Tenor P108 Maintenance Release Notes

Tenor VoIP MultiPath Switch/Gateway Products

P108 Maintenance Release

This document lists all software fixes for VoIP MultiPath Switch/Gateway products running maintenance software P108-09-38 and previous.

Products Affected

These maintenance release notes apply to the following products:

- Tenor DX VoIP MultiPath Switch/Gateway
- Tenor AX VoIP MultiPath Switch/Gateway
- Tenor AF VoIP MultiPath Switch/Gateway
- Tenor BX VoIP MultiPath Switch/Gateway

A software maintenance release is created by Ribbon resolve an inconsistency or "bug" that has been identified within a release.

Maintenance releases (for Second Generation Tenors) can be identified by the 3rd set of digits in the release name as PX-Y-Z, where the Z is the maintenance number. All maintenance releases are built on top of the latest GA release of software. For example, if P102-11-00 is the latest GA software, the first maintenance release of software will be P102-11-01. The notes below are cumulative since the last Generally Available (GA) software release P108-09-00. Maintenance software goes through limited testing from Ribbon, but has been verified as resolved by the customer who identified the problem.

If additional maintenance releases are required for a specific GA release, each new maintenance will contain all previous maintenance fixes. For example, maintenance release P106-11-02 will include all maintenance release changes from P106-11-01.

On a scheduled basis, Ribbon releases a full GA version that includes all previous maintenance released that have undergone full system test.

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Resolved Issues/Feature Enhancements

P108-09-38

Tenor-55 Under View Menu changed menu item name "Whole Database" to just "Database"

In the Tenor Configuration Manager, under View, the sub-menu option "Whole Database" has been renamed to "Database"

Tenor-58 After Tenor Factory Reset, Wizard takes too long to launch

After Tenor Factory Reset, the Wizard took too long to launch. This has been resolved.

Tenor-67 Tenor Config Manager: Add Test Call, Test Dial Tone and Test Line Monitor CAS diagnostics commands

Previously, the ability to perform diagnostics on the CAS Lines with Test call, dial tone, etc., was available only from the CLI. These commands are now available from the Tenor Configuration Manager, as follows:

CAS Signaling Phone (FXS Ports)

A new tab "**Diagnostics**" has been added, which includes the following new command:

- **Test Call:** Attempts to ring the phone connected to the specified FXS port.

CAS Signaling Line (FXO Ports)

A new tab "**Diagnostics**" has been added. This tab includes the following new commands**:**

- **Dial Tone Test:** Performs the dial tone test on the specified FXO port.
- **Calibrate Line:** Performs the line calibration test on the specified FXO port for the impedance specified (Analog Specific Tab) or for All Impedances.
- **Test Call:** Performs a test call on the specified FXO port using the phone number specified with the command.
- **Monitor Line**: Starts to Monitor the specified port and after stopping the test, will report the Voltage and Current observed on that port.

Tenor-68 GUI: Added ability to receive event logs from Tenor Configuration Manager

A new button **Event Logging** has been added to the Tool Bar. Event Logging enables logging for Call Handler (ch), Analog/Digital CAS (cas), and SIP protocol modules. For each module, three levels of logging are offered: **Basic**, **Detailed** or **Verbose**. The event logging buffer can be cleared with the **Clear Logs** button. After performing test calls, click on "**Retrieve Logs**" to view the event logs.

Tenor-75 Tenor Configuration Manager Displayed Un-Authorized (under Basic View), when refresh was clicked multiple times

Under the Basic View, when refresh was clicked quickly multiple times, an Un-Authorized error message was displayed. This has been resolved.

Tenor-76 Tone Profile settings not used to disconnect signal

When the Progress Tone Country was configured to 19 (Germany) in the ProgressToneCountry.xml, the Tenor did not honor the configured Tone Profile for disconnect tone, but instead forced a single frequency tone of 425 Hz with 1000ms on and 4000ms off. The Tenor now uses the tone profile settings in the ProgressToneCountry.xml file.

Tenor-77 SIP Configuration - Added prepend PLUS Configuration to CLI and GUI

Previously, for SIP Calls, the Tenor could be configured to send a '+' in front of the Called or Calling number via a setting in the 'var_config.cfg' file. This can now be configured in the SIP Signaling Group.

Tenor-80 Added a configuration for List of Endpoints allowed to communicate with the Tenor

Previously, the **filtertelnetftp** command, which blocked Telnet and FTP access into the Tenor, was available only by specifying the allowable source IP's defined in a var_config fi**le**. This feature is now configurable under the **Filter IP Directory.** A new configuration option **Access Mode** has been added for each Filter Ip. Access Mode has two options: **Call Access** (default, only those endpoints can make VoIP calls with the Tenor) or **Management Access** (only those end points can have FTP access to the Tenor, and manage the Tenor via Telnet and Tenor Configuration Manager).

Tenor-82 Varconfig gDNSCacheRefreshInterval is now configurable item under DNS Server

The DNS Cache Refresh timer feature was previously implemented using varconfig gDNSCacheRefreshInterval. This feature can now be configured under DNS Server configuration via CLI and Tenor Configuration Manager.

Tenor-83 Varconfig DNSLoadBalance is now configurable under DNS Server Config

The varconfig **DNSLoadBalance** (DNS Load Balancing feature) is now configurable under DNS Server configuration via CLI and Tenor Configuration Manager.

Tenor-84 Varconfig SipUserPhone is now configurable under SIP Sig Group Config

The varconfig **SipUserPhone** command (on outgoing Invites, the Tenor turns on/off user=phone in ONLY the RequestURI and To headers) is now configurable under SIP Signaling Group as **user=phone in URIs**, with the following configuration options:

- **No.** No user=phone in any SIP URI (default).
- **All URIs**. User=phone in all SIP URIs.
- **Request URI, To Header.** User=phone in the Request-URI and To: header of an Outgoing Invite.

The varconfig **SipUserPhone** is no longer used.

Tenor-89 CLI command "cmd sip resolve" added into Tenor Configuration Manager

The CLI command **cmd sip resolve** (performs DNS resolution for a specified FQDN) is now available in the Tenor Configuration Manager (via **View > Tenor Status/Info)**.

Tenor-94 Added the "echoTailSize" parameter into the GUI and CLI

The **DSP Echo Tail Size** parameter (for optimizing echo cancellation) is now configurable in the Configuration Manager (via Gateway) as **Echo Tail Size** (in msec). The echoTailSize options include: 128, 64, 32, or 16 msecs.

Tenor-95 Bypass Filter-IP for all Statically configured Servers in the Tenor

When a list of allowed IPs was configured under Filter IP Directory, the Tenor would exclude filtering out IPs that were configured in the Tenor for DNS Servers, CDR Servers, SysLog Servers, Radius Servers, etc, so that the user does not have to explicitly configure them again in the Filter IP list.

Tenor-97 Basic Config – Improve Call Status view layout / info

In the Tenor Configuration Manager, the call statistics information (**via Basic Config> Basic Status> Call Status**) has been improved to be more intuitive.

Tenor-99 Tenor Configuration Manager - required a "Right" click first before a user was allowed to "Left" click to create a group or new entity

When switching the menu with a Right click, you would need to right-click once again before the content menu was displayed. This has been resolved.

Tenor-100 Rename the "Filter-IP" Directory to Allowed Access List

The Filter IP (for white list) has been renamed to **Allowed Access IP** for consistency.

Tenor-101 Filter-IP: Could not ping Tenor via configured broadcast IP's

When an endpoint's IP was configured under the Filter IP (Allowed Access IP, previously Filter IP) with Management Access, the Tenor could not be pinged. This has been resolved.

Tenor-106 Tenor GUI - Remove Menu items for viewing draft database in the Tenor

Under *Configuration Manager View> Database*, (previously *View> Whole Database*), all options for viewing the draft version of the database have been removed.

Tenor-107 Version Info was not shown when displaying working db via CLI or Configuration Manager

The Tenor version information was printed at the end of **sh -l** command but not for the **sh -w-l** command. This has been resolved.

Tenor-109 Ribbon re-branding of Tenor's GUI

The Tenor Configuration Manager has been rebranded for Ribbon.

Tenor-110 Change Field name Destination in Allowed Access IP (TENOR-80, TENOR-100) to Allowed Access IP Allowed IP Address

Under **Allowed Access IP** (formerly Filter IP), the field name **Destination** for the end point IP has been renamed to **Allowed IP Address**, to be more intuitive.

Tenor-113 Tenor did not sent digit to FXO on DNCM match after enabling local routing with routing table

When Use Routing Table and **Enable Local Routing** are enabled, and a DNCM configured for a FXO port, when the call was made from the FXS and the DN matches the FXO port's DNCM, the Tenor did not dial out the DN from the FXS (when the call was placed on the matching

FXO). This has been resolved.

P108-09-35

Tenor-45 New CAS Diagnostics Commands added to Tenor Configuration Manager

In the Tenor Configuration Manager, a new **Diagnostics** tab panel has been added under **Cas Signaling Group Line** with the following four options:

- **Calibrate Line** - Calibrates the line. Click this option to enter the FXO line number in which the calibration will be completed. Two Impedance options are available: **Configured Impedance** (calibrates for the Impedance value set in ***CAS Signaling Group-line >Analog Specific***) and **All Impedances** (calibrates for all possible Imepedance values).

- **Dial Tone Test** - Dial Tone Test is performed on the FXO line. Click this option to enter the FXO line in which the Dial Tone Test will be completed.

- **Test Call** (FXO or FXS) - Manually initiates a test call on the FXO or FXS port. Click this option to enter the desired FXO or FXS line number and a phone number to dial out. The Tenor then attempts an outgoing call on the chosen FXO or FXS line with the specified destination number.

- **Monitor Line** - Monitors the signal levels on the FXO line. Click this option to enter the desired FXO line number. The Tenor then continuously monitors the signal on the FXO port until the user terminates it from the Configuration Manager.

P108-09-34

Tenor-29 Names for several Configuration options have been changed

In Auto Provisioning (available through ***System-Wide Configuration> Auto Provisioning),*** several configuration options in the Configuration Manager have been updated to the following names:

Table 1: Configuration Options - Updated Names

Original Configuration Name	New Configuration Name	Definition
Profile Rule	Provisioning Server URL	URL of the Tenor Configuration File Location
Provision Enable	Provisioning Profile	Specifies how to apply the configuration
Provision Error Policy	Error Provisioning Policy	Specifies how to handle errors (if any) when the configuration is applied
Forced Resync Delay	Resync Reset Delay	Forces delay before resetting the Tenor, in seconds
Upgrade Rule	Software Location (Server URL)	URL of new Tenor Software file location
Upgrade Enable	Enable Software Upgrade	If enabled, upgrades to "Software Version" by TFTP from "Software Location"
Upgrade Error Retry Delay (in sec.)	Error Delay (in sec.)	Forces this amount of delay time before the Tenor re-tries to retrieve the file
Upgrade Release	Software Version	Specifies the version to which the Tenor should upgrade
Allow New Calls in Forced Resync Delay	Allow New Calls in Resync Delay	Enables/disables whether new calls are allowed in the Tenor

Tenor-31 Resync on Reset command added new option

The **Resync on Reset** option (available through ***System-Wide Configuration>Auto Provisioning> General***) has been updated to include a drop down list whch includes the following:

- Disabled
- After Every Reset
- After Next Reset Only

Tenor-33 Option added to Auto Provisioning to manually sync the Tenor on demand

Auto Provisioning has a new option (available through the CLI and Configuration Manager) which enables you to manually sync the Tenor on demand.

For CLI: The command **cmd apsync** has been added.

For Configuration Manager: The **Sync Now** button (available through ***Auto Provisioning> General***) is available only if:

- The Tenor release is P108-09-33 and higher.
- The Provisioning Server URL field is not empty.

Tenor-35 Commands added to generate db.xml without Auto Provisioning items

New commands are added to view the **db.xml** files in draft and permanent forms (for the Configuration Manager and CLI).

For CLI:

- **sh -xml -noap**
- **sh -w xml -noap**

For Configuration Manager:
Commands are available through ***View>Database*****: Draft Version in XML (no AP)** and **Permanent Version in XML (no AP)**.

Tenor-40/Tenor 41 Default Values updated for Auto Provisioning

The following default values (located in ***System-Wide Configuration> Auto Provisioning***) are updated for Auto Provisioning.

Table 2: Auto Provisioning Default Fields

	Original Default Value	New Default Value
Provisioning Profile	Only for setting Auto Provisioning parameters; Ignore all others	Apply retrieved file on top of current DB; Reset Tenor if necessary
Resync on Reset	After Every Reset Only	After Next Reset Only
Resync Periodic	86400	0 **Note:** When this field is set to "0", a periodic resync will not occur.

Tenor-42 For Auto Provisioning, update to Profile/Software Upgrade URL required reboot

For Auto Provisioning previously, if a URL was newly configured for a Profile or Software Upgrade, the Tenor required a reboot for this change to take effect. Now, these changes take effect dynamically and do not require a reboot.

Tenor-43 For Auto Provisioning, Supplementary Rule integrated into Software Upgrade

For a software upgrade, you are now only required to specify a URL of the folder which contains all the Tenor files required for upgrade (tnrsys.bin and all supplementary files). The SupplementaryRule URL field as now been integrated into the **Software Location (Server URL)** fu nction (available ***System-Wide Configuration> Auto Provisioning> Auto Software Upgrade***) and is no longer required as a separate URL.

Tenor-44 Progress Tone for New Zealand has been added

Support for the Progress tone for New Zealand has been added.

Tenor-50 - For Auto Provisioning, configuration has been simplified

To simplify Auto Provisioning configuration, the **ResyncAfterUpgradeAttempt** command has been removed.

Tenor-54 For FXO Line Test, CMD test did not exit process if an error occurred

The **cmd test** command performs line calibration or dial tone tests, etc. on the FXO ports. When this command ran and an error resulted (i.e., no dial tone detected or a DSP could be allocated), the test task continued to run in the background. As a result, when the command was run again, an error indicated that the test process was still running. This has been fixed.

P108-09-32

4930 SNMP - Tenor AXT/AFT did not return active calls or active channels in certain situations

The Tenor AXT/AFT models did not return active calls or active channels when being queried by the SNMP server. This has been resolved.

4931 SIP Session ID and Session ID changed but version in SDP stayed the same

The Tenor changed the Session ID, while leaving the version number unchanged in the SDP between the 180 ringing and 200 OK responses to the INVITE. This was treated as non-compliance by the TSS, and the call was dropped. This has been resolved and the call is now compliant with the RFC for SDP Session ID and Version Number.

4932 SDP changes in ACK messages were not processing for calls going On Hold

Tenor did not process any SDP changes for calls that were going On Hold (if the SDP changes coming from the IP-PBX / SBC were in the SIP ACK message). As a result, calls were not re-sent out properly after receiving a Music on Hold. This has been resolved.

4933 Modem Bypass setting in LCRG/TCRG was unable to be set when "New Routing" was enabled

When "New Routing" was used in the Tenor, the user was unable to enable/set the Modem Bypass setting to 2 (MoIP) in the LCRG. This setting was restricted and could not be accessed via Configuration Manager or Command Line Interface (CLI). This has been resolved.

4934 Tenor was not interpreting absence of "a=" media direction field

When the peer side put the Tenor on Hold and later sent a Re-invite without the "a=" media direction field, the Tenor did not interpret the absence of the direction field in SDP as a sendRecv. As a result, the call had one way audio. This has been resolved.

4935 Factory default setting for G729 codec's payload size was changed

If a new G729 codec was added or an existing G729 code was changed, the payload size setting defaulted to 40 ms.The default has been changed to 30ms.

4936 Time Server IP in DHCP offer causes the Tenor to reboot continuously

During the boot up / DHCP IP assignment process, if an NTP Time Server IP Address was configured, it caused the Tenor gateway to get stuck in a reboot cycle. This has been resolved.

4937 Caller ID detection issue with the PSTN Line

PSTN lines connected to the Tenor's FXO ports were not detecting Caller ID. This has been resolved.

4938 New configuration item in SIP Signaling Group

To enable a Tenor to negotiate for a ReInvite that puts the Tenor on hold, a new configuration option **Negotiate Media in Hold Reinvite** (check to negotiate media; uncheck to disable negotiating media) has been added to the Configuration Manager under SIP Signaling Group> Advanced Tab. This configuration field was previously included in the varconfig as sipNegotiateMediaInHoldReInviteas (see 4925).

Click the field to enable/disable: **Check** (Enables a Tenor to negotiate negotiate media when it receives a ReInvite that put the Tenor on hold) or **Uncheck** (Disables Tenor from negotiating media when it receives a ReInvite that put the Tenor on hold).

4939 EnableLocalRouting configuration item displayed when New Routing was not used

Configuration parameter **EnableLocalRouting** was displayed even when Routing Table was not used. The EnableLocalRouting field must be displayed only when Routing Table is used. This has been resolved.

4940 New configuration item in Gateway

To support Direct SIP Enhanced Gateway in support of Microsoft® Lync™ Server 2010 deployments, a configuration option **Lync Interworking** (check to enable Tenor to drop short packets, such as STUN and comfort noise; uncheck to disable Tenor from dropping short packets) has been added to the Configuration Manager under Gateway. This configuration item was previously included in the varconfig as lyncInterOp (see 4874). In order to use a Tenor in a Lync Environment, the field must be checked.

Click the field to enable/disable: **Check** (Enables a Tenor to drop short packets, such as STUN and comfort noise) or **Uncheck** (Disables Tenor from dropping short packets, such as STUN and comfort noise).

4942 SIP Trace: Inconcistent output format of SIP messages

In SIP Trace (sproto), some SIP messages (i.e., Invite, 183, 180) were formatted by indenting to the right while some other messages (i.e., Trying, Cancel) did not have the same formatting and indentation.This has been resolved.

4943 Evlogsys now configurable in Configuration Manager and CLI

The varconfig **evsyslog** command is now configurable in the Configuration Manager and CLI; this command is used to control sending evlog output to the Syslog Server. Configure the command as follows:

- **In Configuration Manager:**
  - The **Enable Event Logging Server** checkbox is available under **System-Wide Configuration> SysLog Server > SysLog Server-1** . Click to enable (evlog output is sent to syslog); unclick to disable (evlog output is not sent to syslog). The default is unchecked. This field can be enabled only for the first Sys-Log Server in the list.
- **In CLI:**
  - The **SYSLogEnableEvlog** command is available under **SYSLogServer-1**. Valid entry: **0** (disable, default) or **1** (enable). This field can be enabled only for the first SysLog Server in the list.

P108-09-26

4929 CID feature works in Receive Direction only

The E&M Caller ID feature implemented by MR 4923 worked in Receive Direction only. When the Tenor made outgoing calls, it sent only DNIS, independent of configuration. This has been resolved.

P108-09-25

4928 GUI does not connect to Analog Tenors running P108-09-24

When MR 4923 (E&M Caller ID feature) was added to the Tenor DX, it caused the Tenor Configuration Manager to not connect with the Analog Tenors. As a work around, enable the feature changes in the CLI database for Digital Tenor only, not Analog.

P108-09-24

4920 SIP Decode failed when CallID had % character

When the Tenor decoded the CallID header and found a % character present, it would interpret the character (rather than taking it as a literal character). As a result, the Decode failed. This has been resolved.

4921 No ISDN Connect message for 911 calls

Some T1/ISDN carriers do not provide an ISDN level connect message for 911 calls. As a result, the Tenor never sent a 200Ok message to the SIP side to connect the call. A new feature enables you to configure up to 8 phone numbers for mapping of ISDN Progress to SIP connect. Each phone number handles a maximum of 16 digits. This feature is applicable only for calls coming into SIP and out ISDN (Trunk/ISDN). The DN specified is used to match against the final translated DN (not the original incoming DN) that is sent out on the DN side.

A new varconfig file is available to support this feature: **mapIsdnProgressToSipConnectDN1 DN1** (where DN1 indicates the DN number).

4922 IP to IP Calls: Unsupported Codecs in the SDP caused unexpected behavior

If the Tenor was configured for IP to IP calls and if the originating IP device/phone had any codecs in its profile that the Tenor's DSP did not support, the incoming SIP invite caused the Tenor to crash and reboot. This has been resolved.

4923 Calling Party Number (Caller ID) support in CAS E&M

Although the current CAS E&M implementation does not support receiving and sending of ANI (calling party number), new configuration options are included for E&M to support a modified DNIS that includes ANI.

To support this feature, the following new configuration items are available for the CLI only (not available in the Tenor Configuration Manager) under CASSG-1:

- ENMAniDnisSequence
- ENMStartDigit
- ENMAniDnisDelimiter
- ENMEndDigit

For detailed information about the commands, see the Command Reference Guide.

4924 Three Beeps when making outgoing call via Tenor DX

When a call was made from a Lync Mediation Server, a user would receive three quick beeps when making any outbound call from the Tenor DX before the user on the Lync client side heard any ring back. This has been resolved.

4925 T.38 offered in reInvite to Tenor, but Tenor chose G.711ulaw instead

When a Tenor was making an outbound fax call through a SIP Trunk, and a T.38 in the INVITE was returned, the Tenor chose G.711ulaw instead of T.38 and the fax subsequently failed. The Tenor was unable to re-negotiate media when it received a ReInvite that put the Tenor on hold. A new **var_config** item is now available that configures the Tenor to accept and negotiate media change on a ReInvite even if it indicates Hold.

To enable a Tenor to negotiate for a ReInvite that puts the Tenor on hold, a new varconfig configuration option is added: **sipNegotiateMediaOnHold**.

4927 R2 ANI Collection more than once (Applies to Tenor Digital only)

Under certain conditions the Tenor would attempt to collect ANI more than once, causing an R2 call to fail. This has been resolved.

P108-09-23

4918 Effect of EnableLocalRouting parameter was not retained after Reboot (Analog only)

If EnableLocalRouting was set to 1, DNCM routing took precedence over the Routing Table. However, after a power cycle the local routing did not work. This has been resolved.

P108-09-22

4915 Disable Auto Switch: Ignore AutoSwitch License if found

The Autoswitch feature has been disabled, even if a valid AutoSwitch license is discovered.

4916 PSTN_to_SIP call: Music on Hold not heard when Lync put Call on Hold

When a call was made from the PTSN to Tenor to Lync and then the call was put on hold from the Lync side with a re-INVITE SDP from Lync Mediation contained a=sendonly, the PSTN caller could not hear the music on hold played from the Lync Server. This has been resolved.

4917 UDP Port 17185 (WIN Debug Agent) not to be opened

For security reasons, Port 17185 will not be opened.

P108-09-21

4717 If no media in re-Invite, do not send Hold to remote end

After an IP-IP call was up through the Tenor, the server was sending the Tenor a re-Invite with no SDP. The Tenor took this re-Invite as a trigger to send a hold request to the far end. Instead, a regular re-Invite (with SDP) should have been sent to the far end, which would result in an OK with SDP to the server and a received ACK with SDP from the server. This has been resolved.

4855 Denmark Caller ID generation and detection

Note: A varconfig implemented this feature in P108-09-14 ("DtmfCidStartDelimiter" and "DtmfCidEndDelimiter") but are now removed and replaced with the below configuration options available in the Command Line Interface (CLI).

Denmark Caller ID generation and detection is now included as a feature. There are two new configuration fields available for Caller ID generation and detection: DtmfCidStartDigit and DtmfCidEndDigit. These fields are available via Command Line Interface (CLI) under Gateway.

DtmfCidStartDigit. Valid entry: 0 - 15. Default entry: 13

DtmfCidEndDigit. Valid entry: 0 - 15. Default entry: 12

Entry definitions include the following:

0-9: Digit '0'-'9'

10: Digit 'A'

11: Digit 'B'

12: Digit 'C'

13: Digit 'D'

14: Digit '*'

15: Digit '#'

Guidelines

- For this feature to work, CallerIDDetection must be set to 7 (DTMF, FSK, No Ring) in CASSG-Line and 7 (DTMF, No Ring) in CASSG-Phone.
- When a call comes into the Tenor with CallerID that is restricted or unavailable, and is routed out to the Tenor's FXS, the Tenor will send CallerId "D1#" to the phone.

4874 Tenor certified in support of Microsoft® Lync™ Server 2010

The Tenor DX Series Digital and Analog VoIP MultiPath Switches has been qualified for Direct SIP Enhanced Gateway in support of Microsoft® Lync™ Server 2010 deployments.

A new varconfig file is available to support this feature:

lyncInterOp 1. This enables Tenor to drop short packets (i.e., STUN and Comfort Noise).

lyncInterOp 0. This disables Tenor from dropping short packets (i.e., STUN and Comfort Noise).

Guidelines

- In order to use a Tenor in a Lync Environment, the following varconfig MUST be used: lyncInterOp 1`
- The previous varconfig lyncEarlyMediaDropShortPackets has been removed and replaced by lyncInterOp.

4890 IVR Prompt when MultiPath Call is routed to PSTN

Using an IVR prompt when a Multipath call is routed to the PSTN is now supported in MultiPath routing. This has been fixed to support MultiPath routing.

4895 Tenor did not send PRACK on SIP 182 queued message from IP-PBX

The Tenor did not send a PRACK message on a SIP 182 message from an IP-PBX, which caused the PBX to not open the audio path for the queued call. The Tenor will now respond back with a PRACK message.

4896 Music On Hold did not work in IP to IP calls

When a call was put on hold from a Lync client, the caller did not hear the music on hold coming into the Tenor from Lync. This has been resolved.

4899 Invite message was not properly encoded

When a message was sent to SSG having Request-URI as INVITE, the message was not completely sent through. This has been resolved.

4900 Digit missing for the DNIS (from To header or Request-URI)

A digit was missing for the DNIS number (from the To header or Request URI). This has been resolved.

4902 P108 New Routing; allow use of Routing Table instead of direct DNCM Routing

NOTE:: For upgrades to P108-09-21: When using the New Routing Table and direct DNCM routing, ensure ***EnableLocalRouting*** is configured to ***1 (enabled)***. If this field is not configured properly, the existing direct DNCM routing will not work.

With Multipath routing, if the dialed number matched a DNCM, the Tenor would directly route the call between the analog ports, and not use the Routing Table. A new feature enables the Tenor to use the Routing Table to send the call to a SIP server, and have the call routed back to the Tenor, instead of doing Direct local routing.

A new configuration item, ***EnableLocalRouting*** (available via Command Line Interface (CLI) under Gateway.) can be configured to enable/disable this option. Valid entries: 0: Default, disabled (the Tenor by default will use the Routing Table instead of direct DNCM Routing) or 1: Enabled (direct DNCM routing is used).

4903 NAT IP Directory Feature added the wrong external NAT address

When a NAT IP directory was configured, the Tenor sent out the external NAT IP address (configured under the Ethernet setting) as NAT in the SIP invite and in the SDP to the IP side, rather than sending out its own locally configured Ethernet IP address. This has been resolved.

Tenor-4907 No Ring Back on Calls with Ring No answer Forward to another Lync client

When a Lync client (A) was forwarded with a Ring No Answer to another Lync Client (B) within the Lync environment, the caller heard a partial ring before silence (even though the Lync Client B was still ringing until it went to voice mail). This has been resolved.

4908 Order of parameters causes Tenor to not encode Telephone Event 101 in forwarded Invite

When a Tenor received an Invite with SDP parameters, the forwarded invite from the Tenor was properly encoded. However, if the incoming parameters were not in irder, the forwarded invite from the Tenor was incorrect. This has been resolved.

4910 Tenor does not Ring specific channels/phone numbers in DN-Channel Map table

When the Tenor (using the New Routing Table) received a SIP invite with any of the configured phone numbers in the DN-Channel map table, the Tenor rang the first available FXS channel (like a hunt group) rather than ringing the specific FXS channel based on the phone number that is in the DN-Channel map table. This has been resolved.

4911 Under specific conditions, the Tenor reboot

If a Tenor had both FXO and FXS lines configured for offline operation nd the CallerID detection type configured in CAS (set to Type 7), a call into the Tenor would cause a reboot. This has been resolved.

Tenor-4913 Tenor UAS - UAC Refresh Session Timer

With Session Timer supported in the Tenor, when a SIP End Point sent an Invite indicating that it wanted to be the refresher (refresher=uac), the Tenor was overriding with refresher=uas in its response, instead of honoring the client's request to be the refresher. This has been resolved.

4914 After receiving Call Progress Tenor was not sending 180 ring or 183 SIP message to Lync

When the Tenor BX received a Call Progress, it would not send the 180 Ring or 183 SIP message to the Lync/ Mediation Server. As result, the Tenor would receive a SIP CANCEL message from the Lync/Mediation Server. This has been resolved.

P108-09-19

4884 Tenor reboot when "st ds1" command was issued

When the st ds1 command was issued, the Tenor reboot. This has been resolved.

4885 Messages sent to primary outbound proxy when invite was sent to secondary outbound proxy

When the primary proxy was down, the INVITE messages were send to the secondary proxy, but subsequent messages (i.e., PRACK and BYE) were being sent to the secondary proxy. This has been resolved.

4886 Tenor multiplying Authentication Header

When the Tenor sent out an INVITE message, if the proxy authentication credentials failed, the Tenor would keep adding multiple authentication headers by repeating them (even if the credentials were passed). As a result, this eventually caused the message to grow too large and end the calls.

4888 Tenor did not decode message

The Tenor was not decoding the SIP header message correctly. This has been resolved.

4889 Invite without an SDP caused the call to fail

When the Tenor sent an invite (without an SDP) to an endpoint. and the endpoint sent back the 200 OK message, an exception occurred and the call failed. This has been resolved.

P108-09-18

4865 Failed to decode header in the Refer message

Tenor was unable to decode the header in the Refer message. This has been resolved.

P108-09-16

4882 Tenor did not retrieve parked call

When a Tenor attempted to retrieve a parked call, the call was dropped. This has been resolved.

P108-09-15

4881 Tenor reset on a call transfer scenario

When a call transfer was attempted with early media turned on, the Tenor reset. This has been resolved.

P108-09-14

4849 One way audio

There was a problem with one way audio and the Called party could not hear the caller. This has been resolved.

4855 Denmark Caller ID generation and detection

Note: This feature is now available for CLI configuration, rather than var_config. See P108-09-21.

4861 Reset with Transfer/Redirect

The Tenor reset about every two hours with the following PRI>SIP call flow:

- Far end sent Refer (unattended)
- Tenor sent new Invite, which received a 302 response
- Tenor sent another new Invite, which received another 302 response
- Tenor sent another new Invite which connected

This has been resolved.

4864 One way voice after transfer (Call Relay SP only)

Tenor had an issue with a re-Invite with no SDP after a call was put on hold, and resulted in one way voice after transfer. This has been resolved.

4866 Call transfer: Second Invite Issue (Call Relay SP only)

After an initial call was setup, and a transfer was initiated, there was an issue with a second invite and the call failed. This has been resolved.

4868 One way audio on call transfer

An attended call transfer to a User Agent in the same Tenor resulted in one-way voice from the Tenor to the far end. This has been resolved.

4876 Manipulate cause code 47 to 34

A new command in the var_config.cfg file, mapCauseCode47To34FromSipToCircuit, will now map cause code 47 to 34, and send out 34 back to the calling PRI side. Options to enable/disable this option are as follows:

- mapCauseCode47To34FromSipToCircuit 1. Enabled. Cause code 47 will be sent out as 34 back to the calling PRI side.
- mapCauseCode47To34FromSipToCircuit 0. Disabled. When Cause code 47 is generated from SIP to ISDN, it will be sent out as 47.

4877 T-Mobile *,# in Invite not to be Escaped

A new varconfig sipDoNotEscapeDigitsInPhoneNumbers has been introduced to allow the *, # characters to be escaped. When the varconfig command is used, these characters are not escaped and sent out literally in the Invite Header and To field.

4878 Tenor reboot when multiple Number Tables with many entries were added

The Tenor reboot automatically when multiple NT tables were used. This has been resolved.

P108-09-13

4856 Destination number terminated with a '#' caused the Tenor to crash

If a '#" sign was located at the end of a destination number, it caused the origination Tenor to crash. This has been resolved.

4858 "From" header not being properly decoded

A specific "From" header (in which the From line ends with "cscf") was not being properly decoded. As result, when the Tenor received the Invite, it was rejected because of the decode problem, and the call was dropped. This has been resolved.

4859 With MultiPath Routing, not all CDR fields were populated

When MultiPath routing was used, not all the fields in CDR format 199 were set to proper values. This has been resolved.

P108-09-12

4807 ARP Expiration now configurable

A new ARP (Address Resolution Protocol) expiration varconfig command, arpexpiration, is now configurable in the Tenor to determine how long (in seconds) an ARP table entry should live. Valid entry: 1 - 1200 seconds. Default entry: 1200 seconds (20 minutes).

4828 Help problem in RoutingTable and NumberTable

In CLI, there was a problem with help not being displayed for the RoutingTable or NumberTable. This has been resolved.

4829 Unable to configure SL 2 attributes via auto provisioning

For Tenor Digital Units (Tenor DX and Tenor BX), the attributes of PowerOffBypass and PhantomPowerPassthru (available under Slot-2) were not able to be configured via the auto provisioning feature. This has been resolved.

4836 Incorrect ANI TON/NPI Reported In Event Log (ISDN)

The ANI TON/NPI was being reported incorrectly in the Event Log due to an internal decoding error. This has been resolved.

4840 Allow Only Proxy Calls not working under certain circumstance

AOPC (Allow Only Proxy Calls) was not working when using DNS for registration purposes. This has been resolved.

4845 Auto-Provisioning was not functional

Auto Provisioning was not functional when upgrading from a previous release to P108. This has been resolved.

4846 Host Name and Dial Plan enhancements

Two new enhancements have been implemented in the Tenor: Host Name and Dial Plan
Host Name. In the Tenor, you are now able to configure a Host Name for the Tenor. If the Host Name is configured and the value in SIPSG indicates to send the IP address in the From Header, the host name will now be sent instead. Through the var_config.cfg file, you are able to configure the Host Name: TenorHostName <hostname>.
Dial Plan. A new dial plan option (Supplementary Services 3) is now supported in the Tenor. Access this feature in the Tenor Configuration Manager as follows:

- Select System-Wide Configuration> Dial Plan.
- From the Dial Plan Country drop down box, select Custom.
- Select the UPDP tab.
- Add a Dial Plan.
- From the Type box, select 9: Supplementary Server Type 3.

4848 One way MOH, not passing first reinvite

For IP-IP calls, when a call was put on hold, the hold re-Invite was not being passed through correctly. As a result, MOH (Music on Hold) was not being played. This has been resolved

4851 FAILED to decode Record-Route in the SIP invite

The Tenor gateway was receiving a SIP invite from the IP-PBX, but failing to decode the “Record-Route” field in the invite. Now, the Invite decodes properly and will include the correct Record-Route field.

4854 IP-IP alters Remote Party Id like call has originated from PSTN

The Tenor (New Routing IP-IP calls) altered the outgoing Invite as though the request was from the PSTN, rather than another IP. The Tenor now passes the From and Contact fields without altering these fields.

P108-09-10

4794 Tenor boot directly into Safe Mode upon startup

Tenor should go into safe mode if it reset numerous times in a specific amount of time. However, on one occasion, the Tenor reboot into safe mode upon initial startup. This has been resolved.

4825 Gateway had no way to separate Authentication user name from primary user name

For a Tenor registration with Authorization parameters, the user name in To/From was the same as the user name in the Authorization header. These needed to be different. This has been resolved.

4827 CDR problem with Table Based Routing

If Table based routing was enabled, the Called number was missing from the CDR record. This has been resolved.

P108-09-09

4810 Changes to the map.cfg file required a reset

When any changes were made to the map.cfg file, the Tenor would require a reset. To resolve this, a new command, cmd sip readmap, allows the Tenor to read the values of the map.cfg file without requiring a reset.

4812 IPRG in SIPSG "not set" option caused Tenor to reset while receiving calls

When both source and destination Tenors were registered to a SIP server (in the SIPSG of the destination Tenor, the IPRG was configured to 'not set'), and a call was placed from the source Tenor to the destination Tenor, the destination Tenor reset as soon as it received the Invite. This has been resolved.

4813 Analog phone did not hear disconnect tone after hold

When an Analog phone answered a call, and the call was put on/off hold, if the far end disconnected the call, the Analog phone would not hear the disconnect tone. This has been resolved.

4815 SDP decode failure

A SDP decode error appeared in the Tenor. This has been resolved.

4816 Voice went to wrong MAC address after re-invite (Call Relay only)

On a re-Invite, when the IP address in SDP changed, the Tenor put the correct IP address in the IP header, but the Ethernet header still contained the MAC address of the previous audio destination. This has been resolved.

P108-09-08

4788 DNS Host timer refresh issue

When a call failed, the Tenor was not handling the call failover properly. This had to do with a DNS Host Timer refresh issue, but it has been resolved.

4800 DNIS Command moved to Configuration Manager and CLI

The command, IncomingDNISinTo (previously a varconfig option named SIPUseToAsNumber, now available under SISG in the Tenor Configuration Manager and CLI) is now used as follows:

For the CLI:

Access SIPSG. The two fields are as follows: IncomingDNISinTo 0 (Incoming DNIS is taken from Request URI) and IncomingDNISinTo 1 (Incoming DNIS taken from the To: header).

For the Tenor Configuration Manager:

Click on SIPSG > Advanced. From the Incoming DNIS In drop down box, select Request URI (Incoming DNIS is taken from Request (URI) or Header (Incoming DNIS taken from the To: header).

4802 Q.SIG Protocol Profile Value

A new varconfig parameter QSIGProtocolProfile is now available, which allows you to change the Q.SIG protocol profile to either ISO variant or ECMA variant. Valid entries are as follows:

- QSIGProtocolProfile 0. ISO variant (default value).
- QSIGProtocolProfile 1. ECMA variant.

4803 Tenor Auto Provisioning error on post-provisioning reboot

In AutoProvisioning, following a post-provisioning reboot, the Tenor would not wait the proper amount of time before attempting to download the provisioning file. As a result, the Tenor would continually reboot itself.

4804 Tenor Auto Provisioning error on setting Digital Interface Line Type

In AutoProvisioning, when the Tenor was set to Linetype 0 (Digital Interface-1), it was not taking effect. This has been resolved.

4807 ARP Expiration now Configurable

The ARP expiration time has been made configurable per a new varconfig command arpexpiration. Through this command, you can configure how long (in seconds) an ARP table entry should live. Valid entries are 1 - 1200 seconds; default value is 20 minutes (1200 seconds). For example, arpexpiration 160 will allow the ARP table entry to live for 160 seconds.

4808 Re-Invite with no SDP was not being handled correctly by CallRelay

When a call was up and the CallRelay received a re-invite, it sent an OK with an OFFER. The far ends responded with an ANSWER, but the media was different than the media from the original call. As a result, the CallRelay was unable to resolve this. This has been resolved.

P108-09-07

IP-IP calls did not pass Diversion header end to end

Note: Applies to New Routing or CRSP IP-IP calls only.

The Tenor dropped the Diversion header on the outbound IP call leg. This has been resolved.

Improper Analog Caller ID Privacy Format

When a caller blocked his/her Caller ID from being displayed on the calling party’s phone, the PSTN transmitted just the letter P as the Caller ID. The Tenor added this letter P in the FROM field of the SIP INVITE, which was displayed as the Caller ID on the receiving SIP endpoint. This has been resolved.

4795 Tenor returned a 415 on unsupported T.38

When a call was up and the far end sent the Tenor a re-Invite with T38 fax, the Tenor was not configured for T.38 and sent a 415 (unsupported media type) in response. AT&T testing required a 488 to be sent instead. This has been resolved.

4796 Diversion Header failed to decode

The following Diversion header was not decoded properly:

"Diversion:"AA1 HostedPBXGroup1"<sip:13359013@atyaf.com;user=phone>;privacy=off;reason=deflection;answered;counter=1".

This has been resolved.

4797 Tenor checked for forking even if call was established

When a call was connected and the far end sent a re-Invite (which the Tenor matched on session), the Tenor failed to match the request-uri and declared that there was a forking situation.The Tenor should not have been checking for forking when the call was already established. This has been resolved.

4799 SS7; Category set incorrectly in IP->PSTN direction

If the incoming SIP Invite header did not contain a Category header, the Tenor would incorrectly pass a 255 in the Category IE. Instead, when there was no Category in the Invite, the Category should have been set to the Calling Party Category configured in the ISDNSG. This has been resolved.

4801 New Routing- 2nd route not attempted when terminating channel was unavailable

When there were two routes defined from an incoming SIP call, and no circuits were available, the second route was never being attempted. This has been resolved.

P108-09-06

4785 SIP To header decode failure

When the Tenor received an Invite, it failed to decode the TO header. As a result, the call failed and an exception was generated. This has been resolved.

P108-09-05

4671 CID failed for calls inbound on IPRG

When using the varconfig command cidt2 1 (supports the '.' placeholder), the Tenor failed to translate Caller ID for inbound calls on the IPRG with an associated CID translation directory that included a number pattern and replacement number containing '.' in it. This has been resolved.

4762 Corrupted cli_help_msg.txt caused Tenor to reboot

While downloading the cli_help_msg.txt file to a Tenor, the file may have gotten corrupted due to ftp/network problems. This has been resolved.

4779 IVR field available

In TCRG, when the ivrtype is set at 1 or 8, the following fields are now available:

- PIN
- IVRAccessNumber
- IVRAnswerDelay

4781 IPRG defined in static route was not being applied for inbound SIP calls

When an IPRG was attached to a SIPSG (for inbound SIP calls), if there was a static route match based on destination address, the IPRG in the static route did not take effect. This has been resolved.

4782 Match invalid CIDXD pattern

When a called ID translation was used with positional dots in the Pattern, and a non-matching pattern came in, the translation would still match and then send an invalid caller ID. As a result, a SIP error would occur, resulting in the Contact and From fields missing. To resolve this issue, enter cidt2 1 in the var_config.cfg file.

4783 Referred-by Header failed decode

When the Tenor received a SIP invite, the Tenor could not decode the URI in the Referred-By header. This has been resolved.

4784 IP address in SDP was sent as private when it should have been public

The IP address in the SDP was being sent by the Tenor as private, even when the external NAT was set. The IP address should have been sent as public. This has been resolve

P108-09-04

4758 Dual Register Proxy Failover Problem

In dual registration Proxy failover mode, the Primary Proxy should have attempted proxy fail over mode first, and if that failed, the secondary Proxy should have been attempted. Instead, the Tenor did not failover to the secondary SIP server on a no response from the Primary Proxy. As a result, all subsequent calls failed. This has been resolved.

4760 Routing with DNCM

With FXO ports configured as IVR type 1 (second dial tone), when the user called in and received second dial tone, and then dialed an extension that was configured as a specific channel DNCM, the call was not routed according to the DNCM parameters. This has been resolved.

4763 SS7: Enhance Continuity Check Support

For SS7, when calls were sent to the Tenor for a loopback test on a previous circuit, the Tenor would ignore the call. Now, the Tenor waits for the results of the loop back test to continue or drops the call.

4765 Ringback not working with IVR and EarlyMedia

Ringback was not working with IVR and early media enabled in SIPSG. This has been resolved.

4766 SipInfo Payload Decode Failure

Tenor failed to decode SipInfo payload when 11(*) and 12 (#) were in the payload of the SipInfo received from a ResponsePoint unit. The Response Point unit encoded these characters differently depending on the brand of the phone from which the call originated. For example, if the call originated from a Syspine phone, characters * and # were in the payload. If the call originated from an Astra phone, numbers 11 and 12 were in the payload. Tenor could only decode characters * and # and not numbers 11(*) and 12(#). This has been resolved.

4767 Cancel re-sent even though final response received

When the Tenor received a final response to an ongoing Invite, it re-sent a Cancel message, which made the call fail. Now, the Tenor will send a BYE.

4769 AllowOnlyProxyCalls did not work for TCP calls

The config parameter, AllowOnlyProxyCalls, in SIPSG did not work for TCP calls. This has been resolved.

P108-09-03

4744 Incorrect CName Encoding Under NI2 (Tenor DX only)

Cname was not being encoded properly for outgoing PRI calls when NI2 was used. This has been resolved.

4745 Change syntax of backup and restore commands

The names for two backup commands have been changed (these are general commands available under configuration in CLI):

- try restore database changed to restore database
- try backup database changed to backup database

4746 Tel URI in Diversion header was not being decoded properly

When a Tenor received an Invite with the Diversion header field, the Tenor was unable to decode the Diversion header with a Tel URI, and the Invite was not processed. This has been resolved.

4748 Tenor used "To" header as called party rather than the P-Called-Party-ID header

Tenor routed calls based on the P-Called-Party-ID header as the called number. Now, the Tenor will route the calls based on the "To" header.

4750 New routing - wildcards did not work

In New Routing, a Number Table match was found even if the normalized number was longer or shorter than the configured number pattern using wildcard characters (dots). This has been resolved.

4751 Database directory enhancements

If the directory /hd/cfg/db/ was deleted accidentally, the system could not write any database files. As a result, any changes to the database were lost after the system reboot. Now, as a preventative measure for the files not to be lost, upon startup, the Tenor will check to see if the directory /hd/cfg/db/ is missing. If it is, the system will create a new /hd/cfg/db/ directory in which the database files can be written.

4753 PRI ETSI Status message resulted in call failure

If a Digital Tenor received a STATUS message in response to an outgoing SETUP, it would have dropped the call. This has been resolved.

4756 Incorrect transportType used on Call 2 before attended transfer

When an initial call was made using UDP (because transportType in SIPSG was set to UDP), and the user hit flash hook and dialed another number, the call may have been made using TCP. The call should have been made using UDP. This has been resolved.

4757 CRSP with new routing asked for license

When a Tenor (CRSP) was upgraded to P108 (which includes new IP to IP routing), the Tenor Configuration Manager gave an error/warning message about needing the advanced routing P108 license.This should not have occurred and has been resolved.

P108-09-02

4741 Port mapping was not working in P108

Port mapping using SIP did not work in P108. This is now working.

4742 IVR was not accessible with New Routing

IVR types were not accessible using the New Routing. IVR type 0, 1 and 8 are now exposed.

P108-09-01

4705 REFER-TO header Transport tag was not working

The Tenor did not obey the transport specified in the REFER-TO header of a REFER message. Instead, it was using the transport type configured in the SIP Signaling Group to send the Invite out. This has been resolved.

4717 If no media in re-Invite, a Hold was being sent

After a call is up, the server was sending the Tenor a re-Invite with no SDP. The Tenor took this Re-Invite as a trigger to send a hold request to the far end. Instead, a regular re-Invite (with SDP) should have been sent to the far end. This has been resolved.

4718 When BYE was received at a non-existent session, the Tenor now sends 481 instead of ignoring

If a Tenor received a BYE at a session that did not exist, it would ignore the message. Instead, a 481 will now be sent in response to a BYE with non-existent call session.

4723 For new routing, call was not being re-routed

A call tried to route out PRI, but there were no available channels. The return cause was Busy (17), and an alternate route was not attempted. This has been resolved.

4727 Tenor improperly set "annexb=no" in SDP

If the Tenor received a request with SDP that did not contain annexb, then in response, the Tenor would improperly put annexb in the SDP. This has been resolved.

4732 Error in setting static IP address

Upon startup, there is an option of hitting the "i" key via the RS232 interface to set a static IP address of the Tenor. Occasionally, the Tenor would not save the IP address correctly, and when it reboot, the IP address would show as 0.0.0.0 instead of the keyed-in IP address. This has been resolved.

4733 Transport type overrides entry in var_config file

When the Transport Type in SIPSG was set to UDP, it ignored the varconfig entry "SipTransportDefault" set to 1 (to use TCP for the SIP calls). This has been resolved for the varconfig entry to take precedence, resulting in the call being sent over TCP.

4736 SIP Server field restricted

The PrimarySIPServer was restricted and unable to be configured. This has been resolved.

4737 Tenor stopped working when the command of "st ds1 call" was issued

The Tenor stopped working when the command "st ds1 call" was issued. This has been resolved.

4738 Factory Fresh Setting Static IP Address problem (Tenor CMS only)

There was an IP address configuration problem on a new Tenor CMS. This has been resolved.

4739 Changes over Multipath Routing Enhancements

Several new changes/feature enhancements have been included for the Tenor's new Routing feature.

In the Routing Table, the following configuration options have been changed as follows:

- Inbound TG changed to Incoming From
- OutboundTg changed to OutgoingTo
- InboundMatchCriteria changed to IncomingMatchCriteria
- NumberForDiversion changed to DiversionPilotNumber
- DiversionContent changed to DiversionMatchCriteria

For the DiversionMatchCriteria field, a third value of None (-1) has been added to the other two values available, CalleD (0) and CalliNg (1). The value of None will become the default value.

Currently when you enable the UseRoutingTable field, a warning is given about the permanent loss of non-Routing Table related settings. Two new commands are added for database backup and restore.

- try restore database. Restores db.txt file from db.bak file, if it is available.
- try backup database. Backs up db.txt file as db.bak file.

4740 CLI should put message for hunt group with New Routing

When new routing was enabled, and a user tried to access Hunt LDN, there was not a message stating that the hunt group was disabled. The following message now exists:

Error: This command is inaccessible in the current configuration.
An su login is required to access it.