



Nortel Session Server Lines

Fundamentals

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4 Contents

New in this release

The following sections detail information that is new for the Nortel Session Server Lines documentation.

- "Features" (page 5)
- "Other changes" (page 6)

Features

The following SIP Lines features are new for SN09U. For more information about SIP Lines features for SN09U, see *"Nortel MCS 5200/SSL Feature Overview"* (NN10251-115):

- **Enhanced Network and Centrix Residential Features:** This feature provides for the integration of existing Centrex services on SIP lines. This feature provides enhanced residential and Business client services that are supported directly from the softswitch.
- **Line Provisioning for SIP Lines:** This feature provides SERVORD+ capabilities that are similar to the provisioning capabilities of other VoIP services.
- **SIP Lines - Data Sync Audit:** This feature provides SIP and Session Server Lines data integrity audit capability.
- **SIP Lines - Maintenance Enhancements:** This feature provides the ability to perform the following:
 - Log into OSSGate and post a SIP line and identify the state of the line.
 - Change the state of a SIP line.
 - Highlight the number of SIP lines registered at any one time.
 - Re-register SIP phones as a group or individually.
 - Download new or updated software to SIP phones.
 - Diagnose reasons for unreachable SIP lines.
 - QSIP command is accessible through the client position.

- Session Server Lines Platform: Provide fault monitoring and handling capability of the Session Server Lines platform.

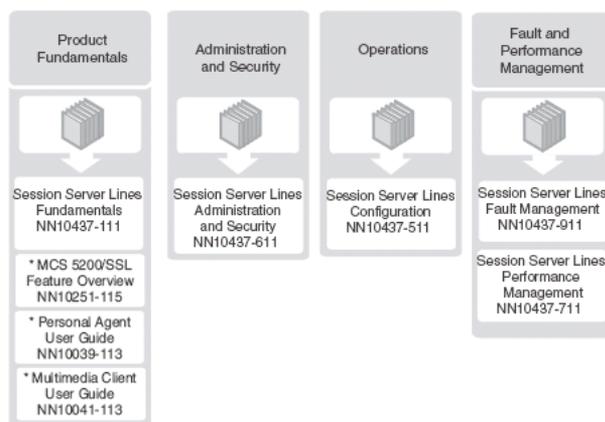
Other changes

The SIP Lines documentation is restructured in SN09U.

- *"Nortel Session Server Lines Fundamentals"* (NN10437-111)
- *"Nortel Session Server Lines Configuration"* (NN10437-511)
- *"Nortel Session Server Lines Administration and Security"* (NN10437-611)
- *"Nortel Session Server Lines Performance Management"* (NN10437-711)
- *"Nortel Session Server Lines Fault Management"* (NN10437-911)

Roadmap

The figure below is a graphical representation of the SIP Lines documentation suite. The figure shows the job function under which each document belongs. The job functions also appear in the Technical Support Portal (TS Portal) at www.nortel.com/support. Use the job function to filter content on the TS Portal and to search for information in this document. An asterisk indicates documents that are updated in this release.



Applicability

Nortel software releases for the SIP Lines product are identified by the CS 2000 product line designations. The SIP Lines software is identified by the letters SN and a two-digit number, such as SN09.

Software developments and refinements between major releases are identified with letters appended to the software designation. The contents of this document apply to the (I)SN09U software release.

(I)SN09U is an extension of the (I)SN09 software release and supersedes both (I)SN09 and (I)SN09FF for CVoIP applications. (I)SN09 continues to be supported for TDM-only applications. The (I)SN09U software release incorporates all capabilities of (I)SN09, (I)SN09FF, and additional corrective content delivered as part of Nortel's (I)SN09 software robustness program. (I)SN09, (I)SN09FF, and (I)SN09U are valid terms for use within the (I)SN09U software release.

Information quality

Documents are delivered with either a Draft, Preliminary, or Standard designation for the entire book.

Status designations are as follows:

- Draft: Information is written and reviewed for technical content, but the feature is still under development and change is expected.
- Preliminary: Development of the feature content is complete and is considered stable.
- Standard: The feature content has been fully reviewed for quality and is suitable for delivery to customers.

Text conventions

The following text conventions explain how text may be formatted throughout the documentation.

angle brackets (< >)

Indicate that you choose the text to enter based on the description inside the brackets. Do not type the brackets when entering the command.

Example: If the command syntax is `ping <ip_address>`, you type `ping 192.32.10.12`

asterisk (*)

In command syntax, the asterisk indicates that you can enter multiple instances of the preceding parameter.

bold Courier text

Indicates command names and options and text that you need to enter.

Example: Use the `debug portal` command.

Example: Enter `show interface`.

braces ({})

Indicate required elements in syntax descriptions where there is more than one option. You must choose only one of these options. Do not type the braces when entering the command.

Example: If the command syntax is: `add loghost [server=] <addr> [[facility=] {local0, local1, local2, local3, local4, local5, local6, local7}]`, you can enter either

```
add loghost server=10.11.12.10 facility=local2
```

or

```
add loghost server=10.11.12.10 facility=local3
```

brackets ([])

Indicate optional elements in syntax descriptions. Do not type the brackets when entering the command. Example: If the command syntax is

```
add user [name=] <username> [epwd=<epwd>] [priv={U, SU, SSU}],
```

you can enter

```
add user name=james epwd=passwd priv=u
```

ellipsis points (...)

Indicate that you repeat the last element of the command as needed.

Example: If the command syntax is

```
ethernet/2/1 [<parameter> <value>] ... you enter
```

`ethernet/2/1` and as many parameter-value pairs as are needed.

italic text

Indicates new terms, book titles, and Web addresses.

plain Courier text

Indicates command syntax and system output, for example, prompts and system messages.

Example: `Set Trap Monitor Filters`

separator (>)

Shows menu paths.

Example: File > SCS configuration identifies the SCS Configuration option in the File menu.

vertical line (|)

Separates choices for command keywords and arguments. Enter only one of the choices. Do not type the vertical line when entering the command.

Example: If the command syntax is

```
check sub [sub]=<subid> | sub=<subname> [isp]=<ispn-  
name>, you can enter
```

```
check sub sub=345 isp=blue or
```

```
check sub sub=james isp=blue
```

Add user james, leaving out the terms shown in brackets.

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Introduction

This document provides an overview description of the CS 2000 Session Server Lines platform and capabilities. Click on any of the following links to display the description:

- ["SIP Lines overview" \(page 15\)](#)
- ["SIP Lines supported services" \(page 19\)](#)
- ["SIP Lines-supported clients" \(page 32\)](#)
- ["Session Server Lines platform" \(page 44\)](#)
- ["PacketCable Multimedia" \(page 51\)](#)
- ["Upgrades" \(page 55\)](#)
- ["Accounting management" \(page 57\)](#)
- ["Security and administration" \(page 59\)](#)

SIP Lines overview

SIP Lines is an application of capabilities across various elements within the CS 2000 Voice over IP (VoIP) solution. The CS 2000 Session Server Lines provides the SIP interface to the various clients and provides SIP-oriented multimedia services.

Functional description

The heart of the SIP Lines program is the re-use of these various elements with software modifications to make use of the existing capabilities and features of each platform. Dozens of Centrex and residential features are made available to SIP Lines by means of the Core software. The SIP-oriented multimedia features are made available to these lines through the use of Multimedia Communication Server (MCS) software on the Session Server Lines (SSL) platform.

Use this document to understand SIP Lines capabilities and to assist you in provisioning SIP-oriented multimedia services.

See the section "[SIP Lines supported services](#)" (page 19) for information on supported network and client services.

SIP standards compliance

Session Initiation Protocol (SIP) is an application control protocol for conferencing, telephony, presence, events notification, and instant messaging. The protocol initiates call setup, routing, authentication, and other feature messages to endpoints within an IP domain. More information on the SIP protocol is available on the IETF website (<http://www.ietf.org/rfc.html>). RFC3261 provides the basis for the SIP protocol.

SIP Lines protocols

SIP Lines supports the User Data Protocol (UDP). Transmission Control Protocol (TCP) is not supported.

SIP Lines application support

CS 2000 SIP Lines applications support various SIP endpoint types. Multimedia capabilities are natively integrated as shown in the following figure.

See the section "[SIP Lines-supported clients](#)" (page 32) for more information.

SIP Lines application support



Hardware

The SIP Lines application uses existing hardware (HW) elements. The required pieces are the CS 2000 and the Session Server Lines platform. The Communication Server includes either XA-CORE or Compact Communication Server 2200, Gateway Controllers (GWC), and Passport 8600 routers. Session Server Lines is deployed on the Session Application Module eXtreme Thin Server (SAM-XTS), which is the Intel Langley Prestonia (OEM'd by Hewlett Packard* as the HP-cc3310). For details, see "[Session Server Lines platform](#)" (page 44). Media Application Servers (MAS) are required for audio conferencing in addition to tone and announcement support. In networks that use IP Virtual Private Networks (VPN) and Network Address Translation (NAT), a Border Control Point (BCP, formerly known as the RTP Media Portal), hosted on the Border Control Point hardware, is required for the SIP voice traffic to navigate around firewalls.

Tools and utilities

The following tools are used to configure the components that make up the SIP voice application:

- CS 2000 Management Console
- Maintenance and Administration Position (MAP) Command Interface (CI)

- SERVORD+ commands over OSSGate and BPT
- MCP System Management Console. For further information see "*Nortel System Management Console User Guide*" (NN10247-111).
- Provisioning Client

SIP Lines capacity

The maximum number of SIP Lines for each CS 2000 is 180,000.

On the Session Server Lines platform, capacity is determined by Session Manager pairs. The number of SIP Lines for each Session Manager pair is 35,805 (based on a multiple of 1023).

Software

All Carrier VoIP (CVoIP) components involved with SIP Lines run SN09 software. The details of the SN09 load lineups are documented in the solution-level basics guides.

Consult installation and configuration procedures available for third-party software through vendor-supplied literature.

OAM strategy

Operations, Administration, and Maintenance (OAM) strategy for the SIP voice application is as described in the product documentation of the involved components.

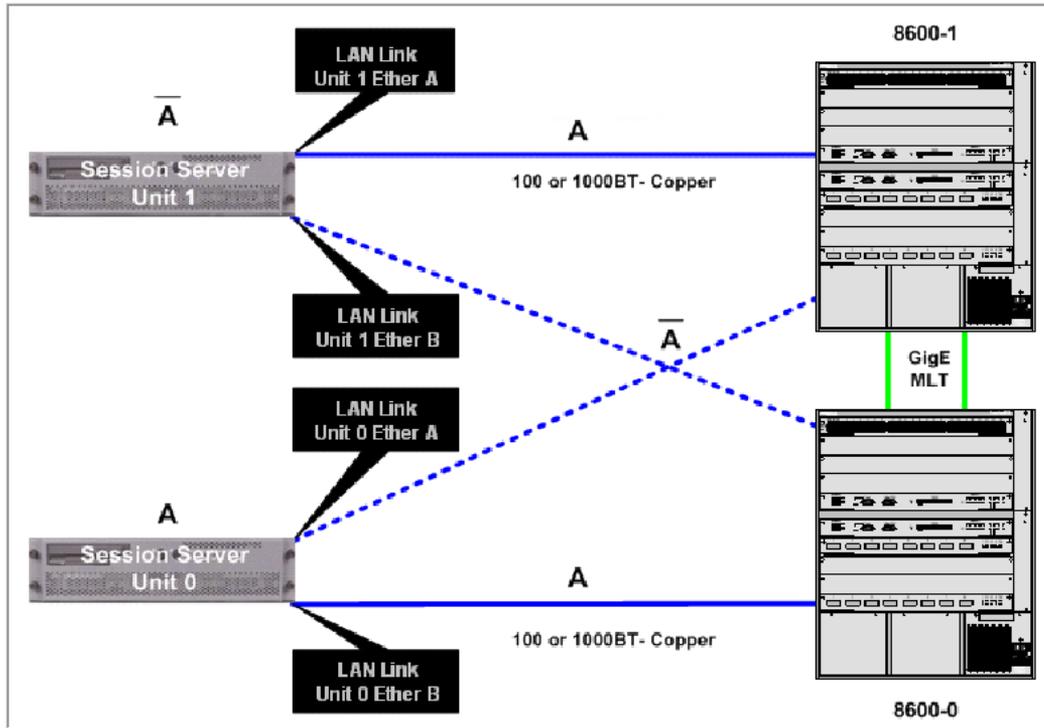
Interfaces

The interfaces of the involved CS 2000 components are as described in their respective product documentation.

Session Server Lines network interfaces

Each Session Server Lines (SSL) server has two GigE Ethernet interfaces, configured as link 0 and link 1 to communicate with the CS-LAN switch. See "["CS 2000 Session Server CS-LAN"](#) (page 18).

CS 2000 Session Server CS-LAN



SIP Lines supported services

CS 2000 SIP Lines provides telephony and multimedia services to subscribers over an Internet Protocol (IP) network. A SIP access client is the subscriber equipment. For example, software in a computer or a telephone. With SIP Lines, some services can be provided directly from the client without the need of a feature programmed in the controlling switch (CS 2000). Examples of services that are often provided by the client are conference calling and call transfer.

SIP provides local call control, which co-exists with CS 2000 core network call control. The SIP access client gives the subscriber control over call screening, routing, and acceptance features.

End-user features are either client-hosted or Core-network hosted. The client-hosted SIP Line feature set differs for each client and in how they are invoked on the client.

See the following sections for the supported features:

- ["CS 2000 Core network telephony services" \(page 19\)](#)
- ["SIP Client voice and multimedia services" \(page 28\)](#)
- ["International tone support" \(page 30\)](#)

CS 2000 Core network telephony services

The CS 2000 handles the basics of the call, such as registering, billing, line provisioning, and network-based features.

The following table lists the CS 2000 Core network telephony services that are available on the PC Client and hard client. As a general rule, network features supported through translations are supported for CS 2000 SIP Lines.

Supported Core network telephony services

Core network telephony service	Notes, restrictions, and limitations
10-digit unconditional Local Number Portability (LNP) trigger	
Advanced Intelligent Network (AIN)	Mid-call digit collection is not supported. Responses such as Send To Resource Intelligent Peripheral (STR IP) (with the IP trying to collect digits) and Collect Info will fail. The same is true for Connect To Resource Intelligent Peripheral (CTR IP) (with the IP trying to collect digits).
Advanced Intelligent Network Directory Number (DN)	
AIN Message Waiting	

Core network telephony service	Notes, restrictions, and limitations
Alternate Service Provider	Similar to Equal Access, this gives support for certain services (for example, voice mail) through alternative providers.
Automatic Alternate Routing	
Automatic Call Back (ACB)	
Automatic Collect Call Barring (ACCB)	
Automatic Flexible Routing (AFR)	
Automatic Identification of Outward Dialing	
Automatic Identified Outward Dialing for Enhanced Private Switched Communications Service (EPSCS) access	
Automatic Message Accounting Test Call Capability	
Automatic Recall (AR)	
Automatic Route Selection (ARS)	
Automatic Time and Charges (ATC)	
Business Group Dialing Plan	
Business Group Direct Outward Dialing	Second dial tone is not supported; the DOD activation code and the digits should be dialed in one string of digits.
Call Forward Variable - Basic Business Group	
Call Forwarding - Busy Line - Incoming Only	
Call Forwarding - Don't Answer - All Calls	
Call Forwarding - Don't Answer - Incoming Only	
Call Forwarding - Incoming Only	
Call Forwarding - Within Group Only	
Call Forwarding Busy (CFB)	
Call Forwarding Busy Line (CFBL)	
Call Forwarding Do Not Answer (RES)	
Call Forwarding Fixed (CFF)	
Call Forwarding Fixed Activation (CFFA)	SERVORD command to activate CFF.
Call Forwarding Intragroup (CFI)	
Call Forwarding over Private Facilities	

Core network telephony service	Notes, restrictions, and limitations
Call Forwarding Remote Activation (CFRA)	CFRA programming is not supported directly from the user's SIP client using feature codes. CFRA programming will be supported using loop-heyaround trunks and a DISA DN to perform digit collection.
Call Forwarding Universal (CFU)	Requires programming, using Virtual Service Codes (VSCs).
Call Forwarding Usage-Sensitive Pricing (CFUSP / CFWUSP)	
Call Park (CPARK)	Active with Music On Hold (MOH). If the ANN selector is chosen for CPARK tuples in table AUDIO, the associated subfield for audible ringing (AR) is disregarded by SIP Lines. Regardless of the setting of the AR field, audible ringback will not be applied to SIP Lines prior to the MOH announcement.
Call Pickup (CPU)	Active with MOH.
Calling Identity Delivery and Suppression Delivery (CIDS DLV)	
Calling Identity Delivery on Call Waiting	
Calling Line Identification (CLI)	
Calling Name Delivery (CNAMD)	
Calling Name Delivery Blocking (CNAB)	
Calling Number Delivery (CND)	
Calling Number Delivery Blocking (CNDB)	
Calling Party Identification (CID (NTS_CID))	
Calling Plan - Incoming	
Carrier Preselection (CARR)	
Carrier Toll Denied (CTD)	
Change Line Termination Group	
Charge Number for Calling Number	Provides another number to charge instead of DN.
Code Restriction and Diversion	

Core network telephony service	Notes, restrictions, and limitations
Common Control Switching Arrangement (CCSA)	Part of base capabilities: Option CCSA identifies an IBN line as a member of a CCSA facility. A CCSA network provides private communications facilities between MDC and private branch exchange (PBX) customer locations. When an IBN line assigned option CCSA originates a call, an Automatic Message Accounting (AMA) record is generated, identified by call code 021. This AMA record is identified by call code 021 only if the call is non-billable (NP in subtable STDPRT) and option CCSA is assigned to the originating IBN line.
Custom Calling Service Access Codes	
Customer Data Change (CDC)	
Customer Originated Trace (COT)	Second-level activation with the CS 2000 prompting the user for digits is not supported.
Customer Originated Trace with AMA	
Deluxe Automatic Route Selection	
Denied Origination	
Denied Termination	
Dial '0' plus 7- or 10-Digits	
Dial Access to Private Facilities	
Dial Plan Management	
Dialed Number Identification Delivery	
Digit Interpretation	
Direct Distance Dialing (DDD)	
Direct Inward Dialing	
Direct Outward Dialing	
Direct Inward System Access (DISA)	Loop-around trunks are required for digit collection.
Directory Assistance	
Do Not Disturb (DND)	Programming of the Subscriber List Editing (SLE) list for enhanced DND is not supported.
Equal Access	Overlap Carrier Selection (OCS) is not supported because this feature starts translations and outpulsing while the subscriber is still dialing digits. With SIP, all the digits reach the CS 2000 in one SIP message request.
Free Number Terminating	

Core network telephony service	Notes, restrictions, and limitations
Full Carrier Toll Deny for International Carriers	
Fully Restricted (Originating)	
High Probability of Completion Network Capability	
IDDD Local Billing Control	
IDDD through an Operator System	
Interface to Directory Assistance System	
Inter-LATA Calls	
Inter-LATA Full Carrier Toll Denied	
International Calls	
International Primary Carrier	
Intra-LATA Calls	
Intra-LATA Full Carrier Toll Denied	
Intra-LATA Primary InterLATA Carrier (PIC)	
INWATS	IBN Class 5 variant for INWATS is supported (only variant of INWATS that was tested.) None of the WATS Line Card Carrier (LCC) is supported with SIP. Refer to the <i>North American DMS-100 Translations Guide Volume 13 of 20</i> , <i>North American DMS-100 Translations Guide Volume 8 of 20</i> .
Lawful Intercept (LI)	LI is supported for SIP lines for North American and international markets. In addition, for (I)SN09U, international support for SIP lines is added through a patch that can be activated. Refer to the appropriate LI documentation for details.
Line Study	
Local Call Detail Recording	
Local Number Portability. Location Routing Number (LRN) based.	
Local Number Portability Test	
Local Service Provider Switch Owner (LSPAO)	The LSPAO option allows subscribers to specify an LSP account owner. Subscribers can also specify if a line is native, unbundled, or resold.
Make Set Busy	Ring splash is not supported.
Make Set Busy Intragroup	Ring splash is not supported.
Manual/Time of Day Routing Control	

Core network telephony service	Notes, restrictions, and limitations
Meet-Me Conference	<p>The conference cannot be locked using Flash. The STD conference type is not supported by CS 2000 and therefore is not supported by SIP. The conference type of FLASHONLY should be used even though flash is not supported. No interaction with the Attendant Console is supported.</p> <p>Conference classes of A (alternate conferee class) and C (secondary conference on another switch) are not supported by CS 2000 and therefore will not be supported by SIP. Pressing # to start the conference before all conferees answer is not supported by CS 2000 and therefore will not be supported by SIP. The field IMMEDIATE in table PRECONF must be set to Y.</p> <p>The ADDON field in table PRECONF is not supported by CS 2000 and therefore will not be supported by SIP. The ATD field in table PRECONF is not supported by CS 2000 and therefore will not be supported by SIP.</p>
Message Waiting	
Message Waiting Indication	Visual indicator.
Multi-Switch Business Group (MBG)	Only provides name, number and translations. Reverse name display and redirection reason display is not supported.
Name Display	
Network Class of Service (NCOS) Restrictions	
NCOS Time of Day Routing	
Network Access Registers	
No Prefix Required for 7- or 10-Digit Calls	
Numbering Plan	
Off-Network-to-On-Network Conversion	
On-Network-to-Off-Network Conversion	
Operator Assistance - Dial 0	
Operator Number Identification	
Operator Services Access	
Outbound Call Dialing	

Core network telephony service	Notes, restrictions, and limitations
Outbound Call Routing	
OUTWATS	IBN Class 5 variant for OUTWATS is supported (only variant of OUTWATS that was tested). None of the WATS LCC is supported with SIP. Refer to North American DMS-100 Translations Guide Volume 13 of 20, North American DMS-100 Translations Guide Volume 8 of 20.
Payment Ceiling Advice	
Plug-Up (Trouble Intercept)	
Prefix '1' Allowed on 10-Digit Calls Only	
Prefix '1' Allowed on 7- or 10-Digit Calls	
Prefix '1' for 10-digit	
Prefix '1' for InterNPA Calls	
Preset Conference	
Prevent Delete Option	
Primary Inter-LATA Carrier	
Private Facility Access	
Public Safety	Basic emergency service is supported. In this release, Automatic Number Identifier (ANI) and Automatic Location Information (ALI) is delivered for all 911 calls from SIP clients. However, due to the flexibility of location with SIP clients, the actual physical location determination from the ALI system may not be accurate. Note that most client services are disabled during a 911 call.
Public Safety - Emergency Ringback/Operator Hold	Emergency Ringback/Operator Hold is not supported in this release.
PVN Priority Line	
Rate Area	
Two-way WATS	ETW (Enhanced Two-way WATS) is supported. None of the WATS LCC are supported with SIP. This is the only type of 2WW tested for SIP.
Remote Activation of Call Forwarding	
Remote Call Forwarding (Access to)	
Requested Suspension	

Core network telephony service	Notes, restrictions, and limitations
<p>Semi-Restricted (Originating)</p> <p>Semi-Restricted (Terminating)</p>	<p>As the Attendant console is not supported in this release, Semi-Restricted Originating is not supported.</p> <p>Only one version of Semi-Restricted Terminating is supported. The scenario is that A is in customer group 1 and B and C are in Customer group 2. C is the Semi-Restricted DN. In case of semi-restricted termination, A cannot call C directly but A can call B and, if B transfers the call to C, then A and C can be in speech path. This is the only version of Semi-Restricted that is supported. This will only work with IBN, as the option DIN is incompatible with RES.</p>
Service Codes N11	
Seven-digit Home Numbering Plan Area (HNPA) Dialing	
Simplified Message Desk Interface	
Simultaneous Ring (SIMRING)- group	
Simultaneous Ring (SIMRING) - personal	
Simultaneous Ringing (SIMRING)	
SMDI-SMDI Calling Number Delivery	
Special Billing	
Station Message Detail Recording (SMDR)	
Station Origination Restriction (SOR)	<p>A SIP client cannot be a SOR controller (SORC). A SOR controller must be able to enter digits interactively in response to prompts from the CS 2000.</p>
Station Restrictions	<p>No override mechanism in the first release, general term for DTM, DOR, SOR, MSB, DND, and so forth.</p>
Subscriber Line Usage	
Subscriber Programmable Ringing (SPRING)	<p>SPRING cannot be programmed directly from the SIP client using feature codes. A loop-around trunk must be used to perform remote programming. The use of UAS/AMS announcements to prompt the user while using SPRING is not supported. Tones are supported.</p>
Suppress Line Identification Information	
Suspended Service	

Core network telephony service	Notes, restrictions, and limitations
Transport Capability Application Process (TCAP)	
Terminating Billing Option	
Toll Essential	
Toll-free Services	
Toll-Restricted Service	
Tracing of Terminating Calls	Calling Line Identification (CLI) and Customer Origination Trace (COT).
Translations Plan	
Virtual Facility Groups (VFG)	The VFG Look Ahead feature is not supported. The field ORIGHOLD in tables VIRTGRPS and VFGDATA is not supported. Operator hold for E911 is not supported in the initial release.
Virtual Private Networks	
Voice mail	Core based for voice, no video voicemail.
Wide Area Centrex	

Limitations and restrictions

The following is a list of limitations and restrictions of the CS 2000 Core network telephony services:

- Flash support must be provided by the SIP client. There is no SIP message to the CS 2000 indicating the user has pressed Flash.
- All digits required for a phone call must be sent in the initial SIP INVITE message. Digits are not reported to the CS 2000 as they are dialed.
- For features that require additional digit collection from the user, a Packet Media Anchor (PMA) is inserted into the call, for example, Call Forwarding.
- If the SIP client supports the dialing plan, Network Call Forwarding supports programming with a single SIP INVITE message in the format of *featureCode#DN#. For example, programming Call Forwarding would require *72#6212100#.
- Network services accessed through SIP clients also inherit any general CS 2000 limitations for that service.
- The use of the octothorpe (#) is not supported by SIP clients at the start of a digit string. It usually indicates the end of the dialled digits. The CS 2000 supports the octothorpe or the asterisk as the leading digit of a feature activation code. For example *72 or #72 can be datafilled on the CS 2000 to activate Call Forwarding. SIP clients will generally not support the #72 example.

- Subscribers are limited to a maximum of ten active call legs (simultaneous sessions). Once they have reached their maximum, they will be unable to make further calls, including 911 calls.
- The current implementation of SIP Lines does not support supplying audible ringback before the application of an announcement or music media.
- Dialable Short Circuit (DSC) is not supported for SIP Lines.

SIP Client voice and multimedia services

Many of the available SIP voice and multimedia features are client controlled. As such, whether they are supported or not will differ depending on the SIP Client used. The table that follows lists the support of the SIP voice and multimedia features on Nortel's Multimedia PC Client and Personal Agent (PA).

For a description of the multimedia features, and features as implemented on Nortel's Personal Agent and Multimedia PC Client, refer to the following documents:

- *"Nortel MCS 5200/SSL Feature Overview"* (NN10251-115)
- *"Nortel Personal Agent User Guide"* (NN10039-113)
- *"Nortel Multimedia Client User Guide"* (NN10041-113)

Supported SIP voice and multimedia client telephone services

Multimedia service / feature	Nortel's SIP Multimedia PC Client	Personal Agent - Web portal
Address book / Directory with support for search and groups	Yes	Yes
Audio Conferencing - Ad hoc using the Media Application Server (MAS)	Yes	N/A
Audio Conferencing - Meet Me on the CS 2000 Core	Yes	N/A
Chat - 3+-person IM sessions using the MAS	Yes	N/A
Call - dial entered digits	Yes - digits entered using keyboard or mouse	No
Click to Call - 2 calls invoked from wireless client	No	N/A
Click to Call - 2 calls invoked with "calling digits" entered AND "called digits" entered by user	No	Yes
Click to Dial - from Address Book / Directory, Call Logs, Friend List	Yes	No

Multimedia service / feature	Nortel's SIP Multimedia PC Client	Personal Agent - Web portal
Click to Dial - from Microsoft Outlook	Yes	N/A
Clipboard Push	Yes	N/A
Customizable voicemail greetings	No	No
Customizable Ringback Tones	No	No
Customizable Ringing Tone - client based	Yes	No
Customizable Ringing Tone - network-based distinctive ringing	No	No
Customizable User Interface - in other words, branding/skins	Yes	N/A
Dynamic Call Handling <ul style="list-style-type: none"> • Answer • Decline ("Decline with reason" is not supported) • Ignore • Redirect • Forward to (for example, Voice Mail) <p>Equivalent DMS Voice feature capabilities mapped in "Voice Features" tab</p>	Yes	N/A
File Transfer	Yes	N/A
Friends List (in Groups)	Yes	Yes
Inbox / Outbox Call Logs (client based)	Yes	Yes
Inbox / Outbox Call Logs (network based)	No	Yes
Instant Messaging (IM)	Yes	No
Internet co-browsing	Yes	No
Microsoft Outlook Contacts Download	Yes	N/A
Picture Call ID	Yes	Upload point only
Presence (automatic and manual)	Yes	View Presence only
Secure Instant Messaging	Yes	N/A
Unified Communications / Voice Mail	Yes - through TDM VMS	No
User-Defined Call Screening - Block List, White List, Time of Day, for example	Yes - through PA-defined rules	Yes
User-Defined Call Routing - SIMRING, Find Me / Follow Me	Yes - through PA-defined rules	Yes

Multimedia service / feature	Nortel's SIP Multimedia PC Client	Personal Agent - Web portal
User Defined Call Routing - Simultaneous Ringing	Yes	Yes
User Defined Call Routing - Sequential Ringing	Yes	Yes
Video Calling - Point to Point (desktop)	Yes	N/A
Video Conferencing - Ad Hoc	Yes - 3-way through MAS	N/A
Video Conferencing - MeetMe	No	N/A
Voice Call Capability - Client controlled	Yes	N/A
Voice Call Capability - Do Not Disturb	Yes	N/A
Voice Call Capability - CEPT Three-Way Call (3WC)	Yes	N/A
Voice Call Capability - Six-Way Call	Yes	N/A
Voice Call Capability - Calling / Called Name / Number Display (before and after answer)	Yes	N/A
Voice Call Capability - Call hold	Yes	N/A
Voice Call Capability - Call Transfer	Yes	N/A
Voice Call Capability - Call Transfer / Call Waiting with Conference Call	Yes	N/A
Voice Call Capability - Call Waiting	Yes	N/A
Voice Call Capability - Voice Mail Indicator (audible / visible)	Yes	N/A
Web Collaboration - application sharing and publish mode, for example	No	No
Web Page Push	Yes	No
White Boarding	Yes	N/A

International tone support

The tone version played in a call between two SIP clients is based on the language set for the call ORIGINATOR. For example, if a subscriber from the United States (US) is placing a call to the United Kingdom (UK), the ring or busy tone inserted will be a US tone.

The following languages and associated country tone sets are supported:

- English NA
- English UK
- French

- German
- Spanish

6-way call limitations and behavior

Port allocation and reservation is done one session at a time. Attempts to join an additional party to the call will fail if all available ports are occupied.

Attempts to add a seventh party to the call will fail. The originator will hear a NACK tone and be reconnected with only the seventh party. The seventh party must either release or be released in order for the originator to rejoin the conference.

Six-way calls will come down if an active Session Manager fails over to the hot standby instance.

SIP Lines-supported clients

Supported clients

The following SIP clients are officially supported in this release:

- Nortel Multimedia PC Client (soft client)
- Nortel web-based Personal Agent
- Cisco* SIP IP Phone 7960 - Cisco Firmware release POS3-07-4-00.sb2. (version 7.4) (See http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7960/index.htm.)

Other clients may work but have not been thoroughly tested.

Nortel Multimedia PC Client and Personal Agent

For information on how to use, and the recommended hardware/software requirements for the Nortel Multimedia PC Client and web-based Personal Agent, refer to the following guides:

- *"Nortel Personal Agent User Guide"* (NN10039-113)
- *"Nortel Multimedia User Guide"* (NN10041-113)

Limitations and restrictions

The Multimedia PC Client has the following restriction:

- All voice calls need to be digit dialed. Using the user@domain format to invoke a voice call is not supported. Call dialing is performed by
 - selecting **Make A Call** and entering a dialing number
 - selecting a user in the Friends Online or an address book. Right-click and select **Call user@domain at <Dialing_number>**.

This method requires that the contact information for subscribers is datafilled with a valid dialing number.

The Personal Agent has the following limitations:

- Click to Call from the PA is not supported if the PA subscriber's line has the Subscriber Activated Call Blocking (SACB) option with ALL restrictions enabled.

With ALL restrictions enabled, a Click-to-Call attempt will result in a message on the PA stating that the Click to Call failed because the user was temporarily unavailable. The following is an example of the message when the PA times out:

```
Originating Party: <originator>
Terminating Party: 98001234567
```

```
Trying <originator> .....  
Click to Call failed  
Call to: <originator>  
Reason: <user temporarily unavailable>
```

Click to Call will only function properly with SACB if the SACB option is activated with restrictions on certain types of calls (such as 900 calls). In the supported scenarios, Click to Call is initiated from the subscriber to a restricted number. The subscriber's client rings and, after the subscriber answers, the client is connected to the packet media anchor so that the subscriber can enter the Personal Identification Number (PIN) to lift the SACB restriction and allow the call. After successful PIN verification, the subscriber is connected to the restricted number.

- Click to Call from a PA to a Core-based Meet Me conference is not supported. The Click-to-Call attempt will result in a message on the PA stating that the Click to Call failed because the user was temporarily unavailable. The following is an example of the message when the PA times out:

```
Click to Call Failed  
Call to: <Meet-Me Conference DN>  
Reason: User temporarily unavailable
```

- Star (*) code or feature access code (through 11+ fac) dialing using the Personal Agent is not supported.

Cisco 7960 hard client



Refer to the product documentation for information on the SIP phone and how to use it (See http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7960/index.htm).

Supported features

The table "Supported features for the 7960" (page 34) lists Cisco 7960 IP phone features supported in this release.

Supported features for the 7960

Feature	Comments
911	Routing provided by the Core.
Basic inbound call delivery	
Call forward locally on 7960	Programmed locally using the softkeys on the phone.
Call Forward through the Personal Agent	
Call Forward through the CS 2000 Core	(if through Feature Activation Code [FAC]) Handled by the Core.
Call return	(if through FAC) Handled by the Core.
Call tracing	(if through FAC) Handled by the Core.
Call waiting	User configurable on the phone.
Call waiting disable	User configurable on the phone.
Caller ID	User configurable on the phone.
CODEC - G711	Set to 20 ms
CODEC - G729	Set to 30 ms
Conference (Ad hoc; 3-way calling only)	Phone provides the audio mixing of G.711 packets only -- maximum 3-way call. Existing legs negotiated to G.729 will attempt to be renegotiated to G.711 when the conference is activated.
Do Not Disturb	User configurable on the phone.
Firewall support	Support for non-symmetric firewalls.
Hold (automatic hold)	Basic hold/retrieve performed on the phone.
Inbox	Phone has "Received Calls" and "Missed Calls" directories.
Lawful intercept (through Core)	Done by the Core.
Meet me Audio Conferencing	
Message Waiting Indicator (MWI)	
Multiple Lines/Users supported	Supports multiple users logged into the same device.
Outbound call dialing	
Outbound call routing	
Outbox	Phone has a "Placed Calls" directory.
Protocol - UDP	
QoS Type Of Service (ToS) Marking	

Feature	Comments
Server selection	Administrator can provision a single Session Manager address at the outbound proxy.
Transfer - Blind	Transfer failure cannot be retrieved.
Transfer - Consult	Transfer failure cannot be retrieved.
User-configurable settings	Requires user knowledge of phone configuration password. See " Limitations and restrictions " (page 35).
User-controlled codec selection	Requires user knowledge of phone configuration password.
Voice mail	

Limitations and restrictions

Certain call processing limitations of the 7960 result in the following services limitations:

- Only firmware release POS3-07-4-00.sb2. (version 7.4) of the Cisco SIP IP Phone 7960 is supported in this release.
- Calling Name/Number Blocking must be done using the star-code activation on the Core. The 7960 SIP implementation of this feature is not compatible with the CS 2000 Session Manager.
- Blind transfers that fail cannot be retrieved.
- Ad-hoc conferencing is accomplished as a 3-way call. The Cisco 7960 provides the audio mixing at the device itself. Thus, the maximum ad-hoc conference size is 3.
- Presence is not supported from the phone. Therefore, all presence indications or updates must be derived from the active call state or registration state. By default, Session Server Lines supports a presence state of "connected" for a registered user and "unavailable offline" for a user who is not registered. In addition to "connected," an On The Phone presence state is provided every time a call is active on the phone. Any additional presence states must be derived as if this device were a "Non SIP Lines device" by the Core.
- A SIP Lines inter-operability issue exists with the Cisco 7960 when using the G.729 codec. The Cisco 7960 client will only support an outgoing packetization rate of 30ms when set for G.729. The CS 2000 solution supports 10-ms and 20-ms packetization rates for G.729. Inbound calls will ring on the Cisco set, but the call will release when answered. Outbound calls will not be accepted by the gateway.

In addition, CS 2000 Session Server Lines does not support initiating a remote reset of the Cisco 7960 phone for this release. Since reset remote is not supported, the phones cannot initiate a firmware upgrade without manual intervention. All firmware upgrades must be done manually through editing the configuration files located on the TFTP server and resetting the phone. Once the phone is reset, it will compare the current firmware load in flash with the load specified on the TFTP server configuration file and initiate a download of the new firmware if the loads do not match. The network administrator is responsible for scheduling and initiating the reset of the phones for firmware upgrade. In addition, the TFTP server will not be integrated with the CS 2000 Session Manager solution but will rather be integrated with a separate server from any other CS 2000 Session Manager component.

Currently, there is no security mechanism in place to authenticate a remote reboot request through NOTIFY.

Likewise, provisioning of a CS 2000 Session Manager SIP Lines user will not automatically generate a phone-specific configuration file on the TFTP server. The administrator is responsible for creating and maintaining all the configuration files on the TFTP server for each phone in the network. See "[Configuration file settings](#)" ([page 37](#)) for more information.

The Cisco phone does not provide a NAT traversal mechanism to indicate in SIP signaling the presence of a firewall between the client and the CS 2000 Session Manager. To compensate for this, the CS 2000 Session Manager will compare the UDP packet source IP address against the SIP through the Header IP address to determine whether or not to treat this client as a firewalled client and replace the contact IP address appropriately.

The Cisco 7960 firmware version P0S3-07-4-00 causes a loss of voicepath during the following call scenario:

- Cisco phone calls a legacy PSET.
- The PSET answers the call and then parks it. The Cisco phone puts the call on hold as well.
- After the recall timer for call park expires, the PSET answers the call. The Cisco phone tries to resume the call by pressing the Resume softkey. A loss of voicepath occurs.

Loss of voicepath does not occur for this scenario in Cisco 7960 firmware version P0S3-08-2-00.

The Cisco 7960 firmware version P0S3-07-4-00 causes a call disconnection during the following call scenario:

- The Cisco client dials 911, and the call is answered by PSAP.
- The Cisco client presses the Hold key.

- After a few seconds the Cisco client presses the Resume key to resume the call.
- The call is reconnected.
- After approximately 2 minutes, the call is disconnected and the PSAP experiences a busy tone.

The 911 call disconnection may not appear in later Cisco 7960 firmware versions.

Interactions

Where applicable, services initiated from the Cisco phone will use existing SIP service implementation to accomplish the desired feature. However, due to the complex suite of services available to SIP Lines on the CS 2000 Core, some feature activation will be done by means of star-code feature activation codes (FACs).

The following services are activated or deactivated by the phone through the configuration menu system:

- Do Not Disturb
- Auto-Completion of numbers
- Call Waiting
- Call Hold Ringback
- Stutter Message Waiting
- Auto-Answer (Intercom)
- Speed Dial (up to 5 numbers)
- Call Forward Unconditional

These services will be permanently disabled on the phone and only available through FACs.

- Caller ID Blocking
- Anonymous Call Reject

Configuration file settings

Device configuration files need to be datafilled to allow proper interoperability between the Cisco 7960 and the Session Server Lines. The files reside in the root directory of an available TFTP server that all the 7960 phones are configured to use. If these default parameters are not applicable to all 7960 phones in the network, then multiple TFTP servers should be used or duplicate parameters included in the phone-specific configuration file.

- *SIPDefault.cnf* file

Contains the common SIP parameters needed by all the 7960 devices.

- *SIP<MACAddress>.cnf* file

Contains the parameters that apply only to the specific 7960 phone identified by the MAC address.

- *Dialplan.xml* file

Needs to be updated to allow proper feature activation code (FAC) star-code processing.

Common SIP parameters in *SIPDefault.cnf* file

The parameters defined in the *SIPDefault.cnf* file control the default behavior of the phone and provide the phone with the Session Server IP address and domain name that should be used. The file should be in the format described below with only the domain name and IP Addresses (highlighted in blue) changed.

The generic parameters to change are

proxy1_address

change to the root domain name

outbound_proxy

change to the IP address of the Session Server

```
# SIP Default Generic Configuration File

# Image Version
# image_version:   POS3-07-4-00

# Proxy Server
proxy1_address:"yourcompany.com"; Can be dotted IP or FQDN
proxy2_address:"                "; Can be dotted IP or FQDN
proxy3_address:"                "; Can be dotted IP or FQDN
proxy4_address:"                "; Can be dotted IP or FQDN
proxy5_address:"                "; Can be dotted IP or FQDN
proxy6_address:"                "; Can be dotted IP or FQDN

# Proxy Server Port (default - 5060)
proxy1_port: 5060
proxy2_port: 5060
proxy3_port: 5060
proxy4_port: 5060
proxy5_port: 5060
proxy6_port: 5060

# Proxy Registration (0-disable (default), 1-enable)
proxy_register: 1
```

```

# Phone Registration Expiration [1-3932100 sec] (Default - 3600)
timer_register_expires: 3600

# Codec for media stream (g711ulaw (default), g711alaw, g729a)preferred_codec: g711ulaw

# TOS bits in media stream [0-5] (Default - 5)tos_media: 5

# Inband DTMF Settings (0-disable, 1-enable (default))dtmf_inband: 1

# Out of band DTMF Settings (none-disable, avt-avt enable (default), avt_always - always avt )
dtmf_outofband: avt

# DTMF dB Level Settings (1-6dB down, 2-3db down, 3-nominal (default), 4-3db up, 5-6dB up)
dtmf_db_level: 3

# SIP Timers
timer_t1: 500 ; Default 500 msec
timer_t2: 4000 ; Default 4 sec
sip_retx: 10; Default 10
sip_invite_retx: 6 ; Default 6
timer_invite_expires: 180 ; Default 180 sec

##### New Parameters added in Release 2.0 #####

# Dialplan template (.xml format file relative to the TFTP root directory)
dial_template: dialplan

# TFTP Phone Specific Configuration File Directorytftp_cfg_dir:
"" ; Example: ./sip_phone/

# Time Server (There are multiple values and configurations see the Admin Guide for Specifics)
sntp_server: "" ; SNTP Server IP Address
sntp_mode: directedbroadcast; unicast, multicast, anycast, or directedbroadcast (default)
time_zone: EST ; Time Zone Phone is in
dst_offset: 1 ; Offset from Phone's
time when DST is in effect
dst_start_month: April ; Month in which DST starts
dst_start_day: "" ; Day of month in which DST starts
dst_start_day_of_week: Sun; Day of week in which DST starts
dst_start_week_of_month: 1 ; Week of month in which DST starts
dst_start_time: 02 ; Time of day in which DST starts

```

```
dst_stop_month: Oct           ; Month in which DST stops
dst_stop_day: ""             ; Day of month in which DST stops
dst_stop_day_of_week: Sunday; Day of week in which DST stops
dst_stop_week_of_month: 8    ; Week of month in which DST
stops 8=last week of month
dst_stop_time: 2             ; Time of day in which DST stops
dst_auto_adjust: 1           ; Enable(1-Default)/Disable(0)
DST automatic adjustment
time_format_24hr: 1         ; Enable(1 - 24Hr
Default)/Disable(0 - 12Hr)

# Do Not Disturb Control (0-off, 1-on, 2-off with no user
control, 3-on with no user control)
dnd_control: 0               ; Default 0 (Do Not
Disturb feature is off)

# Caller ID Blocking (0-disabled, 1-enabled, 2-disabled no user
control, 3-enabled no user control)
callerid_blocking: 2        ; Default 0 (Disable sending
all calls as anonymous)

# Anonymous Call Blocking (0-disabled, 1-enabled, 2-disabled no
user control, 3-enabled no user control)
anonymous_call_block: 2     ; Default 0 (Disable
blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)
dtmf_avt_payload: 96        ; Default 101

# Sync value of the phone used for remote reset
sync: 1                      ; Default 1

##### New Parameters added in Release 2.1 #####

# Backup Proxy Support
proxy_backup: ""             ; Dotted IP of Backup Proxy

#proxy_backup_port: 5060    ; Backup Proxy port (default is 5060)

# Emergency Proxy Support
proxy_emergency: ""         ; Dotted IP of Emergency Proxy

#proxy_emergency_port: 5060 ; Emergency Proxy port (default is 5060)

# Configurable VAD option
enable_vad: 0               ; VAD setting 0-disable (Default), 1-enable

##### New Parameters added in Release 2.2 #####

# NAT/Firewall Traversal
```

```
nat_enable: 1 ; 0-Disabled (default), 1-Enabled

#nat_address: "" ; WAN IP address of NAT box
(dotted IP or DNS A record only)
voip_control_port: 5060 ; UDP port used for SIP
messages (default - 5060)
start_media_port: 16384 ; Start RTP range for
media (default - 16384)
end_media_port: 32766 ; End RTP range for media (default - 32766)

#nat_received_processing: 0 ; 0-Disabled (default), 1-Enabled

# Outbound Proxy Support
outbound_proxy: "xxx.xxx.xxx.xxx" ; restricted to dotted
IP or DNS A record only
outbound_proxy_port: 5060 ; default is 5060

##### New Parameter added in Release 3.0 #####

# Allow for the bridge on a 3-way call to join remaining
parties upon hangup
cnf_join_enable : 0 ; 0-Disabled, 1-Enabled (default)

##### New Parameters added in Release 3.1 #####

# Allow Transfer to be completed while target phone is still ringing
semi_attended_transfer: 0 ; 0-Disabled, 1-Enabled (default)

# Telnet Level (enable or disable the ability to telnet into the phone)
telnet_level: 2 ; 0-Disabled (default), 1-Enabled, 2-Privileged

##### New Parameters added in Release 4.0 #####

# XML URLs
services_url: "" ; URL for external Phone Services
directory_url: "" ; URL for external Directory location
logo_url: "" ; URL for branding logo to be
used on phone display

# HTTP Proxy Support
http_proxy_addr: "" ; Address of HTTP Proxy server
http_proxy_port: 80 ; Port of HTTP Proxy Server (80-default)

# Dynamic DNS/TFTP Support
dyn_dns_addr_1: "" ; restricted to dotted IP
dyn_dns_addr_2: "" ; restricted to dotted IP
dyn_tftp_addr: "" ; restricted to dotted IP
```

```
# Remote Party ID
remote_party_id: 0 ; 0-Disabled (default), 1-Enabled
```

Device-specific parameters in *SIP<MACAddress>.cnf* file

The parameters defined in the *SIP<MACAddress>.cnf* file apply only to a specific 7960 phone identified by the MAC address in the file name. Parameters included in this file take precedence over duplicate parameters in the *SIPDefault.cnf* file. The file should be in the format of the following example with each user-field matching what is provisioned for the subscriber.

```
# SIP Configuration Generic File

# Line 1 appearance
line1_name: user10

# Line 1 short name
line1_shortcode: user10

# Line 1 Registration Authentication
line1_authname: "user10"

# Line 1 Registration Password
line1_password: "1234"

# Line 2 appearance
line2_name: user3

# Line 2 Registration Authentication
line2_authname: "user3"

# Line 2 Registration Password
line2_password: "1234"

##### New Parameters added in Release 2.0 #####

# All user_parameters have been removed

# Phone Label (Text desired to be displayed in upper right corner)
phone_label: "Phoenix SIP Lines"; Has no effect on SIP messaging

# Line 1 Display Name (Display name to use for SIP messaging)
line1_displayname: "user10"

# Line 2 Display Name (Display name to use for SIP messaging)
line2_displayname: "user3"

##### New Parameters added in Release 2.0 #####
```

```
# Phone Prompt (The prompt that will be displayed on console and telnet)
phone_prompt: "SIPPhone" ; Limited to 15 characters (Default - SIP
Phone)

# Phone Password (Password to be used for console or telnet login)
phone_password: "cisco" ; Limited to 31 characters (Default
- cisco)

# User classification used when Registering [ none(default), phone, ip ]
user_info: none
```

Dialplan definition in *dialplan.xml* file

To allow the proper handling of feature activation code (FAC) star-code processing, add the following entry in the *dialplan.xml* file:

```
<DIALTEMPLATE>
  <TEMPLATE MATCH="\*.*" Timeout="5"/> <!--Anything else-- >
</DIALTEMPLATE>
```

Session Server Lines platform

The Session Server Lines platform is a combination of the Session Server Trunks (formerly known as Next Generation Session Server) component running Linux and MCP multimedia software components. Select any of the following links to navigate to the descriptions in this section:

- ["Hardware platform" \(page 44\)](#)
- ["Software components" \(page 45\)](#)

Hardware platform

The Session Server chassis is deployed in a rack-mounted configuration house processing hardware, hard drives, Ethernet interface ports (two currently reserved for inter-Session Server communication for fault tolerance), and redundant power supplies. This hardware platform uses Network Equipment Building Standards (NEBS) Level 3-compliant hardware designed for telecommunications central offices or data centers, based on the Hewlett Packard cc3310 carrier-grade server. With two chassis working together to provide carrier-grade-level fault tolerance, this hardware configuration provides the platform for the SIP Gateway application.

- Minimum configuration:
 - 2 units (1 pair) for provisioning and database server. These units are part of the System Manager (SM) pair of servers.
 - 2 units (1 pair of Session Managers) for SIP Lines session server
- Maximum configuration:
 - 2 units (1 pair) for provisioning and database server. These units are part of the SM pair of servers.
 - 10 units (5 pairs of Session Managers) for SIP Lines session server

The Session Manager is deployed on its own set of redundant servers (servers 3 and 4). The System Manager, Provisioning Manager, and Database Manager are deployed on a single pair of redundant servers.

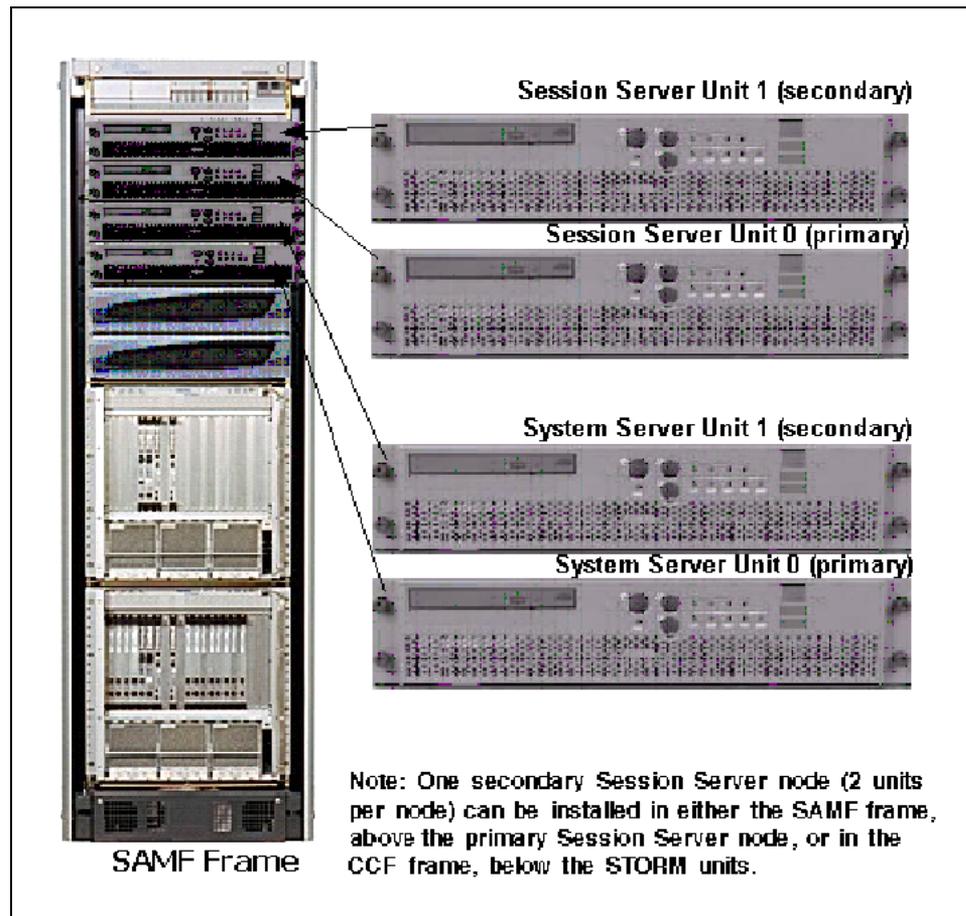
Features of each hardware platform unit include

- Dual 2.4GHz Xeon Processors
- 4GB DDR Memory
- Dual 73GB Hot Swappable Disk Drives
- CD-RW/DVD-R Drive
- Dual Hot Swappable Power Supplies

- Dual GigE Interface network interface cards
- Service Application Module-eXtremely Thin Server (SAM-XTS) platform, used with other Carrier VoIP components like STORM IA, is based on the compact HP cc3310 carrier-grade server, designed specifically for network elements, such as media gateways, signaling gateways, media servers, and soft switches.

The following diagram shows the location of the redundant pairs of Session Server hardware units in the SAM frame (SAMF).

Locating and identifying Session Server hardware in the SAMF frame



Software components

Session Server Lines uses the following software components:

- System Manager
- Provisioning Manager
- Database Manager
- Session Manager

These components are also used in other CVoIP solutions, such as MCS 5200. For more information on each of these software components, see the MCS 5200 product documentation.

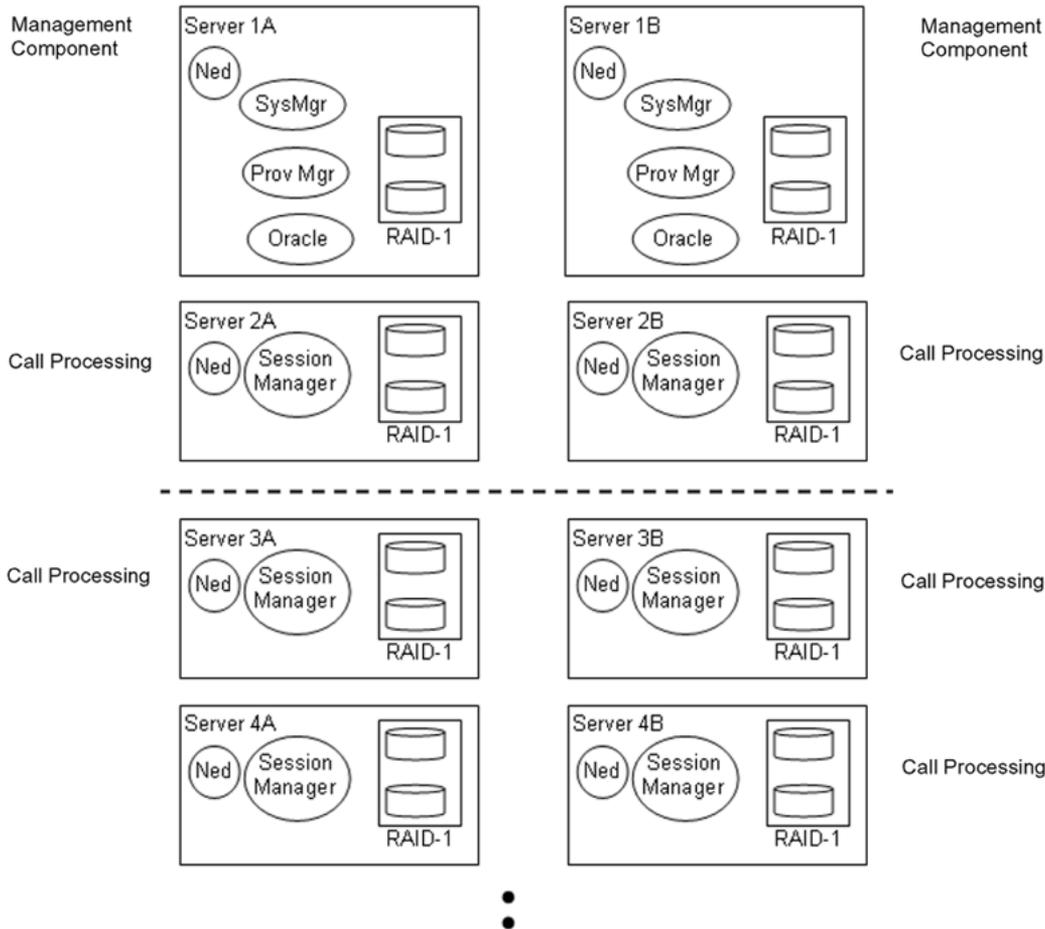
Server pair 1 hosts the following MCP elements:

Component	Description
System Manager	<ul style="list-style-type: none"> • Configures, monitors, and maintains all MCP-based network elements. • Serves as the Fault Performance Manager (FPM), which formats, stores, and forwards logs, alarms, and OMs. • Provides the interface for maintenance and restarts. • Terminates the Non-Call-Associated Signal (NCAS) from the CS 2000 Core. • Supports downstream maintenance links (such as IEMS). • Can stream logs northbound to IEMS/OSS. • The backup system manager runs in a HOT standby state and supports automatic failover. • Scaling factor: one redundant instance for each CS 2000.
Provisioning Manager	<ul style="list-style-type: none"> • Centralized provisioning point for SIP system data and subscriber data in the Session Server Lines. • Provides the Open Provisioning Interface to CS 2000 Management Tools (CMT) for data configuration. • Runs the SunOne web server through which client browsers connect to perform system-level provisioning actions. • Runs in load-sharing mode. • Scaling factor: Initially one redundant instance for each CS 2000. Capacity may need to be increased based on PC Client and Personal Agent usage.
Database	<ul style="list-style-type: none"> • Provides the storage of persistent data for the system. Currently built on the Oracle DBMS. • Redundancy is achieved through Oracle replication. • Scaling factor: One redundant instance for each CS 2000.
Ned (Network Element Daemon)	<ul style="list-style-type: none"> • Ned is a process that runs on each server machine, oversees the lifecycle of the MCP software components, and cleans up logical network interfaces (where applicable) should a component experience an unexpected, catastrophic failure. • CS 2000 SIP Lines are controlled by a Hard Usage SOC for each line. SOC code CS2C0005 (preliminary) must be activated.

Server pair 2 hosts the following MCP components:

Component	Description
SIP Session Manager	<ul style="list-style-type: none">• Runs the main SIP application portion of the CS 2000 Session Server.• Terminates SIP messages and performs conversion to GCP.• Manages registration and knowledge of client availability.• Provides state machine for driving SS-based subscriber services (for example, Personal Agent) and MAS insertion.• Supports presence and instant messaging.• Caches subscriber profiles from Provisioning Server• The backup SIP Session Manager runs in hot standby.• Scaling factor: One redundant instance for each set of 35,000 subscribers.
Ned	As described above.

Session Server Lines configuration



Fault Tolerance

The System Manager, Session Manager, Provisioning Manager, and Database fault tolerance mechanisms and recovery procedures are described in their respective MCS 5200 documents.

Redundant Array of Independent Disks (RAID) Level 1 (disk mirroring) is used by the Session Server Lines servers. RAID-1 allows the continued operation of the system in the event one of the disk drives should fail.

The Session Manager makes use of the call checkpointing. This feature adds the ability to checkpoint Calls and Subscriptions to the hot standby instance of a Session Manager. The checkpointing functionality supports

- Calls in an active stable state are checkpointed. An active stable call is one that has been answered.
- The Presence of those on the watched list are checkpointed. The accuracy of a watched Presence list remains intact after a failover.

- Call logs are checkpointed. The call log information is preserved so that a user will be able to get a record of their calls from the Personal Agent even after a failover.

Session Manager checkpointing limitations and restrictions

There are limitations with checkpointing. For example, if a standby instance becomes active due to a failover, two possible problems exist. Both problems occur where a Checkpoint was in process at the time the failover occurred. The newly active system can have calls active that were in the process of disconnecting and the checkpoint was not transferred before the failover occurred. In addition, calls that were just set up before the failover will not appear on the newly active instance. This leads to inconsistent state information between the SIP Line clients, the Session Manager, and the GWC.

Several audits go active once the standby instance becomes the active instance to correct the inconsistencies. The first audit causes the Session Manager to send a SIP message to all SIP Line clients the Session Manager thinks are active on a call to verify that they are actually in a call. In addition to the audit of the SIP Lines the Session Manager will send a message to the GWC for each of the calls it has as active, thereby bringing the nodes into sync.

Due to the above limitations on checkpointing, not all of the calls to a subscriber will be logged. Any entry in the local call log table in the process of being checkpointed at the time of the fail over will be lost. Unfortunately, there is no way to mediate this loss of information. Subscriptions have the same limitations as the call logs in that there is no way to audit them from the Session Manager. However, any subscription that is lost will be refreshed by the client within one hour, which is the expiration time returned by the Session Manager upon processing a subscription.

Advanced services are not guaranteed to work after the failover of the call. The only service guaranteed to work is the release of the call by either party in the call. Other messages will be sent a 500 service unavailable if the Core is not able to process the message.

For SIP Lines, checkpointing does not support call scenarios that involve call transfer. The following features involve call transfer and therefore do not support checkpointing:

- 6-Way Call
- Ad hoc Conference
- Blind Call Transfer
- Consultative Call Transfer
- Click-to-Call

System Manager automatic failover limitations and restrictions

Automatic failover of the active System Manager to the warm standby instance has the following limitations:

- Log and Operational Measurement (OM) files are written to disk only on the active SM instance. When that instance is no longer active, the files are not available to the newly active instance. Files written to the file system of a failed System Manager instance will remain on that file system and new files will be written to the disk of the newly active instance.
- The current SNMP sequence number for northbound traps will not persist over a failure. The OSS will need to resync their alarm snapshots after a failover.
- The current NCAS Link failover behavior will be maintained.
- An Open Management Interface (OMI) client will get a RemoteException if the System Manager fails during an OMI call.
- Any partially complete database activities will be rolled back by the database. Current operations will not complete.
- An in-progress change to subscriber data through the Provisioning Manager at the time of a System Manager failover may result in the newly active System Manager viewing a Session Manager as being out of sync. This may require a restart of one or more Session Managers.

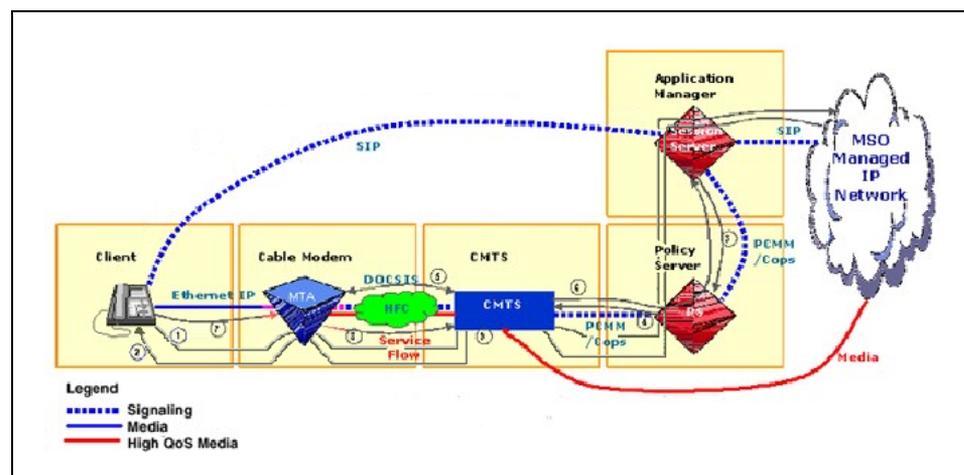
PacketCable Multimedia

This section describes PacketCable Multimedia (PCMM). Click on the following links to navigate to the descriptions:

- ["Emergency support" \(page 52\)](#)
- ["Hardware requirements and dependencies" \(page 52\)](#)
- ["Software requirements and dependencies" \(page 52\)](#)
- ["Limitations and restrictions" \(page 53\)](#)
- ["Interactions" \(page 53\)](#)

PacketCable Multimedia (PCMM) is an architecture developed by CableLabs. PCMM enables multi-service operating companies to deliver Quality of Service (QoS)-enabled multimedia services over Data Over Cable System Interface Specification (DOCSIS) protocol networks. PCMM extends the dynamic QoS architecture developed as part of PacketCable 1.X to allow application managers in the Multiple Service Operator (MSO) network to request and obtain QoS treatment for multimedia traffic flows on behalf of an end user.

The PCMM signaling implementation is compliant with PacketCable specification PKT-SP-MM-102-040930. The following diagram and the supporting list below it illustrate a high-level call flow for a PCMM SIP call.



1. The SIP Client originates a session application (SIP Invite) to the Session server and into the network to a destination client.
2. The destination responds with an OK (200 OK) to the SIP Client through the Session server.
3. The Application Manager on the Session server requests managed QoS on behalf of the SIP client by signaling to the Policy Server.
4. The Policy Server makes policy decisions and sends the request to the CMTS.
5. The CMTS uses DOCSIS signaling to create service flows in the cable modem.
6. The CMTS responds with a gate-id, which is passed through the Policy Server to the Session server.
7. The SIP Client sends data and media to the cable modem.
8. The cable modem puts data on the appropriate managed service flow to the CMTS based on classifier (for example, IP address or port).
9. CMTS enforces policy for data on the service flow and network using the gate.

Emergency support

The priority contained in the PCMM Gate-Set message is raised to the highest level when a SIP call is made to an emergency number. If the CMTS reserves bandwidth for high priority calls and the bandwidth usage is high, then this could allow an emergency call to receive quality of service when other calls do not. Also, the priority setting could be used to prevent other calls from using the bandwidth of an emergency call. The provisioning of emergency numbers is accomplished using the Provisioning Client's Emergency Number and Emergency Alias Lists.

Hardware requirements and dependencies

To deploy the PCMM feature, the Multiple Service Operator (MSO) needs to have PCMM-compliant cable network elements (not supplied by Nortel). These third-party elements include cable modems, CMTSs, and a policy server.

For acceptable voice and video quality when using PC clients, ensure that the computing platform meets the recommended hardware and software configuration.

Software requirements and dependencies

The policy server and CMTSs in the cable network must support PacketCable Multimedia specification PKT-SP-MM-I02-040930. The cable modems must support DOCSIS 1.1 or greater.

Limitations and restrictions

Failover for PCMM does not support the processing of unsolicited messages received from the policy server after the switch of activity. Therefore, the following scenario could occur:

- A SIP session is answered, and it has a PCMM service flow.
- The active Session Manager fails and the standby Session Manager takes over.
- The CMTS detects and reports a loss of communication (through the Policy Server) to the Session Manager for the SIP session.
- The call will lose its PCMM service flow and resort to Best Effort quality of service.

Interactions

The PCMM feature sets up managed quality of service for SIP calls originating from or terminating to the cable network. PCMM operates on a half-call basis. This means that PCMM signaling and managed QoS happen only for the half of the call that is in the cable network. For example, an on-net to off-net call will only do PCMM signaling for the originating line half of the call. The trunk half of the call is not touched by PCMM.

No additional client-side provisioning is required to enable PCMM. The PCMM feature supports the SIP clients as noted in "[SIP Lines-supported clients](#)" (page 32).

For managed QoS, you must add the QoS Packet Scheduler on the PC. For further information on managing QoS on the PC client, see "*Nortel Multimedia Client User Guide*" (NN10041-113).

Basic call

The PCMM feature attempts to set up managed quality of service when the Session Description Protocol (SDP) information is known for both of the media endpoints (originating and terminating). Therefore, application of the PCMM protocol procedures will not cause call failures, but it is possible that under some network error conditions a call might receive "best effort" QoS rather than managed QoS.

Upgrades

SIP Lines uses the patching and upgrade strategies of the network components over which it runs.

No upgrades of the Session Server Lines platforms are required in this release. If a network is running the SN08 version of SIP Lines, the lines will migrate as part of the solution upgrade to SN09 without any issues or special procedures.

For further information on upgrades, please contact your next level of support.

Accounting management

This section describes account management. Click on the following links to navigate to the descriptions:

- ["Accounting management strategy" \(page 57\)](#)
- ["Limitations and restrictions" \(page 57\)](#)
- ["Tools and utilities" \(page 58\)](#)

Accounting management strategy

All billing is performed on the Core. SIP Lines are integrated into the existing accounting methods used for IBN and RES line types. This existing functionality creates records using the Automatic Message Accounting (AMA) billing Module 260 for SIP voice calls.

Module 260 is a new option in the table AMAOPTS called RECORD_MC260. This option identifies when SIP clients are used as originating or terminating agents. When activated, the option does not force billing, but rather collects the additional information for existing billing scenarios.

By default Module 260 is deactivated. This option is activated by changing the option's SCHEDULE to ON.

Limitations and restrictions

The CS 2000 does not control client-based features; therefore, these are billed as simple call originations and/or terminations. For example, a local client feature such as Ad-hoc conferencing cannot be billed or pegged as 3WC, since the CS 2000 has no knowledge of the client actions.

No IPDR records are generated in this release for multimedia features. Flat rate billing must be used.

Activity limited to the Session Server Lines platform does not generate any accounting records.

Tools and utilities

SIP Lines use the existing CS 2000 tools for collecting and managing the AMA records for the voice transactions.

For information on CS 2000 billing systems for lines, see *"North American ATM/IP Solution-level Accounting"* (NN10412-800).

Security and administration

This section describes security and administration. Click on the following link to navigate to the description:

- ["Security and administration management strategy" \(page 59\)](#)

Security and administration management strategy

On the Core, SIP Lines use the existing security and administration management strategy described in the Succession SN09 documents.

For more information on the strategy used by the Gateway Controller see *"Gateway Controller Security and Administration"* (NN10213-611).

The Session Server Lines platform adheres to the security strategy described in the MCS 5200 documents for the System Manager, Provisioning Manager, Database Manager, and Session Manager.

For further information on backup, restore and recovery activities, see *"Nortel Session Server Lines Administration and Security"* (NN10437-611).

Nortel Session Server Lines

Fundamentals

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