SYSTEM RELEASE NOTES

Release 3.10.1.3

Table of Contents

1. Scope

This document provides information regarding the Plexus 9000 system software version 3.10.1.3 for the System Processor (SP). Topics covered are:

- File contents and notes
- New features and product improvements
- New and modified TL1 commands
- Known hardware limitations
- Known software limitations

All hardware supported by release 3.10.1.3 is listed by CLEI code and part number in your copy of the *Planning and Engineering Guide*. This information is listed under the Part Information section in the manual.

2. Upgrade Files

The following files are required to upgrade the Plexus 9000 to version 3.10.1.3:

3.10.1.3_cpu.tar.gz
3.10.1.3_ds1.tar.gz
3.10.1.3_ds3.tar.gz
3.10.1.3_trids3.tar.gz
3.10.1.3_trids3_3.tar.gz
3.10.1.3_octds3.tar.gz
3.10.1.3_octds3_2.tar.gz
3.10.1.3_octds3_3.tar.gz
3.10.1.3_voip.tar.gz
3.10.1.3_voip6.tar.gz
3.10.1.3_ena.tar.gz

3. Upgrade Notes

- The In-Service Upgrade feature from a version lower than 3.10.1.3 is presently not supported.
- The In-Service Upgrade Abort feature is not supported.

4. EMS/Billing/Traffic Server Requirements

- Element Management Server (EMS) version 9.6.2 or higher is required for 3.10.1.3.
- TelicaPlexusDc version 3.10.1.2.1 or later is required for 3.10.1.3.
- Traffic Collection Application (TCA) 2.0.0.78 is required for 3.10.1.3.

5. New Features and Product Improvements

For software version 3.10.1, version 3.8.3 functionality has been merged with version 3.9.0. This provides functionality of Class 5 features, Gateway MSC and Class 4 routing.

This release of system software provides the following features and improvements.

5.1 New Modules

5.1.1 High Performance DS3 Modules

New Triple and Octal DS3 IOMs, part numbers 89-0424 and 89-0425, provide increased performance over current Triples and Octal DS3/STS-1 IOMs, part numbers 89-0397/ 0398/ 0410/ 0411. The actual performance increase is protocol (SS7, ISDN, CAS, GR-303) and interface -configuration (CAS, GR-303) dependent. These high-performance IOMs can be protected by corresponding 89-0424/0425 IOMs as well as by 89-0410/0411 IOMs. Rear Octal DS3 module, 89-0383, and rear Octal Protection module, 89-0386, are required. They are only supported in a chassis with Midplane 2 (85-3004 chassis) and Midplane 3 (85-3007 and 85-3008 chassis).

5.1.2 System Processor 3 (SP-3)

SP-3, 89-0406, provides increased call processing performance over SP-2 (89-0389). Actual call processing increase is dependent on the call model. Call processing capacity for various types and combinations of ingress/egress lines and trunks along with AIN processing and billing will be provided in the future. The associated rear SP-3 module, 89-0417, currently supports two 10/100BT interfaces. There are five physical hardware interfaces on the module, but the additional three will be supported in a future software release. These modules are only supported in a chassis with Midplane 2 (85-3004 chassis) and Midplane 3 (85-3007 and 85-3008 chassis).

5.2 SIP Enhancements

5.2.1 SIP SUBSCRIBE /NOTIFY for DTMF Relay

The use of SUBSCRIBE/NOTIFY messages, which is a standard for transporting events in SIP, is now supported for DTMF relay to an application. This allows for applications to get DTMF tones without having bearer traffic going through it.

5.2.2 SIP REFER

The SIP REFER method is now supported for doing call transfer. It is a more efficient method for doing call transfer than using re-INVITEs that can minimize hairpinning. It can also be used for replacing or collapsing call legs (similar to ISUP RLT or ISDN TBCT). Both attended and unattended call transfers are supported.

5.2.3 SIP Announcements

There is now support for an application that uses SIP to tell the Plexus to play announcements and digit readout. In this release, the Plexus also announces the Directory Number, Digits and Numbers in Spanish.

5.2.4 SIP SUBJECT Header Pass-through

The Plexus will pass the SIP SUBJECT Header transparently from the incoming to the outgoing SIP call. This allows for two end devices to pass information between themselves through the Plexus.

5.2.5 SIP Session Timer

SIP now uses the Session Timer feature to audit active calls to make sure they have not unilaterally gone away. This allows for the cleanup of lost sessions and frees up resources.

5.2.6 SIP PRACK

SIP provisional response acknowledgements (PRACKs) are now supported (e.g., 101-199 responses). This is needed to ensure that provisional responses are not lost.

The Plexus now supports the sending of the supported header field in the INVITE to indicate support of specific functions (i.e., option tags). Some SIP devices require the setting of the supported header with appropriate options before they will support those functions. 100rel is used for the sending and receiving of PRACKs.

5.2.7 SIP Hold (a=inactive/sendonly)

The Plexus previously supported the method of IP address of 0.0.0.0 in the SDP for putting calls on hold. The Plexus now will also support the newer method of putting calls on hold by using a=inactive or sendonly and retrieving a call by using a=sendrecv. The Plexus will also support an IP address of 0.0.0.0 as well.

5.2.8 SIP UDP Fragmentation

If SIP packets exceed 1500 bytes, the Plexus will now fragment the packet into smaller packets. This will minimize the probability of the large packets being discarded by intermediate routers.

5.2.9 SIP-T

This standard for transporting ISUP over SIP is now supported. The Plexus currently supports the standard BICC CS2 (Bearer Independent Call Control Capability Set 2) to transport ISUP over IP today. The support for SIP-T adds additional options for interoperability with third-party softswitches.

5.2.10 SIP - ‘+’ Support (E.164 numbers)

This feature adds E.164 number support to the SIP Uniform Resource Identifiers (URIs) with the ‘+’ pre-pended to the phone numbers. Within the Plexus, the presence of the ‘+’ in the phone number treats the number with a Numbering Plan Indicator (NPI) of ISDN. A lack of '+' should indicate that the number be assigned an NPI of Private. The support for ‘+’ is now configurable.

When the PLUS (‘+’) feature is enabled on the trunk group, if the Plexus receives + in the Request-URI it will make the CDPN’s NOA/NPI of an International Number / ISDN. If there is no +, then it will keep it an UNKNOWN/ UNKNOWN.

If the PLUS feature is enabled and Plexus receives + in the FROM or Remote-Party-ID header, it will make the CGPN’s NOA/NPI a Unique International Number / ISDN. If it does not receive +, it will make CGPN a Non-Unique National Number / ISDN.

5.2.11 SIP Trunk Group Heartbeats

Heartbeats are used to determine if a SIP Trunk Group is up or down. It allows for the Plexus to be able to route advance without the added SIP messaging and delays associated with relying on SIP timeouts on every call. The Plexus uses the OPTION message for heart-beating.

5.2.12 Configurable SIP INVITE Retries

The standard SIP timeout/retries for INVITE takes 32 seconds before the call is timed out. This is too long for maintaining a call setup especially with route advancements to the next trunk group. The timers are already configurable, with a minimum of 500ms, and now the number of retries is also configurable.

5.2.13 SIP-PRI Interworking Corrections

Previously, in the interworking of SIP to PRI calls, there was an incorrect and extra '180 Ringing' message with SDP.

5.2.14 SIP Remote-Party-ID Interworking Corrections

Enhancements have been made for SIP Remote-Party-ID interworking including using 'Anonymous' instead of 'Restricted' in the From field if the presentation is restricted and always including the 'Remote-Party-ID' even if presentation is not restricted.

5.2.15 Multiple SDP Profiles against SIP Proxies

SIP Proxies often front-end different networks/services, where each network/service has different SDP requirements. This feature provides support for different SDP profiles being associated with calls going to a common SIP Proxy, without requiring any changes to the SIP proxy.

For example, Service A only allows G.711 but Service B allows G.711, T.38, G.7.29ab and G.723.1. Previously, in order to support both services simultaneously, the Plexus 9000 would have had to reserve resources for G.729 or G.723.1which would have resulted in lower density, and less flexibility since only G.729 or G.723.1 could be offered.

5.2.16 SIP Call Processing Performance

Call processing performance has been enhanced in this version for the following ingress⇔egress call types:

- SS7⇔SIP
- SIP⇔SS7
- PRI⇔SIP
- SIP⇔PRI

5.3 MGCP NAT Traversal Modes

Support has been added for MGCP through NAT Traversal devices such as Session Border Controllers, VoIP-aware firewall and NAT servers. It also provides support for dynamic IP addressing by using the endpoint name identification instead of the IP address. The MGCP endpoint name is that used in MGCP signaling messages to identify the port (for example, aaln/1@10.0.0.1, aaln/2@foo.bar, aaln/1@03FDE1233212).

5.4 G.723.1 CODEC

- Support for a 60ms packet size for G.723.1 CODECs is now provided as well as 30ms. If other values, such as 10, 20, or 40ms are provisioned, they will be ignored and 60ms will be used by default.
- G.723 annex A (silence suppression) is not supported.

5.5 G.726 Encoding

The Plexus now supports both little-endian (least significant byte first) and big-endian (most significant byte first) encoding for G.726. Some devices, such as the Cisco 5300, support big-endian and others, such as Cisco 2421 IADs, support little-endian.

5.6 T.38 Fax

- The Plexus supports T.38 over UDP only. TCP is not supported.
- The Plexus does not revert to voice at the end of a fax transfer. Once a T.38 fax call is established, the call remains in T.38 mode for the duration of the call.
- The Plexus no longer automatically reverts to G.711 on a fax/modem tone for Fax and Modem Pass-through operation for calls with a compressed CODEC. Upon detection of a fax/modem tone for calls originating with a G.72X CODEC, the Plexus will re-negotiate G.711 via the SDP. A G.711 CODEC must be provisioned in the SDP profile for this re-negotiation to occur.
- Invalid performance monitoring and call traffic statistics are reported for T.38 connections.
- When T.38 is used between the Plexus and any other gateways such as the Cisco AS 5300/5400, the format of the ENA must be provisioned for ENET_802_3.

5.7 Real Time Control Protocol (RTCP)

- Real-time Control Protocol (RTCP) v 2.0 (RFC 1889, updated by RFC 3550) is supported in this version.
- The following RTCP packet types are supported:

    - Sender Report (SR)
    - Receiver Report (RR)
    - Source Description Items (SDES)
    - Bye
- The UDP port number that is assigned to the RTCP session corresponding to a given RTP session is equal to the next higher port number.
- A 5-second average interval is maintained between RTCP packets transmission. The actual calculated interval between RTCP packet transmission varies randomly between 2.5 to 7.5 seconds.
- The following RTCP statistics for each call are calculated and available via TL1 and the PlexView EMS:
    - Octets Transmitted
    - Packets Received
    - SID Packets Transmitted
    - Avg. Inter-Arrival Jitter
    - Octets Received
    - Packets Received
    - Packets Lost
    - SID Packets Received
    - Packets that didn't make it to the peer
    - Peer's Inter-Arrival Jitter
    - SDU Lost Count
    - Current Jitter Buffer Count
    - Round Trip Delay

5.8 DTMF Relay Negotiation Fixes/Enhancements

These are enhancements regarding how the Plexus does DTMF Relay negotiation within the SDP.

5.9 Selective Calling Features

5.9.1 General Requirements

In general, to use one of the selective calling features, the subscriber enters a two-digit vertical service code. He then receives announcements and prompts as follows, but not necessarily in this exact order:

- The name of the service (for example, SCF)
- The current status of the service (for example, active or inactive)

- The current size of the customer's screening list
- The customer's remote forwarding services
- Actions and associated dialing codes available to the user for the particular service:
  - Confirm or change a remote directory number (if applicable)
  - Add an entry to the screening list
  - Delete an entry from the list
  - Review the list
  - Change the status (for example, active to inactive or inactive to active).
- Confirmation of status change

5.9.2 Selective Call Forwarding (SCF)

SCF allows a subscriber to forward an incoming call if the calling party is on the subscriber's selective call forwarding list. The customer can construct or modify a telephone number screening list by dialing an activation code. The Plexus will screen incoming calls against the customer's list and forward only those telephone numbers on the list.

This feature allows the subscriber to create a list of up to 32 telephone numbers. Incoming calls from these numbers may be forwarded to another number instead of being completed at the subscriber's telephone number. All other calls are completed as usual.

The subscriber can provision using the star codes. The star codes are configurable, but by default are *62 to activate and *83 to deactivate.

5.9.3 Selective Call Rejection (SCR)

SCR allows a subscriber to reject calls from any party that is programmed into the subscriber's selective call rejection list. Rejected calls are routed to the selective call rejection intercept treatment. The subscriber can create a list of up to 32 telephone numbers.

The subscriber can provision using the star codes. The star codes are configurable but by default *60 is used to activate and *80 to deactivate this feature. There are recorded instructions to allow set-up, review or changes to the list of phone numbers to be rejected or to turn the feature on/off. The last call received can be added by dialing #01#.

5.9.4 Selective Call Acceptance (SCA)

SCA allows a subscriber to accept only calls from a party that is programmed on the subscriber's selective call acceptance list. The subscriber can create a list of up to 32 telephone numbers. Incoming calls from these numbers are accepted and all other calls are forwarded to an announcement.

The subscriber can provision using the star codes. The star codes are configurable, but by default *64 is used to activate and *84 to deactivate this feature. There are recorded instructions to allow set-up, review or changes to the list of phone numbers to be accepted or to turn the feature on/off. The last call received can be added by dialing #01#.

5.10 Subscriber Provisioning Using Star Codes

Previously, Call Forward Busy Line (CFBL) and Call Forward Don't Answer (CFDA) were subscription services on the Plexus meaning that the forwarding number was not user provisionable. Now the subscriber can activate or deactivate this feature using star codes in the same manner as currently supported for call forward variable (CFV). The codes are *92 to activate and *93 to deactivate.

5.11 Inter-Switch Voice Mail

Previously, the Plexus offered the service of a voice mail system (VMS) to subscribers on the switch. Now this service is extended so that subscribers served by other switches in the network can be served by a VMS on the Plexus. In addition, if the VMS resides on another switch then subscribers on the Plexus are able to receive voice mail services from the VMS of the remote switches. Refer to:

GR866-ISDN Message Service Generic Switching Signaling Requirements
GR1401-Visual Message Waiting Generic Requirements

5.12 4ESS Variants

The Plexus 9000 now supports 4ESS variants for ISUP and ISDN.

5.13 Interworking Enhancements for 4ESS PRI and 4ESS ISUP

The Plexus 9000 now supports interworking between the 4ESS variants for ISUP and ISDN.

5.14 MTX Single Port Release Link Trunk

The Mobile Telephone Exchange (MTX) Single Port Release Link Trunk (SPRLT) enables the Plexus 9000 acting as an enhanced service platform to release a call from itself back to the Nortel MTX after it has completed its task (for example, Directory Assistance). SPRLT optimizes ports that were utilized on the service platform and makes them available for the deployment of additional services.

5.15 Release Link Trunk (RLT) Billing

The existing CDR will be able to handle the enhancement without adding new fields. The primary modifications include the following:

- Passing new Service Logic ID codes and call-handoff (RLT) information (terminating number or original called number) for module 40 population.
- Supporting Facility Accept (FAA) messages for calculating elapsed call times.**Class 4 Subscriber Enhancements** The maximum number of Class4 Subscribers has been increased to 500,000 from 250,000.

5.17 Pseudo STP - Tandem TCAP for CAP & GROW

This feature allows the Plexus (MTP3) to "tandem/transfer" SS7 messages received by, but not destined for the Plexus. Previously, MTP3 rejected all the messages with the DPC not equal to the Plexus Point Code, as defined by the specifications for a node acting as an SSP in the SS7 network. This additional functionality adds one of the properties of an STP to the Plexus.

Note: Complete STP functionality has not been implemented.

5.18 Time of Day (TOD) Routing Enhancements

The resolution for TOD routing has been increased to 15 minutes allowing a new transplan and route to be selected and/or changed every 15 minutes instead of every hour.

5.19 Authorization Codes Enhancements

The maximum number of authorization codes has been increased to 10,000 from 500. The maximum character length for authorization codes and PINs has been increased to 16 from 8.

5.20 Account Codes Enhancements

The maximum character length for account codes has been increased to 16 from 8.

5.21 Spanish Language Support for Variable Announcements

Plexus 9000 announcements contain a fixed part and a variable part. The fixed part is typically a set of pre-recorded phrases that are provisioned to one or more announcement IDs. Variable announcements typically comprise numbers or currency and are selected dynamically and combined with the fixed parts of an announcement. Previously, the Plexus 9000 only supported variable announcements in English. This enhancement adds support for variable announcements in Spanish and English.

5.22 New Commands

Following is a list of routing commands in version 3.10.1.3.

- ED/RTRV-GMSC-ANSISYS, which is used to modify or retrieve the system wide ANSI IS41 network parameters for the Gateway Mobile Switching Center (GMSC) functionality. This command appears under the heading “Global Configuration Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-665, for procedures.
- ED/RTRV-GMSC-GSMSYS, which is used to modify or retrieve the system-wide Global System for Mobile Communication (GSM) network parameters for the GMSC functionality. This command appears under the heading “Global Configuration Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-665, for procedures.
- RTRV-PM-GMSC, which is used to retrieve current and historical performance data related to GMSC messages. This command appears under the heading “Global Configuration Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-666 for procedures.
- INIT-REG-GMSC, which is used to initialize one or more storage registers or event counters for GMSC message statistics. This command appears under the heading “Global Configuration Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-666 for procedures.
- ENT/RTRV/DLT-GMSC-LAESDN, which is used to designate a wireless subscriber as being under CALEA surveillance. This command appears under the heading “CALEA Commands” in the *TL1 Commands Reference Guide*.
- ENT/ED/DLT/RTRV-LIST-TRIGESC, which is used to add, modify, delete or retrieve the range of directory numbers associated with a specified trigger escape list. This command appears under the heading “Intelligent Network Functions Commands” in the *TL1 Commands*

Reference Guide. Refer to the *Maintenance and Troubleshooting Guide*, DLP-619, for procedures.

- ENT/ED/RTRV/DLT-SUB-SCFLIST, which is used to add, modify and retrieve selective call feature number lists. This command appears under the heading “Subscriber Management Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-598, for procedures.
- RTRV-PM-VOIPCKT, which is used to retrieve performance-monitoring data for a VoIP termination circuit. This command appears under the heading “VoIP Connections Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guid*e, DLP-633, for procedures.

5.23 Modified Commands

- The Local Access & Transport Area ID has been re-defined to be either 3 or 5 digits long. Previously, it was defined as ranging from 100 – 99999. This change affects the following commands:
  - DLT/ED/ENT/RTRV-DIGITMOD-PARAM (*LataId*)
  - DLT/ ENT/RTRV-DIGIT-SCREEN (*LataId*)
  - DLT/ ENT/RTRV-RATE-CENTER (*LataId*)
  - DLT/ED/ENT/RTRV-ROUTE-DIGITS (*LataId*)
  - DLT/ED/ENT/RTRV-SWITCH-CFG (*ownLataId*)
  - DLT/ ENT/RTRV-TEST-TRANS (*LataId*)
  - DLT/ED/ENT/RTRV-TRKGRP (*LataId*)

- ED/ENT/RTRV-PRFL-SCHED – Changed granularity for *sun*, *mon*, *tue*, *wed*, *thu*, *fri*, and *sat* from whole hours to quarter hours. This command appears under the heading “Routing Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-601, for procedures.
- ENT/ED/DLT/RTRV-LIST-AUTHCODE - Increased *authCode* length from 8 to 16. Added support for using * as an authcode character (cannot be the first digit of the authcode). This command appears under the heading “Routing Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-578, for procedures.
- ENT/ED/RTRV-DIGITMOD-DN – The insertSrc parameter values now include COUNTRY and REG-{1-6}. REG-{1-3} replaced REG1, REG2 and REG3. NextDgtMod was changed to nextDgtModKey. StripDest and nextDgtModKey parameter values also include REG-{1-6}. Support for over-decadic digits (A-F) for

CDPN, CGPN and GAP primKeys was also added. ENT-DIGITMOD-DN entries are limited to 317,000. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-658, for procedures.

- ENT/ED/RTRV-DIGITMOD-GEN – The insertSrc parameter values now include COUNTRY and REG-{1-6}. REG-{1-3} replaced REG1, REG2 and REG3. NextDgtMod was changed to nextDgtModKey. StripDest and nextDgtModKey parameter values also include REG-{1-6}. Support for over-decadic digits (A-F) for CDPN, CGPN and GAP primKeys was added. ENT-DIGITMOD-GEN entries are limited to 1000. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-658, for procedures.

- ENT/ED/RTRV-DIGITMOD-PARAM – NextDgtMod was changed to nextDgtModKey. ENT-DIGITMOD-PARAM entries are limited to 1000. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-658, for procedures.

- ENT/ED/RTRV-DIGITMOD-TNS – NextDgtMod was changed to nextDgtModKey. ENT-DIGITMOD-TNS entries are limited to 2500. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-658, for procedures.

- ENT/ED/RTRV-ROUTE-DIGITS – Support for over-decadic digits (A-F) for CDPN, CGPN and GAP primKeys and a new primKey, ABGN (Anchor Bearer Group Number), was added. ENT-ROUTE-DIGITS entries are limited to 250,000. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-573, for procedures.

- ENT/ED/DLT/RTRV-DIGIT-SCREEN – A New primKey, ABGN, was added. ENT-DIGIT-SCREEN entries are limited to 250,000. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-660, for procedures.

- ENT/ED/DLT/RTRV-PRFL-PARAMDEFAULTS - Support for over-decadic digits (A-F) for CDPN, CGPN and GAP primKeys was added. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-662, for procedures.

- ENT/ED/RTRV- TRANS-PLAN – A new parameter, ABGN (Anchor Bearer Group Number), was added. ABGN is a string in the format of <intfcType>-<idx>, where intfcType values are ISDNIF, CASIF, TGN, MGCP and GR303 and idx is number of up to five digits. ENT-TRANS-PLAN entries are limited to 100. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-659, for procedures.
- ENT/RTRV-RATE-CENTER – The RATECENTER name was extended to 31 characters. ENT-RATE-CENTER entries are limited to 12,500. This command appears under the heading "Numbering Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-579, for procedures.
- ENT/ED/RTRV-PRFL-PARAMSUPPRESS – The GDP parameter was removed and GAP and GENDGTS parameters were added. This command appears under the heading "Routing Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-662, for procedures.
- RTRV-MSG-USER – The AID of uid is now optional. This command appears under the heading "System Commands, Administrative" in the *TL1 Commands Reference Guide*.
- RTRV-LOG – The LOGNM parameter, PATCH, was removed. This command appears under the heading "Log Commands" in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide,* DLP-555, for procedures.
- ENT/ED/RTRV-SS7SYS – The DPCFWD parameter was added. This command appears under the heading "Global Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-530, for procedures.
- ED/RTRV-SWITCH-CFG – The IGNORESAMELATALNPRSTRN, and Custom Variables parameters were added. This command appears under the heading "Global Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-582, for procedures.

- SET/RTRV-DATASERVER-HOSTCFG – The following connection timers were added: INITRETRYTMRANET1, FAILRETRYTMRANET1, INITRETRYTMRBNET1, FAILRETRYTMRBNET1, INITRETRYTMRANET2, FAILRETRYTMRANET2, INITRETRYTMRBNET2, and FAILRETRYTMRBNET2. This command appears under the heading "Global Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-563, for procedures.
- ED/RTRV-SERVICE-ACCESSCODE – GSMC_IS41, GSMC_MAP, RLT, SCA, SCF and SCR services were added. This command appears under the heading "Equal Access to Carriers Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-591, for procedures.
- ENT/DLT/RTRV-HOME-NPA – The DGTSDIAL parameter was removed and added to CALLING-AREA commands. This command appears under the heading "Numbering Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-574, for procedures.
- ENT/ED/DLT/RTRV-CALLING-AREA – The DGTSDIAL parameter was added and RATE-CENTER range was extended to 31 characters. ENT-CALLING-AREA entries are limited to 1024. This command appears under the heading "Numbering Configuration Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-579, for procedures.
- ENT/ED/DLT/RTRV-PRFL-SDP – G726ENC, FAXPROTOCOL and G723BITRATE parameters were added and the CODEC1 parameter is now mandatory for the ENT-PRFL-SDP command. ENT-PRFL-SDP entries are now limited to 127. This command appears under the heading "Inter-Machine Trunks Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-656, for procedures.
- ENT/ED/DLT/RTRV-TRKGRP – Custom Variable parameters, XFERIN, XFEROUT, TXPAD, RXPAD, SIPHRTBTTMR, SIPPLUS, ECHOCTRLNLP, and ECHOCTRLTYPE parameters and SIPT as a value for TYPE were added. Added NONE to ECHOCTRLTYPE enum list. CIRCULAR_DESCEND was removed as a hunting option. MAXIPCALLS range was extended to include 0 for SIP and SIPT to allow for the disabling of the trunk group. ENT-TRKGRP entries are limited to 9,999. This command appears under the heading "Inter-Machine Trunks Commands" in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-528, for procedures.

- ENT/DLT/RTRV-SCCP-SSN – The range of SSN was changed from 11-255 to 1-255. ENT-SCCP-SSN entries are limited to 255. This command appears under the heading “Intelligent Network Functions Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-590, for procedures.
- ENT/ED/DLT/RTRV-SLHR-SCP – The range of SSN was changed from 11-255 to 1-255. It also allows for non-zero svcProvidingSsn values when ROUTETYPE=RTE_GT. ENT-SLHR-SCP entries are limited to 32. This command appears under the heading “Intelligent Network Functions Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-591, for procedures.
- ENT/ED/DLT/RTRV-LIST-AINTRIGGER - MOBILE_TERMINATION, MOBILE_TERMINATION_ANSI and MOBILE_TERMINATION_GSM were added to values for the TRIGTYPE parameter and the TRIGESCLIST parameter was added. ENT-LIST-AINTRIGGER entries are limited to 1,000. This command appears under the heading “Intelligent Network Functions Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-619, for procedures.
- ENT/ED/DLT/RTRV-SUB-IF – New termSvc parameter with values of Y and N was added. It indicates whether to apply the DNs terminating services to this subscriber. The default is N. Increased *acntCodeLen* max to 16. ENT-SUB-IF entries are limited to 80,000. This command appears under the heading “Subscriber Management Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-598, for procedures.
- ENT/ED/DLT/RTRV-SUBSCRIBER – SCA, SCR, SCF and related parameters were added for selective calling features. CFBLAC, CFBLECC, CFDAAC, CFDAECC parameters were added to enable programmable codes for FCB, CFDA services. CFTOLLFREE was added to specify whether to enable a tollfree number as a forwarding number. CFBCC, CFDCC and CFVCC parameters were added to allow whether activation and deactivation is controlled by service provider, the subscriber or both. Any VMSTYPE can now be configured when CFBL or CFNA are enabled. Previously, you had to use the correct VMSTYPE enum to match whichever of these services were provided. Now there is an understanding that call forwarding will take precedence should both services be enabled. ENT-SUBSCRIBER entries are limited to 80,000. This command appears under the heading “Subscriber Management Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-578, for procedures.

- ED/RTRV-VMS-SYS – The SLHRID parameter was added. This command appears under the heading “Voice Mail System Provisioning Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-646, for procedures.
- TRC-CALL-<EQPT, TRK, LNK> – These commands were modified to add call context and change return parameter names. PROTHDLR was changed to protoHdlr, TRKIF to OE, TRKNMBR to TRKMBR, NUMBCHL to BCHL, and INTFC to IF. DPC and OEDCHL were removed and LNG was added. TRC-CALL entries are limited to 20. This command appears under the heading “Test and Diagnosis Commands” in the *TL1 Commands Reference Guide*.
- ENT/ED/DLT/RTRV-ISDN-IF – Custom Variables and CHRGNCND were added. The ISDNVARIANT parameter has a new value of 4ESS. ENT-ISDN-IF entries are limited to 2,688. This command appears under the heading “ISDN-Primary Rate Interface Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-546, for procedures.
- ENT/ED/DLT/RTRV-PRFL-CAS – EO_OPS_TRUNK was added to values for CALL MODEL parameter. TDGTFIRST, TDGTLONG, TDGTSHORT timer parameters were added. ENT-PRFL-CAS entries are limited to 128. This command appears under the heading “Channel-Associated Signaling Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-602, for procedures.
- ENT/ED/DLT/RTRV-CAS-IF – Custom variable parameters were added. ENT-CAS-IF entries are limited to 30,000. This command appears under the heading “Channel-Associated Signaling Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-603, for procedures.
- ED/RTRV-SIP-SYS – The names of T1TMR and T2TMR were changed to T1 and T2. The ANNCURL, REMOTEANNCREQ, SENDPRACK, SESTMR and INVITERETRY parameters were added. This command appears under the heading “Session Initiation Protocol Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-628, for procedures.
- RTRV-PM-SIPMSG – The REFER_SENT, REFER_RCV, NOTIFY_SENT, NOTIFY_RCV, SUBSCRIBE_SENT, SUBSCRIBE_RCV, INFO_SENT, INFO_RCV, ANNC_RCV, and ANNC_SENT values to MONTYP parameter. This command appears under the heading “Session Initiation Protocol Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-644, for procedures.

- INIT-REG-SIPMSG – The REFER_SENT, REFER_RCV, NOTIFY_SENT, NOTIFY_RCV, SUBSCRIBE_SENT, SUBSCRIBE_RCV, INFO_SENT, INFO_RCV, ANNC_RCV, and ANNC_SENT values to MONTYP parameter. This command appears under the heading “Session Initiation Protocol Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-644, for procedures.
- ENT/ED/DLT/RTRV-MGCP-GW – The DYNAMIC value was added to NAMETYPE and the ENDPTIDENT parameter was added. ENT-MGCP-GW entries are limited to 4,000. This command appears under the heading “Media Gateway Controller Protocol Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-655, for procedures.
- ENT/DLT/ED/RTRV-MGCP-LINE – The value of ENDPTNAME parameter was extended to 32 characters with support for these characters: @, [, and]. ENT-MGCP-LINE entries are limited to 64,000. This allows support for MAC addresses in the end point name. This command appears under the heading “Media Gateway Controller Protocol Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-655, for procedures.
- ENT/ED/DLT/RTRV-PRFL-FAILCND – Support for SIPT protocol was added. ENT-PRFL-FAILCND entries are limited to 10.
- TEST-TRANS – A new parameter, ABGN (Anchor Bearer Group Number), was added. ABGN is a string in the format of <intfcType>-<idx>, where intfcType values are ISDNIF, CASIF, TGN, MGCP and GR-303 and idx is a number of up to five digits. This command appears under the heading “Routing Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-672, for procedures.
- ED/RTRV-CHASSIS-EQPT – A new parameter, dnsName, was added. This is a string of up to 30 characters identifying the domain name of the Plexus. This command appears under the heading “Equipment Commands” in the *TL1 Commands Reference Guide*. Refer to the *Maintenance and Troubleshooting Guide*, DLP-644, for procedures.
- ENT/DLT/RTRV-VMS-CKTID – VMS-CKTID entries can now be provisioned on CAS-TRKs as well as on CAS-LINEs. ENT-VMS-CKTID entries are limited to 5,000. This command appears under the heading “Voice Mail System Provisioning Commands” in the *TL1 Commands Reference Guide*. Refer to the *Installation and Operation Manual*, DLP-646, for procedures.

- RTRV-PM-RC – monTypes was modified to match failure conditions. This command appears under the heading “Test and Diagnosis Commands” in the TL1 Commands Reference Guide.

5.24 Other Command Limitations

- ENT-AUDIO-MSG entries are limited to 16,000.
- ENT-ANNC-MAP entries are limited to 25.
- ENT-AUDIO-ANNC entries are limited to 1,000.
- ENT-LIST-AUTHCODE (2000 lists) entries are limited to 60,000.
- ENT-BICC-TRK entries are limited to 21,500.
- ENT-PRFL-TGBILLING entries are limited to 9,999.
- ENT-CARRIER entries are limited to 10,000.
- ENT-CAS-TRK entries are limited to 30,000.
- ENT-CCC-LIST entries are limited to 386.
- ENT-CDC entries are limited to 5.
- ENT-COUNTRY entries are limited to 580.
- ENT-DIALPLAN entries are limited to 100.
- ENT-DIALPLAN-TRIGGER entries are limited to 100.
- ENT-PC (type=own) entries are limited to 1.
- ENT-PC (type=adj/rem) entries are limited to 255.
- ENT-PC (dpc, BICC) M3UAAS entries are limited to 12.
- ENT-GR303-PORT entries are limited to 8,064.
- ENT-GR303-CRV entries are limited to 80,000.
- ENT-GR303-IF entries are limited to 56.
- ENT-HOME-NPA entries are limited to 25.
- ENT-HOME-NXX entries are limited to 12,500.
- ENT-ISDN-LNK entries are limited to 2,688.
- ENT-LAES-CASE entries are limited to 386.
- ENT-LCC entries are limited to 4,096.
- ENT-NP-POOLED/PORTEDOUT/RESERVED entries are limited to 1,000.
- ENT-NP-POOLED/PORTEDOUT/RESERVED - you can establish no more than 31 blocks of numbers.

- ENT-LNPSCREEN-DIGITS entries are limited to 125,000.
- ENT-MGCP-LNGRP entries are limited to 64,000.
- ENT-CRS-T0 entries are limited to 2,016.
- ENT-NATIONAL-NPA entries are limited to 500.
- ENT-LRN entries are limited to 12,500.
- ENT-SIP-IPADDR entries are limited to 32.
- ENT-PRFL-SCHED entries are limited to 255.
- ENT-SCREEN-LIST entries are limited to 100,000.
- ENT-VMS-LNK entries are limited to 16.
- ENT-SS7-TRK entries are limited to 80,640.
- ENT-PRFL-CIC entries are limited to 128.
- ENT-SUB-ALTDN entries are limited to 10,000.
- ENT-SUB-SPEED entries are limited to 40,000.
- ENT-LIST-SUBSCRNBLK entries are limited to 5,000.
- ENT-ROUTE-SCHED entries are limited to 1,000.
- ENT-TOLLFREE-NPA entries are limited to 8.
- ENT-LIST-AINTRIGGER entries are limited to 1000.
- ENT-LIST-TRIGESC (escape list) entries are limited to 100.
- ENT-DIGITMOD-MAP entries are limited to 10,000.
- ENT-DIGIT-SCREEN entries are limited to 250,000.
- ENT-PRFL-FRAUDTRAP entries are limited to 10.
- ENT-PRFL-SCHED entries are limited to 255.
- ENT-ROUTE entries are limited to 80,000.
- ENT-ROUTE-LIST entries are limited to 80,000. When a route list has ALLOC equal to DISTRIBUTED or RANDOM, it can have no more than 25 routes assigned to it. When a route has ALLOC equal to LEASTCOST, it cannot have a route that points to a route list nor have another route list point to it.
- ENT-ROUTE-SCHED entries are limited to 1,000.
- ENT-TRANS-SCHED entries are limited to 100.
- ENT-TREATMENT entries are limited to 1,000.
- ENT-SUBSCRIBER-CL4 entries are limited to 500,000.

5.25 Removed Commands

The following commands were removed from version 3.10.1.3.

- EXEC-RUN-PATCH
- ED/DLT/RTRV-GR303-CHAN
- ED-NPA-HOME

6. Corrected Defects

Tracking Number	Description
21808	If the Primary IP Address has failed over to the Secondary IP Address and then you fail over the SP, your first SIP call after the SP failover will initially attempt the call on the wrong IP Address. The call will complete on the Secondary IP Address once the SP determines the Primary is down. The Secondary IP Address will then be made "active" and all subsequent calls will be made using the "active" IP Address. This issue has been resolved.
24658	The Gig-ENET module, when provisioned as FULLDUPLEX or HALFDUPLEX, bounces in and out of service if the fiber is dark. Therefore, the Gig-ENET module should only be provisioned as AUTONEG. This issue has been resolved.
24724	When building a CAS profile and selecting Calling Model of Subscriber Line (in order to provide dial tone) and signaling model of E_M, the Glare Resolution field is not accessible. Glare Resolution is now editable when the signaling model is E_M.
25860	Need to be able to use digits A-F. CDPN, CGPN, GAP, REG-x can have A-F in the digit-string. This applies to the DIGITMOD-DN, DIGITMOD-GEN, DIGIT-SCREEN and ROUTE-DIGITS commands. PRFL-PARAMDEFAULTS also accepts these characters for CDPN and CGPN.
26284	The OLI in the IAM is not the same as in the CDR. Now the OLI can be a value of 0-999.
26324	The dialed digits of Module 101 are displayed incorrectly. The display of field 157 has changed so the digits read as one complete number rather than two twelve-digit numbers concatenated together.

Tracking Number	Description
26331	When the Plexus sent a SIP INVITE request and received a 200 OK response with the "To" header missing the tag field, the SP was rebooting. Such a response is now handled correctly.
26412	The Plexus kills the call on the receipt of an unknown 1xx response. The Plexus now accepts any 1xx according to both RFC2543 and RFC3261.
26430	When the Plexus relays a SIP message it does not decrement the Max-Forward value. Instead, it starts it at 70 again. This destroys the loop prevention feature. The Plexus is now decrementing Max-Forward for the INVITE message.
26469	When the Plexus received a SIP INVITE request with an unknown domain or IP address in the Request-URI header, it incorrectly responded with a 500 error response. The Plexus now sends a 404 error response in this case.
26686	When a call comes in on a SIP trunk group and a charge number is assigned in the defaults profile, the outgoing SS7 IAM does not have the charge number. The CDR, however, does show the charge number. The SS7 IAM now does send the charge number.
26762	The Plexus maps a 183 message into an ACM with BCI = subscriber free, when it should be "No Indication". The BCI is now set to No Indication.
27034	Customer needs a new traffic statistics report. This was done for TCA 2.0.0.69 and higher.
27085	When SIP protocol tracing was enabled on the Plexus, ACK messages were being printed to the sipa.out log file with extra characters at the end. ACK messages are now printed correctly.
27182	The Plexus should detect loops in routing and translation table provisioning. A check for a loop between failure condition and treatment table provisioning was missing and has been installed.

Tracking Number	**Description**
27283	If a call is using switch/trunk routing and has an outpulse number with spare NOA, field sixteen is populated incorrectly in billing. If the call is answered, the field is set to "F" instead of "0". If the call is disconnected locally, field sixteen is set to "5" (in case of a 4-digit outpulse number) instead of "0" on an ISUP IMT termination. The overseas indicator (BAF Table 15) was determined by the type of the terminating number. There was a check to see if the terminating number was national in which case the overseas indicator would be set to one of the national values. If the number was not national, the assumption was that it was international. This assumption was incorrect. There are other number types that should result in the overseas indicator being set to 2 (Non-NANP, Non-international). This incorrect assumption was fixed.
27673	If a call to a subscriber number was part of a screen list and the main number of the screen list is forwarded to VMS (using CFV), then when the call to the subscriber is forwarded to VM, it is forwarded with the main numbers' information, not the called party information. For any Call Forwarding Feature if a call comes on a screen list DN, the redirecting party ID and OCN will be filled with screen list DN, not the default DN on the line.
27740	After Init Register on an active (in-service (IS)) LinkSet, the PM stats for that LSET record improperly. After the Init Register, the DURLSETUNAV timer increments even though the LSET is IS. This gives the false impression that the LSET is tracking unavailable time even though it is not. The DURLSETUNAV register should only increment in cases where the LSET is actually out of service, regardless as to if/when the registers are re-initialized. This issue has been resolved.
27747	After the Init Register, the DURRTEUNAV timer increments even though the Routesets are in-service (IS). This gives the false impression that there is a Routeset that is tracking unavailable time even though it is not. The DURRTEUNAV register should only increment in cases where a Routeset associated to a point code is actually out of service, regardless as to if/when the registers are re-initialized. This issue has been resolved.

Tracking Number	Description
27953	The Descending Circular Hunting option was a value enum for the hunting parameter within the `ENT-TRKGRP` command. Descending Circular Hunting is no longer a valid enum for the hunting parameter within the `ENT-TRKGRP` command.
28058	The signaling application failed processing invalid statistics resource data. The signaling application will no longer fail as the result of invalid data.
28360	Calls to a number setup as ALTDN for a subscriber was failing with Cause 26. The incoming trunk group was set with VALIDATEGAP=Y. The reason for the call failure is the validate gap flag initiates a check of the subscriber list for a match. It fails since there is not a subscriber built with the Directory Number (DN). It is only in the ALTDN field under a valid subscriber. The Alternate DN field enables the distinctive ringing feature. The validate GAP will now match against Subscriber and Alternate Directory numbers.
28854	Setting parameters to null in the `ED-PRFL-PARAMDEFAULTS` command causes the data for those parameters to become corrupted instead of set to null. This problem has been resolved.
32171	The SIPA process faults due to lack of text encoding support for echo cancellation and silence suppression.
32191	Incorrect generation of "SDP_ERROR: ecanEnable is not G165 or G168" in signaling logs occurs.
32213	An SP-B to SP-B relay channel 19 communication error occurs due to stack overflow caused by recursive function calls to an incorrect routine.
32410	The Plexus does not send OOB DTMF attributes in the SDP; as a result, DTMF is not working from MGCP->PSTN calls.
24445	A hop counter is not currently implemented for BICC trunk groups.
29021	Hex digits (A-F) need to be permitted anywhere in the digit string for the following: CDPN, CGPN, GAP - secKeyvalue insertDgts (in `DIGITMOD-DN`, `DIGITMOD-GEN`), CDPN-NMBR, CGPN-NMBR in param-defaults primkeyval for REG (`ROUTE-DIGITS` and `DIGITS-SCREEN`, `DIGITMOD-GEN` if REG is supported there).

7. Anomalies

Tracking Number	Description
24441	Recursive linking of ALTPLAN is not supported.
25305	A call is received over an ISDNIF. The terminating number is a 10-digit NANP number prefixed with a '1' and the nature of the address is INTNATNUM. The Plexus has a route defined that strips one digit, changes the nature of address to NATNUM, and routes the call. The CDR incorrectly records with the overseas indicator of 6, a 10-digit international number.
26785	The following is a limitation for TEST-TRANS: InsertSrc=COUNTRY capability available in some DigitMod commands cannot be verified by TEST-TRANS.
29645	When Tone-Relay is enabled, a speaker may hear echo when pressing digits, because the Echo Canceller is not properly canceling those tones.
28335	Over-saturation of input on G.729 and G.726 calls can lead to voice quality degradation during the saturation period. This can occur under extremely high volume voice conditions or with high gain level input signals. It is recommended that interface equipment provide a minimum of -6dB signal attenuation toward the Plexus and normal volume levels be maintained to aid against over-saturation of input signals. This will help minimize this effect. Under normal volume and signal levels, this problem does not occur.
28386	For G.729 calls with packetization of 30ms and 40ms and silence suppression enabled, voice quality may experience transient increases in volume and transitional artifacts during silence/noise transitions. It is recommended that the default G.729 profile packetization be provisioned as 10ms or 20ms if silence suppression is required. If higher packetization is required it is recommended that silence suppression be disabled.
29657	Packetization time is not updated when the Plexus is the originating side of a call. The Plexus will use the locally provisioned value even if the SDP from the remote end specifies a different value.

Tracking Number	Description
29656	For 3.9.X.X and earlier releases, the Plexus would automatically revert to G.711 once it detected fax tone. Now, if a user provisions the SDP profile for PASSTHROUGH or FALLBACK, a PCM CODEC must be provisioned in the SDP profile as one of the CODECs.
30225	If a customer has a Voice Mail System (VMS) connected to the Plexus via a CAS interface, digit modification is not allowed and various scenarios will fail.

8. Hardware Installation and Provisioning Considerations

- A Mid Plane II (85-3004) chassis or Midplane III (85-3007 and 85-3008) chassis requires an Octal rear-protection IOM on DS3 IOMs regardless of whether the front IOMs are Octal or Triple DS3 IOMs (rear-working IOMs can be Triples, but the protection IOM must always be Octal).
- An Octal rev. A can be backed up with either an Octal rev. A or an Octal rev. B; however, an Octal rev. B can only be backed up with an Octal rev. B. IOM failover won’t work if Octal rev. B tries to fail over to an Octal rev. A.
- A Switch Fabric (SF) module must be inserted before its associated SP module.
- SP and SF Rev. B or later are required for operation with Octal DS3 IOMs.
- Chassis Part Number 85-3000, CLEI Code BAM9LJ0GRA, does not support Octal IOMs.
- Modules with the following CLEI codes are not supported in this software version:
    - Quad OC12c Network Adapter Interface Module, CLEI code BAA91Z0GAA
    - Quad OC3/OC12c Network Adapter Interface Module, CLEI code BAA91Z0GAB
    - Quad OC12c Packet Interface Module, CLEI code BA1AAA0AAA
    - E1 IO Front Module, CLEI code BA2AV30GAA

- GR-303 is not supported on the DS1 module or the Triple DS3, part number 89-0365. GR-303 is supported on the Triple DS3, part number 89-0397 and higher, and the Octal DS3, part number 89-0398 and higher.
- CAS is supported on Octal, part number 89-0398 and higher and triple DS3, part number 89-0397 and higher.
- When using the `ENT-EQPT` command, redundancy can be set equal to SEC (redundancy=sec) only if the IOM AID specified is in a protection slot. Attempting to provision an unsupported slot as a SEC redundancy returns a DENY message.
- ENA IOMs can only be provisioned in slot 8 and will only fail over to slot 10. VPS can be provisioned in any slot, but will only fail over to slot 17.
- The ENA IOM does not support complete ARP functionality. It relies on an IP Router to route the IP packets, even when the source and destination are on the same subnet (i.e., packets could be switched by a layer 2 device). Currently, the ENA will ONLY ARP the IP address used as the default gateway. It is recommended that a router be used as your default gateway.
- You cannot provision slot 11 (IOM slot 9) if the ENA port 4 is provisioned, nor can you provision the ENA port 4 if slot 11 (IOM slot 9) contains a provisioned card. This is because there is a bandwidth limitation on the SF card.
- Some limitations exist for three-way calling (TWC) and Add-On Transfer Call (AOTC) on the Octal DS3/STS-1 Front Module (part number 89-0398-A, CLEI code BAA9UVYGAA) and the Triple DS3/STS-1 Front Module with Tone Detect (part number 89-0410-A, CLEI code BA4A60ZFAA).
- If an SP module is manually removed and then reinserted, it will not automatically be restored to service. You must enter the TL1 command, `RST-EQPT::SP-{A|B}`, to initialize and synchronize the previously removed SP.

9. Software Installation and Provisioning Considerations

9.1 Equipment Management Issues

- There is no simulation of action = sub, which can be called out from within *InfoAnalyzedAction*, in a translation plan. This means that "substatus" as a key is not available to Screening.
- There is limited support for the CallType key in routing, screening and digit modification, unless filled by param-defaults associated with a trunk group or provided with this command. The real call type

associated with the simulated call based on number analysis of the calling and called party number is not available to this feature.

- Treatments related to announcements capture the announcement ID. If the announcement takes multiple arguments, those may not be captured.
- Some deletes are occasionally allowed even though dependencies may exist.
- The IP addresses must be correct and unique on both SPs before bringing the switches into service.
- IP addresses for OS, signaling and craft Ethernet ports should not be on the same subnet. If it is desired that they be on the same subnet, contact Technical Services at 1-888-440-8354 for provisioning assistance.
- If the signaling interfaces are not configured with IP addresses, an alarm from each SP, stating “lost link on signaling Ethernet,” will be generated.
- When switching from SS7 to ISDN signaling or vice-versa, the signaling link and interface must be deleted and the Octal DS3/DS1/DS3 IOMs must be rebooted before provisioning can occur.
- If you pull an SP, thus taking it out of service, you must wait 10 seconds before reinserting the SP.
- Once you execute three CPY-MEM commands, the next CPY-MEM command cannot take place until one of the previously executed commands completes. Error response is indicated as “All resources busy”.
- If CPY-MEM fails, the flash update process becomes corrupt. In this event, contact Technical Services at 1-888-440-8354.
- Using the ED-DAT TL1 command to change the time is not required or recommended when the NTP is provisioned. If the new date differs by more than 1000 seconds, then the NTP daemon may shut down. If this happens, you should reset the NTP server to 0.0.0.0 then back to the correct server IP address.
- A single IOM cannot support signaling links of both 56K and 64K. The link speed must be the same for all links on a single IOM. This restriction does not apply to Triple DS3 IOMs, part numbers 89-0397 and higher, and Octal DS3 IOMs, part numbers 89-0398 and higher, as a mix in the link speed creates no problems.
- Currently, Authorization Codes are supported on CAS and ISDN lines only.
- When bulk provisioning T1 ports, it is necessary to wait anywhere from 20 seconds to 3 or 4 minutes before continuing to provision, depending on how many blocks of T1s you are provisioning. For instance, if provisioning T1 blocks on a fully loaded Octal DS3, you

should probably wait 3 or 4 minutes before continuing with provisioning. The fewer blocks of T1 ports you bulk provision, the less time you have to wait before continuing with provisioning. This also prevents the receipt of reply timeouts when adding PCs.

- Line timing must be configured as a protected pair. I/O 1 slots and 2 are paired, as are I/O slots 8 and 10. You cannot have just one IOM in a paired slot. Any Triple or Octal DS3 IOM can be used. **Note:** Line timing is not supported with Midplane I (85-3000 chassis) or SP1.
- CALEA CDC is not supported when the SP is running in dual processor mode. The default mode is dual processor mode enabled. If CALEA CDC is required, please contact Technical Services at 1-888-440-8354.

9.1.1 DS1 IOMs

- Far-end (FEND) loopbacks are only supported in Extended Superframe (ESF) mode.

9.1.2 DS3 IOMs

- FEND loopbacks for DS1 ports of a DS3 IOM are only supported in ESF mode.
- The `ED/ENT-T3` command does not support LINECDE=CCHAN; it only supports B3ZS.
- Operating a FEND loopback on a channel already in a near-end (NEND) loopback cannot be done because of DS3 interface chip limitations. If a NEND loopback command was followed by a FEND loopback command, the DS3 interface chip will only execute the NEND loopback, but will remember the FEND request. If the NEND loopback is released, the FEND request is remembered but will not execute. Issuing a FEND loopback will not work because the DS3 interface chip thinks a loopback is in progress. The FEND must also be released. To insure a T3 is put into a FEND loopback, first send a `RLS-LPBK-T3::<lpbk_id>:::FEND` command followed by a `OPR-LPBK-T3::<lpbk_id>:::FEND` command.
- When using the `RTRV-PM-T3` command, do not set the value to “0-UP” when the value of *montype* is “ALL”, as the large amount of data returned could overload the output buffer of the client application and cause the TL1 session to freeze.

9.1.3 STS-1 IOMs

- GR-253 (R6-372) states that there must be a method provided to detect and report the actual contents of the Received STS Path Trace message. The `RTRV-STS1` command presently only supports the expected Rx Trace message and the Tx Trace message.

- The system reports an STS Trace ID Mismatch when the J1 byte is inaccessible. GR-253 (R6-382) states that STS Path Trace monitoring should be suspended if the J1 byte in the Path Overhead cannot be accessed (e.g., LOS, LOF, LOP-P and AIS-P). This means that the system should not report an STS Trace ID Mismatch just before it declares any of the above mentioned alarm conditions, or after they clear.
- STS reports Trace ID Mismatch events.

9.1.4 Voice Server Modules

When the precedence of RTP packets is changed with `ED-VOIP-SYS`, the voice server module needs to be rebooted for the settings to take effect. This can be done with an IOM failover to the protection voice server module followed by an IOM revert of the voice server module.

9.2 Primary Rate ISDN Line

- Two-way Primary Rate ISDN lines on the Plexus 9000 should be provisioned as National ISDN-2 or 4ESS.
- The Plexus 9000 supports National ISDN-2, 4ESS, 5ESS, and DMS-100 variants for one-way calls leaving the Plexus.

9.3 SS7 and ISDN Signaling

- When editing a T1 out-of-service the mode (OMODE) must be set to AIS in order to bring MTP2 and LAPD down.
- Currently, the signaling point code restart procedure is not supported.
- Alarms for ISUP timer expiry are disabled.
- Alarms are not generated if the initial condition for a remote Point Code is down.
- A maximum of 250 destination point codes total can be configured.
- A signaling link set must contain at least one signaling link with a link priority set to 0. Any additional links, in the link set, must have contiguous priorities.
- Plexus 9000 currently does not support T-321 timers.
- Trunk Group IDs must be unique in the system.
- There can be no more than 100,000 interfaces formed by TRKGRPS+ ISDNIF+CASIF+GR303 in the router. The breakdown of each consists of the following: a maximum of 3808 configurable ISDN interfaces and TRKGRPS each; a maximum of 56 GR303-IF interfaces; and a maximum of 91392 CAS-IF interfaces (currently, however, it is recommended that you not configure more than 64,000 CAS-IFs).

9.4 CAS Signaling

- The maximum CAS ports (T1s) that can be provisioned for one Octal (89-0398, 89-0411, 89-0425) is 150 T1 ports (3600 T0s).
- Parameters in the CAS-IF command cannot be edited after changing the ALLOC parameter equal to CIRCULAR for one CAS-IF entry.

9.5 SIP

- Enabling the SIP session timer can cause some performance degradation at high call rates if a large number of session refresh messages are sent at the same time or after a failover. The recommended value for this timer when enabled is 1800 (30 minutes) -- that way, session refresh messages will not occur in most cases.
- When a 200 OK message is sent and no ACK message is received, the BYE message is not transmitted immediately after the seventh expiry of the T1 timer.

9.6 Intelligent Networks

- LNP ported numbers administration is currently not supported.
- LNP Peg Counts are currently not supported.
- LNP test calls are currently not supported.
- Notification and/or termination messages from the SCP are currently not supported.
- Queries are not sent to the backup SCP. This will not effect routing if Global Title is used.

9.7 Call Processing

- The SS7 signaling links are not protected during an IOM hardware failover.
- Retrieval of Calling Area entries requires knowing the Calling Area ID entered.
- When an IOM is in protection mode, attempting to add, modify or delete ISDN or signaling links from the protected IOM will return DENY messages.
- Calling Areas cannot be removed if they are associated with a subscriber.
- All Local Calling Areas must contain 555 to generate call type 33. Specifically, 555 must be a HOME-NXX on your switch, and 555 must also be in your Local Calling Area. If either of these conditions is not satisfied, then the call is treated as an InterLATA call and call type 110 is generated. The most efficient way to provision 555 into all Local Calling Areas is to designate one HOME-NPA-NXX as 555,

add that HOME-NPA-555 to one Rate Center, and then assign that Rate Center to all Local Calling Areas.

- When a caller with Call Waiting is talking to a second party, and the caller with Call Waiting receives a call from a third party, the caller with Call Waiting can flash over to the third party, placing the second party on hold. Upon disconnecting the call with the third party, the caller with Call Waiting should either be able to wait for normal timer expiry before being reconnected with the second party, or should manually flash over to the second party to continue the call. Currently, the Plexus does not wait for timer expiry or a manual command from the caller, but rather, automatically cuts back to the second party upon disconnect from the third party.
- G.723 PTIME provisioning is not supported. The Plexus defaults to 60ms, unless the endpoint requests 30ms.

9.8 Billing/Statistics

- By default, when Excel imports the ASCII billing “time” fields, it picks a display format that excludes the hour information. The data remains in the column; however, you have to alter the display format to see the information.
- The number of files that can reside in the ASCII Billing server’s data directory is finite and is based on your particular Solaris configuration. Because it is possible to fill the directory to a point where the ‘ls’ command will not display the contents of the data directory, it is suggested that you first remove the previous month’s data files from the data directory during the first week of the current month to prevent this from happening. If you do fill the data directory, use the ‘find .’ command, executed from the data directory, to display the contents of the directory. Move the files out of the data directory using the ‘cp ‘`find ./SOME_PATHNAME*’ SOMEOTHER_PLACE`’, executed from the data directory, until the ‘ls’ command works.
- The ISDN PRI Traffic CCS report does not currently update the available circuit (AVLCIRC) field to reflect ISDN channels that are OOS.
- Because Feature Group D functionality is fully supported for CAS and ISUP in release 3.8 and above, originating carrier information (such as connect date, connect time, and elapsed time) is only valid for Feature Group D trunks. For terminating carrier access calls, the carrier timing information is populated even if the incoming trunk group is not Feature Group D.
- If the physical connection is lost between the Plexus and the BTS server, the Plexus does not generate an alarm due to a limitation of the Lynx TCP protocol stack. However, an alarm will be generated when

the active SP determines that it cannot transmit records to the BTS server.

- The Traffic Statistics Reports chapter of the *Billing and Traffic Guide* states that busy hour reports are supported for SIP, CAS, BICC and SS7 trunk groups. SIP busy hours reports are not supported. This is corrected in Issue 7 of the guide.
- The Traffic Statistics Reports chapter of the *Billing and Traffic Guide* states that Incoming and Outgoing CICs have allowed value types of CAS interface and CAS line. This is incorrect, and is corrected in Issue 7 of the guide. Selections should be:

  0 = (not used) for ISDN, CAS Interface, CAS Trunk Group or SIP Trunk Group

  CIC for SS7 Trunk Group

  CIC for BICC Trunk Group
  CRV for GR-303 interface

9.9 Integrated VoIP

- The adaptive jitter buffer algorithm has temporarily been disabled to provide additional feature support.

  Note: The TL1 `EDIT-VOIP-SYS` command still enables you to set the *jitBufPlMod* parameter to "ADAPTIVE" but the fixed jitter buffer algorithm will be used.
- Echo tail of 64ms is not supported in this version. Supported echo tails are 32ms and 128ms.

  Note: If 64ms is provisioned, using TL1 or the PlexView EMS, then an echo tail of 128ms will be in effect.
- Tone Relay support is limited to DTMF Events per RFC2833, Section 3.10, Table 1. Other types of tones are not supported.
- BICC trunking supports a maximum of 16383 BICC trunks per Destination Point Code.
- The system does not currently verify the IP link connectivity state on IOM-8 prior to reverting. You should verify the status of the IP links on IOM-8 prior to reverting.
- If you are unable to change BICC trunk group configuration settings, you must delete the BICC trunks and the BICC trunk group and then provision the BICC trunk group and trunks with the required modifications.
- Fax/modem support between the Plexus and some third-party equipment (such as gateways, IADs) is only available on G.711 calls. For calls using compressed CODECs (using G.729 protocol), fax/modem calls won't work with these vendors if they use proprietary signaling protocols.

- Currently, the Plexus 9000 does support SIP-to-SIP calls, however, no SIP subscriber features are supported in this release. You must use an external feature server to provide features to SIP subscribers.
- The Plexus 9000 supports G.711, G.726, G.729, G.723.1, CLEARMODE, X-CCD, CISCO-CLEAR-CHANNEL CODECs, each with 10, 20, 30 and 40ms sampling, except for G.723.1, which supports 30 and 60ms sampling
- If you want to modify the endptvoip IP address on a GigE port, you must edit the ENET port Out of Service (OOS) and then use the RMV-EQPT command on the VPS IOM. There is no way to dynamically update the IP addresses without dropping all calls associated with a VPS IOM. For example, information would be entered as such:

  `ED-ENET::IOM-8-ENET-1::::OOS`; (GigE port)
  `RMV-EQPT::IOM-11`; (VPS/VSM IOM)

  `DLT-ENET-ENDPTVOIP::IOM-8-ENET-1`;(Delete Endpoint)

  `ENT-ENET-ENDPTVOIP::IOM-8-ENET-:::IPADDR=10.18.140.211`, `MATEIPADDR=10.18.140.212,SUBNETMASK=255.255.255.0`, `DEFAULTGATEWAY=10.18.140.1:IS`;
  (Enter new IP address to the endpoint)

  `ED-ENET::IOM-8-ENET-1::::IS`;

  `RST-EQPT::IOM-11`;

- When the SDP profile contains multiple deep compression CODECs (G.729, G.723.1 or G.726), only the first available deep compression CODEC in the preference list will be included in the originating offer from the Plexus, e.g., G.729, G.711, G.726. This results in G.711 and G.729 being the offered CODECs in the originating offer (provided G.729 is available; otherwise, G.726 or G.711 would be offered). This limitation applies to CODEC negotiation across BICC, SIP, and MGCP.
- MF Tone detection is not supported on the VPS IOM in this release. In order to support MF Tone detection, 89-0410/0411/0424/0425 IOMs, which have on-board DSPs to perform MF tone detection, must be used.
- The Plexus only supports Payload Type 13 comfort noise for G.711 and G.726.
- RTCP reports sent for T.38 calls are invalid and should be ignored.
- When T.38 is used between the Plexus and any other gateways such as the Cisco AS 5300/5400, the format of the ENA must be provisioned

for ENET_802_3. If provisioned for ENET_DIX_II, T.38 calls will fail to complete with AAL5 CRC errors.

9.10 System Software

- Although the Plexus currently enables the following services to be assigned to subscribers/interfaces with more than one line (DS0), assigning these services to subscribers/interfaces with more than one DS0 is not recommended:
    - Call Forwarding Busy Line
    - Call Waiting
    - Cancel Call Waiting
    - Three-Way Calling
    - Call Forwarding Variable
    - Remote Call Forwarding

  If multiple lines (DS0s) are assigned to an interface and more than one party is off hook, the Plexus has trouble identifying which line gets the call waiting tone or which party pushed the FLASH HOOK.
- UNIX system security does not support password aging or security logging for FTP and remote login access.
- The date/time should be set before configuring the system and adding the IOM. It is advised that GMT time be used. Please contact Technical Services at 1-888-440-8354 for instructions on how to provision the Plexus clock for GMT time.
- If an IOM protection switch occurs, the monitoring function stops. Once an IOM revert occurs, the cross-connect to the test port has to be put out of service (OOS) and restored.
- Currently, there is no way to retrieve the status of a nailed-up DS0 connection unless you know the exact DS0 of one of the connections. `RTRV-CRS-T0` without an AID should return the status of test port settings.
- The `ED-SERVICE-ACCESSCODE` must put the VMS in-service to support voice mail. The command would appear as: `ED-SERVICE-ACCESSCODE::VMS::::IS`;.

10. TL1 Restrictions and Security Errata

10.1 TL1 Restrictions

- Occasionally a “reply timeout” message will be seen while trying to execute an ENT, ED or DLT TL1 command. The command may have still worked and should be validated with a RTRV.

- The number of TL1 retrieve (RTRV) commands is limited to five results per second.
- It is possible that some TL1 commands may timeout before completion, depending on system load. Should this occur, please contact Technical Services at 1-888-440-8354 for assistance.
- Signaling TL1 commands respond “All Resources Busy” while standby SP is synchronizing.
- Logging into the TL1 agent cannot be done during part of SP sync on the standby side.
- When using the `INIT-SYS`, `ED-BILLSYS`, and `EXEC-RESTORE-PLEXUS` commands, the target identifier (TID) must always be used. Therefore, the form of the command must be, as an example, `INIT-SYS:TID::::10; or ED-BILLSYS:TID::::10; or EXEC-RESTORE-PLEXUS:TID::::10`;
- Before restoring a backed up database using the `EXEC-RESTORE-PLEXUS` command, contact Technical Services at 1-888-440-8354 for assistance to ensure a successful database restoration.
- The *T3 Idle* and *T3 Map* parameters in the `ENT/ED/RTRV-T3` command are not supported.
- For the commands `INIT-REG-T1/T3/E1/OC12/OC3/STS1`, NULL is the only accepted value in the *mondat* and *montm* parameters. *ALL* is the only *montype* that will clear the registers.
- The “state” information on the SIP-IPADDR command is not persistent. If the primary IP address has failed over to the secondary IP address, and then you fail over the SP, your first SIP call after the SP failover will initially attempt the call on the wrong IP address. The call will complete on the secondary IP address once the SP determines the primary is down. The secondary IP address will then be made “active” and all subsequent calls will be made using the "active" IP Address.

10.2 Security

An IP packet filtering application that runs on the SPs is available in this build.

Should filtering be required, contact Technical Services at 1-888-440-8354 for assistance.