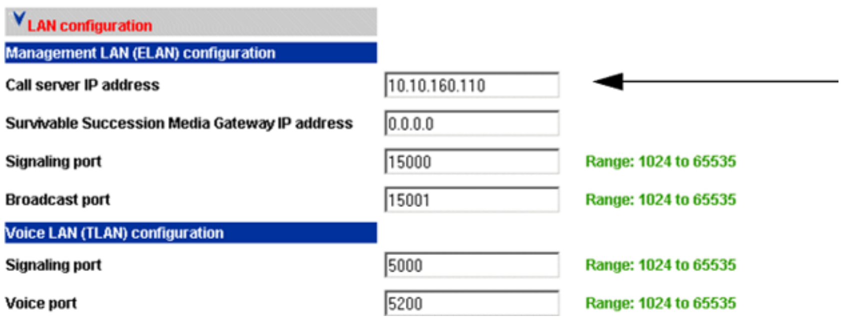
1. At the Element Manager interface, move through Configuration > IP Telephony Configuration > Node Summary and add a node, for example, 220.



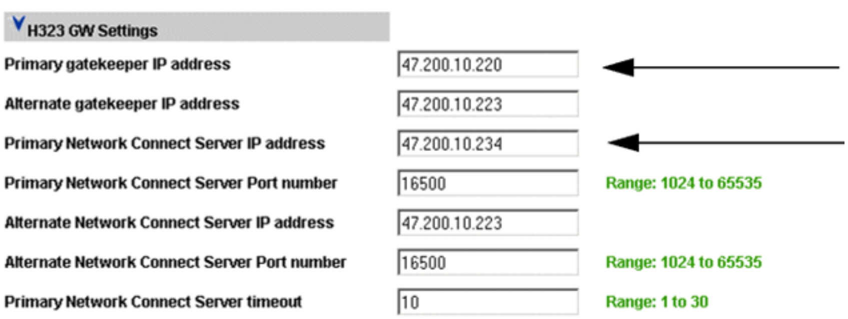
Note:

In the node configuration, the node IP is virtual, and can be pinged through the TLAN interface.

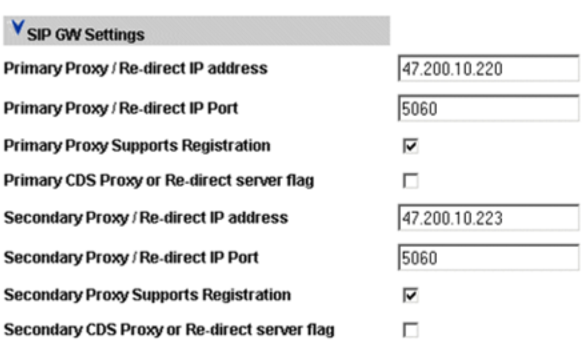
2. In the LAN Configuration field, enter the IP of the call server.



3. In the H.323 Gateway settings, enter the TLAN IP of the signaling server. This is primarily used for the H.323 setup.



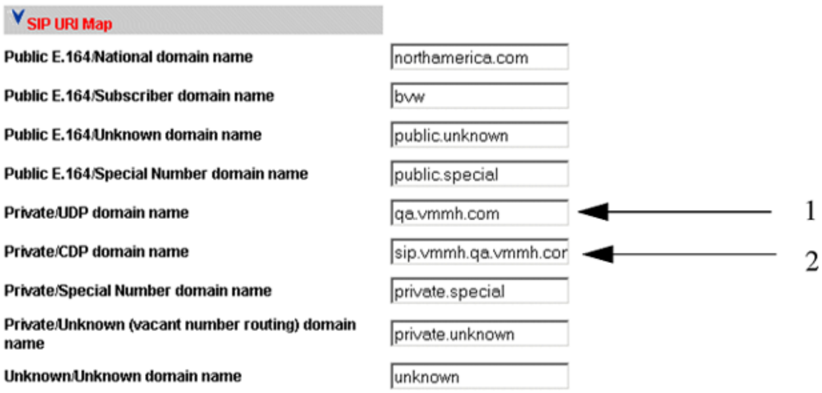
4. In the SIP GW settings field, enter the TLAN IP of the signaling server. The standard SIP UDP port is 5060.



5. In the SIP URI Map, the key entries are the UDP and CDP domain names, as set in the NRS. Examples include:

   • UDP Domain = cs1ksipudp.com (#1 in the following figure)

   • CDP Domain = CSE_NEW.northamerica.com (#2 in the following figure)



6. Move to the Signaling Server section and do the following:

   a. Enter the ELAN IP and MAC address of the signaling server to the H. 323/SIP server (#1 in the following figure).

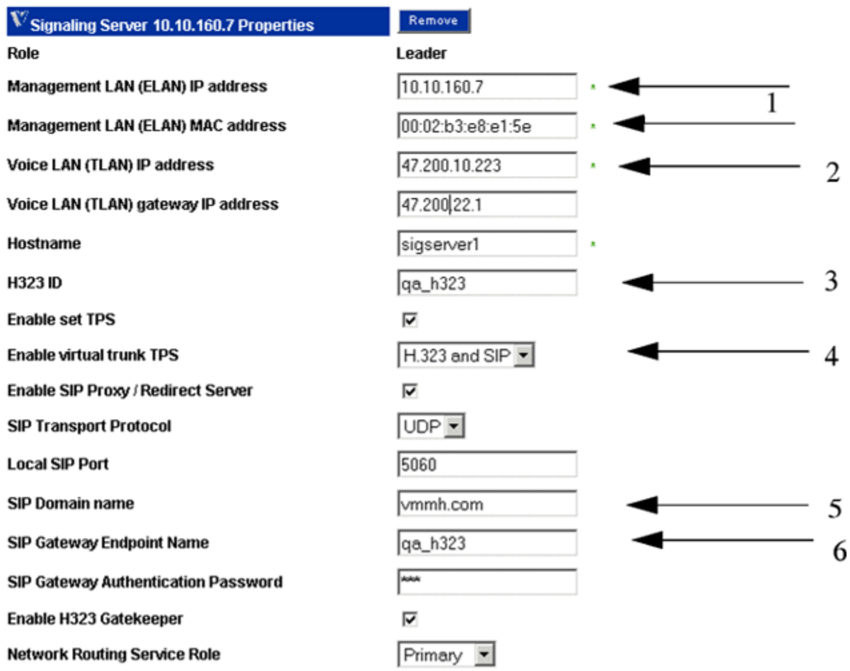
   b. Enter the TLAN physical interface (#2 in the following figure).

   c. Assign H.323 ID as the node ID (#3 in the following figure).

   d. Set TPS to use both H.323 and SIP on the same node (#4 in the following figure).

   e. The SIP Domain name is obtained from the NRS (#5 in the following figure).

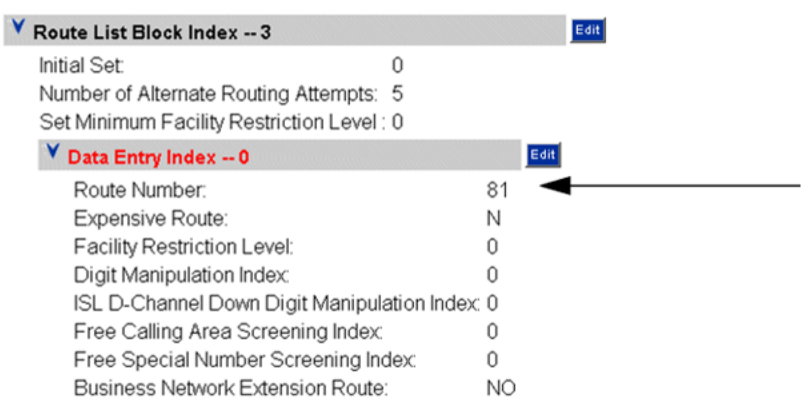
   f. The SIP GW Endpoint name is set in the NRS later (#6 in the following figure).



CS 1000 Numbering Plan Configuration

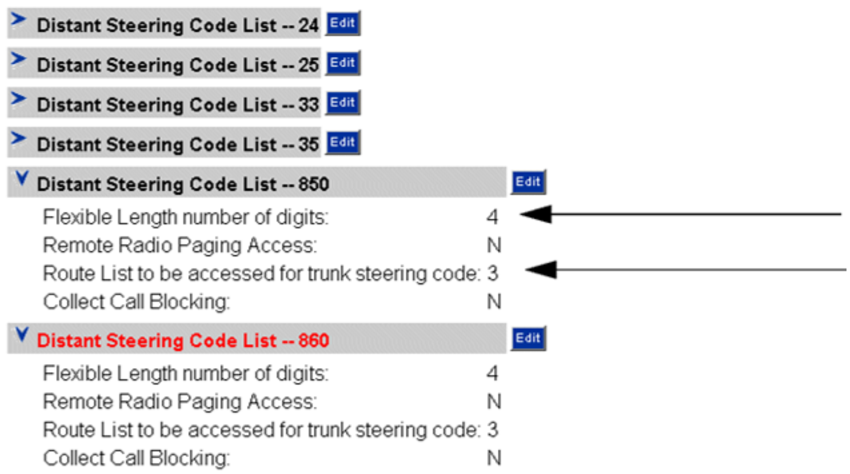
Perform the following steps to configure the numbering plan:

1. Move through Network Numbering Plan > Call Server > Route List Block (RLB). Type an entry (in this example, RLI = 3) that points to the SIP route created earlier.



2. Move through Network Numbering Plan > Call Server > CDP > Distant Steering Code (DSC) and enter respective SIP endpoints. For example, DSC = 860 is used for mpsdssi SIP endpoint and DSC = 850 is used for mpsproxy SIP endpoint.

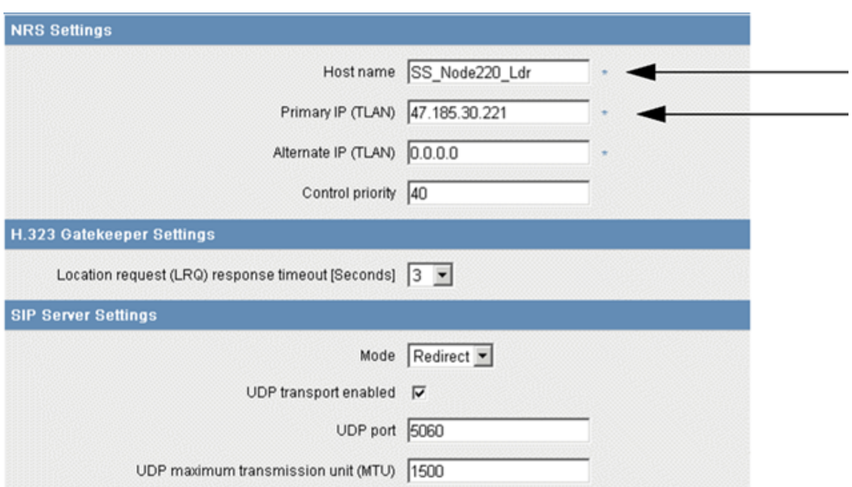
   Both use SIP route 81 (RLI = 3) and are 4 DN in length. 860X reaches mpsdssi and 850X reaches mpsproxy SIP endpoints.



CS 1000 NRS Configuration

Perform the following steps to configure the NRS:

1. From the Element Manager, move through Network Numbering Plan > Network Routing Service to access the NRS.
2. Log onto the NRS Settings interface. Move through Home > NRS Server Settings.
3. Ensure that the Host name matches the SIP GW Endpoint name set earlier and the TLAN IP matches the interface IP.



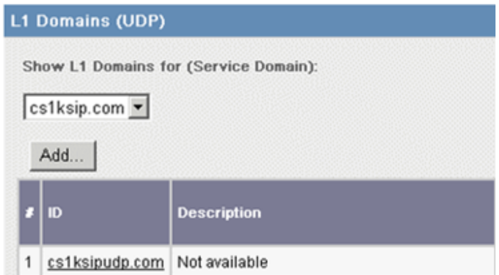
4. Move through Configuration > Service Domains > Standby DB View, and set the Domain name, which must match settings in the Element Manager.

Service Domains

Add...

#	ID	Description
1	cs1ksip.com	Not available

5. Move through Configuration > L1 Domains (UDP) and set name to match the Element Manager configuration.



L1 Domains (UDP)

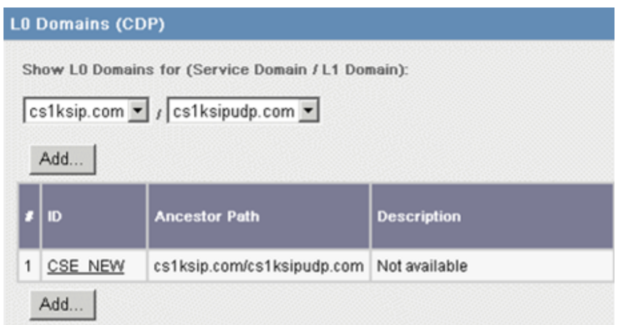
Show L1 Domains for (Service Domain):

cs1ksip.com

Add...

#	ID	Description
1	cs1ksipudp.com	Not available

6. Move through Configuration > L0 Domains (CDP) and set name. Setting the name plays a role in setting the CDP domain.



L0 Domains (CDP)

Show L0 Domains for (Service Domain / L1 Domain):

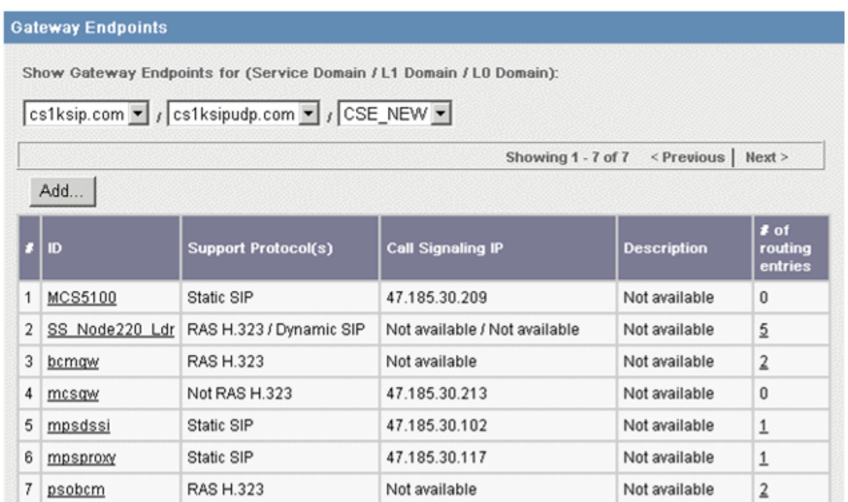
cs1ksip.com / cs1ksipudp.com

Add...

#	ID	Ancestor Path	Description
1	CSE_NEW	cs1ksip.com/cs1ksipudp.com	Not available

Add...

7. Move through Configuration > Gateway Endpoints and set both H.323 and SIP endpoints, as shown in the following diagram.



Gateway Endpoints

Show Gateway Endpoints for (Service Domain / L1 Domain / L0 Domain):

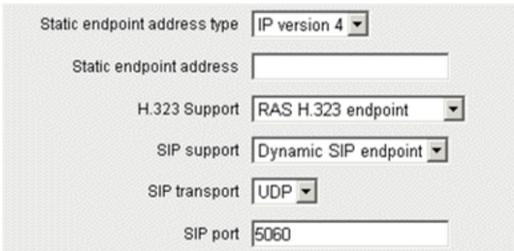
cs1ksip.com / cs1ksipudp.com / CSE_NEW

Showing 1 - 7 of 7 < Previous | Next >

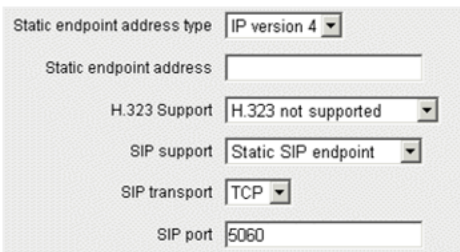
Add...

#	ID	Support Protocol(s)	Call Signaling IP	Description	# of routing entries
1	MCS5100	Static SIP	47.185.30.209	Not available	0
2	SS_Node220_Ldr	RAS H.323 / Dynamic SIP	Not available / Not available	Not available	5
3	bcmgw	RAS H.323	Not available	Not available	2
4	mcsgw	Not RAS H.323	47.185.30.213	Not available	0
5	mpsdssi	Static SIP	47.185.30.102	Not available	1
6	mpsproxy	Static SIP	47.185.30.117	Not available	1
7	psobcm	RAS H.323	Not available	Not available	2

8. Click on endpoint SS_Node220_Ldr. This is the CS 1000 endpoint that registers with the CS 1000 H.323 gatekeeper and points to itself. Because it supports SIP, set this parameter to Dynamic SIP.

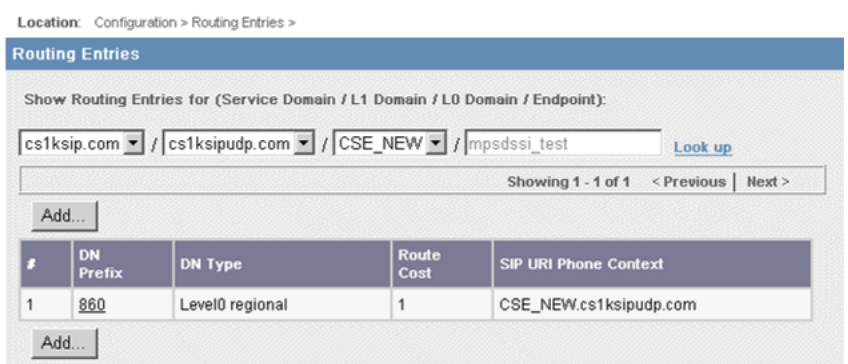


9. Set MPS endpoints, such as mpsdssi (MPS SIP DSSI mode) and mpsproxy (MPS SIP proxy mode) with a static IP entry. Enter information on the IP address of the SIP server.



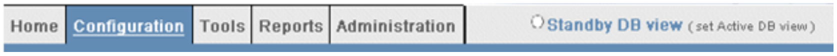
10. Move through Configuration > Routing Entries and enter the DN prefix for each endpoint.

    For example, the DN prefix for mpsdssi is 860. On the CS 1000, the 860 is later set as a four-digit DN. For example, if 860x is dialed, the caller is directed to the mpsdssi SIP server, where a corresponding application carries the process forward.

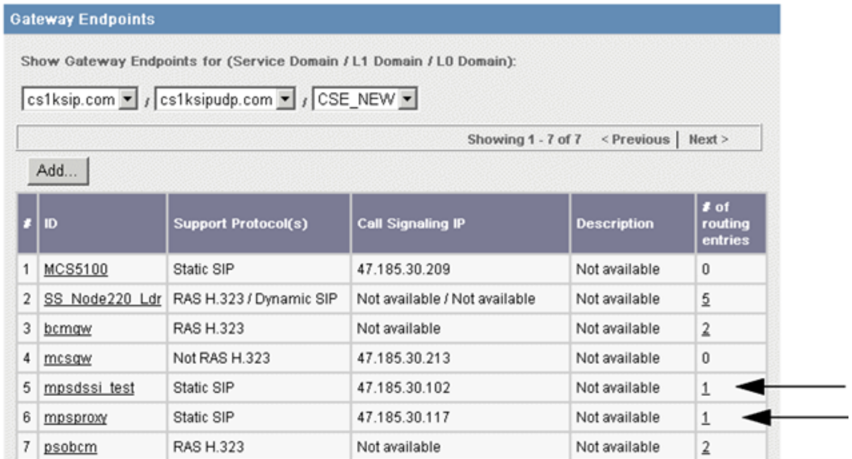


11. Once you select Service, UDP and CDP domains, as well as the G/W endpoints and the routing entries, select Tools > Database Actions and click the Cut Over and Commit button.

    This process transfers standby DB data entered earlier to the active database. Confirm this by checking the Active Database view in the Configuration tab.



12. Ensure that all endpoints, especially the CS 1000 endpoint, are registered.



MPS SIP Server Configuration

Perform the following steps to configure the SIP server:

1. Ensure that the NETWORK_HOSTS file points to the CS 1000 SIP proxy in /opt/ccss/etc/sip.conf, using the service domain name cs1ksip.com defined earlier.

   The primary host name points to the IP assigned to the node ID.

```
NETWORK_HOSTS = {
!-----------      ---   ------------   -------    -------    --   --   -----      --------
!NAME             TYPE  SIP DOMAIN NAME PRIMARY   SECONDARY  PORT SSA  TIMER      REGISTRAR
!                                       HOST NAME HOST NAME       PORT PROFILE    FLAG
!-----------      ---   ------------   -------    -------    --   --   -----      --------
CS1K              UDP   cs1ksip.com    cs1ksip    -          5060 -    SIP_TMR1   true
}
```

2. In the DSSI mode, ensure that the CC_LINEMAP points to TMS spans. In the following example, the CC_LINEMAP points to two TPM-200 spans of 30 channels each.

```
CC_HOSTS = {
!HOST                   HOST
!ID                       PROFILE NAME
!------------           ------------------
!1                      tms1
!
CC_LINEMAP
!HOST                   SPAN        LINE RANGE
!ID                     ID          from:to
!------------           ------------------
```

```
1                    1          1:30
1                    2          1:30
{
```

3. Ensure that the DTCMAP points to sipdssi_proto.cfg for spans 1 and 2.

```
[DTCMAP]
;----------------------------------------------------------------
;    TMS  PLI  Span svc_type  MpsNum Outline  Pool/   Protocol
;    Num  Slot                                class   Pkg
;----------------------------------------------------------------
-
LOAD 1        2    1    -         -        -        -
sipdssi_proto.cfg
LOAD 1        2    1    -         -        -        -
sipdssi_proto.cfg
```

Because the CS 1000 has no registration service, no service name pertaining to CS 1000 is entered in the REGISTRATION_SERVICE section.

4. Enter CS 1000 DNs corresponding to line names in the REG_LINE_CONFIG file.

   The following example indicates that all 8601 calls on CS 1000 hit line 1 and all 8602 calls hit line 2. By default, all other calls are directed to lines 3–30 because the MPSDSSI GW endpoint name is configured with routing entry 860. Hence, all 860x calls are sent to MPSDSSI.

```
REG_LINE_CONFIG = {
!---------     ------------      -------     ----------
!SERVICE       GROUP   HOST      SPAN       LINE
!NAME                  ID        ID         RANGE
!-----------------------------------------------------
8601                   1         1           1
8602                   1         1           2
default                1         1           3-30
```

5. In the proxy mode, ensure that the CC_LINEMAP points to the MPS component ID (2) for mps2 and the lines that point to ADMIN lines (240-270) as real or RTP lines (1-60 for TPM200 or 1-240 for TPM800) are placed in a pool and used by the application as and when needed.

```
CC_HOSTS = {
!HOST             HOST
!ID               PROFILE NAME
!----------       ------------------
!1                ase1
!
CC_LINEMAP
!----------       ------------------
!HOST             COMP        LINE RANGE
!ID               ID          from:to
!----------       ------------------
1               2           240:270
```

Because CS 1000 has no registration service, no service name pertaining to CS 1000 is entered in the REGISTRATION_SERVICE section.

6. Enter CS 1000 DNs corresponding to line names in the REG_LINE_CONFIG file.

   In the following example, 8501 calls on the CS 1000 hit lines 240–250 and 8502 calls hit lines 260–270 on the MPS. This is because the MPSPROXY GW endpoint name is configured with routing entry 850, which means that 850 calls are sent to the MPSPROXY.

```
REG_LINE_CONFIG = {
!----------     ------------       -----      ----
!SERVICE        GROUP   HOST        COMP       LINE
!NAME                   ID          ID         RANGE
!----------------------------------------------------
8501                   1          2         240-250
8502                    1          2         260-270
```

7. Ensure that the DTCMAP points to sipproxy_proto.cfg for spans 1 and 2.

```
[DTCMAP]
;------------------------------------------------------------
---
;    TMS  PLI  Span svc_type  MpsNum Outline  Pool/  Protocol
;    Num  Slot Num                            class  Pkg
;------------------------------------------------------------
----
LOAD 1        2    1    -         -        -        -
sipproxy_proto.cfg
LOAD 1        2    2    -         -       -       -
sipproxy_proto.cfg
```

8. Ensure that the SIP proxy is running as a service, and the RTP line pool is defined in /opt/vps/common/etc/pmgr.cfg. The pool name rtpline.2 is used in the application.

```
vsh#mps.2,vos/mpsap3 {1} > srp -status
...
mpsap3   root   18390 -  RUNNING   Feb 01 15:24:29   sipproxy
```

   RTP LINE POOL ENTRY defpool rtpline.2 cfgrsrc rtpline.2,phone.1-60.mps.*

Calling into the MPS SIP Application

The SIP GW uses the signaling server node IP address and NRS uses TLAN IP address.

To use the MPS SIP client, register the client to the MPS proxy and use the URI CS1K_Endpoint_DN@Node_IP.

For example, 8501 is the DN for MPSPROXY SIP endpoint. Because the SIP GW is linked to the Node IP, dialing 8501@NodeIP impacts MPS applications registered to service name 8501.

IP phones, such as the i2050 and i2004, can also be used on CS 1000 to work with SIP. IP phones are controlled by Call Server and use UNIStim protocol.

1. Enter the node IP address and select CS 1000 for port and server type.
2. When connected, dial 8501 to get to the MPS application.

MPS-CS 2000 SIP Configuration

The following sections details the setup of SIP and CTI for interoperability between the CS 2000 signaling server and an MPS SIP server.

SIP Trunking Configuration

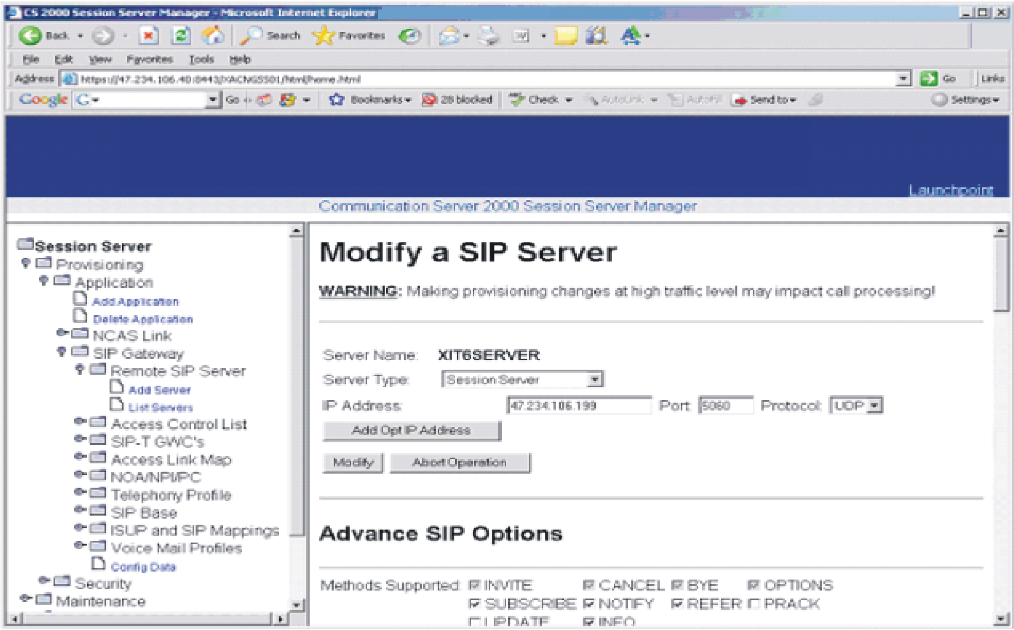
The following sections detail the setup of SIP trunking between the CS 2000 and a MPS SIP server.

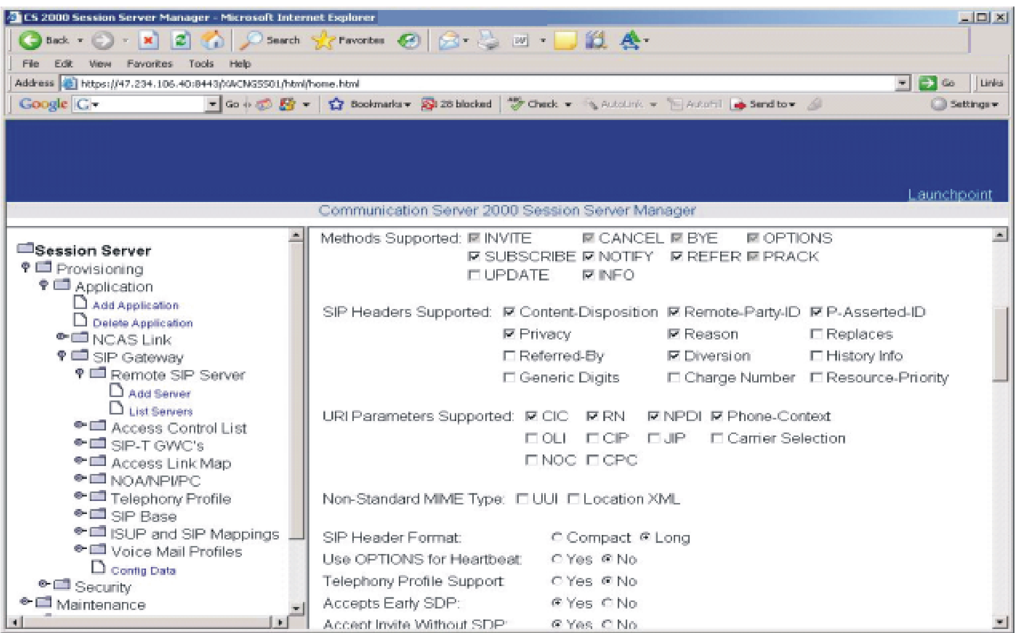
Configuration on CS 2000/CS 2100

1. Access Session Server GUI from Integrated Element Management System. Right click Session Server and click “Launch Session Server”.
2. Following screenshots describes Remote SIP Server configuration. On the Session Server GUI interface, Click on Provisioning □ Application □ SIP Gateway □ Remote SIP Server □ List Severs □ Details.

   Server Name: Valid unique string identifying MPS SIP Server. IP Address: In this screen must be the MPS SIP Server IP Address.

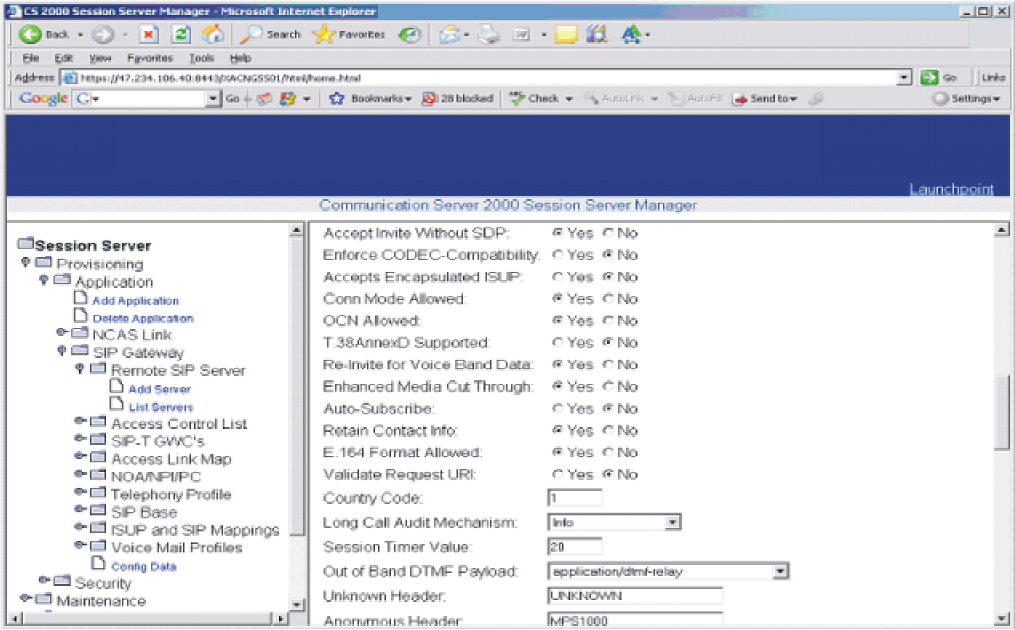
   Server Type field: This field must be set to “Message Server” for UM2000 Application, for other Application like PSP this field must be set to “Session Server”. Protocol Field: must be always set to UDP for connecting to MPS. Other Parameters in this screen and the following screen may be configured as shown in the screen shots images.

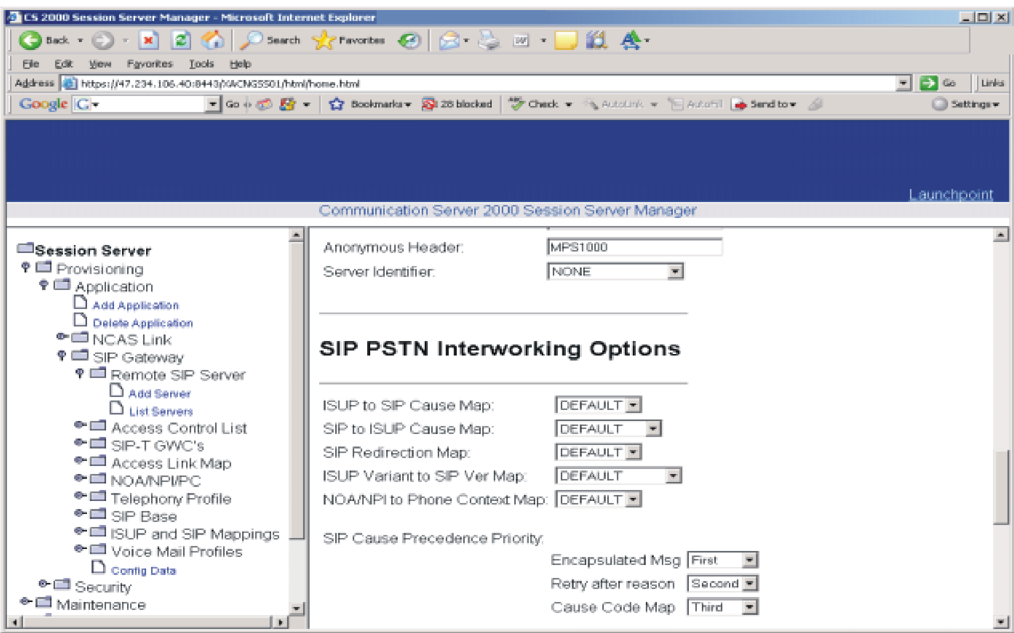


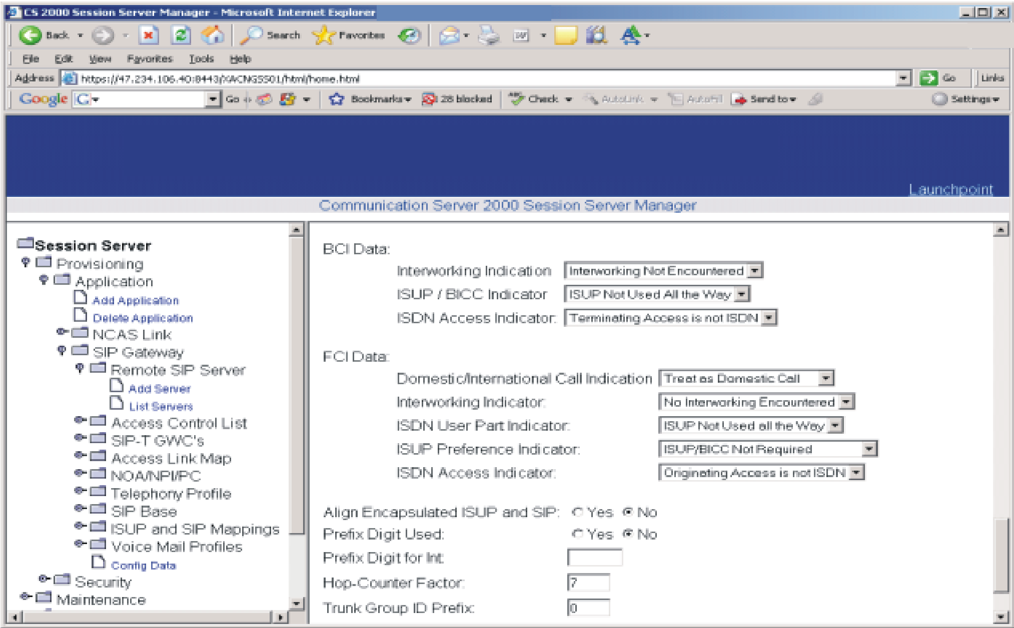


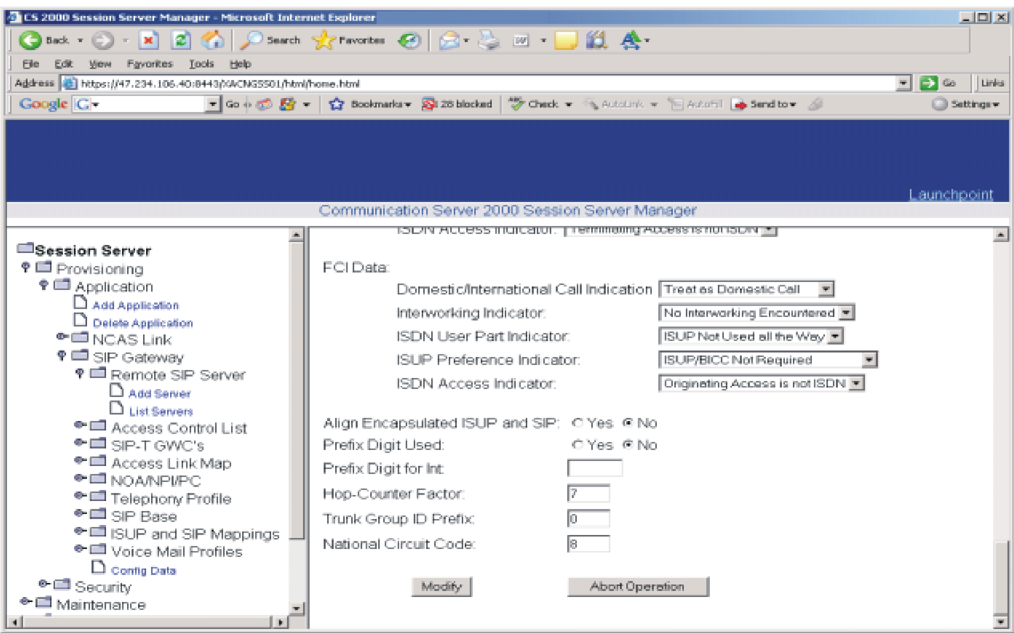
3. If URI parameters (CIC, RN, NPDI, Phone-Context) are present in the incoming URI, MPS SIP server will pass these parameters to application. It is Application's responsibility to interpret it.

   SIP Header Format field can be either "Compact" or "Long". Use OPTIONS for Heartbeat field can be set to either "Yes" or "No". Out Of band DTMF Payload field in this screen shows "application/dtmf-relay" but recommended configuration is "application/telephone-event". If SIP INFO method is used the set this field to "application/vnd.avaya.digits". Set "Long Call Audit Mechanism" to "info".

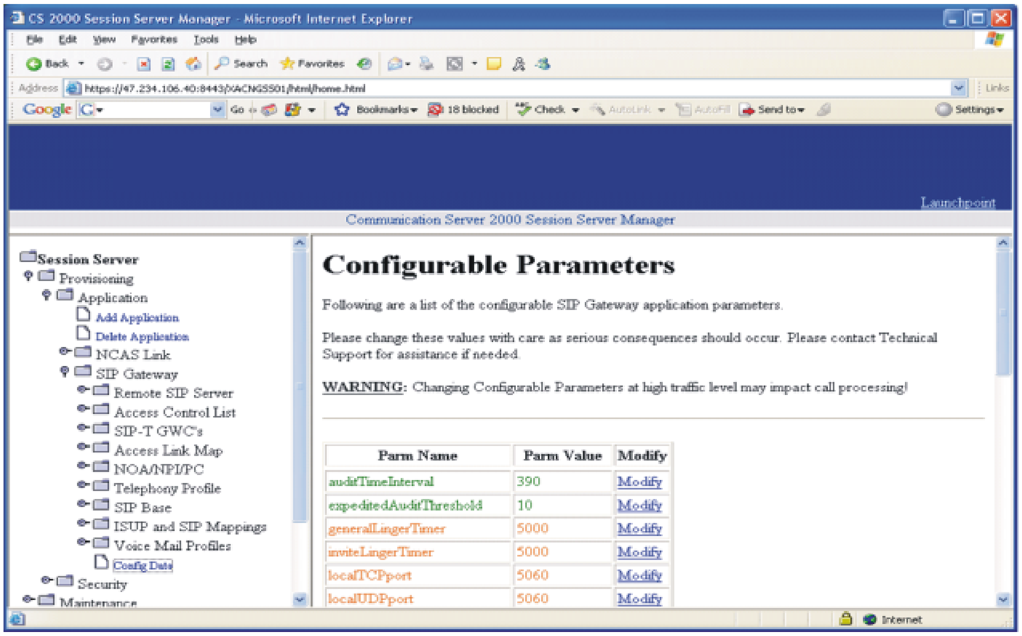

CS 2000 Session Server Manager - Microsoft Internet Explorer
Launchpoint
Communication Server 2000 Session Server Manager
Session Server
Provisioning
Application
Add Application
Delete Application
NCAS Link
SIP Gateway
Remote SIP Server
Add Server
List Servers
Access Control List
SIP-T GWC's
Access Link Map
NOA/NPI/PC
Telephony Profile
SIP Base
ISUP and SIP Mappings
Voice Mail Profiles
Config Data
Security
Maintenance
Accept Invite Without SDP: Yes No
Enforce CODEC-Compatibility: Yes No
Accepts Encapsulated ISUP: Yes No
Conn Mode Allowed: Yes No
OCN Allowed: Yes No
T.38AnnexD Supported: Yes No
Re-Invite for Voice Band Data: Yes No
Enhanced Media Cut Through: Yes No
Auto-Subscribe: Yes No
Retain Contact Info: Yes No
E.164 Format Allowed: Yes No
Validate Request URI: Yes No
Country Code: 1
Long Call Audit Mechanism: Info
Session Timer Value: 20
Out of Band DTMF Payload: application/dtmf-relay
Unknown Header: UNKNOWN
Anonymous Header: MPS1000

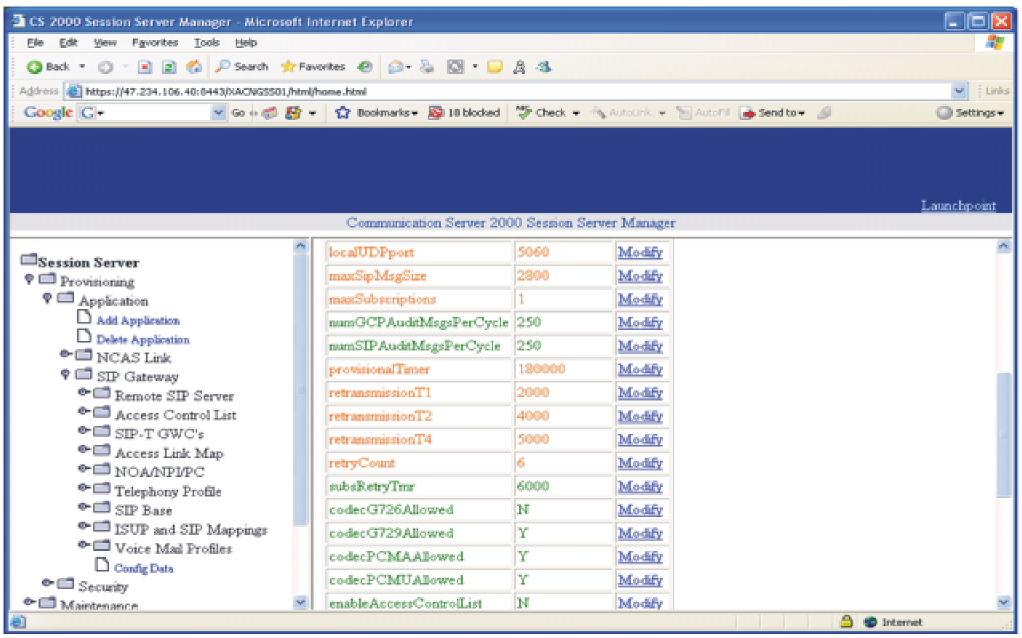

CS 2000 Session Server Manager - Microsoft Internet Explorer
Launchpoint
Communication Server 2000 Session Server Manager
Anonymous Header: MPS1000
Server Identifier: NONE
SIP PSTN Interworking Options
ISUP to SIP Cause Map: DEFAULT
SIP to ISUP Cause Map: DEFAULT
SIP Redirection Map: DEFAULT
ISUP Variant to SIP Ver Map: DEFAULT
NOA/NPI to Phone Context Map: DEFAULT
SIP Cause Precedence Priority
Encapsulated Msg First
Retry after reason Second
Cause Code Map Third



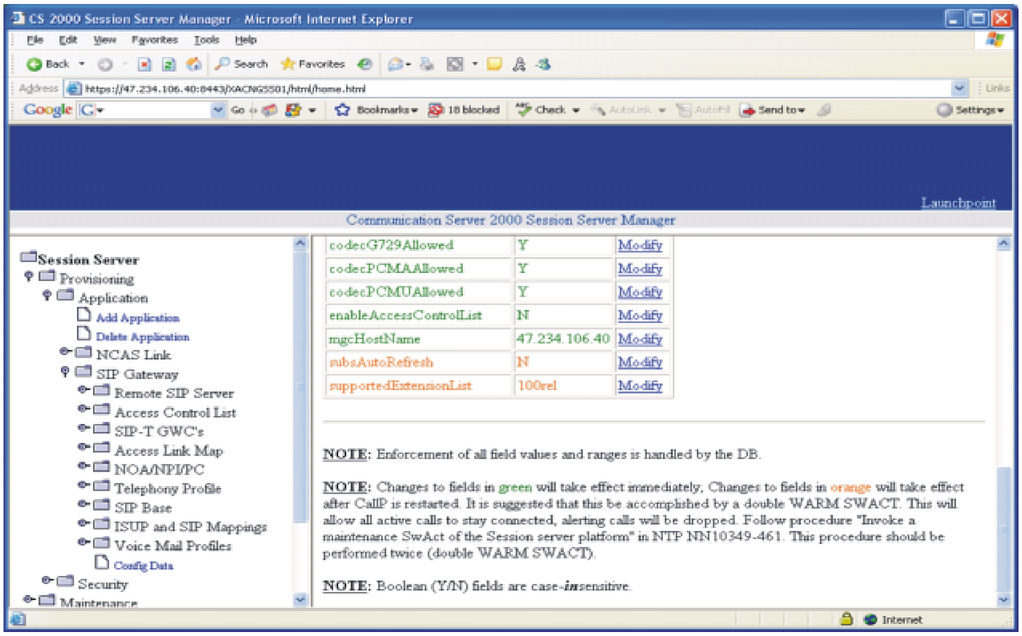


4. Following screenshots describes configuration of SIP Gateway application parameters. Click on Provisioning □ Application □ SIP Gateway □ Config Data, and set the parameters as shown by the following image (for example).





5. mgcHostName field in this screen can be specified a logical HostName instead of IP Address. This requires MPS SIP Server to have patch ccss6.3.118 or higher.

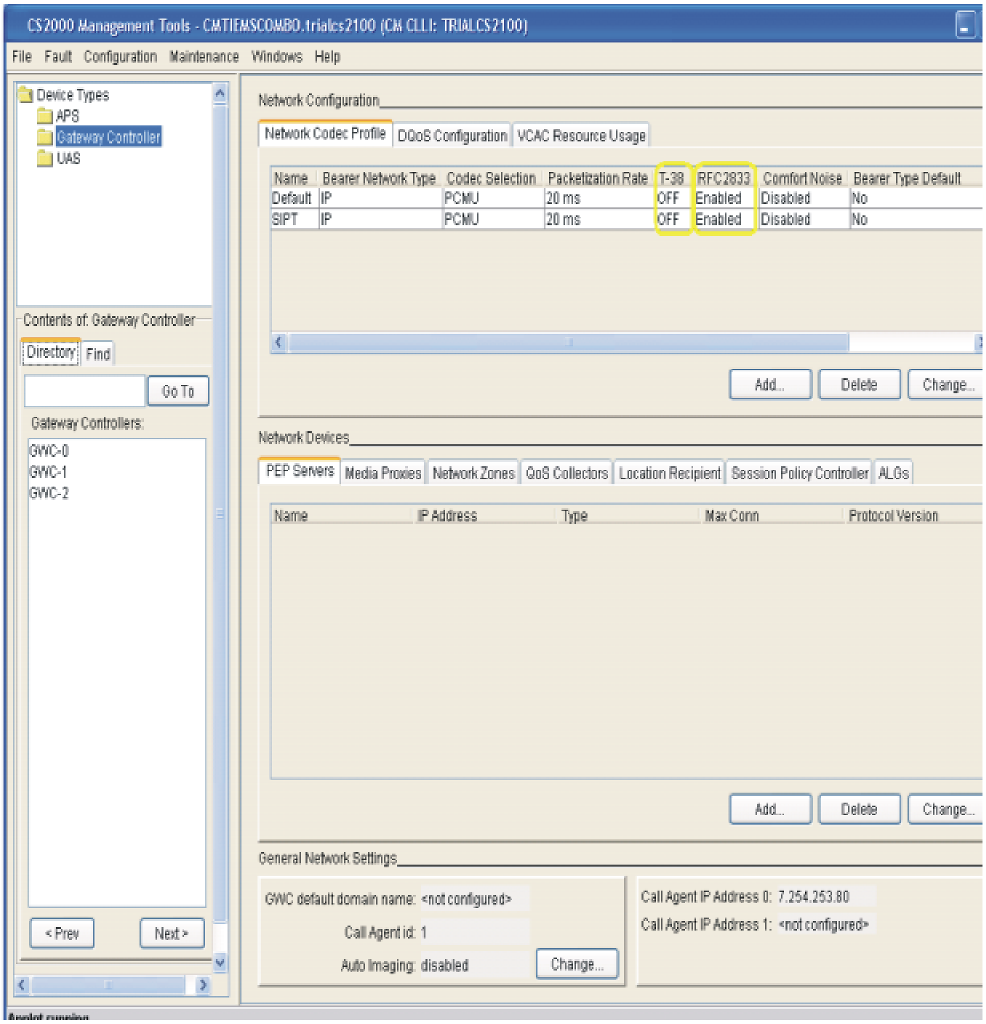


Gateway Controller Configuration

A SIPT Profile must be created within its respective Gateway Controller (GWC), and in turn, this GWC must be associated to the NGSS. In order for the interoperability to function properly, a couple of parameters must be checked within the CMT. To do this, follow these quick steps:

1. Launch the CMT GUI.
2. Login using the proper username and password.
3. Click on CS2000 Management Tools
4. Once running, click on Gateway Controller located in the upper-left-hand window.

The two parameters we are concerned with are T-38 and RFC2833. T-38 must be disabled. MPS does not support T-38. RFC2833 must be enabled and the remote system (MPS) must also be enabled. Make sure that the T-38 and the RFC2833 parameters are set according to the following image:



In addition to the above GUI configuration, some of the tables may need to be filled with data for proper operation. Please refer to CS2100 Session Server Trunks Configuration and Management (NN10338-511) guide for further information.

Configuration on MPS SIP

1. Logon to MPS SIP server.
2. Edit "/opt/ccss/etc/sip.conf" file.
3. Add an entry into NETWORK_HOSTS section of this file for every GWC the MPS SIP server will be using for making outbound calls.

   Let's assume for example, GWC1 domain name is "avaya.com", IP Address 100.100.32.25. GWC2 domain name is "nt.com", IP Address 135.100.32.25.

For MPS SIP Server to use the GWCs in the above example, entries in Network_Hosts section will be the following:

NETWORK_HOSTS = { !-------- ------- -------------------- ----------------------
--------------------- ------ ----- --------- ------------------ ------ ----- --------- ---------------- !
NAME TYPE SIP DOMAIN PRIMARY SECONDARY PORT SSA TIMER
REGISTRAR ! NAME HOST NAME HOST NAME PORT PROFILE FLAG ! --------
------- -------------------- ---------------------- --------------------- ------ ----- ---------
----------------- ------ ----- --------- -------------- CS2100_1 UDP avaya.com
100.100.32.25 - 5060 - SIP_TMR1 false CS2100_2 UDP nt.com 135.100.32.25 -
5060 - SIP_TMR1 false }

“Type” parameter in the above examples identifies the protocol used. Set this parameter to “UDP”, as UDP is the only supported protocol.

“PORT” parameter should be set to the port on which CS2K/2100/GWC receives SIP messages.

MPS Application uses the value (CS2100_1 or CS2100_2) from the above NAME field to select the Gateway Controller for making outbound calls.

For complete description of other parameter in the NETWORK_HOSTS section of the "sip.conf", please refer Media Processing Server SIP 6.3 Features Manual (NN44100-130).

Apart from NETWORK_HOSTS section other sections in the "sip.conf" file are configured normally. Please refer Media Processing Server SIP 6.3 Features Manual (NN44100-130) for configuring other sections of "sip.conf".

SIP Lines with CTI Configuration

The following sections detail the setup of SIP Lines with CTI between the CS 2000 and a MPS SIP server.

SSL domain management

This section describes how to provision domains, subdomains and manage locations.

SSL domain management navigation

- Adding a Trusted Node to the SSL on page 101
- Adding a local root domain on page 104

- Adding a subdomain on page 115
- Deleting a local domain or subdomain on page 116

Adding a Trusted Node to the SSL

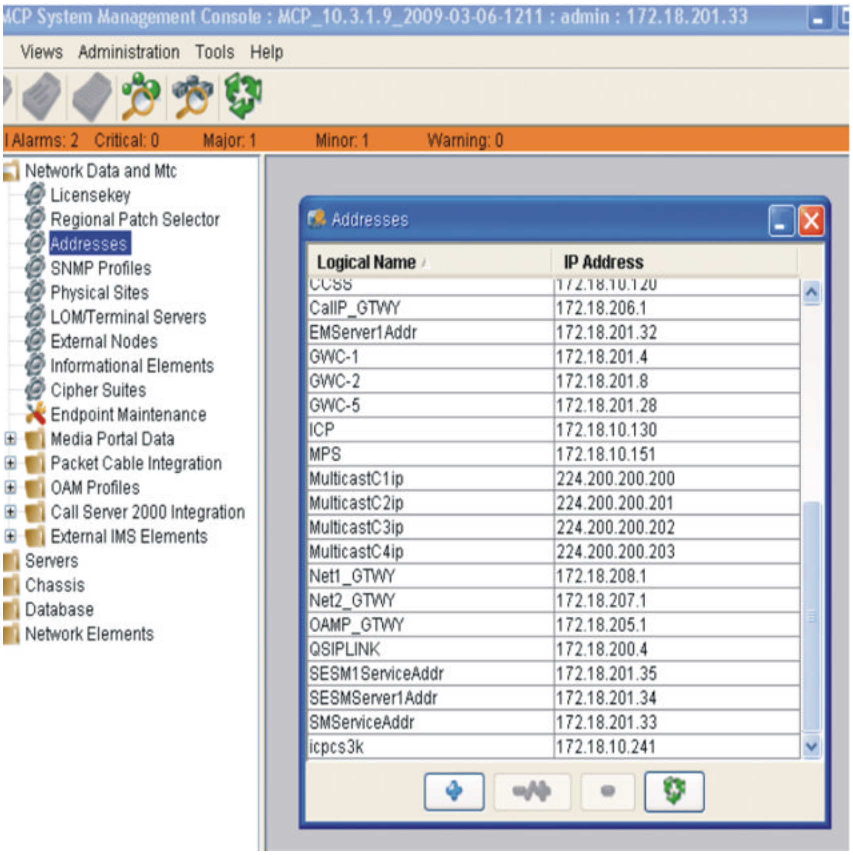
Use this procedure to add the “Trusted“ node to the management console (MCP).

Procedure steps

1. Log on into MCP.

   For example: https://172.18.201.33:12121/

2. In the left menu, click the folder at the top, and click Addresses.

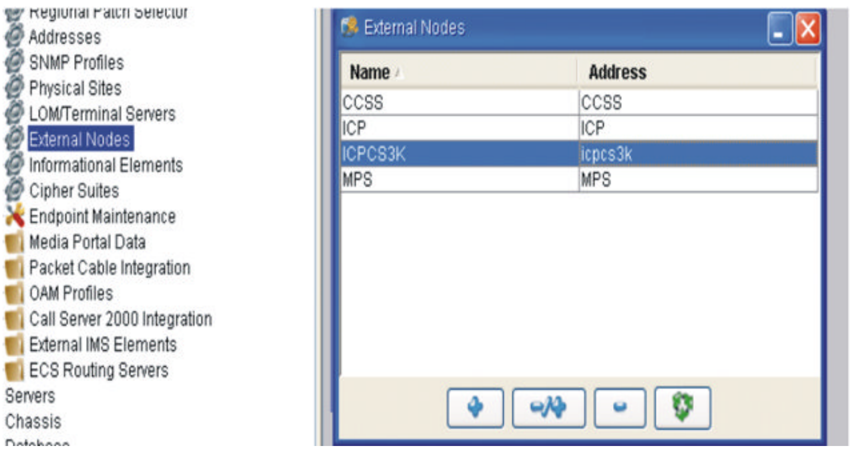


3. Click the "+" button to add an entry for the new trusted node.

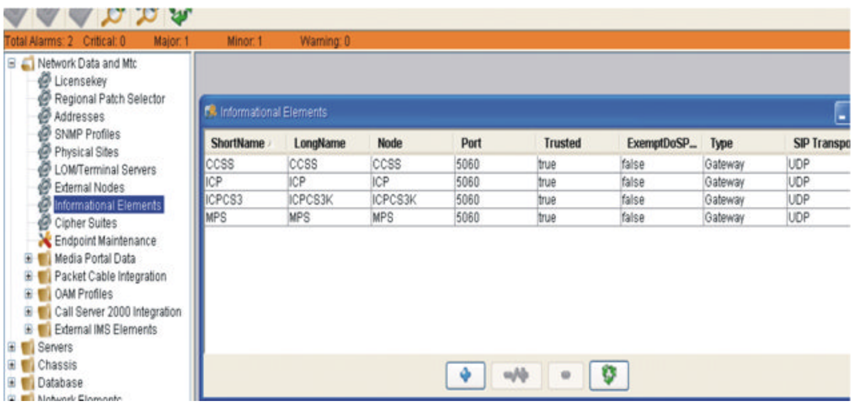
   For example: icpcs3k 172.18.10.241

4. In the menu on the left, click External Nodes and add an entry for the new trusted node.

   For example: Add an entry for ICPCS3K.



5. In the menu on the left, click Information Elements, widen the window and add an entry for the new node.



6. Go to the Provisioning GUI.

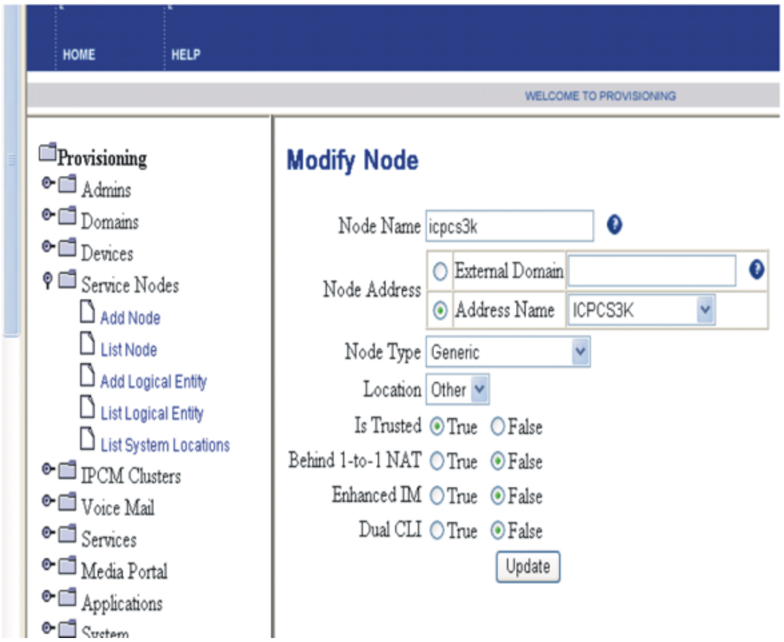
   For example: https://172.18.201.33:8443

7. On the left, click Service Nodes, List Node, Add Node.

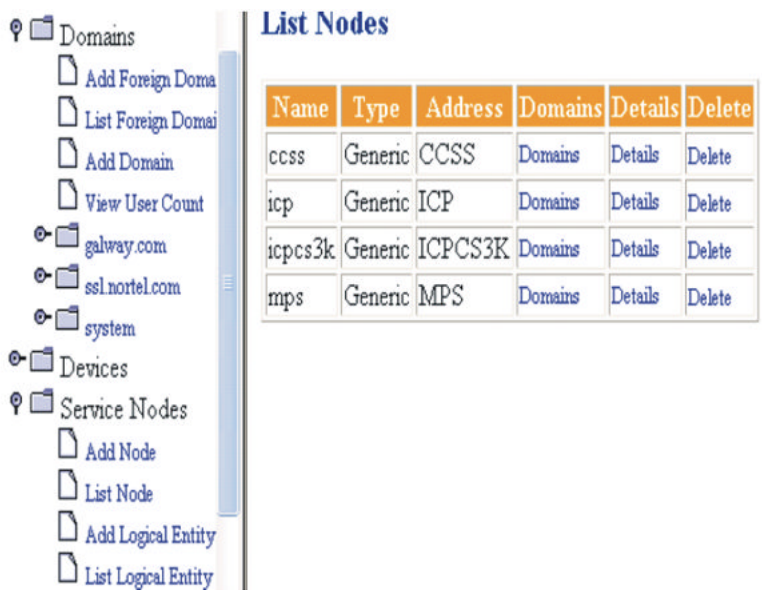
8. For address, select Address Name and use the drop-down menu to select the new trusted node you added in MCP.

   For example: ICPCS3K

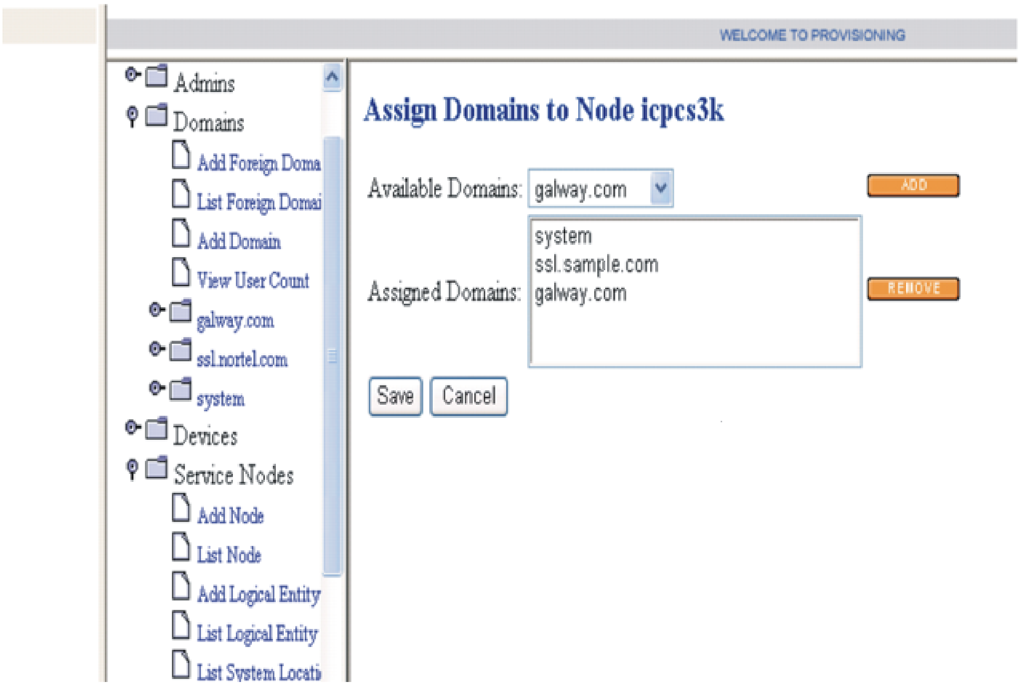
   Note that this is picked up from the entries you've just made on the MCP set as trusted. Node type can be generic.



9. On the line of the trusted node you added, click on the link Domains.



10. Add the Domains that you require. Click Save.



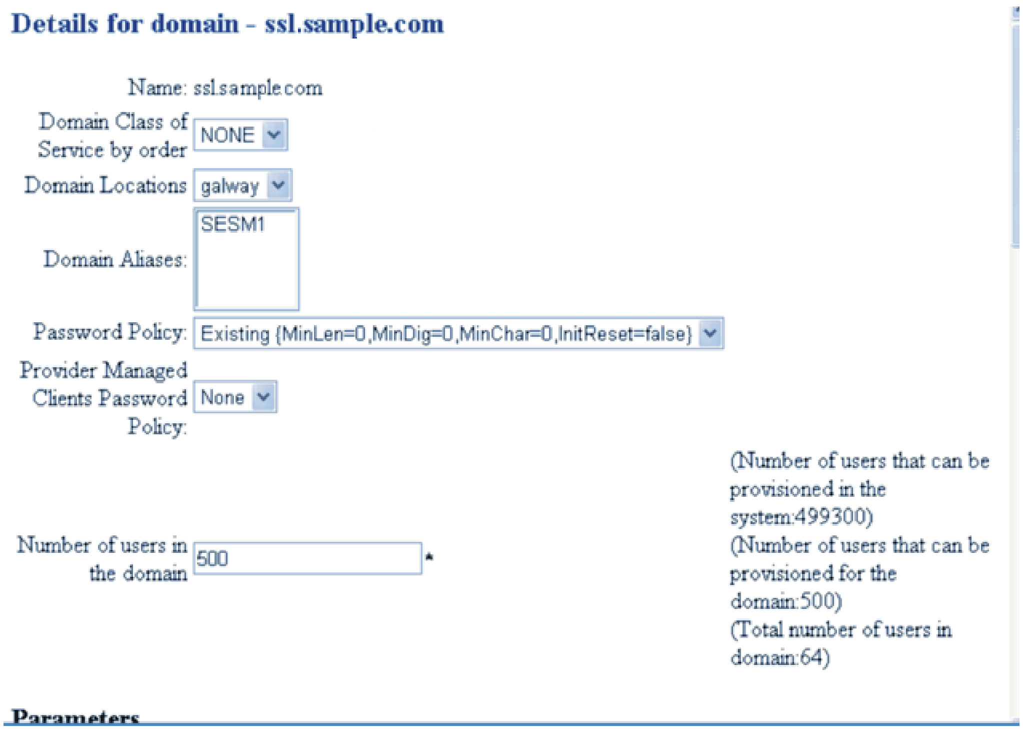
Adding a local root domain

Use this procedure to add a local root domain.

Procedure Steps

1. Click the Domains menu option.
2. Click Add Domain.

   The Create new domain window appears.



Details for domain - ssl.sample.com

Name: ssl.sample.com

Domain Class of Service by order: NONE

Domain Locations: galway

Domain Aliases: SESM1

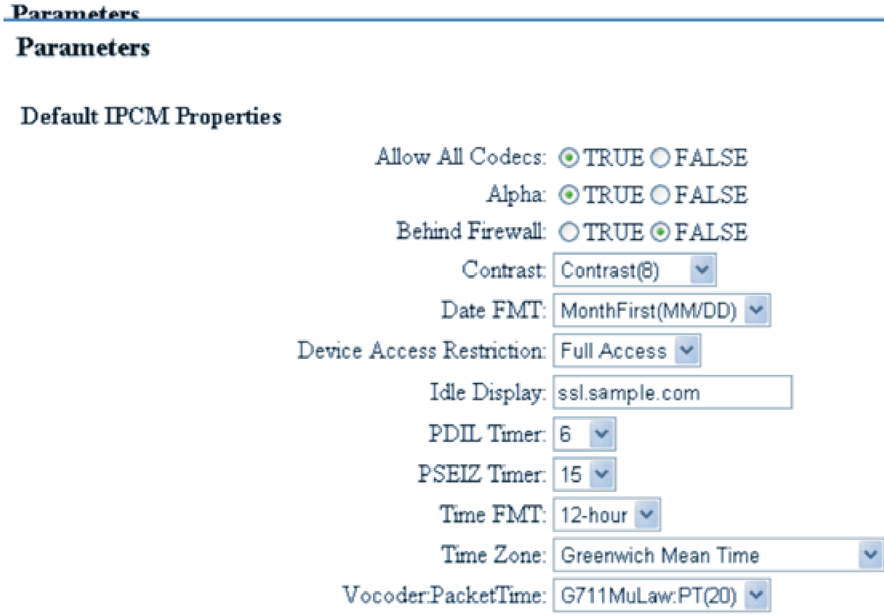
Password Policy: Existing {MinLen=0,MinDig=0,MinChar=0,InitReset=false}

Provider Managed Clients Password Policy: None

Number of users in the domain: 500 *

(Number of users that can be provisioned in the system:499300)
(Number of users that can be provisioned for the domain:500)
(Total number of users in domain:64)

Parameters



Parameters

Default IPCM Properties

Allow All Codecs: TRUE FALSE

Alpha: TRUE FALSE

Behind Firewall: TRUE FALSE

Contrast: Contrast(8)

Date FMT: MonthFirst(MM/DD)

Device Access Restriction: Full Access

Idle Display: ssl.sample.com

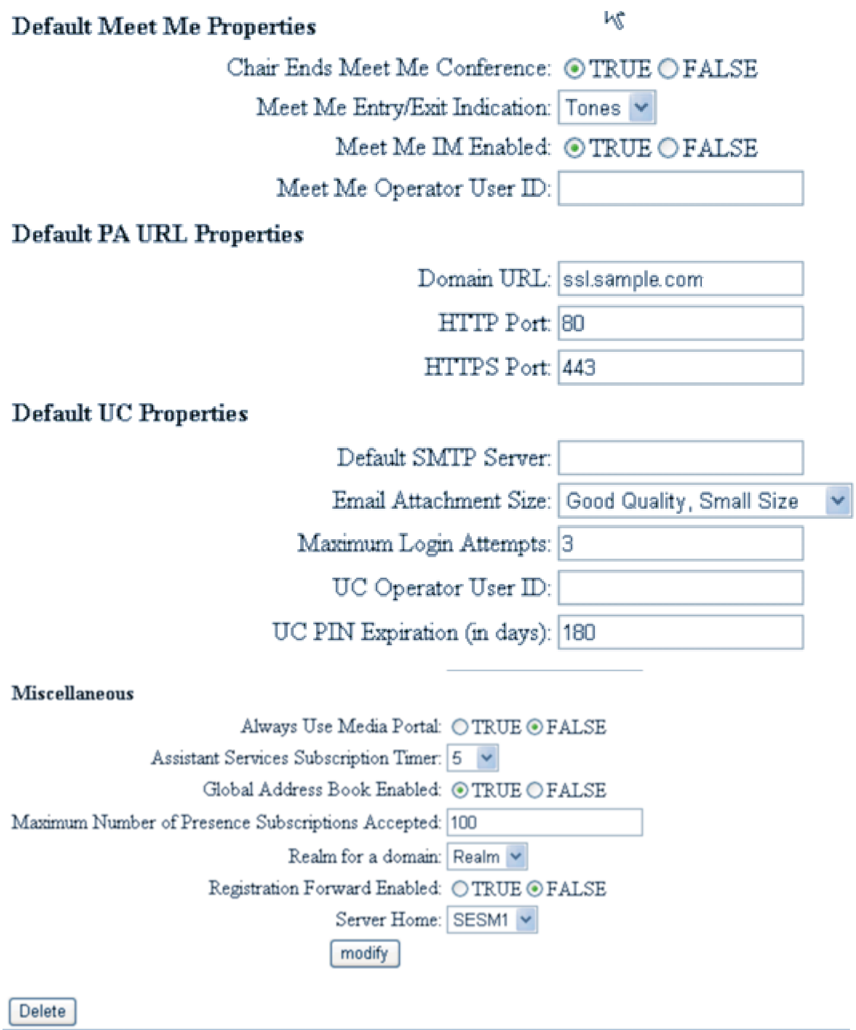
PDIL Timer: 6

PSEIZ Timer: 15

Time FMT: 12-hour

Time Zone: Greenwich Mean Time

Vocoder:PacketTime: G711MuLaw:PT(20)



3. Enter the name of the new local domain.

   The domain name must not be more than 64 characters in length. It can only contain letters, numbers, and limited symbols and cannot contain specific symbols or characters as described in Table 4: Domain name restrictions on page 107 and Table 5: Character limitations for domains on page 107.

4. Enter an alias for the domain from the drop-down list. The domain alias can be the name of the address it represents. It can either be the name of the Session Manager or an Informational Element of type General.

5. Allocate the number of users for the local root domain. The total number of users that can be provisioned is listed in this window.

6. Enter parameter information in the fields provided.

   For domain parameter values and descriptions, see Job aid on page 107.

7. Click Add.

Important:

When you create a root domain that will be used only for SIP-enabled VoIP VPN, a pop-up window with the following error message may appear: "The number of users for a domain cannot exceed the license key enforced system remaining limit of 0." This error message is specific to user resources and does not apply to SIP-enabled VoIP VPN. Set the number of users to 0 and the root domain will be provisioned successfully.

Important:

To see the new root domain added to the list of domains, you must close and reopen the domains folder to refresh the list of domains.

Job aid

Use the data in the following tables for domain name restrictions.

Table 4: Domain name restrictions

For a domain name, use...	Ranging from...
Letters	• a to z • A to Z
Numbers	0 to 9
Symbols	• period (.) • hyphen (-)

Use the data in the following table for character restrictions for domains.

Table 5: Character limitations for domains

<table>
<tr><th>Special character limitations for a domain name</th><th colspan="4"></th></tr>
<tr><td>'</td><td>$</td><td>( )</td><td>|</td><td>;</td></tr>
<tr><td>~</td><td>%</td><td>_</td><td>[ ]</td><td>/</td></tr>
<tr><td>!</td><td>^</td><td></td><td>{ }</td><td>?</td></tr>
<tr><td>@</td><td>&</td><td>+</td><td>"</td><td>,</td></tr>
<tr><td>#</td><td>*</td><td>=</td><td>:</td><td>< ></td></tr>
</table>

Use the data in the following table to provision Domain parameters.

Table 6: Domain parameters

Parameter	Type	Range	Default	Description
Default IPCM Properties				
Allow All Codecs	Boolean	True/False	False	Allow all codecs (true) or only the selected codec (false). See the Vocoder parameter.
Alpha	Boolean	True/False	False	Enables alphanumeric dialing as the default dialing style. If False, the Avaya IP Phone 2002 or Avaya IP Phone 2004 defaults dialing style to numeric.
Behind Firewall	Boolean	True/False	True	Boolean indicating whether there is a firewall between the IPCM and the Avaya IP Phone 2002 or Avaya IP Phone 2004.
Contrast	Integer	0 - 15; select from drop-down list	8	Avaya IP Phone 2002 and Avaya IP Phone 2004 display contrast settings.
Date FMT	Numeric Standard or Numeric Inverse	MonthFirstMM/DD FirstMonthDD/MM	2	2-numeric standard (09/16) 3-numeric inverse (16/09)
Device Access Restriction		Full Access Hands Free Disabled Restricted		Provides the ability to restrict functionality of an Avaya IP Phone 2002 or Avaya IP Phone 2004 respective to the subscribed user who is registered on the device. If multiple people are logged on, they can all receive a call on that phone and can place calls from their account by selecting their line. Being registered on a device is the same as

Parameter	Type	Range	Default	Description
				being logged on. A user can in theory be "logged on" but not registered properly in the following scenario: The user has at one time successfully registered and then for some reason the registration failed due to an error with the Session Manager. The registration is retried periodically. The user is logged on, so, if the IPCM is restarted or the user replugs the device, an attempt is made to reregister the user.
Idle Display	Integer (seconds)	N/A	4	Interdigit time-out (seconds).
PDIL Timer	Integer	N/A	15	Time-out for first dialed digit.
Time FMT	Integer	Select from drop-down list: • 12-hour • French • 24-hour	0	The time format of the proxy used as default for the Avaya IP Phone 2002 or Avaya IP Phone 2004 0 -> 12 hour clock.
Time Zone	Integer	Up to 30 characters; select from drop-down list	N/A	Time zone of the user.
Vocoder: PacketTime	Integer	Select from drop-down list: 0 – G711 Mu-law 4 – G723\6.3kbps 8 – G711\A-law 10 – L16 18 – G729A	0	Default Codec Setting.
Default MeetMe Properties				

Parameter	Type	Range	Default	Description
Chair Ends Meet Me Conference	Boolean	True/False	True	When set to True, the conference will end when the chairman exits the conference. There are two places to set the "Chair Ends Meet Me Conference" property: • At the root domain/ subdomain level, you can set the default value for all users for that domain/ subdomain. If you do not set it for an individual user, then each user receives the value set at its immediate domain or subdomain. • You can change it for individual user using the Provisioning Client (Meet Me Properties link) or using the Personal Agent.
Meet Me Entry/Exit Indication	Boolean	Tones/None	True	When set to True, attendees hear a tone whenever a person enters or exits the conference.
Meet Me IM Enabled	Boolean	True/False	True	When set to True, an IM shows participants as they enter or exit the conference.
Meet Me Operator User ID	String			N/A
Default PA URL Properties				
Domain URL (Required parameter)	String	N/A	N/A	This is the host name of the machine that hosts the Provisioning

Parameter	Type	Range	Default	Description
				Manager. The Multimedia PC Client uses it to retrieve data required through the Provisioning Manager. Note that the Domain URL has to be a DNS-resolvable hostname and not an IP address.
HTTP Port	String	N/A	80	This is the port information tagged onto the Domain URL information, which is sent to the Multimedia PC Client. The Multimedia PC Client uses it to retrieve data from the Provisioning Manager in a non-secure or HTTP mode.
HTTPS Port	String	N/A	443	This is the port information tagged onto the Domain URL information, which is sent to the Multimedia PC Client. The Multimedia PC Client uses it to retrieve data from the Provisioning Manager in a secure or HTTPS mode using Secure Socket Layer (SSL).
Default UC Properties (for Carrier markets only)				
Default SMTP Server	String	N/A	N/A	his is the hostname or IP address of the SMTP server that the Unified Communications service will use when sending email. Example: 46.49.47.70

Parameter	Type	Range	Default	Description
Email Attachment Size	String	N/A	N/A	This is how the Unified Communications service encodes the voicemail message attachment included in an email. Example: Good Quality, Small Size Remember, when the Highest Quality, Large Size option is selected, this does not guarantee that all voicemail messages will be of highest quality. The quality of the encoded voicemail recording cannot exceed the original quality that was received during initial recording. For example, if a user leaves a voicemail message while using a low-quality codec, then the voicemail message is delivered with no increase in quality, even though it is encoded using the Highest Quality, Large Size encoding option.
Maximum Login Attempts	String	N/A	N/A	This is the number of incorrect logons a user can attempt before the Unified Communications service locks the user's mailbox. Example: 3
UC Operator User ID	String	N/A	N/A	This is the username of the Unified Communications service operator. Example: uc_operator

Parameter	Type	Range	Default	Description
UC PIN Expiration (in days)	String	N/A	N/A	This is the number of days that a user's mailbox password is valid. Example: 180
Miscellaneous				
Always Use Media Portal (Required parameter)	String	True/False	False	Directs the Application Server to use the BCP 7000 series (formerly known as RTP Media Portal) when set to True. The purpose is to use the BCP 7000 series function in dealing with special SIP scenarios. For example, a domain that straddles multiple sites contains users who do not have Internet Protocol (IP) connectivity between them due to firewalls at different sites. This domain requires the Always Use Media Portal to be set to True to negotiate between the firewalls to set up SIP sessions.
Assistant Services Subscription Timer	Integer	Select from drop-down list: 1 - 15 minutes	5 minutes	Specifies the amount of time that a client running as an Assistant Console will wait before re-subscribing to the services needed for an Assistant to provide assistant support for another user.
Maximum Number of Presence Subscriptions Accepted (Required parameter)	Integer	0 - 100	0	Maximum number of inbound subscriptions to any given user in this domain. The Personal Agent, the Multimedia PC Client, and the Multimedia Web Client

Parameter	Type	Range	Default	Description
				will use this limit when allowing a particular username to be added to a subscriber's list of friends.
Password Policy (Required parameter)	String/ Integer	Select from drop-down list	Default or Existing	Specifies the subscriber password complexity rules for the domain. Subscriber password complexity rules can be defined to enforce such things as the minimum length of a password, the minimum number of digits that a password must contain, and the minimum number of characters a password must contain. The relationship between these three password complexity controls is such that the sum of the minimum number of digits and the minimum number of characters must be less than or equal to the minimum length of the password.
Public Name	String	Up to 64 characters: • a to z, A to Z • 0 to 9 • period (.) • hyphen (-) • underscore (_)	N/A	After provisioned against a domain or subdomain, the Public Name allows an originator’s calling name to be replaced with the Public Name provisioned against the domain based on a set of rules.
Public Number	String	E164 number	N/A	After provisioned against a domain or subdomain, the Public Number allows an originator’s calling number to be replaced with the Public Number

Parameter	Type	Range	Default	Description
				provisioned against the domain based on a set of rules.
Server Home (Required parameter)	String (60)	UP to 60 characters	N/A	Select a Session Manager from the list of provisioned system managers for the domain.
Registration Forward Enabled	String	True/False	False	Specifies if registrations coming in from subscribers in this domain are handled by the Session Manager or are forwarded to another entity for processing.
Realm for a domain (Required parameter)	String (120)	Up to 120 characters	Realm	Identifies the domain for subscribers when they are being authenticated. Subscribers may register in multiple domains. When they are required to enter a password during authentication, they need to know which domain it is so that they can enter the appropriate password.

Adding a subdomain

Use this procedure to add a subdomain.

Procedure steps

1. Click the Domains menu option.
2. Under the root domain where you want to add a subdomain, click Sub-Domains, Add Sub Domain.

   The Create new subdomain window appears.
3. Enter a name for the subdomain.
4. Enter information in the parameter fields.
5. Click Add.

   A new window appears saying that the domain was added successfully.

Deleting a local domain or subdomain

Use this procedure to delete a local domain.

Important:
Deleting a domain deletes everything it contains (including service packages and subscribers assigned within).

Procedure steps

1. Click the Domains menu option.
2. Click the domain or subdomain you want to delete from the list of domains.

   Important:
   To delete a domain or subdomain, first delete all of its child subdomains.
3. Click Delete to delete the domain or subdomain.
4. Enter a valid admin password in the confirmation window to delete the domain or subdomain, or Cancel to cancel the delete operation.

Service node provisioning

This section describes how to provision a service node.

Adding a node

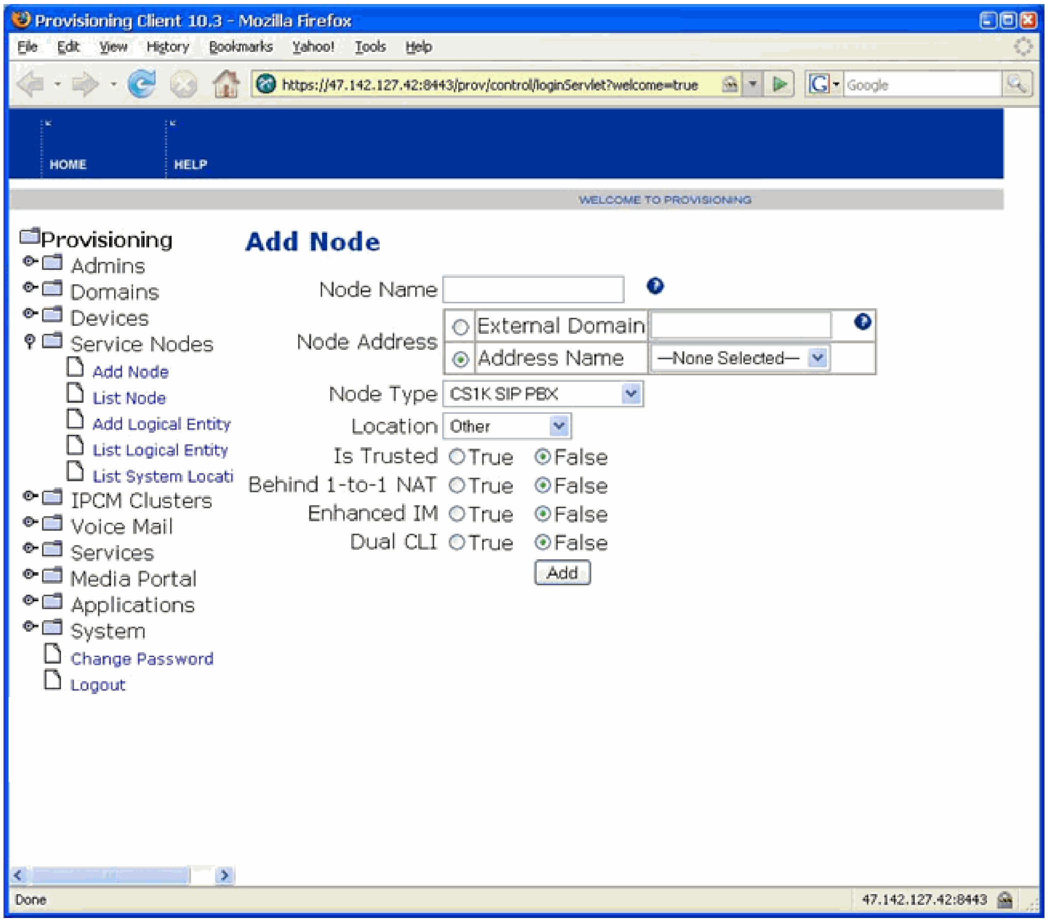
Use this procedure to add a node.

From the Service Nodes menu option, an administrator can add a physical node, such as a gateway, a Media Application Server (MAS), or any other third-party resource, and assign it the necessary attributes, such as its location and node type.

Procedure steps

1. Click the Service nodes, Add Node menu option.

   The Add Node page appears.



2. From the Add Node page, select the fields as shown in the job aid at the end of this section.
3. Click Add.

Job aid

The following table describes the field for the Add Node page.

Table 7: Add Node fields

Field	Description
Node Name	Add the name of the node you wish to add, such as the Audio Codes gateway, H.323 Gateway, or the Media Application Server. Ensure that a node is less than 30 characters. It can only contain letters, numbers, and specific symbols: • Letters a to z, A to Z • Numbers 0 to 9 • Symbols underscore (_)

Field	Description
Node Address	Select either the Address Name or External Domain. Both these fields are mutually exclusive and cannot be selected simultaneously. Address Name. From the drop-down list, select one of the elements on the System Manager: an Audio Codes gateway, a MAS server, an H.323 Gatekeeper, a long name of the Sip-enabled VoIP VPN, an Informational Element of type General or Gateway or Pooled Media resource, or any other element. External Domain. Select this field if an IP address is unavailable. The External Domain address is used during an upgrade. If the host field on the gateway or the route on the pooled entity cannot be parsed properly, the entire route is entered in this field, enabling all calls to work, and a log is generated for all these types. Any data available in this field is not validated and cannot be re-IP'ed.
Node Type	Select the node type from the drop-down list. For example, select the PRI gateway.
Location	Select the location of the node from the drop-down list.
Is Trusted	Select True if the node is trusted. Always select True when configuring SIP-enabled VoIP VPN.
Behind 1-to-1 NAT	Select whether the behind 1-to-1 NAT is available. Always select the True radial button to activate video support on MeetMe Conference on Session Server Lines. This option does not apply to Sip-enabled VoIP VPN.
Enhanced IM	Select whether enhanced IM is available.
Dual CLI	Select True radial button to activate Dual CLI option. Always select the True radial button, when configuring SIP-enabled VoIP VPN.

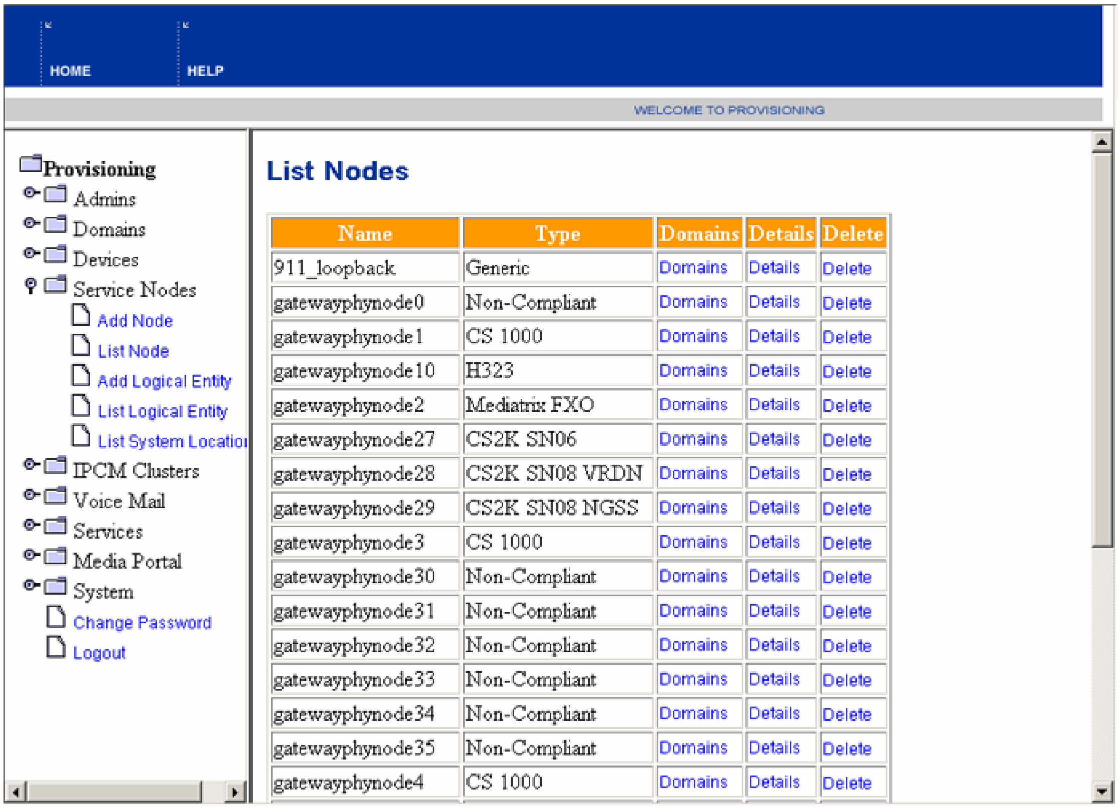
Listing a node

Use this procedure to list a node. Administrators can view all the nodes provisioned on the system.

Procedure steps

1. Click Service nodes.
2. Click List Node.

   The List Nodes page appears, showing the node or all the nodes that were provisioned from the Add Nodes page.



Assigning a domain to a node

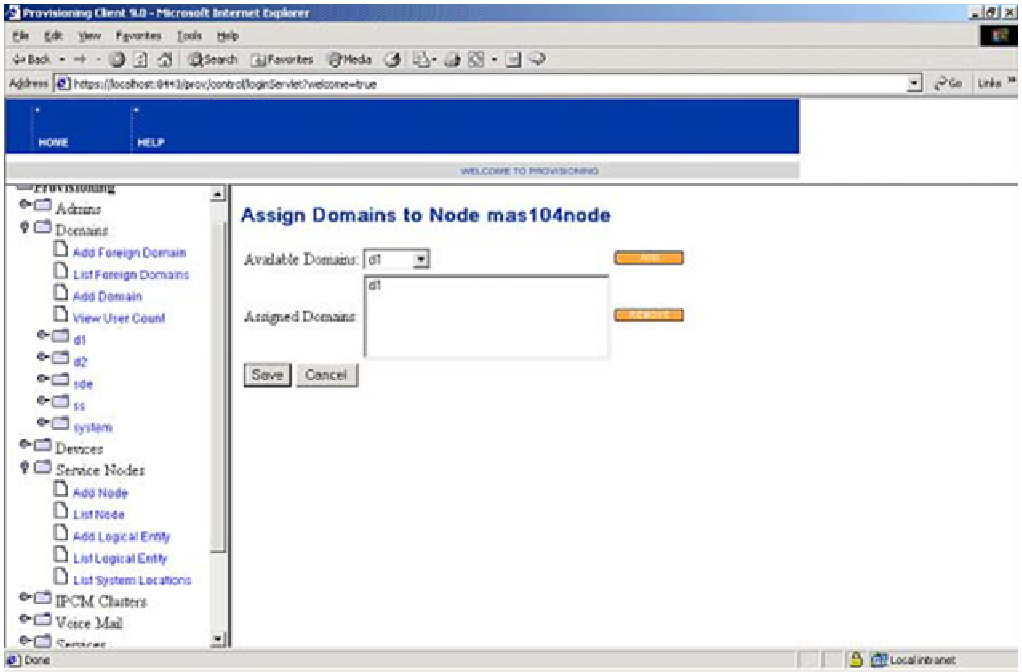
Use this procedure to assign a domain to a node.

When a service node is added, an administrator can assign a domain to a node, such as a gateway or a media application server.

Procedure steps

1. Click the Service nodes, List Node menu option.
2. Click the Domains link of the node name.

   The Assign Domains page appears.



3. Select the domain you wish to associate with the node from the Available Domains drop-down list.
4. Click Add. The Assigned Domains field shows the domain you added.
5. Select the domain from the Assigned Domains field and click Remove to remove the domain.
6. Click Save to save your changes or Cancel to abort the action.

Important:

After a domain is assigned to a node, the domain can use the node to create gateway routes, Emergency Response Locations (ERLs), or assign to Routable Services.

Adding a logical entity

Use this procedure to add a logical entity.

An administrator can add logical nodes from the Services nodes menu option. Logical nodes consist of one or more physical nodes, enabling the administrator to define more than one physical node to associate parameters with a specific physical node.

An administrator is allowed to associate parameters with a specific physical node. For example, an administrator may have provisioned one physical gateway, but may want to route to different trunk groups owned by this gateway. To accomplish this, the administrator has to create a logical node for each trunk group by picking the physical node and then associating a "avayatrkgrp" parameter with it.

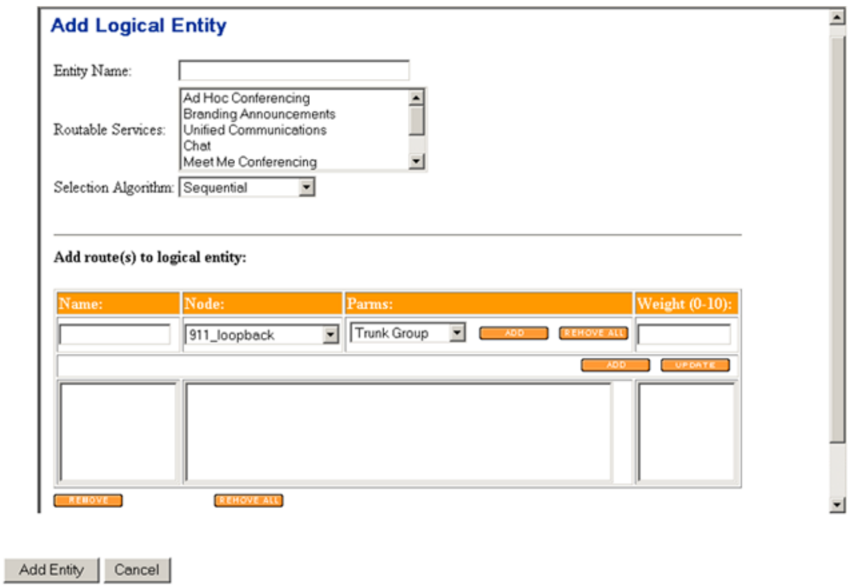
Administrators can also define more than one physical node as part of the logical node definition.

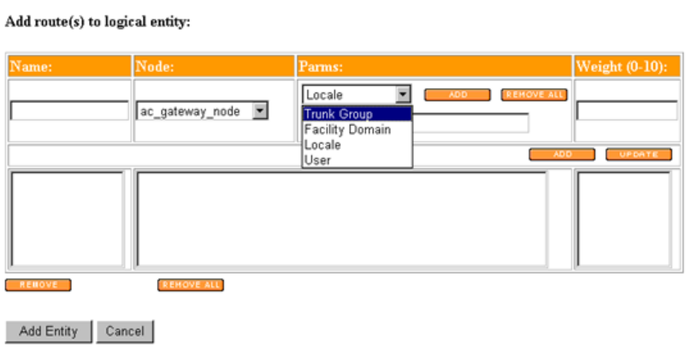
Important:
After a logical entity has been added, it can be assigned to root domains. Only after a domain is assigned to a node, can the domain use the node to create gateway routes, Emergency Response Locations (ERLs), or assign to Routable Services.

Procedure steps

1. Click the Service nodes, Add Logical Entity menu option.

   The Add Logical Entity page appears.





2. Add the parameters for the Logical Entity fields as shown in job aid at the end of this section.
3. Click Add Entity to add the new or modified logical entity, or click Cancel to cancel your changes.

Job aids

The following table describes the parameters associated with the Logical Entity field.

Table 8: Logical Entity field descriptions

Logical Entity field	Description
Entity name	Type a name for the logical entity. A logical entity name cannot be more than 30 characters in length and can only contain the following letters, numbers, and specific symbols: • Letters: a to z and A to Z • Numbers: 0 to 9 • Symbols: underscore (_)
Routable Services	Select the appropriate type from the drop-down list as determined by the type of logical entity being added. Remember, the Routable Services are optional for a logical entity, which is a collection of gateways only.
Selection Algorithm	Select Sequential, Weighted Average, or Round Robin drop-down list. Sequential is the default option and is used for pooled gateways.
Add route(s) to logical entity	Assign one or more routes and associated weights (0 – 10; default is 1) to the logical entity. For field descriptions, see Table 9: Add route(s) to logical entity fields on page 122.

The following table provides field descriptions of the Add route(s) to logical entity fields.

Table 9: Add route(s) to logical entity fields

Name	Type the name of the route in this field. This field can have up to 30 alphanumeric characters.
Node	Select the appropriate node from the drop-down list.
Parms	Select the appropriate element from the drop-down list: • Trunk Group: This parameter associates a node with a trunk group. • Facility Domain: This is the domain associated with the outgoing requested URL. • User: The field is used in conjunction with SIP and PSTN-type gateways. The only option currently available in the drop-down list is phone. The phone is added to requested URL in the outgoing SIP INVITE. • Locale: This is the location of the device. Populate the field to add a locale to the outgoing SIP message.
Weight	Modify existing route weights by highlighting the desired route and changing the weight value in the Weight (0 – 10) field. Enter the associated weight of the route in this field. Remember the following guidelines:

	• Routes with a zero weight are disabled. • At least one route must be defined before the logical entity can be added. • The total weight of all routes may be zero. • The percentage of the total weight for each route is calculated automatically. • Weight field has no meaning when Sequential Selection Algorithm is selected.

Listing, modifying, deleting a logical entity

Use this procedure to list, modify, or delete a logical entity.

The List logical Entity menu option allows an administrator to modify or delete information for existing logical entities provisioned in the system.

Procedure steps

1. Click the Service nodes, List Logical Entity menu option.

   The Logical Entities page appears.

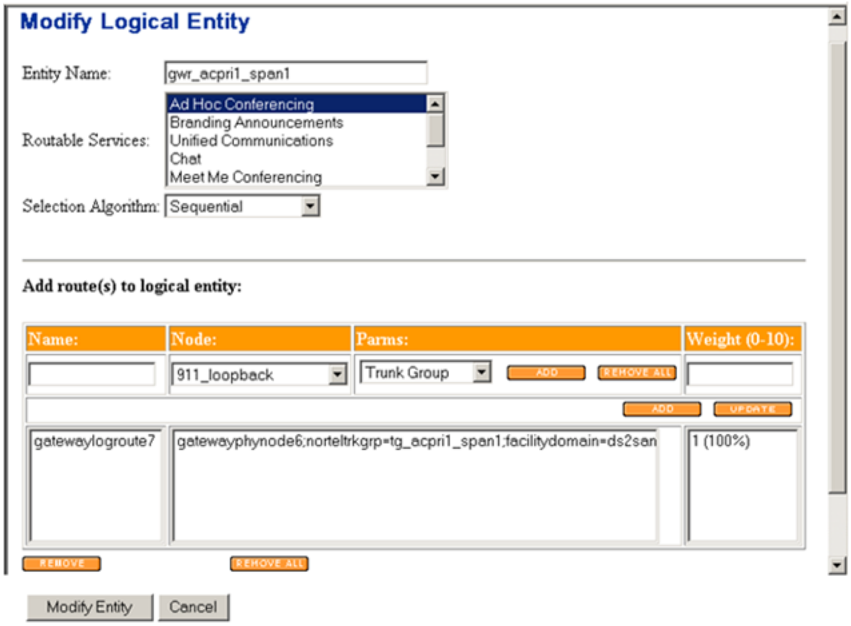
Logical Entities

Name	Domains	Details	Delete
gwr_acpri1_span1	Domains	Details	Delete
gwr_acpri1_span2	Domains	Details	Delete
gwr_acpri1_span_all_ds2sanity1	Domains	Details	Delete

2. To delete a logical entity click the Delete link. Enter a valid admin password in the confirmation window to permanently delete the logical entity or Cancel to abort the action.
3. Click the Domains link to assign domains to a logical entity.

   The Assign Domains page appears.
4. Select the domain you wish to assign from the Available Domains drop-down list.
5. Click Add. The Assigned Domains field shows the domain you added.
6. Select the domain from the Assigned Domains field and click Remove to remove the domain.
7. Click Cancel to abort the action.

   The Logical Entities page appears.
8. To view details and modify the parameters of a logical entity, click the Details link.



Session Server Line subscribers provisioning

Session Server Line subscribers are provisioned using SERVORD+ commands at the CS 2000 OSSGATE interface or using the Bulk Provisioning Tool. The subscriber provisioning information flows through to the Session Server Lines.

This section describes the Session Server Lines-specific options in the SERVORD commands. For Session Server Lines, only NEW, OUT, and CHF provisioning commands are supported.

Subscribers are assigned a service package(s) during the flow through provisioning using SERVORD+ or OSSGATE. Multiple service packages can be assigned to the subdomains supporting the Session Server Line subscribers.

The information contained in this section does not require that you perform any of the operations in any particular order.

- The hardware and software components needed by the Session Server Lines application are configured and operational.
- The basic provisioning has been completed using the Provisioning Client.

For more information on using SERVORD+ commands and the OSSGate interface, see the SERVORD Reference Manuals and the OSSGate Users Guide (NE10004-512).

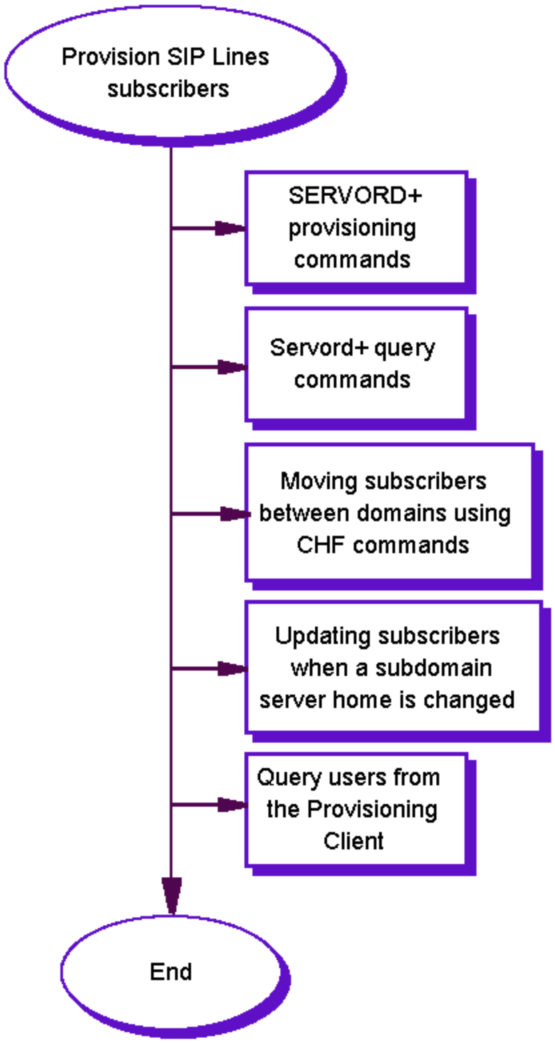


Figure 1: Provision Session Server Lines subscribers procedures

- SERVORD+ provisioning commands on page 125
- Servord+ query commands on page 128
- Moving subscribers between domains using CHF commands on page 129
- Updating subscribers when a subdomain server home is changed on page 129

Provisioning commands

This section contains commands one is able to use for provisioning.

SERVORD+ provisioning commands

The supported SERVORD+ commands use the following three additional options for Session Server Lines provisioning:

DPL

DPL is the option for a dynamic packet line and sets it for use as a Session Server Line. The DPL command contains two sub-options as described in the table. Example: DPL Y 10. The DPL data is proxied to the CS 2000 Core and is visible as read-only data in Table IBNFEAT.

Table 10: DPL sub-options

Sub-option	Format	Description
SIP	boolean [Y]	The first sub-option enables the DPL to act as a SIP line and is set to Y.
Maximum number of simultaneous sessions	Integer [1-10]	The second sub-option sets the maximum number of simultaneous sessions allowed for the subscriber. Avaya recommends that you set this sub-option to the default value of 10.

AGNTPCL

AGNTPCL (Agent Protocol) - This option provides provisioning functionality for (IMS Service Control) ISC SIP lines. Using the PROTOCOL and VARIANT fields provided by this option, the AS2000 Core distinguishes between IETF and ISC SIP lines. The AGNTPCL option is only available to IBN and RES (1FR and 1MR only) Line Class Codes (LCCs).

The AGNTPCL data is proxied to the CS2000/AS2000 Core and is visible as read-only data in Table IBNFEAT.

Sub-option	Format	Description
Protocol	NONE/SIP	The first sub-option specifies if SIP is provisioned on the line. NONE - specifies that SIP is not provisioned on the line. SIP - specifies that the line is to be provisioned as a SIP line.
Variant	NONE/IETF/ISC	The second sub-option specifies how the SIP line is to be provisioned. NONE - specifies that the line is not provisioned as IETF or ISC SIP. IETF - specifies that the line is to be provisioned as an IETF SIP line. ISC - specifies that the line is to be provisioned as an ISC SIP line.

The following is an example of the SERVORD+ command with the AGNTPCL option:

```
> new $ 6137550102 ibn bnr 0 0 nillata 0 SS 00 1 02 01 SIP_DATA
SIP_PACKAGE 6wcpkg SIP_URI+
user101Qrtp7ssb.com SIP_CLIENT_TYPE ONT SIP_LOCATION other 2
SIP_PASSWD u1234 $ dpl y 10 agntpcl sip isc $
```

See the Avaya OSSGate User Guide (NN10004-512) for more information about the AGNTPCL option.

SIP_PASSWORD

This is the subscriber's password that is proxied to the Session Server Lines.

SIP_DATA

This is the set of SIP-specific parameters that are proxied to the Session Server Lines. The data from this option is returned as part of QLEN and QDN queries performed through OSSGATE. The sub-options are the following:

- SIP_PACKAGE
- SIP_CLIENT_TYPE
- SIP_URI
- SIP_LOCATION
- SIP_PASSWD
- SIP_SUBDOMAIN

Important:

The package, client type, and location information must match exactly what was provisioned using the Provisioning Client. This includes case and spacing. Moreover, define the client type in lowercase in the Provisioning Client and do not use the default client type ONT. For example, if the following is the service package name provisioned in the Provisioning Client:

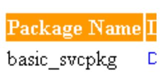


Figure 2: Package name screen

then provision SIP_PACKAGE with basic_svcpkg.

Important:

Include the subscriber's subdomain when provisioning the SIP_URI. Users are provisioned into subdomains, not the root domain. Each subdomain is associated with a specific Session Manager. By associating subdomains (and, therefore, subscribers) with a Session Manager, you can properly manage scaling and performance of the Session Server Lines.

Important:

Subscriber usernames and directory numbers must be different.

For example, a subscriber with a DN = 7057420256 cannot have a username of 7057420256@yourcompany.com.

The vmgs/endpoints found within the supported NEW, OUT, and CHF commands trigger the data distribution to Session Server Lines from SESM SERVORD+.

Examples using NEW command (establish service)

The following are two examples of the NEW command being used to provision a Session Server Lines endpoint:

```
NEW $ 6195209998 1FR LATA1 0 SCOT 00 0 00 00 dgt + SIP_DATA SIP_PACKAGE SIP Lines
SIP_URI + sdoe@yournetwork.com SIP_CLIENT_TYPE SIP Line + SIP_LOCATION
yournetwork.RT.NC0 SIP_PASSWD scott11 + SIP_SUBDOMAIN subdomain.yournetwork.com $
DPL Y 10 $
```

```
NEW $ 6195209998 IBN IBNTST 0 0 LATA1 0 SS 000 6 00+ 04 SIP_DATA SIP_PACKAGE SIP
Lines SIP_URI + sdoe@yournetwork.com SIP_CLIENT_TYPE SIP Line + SIP_LOCATION
yournetwork.NC0 SIP_PASSWD scott11 + SIP_SUBDOMAIN subdomain.yournetwork.com $ DPL
Y 10 DGT $
```

Note that the DGT line option is required for Session Server Lines. It will be added automatically if not specified in the commands.

Examples using CHF command (change feature)

Important:

CHF requires the use of gateway/termination names (or the equivalent LEN) to provide flow through to the Session Server Lines. Use of a DN will result in the command being processed only by the Core Manager.

The following are two examples of the CHF command being used to change the Session Server Lines options of an provisioned endpoint:

```
CHF $ SCOT  00 0 00 00 SIP_DATA SIP_PACKAGE siplines SIP_PASSWD scott11
SIP_CLIENT_TYPE IBN SIP_LOCATION IBM.RTP $ $
```

```
CHF $ vmg1 SCOT/000/0/0000 SIP_DATA SIP_PACKAGE siplines SIP_PASSWD scott11
SIP_CLIENT_TYPE IBN SIP_LOCATION IBM.RTP $ $
```

Examples using OUT command (remove service)

The following are two examples of the OUT command being used to remove provisioned endpoints:

```
OUT $ 6195209998 SCOT  00 0 00 00 BLDN
OUT $ 6195209998 vmg1 SCOT/000/0/0000 BLDN
```

Servord+ query commands

The query commands QLEN/QTP and QDN return the SIP_DATA options provisioned using the NEW and CHF commands. The SIP_DATA option information appears at the end of the existing QLEN/QDN output, under the header END POINT DATA. The endpoint information lists the following:

- SIP_CLIENT_TYPE
- SIP_EP_NAME
- SIP_VMG_NAME
- SIP_DN
- SIP_LOCATION
- SIP_PACKAGE
- SIP_URI
- SIP_SUBDOMAIN

Moving subscribers between domains using CHF commands

Much of provisioned Session Server Lines information for subscribers is directly related to the subdomain of the subscriber. Moving a subscriber to a different subdomain using the CHF command can result in a loss of the subscriber data if the provisioned data does not match. No warning is given when this occurs.

For example, the name of the Session Server Lines service package may differ between subdomains. If a subscriber is moved between domains and the provisioned service package name does not match, the subscriber is assigned the default package provisioned for the new subdomain.

To prevent the loss of a subscriber's Session Server Lines data, execute a QLEN/QDN before and after moving the subscriber between subdomains to ensure that there was no data mismatch or loss.

Updating subscribers when a subdomain server home is changed

Virtual Media Gateways are associated with both a Session Manager and subscribers. Subscribers are homed on a Session Manager through the server home attribute associated with the subdomain that the subscriber belongs to. If there is a change in the Session Manager of a subdomain, then the virtual media gateway information associated with the subscriber is no longer valid.

When the subdomain server home is changed, the following has to be performed to associate the subscriber with the correct virtual media gateway:

- Using SESM, change the end point ID for each affected subscriber on the Core.
- Using the Provisioning Client, change the end point ID and the VMG of the subdomain.

Adding a user to a domain or a subdomain

Use this procedure to add a user to a domain or subdomain.

Prerequisites

If a service package changes while you are modifying a user, and if the service package has a license-keyed service, the resources are added and removed accordingly.

Procedure steps

1. Click the Domains option.
2. Under the domain or subdomain, click the User, Add User menu option.

   The Add new user window appears.
3. Enter information about the new user in the parameter fields as described in the following job aid.
4. Click Save.

   Remember that the save operation succeeds only if the maximum number of users that can be added to the domain has not been reached.

Job aid

Use the date in the following table for values of the User parameters.

Table 11: User parameters

Parameter	Type	Range	Default	Description
User Name	String (60	Up to 60 characters	N/A	Username of the user. Preferably use lowercase. Does not have to be the same name as the First Name. Remember, a user name can only contain letters, numbers, and specific symbols: Letters: a to z, A to Z Numbers: 0 to 9 Symbols: period (."), hyphen (-) and underscore (_)
First Name	String (30)	Up to 30 characters	N/A	First name of the user. Case sensitive. The first name combines with the last name to form the display name on the Avaya IP Phone 2002 or Avaya IP Phone 2004. Either first name or last name is a required field.
Last Name	String (30)	Up to 30 characters	N/A	Last name of the user. Case sensitive. The last name

Parameter	Type	Range	Default	Description
				combines with the first name to form the display name on the Avaya IP Phone 2002 or Avaya IP Phone 2004. Either first name or last name is a required field.
Password	String (20)	Up to 20 characters	N/A	Password of the user. Must match the password policy of the domain.
Confirm Password	String (20)	Up to 20 characters	N/A	Password of the user.
Type	Select from drop-down list	N/A	N/A	
Service package	Select from drop-down list	N/A	N/A	The list of service packages assigned to the user's domain or subdomain.
Aliases	String (40)	N/A	N/A	The list of aliases for the user. Aliases associate a PSTN phone number with a username. **Important:** Make certain that the aliases of the user are not the same as the Private Charge ID, Public Charge ID, or the user name of this user, and vice versa.
Status Reason	Select from drop-down list	N/A	N/A	The status value of the user.
email	String (60)	Up to 60 characters	N/A	Email address of the user.
Business Phone	String (60)	Up to 60 characters	N/A	Business phone number.
Home Phone	String (30)	Up to 30 characters	N/A	Home phone number.
Cell Phone	String (30)	Up to 30 characters	N/A	Cell phone number.

Parameter	Type	Range	Default	Description
Pager	String (30)	Up to 30 characters	N/A	Pager number.
Fax	String (30)	Up to 30 characters	N/A	Fax number of the user.
Directory Number	Number	Up to 64 characters	N/A	The number that identifies a CS 2000 user and is used in call termination attempts.
Private Charge ID	String (30)	Up to 30 characters	N/A	The Private Charge ID of the user. The private charge ID, typically consists of a public or private directory number. Your charge ID may contain other numbers that are not directory numbers. They can be account numbers, your lucky number, any number the administrator wants to put in there to identify the user for accounting purposes. **Important:** Make certain that the Private Charge ID is not the same as the user alias or the user name of this user, and vice versa.
Public Charge ID	String (30)	Up to 30 characters	N/A	The 10-digit national dial plan number associated with the subscriber. This is required for the interworking with Public Switched Telephone Network (PSTN) switches in the Time Division Multiplex (TDM) network. The public charge ID typically consists of a public or private directory number. Your charge ID may contain other numbers that are not directory numbers. They can be account numbers, your lucky number, any number the administrator wants to put in there to identify the user for accounting purposes. **Important:** Make certain that the Public Charge ID is not the same as the user alias or the user name of this user, and vice versa.

Parameter	Type	Range	Default	Description
Location	Select from drop-down list	Up to 30 characters	Use Domain Default "Other"	The location where the user resides.
Class of Service	Select from drop-down list	Up to 30 characters	None selected	The name of the class of service assigned to the domain and user. This appears to the administrator when modifying or assigning the COS to a domain/subdomain/ user. Select None. Selected to limit the user to IP-only calls.
Redirection Class of Service (COS)	Select from drop-down list	Up to 30 characters	None selected	The class of service value associated with a redirected call attempt. For a user who redirects a call, this COS value is used in place of the user's normal COS for such things as gateway routing.
Time Zone	Select from drop-down list	N/A	N/A	The time zone that the user belongs to.
Locale	Select from drop-down list	N/A	N/A	Locale of the user.

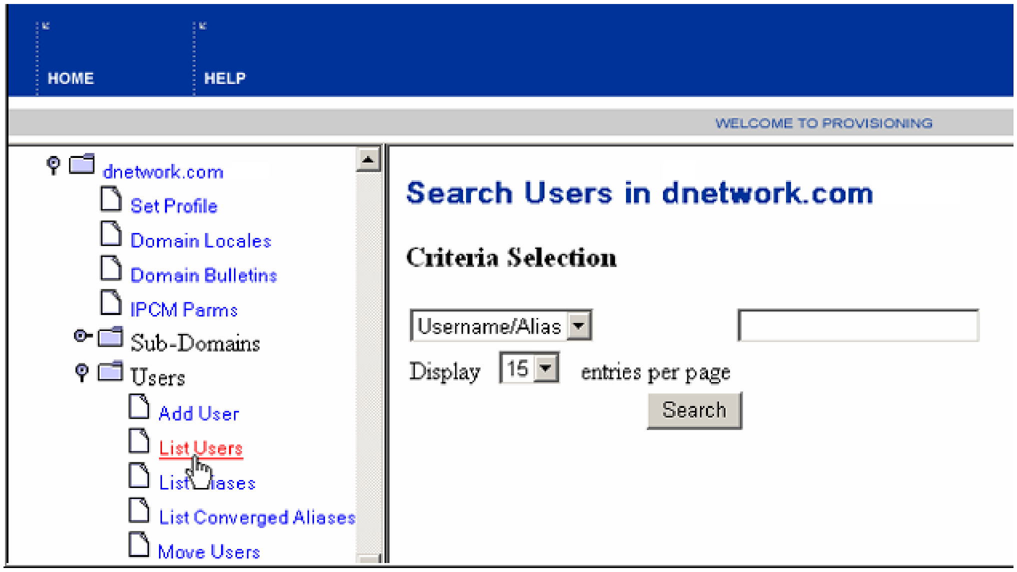
Listing, modifying, and deleting users

Use this procedure to list, modify, or delete users.

Procedure steps

1. Click the Domains menu option.
2. Under the domain or subdomain, click the User, List User menu option.

   The Search Users window appears.



3. Enter the Username or Alias or leave the field empty to search all users.
4. Define the size of your search list from the drop-down list.

   Important:
   If the number of users and user aliases is more than 1000, the search will not be successful. Refine your search criteria to get the user list.
5. Click Search to begin the search.

   The Search results window appears.
6. Click the Details link to modify user information.

   The User details window appears.
7. Click the Delete link to delete the user from the domain.

   Important:
   Before deleting a user, ensure that the user is logged off from the Multimedia PC Client and Multimedia Office Client (MOC) because the presence status of a user is not updated automatically. If users are not logged off before they are deleted from the Provisioning Client, their presence status will show up asConnected on the Multimedia PC Client and MOC.
8. Enter a valid admin password in the confirmation window to delete the user, or Cancel to cancel the operation.

   Note that the Meet Me Properties link appears on the user details page only if this service option is enabled in the user's service package.
9. Click Logout Contacts to log off the user’s registered contacts.
10. Click Save.

    A User Updated Successfully message appears.

MPS Configuration

Follow the Media Processing Server SIP 6.3 Features Manual (NN44100-130) to configure MPS for SIP. Below is the procedure to configure the changes that are required for the CS2K SIPCTI SSL support.

Configure CCSS SIP sip.conf file

Procedure steps

1. Logon to MPS SIP server.
2. Edit "/opt/ccss/etc/sip.conf" file.
3. Add an entry into "NETWORK_HOSTS" section of this file for the CS2K SSL Server.

   Let’s assume for example, CS2K SSL Server is GLYSSLSESM. Its domain name is “ssl.avaya.com” and IP Address 172.18.201.35. For MPS SIP Server to use the CS2K SSL Server in the above example, entries in Network_Hosts section will be the following.

   ```
   NETWORK_HOSTS = { !-------- ------- ------------------- ---------------------
   -------------------- ------ ----- --------- ----------------- ------ ----- --------- ---------------- !
   NAME TYPE SIP DOMAIN PRIMARY SECONDARY PORT SSA TIMER
   REGISTRAR ! NAME HOST NAME HOST NAME PORT PROFILE FLAG ! --------
   ------- ------------------- --------------------- -------------------- ------ ----- ---------
   ---------------- ------ ----- --------- ------------- GLYSSLSESM UDP ssl.avaya.com
   172.18.201.35 - 5060 - SIP_TMR1 true }
   ```

   “Type” parameter in the above examples identifies the protocol used. Set this parameter to “UDP”, as UDP is the only supported protocol. “PORT” parameter should be set to the port on which CS2K/2100/SSL receives SIP messages.

   For complete description of other parameter in the NETWORK_HOSTS section of the sip.conf, please refer Media Processing Server SIP 6.3 Features Manual (NN44100-130).

4. In the REGISTRATION_CONTROL section all SIP CTI agents must be registered for correct routing of the calls.

   Below is the example of the REGISTRATION_CONTROL section of the sip.conf.

   ```
   [REGISTRATION_CONTROL] ! REGISTRATION_SERVICE = { ! --------
   ------------------ ------------- ----------- -------------- ! SERVICE DOMAIN CONTACT
   KEEP REGISTERED ! NAME NAME EXPIRES PROFILE ! -------- ------------------
   ------------- ----------- -------------- 09726911050 ssl.avaya.com 60000
   CONTACT_01 Y 09726911051 ssl.avaya.com 60000 CONTACT_02 Y
   09726911052 ssl.avaya.com 60000 CONTACT_03 Y 09726911053
   ssl.avaya.com 60000 CONTACT_04 Y 09726911054 ssl.avaya.com 60000
   ```

CONTACT_05 Y 09726911055 ssl.avaya.com 60000 CONTACT_06 Y 09726911056 ssl.avaya.com 60000 CONTACT_07 Y 09726911057 ssl.avaya.com 60000 CONTACT_08 Y 09726911058 ssl.avaya.com 60000 CONTACT_09 Y 09726911059 ssl.avaya.com 60000 CONTACT_10 Y 09726911060 ssl.avaya.com 60000 CONTACT_11 Y 09726911061 ssl.avaya.com 60000 CONTACT_12 Y 09726911062 ssl.avaya.com 60000 CONTACT_13 Y 09726911063 ssl.avaya.com 60000 CONTACT_14 Y 09726911064 ssl.avaya.com 60000 CONTACT_15 Y 09726911065 ssl.avaya.com 60000 CONTACT_16 Y 09726911066 ssl.avaya.com 60000 CONTACT_17 Y 09726911067 ssl.avaya.com 60000 CONTACT_18 Y 09726911068 ssl.avaya.com 60000 CONTACT_19 Y 09726911069 ssl.avaya.com 60000 CONTACT_20 Y 09726911070 ssl.avaya.com 60000 CONTACT_21 Y 09726911071 ssl.avaya.com 60000 CONTACT_22 Y 09726911072 ssl.avaya.com 60000 CONTACT_23 Y 09726911073 ssl.avaya.com 60000 CONTACT_24 Y 09726911074 ssl.avaya.com 60000 CONTACT_25 Y 09726911075 ssl.avaya.com 60000 CONTACT_26 Y 09726911076 ssl.avaya.com 60000 CONTACT_27 Y 09726911077 ssl.avaya.com 60000 CONTACT_28 Y 09726911078 ssl.avaya.com 60000 CONTACT_29 Y 09726911079 ssl.avaya.com 60000 CONTACT_30 Y 09726911080 ssl.avaya.com 60000 CONTACT_31 Y 09726911081 ssl.avaya.com 60000 CONTACT_32 Y 09726911082 ssl.avaya.com 60000 CONTACT_33 Y 09726911083 ssl.avaya.com 60000 CONTACT_34 Y 09726911084 ssl.avaya.com 60000 CONTACT_35 Y 09726911085 ssl.avaya.com 60000 CONTACT_36 Y 09726911086 ssl.avaya.com 60000 CONTACT_37 Y 09726911087 ssl.avaya.com 60000 CONTACT_38 Y 09726911088 ssl.avaya.com 60000 CONTACT_39 Y 09726911089 ssl.avaya.com 60000 CONTACT_40 Y 09726911090 ssl.avaya.com 60000 CONTACT_41 Y 09726911091 ssl.avaya.com 60000 CONTACT_42 Y 09726911092 ssl.avaya.com 60000 CONTACT_43 Y 09726911093 ssl.avaya.com 60000 CONTACT_44 Y 09726911094 ssl.avaya.com 60000 CONTACT_45 Y 09726911095 ssl.avaya.com 60000 CONTACT_46 Y 09726911096 ssl.avaya.com 60000 CONTACT_47 Y 09726911097 ssl.avaya.com 60000 CONTACT_48 Y 09726911098 ssl.avaya.com 60000 CONTACT_49 Y 09726911099 ssl.avaya.com 60000 CONTACT_50 Y 09726911100 ssl.avaya.com 60000 CONTACT_51 Y 09726911101 ssl.avaya.com 60000 CONTACT_52 Y 09726911102 ssl.avaya.com 60000 CONTACT_53 Y 09726911103 ssl.avaya.com 60000 CONTACT_54 Y 09726911104 ssl.avaya.com 60000 CONTACT_55 Y 09726911105 ssl.avaya.com 60000 CONTACT_56 Y 09726911106 ssl.avaya.com 60000 CONTACT_57 Y 09726911107 ssl.avaya.com 60000 CONTACT_58 Y 09726911108 ssl.avaya.com 60000 CONTACT_59 Y 09726911109 ssl.avaya.com 60000 CONTACT_60 Y }

Below is the example of the corresponding CONTACT_PROFILE section for the above agents.

CONTACT_PROFILE = { ! ---------- ---------- ---------- ------ --------- ! PROFILE USER ! NAME NAME TRANSPORT PORT EXPIRES ! ---------- ---------- ---------- ------ --------- CONTACT_01 09726911050 UDP 5060 60000 CONTACT_02 09726911051 UDP 5060 60000 CONTACT_03 09726911052 UDP 5060 60000

CONTACT_04 09726911053 UDP 5060 60000 CONTACT_05 09726911054 UDP 5060 60000 CONTACT_06 09726911055 UDP 5060 60000 CONTACT_07 09726911056 UDP 5060 60000 CONTACT_08 09726911057 UDP 5060 60000 CONTACT_09 09726911058 UDP 5060 60000 CONTACT_10 09726911059 UDP 5060 60000 CONTACT_11 09726911060 UDP 5060 60000 CONTACT_12 09726911061 UDP 5060 60000 CONTACT_13 09726911062 UDP 5060 60000 CONTACT_14 09726911063 UDP 5060 60000 CONTACT_15 09726911064 UDP 5060 60000 CONTACT_16 09726911065 UDP 5060 60000 CONTACT_17 09726911066 UDP 5060 60000 CONTACT_18 09726911067 UDP 5060 60000 CONTACT_19 09726911068 UDP 5060 60000 CONTACT_20 09726911069 UDP 5060 60000 CONTACT_21 09726911070 UDP 5060 60000 CONTACT_22 09726911071 UDP 5060 60000 CONTACT_23 09726911072 UDP 5060 60000 CONTACT_24 09726911073 UDP 5060 60000 CONTACT_25 09726911074 UDP 5060 60000 CONTACT_26 09726911075 UDP 5060 60000 CONTACT_27 09726911076 UDP 5060 60000 CONTACT_28 09726911077 UDP 5060 60000 CONTACT_29 09726911078 UDP 5060 60000 CONTACT_30 09726911079 UDP 5060 60000 CONTACT_31 09726911080 UDP 5060 60000 CONTACT_32 09726911081 UDP 5060 60000 CONTACT_33 09726911082 UDP 5060 60000 CONTACT_34 09726911083 UDP 5060 60000 CONTACT_35 09726911084 UDP 5060 60000 CONTACT_36 09726911085 UDP 5060 60000 CONTACT_37 09726911086 UDP 5060 60000 CONTACT_38 09726911087 UDP 5060 60000 CONTACT_39 09726911088 UDP 5060 60000 CONTACT_40 09726911089 UDP 5060 60000 CONTACT_41 09726911090 UDP 5060 60000 CONTACT_42 09726911091 UDP 5060 60000 CONTACT_43 09726911092 UDP 5060 60000 CONTACT_44 09726911093 UDP 5060 60000 CONTACT_45 09726911094 UDP 5060 60000 CONTACT_46 09726911095 UDP 5060 60000 CONTACT_47 09726911096 UDP 5060 60000 CONTACT_48 09726911097 UDP 5060 60000 CONTACT_49 09726911098 UDP 5060 60000 CONTACT_50 09726911099 UDP 5060 60000 CONTACT_51 09726911100 UDP 5060 60000 CONTACT_52 09726911101 UDP 5060 60000 CONTACT_53 09726911102 UDP 5060 60000 CONTACT_54 09726911103 UDP 5060 60000 CONTACT_55 09726911104 UDP 5060 60000 CONTACT_56 09726911105 UDP 5060 60000 CONTACT_57 09726911106 UDP 5060 60000 CONTACT_58 09726911107 UDP 5060 60000 CONTACT_59 09726911108 UDP 5060 60000 CONTACT_60 09726911109 UDP 5060 60000 }

The following REG_LINE_CONFIG section is required. It maps the SIP ACD agents to unique phone lines. It enables the CTI server to correctly receive the CTI events. The CTI configuration must match the phone lines numbers assigned in the sip.conf. Please note that the span 2 line 1 will correspond to line 31 in CTI configuration. Span 3 line 1 will correspond to line 61 in CTI configuration and so on.

REG_LINE_CONFIG = { ! ------------------ -------- ------ ---------- ! SERVICE GROUP HOST SPAN LINE ! NAME ID ID RANGE ! ------------------ -------- ------ ---------- 09726911050 1 1 1 09726911051 1 1 2 09726911052 1 1 3 09726911053 1 1 4 09726911054 1 1 5 09726911055 1 1 6 09726911056 1 1 7 09726911057 1 1 8 09726911058 1 1 9 09726911059 1 1 10 09726911060 1 1 11 09726911061 1 1 12 09726911062 1 1 13 09726911063 1 1 14 09726911064 1 1 15 09726911065 1 1 16 09726911066 1 1 17 09726911067 1 1 18 09726911068 1 1 19 09726911069 1 1 20 09726911070 1 1 21 09726911071 1 1 22 09726911072

1 1 23 09726911073 1 1 24 09726911074 1 1 25 09726911075 1 1 26 09726911076 1 1 27 09726911077 1 1 28 09726911078 1 1 29 09726911079 1 1 30 09726911080 1 2 1 09726911081 1 2 2 09726911082 1 2 3 09726911083 1 2 4 09726911084 1 2 5 09726911085 1 2 6 09726911086 1 2 7 09726911087 1 2 8 09726911088 1 2 9 09726911089 1 2 10 09726911090 1 2 11 09726911091 1 2 12 09726911092 1 2 13 09726911093 1 2 14 09726911094 1 2 15 09726911095 1 2 16 09726911096 1 2 17 09726911097 1 2 18 09726911098 1 2 19 09726911099 1 2 20 09726911100 1 2 21 09726911101 1 2 22 09726911102 1 2 23 09726911103 1 2 24 09726911104 1 2 25 09726911105 1 2 26 09726911106 1 2 27 09726911107 1 2 28 09726911108 1 2 29 09726911109 1 2 30 }

Please refer Media Processing Server SIP 6.3 Features Manual for other configuring other sections of sip.conf.

5. Using SIPUI, create password for all phone lines entries in the REG_LINE_CONFIG.

   Below are the steps to create password using SIPUI.

   a. sipui
   b. cd mps
   c. cd register
   d. passwd new

6. Enter the username (phone number), password and domain when prompted.
7. Configure CCTIVR.

   This is described in MPS with CCTIVR Configuration and Interfaces Release Notes (NN44100-407).

CTI TDM Configuration

The following tables contain the CSS configuration files used for CS 2000 testing.

SCAIGRP Table Sample

Table 12: SCAIGRP Table Sample

TABLE: SCAIGRP SCAIGNAM PASSWORD NETNODID BGID ____ VERIF_SCAI_1 TESTER 25 LOCAL 2 (LINKSET (VERIF1)(VERIF2)(VERIF3)(VERIF4)(VERIF5)$)$

Legend:

- SCAIGNAM - Linkset Name
- PASSWORD - Password
- NETNODID - Network ID
- BGID - Business Group ID

SCAIPROFS Table Sample

Table 13: SCAIPROFS Table Sample

TABLE: SCAIPROF LinkSet: VERIF2 used when going through Link Plexer. VERIF4 used for Direct connect to the switch. VERIF_SCAI_1 1 (CTXEVENT10$) (ACDEVENT11$) (ROUTING35$) (TPCC08$) (RESOURCE35$) (TPAC36$) (CALLINIT07$) (SCAI3WC07$) (SCAIMWTI07$) (DNQUERY07$) (SCAICC08$) (TPQC10$) (ICCM10$) $ CTEXPO 1 (CTXEVENT10$) (ACDEVENT36$) (ROUTING35$) (TPCC08$) (RESOURCE35$) (TPAC36$) (CALLINIT07$) (SCAI3WC07$) (SCAIMWTI07$) (DNQUERY07$) (SCAICC12$) $ VERIF2 1 (CTXEVENT13$) (ACDEVENT12$) (ROUTING35$) (TPCC11$) (RESOURCE11$) (TPAC12$) (CALLINIT07$) (SCAI3WC09$) (SCAIMWTI13$) (DNQUERY07$) (SCAICC08$) (TPQC10$) (ICCM10$) $ VERIF2 1 (Updated Version – used for LinkPlexer) (CTXEVENT13$) (ACDEVENT12$) (ROUTING35$) (TPCC11$) (RESOURCE11$) (TPAC13$) (CALLINIT07$) (SCAI3WC09$) (SCAIMWTI13$) (DNQUERY07$) (SCAICC08$) (TPQC10$) (ICCM10$) $ VERIF3 1 (CTXEVENT10$) (ACDEVENT11$) (ROUTING35$) (TPCC08$) (RESOURCE35$) (TPAC36$) (CALLINIT07$) (SCAI3WC07$) (SCAIMWTI07$) (DNQUERY07$) (SCAICC08$) (TPQC10$) (ICCM10$) $ CTEXPO_TCP 1 (CTXEVENT10$) (ACDEVENT36$) (ROUTING35$) (TPCC08$) (RESOURCE35$) (TPAC36$) (CALLINIT07$) (SCAI3WC07$) (SCAIMWTI07$) (DNQUERY07$) (SCAICC08$) $ VERIF4 1(Original for LinkPlexer connections) (CTXEVENT10$) (ACDEVENT11$) (ROUTING35$) (TPCC08$) (RESOURCE35$) (TPAC36$) (CALLINIT07$) (SCAI3WC07$) (SCAIMWTI07$) (DNQUERY07$) (SCAICC08$) (TPQC10$) . .

Legend:

- VERIF_SCAI_1 - Linkset Name
- 1 - Service ID
- (SCAICC12$) - Service Version

SCAISSRV Table Sample

Table 14: SCAISSRV Table Sample

TABLE: SCAISSRV TOP SUBSERV SPROFILE --- CTXEVENT34$ CTXEVENT (CALLOFFR Y Y N N Y Y Y N N Y Y Y Y N) (CALLANSWR Y Y Y N N Y Y Y N N Y Y Y N N) (CALLREL Y Y Y N) $ ACDEVENT34$ ACDEVENT (CALLQUED Y Y Y Y Y Y Y Y Y N) (CALLOFFR Y Y Y Y Y Y Y Y Y N) (CALLANSWR Y Y Y Y Y Y Y Y Y N) (CALLREL Y Y Y Y Y N) $ ROUTING34$ ROUTING (CALLRECDC Y Y Y Y Y Y Y Y Y N) (CALLREDIR Y Y Y)$ TPCC34$ TPCC (ADDPTY Y Y Y N) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) (MAKECALL Y Y) $ CTXEVENT35$ CTXEVENT (CALLOFFR Y Y N N Y Y Y N N Y Y Y Y N) (CALLANSWR Y Y Y N N Y Y Y N N Y Y Y N N) (CALLREL Y Y Y N) $ ACDEVENT35$ ACDEVENT (CALLQUED Y Y Y Y Y Y Y Y Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y Y Y N) $ ROUTING35$ ROUTING (CALLRECDC Y Y Y Y Y Y Y Y Y Y) (CALLREDIR Y Y Y)$ TPCC35$ TPCC (ADDPTY Y Y Y N) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) (MAKECALL Y Y) $ RESOURCE35$ RESOURCE (ACDQUERY)$ ACDEVENT36$ ACDEVENT (CALLQUED Y Y Y Y Y Y Y Y Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y Y Y N) (AGTLGDIN Y Y Y) (AGTLGDOUT Y Y) (AGTREADY Y Y) (AGTNREADY Y Y N) $ TPAC36$ TPAC (LOGINAGT Y Y N) (LOGOUTAGT Y) (READYAGT Y) (NREADYAGT Y N) $ CTXEVENT07$ CTXEVENT (SETOFFHK Y Y N) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y Y N N) (CALLREL Y Y Y N) $ RESEVENT07$ RESEVENT (SETOFFHK Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLREL Y Y Y) $ CALLINIT07$ CALLINIT (MAKECALL Y Y)$ SCAI3WC07$ SCAI3WC (ADDPTY Y Y Y) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) $ SCAIMWTI07$ SCAIMWTI (MSGWAIT Y Y Y)$ DNQUERY07$ DNQUERY (DNQUERY Y)$ SCAICC08$ SCAICC (HOLDCALL) (UNHOLDCALL) (CALLUNHELD Y) (ANSWCALL) (RELSCALL Y) $ TPCC08$ TPCC (HOLDCALL) (UNHOLDCALL) (CALLUNHELD Y) (ANSWCALL) (RELSCALL Y) (ADDPTY Y Y Y N) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) (MAKECALL Y Y) $ TPCC09$ TPCC (HOLDCALL) (UNHOLDCALL) (CALLUNHELD Y) (ANSWCALL) (RELSCALL Y) (CONSULTEV Y) (CONFEVNT Y) (TRANSFEREV Y) (ADDPTY Y Y Y N) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) (MAKECALL Y Y) $ SCAI3WC09$ SCAI3WC (CONSULTEV Y) (CONFEVNT Y) (TRANSFEREV Y) (ADDPTY Y Y Y) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y) $ TPQC10$ TPQC (ROUTECALL) (GIVETRMT) (TRMTCOMP)$ ICCM10$ ICCM (SETCDNST)$ CTXEVENT10$ CTXEVENT (SETOFFHK Y Y N) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y N N) (CALLREL Y Y Y N) (CALLNAME Y Y Y Y Y) $ RESEVENT10$ RESEVENT (SETOFFHK Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y N) (CALLREL Y Y Y) (CALLNAME Y Y Y Y Y) $ ACDEVENT11$ ACDEVENT (CALLQUED Y Y Y Y Y Y Y Y Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y Y Y N) (AGTLGDIN Y Y Y) (AGTLGDOUT Y Y) (AGTREADY Y Y) (AGTNREADY Y Y N) (LOBEVENT Y Y Y) (EMKEVENT Y Y Y Y Y Y) $ TPCC11$ TPCC (HOLDCALL) (UNHOLDCALL) (CALLUNHELD Y) (ANSWCALL) (RELSCALL Y) (CONSULTEV Y) (CONFEVNT Y) (TRANSFEREV Y) (ADDPTY Y Y Y Y) (CONFPTY Y) (DROPPTY Y Y) (TRANPTY Y)

```
(MAKECALL Y Y) $ RESOURCE11$ RESOURCE (ACDQUERY ) (APPSTQRY Y Y Y Y Y)
$ TPAC12$ TPAC (LOGINAGT Y Y N) (LOGOUTAGT Y) (READYAGT Y) (NREADYAGT Y
N) (RESERVEAGT Y Y) (UNRESERVEAGT Y) $ ACDEVENT12$ ACDEVENT
(CALLQUED Y Y Y Y Y Y Y Y Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y) (CALLANSWR
Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y Y Y Y) (AGTLGDIN Y Y Y) (AGTLGDOUT Y Y)
(AGTREADY Y Y) (AGTNREADY Y Y N) (LOBEVENT Y Y Y) (EMKEVENT Y Y Y Y Y Y)
(AGTSETACT Y Y N) $ SCAIMWTI13$ SCAIMWTI (MSGWAIT Y Y Y) ( MWTACT Y Y Y)
$ CTXEVENT13$ CTXEVENT (SETOFFHK Y Y Y) (CALLOFFR Y Y Y Y Y Y Y Y Y Y Y Y
Y Y Y) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y Y) (CALLNAME
Y Y Y Y Y) $ RESEVENT13$ RESEVENT (SETOFFHK Y Y) (CALLOFFR Y Y Y Y Y Y Y
Y Y Y Y Y Y Y) (CALLANSWR Y Y Y Y Y Y Y Y Y Y Y Y Y Y) (CALLREL Y Y Y) (CALLNAME Y
Y Y Y Y) $ TPAC13$ TPAC (LOGINAGT Y Y N) (LOGOUTAGT Y) (READYAGT Y)
(NREADYAGT Y N) (RESERVEAGT Y Y) (UNRESERVEAGT Y) (CHGVWRAP Y Y)
(CHGFORCE Y Y) $ ICCM13$ ICCM (SETCDNST ) (REASNAGT Y Y)$ TPAC09$ TPAC
(LOGINAGT Y Y Y) (LOGOUTAGT Y) (READYAGT Y) (NREADYAGT Y N) $ ROUTING09$
ROUTING (CALLRECDC Y Y Y Y Y Y Y Y Y Y) (CALLREDIR Y Y N)$
```

SCAICOMS Table Sample

Table 15: SCAICOMS Table Sample

```
>table scaicoms TABLE: SCAICOMS >list all TOP LINKSET SCAILNKS OPTIONS
-------------------------------------------------------------------------- JHANSEN X25 (MPC ( 1 3
11111111 0 0 0 0 SVC) $)$ MN MN $ VERIF1 X25 (MPC ( 2 3 11111111 0 0 0 0 SVC) $)$
MN MN $ CTEXPO X25 (MPC ( 3 3 11111111 0 0 0 0 SVC) $)$ MN MN $ VERIF_SCAI_1
TCP 47 106 100 141 N $ JHANSEN2 TCP 47 106 100 143 N $ JHANSEN3 TCP 47 105
192 165 N $ VERIF3 TCP 47 106 100 141 N $ CTEXPO_TCP TCP 47 105 193 80 N $
VERIF4 TCP 47 106 100 142 N $ VERIF5 TCP 47 105 193 101 N $ JHANSEN4 TCP 47
105 193 168 N $
```

Legend:

- VERIF_SCAI_1 - TCP Linkset Name
- 47 106 100 141 - IP Address

Sample Configuration of IVR Ports Device and Position ID

Directory Number	LEN	Agent/Position ID
9321801	02 0 06 00	1801
9321802	02 0 06 01	1802

Directory Number	LEN	Agent/Position ID
9321803	02 0 06 02	1803
9321804	02 0 06 03	1804
9321805	02 0 06 04	1805
9321806	02 0 06 05	1806
9321807	02 0 06 06	1807

-- LEN: (See Above Table - LEN entry) TYPE: SINGLE PARTY LINE SNPA: 612 DIRECTORY NUMBER: (See Above Table entry – Directory Number) (NON-UNIQUE) LINE CLASS CODE: IBN IBN TYPE: STATION CUSTGRP: VERIF1 SUBGRP: 0 NCOS: 0 SIGNALLING TYPE: DIGITONE CARDCODE: 5D11AA GND: N PADGRP: NPDGP BNV: NL MNO: N PM NODE NUMBER : 19 PM TERMINAL NUMBER : 193 DNGRPS OPTIONS: NETNAME:PUBLIC NONUNIQUE OPTIONS: CWT 3WC DGT CNAMD NOAMA CLI ACD JIM_PERI_1_IVR 0 Y (See Above Table entry – Agent ID) ACDNR

Directory Number	LEN	Agent/Position ID
9321820	02 0 06 07	1810
9321821	02 0 06 08	1811
9321822	02 0 06 09	1812
9321823	02 0 06 10	1813
9321824	02 0 06 11	1814
9321825	02 0 06 12	1815
9321826	02 0 06 13	1816

-- LEN: (See Above Table - LEN entry) TYPE: SINGLE PARTY LINE SNPA: 612 DIRECTORY NUMBER: (See Above Table entry – Directory Number) (NON-UNIQUE) LINE CLASS CODE: IBN IBN TYPE: STATION CUSTGRP: VERIF1 SUBGRP: 0 NCOS: 0 SIGNALLING TYPE: DIGITONE CARDCODE: 5D11AA GND: N PADGRP: NPDGP BNV: NL MNO: N PM NODE NUMBER : 19 PM TERMINAL NUMBER: 193 DNGRPS OPTIONS: NETNAME:PUBLIC NONUNIQUE OPTIONS: CWT 3WC DGT CNAMD NOAMA CLI ACD JIM_PERI_2_IVR 0 Y (See Above Table entry – Agent ID) ACDNR

Sample Configuration for Agent Position ID on CS 2000

Directory Number	LEN	Agent/Position ID
9321827	02 0 06 14	1817
9321830	02 0 06 15	1830

-- LEN: (See Above Table - LEN entry) TYPE: SINGLE PARTY LINE SNPA: 612 DIRECTORY NUMBER: (See Above Table entry – Directory Number) (NON-UNIQUE) LINE CLASS CODE: IBN IBN TYPE: STATION CUSTGRP: VERIF1 SUBGRP: 0 NCOS: 0 SIGNALLING TYPE: DIGITONE CARDCODE: 5D11AA GND: N PADGRP: NPDGP BNV: NL MNO: N PM NODE NUMBER : 19 PM TERMINAL NUMBER: 193 DNGRPS OPTIONS: NETNAME:PUBLIC NONUNIQUE OPTIONS: CWT 3WC DGT CNAMD NOAMA CLI ACD JIM_PERI_3_IVR 0 Y (See Above Table entry – Agent ID) ACDNR

Sample Configuration for ACD Group on CS 2000

ACDSHOW>>status group jim_peri_1_ivr
== Status For ACD Group JIM_PERI_1_IVR Call Transfer Recall Time: 45 Secs Call Park Recall Time: 0 Secs Current Call Queue Size: 0 Max Call Queue Size: 8 Current Call Wait Time: 0 Secs Max Call Wait Time: 1800 Secs Number of Agent Positions Assigned: 7 Agents Logged In: 0 Agents Not Logged In: 76 Agtpos On ACD Calls: 0 Agtpos Idle: 0 Agtpos In Not Ready Mode: 0

>status group jim_peri_2_ivr
== Status For ACD Group JIM_PERI_2_IVR Call Transfer Recall Time: 45 Secs Call Park Recall Time: 0 Secs Current Call Queue Size: 0 Max Call Queue Size: 8 Current Call Wait Time: 0 Secs Max Call Wait Time: 1800 Secs Number of Agent Positions Assigned: 8 Agents Logged In: 0 Agents Not Logged In: 8 Agtpos On ACD Calls: 0 Agtpos Idle: 0 Agtpos In Not Ready Mode: 0

> status group jim_peri_3_ivr
== Status For ACD Group JIM_PERI_3_IVR Call Transfer Recall Time: 45 Secs Call Park Recall Time: 0 Secs Current Call Queue Size: 0 Max Call Queue Size: 8 Current Call Wait Time: 0 Secs Max Call Wait Time: 1800 Secs Number of Agent Positions Assigned: 2 Agents Logged In: 0 Agents Not Logged In: 2 Agtpos On ACD Calls: 0 Agtpos Idle: 0 Agtpos In Not Ready Mode: 0

Chapter 7: Protocols

This chapter covers:

1. Introduction
2. T1 Protocols
3. E1 Protocols

Introduction

The MPS and the Avaya PBX function with a number of common protocols. The most important protocols the MPS and the Avaya PBX use to communicate are T1 Protocols and E1 Protocols.

T1 Protocols

T1 is a digital transmission link with a capacity of 1.544 Mb/s. T1 is the standard for digital telephony in North America. One T1 span normally carries 24 voice channels. The most common method of encoding voice data into a digital bit stream is Pulse Code Modulation (PCM).

Lineside T1 Protocol

The Lineside T1 supports D4 or ESF channel framing formats as well as AMI or B8ZS coding. D4 or ESF can be used with either of the coding formats. Robbed bit A/B bit signaling is used, and loopstart and groundstart are supported. These protocols are DIP switch configurable.

The Lineside T1 supports up to 24 channels, and occupies two line card slots. The ports in the Avaya M1/ CS 1000 PBX are configured as Analog Line Cards. The Supervise Analog Line Feature of the Avaya M1/CS 1000 PBX is used to support cut-off on disconnect supervision.

For more information on the Lineside T1 protocol and its configuration, refer to the Avaya Media Processing Server Series Telephony Reference Manual.

To ensure accurate functionality, the Lineside T1 (switch) and the MPS must be configured as follows.

Super Frame Configuration in Loopstart Mode

This is the [D4(SF)/AMI/loopstart] mode:

Lineside T1 Switch Configuration

On the Lineside T1 card, set the appropriate DIP switches to:

1. Configure the Lineside T1 switch for a Framing/Coding combination of D4(SF)/AMI.
2. Set the mode to Loopstart.

MPS Configuration for Loopstart Signaling—Super Frame

To configure the MPS with the Lineside T1 protocol (for a single span), make the following entry in the tms.cfg file:

```
[DTCMAP]
;-----------------------------------------------------------------
;    TMS  PLI  Span svc_type  MpsNum Outline  Pool/  Protocol
;    Num  Slot Num                            class  Pkg
;------------------------------------------------------------------
LOAD 1        4    1    CAS       -        -        -
linesideT1proto.cfg
```

Note:

No other configuration apart from this is essential because the MPS defaults to a Framing/Coding combination of D4(SF)/AMI with the mode set to Loopstart.

Super Frame Configuration in Groundstart Mode

This is the [D4(SF)/AMI/groundstart] mode:

Lineside T1 Switch Configuration

On the Lineside T1 switch, set the appropriate DIP switches to:

1. Configure the Lineside T1 switch for a Framing/Coding combination of D4(SF)/AMI.
2. Set the mode to Groundstart.

MPS Configuration for Groundstart Signaling—Super Frame

For information on configuration, go to MPS Configuration for Loopstart Signaling—Extended Super Frame on page 147.

To configure the MPS for Groundstart signaling, modify the linesideT1_proto.cfg file as follows:

```
;
; Mode of Operation (Groundstart = 1;
; Loopstart = 0; Default 0)
;
;PARAM TIM_param_10 = 0
PARAM TIM_param_10 = 1 (add this line)
;
```

Extended Super Frame Configuration in Loopstart Mode

This is the [ESF/B8ZS/Loopstart] mode:

Lineside T1 Switch Configuration

On the Lineside T1 switch, set the appropriate DIP switches to:

1. Configure the Lineside T1 switch for a Framing/Coding combination of ESF/B8ZS.
2. Set the mode to Loopstart.

MPS Configuration for Loopstart Signaling—Extended Super Frame

To configure the MPS with the Lineside T1 protocol (for a single span), make the following entry in the tms.cfg file:

```
[DTCMAP]
;----------------------------------------------------------------
;    TMS  PLI  Span svc_type  MpsNum Outline  Pool/  Protocol
;    Num  Slot Num                            class  Pkg
;-----------------------------------------------------------------
LOAD 1        4    1    CAS       -        -        -
linesideT1proto.cfg
```

To configure the MPS for a Framing/Coding combination of ESF/B8ZS, modify the linesideT1_proto.cfg file as follows:

Span Class

The span class is a special class of resource for the proto.cfg file. It specifies the information used to load the span. If more than one span class section is specified the first one found is used and subsequent specifications are ignored

```
;
;[SPAN_CLASS]
CLASS_NAME = linesideT1
;CDF = t1_sf_cas.cdf  ; what .cdf to load
CDF = t1_esf_cas.cdf  ; what .cdf to load  (add this line)
STATE_TBL= linesideT1.bin
;
```

Note:
No configuration is necessary on the MPS for the Loopstart because this is the default operational mode.

Extended Super Frame Configuration in Groundstart Mode

Lineside T1 Switch Configuration

On the Lineside T1 switch, set the appropriate DIP switches to:

1. Configure the Lineside T1 switch for a Framing/Coding combination of ESF/B8ZS.
2. Set the mode to Groundstart.

MPS Configuration for Groundstart Signaling—Extended Super Frame

For information on configuration, go toMPS Configuration for Groundstart Signaling—Super Frame on page 146.

To configure the MPS for Groundstart signaling, modify the linesideT1_proto.cfg file as follows:

```
;
; Mode of Operation (Groundstart = 1;
; Loopstart = 0; Default 0)
;
;PARAM TIM_param_10 = 0
PARAM TIM_param_10 = 1 (add this line)
```

E1 Protocols

E1 or Conference of European Postal and Telecommunications (CEPT) is a major telephony standard used in Europe and parts of Asia. The CEPT standard refers to the 30 Channel Pulse Code Modulation (PCM-30) interface protocols used in Europe. One E1 span carries 30 simultaneous voice channels with a bit-transmission rate of 2.048 Mb/s. There are actually 32 slots carried, but two are used for signaling and other control functions.

Lineside E1 Protocol

The Lineside E1 requires two card slots and supports up to 30 channels, configured as two analog line cards in the Avaya M1/CS 1000 PBX. The E1 interface can be either a 120 ohm or a 75 ohm coaxial connection. The Supervise Analog Line Feature of the Avaya M1/CS 1000 PBX is used to support cut-off on disconnect supervision.

Loopstart and Groundstart are supported. The Australian P2 signaling scheme is also supported. You can set the DIP switches to make the signaling scheme custom configured to meet the requirements of the device to which it connects. All protocols are DIP switch configurable, with the custom configuration needing Man Machine Interface (MMI) commands to define the protocol.

The Lineside E1 can support CRC-4 or FAS framing with line coding of either AMI or HDB3. Both interface cards can support a direct connection to a device from 0 to 655 feet (199.6 meters).

For more information on the Lineside E1 protocol and its configuration, refer to the Avaya Media Processing Server Series Telephony Reference Manual.

Non-CRC4(MF/FAS) Configuration in Loopstart Mode

This is the [MF(FAS)/HDB3/Loopstart] mode:

Lineside E1 Switch Configuration

On the Lineside E1 switch, set the appropriate DIP switches to:

1. Configure the Lineside E1 switch for a Framing/Coding combination of FAS/HDB3.
2. Set the mode to Loopstart.

MPS Configuration

To configure the MPS with Lineside E1 protocol (for a single span), make the following entry in the tms.cfg file:

```
[DTCMAP]
;-----------------------------------------------------------------
;    TMS  PLI  Span svc_type  MpsNum Outline  Pool/  Protocol
;    Num  Slot Num                           class  Pkg
;------------------------------------------------------------------
LOAD 1        4    1    CAS       -        -        -
linesideE1proto.cfg
```

Note:
No other configuration is essential because the MPS defaults to a Framing/ Coding combination of MF/ HDB3 with the mode set to Loopstart.

Non-CRC4(MF/FAS) Configuration in Groundstart Mode

This is the [MF(FAS)/HDB3/Groundstart] mode:

Lineside E1 Switch Configuration

On the Lineside E1 switch, set the appropriate DIP switches to:

1. Configure the Lineside E1 switch for a Framing/Coding combination of FAS/HDB3.
2. Set the mode to Groundstart.

MPS Configuration

For information on configuration, see the preceding Non-CRC4(MF/FAS) Configuration in Loopstart Mode on page 149.

In addition, to configure the MPS for Groundstart signaling, modify the linesideE1_proto.cfg file as follows:

```
;
; Mode of Operation (Groundstart = 1;
; Loopstart = 0; Default 0)
;
;PARAM TIM_param_10 = 0
PARAM TIM_param_10 = 1  (add this line)
;
```

M1/CS 1000 PBX Settings for MPS E1 Lineside Card

The E1 Lineside card (NT5D33/ NT5D34) is an expansion capability for Avaya M1/CS 1000 PBXs, which allows a digital E1 span of 30 lines to emulate 30 ports of analog telephony from the switch perspective. As with any CAS-based signaling, call control in the Lineside card is provided through a combination of line signals and register signals. In general, CAS seeks to provide an approximate emulation of the functionality that the analog line provides.

Line signals provide information on the state of each end of the emulated analog line, such as off-hook, originating call, answered call, and so on. This information is conveyed through a series of signaling bits sent for each individual line or channel on a 30-channel E1 system.

Procedure to Access E1 Lineside Card Configuration

Perform the following steps to access E1 Lineside card configuration:

1. Connect a VT100 from the card to the P5 cable. (Serial settings are usually 2400 7 M 1)
2. Standby for an LE1 prompt.
3. Type L to login.
4. Type LE1LINK for password (default).
5. Select S S.

6. Disable simple call clearing.
7. Select S M for mode.
8. Configure the bit pattern as indicated in the following table to get the span to work with the IVR correctly.

Call State	Bit Pattern	Note
Idle Send	1101	
Idle Receive	1101	
Blocking Receive	no	
Ringer On	1011	MELCAS does not support "intermittent" ringer
Ringer Off	1001	Ring-count may be controllable by setting this to 1101
Offhook Receive	0101	
Connect Send	0101	
Disconnect Send	1101	
CPE Disconnect Receive	1101	
Seize Receive	0101	
Seize Acknowledge	no	MELCAS has no seizure acknowledge
dial make	0101	
dial break	0001	
answer send	0101	
disconnect receive	1101	
CPE disconnect send	1101	
Intercall Timer	800ms	May be set to a minimum of 100ms, which can be increased to at least 250-300. General range may go up to 2000 ms.
Disconnect Timer	200ms	May be set to up to 800ms

Note:

The number of significant receiving bits is 4, and this enables all bits—A, B, C, D. All 4 bits must be enabled for the protocol to work.

Sample procedure to access E1 Lineside card configuration

```
Frederick Engineering FELINE WinXL 3.00 Build 46 Release
Buffer Printout: 3rd trace.buf
Configuration:
Protocol: BIT ORIENTED
Bit Rate (BPS): 9600 Clock Source: External
Code Set: ASCII 8 Suppress: None
Parity: None Mode: Sync NRZ
Sel. Addr.: All
Timestamp Mode: None Buffer Mode: Continuous
Timestamp Resolution10 usec. Disk Mode Continuous
Timestamp Format: Relative File Sized: All
Mon / Sim: Monitor
Lead Names
-----------
```

IN 1 = In 1	OUT 1 = Out 1
IN 2 = In 2	OUT 2 = Out 2
IN 3 = In 3	OUT 3 = Out 3
IN 4 = In 4	OUT 4 = Out 4

```
Start Date: 12/30/2004 Start Time: 13:14:08.000
End Date: 12/30/2004 End Time: 13:15:15.000
National ISDN-1 PRI Detail
DTE Frame 1 (42 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 1
Frame Type = INFO
N(S) = 1, N(R) = 1, P/F = 0
NISDN:
Protocol Discriminator = 8 (Q.931)
Call Reference: Length = 1, Flag = Origination, Value = 21
Message Type = SETUP
Info Element = Bearer Capability
IE Length = 3
Transfer Capability = Speech
Transfer Mode/Rate = 64 kbps, Circuit mode
Layer 1 Protocol = mu-law (Rec G.711)
Info Element = Channel Identification
IE Length = 4
Interface Id Present = Interface explicitly defined
Interface Type = Primary Rate Interface
Preferred/Exclusive = Exclusive, only indicated channel     acceptable
D-Channel Indicator = No
Channel Select = As indicated
Interface Id = 0
Number/Slot Map = Channel Number
Channel Number = 21
```

```
Info Element = Progress Indicator
IE Length = 2
Coding Standard = ITU-TS Standard
General Location = Private Network serving Local User
Progress Descriptor = Call is not end-to-end ISDN
Info Element = Calling Party Number
IE Length = 7
Number Type/Plan = Abbreviated number in private plan
Origin/Presentation = User number allowed; not screened
Calling Number = 32555
Info Element = Called Party Number
IE Length = 6
Number Type/Plan = Abbreviated number in private plan
Called Number = 33000
=========================================================
DCE  Frame 2 (15 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = INFO
N(S) = 1, N(R) = 2, P/F = 0
NISDN:
Protocol Discriminator = 8 (Q.931)
Call Reference: Length = 1, Flag = Destination, Value = 21
Message Type = RELease COMplete
Info Element = Cause
IE Length = 3
Coding Standard = ITU-TS Standard
General Location = User
Cause Class = Protocol error
Cause Value = Invalid information element contents
Diagnostics:
Info Element = Change Status
=========================================================
DTE  Frame 3 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 0
=========================================================
DTE Frame 4 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 1
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DCE Frame 5 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 1
National ISDN-1 PRI Detail
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DCE Frame 6 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DTE Frame 7 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DCE Frame 8 (6 bytes) was captured with no errors.
```

```
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DTE Frame 9 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DCE Frame 10 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DTE Frame 11 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DCE Frame 12 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
DTE Frame 13 (6 bytes) was captured with no errors.
LAPD:
SAPI = 0, TEI = 0, C/R = 0
Frame Type = RR
N(R) = 2, P/F = 1
=========================================================
```

Chapter 8: Status Monitoring

This chapter covers:

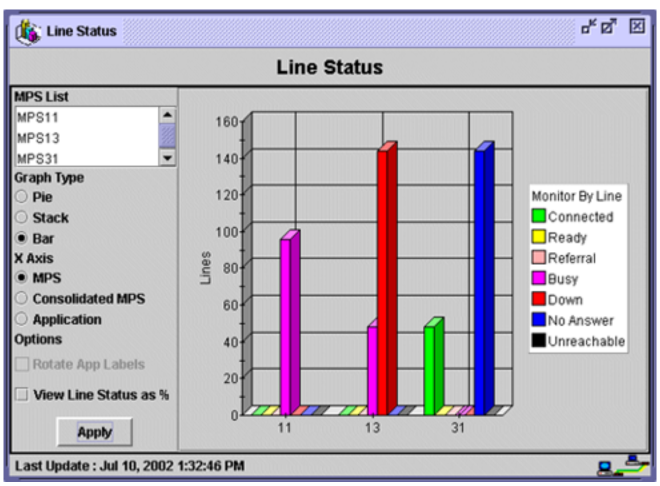
1. Status Monitoring with MPS Manager
2. Controlling and Monitoring MPS Span Status

Status Monitoring with MPS Manager

The MPS Manager graphical utility tool is used to administer, operate, and control the MPS 500 or the MPS 1000. The tool can be used to monitor the activity of the MPS as well as the Avaya PBX in the following ways.

Using Phone Line Status

Phone Line Status refers to the physical status of the phone lines tracked over a period of time. MPS Manager is used to monitor current phone line status for the MPS and the applications. The MPS Manager Activity Monitor quantifies and displays the following seven phone line status attributes, which are color-coded.



The following table displays the different phone line status attributes and their color codes.

Phone Line State	Color
Connected	Green
Ready	Yellow
Referral	Orange
Busy	Purple
Down	Red
No Answer	Blue
Unreachable	Black

Phone line status can also be displayed in the following pictorial ways:

Example

- Graph Display—Monitors the physical activity state of phone lines for MPS components and applications. A phone line can be in any one of the seven activity states described previously.
- Linked Application Status Bar Graph—Displays the phone line status of a currently executing main application and any linking activity, which depicts the working between the main and linked applications, during its execution cycle.

Controlling and Monitoring MPS Span Status

MPS span status is controlled by the Diagnostic Logging and Tracing (DLT) utility. DLT commands are executed from the DLT shell. The DLT shell is entered by invoking the dlt utility in the current shell (Solaris) or in a Command Prompt window (Win32).

The DLT command devlist displays a comprehensive hierarchical list of physical devices and logical instances residing on the current component (TMS). Physical devices, such as phone lines and DSPs, are identified by chassis, TMS slot, card, and device number (for example, span, DSP, etc.). Logical instances are lines on a span or resource classes loaded on a DSP. Manufacturing information for the TMS and cards installed (revision level, serial number, and date) is also provided. Certified Avaya Support Personnel can use this data for troubleshooting or compatibility resolution.

The following is an example of a devlist command list.

```
dlt#mps.1,vos/mpsap1 {2} -> spandisable 3! disable 3rd span ! Span disabled
dlt#mps.1,vos/mpsap1 {3} -> spanenable 3! enable 3rd span ! Span enabled
dlt#mps.1,vos/mpsap1 {5} -> spanstatus all

------------------------------------------------------------------
Alarm             Time  Span   Alarms    Alarms   Span
```

```
Sp   State  Status     Src   Type   Received  Xmitted  Conditions
-------------------------------------------------------------------
1   GREEN   InService  No    ???   (None)     (None)   (None)
2   GREEN   InService  Yes   T1    (None)     (None)   (None)
3   GREEN   InService  No    T1    (None)     (None)   (None)
```

Linking to the Avaya PBX

MPS Manager enables you to set up a telephony link server to connect to the Avaya PBX.

See the MPS Manager Reference Manual for information on setting up and linking the telephony link server.

Index