



SIP Support

in

Avaya Communication Manager 2.0

running on the

Avaya S8300, S8500, or S8700 Media

Server

555-245-206
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Notice

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Warranty

Avaya Inc. provides a limited warranty on this product. Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language as well as information regarding support for this product, while under warranty, is available through the following Web site: <http://www.avaya.com/support>.

Preventing Toll Fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there may be a risk of toll fraud associated with your system and that, if toll fraud occurs, it can result in substantial additional charges for your telecommunications services.

Avaya Fraud Intervention

If you suspect that you are being victimized by toll fraud and you need technical assistance or support, in the United States and Canada, call the Technical Service Center's Toll Fraud Intervention Hotline at 1-800-643-2353.

How to Get Help

For additional support telephone numbers, go to the Avaya support Web site: <http://www.avaya.com/support>. If you are:

- Within the United States, click the *Escalation Management* link. Then click the appropriate link for the type of support you need.
- Outside the United States, click the *Escalation Management* link. Then click the *International Services* link that includes telephone numbers for the international Centers of Excellence.

Providing Telecommunications Security

Telecommunications security (of voice, data, and/or video communications) is the prevention of any type of intrusion to (that is, either unauthorized or malicious access to or use of) your company's telecommunications equipment by some party.

Your company's "telecommunications equipment" includes both this Avaya product and any other voice/data/video equipment that could be accessed via this Avaya product (that is, "networked equipment").

An "outside party" is anyone who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf. Whereas, a "malicious party" is anyone (including someone who may be otherwise authorized) who accesses your telecommunications equipment with either malicious or mischievous intent.

Such intrusions may be either to/through synchronous (time-multiplexed and/or circuit-based) or asynchronous (character-, message-, or packet-based) equipment or interfaces for reasons of:

- Utilization (of capabilities special to the accessed equipment)
- Theft (such as, of intellectual property, financial assets, or toll facility access)
- Eavesdropping (privacy invasions to humans)
- Mischief (troubling, but apparently innocuous, tampering)
- Harm (such as harmful tampering, data loss or alteration, regardless of motive or intent)

Be aware that there may be a risk of unauthorized intrusions associated with your system and/or its networked equipment. Also realize that, if such an intrusion should occur, it could result in a variety of losses to your company (including but not limited to, human/data privacy, intellectual property, material assets, financial resources, labor costs, and/or legal costs).

Responsibility for Your Company's Telecommunications Security

The final responsibility for securing both this system and its networked equipment rests with you - Avaya's customer system administrator, your telecommunications peers, and your managers. Base the fulfillment of your responsibility on acquired knowledge and resources from a variety of sources including but not limited to:

- Installation documents
- System administration documents
- Security documents
- Hardware-/software-based security tools
- Shared information between you and your peers
- Telecommunications security experts

To prevent intrusions to your telecommunications equipment, you and your peers should carefully program and configure:

- Your Avaya-provided telecommunications systems and their interfaces
- Your Avaya-provided software applications, as well as their underlying hardware/software platforms and interfaces
- Any other equipment networked to your Avaya products

TCP/IP Facilities

Customers may experience differences in product performance, reliability and security depending upon network configurations/design and topologies, even when the product performs as warranted.

Standards Compliance

Avaya Inc. is not responsible for any radio or television interference caused by unauthorized modifications of this equipment or the substitution or attachment of connecting cables and equipment other than those specified by Avaya Inc. The correction of interference caused by such unauthorized modifications, substitution or attachment will be the responsibility of the user. Pursuant to Part 15 of the Federal Communications Commission (FCC) Rules, the user is cautioned that changes or modifications not expressly approved by Avaya Inc. could void the user's authority to operate this equipment.

Product Safety Standards

This product complies with and conforms to the following international Product Safety standards as applicable:

Safety of Information Technology Equipment, IEC 60950, 3rd Edition including all relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE) CB-96A.

Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-00 / UL 60950, 3rd Edition

Safety Requirements for Customer Equipment, ACA Technical Standard (TS) 001 - 1997

One or more of the following Mexican national standards, as applicable: NOM 001 SCFI 1993, NOM SCFI 016 1993, NOM 019 SCFI 1998

The equipment described in this document may contain Class 1 LASER Device(s). These devices comply with the following standards:

- EN 60825-1, Edition 1.1, 1998-01
- 21 CFR 1040.10 and CFR 1040.11.

The LASER devices operate within the following parameters:

- Maximum power output: -5 dBm to -8 dBm
- Center Wavelength: 1310 nm to 1360 nm

Luokan 1 Laserlaite

Klass 1 Laser Apparat

Use of controls or adjustments or performance of procedures other than those specified herein may result in hazardous radiation exposures. Contact your Avaya representative for more laser product information.

Electromagnetic Compatibility (EMC) Standards

This product complies with and conforms to the following international EMC standards and all relevant national deviations:

Limits and Methods of Measurement of Radio Interference of Information Technology Equipment, CISPR 22:1997 and EN55022:1998.

Information Technology Equipment – Immunity Characteristics – Limits and Methods of Measurement, CISPR 24:1997 and EN55024:1998, including:

- Electrostatic Discharge (ESD) IEC 61000-4-2
- Radiated Immunity IEC 61000-4-3
- Electrical Fast Transient IEC 61000-4-4
- Lightning Effects IEC 61000-4-5
- Conducted Immunity IEC 61000-4-6
- Mains Frequency Magnetic Field IEC 61000-4-8
- Voltage Dips and Variations IEC 61000-4-11
- Powerline Harmonics IEC 61000-3-2
- Voltage Fluctuations and Flicker IEC 61000-3-3

Federal Communications Commission Statement

Part 15:

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Part 68: Answer-Supervision Signaling

Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 rules. This equipment returns answer-supervision signals to the public switched network when:

- answered by the called station,
- answered by the attendant, or
- routed to a recorded announcement that can be administered by the customer premises equipment (CPE) user.

This equipment returns answer-supervision signals on all direct inward dialed (DID) calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

Avaya attests that this registered equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

REN Number

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

This equipment complies with Part 68 of the FCC rules. On either the rear or inside the front cover of this equipment is a label that contains, among other information, the FCC registration number, and ringer equivalence number (REN) for this equipment. If requested, this information must be provided to the telephone company.

For G350 and G700 Media Gateways:

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the rear of this equipment is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. The digits represented by ## are the ringer equivalence number (REN) without a decimal point (for example, 03 is a REN of 0.3). If requested, this number must be provided to the telephone company.

For all media gateways:

The REN is used to determine the quantity of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in devices not ringing in response to an incoming call. In most, but not all areas, the sum of RENs should not exceed 5.0. To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company.

REN is not required for some types of analog or digital facilities.

Means of Connection

Connection of this equipment to the telephone network is shown in the following tables.

For MCC1, SCC1, CMC1, G600, and G650 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/A.S. Code	Network Jacks
Off premises station	OL13C	9.0F	RJ2GX, RJ21X, RJ11C
DID trunk	02RV2-T	0.0B	RJ2GX, RJ21X
CO trunk	02GS2	0.3A	RJ21X
	02LS2	0.3A	RJ21X
Tie trunk	TL31M	9.0F	RJ2GX
Basic Rate Interface	02IS5	6.0F, 6.0Y	RJ49C
1.544 digital interface	04DU9-BN	6.0F	RJ48C, RJ48M
	04DU9-IKN	6.0F	RJ48C, RJ48M
	04DU9-ISN	6.0F	RJ48C, RJ48M
120A4 channel service unit	04DU9-DN	6.0Y	RJ48C

For G350 and G700 Media Gateways:

Manufacturer's Port Identifier	FIC Code	SOC/REN/A.S. Code	Network Jacks
Ground Start CO trunk	02GS2	1.0A	RJ11C
DID trunk	02RV2-T	AS.0	RJ11C
Loop Start CO trunk	02LS2	0.5A	RJ11C
1.544 digital interface	04DU9-BN	6.0Y	RJ48C
	04DU9-DN	6.0Y	RJ48C
	04DU9-IKN	6.0Y	RJ48C
	04DU9-ISN	6.0Y	RJ48C
Basic Rate Interface	02IS5	6.0F	RJ49C

For all media gateways:

If the terminal equipment (for example, the media server or media gateway) causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact the Technical Service Center at 1-800-242- 2121 or contact your local Avaya representative. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. It is recommended that repairs be performed by Avaya certified technicians.

The equipment cannot be used on public coin phone service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

This equipment, if it uses a telephone receiver, is hearing aid compatible.

Canadian Department of Communications (DOC) Interference Information

This Class A digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Declarations of Conformity

United States FCC Part 68 Supplier's Declaration of Conformity (SDoC)

Avaya Inc. in the United States of America hereby certifies that the equipment described in this document and bearing a TIA TSB-168 label identification number complies with the FCC's Rules and Regulations 47 CFR Part 68, and the Administrative Council on Terminal Attachments (ACTA) adopted technical criteria.

Avaya further asserts that Avaya handset-equipped terminal equipment described in this document complies with Paragraph 68.316 of the FCC Rules and Regulations defining Hearing Aid Compatibility and is deemed compatible with hearing aids.

Copies of SDoCs signed by the Responsible Party in the U. S. can be obtained by contacting your local sales representative and are available on the following Web site: <http://www.avaya.com/support>.

All Avaya media servers and media gateways are compliant with FCC Part 68, but many have been registered with the FCC before the SDoC process was available. A list of all Avaya registered products may be found at: <http://www.part68.org> by conducting a search using "Avaya" as manufacturer.

European Union Declarations of Conformity



Avaya Inc. declares that the equipment specified in this document bearing the "CE" (*Conformité Européenne*) mark conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (1999/5/EC), including the Electromagnetic Compatibility Directive (89/336/EEC) and Low Voltage Directive (73/23/EEC). This equipment has been certified to meet CTR3 Basic Rate Interface (BRI) and CTR4 Primary Rate Interface (PRI) and subsets thereof in CTR12 and CTR13, as applicable.

Copies of these Declarations of Conformity (DoCs) can be obtained by contacting your local sales representative and are available on the following Web site: <http://www.avaya.com/support>.

Japan

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 actions.

この装置は、情報処理装置等電波障害自主規制協議会（VCCI）の基準に基づくクラスA情報技術装置です。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講ずるよう要求されることがあります。

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Write: Globalware Solutions
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E-mail: totalware@gwsmail.com

For the most current versions of documentation, go to the Avaya support Web site: <http://www.avaya.com/support>.

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SIP Support in Communication Manager

This chapter describes the support for SIP (Session Initiated Protocol) incorporated into Avaya Communication Manager 2.0 or later running an Avaya S8300, S8500 or S8700 Media Server.

Introduction to SIP

What is SIP?

SIP is the Session Initiated Protocol, an endpoint-oriented messaging standard defined by the [Internet Engineering Task Force \(IETF\)](#). As implemented by Avaya for release 2.0 of Communication Manager, SIP "trunking" functionality will be available on any of the Linux-based media servers (S8300, S8500 or S8700). These media servers will function as [Plain Old Telephone Service \(POTS\)](#) gateways, and they will also support name/number delivery between and among the various non-SIP endpoints supported by Communication Manager (analog, DCP or H.323 stations and analog, digital or IP trunks) and new SIP-enabled endpoints, such as the Avaya 4602SIP Telephone. In addition to its calling capabilities, IP Softphone R5 and later also will include instant-messaging client software, which is a SIP-enabled application, while continuing its full support of the existing [H.323](#) standard for call control.

How does SIP fit into Your System?

The following are attributes of the support for SIP which is built into [Avaya Communication Manager](#) that also help it fit easily into your system:

- It is built around published standards (e.g., SIP Drafts, etc.).
- It integrates traditional circuit-switched interfaces and IP-switched interfaces. This integration allows the user to evolve easily from the current circuit-switched telephony infrastructures to next generation IP infrastructures, including SIP.
- It positions customers to leverage as needed the increasing number and power of SIP-enabled applications, like instant messaging and presence.

NOTE:

Building SIP support into Avaya MultiVantage Software adds another element to the modular family of Avaya components, which seamlessly delivers a business's voice and messaging capabilities over an IP network. Avaya continues enhancing the value it provides to customers in a standards-based, IP communications infrastructure.

The modular and extensible system architecture that Avaya has chosen for offering SIP support has a unique benefit for Avaya customers: the set of features supported by SIP itself is augmented by those supported by Avaya Communication Manager 2.0. A media server running Communication Manager becomes, in effect, a telephony feature server, which is accessible from any SIP endpoint and provides access transparently to many telephony features that the SIP standard currently does not address.

SIP-Related Support

The additions made to support SIP in Avaya Communication Manager 2.0 or later running on S8300, S8500 or S8700 media servers include what is described in the following section.

Trunking

Support for SIP trunks allows an enterprise to connect its media server(s) to the SIP-enabled proxy server, the Avaya [Converged Communications Server \(CCS\)](#), and through this proxy, optionally to an external SIP service provider, if desired. The trunk support in Communication Manager complies with SIP standards (e.g., IETF RFC 2543bis) and thereby will interoperate with any SIP-enabled endpoint/station which also complies with the standard.

In complex configurations with Avaya S8700 Media Server(s), the signaling-group properties in Communication Manager must be administered to match in certain ways. See [Administrative Requirements](#) on page 12.

Stations

Support for SIP stations (which are administered as [off-PBX station \(OPS\)](#)) using SIP trunks allows any fully compliant SIP phone to interoperate with Avaya phones. This means any SIP phone, from Avaya or a third party, that complies with the appropriate RFC or draft service standards will be able to:

- Dial and be dialed as an extension in the enterprise dial plan. OPS stations support additional features as well, like bridging. For more details, refer to the *Avaya Extension to Cellular User's Guide*, Issue 5, DocID 210-100-700, and the *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide*, Issue 6, DocID 210-100-500.
- Put calls on hold and participate in transfers and conference calls.

NOTE:

These features are available using SIP stations managed by media servers running Avaya Communication Manager 2.0 or later, but not using SIP stations which may register with a SIP proxy server and are not managed directly through Communication Manager.

CDR

Support for complete call detail records for all SIP calls, if desired.

Access Control

Support is provided for full access control to external trunks from any phone. Note that some other means of access control, such as a firewall, typically would be required to control access to SIP trunks from SIP endpoints that are outside of the enterprise.

Requirements for SIP

Software

Support for SIP can be enabled in the 2.0 and later releases of Avaya Communication Manager running on any of the available Linux-based media servers. The appropriate Avaya licensing is also required.

Hardware

The SIP-enabled release of Avaya Communication Manager runs on the following media servers for the release in November 2003:

- Avaya S8300 Media Server
- Avaya S8500 Media Server (new for release 2.0)
- Avaya S8700 Media Server.
- NOTE: Any of these media servers may control one or more Avaya G350 Media Gateways.

All CLAN or processor CLAN IP interfaces must be configured correctly. For more details, refer to "*Administration for Network Connectivity for Avaya Communication Manager, 555-233-504*".

Refer to the *Converged Communications Server Installation and Administration* document, Issue 1, DocID 555-245-705, for details on Avaya's SIP proxy, registration and instant-messaging (IM) server, which connects to the media server(s) running Communication Manager and provides SIP-enabled applications such as enterprise IM using the Avaya IP SoftPhone R5 or later software clients.

Firmware

It is important to note that SIP standards do (and hence, potential interoperability with certain third-party SIP-enabled devices may) dictate that Dual-Tone Multifrequency (DTMF) tones be supported within the Real-Time Transfer Protocol (RTP) data stream. This requirement further demands, for example, that the newest versions of Avaya's VoIP engine be installed throughout your system.

NOTE:

For example, any TN2302AP circuit packs that are present in your system must have the correct firmware version to support DTMF tones in the RTP data stream.

TN2302AP hardware/ firmware version	Circuit Pack w/1-digit HW vintage	Circuit Pack w/2-digit HW vintage
Minimum for SIP	V71	V71
Highly Recommended	V72	V72 (<i>required</i> w/G.729 fax)
In G700/G350 media gateways	V21 (or V22, with full support for facsimile in G.729 mode)	

Administrative Requirements

The SIP signaling group administered on Communication Manager defines the characteristics of a signaling connection.

There is no physical "trunk" when using SIP. Therefore, there is no physical limit on how many calls (trunk members) can be set up using a particular signaling connection. (It is limited only by the total bandwidth available).

However, using the signaling group and trunk group administrative interface in Communication Manager also for SIP is useful, because doing so helps extend several Communication Manager features to SIP easily. Although Communication Manager normally limits signaling groups to 255 trunk members, thereby limiting each signaling connection to 255 calls, for SIP groups Avaya has removed the restriction that each combination of far-end and near-end IP address/port must be unique for each signaling group.

More than one signaling group may be administered to share exactly the same properties of:

- far-end node-name (fe-nn)
- far-end port (fe-pt)
- near-end node-name (ne-nn)
- near-end port (ne-pt).

This supports more than 255 calls on the same SIP-based signaling connection, where a signaling connection is defined as <near-end node-name, near-end port, far-end node-name, far-end port>.

Communication Manager sequentially checks SIP signaling groups and chooses the first available trunk member for every new, incoming call on that signaling connection. Therefore, it is *highly recommended* that all these SIP signaling groups which share node names/ports, as well as the SIP trunks groups using each of the signaling groups, be administered with identical properties (i.e., matching entries in the administrative screen fields). Of course, different SIP signaling connections will differ with respect to their near-end and/or far-end node name(s)/port number(s), and they should therefore have their SIP trunk's signaling groups administered accordingly. It is not appropriate to administer them the same way.

In release 2.0 of Communication Manager, the number of simultaneous SIP signaling connections is limited to three. You may administer more than three (e.g., for purposes of supporting back-up SIP proxy servers), but the run-time limit of simultaneous signaling connections is three. Remember that a signaling connection is not the same thing as a signaling group, and that more than one SIP signaling group can share the same signaling connection.

Example Media Server 1 (named ACM1) has three Converged Communication Servers (Proxies A, B, and C) communicating with it. Proxy server A is the most-used proxy, and it requires 355 members. ACM1 needs to administer three backup proxies (BA, BB, BC), one backup for each primary proxy host. The following table outlines the administration of such a system.

Sig group	<fe-nn, fe-pt, ne-nn, ne-pt>	trunk-group	<number of members, siggroup>
1	<A, 5061, ACM1-CLAN1, 5061>	1	<255, 1>
2	<A, 5061, ACM1-CLAN1, 5061>	2	<100, 2>
3	<B, 5061, ACM1-CLAN1, 5061>	3	<10, 3>
4	<C, 5061, ACM1-CLAN1, 5061>	4	<10, 4>

Sig group	<fe-nn, fe-pt, ne-nn, ne-pt>	trunk-group	<number of members, siggroup>
5	<BA, 5061, ACM1-CLAN2, 5061>	5	<255, 5>
6	<BA, 5061, ACM1-CLAN2, 5061>	6	<100, 6>
7	<BB, 5061, ACM1-CLAN2, 5061>	7	<10, 7>
8	<BC, 5061, ACM1-CLAN2, 5061>	8	<10, 8>

The above administered signaling groups define a total of six signaling connections (note that signaling groups 1 and 2 use the same signaling connection and 5 and 6 use the same signaling connection, for a total of six connections). Out of these six connections, only three of them can be active at any time.

Generally, the following shows administration of the route pattern forms for these connections:

Route pattern	Trunk group in entry 1	Trunk group in entry 2
1	1	5
2	2	6
3	3	7
4	4	8

NOTE:

The backup SIP trunks will be used only when the signaling connection to the main SIP proxy server is not available.

Related Systems

Refer to the *Converged Communications Server Installation and Administration* document, Issue 1, DocID 555-245-705, for details on the server hardware requirements for the SIP-based instant-messaging application. Refer to the online help which came with Avaya IP SoftPhone R5.x for details on setting up and using it for SIP instant messaging. Refer to the documentation for your Avaya 4602SIP Telephone for details on setting up and using it as a SIP station for voice calling. The latter includes both the:

- *4602/4602SW SIP IP Telephone User's Guide* (16-300035, Issue 1, February 2004), and the
- *4602/4602SW SIP IP Telephone Administrator's Guide* (16-300037, Issue 1, February 2004).

Setup and Configuration

Initial Setup

The CLAN (or processor CLAN) IP interface of your media server running Avaya Communication Manager 2.0 must be configured correctly. For information on network assessment and readiness testing, refer to *Administration for Network Connectivity for Avaya Communication Manager*, 555-233-504.

Administration

To administer SIP trunks in Avaya Communication Manager 2.0, you must perform the following tasks:

- 1** Verify that your system supports and is correctly configured for IP connectivity.
- 2** On the [System-Parameters Customer-Options screen](#) on page 36:
 - a** Verify that the "IP Trunks?" field on page 4 is set to **y** (controlled by the license file).
 - b** Verify the license-authorized value from **0-100** (for S8300) or **0-800** (for S8500) or **0-1000** (for S8700) in the "Maximum Administered SIP Trunks:" field on page 2.
- 3** Enter the valid hostname and IP Address for the SIP proxy server for your network in the corresponding fields on the **IP Node Names** screen. See the *ANC* document, 555-233-504.
- 4** Enter the fully qualified name or IP address of the domain for which this network region applies in the "Home Domain:" field on the [IP Network Region screen](#) on page 32. A valid entry in this field is required for SIP endpoints on Communication Manager to call the public network. Note that this Home Domain must exactly match the Domain setup on the Edit System Properties screen using the Master Administrator web interface on the SIP proxy server.
- 5** On the [Feature-Related System Parameters screen](#) on page 31, select **all** in the "Trunk-to-Trunk Transfer?" field to enable the transfer and conference features to operate correctly with SIP.
- 6** On the [Signaling Group screen](#) on page 16:
 - a** Enter **sip** in the "Group Type:" field . A new screen for SIP groups will be displayed.
 - b** Verify that the "Transport Method:" field displays the default value of **tls** (TLS).
 - c** The "Trunk Group for Channel Selection:" field may be set to the default of **BLANK**.
 - d** Enter the name of the IP interface at the near (local) end of the SIP trunk signaling group in the "Near-end Node Name:" field. For S8300 media servers, the value of this entry is typically **procr**. For the S8700, it is the node name for the selected CLAN interface.
 - e** Ensure the recommended TLS port value **5061** is set in the "Near-End Listen Port:" field.
 - f** Enter the name of the node you administered for the SIP proxy server in step 3 in the "Far-end Node Name:" field.
 - g** Enter the recommended TLS port value of **5061** in the "Far-End Listen Port:" field.
 - h** If you want the SIP proxy server you administered in step 3 to use the codec set and/or parameters specified for an IP network region different from that of the LAN IP interface, then enter the SIP proxy's region in the "Far-End Network Region:" field.

Transport Method

Appears only when the value of the entry in the "Group Type:" field is **sip**. Ensure that the default of TLS is selected in this field. If not, results will be unpredictable.

Valid entries	Usage
tls	Default and recommended (secure) transport method is TLS.
tcp	WARNING: TCP may be used only for debugging purposes.

Near-end Node Name

Appears when the value of the entry in the "Group Type:" field is **h.323** or **sip**. Enter the node name for the C-LAN IP interface in this media server. The node name must be administered on the IP Node Names screen and the IP Interfaces screen.

Valid entries	Usage
Name of an administered IP node.	Describe the near-end node.

Far-end Node Name

Appears when the value of the entry in the "Group Type:" field is **h.323** or **sip**. Enter the node name for the SIP proxy server used for trunks assigned to this signaling group. The node name must be administered on the IP Node Names screen.

Valid entries	Usage
Name of an administered IP node.	Describe the far-end node.



Tip:

If the either node name or port differs for each SIP signaling group, you have different SIP signaling connections, and you should administer a maximum of three using TLS. If you administer more than three TLS signaling connections and they are all in use at the same time, the results may be unpredictable. Note that if the node names and ports do match, you may administer as many identical SIP signaling groups using TLS as desired.

Near-end Listen Port

Appears when "Group Type:" is **h.323** or **sip**. Defaults to 5061 for SIP over TLS.

Valid entries	Usage
5000-9999	Enter an unused port number. The recommended port for SIP over TLS is 5061.

Far-end Listen Port

Appears when the Group Type field is **h.323** or **sip**.

Valid entries	Usage
1-65535	Enter the same number as entered in the Near-end Listen Port field. The recommended port entry is 5061 for SIP over TLS.

Far-end Network Region

Appears when the Group Type field is **h.323** or **sip**. The number of the network region that is assigned to the far-end of the trunk group.

Valid entries	Usage
1-250 or blank	Enter the network region number that is assigned to the far end of the trunk group. The region is used to obtain the codec set used for negotiation of trunk bearer capability. Leave blank to select the region of the near-end node by default.

Far-end Domain

Appears only when the value of the entry in the "Group Type:" field is **sip**.

Valid entries	Usage
Max. 40 character string, or leave blank	Enter the fully qualified domain name or IP address for the destination proxy server. For example, to route SIP calls within your enterprise, enter the domain assigned to your proxy server; for external SIP calling, it could be that of your SIP service provider. If blank, the far-end IP address is used.

Bypass If IP Threshold Exceeded

Appears when the Group Type field is **h.323** or **sip**.

Valid entries	Usage
y/n	Enter y to automatically remove from service the trunks assigned to this signaling group when IP transport performance falls below limits administered on the Maintenance-Related System Parameters screen.

DTMF over IP

Appears when the value of the entry in the "Group Type:" field is **h.323** or **sip**.

Valid entries	Usage
rtp-payload	SIP Trunks require the entry of rtp-payload .

IP Audio Hairpinning

Appears when the Group Type field is **h.323** or **sip**. Allows the option for H.323 and SIP-enabled endpoints to be connected through the IP circuit pack in the media server or switch, without going through the TDM bus Available for SIP in 2.0.1.

Valid entries	Usage
y/n	Enter y to enable hairpinning for H.323 or SIP trunk groups.

Trunk Group screen, page 1

The following screen is displayed when **sip** is entered in the "Group Type:" field on page 1:

```

add trunk-group next                                     Page 1 of x
                                     TRUNK GROUP
Group Number: ____          Group Type: _____          CDR Reports: _
Group Name: _____          COR: ____          TN: ____          TAC: ____
Direction: _____          Outgoing Display? _
Dial Access? _          Busy Threshold: ____          Night Service: _____
Queue Length: ____
Service Type: _____          Auth Code? _
                                     Signaling Group: ____
                                     Number of Members: ____
TRUNK PARAMETERS
                                     Digital Loss Group: ____

```

Trunk Group field descriptions

The following fields are displayed when **sip** is entered in the "Group Type:" field on page 1:

Group Number

This field displays the group number assigned when the trunk group was added.

NOTE:

For trunk groups connecting 2 media servers or switches in Distributed Communication System networks, Avaya recommends that you assign the same group number on both servers or switches.

Group Type

Enter the type of trunk group.

Busy-out the trunk group before you change the group type. Release the trunk group after you make the change. For more information about busying out and releasing trunk groups, see your system's Maintenance documentation.

Valid entries	Usage
SIP	Use SIP trunks to connect a media server running Communication Manager to a SIP proxy server (Avaya Converged Communications Server).

CDR Reports

Valid entries	Usage
y	All outgoing calls on this trunk group will generate call detail records. If the Record Outgoing Calls Only field on the <i>CDR System Parameters</i> screen is n , then incoming calls on this trunk group will also generate call detail records.
n	Calls over this trunk group will not generate call detail records.
r (ring-intvl)	CDR records will be generated for both incoming and outgoing calls. In addition, the following ringing interval CDR records are generated: Abandoned calls: The system creates a record with a condition code of "H," indicating the time until the call was abandoned. Answered calls: The system creates a record with a condition code of "G," indicating the interval from start of ring to answer. Calls to busy stations: The system creates a record with a condition code of "I," indicating a recorded interval of 0.

NOTE:

For ISDN trunk groups, the Charge Advice field affects CDR information. For CO, DIOD, FX, and WATS trunk groups, the Analog PPM field affects CDR information.

Group Name

Valid entries	Usage
1 to 27 characters	Enter a unique name that provides information about the trunk group. Don't use the default entry or the group type (DID, WATS) here. For example, you might use names that identify the vendor and function of the trunk group: USWest Local; Sprint Toll, Level3 SIP, etc.

COR

Valid entries	Usage
0 to 95	Enter a class of restriction (COR). Classes of restriction control access to trunk groups, including trunk-to-trunk transfers.



Tip:

Remember that facility restriction levels (FRL) are assigned to classes of restriction. Even if 2 trunk groups have classes of restriction that allow a connection, different facility restriction levels may prevent operations such as off-net call forwarding or outgoing calls by remote access users.

TN

Valid entries	Usage
1 to 100 (DEFINITY R, S8700 IP-Connect)	Enter a Tenant Partition number to assign this trunk group to the partition.



Tip:

Double-check your entry. If you accidentally enter an unassigned tenant partition number, the system will accept the entry but no calls will go to the trunk group.

TAC

Enter the trunk access code (TAC) for each trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.

Valid entries	Usage
1- to 4-digit number	Enter any number that fits the format for trunk access codes or dial access codes defined in your dial plan. NOTE: Although this field is required, trunk groups of type SIP cannot be dialed via TAC; the TAC you enter here is used only to identify them on CDR reports.
*, #	* and # may be used as the first character in a TAC.

Direction

Enter the direction of the traffic on this trunk group. The entry in this field affects which timers appear on the Administrable Timers page. This field appears for all trunk groups except DID and CPE.

Valid entries	Usage
incoming	Enter for trunk groups with incoming traffic only.
outgoing	Enter for trunk groups with outgoing traffic only.
two-way	Enter two-way for Network Call Redirection.

Outgoing Display

This field allows display phones to show the name and number of the trunk group used for an outgoing call before the call is connected. This information may be useful to you when you're trying to diagnose trunking problems.

Valid entries	Usage
y	Displays the trunk group name and number.
n	Displays the digits the caller dials.

Dial Access

This field controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS. Dial access to SIP trunks is not allowed.



SECURITY ALERT:

Calls dialed with a trunk access code over WATS trunks bypass AAR/ARS and aren't restricted by facility restriction levels. For security, you may want to leave the field set to n unless you need dial access to test the trunk group.

Valid entries	Usage
y	Allows users to access the trunk group by dialing its access code. NOTE: This entry is not available for SIP trunks.
n (Display only for SIP trunks)	Does not allow users to access the trunk group by dialing its access code. Attendants can still select this trunk group with a Trunk Group Select button. This is the default entry.

Busy Threshold

Use this field if you want attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code will be connected to the attendant, and the attendant can prioritize outgoing calls for the last remaining trunks. Calls handled by AAR and ARS route patterns go out normally.

Valid entries	Usage
1 to 255 (DEFINITY R, S8700 IP-Connect)	Enter the number of trunks that must be busy in order to light the warning lamp on the Attendant Console. For example, if there are 30 trunks in the group and you want to alert the attendant whenever 25 or more are in use, enter 25 .

Night Service

This field sets the destination to which incoming calls go when Night Service is in operation. If a Night field on the Group Member Assignments page is administered with a different destination, that entry will override the group destination for that trunk. CPE, DID, and DIOD trunk groups do not support night service.



Tip:

Whenever possible, use a night service destination on your switch: otherwise some features won't work correctly, even over a DCS network.

Valid entries	Usage
blank	Leave this field blank if the Trunk Type (in/out) field is not auto/...
An extension number (can be a VDN)	Enter the extension of your night service destination.
attd	Calls go to the attendant and are recorded as Listed Directory Number (LDN) calls on call detail records.

Queue Length

Outgoing calls can wait in a queue, in the order in which they were made, when all trunks in a trunk group are busy. If you enter 0, callers receive a busy signal when no trunks are available. If you enter a higher number, a caller hears confirmation tone when no trunk is available for the outgoing call. The caller can then hang up and wait: when a trunk becomes available, Communication Manager will call the extension that placed the original call. The caller will hear 3 short, quick rings. The caller doesn't need to do anything but pick up the handset and wait: Communication Manager remembers the number the caller dialed and automatically completes the call.

This field appears when the Direction field is **outgoing** or **two-way**.

Valid entries	Usage
0 to 100	Enter the number of outgoing calls that you want to be held waiting when all trunks are busy.
0	Enter 0 for DCS trunks.

Service Type

Indicates the service for which this trunk group is dedicated. The following table provides a listing of predefined entries. In addition to the Services/Features listed in this table, any user-defined Facility Type of 0 (feature) or 1 (service) on the Network Facilities screen is allowed. For SIP trunks, only "public-ntwrk" and "tie" are valid.

As many as 10 (for Avaya DEFINITY Server SI or CSI) or 200 (for DEFINITY Server R) ISDN trunk groups can have this field administered as **cbc**.

Valid entries	Usage
public-ntwrk	Public network calls — It is the equivalent of CO (outgoing), DID, or DIOD trunk groups. If Service Type is public-ntwrk and the trunk is not a SIP trunk, then Dial Access can be set to y .
tie	Tie trunks — general purpose

Auth Code

This field affects the level of security for tandemed outgoing calls at your server running Communication Manager. This field appears if the Direction field is incoming or two-way, and it can only be **y** if the Authorization Codes field is **y** on the [System-Parameters Customer-Options screen](#) on page 36.

Valid entries	Usage
y/n	Enter y to require callers to enter an authorization code in order to tandem a call through an AAR or ARS route pattern. The code will be required even if the facility restriction level of the incoming trunk group is normally sufficient to send the call out over the route pattern.

Signaling Group

Appears only when the value of the entry in the "Group Type:" field is **sip**.

Valid entries	Usage
1-255	Enter the number of the SIP signaling group number you wish to assign to the trunk group.

Number of Members

Appears only when the value of the entry in the "Group Type:" field is **sip**.

Valid entries	Usage
1-100 (S8300), 1-800 (S8500),or 1-1000 (S8700)	Enter the number of SIP trunks that are members of the trunk group. All members of a SIP trunk group will have the same characteristics. NOTE: Member pages for SIP trunk groups are completed automatically based on this entry and are not individually adminstrable.

Digital Loss Group

This field determines which administered 2-party row in the loss plan applies to this trunk group if the call is carried over a digital signaling port in the trunk group.

Valid entries	Usage
1 to 17	Shows the index into the loss plan and tone plan.

Trunk Group screen, page 2

The following screen is displayed when **sip** is entered in the "Group Type:" field on page 1:

```

add trunk-group next                                     Page 2 of x
          TRUNK FEATURES
ACA Assignment? _      Measured: none
                                Maintenance Tests? y
                                Send Name: n      Send Calling Number: y
                                Numbering Format: public_
                                Replace Unavailable Numbers? n
                                Send Connected Number: n
  
```

Trunk Group field descriptions, cont.

The following fields are displayed on page 2 when **sip** is entered in the "Group Type:" field on page 1:

ACA Assignment

<u>Valid entries</u>	<u>Usage</u>
y/n	Enter y if you want Automatic Circuit Assurance (ACA) measurements to be taken for this trunk group. If y is entered, complete the Long Holding Time, Short Holding Time, and Short Holding Threshold fields.

Measured

Indicates if the system will transmit data for this trunk group to the Call Management System (CMS). You cannot use **internal** and **both** unless either the BCMS (Basic) or the VuStats field is **y** on the *System-Parameters Customer-Options screen*. If the ATM field is set to **y** on the *System-Parameters Customer-Options screen*, this field accepts only **internal** or **none**. If this field contains a value other than **internal** or **none** when ATM is **y**, **none** appears.

<u>Valid entries</u>	<u>Usage</u>
internal	Enter internal if the data can be sent to the Basic Call Management System (BCMS), the VuStats data display, or both.
external	Enter external to send the data to the CMS.
both	Enter both to collect data internally and to send it to the CMS.
none	Enter none if trunk group measurement reports are not required. NOTE: This is the default for SIP trunk groups.

Maintenance Tests

Appears when the value of the Group Type field is **aplt**, **isdn**, **sip** or **tie**.

<u>Valid entries</u>	<u>Usage</u>
y/n	Enter y (the default) if hourly maintenance tests will be made on this trunk group. One or more trunk members must be administered for this entry to be saved.

Send Name

Specifies whether the calling/connected/called/busy party's administered name is sent to the network on outgoing/incoming calls. In general, the valid entries are **y**, **n**, or **r** (restricted). For SIP, if the value in this field is set to the default of **n** or to **r**, then the calling/connected name of "anonymous" will be sent by Avaya Communication Manager for the display name.

NOTE:

If name information is not administered for the calling station or the connected/called/busy station, then the extension number is sent in place of the name.

Send Calling Number

Specifies whether the calling party's number is sent on outgoing or tandemed calls. Valid entries are **y**, **n**, or **r** (restricted). If the value is **r**, the calling number for display is "presentation restricted". For SIP trunks, entry of the default value of **n** or of **r** behave identically. If it has been set to **y**, then the *Numbering - Public/Unknown Format* screen is accessed to construct the actual number to be sent, or the *Numbering-Private* screen (based on the entry in the Numbering Format field) is used. The *Numbering - Public/Unknown Format* screen is used to manipulate only when Send Calling Number is set to **y** for any administrable block of extensions.

Numbering Format

This field appears if the Send Calling Number field is **y** or **r** or the Send Connected Number field is **y** or **r**. This specifies the encoding of Numbering Plan Indicator for identification purposes in the Calling Number and/or Connected Number IEs, and in the QSIG Party Number. Valid entries are **public**, **unknown**, **private**, and **unk-pvt**. **Public** indicates that the number plan according to CCITT Recommendation E.164 is used and that the Type of Number is national. This is the default entry for SIP trunks. **Unknown** indicates the Numbering Plan Indicator is unknown and that the Type of Number is unknown. **Private** indicates the Numbering Plan Indicator is PNP and the Type of Number is determined from the Private-Numbering screen. An entry of **unk-pvt** also determines the Type of Number from the Private-Numbering screen, but the Numbering Plan Indicator is unknown.

Replace Unavailable Numbers

Appears when the Group Type field is **isdn** or **sip**. Indicates whether to replace unavailable numbers with administrable strings for incoming and outgoing calls assigned to the specified trunk group. This field applies to BRI/PRI and SIP trunks.

Valid entries	Usage
y/n	Enter y for the display to be replaced regardless of the service type of the trunk. The default entry for SIP trunks is n .

Trunk Group field descriptions, cont.

The following fields are displayed on page 3 when **sip** is entered in the "Group Type:" field on page 1:

Service/Feature

Specifies the Services/Features for an incoming call type. See the Service Type field description for a list of predefined Services/Features that can be received. Or, use Type 0, Type 1, and Type 2 user defined services. The identifier **other** can be used for any Services/Features not explicitly specified, for example, for SIP trunk groups.

Called Len

Specifies the number of digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any length of digits associated with the specified Service/Feature can match in this field. Valid entries are **0** to **21**, or leave blank.

Called Number

Specifies the leading digits received for an incoming call. A blank entry is used as a "wild card" entry and, when used, means that any number associated with the specified Service/Feature can match in this field. Valid entries are up to 16 digits, or leave blank.

Del

Specifies the number of leading digits to be deleted from the incoming Called Party Number. Calls of a particular type may be administered to be routed to a single destination by deleting all incoming digits and then administering the Insert field with the desired extension. Valid entries are **1** to **21**, **all**, or leave blank.

Insert

Specifies the digits to be prepended to the front of the remaining digits after any (optional) digit deletion has been performed. The resultant number formed from digit deletion/insertion is used to route the call, provided night service is not in effect. Valid entries are up to 16 characters consisting of a combination from the following: **0** to **9**, *****, **#**, or leave blank.

Trunk Group screen, page 4

The following screen is displayed when **sip** is entered in the "Group Type:" field on page 1:

```
add trunk-group next                                     Page 4 of 10
                                                         TRUNK GROUP
                                                         Administered Members(min/max): xxx/yyy
                                                         Total Administered Members: xxx

GROUP MEMBER ASSIGNMENTS
  Port      Name
1: _____
2: _____
3: _____
4: _____
5: _____
6: _____
7: _____
8: _____
9: _____
10: _____
11: _____
12: _____
13: _____
14: _____
15: _____
```

NOTE:

For SIP trunks, the group member-assignment pages are *not* individually administrable. These fields are automatically populated and displayed based on the number of members of SIP trunk groups specified on page 1 of this screen. Note that these display-only group member-assignment pages of the trunk group screen are repeated, as needed, to support all the trunk group's members.

Trunk Group field descriptions, cont.

The following fields are displayed on page 4 when **sip** is entered in the "Group Type:" field on page 1:

Administered Members (min/max)

This field shows the minimum and maximum member numbers that have been administered for this trunk group. For SIP trunks, this field is display only.

Total Administered Members

This field shows the total number of members administered in the trunk group. For SIP trunks, this field is display only.

Port

This field shows the port number assigned to each of the members administered in the trunk group. For SIP trunks, this field is display only.

Changed Administrative Screens

The following screens in Communication Manager have added or changed fields related to SIP:

- [Feature-Related System Parameters screen](#) (change to field name)
- [IP Network Region screen on page 32](#) (added "Home Domain:" field)
- [Locations screen on page 32](#) (added "Proxy Selection Route Pattern" fields)
- [Numbering -- Private screen](#) on page 33 (changed name from "ISDN Numbering -- Private")
- [Numbering -- Public/Unknown screen on page 33](#) (changed name from "ISDN Numbering -- Public/Unknown")
- [Route Pattern screen](#) on page 34 (added "SIP Secure?" field)
- [Signaling Group screen](#) (new sip value in Group Type field; see [New Administrative Screens](#) on page 16)
- [System Capacity screen](#) on page 35 (added a row for "SIP Trunks (included in 'Trunk ports'):")
- [System Parameters Call Coverage / Call Forwarding screen](#) on page 35 (added "Disable call classifier for CCRON over SIP trunks?" field to page 2).
- [System-Parameters Customer-Options screen on page 36](#) (added "Maximum Administered SIP Trunks:" field)
- [Trunk Group screen, page 1](#) (new sip value in Group Type field; see [New Administrative Screens](#) on page 16).

Feature-Related System Parameters screen

The change is highlighted in **bold** on this screen:

```

change system-parameters features                                     page 1
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer? all
Automatic Callback - No Answer Timeout Interval (rings): 4_
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20_
      AAR/ARS Dial Tone Required? y
      Music/Tone On Hold: music      Port: _____
      Music (or Silence) On Transferred Trunk Calls: all
      DID/Tie/ISDN/SIP Intercept Treatment: attd

Internal Auto-Answer of Attd-Extended/Transferred Calls? y
      Automatic Circuit Assurance (ACA) Enabled? y
      ACA Referral Calls: local
      ACA Referral Destination: _____
      ACA Short Holding Time Originating Extension: _____
      ACA Long Holding Time Originating Extension: _____
      Abbreviated Dial Programming by Assigned Lists:
      Auto Abbreviated/Delayed Transition Interval(rings):
      Protocol for Caller ID Analog Terminals: Bellcore
      Display Calling Number for Room to Room Caller ID Calls?
    
```

NOTE:

The "Trunk-to-Trunk Transfer?" field must be set to **all** to support the SIP transfer feature.

Numbering -- Private screen

This screen supports Private Numbering Plans (PNP) and now applies to both ISDN and SIP. It allows you to specify the digits to be put in the Calling Number information element (IE), the Connected Number IE, and the QSIG Party Number for extensions in the Private Numbering Plan.

Avaya Communication Manager supports private-network numbers as long as 15 digits. If the complete number — including the level 1 and 2 prefixes, the PBX identifier, and the extension — is more than 15 digits long, then Communication Manager will neither create nor send the QSIG Party Numbers or information elements.

```
change private-numbering                                     Page 1 of 1

                                NUMBERING - PRIVATE FORMAT

Network Level:  _                                           PBX Identifier:  ____
Level 2 Code:  ____                                         Deleted Digits:  _
Level 1 Code:  ____
```

Numbering -- Public/Unknown screen

This screen supports the Call Identification Display feature for ISDN and SIP. The feature provides a name/number display for display-equipped stations within an ISDN or SIP network. The system uses the caller's name and number and displays it on the called party's display. Likewise, the called party's name and number can be displayed on the caller's display.

The screen allows you to specify the desired digits for the Calling Number IE and the Connected Number IE (in addition to the QSIG Party Number) for any extension in the Public and/or Unknown Number Plans.

Administer these screens if the value entered in either the **Send Calling Number** or the **Send Connected Number** field is **y**, or the value entered in the **Supplementary Service Protocol** field is **b**, on the *Trunk Group* screen.

NOTE:

If the table is not properly administered, but the **Send Calling Number** or the **Send Connected Number** field is **public** or **unknown**, the Identification Number (PartyNumber data type) is not sent for QSIG PartyNumbers. In this case, the ASN.1 data type containing the PartyNumber (PresentedAddressScreened, PresentedAddressUnscreened, PresentedNumberScreened, or PresentedNumberUnscreened) will be sent marked as *PresentationRestricted* with NULL for the associated digits.

System Capacity screen

The change is highlighted in **bold** on this screen:

```

display capacity
                                                    Page 7 of 12
                SYSTEM CAPACITY
                Used Available System
                -----
                TRUNKS
                DS1 Circuit Packs:           10      390      400
                DS1 With Echo Cancellation:   0        400      400
                ICHT For ISDN Trunks:         0        576      576
                ISDN CBC Service Selection Trunks: 1        199      200
                Trunk Groups:                 34      1966      2000
                Trunk Ports:                  608     7392     8000
                H.323 Trunks (included in 'Trunk ports'): 604     3396     4000
                Remote Office Trunks (included in 'Trunk ports'): 0        4000     4000
                SBS Trunks (included in 'Trunk ports'): 0        1000     1000
                SIP Trunks (included in 'Trunk ports'): 764      236     1000
    
```

Note that system trunking capacity varies, based on media server. Refer to the "Capacities Table" for more information.

System Parameters Call Coverage / Call Forwarding screen

The change is highlighted in **bold** on this screen:

```

change system-parameters coverage-forwarding
                                                    page 2
                SYSTEM PARAMETERS -- CALL COVERAGE / CALL FORWARDING
                COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)
                Coverage Of Calls Redirected Off-Net Enabled? y
                Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                Ignore Network Answer Supervision? y
                Disable call classifier for CCRON over ISDN trunks? n
                Disable call classifier for CCRON over SIP trunks? n
    
```

For more details on completing the fields on this screen, refer to the "Administrator's Guide for Avaya Communication Manager R2.0".

System-Parameters Customer-Options screen

The change is highlighted in **bold** on this screen:

```
display system-parameters customer-options                page 1 of __
                                OPTIONAL FEATURES
                                Used
G3 Version: V12                                Maximum Ports: 2800  1041
Location: 1                                Maximum XMOBILE Stations: 0  0
Platform: 2

IP PORT CAPACITIES
                                Maximum Administered H.323 Trunks: 100  96
                                Maximum Concurrently Registered IP Stations: 10  10
                                Maximum Administered Remote Office Trunks: 0  0
Maximum Concurrently Registered Remote Office Stations: 0  0
                                Maximum Concurrently Registered IP eCons: 0  0
                                Maximum Administered SIP Trunks: 100  50

Maximum Number of DS1 Boards with Echo Cancellation: 0  0
                                Maximum TN2501 VAL Boards: 1  0
                                Maximum G700 VAL Sources: 0  0

(NOTE: You must logoff & login to effect the permission changes.)
```

Many fields on this screen are controlled by the Avaya license file. The web-based RFA process is used to generate these license files for customers. For more detailed descriptions of the fields on this screen, refer to the *"Administrator's Guide for Avaya Communication Manager R2.0"*.

Glossary

A

access code

A dial code of 1 digit to 3 digits that is used to activate a [feature](#), cancel a feature, or access an outgoing [trunk](#).

Access Security Gateway (ASG)

An optional interface that can be used to secure the administration and maintenance [ports](#) on the system.

American National Standards Institute (ANSI)

A professional technical association that supports standards for transmission, [protocol](#), and high-level languages, and that represents the US in the [International Organization for Standards](#). ANSI standards are for voluntary use in the US.

Avaya Communication Manager

An open, scalable, highly reliable, and secure telephony application. Communication Manager provides user functionality and system management functionality, intelligent call routing, application integration and extensibility, and Enterprise Communications networking.

B

bearer channel (B-channel)

A 64-kbps channel or a 56-kbps channel that carries a variety of [digital](#) information streams. A B-channel carries voice at 64 kbps, data at up to 64 kbps, [wideband](#) voice encoded at 64 kbps, and voice at less than 64 kbps, alone or combined. See also [data channel \(D-channel\)](#).

bus

A multiconductor electrical path that is used to transfer information over a common connection from any of several sources to any of several destinations. *See also* [packet bus](#); [time-division multiplex \(TDM\) bus](#).

C

Call Detail Recording (CDR)

A [feature](#) that uses software and hardware to record call data. CDR was formerly called Station Message Detail Recording (SMDR). *See also* [Call Detail Recording utility \(CDRU\)](#).

Call Detail Recording utility (CDRU)

Software that collects, stores, filters, and provides output of call detail records. *See also* [Call Detail Recording \(CDR\)](#).

carrier

An enclosed shelf that contains vertical slots that hold [circuit packs](#).

central office (CO)

Telephone switching equipment that provides local telephone service and access to toll facilities for long distance calling.

channel

(1) A [circuit](#)-switched call. (2) A communications path that is used to transmit voice and data. (3) In [wideband](#) transmission, all the contiguous [time slots](#) or noncontiguous time slots that are necessary to support a call. For example, an H0-channel uses six 64-kbps time slots. (4) A digital signal-0 (DS0) on a T1 facility or an E1 facility that is not specifically associated with a logical circuit-switched call. *See also* [data channel \(D-channel\)](#).

circuit

(1) An arrangement of electrical elements through which electric current flows. (2) A [channel](#) or a transmission path between two or more points.

circuit pack

A circuit card on which electrical [circuits](#) are printed, and integrated circuit (IC) chips and electrical components are installed. A circuit pack is installed in a [switch carrier](#). One example is the TN2302.

Class of Restriction (COR)

A [feature](#) that allows up to 96 classes of call-origination restrictions and call-termination restrictions for telephones, telephone groups, [data modules](#), and [trunk groups](#). *See also* [Class of Service \(COS\)](#).

Class of Service (COS)

A [feature](#) that uses a number to specify whether telephone users can activate the Automatic Callback (ACB), Call Forwarding All Calls, Data Privacy, or Priority Calling features. *See also* [Class of Restriction \(COR\)](#).

CCITT

Comite Consultatif International Telephonique et Telegraphique. *See* [International Telecommunications Union \(ITU\)](#).

communications system

A software-controlled processor complex that interprets dial pulses, tones, and keyboard characters, and makes the proper connections within the system and externally. The communications system consists of a [digital](#) computer, software, storage devices, and [carriers](#), with special hardware to perform the connections. A communications system provides communications services for the telephones on customer premises and the [data terminals](#) on customer premises, including access to [public networks](#) and [private networks](#). *See also* [switch](#).

Controlled Local Area Network (CLAN) circuit pack

A [circuit pack](#) (TN799B) in an Avaya DEFINITY port network (PN) that provides [TCP/IP](#) connectivity to adjuncts over Ethernet or Point-to-Point Protocol (PPP). The CLAN circuit pack serves as the network interface for a DEFINITY server. The CLAN terminates [Internet Protocol \(IP\)](#) ([TCP](#) and [UDP](#)), and relays those sockets and connections up to the Avaya DEFINITY server.

Converged Communications Server (CCS)

Avaya's proxy server for [Session Initiated Protocol \(SIP\)](#), initially supporting instant messaging.

CPN

called-party number

CPN/BN

calling-party number/billing number

customer-premises equipment (CPE)

Equipment that is connected to the telephone [network](#), and that resides on a customer site. CPE can include telephones, modems, fax machines, video conferencing devices, switches, and so on.

D

data channel (D-channel)

A 16-kbps channel or a 64-kbps channel that carries signaling information or data on an [Integrated Services Digital Network Basic Rate Interface \(ISDN-BRI\)](#) or an [Integrated Services Digital Network Primary Rate Interface \(ISDN-PRI\)](#). *See also* [bearer channel \(B-channel\)](#); [data channel \(D-channel\)](#).

data communications equipment (DCE)

Equipment on the [network](#) side of a communications link that makes the binary serial data from the source or the transmitter compatible with the communications [channel](#). DCE is usually a modem, a [data module](#), or [packet assembly/disassembly \(PAD\)](#) device.

data module

An interconnection device between an [Integrated Services Digital Network Basic Rate Interface \(ISDN-BRI\)](#) or a [Digital Communications Protocol \(DCP\)](#) interface of the [switch](#), and [data terminal equipment \(DTE\)](#) or [data channel \(D-channel\)](#).

data terminal

An input/output (I/O) device that has either switched access or direct access to a [host computer](#) or to a processor interface.

data terminal equipment (DTE)

Equipment that comprises the endpoints in a connection over a data [circuit](#). In a connection between a [data terminal](#) and a host, the terminal, the host, and the associated modems or [data modules](#) comprise the DTE.

digital

The representation of information by discrete steps. *Compare with* **analog**.

Digital Communications Protocol (DCP)

A proprietary [protocol](#) that is used to transmit both digitized voice and digitized data over the same communications link. A DCP link consists of two 64-kbps information (I) [channels](#), and one 8-kbps signaling (S) channel. The DCP protocol supports two information-bearing channels, and thus two telephones or [data modules](#). The I1 channel is the DCP channel that is assigned on the first page of the 8411 Station screen. The I2 channel is the DCP channel that is assigned on the analog adjunct page of the 8411 Station screen, or on the data module page.

dual-tone multifrequency (DTMF)

The touchtone signals that are used for in-band telephone signaling.

Dynamic Host Configuration Protocol (DHCP)

An IETF [protocol](#) (RFCs 951, 1534, 1542, 2131, and 2132) that assigns IP addresses dynamically from a pool of addresses instead of statically.

E

extension

A number from 1 digit to 5 digits that is used to route calls through a [communications system](#). With a Uniform Dial Plan ([UDP](#)) or a main-satellite dialing plan, extensions are also used to route calls through a [private network](#).

F

feature

A specifically defined function or service that the system provides.

H

H.323

An [International Telecommunications Union \(ITU\)](#) standard for switched multimedia communication between a [local area network \(LAN\)](#)-based multimedia endpoint and a gatekeeper. *See also* [Session Initiated Protocol \(SIP\)](#).

host computer

A computer that is connected to a [network](#), and that processes data from data-entry input devices.

I

IE

See [information element \(IE\)](#).

IEEE

See [Institute of Electrical and Electronics Engineers \(IEEE\)](#).

IETF

See [Internet Engineering Task Force \(IETF\)](#).

IM

Instant Messaging. The instant-messaging client software required for the [Avaya Communication Manager](#) release 2.0 or later is a version of the Avaya IP Softphone R5 or later.

information element (IE)

The name for the data fields within an [Integrated Services Digital Network \(ISDN\)](#) Layer 3 message.

Institute of Electrical and Electronics Engineers (IEEE)

An organization that, among other things, produces standards for [local area network \(LAN\)](#) equipment.

Integrated Services Digital Network (ISDN)

A [public network](#) or a [private network](#) that provides end-to-end [digital](#) communications for all services to which users have access. ISDN uses a limited set of standard, multipurpose user-network interfaces that are defined by the [CCITT](#). Through internationally accepted standard interfaces, an ISDN provides digital [circuit-switching](#) communications or [packet switching](#) communications within the network. An ISDN provides links to other ISDNs to provide national digital communications and international digital communications. *See also* [Integrated Services Digital Network Basic Rate Interface \(ISDN-BRI\)](#); [Integrated Services Digital Network Primary Rate Interface \(ISDN-PRI\)](#).

Integrated Services Digital Network Basic Rate Interface (ISDN-BRI)

The interface between a communications system and terminal that includes two 64-kbps [bearer channel \(B-channel\)s](#) for transmitting voice or data, and one 16-kbps [data channel \(D-channel\)](#) for transmitting associated B-channel call control and out-of-band signaling information. ISDN-BRI also includes 48 kbps for transmitting framing and D-channel contention information, for a total interface speed of 192 kbps. ISDN-BRI serves [Integrated Services Digital Network \(ISDN\)](#) terminals and [digital](#) terminals that are fitted with ISDN terminal adapters. *See also* [Integrated Services Digital Network Primary Rate Interface \(ISDN-PRI\)](#).

Integrated Services Digital Network Primary Rate Interface (ISDN-PRI)

The interface between multiple communications systems that in North America includes 24 64-kbps channels that correspond to the North American digital signal level-1 (DS1) standard rate of 1.544 Mbps. The most common arrangement of channels in ISDN-PRI is 23 64-kbps [bearer channel \(B-channel\)s](#) for transmitting voice and data, and one 64-kbps [data channel \(D-channel\)](#) for transmitting

associated B-channel call control and out-of-band signaling information. With nonfacility-associated signaling (NFAS), ISDN-PRI can include 24 B-channels and no D-channel. *See also* [Integrated Services Digital Network \(ISDN\)](#); [Integrated Services Digital Network Basic Rate Interface \(ISDN-BRI\)](#).

International Organization for Standards

A worldwide federation of standards bodies who issue International Standards for technological, scientific, intellectual, and economic activity. The federation is called *ISO*, and the US representative to the federation is the [American National Standards Institute \(ANSI\)](#).

International Telecommunications Union (ITU)

An international organization that sets universal standards for data communications, including [Integrated Services Digital Network \(ISDN\)](#). ITU was formerly known as International Telegraph and Telephone Consultative Committee ([CCITT](#)).

International Telegraph and Telephone Consultative Committee

See [International Telecommunications Union \(ITU\)](#).

Internet Engineering Task Force (IETF)

One of two technical working bodies of the Internet Activities Board. The IETF develops new [Transmission Control Protocol \(TCP\)/Internet Protocol \(IP\)](#) (i.e. [TCP/IP](#)) standards for the Internet.

Internet Protocol (IP)

A connectionless [protocol](#) that operates at Layer 3 of the [Open Systems Interconnect \(OSI\)](#) model. IP protocol is used for Internet addressing and routing [packets](#) over multiple [networks](#) to a final destination. IP works in conjunction with [Transmission Control Protocol \(TCP\)](#), and is usually identified as [TCP/IP](#).

L

local area network (LAN)

A networking arrangement that is designed for a limited geographical area. Generally, a LAN is limited in range to a maximum of 6.2 miles, and provides high-speed carrier service with low error rates. Common configurations include daisy chain, star (including [circuit](#)-switched), ring, and bus.

N

narrowband

A [circuit](#)-switched call at a data rate of 64 kbps or less. All switch calls that are not [wideband](#) are considered to be narrowband.

network

A series of points, [nodes](#), or stations that are connected by communications [channels](#).

node

A switching point or a control point for a [network](#). Nodes are either tandem or terminal. Tandem nodes receive signals, and pass the signals on. Terminal nodes originate a transmission path, or terminate a transmission path.

O

off-PBX station (OPS)

A telephone that [Avaya Communication Manager](#) does not control, such as a cellular telephone or the home telephone of a user. The features of Communication Manager can be extended to an OPS through switch administration by associating the extension of the office telephone with the off-site telephone. NOTE: [Session Initiated Protocol \(SIP\)](#) endpoints are administered on Communication Manager as OPS.

Open Systems Interconnect (OSI)

A system of seven independent communication [protocols](#) defined by the [International Organization for Standards](#) or ISO. Each of the seven layers enhances the communications services of the layer below, and shields the layer above from the implementation details of the lower layer. In theory, this structure can be used to build [communications systems](#) from independently developed layers.

P

packet

A group of bits that is used in [packet switching](#) and that is transmitted as a discrete unit. A packet includes a message element and a control [information element \(IE\)](#). The message element is the data. The control IE is the header. In each packet, the message element and the control IE are arranged in a specified format.

packet assembly/disassembly (PAD)

The process of packetizing control data and user data from a transmitting device before the data is forwarded through the packet network. The receiving device disassembles the [packets](#), removes the control data, and then reassembles the packets, thus reconstituting the user data in its original form.

packet bus

A [bus](#) with a wide bandwidth that transmits [packets](#).

packet switching

A data-transmission technique that segments and routes user information in discrete data envelopes that are called [packets](#). Control information for routing, sequencing, and error checking is appended to each packet. With packet switching, a [channel](#) is occupied only during the transmission of a packet. On completion of the transmission, the channel is made available for the transfer of other packets.

PBX

private branch exchange. *See* [switch](#).

port

A data-transmission access point or voice-transmission access point on a device that is used for communicating with other devices.

Plain Old Telephone Service (POTS)

Basic voice communications with standard, single-line phones accessing the [public switched telephone network \(PSTN\)](#).

private network

A [network](#) that is used exclusively for the telecommunications needs of a particular customer.

protocol

A set of conventions or rules that governs the format and the timing of message exchanges. A protocol controls error correction and the movement of data.

public network

A [network](#) to which all customers have open access for local calling and long distance calling.

public switched telephone network (PSTN)

The public, worldwide, voice telephone [network](#).

R**Real Time Transfer Protocol (RTP)**

An [Internet Engineering Task Force \(IETF\) protocol](#) (RFC 1889) that addresses the problems that occur when video and other exchanges with real-time properties are delivered over a [local area network \(LAN\)](#) that is designed for data. RTP gives higher priority to video and other real-time interactive exchanges than to connectionless data.

RFC

Request For Comment designates [Internet Engineering Task Force \(IETF\)](#) standards that are Drafts.

RPM

RedHat Package Manager

RTC

Real Time Communication

RTCP

Real Time Control Protocol

S**Session Initiated Protocol (SIP)**

A signaling [protocol](#) for Internet conferencing, telephony, presence, events notification, and instant messaging. SIP initiates call setup, routing, authentication, and other feature messages to endpoints within an IP domain. *See also* [H.323](#); [Voice over IP \(VoIP\)](#).

switch

Any kind of telephone switching system. *See also* [communications system](#).

T**TAC**

trunk-access code

TCP

See [Transmission Control Protocol \(TCP\)](#).

TCP/IP

See [Internet Protocol \(IP\)](#). *See also* [Transmission Control Protocol \(TCP\)](#).

tie trunk

A telecommunications [channel](#) that directly connects two private switching systems.

time-division multiplex (TDM) bus

A [bus](#) that is time-shared regularly by preallocating short [time slots](#) to each transmitter. In a [switch](#), all [port circuits](#) are connected to the [time-division multiplex \(TDM\) bus](#), and any port can send a signal to any other port. *See also* [time-division multiplexing \(TDM\)](#).

time-division multiplexing (TDM)

A form of multiplexing that divides a transmission [channel](#) into successive [time slots](#). *See also* [time-division multiplex \(TDM\) bus](#).

time slot

In the [switch](#), a time slot refers to either a digital signal-level 0 (DS0) on a T1 facility or an E1 facility, or a 64-kbps unit on the [time-division multiplex \(TDM\) bus](#) or fiber connection between [port networks \(PNs\)](#) that is structured as 8 bits every 125 microseconds.

Transmission Control Protocol (TCP)

A connection-oriented transport-layer [protocol](#), IETF STD 7. RFC 793, that governs the exchange of sequential data. Whereas the [Internet Protocol \(IP\)](#) deals only with [packets](#), TCP enables two hosts to establish a connection and exchange streams of data. TCP guarantees delivery of data, and also guarantees that packets are delivered in the same order in which the packets are sent.

Transport Layer Security (TLS)

An IETF standard (RFC 2246) to supersede Netscapes' Secure Socket Layer (SSL) and provide host-to-host data connections with encryption and certification at the transport layer, as the name implies.

trunk

A dedicated telecommunications [channel](#) between two [communications systems](#) or [central office \(CO\)s](#).

trunk group

Telecommunications [channels](#) that are assigned as a group for certain functions, and that can be used interchangeably between two [communications systems](#) or [central office \(CO\)s](#).

U

UDP

(1) [User Datagram Protocol \(UDP\)](#); (2) Uniform Dial Plan.

universal serial bus (USB)

A high-speed serial interface that is used primarily to add a printer, a modem, a keyboard, a mouse, or another peripheral device to a personal computer.

User Datagram Protocol (UDP)

A [packet](#) format that is included in the [TCP/IP](#) suite of [protocols](#). UDP is used for the unacknowledged transmission of short user messages and control messages.

V

Voice over IP (VoIP)

A set of facilities that use the [Internet Protocol \(IP\)](#) to manage the delivery of voice information. In general, VoIP means to send voice information in digital form in discrete [packets](#) instead of in the traditional [circuit-committed protocols](#) of the [public switched telephone network \(PSTN\)](#). Users of VoIP and Internet telephony avoid the tolls that are charged for ordinary telephone service.

W

wideband

A [circuit](#)-switched call at a data rate that is greater than 64 kilobits per second. A circuit-switched call on a single T1 facility or a single E1 facility with a bandwidth that is between 128 kilobits per second and 1536 kilobits per second (T1) or 1984 kilobits per second (E1) in multiples of 64 kilobits per second. H0, H11, H12, and N x digital signal-level 0 (DS0) calls are wideband. *Compare with [narrowband](#).*

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