Release Notes

AudioCodes SBC, Media Gateway, and MSBR Series

Session Border Controllers
Multi-Service Business Routers
Analog & Digital Media Gateways

Version 7.2

audiocodes

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Notice

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WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Throughout this manual, unless otherwise specified, the term *device* refers to the AudioCodes products.

Related Documentation

Document Name
Gateway and SBC Product Series
Mediant 500L Gateway and E-SBC Hardware Installation Manual
Mediant 500L Gateway and E-SBC User's Manual
Mediant 500 E-SBC Hardware Installation Manual
Mediant 500 E-SBC User's Manual
Mediant 800 Gateway and E-SBC Hardware Installation Manual
Mediant 800 Gateway and E-SBC User's Manual
Mediant 1000B Gateway and E-SBC Hardware Installation Manual
Mediant 1000B Gateway and E-SBC User's Manual

Document Name
MP-1288 Hardware Installation Manual
MP-1288 High-Density Analog Media Gateway User's Manual
Mediant 2600 E-SBC Hardware Installation Manual
Mediant 2600 E-SBC User's Manual
Mediant 4000 SBC Hardware Installation Manual
Mediant 4000 SBC User's Manual
Mediant 9000 SBC User's Manual
Mediant 9000 SBC Hardware Installation Manual
Mediant SE SBC Installation Manual
Mediant Virtual Edition SBC Installation Manual
Mediant Virtual Edition SBC for Microsoft Azure Installation Manual
Mediant Virtual Edition SBC for Amazon AWS Installation Manual
Mediant CE SBC Installation Manual
Mediant Software SBC User's Manual
MSBR Product Series
Mediant 500L MSBR Hardware Installation Manual
Mediant 500L MSBR User's Manual
Mediant 500 MSBR Hardware Installation Manual
Mediant 500 MSBR User's Manual
Mediant 800 MSBR Hardware Installation Manual
Mediant 800 MSBR User's Manual

Document Revision Record

LTRT	Description
26957	Initial document release for Version 7.2.
26963	Capacity updated for Mediant 9000, Mediant 4000/B detection features, and Mediant 9000 with Media Transcoders.
26968	Mediant VE High-Capacity VMware capacity; Mediant 500L Gateway & E-SBC capacity (hybrid).
26969	Ver. 7.20A.001; Typo in Mediant 4000B SBC capacity table.
26970	Ver. 7.20A.001 updates: Mediant VE SBC virtual platforms (Amazon EC2 and SR-IOV); Registered users capacity updated for 1/2/4 vCPU 4 GB RAM Hyper-V; Capacity added for Amazon EC2 and SR-IOV.
26980	VoIPerfect updates; Capacity table updates; RFCs added.
26983	Ver. 7.20A.002; G.722.2 added to AMR-WB.
26987	Ver. 7.20A.100.

LTRT	Description
26990	Capacity updates - MP-1288; Mediant VE (c4.2xlarge, c4.8xlarge, with Media Transcoders); Mediant 9000.
26994	Update to section 'Multiple SRSs and SRS Redundancy for SIPRec'; new feature 'Debug for Remote Web (HTTP) Services'; constraint VI-140547 added; SRTP-RTP capacity updated for Mediant SE DL360p G8 20-cores and DL360 G9 8-cores; Mediant VE c4.8xlarge removed.
26998	OVR capacity; WebRTC capacity.
26999	Note added to Gateway and SBC Capacity for Mediant VE SBC and vMT-type Media Transcoder.
27083	Ver. 7.20A.104.001.
27084	Typos.
27090	Ver. 7.20A.150.004.
27091	▪ New 7.20A.150.004 features: Analog Voice Interface Support on Mediant 500L E-SBC/Gateway; Ringback and Held Tones per User; Same VLAN ID for Multiple Ethernet Devices ▪ Updated 7.20A.150.004 sections: Routing Back to Sender; IP Group Parameter Representation in Call Setup Rules; Skype User Presence Notification for Non-Skype Endpoint Devices; FXS Line Testing (typo); Resolved Constraints (VI 143342); MSBR Known Constraints (VI 141108); SIP Signaling, Media and User Capacity (Mediant 500 E-SBC, Mediant 500 MSBR, Mediant 800 MSBR) ▪ Updated 7.20A.150.004 tables: Mediant 500 Hybrid E-SBC (with Gateway) Media & SBC Capacity; Mediant 800/B Gateway & E-SBC SBC Session Capacity per Capabilities (SBC Only)
27092	▪ Ver. 7.20A.106.003. ▪ Modifications to Ver. 7.20A.150.004: ✓ Updated section: Session Capacity Increases ✓ New feature: Prefix String for External Line Enhancement ✓ Capacity tables: Mediant VE on KVM; Mediant 800/B Gateway & E-SBC Channel Capacity per Capabilities
27095	▪ Ver. 7.20A.152.009 (SBC/Gateway) ▪ Ver. 7.20A.152.003 (SBC/Gateway) ▪ Updates to Ver. 7.20A.150.004: New feature - High-Availability Disconnect; modified feature description - Console Access Mode; modified feature description - Snapshot Load through CLI; modified feature description - Routing Back to Sender ▪ Capacity updates: SIPRec; Mediant 500 MSBR; Mediant 800 MSBR
27099	Capacity updated for 8-vCPU Mediant VE SBC on OpenStack/VMware.
27241	▪ Ver. 7.20A.154.007 (SBC/Gateway) ▪ Updates to 7.20A.150.004: Modified feature description - OVR Support in High-Availability Mode
27242	▪ Ver. 7.20A.154.025 (MSBR) ▪ Updates to 7.20A.154.007: Compatible AudioCodes One Voice Operations Center version
27246	▪ Ver. 7.20A.154.052 (SBC/Gateway) ▪ Global replacement of "EMS" and "SEM" with "One Voice Operations Center"
27247	▪ Ver. 7.20A.154.059 (SBC/Gateway) ▪ VI 143930 added to Ver. 7.20A.150.004

LTRT	Description
27249	▪ Ver. 7.20A.156.009 (SBC/Gateway) ▪ Capacity figures
27250	▪ Additional resolved constraints for Ver. 7.20A.156.009.
27251	Ver. 7.20A.154.061 (MSBR).
27252	Ver. 7.20A.156.023 (SBC/Gateway).
27254	Ver. 7.20A.156.041 (Mediant 9000 Only).
27256	Ver. 7.20A.158.009
27257	▪ Ver. 7.20A.154.078 (MSBR). ▪ Transcoding Mode parameter name change (7.20A.158.009)
27258	Ver. 7.20A.158.012 (SBC and Gateway)
27260	Ver. 7.20A.200.019 (SBC and Gateway) Ver. 7.20A.158.035 (SBC and Gateway)
27261	▪ Ver. 7.20A.200.038 (MSBR) ▪ Resolved constraint 148119 added to Ver. 7.20A.158.035 (SBC/Gateway) ▪ New features added to Ver. 7.20A.200.019 (SBC and Gateway): Performance Monitoring Thresholds Included in ini File; ENUM Query Enhancement for Call Setup Rules
27265	▪ Ver. 7.20A.162.001 (Mediant 9000 SBC only) ▪ Ver. 7.20A.158.056 (SBC and Gateway) ▪ Updates to 7.20A.200.019: NGINX alarms and CLI commands; SFTP for Packaged Configuration file; new feature - IDS alarm improvements feature added; new feature - alarm tables enhancements; new feature - Retry-After header; new feature - enhanced cross validation for UDP ports; new feature - randomized expire time; new feature - variables for Message Manipulation; new feature – parameter name changes from "Prefix" to "Pattern"
27267	▪ Ver. 7.20A.200.550 (SBC and Gateway) ▪ Ver. 7.20A.158.056: VI150995 and VI152028 added as resolved constraints ▪ Ver. 7.20A.200.019: Update (TFTP) to Packaged Configuration File ▪ RFCs added (7866, 7245, 8068, 7865, 6341)
27268	▪ Ver. 7.20A.162.017 (SBC and Gateway) ▪ Ver. 7.20A.100: New feature - acCertificateExpiryNotification
27270	▪ Ver. 7.20A.202.112 (SBC, Gateway and MSBR series) ▪ Note added for SbcPerformanceProfile parameter
27272	▪ MTC resolved constraint added to Ver. 7.20A.162.017 ▪ Note added for SbcPerformanceProfile ▪ VI-147159 added to Ver. 7.20A.202.112
27275	▪ Ver. 7.20A.202.141 (SBC and Gateway) ▪ Ver. 7.20A.158.065 (SBC and Gateway) ▪ Resolved constraints added to 7.20A.202.112 (153341, 153888, and snapshot)
27276	▪ Ver. 7.20A.202.203 (SBC and Gateway) ▪ Resolved constraint added to 7.20A.202.112 (151502 / 151184)

LTRT	Description
27340	▪ Ver. 7.20A.204.015 (SBC and Gateway) ▪ New feature for 7.20A.150.004 (SBC and Gateway) - Configurable Failed Restarts for Triggering Automatic Recovery ▪ Constraint VI 154386 added to 7.20A.156.009 (SBC and Gateway) ▪ Feature updated for CDR Local Storage Value Changes (7.20A.158.009 - SBC and Gateway) ▪ Note added to feature NGINX for HTTP Proxy Server Configuration (7.20A.200.019 - SBC and Gateway) ▪ Resolved constraint added VI 150950 (7.20A.200.019 - SBC and Gateway) ▪ Resolved constraint added VI 153025 (7.20A.202.112 - SBC and Gateway)
27342	Added RFC 6337 and RFC 6442; Privacy header supported; resolved constraint VI 153631 added.
27343	Updates to Ver. 7.20A.204.015 (SBC and Gateway): 'Advice of Charge Enhancements' feature modified; new feature added - 'New Mediant 9000 Product Offering' (Mediant 9030/9080); Mediant 9030/9080 added to capacity table; VI's 152338 and 146443 added to known constraints; VI's 155065/151078/153846 added to resolved constraint. New feature added 'ENUM Query Enhancement for Call Setup Rules' to 7.20A.156.023 (SBC-Gateway).
27344	Ver. 7.20A.202.307 (MSBR)
27345	Ver. 7.20A.204.108 (SBC and Gateway)
27347	Description of feature "Hitless License Key Installation for HA" updated; resolved constraint Incident #155182 added; Mediant 800C capacity update
27348	Ver. 7.20A.204.127 (SBC and Gateway) Description updated for VI-154025; VI 154437 added to 7.20A.204.108;
27349	Ver. 7.20A.204.128 (SBC and Gateway) VI-150004 added to Ver. 7.20A.200.019; SBC-10110 added to Ver. 7.20A.204.127
27351	Ver. 7.20A.204.132 (SBC and Gateway)
27353	▪ Ver. 7.20A.204.222 (SBC and Gateway) ▪ VI-153524 removed
27355	▪ Ver. 7.20A.204.233 (SBC and Gateway) ▪ Ver. 7.20A.250.003 (SBC and Gateway) ▪ 7.20A.204.222 – resolved constraint added (SBC-9969) ▪ Mediant SE DL360 Gen10 added to Capacity Table ▪ Configuration Tables Capacity updated
27356	▪ Ver. 7.20A.204.337 (SBC and Gateway) ▪ Constraint SBC-11568 / SBC-11603 added to 7.20A.250.003
27357	▪ Ver. 7.20A.250.028 (MSBR) ▪ New features added to Ver. 7.20A.250.003: Mediant CE in Microsoft Azure Environment; Handling Advice of Charge Information in XML Format ▪ Table of Configuration Table Capacity updated for Proxy Address table
27358	▪ Resolved constraint (SBC-9536) added to 7.20A.250.028 (MSBR)
27359	Ver. 7.20A.204.362 (SBC and Gateway)

LTRT	Description
27360	Ver. 7.20A.204.237 (SBC and Gateway) Resolved constraint SBC-10320 added to 7.20A.250.003.
27362	Ver. 7.20A.250.256 (SBC and Gateway) SBC-10819 updated with applicable product
27363	Ver. 7.20A.204.241 and 7.20A.204.433 (SBC and Gateway) Annex A and AB added to G.729.
27365	Ver. 7.20A.204.442 (SBC and Gateway) Resolved constraint SBC-12352 added to Ver. 7.20A.204.443
27366	Ver. 7.20A.250.273 (SBC and Gateway)
27368	Ver. 7.20A.252.011 (SBC and Gateway) Configuration tables capacity updated; Mediant 9030 transcoding capacity updated. SNMP Alarm for No Configured Proxy feature added to 7.20A.158.009
27369	Ver. 7.20A.204.510 (SBC and Gateway) Ver. 7.20A.252.062 (MSBR)
27371	Ver. 7.20A.204.735 (SBC and Gateway) Ver. 7.20A.250.413 (SBC and Gateway)
27372	Ver. 7.20A.252.023 (SBC and Gateway) Resolved constraint SBC-13746 added to Ver. 7.20A.204.735 SIPRec capacity for MSBR updated
27373	Capacity updated: Mediant CE on Azure; Mediant VE on Azure (DS1/2/3_v2); Mediant VE on AWS/EC2 r.4large (SIP sessions).
27376	Ver. 7.20A.252.261 (SBC and Gateway)
27378	Typo in SIP Signaling and Media Capacity section for Mediant 9030.
27379	Max. SIPRec sessions updated for Mediant 500L Gateway & E-SBC
27380	▪ Ver. 7.20A.252.269 (SBC and Gateway) ▪ SBC-14187 constraint added to 7.2 GA ▪ Capacity updated for Mediant 800A/B Gateway & E-SBC SBC Session Capacity per Capabilities (SBC Only) ▪ Opus-WB added to Transcoding Capacity per Profile for Mediant VE SBC with Single MT table
27381	Ver. 7.20A.252.078 (MSBR)
27382	Ver. 7.20A.204.759 (SBC and Gateway)
27383	Ver. 7.20A.204.521 (SBC and Gateway) New feature added to Ver. 7.20A.162.001: Max. RADIUS-Accounting Attributes for CDR Customization SBC with Gateway capacity updated for Mediant 800A/B Gateway & E-SBC
27384	Ver. 7.20A.254.202 (SBC and Gateway)

LTRT	Description
27386	SW revision record table updated; typos
27390	▪ Ver. 7.20A.204.523 (SBC and Gateway) ▪ Resolved constraint VI-150950 (Ver. 7.20A.200.019) updated; resolved constraint re SIPRec and MTC added to Ver. 7.20A.204.759; Voice.AI Gateway removed; new 'MSRP Empty Message Format' parameter added to Ver. 7.20A.254.202; new feature Microsoft Teams License Included in Evaluation License Key added to Ver. 7.20A.254.202; OVR capacity updated for Mediant 800C; Dial Plan rule capacity updated for Mediant 90xx/Mediant SE
27391	Ver. 7.20A.254.026 (MSBR)
27392	▪ Ver. 7.20A.254.375 (SBC and Gateway) ▪ Typo Capacity Table re Mediant VE VMware (4/8 vCPU)
27393	▪ Ver. 7.20A.204.789 (SBC and Gateway) ▪ CDR filename with hostname for local storage feature added to Ver. 7.20A.252.011

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our website at https://online.audiocodes.com/documentation-feedback.

1 Introduction

This document describes the release of Version 7.2. This includes new products, new hardware features, new software features, known constraints, and resolved constraints.

Note:

- Some of the features mentioned in this document are available only if the relevant software License Key has been purchased from AudioCodes and is installed on the device. For a list of available License Keys that can be purchased, please contact your AudioCodes sales representative.
- Open source software may have been added and/or amended. For further information, visit AudioCodes website at https://www.audiocodes.com/services-support/open-source or contact your AudioCodes sales representative.
- Updates to this document may be made due to significant information discovered after the release or too late in the release cycle to be otherwise included in this release documentation. You can check for an updated version on AudioCodes website at https://www.audiocodes.com/library/technical-documents.

1.1 Software Revision Record

This section lists the software versions released in Version 7.2.

Note: The latest software versions can be downloaded from AudioCodes' Services Portal (registered users only) at https://services.audiocodes.com.

1.1.1 SBC and Media Gateway Series

The following table lists the SBC and Media Gateway software versions released in Version 7.2.

Table 1-1: SBC and Media Gateway Software Revision Record

Software Version	Date
Beta Version (7.20A.000.042)	April 2016
7.20A.001	July 2016
7.20A.002	November 2016
7.20A.100	December 2016
7.20A.104.001	March 2017
7.20A.150.004	May 2017
7.20A.106.003	June 2017
7.20A.152.003	June 2017
7.20A.152.009	July 2017
7.20A.154.007	August 2017
7.20A.154.044	September 2017

Software Version	Date
7.20A.154.052	October 2017
7.20A.154.059	October 2017
7.20A.156.009	November 2017
7.20A.156.023	November 2017
7.20A.156.041	December 2017
7.20A.158.009	January 2018
7.20A.158.012	February 2018
7.20A.200.019	February 2018
7.20A.158.035	March 2018
7.20A.162.001	April 2018
7.20A.158.056	April 2018
7.20A.200.550	May 2018
7.20A.162.017	May 2018
7.20A.202.112	June 2018
7.20A.158.065	July 2018
7.20A.202.141	July 2018
7.20A.202.203	17 July 2018
7.20A.204.015	13 September 2018
7.20A.204.108	15 October 2018
7.20A.204.127	31 October 2018
7.20A.204.128	7 November 2018
7.20A.204.132	14 November 2018
7.20A.204.222	6 December 2018
7.20A.204.233	8 January 2019
7.20A.250.003	8 January 2019
7.20A.204.337	16 January 2019
7.20A.204.362	31 January 2019
7.20A.204.237	7 February 2019
7.20A.250.256	18 February 2019
7.20A.204.241	4 March 2019
7.20A.204.433	4 March 2019
7.20A.204.442	19 March 2019
7.20A.250.273	25 March 2019
7.20A.252.011	17 April 2019
7.20A.204.510	30 April 2019
7.20A.204.735	16 May 2019

Software Version	Date
7.20A.250.413	16 May 2019
7.20A.252.023	20 May 2019
7.20A.252.261	3 June 2019
7.20A.252.269	30 June 2019
7.20A.204.759	11 July 2019
7.20A.204.521	28 July 2019
7.20A.254.202	6 August 2019
7.20A.204.523	29 August 2019
7.20A.254.375	16 September 2019
7.20A.204.789	22 September 2019

1.1.2 MSBR Series

The following table lists the MSBR software versions released in Version 7.2.

Table 1-2: MSBR Software Revision Record

Software Version	Date
7.20A.150.004	May 2017
7.20A.154.025	August 2017
7.20A.154.061	November 2017
7.20A.154.078	February 2018
7.20A.200.038	March 2018
7.20A.202.112	June 2018
7.20A.202.307	October 2018
7.20A.250.028	24 January 2019
7.20A.252.062	30 April 2019
7.20A.252.078	1 July 2019
7.20A.254.026	9 September 2019

1.2 Supported Products in Release 7.2

This section lists the products that are supported in this release.

Note:

- Product support and hardware configurations may change without notice. Currently available hardware configurations are listed in AudioCodes Price Book. For further enquiries, please contact your AudioCodes sales representative.
- Figures shown in the tables in this section are maximum values per interface. For available hardware configurations including combinations of supported interfaces, contact your AudioCodes sales representative.

1.2.1 SBC and Media Gateway Series

The following table lists the SBC and Media Gateway products supported in this release.

Table 1-3: SBC and Media Gateway Products Supported in Release 7.2

Product	Telephony Interfaces			Ethernet Interfaces	USB	OSN
	FXS/FXO	BRI	E1/T1			
Mediant 500 Gateway & E-SBC	-	-	1/1	4 GE	2	-
Mediant 500L Gateway & E-SBC	4/4	4	-	4 GE	1	-
Mediant 800B Gateway & E-SBC	12/12	8	2	4 GE / 8 FE	2	√
Mediant 800C Gateway & E-SBC	12/12	8	4	4 GE / 8 FE	2	√
Mediant 1000B Gateway & E-SBC	24/24	20	6/8	7 GE	-	√
MP-1288 Gateways & E-SBC	288/0	-	-	2 GE	1	-
Mediant 2600 E-SBC	-	-	-	8 GE	-	-
Mediant 4000 SBC	-	-	-	8 GE	-	-
Mediant 4000B SBC	-	-	-	8 GE	-	√
Mediant 9030 SBC	-	-	-	12 GE	-	-
Mediant 9080 SBC	-	-	-	12 GE	-	-
Mediant SE SBC	-	-	-	12 GE	-	-
Mediant VE SBC	-	-	-	12 GE	-	-
Mediant CE SBC	-	-	-	12 GE	-	-

1.2.2 MSBR Series

The following table lists the MSBR products supported in this release.

Table 1-4: MSBR Products Supported in Release 7.2

Product	Telephony Interfaces			Ethernet Interfaces	USB	OSN	WAN
	FXS/FXO	BRI	E1/T1				
Mediant 500 MSBR	4/4 (or 8 FXS)	2	1	4 GE	2	-	GbE; Fiber; ADSL2+/VDSL2; SHDSL; 3G Cellular (USB)
Mediant 500L MSBR	4/4	4		4 GE	1	-	GbE; Fiber; ADSL2+/VDSL2; 3G Cellular (USB)
Mediant 800B MSBR	12/12	8	2	4 GE / 8 FE (Optional PoE)	2	√	GbE; Fiber; 4 E1/T1 WAN; ADSL2+/VDSL2; SHDSL; 3G Cellular (USB)
Mediant 800C MSBR	12/12	8	4	4 GE / 8 FE (Optional PoE)	2	√	GbE; Fiber; 4 E1/T1 WAN; ADSL2+/VDSL2; SHDSL; 3G Cellular (USB)

1.3 Documentation Convention for Indicating Products

Throughout this guide, the following terms are used to refer to groups of AudioCodes products for indicating applicability. Where applicability is specific to a product, the name of the product is used.

Table 1-5: Terms Representing Product Groups

Term	Product
Analog	Products with analog interfaces (FXS or FXO): ▪ MP-1288 ▪ Mediant 500L Gateway & E-SBC ▪ Mediant 800 Gateway & E-SBC (Rev. B and C) ▪ Mediant 1000B Gateway & E-SBC ▪ MSBR
Device	All products
Digital	Products with digital PSTN interfaces (e.g., ISDN BRI or PRI): ▪ Mediant 500 Gateway & E-SBC ▪ Mediant 500L Gateway & E-SBC ▪ Mediant 800 Gateway & E-SBC (Rev. B and C) ▪ Mediant 1000B Gateway & E-SBC ▪ MSBR
Mediant 90xx	▪ Mediant 9000 ▪ Mediant 9000 Rev. B ▪ Mediant 9030 ▪ Mediant 9080
Mediant Software	Software-based products: ▪ Mediant SE SBC ▪ Mediant VE SBC ▪ Mediant CE SBC
MSBR	▪ Mediant 500 MSBR ▪ Mediant 500L MSBR ▪ Mediant 800 MSBR (Rev. B and C)

2 Gateways and SBCs

This chapter describes new features, known constraints and resolved constraints relating to Gateway and SBC functionalities.

2.1 Version GA

This section describes new features, known constraints and resolved constraints for the GA version.

2.1.1 New Features

New features introduced in the GA version include the following:

2.1.1.1 New Product - Media Transcoder Device

AudioCodes' Media Transcoder (MT) delivers high-capacity DSP-based transcoding in conjunction with AudioCodes' field-proven SBC product family (currently, supported only by Mediant 9000 SBC) enabled with the Media Transcoding Cluster feature. AudioCodes MT is a modular solution, supporting up to three field-upgradable transcoding modules in a single 1-U chassis. As transcoding needs increase, multiple AudioCodes MT devices can be added to form a cluster configuration giving virtually unlimited scalability along with HA cluster redundancy.

The main hardware specifications of the Media Transcoder include:

- 1U chassis design, suitable for 19-inch rack mounting
- Eight 100/1000Base-T Ethernet ports, supporting 1+1 Ethernet port redundancy
- Dual Power Supply modules, providing power load sharing and AC power redundancy
- Modular scalability from one to up to three MPM12B DSP modules

For more information on the Media Transcoding Cluster feature, see Section 2.1.1.26 on page 43.

2.1.1.2 New GUI for Web-based Management Tool

This feature introduces a new graphical user interface (GUI) for the device's Web-based management tool (Web interface). The new GUI offers the following new features:

- New modern look-&-feel design, making configuration more intuitive and improving user experience.
- Topology view showing a graphical display of the core SIP configuration entities (IP Groups, SIP Interfaces, Media Realms, and Trunk Groups), enabling the administrator to easily build and view the SIP topology.
- Network view showing a graphical display of the core networking entities (IP interfaces, Ethernet Devices, Ethernet Groups, and Physical Ethernet ports), enabling the administrator to easily build and view the main network topology.
- Improved navigation to Web pages, facilitating configuration.
- Indication icons of configured table rows. Navigation pane and tables display icons indicating the number of configured table rows, invalid row configuration, and invalid associations with other table rows.
- Easy access to associated configuration entities while configuring an entity.
- Fewer user clicks to save configuration and reset device.
- Quick access to vital call statistics.
- Search based on strings and IP address.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.3 New CLI Structure

This feature introduces a new structure of the CLI that is more aligned with the hierarchical structure of the navigation tree of the new Web GUI launched in this version. The modified structure allows faster and easier navigation between commands in the CLI. The CLI provides fewer folders, allowing the administrator to access commands with fewer key strokes. Many command names have also been made more concise to eliminate visual "clutter".

The CLI commands are now organized under the following main folders:

- `configure system`: Contains system-related commands (e.g., `clock`, `snmp settings` and `web`)
- `configure network`: Contains IP network-related commands (e.g., `interface`, `dhcp-server` and `nfs`)
- `configure voip`: Contains voice-over-IP related commands (e.g., `ip-group`, `sbc`, `gateway` and `media`)
- `configure troubleshoot`: Contains logging-related commands (e.g., `syslog`, `logging` and `test-call`)

The debugging-related commands are located under the root directory for quick access.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.4 Interworking between SIP and SIP-I Endpoints

This feature provides support for interworking between SIP and SIP-I endpoints for SBC calls. SIP-I is a flavor of the SIP protocol, which carries a message body consisting of the User Part of the ISDN protocol (or ISDN User Part - ISUP) over IP networks. SIP-I endpoints are entities that are connected to the SS7 network, referred to as the ISDN user part (ISUP) domain. The device supports the SIP-I Application-layer signaling protocol, which is a standard for encapsulating a complete copy of the SS7 ISUP message in SIP messages, according to ITU-T Q.1912.5, *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part*.

For the interworking process, the device maps between ISUP data and SIP headers. For example, the E.164 number in the Request-URI of the outgoing SIP INVITE is mapped to the Called Party Number parameter of the IAM message and the From header of the outgoing INVITE is mapped to the Calling Party Number parameter of the IAM message. The ISUP data is included in SIP messages using the Multipurpose Internet Mail Extensions (MIME) body part,

The feature also introduces support for manipulating ISUP data, using the existing Message Manipulations table. For a complete description of the ISUP manipulation syntax, refer to the *SIP Message Manipulation Reference Guide*.

To support the feature, the following new parameter has been added:

ISUP Body Handling sbc-isup-body-handling [IpProfile_SBCISUPBodyHandling]	Defines the handling of ISUP data for interworking between SIP and SIP-I. ▪ [0] Transparent = (Default) ISUP data is passed transparently (as is) between endpoints (SIP-I to SIP-I calls). ▪ [1] Remove = Delete the ISUP body from the INVITE message. ▪ [2] Create = Adds ISUP body to outgoing INVITE message.

Note: For more information on the feature, please contact your AudioCodes sales representative.

Applicable Products: All.

Applicable Application: SBC.

2.1.1.5 Maximum Call Duration per Gateway and SBC Calls

This feature provides support for configuring the maximum call duration for SBC and Gateway calls. Up until this release, maximum call duration could only be configured globally and applied to all calls for both applications—Gateway and SBC—using the MaxCallDuration parameter (which is now obsolete).

The feature allows the administrator to configure maximum call duration for the following:

- SBC calls:
  - All SBC calls (i.e., globally)
  - Specific SBC calls (using IP Profiles)
- Gateway calls: All Gateway calls (globally) only

The feature is useful for ensuring that calls are properly terminated, making device resources available for new calls.

To support the feature, the following new parameters have been added:

SBC Max Call Duration `sbc-mx-call-duration` [SBCMaxCallDuration]	Defines the maximum duration (in minutes) for each SBC call (global). If the duration is reached, the device terminates the call. The valid range is 0 to 35,791, where 0 is unlimited duration. The default is 0. Note: The parameter replaces the MaxCallDuration parameter.
Max Call Duration `sbc-max-call-duration` [IpProfile_SBCMaxCallDuration]	Defines the maximum duration (in minutes) for each SBC call that is associated with the IP Profile. If the duration is reached, the device terminates the call. The valid range is 0 to 35,791, where 0 is unlimited duration. The default is the value configured for the SBCMaxCallDuration parameter.
GW Max Call Duration `gw-mx-call-duration` [GWMaxCallDuration]	Defines the maximum duration (in minutes) for each Gateway call (global). If the duration is reached, the device terminates the call. The valid range is 0 to 35,791, where 0 is unlimited duration. The default is 0. Note: The parameter replaces the MaxCallDuration parameter.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.6 Protection against Known Malicious Attacks

This feature provides support for protecting the device against malicious attacks on SBC calls using a Malicious Signature database. The feature allows the administrator to configure a database of malicious signature patterns which identify specific scanning tools used by attackers to search for a SIP server in a network. The feature identifies and protects against SIP (Layer 5) threats by examining any new inbound SIP dialog message. Once the device identifies an attack based on the configured malicious signature patterns, it marks the SIP message as invalid and discards it or alternatively, rejects it with a SIP response (by default 400).

The malicious signatures are based on the SIP User-Agent header and employ the same syntax used for Message Manipulation rules. For example:

- Malicious signature is defined as follows for a malicious scanner:

```
header.user-agent.content prefix "malicious scanner"
```

- Malicious signature is defined as follows for the scanning tool "sip-scan":

```
Header.User-Agent.content prefix 'sip-scan'
```

The protection applies only to new dialogs (e.g., INVITE messages) and unauthenticated dialogs. The Malicious Signature database does not apply to the following:

- Calls from IP Groups where classification is by Proxy Set.
- In-dialog SIP sessions (such as refresh REGISTER requests, re-INVITE etc.)
- Calls from users that are registered with the device.

By default, the device is installed with a list of known attackers, called the Malicious Signature Database. The Malicious Signature database is presented in table format. The administrator can add, edit or delete entries. As a safety mechanism, if all entries are deleted and the device is subsequently reset, the table is populated again with all the signatures. In addition, the administrator can export or import a Malicious Signature database through HTTP, HTTPS, or TFTP.

The feature is enabled by a new global parameter (see below). The existing Message Policy table provides an additional default Message Policy rule for the Malicious Signature database ("MaliciousSignatureDBProtection"). To apply the Malicious Signature database to calls, the administrator needs to associate this default Message Policy rule to an SBC SIP Interface in the existing SIP Interface table.

The Malicious Signature database can also be used with the existing Intrusion Detection System (IDS) feature. A new IDS reason has been added to denote Malicious Signature detections (Signature DB invalid). This allows the administrator to enable SNMP alarm generation ("Dialog establishment failure") if any signature is detected by the device.

To support the feature, the following new parameters have been added:

Malicious Signature Table [MaliciousSignatureDB]	Defines up to 30 malicious signature patterns (rows). [MaliciousSignatureDB] FORMAT MaliciousSignatureDB_Index = MaliciousSignatureDB_Name, MaliciousSignatureDB_Pattern; [\MaliciousSignatureDB]
Message Policy Table [MessagePolicy_UseMaliciousSignatureDB]	New parameter: Malicious Signature Database [MessagePolicy_UseMaliciousSignatureDB] = Enables the use of the Malicious Signature database for SIP Interfaces that are assigned the Message Policy.
`configure voip > sbc malicious-signature-database <export-csv-to | import-csv-from> <URL>`	Exports/imports a Malicious Signature database file (in *.csv format) to/from a server (HTTP, HTTPS, or TFTP).

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000B; Mediant 2600; Mediant 4000; Mediant 9000; Mediant SE/VE.

Applicable Application: SBC.

2.1.1.7 Block SIP Requests from Registered Users when Address Different

This feature provides support for blocking (rejecting) SIP dialog-initiating requests (such as INVITE messages) from a user that is registered with the device, but where the source address (IP address and/or port) and transport type (e.g., UDP) is different to that registered for the user (during the REGISTER message process). When the device rejects a request, it reports the rejection (Classification failure) through the already supported Intrusion Detection System (IDS), by sending an SNMP trap.

The device can verify whether the IP address and port are different only if the transport protocol is UDP; otherwise, the device verifies only the IP address. The verification is performed before any of the device's call handling processes (i.e., Classification, Manipulation and Routing) and applies only to User-type IP Groups.

Note that the feature does not apply to registration refreshes. These requests are accepted even if their source address is different to that registered for the user.

To support the feature, the following existing parameters have been modified:

User Security Mode	Parameter name and optional values modified:

[SRD_BlockUnRegUsers]	Defines the blocking (reject) policy of incoming SIP dialog-initiating requests from users (except REGISTER requests). When the device rejects a request, it sends a SIP 500 "Server Internal Error" response to the user. ▪ [0] Accept All = (Default) Accepts requests from registered and unregistered users. ▪ [1] Accept Registered Users = Accepts requests from registered users only and rejects requests from users not registered with the device. ▪ [2] Accept Registered Users from Same Source = Accepts requests only from registered users whose source address is the same as that registered with the device. All other requests are rejected.
User Security Mode [SIPInterface_BlockUnRegUsers]	Parameter name and optional values modified: Defines the blocking (reject) policy of incoming SIP dialog-initiating requests from users (except REGISTER requests). When the device rejects a request, it sends a SIP 500 "Server Internal Error" response to the user. ▪ [-1] Not Configured (default) ▪ [0] Accept All = Accepts requests from registered and unregistered users. ▪ [1] Accept Registered Users = Accepts requests from registered users only and rejects requests from users not registered with the device. ▪ [2] Accept Registered Users from Same Source = Accepts requests only from registered users whose source address is the same as that registered with the device. All other requests are rejected.

Applicable Products: All.

Applicable Application: SBC.

2.1.1.8 Enhanced Dialog Classification Based on Proxy Set

This feature provides support for enhanced classification of incoming SIP dialogs to IP Groups, based on Proxy Set when multiple Proxy Sets are configured with the same IP address. For more information, refer to the *User's Manual*.

Applicable Products: All.

Applicable Application: SBC.

2.1.1.9 Wildcard Denoting 18x Responses in Message Manipulation Rules

The feature provides support for using the 'x' wildcard in SIP message manipulation rules to denote all SIP 18x responses (e.g., 180, 181, 182 and 183). The wildcard is used in the 'Message Type' field, which defines the type of message to which the manipulation is applied. For example, to configure a rule that applies to any SIP 18x in response to an INVITE message, the following syntax is used in the 'Message Type' field:

```
invite.response.18x
```

Up until this release, the exact 18x response (e.g., 180, 181, 182 or 183) had to be specified. For example, if the administrator wanted to apply the same message manipulation to all 18x responses, multiple rules with the same syntax except for the specified 18x response had to be configured.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.10 Increase in Maximum SIP Message Size

This feature provides support for configuring the existing parameter, MaxSIPMessageLength to up to 100 KB. The device rejects SIP messages exceeding the configured size. Up until this release, the maximum SIP message size could be configured to 50 KB.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.11 IP Group Keep-Alive Connectivity Status Indication

This feature provides support for displaying the connectivity status of Server-type IP Groups. As the Proxy Set defines the actual address of the IP Group, the connectivity check (or keep-alive) by the device is done to this address. Note that for the feature to be relevant, the keep-alive mechanism must be enabled for the associated Proxy Set (using the existing parameter, ProxySet_EnableProxyKeepAlive).

The connectivity status is indicated as follows:

- Topology View: The status is displayed as a color-coded icon in the IP Group element:
  - Green: Keep-alive is successful (i.e., connectivity with IP Group). Note that if the device rejects calls destined to this IP Group due to low QoE (e.g., low MOS), the indication still appears green.
  - Red: Keep-alive failure (i.e., no connectivity with IP Group).

  An example of these icons is shown below:

- IP Group table: The status is displayed in the new read-only field, 'Proxy Set Connectivity' (IPGroup_ProxySetConnectivity ini parameter or `show voip proxy sets status` CLI command):
  - "NA": Functionality is not applicable in the following cases:
    - If Server-type IP Group and the Proxy Keep-Alive mechanism is disabled
    - If User-type IP Group
  - "Not Connected": Keep-alive failure (i.e., no connectivity with IP Group)
  - "Connected": Keep-alive is successful (i.e., connectivity with IP Group)

Applicable Products: All.
Applicable Application: Gateway and SBC.

2.1.1.12 Enhanced Configuration of Allowed Coder Groups

This feature provides support for enhanced configuration design of Allowed Audio Coder Groups and Allowed Video Coder Groups:

- Allowed Audio Coder Groups: User-defined coders can now be configured through the Web interface. Up until now, it could only be configured through ini file and CLI. In addition, configuration now consists of two tables – parent and child. The parent table configures the ID and name; the child configures the coders of the selected group.
- Allowed Video Coder Groups: Now configurable through the Web interface. Up until this release, Allowed Video Coders Groups could only be configured through ini file and CLI.

Allowed Audio Coders Groups `configure voip > coders-and-profiles allowed-audio-coders-groups` [AllowedAudioCodersGroups]	Parent table that defines the names of the Allowed Audio Coder Groups. [AllowedAudioCodersGroups] FORMAT AllowedAudioCodersGroups_Index = AllowedAudioCodersGroups_Name; [\AllowedAudioCodersGroups]

Allowed Audio Coders `coders-and-profiles allowed-audio-coders <group index/coder index>` [AllowedAudioCoders]	Child table of the Allowed Audio Coders Groups that defines the audio coders of the group. [AllowedAudioCoders] FORMAT AllowedAudioCoders_Index = AllowedAudioCoders_AllowedAudioCodersGroupName, AllowedAudioCoders_AllowedAudioCodersIndex, AllowedAudioCoders_CoderID, AllowedAudioCoders_UserDefineCoder; [\AllowedAudioCoders]
Allowed Video Coders Groups `configure voip > coders-and-profiles allowed-video-coders-groups` [AllowedVideoCodersGroups]	Parent table that defines the names of the Allowed Video Coder Groups. [AllowedVideoCodersGroups] FORMAT AllowedVideoCodersGroups_Index = AllowedVideoCodersGroups_Name; [\AllowedVideoCodersGroups]
Allowed Video Coders `coders-and-profiles allowed-video-coders <group index/coder index>` [AllowedVideoCoders]	Child table of the Allowed Video Coders Groups that defines the video coders of the group. [AllowedVideoCoders] FORMAT AllowedVideoCoders_Index = AllowedVideoCoders_AllowedVideoCodersGroupName, AllowedVideoCoders_AllowedVideoCodersIndex, AllowedVideoCoders_UserDefineCoder; [\AllowedVideoCoders]

Applicable Products: All.

Applicable Application: SBC.

2.1.1.13 Enhanced Audio Coder Groups Configuration

The feature provides the following enhancements:

- The Coders table is obsolete and has been replaced by the existing Coder Groups table (formerly known as Coder Group Settings table), facilitating configuration.
- Coder Group configuration through ini file is now done using two ini file tables:
  - AudioCodersGroups: Defines the Coder Group name/index
  - AudioCoders: Defines the coders for the Coder Groups
- Enumerations are now used for coder names, packetization times, and rate.
- Deletion of Coder Groups through the Web interface is now possible by the Delete Group button, which when clicked, deletes the currently displayed Coder Group. Up until this release, to delete a Coder Group, the administrator had to remove all its coders one by one.

Applicable Products: All.

Applicable Application: All.

2.1.1.14 Enhanced Dial Plan Tagging

This feature provides the following Dial Plan Tagging enhancements:

- CDR fields for source and destination dial plan tags (see Section 2.1.1.18 on page 40)
- Exporting and importing Dial Plan rules in CSV file format to a local folder on the PC running the Web client, through the Web interface (already supported through CLI)
- Increased capacity:
  - Max. Dial Plans:
    - Mediant 2600/4000: 25
    - Mediant VE: 50

- Others: 10
- Max. dial plan rules:
  - Mediant 2600/4000: 10,000
  - Mediant VE (< 16G): 2,000
  - Mediant VE (> 16G incl.): 20,000
  - Others: 2,000

Applicable Products: All.

Applicable Application: SBC.

2.1.1.15 Increase in Maximum Network Interfaces

This feature provides support for an increase in the maximum number of IP network interfaces that can be configured in the IP Interfaces table (InterfaceTable). The increase is from 100 to 1024 network interfaces. The maximum capacity of Media Realms and Ethernet Devices that can be configured in the Media Realms table (CpMediaRealm) and Ethernet Devices table (DeviceTable) were also increased to 1,024.

Applicable Products: Mediant 9000; Mediant VE/SE.

Applicable Application: SBC.

2.1.1.16 CDR Local Storage for Gateway Calls

This feature provides support for CDR local storage for Gateway calls. Up until now, CDR local storage was supported only for SBC calls. Configuration for CDR local storage is the same as SBC (CDRLocalMaxFileSize, CDRLocalMaxNumOfFiles, and CDRLocalInterval) and Logging Filters table for selectively enabling the feature.

Due to the feature, customization of locally stored Gateway CDRs is also supported. As a result, the new optional value Local Storage Gateway [9] has been added to the 'CDR Type' (GWCDRFormat_CDRType) parameter in the Gateway CDR Format table.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000B.

Applicable Application: Gateway.

2.1.1.17 Historical CDRs Display for SBC Calls

This feature provides support for displaying historical CDRs (last 4,096 CDRs) for SBC calls in the device's management interfaces. Up until now, historical CDRs were displayed for Gateway calls only.

To support the feature, the new table, SBC CDR History has been added:

- Web: Monitor menu > Monitor tab > VoIP Status folder > SBC CDR History
- CLI: `show voip calls history sbc`

The table includes the following CDR fields: Call End Time, IP Group, Caller, Callee, Direction, Remote IP, Duration, Termination Reason, and Session.

The name of the existing CDR History table for Gateway calls has been changed to Gateway CDR History:

- Web: Monitor menu > Monitor tab > VoIP Status folder > GW CDR History
- CLI: `show voip calls history gw`

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.18 New CDR Fields

This feature introduces the following new CDR fields:

- **LegId:** Identifies each leg by a unique ID number within a specific call session. The

field is assigned a unique number for each leg in the call session. This unique identification enhances the ability of applications such as AudioCodes One Voice Operations Center to analyze call data according to various segments in the call session.

- **Trigger:** Describes the reason of the call. The field name can be customized, using the Gateway CDR Format and SBC CDR Format tables. The tables show the field as "Trigger" (ini file enumeration 439) in the 'Field Type' field. The field can have one of the following values:
  - "Normal": regular call
  - "Refer": call as a result of call transfer
  - "AltRoute": call as a result of alternative routing
  - "Forward": call as a result of forwarded call
  - "Reroute": call re-routed due to a voice issue (e.g., broken RTP connection)
  - "Forking": call as a result of call forking
- **SrcDialPlanTags / DestDialPlanTags:** Indicate Dial Plan tags (source and destination) used for the call (if the Dial Plan Tagging feature is implemented). The field name can be customized using the SBC CDR Format table. The table shows the field as "Source Dial Plan Tags" (ini file enumeration 816) and "Destination Dial Plan Tags" (ini file enumeration 817) in the 'Field Type' field.

Note that the ini file enumerations of the optional values in the 'Field Type' field of the Gateway and SBC CDR Format tables have changed.

Applicable Products: All.
Applicable Application: Gateway and SBC.

2.1.1.19 Maximum RADIUS Requests

The feature provides support for an increase in the maximum number of RADIUS requests that the device can send simultaneously to a RADIUS server. Up until this release, the device could send only up to 254 concurrent RADIUS requests (RADIUS Accounting and Authentication together).

This feature provides the following support:

- All Products: Up to 201 concurrent RADIUS requests **per** RADIUS service type (Accounting or Authentication) and per RADIUS server (up to three servers per service type).
- Mediant 2600, Mediant 4000, Mediant 9000 and Mediant Software Only: Up to 201 concurrent RADIUS requests per RADIUS service type (Accounting or Authentication), per RADIUS server and per local port, which has been increased from one port to the following:
  - Mediant 2600/4000: two local ports
  - Mediant 9000/SW: four local ports
  - For all other products: only one port is supported.

  For example, for Mediant 4000, 402 (201 * 2) concurrent RADIUS requests can be sent for Authentication and 402 (201 * 2) for Accounting. These numbers are per RADIUS server.

Applicable Products: All.

Applicable Applications: SBC and Gateway.

2.1.1.20 Increase in Maximum Network ACL Rules

This feature provides support for an increase in the maximum number of network Access Control List (ACL) or firewall rules that can be configured in the Firewall table (AccessList). The increase is from 50 to 500 rules.

Applicable Products: Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.
Applicable Application: SBC.

2.1.1.21 Enhanced TLS Certificate Support

This feature provides support for the following TLS enhancements:

- Private Key size (in bits): The private key size can now be configured to 4096 bits, which provides very high strength key. Up until this release, the key size options were 512, 768, 1024, and 2048. The private key size is configured by the existing parameter, Private Key Size (Web - TLS Contexts page > TLS Context Certificate link; CLI - configure network > tls > private-key generate).
- Signature algorithm for certificates: The signature algorithm can now be configured to SHA-256 or SHA-512. Up until this release, the device supported only the SHA-1 algorithm (default). The algorithm is configured by the new parameter, Signature Algorithm (Web - TLS Contexts page > TLS Context Certificate link; CLI - configure network > tls > certificate signature-algorithm).
- Enabling validation of extensions (keyUsage and extentedKeyUsage) of peer certificates is now configured per TLS Context. Up until this release, it was configured globally. To support the feature, the global parameter, RequireStrictCert has been replaced by the new TLS Context table parameter, TLSContexts_RequireStrictCert.
- Configuring the TLS Server Certificate Expiry Check feature per TLS Context. Up until this release, it was configured globally for all TLS Contexts. (No change in parameters.)

Applicable Products: All.
Applicable Application: Gateway and SBC.

2.1.1.22 TLS Certificate Verification

This feature provides a change in support for verifying the address in the TLS certificate received from a Server-type IP Group whose Proxy Set is configured as an FQDN. Up until now, the device verified that the DNS-resolved IP address of the FQDN matched the IP address in the certificate. Now, the device verifies that the FQDN of the Proxy Set matches the FQDN in the certificate. The feature is enabled by the existing parameter, PeerhostNameVerificationMode.

Applicable Products: All.

Applicable Application: SBC.

2.1.1.23 Disable Reuse of TLS Connections

This feature provides support for disabling the use of the same TLS connection for new SIP requests between the device and a SIP user agent (UA). Up until this release, the device always used the same TLS connection (successful handshake) that was established in the initial SIP dialog request, for subsequent requests (e.g., INVITE or REGISTER) sent to the UA. The feature is supported by the existing parameter, EnableTCPConnectionReuse, which up until this release, was applicable only to TCP.

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.1.24 UDP Port Spacing by Four

This feature provides support for local UDP port allocation in "jumps" (*spacing*) of four. Up until this release, UDP port spacing could be configured to 5 or 10.

The device allocates ports for a media channel (leg) from a pool of UDP ports. The pool starts from a port configured by the existing parameter, BaseUDPPort and each leg is assigned several consecutive ports for its usage (e.g. RTP, RTCP, and T.38). The spacing between ports per leg is configured by the existing parameter, UdpPortSpacing. For example, if port

spacing is configured to four and BaseUDPPort to 6000, the allocated ports are 6000 for the first leg, 6004 for the second leg, 6008 for the third leg, and so on.

(For all other products, UDP port spacing is 10 as supported in previous releases).

Applicable Products: Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.
Applicable Application: SBC.

2.1.1.25 Sending of Silence RTP Packets to SIP Trunks

The feature provides support for the device to interoperate with SIP entities (e.g., SIP Trunks) that wait for the first incoming packet before sending RTP (e.g., early media used for ringback tone and IVR) during media negotiation. The feature enables the device to generate "silence" RTP packets to the SIP entity upon receipt of a SIP response (183 with SDP) from the SIP entity. In other words, these packets serve as the first incoming packets for the SIP entity. The device stops sending the silence packets when it receives RTP packets from the peer side (which it then forwards to the SIP entity).

Note: To generate silence packets, DSP resources are required (except for calls using G.711).

Generate RTP `sbc-generate-rtp` [IPProfile_SBCGenerateRTP]	Enables generation of silence RTP packets until audio RTP packets are detected. ▪ [0] None (Default) = No silence packets are generated. ▪ [1] Until RTP Detected = Silence packets are generated

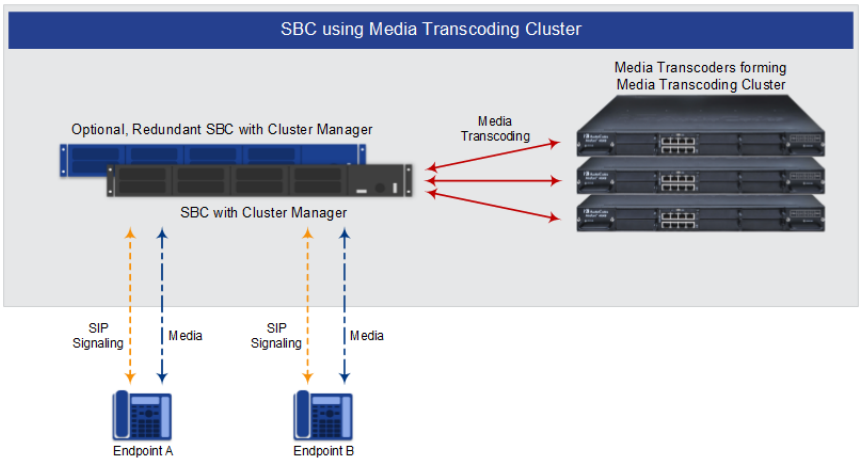
Applicable Products: All.

Applicable Application: SBC.

2.1.1.26 Media Transcoding Cluster Feature

The feature provides support for the SBC device (Mediant 9000) to use an external source of DSP resources for media-related features requiring DSPs, for example, vocodec transcoding, fax transcoding, and DTMF detection. The external farm (*cluster*) of DSP resources is provided by AudioCodes transcoding devices (up to six), called *Media Transcoders*. The SBC device itself functions as the cluster manager and does not perform any transcoding (does not utilize any of its local DSP resources). The Media Transcoders provide only DSP functionality (i.e., no SIP routing functionalities) and a few system functionalities such as debugging through Syslog. The Media Transcoders are "hidden" from the endpoints being serviced by the device. The Media Transcoding Cluster feature is a licensed feature, requiring the SBC device to be installed with a suitable License Key.

The device with the Cluster Manager functionality can still operate as a High-Availability (HA) system. If a switchover occurs, transcoding sessions handled by the Media Transcoding Cluster are maintained.



The main benefit of the Media Transcoding Cluster feature is scalability. The Media Transcoder doesn't require licensing of its transcoding resources and allows utilization of all its DSP resources. However, the maximum possible transcoding capacity by the SBC device is according to the License Key of the SBC device, regardless of the number of deployed Media Transcoders.

After initial configuration of the Media Transcoders through their Web interfaces, subsequent management is through the device's Web interface. The Cluster Manager running on the SBC device can perform various actions on the Media Transcoders such as software upgrade, resetting, and locking (to stop allocating transcoding sessions).

The Media Transcoding Cluster feature provides load-sharing and cluster redundancy between multiple Media Transcoders. Load sharing attempts to distribute the transcoding sessions load between the Media Transcoders. For cluster redundancy, the following modes can be configured:

- HA (default): The Cluster Manager guarantees that in case of a failure in a Media Transcoder, sufficient DSP resources are available on other Media Transcoders to take over the active transcoding sessions of the failed Media Transcoder.
- Best Effort: The Cluster Manager allocates sessions for transcoding to the Media Transcoder without guaranteeing availability of DSP resources on other Media Transcoders should the Media Transcoder fail. Therefore, Media Transcoders utilize all their DSP resources, if required.

The following SNMP alarms have been added for the Media Transcoding Cluster feature:

- AcMtcmClusterHaAlarm: Cluster HA usage exceeds 100% (insufficient DSP resources available on other Media Transcoders to take over active transcoding sessions of a failed Media Transcoder).
- acMtceNetworkFailureAlarm: Connectivity failure between Media Transcoder and Cluster Manager.
- acMtceSwUpgradeFailureAlarm: Software upgrade or Auxiliary file load failure on Media Transcoder.
- acMtceHwTemperatureFailureAlarm: Media Transcoder chassis temperature reaches critical threshold.
- acMtceHwFanTrayFailureAlarm: Media Transcoder Fan Tray module failure.
- acMtcePsuFailureAlarm: Media Transcoder Power Supply module failure.

Note:

- A Media Transcoding Cluster cannot be shared by multiple devices.
- Each Ethernet port on the SBC device associated with the cluster network interface ("Cluster-Media-Control"), communicates with a single Media Transcoder and supports up to 5,000 media transcoding sessions.

Cluster Manager Management Interface	
Cluster Manager Functionality configure network > mtc settings > enable-mtc-sbc [EnableMtcSbc]	Enables the Cluster Manager feature.
MTC Redundancy Mode [MtcRedundancyMode]	Defines the redundancy mode for the Media Transcoding Cluster. ▪ HA Mode (Default) ▪ Best Effort
Application Type [InterfaceTable_ApplicationTypes]	New option: [23] Cluster Media + Control = IP interface for interfacing between the Cluster Manager and Media Transcoders.
MTC Graceful Timeout configure network > mtc settings > graceful-timeout [MtcGracefulTimeout]	Defines the graceful period (in seconds).

Media Transcoders Table configure network > mtc entity [MtcEntities]	Defines Media Transcoders associated with the Cluster Manager.
Transcoding Cluster Log	Displays logged activities of Media Transcoders and Cluster Managers.
Media Transcoders Management Interface	
Cluster Manager IP Address [ClusterManagerIpAddress]	Defines the Cluster Manager by IP address of the corresponding cluster interface (Cluster Media + Control network interface).

Applicable Products: Mediant 9000.

Applicable Application: SBC.

2.1.1.27 New Quality of Service PMs and Alarms

The feature provides support for new quality-of-service performance monitoring (PM) call metrics that can be calculated by the device. The metrics measure network quality and call success rates and are calculated globally, per SRD and per IP Group.

- Answer-seizure ratio (ASR): The number (in percentage) of answered calls (i.e. number of seizures resulting in an answer signal) out of the total number of attempted calls (seizures). The metric is calculated for the outgoing call leg. Note that forwarded calls are not considered in the calculation. The PMs related to the metric include:
  - PM_gwSBCASR: ASR for all (global) entities (i.e., all IP Groups and SRDs)
  - PM_gwSBCIPGroupASR: ASR per IP Group
  - PM_gwSBCSRDASR: ASR per SRD
- Network Effectiveness Ratio (NER): The number (in percentage) of successfully connected calls out of the total number of attempted calls (seizures). The metric measures the ability of the network to deliver a call to the called terminal. In addition to answered calls, the following response codes are regarded as successfully connected calls: 408 (Request Timeout), 480 (Temporarily Unavailable), and 486 (Busy Here). The metric is calculated for the outgoing call leg. Note that forwarded calls are not considered in the calculation. The PMs related to the metric include:
  - PM_gwSBCNER: NER for all (global) entities (i.e., all IP Groups and SRDs)
  - PM_gwSBCIPGroupNER: NER per IP Group
  - PM_gwSBCSRDNER: NER per SRD
- Average Call Duration (ACD): The ACD plus the session disconnect time (SDD) is the time from when the SIP 200 OK is received to when the SIP Bye message is sent. The metric is calculated for both the incoming and outgoing call legs. The PMs related to the metric include:
  - PM_gwSBCACD: ACD for all (global) entities (i.e., all IP Groups and SRDs)
  - PM_gwSBCIPGroupACD: ACD per IP Group
  - PM_gwSBCSRDACD: ACD per SRD

Minor and major thresholds can be configured per metric (in the new table, Performance Profile table - see below) that if crossed, minor and major severity alarms are generated. The following new SNMP alarms are supported:

- acASRThresholdAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.111): The alarm is raised when the configured ASR minor and major thresholds are crossed.
- AcNERThresholdAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.113): The alarm is raised when the configured NER minor and major thresholds are crossed.
- acACDThresholdAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.112): The alarm is raised when the configured ACD minor and major thresholds are crossed.

To support the feature, the following new table has been added:

Performance Profile table `configure system > performance-profile` [PerformanceProfile]	Defines alarm thresholds per metric (ASR, ACD and NER). [PerformanceProfile] FORMAT PerformanceProfile_Index = PerformanceProfile_Entity, PerformanceProfile_IPGroupName, PerformanceProfile_SRDName, PerformanceProfile_PMType, PerformanceProfile_MinorThreshold, PerformanceProfile_MajorThreshold, PerformanceProfile_Hysteresis, PerformanceProfile_MinimumSample, PerformanceProfile_WindowSize; [\PerformanceProfile]

Applicable Products: All.

Applicable Application: SBC.

2.1.1.28 Actions upon Poor Voice Quality Detections

The feature supports configuration of actions that must be performed if poor quality of experience is detected. Configuration is based on Quality of Service rules, using the new Quality of Service Rules table. The following actions can be performed:

- Reject calls to an IP Group for a user-defined duration if a user-defined threshold (major or minor) of a specified metric is crossed. The metric can be voice quality (i.e., MOS), bandwidth (supported in the previous release), ASR, NER, or ACD.

  When the device rejects calls to an IP Group based on a QoS rule, the device raises the new SNMP alarm, acIpGroupNoRouteAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.114).

  When the device rejects a call due to an ASR, NER or ACD threshold crossing, it sends the new SIP response, 850 (Signaling Limits Exceeded). This SIP response code has been added to the Alternative Routing Reasons table (SBCAlternativeRoutingReasons). If it is configured and the device rejects a call due to threshold crossing, it searches in the IP-to-IP Routing table for an alternative routing rule.
- Use an alternative IP Profile for the IP Group upon threshold crossings of voice quality or bandwidth. The alternative IP Profile can be used:
  - For all new calls: If poor voice quality or bandwidth threshold is crossed, the alternative IP Profile is used for all **new** calls. All the parameters of the alternative IP Profile can be configured.

As a result of the feature, the MediaEnhancementProfile and MediaEnhancementRules tables are now obsolete.

Quality of Service Rules Table `configure voip > qoe quality-of-service-rules` [QualityOfServiceRules]	Defines Quality of Service rules. [QualityOfServiceRules] FORMAT QualityOfServiceRules_Index = QualityOfServiceRules_IPGroupName, QualityOfServiceRules_RuleMetric, QualityOfServiceRules_Severity, QualityOfServiceRules_RuleAction, QualityOfServiceRules_CallsRejectDuration, QualityOfServiceRules_AltIPProfileName; [\QualityOfServiceRules]

Applicable Products: All.

Applicable Application: SBC.

2.1.1.29 Bitrate Configuration for SILK and Opus Coders

The feature provides support for configuring the bitrate of the Opus coder. In addition, the default of the existing SilkMaxAverageBitRate parameter, which configures the bitrate for the SILK coder has changed to 50,000.

Opus Max Average Bitrate `configure voip > sip-definition settings > opus-max-avg-bitrate` [OpusMaxAverageBitRate]	Defines the maximum average bit rate (bps) for the Opus coder. The valid value range is 6000 to 50,000. The default is 50,000.

Applicable Products: All.

Applicable Application: SBC.

2.1.1.30 Core Dump File Deletion

This feature provides support for deleting the core dump file from the device's flash memory through CLI. As supported in the previous release, the core dump file is created by the device upon device crash (enabled by the EnableCoreDump parameter) and is a copy of the memory image of the device at the time of the crash.

To support the feature, the following new command has been added under the root CLI directory (enable mode):

```
# clear debug-file
```

Applicable Products: All.

Applicable Application: Gateway and SBC.

2.1.2 Known Constraints

This chapter lists known constraints in Release 7.2.

Table 2-1: Known Constraints in Release 7.2

Incident	Description	Status
-	Maximum SRTP-RTP sessions with voice transcoding is limited to 4,600 sessions when the SRTP leg uses a non-compressed vocoder (G.711). **Applicable Products:** Mediant 4000B.	-
134449	RADIUS-based authentication of SIP users and RADIUS-based authentication of login username and password for management users are currently not supported. **Applicable Products:** Mediant 2600; Mediant 4000.	Resolved in Version 7.20A.100 (See Section 2.4.3)
-	The SIPRec feature is not supported when the Media Transcoding Cluster feature is used. **Applicable Products:** Mediant 9000.	Resolved in Ver. 7.20A.204.759 (See Section 2.46.1)
132977	To upgrade from software version 7.0 to 7.2, the device must first be upgraded to the latest 7.0 version (later than 7.00A.058.002) and only then to version 7.2. **Applicable Products:** Mediant 2600; Mediant 4000; Mediant 9000; Mediant SE/VE.	Resolved in Version 7.20A.100 (See Section 2.4.3)
133943	SRTP with ARIA encryption is not supported for SBC sessions. **Applicable Products:** All.	-
-	ARM is not supported. **Applicable Products:** All.	Resolved in Version 7.20A.100 (See Section 2.4.3)
131889	When importing a Dial Plan file (*.csv file), it is recommended to configure the SyslogDebugLevel parameter to **No Debug**. **Applicable Products:** All.	Resolved in Version 7.20A.100 (See Section 2.4.3)
116756	The device interworks with devices that support RTP bundling. However, it does not support receipt of bundled multimedia sessions on the same port and instead, it uses different ports for each media type (audio and video). By default, the device removes all bundle-related attributes ('a=group:BUNDLE' and 'a=ssrc') from the SDP offer and answer. **Applicable Products:** All.	-
-	CLI scripts used in Version 6.8 are not fully supported and need to be modified in order to be fully compatible in Version 7.2. **Applicable Products:** All.	-
-	Downgrade from Version 7.2 to a previous software version only works if the device was upgraded to Version 7.2 and no configuration changes were done after the upgrade. **Applicable Products:** All.	-

Incident	Description	Status
-	The combination of SBC direct media and termination features such as the handling of 3xx, REFER, and INVITE with Replaces is supported only if all SIP user agents support INVITE/re-INVITE without SDP, and terminations of semi-attendant transfer and INVITE with Replaces during call ringing is not supported with direct media. **Applicable Products:** All.	-
-	SBC Delayed SDP offer is supported only by devices that support DSP transcoding. **Applicable Products:** All.	-
-	High Availability (HA) for One-Voice Resiliency (OVR) is not fully supported (signaling may not function correctly in certain scenarios). **Applicable Products:** HA-Supporting Devices.	Resolved in Version 7.20A.150 (See Section 2.7.3)
-	High Availability (HA) for WebRTC is not fully supported (signaling may not function correctly in certain scenarios). **Applicable Products:** HA-Supporting Devices.	-
-	The SBC User Info table limits the maximum number of users that can be configured (half of the maximum per device). **Applicable Products:** All.	-
-	Out-of-dialog SIP REFER message for SBC calls is forwarded transparently; the subsequent NOTIFY message is not fully supported. **Applicable Products:** All.	-
-	Transrating of G.711, G.726, and G.729 for SBC calls from packetization time (ptime) 100/120 msec to 10/30/50 msec is not supported. **Applicable Products:** Mediant 1000B.	-
-	When SBC termination features are used so that the device handles them locally (i.e., 'Remote Can Play Ringback', 'Play Held Tone', and 'Play RBT To Transferee'), Extension Coders Group ID must be configured, even if only one coder is used. This is especially relevant for the RBT to transferee feature. **Applicable Products:** All.	-
-	Ring to Hunt Group feature does not function when early media is used. **Applicable Products:** Mediant 8xx.	-
-	For the Tel-to-IP Call Forking feature (supported by the Gateway application), if a domain name is used as the destination in the Tel to IP Routing table, the maximum number of resolved IP addresses supported by the device's internal DNS that the call can be forked to is three (even if four IP addresses are defined for the domain name). **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B.	-

Incident	Description	Status
-	The AT&T toll free out-of-band blind transfer for trunks configured with the 4ESS ISDN protocol can only be configured using *ini* file parameters. **Applicable Products:** Mediant 8xx; Mediant 1000B.	-
-	When using the DSP Cluster feature, the local DSP resources on the SBC cannot be utilized. **Applicable Products:** Mediant 9000; Mediant VE.	-
-	When SRTP is enabled, RTP Redundancy and M-factor cannot operate together. In other words, SRTP can operate with RTP Redundancy greater than 0 or with m-factor greater than 1, but not with both. **Applicable Products:** Mediant 1000B.	-
-	When IP-to-IP or IP-to-PSTN calls use SRTP with ARIA encryption, the number of simultaneous calls is limited to 31. **Applicable Products:** Mediant 5xx; Mediant 8xx.	-
-	SBC RTP call forwarding using the SRTP tunneling feature cannot provide RTCP XR monitoring parameters (such as MOS) required for the QoE feature on the following variable bit rate coders: G.723, GSM FR, GSM EFR, MS RTA, EVRC, AMR, QCELP, and Speex. A workaround is to use SRTP full encryption / decryption on the forwarding calls. **Applicable Products:** Mediant 1000B GW & E-SBC.	-
-	Ethernet packets received on the RTP side of SRTP-RTP SBC sessions must not exceed 1500 bytes. Packets exceeding this size are dropped. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B; Mediant Non-Hybrid SBC.	-
	The device does not support the sending of RFC 2198 RTP redundancy packets as an operation if the configured packet loss threshold is exceeded; this is configured in the Quality Of Experience Web page. **Applicable Products:** All.	-
-	The Transparent coder (RFC 4040) poses the following limitations: ▪ The coder can be used only when using physical terminations ▪ No detection of IBS (e.g., DTMF) ▪ Generation of IBS is only toward the network ▪ No fax/modem detection or generation (i.e., no support for T.38 and Bypass) A workaround for this constraint is to use the G.711 coder instead. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B.	-

Incident	Description	Status
-	The RFC 2198 Redundancy mode with RFC 2833 is not supported (i.e., if a complete DTMF digit is lost, it is not reconstructed). The current RFC 2833 implementation supports redundancy for lost inter-digit information. Since the channel can construct the entire digit from a single RFC 2833 end packet, the probability of such inter-digit information loss is very low. **Applicable Products:** Mediant 5xx; Mediant 8xx.	-
-	The duration resolution of the On and Off time digits when dialing to the network using RFC 2833 relay is dependent on the basic frame size of the coder being used. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B.	-
-	The Calling Tone (CNG) detector must be set to Transparent mode to detect a fax CNG tone received from the PSTN using the Call Progress Tone detector. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B.	-
18743	EVRC Interleaving according to RFC 3558 is supported only on the receiving side. Supporting this mode on the transmitting side is not mandatory according to this RFC. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B.	-
-	The SILK coder is currently not supported. **Applicable Products:**Mediant 500L Gateway & E-SBC.	-
-	The ISDN BRI American variants (NI2, DMS100, 5ESS) are partially supported by the device. Please contact your AudioCodes representative before implementing this protocol. **Applicable Products:**Mediant 5xx; Mediant 8xx; Mediant 1000B.	-
-	All the device's trunks must belong to the same Protocol Type (i.e., either E1 or T1). **Applicable Products:** Mediant 8xx; Mediant 1000.	-
-	After changing the trunk configurations from the initial factory default (i.e., trunks are of Protocol Type 'None'), a device reset is required (i.e., the change cannot be made on-the-fly). **Applicable Products:** Mediant 8xx; Mediant 1000B.	-
-	When configuring the framing method to 'Extended Super Frame' (0) or 'Super Frame' (1), the framing method is converted to another framing method. The correct value that is updated in the device is displayed in the Web interface: ▪ For E1: 'Extended Super Frame' (0) and 'Super Frame' (1) are converted to 'E1 FRAMING MFF CRC4 EXT' (c). ▪ For T1: 'Extended Super Frame' (0) is converted to 'T1 FRAMING ESF CRC6' (D). In addition, 'Super Frame' (1) is converted to 'T1 FRAMING F12' (B). **Applicable Products:** Mediant 8xx; Mediant 1000B.	-
-	Core Dump to the internal flash device may take up to 30 minutes. During this period, a red alarm LED is lit. **Applicable Products:** Mediant 2600; Mediant 4000.	-

Incident	Description	Status
-	Hyper-Threading (HT) is supported for Mediant VE in a VMWare environment only and with special configuration (refer to the *Mediant VE SBC Installation Manual*). For all other environments of Mediant Software, HT should be disabled in the BIOS setting of the server. **Applicable Products:** Mediant Software.	-
70318	The following parameters do not return to their default values when attempting to restore them to defaults using the Web interface or SNMP, or when loading a new *ini* file using BootP/TFTP: ▪ VLANMode ▪ VLANNativeVLANID ▪ EnableDHCPLeaseRenewal ▪ IPSecMode ▪ CASProtocolEnable ▪ EnableSecureStartup **Applicable Products:** All.	-
79630	Files loaded to the device must not contain spaces in their file name. Including spaces in the file name prevents the file from being saved to the device's flash memory. **Applicable Products:** All.	-
-	Configuration file constraints when upgrading from 6.8 to 7.2: ▪ CLI Script file of 6.8 cannot be loaded to a 7.2 device ▪ Incremental ini file of 6.8 cannot be loaded to a 7.2 device **Applicable Products:** All.	-
-	The 'Monitor Destination Status' read-only field on the HA Settings page does not refresh automatically. **Applicable Products:** Mediant 4000 HA.	-
-	An unnecessary scroll bar appears on many of the Web pages when using 1280 x 1024 screen resolution. **Applicable Products:** All.	-
-	After manual switchover in HA Revertive Mode, the Web Home page isn't refreshed. A workaround is to refresh the Home page to get the updated status. **Applicable Products:** Mediant 2600; Mediant 4000.	-
-	When using the Software Upgrade Wizard, if the Voice Prompt (VP) file is loaded and the **Next** button is clicked while the progress bar is displayed, the file is not loaded to the device. Despite this failure, the user receives a message that the file has been successfully downloaded. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B; Mediant 2600; Mediant 4000.	-
87767	The Web Search feature may produce incorrect search results. **Applicable Products:** All.	Resolved in Version 7.20A.100

Incident	Description	Status
-	The fax counters, 'Attempted Fax Calls Counter' and 'Successful Fax Calls Counter' in the Status & Diagnostics page do not function correctly. **Applicable Products:** Mediant 8xx; Mediant 1000B.	-
-	From Release 7.2, configuration through SNMP is not supported. **Applicable Products:** All.	-
-	The MIB-II ifTable, ifxTable, and entPhysicalTable are not supported. **Applicable Products:** Mediant 9000; Mediant Software.	-
58872	When defining or deleting SNMPv3 users, the v3 trap user must not be the first to be defined or the last to be deleted. If there are no non-default v2c users, this results in a loss of SNMP contact with the device. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B; Mediant 2600; Mediant 4000.	-
-	Only the CLI commands explicitly mentioned in the *Installation Manual* are supported. **Applicable Products:** Mediant 9000; Mediant Software.	-
131651	Before upgrading a new firmware, the number of system snapshots should be reduced to maximum five snapshots. If the number of snapshots is above five, the user should delete some of the snapshots to free the disk space required for the burn & upgrade process. **Applicable Products:** Mediant 9000; Mediant VE/SE.	-
SBC-14187	If there no IP Groups (in the IP Groups table), the device rejects all Gateway calls (even if they don't use an IP Group). If IP Groups are not being used, IP Group Index #0 must exist for calls to be processed. **Applicable Products:** Gateway.	

2.1.3 Resolved Constraints

This chapter lists constraints from previous releases that have now been resolved.

Table 2-2: Resolved Constraints in Release 7.2

Incident	Description
124526	When upgrading the device from Version 6.8 to 7.2, the RADIUS Accounting server IP address and port (configured by the RADIUSAccServerIP and RADIUSAccPort parameters in Version 6.8) do not migrate to the new RADIUS Servers table (RadiusServers) in Version 7.2. The administrator is recommended to configure the Accounting server's IP address and port in the new table after the device has been upgraded. **Applicable Products:** Mediant Software.

2.2 Patch Version 7.20A.001

This patch version includes only new features.

2.2.1 New Features

New features introduced in this patch version include the following.

2.2.1.1 New Virtualized Platforms for Mediant VE SBC

This feature provides support for the following new virtualized platforms for the Mediant VE SBC:

- Amazon Web Service (AWS) - Elastic Compute Cloud (EC2): The device now supports Amazon cloud computing services (AWS EC2). The device needs to run on EC2 instance type c4.2xlarge. This platform also provides transcoding services.
- SR-IOV: Mediant SBC VE can now utilize SR-IOV acceleration of Intel NICs to reach even higher capacity than before. The Virtual Function (VF) of the SR-IOV capable Intel NICs should be mapped to the Ethernet ports used by the device's media IP network interfaces. SR-IOV acceleration has been verified by AudioCodes on OpenStack platform with 8 vCPUs, 64-GB RAM and Intel® 82599 NICs.

Applicable Products: Mediant VE SBC.

Applicable Application: SBC.

2.2.1.2 Enhanced Dial Plan Tags and Call Setup Rules

This feature provides support for enhanced use of Dial Plan tags:

- Dial Plan queries by Call Setup Rules (CSR): Up until now, CSR was executed only during the routing process where a CSR was assigned to an IP-to-IP Routing rule. Now, the CSR can be executed for a classified source IP Group immediately before the routing process (i.e., Classification > Manipulation > Dial Plan table > CSR > Routing) and therefore, the result of the CSR (i.e., source and/or destination tag) can be used as the matching characteristics for locating a suitable IP-to-IP Routing rule. The CSR can query the Dial Plan table for a specified search key in a specified Dial Plan to obtain the corresponding tag. The CSR can also change (modify) the name of the obtained tag.

  Multiple tags for complex routing schemes. This is typically required when the source and/or destination of the call needs to be categorized with more than one characteristics. For example, tags can be used to categorize calls by department (source user) within a company, where only certain departments are allowed to place international calls.
- LDAP queries by CSR: A specific LDAP server (LDAP Servers Group) can now be configured for the CSR.
- Message Manipulation: Source and destination tags (*srctags* and *dsttags*) can now be used in Message Manipulation rules. For example, a rule can use a specific source tag as a condition for adding a specific header to outgoing SIP messages. Note that message manipulation cannot be used to modify tags.

The following parameter changes have been made to support the feature:

- A new parameter 'Call Setup Rules Set ID' in the IP Group table that associates a CSR with the IP Group.
- Call Setup Rules table:
  - New parameter: 'Query Type' to choose between a Dial Plan and LDAP query.
  - New parameter: 'Query Target' to specify the Dial Plan name in which to search for the prefix or to specify the LDAP server (LDAP Servers Group) for LDAP queries by the CSR.

- The 'Attributes To Query' parameter (in the Web interface) has been changed to 'Search Key' as it can now be used for Dial Plan queries (prefix number) as well as LDAP queries (Attribute).

- New arguments (*dialplan.found* and *dialplan.result*) for the 'Condition' parameter in the Call Setup Rules table (e.g., *dialplan.found exists and dialplan.result=='uk'*).

Applicable Products: All.

Applicable Application: SBC.

2.2.1.3 Enhanced SIP-SIP-I Interworking

This feature provides the following enhancements for interworking SIP and SIP-I endpoints:

- Support for additional ISUP fields and corresponding Message Manipulation capabilities.
- Support for attaching any ISUP body to any SIP message, using Message Manipulation rules.
- Support for the French (France) specification, SPIROU (Système Pour l'Interconnexion des Réseaux OUverts), which regulates Telecommunication equipment that interconnect with networks in France. Therefore, a new IP Profile parameter ('ISUP Variant') has been added that allows the administrator to configure the ISUP variant to SPIROU or ITU-92 (default). For ITU—92, the device sets the Content-Type header to "version=itu-t92+; base=itu-t92+"; for SPIROU, it sets it to "version=spirou; base=itu-t92+".
- Support for configuring the SIP Content-Type and Content-Disposition header values, using Message Manipulation rules.
- Handling SIP-I suspend-resume messages (on-hook or on-hold), using a proprietary SIP header (X-Ac-Action) in SIP messages, using Message Manipulation rules.

Applicable Products: All.

Applicable Application: SBC.

2.2.1.4 Triggering Special Call Actions using X-AC-Action SIP Header

This feature provides support for triggering the device to perform special call actions. For example, it can be used for disconnecting a call when interworking SIP-I and SIP endpoints, and an ISUP SUS (suspend) message is received. This is configured using Message Manipulation rules with AudioCodes' proprietary X-AC-Action SIP header. The actions that can be performed include:

- Disconnect a call (optionally, after a user-defined time):
  disconnect[;delay=<time in ms>]
- Resume previously suspended call:
  abort-disconnect

  Example:

  ```
  X-AC-Action: abort-disconnect
  ```
- Reply to the message with a SIP response without forwarding the response to the other side:
  reply[;response=<response code, e.g., 200>]
- Switch IP Profile for the call (re-INVITE only), as defined in the IP Group:
  switch-profile [;reason=<reason - PoorInVoiceQuality or PoorInVoiceQualityFailure >]

For example, the below rule disconnects a call after 3 sec if the received SIP INFO message contains the ISUP SUS field:

```
MessageManipulations 2 = "INFO suspend", 2, "info.request",
"body.isup.sus exists", "header.x-ac-action", 0,
"'disconnect;delay=3000,reply'", 0;
```

Applicable Products: All.

Applicable Application: SBC.

2.2.1.5 VoIPerfect Feature

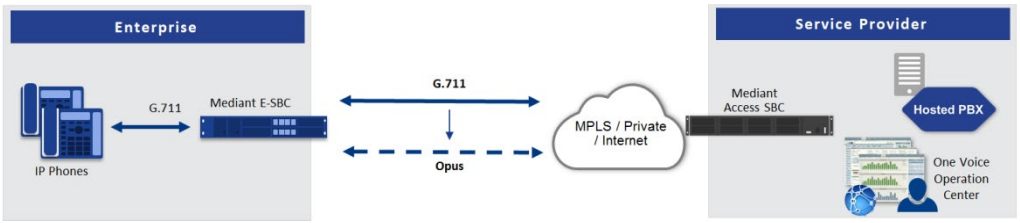
This feature provides support for a new application called VoIPerfect™ that combines AudioCodes' access and enterprise SBC technology. VoIPerfect ensures high call quality (MOS) between the Enterprise SBC and the Access SBC (located at the Internet service provider / ISP) during periods of WAN network issues (packet loss and bandwidth reduction).

VoIPerfect also guarantees that 95% of calls will achieve a Perceptual Evaluation of Speech Quality (PESQ) score greater than or equal to 3.6, if the summation of bandwidth overuse and packet loss is less than or equal to 25%. ISPs can therefore offer such service level agreements (SLAs) to their customers. For more information, contact your AudioCodes sales representative.

By ensuring high call quality even in adverse network conditions, VoIPerfect can reduce costs for ISPs such as SIP trunk providers and Unified Communications as a Service (UCaaS) by eliminating the need for dedicated WAN links (such as MPLS and leased links) and instead allow the use of standard broadband Internet connections. However, it can also be used in tandem with existing infrastructure.

The feature is applicable only to G.711 calls and uses the Opus coder for ensuring call quality. VoIPerfect can be implemented in one of the following modes:

- **Smart Transcoding:** If the SBC (Enterprise or Access) detects WAN network impairments during a call between the Enterprise SBC and Access SBC, the SBC employs voice transcoding by switching the coder from G.711 to Opus for that specific call only. Transcoding is done only on the path between the Enterprise SBC and Access SBC. As Smart Transcoding is applied only on a per call basis, it preserves valuable DSP resources that may be required for other functionalities. An advantage of using the Opus coder is that it consumes less bandwidth than G.711 and overcomes packet loss (by dynamic packet redundancy), allowing the SBC to support more concurrent calls than with G.711 for the same bandwidth. This mode is useful for WAN networks that are relatively stable, allowing the use of G.711 whenever possible and switching to Opus only during adverse network conditions.



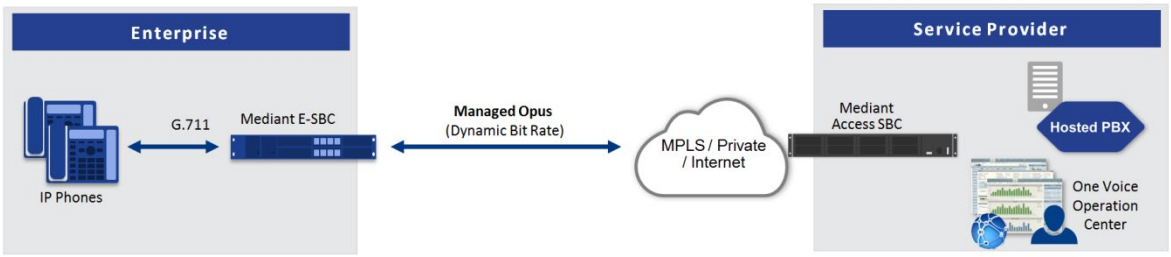
Configuration of the Enterprise SBC:

- Device's License Key includes the SBC transcoding feature
- Coder Groups:
  - Coders Group with G.711
  - Coders Group with Opus
- Allowed Audio Coders Groups:
  - Allowed Audio Coders Group with G.711
  - Allowed Audio Coders Group with Opus
- Main IP Profile:
  - Extension Coders Group: Coders Group with G.711

- Allowed Audio Coders: Allowed Audio Coders Group with G.711
- Allowed Coders Mode: Restriction
- RTCP Feedback: Feedback On
- Voice Quality Enhancement: Enable

- Alternative IP Profile:
  - Extension Coders Group: Coders Group with Opus
  - Allowed Audio Coders: Allowed Audio Coders Group with Opus
  - Allowed Coders Mode: Restriction
  - RTP Redundancy Mode: Enable
  - RTCP Feedback: Feedback On
  - Voice Quality Enhancement: Enable
  - Max Opus Bandwidth: 80000
- Quality of Service Rules:
  - Rule Metric: Poor InVoice Quality
  - Alternative IP Profile Name: name of Alternative IP Profile (above)

- **Managed Opus:** If the SBC detects WAN network impairments during a call using the Opus coder between the Enterprise SBC and Access SBC, it can adjust the Opus coder's attributes (e.g., bit rate) for that specific call to ensure high voice quality is maintained. The advantage of the Opus coder is that its' bit rate can change dynamically according to bandwidth availability. This mode is useful for unstable networks, allowing Opus to dynamically adapt to adverse network conditions.



Configuration of the Enterprise SBC:

- Coders Group with Opus
- Allowed Audio Coders Group with Opus
- IP Profile:
  - Extension Coders Group: Coders Group with Opus
  - Allowed Audio Coders: Allowed Audio Coders Group with Opus
  - Allowed Coders Mode: Restriction
  - Voice Quality Enhancement: Enable
  - RTCP Feedback: Feedback On
  - Max Opus Bandwidth: 0

Configuration of the Access SBC for both methods:

- Coder Groups:
  - Coders Group with G.711 and Opus
  - Coders Group with Opus
- Allowed Audio Coders Group with Opus
- IP Profile:
  - Extension Coders Group: Coders Group with G.711 and Opus
  - Voice Quality Enhancement: Enable
  - RTP Redundancy Mode: Enable

- RTCP Feedback: Feedback On
- Max Opus Bandwidth: 0

- Alternative IP Profile:
  - Extension Coders Group: Coders Group with Opus
  - Allowed Audio Coders: Allowed Audio Coders Group with Opus
  - Allowed Coders Mode: Restriction
  - Voice Quality Enhancement: Enable
  - RTP Redundancy Mode: Enable
  - RTCP Feedback: Feedback On
  - Max Opus Bandwidth: 0
- Quality of Service Rules:
  - Rule Metric: Poor InVoice Quality
  - Alternative IP Profile Name: name of Alternative IP Profile (above)

To support VoIPerfect, the device now supports the negotiation of Temporary Maximal Media Stream Bit Rate (TMMBR) for Opus coders. Through TMMBR, the device can receive indications of network quality and dynamically change the coder's payload bit rate accordingly during the call to improve voice quality. TMMBR is an RTCP feedback message (per RFC 4585) which enables SIP users to exchange information regarding the current bit rate of the media stream. The information can be used by the receiving side to change the media stream parameters (e.g., coder rate or coder) to enhance voice quality. TMMBR is negotiated in the SDP Offer/Answer model using the 'tmbr' attribute and following syntax:

```
a=rtcp-fb:<payload type> ccm tmmbr smaxpr=<sent TMMBR packets)
```

The device also supports another new SDP attribute, 'a=rtcp-rsize' that reduces the RTCP message size (as defined in RFC 5506). As feedback messages are frequent and take a lot of bandwidth, the attribute attempts to reduce the RTCP size. The attribute can only be used in media sessions defined with the AVPF profile. In addition, it must be included with sessions supporting TMMBR; otherwise, the call is rejected.

Note:

- VoIPerfect is applicable only to G.711 calls.
- If you are deploying a third-party device between the Enterprise SBC and Access SBC, make sure that the third-party device adheres to the following:
  - √ Enable RFC 2198 in SDP negotiation
  - √ Enable TMMBR in SDP negotiation
  - √ Forward the SDP with feedback (SAVPF) as is
  - √ Forward TMMBR messages as is
  - √ Forward RTCP messages as is (not terminate them)
  - √ (Smart Transcoding only) Forward re-INVITE messages for using Opus as is
  - √ (Smart Transcoding only) Forward the SIP header, X-Ac-Action as is

Applicable Products: Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

Applicable Application: SBC.

2.3 Patch Version 7.20A.002

This patch version includes new features, resolved constraints and known constraints.

2.3.1 New Features

New features introduced in this patch version are described in this section.

2.3.1.1 Load-Balancing of SBC Calls between Destination IP Groups

This feature provides support for load balancing of calls, belonging to the same source, to a set of call destinations known as an *IP Group Set*, which can include up to five IP Groups (Server-type and/or Gateway-type). The selected destination IP Group for each call depends on the configured load-balancing policy, which can be Round Robin, Random Weights, or Homing. Alternative routing within the IP Group Set is also supported whereby if a destination IP Group responds with a reject SIP response that is configured as a reason for alternative routing, or doesn't respond at all (i.e., keep-alive with its Proxy Set fails), the device attempts to send the call to the next IP Group (according to the policy). For example, for round-robin load-balancing, call 1 is sent to IP Group #1, call 2 to IP Group #2, and call 3 to IP Group #3. If the call sent to IP Group #1 is rejected, the device employs alternative routing and sends it to IP Group #4.

As a result of the feature, the following new parameters have been added:

- New tables:
  - IP Group Set (parent): Defines an IP Group Set (with a policy) for load balancing.
  - IP Group Set Member (child): Assigns IP Groups to the IP Group Set.
- IP-to-IP Routing table:
  - New optional value for 'Destination Type' field: "IP Group Set"
  - New field to assign an IP Group Set: 'IP Group Set'

Applicable Products: All.

Applicable Application: SBC.

2.3.1.2 Configurable FXS Off-hook Current

This feature provides support for configuring the FXS off-hook current for specific ports. FXS off-hook current is the current that the device supplies to the analog line when it is in off-hook state. Up until now, the FXS off-hook current was not configurable and fixed to 20 mA. Now, the administrator can increase the current to 35 mA using the new ini file parameter EnhancedFXSLineCurrent, where the value "0" is 20 mA (default) and "1" is 35 mA. A device reset is required for the parameter's settings to take effect. Configuration can be done only on the first (1) and last (24) ports per FXS connector.

Note that for the first FXS connector on FXS blade 1, the first port in the ini file is denoted as 0 and the last port as 23. The following configuration example sets specific first and last ports to 35 mA:

```
EnhancedFXSLineCurrent_0 = 1      ; Port 1 on FXS Blade 1
EnhancedFXSLineCurrent_23= 1      ; Port 24 on FXS Blade 1
EnhancedFXSLineCurrent_24 = 1     ; Port 25 on FXS Blade 1
EnhancedFXSLineCurrent_47 = 1     ; Port 48 on FXS Blade 1
EnhancedFXSLineCurrent_48 = 1     ; Port 49 on FXS Blade 1
EnhancedFXSLineCurrent_71 = 1     ; Port 72 on FXS Blade 1
EnhancedFXSLineCurrent_72 = 1     ; Port 1 on FXS Blade 2
```

Applicable Products: MP-1288.

2.3.2 Known Constraints

This section lists known constraints.

Table 2-3: Known Constraints in Version 7.20A.002

Incident	Description	Status
138581	Mediant Virtual Edition SBC with Microsoft Hyper-V hypervisor with 4 GB is not supported. **Applicable Products:** Mediant VE.	Resolved in Version 7.20A.100 (See Section 2.4.3)

2.3.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-4: Resolved Constraints in Version 7.20A.002

Incident	Description
138447	When the ini file includes parameter values that are over 50 characters, searching values in the Web interface causes the device to crash (reset). **Applicable Products:** All.
138371	When the device forwards a SIP re-INVITE message with more than one media (e.g. voice and fax) and receives a 200 OK with one media (e.g. RTP only), it sends a 488 response to the party that initiated the re-INVITE. As a result, the call fails. A workaround is to configure the fax parameters to send only one media. **Applicable Products:** SBC.
137859	When the UseSiptgrp parameter is configured to "Send & Receive", IP-to-Tel alternative routing does not function and the call fails. **Applicable Products:** Gateway.
137394	For PRI and BRI protocol-based calls, when a call is received from the PSTN with an empty display name, the call is sent to the IP with invalid display name. As a result, the call fails (rejected by IP side). **Applicable Products:** Gateway.
137384	When editing an IP Profile and the View button is clicked for the Extension Coder Group parameter, an error message appears. **Applicable Products:** SBC.
137356	Syslog displays responses to SIP OPTIONS messages with different SIDs compared to the OPTIONS, causing in problems with tracking messages and debugging. **Applicable Products:** All.
136808	The IPG field in the CDR displays the IP Group name only (instead of ID as well). **Applicable Products:** All.
136441	If configuration includes an invalid license pool service and host parameter, when trying to remove it, the device crashes (and resets). **Applicable Products:** SBC.
135501	The primary and secondary NTP server cannot be configured through CLI. **Applicable Products:** All.

2.4 Patch Version 7.20A.100

This patch version includes new features, resolved constraints and known constraints.

2.4.1 New Features

New features introduced in the GA version include the following:

2.4.1.1 Capacity Updates

This release introduces capacity updates to the following products:

- Mediant 1000 Gateway & E-SBC (Profile #6 for E1/T1)
- MP-1288: 588 (signaling and RTP-to-RTP; 350 registered users)
- Mediant 9000 SBC (180,000 registered users)
- Mediant 9000 SBC with Media Transcoder (180,000 registered users)
- Mediant VE SBC with Media Transcoders (new)

For more information, see Section 0.

Applicable Products: MP-1288; Mediant 9000 SBC; Mediant 9000 SBC with MT; Mediant VE SBC with MT/vMT.

2.4.1.2 OpenSSL Library Update

This version uses the latest OpenSSL library, which may have removed certain cipher suites from the default subset due to new vulnerabilities, hacks and computation advances. If you are using encrypted communications, you should verify that the cipher suites of both client and server roles are configured correctly in the TLS Contexts table so that they match peer abilities and desired security level. For a list of cipher suite configuration syntax, please visit the OpenSSL online documentation at https://www.openssl.org/docs/man1.0.2/apps/ciphers.html.

Applicable Products: All.

2.4.1.3 Integrated SBC Configuration Wizard in Web Interface

This feature provides **beta** support for the integration of the SBC Configuration Wizard into the device's Web interface. Up until now, the SBC Configuration Wizard was a stand-alone utility offered by AudioCodes.

The SBC Wizard provides a quick-and-easy method for initial configuration of your <device>. It guides you through a sequence of pages, assisting you in defining your setup and then finishing with a suitable configuration. The wizard is based on partial as well as fully-tested interoperability setups (configuration templates) between AudioCodes devices and a wide range of vendors, including SIP trunking providers, IP PBXs, and contact centers.

The wizard is accessed using the new Configuration Wizard button, located on the menu bar. The Auxiliaries page allows the upload of additional SBC configuration templates using the new Browse button for SBC Wizard Template files.

Applicable Products: SBC.

2.4.1.4 MP-1288 Support for SBC Application

MP-1288 now supports the SBC application. Up until now, it supported only the Gateway application (FXS interfaces). The device supports up to 300 concurrent SBC call sessions, 350 registered SBC users, and 800 SIP SUBSCRIBE messages. (Transcoding is not supported.)

Applicable Products: MP-1288.

2.4.1.5 AudioCodes One Voice Operations Center Support for MP-1288

AudioCodes One Voice Operations Center now supports the MP-1288 Analog Media Gateway.

Applicable Products: MP-1288.

2.4.1.6 MP-1288 Support for Cloud Resilience Package Application

MP-1288 now supports the Cloud Resilience Package (CRP) application. Up until now, this application was supported only by Mediant 5xx, Mediant 8xx, Mediant 1000B, Mediant 2600, Mediant 4000, and Mediant VE/SE SBCs.

Applicable Products: MP-1288.

2.4.1.7 New SNMP Alarms for MP-1288

This feature provides support for the following new SNMP alarms:

- AcModuleServiceAlarm – sent when multiple FXS ports on a specific FXS blade are out-of-service or a hardware fault occurs on the FXS blade.
- AcModuleOperationalAlarm - sent when an operational hardware failure occurs on the FXS ports or on the FXS blades (DSP and CPU).
- acPortServiceAlarm - sent is raised when an FXS port is out of service due to one of the following:
  - The Serial Peripheral Interface (SPI) connection with the port is lost.
  - The temperature of the port has exceeded the temperature threshold.
  - The port is inactive due to a ground fault.

Applicable Products: MP-1288.

2.4.1.8 New SNMP Alarm for License Pool Over-Allocation

This feature provides support for the new SNMP alarm, acLicensePoolOverAllocationAlarm, which the device sends when the SBC license received from the License Pool Manager has exceeded the maximum capacity supported by the device.

Note that the functionality of the alarms, acLicensePoolApplicationAlarm and acLicensePoolInfraAlarm were slightly modified in this release. For more information, refer to the *SNMP Reference Guide*.

Applicable Products: All.

2.4.1.9 New SNMP Alarm for TLS Certificate Expiration

This feature provides support for the new SNMP alarm, acCertificateExpiryAlarm, which the device sends when the TLS certificate of a configured TLS Context is about to expire or has expired. This alarm replaces the now obsolete trap, acCertificateExpiryNotification.

Applicable Products: All.

2.4.1.10 SNMP Version in Keep-Alive Trap

This feature provides support for indicating the device's SNMP version in the acKeepAlive trap. The version is shown in the trap Varbind, acBoardTrapGlobalsAdditionalInfo2 (SNMPVersion=SNMPv3 or SNMPv2c).

Applicable Products: All.

2.4.1.11 New SNMP Varbind for Serial Number

This feature provides support for including the device's serial number in the Variable Binding list (Varbind) of raised SNMP traps. A new Varbind, acBoardTrapGlobalsSystemSerialNumber has been added to include the serial number.

Applicable Products: All.

2.4.1.12 DH Key Size per TLS Context

This feature provides support for configuring the Diffie-Hellman (DH) key size per TLS Context. Up until now, the DH key size was a hard-coded, globally set 1024-bit key. The new feature gives administrators the option to select a 1024- or 2048-bit key size for DH. DH is an algorithm used chiefly for exchanging cryptography keys used in symmetric encryption algorithms like AES. To support the feature, a new parameter, 'DH Key Size' (TLSContexts_DHKeySize) with optional values 1024 (default) and 2048 has been added to the TLS Contexts table.

Applicable Products: All.

2.4.1.13 DTLS Version per TLS Context

This feature provides support for configuring the Datagram Transport Layer Security (DTLS) protocol version per TLS Context. The new feature gives administrators the option to select any version, Version 1.0, or Version 1.2. DTLS key negotiation protocol secures UDP-based media streams (according to RFC 5763 and 5764). To support the feature, a new parameter, 'DTLS Version' (TLSContexts_DTLSVersion) with optional values Any (default), DTLSv1.0, and DTLSv1.2 has been added to the TLS Contexts table.

Applicable Products: All.

2.4.1.14 RSA Public Key for SSH Authentication per Management User Account

This feature provides support for configuring a secure socket shell (SSH) public key per management-user account for accessing the CLI. Up until now, only one SSH public key could be configured (using the SSHAdminKey parameter), which applied to all user accounts. The feature is made possible by a new parameter in the Local Users table, called SSH Public Key (WebUsers_SSHPublicKey). The public key is used for authenticating remote users logging into the device's management interface through SSH (PKI). Connection to the management interface is established only when a successful handshake with the user's private key occurs.

Applicable Products: All.

2.4.1.15 Increase in IP Network Interfaces, VLANs and Media Realms

This feature provides support for an increase in the maximum number of IP network interfaces (IP Interfaces table), VLANs (Ethernet Devices table), and Media Realms (Media Realms table) that can be configured, from 48, 48 and 64 respectively to 1,024.

Applicable Products: Mediant 2600; Mediant 4000.

2.4.1.16 Online Detection for Proxy Set Load Balancing

This feature provides support for configuring the minimum number of online proxies, in a Proxy Set, for the Proxy Set to be considered as online when Proxy Load Balancing is used. The feature is configured using the new Proxy Set table parameter, 'Min. Active Servers for Load Balancing' (ProxySet_MinActiveServersLB).

Applicable Products: All.

2.4.1.17 LED Indication for Software Upgrade

This feature provides support for the device's STATUS LED, located on the front panel of the chassis, to indicate that the device is currently upgrading its software (.cmp file). During an upgrade, the LED flashes green.

Applicable Products: Mediant 800.

2.4.1.18 Media Transcoding Cluster Enhancements

This feature provides the following enhancements for the Media Transcoding Cluster feature:

- Media Transcoders can be connected to the Cluster Manager through an Ethernet switch. Up until now, they could only be connected directly (not through a switch) to the Cluster Manager (i.e., port to port).
- Multiple Media Transcoders can be associated with the same Cluster interface. Up until now, each Cluster interface could be associated with only one Media Transcoder.
- The Cluster Manager can also be a Mediant VE SBC (currently, supported only by Mediant VE SBC based on OpenStack). Up until now, only Mediant 9000 could serve as a Cluster Manager.
- When the Cluster Manager is a Mediant VE SBC, the Media Transcoder can also be a virtualized machine (VM), referred to as "vMT" (virtualized Media Transcoder). Up until now, the Media Transcoder was based only on a hardware appliance, referred to as "MT". Note that the Media Transcoders can only be of one type (all MT or all vMT; a combination is not allowed).
- Maximum number of Media Transcoders:
  - MT (Hardware-based appliance): increased from six to eight
  - vMT: 5
- Configurable maximum bandwidth for Cluster interfaces. The bandwidth applies to each Cluster interface. The new parameter MtcClusterNetworkMaxBandwidth has been added to the Cluster Manager Settings page (Setup menu > IP Network tab > Transcoding Cluster folder > Cluster Manager Settings) and CLI (configure network > mtc settings > cluster-network-max-bandwidth). The range is 1 to 10,000 Mbps (default is 1,000 Mbps).
- SNMP alarm for bandwidth over-utilization of a Cluster interface. To support the feature, a new SNMP alarm, acClusterBandwidthAlarm has been added. The device generates the alarm with one of the following severity levels:
  - Minor: bandwidth utilization is between 85 and 90%
  - Major: bandwidth utilization is above 90%

Applicable Products: Mediant 9000; Mediant VE.

2.4.1.19 Register-Unregister per Trunk Group

This feature provides support for initiating register and un-register actions per Trunk Group in the Trunk Group Settings table (TrunkGroupSettings), using the Register and Un-Register commands. Up until now, when these actions were done in the Trunk Group Settings table, all Trunk Groups were affected.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000.

2.4.1.20 Enhanced Row-Pointer Feature

This feature provides enhanced capabilities of row-pointer fields, which are used to assign index rows of other configuration tables to a current table row:

- The row-pointer field (drop-down list) allows the administrator to search by name for a referenced-table row.
- The row-pointer field displays the status of the referenced-table rows (e.g., invalid row), using icons.
- The row-pointer field provides an "Add Row" button that allows the administrator to add a new row in the referenced table.
- When in the referenced table (after the View or Add New button has been clicked), the administrator can select the required row using the new "Use selected row" button.
- Multiple display of "View" capability.

Applicable Products: All.

2.4.1.21 Multiple SRSs and SRS Redundancy for SIPRec

This feature provides support for sending copies of call sessions traversing the device to multiple Session Recording Servers (SRS) for the SIPRec feature. Up until this version, only one SRS could be configured. Now, the administrator can configure up to three groups of SRSs, where each group can contain one standalone SRS, or two SRSs for 1+1 (active-standby) SRS redundancy.

Note:

- The feature is applicable only to the SBC application (only one SRS can be configured for the Gateway application).
- SRS redundancy is a license-dependent feature, defining the maximum number of SIPRec sessions that can be copied to the redundant (standby) SRS. (This is in addition to the regular SIPRec feature key.)

Applicable Products: MP-1288; Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.4.1.22 Product Key for Enhanced Product Identification

This feature provides support for aligning the device's serial numbers displayed on the chassis' product label and management interfaces (Web and One Voice Operations Center). Up until now, the product label displayed the chassis' serial number while the management interfaces displayed the CPU serial number. These serial numbers are now displayed on both the product label and management interfaces, as follows:

Serial Number Type	Product Label	Management Interface
Chassis	"S/N(Product Key)"	"Product Key"
CPU	"CPU S/N"	"Serial Number"

The Web interface displays the Serial Number and Product Key on the License Key page and Device Information page, in the new fields 'Serial Number' and 'Product Key', respectively.

For new product purchases as well as for each new License Key upgrade, the License Key includes the Product Key, which will be displayed automatically in the 'Product Key' field when the License Key is installed on the device. For existing customers who have upgraded their device's firmware but not License Key, the 'Product Key' field will appear empty.

Note that the Product Key is already supported by Mediant 9000 SBC and Mediant SE/VE SBC.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000B Mediant 2600; Mediant 4000.

2.4.1.23 CLI Startup Script for Non-MSBR Products

This feature provides support for the CLI Startup script file by non-MSBR devices. Up until now, the file was supported only by AudioCodes MSBR product line. The file contains only CLI-based configuration and when loaded to the device, applies its settings and restores all other parameters not included in the file to factory defaults. The file causes the device to undergo two resets to apply the settings and thus, typically contains the Automatic Update settings and other settings that require a <device> reset.

Applicable Products: All.

2.4.1.24 Saving and Loading CLI-based Configuration Files in Web Interface

This feature provides support for saving and loading CLI Script and CLI Startup Script files through the device's Web interface. This is done in the existing Configuration File page (Setup menu > Administration tab > Maintenance folder > Configuration File).

Applicable Products: All.

2.4.1.25 Hitless License Upgrade from Pool Manager

This feature provides support for devices operating in High-Availability (HA) mode to receive a new SBC license from the License Pool Manager without affecting traffic (i.e., current calls are maintained). Up until now, each device, including the active device was reset, thereby disconnecting currently active calls. The new feature employs a "hitless" license upgrade mechanism, whereby the License Pool Manager first downloads the license to the redundant unit, resets it, and then triggers an HA switchover. It then downloads the same license to the previously active device, resets it and then triggers another HA switchover.

Applicable Products: All.

2.4.1.26 Debug for Remote Web (HTTP) Services

This feature provides support for enabling the device to generate debug messages for remote Web (HTTP) services and send them to a Syslog server. The feature is enabled by the new parameter, 'HTTP Proxy Debug Level' (ini – HTTPProxySyslogDebugLevel; CLI - configure network > http-proxy settings > http-proxy-debug-level). The debug level can be configured to 0 (No Debug), 1 (Basic) or 3 (Detailed).

Applicable Products: All.

2.4.2 Known Constraints

This section lists known constraints.

Table 2-5: Known Constraints in Version 7.20A.100

Incident	Description	Status
139442	When the device is operating in High-Availability (HA) mode and a hitless software upgrade from an earlier version to Version 7.2.100 is done through the Web interface, the Web interface sometimes erroneously displays an upgrade failure message and that a reset must be done, even though the devices were upgraded successfully. If this occurs, refresh the browser and then log in again to the Web interface. **Applicable Products:** Mediant 500 E-SBC HA; Mediant 800 Gateway & E-SBC HA.	Resolved in Version 7.20A.150 (See Section 2.7.3)
133294	Creating or deleting of virtual machine snapshots using the hypervisor tools sometimes causes the SBC HA system to reset. A workaround is to first shutdown the virtual machine (active or redundant SBC) and only then create or delete the snapshot. **Applicable Products:** Mediant VE SBC.	
139964	When the Firewall table (AccessList in file parameter) is configured with a firewall rule that blocks (denies) traffic from source port 53, start port 0 and end port 0, incoming Standard query responses from DNS port 53 is erroneously allowed by the firewall rule. **Applicable Products:** All.	
140547	Transcoding of G.711 to G.729 for loopback calls fail (disconnect) after an HA switchover. **Applicable Products:** HA Products.	Resolved in Version 7.20A.150 (See Section 2.7.3)

2.4.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-6: Resolved Constraints in Version 7.20A.100

Incident	Description
137821	When the LDAPDebugMode parameter is set to 3, the LDAP passwords erroneously appear in syslog, posing security risks. A workaround is to set the parameter to a lower value. **Applicable Products:** All.
139470	If the value of the Source IP field in the Firewall table contains an asterisk (*), the device crashes (resets). **Applicable Products:** All.
138984	Three-way conference calls cannot be made when the device's License Key includes "DSPCh = 288". A workaround is to set DSPCh to 72. **Applicable Products:** MP-1288.
138889	If ICE parameters are changed during a WebRTC session, the device rejects the incoming STUN binding requests and as a result, the call cannot be established. **Applicable Products:** WebRTC-supporting products.

Incident	Description
138751	DTMF transcoding fails when the SBC call uses the G.729 coder. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000.
138731	The device crashes (resets) when processing Call Setup rules. **Applicable Products:** SBC.
138703	Graceful lock cannot be cancelled during the lock timeout. **Applicable Products:** All.
138701	Processing some values in the ini file causes the device to crash (reset). **Applicable Products:** All.
138551	When configuring IP-to-Tel routing rules, the Source IP Group is erroneously used as a matching input for the table and not as the output, causing incorrect routing. **Applicable Products:** MP-1288.
138517	When the NTP server is configured as an FQDN, the HA redundant device crashes during the ini file upload and as a result, it returns to its former configuration. A workaround is to use an IP address for the NTP server. **Applicable Products:** HA-supporting products.
138495	When processing a Call Setup rule that requires an LDAP query, connectivity with the LDAP server fails, causing a device crash (reset). **Applicable Products:** SBC.
138386	Blind call transfers (SBC call) fail during an HA switchover (device rejects re-INVITE). **Applicable Products:** HA-supporting products.
138371	One leg sends Re-INVITE with a=sendonly and the device sends the re-INVITE with added T.38 media line to the second leg. As a result, the remote party rejects the re-INVITE and the call fails. **Applicable Products:** SBC.
137317 / 137318	The following parameters do not appear in the Web interface: SBCKeepContactUserinRegister and UdpPortSpacing. **Applicable Products:** All.
138277	The device does not support the maximum number of transcoding sessions as defined in the License Key and any calls above a certain number are dropped. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000.
138015	Some Gateway parameters are missing from the CLI. **Applicable Products:** MP-1288.
138014	Some Message Manipulation parameters do not appear in the CLI. **Applicable Products:** MP-1288.
137384	The Extension Coders Group in the IP Profile table cannot be accessed through the View button. **Applicable Products:** All.
136948	Daylight Saving Time settings are not saved after an HA switchover and therefore, the device is set with the incorrect time. **Applicable Products:** Mediant Software SBC HA.
135434	GRUB logging timeout during device reset is too long. **Applicable Products:** Mediant VE SBC.

Incident	Description
134449	RADIUS-based authentication of SIP users and RADIUS-based authentication of login username and password for management users are currently not supported. **Applicable Products:** Mediant 2600; Mediant 4000.
-	The SIPRec feature is not supported when the Media Transcoding Cluster feature is enabled. **Applicable Products:** Mediant 9000.
132977	To upgrade from Software Version 7.0 to 7.2, the device must first be upgraded to the latest 7.0 version (later than 7.00A.058.002) and only then to Version 7.2. **Applicable Products:** Mediant 2600; Mediant 4000; Mediant 9000; Mediant SE/VE.
133943	SRTP with ARIA encryption is not supported for SBC sessions. **Applicable Products:** All.
-	ARM is not supported. **Applicable Products:** All.
131889	When importing a Dial Plan file (*.csv file), it is recommended to configure the SyslogDebugLevel parameter to No Debug. **Applicable Products:** All.
116756	The device interworks with devices that support RTP bundling. However, it does not support receipt of bundled multimedia sessions on the same port and instead, it uses different ports for each media type (audio and video). By default, the device removes all bundle-related attributes ('a=group:BUNDLE' and 'a=ssrc') from the SDP offer and answer. **Applicable Products:** All.
87767	The Web interface's search feature may produce incorrect search results. **Applicable Products:** All.
138581	Mediant Virtual Edition SBC with Microsoft Hyper-V hypervisor with 4 GB is not supported. **Applicable Products:** Mediant VE.

2.5 Patch Version 7.20A.104.001

This patch version includes only resolved constraints.

2.5.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-7: Resolved Constraints in Version 7.20A.104.001

Incident	Description
-	AMR narrowband bug.
141854	When IP Phones register through OVR, they are unable to process all the XML in the SUBSCRIBE response (which includes the user number). As a result, they are not registered. **Applicable Products:** Mediant 800 with OVR.
141828	Incorrect routing – the device matches the suffix string (objectif-54) in the IP-to-IP Routing table for user +33467670000, even though the destination/source numbers do not contain the string objectif-54. **Applicable Products:** SBC.
141659	The device crashes (resets) upon receipt of a REGISTER when its License Key doesn't include FEU. **Applicable Products:** SBC.
141640	When the power source of the device is unstable, some ports experience a problem and many error messages are generated. As a result, the ports become disabled and the Syslog is flooded with errors. A workaround is to make sure that the power source is stable. **Applicable Products:** MP-1288.
141535	When the parameter HAPingEnabled is enabled, the device crashes (resets). **Applicable Products:** HA SBC.
141527	When the device receives an RTP without payload, it crashes (resets). **Applicable Products:** All.
141499	When debug recording is enabled to record media, the device crashes (resets). **Applicable Products:** Mediant VE.
141446	The CPT file cannot be loaded through HTTP or HTTPS. A workaround is to use the Web to load the file. **Applicable Products:** MP-1288.
141435	When an UPDATE for AMR is received without octet-aligned attribute, the device rejects the UPDATE and the call fails. A Workaround is to use message manipulation to add it. **Applicable Products:** SBC.
141419	If a Startup Script CLI file or ini file is loaded through the Web interface, the configuration is not correctly reflected on the device. **Applicable Products:** All.
141397	When the user part of the To\From header is greater than 500 characters, the device crashes (resets). **Applicable Products:** SBC.

Incident	Description
141336	When the device authenticates users through the User Info file and the password of a user is changed, the device does not challenge the user, exposing a security risk. **Applicable Products:** All.
141335	When the device receives some non-standard packets from the IP, it does not respond. **Applicable Products:** All.
141326	When the device receives a REFER before the call is connected, it rejects it and the call transfer fails. **Applicable Products:** SBC.
141243	If the first entry in the Ethernet Devices table is tagged, the redundant device does not operate (no HA). A workaround is to set the first entry to tagged. **Applicable Products:** HA.
141187	Some scenarios during trans-rating cause exceptions and as a result, the device resets. **Applicable Products:** Mediant VE.
141132	A Major alarm permanently appears that indicates resetting the device after allocating licenses to the device from the AudioCodes One Voice Operations Center license pool. **Applicable Products:** All.
141044	The device does not allow more than 60,000 registered users (expected is 75,000). **Applicable Products:** Mediant VE/SE; Mediant 9000.
141039	The device crashes (resets) in the following scenario: 1) device sends INVITE and the first 180 response doesn't have a Contact 2) another 180 response with a different To tag is received and forking is recognized. **Applicable Products:** SBC.
141016	When using the CLI command **show ntp**, there is no NTP reference line in the result. **Applicable Products:** All.
140881	When user information is removed the User Info table, registration is not automatically removed in the SBC Registered Users database and the device keeps replying with 200 OK. **Applicable Products:** SBC.
140812	When the Syslog line for RAISE or CLEAR alarm ends with "Unique ID:1", "Unique ID:2" or "Unique ID:3", no UTC time is printed at the end of the Alarm syslog line. **Applicable Products:** All.
139730	When connection to the NTP server is lost, no alarm is raised **Applicable Products:** All.
140696	When an IP Profile is assigned in the Classification table, the device does not de-allocate resources when the call ends and as a result, new calls cannot be processed. **Applicable Products:** SBC.
140561	Modifying the parameters DenyAccessOnFailCount and DenyAuthenticationTimer does not take effect. **Applicable Products:** All.

2.6 Patch Version 7.20A.106.003

This patch version includes only resolved constraints.

2.6.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-8: Resolved Constraints in Version 7.20A.106.003

Incident	Description
144742	A bug in RTCP fraction lost calculation causes the device to crash (reset). **Applicable Products:** SBC.
143903	When in HA mode, even though the two License Keys are identical between active and redundant device, an alarm is raised indicating there's a License Key mismatch. **Applicable Products:** Mediant VE SBC.
142336	When using message manipulation to remove the prefix ('+') of non-existing Request-URI's user-part, the device crashes (resets). **Applicable Products:** SBC.
143127	The device is unable to load (when automatic update is employed) an ini file through HTTPS when the URL for the ini file is greater than 512 bytes. **Applicable Products:** All.
141935	HA keep-alive packets are not received correctly, causing the device to perform HA switchovers (and resets). **Applicable Products:** Mediant VE/SE; Mediant 9000.

2.7 Patch Version 7.20A.150.004

This patch version includes only new features, known constraints and resolved constraints.

2.7.1 New Features

New features introduced in this version include the following:

2.7.1.1 Session Capacity Increases

This feature provides an increase in capacity for the following products:

- **Mediant 500 E-SBC:**
  - RTP-RTP / RTP-TDM sessions (with registrations): 250
    SRTP-RTP / SRTP-TDM sessions (with registrations): 200
  - Registered users: 1,500 (with registration refresh rate of 3,600 seconds and without SUBSCRIBES)
- **Mediant 800B E-SBC & Gateway:**
  - Signaling sessions:
    - Without registrations: 400
    - With registrations: 300
  - RTP-RTP / RTP-TDM sessions:
    - Without registrations: 400
    - With registrations: 300
  - SRTP-RTP / SRTP-TDM sessions:
    - Without registrations: 250
    - With registrations: 200
  - Registered users: 1,500 (with registration refresh rate of 3,600 seconds and without SUBSCRIBES)
- **Mediant 9000 SBC** (with at least one 10 GbE NIC)**:** RTP sessions increased to 50,000
- **Mediant VE SBC (OpenStack KVM and SR-IOV Intel NICs):**
  - Signaling sessions: 24,000
  - RTP-RTP sessions: 24,000
- **Mediant VE SBC with Media Transcoders (OpenStack):**
  - Signaling sessions: 24,000
  - RTP-RTP sessions: 24,000
  - SRTP-RTP sessions: 12,000

For detailed session capacity, see Section 4.1.

Applicable Applications: SBC.

Applicable Products: Mediant 500; Mediant 800; Mediant 9000; Mediant VE/SE.

2.7.1.2 Analog Voice Interface Support on Mediant 500L E-SBC/Gateway

The Mediant 500L E-SBC and Media Gateway now supports analog voice interfaces (FXS and FXO). This support is currently offered with up to four FXS ports and four FXO ports. FXS Analog Lifeline is also supported, whereby during a power outage, calls can be received / made from / to the PSTN (FXO) by the FXS lifeline telephone.

Applicable Applications: Gateway.

Applicable Products: Mediant 500L E-SBC & Gateway.

2.7.1.3 Bulk TLS Root Certificate Import

This feature provides support for importing multiple TLS root certificates into the device's Trusted Root Certificate store from a single file. The file must have the *.PEM extension and each X.509 certificate in the file must be Base64 encoded (PEM). When copying-and-pasting the certificates into the file, each certificate must be enclosed by the first line "-----BEGIN CERTIFICATE-----" and the last line "-----END CERTIFICATE-----".

The file is imported using the existing mechanism – Import button in the Trusted Certificates page (TLS Contexts page > Trusted Root Certificates link).

Applicable Products: All

2.7.1.4 Base64 (PEM) Encoded String Included in Certificate Display

This feature provides support for including the Base64-encoded format of the certificate, associated with a TLS Context, in the certificate information display in the Web interface (by clicking the Certificate Information link in the TLS Contexts page). Up until now, the certificate information option displayed general information, for example, Issuer, Subject, and Validity. The feature may be useful, for example, by allowing the administrator to select the encoded string and then copy-and-paste it in a text-based file for backup.

Applicable Applications: All.

Applicable Products: All.

2.7.1.5 Generation of Encrypted Private Key File

This feature provides support for configuring a password (passphrase) for a private key file generated by the device for a specified TLS Context. The passphrase provides secondary security, for example, if the encrypted private key is stolen the key cannot be viewed without the passphrase. The feature is supported by a new parameter, 'Private key pass-phrase' on the Change Certificates page (TLS Contexts table > Change Certificate link). If left blank, the private key will not encrypted.

Applicable Applications: All.

Applicable Products: All.

2.7.1.6 Token-based Authentication for Accessing Web Interface

This feature provides support for the integration of the device's Web interface with third-party products. The authentication token can be retrieved using the REST API and appended to the device's URL (<IP address>/api/v1/actions/authToken), thus enabling direct access to the device's Web interface without the need to enter a username and password.

Applicable Applications: All.

Applicable Products: All.

2.7.1.7 TLS Certificate Management through REST

This feature provides support for managing (GET, PUT and POST actions) the device's TLS Certificates (TLS Contexts) from a REST client through AudioCodes REST API. The feature is supported by the following new REST URL path:

```
/api/v1/files/tls
```

The feature can be used, for example, to retrieve information of a certificate, upload or download a certificate, and generate a CSR. For more information, refer to the document *REST API for Mediant Devices*.

Applicable Applications: All.

Applicable Products: All.

2.7.1.8 Routing Based on QoS by Routing Server

This feature provides support for the routing server (for example, AudioCodes ARM) to route calls based on QoS metrics (media and signaling) collected by the SBC/Gateway device. The device collects QoS metrics (e.g., packet loss, MOS, audio bandwidth) per IP Group that is configured to operate with the routing server (Used by Routing Server parameter set to "Used"). Each QoS report can contain the status of up to 100 IP Groups. If more than 100 IP Groups exist, multiple QoS reports are sent.

To enable the device to send QoS reports for these IP Groups:

- The new global parameter, Quality Status (RoutingServerQualityStatus) must be set to "Enable".
- The new global parameter, Quality Status Rate (RoutingServerQualityStatusRate) can optionally be configured, which defines the rate (sec) at which QoS reports are sent (15-3600, default 60).
- The exiting parameter, Type (HTTPRemoteServices_HTTPType) in the Remote Web Services table must be set to the new optional value "QoS", for the Web service configured for the routing server.
- Voice quality monitoring and RTCP-XR must be enabled (using the exiting parameter Enable RTCP XR (VQMonEnable).

The Quality Status and Quality Status Rate parameters can be read and modified in REST using the new REST API parameters, RoutingServerQualityStatus and RoutingServerQualityStatusRate, under the URL resource /api/v1/rmConfig/globals.

The feature is supported by the following new REST API URL resource:

```
POST <Route_Server_Path>/qualityStatus
```

Note:

- For media metrics calculations, the device's License key must include voice quality monitoring and RTCP-XR.
- If there is no service configured with the type “QoS”, reports are sent to the Topology server.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.9 Tag-Based Routing Enhancement

This feature provides support for tag-based routing, whereby the destination in the IP-to-IP Routing table is based on a Dial Plan tag. A summary (skips steps for ease of clarity) of the call processing for the feature is as follows: Once the incoming SIP dialog is classified to an IP Group, the device searches the Dial Plan that is associated with the IP Group, for a Dial Plan rule that matches the destination (called) prefix number. The device then searches the IP-to-IP Routing table for a matching routing rule. If the destination of the matched rule is based on a tag (see parameters below), it performs some logic to use one of the tags in the matched Dial Plan rule and then searches the IP Groups table and IP Group Set table for an IP Group or IP Group Set that is configured with this "destination" tag and if found, routes the dialog to that IP Group.

The feature is supported by the following new optional values and parameters:

- IP Groups table: New parameter – 'Tags' (IPGroup_Tags)
- IP Group Set table: New parameter – 'Tags' (IPGroupSet_Tags)
- IP-to-IP Routing table:
  - 'Destination Type' parameter has a new optional value - "Destination Tags" (12)
  - New parameter – 'Routing Tag Name' (IP2IPRouting_RoutingTagName)

Applicable Applications: SBC.

Applicable Products: All.

2.7.1.10 Fax Rerouting for SBC Calls

This feature provides support for rerouting incoming SBC calls that are identified as fax calls to a new IP destination. The device identifies a fax call if it detects a calling (CNG) tone on the incoming call (originator side). Detection is done within a user-defined interval, configured by the existing parameter, SBCFaxDetectionTimeout.

Once the device detects a fax call, it terminates the initial call and reroutes it using a new INVITE to the new fax destination (new IP Group). If the initial INVITE used to establish the voice call was already sent, the device sends a SIP CANCEL (if not connected yet) or a BYE (if already connected) to release the call (with the internal disconnect reason RELEASE_BECAUSE_FAX_REROUTING, translated to Q.850 reason GWAPP_NORMAL_UNSPECIFIED 31).

The feature is configured using two IP-to-IP Routing rules in the IP-to-IP Routing table, where the second rule is configured to the new optional value "Fax Rerouting" (6) for the 'Call Trigger' parameter (IP2IPRouting_Trigger). In addition, the IP Profile of the terminating fax side is configured with the new parameter 'Fax Rerouting Mode' (IpProfile_SBCFaxReroutingMode) to "Rerouting without delay".

Applicable Products: SBC.

2.7.1.11 Routing Back to Sender

This feature provides support for configuring the device to reply to the sender (source) of an incoming SIP dialog, instead of routing the call to another SIP entity. The device can reply with a SIP response code (e.g., 200 OK) or a 3xx redirection response (with an optional Contact field indicating to where the sender must re-send the message). For example, if the incoming call matches the routing rule, the rule can be configured to send a SIP 200 OK response to the sender of the incoming call. The feature can be used for normal and alternative routing. The feature is supported by the following new option and parameter in the IP-to-IP Routing table:

- New optional value "Internal" for 'Destination Type' (IP2IPRouting_DestType): Enables the feature.
- New parameter 'Internal Action' (IP2IPRouting_InternalAction): Defines the response code or redirect response, using the following syntax:
  - Response codes:
    ```
    Reply(response='<code>')
    ```
  - Redirect:
    ```
    Redirect (response='<code>', contact='sip:'+….)
    Redirect (contact='…', response = '<code>')
    Redirect (contact = 'sip:user@host')
    ```
    The response code for redirect messages can only be 3xx.

  The string value "Reply" on its own depicts a 200 OK; the value "Redirect" depicts a 302 Redirect.

  Examples:
  - The device responds to incoming dialog with SIP 200:

```
Reply(response='200')
```

- The device responds to incoming dialog with SIP 300:

```
Redirect (response='300', contact='sip:102@host')
```

- The device redirect calls from the sender to a SIP Recording server (SRS), by sending the sender the following redirect message:

```
Redirect(response='302',contact='sip:'+header.to.url.user+'
@siprecording.com')
```

Applicable Applications: SBC.

Applicable Products: All.

2.7.1.12 String Concatenation in Message Conditions

This feature provides support for using the plus "+" operator as part of the value in message conditions ('Condition' field) for concatenating strings. Conditions are used in the Message Manipulations table, Message Conditions table, and Call Setup Rules table. Below shows two examples of a Condition using the "+" operator (bolded):

```
header.from contains 'sip:' + header.REQUEST-URI.url.user AND
header.to contains var.global.0 + var.global.1
ldap.attr.msRTCSIP-Line contains
'tel:'+param.call.dst.user+':ext='
```

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.13 Pre-Parsing SIP Message Manipulation

This feature provides support for manipulating incoming SIP messages before they are parsed (as an object) by the device. In other words, messages can now be manipulated in their original format (plain text) as received from the network. This may be useful, for example, to overcome parser strictness or to “allow” possible parsing errors.

Pre-parsing message manipulation rules are configured using the new parent-child tables, Pre-Parsing Manipulation Sets table (PreParsingManipulationSets) and Pre-Parsing Manipulation Rules table (PreParsingManipulationRules), respectively. The rule set is associated with specific calls by assigning it to the relevant SIP Interface, using a new parameter in the SIP Interfaces table called Pre-Parsing Manipulation Set (SIPInterface_PreParsingManSetName).

Pre-parsing message manipulation rules are defined by SIP message element to manipulate (for example, INVITEs), pattern based on regular expression (REGEX) to search for (match) in incoming messages, and the regex pattern that will replace the matched pattern.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.14 Message Manipulation and Carriage Returns

This feature provides support for indicating carriage returns (new lines) in literal strings for all SIP message elements (request URI, headers and body), in Message Manipulation rules, Message Condition rules, and Call Setup rules. Up until now, this was supported only for the addition of SIP message bodies, for example, SDP ('Action Type' field set to **Add**).

The double-backslash (\\) is used to indicate a carriage return within a string (enclosed by a single apostrophe), for example:

```
body.sdp contains 'a=bbb\\a=ccc'
```

The above example shows a Condition value where the condition is an SDP that includes the following two lines of strings:

```
a=bbb
```

```
a=ccc
```

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.15 IP Group Parameter Representation in Message Manipulation

This feature provides support for using the following additional IP Group table parameters in SIP message manipulation rules:

- 'Tags' parameter:
  - param.ipg.src.tags
  - param.ipg.dst.tags
  - param.ipg.src.tags.<tag name>
  - param.ipg.dst.tags.<tag name>
- 'Name' parameter:
  - param.ipg.src.name
  - param.ipg.dst.name

Applicable Products: All.

2.7.1.16 Message Manipulation for SDP Origin Username

This feature provides support for using the username in the Origin field ("o=") of the SDP body in SIP messages, for message manipulation. The new syntax is:

```
param.message.sdp.originusername
```

Applicable Products: All.

2.7.1.17 Enhanced ISUP Body Message Manipulation

This feature provides the following enhanced manipulation support for SIP-ISUP interworking:

- SIP 200 OK with the CON (Connect) ISUP message type. This is applicable to the Spirou variant.
- Additional SIP INVITE with the IAM (Initial Address Message) ISUP message type:
  - Access transport (See 4.5.19 of Recommendation Q.931)
  - User service information (see 3.57 of Q.763)

Applicable Applications: All.

Applicable Products: All.

2.7.1.18 IP Group Parameter Representation in Call Setup Rules

This feature provides support for using the below additional IP Group table parameters in Call Setup Rules (CSR). These parameters can be used to specify values in the 'Search Key', 'Conditions' and 'Action Value' fields of Call Setup Rules to represent the IP Group of the incoming call.

- param.ipg.src.user
- param.ipg.src.host
- param.ipg.src.type
- param.ipg.src.id
- param.ipg.src.tags
- param.ipg.src.name

- param.ipg.src.user-defined.0
- param.ipg.src.user-defined.1

Note that this feature is already supported for manipulation rules.

Applicable Applications: SBC.

Applicable Products: All.

2.7.1.19 Maximum Characters for "o" Field in SDP Body

This feature provides support for configuring the maximum number of characters allowed in the SDP body's "o=" (originator and session identifier) field for the session ID and session version values. The feature is supported by the new ini file parameter, MaxSDPSessionVersionId - valid range 1,000 to 214,748,3647 (default). An example of an "o=" line with session ID and session version values is shown below:

```
o=jdoe 2890844526 2890842807 IN IP4 10.47.16.5
```

Applicable Applications: All.

Applicable Products: All.

2.7.1.20 Detection of Pulse Dialing

This feature provides support for detecting pulse (rotary) dialing from analog equipment (e.g., telephones) that are connected to the device's FXS ports. The new parameter, EnablePulseDialDetection enables the feature.

Note that the feature is already supported in earlier releases by MP-1xx.

Applicable Applications: Gateway.

Applicable Products: MP-1288.

2.7.1.21 Prefix String for External Line Enhancement

This feature provides enhanced support for the prefix string used to access an external line (configured by the existing Prefix2ExtLine parameter). An additional option has been added (2) to the existing AddPrefix2ExtLine parameter that enables the device to not only include this prefix string in the called number sent to the IP destination, but also to use the prefix string in Digit Maps and Dial Plans. This is useful in that it allows the configuration of separate digit map and/or dial plan patterns for internal and external dialing (where the first digit of the pattern is the prefix string). For more information, refer to the *User's Manual*.

Applicable Applications: Gateway (FXS).

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000; MP-1288.

2.7.1.22 MWI Notification Timeout on Endpoint Equipment

This feature provides support for configuring the maximum duration (timeout) that a message waiting indication (MWI) is displayed on endpoint equipment (i.e., phones' LED, screen notification or voice tone). If the timeout has not yet expired for an MWI and the endpoint receives a new MWI, the timeout restarts its countdown. The feature is supported by the following new parameters:

- Global parameter - 'MWI Notification Timeout' (MWINotificationTimeout)
- Tel Profile - - 'MWI Notification Timeout' (TelProfile_MWINotificationTimeout)

The value range is 1 to 2,000,000 seconds. The default is 0 (i.e., unlimited).

Applicable Applications: Gateway (FXS).

Applicable Products: MP-1288; Mediant 5xx; Mediant 8xx; Mediant 1000B.

2.7.1.23 Ringback and Held Tones per User

This feature provides support for playing different user-defined ringback and held tones per user (i.e., IP Group). This is achieved by loading the device with a Prerecorded Tone file (PRT) with the different tones, and then configuring an IP Profile (associated with the IP Group) with the index of the required ringback and/or held tone as defined in the PRT file.

To support the feature, the following new parameters have been added to the IP Profile table:

- Local RingBack Tone Index (IPProfile_LocalRingbackTone): Defines the ringback tone that you want to play from the PRT file. The tone is configured by the index number (0-79) where it is defined in the PRT file. By default, the device plays a default ringback tone.
- Local Held Tone Index (IPProfile_LocalHeldTone): Defines the held tone that you want to play from the PRT file. The tone is configured by the index number (0-79) where it is defined in the PRT file. By default, the device plays a default held tone.

Up to 80 user-defined tones can be created in the PRT file. The prerecorded tones can be created using a standard third-party, recording utility (such as Adobe Audition), and then combined into a single and loadable file (PRT file), using the latest version of AudioCodes DConvert utility.

Applicable Applications: SBC and Gateway.

Applicable Products: All.

2.7.1.24 Retry Time Enhancement for Registration Failures

This feature provides enhanced support regarding registration failure and a dynamic interval (time to wait) between the device's subsequent registration attempts. The feature is applicable only to registrations initiated by the device on behalf of SIP entities (for example, User Info, Accounts, Endpoints or the device itself) with a SIP proxy server (registrar).

Up until now, the interval between registration attempts due to a registration failure could only be configured (by the RegistrationRetryTime parameter) as a fixed interval (e.g., every 30 seconds). The new feature now enables the device to perform registration attempts at intervals that increase for each failed subsequent registration attempt (per RFC 5626, Section 4.5) for the specific registration flow.

The feature is supported by the new parameter, 'Max Registration Backoff Time' (MaxRegistrationBackoffTime), which operates together with the existing RegistrationRetryTime parameter. When the MaxRegistrationBackoffTime parameter is configured, the wait-time before another registration attempt increases after each failed registration, until it reaches the maximum value specified by the parameter. The device uses the following algorithm to calculate an incremental augmented wait-time between each registration attempt:

```
Wait Time = min (max-time, (base-time * (2 ^ consecutive-
failures)))
```

Where:

- *max-time* is the value configured by MaxRegistrationBackoffTime
- *base-time* is the value configured by RegistrationRetryTime

For example, if *max-time* is 1800 seconds and *base-time* is 30 seconds, and there were three registration failures, then the upper-bound wait time is the minimum of (1800, 30*(2^3)), which is (1800, 240) and thus, the minimum of the two values is 240 (seconds). The actual time the device waits before retrying registration is computed by a uniform random time between 50% and 100% of the upper-bound wait time (e.g., for 240, the actual wait-time is between 120 and 240 seconds). As can be seen from the algorithm, the upper-bound wait time never exceeds the value of the MaxRegistrationBackoffTime parameter.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.25 Random IDs in Contact Header User Part for New Registrations

This feature provides support for enabling the device to assign a random ID string value to the user part of the SIP Contact header for new user account registrations with the device. The assigned ID is unique to each user account. An example of a random ID is shown (in bold) below:

```
Contact: <sip:HRaNEmZnfX6xZl4@pc33.atlanta.com>
```

The feature is supported by the new ini file parameter, UseRandomUser, where 0 is disable (default) and 1 is enable. When enabled, all INVITE messages for these new user accounts are sent with their unique ID. The IDs are also used for registration refreshes and for un-registering these accounts. The IDs apply until the parameter is disabled. When enabled again, new random ID strings are assigned.

Applicable Applications: All.

Applicable Products: All.

2.7.1.26 Unregistration of User Accounts upon Device Reset

This feature provides support for deregistering all user accounts that were registered with the device, upon a device reset. However, during device start-up, each account sends a REGISTER message (containing "Contact: *") to unregister all contact URIs belonging to its Address-of-Record (AOR), and then a second after they are unregistered, the device re-registers the account. The feature is supported by the new ini parameter, UnregisterOnStartup, where 0 is disable (default) and 1 is enable.

Applicable Applications: All.

Applicable Products: All.

2.7.1.27 Register "Stickiness" to Registrar Server

This feature provides support for configuring the device to always route SIP requests of a registered Account to the same registrar server to where the last successful REGISTER request was routed. In other words, once initial registration of the Account to one of the IP addresses in the Proxy Set is successful (i.e., 200 OK), binding ("stickiness") occurs to this specific address (registrar). All future SIP requests (INVITEs, SUBSCRIBEs and REGISTER refreshes) whose source and destination match the Account are sent to this registrar only. This applies until the registrar is unreachable or the register refresh fails, for whatever reason.

Up until now (and when the feature is disabled), after a successful initial registration, whenever the device received a SIP request or registration refresh no binding happened to any specific IP address in the Proxy Set and the device simply sent the request to the currently working registrar. In the case of proxy load-balancing, there was no certainty to which IP address in the Proxy Set the request would be routed.

The feature applies to Accounts and is enabled in the Accounts table using the new parameter, 'Registrar Stickiness' (Account_RegistrarStickiness). For the feature to function, the existing 'Register' parameter must also be enabled (Regular or GIN) in the Accounts table.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.28 Registrar Search Method for Registrar "Stickiness"

This feature provides support for configuring the method for choosing an IP address (registrar) in the Proxy Set to which the Account initially registers and refreshes registration, when the Register Stickiness feature is enabled. Once chosen, this is the IP address to which the Account is then bound for subsequent SIP requests.

- Current Working Server: For each initial and refresh registration request, the device

routes to the currently working server in the list of IP addresses (configured or DNS-resolved IP addresses) of the Proxy Set. In the case of proxy load-balancing, the chosen IP address is according to the load-balancing mechanism.

- According to IMS Specifications: For the initial registration request, the device performs DNS resolution if the address of the Proxy Set is configured with an FQDN. It then attempts to register sequentially to the list of DNS-resolved addresses (or configured IP addresses). If an address results in an unsuccessful registration, the device immediately tries the next address (without waiting any retry timeout). The device goes through the list of addresses until an address results in a successful registration. If the registration process is unsuccessful for all the addresses, the device waits a configured retry time and then goes through the list again. Once initial registration is successful, periodic registration refreshes are performed as usual. In addition to the periodic refreshes, immediate register refreshes are done upon the following triggers according to the IMS specification:
  - The device receives a SIP 408, 480, or 403 response from the serving IP Group in response to an INVITE.
  - The transaction timeout for an INVITE sent to the serving IP Group expires.
  - The device receives an INVITE from the serving IP Group from an IP address other than the address to which it is currently registered. In this case, it also rejects the INVITE with a SIP 480 response.

The feature applies to Accounts and is enabled in the Accounts table using the new parameter, 'Registrar Search Mode' (Account_RegistrarSearchMode).

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.29 Registration Event Package Subscription for Registrar "Stickiness"

This feature provides support for the device to subscribe to the registration event package service (RFC 3680) with a registrar server to which an Account has been successfully registered, when the Registrar Stickiness feature is enabled. This subscription allows the device to receive the registration state of the Account registered with the server.

When enabled, the device subscribes to this service by sending a SUBSCRIBE message containing the Event header set to "reg" (Event: reg). Whenever a change occurs in the registration binding state, the server notifies the device by sending a SIP NOTIFY message.

The feature is enabled by the new parameter, 'Reg Event Package Subscription' (Account_RegEventPackageSubscription).

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.30 High-Availability Disconnect

This feature provides support for disconnecting two devices operating as a 1+1 High-Availability (HA) system, returning them to stand-alone devices. The feature is enabled from the active device, using the following CLI command:

```
# debug HA disconnect-system < new OAMP address of redundant
device>
```

For more information, refer to the *User's Manual*.

Applicable Applications: SBC/Gateway.

Applicable Products: HA Systems.

2.7.1.31 Enhanced HA Keep-Alive

This feature provides support for an enhanced HA keep-alive mechanism, which improves the keep-alive mechanism.

Note: UDP ports 670 and 680 have been added for HA operation on the Maintenance interface. Make sure that these two ports are opened (allowed) between the Active and Redundant Maintenance interfaces.

Applicable Applications: SBC/Gateway.

Applicable Products: HA Systems.

2.7.1.32 OVR Support in High-Availability Mode

This feature provides support for One-Voice Resiliency (OVR) application when the device operates in High-Availability (HA) mode. OVR is supported in HA mode in both Normal and Survivability (Limited Service) modes.

Note that besides the usual OVR and HA configuration, the only special configuration for OVR support in HA is to configure the IpProfile_SBCSessionExpiresMode parameter to "Observe" for the IP phone's IP Profile. This is needed to avoid the scenario of calls being "stuck" (never released by receiving BYE from phone or Microsoft server) for phones that were in a call before the HA switchover and that fail to register after the switchover.

Following HA switchover, all the IP Phones in the OVR network register again and normal operation resumes within 90 seconds.

Applicable Products: Mediant 800B with OVR.

2.7.1.33 SIPRec Session Capacity Increase

This feature provides for an increase in the maximum number of supported concurrent SIPRec sessions to 20,000 for Mediant 9000 and 12,000 for Mediant VE/SE.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.7.1.34 Skype User Presence Notification for Non-Skype Endpoint Devices

This feature provides support for the device to notify the Microsoft Skype for Business Server of the presence status ("on-the-phone") of Skype for Business users when making and receiving calls using third-party (non-Skype for Business) endpoint devices (such as mobile phones and PBX phones). Up until now, presence status was handled solely by the Skype for Business Server to reflect calls between Skype for Business endpoints only (e.g., Skype for Business desktop clients).

The device uses SIP PUBLISH messages to update the Skype for Business Server of presence status changes. The Skype for Business Server then publishes the presence status to all the Skype for Business users, which is displayed on their native Skype for Business endpoints.

To support the feature, the following new configuration items have been added:

- Global parameters:
  - Presence Publish IP Group ID (PresencePublishIPGroupId) – indicates the IP Group (by ID) of the Skype for Business Server (presence server)
  - Enable MsPresence message (EnableMSPresence) - enables the feature
- Call Setup Rules table - new parameters added for 'Action Subject' field (read/write):
  - presence.src – retrieves source (caller) user's Skype for Business URI through LDAP query (used for the Request URI, and From/To headers in the PUBLISH message)

- presence.dst – retrieves destination (called) user's Skype for Business URI through LDAP query (used for the Request URI, and From/To headers of the PUBLISH message)

Note:

- The support is also applicable to Lync Server 2013 (Version 5.0.8308.866 and later).
- The feature requires that the “Presence gateway service” be enabled on the Skype for Business Server.
- This feature is provided by default on all products, except Mediant 500 E-SBC and Mediant 500L Gateway & E-SBC for which it is a licensed feature (needs to be purchased).

Applicable Applications: SBC/Gateway (Tel-to-IP Calls).

Applicable Products: All.

2.7.1.35 SIP-based Private Wire Interworking

This feature provides support for interworking signaling for Private Wire services, where one side is a legacy digital PSTN equipment using E1/T1 CAS, and the other side an IP-based Private Wire session manager using SIP. The feature enables private wire services to migrate to IP-based private wires without replacing existing, legacy TDM networks.

Private Wire is a generic term used to describe static point-to-point voice connections between two locations. Private Wires are used by a number of communities such as military, railways, and financial services (e.g., turret trading systems). The telephone lines between users are "always" connected and no dialing is necessary. The private-wire signaling standard allows users to signal certain events to one another using the SIP INFO message (with an XML schema in the body). These events include Hook Switch (On/Off) states and Ringdown states (No Ring/Ring).

The feature is supported by the new optional value, "Private Wire" for the existing global parameter 'Enable TDM Tunneling' (EnableTDMoverIP). TDM tunneling can now also be enabled per trunk, using the new ini file parameter, EnableTDMOverIPforTrunk.

Applicable Applications: Gateway (E1/T1 CAS and IP-to-Tel calls).

Applicable Products: Mediant 500 E-SBC; Mediant 500 MSBR; Mediant 8xx; Mediant 1000B.

2.7.1.36 Configurable Maximum Transmission Unit

This feature provides support for configuring the Maximum Transmission Unit (MTU) in bytes per VLAN (Ethernet Device). The feature is supported by the new parameter in the Ethernet Devices table, 'MTU' (DeviceTable_MTU), where the value can be 68 to 65,535 for Mediant 9000 and Mediant VE/SE, and 68 to 1,500 for all other products. The default is 1,500.

Applicable Applications: All.

Applicable Products: All.

2.7.1.37 Same VLAN ID for Multiple Ethernet Devices

This feature provides support for using the same VLAN ID for more than one Ethernet Device. Up until now, each Ethernet Device had to be configured with a unique VLAN ID (in the Ethernet Devices table).

Applicable Applications: SBC.

Applicable Products: Mediant 9000 SBC; Mediant VE/SE SBC.

2.7.1.38 SFP+ 10G Support for Network Interface

This feature provides support for optional, small form-factor pluggable (SFP+) 10Gb network cards with LX or SX transceivers. Up to two network cards can be installed in the device

instead of the existing copper GbE NICs. Each SFP card provides four SFP port pairs. Up until now, the device's support for fiber included only SFP (1Gb) network cards.

Applicable Applications: SBC.

Applicable Products: Mediant 9000 SBC.

2.7.1.39 Disable Periodic DNS Queries

This feature provides support for disabling periodic DNS queries with a DNS server performed by the device for resolving FQDNs into IP addresses. DNS queries are used, for example, for Proxy Sets that are defined with FQDNs. When disabled, DNS resolution is done only once (upon device reset, power up, or new and modified configuration) and the DNS-resolved IP addresses are then used all the time (i.e., not refreshed).

The feature is configured by the existing parameter, 'Proxy IP List Refresh Time' (ProxyIPListRefreshTime), which now can be set to 0 to disable it. Up until now, the parameter could not be disabled (periodic DNS queries was always enabled).

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.40 SBC Application Enabled by Default

This feature changes the default setting of the SBC application (EnableSBCApplication) to enabled for all devices. Up until now, the application was enabled by default for all products except the ones listed under Applicable Products (below).

Note:

- The SBC application is enabled by default only if the License Key contains at least one of the SBC-related capacity features (e.g., "SBC-Signaling"). If the License Key does not contain any SBC-related capacity values, the application is disabled.
- When upgrading the device to 7.2.150, the SBC application is enabled (even if it was disabled on the device when running the previous version).

Applicable Applications: SBC.

Applicable Products: MP-1288; Mediant 5xx; Mediant 8xx; Mediant 1000B.

2.7.1.41 Web GUI Enhancements

This feature provides support for the following main Web GUI enhancements:

- License Key page: New design, providing improved readability of features as well as icon indications of feature and capacity changes between previous License Key and newly loaded License Key.
- SBC Configuration Wizard: Accessed now from Actions drop-down list on toolbar and/or Navigation tree (Setup menu > Administration tab > Maintenance folder > Configuration Wizard). Miscellaneous changes in wizard.

Applicable Products: All

2.7.1.42 Console Access Mode

This feature provides support for configuring the access mode (VGA or RS-232) to the device's console for accessing the CLI. Up until now, the mode could be configured only through the GRUB boot-loader menu (but after a software upgrade, the mode reverted back to VGA). This feature allows you to configure the mode through CLI (configure troubleshoot > startup-n-recovery > system-console-mode) and ini file (SystemConsoleMode). The default mode is VGA. The mode is unaffected by a device reset.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant SE/VE.

2.7.1.43 Single Sign-On to Web Interface from OVOC and Mediant CCE

This feature provides support for single sign-on access to the device's Web interface from the Web-based management interfaces - AudioCodes One Voice Operations Center (OVOC) and Mediant Cloud Connector Edition Appliance. The device's Web interface layout has also been re-designed for the requirements of OVOC and Mediant CCE Appliance.

Applicable Applications: All.

Applicable Products: All.

2.7.1.44 Broadcast Indication of Firmware Upgrade

This feature provides support for displaying a message in all active CLI sessions pertaining to a specific device, notifying all the users that the device is currently uploading firmware (.cmp). Up until now, the message was displayed only in the CLI session of the user that initiated (**copy firmware** command) the firmware upload. The purpose of the message is to prevent users that are connected to the same device from resetting or powering off the device during firmware upgrade, thereby disrupting the upgrade process.

The message not only displays the upload progress, but also displays the username of the management user who initiated the upgrade and the IP address of the user's PC (or "local" if the user is connected through serial interface). Regardless of which actions the users are performing in their CLI session prior to the upgrade, the message is forcibly displayed on their CLI consoles.

Below shows an example of such a message:

```
# copy firmware from http://10.3.1.52:1400/tftp/SIP_F7.20A.335.cmp
 % Total    % Received % Xferd  Average Speed   Time    Time
Time  Current Dload  Upload   Total   Spent    Left  Speed
100 40.7M  100 40.7M    0     0  1288k      0  0:00:32  0:00:32 --
:--:-- 1979k
Firmware file http://10.3.90.52:1400/tftp/SIP_F7.20A.335.cmp was
loaded.  (user: Admin, IP local)
The system will reboot when done
DO NOT unplug/reset the device
Firmware process done. Restarting now...
Restarting......
```

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.45 Free Product Evaluation Enhancements

This feature provides support for the following enhancements concerning the free product evaluation offering:

- Up to three user registrations (far-end users) are now supported (in addition to the already supported three SBC sessions) in the default License Key. Up until this release, user registration was not allowed.
- Transcoding capabilities with the three SBC sessions are now allowed, but this requires the administrator to install a special evaluation License Key.

For more information, refer to the *Mediant Virtual Edition SBC Installation Manual*.

Applicable Applications: SBC.

Applicable Products: Mediant VE SBC.

2.7.1.46 Hitless License Key Installation for HA

This feature provides support for hitless License Key installation through the Web interface for devices in HA mode. The installation method is non-traffic affecting, employing the HA switchover mechanism to ensure that current calls are maintained. The support is provided

by a new design of the License Key page, which provides hitless and non-hitless installation options.

A new alarm has been introduced, acLicenseKeyHitlessUpgradeAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.129), which is sent when the hitless License Key update fails.

Note:

- Hitless Upgrade of the License Key is applicable only if the new License Key includes changes in **only** one or more of the following licenses: FEU, SBC, Coder Transcoding, SBC Signaling.
- A License Key sent from the License Pool Manager Server is automatically installed using the Hitless Upgrade method.

Applicable Applications: All (HA).

Applicable Products: Mediant 500 E-SBC; Mediant 800 E-SBC; Mediant 2600 E-SBC; Mediant 4000 SBC; Mediant 9000 SBC; Mediant VE/SE SBC.

2.7.1.47 SNMP Proprietary Trap Variable Bindings

This feature provides support for new variable bindings (varbinds) for proprietary SNMP traps (acTrap). Each trap is now sent with 16 varbinds (instead of 13 in previous releases). The new varbinds include:

- acBoardTrapGlobalsDeviceName (13)
- acBoardTrapGlobalsDeviceInfo (14)
- acBoardTrapGlobalsDeviceDescription (15)

Note that the device sends these varbinds with empty values; OVOC provides the proper values when sending the traps northbound.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.7.1.48 Debug for Remote Web Services

This feature provides support for debugging remote Web (HTTP) services, specifically the HTTP client. The debug level can be configured from 1 to 3 (detailed). The debug messages are sent to the Syslog server. The feature is supported by the new parameter Debug Level (RestDebugMode), where 0 disables (default) and 1 through 3 is the debug level.

Applicable Applications: All.

Applicable Products: All.

2.7.1.49 FXS Line Testing

This feature provides support for FXS line testing, where the output displays various line statuses and electrical measurements per FXS port and country coefficient USA (70) or TBR21 (66). Some of the FXS line measurements supported by this feature include:

- Hazardous Potential Tests (HPT) - hazardous AC or DC voltage is present on the tip and ring or both.
- Foreign Electromotive Force Tests (FEMT) - foreign voltage is present on the tip, ring or both
- Resistive Fault Tests (RFT) - tip or ring is shorted to ground, or they are shorted to each other
- Receiver Off-hook Tests (ROH) - one or more phones are off hook on phone line during test.
- Ringer Impedance Tests (RIT)
- AC/DC line voltage

- AC/DC line current
- Line resistance
- Line capacity

The feature is supported by the following new command:

```
# configure voip
(config-voip)# interface fxs-fxo
(fxs-fxo)# fxs-line-testing {module/port} {66|70}
```

Note:

- For Mediant 1000B, the feature is supported only on the FXS module type (GTPM01046) that supports outdoor FXS cabling.
- For Mediant 800, please contact your AudioCodes sales representative regarding the hardware versions that support this feature.

Applicable Applications: Gateway (FXS).

Applicable Products: Mediant 800; Mediant 1000B; MP-1288.

2.7.1.50 Persistent Logging of Syslog Messages on Device

This feature provides support for automatic logging of system event messages to the device's storage, where they persist even if the device undergoes a reset or powers off. Persistent logging is done by default (cannot be disabled). The feature eliminates the need for sending logged messages to third-party, logging servers (such as a Syslog server) and offers an efficient tool for troubleshooting the device. When the persistent logging storage is full, older messages are overwritten by new messages.

The device organizes the stored logged events into the following groups (categories):

- "Conf" – log messages relating to device “boot up” and application initialization, including configuration file parsing
- "Err" - log messages relating to warnings, errors, and critical severity levels
- "Ha" – log messages relating to High Availability (HA)
- "Init" - log messages relating to device startup
- "Other" - log messages that do not belong to any category above or that are system logs

The administrator can view all the logged messages or filter the logged messages by category, time, and number of last logged messages.

The feature is supported by the following CLI commands:

```
# debug persistent-log show category-list
{conf|err|ha|init|other|sip} start-date <YYYY-MM-DD> end-date
<YYYY-MM-DD> count <Number of Logs> offset <Log Index>
```

The administrator can view statistics of the persistent logging database, which displays number of incoming logs, number of logs sent to the database, and dropped logs (due to various reasons):

```
# debug persistent-log show stats
```

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant SE/VE.

2.7.1.51 Customization of Remote SIP User Agent Field in SBC CDRs

This feature provides support for customizing the title of the “Remote SIP User Agent” field in CDRs for SBC calls. CDR customization is done in the existing SBC CDR Format table. The field represents the SIP User-Agent header, which identifies the source of the SIP message. By default, the field is excluded from the CDR.

The feature is supported by the following new optional value in the SBC CDR Format table's 'Field Type' field: "Remote SIP User Agent" (818).

Applicable Applications: SBC.

Applicable Products: All.

2.7.1.52 Snapshot Load through CLI

This feature provides support for loading a snapshot of the device's system through CLI. Snapshots provide the capability of returning the device to a previous state, which is used as a rescue option if a system malfunction occurs. Up until now, a snapshot could only be loaded through the GRUB menu.

The feature is supported by the new CLI command, load-from-snapshot:

```
(config-troubleshoot)# startup-n-recovery
(startup-n-recovery)# load-from-snapshot <Name of Snapshot>
```

Note: This feature is currently not supported for HA mode.

Applicable Applications: All.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.7.1.53 Configurable Failed Restarts for Triggering Automatic Recovery

The number of consecutively failed device restarts (reboots) to trigger an automatic recovery process whereby the device is restored to the default System Snapshot, can now be configured. The feature is configured by the new parameter, MaxStartupFailAttempts.

Applicable Applications: All.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.7.1.54 Log of Loaded CLI Script File

This feature provides support for viewing the contents of the latest CLI Script file that was loaded (i.e., copy cli-script from) to the device. The device always keeps a log file of the most recently loaded CLI Script file. The feature is supported by the following new CLI command:

```
# show last-cli-script-log
```

Note that if the device resets (or powers off), the log is deleted.

Applicable Applications: All.

Applicable Products: All.

2.7.1.55 CLI Show Run Enhancements

This feature provides support for displaying the current configuration (show running-config) of the device through CLI, according to a selected main command set (Network, Troubleshoot, System, VoIP, and Data). Up until now, only the current configuration of the command sets System, VoIP, and Data could be displayed (or full configuration).

To support the feature, the **show running-config** now provides the following optional arguments:

```
# show running-config {data|full|network|system|troubleshoot|voip}
```

Applicable Applications: All.

Applicable Products: All.

2.7.2 Known Constraints

This section lists known constraints.

Table 2-9: Known Constraints in Version 7.20A.150.004

Incident	Description
-	When upgrading the device to Version 7.2.150, syslog is enabled (instead of disabled). A workaround is to manually configure the debug level to 0. **Applicable Products:** All.
-	Hitless software downgrade from Version 7.2.150 to an earlier version is not supported (the non-hitless method must be used). **Applicable Products:** HA.
-	The device does not support Hyper-Threading (HT) for Hyper-V environments (and therefore, HT must be disabled in the device's BIOS settings). **Applicable Products:** Mediant VE.
143292	Software Hitless Upgrade from any version earlier than 7.00A.082.007 to Version 7.2.150 fails (the device crashes and resets). For devices running a version earlier than 7.00A.082.007, the device must first be upgraded to Version 7.00A.082.007 and only then to Version 7.2. **Applicable Products:** HA.
144353	The On-board, Three-way Conferencing feature for Gateway calls (3WayConferenceMode =2) functions only if the device's License Key includes an SBC license. **Applicable Products:** MP-1288.

2.7.3 Resolved Constraints

This section lists constraints from previous versions that have now been resolved.

Table 2-10: Resolved Constraints in Version 7.20A.150.004

Incident	Description
143930	The time stamp in SIP PUBLISH messages are not according to the RFC 6035, resulting in incorrect reports. **Applicable Products:** SBC.
-	The device does not support Hyper-Threading (HT) for KVM-OpenStack and VMWare environments. (Now supported.) **Applicable Products:** Mediant VE.
-	High Availability (HA) for One-Voice Resiliency is not fully supported (signaling may not function correctly in certain scenarios). **Applicable Products:** HA-Supporting Devices.
124743	The device allows a certificate to be loaded even though it already exists in the device's Trusted Root Certificate store. As a result, duplicated certificates appear on the device. **Applicable Products:** All.
135242	The maximum number of characters that can be configured for SNMP community string is limited to 19 characters. (Resolved by an increase to 30.) **Applicable Products:** All.

Incident	Description
137574	When loading an incremental ini file through REST, the device crashes (and resets). **Applicable Products:** All.
137601	When an HA switchover occurs, the new active device always switches to the first entry in the RADIUS Servers table, regardless of which servers were used by the previously active device. **Applicable Products:** HA.
138854	If the device receives a SIP REFER message before the call is connected, the device rejects the message. As a result, call transfer fails. **Applicable Products:** SBC.
139371	Sometimes when the device uses DSPs, it attempts to activate the acoustic echo canceller even though it is disabled. As a result, the open channel fails (no voice). **Applicable Products:** All.
139442	When the device operates in High-Availability (HA) mode and a hitless software upgrade from an earlier version to Version 7.2.100 is done through the Web interface, the Web interface sometimes erroneously displays an upgrade failure message and that a reset must be done, even though the devices were upgraded successfully. If this occurs, refresh the browser and then log in again to the Web interface. **Applicable Products:** Mediant 500 E-SBC HA; Mediant 800 Gateway & E-SBC HA.
139544	When five three-way conferences are needed, channels for regular Gateway calls are lost (i.e., insufficient resources). **Applicable Products:** Mediant 500L.
139988	The 'Board Type' field in the Web interface's Device Information page does not reflect the modified UseRProductName parameter value. **Applicable Products:** All.
140061	SNMP walk on SysDataStatus causes syslog errors. **Applicable Products:** All.
140081	The order of configured rows in the Proxy Sets table is incorrect. **Applicable Products:** SBC.
140113	When the SNR is low, the device does not detect DTMF digits. As a result, Gateway calls fail. **Applicable Products:** MP-1288.
140547	Transcoding of G.711 to G.729 for loopback calls fail (disconnect) after an HA switchover. **Applicable Products:** HA Products.
141398	The device often crashes (and resets) due to a problem in memory handling. **Applicable Products:** Mediant VE.
141587	REST API does not support the sending (PUT method) of an incremental CLI Script file or incremental ini file to the device. **Applicable Products:** All.
141698	When the device sends a re-INVITE, it uses only the Extended\Allowed coders (instead of also the coder that it used before the re-INVITE). **Applicable Products:** SBC-WebRTC.

Incident	Description
141903	A registration failure occurs when the device's 288 endpoints attempt to register. This is due to an overload on the device. **Applicable Products:** MP-1288.
141933	Incorrect SIP REFER message handling: When the device forwards the SIP REFER message to the proxy server and the Refer-To header value is changed to the LAN address (of the proxy side), the routing fails. (Bug resolved by new "Local Host" option of the IPProfile SBCRemoteReferBehavior parameter.) **Applicable Products:** SBC.
142150	When using the SBC Configuration Wizard, the NAT Translation table is created with only one media port even though multiple users are configured. **Applicable Products:** SBC.
142222	The device does not forward the "opaque" field if it has an empty value in the received SIP Authenticate header. Instead, it removes the field, which may cause unsuccessful authentication. **Applicable Products:** All.
142440	The device can handle only up to 100 concurrent SUBSCRIBE messages. As a result, calls fail when this number is exceeded. **Applicable Products:** MP-1288.
142494	When the device sends SIP PUBLISH messages for QoS, the body of the message is removed when alternative routing is done. As a result, incorrect QoS reports are sent. **Applicable Products:** SBC.
142504	If a VLAN ID of any interface is modified to three digits (in the Ethernet Devices table), the device continually crashes (and resets). **Applicable Products:** Mediant 9000; Mediant VE/SE.
142528	In SIP-I, if the device receives a SIP 200 OK without 18x, the device erroneously attaches the SIP-I ANM message (instead of the SIP-I CON message). **Applicable Products:** SBC.
142633	In certain call-forking scenarios to five destinations, the device does not establish voice toward the correct destination. **Applicable Products:** SBC.
142665	When using REST API to query HA status, the returned value is the serial number instead of the HA status. **Applicable Products:** HA.
142924	Even though a license from the License Pool Manager is successfully applied to the device, the device incorrectly reports to AudioCodes One Voice Operations Center that the apply process is still in progress. **Applicable Products:** SBC.
142938	If the device receives in the media line ("m=") "AVP" with crypto (indicating SRTP), the device forwards it as RTP (as it was not received with "SAVP"). As a result, no voice occurs. **Applicable Products:** SBC.
143030	If the device sends an INVITE with VBD and the response does not include VBD but the same payload type, the device rejects the response. As a result, the call fails. **Applicable Products:** SBC.

Incident	Description
143054	The device is unable to parse a user part of a SIP Contact header that is greater than 300 characters. As a result, registration fails. **Applicable Products:** SBC.
143185	The automatic update feature fails when the IniFileUrl parameter is configured with a long string (greater than 512). As a result, the file does not load to the device. **Applicable Products:** All.
143245	During an HA switchover, the TCP session between the device and AudioCodes One Voice Operations Center ends and a new session is started by the Active device. However, all the calls that were active before the switchover are not reflected in AudioCodes One Voice Operations Center after the switchover. **Applicable Products:** HA.
143269	The CLI command to display active calls (show voip calls active) displays incorrect session IDs. As a result, calls cannot be tracked. **Applicable Products:** All.
143291	The legacy (old) Dial Plan configuration method does not function. As a result, related calls fail. **Applicable Products:** Mediant 9000; Mediant VE/SE.
143450	If the SIP Interface ports are modified, the device stops sending SIP OPTIONS message to the proxy server. As a result, connection to the proxy server is lost. A workaround is to reset the device. **Applicable Products:** SBC.
143525	A certain problem in the Linux kernel causes the device to reset. **Applicable Products:** Mediant VE.
143687	If the two units in an HA system (Active and Redundant) have the same License Key but where each License Key has a different Product Key, HA fails. **Applicable Products:** HA.
143696	When the management user clicks the Monitor tab in the Web interface, the device crashes (and resets). **Applicable Products:** Digital Gateways.
143768	For call forking, when the SBCRemoteMultipleEarlyDialogs parameter is enabled, the device does not forward the SIP 200 OK. As a result, the call fails. **Applicable Products:** SBC.
143808	If the 'Channel Select Mode' parameter in the Trunk Group Settings table is configured to "Ring to Hunt Group", calls from the SBC to a Gateway-type IP Group fails. A workaround is to configure the parameter with a different select mode. **Applicable Products:** SBC and Hybrid.
143813	If the OAMP interface is configured with untagged VLAN ID 1, modifying the VLAN ID (in the Ethernet Devices table) results in a loss of management connection to the device. **Applicable Products:** All.
143999	UDP port spacing can be configured to 0 in the Web interface, which is an invalid configuration. **Applicable Products:** Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

Incident	Description
144044	When the device sends SIP PUBLISH messages at the end of calls, all call-quality metric values (reported to AudioCodes One Voice Operations Center) are zero ("0"). **Applicable Products:** SBC.
144241	Under certain high-load session conditions, the device crashes (and resets). **Applicable Products:** SBC.
143342	When loading a License Key separately for each unit in the HA system, the device sends the acHASystemConfigMismatchAlarm alarm even though no mismatch exists between the License Keys of the active and redundant device. **Applicable Products:** HA.

2.8 Patch Version 7.20A.152.003

This patch version includes new features, known constraints and resolved constraints.

2.8.1 New Features

New features introduced in this version include the following:

2.8.1.1 User "Stickiness" to Registrar Server for IP Groups

This feature provides support for configuring the device to always route SIP requests of a user (belonging to a User-type IP Group) to the same registrar server in a Proxy Set (associated with a Server-type IP Group) to where the last successful REGISTER request was routed. In other words, once initial registration of the user to one of the IP addresses in the Proxy Set is successful (i.e., 200 OK), binding ("stickiness") occurs to this specific address (registrar). All future SIP requests (INVITEs, SUBSCRIBEs and REGISTER refreshes) from the user are routed (based on matched routing rule) only to this registrar.

When the feature is disabled, after a successful initial registration, whenever the device receives a SIP request or registration refresh from the user, the device sends the request to the currently active registrar. In the case of proxy load-balancing, there is no certainty to which IP address in the Proxy Set the request is routed.

This new feature applies to users belonging to a User-type IP Group that are routed to a Server-type IP Group configured for "stickiness". The "stickiness" is configured using a new IP Group table parameter, 'User Stickiness' (IPGroup_SBCUserStickiness) with optional values, "Enable" and "Disable".

The feature also functions when IP Group Sets are configured. If a user is bound to a registrar associated with a Server-type IP Group that belongs to the IP Group Set, IP Group Set logic of choosing an IP Group is ignored and requests are routed to this same registrar (associated with the IP Group).

The feature supports devices operating in HA mode. Registrar "stickiness" is retained even after an HA switchover.

Note:

- The Proxy Set associated with the Server-type IP Group must be configured with multiple IP addresses (or an FQDN that resolves into multiple IP addresses).
- The Proxy Set Hot-Swap feature (for proxy redundancy) is not supported for users that are already bound to a registrar. However, Proxy "hot-swap" can be achieved for failed initial (non-bounded) REGISTER requests. If a failure response is received for the REGISTER request and the response's code appears in the Alternative Routing Reasons table, "hot-swap" to the other IP addresses of the Proxy Set is done until a success response is received from one of the addresses. In the case of failed REGISTER refresh requests from users already bound to a registrar, no "hot-swap" occurs for that request; only for subsequent refresh requests.
- When using the User Info table, registrar "stickiness" is supported only when the user initiates the REGISTER request (i.e., the User-type IP Group's 'Registration Mode' parameter must be configured to "User Initiates Registration").
- A user's registrar "stickiness" to a specific Proxy Set's IP address ends upon the following scenarios:
  - Proxy Set modification
  - If the Proxy Set is configured with an FQDN and a DNS resolution refresh removes the IP address to which the user is bound.
  - User registration expires or the user initiates an unregister request.

Applicable Applications: SBC.

Applicable Products: All.

2.8.1.2 Trapezoid Ring Waveform Support

This feature provides support for generating trapezoid ringing for third-party, FXS analog phones that are connected to the device's FXS ports. As opposed to the normal ringing signal, which uses sinusoid waveform, some telephones require a trapezoid waveform, which provides a higher ringing signal voltage. The supported trapezoid ringing provides a ring voltage of 85Vrms and ring frequency of 20 Hz.

To support this feature, a new ini file parameter, EnableTrapezoidRing has been introduced with optional values "Enable" (trapezoid) and "Disable" (sinusoid). A device reset is required for the parameter to take effect.

Note:

- This feature is supported only on MP-1288 with Hardware Revision 2.0 or later.
- Each segment (12 ports) of an FXS Telco connector (on an FXS blade) supports up to six concurrent trapezoid ringing. If ringing is done for calls on more than this number of ports, these additional calls are rejected with a SIP response code of 503 (Service Unavailable) and a PSTN release cause code of 43 (Access information discarded).

Applicable Applications: Gateway (FXS and IP-to-Tel).

Applicable Products: MP-1288.

2.8.2 Known Constraints

This section lists known constraints.

Table 2-11: Known Constraints in Version 7.20A.152.003

Incident	Description
145029	In the Web interface, values of table fields that reference fields of other tables are not searchable using the Web interface's table search feature (i.e., the search field that appears on the same page as the table). **Applicable Products:** All.

2.8.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-12: Resolved Constraints in Version 7.20A.152.003

Incident	Description
144868	SBC call preemption (911) does not function - emergency calls fail as the device does not free required resources. **Applicable Products:** SBC.
144864	When the device forwards T.38 fax calls without DSPs, and voice quality monitoring is enabled as well as device connected to AudioCodes One Voice Operations Center, the device crashes (and resets). A workaround is to disable quality monitoring. **Applicable Products:** SBC.
144650	For the Media Transcoding Cluster application, the device can perform transcoding with coders that are not listed in the installed License Key. **Applicable Products:** MTC.
144384	Sometimes DiffServ values of SIP packets are incorrect and as a result, incorrect traffic priority is applied to SIP packets. **Applicable Products:** SBC.
144185	The device reports packet loss even when the channel is on-hold (no packets are received). **Applicable Products:** SBC.
143974	When the devices are in HA mode and a switchover is initiated, the redundant device becomes the active device, but the initial active device remains in reset mode for a long time due to a clock synchronization issue. **Applicable Products:** HA.
142978	In some scenarios, an incorrect IP Profile ("-1") is displayed on the SBC Registered Users page of the Web interface. **Applicable Products:** SBC.

2.9 Patch Version 7.20A.152.009

This patch version includes only resolved constraints.

2.9.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-13: Resolved Constraints in Version 7.20A.152.009

Incident	Description
146000	During an HA switchover, TLS certificates are removed. As a result, calls cannot be processed and the device's management interface cannot be accessed. **Applicable Products:** SBC HA.
145933	When the device is in HA mode and resets or performs an HA switchover, all "allow" rules in the Firewall table are removed. As a result, access to the device's management interface is blocked. **Applicable Products:** Mediant VE/SE HA; Mediant 9000 HA.

2.10 Patch Version 7.20A.154.007

This patch version includes new features, known constraints, and resolved constraints.

Note: This patch version is compatible with AudioCodes EMS/SEM Version 7.2.3083.

2.10.1 New Features

New features introduced in this version include the following:

2.10.1.1 Increase in CDR Fields Sent to RADIUS Server

This feature provides support for an increase in the number of CDR fields—from 40 to 128—that can be configured and sent to a RADIUS server. The fields are configured in the SBC CDR Format table (SBCCDRFormat) and therefore, up to 128 table rows can now be configured when the 'CDR Type' parameter is set to **RADIUS SBC**.

Note: The maximum RADIUS packet size is 4,096 bytes (RFC 2865). If the packet size is greater than this due to the inclusion of many CDR fields with long customized title strings, the device removes the last CDR fields and sends the Accounting-Request packet with the CDRs that meet the packet size restriction. The removed CDR fields can be viewed in the Syslog.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.10.1.2 Call Preemption for Emergency Calls by Routing Server

This feature provides support for implementing call preemption for emergency calls (such as 911) by the routing server (for example, AudioCodes ARM). If the device is enabled for call preemption for emergency calls (SBC and/or Gateway), the routing server determines whether the incoming call is an emergency call or not and if so, handles the routing decision accordingly (i.e., preempts a non-emergency call if the maximum call capacity of the device is reached in order to allow the emergency call to be routed).

The feature is supported by the following REST API enhancements:

- The REST API resource GetRoute now includes the new parameter "emergency", whose value indicates to the device whether or not ("yes" or "no") the call is an emergency call.
- The REST API URL resource */api/v1/rmConfig/globals* now includes the new parameters "preemptionmode" (enables call preemption for SBC) and "callprioritymode" (enables call preemption for Gateway). These parameters are supported by both GET and PUT methods.

Applicable Applications: SBC; Gateway (IP-to-Tel).

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 1000B; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.10.1.3 Display of Active SIPRec Sessions in CLI

This feature provides support for displaying the number of currently active SIPRec signaling sessions through the device's CLI. An active session implies that the device has sent a SIP INVITE message to the SIPRec server (SRS).

The feature is supported by the new CLI option, **siprec**:

```
show voip calls statistics siprec
SIPRec number of active sessions: 8 (redundant sessions: 0)
```

If active-standby SRS is implemented, the SIPRec sessions with the redundant (standby) SRS is shown in parenthesis.

Applicable Applications: SBC/Gateway.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.10.1.4 Number of Displayed Output Lines in CLI Terminal Window

This feature provides support for configuring the maximum number of lines (window height) displayed in the CLI terminal window (SSH and Telnet sessions) for the output of CLI commands. This settings applies to all new CLI sessions and is preserved after device resets. Up until now, the number of output lines could only be configured per CLI session (using the **window-height** command).

The feature can be configured using the following new configuration settings:

- CLI:

  ```
  configure system > cli-settings > default-window-height
  <value>
  ```

- Web interface: 'Default Terminal Window Height' (**Setup** menu > **Administration** tab > **Web & CLI** folder > **CLI Settings**)
- ini file: DefaultTerminalWindowHeight

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.10.1.5 Increase in Maximum IP Groups and Proxy Sets

This feature provides support for an increase in the maximum number of IP Groups and Proxy Sets—to 5,000—that can be configured in the IP Groups table and Proxy Sets table, respectively. The feature is applicable only to the applicable products (below) with 32-GB or 64-GB RAM. For less than 32-GB RAM, the maximum number is 1,500 IP Groups and Proxy Sets (as supported in the previous version).

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.10.1.6 Static UDP Port Assignment for SIP Signaling

This feature provides support for configuring the device to use specific, local UDP ports for SIP signaling for each SIP entity (e.g., PBX) communicating (transmit and receive) with a specific proxy server. This applies to the device's local ports on the leg interfacing with the proxy server. Up until now (and still supported if this feature is disabled), the device used the same local UDP (as well as TLS and TCP) port for all communications with the proxy server.

This feature can be used, for example, when multiple SIP entities (IP Groups) route calls to the same proxy server. In such a scenario, the device can use a different local port for each SIP entity on the leg interfacing with the proxy server. With this set up, the proxy server is thus able to identify each SIP entity based on their unique Layer-3 address (i.e., IP address + port).

The feature is configured by a new parameter in the SIP Interfaces table, 'Additional UDP Ports' (SIPInterface_AdditionalUDPPorts). The parameter is configured for the SIP Interface that is associated with the proxy server. The valid value is a range from 1025 to 65535 using the syntax *x-y* (e.g., 2000-6000). By default, the parameter is not configured. The port range must adhere to the following:

- The parameter's port range must not overlap with the UDP port configured by the 'UDP Port' parameter (SIPInterface_UDPPort).
- The parameter's port range must not overlap with UDP port ranges of other SIP Interfaces that are associated with the same network interface.
- The parameter's port range must not overlap with UDP port ranges of Media Realms that are associated with the same network interface.
- The maximum number of ports in the range is limited to the maximum number of IP Groups that can be configured.
- Only for Mediant 1000B: the end port in the port range must be less than the value of the global parameter, BaseUDPPort.

In addition, to assign a specific (static) local UDP port from the configured range to each SIP entity communicating with the proxy server, tags and Call Setup Rules are employed, using the following new Message Manipulation keywords:

- *message.incoming.local-port*: (Read-only) Contains the local port on which the SIP message is received.
- *message.outgoing.local-port*: Configures the local port on which outgoing SIP messages are sent. It can be used for "write" operations in Call Setup Rules and read-only operations for Message Manipulations.

Note that the existing message manipulation keywords "param.message.address.src.port" and "param.message.address.dst.port" have been replaced with "message.incoming.remote-port" and "message.outgoing.remote-port", respectively.

For more information on configuring this feature, refer to the *User's Manual*.

Applicable Applications: SBC.

Applicable Products: All.

2.10.1.7 Sending DTMF using both SIP INFO and RFC 2833

This feature provides support for sending DTMF digits out-of-band (not with audio stream) using both the SIP INFO and RFC 2833 methods on the same leg for the same call. Up until now, the device could use only one method for sending the DTMF digits - SIP INFO or RFC 2833 (not both). The RFC 2833 method sends out-of-band DTMF digits using the RTP protocol while the SIP INFO method sends the digits using the SIP protocol.

The feature is configured by a new parameter in the IP Profiles table, 'Send Multiple DTMF Methods' (IPProfile_SBCSupportMultipleDTMFMethods), with optional values **Enable** and **Disable** (default). To implement the feature, not only must this parameter be enabled, but the existing parameter 'Alternative DTMF Method' (IPProfile_SBCAlternativeDTMFMethod) must be configured to one of the SIP INFO values (**INFO – Cisco**, **INFO – Nortel**, or **INFO – Lucent**). In addition, sending of DTMF digits using the RFC 2833 method must be enabled (**As Is** or **Extend**), using the existing parameter 'RFC 2833 Mode' (IpProfile_SBCRFC2833Behavior).

This feature also introduces a method to stop sending the DTMF digits using the SIP INFO method when a re-INVITE is received (and keep sending the DTMF digits using the RFC 2833 method). This is done using AudioCodes proprietary SIP header, X-AC-Action in Message Manipulation rules to switch to a different IP Profile that is configured to disable the sending of DTMF digits using both methods (i.e., 'Send Multiple DTMF Methods' configured to **Disable**):

```
X-AC-Action:'switch-profile;profile-name="IP Profile Name"'
```

Note:

- It is recommended that the settings of the switched IP Profile are identical (except for the 'Send Multiple DTMF Methods' parameter) to the initial IP Profile. Different settings may adversely affect the processing of the call.
- The feature requires DSP resources (for detection and generation of RFC 2833).

Applicable Applications: SBC.

Applicable Products: All.

2.10.1.8 Termination of Call Hold and Retrieve SIP Requests

This feature provides support for terminating call hold and call retrieve (resume) SIP requests (re-INVITE or UPDATE) for SIP entities that don't support call hold. Termination is done on the device's leg interfacing with the initiator of the call hold/retrieve. Instead of forwarding the request to the SIP entity that doesn't support call hold/retrieve, the device terminates the request and replies to the initiator of the call hold/retrieve with a SIP 200 OK. Up until now, the device supported an option to terminate call hold requests only; call retrieve requests were forwarded to the SIP entity that did not support call hold.

The feature is supported by the new optional value—**Hold and Retrieve Not Supported**—for the existing parameter 'Remote Hold Format' (IPProfile_SBCRemoteHoldFormat) in the IP Profiles table.

Applicable Applications: SBC.

Applicable Products: All.

2.10.1.9 Multiple Management Interfaces

This feature provides support for configuring multiple management network interfaces for the device, allowing access to the device's Web-based management tool through different IP addresses. Each management interface can be configured to use a specific network interface (Control and/or Media type) and TLS Context, and can be configured to restrict access through HTTPS.

Up until now, the device's management interfaces could be accessed through either one IP network interface ("OAMP"), or all the network interfaces listed in the IP Interfaces table if the EnableWebAccessFromAllInterfaces parameter was configured to 1. However, this was a global setting that applied to all the network interfaces, and specific TLS Contexts and HTTP or HTTPS connectivity could not be specified.

The feature is supported by the new configuration table, Additional Management Interfaces table (AdditionalManagementInterfaces ini file parameter; CLI command - configure system > **additional-mgmt-if**), located in the Web interface under **Setup** menu > **Administration** tab > **Web & CLI**.

Note:

- This feature will be supported by AudioCodes ARM (REST API) in a future release.
- Additional management interfaces can be associated only with Media and/or Control network interface types (not OAMP).
- This feature will be supported for SNMP, LDAP, RADIUS and CLI access in a future release.

Applicable Applications: All.

Applicable Products: MP-1288; Mediant 500 Gateway & E-SBC; Mediant 500L Gateway & E-SBC; Mediant 800B Gateway & E-SBC; Mediant 1000B Gateway & E-SBC; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.10.1.10 Increased Value Ranges for Proxy Online Detection

This feature provides support for an increased value range for the existing parameters 'Success Detection Retries' (ProxySet_SuccessDetectionRetries) and 'Success Detection Interval' (ProxySet_SuccessDetectionInterva). These parameters are used to ensure that connectivity with the proxy has indeed been restored.

The 'Success Detection Retries' parameter can now be configured to up to 100 retries. The 'Success Detection Interval' parameter can now be configured to up to 200 seconds.

Applicable Applications: All.

Applicable Products: All.

2.10.1.11 User Account Re-registration after Physical Link Restored

This feature provides support for the device to re-register an Account (in the Accounts table) that is configured for IMS-based registration ('Registrar Search Mode' parameter set to **According to IMS Specifications**), when the device's physical Ethernet link to the proxy is restored after a failure, even if proxy keep-alive (using SIP OPTIONS) is disabled. Up until now, re-registration due to Ethernet link restoration occurred only if keep-alive was enabled (i.e., when the link was restored the device would re-register due to successful keep-alive response).

Applicable Applications: All.

Applicable Products: All.

2.10.1.12 Enhanced SIP REFER Handling

This feature provides enhanced support for handling SIP REFER messages (used for SBC call transfer). The device can now forward a received SIP REFER message between SIP entities without changing the host part in the SIP Refer-To header. This applies to all types of call transfers (e.g., blind and attendant transfer).

The feature is supported by a new optional value—**Keep Host** (5)—for the existing 'Remote REFER Mode' (IpProfile_SBCRemoteReferBehavior) parameter.

Applicable Applications: SBC.

Applicable Products: All.

2.10.2 Known Constraints

This section lists known constraints.

Table 2-14: Known Constraints in Version 7.20A.154.007

Incident	Description
146495	The links to the "child" tables in the TLS Contexts table are not displayed when using Mozilla Firefox Web browser. **Applicable Products:** All.
145104	If any of the physical LAN cables are disconnected from the redundant device, the active device doesn't raise an alarm to indicate this. **Applicable Products:** HA.

2.10.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-15: Resolved Constraints in Version 7.20A.154.007

Incident	Description
146330	The user is unable to log in to the CLI when login authentication is RADIUS-based, as the RADIUS access level is incorrect. **Applicable Products:** All.
146202	When a few Web sessions exist with the device, under some conditions the device crashes (resets). **Applicable Products:** All.
146135	After an HA switchover, the private key no longer matches the certificate. As a result, login to the management interface (Web) and calls fail. **Applicable Products:** Mediant 9000; Mediant VE/SE.
146110	The device changes the SIP Refer-To header (host part) even though it is configured not to. As a result, call processing is incorrect. (Resolved by a new optional value "Keep Host" (5) for the SBCRemoteReferBehavior in the IP Profile table.) **Applicable Products:** SBC.
146068	The Proxy Set alarm is not cleared after connectivity with the lost proxy is restored **Applicable Products:** All.
146053	The value range of the ProxySet_SuccessDetectionRetries parameter is insufficient (detects primary proxy is online again before reverting to it). (Parameter range has been increased). **Applicable Products:** SBC.
146034	After the device restarts, the alarms it sent previously to the AudioCodes One Voice Operations Center show a different timestamp between pre- and post- reset. **Applicable Products:** All.
145973	After an HA switchover, the default self-signed certificate is corrupted and as a result, calls fail. **Applicable Products:** HA.

Incident	Description
145969	When employing LDAP-based login authentication, Index 0 cannot be configured in the LDAP Management Server Group table. As a result, login to the device could not be done. A workaround is to use Index 1. **Applicable Products:** All.
145946	The TDM-to-SBC feature is not functional; the device does not use licenses from the TDM channel. As a result, calls fail. **Applicable Products:** Hybrid (Gateway with SBC).
145920	The device cannot be accessed through SSH (enabled), which is caused by a memory leak. **Applicable Products:** All.
145734	When using the CLI Script file for configuration backup and restore, Dial Plan Rules are not restored. **Applicable Products:** SBC.
145695	When there are calls between two channels that are handled by two different cores, the device crashes (resets). **Applicable Products:** Mediant VE SBC.
145582	When the Destination IP Group is not specified in the IP-to-IP Routing table, the wrong IP Group is chosen and as a result, SBC call routing fails. **Applicable Products:** SBC.
145577	LDAP-based routing authentication fails after a while. As a result, call routing fails. A workaround is to reset the device. **Applicable Products:** All.
145563	The device does not update the DNS IP address obtained from DHCP. A workaround is to use a static IP address. **Applicable Products:** All.
145471	The "Match Count" statistics in the Firewall table displays incorrect values. As a result, access to the device is blocked. **Applicable Products:** Mediant 9000; Mediant VE/SE.
145404	The Remote Web Services page in the Web interface is displayed corrupted. **Applicable Products:** All.
145351	IP Groups are displayed as offline in the Topology View of the Web interface after a version upgrade. **Applicable Products:** All.
145282	If the IP Group is configured as a "Gateway" type in the IP Groups table and the IP Group sends an unregister request or the IP Group type is changed (for example, to “User"), the “GW GROUP STATUS” fields in the IP Groups table is not updated and shows "registered" (even after a device reset). **Applicable Products:** SBC
145018	The SIPRec application does not respond to session timer re-INVITE or UPDATE. As a result, calls fail. **Applicable Products:** All.
144995	No corresponding CLI command for the ‘Publication IP Group ID' parameter. (Resolved - configure voip > media rtp-rtcp > publication-ip-group-ID.) **Applicable Products:** SBC.

Incident	Description
144979	During silence period from the IP side, the device sends noise to the PSTN side. A workaround is to configure the ECEnableComfortNoiseGeneration parameter to 0. **Applicable Products:** Gateway.
144634	The device's management (Web and CLI) user password cannot be configured (in clear text) in the ini file (and then loaded to the device). **Applicable Products:** All.
144600	The device does not correlate between the incoming call and the received SMDI message. As a result, the call fails. **Applicable Products:** Gateway (with SMDI).
144586	The device doesn't forward RTP to a user that is located behind NAT when the NATMODE parameter is configured to 3 (By Signaling). As a result, no voice occurs. **Applicable Products:** SBC.
144504	When adding a new Web user to the active device (in the Local Users table), the new user is not added to the redundant device after an HA switchover. As a result, the user is unable to log in to the device. **Applicable Products:** HA.
144230	In high load conditions (CPU overload), the device sends TCP window of 0. As a result, requests were dropped. **Applicable Products:** SBC.
144041	When the device rejects a call due to Classification failure, the device does not include the session ID in the report sent to AudioCodes One Voice Operations Center at the end of the call. **Applicable Products:** SBC.
143394	The user is unable to configure the 'Silence Detection Method' (FarEndDisconnectSilenceMethod) parameter. (This parameter erroneously appears in the Web interface and is not applicable). **Applicable Products:** MP-1288; Mediant 5xx; Mediant 8xx; Mediant 2600/4000; Mediant 9000; Mediant VE/SE.

2.11 Patch Version 7.20A.154.044

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes EMS/SEM Version 7.2.3088.

2.11.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-16: Resolved Constraints in Version 7.20A.154.044

Incident	Description
145877	If during an established call the call mode changes from RTP forwarding to transcoding (trans-rating), the device crashes (and resets). **Applicable Products:** SBC.
146815	The remote side opens TCP connections (per SIP dialog), but does not close them. When the device tries to close the connections, it crashes (and resets). **Applicable Products:** All.
147019	When the device functions as a DHCP server, under a high load of DHCP requests, the device crashes (and resets). **Applicable Products:** All.
146955	After the device is upgraded, it sends an alarm indicating that the IP Group is blocked (which it is not). **Applicable Products:** All.
146955	When using quality of service (QoS) rules with SNMP, the device crashes (and resets). **Applicable Products:** All.
146865	Message Manipulation rules (MessageManipulations) with "\" do not load when loading an ini file. A workaround is to use "\\" (double backslash). **Applicable Products:** All.
146969	The device rejects a user registration if the Contact header of the new REGISTER request already exists for another AOR. **Applicable Products:** SBC.
146809	If a SIPRec re-INVITE is sent after a REFER failed, the device crashes (resets). **Applicable Products:** SIPRec Supporting Devices.
146792	Ethernet port information is not displayed in the Web interface. **Applicable Products:** All.
146749	The device crashes (and resets) due to timing issues between device processes. **Applicable Products:** SBC.

Incident	Description
146791	No voice occurs in the following scenario: The device is defined to play RBT. When it receives 180 without SDP, it disconnects the voice stream and plays RBT. When the 200 OK is received with the same SDP version, the device stops playing the RBT, but does not reconnect the voice stream. A workaround is to configure the 'SBC Remote Can Play Ringback' parameter to Yes. **Applicable Products:** SBC.
146827	Due to certain operations in debug recording, the device crashes (and resets). **Applicable Products:** All.
146614	The CLI command to copy the CLI Script file to FTP does not function. **Applicable Products:** Mediant 1000.
145136	Configuration transfer from AudioCodes One Voice Operations Center to the device through HTTPS fails. A workaround is to load the configuration manually. **Applicable Products:** Mediant 4000.
146485	When there is redundant AudioCodes One Voice Operations Center License Pool Manager server, the device fails to communicate with it. A workaround is not to use a redundant License Pool Manager server. **Applicable Products:** SBC.

2.12 Patch Version 7.20A.154.052

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.321 and EMS/SEM Version 7.2.3088.

2.12.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-17: Resolved Constraints in Version 7.20A.154.052

Incident	Description
146684	When an HA switchover occurs, no voice occurs on existing calls due to a problem in Generic Attribute Registration Protocol (GARP) timing. **Applicable Products:** SBC HA.
147408	When a user is located behind NAT and a re-INVITE changes the media port, the device sends the RTP to an incorrect destination. As a result, one-way voice occurs. **Applicable Products:** SBC.
147398	When using the SIPRec feature, certain SIPRec calls do not disconnect correctly and as a result, the device crashes (resets). **Applicable Products:** SBC.
145877	If an ongoing call changes from RTP forwarding to transcoding (transrating), the device crashes (resets). **Applicable Products:** SBC.

2.13 Patch Version 7.20A.154.059

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.321 and EMS/SEM Version 7.2.3088.

2.13.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-18: Resolved Constraints in Version 7.20A.154.059

Incident	Description
147735	The device does not clear registration entities from its database after un-REGISTER requests. As a result, it cannot register new users due to database being at maximum registration entries. **Applicable Products:** SBC.

2.14 Patch Version 7.20A.156.009

This patch version includes new features, known constraints, and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.2.3083.

2.14.1 New Features

New features introduced in this version include the following:

2.14.1.1 Port Assignment per Registered User

This feature provides support for assigning a unique, local UDP port (for SIP signaling) per registered user. Up until now, SIP messages received from all the registered users (User-type IP Group) were sent to and received from the proxy server (Server-type IP Group) on the same local UDP port configured for the SIP Interface (UDP Port parameter) associated with the Proxy Set of the proxy server.

With this feature, the device assigns each registered user a unique local port from a configured port range, and traffic between the user and proxy server is sent and received on the unique port (on the leg interfacing with the proxy server).

To support this feature, a new parameter—User UDP Port Assignment (IPGroup_UserUDPPortAssignment; user-udp-port-assignment)—has been added to the IP Groups table. The parameter must be enabled for the IP Group of the proxy server. In addition, the port range from which the device allocates unique ports to each user is configured by the existing parameter, Additional UDP Ports of the SIP Interfaces table (for the SIP Interface associated with the proxy server).

The device assigns a unique port upon the first REGISTER request received from the user. Subsequent SIP messages other than REGISTER messages (e.g., INVITE) received from the user are sent to the proxy server on this unique local port. The device rejects the SIP request if there are no free ports available for use (due to the number of registered users exceeding the configured port range). The unique port is also used for registration refreshes. A registration expiry de-allocates the unique port. For SIP requests received from the proxy server and destined to the user, the local port on which they are received is irrelevant (unique port or any other port); the device does not use this port to identify the registered user.

Note:

- The feature does not apply to SIP requests received from non-registered users. For these users, the device sends all requests to the proxy server on the single port configured for the SIP Interface (UDP Port parameter).
- For HA systems, the unique port assigned to a registered user is used after an HA switchover.
- This feature is applicable only if the user initiates registration (i.e., user sends the REGISTER request). In other words, the Registration Mode parameter of the IP Group of the user must be configured to User Initiates Registration.

Applicable Applications: SBC.

Applicable Products: All.

2.14.1.2 Multiple AORs with Same Contact User

This feature provides support for handling registration and call routing when multiple AORs have the same URI in the Contact header, as shown in the example below. Such a scenario

typically occurs when two SIP endpoints reside in separate private networks and both are assigned the same local IP address.

- User 1 Registration:

```
REGISTER sip:300@10.33.4.140;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.40;branch=OTGHREPCXDBIWECOCPJK
From: <sip:300@domain1;user=phone>;tag=ULYEYCGXHXMBPSOCXVWH
To: <sip:300@domain1;user=phone>
Call-ID: XDRXGAAWNVBTFHBMQCKE@10.33.2.38
CSeq: 1 REGISTER
Contact: <sip:300@10.33.2.40>
```

- User 2 Registration:

```
REGISTER sip:300@10.33.4.140;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.40;branch=YHDWUJRMMOEIJRXVYKHD
From: <sip:300@domain2;user=phone>;tag=CVYTCHLIVMPBCGNGRTUA
To: <sip:300@domain2;user=phone>
Call-ID: INRNGFCHFHETRXAQNAIT@10.33.2.38
CSeq: 1 REGISTER
Contact: <sip:300@10.33.2.40>
```

In the above example, the device adds two AORs ("300@domain1" and "300@domain2") to its registration database, where each AOR is assigned the same Contact URI ("300@10.33.2.40").

To support this feature, the device needs to search for the user in its registration database using the full URI (user@host parts) - . Therefore, the existing parameter, SBCDBRoutingSearchMode must be configured to 1.

Applicable Applications: SBC.

Applicable Products: All.

2.14.1.3 Syntax Enhancement for Dial Plan Tags

This feature provides support for using the dot symbol (.) in Dial Plan tag values. For example, the tag can now be configured as an IP address in dotted-decimal notation (10.1.1.2). Note that the configured tag cannot start with a dot; it can be located anywhere after the first character, for example, "Country=USA.NY".

Applicable Applications: SBC.

Applicable Products: All.

2.14.1.4 DHCP Option 160 for Automatic Provisioning

This feature provides support for DHCP Option 160, which the device, as a DHCP client, can use to download software (.cmp) and configuration (.ini) files from a provisioning server. Option 160 defines the location (URL address) of the provisioning server and optionally, the names of the required files and their folder location on the server.

Upon device reset or power up, the device sends a DHCP request to a DHCP server for networking parameters (e.g., IP address). The response from the DHCP server can include the networking information as well as Option 160.

The following syntax is supported for defining the URL and configuration/firmware filenames in DHCP Option 160:

- <protocol>://<server IP address or hostname>
- <protocol>://<server IP address or hostname>/<software filename>
- <protocol>://<server IP address or hostname>/;<configuration filename>
- <protocol>://<server IP address or hostname>/<software filename>;<configuration filename>

Where *protocol* can be HTTP, HTTPS, FTP or TFTP. As shown above, a URL can include both the software and configuration filenames. In this case, they must be separated by a semicolon (;) and without spaces.

If the URL does not specify a configuration filename or the file does not exist on the provisioning server, the device requests from the server a "default" configuration file whose name includes the device's product name and MAC address (<Product><MAC>.ini, for example, "M800B00908f5b1035.ini"). If this "default" file also does not exist on the server, the device attempts to retrieve another "default" configuration file whose name includes only the device's product name (<Product>.ini, for example, "M800B.ini"). The device makes up to three attempts to download the configuration file if a failure occurs (i.e., file not exist or any other failure reason). This applies to each of the configuration files, as mentioned previously.

If the URL specifies a software file, the device makes only one attempt to download the file (even if a failure occurs). If the URL does not specify a software file, the device does not make any attempt to download a software file.

Once the device downloads the file(s), it undergoes a reset to apply the configuration and/or software. In addition, once the file(s) has been downloaded, the device ignores all future DHCP Option 160 messages. Only if the device is restored to factory defaults will it process Option 160 again (and download any required files).

To support the feature, the new parameter, DhcpOption160Support has been introduced, with optional values 0 to disable (default) and 1 to enable DHCP Option 160 handling. A device reset is required for the parameter to take effect.

Applicable Applications: SBC & Gateway.

Applicable Products: All.

2.14.1.5 ENUM Queries for Call Setup Rules

This feature provides support for configuring Call Setup rules to query ENUM servers and to handle responses from ENUM servers. ENUM translates ordinary telephone numbers (E.164 telephone numbers) into Internet addresses (SIP URIs), using the ENUM's DNS NAPTR records. Once resolved into a URI, the device can route the call to this destination address.

To support the feature, the Call Setup Rules table's 'Query Target' parameter has a new optional value, **ENUM** (3). In addition, in order to use the query result, the new Call Setup rule keywords have been introduced: "enum.result.url" and "enum.found" (if condition for ENUM located for the number). The ENUM server's address is defined for the IP Interface used for the call.

Applicable Applications: SBC.

Applicable Products: All.

2.14.1.6 Message Conditions for Starting/Stopping SIPRec Sessions

This feature provides support for assigning a Message Condition rule (configured in the Message Conditions table) to a SIP Recording rule (SIPRec) in the SIP Recording Rules table. The Message Condition rule defines the condition for starting and stopping a SIPRec session. Only if the condition is met will the device start the SIPRec session.

To support the feature, a new parameter, 'Condition' (SIPRecRouting_ConditionName) has been added to the SIP Recording Rules table, which assigns a Message Condition rule. For this feature, only the following keywords can be used in the syntax in Message Condition rules:

- var.global
- var.session.0
- srctags/dsttags (only SBC too).

The feature is typically configured using Message Condition rules together with Call Setup rules (CSR). For example, the CSR can assign the "srctags" tag with the value "record" if the SIP message contains a header "X-Record:yes". The Condition rule can then define the

condition srctags=='record'. If the condition is met, the device will start a SIPRec session for the SIP dialog session.

Applicable Applications: SBC & Gateway.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.14.1.7 SIP Classification by IP Address and Contact Header

This feature provides support for classifying incoming SIP dialog messages based on the combination of source IP address and URI of the Contact header. The feature applies to User-type IP Groups that represent multiple users. Therefore, multiple users can now be registered from a single IP address when using SIP Connect. Up until now, SIP Connect classification was based only on IP address.

To support this feature, the existing IP Group table parameter, 'SIP Connect' (IPGroup_SIPConnect) has a new optional value, **Classify by IP and Contact** (2). The existing optional value, **Yes** (1) has been renamed to **Classify by IP**.

For initial user registration:

- If configured to **Classify by IP**, the device adds a key representing the user to its registration database based on the REGISTER request's source IP address, port (if UDP) and SIP Interface ID (e.g., "10.33.3.3:5010#1"). The device rejects initial registration requests that have the same IP address, as the necessary key is already used for another registration.
- If configured to **Classify by IP and Contact**, the device adds a key representing the user to its registration database based on the URI of the Contact header, source IP address, port (if UDP) and SIP Interface ID (e.g., "user@host.com#10.33.3.3:5010#1"). The device rejects initial registration requests that have the same IP address and Contact URI, as the necessary key is already used for another user registration.

Applicable Applications: SBC.

Applicable Products: All.

2.14.2 Known Constraints

This section lists known constraints.

Table 2-19: Known Constraints in Version 7.20A.156.009

Incident	Description
147612	For the SIP call-flow feature where the device sends SIP messages to OVOC, for messages that undergo authentication, the device only sends OVOC the SIP messages from the INVITE that is sent with the user's credentials (i.e., initial INVITE and subsequent SIP 4xx authentication responses are not sent). **Applicable Products:** All.
148119	Mediant VE SBC with 1 vCPU / 2-GB RAM is not supported. **Applicable Products:** Mediant VE SBC.
147892	The device does not support HA mode. **Applicable Products:** Mediant 800.
148296	The configured value of the 'Name' field in the IP Groups table and Proxy Sets table cannot end with a space or tab. **Applicable Products:** All.
154386	For SBC calls that have failed the device's routing stage, OVOC's SIP Call Flow diagram displays the initial SIP INVITE and 100 Trying lines, but not the final error message "404 Not Found". This applies only to calls which invoke Call Setup Rules involving LDAP, HTTP or ARM queries. **Applicable Products:** All.

2.14.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-20: Resolved Constraints in Version 7.20A.156.009

Incident	Description
148524	In certain scenarios, the license (with license pool) is not synchronized correctly between the two devices in the HA system after an HA switchover or a reset. **Applicable Products:** HA.
143959	After a power off and then on, the device requests a license from the EMS (OVOC), as due to the connection lost with the EMS, the device lost its current license. A workaround is to assign the license manually from the EMS. **Applicable Products:** SBC (with License Pool).
146860	If a user sends a SIP REGISTER message to a server through the device and the server responds with a SIP 401, the user sends a new REGISTER, but when the device forwards it to the server, it removes the proprietary body from the message. As a result, incorrect information is sent to the server. **Applicable Products:** SBC.
148136	For direct-media calls, the device does not release allocated resources. As a result, calls fail. **Applicable Products:** SBC.

Incident	Description
147262	In scenarios where the device receives a call from the Tel side and sends forking INVITES to multiple destinations (belonging to the same IP Group) and implements CSR for manipulation, the device uses the incorrect CSR rule, resulting in the calls being sent with an incorrect destination number. **Applicable Products:** Gateway.
148519	The ENUM feature with Call Setup Rules does not function with Gateway calls (only SBC calls). **Applicable Products:** All.
148366	When the device receives a large INVITE (greater than 4k), it does not send a route request (getRoute) to ARM and as a result, the call fails. **Applicable Products:** All (SBC with ARM).
148309	In an SRTP call, if the crypto key changes before an HA switchover, then after the switchover the device uses the old keys. As a result, no voice occurs. **Applicable Products:** HA.
147402	For sent Syslog CDRs, the device truncates CDR field values that contain many characters. **Applicable Products:** All.
147975	The CLI command **clear voip calls** does not function properly and when run the CLI "freezes" and need to press CTRL+C to release it. **Applicable Products:** All.
147494	The parameter **asserted-identity-m** does not appear in the CLI. **Applicable Products:** All.
147877	When the device receives an SNMP Get for the MIB of Channel Status for a specific CID, it includes the "invalid input parameter" message in the sent Syslog. **Applicable Products:** Mediant VE/SE; Mediant 9000.
147708	When forwarding SBC messages, the device does not change the media IP address in the SDP body to its own IP address. As a result, no voice occurs. **Applicable Products:** All.
147508	When connection to the LDAP server is lost, the device responds very late for implementing SBC alternative routing. **Applicable Products:** All.
147491	When the query response from the LDAP server is delayed, the device switches to SBC alternative routing after 10 seconds instead of waiting 16 seconds, resulting in incorrect routing. **Applicable Products:** All.
147463	When the device performs SBC outbound manipulation and it receives a SIP NOTIFY request with an MWI body that contains “Messages-Waiting: Yes”, it erroneously forwards the NOTIFY request with “Messages-Waiting: No”. As a result, MWI does not occur. **Applicable Products:** All.
147415	If the plus (+) character is included in an SBC tag name, the "dialplan.result" cannot be used. **Applicable Products:** All.
147402	Tags defined with many characters are truncated in generated customized CDR. **Applicable Products:** All.

Incident	Description
147372	The displayed coder in the Web interface's Monitor page is incorrect. **Applicable Products:** MP-1288.
147341	When running the CLI command **show voip dsp status** when there are no DSPs causes the device to crash. **Applicable Products:** Mediant 4000.
147298	When the device's ports are configured to Gigabit speed, the MAC addresses of the ports are not detected and as a result, no voice occurs. A workaround is to use auto-negotiating. **Applicable Products:** All.
147197	When filtering display (**show** command) in the CLI, filtering using **grep** does not function properly. **Applicable Products:** All.
147087	During a test call, the device crashes (and resets). **Applicable Products:** All.
146060	When the device acts as DHCP server and there is a high load of DHCP and ARP requests, it crashes (and resets). **Applicable Products:** All.
147017	If the device is configured in proxy mode and it receives a SIP 18x for forking with Record-Route header, it sends the response without the header. As a result, the SBC call fails. **Applicable Products:** All.
147015	If the device receives a SIP 491 response to an INVITE, it sends the new INVITE with the same CSEQ as the previous INVITE. As a result, the SBC call fails. **Applicable Products:** All.
146984	During certain LDAP operations, the device crashes (and resets). **Applicable Products:** All.
146956	The device does not assign an IP Profile to an incoming REGISTER message received from an IP Group for SBC calls. **Applicable Products:** All.
146926	For IP-to-Tel calls, if the device receives an ISDN FACILITY message, the device uses the wrong contact in the outgoing SIP 302. As a result, the call is not routed correctly. **Applicable Products:** Digital Gateway.
146888	The device sends "Realm" in the SIP WWW-authenticate header, but the remote side expects "realm" (lower case). As a result, authentication fails for SBC calls. **Applicable Products:** All.
146869	The ini file parameter AGGRESSIVEDTMFERASURE has no corresponding command in the CLI. **Applicable Products:** Gateway.
146842	The device discards incoming messages that are larger than 260 bytes. As a result, calls fail. **Applicable Products:** Digital Gateway.

Incident	Description
146823	When the parameter BrokenConnectionEventTimeout is configured on the HA system, the configuration change is not sent to the redundant device. **Applicable Products:** HA.
146813	If any channel command (such as Open-Channel or Avtivate-RTP) fails during the Hitless Software Upgrade, the process fails. As a result, HA fails. **Applicable Products:** HA.
146808	The IDS Policy blacklist does not function - blocked users can access the device when using TCP. **Applicable Products:** Mediant 4000.
146805	When working with the SBC Configuration Wizard, a problem exists when selecting a template. **Applicable Products:** All.
146772	If the LDAP server doesn't return a value for the attribute, the device considers the string as 'NULL' and returns FALSE. As a result, incorrect LDAP handling. **Applicable Products:** All.
146587	The device responds with the incorrect crypto tag in the INVITE with SRTP, causing a voice problem in SBC calls. **Applicable Products:** All.
146325	Incorrect handling of incoming SBC packet causes the device to crash (reset). **Applicable Products:** All.
146303	When the device receives an INVITE with "a=maxptime:40", it responds with the incorrect ptime (ptime = 40) for SBC calls. **Applicable Products:** All.
146189	When CRP is configured with SBCKeepContactUserinRegister = 2 and a call is forked to two IP phones (same AOR but different contact), the call is disconnected after 30 seconds. **Applicable Products:** CRP.
146096	The device reports overload even when there is no SBC traffic. **Applicable Products:** All.
146072	If the device sends an un-REGISTER due to a session expire and receives a SIP 401 in response, when the new REGISTER arrives, the device does not route SBC messages correctly. **Applicable Products:** All.
145848	The device fails in some vulnerability scan, causing a security risk. **Applicable Products:** All.
145765	If the user attempts to import an invalid TLS certificate, the load fails without any notification of the reason to the user. **Applicable Products:** All.
145401	The Details tab of some tables in the Web interface is not displayed. A workaround is to delete the Web browser cache. **Applicable Products:** All.
144652	In certain situations when one of the device crashes, HA mode does not recover. **Applicable Products:** HA Devices.

Incident	Description
144187	CLI erroneously displays that the **ping** command can ping a hostname (but, it can only ping an IP address). **Applicable Products:** All.
143766	The device sends an alarm to the EMS that its fan is not operating, even though it is working. **Applicable Products:** Mediant 4000.
148054	After performing a hitless software upgrade and the device uses a License Pool, the Remote Web Services table shows incorrect configuration even though the device operates normally. A workaround is to wait 15 minutes after the upgrade completes, and then in EMS/OVOC license pool page, right-click the device, and choose Update MG to download a new License Pool. **Applicable Products:** HA Devices.

2.15 Patch Version 7.20A.156.023

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.2.3104.

2.15.1 New Features

New features introduced in this version include the following:

2.15.1.1 ENUM Query Enhancement for Call Setup Rules

The address of the ENUM server to which the Call Setup Rule performs the ENUM query can now be specified. This is done by specifying an IP Interface in the 'Query Target' field in the Call Setup Rules table. The ENUM server's address is the address configured for the 'Primary DNS Server' or 'Secondary DNS Server' fields of the specified IP Interface.

Applicable Applications: All.

Applicable Products: All. -

2.15.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-21: Resolved Constraints in Version 7.20A.156.023

Incident	Description
148696	When the device is in HA mode and an SBC call's SDP parameters (e.g., IP address and port) change, when an HA switchover occurs, the device erroneously uses the old SDP parameters. As a result, no voice is heard for a few minutes. **Applicable Products:** HA.
148275	If the device adds to the coder list of a call an extended coder for the image (T.38 fax), if the remote side rejects the T.38 (i.e., no supported), the device crashes (resets). **Applicable Products:** SBC.
148555	For a WebRTC call done from a Firefox browser, if a second re-INVITE SIP message occurs, the device crashes (resets). **Applicable Products:** SBC with Web RTC.
146960	When using CAS and SRTP and the call destination changes, the device does not change the SSRC and the sequence number is lowered and thus, the remote side drops the packets. As a result, one-way voice occurs. **Applicable Products:** CAS Gateway.

2.16 Patch Version 7.20A.156.041

This patch version includes only resolved constraints.

Note:

- This patch is applicable only to Mediant 9000 SBC.
- This patch version is compatible with AudioCodes One Voice Operations Center Version 7.2.3104.

2.16.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-22: Resolved Constraints in Version 7.20A.156.041

Incident	Description
149346	For the Media Transcoding Cluster (MTC) feature, when performing an HA switchover on the device, the Media Transcoders (MT) reset and connection is lost (for a few minutes) between the device and the Media Transcoders. **Applicable Products:** Mediant 9000.
149345	When an ini file is loaded to the device that changes the VLAN of the OAMP interface, connection with the device is lost. **Applicable Products:** Mediant 9000 HA.
149302	VI - M9K HA Switch-over Description Race condition during DSP restart causes the device to crash (reset). **Applicable Products:** Mediant 9000 HA.

2.17 Patch Version 7.20A.158.009

This patch version includes new features, known constraints, and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.1079, and EMS/SEM Version 7.2.3104.

2.17.1 New Features

New features introduced in this version include the following:

2.17.1.1 Sending SIP Messages to OVOC for SIP Call Flow Diagrams

This feature provides support for the device to send SIP messages (in XML format) of SIP call dialogs to AudioCodes One Voice Operations Centers so that One Voice Operations Center management users can view the call dialog sessions as call flow diagrams. The call flow is displayed in One Voice Operations Center using vertical and horizontal lines, where the vertical lines represent the SIP entities (including AudioCodes device) involved in the dialog and where the horizontal lines represent the SIP requests and responses.

SIP call flow diagrams may be useful for debugging and for better understanding of the SIP call. The call flow displays all the SIP messages related to the call session, including requests (e.g., INVITEs) and responses (e.g., 200 OK). For SBC calls, the call flow reflects messages as sent "over the wire" - incoming messages before manipulation and outgoing messages after manipulation, For Gateway calls, the call flow reflects incoming messages after Pre-Parsing Manipulation (if configured) but before general Message Manipulation, and outgoing messages after manipulation.

To support this feature:

- The new parameter has been introduced to enable or disable the feature: 'SIP Call Flow Report Mode' (configure voip > qoe call-flow-report) with optional values “Enable” and “Disable” (default).
- To send SIP call flow messages for specific calls only, the existing Logging Filters table can be used. For specifying these messages, the table's 'Log Destination' parameter must be configured to the new optional value, "Call Flow Server" and the 'Log Type' parameter to the new optional value, "Call Flow". If the table does not include any filtering rule for SIP call flow, the device sends One Voice Operations Center call flow messages for all calls.

Note:

- The feature does not support SIPRec messages and REGISTER messages.
- For HA systems, during a switchover the device stops sending the SIP call flow messages of current SIP dialogs and continues sending them after the switchover (even though OVOC does not display the continuation of the call after switchover).
- If the device experiences a CPU overload, it stops sending SIP call flow messages to the One Voice Operations Center until the CPU returns to normal levels.

Applicable Applications: SBC and Gateway.

Applicable Products: All.

2.17.1.2 Configurable Unit of Measurement for Call Duration in CDRs

This feature provides support for configuring the unit of measurement for call duration in CDRs ('Call Duration' field) generated by the device. The unit of measurement can be configured to seconds (default), deciseconds, centiseconds, or milliseconds. Up until now, call duration was displayed only in seconds.

This feature is configurable by the following new parameter:

- Web: 'Call Duration Units'
- CLI: configure troubleshoot > cdr > call-duration-units
- ini: CallDurationUnits

Applicable Applications: All.

Applicable Products: All.

2.17.1.3 New Customized CDR Field "Call End Sequence Number"

This feature provides support for a new CDR field—"Call End Sequence Number" [442]—that can be added to CDRs (customizable), using the Gateway CDR Format table or SBC CDR Format table.

The feature applies to Syslog, RADIUS, and local-storage CDRs. The field is added only to CDRs that are generated at the end of calls. For each CDR, the value is assigned the next consecutive number. For example, for the first terminated call processed by the device, the field is assigned the value "1"; for the second terminated call, the field is assigned the value "2", and so on. The field value resets to 1 upon a device reset, an HA switchover (for HA-supporting products), or when it reaches the value FFFFFFFF (hexadecimal).

As this CDR field value is consecutive, the feature can be useful for checking whether there are any missing CDRs.

Applicable Applications: SBC and Gateway.

Applicable Products: All.

2.17.1.4 CDR Local Storage Enhancements

This feature provides the following enhanced support for the local storage of CDRs:

- At full session capacity, CDRs can be stored for at least seven days (depending on local storage configuration).
- CDR files can be compressed to a ZIP or GZIP file, using the new CDRLocalCompression parameter.
- The name of the CDR file can be configured, using the new CDRLocalFileName parameter. The configuration supports format specifiers. For example, "CDR_%y.%m.%d-%H.%M.%S_%qqqqq.csv" creates the filename "CDR_17.12.25-14.20.02_00010.csv" (i.e., 25 December 2017, 14:20:02).
- Each CDR is automatically assigned a unique sequence number, which is appended (by default) at the end of the filename.
- CDRs can be accessed through SFTP, allowing the SFTP client to rename CDR files or download them. Regular CDR files are stored in the */cdr folder and SBC test call CDRs are stored in the /cdr-gw* folder. The SFTP client needs to authenticate itself with the SFTP server (device). Access is granted only to users with Security Administrator level.
- The names of the following existing parameters have been modified:
  - 'Local Storage Max File Size' (CDRLocalMaxFileSize) has been renamed 'File Size'
  - 'Local Storage Max Number of Files' (CDRLocalMaxNumOfFiles) has been renamed 'Number Of Files'
  - 'Local Storage File Creation Interval' (CDRLocalInterval) has been renamed 'Rotation period'

Note:

- Devices running more than 100 calls per second must use a file size (CDRLocalMaxFileSize) that is greater than 100 MB.
- When upgrading a device that already uses CDR local storage, the default maximum

file size (CDRLocalMaxFileSize) is 100 MB.

- When upgrading a device that already uses CDR local storage, the filename of the CDRs will be changed using the default filename format specifiers (CDR__%y.%m.%d-%H.%M.%S_%qqqqq.csv).

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.17.1.5 CDR Local Storage Value Changes

This feature provides the following changes in the supported values of existing CDR local storage parameters:

- CDRLocalMaxFileSize (min., max., default):
  - Mediant 9000/Mediant VE/SE: 1 MB, 1 GB, 100 MB
  - Mediant 2600/Mediant 4000: 1 MB, 10 MB, no change (1 MB)
  - All other products: no change (100 KB), 10 MB, no change (1 MB)
- CDRLocalMaxNumOfFiles (max):
  - Mediant 9000/Mediant VE/SE: 65535

Applicable Products: All.

2.17.1.6 Enhanced HA Network Monitor Feature

This feature provides enhanced support for the HA Network Monitor feature, applicable to devices operating in HA mode. This new feature enables the monitoring (using pings) of multiple network entities (destination addresses). Up until now, only a single network entity could be monitored. This enhancement is especially important for deployments that use multiple network interfaces and thus, different network entities in different networks can be selected for monitoring. The feature also allows the administrator to configure the minimum number of failed monitored network entities in order to trigger an HA switchover.

To support the feature, the following new configuration entities have been introduced:

- HA Network Monitor table (HaNetworkMonitor; configure network > high-availability network-monitor), which defines up to 10 rows, each with up to 5 destinations (IP addresses) to ping.
- HA Network Monitor Peers Status table: Displays the status of each destination (IP address) of a selected row in the HA Network Monitor table.
- 'Failed Monitored Rows for Switchover' parameter (HaNetworkMonitorThreshold or configure network > high-availability settings > network-monitor-threshold), which defines the number of failed monitored network entries (ping destinations) required to trigger an HA switchover.

This feature also introduces a new SNMP alarm, acHANetworkMonitorAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.55), which is raised (Major severity) when all previously reachable destinations configured for a specific row in the HA Network Monitor table become unreachable.

Note:

- Existing standalone HA parameters in CLI have been relocated from the “high-availability” folder to the “high-availability settings” folder.
- The following parameters are now obsolete: HAPingDestination, HAPingSourceIfName, HAPingRetries, and HAPingTimeout.

Applicable Applications: SBC/Gateway.

Applicable Products: Mediant 500 Gateway & E-SBC; Mediant 800 Gateway & E-SBC; Mediant 2600 E-SBC; Mediant 4000 SBC; Mediant 9000 SBC; Mediant VE/SE.

2.17.1.7 LDAP-based Management Services

This feature provides support for configuring an LDAP-based management service account (Management Service-type LDAP server). This LDAP service account has an always-on connection with the device, using a configured LDAP username (Bind Name) and password, and which performs **only user authorization** for users attempting to log in to the device. This account operates together with the already supported LDAP-based management account (Management-type LDAP server), which in this setup, is used only for authenticating the user's login username and password.

The device connects to the Management-type LDAP server only when users attempt to log in to the device. If authentication is successful, the device then queries the Management Service-type LDAP server for user authorization (i.e., the user's management access level and privileges). Therefore, having two separate LDAP-based management accounts —one for user authentication and one for user authorization—whereby authorization is performed only by an LDAP "administrator", may provide additional security to the network by preventing users from accessing the authorization settings of the LDAP server.

To support the feature, the existing 'Type' (LdapServerGroups_ServerType) parameter in the LDAP Server Groups table provides an additional optional value—Management Service (2)—for configuring the LDAP Server Group for LDAP management services.

Applicable Applications: All.

Applicable Products: All.

2.17.1.8 Ping by Hostname

This feature provides support to ping a destination by hostname. Up until now, the device could only ping a destination by IP address. The feature is supported by the existing CLI command, **ping**. For example:

```
ping corp.abc.com source voip interface vlan 1
```

Applicable Applications: All.

Applicable Products: All.

2.17.1.9 User Account Registration Based on IP Group Connectivity Status

This feature provides support for enabling the device to forward register requests from a SIP entity (Served IP Group) to a SIP registrar (Serving IP Group) only if the Served IP Group is online. The IP Group's connectivity status is determined by the keep-alive mechanism of its associated Proxy Set. The feature is applicable only to Accounts where registration is initiated by the device (i.e., 'Register' parameter is set to any value other than **No**), configured in the Accounts table.

This feature is configured by the following new parameter in the Accounts table:

- Web: 'Register by Served IP Group Status'
- CLI: reg-by-served-ipg-status
- ini: Account_RegByServedIPG

When configured to **Register Only if Online** [1], the device performs registration depending on the connectivity status of the Served IP Group. It sends a registration request to the Serving IP Group only if the Served IP Group is online. In addition, if the Served IP Group was registered but then later goes offline, the device unregisters it. If it becomes online again, the device re-registers it.

By default (**Register always** [0]), the registration by the device does not depend on the status of the Served IP Group.

Applicable Applications: SBC.

Applicable Products: All.

2.17.1.10 Enhanced Behavior for Account Registration

This feature provides enhanced support for user registration and authentication, whereby the device uses the username and password configured in the IP Groups table for the Serving IP Group (registrar server) for user registration and authentication, in the following scenarios:

- If there is no Account configured for the Served IP Group and Serving IP Group in the Accounts table.
- If there is an Account configured for the Served IP Group and Serving IP Group in the Accounts table, but without a username and password.

For this mode of operation, the 'Authentication Mode' parameter in the IP Groups table for the Serving IP Group must be configured to **SBC As Client**.

Applicable Applications: SBC.

Applicable Products: All.

2.17.1.11 Dynamic SIP UDP Port Assignment for Registration Accounts

This feature provides support for enabling the device to dynamically allocate local SIP UDP ports to Accounts on the interface facing the Serving IP Group (i.e., registrar server). Each Account is allocated a unique port taken from a port range configured for the SIP Interface (existing 'Additional UDP Ports' parameter - SIPInterface_AdditionalUDPPorts) associated with the Proxy Set of the Accounts' Serving IP Group. This feature is applicable only to Accounts where the device initiates registration (i.e., the 'Register' parameter is set to any value other than **No**).

Up until now (and still supported if this feature is disabled), the device used the same local UDP port for all Accounts communicating with the same Serving IP Group.

This feature is configured by the following new parameter in the Accounts table:

- Web: 'UDP Port Assignment'
- CLI: udp-port-assignment
- ini: Account_UDPPortAssignment

Applicable Applications: SBC.

Applicable Products: All.

2.17.1.12 Parameter Name Change for 'Transcoding Mode'

The Web interface's parameter 'Transcoding Mode' (IpProfile_TranscodingMode) has been renamed 'Mediation Mode', It's optional values have also been renamed as follows:

- **Only if Required** -- to **RTP Mediation**
- **Force** -- to **Force Transcoding**
- **RTP Forwarding** (new)

Applicable Applications: SBC.

Applicable Products: All.

2.17.1.13 IP Group Parameter Representation in Message Manipulation

This feature provides the following enhanced support for IP Group representation in SIP message manipulation rules:

- Up until now, the manipulation syntax only allowed the administrator to specify the source or destination IP Group (e.g., *param.ipg.src.4*). This new feature allows the administrator to specify any IP Group regardless of the call's source and destination IP Group:
  - *param.ipg.<ID>*

    An example of the syntax *param.ipg.<ID>.host*:

    ```
    param.ipg.5.host
    ```

- *param.ipg.<Name>*
  An example of the syntax *param.ipg.<Name>.host*:
  ```
  param.ipg.ITSP-WORLD.host
  ```

This syntax is applicable to all configuration tables that can be configured or associated with manipulation syntax (e.g., Call Setup Rules and Message Conditions tables).

Note: The IP Group name is case-sensitive and cannot contain spaces or dots (.).

- A new manipulation syntax element, "is-alive" represents the IP Group's connectivity status - online or offline (typically used when the associated Proxy Set is configured with keep-alive functionality):
  - param.ipg.<ID>|<Name>.is-alive
  - param.ipg.src|dst.is-alive

  The "is-alive" parameter uses the keywords "true" and "false" to indicate whether the specified IP Group is online or offline, respectively. For example:
  ```
  param.ipg.4.is-alive == 'true'
  param.ipg.4.is-alive == 'false'
  param.ipg.ITSP-WORLD.is-alive == 'true'
  param.ipg.ITSP-WORLD.is-alive == 'false'
  ```
  Note: The 'true' and 'false' keywords are case-sensitive.

  An example would be to use this manipulation syntax as a condition for a routing rule, where the status of an IP Group (instead of the destination IP Group) is checked. If the IP Group is online, then apply the routing rule. If the IP Group is offline, then route the call to an alternative destination. For an example, refer to the *SIP Message Manipulation Reference Guide*.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.17.1.14 SNMP Alarm for No Configured Proxy

The existing SNMP alarm acIpGroupNoRouteAlarm is now also sent by the device if a Server-type IP Group is not associated with a Proxy Set or it's associated with a Proxy Set that is not configured with any address, or the associated Proxy Set experiences a proxy keep-alive failure.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.17.1.15 Enhanced Message Manipulation Syntax for User-to-User Header

This feature provides support for referring to parameters (known or unknown) in the User-to-User (or X-User-to-User) SIP header. This is supported by the following syntax:

```
header.user-to-user.param
```

For example, the following syntax adds the parameter "purpose" with value "isdn-network" to the User-to-User header:

Action Subject	Action Type	Action Value
header.user-to-user.param.purpose	Add	'isdn-network'

Applicable Applications: All.

Applicable Products: All.

2.17.1.16 Enabling Global Session ID through REST API

This feature provides support for a REST client to enable the global session ID through AudioCodes REST API (GET and PUT actions). Up until now, the global session ID could only be configured through ini file (SendAcSessionIDHeader parameter). When enabled, the global session ID is included in SIP messages in the AudioCodes proprietary SIP header, AC-Session-ID. It is a unique identifier of the call session and is maintained even if the SIP dialog traverses multiple devices. This is useful for keeping track of a specific call

The feature is supported by the following new REST URL path:

```
/api/v1/rmConfig/globals/sendAcSessionIDHeader
```

The possible values are "yes" (enable) and "no". For more information, refer to the document, *REST API for Mediant Devices*.

Applicable Applications: All.

Applicable Products: All.

2.17.1.17 Web Interface Updated with New AudioCodes Corporate Logo

The device's Web interface has been updated with AudioCodes' new corporate logo. This includes the logo that appears on the Web Login screen and the main menu bar. The Web browser's favicon has also been updated with the new logo.

Applicable Applications: All.

Applicable Products: All.

2.17.1.18 Customization of Web Browser's Tab Label

This feature provides support for customizing (private labeling) the label that appears on the tab of the Web browser used to open the device's Web interface. The default label is "AudioCodes", which can either be replaced by different text or with the device's IP address. Up until now, the tab displayed "AudioCodes" and couldn't be customized.

Applicable Applications: All.

Applicable Products: All.

2.17.1.19 Invalid RTCP Packet Handling

This feature provides support for configuring the device's handling of invalid incoming RTCP packets. Up until now, the device supported configuration of invalid incoming RTP packet handling. The parameter used for RTP invalid packet handling now also applies to RTCP invalid packet handling (i.e., RTPFWInvalidPacketHandling).

Applicable Applications: All.

Applicable Products: All.

2.17.1.20 OVR Support on Mediant VE SBC

This feature provides support for the One-Voice Resiliency (OVR) application on the Mediant VE SBC and supports up to 2,000 users.

Applicable Applications: OVR.

Applicable Products: Mediant VE SBC.

2.17.2 Known Constraints

This section lists known constraints.

Table 2-23: Known Constraints in Version 7.20A.158.009

Incident	Description
149163	When configuring an SBC routing rule in the IP-to-IP Routing table and the 'Destination Type' parameter is configured to either Dest Address, Request URI, ENUM, Dial Plan, or LDAP, a destination IP Group must be specified in the 'Destination IP Group' parameter. The destination IP Group is not used for the actual destination (i.e., associated Proxy Set), but its associated configuration elements are used such as the IP Profile. If not specified, the device uses an IP Profile and other elements according to its own logical processes. **Applicable Products:** All.

150025	When upgrading the Mediant 9000 or Mediant VE (device) from Version 7.20A.156.41, current calls are disconnected. To avoid this, prior to starting the upgrade process, run the following CmdShell commands on the device for each MT:
	`IgnoreMtceHBTimeout <OAMP IP address of MT> 1`
	`IgnoreMtceTpncpTimeout <OAMP IP address of MT> 1`
	The IP addresses appear in the Media Transcoders table. These commands temporarily disable the MT keep-alive mechanism (until the device resets) and thus, the upgraded device won't disconnect the calls due to disconnected MTs.
	Applicable Products: Mediant 9000 with MT; Mediant VE SBC with MT.

148040	For Mediant VE SBC devices running on Microsoft Hyper-V, the device exposes the same serial number for all virtual machines, in SNMP (e.g., to OVOC). **Applicable Products:** Mediant VE SBC.

2.17.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-24: Resolved Constraints in Version 7.20A.158.009

Incident	Description
149860	DNS query results appear backward in the CLI (for example, "2.174.18.172" instead of "172.174.18.2"). **Applicable Products:** All.
149579	The NAT Traversal parameter setting to **Force NAT** does not function. In some cases, the device sends RTP to the private destination address that is identified as being behind NAT instead of waiting for the first incoming packet. As a result, the RTP is sent to the wrong destination. **Applicable Products:** SBC.
149646	An outage (device resets) is experienced for a few minutes during Hitless Software Upgrade. This occurs when the call opened on a CID which is not valid on the second device. **Applicable Products:** HA SBC.
149490	The device loses the TLS certificate when the same ini file is uploaded to the device. As a result, secured calls cannot be established. **Applicable Products:** All.

Incident	Description
149697 / 148949	When using TLS, keep-alive messages are not sent from the device to ARM and there is a loss of connection. **Applicable Products:** SBC with ARM.
149588	In some scenarios, the SIP Connect feature does not correctly classify users. **Applicable Products:** SBC.
149342	If an endpoint restarts (i.e., a new call ID in REGISTER messages), the device keeps sending the REGISTER to the proxy using the same call ID and CSEQ. As a result, the proxy is unaware that the endpoint restarted. **Applicable Products:** SBC.
149599	If the device is inundated by REGISTER requests that should be replied with a SIP 401, the device crashes (resets). **Applicable Products:** SBC.
149595	If the global parameter SBCDirectMedia is enabled and an UPDATE message with SDP is sent after the initial SDP offer-answer, the device crashes (resets). **Applicable Products:** SBC.
148520	The ACD (Average Call Duration) value displayed in the Web interface is incorrect. **Applicable Products:** SBC.
149545	The device crashes (and resets) in the following scenario: An SBC call is answered by a 302. The device creates a new call that is answered with a 401. The device forwards it to the calling side, which then responds with a new INVITE without credentials. **Applicable Products:** SBC.
149409	The VLAN priority of outgoing media packets are not calculated correctly. **Applicable Products:** All.
148253	Invalid characters in the 'SSH Public Key' parameter in the Local Users table causes partial Web GUI display. **Applicable Products:** All.
149073	If two call legs are WebRTC and direct media is enabled, the device sends incorrect headers. As a result, the message may fail. **Applicable Products:** SBC.
149129	If a user registers using TLS, the device uses TLS even for dialogs intended for TCP transport. As a result, the message fails. **Applicable Products:** SBC.
149395	The Dial Plan does not function correctly for prefixes with digits and letters, causing routing problems. **Applicable Products:** SBC.
149209	If an ini file with configuration of four LDAP servers (even though only two are supported) are loaded to the device, no validation is performed and after editing and applying changes, a Web error occurs. **Applicable Products:** All.
149387	For HA devices, when one device has a Product Key in the License key and the second device does not, the device erroneously issues the alarm, "FK mismatch alarm". **Applicable Products:** HA SBC.

Incident	Description
149163	IP Groups are selected according to SRD instead of SIP Interface. As a result, calls fail. **Applicable Products:** SBC.
149217	After an HA switchover, the device uses the IP Profile from the IP Group (instead of the IP Profile saved with the registered user). As a result, the incorrect IP Profile is used for the call and calls are rejected with a SIP 488. **Applicable Products:** HA SBC.
149183	If the device has to forward an INVITE that includes a very long User-To-User header, the device crashes (resets). **Applicable Products:** SBC.
149305	Media CDRs include the 'Fraction lost' value from RTCP instead of the 'Cumulative number of packets lost' **Applicable Products:** All.
148661	After an HA switchover, the SAVE button is red indicating that the user needs to perform a save (which is incorrect). **Applicable Products:** HA SBC.
146079	In certain scenarios, some buffers are lost over time, resulting in one-way voice. **Applicable Products:** SBC.
148663	The help description for the same command (idle-timeout) is not the same for Telnet and CLI. **Applicable Products:** All.
141323	A problem in initiating ports causes the flooding of error messages and as a result, the device crashes (resets). **Applicable Products:** MP-1288.
149153	If the device sends a CANCEL message to a destination to cancel a call, no SIP 487 response is received and the device erroneously performs alternative routing to the next proxy. **Applicable Products:** SBC.
149050	When the device is configured for LDAP authentication for Web-based management, LDAP users can log in, but local users cannot. **Applicable Products:** All.
148950	The device does not correctly handle incoming messages with a crypto line containing "UNENCRYPTED_SRTP UNENCRYPTED_SRTCP UNAUTHENTICATED_SRTP". As a result, calls failed. **Applicable Products:** All.
148746	The redundant device sends audit messages to Syslog. **Applicable Products:** SBC HA.
148994	If the display name in the From header includes a backslash "\", the source number is removed. **Applicable Products:** SBC.
148976	Messages are sent to the public IP address destination instead of the private address (even though Syslog erroneously indicates the private). **Applicable Products:** SBC.

Incident	Description
148441	A space in the incoming user name causes a parsing error (should be replaced with an escape character). As a result, the REGISTER fails. **Applicable Products:** Gateway.
148815	When performing CSR on incoming call, call forking does not function and calls are not routed correctly. **Applicable Products:** Gateway.
148717	The Monitor page does not display OVR calls after an HA switchover. **Applicable Products:** SBC HA with OVR.
148555	In certain scenarios, during a re-INVITE when video media is added, the device crashes (resets). **Applicable Products:** SBC.
148845	If two blanks occur in the Request-URI (between 400 and "not found"), Pre-Parsing Manipulation does not function. As a result, the message fails. **Applicable Products:** SBC.
148686	No voice for a few minutes after an HA switchover. The voice stream was opened with transcoding (DSP) and due to a bug, when channels opened with DSP, the RTP sequence jumped to the beginning after switchover. **Applicable Products:** SBC HA.
148727	IAM message for SIP-I is not parsed correctly, which causes manipulation failure. **Applicable Products:** SBC.
148174	If the SBCKEEPCONTACTUSERINREGISTER is set to 1 and the REGISTER is received from WebRTC, the device does not release resources. As a result, the device rejects the registration. **Applicable Products:** SBC with WebRTC.
147955	Under some conditions, the ini file cannot be loaded using the Automatic Update feature (IniFileURL parameter). **Applicable Products:** All.
148559	When creating PRT files, the user is unable to track the index of a tone within the file (Local Held Tone Index parameter), resulting in difficulties in playing tones. **Applicable Products:** All.
148630	Some resources are not released when the device sends multiple Call Setup Rules requests to the LDAP server which is offline. As a result, calls are dropped. **Applicable Products:** SBC.
148016	When the device receives more than four crypto suites and the supported crypto suites were indexed at greater than four, the device does not use the supported crypto. As a result, no voice is experienced. **Applicable Products:** All.
147430	Under some conditions, the device loses the DNS resolution and thus, failed to classify calls from the proxy. As a result, calls failed. **Applicable Products:** SBC.
148189	When collecting the debug file from the device, the device crashes (resets). **Applicable Products:** SBC.
148366	When the device receives a large INVITE message (greater than 4k), it fails to send the route request (getRoute) to ARM. As a result, call routing failure occurs. **Applicable Products:** SBC with ARM.

Incident	Description
146960	For CAS calls, when the destination (IP:port) changes (simultaneous ring is enabled), the device does not change the SSRC. As a result, one-way voice occurs. **Applicable Products:** CAS Gateway.
145870	CAS issues result in calls not being processed. CAS starts to work only after the trunk is stopped and then started. **Applicable Products:** CAS Gateway.
148326	When the device receives an incoming WebRTC call with the Opus coder, the device erroneously uses G.711 coder for the call. As a result, no voice occurs. **Applicable Products:** SBC with WebRTC.
148161	Static IP routes disappear from the Static Routes table after changing the Application Type from OAMP + Media + Control to OAMP only, causing routing problems. **Applicable Products:** All.
147110	Some performance monitoring calculations are incorrect. **Applicable Products:** SBC.
148276	The device rejects180 messages that have an IPv6 address enclosed by brackets in the "received" parameter of the Via header (this format contradicts the RFC). As a result, such calls fail. **Applicable Products:** SBC.
147244	During T.38 fax transmission, the device erroneously sends HA warning messages to Syslog. **Applicable Products:** HA SBC.
147244	The device fails to download ini files from the EMS. **Applicable Products:** SBC.
147483	When the License Key includes only 1 PRI trunk, it is not synchronized. As a result, Gateway calls cannot be processed. **Applicable Products:** Digital Gateway.
148130	The CLI is missing the "asserted-identity-m" command. **Applicable Products:** All.
150023	If the IP Profile parameter 'SBC Remote Update Support' is configured to 1, the fax detection feature for SBC calls does not function and as a result, faxes fail. **Applicable Products:** All.
149890	When Quality of Experience (QoE) is enabled, during the calculation of QoE, a "divide by 0" occurs, causing the device to crash (reset). **Applicable Products:** All.
150044	For the Media Transcoding Cluster (MTC) system, the MT drops the calls during a switchover. **Applicable Products:** MTC.
149837	On the Web interface's SBC CDR History page, when defining the display of more rows than the default, no scrolling option occurs and table sorting cannot be done. **Applicable Products:** All.
149536	When all CAS trunks are up and running and the Web interface's Trunks & Channels Status page is opened, the trunk becomes inactive. **Applicable Products:** CAS Gateways.

Incident	Description
149796	The device crashes (resets) when it attempts to add a new user registration to the list of IP Groups in its registration database. **Applicable Products:** All.

2.18 Patch Version 7.20A.158.012

This patch version only includes a resolved constraint.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.1079, and EMS/SEM Version 7.2.3104.

2.18.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-25: Resolved Constraints in Version 7.20A.158.012

Incident	Description
150342	If the device (SBC application) receives a SIP message that contains a character in the beginning that is not allowed, for example CRLF or space, and the device performs Pre-Parsing Manipulation, the device crashes (and resets). **Applicable Products:** All.

2.19 Patch Version 7.20A.158.035

This patch version only includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.1083, and EMS/SEM Version 7.2.3106.

2.19.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-26: Resolved Constraints in Version 7.20A.158.035

Incident	Description
148119	Mediant VE SBC with 1 vCPU / 2-GB RAM is not supported. (Now it is.) **Applicable Products:** Mediant VE SBC.
151024	The device reports incorrect MOS values when RTCP-XR is enabled. **Applicable Products:** All.
150877	When the device does transcoding between two legs, the outgoing DTMF stream is different from the incoming DTMF stream. As a result, the user cannot join the conference. **Applicable Products:** All.
151243	INVITE dialog-initiating requests to the "sticky" proxy server do not receive any response and therefore, the server is marked as offline. This causes the Account to stop working with it and registers with another server which causes the calls to be dropped. **Applicable Products:** All.
150821	Fragmented packets are not handled correctly and as a result, large SIP messages are dropped. **Applicable Products:** All.
151164	The Account Registrar Stickiness feature does not function in HA mode (and calls are thereby dropped). **Applicable Products:** HA Devices.
151100 / 150483	When the device communicates with a user located behind NAT, after an HA switchover, the device sends the RTP to the wrong IP:port. As a result, no voice occurs. **Applicable Products:** HA Devices.
150970	When using the dynamic port mapping feature in CDR, the device erroneously sends the first port (i.e. incorrect CDR). **Applicable Products:** All.
149683	When communicating with a user located behind NAT and a switch to T.38 occurs, the remote user does not send T.38 packets and therefore, the device is unable to latch and send T.38 to the correct destination. As a result, fax failure occurs. **Applicable Products:** All.

Incident	Description
150937	The device incorrectly handles an invalid Application-Defined RTCP packet, which causes the device to reset. **Applicable Products:** All.
150993	When using the port per user feature and the user adds an IPv6 interface after the device powers up, after an HA switchover the device sends different ports per user. As a result, calls are dropped. **Applicable Products:** HA Devices.
151087	If the device's License Key includes a license for a small number of SIPRec sessions, the device crashes (and resets). **Applicable Products:** All.
151049	The **traceroue** CLI command does not function for IPv6. **Applicable Products:** All.
150994	Even when the SIPCHALLENGECACHINGMODE parameter is configured to 0, the device caches authentication challenges. As a result, calls fail. **Applicable Products:** All.
150872	The Dial Plan table cannot be exported from the Web interface. **Applicable Products:** All.
150981	Device crashes (resets) resulted in the device not being in HA mode. **Applicable Products:** HA Devices.

2.20 Patch Version 7.20A.158.056

This patch version only includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.1083, and EMS/SEM Version 7.2.3106.

2.20.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-27: Resolved Constraints in Version 7.20A.158.056

Incident	Description
152322	The device resets due to the overrunning of a buffer causing a system memory leak. **Applicable Products:** SBC.
152259	The Media Transcoder (MT) displays active alarms (Ethernet link alarms) even though no SIP is configured. **Applicable Products:** Mediant 9000; Mediant Software; Media Transcoder.
152209	In some scenarios, the device sends an error message to the Syslog when object is incorrectly configured. **Applicable Products:** All.
152085	When the device is configured with an invalid Condition in the IP-to-IP Routing table, matching the incoming SIP message to this IP-to-IP Routing rule causes the device to reset. **Applicable Products:** SBC.
152069	When the time configured for the SBCPROXYREGISTRATIONTIME parameter expires and the device sends the user's REGISTER to the server, the server replies with a SIP 401. However, after a few seconds, the device stops tracking the refresh failure and if a subsequent REGISTER is received, the device handles it as if the previous REGISTER succeeded and terminates it. As a result, the user does not get registered with the server. **Applicable Products:** SBC.
151953	When the device processes a large INVITE message and the buffer is full, if an alternative route is located for this INVITE, the buffer overruns and after several repeats, the device crashes (resets). **Applicable Products:** SBC.
151943	When processing the application/BroadsoftDocument+xml XML body that has an alias whose length is 37 characters, a memory corruption occurs due to unsafe copying. After some repeats of this corruption, the device crashes (resets). **Applicable Products:** SBC.
151694	When the device sends a SUBSCRIBE with the base UDP port instead of the configured UDP port (configured in the SIP Interfaces table's 'Additional UDP Ports' parameter), the SUBSCRIBE is denied by the server (403 Forbidden response). **Applicable Products:** SBC.

Incident	Description
151616	Message manipulation cannot not be done on SIP Authentication headers. As a result, authentication fails. **Applicable Products:** SBC.
150995	The connection between the PuTTY client and the device is terminated abnormally, causing the login resource to not be deallocated. As a result, access to the device's management interfaces (Web and CLI) is blocked. **Applicable Products:** SBC.
152028	When the device has no available SIP Socket resources (due to incorrect configuration which has exceeded the recommended), the device crashes (resets). **Applicable Products:** SBC.

2.21 Patch Version 7.20A.158.065

This patch version only includes a resolved constraint.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.1083, and EMS/SEM Version 7.2.3106.

2.21.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-28: Resolved Constraints in Version 7.20A.158.065

Incident	Description
154152	When SIP Connect is enabled, a maximum of four contacts per user (AOR) can be registered in the device's registration database. Now, it supports up to eight contacts per AOR. **Applicable Products:** SBC.

2.22 Patch Version 7.20A.162.001

This patch version only includes new features and known constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.2101, and EMS/SEM Version 7.2.3106.

2.22.1 New Features

2.22.1.1 New Mediant 9000 Hardware Revision

A new hardware revision—Rev. B--has been released for Mediant 9000 SBC. This new hardware includes the following main features:

- Latest (Gen10) server architecture
- Upgrade of the Integrated Lights Out (iLO) module to iLO 5, including advanced features

Note: For High-Availability (HA) systems, the participating HA pairs (active and redundant units) must be of the same hardware revision.

Applicable Applications: All.

Applicable Products: Mediant 9000.

2.22.1.2 Max. RADIUS-Accounting Attributes for CDR Customization

The maximum number of RADIUS Accounting attributes that can be customized (and sent) in the CDR generated by the device has been increased from 40 to 70.

Applicable Application: SBC.

Applicable Products: MP-1288; Mediant 500; Mediant 500L; Mediant 800; Mediant 1000B; Mediant 2600; Mediant 4000.

2.22.2 Known Constraints

This section lists known constraints.

Table 2-29: Known Constraints in Version 7.20A.162.001

Incident	Description
-	This software version is applicable only to Mediant 9000 Rev. B. Earlier hardware revisions should not upgrade to this version. **Applicable Products:** Mediant 9000
-	Media Transcoding Cluster feature is not supported in this release. **Applicable Products:** Mediant 9000; Mediant VE; Media Transcoder (MT).

2.23 Patch Version 7.20A.162.017

This patch version includes only resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.3082 and EMS/SEM Version 7.2.3106.

2.23.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-30: Resolved Constraints in Version 7.20A.162.017

Incident	Description
152162	The device sends a CDR with transport type "unknown" for an SBC call when the device undergoes an HA switchover during the call and the call disconnects when the redundant device becomes active. **Applicable Products:** HA Devices.
152508	When port redundancy occurs for the Media Transcoder (Media Transcoding Cluster feature), the Media Transcoder resets itself. **Applicable Products:** Mediant 4000.
151612	CDR files that are generated by the device for SBC calls have the incorrect size and time (not as configured). **Applicable Products:** All.
150867	For the Media Transcoding Cluster feature, a memory overrun in the Media Transcoder (MT) causes it to crash (reset). **Applicable Products:** Mediant 4000.
-	Media Transcoding Cluster feature is not supported in this release. Now, it is supported. **Applicable Products:** Mediant 9000; Mediant VE; Media Transcoder (MT).

2.24 Patch Version 7.20A.200.019

This patch version includes new features, known constraints, and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center Version 7.4.2094, and EMS/SEM Version 7.2.3106.

2.24.1 New Features

New features introduced in this version include the following:

2.24.1.1 Entity Names Added to SNMP Alarm Descriptions

Names of configuration entities are now included in SNMP alarm descriptions. Entity names such as Proxy Set, IP Group, SIP Interface, and SRD now appear in the alarm descriptions. Previously, alarm descriptions only included the entity's table row index (ID), for example: "Proxy Set Alarm Proxy Set 37: Proxy lost. looking for another proxy". Now, the Proxy Set name is also included and shown in parenthesis, for example, "Proxy Set Alarm Proxy Set 37 (ITSP): Proxy lost. looking for another proxy".

Applicable Applications: All.

Applicable Products: All

2.24.1.2 Performance Monitoring Thresholds Included in ini File

The downloaded ini file now includes SNMP performance monitoring MIBs whose thresholds (low and/or high) have been changed from default values. The ini file displays the performance monitoring MIB with the modified low and high threshold values. This feature can be useful for applying the same thresholds to other devices (by simply loading the same ini file to them).

Applicable Applications: All.

Applicable Products: All

2.24.1.3 Proxy Set Name in Proxy Set Status Display

The name of the Proxy Set is now displayed in the Web interface's Proxy Sets Status table (Monitor menu > Monitor tab > VoIP Status folder > Proxy Sets Status), in a new column called 'Name'. The Proxy Set name also appears in the output of the CLI command, `show voip proxy sets status`.

Applicable Applications: All.

Applicable Products: All

2.24.1.4 Restoring Defaults while Preserving Network Settings in CLI

This feature provides support for restoring the device to factory defaults while preserving network settings, through CLI. Preserving network settings ensures connectivity to the device's management interfaces using the same OAMP IP address after the device has been restored to defaults. Up until now, this option was supported only through the device's Web interface. To support this feature, the following new CLI command has been added:

```
# write factory keep-network-and-users-configuration
```

Applicable Applications: All.

Applicable Products: All.

2.24.1.5 Tail Filter for CLI Command Output

All CLI command outputs can now be filtered to display a user-defined number of lines from the end (*tail*) of the output. To support this feature, the following new command syntax needs to be added to the command to which the filter is applied:

```
<command> | tail <number of lines (1-1000) to display>
```

Below shows an example where the last two log messages (lines) in the output of the show system log command are displayed:

```
# show system log | tail 2
Jan  3 00:35:54 local0.warn [S=147146] [BID=5b1035:250]  SNMP
Authentication Failure - source: IP = 172.17.118.219, Port = 1161,
failed community string = public.
Jan  3 00:35:55 local0.notice [S=147147] [SID=5b1035:250:36462]  (
sip_stack)(     84788)   AcSIPDialog(#170)::TransactionFail -
ClientTransaction(#175) failed sending message with CSeq 1
OPTIONS, the cause is Transport Error
```

Note that the existing **show system log tail** command has now been replaced with **show system log | tail**.

Applicable Applications: All.

Applicable Products: All.

2.24.1.6 Enhanced SBC User Registration Request Handling

The device now provides enhanced handling of the 'expires' parameter of the SIP Contact header, for REGISTER requests that are received from User-type IP Groups:

- If the Contact header in the incoming REGISTER request contains the 'expires' parameter, the device now forwards the request with the parameter (if the destination is a Server-type IP Group). Up until now, the device removed the parameter (sending only the Expires header).
- The device now always adds the ‘expires’ parameter to the Contact header in the SIP response (200 OK) that it sends to the user. Up until now, the device removed the parameter if it existed. Both the Expires header and the Contact 'expires' parameter are sent to the user with the same value.

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.7 Enabling SBC and CRP Applications Removed from Web Interface

Enabling the SBC or CRP application through the Web interface has been removed as these applications are determined by the installed License Key. The SBC application is enabled if at least one SBC-related feature (for example, "SBC Signaling Sessions") is defined in the License Key. The CRP application is enabled if the License Key includes the "CRP" license.

As such, the Applications Enabling page (Setup menu > Signaling & Media tab > Core Entities folder > Applications Enabling) has been removed from the Web GUI. However, the administrator can disable the SBC application (for whatever reason) through CLI and ini file.

Applicable Applications: All.

Applicable Products: All.

2.24.1.8 Faster Upload of CMP Software File

The device's internal processing capabilities have been improved to make software file (.cmp) upload much faster.

Applicable Applications: All.

Applicable Products: All.

2.24.1.9 Enhanced File Management through REST API

The following file management support through AudioCodes REST API (GET, PUT and POST actions) has been added:

- Configuration Package file:
  ```
  api/v1/files/configurationPackage
  ```
- SBC Wizard Template Package file:
  ```
  api/v1/files/sbcWizard
  ```

For more information, refer to the document *REST API for Mediant Devices*.

Note: The following REST API URL paths have been removed: api/v1/files/voicePrompts and api/v1/files/coderTable.

Applicable Applications: All.

Applicable Products: All.

2.24.1.10 New Alarm for Ethernet Group Down of HA Maintenance Interface

A new SNMP alarm, acHAEthernetGroupAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.137) has been added. This alarm is sent when the Ethernet link of at least one of the ports in the Ethernet Group that is associated with the HA Maintenance interface is down.

Applicable Applications: All.

Applicable Products: Mediant 500 Gateway & E-SBC; Mediant 800 Gateway & E-SBC; Mediant 2600 E-SBC; Mediant 4000 SBC; Mediant 9000 SBC; Mediant VE/SE.

2.24.1.11 Subject Alternative Name (SAN) Field for TLS Certificates

Subject Alternative Name (SAN) fields can now be configured when creating certificate signing requests (CSR) and self-signed certificates. The SAN field is an X.509 Version 3 extension providing additional information (multiple subject names) for identifying the device, which can be an e-mail address, DNS hostname, URI, or IP address. Up to five SAN fields can be configured per certificate.

The following configuration parameters have been added for this feature:

- Web Interface: '1st-5th Subject Alternative Name [SAN]' fields on the Change Certificates page (Setup menu > IP Network tab > Security folder > TLS Contexts > Change Certificate link)
- CLI: configure network > tls > certificate alternative-name-add {dns|email|ip-addr|uri}

Applicable Applications: All.

Applicable Products: All.

2.24.1.12 Fullband Coder for SDP Telephone-Event

The device now provides enhanced support for displaying the DTMF sampling rate of the voice coder in the "a=rtpmap" field for the 'telephone-event' in SDP, using the RFC 2833 method. The following DTMF sampling rates can now be displayed: 8,000 kHz (narrowband coders), 16,000 kHz (wideband coders) and 48,000 kHz (fullband coders). Up until now the "a=rtpmap" field only displayed 8,000 kHz.

Note that the feature requires DSP resources (for detection and generation of RFC 2833).

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.13 NGINX for HTTP Proxy Server Configuration

This feature integrates the NGINX (pronounced *engine x*) platform on the device. NGINX is a widely used open source HTTP proxy server with enhanced functionality and customization capabilities. This mechanism replaces the previously supported HTTP proxy server configuration.

Note: The NGINX proxy is not backward compatible with the previous HTTP Proxy and must be re-configured to activate the service.

The following new SNMP alarms related to NGINX configuration have been introduced:

- acNGINXConfigurationIsInvalidAlarm
- acNGINXPprocessIsNotRunningAlarm

The HTTP Proxy configuration tables have also been changed to enhance HTTP Proxy configuration.

The following new CLI commands related to

- To send the NGINX configuration files to a remote destination:

```
# copy nginx-conf-files to
<Protocol>://<Address>/<filename>.tar
```

- To view the NGINX configuration files:

```
show network http-proxy conf active|errors|new
```

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.14 Default DNS Servers

The device is now configured with default DNS server addresses (primary and secondary), which can be modified. This ensures that applications that may require DNS lookups run seamlessly when DNS servers have not been configured in the Internal DNS table and IP Interfaces table (i.e., last resort). Currently, the default DNS servers are used only for certain applications – Auto-Update mechanism (for loading files through SNMP and CLI copy), pinging remote hosts (CLI ping command), and the updating SBC Configuration Wizard template.

The following configuration parameters have been added for this feature:

- Web Interface: New DNS Settings page (Setup menu > IP Network tab > DNS folder > DNS Settings) - 'Default Primary DNS Server IP' (default 8.8.8.8) and 'Default Secondary DNS Server IP' (default 8.8.4.4)
- ini parameters: DefaultPrimaryDnsServerIp and DefaultSecondaryDnsServerIp
- CLI: (configure network > dns settings > dns-default-primary-server-ip / dns-default-secondary-server-ip

Applicable Applications: All.

Applicable Products: All.

2.24.1.15 Music-on-Hold from External Audio Streamer via FXS Gateway

The device's Gateway application now supports playing music-on-hold (MoH) whose source is from an external, third-party media player. The device can then play this media to any external IP system (for example, a softswitch, media gateway or SBC) or use it for calls that it processes (only SBC application). Thus, the device functions like an IP media server, except that the original source of the media is from an external player.

The external media source is connected to the device's FXS port through a telephone adapter (for FXS emulation). The FXS port is always in off-hook state, continuously receiving media (for example, music or advertisements) from the external media source. Up to two FXS ports can be used for this feature, where each port can play the media to up to 20 concurrent call sessions. In addition, each FXS port can be dedicated for a different purpose, for example, one port can play MoH to normal users while the other port can play MoH to contact centers.

The feature is enabled by configuring the existing parameter 'IP2Tel CutThrough Call Behavior' parameter (TelProfile_IP2TelCutThroughCallBehavior) to the new optional value, **CutThrough+Streaming** (3). The maximum number of concurrent calls that can be established for this MoH feature (for all FXS ports) is configured by the new parameter, MaxStreamingCalls (gw-analog-fxs > max-streaming-calls).

Applicable Applications: Gateway (FXS).

Applicable Products: MP-1288, Mediant 5xx; Mediant 8xx; Mediant 1000.

2.24.1.16 Music-on-Hold from External Audio Streamer for SBC Calls

The device can now play Music on Hold (MoH) from an external IP-based media (audio) source (streamer) for SBC calls that have been placed on hold. Up until now, the device could only be configured to play the local default hold tone or a tone defined in an installed PRT file.

Only one external media source can be configured for MoH. The device can play MoH from the external streamer to up to 20 concurrent call sessions (on-hold parties).

The feature is configured using the following new parameters/options:

- New table parameter: External Media Source table (ExternalMediaSource) – specifies the IP Group of the external media source
- New optional value, **External** for the 'Play Held Tone' (IpProfile_SBCPlayHeldTone) parameter. The existing optional value **Yes** has been renamed **Internal** (to play hold tone from PRT or default)

The feature supports HA switchover, whereby the device continues playing MoH to the calls that were placed on hold before the switchover.

Applicable Applications: SBC.

Applicable Products: MP-1288; Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.24.1.17 Dial Plans for Routing Gateway Calls

Dial plans (and their tags) can now be used in the routing mechanism for Gateway calls (Tel-to-IP and IP-to-Tel). The Dial Plans are used as input criteria for locating a matching routing rule.

To support this feature, the following new configuration parameters have been added:

- 'Tel-to-IP Dial Plan Name' (Tel2IPDialPlanName) - global parameter that selects the Dial Plan for Tel-to-IP routing
- 'IP-to-Tel Dial Plan Name' (IP2TelDialPlanName) - global parameter that selects the Dial Plan for IP-to-Tel routing
- 'Source Tag' (PstnPrefix_SrcTags) and Destination Tag (PstnPrefix_DestTags) – selects the Dial Plan tag for a specific IP-to-Tel routing rule
- 'Source Tag' (Prefix_SrcTags) and Destination Tag (Prefix_DestTags) – selects Dial Plan tags for a Tel-to-IP routing rule

Applicable Applications: Gateway.

Applicable Products: MP-1288; Mediant 5xx; Mediant 8xx; Mediant 1000.

2.24.1.18 Enhanced Packet Loss Concealment

The device now supports packet loss concealment (PLC) for SBC legs using voice coder G.711 with 20-msec packet interval. This technique is used to mask the effects of lost or discarded packets. Therefore, enabling PLC may enhance the device's Quality of Experience (QoE) capabilities by improving MOS scores when packet loss rate is less than 10.

To enable PLC, the following parameter has been added to the IP Profile table:

- CLI: configure voip > coders-and-profiles ip-profile > sbc-enhanced-plc

- ini File: IpProfile_SBCEnhancedPlc
- Web: Enhanced PLC

Note that this feature requires DSP resources.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.24.1.19 SBC User Info Table Activation Changes

The maximum number of allowed rows in the SBC User Info table is according to the number of far-end users that is defined in the device's License Key ("Far End Users (FEU)"). However, as each product supports a maximum number of rows software-wise, the licensed FEU cannot exceed this. For example, if the licensed FEU is 20 and the maximum number of supported rows by the device is 10, the maximum rows will be 10. If the FEU is 5, then the maximum number of rows will be 5.

For the table to be supported, the EnableUserInfoUsage parameter must be enabled and the FEU in the License Key must have a value greater than 0. This also applies to LAD and OVR applications.

Note that the FEU license also refers to the number of allowed registered users (however, this cannot exceed the device's inherent maximum number of supported registered users).

Applicable Applications: All.

Applicable Products: All.

2.24.1.20 Enhanced User Info File Handling

The following enhancements were made to the User Info functionality:

- The SBC User Info table now supports up to 20,000 users (previously, it was up to 3,000). This applies only to Mediant 2600, Mediant 4000, Mediant 9000 and Mediant Software products providing 8-GB RAM or more
- User Information files can now be imported and exported as .csv files, in the SBC User Info table and Gateway User Info table. This is done using the following new configuration updates:
  - Web Interface: New Import and Export commands in the existing Action drop-down list located on the toolbar of these tables.
  - CLI (under configure voip > sip-definition proxy-and-registration):
    - user-info gw-user-info|sbc-user-info export-csv-to <URL>
    - user-info gw-user-info|sbc-user-info import-csv-from <URL>
    - New commands for the Auto-Update mechanism – configure system > automatic-update > gw-user-info | sbc-user-info
  - ini File: SBCUserInfoFileUrl and GWUserInfoFileUrl

  Previously, these files could only be imported (loaded) to the device (as auxiliary files using the Auxiliary Files page). As loading the files using the Auxiliary File page is being phased out, it is recommended to import the tables using the new method.

Applicable Applications: All.

Applicable Products: All.

2.24.1.21 Dial Plan and User Info Table Parameters Exposed in ini File

The following tables are now exposed (previously were hidden) in downloaded ini configuration files as well as in the output of the show running-config CLI command:

- DialPlanRule
- SBCUserInfoTable

- GWUserInfoTable

Applicable Applications: All.

Applicable Products: All.

2.24.1.22 Call Preemption for Emergency Calls by Routing Server

The REST API now supports obtaining (GET) and configuring (PUT) the 'SBC Preemption Mode' (SBCPreemptionMode) parameter for SBC calls, and the 'Call Priority Mode' (CallPriorityMode) parameter for Gateway calls.

This feature provides support for implementing call preemption for emergency calls (such as 911) by the routing server (for example, AudioCodes ARM). If the device is enabled for call preemption for emergency calls (SBC and/or Gateway), the routing server determines whether the incoming call is an emergency call or not and if so, handles the routing decision accordingly (i.e., preempts a non-emergency call if the maximum call capacity of the device is reached to allow the emergency call to be routed).

The REST API URL resource */api/v1/rmConfig/globals* now includes the new parameters "preemptionmode" (enables call preemption for SBC) and "callprioritymode" (enables call preemption for Gateway):

```
<OAMP IP>/api/v1/rmConfig/globals/preemptionMode
<OAMP IP>/api/v1/rmConfig/globals/callPriorityMode
```

Applicable Applications: SBC; Gateway (IP-to-Tel).

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 1000B; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.24.1.23 ENUM Query Enhancement for Call Setup Rules

This feature provides enhanced keyword support for using the results of ENUM queries in Call Setup rules ('Query Target' parameter configured to **ENUM**). Up until now, only the 'enum.result.url' keyword could be used in the Action and Subject fields. Now, the ENUM result can be drilled down to specific parts of the URL using the syntax *enum.result.url.<x>*, where *x* can be 'user', 'host', 'type', 'mhost', 'userphone', 'looseroute', 'bnce', 'cause', 'user', 'transport-type', 'ac-int', and 'param' (for example, enum.result.url.user).

Applicable Applications: SBC and Gateway.

Applicable Products: All.

2.24.1.24 Enhanced Call Admission Control

Call admission control (CAC) configuration has undergone the following enhancements:

- The Admission Control table has been renamed and changed to a parent-child table structure to allow the configuration of CAC profiles, where each profile can have multiple CAC rules:
  - Call Admission Control Profile table (SBCAdmissionProfile) - defines a CAC profile name
  - Call Admission Control Rule table (SBCAdmissionRule) - defines multiple CAC rules per profile
- CAC rules (profiles) are now assigned to IP Groups, SRDs and SIP Interfaces in their respective tables, using a new table parameter "CAC Profile". Previously, these SIP entities were assigned to the CAC rule within the former Admission Control table.
- CAC rules can now be configured (rate and max. burst) per user, which limits the number of calls made per user.

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.25 IDS Blacklist Display in Web Interface

Remote hosts that are currently blacklisted (considered malicious) by the device's Intrusion Detection System (IDS) feature are now displayed in the Web interface. Up until now, this was supported only through CLI (show voip ids blacklist active).

This new feature is supported by the new read-only Web page, IDS Active Black List (Monitor menu > Monitor tab > Network Status folder > IDS Active Black List).

In addition, a new CLI command has been added to remove entries from the IDS Active Black List table:

```
# clear voip ids blacklist {all|<Removal Key>}
```

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.26 Improved IDS SNMP Alarm Descriptions

IDS SNMP alarms now include a more detailed description of the reason why the alarm was raised.

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.27 High-Availability for AWS Environments

High-Availability is now supported when the device operates in AWS environment. Previously, it could only be in standalone.

AWS automatic NAT Elastic IP Addresses configuration is now supported.

Applicable Applications: SBC.

Applicable Products: Mediant VE.

2.24.1.28 Initial HA Configuration from Single INI File

Quick-and-easy initial HA setup is now supported, by loading the same configuration (.ini) file with special configuration to both standalone devices. The active (local) and standby (remote) devices are identified by MAC address. The feature is also useful for HA backup configuration. Once HA is up and running, a backup of the ini file from the active device can be done and when HA failure occurs (for whatever reason), the file can be re-loaded to the devices to restore HA.

The following new ini file parameters configure this feature:

- First device:
  - HALocalMAC - MAC of device
- Second device:
  - HARemoteMAC - MAC of device
  - HARemoteUnitIdName –name of device
  - HARemotePriority – preempt mode

Applicable Applications: All.

Applicable Products: Mediant 500; Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.24.1.29 Changes in Offline HA Parameters

The following parameters no longer require a device reset for their settings to take effect:

- 'Preempt Priority' (HAPriority)
- 'Redundant Preempt Priority' (HARemotePriority)

- 'Preempt Mode' (HARevertiveEnabled)

Applicable Applications: All.

Applicable Products: Mediant 500; Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant VE/SE.

2.24.1.30 Packaged Configuration File Load and Save

A single packaged file containing multiple configuration-related files can now be loaded to or saved from the device. The packaged file is a TAR (Tape ARchive) file (.tar) compressed with gzip. The feature can be used for backing up full configuration and then restoring it to the device in case of device configuration failure (for whatever reason), or for loading the backed-up configuration file to other devices requiring similar configuration.

The packaged file can include the following files:

- ini.ini (ini configuration file)
- LOGO.dat (image file used as the logo in the Web interface)
- FAVICON.dat (favicon file used for Web browsers)
- CPT.dat (Call Progress Tone file) - present only if a CPT file was previously loaded to the device
- PRT.dat (Pre-recorded Tone file) - present only if a PRT file was previously loaded to the device
- AMD.dat (Answer Machine Detection file) - present only if a CPT file was previously loaded to the device
- SBC_Wizard.dat (SBC Configuration Wizard template file)
- CAS_<ID>.dat (CAS file) - present only if a CAS file was previously loaded to the device
- DPLN.dat (Dial Plan file) - only for backward compatibility of previous versions that supported a Dial Plan file; for current versions, the Dial Plan is included in the ini file
- Certificate files (<ctx_id>.crt, <ctx_id>.root, <ctx_id>.pkey)
- DialPlanRule.csv (Dial Plan file) - present only if the device was configured with Dial Plan rules
- CSV files (for example, for Dial Plans and User Info)

This feature can be done through the following management interfaces:

- SFTP: The packaged configuration file (configuration-package.tar.gz) can be downloaded (Get) from the device through SFTP. The file is located in the device's root (/) directory. The SFTP client needs to authenticate itself with the SFTP server (i.e., the device). Access is granted only to users with Security Administrator level.
- Web interface: Existing Configuration Files page, using the new **Save Configuration Package** and **Load Configuration Package** buttons
- CLI: The following new command has been added to the copy command:

```
# copy configuration-pkg from|to <URL>
```

- Ini File: ConfPackageURL

The packaged file is saved with the filename "ConfBackupPkg<Serial Number>.tar.gz". After loading a package file, the device automatically resets with a save to flash.

Note:

- Software file (.cmp) is not supported.
- CAS files cannot be replaced when there are active calls; all trunks must be stopped before CAS files can be replaced.

Applicable Applications: All.

Applicable Products: All.

2.24.1.31 Voltage Configuration for FXS MWI and Phone Lamp

The voltage level mode (low or high) that the FXS port generates to a connected phone for lighting the phone's lamp (LED or NEON type) used for indicating a message in waiting (MWI) can now be configured. This is supported by the following new parameters:

- EnableLowVoltageMwiGeneration
- LedMwiOnDurationTime
- LedMwiOffDurationTime
- NeonMwiOnDurationTime
- NeonMwiOffDurationTime

Applicable Applications: Gateway (FXS).

Applicable Products: MP-1288.

2.24.1.32 Auto-Completion for Message Syntax

Auto-completion for parameters whose values are configured using special syntax is now supported in the Web interface. For these parameters, an Editor button is displayed alongside their fields, which when clicked, opens a syntax editor. As text is typed in the field, the user is prompted with optional syntax.

The feature is supported in the following configuration tables:

- Malicious Signature ('Pattern' field)
- Call Setup Rules ('Search Key', ' Condition', 'Action Subject' and 'Action Value' fields)
- Message Manipulations ('Message Type', 'Condition', 'Action Subject' and 'Action Value' fields)
- IP-to-IP Routing ('Internal Action' field)
- Message Conditions ('Condition' field)
- Pre-Parsing Manipulation Sets ('Message Type' and 'Replace-With' fields)

Applicable Applications: All.

Applicable Products: All.

2.24.1.33 Select All Check Box for Selecting All Activity Types to Report

A 'Select All' check box has been added under the Activity Types to Report group on the Web interface's Syslog Settings page (Troubleshoot menu > Troubleshoot tab > Logging folder > Syslog Settings). This enables the administrator to select (or deselect) all activity types with one click.

Applicable Applications: All.

Applicable Products: All.

2.24.1.34 SSH Server Enabled by Default

The device's embedded SSH server is now enabled by default. It can be disabled using the existing parameter 'Enable SSH Server' (SSHServerEnable).

Applicable Applications: All.

Applicable Products: All.

2.24.1.35 TDM-to-SBC License Displayed in Management Interfaces

The TDM-to-SBC license is now shown in the License Key that is displayed in the management interfaces (for example, Web interface's License Key page). It is displayed on two lines – one showing the number of SBC sessions that can currently be derived from Gateway resources ("TDM-to-SBC Sessions") and one showing whether the feature is licensed ("TDM-to-SBC").

Applicable Applications: All.

Applicable Products: All.

2.24.1.36 License Key Mode Indication

The Web interface now displays the type of License Key installed on the device (locally installed, obtained from the OVOC license pool, or Floating License). This is displayed in the new 'Mode' field on the License Key page.

Applicable Applications: All.

Applicable Products: All.

2.24.1.37 Core Allocation Optimization for Services

The device's CPU cores usage can be optimized for a specified profile to achieve maximum capacity for that profile. The supported profiles include:

- SIP profile –improves SIP signaling performance, for example, SIP calls per second (CPS)
- SRTP profile –improves maximum number of SRTP sessions
- Transcoding profile – enables all DSP-required features, for example, transcoding and voice in-band detectors

Once a profile is selected, the device optimizes cores allocation to achieve maximum performance according to the selected profile. For example, if the deployment involves mainly SRTP sessions (and doesn't require DSP-required features), it is recommended to optimize core allocation for SRTP.

Configuration is done by the new parameter, SBC Performance Profile (SbcPerformanceProfile).

Note:

From Version 7.2.202 (incl.), when using any transcoding/DSP functionality, the device must be configured to use the "Optimized-for-Transcoding" profile, by the SBCPerformanceProfile parameter. However, the parameter's value is not automatically set to this profile in all upgrade scenarios:

- If you are upgrading from a version earlier than 7.2.158, for the parameter to be automatically set to this profile, first upgrade the device to Version 7.2.158 before upgrading it to Version 7.2.2xx.
- If you are upgrading directly to Version 7.2.2xx from a version earlier than 7.2.158, then configure the device to this profile, using any of the below management methods. Once you have configured the parameter, you must save your configuration to flash with a device reset.
  - √ ini file: SBCPerformanceProfile = 2
  - √ CLI: configure system > sbc-performance-settings > sbc-performance-profile optimized-for-transcoding
  - √ Web: 'SBC Performance Profile' = Optimized-for-Transcoding (SBC General Settings page - Setup menu > Signaling & Media tab > SBC folder > SBC General Settings)

Note:

From Version 7.2.202 (incl.), when upgrading the License Key to enable any DSP functionality (including transcoding), the SBCPerformanceProfile parameter must be configured to the Optimized-for-Transcoding profile, as the **parameter's value is not automatically set after installing such a License Key. If not configured, DSP functionality will remain disabled.**

Therefore, to ensure DSP functionality, load the new License Key **without resetting** the device, and then verify that the device is configured to this profile, using any of the following management methods:

- ini file: SBCPerformanceProfile = **2**
- CLI: configure system > sbc-performance-settings > sbc-performance-profile optimized-for-transcoding
- Web: 'SBC Performance Profile' = Optimized-for-Transcoding (SBC General Settings page - Setup menu > Signaling & Media tab > SBC folder > SBC General Settings)

If the device is not configured to this profile, then configure it as shown above. Once configured, reset the device with a save-to-flash for the parameter and the new License Key to take effect.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant VE/SE.

2.24.1.38 Default OAMP Interface Changes

The following changes (shown highlighted) have been made to the default OAMP network interface address per product:

Product	IP Address	Prefix Length	Default Gateway
MP-1288	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 500L SBC/Gateway	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 500 SBC/Gateway	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 800 SBC/Gateway	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 1000 SBC/Gateway	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 2600 SBC	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 4000 SBC	192.168.0.2	24	0.0.0.0 (instead of 192.168.0.1)
Mediant 9000 SBC	192.168.0.2 (instead of 192.168.0.1)	24	0.0.0.0 (instead of 192.168.0.1)
Mediant VE/SE SBC	192.168.0.2 (instead of 192.168.0.1)	24	0.0.0.0 (instead of 192.168.0.1)

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.24.1.39 Alarms Tables Enhancements

The Active Alarms table and Alarms History table in the Web interface now display alarms from newest to oldest. In other words, the most recently raised alarm is shown first in the list. In addition, the Active Alarms table is now automatically refreshed every 60 seconds.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.24.1.40 Handling of Retry-After Header in SIP 503 Responses

The device can be configured for applying different behaviors when SIP 503 (Service Unavailable) containing a Retry-After header is received in response to a SIP message (e.g., REGISTER) sent to the proxy server. The configuration supported by a new global ini file parameter, RetryAfterMode. In certain scenarios (depending on the value of the parameter), the device considers the proxy as offline (down) for the number of seconds specified in the Retry-After header. During this timeout, the device does not send any SIP messages to the proxy.

Applicable Applications: SBC/Gateway.

Applicable Products: All.

2.24.1.41 Enhanced Cross Validation for UDP Port Configuration

The device's cross validation for conflicting configuration has been enhanced to include port (UDP/TCP/TLS) settings between Media Realm Extensions and other configuration entities (i.e., Media Realms and SIP Interfaces) using the same IP network interface. (Ports configured for such entities must not overlap.)

Applicable Applications: SBC/Gateway.

Applicable Products: All (Except Mediant 1000).

2.24.1.42 Improved Distribution of REGISTER and SUBSCRIBE Requests

The device's behavior has been slightly modified when configured to overwrite the expiry time in the Expires header of SIP 200 OK responses for user registration or subscription requests. The existing parameter that configures this feature—SBCRandomizeExpires—is now an enabled-disabled parameter and when enabled, the new expiry time generated by the device is based on fixed algorithms (for more information, refer to the User's Manual). This change has resulted in an improved distribution of registration and subscription requests over time.

Applicable Applications: SBC.

Applicable Products: All.

2.24.1.43 Variable Usage Enhancements for Message Manipulations

Using variables to store SIP message data for Message Manipulation has been enhanced:

- Variables can now be used for each registered far-end user:

  ```
  var.user|peer-user.<Variable Name>
  ```

  Where *user* and *peer-user* denote the two users in the session – current leg and peer leg.
- Instead of indices (numbers), a variable name is used for all variables types.

  ```
  var.global|session|call.src|call.dst.<Variable Name
  ```

  The variable name can include alphanumeric and hyphens (-).

  Previous syntax is still supported (variable indices are considered as names).
- All types allow up to 10 variables, where all 10 can have a summation of 690 characters.
- Variable value can be any string.
- Outbound Message Manipulation applied to a Server-type IP Group (i.e., proxy) can now access the param.ipg.src|dst.<x> syntax when the device sends an un-REGISTER. This is done using var.|peer-user.<Variable Name>

Applicable Applications: All.

Applicable Products: All.

2.24.1.44 Parameter Name Change from "Prefix" to "Pattern"

For parameters in configuration tables whose name contains "Prefix", the "Prefix" string has been replaced with "Pattern". This was done to accurately reflect the functionality of these parameters, which handle not only the prefix of numbers and SIP URIs, but also the suffix, etc. The supported patterns (notations) are documented in the User Manuals.

Applicable Applications: All.

Applicable Products: All.

2.24.2 Known Constraints

This section lists known constraints.

Table 2-31: Known Constraints in Version 7.20A.200.019

Incident	Description
145291	Sometimes after a Web session timeout or an HA switchover (for devices in HA), the user may be redirected to a URL such as http://x.x.x.x/PressLogOff or http://x.x.x.x/HostedTPFrontPanel. A workaround is to refresh the Web session (F5). **Applicable Products:** All.
148040	When using Mediant SBC VE in Hyper-V environments together with EMS or OVOC, the SBC identifier in the EMS / OVOC server (also known as "Serial Number") changes to a new value. If the KeepAlive functionality from the Hyper-V SBC VE to EMS/OVOC server is not enabled, EMS/OVOC will automatically detect the change of the SBC identifier for the existing SBC, within a short time, and there is no need for additional actions. If there is NAT between the SBC and EMS/OVOC server and the KeepAlive functionality from the SBC to the EMS/OVOC server is enabled, the SBC will be detected by the EMS/OVOC server as a completely new SBC device and it should be treated by the Administrator this way (meaning, updating configuration files, updating License Pool allocation to the SBC, etc.). In addition, the old SBC identifier used by the Mediant SBC should be manually removed from the EMS/OVOC server as it will no longer be used. **Applicable Products:** Mediant VE SBC.
150207	When the device is running on Amazon Web Services (AWS) cloud platform and a user manually modifies the IP Interfaces table prior to upgrading the device's software to Ver. 7.20A.200, the device sometimes becomes non-operational. **Applicable Products:** Mediant VE SBC.
150364	The Media Transcoding Cluster feature (MTC) is not supported in this release. **Applicable Products:** Mediant 9000 SBC; Mediant VE.
150850	Devices running software version 7.2.106 must first be upgraded to 7.20A.158.009 before being upgraded to 7.20A.200. **Applicable Products:** Mediant 9000 SBC; Mediant VE/SE.
151219	Devices running software version 7.0 must first be upgraded to 7.20A.158 before being upgraded to 7.20A.200. **Applicable Products:** Mediant 9000 SBC; Mediant VE/SE.
150887	Running devices in Hyper-V environments is currently not supported. **Applicable Products:** Mediant VE SBC.
-	Devices running on Amazon Web Services (AWS) cloud platform do not support HA. **Applicable Products:** Mediant VE SBC.
-	The device does not support the GEN 10 HP server. **Applicable Products:** Mediant 9000 SBC.
150004	Core Dump file is not saved to the device's flash memory due to lack of memeory resources. **Applicable Products:** Mediant 800.

2.24.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-32: Resolved Constraints in Version 7.20A.200.019

Incident	Description
150950	The device generates a warning message notifying that the TLS certificate expired in 1970, even though it hasn't expired. **Applicable Products:** All.
150823	If the device receives an SBC call whose destination number is "*", the device replies with a SIP 180 response that contains "Contact: *". As a result, the call fails. **Applicable Products:** All.
150746	If more than one user has logged in to the device from the OVOC management interface, the device's Web interface displays the incorrect logged-in user. **Applicable Products:** All.
150652	The device is configured with a Proxy Set that has four addresses (alt. routing between addresses). If a new request is received before the next proxy responds to the OPTIONS request, the device rejects the new request (alternative route is not done on the new request) until the next working proxy responds. As a result, the online proxy is incorrectly identified as offline. **Applicable Products:** All.
150640	Two devices report the same serial number to the EMS (SNMP). As a result, the EMS is unable to provide management to the device. **Applicable Products:** Mediant VE/SE.
148040	Two devices report the same UUID to the EMS (SNMP). As a result, EMS cannot distinguish between the two. **Applicable Products:** Mediant VE/SE.
150588	The **show active calls** CLI command erroneously displays a duration of "0" for all calls. **Applicable Products:** All.
150492	Exporting the Dial Plan fails. **Applicable Products:** All.
150482	If during a call disconnect, the device receives a re-INVITE to the same call (To-tag present), it erroneously reports the call to SEM without the call ID. **Applicable Products:** All.
150480	Authentication of SIP PUBLISH messages does not function correctly and as a result, is not authenticated. **Applicable Products:** All.
150471	When fragmented packets are received on a port that is monitored by the Access List, the Access List drops the second part of the packet. As a result, the call fails. **Applicable Products:** All.
150291	When a DNS query responds with no (0) proxies, the device uses the wrong proxy IP address and Classification fails. As a result, the call fails. **Applicable Products:** All.
150231	If the prefix in the Dial Plan is configured with the range 0-9, the Dial Plan does not function. **Applicable Products:** All.

Incident	Description
150223	If the PSAP does not answer an emergency call, resources are not made available and after a while, no more ELIN calls can be processed. A workaround is to reset the device. **Applicable Products:** Gateway.
150190	For the Gateway application, a problem in the detection of DTMF-based caller ID causes no voice from the PSTN to the IP (one-way voice). A workaround is to disable caller ID. **Applicable Products:** Mediant 5xx, Mediant 8xx; MP-1288.
150140	When using an LDAP server for login authentication and the LDAP server responds with multiple groups which the user is a member of, the device uses the access level of the first member instead of the highest access level of all the members. As a result, the user cannot log in. **Applicable Products:** All.
150056	PSTN mapping of Unicode characters do not support 3 and 4 bytes ASCII characters. As a result, the device does not forward the calling name. **Applicable Products:** Digital Gateway.
150033	If the device is configured to send SIP OPTIONS to check the Tel-to-IP routing connectivity (ALTROUTINGTEL2IPENABLE parameter), an error message erroneously appears. **Applicable Products:** Gateway.
149792	When the device performs alternative routing of a REGISTER due to receiving a SIP 302 with two contacts and the two receive a 4xx response, the device does not forward the 4xx to the originator. As a result, a problem in registration occurs. **Applicable Products:** SBC.
149755	The device reports incorrect call quality to SEM **Applicable Products:** SBC.
149454	The CLI command **traceroute** does not function for IPv6 addresses. **Applicable Products:** SBC.
147996	When loading CLI configuration to a new device, the order of the commands are not maintained, which causes an error in configuration. **Applicable Products:** All.
147590	HTTPs only mode cannot be configured (only HTTP) for HTTP Proxy. **Applicable Products:** SBC.
147556	Upon receipt of a SIP 503 from a proxy in response to a REGISTER with a Retry-After header, the device marks the proxy as offline (alarm raised). **Applicable Products:** SBC.
146780	When loading a User Info file without a password, the device adds the user to IP Group 0. **Applicable Products:** SBC.
146643	If the device receives an ALERT with two PIs and only one of them is "= 8", it does not play early media. **Applicable Products:** Digital Gateway.
143381	When trying to create a snapshot, an error message appears. **Applicable Products:** Mediant VE.

Incident	Description
140194	When using the virtual console, if Ctrl+Alt+Delete key combination is pressed, the device resets. **Applicable Products:** Mediant VE.

2.25 Patch Version 7.20A.200.550

This patch version includes only resolved constraints.

2.25.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-33: Resolved Constraints in Version 7.20A.200.550

Incident	Description
153075	If the device has raised a major alarm due to the crossing of an IDS threshold and it then needs to raise a new minor IDS alarm before the major IDS alarm has cleared, the device is unable to provide an alarm description for this new event. As a result, the device crashes (resets). **Applicable Products:** SBC.

2.26 Patch Version 7.20A.202.112

This patch version includes new features, known constraints, and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3082, and EMS/SEM Version 7.2.3106.

2.26.1 New Features

New features introduced in this version include the following:

2.26.1.1 New Mediant Software SBC Product - Mediant Cloud Edition (CE) SBC

AudioCodes has launched a new software-based SBC product, named *Mediant Cloud Edition (CE) SBC*, specifically designed to be hosted in cloud computing environments (currently supported on Amazon Web Services, with planned support for additional environments). The main benefit of the Mediant CE which differentiates it from other AudioCodes software-based SBCs (like the Mediant VE SBC) is that it's deployed as a multi-virtual machine (VM) SBC stack, providing elasticity using an Elastic Media Cluster architecture.

The Mediant CE includes a Signaling Component (SC) which handles the SIP signaling, while offloading all media traffic handling (RTP/SRTP) to Media Components (MC). A Cluster Manager embedded in the Mediant CE SBC determines which MC handles each media flow, allowing load balancing of media between the MCs. MCs are responsible for handling all aspects of their allocated media, including transcoding and forwarding.

Deployment and management of the Mediant CE is facilitated by AudioCodes new Stack Manager application, also introduced in this release. The Stack Manager is deployed in the Amazon VPC created for Mediant CE. The Stack Manager is used for managing the SBC software stack and implements a virtual network function manager functionality. Using the Stack Manager allows for horizontal scalability of the SBC based on changing traffic. The Stack Manager also provides a standard, open API allowing external DevOp tools and orchestrators to easily manage the Mediant CE operation and elastic scalability.

Mediant CE supports the full feature set of AudioCodes SBCs, maintaining familiar operation, management and APIs of the Mediant SBC product line.

The following main management features have been added to support the feature:

- CLI:
  - Auto-Update of CMP file for Media Components (MC):

```
(config-system)# automatic-update
(auto-update)# vmc-firmware <URL>
```

  - Copy CMP of MC:

```
# copy vmc-firmware {from|to} <URL>
```

  - The CMP file is displayed in the output of the dir command.
- Web / INI / CLI:
  - EnableMtcSbc has been replaced by a new parameter that defines the Cluster mode: 'Cluster Mode' / SbcClusterMode / configure network > mtc settings > sbc-cluster-mode)
  - 'Device Role' / SbcDeviceRole / configure network > mtc settings > sbc-device-role

Applicable Applications: SBC.

Applicable Products: Mediant CE.

2.26.1.2 New Mediant 800 Hardware Revision – Mediant 800C

A new hardware revision—Rev. C—has been released for the Mediant 800. In addition to the interfaces and features offered by Mediant 800B, Mediant 800C also supports the following:

- Dual power supply – in addition to the AC power supply (provided by default), the device can be ordered for DC power supply, which is typically used for power redundancy (if AC power supply fails, fallback to the DC power supply is done).
- Up to 4 x E1/T1 (PRI) port interfaces.
- Single SBC capacity profile, supporting up to 2,000 users and 400 sessions.
- 110 transcoding sessions.

Applicable Applications: All.

Applicable Products: Mediant 800.

2.26.1.3 SBC Capacity Licensing Model - Floating License

AudioCodes offers a new SBC capacity licensing model for the device, called "Floating License". The Floating License is a network-wide capacity license that is dynamically shared among multiple devices. It is initially purchased as a set of licenses based on your estimated requirements, providing a pool of SBC-related capacity resources—SBC sessions, SBC signaling sessions, SBC media sessions, far end users (user registrations), and transcoding sessions—for all your deployed devices. The key benefit of the Floating License is that it allows the devices to exceed this initially ordered capacity, typically due to business growth or underestimation of initial requirements, and bills the Customer at the end of the month for the additional SBC resources that were consumed.

The Floating License is managed by AudioCodes OVOC tool and AudioCodes Cloud License Manager system. The devices participating in the Floating License need to be configured in OVOC and on each device. Once connected to OVOC, these devices are "open" to use any capacity. However, capacity is limited by the device's inherent capacity support and by an optional user-defined limit called Allocation Profile, which specifies the capacity per SBC license type.

The devices report their SBC resource consumption of each SBC license type to OVOC at set intervals (typically, every five minutes). OVOC sends a report to the Cloud License Manager every 24 hours with all the SBC resource usage in the last 24 hours by the devices. At the end of the month, AudioCodes analyzes the resource consumption reports in the Cloud License Manager. If consumption has exceeded the capacity as specified by the Floating License, AudioCodes bills the Customer for the extra capacity used.

The following main management features have been added to support the feature:

- Web interface:
  - The feature is enabled and an Allocation Profile is configured on a new page, Floating License (Setup menu > Administration tab > Maintenance folder > Floating License)
  - The existing License Key page displays the SBC capacity specified on the device to obtain from the Floating License
- CLI and ini file: Support the enabling of the Floating License and configuration of the Allocation Profile
- SNMP has two new alarms:
  - acCloudLicenseManagerAlarm – sent for certain OVOC-device connectivity issues and other related issues.
  - acFloatingLicenseAlarm – sent when the device has insufficient memory resources for the capacity configured for the Allocation Profile

Applicable Applications: SBC.

Applicable Products: All.

2.26.1.4 License Key of Redundant Device Included in INI File

The License Key of the Redundant device is now also included in the ini file of the Active device, for High-Availability systems. Up until now, the License Key was included only in the Redundant device's ini file.

Applicable Applications: Gateway/SBC.

Applicable Products: HA Devices.

2.26.1.5 Enhanced Media Transcoding Cluster Feature

The following enhancements have been made to the Media Transcoding Cluster feature:

- CLI command to copy software file (.cmp) for Media Transcoder (MT):

```
copy mt-firmware {from|to} <URL>
```

- Web interface: The folder in the Navigation pane containing the related page items has been renamed to "Media Cluster".

Applicable Applications: SBC.

Applicable Products: Mediant CE.

2.26.1.6 Triggering SBC Actions using SIP NOTIFY Messages

For the SBC application, incoming SIP NOTIFY messages from configured proxy servers can trigger various device actions if the SIP Event header contains certain values:

- 'check-sync;reboot=false': Triggers the Automatic Update feature (if configured)
- 'check-sync;reboot=true': Triggers a device reset.

Up until now, the above was supported only by the Gateway application.

The device now also supports remote trigger by NOTIFY messages for disconnecting all current calls (SBC and Gateway). This occurs if the message's Event header contains the 'soft-sync' value.

The feature is supported by the existing parameter, EnableSIPRemoteReset.

Applicable Applications: SBC.

Applicable Products: All.

2.26.1.7 SIP Session Time Refreshes using UPDATE Messages

For refreshing the timer of currently active SIP sessions, the device can now be configured to send session refreshes using SIP UPDATE messages if the SIP Allow header in the last SIP message received from the user, contains the value "UPDATE".

This is enabled by configuring the existing parameter, 'Remote Update Support' (IpProfile_SBCRemoteUpdateSupport) to the new optional value, "According Remote Allow" (3). If the Allow header does not contain the "UPDATE" value (or if the parameter is not configured to 3), the device uses INVITE messages for session refreshes.

Applicable Applications: SBC.

Applicable Products: All.

2.26.1.8 Enhanced Randomly Assigned SIP Contact User Part

When using Accounts (configured in the Accounts table) and the device is configured to automatically create a random string for the user part of the SIP Contact header in the generated SIP REGISTER request (using the existing UseRandomUser parameter), the string now always begins with a letter (and not a number). For example, the below random string begins with the letter "H":

```
Contact: <sip:HRaNEmZnfX6xZl4@pc33.atlanta.com>
```

Once the Account is registered with the registrar, register refreshing uses this random user part.

In addition, the device stops using the random user part in the following scenarios:

- If the user performs an unregister.
- If the REGISTER request send by the device is not responded by a SIP 200 OK. (Currently, it also includes SIP 3XX responses.)

Applicable Applications: All.

Applicable Products: All.

2.26.1.9 Enhanced Handling of SIP Dialog-initiating INVITE Messages

Optional values 2 and 3 have now been added to the existing global configuration parameter (ini file), VerifyRecievedRequestUri:

- [0] – Does not verify - default (existing value)
- [1] - Verify Request-URI user part for in-call requests (existing value)
- [2] - Verify dialog-initiating INVITE for all required conditions (Via, Source IP and user in Request-URI)
- [3] - Verify both dialog-initiating INVITE as well as in-call requests

In addition, a new global configuration parameter has been added, RegistrarProxySetID (ini file), which configures the Proxy Set associated with the registrar (assuming the SIP Interface has only one registrar). The default value is undefined (-1).

- **Handling Dialog-Initiating INVITEs:** If the VerifyRecievedRequestUri parameter is configured to 2 or 3, and the RegistrarProxySetID is configured, dialog-initiating INVITE requests are allowed from the registrar at which the Accounts (configured in the Accounts table) are registered. For incoming dialog-initiating INVITEs from a specific SIP Interface (from the registrar), the following rules apply (listed according to priority):
  - The top-most Via header must contain a host-resolved IP address of the registrar; otherwise, the device drops the INVITE request.
  - The source IP address must be the same as the IP address of the registrar; otherwise, the device rejects the requests and sends a 403 (Forbidden) response to the registrar.
  - The user, specified in the Request-URI header, must be identical to the Contact user part configured for the associated Account and the Account must be registered. Otherwise, the device rejects the request with SIP 404 (Not Found) response.

  If the RegistrarProxySetID parameter is not configured or no Accounts are configured, the device accepts the dialog-initiating INVITE request.

  This feature is applicable only to the SBC application.
- **Handling In-call Requests:** If the VerifyRecievedRequestUri parameter is configured to 1 or 3, for all received in-call requests (including ACK and CANCEL), the device checks if the Request-URI user part matches the remote Contact user part (i.e., the same as the Contact user configured for the Account). If there is no match, the device rejects the request and sends a SIP 481 response for requests such as BYE and CANCEL or a SIP 404 for other requests, and for ACK it does not send any response.

  This feature is applicable to the Gateway and SBC applications.

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.10 Increase in Maximum Characters for User Part of SIP Messages

The device can now process SIP messages whose user part is up to 60 characters without truncating the characters. In previous releases, the maximum was 50 characters. If the user part has more than 60 characters, the device removes (truncates) the extra characters from the right until there are 60 characters. An example of a lengthy user part (55 characters) is shown below in bold:

```
INVITE sip:0433551202;tgrp=grp001;trunk-
context=98912345678901@10.15.50.31;user=phone SIP/2.0
```

Applicable Applications: All.

Applicable Products: All.

2.26.1.11 SIP Digest URI Handling for Authentication and Authorization

When the device sends a request in response to a SIP 401 (Unauthorized) or 407 (Proxy Authentication Required), it can include or exclude the URI parameters for the Digest URI in the Proxy-Authorization or Authorization header. If configured to include the URI, it sets the Digest URI to the same URI as that received in the original Request-URI. This feature is configured by the new global ini file parameter, SIPDigestAuthorizationURIMode. By default, the Digest URI is sent without URI parameters. Below shows an example of a request with an Authorization header containing a Digest URI (shown in bold):

```
Authorization: Digest username="alice at
audiocodes.com",realm="audiocodes.com",nonce="",response="",uri="s
ip:audiocodes.com"
```

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.12 Handling SIP Messages with Unknown Cryptographic Suites

The device can now be configured to keep or remove unknown cryptographic suites (encryption and authentication algorithms) that may be present in the SDP's 'a=crypto' attribute of incoming SIP messages. In previous releases, the device simply forwarded the SIP message with the unknown cryptographic suites and if the endpoint selected one of the unknown suites (i.e., not supported by the device), the device rejected the call.

To support the feature, a new parameter was added to the IP Profile table, IpProfile_SBCRemoveUnKnownCrypto, which determines whether the device keeps or removes the unknown cryptographic suites before forwarding the SIP message.

Applicable Applications: SBC.

Applicable Products: All.

2.26.1.13 Handling both SDES and DTLS Security in SDP Negotiations

The device can be configured to handle both SDES and DTLS for media security. The feature is configured by setting the existing parameter 'SBC Media Security Method' (IpProfile_SBCMediaSecurityMethod) to the new optional value "Both" (2).

Applicable Applications: SBC.

Applicable Products: All.

2.26.1.14 Graceful Period for Automatic Update of ini File

The Automatic Update feature can now be configured to apply the settings of a downloaded ini file only after a graceful timeout. When the Automatic Update feature is triggered (for example, by a device reset) and the device downloads the ini file from a remote server, the graceful timeout starts and new calls are not accepted during this period. Once the timeout expires, the device terminates all current calls and then applies the configuration according to the downloaded ini file.

The feature is enabled by the new ini file parameter, AupdGracefulShutdown (disabled by default). The graceful timeout is configured by the Web interface's 'Graceful Option' parameter (Maintenance Actions page) or ini file parameter AdminStateLockControl (default is no timeout).

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.15 Enhanced SNMP Alarms for HA Redundant Device

SNMP alarms raised by the Redundant device in a High-Availability (HA) system now provide the following enhancements:

- Alarms of the Redundant device are now reflected correctly in OVOC as alarms (and not as events, as in earlier versions).
- Alarms of the Redundant device are now sent to the Active device and displayed in the Active Alarms table (and History Alarms table) with the alarms of the Active device. Alarms pertaining to the Redundant device are displayed with the alarm source prefix, "Redundant#1/" (e.g., "Redundant#1/EthernetLink#4").
- The Web interface's alarm bell icon (located on the top-right of the GUI window) now displays the number of active alarms raised by both Active and Redundant devices and displays the highest severity of all these active alarms raised.

Applicable Applications: Gateway/SBC.

Applicable Products: HA Devices.

2.26.1.16 Serial Number of Redundant Device in Keep-alive SNMP Traps

SNMP keep-alive traps (acKeepAlive) sent by the active device in a High-Availability system now include the redundant device's serial number. This appears in the acBoardTrapGlobalsAdditionalInfo3 SNMP varbind.

Applicable Applications: All.

Applicable Products: HA Devices.

2.26.1.17 Web Interface's Logo Hyperlinked to Topology View Page

The logo of the Web interface (located in the top-left corner) is now hyperlinked to the Topology View page (Setup menu > Signaling & Media tab > Topology View home icon). The feature offers administrators a quick-and-easy way to navigate to this informative and often needed page.

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.18 Mediant 9000 Hardware Status Display in Management Interfaces

The device's management interfaces now provide more information about the status of the device's hardware components.

- Web interface:

- The Monitor page displays status icons for the Fan and Power modules. If the device is in HA mode, the status is also displayed for the Redundant device. When these icons are clicked, the new Components Status page opens (see below).
- A new page has been added—Components Status page (Monitor menu > Monitor tab > Summary folder > Components Status) —displaying detailed status information of the following hardware components (Redundant device as well for HA mode): fans, power supplies and CPU temperature.

- CLI: The existing command, **show system assembly** now also displays the status of the fans, power supplies and CPU temperature.
- SNMP: The device now supports the following alarms – acFanTrayAlarm, acPowerSupplyAlarm, acBoardTemperatureAlarm.
- INI file:
  - The temperature threshold for raising the acBoardTemperatureAlarm can be configured by the new ini file parameter, HighTemperatureThreshold (default is 70°C or 158°F)
  - The acPowerSupplyAlarm is enabled by a new ini file parameter, DualPowerSupplySupported

Note: The feature will be supported by Mediant 9000 Hardware Rev. B, Mediant 9030 and Mediant 9080 in the next applicable release.

Applicable Applications: SBC.

Applicable Products: Mediant 9000.

2.26.1.19 OVOC Product Key and ID Displayed in Web Interface

The OVOC Product Key and OVOC Product ID are now displayed in the Web interface when the licensing model Floating License or Fixed Pool License is employed. The OVOC Product Key is displayed in the License Key page and the OVOC Product ID in the Floating License page.

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.20 New Default Name for IP Interfaces

When configuring an IP Interface in the IP Interfaces table and no name is configured, the device assigns the IP Interface with a default name in the following format: "Interface_n", where *n* is the row index number. Previously, the format was "InterfaceTable_n".

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.21 No Reset for Number of Media Channels Parameter

The Number of Media Channels parameter (MediaChannels) no longer requires a device reset for its settings to take effect. In addition, if the parameter value is modified to a value that is less than the number of DSPs currently allocated to current calls, these calls are not forcibly terminated by the device (as in previous releases).

Applicable Applications: All.

Applicable Products: All.

2.26.1.22 New CLI Command Structure for Parent-Child Configuration Tables

Configuration tables with parent-child relationships have been restructured in the CLI. Child tables are now accessed from within parent tables. In addition, access to tables can now be done by entity name (as well as by row index, supported in earlier versions).

For example, in earlier versions, to access the child table **dial-plan-rule** Index 1 of **dial-plan** Index 0, the following syntax was used:

```
(config-voip)# sbc dial-plan-rule 0/1
```

Now, it is accessed from within the parent table:

```
(config-voip)# sbc dial-plan 0
(dial-plan-0)# dial-plan-rule 1
(dial-plan-rule-0/1)#
```

Applicable Applications: Gateway/SBC.

Applicable Products: All.

2.26.1.23 Enhanced Filtering for CLI show Commands

Certain **show** CLI commands now support enhanced filtering of their displayed output:

- first <x>: Displays the first *x* number of entries
- last <x>: Displays the last x number of entries
- range <x-y>: Displays a range of entries from *x* to *y*
- descending: Displays the output in descending order

These filters are supported by following **show** commands:

```
show voip calls active
show voip register
show voip proxy sets status
```

Applicable Applications: All.

Applicable Products: All.

2.26.1.24 Improved Location of CLI Command time-zone-format

The time-zone-format command (TimeZoneFormat) has been relocated in the CLI and is now with all the other CDR-related commands -- configure troubleshoot > cdr > time-zone-format.

Applicable Applications: All.

Applicable Products: All.

2.26.1.25 Improved Location of CLI Commands for System Snapshots

The CLI commands for the System Snapshot feature have been relocated in the CLI to the root level (privileged user mode – "#"), facilitating configuration by their fast-and-easy access instead of drilling down CLI commands as in previous releases. In addition, the names of the snapshot commands have also changed slightly, as shown below:

```
# system-snapshot {create|default|delete|help|load|show}
```

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.26.1.26 CLI Command Name Change from "prefix" to "pattern"

For CLI commands in configuration tables whose name contains "prefix", the "prefix" string has been replaced with "pattern". This was done to accurately reflect the functionality of these commands, which handle not only the prefix of numbers and SIP URIs, but also the suffix, etc. The supported patterns (notations) are documented in the User Manuals.

Applicable Applications: All.

Applicable Products: All.

2.26.1.27 New CLI Commands for Assigning Dial Plans to Gateway Routing

Assigning Dial Plans to Gateway routing rules can now be done through the CLI:

- configure voip > gateway routing settings > **ip-dial-plan-name**: IP-to-Tel Dial Plan Name (IP2TelDialPlanName)
- configure voip > gateway routing settings > **tel-dial-plan-name**: Tel-to-IP Dial Plan Name (Tel2IPDialPlanName)

Applicable Applications: Gateway.

Applicable Products: MP-1288; Mediant 500; Mediant 500L; Mediant 800; Mediant 1000.

2.26.1.28 New CLI Commands for Locking and Unlocking Device

Locking the device can now be done through the CLI (root level):

- Locking the device with or without graceful timeout:

```
admin state lock {graceful <Timeout>|no-graceful}
```

- Unlocking the device:

```
admin state unlock
```

The administrative state (locked or unlocked) of the device can be viewed using the following new command:

```
show admin state
```

Applicable Applications: All.

Applicable Products: All.

2.26.1.29 New CLI Commands for Displaying Activity Reports

A logged report of all CLI activities can now be viewed using the following new CLI command:

```
show activity-log [> <URL of Remote Server>]
```

The logged report (output) can also be sent to a remote server (TFTP or HTTP/S), which is defined by URL.

Note that the command displays logged activities only if activity logging has been enabled (configure troubleshoot > activity-log).

Applicable Applications: All.

Applicable Products: All.

2.26.1.30 New CDR Field -- 'Alerting Time'

CDRs can now optionally include the Alerting Time field, which indicates the duration (in milliseconds) between ringing (SIP 180 Ringing) and call answered (SIP 200 OK) or unanswered (Cancel). The field can be added by customizing the CDR format using the CDR Format tables (SBC and Gateway). (The field is included by default in CDRs that are sent to ARM.)

Applicable Applications: All.

Applicable Products: All

2.26.1.31 Name and ID of Media Components (MCs) Included in CDRs

For the Media Cluster feature, CDRs that are sent to OVOC for reporting Quality of Experience, now include the Media Component's name and index number.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.26.1.32 Automatic IP Address for NAT Traversal in AWS Environments

The device's management interface now supports configuration of a NAT IP address mode when the device is deployed in an Amazon Web Services (AWS) cloud-computing environment. The mode can be set to Automatic or Manual. The Automatic mode is used when the AWS environment is configured with an Elastic IP address and enables the device to automatically associate it with the selected source IP interface as the global (public) IP address. To support the feature, the following new parameters have been added to the NAT Translation table: 'Target IP Mode' and 'Automatic Target IP Address' (read-only).

Applicable Applications: SBC.

Applicable Products: Mediant CE.

2.26.1.33 New VoIPerfect Support for Managed G.729 Coder

The device's VoIPerfect feature now also supports Managed G.729 coder. Previously, only Managed Opus was supported. If the Enterprise SBC detects WAN network impairments during a call between the Enterprise SBC and Access SBC, it can adjust the coder's attributes (e.g., bit rate) for that specific call to ensure voice quality is maintained.

This feature also adds support when Managed G.729 operates in MPLS networks. In such scenarios, a new ini file parameter, MPLSMode needs to be enables.

Applicable Applications: SBC.

Applicable Products: Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.26.2 Known Constraints

This section lists known constraints.

Table 2-34: Known Constraints in Version 7.20A.202.112

Incident	Description
-	From Version 7.2.202, when using any transcoding/DSP functionality, the device must be configured to use the "Optimized-for-Transcoding" profile, using the SBCPerformanceProfile parameter. The parameter's value is not automatically set in all upgrade scenarios. If you are upgrading from a version earlier than 7.2.158 and employing transcoding/DSP capabilities, to preserve your transcoding/DSP functionality it is essential that you first upgrade the device to Version 7.2.158 before upgrading it to Version 7.2.2xx. For more information, see Section 2.24.1.37 on page 153. **Applicable Products:** Mediant 9000; Mediant VE.
-	From Version 7.2.202, when upgrading the License Key to enable any DSP functionality (including transcoding), the SBCPerformanceProfile parameter must be configured to the Optimized-for-Transcoding profile option, as the parameter's value is not automatically set after installing such a License Key. If not configured, DSP functionality will remain disabled. For more information, see Section 2.24.1.37 on page 153. **Applicable Products:** Mediant 9000; Mediant VE.
-	The Media Transcoding Cluster (MTC) feature is not supported in this release. **Applicable Products:** Mediant 9000; Mediant VE.
-	The Cluster Mode parameter is supported only by Mediant Cloud Edition (running in an AWS environment). For all other products listed below, it is not supported. **Applicable Products:** Mediant 9000; Mediant VE.
152903	Firewall rules using a specific IP Interface (' Interface Name' parameter) as a matching criterion, do not function if the interface uses a tagged VLAN and/or operates on an Ethernet Group with two ports ("members"). **Applicable Products:** Mediant 9000; Mediant Software.
152811	The device does not accept incoming media traffic if the ingress media interface uses a tagged VLAN and operates on an Ethernet Group with two ports ("members"). (Media traffic also includes traffic from Media Components for the Media Cluster feature). **Applicable Products:** Mediant 9000; Mediant Software.
-	Hardware status indications (fan, power and CPU temperature) are not supported by Mediant 9000 Hardware Rev. B. **Applicable Products:** Mediant 9000.
152859	Music-on-hold played from an external media source (ExternalMediaSource parameter) is not supported by Mediant Cloud Edition. **Applicable Products:** Mediant CE.
-	If the Floating License mode is enabled and the device needs to be downgraded to Version 7.2.200, the Floating License mode must be disabled prior to the downgrade (as it is not supported in this version). **Applicable Products:** All.
153048	Mediant CE does not support IPv6. **Applicable Products:** Mediant CE.

Incident	Description
152094	Incorrect MOS scores are reported at the end of an SBC call between AudioCodes IP Phones (MOS measured during the call is correct). This occurs only in certain scenarios (regarding RTCP and DSPs). **Applicable Products:** MP-1288; Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.
153454	When the 'Mode' parameter (EtherGroupTable_Mode) in the Ethernet Groups table is configured to "2RX/1TX", detection by the device of a failure in the transmitting port is based only on physical link failure (e.g., cable disconnected or port damaged) and not Ethernet link failure. **Applicable Products:** Mediant Software.
-	For Mediant VE, the AWS must have an IAM role and therefore, existing instances must be attached with an IAM role as described in the *Mediant VE Installation Manual*. **Applicable Products:** Mediant VE (AWS).
153614	When the device is configured for HA, ARM is not supported. **Applicable Products:** HA.
153759	When performing a restore-to-default action through the Web interface, the device doesn't reset and therefore, the operation fails. **Applicable Products:** All.
153837	When using the SIPRec feature for the Gateway application, the IP Group representing the SRS must be assigned a Media Realm (in the IP Groups table). **Applicable Products:** Gateway.
153341	When the session timeout (WebUsers_SessionTimeout or WebSessionTimeout) for the Web interface is configured to a value that is less than the duration it takes for the device to reset (4 min.), the Web interface does not display the login screen after reset. **Applicable Products:** All.
-	Loading a system snapshot under traffic is not allowed. **Applicable Products:** Mediant 9000; Mediant Software.
153888	The Product Key displayed in the Web interface is cut off after the "@" symbol. **Applicable Products:** All.

2.26.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-35: Resolved Constraints in Version 7.20A.202.112

Incident	Description
153025	Upon an HA switchover/hitless, a warning message is generated notifying that the TLS certificate expired in 1970, even though it hasn't expired. **Applicable Products:** All HA.
147159	If the IP Interfaces table includes multiple IPv4 IP interfaces that are associated with the same Ethernet Device and each of them has a different Default Gateway, if the last configured of these IP interfaces is deleted, connectivity to the other mentioned IP interfaces is lost. **Applicable Products:** All.

Incident	Description
151962	When importing a partial Dial Plan, new and modified rules were applied, but were not activated. **Applicable Products:** All (SBC).
151998	When the device's Web interface is accessed through OVOC (HTTPS) using the Single Sign On feature, clicking the Troubleshooting tab causes a disconnection with the Web interface. **Applicable Products:** All (SBC).
152749	When the device rejects a call due to CAC, the CDR for this call is not sent to the local storage. **Applicable Products:** All (SBC).
152810	The device merges multiple SIP Reason headers that are received in the incoming leg into one Reason header in the outgoing leg. **Applicable Products:** All (SBC).
152820	The Configuration Package which is downloaded through SFTP doesn't reflect recent configuration changes. **Applicable Products:** All (SBC).
152903	Upgrading an HA system from 7.20A.156.009 to this version, causes the system to become standalone (i.e., no HA). **Applicable Products:** HA Devices.
153358	When an endpoint dials from the PSTN to the device, the endpoint hears the voice from another call processed by the device (instead of hearing a dial tone or ringback tone from the device). **Applicable Products:** Gateway.
153533	When the device is configured to perform Tel-to-IP call forking to two destinations with a delay of 15 seconds between each INVITE (i.e., ForkingDelayTimeForInvite =15), if the first call is disconnected after the selfcheck, the device resets. **Applicable Products:** Gateway.
153635	The Syslog indicates that the LDAP timer has been cancelled when in fact it hasn't. As a result, the LDAP process fails. **Applicable Products:** All.
153645	A "frozen" socket causes the device not to send call statistics to SEM. **Applicable Products:** All.
151056	When using the Pattern Editor in the Web interface, if an incorrect value is entered and the back arrow is used to try and fix the entry, the auto-completion feature does not function correctly. Now, the auto-completion feature is suspended so that the incorrect value can be fixed. To resume auto-completion hints, the cursor must be placed at the end of the field. **Applicable Products:** All (SBC).
152239	In the NAT Translation table, the read-only field 'Automatic Target IP Address' displays the local IP address instead of the global IP address when the 'Target IP Mode' field is configured to Automatic (for AWS environments). **Applicable Products:** Mediant Software.
144263	The SNMP performance monitoring MIB, acPMSIPIPGroupInviteDialogsTotal displays the wrong value. **Applicable Products:** All (SBC).

Incident	Description
148401	The device's SBC application doesn't support SIP NOTIFY with "Event: CheckSync". **Applicable Products:** All (SBC).
149096	Video is not functioning for WebRTC calls. As a result, the call fails. **Applicable Products:** All (SBC).
149304	The device does not send an SNMP alarm for SBC CAC when the limit or reservation threshold is exceeded. **Applicable Products:** All (SBC).
149846	If the device sends an INVITE on the outgoing leg and doesn't receive a 1xx in response, when it receives a CANCEL on the incoming leg, it doesn't forward the CANCEL to the outgoing leg. As a result, the SBC call does not get released. **Applicable Products:** All (SBC).
149925	If the filename of the Web interface's logo file contains spaces, HA synchronization fails. **Applicable Products:** HA.
150353	The maximum length of the user part is only 50 characters and therefore, incoming REGISTER messages with long AOR\Contact values are recorded incorrectly in the device's registration database. **Applicable Products:** All (SBC).
150624	When the device is reset, it sends the wrong MAC address upon bootup. **Applicable Products:** MP-1288; Mediant 5xx; Mediant 8xx; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.
150731	The device resets due to memory allocation overrun. **Applicable Products:** Mediant 1000.
150735	When trying to load an incremental ini file or an incremental CLI script using the REST API (over CURL), the HTTP response 405 (Method Not Allowed) is received and the upload fails. **Applicable Products:** All.
150791	When running debug recording (DR) for the device and it utilizes DSPs, the wrong "ConnectionArray::AuditConnections" messages are displayed in the Syslog after the call ends. **Applicable Products:** All.
150811	The device doesn't respond to SIP NOTIFY messages received from the Application server for the BLF service (fragmented massages); the device sends only ICMP timeout messages. As a result, the call fails. **Applicable Products:** All.
150831	During a direct-media SBC call and in a specific scenario, the device sends a SIP 200 OK with "a=inactive". As a result, no voice occurs. **Applicable Products:** All.
150893	When the device receives a SIP INVITE with a Call-ID greater than 129 characters, the device truncates the Call-ID to 129 characters. As a result, the SBC call fails. **Applicable Products:** All (SBC).
150958	If the OSNInternalVlan is configured to 1, when an HA switchover occurs, the switchover fails and as a result, HA mode fails. **Applicable Products:** Mediant 800 (HA).

Incident	Description
151070	The Dial Plan cannot be exported using the Web interface ("Unauthorized" page appears). **Applicable Products:** All.
155065 / 151078 / 153846	When the device sends a SIP INVITE over UDP and receives "error: Network is unreachable(101)", it keeps trying the retransmission of UDP instead of immediately falling back to the alternative route, which it does only after a few seconds, causing a call delay. **Applicable Products:** All.
151114	If the License Key has no licensed SBC sessions (SBC = 0) and the TDM-to-SBC license is enabled (TDM2SBC > 1), the SIP Interface cannot be modified to SBC application type. As a result, SBC functionality is disabled. **Applicable Products:** Hybrid.
151128	When using the Web interface with the Internet Explorer browser, the SRD filter feature doesn't function. **Applicable Products:** All.
151201	The device continuously reboots (resets) due to voice quality monitoring when trying to connect to OVOC over TLS. **Applicable Products:** All.
151245	The device doesn't handle big or fragmented SIP packets received on network interfaces that are not "OAMP + Control + Media", and resets as a result. **Applicable Products:** All.
151329	When the device runs a Message Manipulation rule to clear the display name from the SIP From header, it crashes (resets) **Applicable Products:** All (SBC).
151334	The CLI command **show voip calls statistics sbc media** fails. **Applicable Products:** All (SBC).
151346	During memory allocation for HA, the Redundant unit resets and HA becomes disabled. **Applicable Products:** HA.
151367	The device fails to switch to the alternative SBC routing rule when the next rule is an IP Group Set. As a result, the call fails. **Applicable Products:** All.
151433	The Fax Re-Routing feature fails. **Applicable Products:** All.
151440	When the parameter DisplayLoginInformation is configured to 1 (enabled), incorrect login information is displayed when accessing the Web interface over HTTPS. **Applicable Products:** All.
151449	A problem occurs with voice in the decoder direction on modem clear-mode calls. As a result, modem calls fail. **Applicable Products:** Gateway.
151556	The device doesn't send SRTP packets in a transcoding call after a hold-retrieve scenario. As a result, one-way voice occurs. **Applicable Products:** Gateway.
151564	The SNMP tables, ifTable and dsx1ConfigTable provide incorrect statistics. **Applicable Products:** Mediant 500L.

Incident	Description
151646	In the early media stage of an SBC call, the device adds silence suppression to the SDP answer. As a result, the call disconnects. **Applicable Products:** All.
151664	The device receives many SIP 18x before DNS resolution is done and therefore, is unable to allocate the necessary resources for each 18x. As a result, the call fails. **Applicable Products:** All.
151671	When upgrading the device to Version 7.20A.200.019, the Ethernet Groups table configuration changes. As a result, the device becomes inaccessible. **Applicable Products:** All.
151690	The device sends a SIP 488 response after receiving a 200 OK to an outgoing re-INVITE due to incorrect fax coder calculations. As a result, the call fails. **Applicable Products:** All.
151699	The SNMP trap community string supports only up to 19 characters. **Applicable Products:** All.
151726	The SNMP fan tray alarm is sent by the device even though the fan tray is ok. **Applicable Products:** All.
151757	The device does not remove unsupported crypto suites (e.g., AES_256) in the outgoing SIP INVITE message and if the remote side chooses an unsupported crypto suite, the SBC call therefore fails. **Applicable Products:** All.
151795	Locally stored CDRs cannot be downloaded through SFTP. **Applicable Products:** All.
151803	The device does not consider the security protocol when creating a re-INVITE for fax calls (proto as RTP/AVP instead of RTP/SAVP). As a result, the fax fails. **Applicable Products:** Gateways.
151885	The "/" character is invalid in some configuration table names (e.g., name for SIP Interface). **Applicable Products:** All.
151897	A memory leak occurs due to an endless recursion of some tasks. As a result, the HA system crashes (resets). **Applicable Products:** HA.
151907	When the device performs an LDAP query for an IP-to-Tel call and gets a result, IP-to-Tel number manipulation doesn’t work if based on source IP Group. **Applicable Products:** Gateways.
151922	Memory utilization value through SNMP displays incorrect results (should be in percentages as displayed in CLI). **Applicable Products:** All.
151951	The Web interface's Port Status page does not display correct SIP information for the call. **Applicable Products:** MP-1288.
152053	If an RTP stream is opened with SRTP for an SBC call and then re-negotiation changes it to RTP, the old crypto suites are not updated and as a result, the call remains in SRTP and no voice occurs. **Applicable Products:** All.

Incident	Description
152098	The Topology View page of the Web interface cannot display more than two digits for an IP Group ID (e.g., IP Group #142 is displayed as IP Group #14). **Applicable Products:** All.
152109	BRI parameters erroneously appear in the Web interface even though BRI is not supported. **Applicable Products:** MP-1288.
152147	When the device re-routes an SBC call upon CNG detection in the early media phase, it sends the new INVITE to the fax server. However, when fax server answers with a 200 OK, the device doesn't forward it to the source leg. As a result, the fax fails. **Applicable Products:** All.
152324	When establishing a CLI connection (Telnet\SSH) with the device without entering the password, device resources are used and never released. As a result, logging in to the device's CLI fails. **Applicable Products:** All.
152328	The MgmtUseLocalUsersDatabase parameter requires a reset for its settings to take effect (instead of on-the-fly). **Applicable Products:** All.
152330	After a manual HA switchover, the registrations are removed and calls fail due to a loss of TPNCP connection and traffic on the HA Maintenance interface. **Applicable Products:** HA.
152354	When using the REST API to load a specific CLI script that includes a command to import a Dial Plan, the device's Web interface freezes and only a device reset resolves the issue. **Applicable Products:** All.
152426	If the device receives an INVITE message containing many large SIP headers, a loss of HDP3 resources occurs and as a result, the SBC call fails. **Applicable Products:** All.
152482	After a software upgrade, the Product Key is no longer displayed in the Web interface. **Applicable Products:** All.
152610	Sometimes ARM tries to connect to the device, but gets an exception "javax.net.ssl.SSLKeyException: Invalid signature on ECDH server key exchange message" and can't connect. **Applicable Products:** All.
152683	The Web interface's Success/Failure Ratio page can only be selected to display up to IP Group ID 32. **Applicable Products:** All.
152716	Number manipulation is performed twice (instead of once) for Tel-to-IP Gateway calls in the following scenario: 1) The device adds NPI and TON and then performs number manipulation; 2) as the routing rule is configured for "Routing Before Manipulation", the device sends the original number to CSR; 3) upon a result from CSR, the device performs number manipulation again, but for the original number (without NPI and TON). As a result, the source number is manipulated to the wrong number. **Applicable Products:** Gateway.
152803	Received analog calls that were transferred by Skype fail due to lack of DSP resources. **Applicable Products:** Gateway.

Incident	Description
152837	The device resets for SBC direct media calls in the following scenario: The device is configured for Direct Media (SBCDirectMedia = 1). A calls B and a direct media call is established. A sends an empty re-INVITE, B sends an offer, and then A rejects the offer by sending an ACK with port 0. **Applicable Products:** All.
152879	Running penetration testing has found a cross-site scripting (CSS) bug (security vulnerability) **Applicable Products:** All.
152992	A large memory leak on the Redundant unit (due to TPNCP restart connection) causes the Redundant unit to reset. **Applicable Products:** HA.
153075	The device experiences a reset loop when the IDS Threshold alarm is changed from higher to lower. As a result, the device crashes (resets). **Applicable Products:** All.
153135	After upgrading to this version, SBC calls experience no audio if additional target ports in the NAT Translation table are not configured. **Applicable Products:** All.
153146	After upgrading to this version, all default (not configured) UDP\TCP ports become 0 in the SIP Interfaces table. As a result, SBC calls fail. **Applicable Products:** All.
153228	On a specific TCP connection call scenario, the device got a SIP BYE, but kept sending retransmissions which caused it to reset. **Applicable Products:** All.
153592	The device crashes (resets) when it receives a REST query from ARM and then sends a response containing more IP Groups than can be sent in the response (currently, up to 200 IP Groups can be sent in the response). **Applicable Products:** All.
151502 / 151184	When the device rejects incoming connections (e.g., due to device overload conditions), it uses a temporary socket. This socket is erroneously attached to the CEMT task, which causes a race condition between SPLB and CEMT threads. In some cases, this leads to a crash (device reset). The bug has been resolved and the temporary socket is now attached to the CEM of SPLB (as done for all other SIP sockets). **Applicable Products:** All (SBC).

2.27 Patch Version 7.20A.202.141

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3082, and EMS/SEM Version 7.2.3106.

2.27.1 New Features

New features introduced in this version include the following:

2.27.1.1 Enhanced Handling of Registered Users and URI Parameters

When a dialog-initiating SIP request (e.g., INVITE) is received, the device performs the Classification, Manipulation, and Routing processes. During Classification, the device searches its Registration database to check if the source of the request is a registered user. The user is registered if both the request's From header matches the registered AOR and the URI in the request's Contact header matches the URI in the Contact of the registered AOR.

By default, the device excludes all URI parameters and ports when comparing two URIs. Therefore, two different registrations of the same user whose Contacts are differentiated only by ports and/or a proprietary parameter are considered by the device to be the same single registration, even though they are different registrations.

To overcome this, a new parameter has been added--URI Comparison Excluded Parameters (Web) / SBCURIComparisonExcludedParams (ini) / config-voip > sbc settings > uri-comparison-excluded-params (CLI). The parameter is used to determine which URI parameters are excluded when comparing two URIs when checking if the incoming request is from a registered user, during Classification.

The value of the parameter is a free-text string, which cannot be empty. It can be configured to any sequence of parameters, separated by commas (e.g., "transport, maddr, ttl"). Alternatively, it can be configured to one of the following values:

- All: (Default) Defines that all URI parameters (except the gruu parameter "gr" and AudioCodes' proprietary parameter "ac-int") and ports are disregarded when comparing the two URIs.
- None: Defines that all URI parameters and ports are included in the comparison of the two URIs.
- Port: Defines that the ports of the URIs are excluded when comparing the two URIs, yet all other URI parameters are included in the comparison. "port" can be combined with other URI parameters to exclude (e.g., "port, transport, proprietary-param").

For example, if two requests are received with different Contact header values as shown below (bolded), if the parameter is configured to All, then the device considers these requests as received from the same registered user as it disregards the port (5060 and 5070), transport, and ttl parameters in its comparison. If configured to None, the device considers these requests as received from two different registered users.

```
Contact: <sip:1000@172.17.142.105:5060;transport=tcp;ttl=10>
Contact: <sip:1000@172.17.142.105:5070;transport=tls;ttl=20>
```

Note: AudioCodes proprietary "feu" string value for the user part must be included in the Contact header of REGISTER requests that the device forwards to the registrar server when the parameter is configured to a non-default value (i.e., not ALL). Therefore, if the parameter is configured to a non-default value, the SBCKeepContactUserInRegister must not be configured to "Keep User Without Unique Identifier" (1).

Applicable Applications: SBC.

Applicable Products: All SBC.

2.27.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-36: Resolved Constraints in Version 7.20A.202.141

Incident	Description
154043	Configured addresses for a Proxy Set are shown associated in the ini file with the incorrect Proxy Set. For example, when addresses are configured (in the Proxy Address table) for Proxy Set ID #2, the ini file shows these addresses as configured for Proxy Set ID #3. **Applicable Products:** All.

2.28 Patch Version 7.20A.202.203

This patch version includes known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3082, and EMS/SEM Version 7.2.3106.

2.28.1 Known Constraints

This section lists known constraints.

Table 2-37: Known Constraints in Version 7.20A.202.203

Incident	Description
154201 (a)	The maximum historical CDRs that can be stored on the device has been reduced -- for SBC CDRs, from 8,192 to 500; for Test Call CDRs, from 4,096 to 500. **Applicable Products:** Mediant VE 1 vCPU 2G.
154201 (b)	The device sometimes erroneously generates the warning message, "Memory Consumption is very high. …", which can be ignored. **Applicable Products:** Mediant VE 1 vCPU 2G.
149873	The Monitor page of the Web interface incorrectly displays the power supply type (AC/DC) used by the devices in HA mode. **Applicable Products:** Mediant 800C.
153930	When a Hitless Software Upgrade is done, WebRTC calls are sometimes terminated. **Applicable Products:** WebRTC.

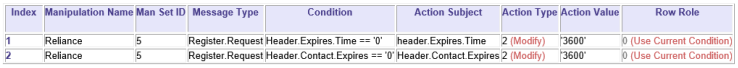
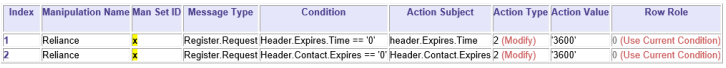
2.28.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-38: Resolved Constraints in Version 7.20A.202.203

Incident	Description
152391	When using the CDR local storage feature, in some scenarios, the device stops collecting CDRs and reports the wrong number of CDR files. **Applicable Products:** All.
152546	The device raises an alarm concerning a memory issue: "Uncorrectable Memory Error ((Processor 1, Memory Module 12))". **Applicable Products:** Mediant 9000.
152624	If the device receives two SBC registrations with the same AOR and the same Contact, but different ports, it considers them as the same user as it ignores the port in the Contact (which is not per RFC). To resolve the issue, a new parameter was added: URI Comparison Excluded Parameters. **Applicable Products:** SBC.

Incident	Description
152875	A WAN IPv6 interface cannot be associated with a Media Realm and the following error message is generated: "User configured a non IPv4 interface for. Interface=1000, Family=10 MATRIX CpMediaRealm: Unable to Activate Line(0) since it is Invalid" **Applicable Products:** All.
153097	The device sends a false alarm for a faulty fan tray (even though it operates ok). **Applicable Products:** MP-1288.
153722	The SNMP displays wrong values for objects ifNumber and ifType (Ethernet ports). **Applicable Products:** Mediant 1000.
153753	When activating both Play RBT and Play-Silence features together, the device crashes (resets). **Applicable Products:** SBC.
153816	The search field in the Web interface displays "[object Object]", which disables the search feature. **Applicable Products:** All.
153917	The SIP Call-ID header value is limited (and truncated if longer) to 129 characters. **Applicable Products:** SBC.
153922	The device (10G interface with fiber) generates a warning message during boot-up about unsupported platform. **Applicable Products:** Mediant 9000.
153937	In a certain scenario, when IP Inbound Message Manipulation is done and afterwards IP Outbound Message Manipulation, the wrong URI is set by the IP Outbound Message Manipulation. **Applicable Products:** SBC.
153968	When an SNMP Performance Monitoring Threshold Crossing alarm is sent by the device, the syslog doesn't display the alarm message. **Applicable Products:** SBC.
153970	If the SIP Contact header contains a "user=phone" parameter, the device ignores the host part (matching based on user part only). As a result, the device does not differentiate between two different users with same user part but different host part. **Applicable Products:** All SBC.
154009	For devices in HA mode, when SNMP changes/sets the MIB of a Performance Monitoring threshold on the Active device, it is not automatically updated on the Redundant device. **Applicable Products:** HA.

Incident	Description
154025	Upon a Hitless Upgrade process, the device sends an unregister message, which removes the specific user from the registration database. For hitless upgrades from software versions that are earlier than 7.20A.202.203, do the following: **For each IP Group that is used as a Serving IP Group and has an associated Outbound Message Manipulation Set:** 1 Add the following Message Manipulation rules to the Set:  2 Save configuration. 3 Perform a Hitless Upgrade. 4 After the upgrade completes, remove the two Message Manipulation rules from the Set. **For each IP Group that is used as a Serving IP Group and doesn't have an associated Outbound Message Manipulation Set:** 1 Through the AdminPage, configure the GWOutboundManipulationSet parameter to "x" (default is "-1"). 2 Add the following Message Manipulation rules:  3 Save configuration. 4 Perform a Hitless Upgrade. 5 After the upgrade completes, delete the Message Manipulation rules, and then configure the GWOutboundManipulationSet parameter to "-1". **Note:** ▪ If the parameter UnregistrationMode is configured to **1**, the un-REGISTER message is sent with "Contact: ***", and un-registration may occur despite Message Manipulation. Therefore, it's recommended to configure the parameter to **0**. ▪ During the first stage of Hitless Upgrade (stage 1\3), the device does not accept new calls for the users in the Accounts table (rejects them with a SIP 500 response). **Applicable Products:** HA.
154030	The device does not handle RTCP stream events correctly and as a result, it resets upon a new SIP INVITE and with the old RTCP event. **Applicable Products:** SBC.
154039	The value of the Level (MgmntLDAPGroups_Level) parameter (in the Management LDAP Groups table) wasn't retained correctly upon an upgrade from 7.2.158 to 7.2.202. **Applicable Products:** SBC.
154062	If the device is configured through the Web interface with incorrect settings in the Physical Ports table and then the IP Network tab is clicked, the device crashes (resets). **Applicable Products:** SBC.
154063	The Reset button in the Web interface does not respond when it is clicked (i.e., no reset occurs) **Applicable Products:** SBC.
154089	When no cable is plugged into PRI port #4, the port's LED is lit "green". **Applicable Products:** Gateway (PRI).

Incident	Description
154090	Significant reordering occurs of the CLI commands in the output of the "show running config" command occurs, when upgrading to Version 7.2.202. **Applicable Products:** Gateway.
154278	A false alarm is sent by the device ("SYS_HA: Active and Redundant modules have different feature keys."). **Applicable Products:** HA.
154337	The device crashes (and resets) due to internal memory overrun in the device's DSP. **Applicable Products:** SBC.

2.29 Patch Version 7.20A.204.015

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3093, and EMS/SEM Version 7.2.3106.

2.29.1 New Features

This section describes the new features introduced in this version.

2.29.1.1 New Mediant 9000 Product Offering

The Mediant 9000 product line now includes the following two new product offerings:

- Mediant 9030 (up to 30,000 SBC sessions)
- Mediant 9080 (up to 70,000 SBC sessions)

Both products are based on the latest HPE ProLiant DL360 Gen10 Server generation and provide the full feature set as the previous Mediant 9000.

Mediant 9030 and Mediant 9080 are supported from Version 7.2.204 and later.

For supported capacity of the Mediant 9000 product line, see Section SIP Signaling and Media Capacity on page 319.

Note: Existing Mediant 9000 customers will continue to benefit from software updates and support services.

Applicable Applications: SBC.

Applicable Products: Mediant 90xx.

2.29.1.2 CSRF Protection of Embedded Web Server

The device's embedded Web server now provides cross-site request forgery (CSRF) protection. CSRF prevents malicious exploits of a website, whereby unauthorized commands are transmitted from a user that the website trusts (i.e., authenticated user). Whenever a user opens (i.e., GET method) one of the device's Web pages, the device automatically generates a CSRF "token" (unique number). When the user performs actions (i.e., POST method) on the page (e.g., configures parameters), the token is included to verify that the authenticated user is the one performing the actions.

The feature is configured by the new global parameter, CSRFProtection (only ini file), which is enabled by default.

Applicable Application: All.

Applicable Products: All.

2.29.1.3 Rate Regulation of TLS Connections

The device now regulates the number of new concurrent TLS connections that can be established per second. This helps protect the device from flooding (avalanches) of TLS connections, which may be caused from TLS-based malicious attacks. The feature is part of the device's protection suite against distributed denial-of-service (DDoS) attacks.

Applicable Application: SBC.

Applicable Products: All.

2.29.1.4 Rate Regulation of User Registrations

The device now regulates the number of new concurrent user registration requests (REGISTER). This helps protect the device from flooding (avalanches) of registrations, causing CPU overload. The feature is part of the device's protection suite against distributed denial-of-service (DDoS) attacks.

Applicable Application: SBC.

Applicable Products: All.

2.29.1.5 Mediant VE SBC Support for Microsoft Azure

Mediant VE SBC can now be deployed in a Microsoft Azure cloud environment.

Applicable Applications: SBC.

Applicable Products: Mediant VE.

2.29.1.6 Cluster Redundancy for Elastic Media Cluster

The device's Elastic Media Cluster feature now supports (by default) cluster redundancy between multiple Media Components. If a failure occurs in a Media Component (e.g., disconnects from the Cluster Manager), its' traffic is transferred to other Media Components in the cluster that have available DSP resources. Traffic (sessions) for which DSP resources are unavailable are terminated. In other words, this feature provides "best-effort" cluster redundancy.

Note: The feature is supported only if both communicating SIP entities support re-INVITE (IpProfile_SBCRemoteReinviteSupport) and/or UPDATE (IpProfile_SBCRemoteUpdateSupport) messages.

Applicable Application: SBC.

Applicable Products: Mediant CE.

2.29.1.7 Session and DSP Utilization Display of Media Components

Information on media sessions and DSP resource utilization for Media Components has been enhanced to also include the number of "legs". This information is displayed in the Media Components table.

Applicable Application: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.29.1.8 SNMP Alarm for Indicating IAM Configuration in AWS

A new SNMP alarm acAWSSecurityRoleAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.139) has been introduced, which is sent (Major severity) when the Amazon Web Services (AWS) instance has not been configured with the required IAM role to access AWS services and resources.

Applicable Application: SBC.

Applicable Products: Mediant VE/CE (AWS Environments).

2.29.1.9 Display of Virtual Networks for Hyper-V Platforms in CLI

The output of the CLI command **show system assembly** now also includes the virtual NICs for devices running on the Microsoft Hyper-V platform. The virtual NICs are displayed under the "Virtual Network" group in the output.

Applicable Application: SBC.

Applicable Products: Mediant VE/CE.

2.29.1.10 Restoring Factory Defaults for Devices in Cloud Environments

For devices (virtual machines / instances) running in cloud environments (Amazon Web Services, Azure, or OpenStack), if the administrator cannot access or log in to the virtual SBC machine (for whatever reason), the SBC can be restored to factory defaults through cloud-init. This is done by placing the `#write-factory` hashtag in the instance user data on the cloud metadata service.

Applicable Application: SBC.

Applicable Products: Mediant VE/CE.

2.29.1.11 Enhanced G.722 Coder Support

The device supports additional packetization times, bit rates and payload types for the G.722 coder:

- For all products except Mediant 9000 and Mediant Software, support for the following packetization times (msec) have been added: 10 (default), 20, 30, and 50
- Rates (kbps) of 48 and 56 have been added
- Payload types of 66 and 67 have been added

For more information, refer to the *User's Manual*.

Applicable Applications: Gateway and SBC.

Applicable Products: All.

2.29.1.12 Increase in Addresses per Proxy Set and DNS-Resolved IP Addresses

The maximum number of addresses (IP addresses or FQDNs) that can be configured per Proxy Set (Proxy Sets table > Proxy Address table) has been increased to 50 for Mediant 9000 and Mediant VE/SE/CE SBCs. In addition, the maximum number of supported DNS-resolved IP addresses for the Proxy Sets has been increased:

- Mediant 9000 and Mediant VE/SE/CE:
  - 32–64 GB RAM: 15,000
  - 2-16 GB RAM: 4,500
- Mediant 2600 and Mediant 4000: 4,500
- MP-1288, Mediant 500, Mediant 500L, Mediant 800, and Mediant 1000: 500

Applicable Applications: SBC.

Applicable Products: All.

2.29.1.13 Keep-Alive SIP OPTIONS for All Proxy Servers

The device sends keep-alive SIP OPTIONS to all proxy servers configured for a Proxy Set, regardless of Proxy Set mode (load balancing, homing, or parking), or priority and weight. Up until now, keep-alive messages were sent only to the proxy currently being used. In addition, for the Homing mode, the device now always attempts to reconnect to an online proxy server with the highest priority (and not to a specific "primary" proxy server as in previous releases).

The benefit of this feature is that as the device does keep-alive with all proxy servers, when it needs to choose a different proxy server (for whatever reason), it already knows which are online and can make its decision quickly.

Applicable Application: All.

Applicable Products: All.

2.29.1.14 IP Group Settings Applied to Proxy Keep-Alive SIP OPTIONS

Various settings of an IP Group can now be applied to keep-alive SIP OPTIONS messages that the device sends to the proxy server (Proxy Set associated with the IP Group). The following IP Group settings can be applied to these keep-alive messages:

- The IP Group's 'SIP Group Name' parameter value is used in the OPTIONS message.
- The IP Group's 'Outbound Message Manipulation Set' for outbound message manipulation is applied to the OPTIONS message (instead of manipulations configured by the GWOutboundManipulationSet parameter). In addition, the syntax 'param.ipg.dst' can be used to access the IP Group's parameters.
- When filtering logs (configured in the Logging Filters table), OPTIONS messages are now filtered by IP Group.

The feature is enabled by the new parameter in the IP Group table – 'Proxy Keep-Alive using IP Group's Settings' (ProxyKeepAliveUsingIPG ini file parameter; proxy-keepalive-use-ipg CLI command).

Note: When multiple IP Groups share the same Proxy Set, the feature can be enabled only on one of them.

Applicable Applications: All.

Applicable Products: All.

2.29.1.15 Priority and Weight Configurable for Proxy Server Addresses

Proxy servers of Proxy Sets can now be configured with priority and weight. Priority and weight apply to all Proxy Set modes (load balancing, homing, and parking). For all these modes, when a proxy server goes offline, the device attempts to connect to an online proxy server with the highest priority.

Weights are used when Proxy Load Balancing Method is configured to Random Weights. Up until now, the weights were determined by the DNS server. Now, the weights can be locally configured per proxy server address (IP and FQDN). In addition, this mode can now operate with multiple proxy server addresses (previously, only a single address was supported, which had to be an FQDN).

The feature is configured by the following new parameters:

- Proxy Priority (ProxyIp_Priority) – configures the priority
- Proxy Random Weight (ProxyIp_Weight) – configures the weight

Note:

- If weight and priority are not configured for any proxy server in the Proxy Set, the order in which the addresses (IP addresses and FQDNs) are listed in the table determine their priority (top-listed address has highest priority). For FQDNs, weight and priority of DNS-resolved IP addresses are determined by the DNS (as in previous releases).
- If at least one of the proxy servers in the Proxy Set is configured with weight and priority, prioritization applies to all configured proxy servers. In this case, proxy servers that are not configured with priority (i.e., 0) are considered as proxy servers with the highest priority.
- Priority and weight received from the DNS server is overridden by the device configuration.
- Both priority and weight (or none of them) must be configured for a proxy server.

Applicable Application: All.

Applicable Products: All.

2.29.1.16 SIP Sockets Opened Only when Needed

The device can be configured to open sockets (ports) for signaling only when needed. The feature applies to the SIP Interface's Additional UDP Port feature with dynamic port allocation. This provides flexibility when configuring the additional UDP port ranges of SIP Interfaces, where the administrator does not have to make sure that the total number of configured ports are within the maximum, as defined by the device's License Key.

The feature is supported by a new parameter in the SIP Interfaces table—'Additional UDP Ports Mode' (AdditionalUDPPortsMode) / additional-udp-ports-mode), with the following optional values:

- Always Open: (Default) The configured ports (sockets) are always open.
- Open When Used: A port is only opened when used. It is used when the device initiates registration with an external SIP entity for a SIP Account (sent to the Account's Serving IP Group) or sends a registration request from a user (IP Group) to a Server-type IP Group. This option is only for dynamic port allocation, where a port is allocated on the outgoing REGISTER (for a SIP Account or IP Group user) and closed when the registration expires. For HA systems, upon a switchover, all the ports used in the active device are also opened on the redundant device (now active), so that the SIP entity will be reachable.

Ports that are not configured by the 'Additional UDP Port' parameter are closed.

The feature is applicable only when 'Additional UDP Ports' is configured for the SIP Interface and enabled for a Server-type IP Group ('User UDP Port Assignment') and/or SIP Account ('UDP Port Assignment').

Applicable Application: SBC.

Applicable Products: All.

2.29.1.17 Close and Reject TLS/TCP Client Connections in Locked State

The device can now be configured to terminate (close) existing TCP/TLS client connections and reject new incoming ones when the device is in locked state. The feature is enabled by the new parameter 'Disconnect Client Connections' (AdminStateRestrictConnections ini file parameter; disconnect-client-connections CLI) on the Maintenance Actions page (Setup menu > Administration tab > Maintenance folder > Maintenance Actions). The parameter is disabled by default (i.e., existing client connections remain and incoming ones are accepted when the device is in locked state).

Applicable Applications: All.

Applicable Products: All.

2.29.1.18 Grouping and Priority of Routing Hosts for Routing Servers

When implementing a third-party routing server (or AudioCodes ARM) whose routing hosts are in different geographical locations, the remote hosts configured on the device for a specific Remote Web Service (table) can be assigned to groups and assigned with different priorities. The hosts closest to the device, for example, can be configured as a high-priority group while the hosts further away, as a low-priority group. The device can then send Get Route requests to the highest priority hosts and only if they fail, send to lower priority hosts.

Up to five groups of hosts can be configured per Remote Web Service. The priority of the group depends on the Group ID (0 highest; 4 lowest). The priority of the host within each group can be configured (0 highest; 9 lowest).

The routing policy between hosts within each group can also be configured (round robin, sticky primary, or sticky next). The routing policy between groups can also be configured (sticky primary or sticky next).

To support the feature, the following new parameters have been added:

- Remote Web Services table (HTTPRemoteService):

- 'Policy Between Groups' (BetweenGroupsPolicy) – configures the policy that determines which group of hosts to use
  - Sticky-Primary (default) - Group 0 is the preferred group, as it is the closest to the device. Upon every user request, the availability of Group 0 is checked, regardless of the current group.
  - Sticky-Last-Available - Uses the current group unless all the hosts in the group fail. In this case, the device searches for an available group from the start (Group 0).

- HTTP Remote Hosts table (HTTPRemoteHost):
  - 'Group ID' – assigns the host to a group. Group 0 is the default group and the highest priority.
  - 'Priority in Group' – assigns the priority of the host. If more than one host has the same priority, the order of the host as listed in the table determines priority.

Applicable Application: All.

Applicable Products: All.

2.29.1.19 New Remote Web Service Type – "General"

A new type of HTTP/S-based (Web) servers can be configured to support the following new services:

- Generating and sending CDRs to a REST server through REST API (see Section 2.29.1.36)
- Querying (GET) HTTP servers using Call Setup Rules (see Section 2.29.1.20)

The feature is configured by the new optional value—General (8)—of the existing 'Type' parameter in the Remote Web Services table.

Applicable Applications: All.

Applicable Products: All.

2.29.1.20 Call Setup Rules for Querying HTTP Servers

Call Setup Rules can now be configured to query (HTTP GET) HTTP-based servers. The response from the HTTP server can be used for various functionality such as tag-based classification and routing.

To support the feature, the new optional value HTTP GET (4) has been added to the 'Query Type' parameter in the Call Setup Rules table. In addition, new syntax options (*http.found* and *http.response*) have been added for configuring the search strings used for querying the HTTP servers, which are configured in the Remote Web Services table (see Section 2.29.1.18).

Applicable Applications: All.

Applicable Products: All.

2.29.1.21 Parameter Changes for Remote Web Services Table

The following parameter changes have been done in the Remote Web Services table:

- 'Persistent Connection' Web parameter renamed 'Automatic Reconnect'
- 'Number of Sockets' (HTTPRemoteServices_NumOfSockets) is now obsolete
- 'Login Needed' (HTTPRemoteServices_LoginNeeded) is now obsolete

Applicable Application: All.

Applicable Products: All.

2.29.1.22 SIPRec Enabled by License Key Only

The SIPRec feature is now enabled only by License Key. Up until now, it was enabled by both License Key and configuration. Consequently, the 'SIP Recording Application' (EnableSIPRec) parameter is now obsolete.

Applicable Applications: All.

Applicable Products: All.

2.29.1.23 Generated SIPRec Metadata in Compliance with RFC 7865

The device's SIPRec feature now complies with RFC 7865 "Session Initiation Protocol (SIP) Recording Metadata" and generates the XML-based body (recording metadata) where all IDs (e.g., participant ID) are in Base64 format. The metadata also includes additional XML tags with association information (e.g., "<participantsessionassoc>"). This compliancy is crucial for Session Recording Servers (SRS) that only accept SIPRec metadata according to RFC 7865. Up until now, the user part of the participant URI (source or destination) was used as the ID.

The feature is enabled by the new parameter, 'SIP Recording Metadata Format' (SIPRecMetadataFormat; siprec-metadata-format). The parameter allows the administrator to choose between RFC 7865 or the legacy metadata format ("Legacy"), which is the default.

Applicable Applications: All.

Applicable Products: All.

2.29.1.24 SBC Call Routing Decision Timeout

The device can be configured with a timeout (in seconds) on call routing decisions that require replies from external servers (e.g., LDAP and ENUM servers). If the timeout expires before the device receives a response from the server, the device sends a routing failure message (SIP 500) to the caller, or uses an alternative routing rule (if configured).

The feature is configured by the new global parameter 'Routing Timeout' (ini file parameter - SbcRoutingTimeout; CLI – configure voip > sbc settings > sbc-routing-timeout).

Applicable Application: All.

Applicable Products: All.

2.29.1.25 Increase in Maximum Number of Dial Plan Rules

The device now supports up to 100,000 Dial Plan rules, which are configured in the Dial Plan Rule table. This is supported on devices with greater than 16-GB RAM.

Applicable Application: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.29.1.26 Increased Characters for Fields Assigned with Dial Plan Tags

The maximum number of characters that can be configured for fields in which Dial Plan tags are assigned have been increased to 70.

Applicable Application: All.

Applicable Products: All.

2.29.1.27 Call Classification to IP Groups by Tags

Incoming calls can now be classified to IP Groups based on the calls' source tags, which are determined by Call Setup Rules associated with SIP Interfaces on which the calls are received. This feature significantly reduces the number of required Classification rules. In some scenarios, a single Classification rule may even suffice.

Classification based on tags includes the following stages:

1. The device determines the tag of the incoming SIP dialog by running a Call Setup Rule that is associated with the SIP Interface on which the dialog is received. The Call Setup Rule can be based on any query type (e.g., LDAP, Dial Plan or HTTP).
2. The device searches the Classification table for a matching rule based on the SIP Interface (and optionally, any other existing matching properties) as well as the tag. The tag can be a name (e.g., "Country") or "default" if the tag only has a value (e.g., "Ireland").
3. The device searches the IP Groups table for an IP Group that is configured with the tag (name or name=value) and if found, classifies the dialog to that IP Group.

To support this feature, the following new parameters have been added:

- Classification table:
  - 'IP Group Selection' - can be set to "Source IP Group" (if an IP Group is specified) or "Tagged IP Group" for using tags as described in this section
  - 'IP Group Tag Name' – defines the tag
- SIP Interfaces table: 'Call Setup Rules ID' – assigns a CSR to determine the tag

Note:

- If multiple IP Groups are configured with the same tag, the device uses the first matching IP Group for classifying the call.
- The IP Group Set table is not used for classification (ignores tags).

Applicable Application: SBC.

Applicable Products: All.

2.29.1.28 Pre-defined Functions for Message Manipulation and CSR

The device provides pre-defined functions that can be used for message manipulation to perform special operations such as changing a returned value from lower to upper case (see example below). The function uses the following syntax:

```
Func.<FunctionName>(<Manipulation Term>)
```

For example, the following Message Manipulation rule adds a header "My-Host" to the outgoing SIP message, whose value is set to the source host, which is converted into upper case, using the function To-Upper:

Action Subject	Action Type	Action Value
Header.My-Host	Add	Func.To-Upper(Param.Call.Src.Host)

If the above rule is used and the host part in the From header of the SIP message is "JohnB":

```
From: <SIP:1000@JohnB>; tag-1c1000228485
```

After manipulation, the following header with the host value in upper case ("JOHNB") is added to the outgoing message:

```
From: <SIP:1000@JohnB>; tag-1c1000228485
My-Host: JOHNB
```

Applicable Application: All.

Applicable Products: All.

2.29.1.29 SIP Message Normalization for Privacy Header

The device supports the normalization of the SIP Privacy header, by removing unknown parameters from the header before forwarding the SIP message. Configuration is done in the Message Manipulations table, by configuring the 'Action Subject' field to "Privacy" and the 'Action Type' field to "Normalize".

Applicable Application: All.

Applicable Products: All.

2.29.1.30 IPv6 Address for SIP Message Manipulations

SIP messages can now be manipulated when the destination IP address is Version 6 (IPv6), using Message Manipulation rules. Up until now, only IPv4 was supported. To support the feature, the param.message.address.[dst|src].[**ip-for-url**|**ip**] has been added. Using "ip" simply returns the IP address as is; using ip-for-url encloses the IPv6 address in square brackets and leaves IPv4 addresses as is. The existing syntax *param.message.address.[dst|src].address* is now obsolete (but still supported for backward compatibility).

Applicable Application: All.

Applicable Products: All.

2.29.1.31 Media Latching on Signaling IP Address for NAT Traversal

For users located behind NAT, the device can be configured to latch on to the first incoming packet that is from the user's signaling IP address. This feature enforces the device to use the signaling IP address (source address of the SIP INVITE message) for sending media to UAs that are located behind NAT. In addition, under these circumstances, the Media Latch Mode is changed automatically from "Strict" to "Dynamic".

The feature is configured by the NAT Traversal (NatMode) parameter's new optional value, NAT by Signaling Restricted IP (4).

Applicable Application: SBC.

Applicable Products: MP-1288; Mediant 500; Mediant 500L; Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.29.1.32 RFC 2833 Generation and Detection without DSPs

For RFC 2833 to SIP INFO interworking, DTMF packet (RFC 2833) generation and detection by the device no longer require the use of DSP resources. This feature affects the parameters IpProfile_SBCRFC2833Behavior, IPProfile_SBCSupportMultipleDTMFMethods, and IpProfile_SBCAlternativeDTMFMethod.

Applicable Application: SBC.

Applicable Products: Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.29.1.33 Advice of Charge Enhancements

Configuration of the Advice of Charge (AOC) supplementary service has been enhanced:

- The currency type (e.g., USD) of the charged call unit can now be configured, using the new 'Currency' parameter in the Charge Codes table.
- The charged call unit can be multiplied by a configurable multiplier number, using the new 'Multiply of Amount' parameter in the Charge Codes table.
- The 'Pulse Interval' parameter has been renamed 'Interval' in the Charge Codes table.
- The 'Pulses Amount on Answer' parameter has been renamed 'Amount on Answer' in the Charge Codes table.

The following new global parameters have been added:

- ISDNAoCMinIntervalGeneration: configures the interval at which the device sends AOC messages (does not affect the charging interval; only the sending of the messages).
- ISDNAoCAmountPerInterval – configures the amount charged per interval.

Applicable Application: Gateway (Tel-to-IP).

Applicable Products: All.

2.29.1.34 Parameters no Longer Requiring Device Reset Parameters

The configuration of the following parameters is applied on the fly and no longer requires a device reset for their settings to take effect:

- EnableIDS
- SIPSRequireClientCertificate
- EnableSilenceDisconnect
- EnableDigitDelivery
- EnableDigitDelivery2IP
- TelProfile_EnableDigitDelivery
- StaticNATIP

Applicable Application: All.

Applicable Products: All.

2.29.1.35 CLI Command Outputs in JSON Format

The output of some `show` CLI commands (e.g., `show system alarms`) can now be displayed in JSON format, using the following new CLI command:

```
output-format json
```

The output is returned to plain text format using the following command:

```
output-format plain
```

Applicable Application: All.

Applicable Products: All.

2.29.1.36 Enhanced Syslog Message Display in Web Interface

The display of device-generated Syslog messages on the Web interface's Message Log page (Troubleshoot menu > Troubleshoot tab > Message Log) has been enhanced. The messages are now displayed in proper alignment and in colors based on message type (Notice, Warning, Error, Critical, Alert, Emergency, Info, and Debug). In addition, buttons (**Start**, **Stop** and **Clear**) have been added, allowing the administrator to stop, start, and clear the message log display.

Applicable Applications: All.

Applicable Products: All.

2.29.1.37 System Snapshots Configuration through Web Interface

System Snapshots can now be configured through the Web interface. Up until now, it could only be configured through CLI and GRUB. This is supported by the new System Snapshots page (Setup menu > Administration tab > Maintenance folder > System Snapshots). The page allows the administrator to create, edit (name), load, and delete System Snapshots, as well as set a specific System Snapshot as default.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.29.1.38 Maximum Characters Increased for System Snapshot Name

When creating or editing a System Snapshot, the name can be up to 64 characters.

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.29.1.39 System Snapshot Name Modifiable through CLI

System Snapshot names can now be modified through CLI, using the following new command:

```
# system-snapshot rename <existing name> <new name>
```

Applicable Applications: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.29.1.40 Download of Redundant Device's Debug File from Active Device

The debug file of the redundant device in a High-Availability (HA) system can now be downloaded through the active device. To support the feature, the **Save Redundant Device's Debug File** button has been added to the Web interface's Debug Files page (Troubleshoot menu > Troubleshoot tab > Debug folder > Debug Files) and the *command copy redundant-debug-file to <URL>* has been added to the CLI.

Applicable Applications: SBC (HA).

Applicable Products: Mediant 500; Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.29.1.41 User Privilege Level per REST API Resource

Each API URL resource (e.g., alarms/active) now has a minimum access level per supported HTTP method (GET, PUT, POST or DELETE). For example, only REST users with Security Administrator privilege level can replace (PUT) the device's License Key. Up until now, only users with Security Administrator levels could use REST API.

REST users and their privilege levels (Monitor, Administrator, or Security Administrator) are configured like for other management interfaces (Web interface's Local Users table). REST users accessing through LDAP or RADIUS must have a minimum access level of 50 (read-only). For prohibited user access, the device responds with a 403 Forbidden Status.

Access to the REST API directory level depends on privilege level:

- Monitor: alarms, performanceMonitoring, status
- Administrator and Security Administrator: actions, alarms, files, license, performanceMonitoring, status

Actions (GET, PUT, POST or DELETE) that can be performed on each API resource depends on the user's privilege level.

For a supported HTTP method, if access is not allowed based on user level, a 403 Forbidden Status or 405 Method Not Allowed code is returned. For requested resource that does not have content, a 400 Bad Request code is returned.

REST API supports Basic Authentication only.

Applicable Application: SBC.

Applicable Products: All.

2.29.1.42 CDRs in JSON Format Sent to REST Server through REST API

The device can now send Call Data Records (CDRs) in JSON format to a REST server through AudioCodes REST API. The REST server is configured in the existing Remote Web Services table as an HTTP-based server, where it is configured with the new 'Type' optional value, General (see Section 2.29.1.20). In addition, the following new parameters have been added to the Call Detail Record Settings page:

- 'REST CDR Report Level' (RestCdrReportLevel ini file parameter; configure system > cdr > rest-cdr-report-level) – defines the stage of the call at which the CDR is generated

- 'REST CDR HTTP Server Name' (RestCdrHttpServer ini file parameter; configure system > cdr > rest-cdr-http-server) – defines the REST server (configured in the Remote Web Services table) where the CDRs are sent

Applicable Applications: All.

Applicable Products: All.

2.29.1.43 Configuration Package File Download through REST API

The Configuration Package file can now be downloaded (GET) from the device through the device's REST API. The file is accessed using the REST API resource URL, api/v1/files/configurationPackage.

Applicable Application: All.

Applicable Products: All.

2.29.1.44 Display of Floating License Reports

The device's SBC resource consumption (signaling sessions, media sessions, transcoding sessions, and user registrations) reports of the Floating License that the device sends to OVOC can now be viewed through the Web interface on the new Floating License Reports page (Setup menu > Administration tab > License folder > Floating License Reports) and through CLI using the new command *show system floating-license reports*.

Applicable Application: All.

Applicable Products: All.

2.29.1.45 Access to Redundant Device from Active Device through SSH

The redundant device in a High-Availability (HA) system can now be accessed from the active device through SSH (or SFTP). SSH can then be used to download files (e.g., debug file, locally stored CDR file, and Configuration Package file) that are stored on the redundant device. This eliminates the need to perform an HA switchover for changing the device from redundant to active for file transfer operations through SSH.

Access to the redundant device through SSH uses a proxy SSH server port on the active device, which is configured by the new parameter, 'Redundant Device Server Port' (SSHRedundantProxyPort / configure system > cli-settings > ssh-redundant-proxy-port). This proxy port must be different to the regular SSH server port (SSHServerPort) of the active device. SSH must be enabled (SSHServerEnable) for this feature.

Applicable Application: All.

Applicable Products: HA.

2.29.1.46 Improved Organization of Files for SFTP

The directory and subdirectories in which files are located for SFTP access has been re-organized to make it easier to locate files. One of the changes is that the Configuration Package file (configuration-package.tar.gz) is now located under the */configuration* directory (instead of the root directory).

Applicable Application: All.

Applicable Products: All.

2.29.1.47 Debug File Downloadable through SFTP

The device's debug file can now also be downloaded from the device through SFTP (SSH must be enabled). The file is located in the /debug folder.

Applicable Application: All.

Applicable Products: All.

2.29.2 Known Constraints

This section lists known constraints.

Table 2-39: Known Constraints in Version 7.20A.204.015

Incident	Description
-	HA is currently not supported by Mediant VE when deployed in a Microsoft Azure environment. **Applicable Products:** Mediant VE.
-	Downloading or uploading the Dial Plan file cannot be done during periods of high traffic. **Applicable Products:** All.
-	The "New Packet loss report" format in Syslog has changed. **Applicable Products:** All.
154109	Downgrading from Ver. 7.20A.204 when 100,000 Dial Plan rules are configured is not supported. (Specifically, if the size of the ini file is greater than 10 MB, downgrade is not supported.) **Applicable Products:** Mediant VE/CE.
154465	To upgrade the device to Ver. 7.20A.202 or later, the device must be running an ISO image that is based on Ver. 6.80.248.006 or later. **Applicable Products:** Mediant Software; Mediant 9000.
154781	Hitless software upgrade from Ver. 7.2.202 to Ver. 7.2.204 is not supported. **Applicable Products:** Mediant CE.
154843	A maximum of three Web sessions (same or different users) can view the Message Log page (Troubleshoot menu > Troubleshoot tab > Message Log) simultaneously. **Applicable Products:** All.
155023	ARM and CDR cannot operate together. **Applicable Products:** All.
155154	The CLI command copy debug file from redundant through TFTP over different subnets fail. **Applicable Products:** HA.
152338	The Intelligent Platform Management (IPMI) chassis indicators (i.e., status of fans, chassis temperature and power supply) are currently unavailable from the management interfaces of the SBC application. However, these indicators can be viewed directly from the Integrated Lights Out (iLO 5) interface (Web, SNMP or REST). **Applicable Products:** Mediant 9000 Rev. B, Mediant 9080, Mediant 9030.
146443	For the WebRTC feature, if the 'SBC Media Security Mode' parameter is configured to Both, and the device receives a SIP INVITE from a browser (Chrome), the device sends a valid outgoing INVITE, but the browser (Chrome) responds with a SIP 488, since it fails to handle both secured and unsecured media. To overcome this, it is recommended to configure the 'SBC Media Security Mode' parameter to SRTP. **Applicable Products:** All.

2.29.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-40: Resolved Constraints in Version 7.20A.204.015

Incident	Description
153631	A syslog message is sent per lost packet report ("Packets-Loss report"), up to the syslog's throttling mechanism. A syslog message is sent for the total lost packets at the last half second per channel (for the first few channels with lost packets). A syslog is also sent for the entire device for lost packets. All these syslog messages are no longer sent. The device sends a maximum of one syslog message every 15 seconds.
154773 / 153858 / 153740	SIP re-INVITE SDP offers sent to the SIPRec Server (SRS) are not properly prepared, causing errors and a device crash (reset). **Applicable Products:** All.
153025	Upon an HA switchover/hitless, a warning message is generated indicating that the TLS certificate expired in 1970, even though it hasn't expired. **Applicable Products:** All HA.
153463	In the device's Web interface, when defining the number of table rows to display per page (default is 10) and then moving to another page and returning to the previous page, the number of records returns to 10. **Applicable Products:** All.
154761	During an HA switchover when the device is running in a VMware environment, the dying active device crashes during the closing of the network ports. **Applicable Products:** Mediant VE/CE.
154706	Upon DTMF transcoding, the device detects each digit twice (on incoming and on outgoing legs) and reports two digits to the SIP INFO side (DTMF duplication). **Applicable Products:** All.
154699	The device rejects calls with the "ac-feu" parameter in the SIP Request-URI. This occurs when the SbcKeepContactUserInRegister parameter is set to 2. **Applicable Products:** All.
154672	The Maintenance interface of the device causes it to drop the last node in the middle of a switchover ("HA: missing last module node in HA DB tree, adding it now"). **Applicable Products:** HA.
154671	For SRTP with SIPRec, the device doesn't send the Avaya UCID in the SIPRec metadata XML body. **Applicable Products:** All.
154667	The device crashes (resets) due to QoS calculation failure over T.38 fax. **Applicable Products:** All.
154665	For SRTP with SRS, if the SRS changes the crypto key upon a SIP re-INVITE, the device fails to change the key. As a result, there is no voice. **Applicable Products:** All.
154661	When the IP address of the CDR server is changed, the device keeps sending CDRs to the old IP address. **Applicable Products:** All.

Incident	Description
154626	The Source and Destination Tags in the CDR field Type changes to "Unknown" after upgrading to Ver. 7.2.202, causing in an incorrect CDR. **Applicable Products:** All.
154573	When the device receives an invalid packet(s) (RTP\RTCP), it crashes (resets). **Applicable Products:** All.
154450	The Name field (AllowedAudioCodersGroups_Name) in the Allowed Coders Groups table doesn't support the comma. If used, the pointer to the AllowedAudioCoders disappears. **Applicable Products:** All.
154341	The CLI command, *configure network* > *network-dev display* shows the tagging mode, but the command, *do show network network-dev* doesn't. **Applicable Products:** All.
154333	When auto-provisioning the device, when it receives an HTTP 302 response from the redirect server with a new location, if the location has "<MAC>", the device doesn't recognize it correctly and sends an HTTP GET request as is, without parsing the "<MAC>". As a result, auto-provisioning fails. **Applicable Products:** All.
154272	NTP loses the NTP sync on the device and as a result, the wrong clock timing occurs. **Applicable Products:** All.
154255	When configuring the device through the Web interface's SBC Configuration Wizard, the resultant configuration is different than configured. **Applicable Products:** All.
154084	When the destination number is greater than 20 digits, the device should send the first 20 digit in the SETUP message and after receiving a SETUP ACK from PSTN side, it sends the rest. However, the device sends all the digits in the SETUP message, which results in a call reject. **Applicable Products:** Gateway.
153818	When downloading CLI-Script file, the User Info file parameters are duplicated **Applicable Products:** All.
153805	The high and low performance monitoring thresholds for SIP are not saved after a device reset. **Applicable Products:** All.
153780	When using force transcoding and changing the legs from G.711-to-G.711 to G.711-to-G.722, the DSP creates echo. **Applicable Products:** All.
153734	The device triggers a hard reset instead of a warm reset in each reset procedure when changing GCT parameter (reset via CLI, WEB etc.). **Applicable Products:** All.
153706	After deleting a Condition rule through the Web interface, the device runs the rule as if it wasn't deleted. **Applicable Products:** All.
153681	Running SIP commands in the CLI of the Media Transcoder causes it to reset. **Applicable Products:** Mediant 9000; Mediant Software.

Incident	Description
153636	Message Manipulation to normalize the SIP message removes the Privacy header. **Applicable Products:** All.
153538	After a manual switchover, the redundant device becomes the active device, but the previously active device becomes unreachable. **Applicable Products:** HA.
153498	The device does not comply with RFC 4566, where the device should add the 'a=rtcp' attribute if the RTP port is an odd number. (Relevant parameter is UDPPortSpacing – 5.) **Applicable Products:** All.
153474	For Tel-to-IP calls (overlap dialing), the device sends to the TDM side a voice stream of another call instead of a ringback tone. As a result, duplicated voice occurs. **Applicable Products:** Gateway.
153456	When the device gets a bulk of calls from ARM with alternative routing, it allocates two resources for each call instead of one. **Applicable Products:** Mediant VE.
153343	The device undergoes an HA switchover due to a Kernel Panic and crashes (resets): "<2>UNSUPPORTED HARDWARE DEVICE: Intel CPU model" and "System crashed due to Kernel Panic". **Applicable Products:** Mediant VE.
153319	The device's CLI displays continuous debug messages: "ACL: cpu 19 in Rx interrupt context number_of_acl_instances". **Applicable Products:** All.
153261	The device raises a critical alarm "License Pool Alarm. The device license has expired! Use of this device is strictly prohibited" even though the License Pool is updated. **Applicable Products:** All.
153225	Message Manipulation rules don't function for IPv6 addresses. **Applicable Products:** All.
153090	During installation of SW-SNC on VMware using OVF, the device alerts about unsupported hardware, and fails as a result. **Applicable Products:** Mediant SE/CE.
152993	Changing the Default Gateway and DNS servers on the active device and then performing an HA switchover causes the redundant device to enter a reset loop. As a result, HA becomes unavailable. **Applicable Products:** HA.
152990	The parameter CSRFProtection is not enabled by default. **Applicable Products:** All.
152881	Changing the default Gateway of the OAMP network interface causes a loss of the static route defined for this IP address. As a result, connectivity with the device is lost. **Applicable Products:** All.
152824	If the device receives RTCP packets from one leg and is configured to block RTCP on the other leg, it still sends RTCP packets on both legs and reports the call (incorrectly) as a good call to the OVOC. **Applicable Products:** All.

Incident	Description
152811	When the device's Ethernet mode on both Ethernet Groups are configured to REDUN_2RX_1TX, the device doesn't send RTP packets and no voice occurs. **Applicable Products:** All.
152653	The device fails the penetration testing for security. **Applicable Products:** All.
152431	When using debug recording (DR) through the Web interface's Logging Filter table, the Stop action isn't applied immediately (only on the next call). **Applicable Products:** All.
152337	The device's locally stored CDR files are generated in the wrong format of size and time frequency. **Applicable Products:** All.
152276	When the device receives a keep-alive UDP packet of length 0, it triggers an IDS alarm indicating a malformed SIP message. **Applicable Products:** All.
152269	The locally stored CDR file on the redundant device in the HA system cannot be downloaded. **Applicable Products:** HA.
151083	Applying Syslog filters on a specific user (number(s)) fails to apply to SIP REGISTER and OPTIONS messages. **Applicable Products:** All.
151022	LDAP FQDN is resolved in to two IP addresses, where only one is valid (active and responds), causes the device to leak in the request messages when search DN is done in parallel. As a result, LDAP fails. **Applicable Products:** All.
149967	When the Media Transcoder's cluster redundant Ethernet port is down, no Ethernet disconnection alarm is forwarded to the device or OVOC (EMS). **Applicable Products:** Mediant 9000; Mediant Software.
147413	The SBC application is enabled by default even though there is no SBC License Key (should be disabled). **Applicable Products:** All.
154883	Calls that fail the classification, manipulation and routing stages don't increment the Performance Monitoring MIBs for CAPs and INVITE dialogs. **Applicable Products:** All.
154881	From Ver. 7.20A.202, the device doesn't get SNMP information on octets (ifOutOctets and ifInOctets). **Applicable Products:** All.
154870	After the device recovers from a reset, it shows that the Media Transcoder is in connecting mode (which is incorrect). **Applicable Products:** Mediant 9000; Mediant Software.
154040	The device fails to reset when a reset is done through the Web interface (Reset button). **Applicable Products:** All.

Incident	Description
153935	When the Web interface sessions expires and a pop-up window appears, the Web interface becomes unavailable and becomes available only if the Web page is refreshed. **Applicable Products:** All.
153619	When the device implements port redundancy and the second port member in the Ethernet Group is removed from the redundancy group, no alarm is sent. **Applicable Products:** All.
153404	An SNMP walk on entPhysicalEntry and ifType doesn't show the device's Ethernet ports. **Applicable Products:** All.
152942	Penetration testing has found that the REST API uses only HTTP Basic Authentication. **Applicable Products:** All.
152940	Penetration testing has found an issue with HTTP Strict Transport Security Not Enforced. **Applicable Products:** All.
152939	Penetration testing has found an issue with the default service banners in the HTTP header in HTTP messages that the device sends, leaks the device's version. **Applicable Products:** All.
152880	The Web interface's users' passwords are stored in the ini file in an unsecured way. **Applicable Products:** All.
152535	The display of the IP Group in the Web interface's Topology View page has changed from centered to left aligned. **Applicable Products:** All.
152003	A mismatch between the Web interface's Logging Filters table and the Command Shell DR folder causes the debug recording to function partially. **Applicable Products:** All.
151354	The device fails to handle SIP re-INVITE for session timer expires from SIP side toward WebRTC side (rejects it with a SIP 488). As a result, the call fails. **Applicable Products:** All.
150894	When the device receives a SIP REFER with an FQDN in the Refer-To header, it tries resolving the FQDN from the DNS, but if it doesn't receive an immediate response, the call fails. **Applicable Products:** All.

2.30 Patch Version 7.20A.204.108

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3091 and EMS/SEM Version 7.2.3106.

2.30.1 New Features

This section describes the new features introduced in this version.

2.30.1.1 IP Group Type Retrieved through REST API

AudioCodes REST API used by third-party routing servers can now retrieve (GET) information on an IP Group regarding whether it is a User-type or Server-type IP Group. The REST API URL resource that is used to retrieve IP Group information is:

```
GET http://<device's IP
address>/api/v1/rmConfig/ipGroups/IpGrp<ID>
```

The example below shows the IP Group type as "Server":

```
Content-type: application/json
HTTP/1.1 200 OK
Content-Type: application/json
X-Frame-Options: SAMEORIGIN
X-XSS-Protection: 1; mode=block
Expires: Thu, 26 Oct 1995 00:00:00 GMT
Content-Length: 413
Server: Web-Server
Set-Cookie: C6=deleted; Expires=Thu, 01 Jan 1970 00:00:01 GMT;
path=/; HttpOnly
{
    "srdId": "SRD-2",
    "type": "Server",
…
}
```

Applicable Application: All.

Applicable Products: All.

2.30.1.2 Tag Value Generation in SIP To Header

The device can generate the 'tag' parameter's value in the SIP To header. This is applied to the first SIP response, received from the called party, which the device sends to the dialog-initiating SIP user agent (caller). In other words, the device-generated To tag overwrites the original To tag generated by the called party. All SIP messages between the device and caller use this generated To tag, while all SIP messages between the device and called party use the To tag generated by the called party.

Up until now, the device forwarded the To tag as is between the SIP UAs (default). As the device-generated To tag value is short (up to 12 characters), this feature may be useful for SIP UAs that cannot handle long tag values.

An example of the To tag:

```
To: Alice@company.com; tag = 9777484849@10.10.1.110
```

To enable generation of the To tag, the following new parameter has been added:

- INI file parameter: ForceGenerateToTag (Disable (default) / Enable)
- CLI: configure voip > sip-definition settings > force-generate-to-tag

Note: The feature is applicable only if the 'SBC Operation Mode' parameter is configured to **B2BUA**. This can be configured in the SRD or IP Group. However:

- The IP Group's 'SBC Operation Mode' parameter takes precedence over the SRD's 'SBC Operation Mode' parameter. For example, if the IP Group is configured for B2BUA but its' associated SRD is not, then the tag-generation feature can function.
- If the IP Group's 'SBC Operation Mode' parameter is not configured (-1), the tag-generation feature for the IP Group is functional only if its' associated SRD is configured for B2BUA.
- For call routing between IP Groups, the feature can only function if both IP Groups are configured for B2BUA, or if one or both are not configured (-1), but the associated SRD is configured for B2BUA.

Applicable Application: SBC.

Applicable Products: All.

2.30.1.3 Interworking SIP 18x and ISDN Q.931 Enhancements

For enhanced interworking SIP 18x without SDP and ISDN Q.931 messages, a new parameter has been added, ISDNIgnore18xWithoutSDP (gateway digital settings > isdn-ignore-18x-without-sdp). When enabled, all incoming SIP 18x messages without SDP are replied by the device by PRACK (if required), but the device doesn't interwork these SIP messages with Q.931 Progress or Alerting messages (i.e., doesn't send to PSTN). When disabled (default), the device interworks these SIP messages with Q.931 Progress and Alerting messages (if required) and sends them to the PSTN.

Applicable Application: Gateway.

Applicable Products: All.

2.30.1.4 ISDN Q.931 Progress Messages Only to Network Side (NT) Trunk

For IP-to-Tel calls, the device can be configured to send Q.931 Progress messages to the ISDN trunk only if the trunk is configured as Network (NT) side ('ISDN Termination Side' parameter configured to **Network side**). To configure this, the new parameter, ISDNSendProgressForTE (gateway digital settings > isdn-send-progress-for-te) has been added. If enabled (default), the device sends Q.931 Progress messages to the ISDN trunk if the trunk is configured as User side (TE) or Network side (NT). If disabled, it sends Progress messages to the trunk only if the trunk is configured as Network side (NT).

Applicable Application: Gateway.

Applicable Products: All.

2.30.1.5 Prefix Length 31 for IP Interfaces

The device supports the configuration of IP interfaces (in the IP Interfaces table) with prefix lengths of 31 (i.e., 255.255.255.254 subnet mask), thereby allowing subnets to contain two IP addresses. To employ this feature, an IP Interface (e.g., 1.1.1.2) needs to be configured with prefix length 31 and its Default Gateway with the remaining (peer) IP address (e.g., 1.1.1.3).

The benefit of this feature is that it saves IP address space usage. Typically, for every subnet, two IP addresses are never assigned to specific network hosts - one all-zeros network and one all-ones broadcast (e.g., for 10.4.0.0/16 subnet the address 10.4.0.0 is the network

address and 10.4.255.255 is the broadcast address). However, for point-to-point connections, these two addresses are not needed. Therefore, using a /31 prefix length assigns a subnet of two addresses - one for each peer without network/broadcast addresses.

Applicable Application: All.

Applicable Products: MP-1288; Mediant 500L; Mediant 500; Mediant 800; Mediant 1000B.

2.30.1.6 SIP REFER Message Handling Based on X-AC-Action Header

AudioCodes proprietary SIP X-AC-Action header has been enhanced and can now be added to incoming SIP REFER requests using Message Manipulation rules to override the device's handling of REFERs configured by the 'Remote Refer Mode' (IpProfile_SBCRemoteReferBehavior).

This is useful if you don't want the settings of this parameter to apply to all calls associated with the IP Profile. For example, if the 'Remote Refer Mode' parameter is configured to **Handle Locally**, all SIP REFER requests associated with the IP Profile are terminated at the device. However, for calls with a specific URI, for example, the device forwards the REFER requests.

Message Manipulation rules can add the X-AC-Action header with one of the following values to the REFER message:

- The device forwards the REFER as is regardless of the 'Remote Refer Mode' parameter settings:
  ```
  X-AC-Action: 'use-config;refer-behavior=regular'
  ```
- The device handles (terminates) the REFER request regardless of the 'Remote Refer Mode' parameter settings:
  ```
  X-AC-Action: 'use-config;refer-behavior= handle-locally'
  ```

Applicable Application: SBC.

Applicable Products: All.

2.30.1.7 Single Username-Password for Authenticating Users

When the device is configured as an Authentication server (User-type IP Group's 'Authentication Mode' parameter is configured to **SBC as Server**), the device can authenticate incoming SIP requests from users belonging to this IP Group, using a single username-password combination that is configured for this IP Group in the IP Groups table ('Username' and 'Password' parameters).

Note: The device always uses the IP Group's credentials for authentication, except when the source of the incoming SIP request is matched to a user that also has an entry in the SBC User Info table and at least one of the 'Username' and 'Password' parameters in the table has a configured value (i.e., credentials in the SBC User Info table take precedence over the credentials in the IP Groups table).

Applicable Application: SBC.

Applicable Products: All.

2.30.2 Known Constraints

This section lists known constraints.

Table 2-41: Known Constraints in Version 7.20A.204.108

Incident	Description
155693	The HTTP PUT request by the Security Administrator user on the device's Configuration Package through REST API fails. **Applicable Products:** All.

2.30.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-42: Resolved Constraints in Version 7.20A.204.108

Incident	Description
152003	A mismatch exists between the Web interface's logging filters and the DR that is run through the command shell page (FAE page). **Applicable Products:** All.
152431	When the DR is stopped thorough the Web interface's Logging Filters page, the DR doesn't stop immediately. **Applicable Products:** All.
153319	After the device resets, "ACL: cpu 19 in Rx interrupt context number_of_acl_instances" messages appear in the serial interface. **Applicable Products:** Mediant 9000.
154093	The device sends many "FIO_DXT_GPIO_RELEASE ioctl failed (0x60386019)" messages to the syslog when submitting any configuration change. This caused a CPU overload (CPU overload alarms). **Applicable Products:** MP-1288.
154168	Connectivity between active and redundant devices is lost after long operations such as burning configuration or loading INI file. **Applicable Products:** All HA.
154678	For a mobile call from SIP Trunk to Skype for Business response group, when the call is answered after call transfer, no voice occurs. **Applicable Products:** All.
154804	The device fails to add a user to the registration database (i.e., registration failure) after an HA switchover. The following message is generated: "[ERROR] RegistrationDB::CreateOrFindEntry - Cannot add entry. AOR and Contact do not match". This occurred as the parameter SBCURIComparisonExcludedParams wasn't saved after the switchover. **Applicable Products:** All HA.
154806	The Floating License for transcoding is not loaded correctly to the device. **Applicable Products:** All.

Incident	Description
155017	The device's Web interface keeps refreshing automatically without any option to connect to the device. This occurs with remote user authentication (RADIUS/LDAP) when the login password contains "\". **Applicable Products:** All.
155050	The device's Web interface cannot be accessed. **Applicable Products:** All.
155067	When installing the SWX module, the device generates many error messages of "No CPLD on this SWX so we cannot RESET the modules". As a result, there is no WAN connectivity. **Applicable Products:** Mediant 1000B.
155069	When operating with the LAD license, the LAD is not added to the SBC media leg pool. As a result, no voice is experienced. **Applicable Products:** All (SBC).
155117	For a WebRTC to SIP call, no voice is experienced on both sides. **Applicable Products:** All (SBC).
155118	If the maximum coder transcoding sessions in the License Key is exceeded, the device crashes (resets). **Applicable Products:** All (SBC).
155150	When the device receives INVITE with Replaces, it ignores the EnableSBCMediaSync parameter. As a result, call transfer fails. **Applicable Products:** All (SBC).
155164	After the device performs a call transfer, no audio (voice) is experienced between the transferee and the transferor. **Applicable Products:** All (SBC).
155182	The device's CLI does not provide the **`insert`** command option (which adds a new row to any specified index) for the IP-to-IP Routing table (**`configure voip > sbc routing ip2ip routing <index> insert`**) **Applicable Products:** All (SBC).
155196	The device crashes (resets) when sending a Debug Recording (DR) to a file **Applicable Products:** All.
155202	The device crashes (resets) when the Ethernet Group is configured incorrectly (includes an upper-row port and a lower-row port). **Applicable Products:** Mediant 2600/4000.
155206	The device sends call reroute Facility message to the PSTN with a different transfer capability as in the incoming call (ServiceCap) in the BC (Bearer Capability) information element (Facility's Q.931 Info Element field). As a result, call re-route fails. **Applicable Products:** Gateway.
155240	When the device tries to add another user to an existing AOR, it generates the error "[ERROR] RegistrationDBEntry(#184478)::SetAORKey - AORKeyList is full". As the AOR is full, additional users cannot be added to the AOR. **Applicable Products:** All (SBC).
155256	When the device is licensed form the License Pool and does not have a local License Key, upon a reset, SIP validation fails (SIP Interface cross-validation). **Applicable Products:** All (SBC).

Incident	Description
155315	The device fails to activate voice after a Microsoft Teams Unattended Call Transfer. This is due to the EnableSbcMediaSync parameter which is disabled, media sync is not done on receiving the SIP 200 OK from the transfer target. **Applicable Products:** All (SBC).
155336	The device fails to run a condition configured in a Message Manipulation rule for vendor-proprietary SIP headers. **Applicable Products:** All (SBC).
155344	The device rejects REGISTER messages from users due to “no more free ID’s” for the gwHashTableItem resource. As a result, registration fails. **Applicable Products:** All (SBC).
155347	For the One Voice Resiliency (OVR) feature in Survivability mode, the device cannot route calls between IP Phones with the same extension. **Applicable Products:** All (OVR).
155365	The device has a limitation (that generates an error) when configuring an IPv6 unique local address (ULA) for the Default Gateway in the IP Interfaces table. **Applicable Products:** All.
155380	When a call is put on hold, the device attempts to allocate coder transcoding resources even though it is not necessary. As a result, the call fails. **Applicable Products:** All.
155402	When the active device in an HA system experiences a CPU overload, no HA switchover occurs (as it should). As a result, the HA system fails. **Applicable Products:** All HA.
155465	For the SIPRec feature, if the device receives a REFER from any leg while still waiting for a SIP 200 OK from the SRS, it cannot handle the REFER and the call fails. **Applicable Products:** All.
155476	If the device receives a SIP re-INVITE that has different authentication and encryption algorithms than those negotiated in the previous INVITE, then the device rejects the call. **Applicable Products:** All (SBC).
155490	DSP errors causes the device to take down the D-Channel **Applicable Products:** Gateway (Digital).
155510	For WebRTC to SIP calls (through the device), no voice (media) is sometimes experienced **Applicable Products:** All (SBC).
155594	The Web login password does not update when it is configured in clear text in an ini file that is uploaded through the Automatic Update mechanism using the IniFileURL parameter. **Applicable Products:** All.
155622	The device's NAT traversal feature fails due to an error in checking for the default SIP interface. **Applicable Products:** All.
155655	The device's IP Group Set policy (configured in the IP Group Set table) does not function and as a result, the call fails. **Applicable Products:** All.

Incident	Description
VI 154437 (SBC-8876)	The device ignores SIP OPTIONS keep-alive messages from users that are not associated with the SIP Connect feature. **Applicable Products:** All.

2.31 Patch Version 7.20A.204.127

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3091 and EMS/SEM Version 7.2.3106.

2.31.1 Known Constraints

This section lists known constraints.

Table 2-43: Known Constraints in Version 7.20A.204.127

Incident	Description
-	For the Media Transcoding Cluster (MTC) feature, the Mediant 4000 cannot be upgraded through the Mediant 4000 management interface itself. It can only be upgraded through the Cluster Manager. For more information, refer to the *Mediant 9000 SBC User's Manual*. **Applicable Products:** Mediant 9000 (and MTC).
-	Sometimes SFTP/SSH is not functioning. A workaround is to disable SSH and then enable it again. **Applicable Products:** All.
SBC-9842	If the 'Mediation Mode' (IpProfile_TranscodingMode) parameter is configured to Force Transcoding, the device doesn't send hook-flash events in RFC 2833. **Applicable Products:** All (SBC).
SBC-6663	When CDRs local storage configuration is changed (through the Logging Filters table), the CDRs are no longer saved to the local storage unless the device is reset. **Applicable Products:** All (SBC).
SBC-10037	The 'Value' field in the Logging Filters table must begin with a capital letter (if the value begins with a letter). **Applicable Products:** All.
SBC-10110	Time and NTP settings should be configured only on the SBC and not on the Media Transcoders (MT) / Media Components (MC). The local time of the MT / MC is synchronized according to the SBC's local time upon initial handshake. Once synchronized, the time used for all MT/MC events (such as alarms) is automatically obtained from the local time of the SBC. **Applicable Products:** Mediant 9000 (with MTs); Mediant CE (with MCs).

2.31.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-44: Resolved Constraints in Version 7.20A.204.127

Incident	Description
SBC-9814	The device doesn't release SSH connections when the SSH application is abnormally terminated (clicking the "x" icon). **Applicable Products:** All.
SBC-9973	SSH stops functioning after an HA switchover (as SSHS tasks was not alive). **Applicable Products:** HA.
VI-151614 (SBC-7437)	Incorrect validation of the 'Cumulative number of packets lost' field in received RTCP packets causes the device to report incorrect Tx\Rx packet loss. **Applicable Products:** All.
VI-153434 (SBC-8289)	The Media Transcoder (MT) resets while closing a channel due to a race condition of using a deleted resource. **Applicable Products:** Mediant 9000; Mediant 4000 (MT).
VI-153455 (SBC-8301)	When the device processes many concurrent calls and the CLI command **`show voip calls history sbc`** is run, the device sends an alarm indicating a CPU overload. **Applicable Products:** All.
VI-154210 (SBC-8724)	The SNMP event indicating a disconnected MT is not cleared on the redundant device after the SBC device undergoes an HA switchover. (MT alarms should only be raised on the active device.) **Applicable Products:** Mediant 9000; Mediant 4000 (MT).
VI-155558 (SBC-9610)	The device doesn't send a hold announcement (music on hold) from one leg to another on hold/unhold transcoding calls. As a result, voice is not heard. **Applicable Products:** All.
VI-155562 (SBC-9612)	The call becomes mute when a hold/un-hold operation is done and it's a transcoding call. **Applicable Products:** All.
VI-155314 (SBC-9466)	When upgrading the device using the Hitless Upgrade method, current calls become muted. **Applicable Products:** HA.
VI-155076 (SBC-9314)	The Media Transcoder (MT) sends the alarm "Alarm: Network element operational state change alarm. Operational state is disable", even though it is operating normally. **Applicable Products:** Mediant 9000; Mediant 4000 (MT).
VI-155248 (SBC-9427)	After incorrect IP Interface configuration which is then deleted, the device produces unusual behavior and incorrect counting of transcoding sessions in the Performance Monitoring (PM) reports. **Applicable Products:** All.
VI-155517 (SBC-9585)	The MT status is not available in the EMS. **Applicable Products:** Mediant 9000; Mediant 4000 (MT).
VI-155522 (SBC-9588)	If the SIP Termination Description field in the CDR for certain calls exceeds the maximum character length (70), CDR processing problems are experienced by third-party tools.

Incident	Description
	Applicable Products: All.
VI-155547 (SBC-9602)	When trying to read the locally stored CDR file through SFTP, the device crashes (resets), due to a wrong client ID. **Applicable Products:** All.
VI-155556 (SBC-9608)	During an MT upgrade, sometimes the MT isn't notified of the upgrade due to race condition and instead, attempts to reconnect to the CM (the MT status toggles between Connect and Disconnecting). **Applicable Products:** Mediant 9000; Mediant 4000 (MT).
VI-155557 (SBC-9609)	The EMS displays the wrong Ethernet port number of the active device in the HA system (redundant is correct). **Applicable Products:** HA.
155600 (SBC-9631)	When the **`show ini-file`** CLI command is run on the MT, the CLI freezes and fails. **Applicable Products:** Mediant 9000; Mediant 4000 (MT).

2.32 Patch Version 7.20A.204.128

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.4.3093 and EMS/SEM Version 7.2.3106.

2.32.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-45: Resolved Constraints in Version 7.20A.204.128

Incident	Description
VI-154196 (SBC-8712)	The redundant power supply unit is not displayed in the Monitor page of the Web interface. **Applicable Products:** Mediant 9000.
VI-154232 (SBC-8741)	When upgrading the device to 7.20A.202.203, the Cluster freezes in the boot screen. As a result, the upgrade fails. **Applicable Products:** Mediant 9000; Mediant 4000.
VI-155017 (SBC-9280)	The device's Web interface keeps refreshing automatically without any option to connect to the device. This occurs with remote user authentication (RADIUS/LDAP) when the login password contains "\". **Applicable Products:** All.
VI-155391 (SBC-9506)	During periods of high traffic, the device loses connectivity with OVOC **Applicable Products:** All.
VI-155459 (SBC-9542)	Even though configuration is correct and the device has no alarm, the MT displays the alarm, "MTC operational state is disabled". **Applicable Products:** Mediant 4000 (MT).
VI-155570 (SBC-9616)	The device has sporadic one-way voice on Tel-to-IP SRTP calls. **Applicable Products:** Gateway.
VI-155622 (SBC-9645)	When upgrading the device to 7.2.202, the Global Static NAT IP address is not propagated to the new release and therefore, NAT does not occur. **Applicable Products:** All.
VI-155654 (SBC-9662)	Call transfer that includes Replaces fails (database fails due to wrong relationship between it and the user in the Replaces header). **Applicable Products:** All.
VI-155494 (SBC-9569)	The device's E1 ports became out of order and return to service only if the trunk is restarted (stop and start). This is caused by a DSP failure. **Applicable Products:** Gateway (Digital).
VI-155329 (SBC-9478)	During a Hitless Upgrade, one of the SIPRec calls returns the wrong value, which causes the device to reset. **Applicable Products:** HA.

2.33 Patch Version 7.20A.204.132

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes EMS Version 7.2.3106.

2.33.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-46: Resolved Constraints in Version 7.20A.204.132

Incident	Description
SBC-10107	HA switchover occurs due to TCP connection failure between the two devices in HA. **Applicable Products:** HA.
SBC-7542	The following Syslog event is displayed after a software upgrade (CTC Analytics AG-RTP Mediation). **Applicable Products:** HA.
SBC-8204	The active device crashes (resets) in the following scenario: Redundant Proxy Port is configured, a switchover occurs, the redundant resets, and then a new SSH/SFTP connection to the active device is done. **Applicable Products:** HA.

2.34 Patch Version 7.20A.204.222

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.143 and EMS/SEM Version 7.2.3113 and 7.2.3106.

2.34.1 New Features

This section describes the new features introduced in this version.

2.34.1.1 CLI Access to Monitor Users

Device management users with "Monitor" user level can now access the device's CLI (basic mode **only** – not privileged mode).

Applicable Application: All.

Applicable Products: All.

2.34.1.2 Chassis Temperature Indication

The temperature of the device's CPU as well as DSPs (on the Media Processing Module / MPM) can now be viewed through the CLI, using the following new command:

```
show system temperature
```

Applicable Application: SBC.

Applicable Products: Median 4000B.

2.34.2 Known Constraints

This section lists known constraints.

Table 2-47: Known Constraints in Version 7.20A.204.222

Incident	Description
SBC-10430	Software upgrade fails due to insufficient memory caused by System Snapshots. A workaround is to delete one or more System Snapshots to free up memory. **Applicable Products:** All.
SBC-10676	After upgrading the device from any previous version to 7.2.204, CPU overload alarms may be raised. This is due to the 7.2.204 feature that sends SIP OPTIONS keep-alive messages to all proxy IP addresses. Depending on the number of proxies (IP addresses and DNS-resolved IP addresses), a CPU overload can occur when the device tries sending SIP OPTIONS keep-alive to all the IP addresses at once. **Applicable Products:** All.
SBC-10577	Modifying the 'Underlying Interface' field (Ethernet Group) in the Ethernet Devices table is not applied on-the-fly. **Applicable Products:** All.
SBC-10717	The CLI commands for handling System Snapshots (default, create, delete, load, rename, show, and help) were moved to a new CLI folder (system-snapshot) under the root folder. However, even though the old snapshot folder is "hidden", it wasn't deleted and therefore, running any of the old snapshots commands (and script) is faulty. **Applicable Products:** Mediant 9000; Mediant Software.

2.34.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-48: Resolved Constraints in Version 7.20A.204.222

Incident	Description
SBC-10655	PCAP file is not present in the SBC directory after connecting to the device from an SFTP client. **Applicable Products:** All.
SBC-7542 (VI-151881)	For HA systems, when a call is put on hold, the following messages erroneously appear in the Syslog and should be ignored: "SYS_HA Redundant message [F8022]: [S=59] _IsRTPMediationFunctionalityPossible - RTPMediationFaxTranscoding should be configured the same value for both channels HA Redundant message [F8022]: [S=60] _HandleRTPMediationFunctionality - RTP Mediation functionality configuration is wrong, some features won't work." **Applicable Products:** HA.
SBC-9690 (VI-155694)	The lack of blkid binary that WALinuxAgent relies on causes issues during the device's installation in an Azure environment. **Applicable Products:** Mediant VE (Azure).

Incident	Description
SBC-9762	The DNS query cache doesn't contain the query results, producing the following Syslog message: "dns query cache is full and delete can't be done". As a result, DNS functionality fails. **Applicable Products:** All.
SBC-9881	For a Microsoft Teams media bypass SBC call, the device sends the wrong crypto key in the response to the re-INVITE message. As a result, the call fails. **Applicable Products:** All.
SBC-9894	The device changes the order of the P-Preferred-Id SIP header between the incoming and outgoing SBC message. Resolved by the new parameter, PPreferredIdListMode (must be set to 1). **Applicable Products:** All.
SBC-9968	When the device opens a channel with "send_rtp_silence", it fails to send T.38 packets. As a result, the fax fails. **Applicable Products:** All.
SBC-10040	The device fails to close files properly, creating a memory overrun which causes a device reset. **Applicable Products:** All.
SBC-10050	The device loses the HTTP-based connection to OVOC for the License Pool, upon a software upgrade to Version 7.2.204. **Applicable Products:** All.
SBC-10062 / SBC-10386 / SBC-10449	After a software upgrade to Version 7.2.204, the device in an Azure environment has missing RTP voice (no voice) in Microsoft Teams. **Applicable Products:** Mediant VE (Azure).
SBC-10076	Password expiration in HA systems results in wrong synchronization between the active and redundant devices. **Applicable Products:** HA.
SBC-10105	When the active and redundant devices in an HA system have different License Keys, the HA system resets. **Applicable Products:** HA.
SBC-10115	On specific SBC calls (where both channels are configured with SendRTPSilence and RTPMediationTransRedundancy enabled), one-way voice occurs. **Applicable Products:** All.
SBC-10278	Due to a DSP restart, no voice occurs in the call for about 5 sec. **Applicable Products:** All.
SBC-10417	When connection to the LDAP server is lost during an LDAP search, the device resets. **Applicable Products:** All.
SBC-10420	When the device's Classification-Manipulation-Routing process fails due to connectivity with the destination IP Group, the CDR of the call doesn't reflect the reason for the failure ("GWAPP_UKNOWN_ERROR" instead of routing error). **Applicable Products:** All.

Incident	Description
SBC-10423	The device adds a crypto line to the SDP of a SIP 200 OK for UPDATE messages, even though the call is RTP only. **Applicable Products:** All.
SBC-10461	When a call fails due to a parsing error, the CDR of the call doesn't reflect the reason for the failure. **Applicable Products:** All.
SBC-10541	Race condition occurs during device boot-up causes IPv6 network connectivity "hang" issues. **Applicable Products:** All.
SBC-10602	For HA systems, a manual switchover during traffic causes the device to reset. **Applicable Products:** HA.
SBC-10744	When the device has DSPs, but has no license for transcoding, it sends the SDP without coders. As a result, the call fails. **Applicable Products:** All.
SBC-8181 (VI-153213) \ SBC-8584 (VI-153966) \ SBC-8896 (VI-154471) \ SBC-8902 (VI-154478) \ SBC-8978 (VI-154600) \ SBC-9396 (VI-155201) \ SBC-9514 (VI-155405) \ SBC-9697 (VI-155709) \ SBC-9739 (VI-155772) \ SBC-10344 \ SBC-10382 \ SBC-10551 \ SBC-10555	A software exception might occur as result of software synchronization issues that very rarely occur. **Applicable Products:** Mediant 4000; Media Transcoder (MT).
SBC-7740 (VI-152282) \ SBC-8111 (VI-153054) \ SBC-8500 (VI-153828) \ SBC-8503 (VI-153832) \ SBC-8862 (VI-154415) \ SBC-8968 (VI-154587)	A memory overrun of the kernel stack pointer causes the device to reset. **Applicable Products:** Mediant 4000.
SBC-9969	After upgrading to Version 7.2.204, the following Syslog message appears: "UpdateDispatcher RemoveChannel Failed!" **Applicable Products:** All.

2.35 Patch Version 7.20A.204.233

This patch version includes known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes EMS/SEM Version 7.2.3106.

2.35.1 Known Constraints

This section lists known constraints.

Table 2-49: Known Constraints in Version 7.20A.204.233

Incident	Description
SBC-11263	Modifying the IPv6 address in the IP Interfaces table when the Ethernet link is down, results in a DAD (duplicate address) alarm and the interface becomes unreachable. **Applicable Products:** All.

2.35.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-50: Resolved Constraints in Version 7.20A.204.233

Incident	Description
SBC-10932	If the device is configured with IPv6 addresses on a 10-GB NIC, after a device reset, there is no network connectivity to the IPv6 addresses from the local subnet. **Applicable Products:** Mediant 9000.
SBC-10820	The IDS feature cannot be enabled due to insufficient memory. **Applicable Products:** All (SBC).
SBC-10819	The device has one-way voice towards the IMS side. **Applicable Products:** Mediant 9000; Mediant Software.
SBC-10803	Activating the CLI **stop** command causes the device to reset (SSSH exception). **Applicable Products:** All (SBC).
SBC-11063	When performing a manual High-Availability (HA) switchover during high traffic, the device resets. **Applicable Products:** Mediant 500; Mediant 800B; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.36 Patch Version 7.20A.204.237

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.36.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-51: Resolved Constraints in Version 7.20A.204.237

Incident	Description
SBC-11107	Video call quality degrades when the device's call volume increases. **Applicable Products:** All (SBC).
SBC-10679	Packet loss occurs when the device pings its own IP address **Applicable Products:** Mediant 9000.
SBC-11454	If the MT is reset when the Cluster Manager has a different software version (.cmp) for the MT, upon bootup the MT tries to upload the new .cmp file and loses connection with the Cluster Manager. **Applicable Products:** MT.
SBC-11724	The device crashes (resets) when trying to create a new network interface. **Applicable Products:** All (SBC).

2.37 Patch Version 7.20A.204.241

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.37.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-52: Resolved Constraints in Version 7.20A.204.241

Incident	Description
SBC-12269	When changing passwords in the device, the new passwords appear in the Syslog message (which should be hidden for security). **Applicable Products:** All.
SBC-12299	When an MT is deleted in the device (Cluster Manager) in a specific scenario, the device crashes (resets). **Applicable Products:** Mediant 9000; Mediant VE.

2.38 Patch Version 7.20A.204.337

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.38.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-53: Resolved Constraints in Version 7.20A.204.337

Incident	Description
SBC-11131	When the device sends an INVITE message and receives a SIP 3xx response containing multiple Contact headers, if it then receives a failed SIP response for the new INVITE, the device automatically tries the next Contact in the 3xx, and so on. Now, the device tries the next Contact only if the failed SIP response is defined in the Alternative Routing Reasons table. **Applicable Products:** All (SBC). Boaz
SBC-9017 (VI-154655)	When the parameter EnableSIPS is enabled, direct SRV resolution results in the wrong transport type (the device sends calls as TCP instead of TLS in the Via SIP header). **Applicable Products:** All (SBC).
SBC-10185	When the device is configured to operate with two LDAP servers and the first server returns a "NOT FOUND" for an LDAP query, its timer is not cancelled when the request is sent to the second server. This results in a resource leak that eventually causes LDAP to stop functioning. **Applicable Products:** All (SBC).
10279	Packet loss occurs when the device pings its own IP address **Applicable Products:** Mediant 9000.
SBC-10383	When the device is deployed in a Microsoft Teams environment on the Microsoft Azure platform, no RTP packets (no voice) are sent from the device. **Applicable Products:** Mediant 9000.
SBC-10559 / SBC-10905	After running traffic during a Hitless software upgrade, the device produces wrong SNMP performance monitoring values for the IPGroupInviteDialogs MIB counter. **Applicable Products:** All (SBC).
SBC-10689	When the device terminates a SIP REFER message and then sends two forking INVITEs, the call fails (it doesn't send ACK for the 200 OK). **Applicable Products:** All (SBC).
SBC-10760	For SRTP-to-RTP calls, the device fails to allocate DSP resources for DTMF transcoding (RFC 2833 to In-Band). **Applicable Products:** All (SBC).

Incident	Description
SBC-10784	Downloading a CLI Script file through the REST API results in an incomplete CLI script. **Applicable Products:** All (SBC).
SBC-10785	Upon a call transfer scenario in a Microsoft Teams environment, the device tries allocating DSP resources for coder transcoding during early media even though it's not needed. As a result, the call fails. **Applicable Products:** All (SBC).
SBC-10813	The CLI command **show network access-list** always displays "Allow" even when it's blocked. **Applicable Products:** All (SBC).
SBC-11065	If the device receives ISO 8859-1 characters, it forwards the character to ARM as is, instead of changing the format to UTF-8 (ARM does not support ISO 8859-1) **Applicable Products:** All (SBC).
SBC-11131	When the device receives a SIP 3xx response with multiple Contact headers, upon a reject reason from the first header, the device should check if the reason exists in the Alternative Routing Reasons table, instead of automatically trying the next Contact header. **Applicable Products:** All (SBC).
SBC-11136	When upgrading the License Key for an HA system using the hitless method, after the upgrade is done, the active unit still shows an alarm about a License Key mismatch. **Applicable Products:** HA.
SBC-11148	When the device receives a SIP REFER with the Referred-By header that exceeds the 2,998 characters, it fails to create an INVITE to terminate the REFER. As a result, call transfer fails. **Applicable Products:** All (SBC).

2.39 Patch Version 7.20A.204.362

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.39.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-54: Resolved Constraints in Version 7.20A.204.362

Incident	Description
SBC-11288	During a Hitless Upgrade to Ver. 7.20A.204.222, the network driver crashes when handling large packets (jumbo packets or LRO). As a result, the device resets. **Applicable Products:** Mediant VE.
SBC-11549	The device doesn't forward the SIP History-Info header as is (when it receives one History-Info header it sends two History-Info headers on the outgoing leg). **Applicable Products:** All (SBC).

2.40 Patch Version 7.20A.204.433

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.40.1 New Features

This section describes the new features introduced in this version.

2.40.1.1 Obscure Password Configuration Enforcement for Management Users

The device can now be configured to enforce obscured (i.e., encrypted) passwords whenever a new device management user is created or a password of an existing user is changed, through CLI.

This feature is enabled by the new ini file parameter, CliObscuredPassword (and CLI, configure system > mgmt-auth > obscure-password-mode). To obtain an encrypted password, the user first needs to enable the parameter, configure the password through the Web interface (Local Users table), save the CLI Script file to the local PC, and then copy the encrypted password from the file.

Applicable Application: All.

Applicable Products: All.

2.40.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-55: Resolved Constraints in Version 7.20A.204.433

Incident	Description
SBC-10546	The device fails to transcode crypto suites for SRTP calls (one side supports crypto suite 80 and the other side supports crypto suite 32). As a result, the call fails. **Applicable Products:** All.
SBC-10579	The device attempts to open a voice channel using an unsupported coder (i.e., not included in the License Key) and generates an error "Lack of Resources: problem with allocation of DSP channel". As a result, the call fails. **Applicable Products:** All.
SBC-11195	The device crashes (resets) due to an error related to the Web interface ("Task WEBS") when receiving a "problematic" packet. **Applicable Products:** All.
SBC-11277	When the HA system is upgraded using the Hitless option, all calls fail. **Applicable Products:** HA.
SBC-11453	The CLI command **show last-cli-script-log** doesn't display errors after applying invalid configuration from the CLI Script file. **Applicable Products:** All.

Incident	Description
SBC-11694	The HA system's ping to the IP addresses of the Network Monitor fails (unreachable) and as a result, performs a switchover (and a rest). **Applicable Products:** HA.
SBC-11871	The Remote Web Services table's (HTTPRemoteServices) 'Login Needed' parameter became hardcoded as enabled, preventing customers that don't work with ARM to connect to it. **Applicable Products:** All.
SBC-11873	In a specific Direct Media scenario, the device breaks the suggested coder list, resulting in no voice. **Applicable Products:** All.
SBC-11889	The stream sent by the device to the SIPRec Server (SRS) is distorted when using Opus and SRTP. **Applicable Products:** All.
SBC-11897	In a specific scenario, the device fails to play a RBT using the PRT file. **Applicable Products:** All.
SBC-11982	The device crashes (reset) when receiving a SIP CANCEL before receiving an LDAP reply. **Applicable Products:** Gateway.
SBC-12036	The device shows incorrect packet loss values in the OVOC report. **Applicable Products:** All.
SBC-12041	The device's Web interface displays both the **Save** and **Reset** buttons (on the toolbar) as highlighted (red border) when the device receives an incremental ini file from OVOC. **Applicable Products:** All.
SBC-12057	The TrunkStatusReportingMode parameter doesn't function as expected. **Applicable Products:** Gateway.
SBC-12130	The device has one-way audio with Direct Media and SIP 183 with SDP. **Applicable Products:** All.
SBC-12186	The device changes its' IP address settings in ARM. **Applicable Products:** All.
SBC-12303	When the device receives a SIP 607, it erroneously translates it into a SIP 600 on the outgoing leg. **Applicable Products:** All.
SBC-12145	When the device receives a SIP 487 response, it makes the TransactionUser "forget" the remote tag (of the To header). Later, when the re-INVITE is received, it matches the TransactionUser without the remote tag. As a result, the call fails. **Applicable Products:** Gateway.
SBC-12352	The device changes the RTP stream direction and codec after it receives a SIP re-INVITE. **Applicable Products:** All.

2.41 Patch Version 7.20A.204.442

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164, and EMS/SEM Version 7.2.3113 and later.

2.41.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-56: Resolved Constraints in Version 7.20A.204.442

Incident	Description
SBC-11263	Modifying the IPv6 address in the IP Interfaces table when the Ethernet link is down, results in a DAD (duplicate address) alarm and the interface becomes unreachable. **Applicable Products:** All.
SBC-11717	The CLI shows duplicated entries for the Call Admission Control Profile tableC - one under "configure voip" and the other under “sbc”. **Applicable Products:** All.
SBC-11921	When unregistering a user in the Accounts table by choosing the Un-Register command in the Action drop-down list, the device unregisters the user, but then re-registers the user. **Applicable Products:** All.
SBC-12141	An HA switchover causes incorrect SNMP MIB performance monitoring values. **Applicable Products:** HA.
SBC-12199	The device fails to make G.711-G.726 transcoding SBC calls, since G.726 does not support dynamic payload type. **Applicable Products:** All.
SBC-12302	The device does not validate CLI scripts received through REST API, allowing invalid parameters and configuration. **Applicable Products:** All.
SBC-12457	When sending CDRs to a REST API server, the JSON formatted body of the CDR record is invalid (double quotes in SIPTermDesc field value). **Applicable Products:** All.
SBC-12509	The device resets when receiving 18x containing SDP with the same SDP version, when the 'Remote Early Media RTP Detection Mode' parameter is configured to **By Media**. **Applicable Products:** All.

2.42 Patch Version 7.20A.204.510

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.42.1 New Features

This section describes the new features introduced in this version.

2.42.1.1 User-Defined Performance Monitoring SNMP MIBs

Up to 26 user-defined Performance Monitoring (PM) MIB groups can be configured to count specified SIP failure responses (e.g., SIP 408) or responses generated internally by the device (e.g., CAC limit reached or NER threshold crossed). The PMs can be configured to count the responses due to SIP INVITE or REGISTER messages.

Each user-defined PM group includes the following PM MIBs:

- acPMSBCInUserDefinedFailures<1-26>Table: Counts the total incoming responses
- acPMSBCOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses
- acPMSBCSRDInUserDefinedFailures<1-26>Table: Counts the total incoming responses per SRD
- acPMSBCSRDOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses per SRD
- acPMSBCIPGroupInUserDefinedFailures<1-26>Table: Counts the total incoming responses per IP Group
- acPMSBCIPGroupOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses per IP Group

The feature is configured in the new table, User Defined Failure PM (Monitor menu > Monitor tab > Performance Monitoring folder > User Defined Failure PM).

Applicable Application: SBC.

Applicable Products: All.

2.42.1.2 IP Subnet Conditions for Message Manipulation Rules

Message Manipulation rules can now be configured with conditions that check if an IP address (IPv4 or IPv6) in a SIP header (e.g., From and To) belongs to a specific subnet. The feature is configured using the new operand, "insubnet" (or "!insubnet" for not in subnet) in the 'Condition' field. The subnet is expressed in CIDR (Classless Inter-Domain Routing) notation. A few examples are shown below:

```
Header.From.URL.Host insubnet '10.8.0.0/8'
Header.To.URL.Host !insubnet ' 172.0.0.0/10'
Header.From.URL.Host insubnet 'ffff:a08:705:0:0::/32'
```

Applicable Application: All.

Applicable Products: All.

2.42.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-57: Resolved Constraints in Version 7.20A.204.510

Incident	Description
SBC-13440	When loading an incremental CLI Script file to the device through REST API, the Dial Plan rule definition in the file is not applied. **Applicable Products:** All.
SBC-13107 / SBC-13112	When upgrading a Media Transcoder (through the SBC – Cluster Manager) during high traffic, the SBC resets. **Applicable Products:** Mediant 9000; Mediant VE.

2.43 Patch Version 7.20A.204.521

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.43.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-58: Resolved Constraints in Version 7.20A.204.521

Incident	Description
SBC-13662	Deleting the default TLS Context (Index #0) through CLI, causes a device reset. **Applicable Products:** All.
SBC-13691	The ini file parameter ShortCallSeconds doesn't have a corresponding Web GUI field and CLI command. **Applicable Products:** All.
SBC-14043	IPv6-based Static Route rules are not active after a device reset. **Applicable Products:** All.
SBC-14048	Running the CLI command **show voip calls** on MT causes it to reset. **Applicable Products:** Mediant 9000 (MT); Mediant VE (MT).
SBC-14161 / SBC-14423 / SBC-14706 / SBC-14792	A CPU overload occurs upon an HA switchover. **Applicable Products:** HA.
SBC-14411	Upon an HA switchover, the STWR task gets stuck in read Linux file, causing a CPU overload. **Applicable Products:** HA.

2.44 Patch Version 7.20A.204.523

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.44.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-59: Resolved Constraints in Version 7.20A.204.523

Incident	Description
SBC-14973	The device rejects an SBC call related to an Account upon the receipt of a SIP INVITE message with a port that is different from the port used for the REGISTER message. **Applicable Products:** All.
SBC-15052	The device shows different NER values between the Performance Monitoring MIB (acPMSBCNerAverage) and the CDR. This is due to incorrect average calculation for the Performance Monitoring MIB. **Applicable Products:** All.
SBC-15180	After a switchover, the MT disconnects due to a timeout of more than 30s. **Applicable Products:** HA.

2.45 Patch Version 7.20A.204.735

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1116 and EMS/SEM Version 7.2.3113.

2.45.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-60: Resolved Constraints in Version 7.20A.204.735

Incident	Description
SBC-10688	The device sends a SIP 200 OK with multi-coder support, even though it supports only one coder. As a result, one-way voice occurs. **Applicable Products:** SBC.
SBC-11556	The word "AudioCodes" still appears for Nuera private labeling (welcome message). **Applicable Products:** Mediant Software.
SBC-11674	The Master user level can log in to the device, but cannot change its password. **Applicable Products:** All.
SBC-11677	When trying to save a debug file through the Web interface, by clicking the Save Debug File button on the Debug Files page (Troubleshoot > Troubleshoot > Debug folder > Debug Files), and a debug file doesn't exist, the pop-up dialog doesn't have a button to abort and close the dialog. **Applicable Products:** Mediant Software.
SBC-12003	When the device sends a voice quality report (to OVOC), some calls are erroneously reported as “Call Quality: Unknown” **Applicable Products:** All.
SBC-12126	LDAP authentication fails when the Management LDAP Group is configured in Index 0 (Management LDAP Groups table). **Applicable Products:** All.
SBC-12214	The Local Jitter value is greater than the Remote Jitter value in the QOE MDR packets that are sent to OVOC at the end of each call. **Applicable Products:** All.
SBC-12248	Connection to the device through SSH fails ("SSH: Too many active sessions"). **Applicable Products:** All.
SBC-12296	The device fails to handle incoming Test Calls. **Applicable Products:** All.
SBC-12393	The device resets due to incorrect configuration in the Static Route table. **Applicable Products:** All.
SBC-12530	The device's RTP events are not relayed after a SIP REFER, resulting in call failure. **Applicable Products:** All.

Incident	Description
SBC-12570	Loading a downloaded ini file to the device fails when the OSNInternalVLAN parameter is configured to 1. **Applicable Products:** All.
SBC-12575	The device fails to obtain NTP information from the NTP server when DNS fails upon device startup. **Applicable Products:** All.
SBC-12657	The Automatic Update mechanism triggers the Web interface's **Save** button to indicate a save is needed (displayed with a red border) even when no changes are done. **Applicable Products:** All.
SBC-12669	The device sends incorrect MoS values for cals using G.722 with ptime of 10ms. **Applicable Products:** All.
SBC-12842 / SBC-13246	Loading an incremental ini file through OVOC requires a reset (should be on the fly). **Applicable Products:** All.
SBC-12871	The device experiences a race condition between SIP UPDATE and 200 OK for INVITES (receiving the 200 OK for the INVITE before receiving the 200 OK for the UPDATE). As a result, the call fails. **Applicable Products:** All.
SBC-12938	The device resets upon the receipt of Fax over IP due to a bug in the process of QoE over T.38. **Applicable Products:** All.
SBC-13042	The device marks the RTP Event packet as "true" (instead of "false", as it was received from the other leg). Therefore, the peer side considers the RTP Event stream as new and presents duplicated digits (i.e., false DTMG report). **Applicable Products:** All.
SBC-13048 / SBC-13603	An SBC call for Microsoft Teams Direct Routing with media bypass results in no media (no voice). **Applicable Products:** All.
SBC-13049	CRP configuration from scratch doesn’t allow modifications of the predefined IP Groups (adding Proxy Sets or IP Profiles, etc.). **Applicable Products:** CRP.
SBC-13060	The device resets with the exception "Task LIBT". **Applicable Products:** All.
SBC-13087	The device's message manipulation feature stops functioning after it is upgraded from Version 7.0. **Applicable Products:** All.
SBC-13186	Call transfer fails due to a race condition: A calls B, puts B on hold, calls C, sends a REFER, then A sends a re-INVITE before the device sends a BYE to A. **Applicable Products:** All.
SBC-13189	The device resets upon a specific scenario involving an SRTP-RTP SIPRec call. **Applicable Products:** All.
SBC-13225	The device fails to perform fax transcoding over SRTP. **Applicable Products:** All.

Incident	Description
SBC-13241	The TDM-to-SBC feature (TDMtoSBC) fails when the License Key includes "SBC=0". **Applicable Products:** All.
SBC-13287	The DispWebUsers CmdShell command displays the passwords of all users that are connected to the device, exposing the device to security threats. **Applicable Products:** All.
SBC-13374	The device stops sending RTCP XR reports after a call hold and retrieve. **Applicable Products:** All.
SBC-13399	The device's analog ports configuration for automatic dialing (**gateway analog automatic-dialing**) through CLI results in a "Command Failed" after entering the **activate** command. **Applicable Products:** Gateway.
SBC-13431	The CLI command **no user-info sbc-user-info all** fails to remove entries from the device's internal hash table. **Applicable Products:** SBC.
SBC-13605	The device resets in a specific scenario where the device terminates the SIP REFER and sends a new INVITE with Replaces. **Applicable Products:** SBC.
SBC-13619	The device resets in a transfer scenario when sending a re-INVITE with SDP and receiving a 200 OK without SDP. **Applicable Products:** SBC.
SBC-13728	The device remains locked (Admin State = Locked) after an Automatic Update with a graceful shutdown. **Applicable Products:** All.
SBC-13746	After enabling RADIUS functionality ('Enable RADIUS Access Control' parameter configured to Enable), subsequent logins fail (device gets stuck on the Web Login page). **Applicable Products:** All.

2.46 Patch Version 7.20A.204.759

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1133 and EMS/SEM Version 7.2.3113.

2.46.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-61: Resolved Constraints in Version 7.20A.204.759

Incident	Description
SBC-12497	Loss of connectivity to the device's Web interface due to losing some buffer resources. **Applicable Products:** All.
SBC-12874	The device sends incorrect reports to OVOC (leg is green despite packet loss) **Applicable Products:** All.
SBC-13811	The device's REST API ASR shows incorrect data upon no traffic. **Applicable Products:** All.
SBC-13858	The device's AMD feature doesn't function for SBC calls when SDP Delayed Offer Support is disabled. **Applicable Products:** All.
SBC-13960	The device's SIPRec XML metadata streams are flipped when recording the outbound leg. As a result, SIPRec fails. **Applicable Products:** All.
SBC-14143	The device rejects SIP NOTIFY requests with 481 because of incorrect IP-to-IP Routing Table rules. As a result, calls fail. **Applicable Products:** All.
SBC-14258 / SBC14277	The device opens voice towards IP address 1.1.1.1 in Direct Routing for Microsoft Teams. As a result, calls fail. **Applicable Products:** All.
SBC-14279	The device's kernel is exposed to security vulnerabilities CVE-2019-11477, CVE-2019-11478, and CVE-2019-11479. **Applicable Products:** All.
-	The SIPRec feature is not supported when the Media Transcoding Cluster feature is used. **Applicable Products:** Mediant 9000.

2.47 Patch Version 7.20A.204.789

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.2125 and EMS/SEM Version 7.2.3113.

2.47.1 New Features

This section describes the new features introduced in this version.

2.47.1.1 SDP Body with Multiple Coders Support

The device can be configured to support multiple coders in the SDP answer for specific SIP user agents (IP Profiles). If not supported (default) and the device receives an SDP answer with multiple coders ('m=' line) from the peer side, it uses only the first supported coder in the list for the RTP media. If supported, the device's behavior depends on whether DSP resources are required for the call. For further information, refer to the device's *User's Manual*. This feature is configured by the new IP Profile parameter 'SBC Multiple Coders' (IpProfile_SBCMultipleCoders).

Applicable Application: SBC.

Applicable Products: All.

2.47.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-62: Resolved Constraints in Version 7.20A.204.789

Incident	Description
SBC-10688	The device sends a SIP 200 OK with multi-coder support, even though it supports only one coder. As a result, one-way voice occurs. **Applicable Products:** SBC.
SBC-14634	The Web interface's CAS buttons are not displayed when using Google Chrome. **Applicable Products:** Gateway.
SBC-14640	The device is exposed to security vulnerability CVE-11478. **Applicable Products:** SBC.
SBC-14641	The device is exposed to security vulnerability CVE-11479. **Applicable Products:** SBC.
SBC-14687	The device's Message Manipulation logic for multiple AND/OR operands should be calculated from left to right. **Applicable Products:** SBC.
SBC-14697	The device replies with a 491 even though the previous transaction ended. As a result, the call fails. **Applicable Products:** SBC.

Incident	Description
SBC-14701	The Web interface's "Active Calls" field displays incorrect values. **Applicable Products:** Gateway.
SBC-14708	The device's Message Manipulation editor allows incorrect expression options for header.Privacy.Privacy.Identity. **Applicable Products:** SBC.
SBC-14890	The device experiences one-way voice for hold-unhold Microsoft TEAMs calls. **Applicable Products:** SBC.
SBC-14891	The device resets upon an RTP-to-SRTP call when the RTP side sends RTP packets with RTP version 0. **Applicable Products:** SBC.
SBC-15007	The device sends the incorrect IP address in SIP 200 OK for forked calls. As a result, no voce occurs. **Applicable Products:** SBC.
SBC-15039	The device's LDAP service doesn't resolve DNS for the Internal DNS Table. **Applicable Products:** All.
SBC-15307	The device reports a fax transcoding call as a voice call to OVOC. **Applicable Products:** All.
SBC-15356	The device experiences one-way voice after playing a RBT and stops playing a RBT. **Applicable Products:** All.

2.48 Patch Version 7.20A.250.003

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.48.1 New Features

This section describes the new features introduced in this version.

2.48.1.1 24-FXS Ports Support

The device now supports up to 24 FXS ports (i.e., six FXS modules, each providing four ports). This applies to the indoor-outdoor FXS module (CPN M1KB-VM-4FXS-O). Up until now, up to 20 FXS ports were supported.

Note: When installed with 24 FXS ports, only up to 20 FXS ports can be active (process voice traffic) simultaneously.

Applicable Application: Gateway.

Applicable Products: Mediant 1000B.

2.48.1.2 WebRTC License Key Update

The License Key for WebRTC now defines the ordered maximum number of concurrent WebRTC sessions. Up until now, the License Key simply enabled WebRTC. The License Key that is displayed in the device's management interfaces (e.g., License Key page) indicates the number of WebRTC sessions.

Note: For Customers with an old License Key, the management interfaces will display the WebRTC support (enabled or disabled) as in previous releases (without any indication about sessions).

Applicable Application: All.

Applicable Products: All.

2.48.1.3 New License Key for Microsoft Teams Support

A new License Key has been introduced ("SW/TEAMS") that enables Microsoft Teams Direct Routing support on the device. The License Key enables the following:

- Connecting and pairing of the device with Microsoft Teams Hub for Direct Routing
- Connecting the device to any SIP trunk, PSTN line or customer-owned telephony equipment such as third-party PBXs, analog devices, and Microsoft Phone System

The Microsoft License Key ("MSFT") must also be enabled on the device to activate the Microsoft Teams feature (most of AudioCodes devices are shipped by default with this license, except Mediant 500 Gateway & E-SBC, and Mediant 500L Gateway & E-SBC).

Note:

- An optional license to support HA-pair with Microsoft Teams can also be purchased ("SW/TEAMS/R").
- The "SW/TEAMS" license automatically enables the following voice coders:
  - SILK Narrowband
  - SILK Wideband

- OPUS Narrowband
- OPUS Wideband

- Number of required SBC sessions must be ordered separately.
- If media transcoding is required, the appropriate transcoding license sessions must be ordered separately.

Applicable Application: SBC.

Applicable Products: All.

2.48.1.4 Mediant CE in Microsoft Azure Environment

Mediant CE now supports deployment in a Microsoft Azure environment. Up until now, this type of deployment was available for evaluation purposes only.

Applicable Application: SBC.

Applicable Products: Mediant CE.

2.48.1.5 Unlimited Multiple Registrations per User with Same Contact

The device supports multiple user registrations containing identical SIP Contact headers. Up until now, the number of such multiple registrations (each identified by a unique AOR) per user was limited. It is now unlimited.

Applicable Application: SBC.

Applicable Products: All.

2.48.1.6 Enhanced Registrar Server Stickiness Feature

The Registrar Stickiness feature has been enhanced for handling refresh REGISTER messages. Up until now, when enabled ('Registrar Stickiness' parameter in the Accounts table configured to **Enable**), refresh REGISTER messages for the Account was always sent to the registrar server that accepted the previous REGISTER request. The 'Registrar Stickiness' parameter now provides an additional optional value—**Enable for Non-Register Requests**—which re-starts the registration process for refresh REGISTER messages, sending the message to one of the registrar servers according to the settings of the Proxy Set associated with the Account's Serving IP Group. For non-REGISTER messages, the behavior is the same as for the existing **Enable** option - once registered to the registrar, these messages are always sent to this registrar.

Applicable Application: Gateway & SBC.

Applicable Products: All.

2.48.1.7 SIP Account Re-registration upon INVITE Failure

The device can now be configured to re-register an Account upon the receipt of specific SIP response codes for a failed INVITE message. This is configured by the new 'Re-Register on Invite Failure' (**Enable** / **Disable**) parameter in the Accounts table. The response codes are configured by the new global parameter, AccountInviteFailureTriggerCodes.

Applicable Application: Gateway & SBC.

Applicable Products: All.

2.48.1.8 Resolution of DNS-A and SRV Queries per Proxy Set Address

When a Proxy Set includes an address that is configured with an FQDN, the maximum number of resolved domain names and IP addresses for SRV and DNS-A queries respectively, have been increased per proxy address:

- An SRV query sent by the device can return up to 50 hostnames (instead of 4 as in previous releases)
- A DNS-A (of a hostname) query sent by the device can resolve into up to 50 IP addresses (instead of 15 as in previous releases)

Applicable Application: All.

Applicable Products: All.

2.48.1.9 SIP 3xx Redirect Response Handling Enhancement

The device's handling of SIP 3xx responses (for INVITE requests) has been enhanced. The configuration of this handling is done by the existing parameter, 'Remote 3xx Mode' (SBCRemote3xxBehavior) in the IP Profile table. This parameter now provides two new optional values:

- IP Group Name: If the 'SIP Group Name' parameter of the dialog-initiating UA is configured with a non-empty value, the device changes the host part of the Contact header in the 3xx response to this value, before forwarding the 3xx response to the dialog-initiating UA.
- Local Host: The device changes the host part of the Contact header in the 3xx response before forwarding the 3xx response to the dialog-initiating UA. If the 'Local Host Name' parameter of the IP Group of the dialog-initiating UA is configured with a non-empty value, the device changes the host part of the Contact header to this value. If the 'Local Host Name' is empty, the device changes the host part to the device's IP address (the same IP address used in the SIP Via and Contact headers of messages sent to the IP Group).

Applicable Application: SBC.

Applicable Products: All.

2.48.1.10 Handling Advice of Charge Information in XML Format

The device can now handle SIP INFO messages containing advice-of-charge (AOC) information in XML format ('application/vnd.etsi.aoc+xml' body), for IP-to-Tel calls. Message Manipulation rules must be applied to the SIP INFO message to add a SIP AOC header with the AOC information from the XML body, and to remove the XML body.

Applicable Application: Gateway (Digital).

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 1000.

2.48.1.11 SIP Response Codes Exclusion from IDS

The device's Intrusion Detection System (IDS) feature can be configured to ignore specified SIP response codes as reasons for SIP-dialog establishment failures. If a specified SIP response code is received, the IDS feature does not include them in its' count of establishment failures.

This is configured by the new parameter:

- INI: IDSExcludedResponseCodes
- Web: 'Excluded Response Codes' (Signaling & Media > Intrusion Detection > IDS General Settings)
- CLI: configure voip > ids global-parameters > excluded-responses

Note:

- The parameter applies only to rejected responses received from the upstream server; not rejected responses generated by the device (except for 404).
- The response codes 401 and 407 are considered authentication failures and thus, are not applicable to this parameter.

The new event type—"establish-cac-reject" (Syslog) and "CAC rejection" (SNMP)—has been added to indicate "Dialog Establishment Failure" due to Call Admission Control (CAC) rejections.

The existing IDSAlarmClearPeriod ini file parameter has been added to the other management platforms:

- CLI: configure voip > ids global-parameters > alarm-clear-period
- Web: 'Alarm Clear Period' (Signaling & Media > Intrusion Detection > IDS General Settings)

Applicable Application: All.

Applicable Products: All.

2.48.1.12 SIPRec of SRTP-to-SRTP Calls Decrypted to RTP for SRS

For SIP-based media recording (SIPRec) of SRTP calls, the device can now send the recorded packets to the Session Recording Server (SRS) as RTP packets (i.e., decrypted). This is useful for SRS's that don't support the handling of recorded SRTP media. The feature is supported by adding an IP Profile for the SRS and configuring its' 'SBC Media Security Mode' parameter to **RTP**.

Applicable Application: SBC & Gateway.

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 2600/4000; Mediant 9000; Mediant Software.

2.48.1.13 NAT Traversal for NGINX with OVOC

When the device is located behind NAT, OVOC can only communicate with the device's embedded NGINX HTTP-based proxy using the device's public static NAT address. However, by default, the device sends OVOC its private address (in the proprietary X-AC-Proxy-URL header). To send its public (global) address, the new parameter, HttpProxyGlobalAddress (http-proxy-global-address) has been added. This parameter defines the global address (dotted-decimal notation) that it uses in the X-AC-Proxy-URL header for HTTP requests that the device sends to OVOC.

Applicable Application: All.

Applicable Products: All.

2.48.1.14 Default UDP Port Spacing

The default for UDP port spacing, configured by the UdpPortSpacing parameter, has been changed to 4 (instead of 5).

Applicable Application: All.

Applicable Products: Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.48.1.15 24-Hour Support for UTC Offsets

The value range for configuring the Universal Time Coordinate (UTC) offset (in seconds) from the local time (configured by the NTPServerUTCOffset parameter) has been changed. The new range is from -86400 seconds (-24 hours) to +86400 seconds (+24 hours). Therefore, the offset is no longer limited to +/-12 hours, but now supports +/-24-hour offsets.

Applicable Application: All.

Applicable Products: All.

2.48.1.16 Non-Operational HA Reduction for Switchovers and Upgrades

Hitless software upgrades and active-redundant switchovers for the device-pair in High-Availability (HA) mode has been optimized to reduce the duration that the HA system is non-operational during these operations.

Applicable Application: All.

Applicable Products: HA.

2.48.1.17 CLI Command Path and Name Changes

The following changes have been done to the CLI:

- CLI paths:
  - charge-code (Charge Codes table): configure voip > gateway dtmf-supp-service charge-code
  - inbound-map-set (GWInboundManipulationSet): configure voip > message > settings > inbound-map-set
  - outbound-map-set (GWOutboundManipulationSet): configure voip > message > settings > outbound-map-set
- CLI command names:
  - ssh-redundant-device-port (instead of ssh-redundant-proxy-port)

Applicable Application: All.

Applicable Products: All.

2.48.1.18 ISDN Progress Indicator and SDP Body for Tel-to-IP Calls

The device can now be configured to send SIP 180 messages without an SDP body depending on the value of the Progress Indicator (PI) information element (IE) in the received ISDN Progress message. This is configured by the new ini file parameter, NoSdpForIsdnPi (or CLI command, no-sdp-for-isdn-pi). The configuration value is a bit field, allowing you to specify more than one PI. The default is 0, meaning that SDP is included in the outgoing SIP 180 message, regardless of PI value.

Applicable Application: Gateway (PRI).

Applicable Products: Mediant 500; Mediant 800B; Mediant 1000B.

2.48.1.19 Name Field for Various Configuration Tables

The 'Name' field, which allows the administrator to configure a descriptive name for the configuration entity, has been added to the following configuration tables:

- Alternative Routing Reasons table
- Accounts table
- Call Setup Rules table

Applicable Application: All.

Applicable Products: All.

2.48.1.20 AES-256 SRTP Cipher Suites

The device now supports the following additional cipher encryption algorithms (crypto suites) according to RFC 6188 for SRTP:

- AES-CM encryption with a 256-bit key and HMAC-SHA1 message authentication with a 32-bit tag (AES_256_CM_HMAC_SHA1_32)
- AES-CM encryption with a 256-bit key and HMAC-SHA1 message authentication with

an 80-bit tag (AES_256_CM_HMAC_SHA1_80)

The feature is supported by the existing parameter, SRTPofferedSuites, which now provides two additional optional values – AES-256-CM-HMAC-SHA1-32 and AES-256-CM-HMAC-SHA1-80.

Applicable Application: All.

Applicable Products: All.

2.48.1.21 OAuth2 Token-based SIP Authentication

The device can authenticate any incoming SIP requests (e.g., REGISTER and INVITE) from client applications, based on access tokens with an OAuth2 Authorization Server (RFC 7662 and Internet Draft draft-ietf-sipcore-sip-authn-02 "Third-Party Authentication for Session Initiation Protocol (SIP)).

When the device receives a SIP request (with an OAuth access token) from a client application (e.g., WebRTC client), the device introspects the token with the OAuth Authorization server (HTTP server). Upon successful introspection, the device allows the client access to the device's resources (e.g., registration and calls) and continues to handle and process the SIP request as usual.

To support this feature, the following configuration updates have been introduced:

- New parameter added to the IP Groups table - 'SBC Server Authentication' (IPGroup_TypeSBCServerAuthType), which defines the authentication method:
  - [-1] According to Global Parameter (default, according to SBCServerAuthMode)
  - [0] Authentication is performed locally
  - [2] According to draft-sterman-aaa-sip-01
  - [3] Authenticate with OAuth authorization
- New parameter added to the IP Groups table - 'OAuth HTTP Service' (OAuthHTTPService_IPGroup), which assigns a Remote Web Service that is configured as the Authorization server when authenticating by OAuth.
- Existing IP Group table parameter, 'Authentication Method List' now supports the value "setup-invite" to support authenticating only setup INVITE requests and not re-INVITE requests

Applicable Application: SBC.

Applicable Products: All.

2.48.1.22 Message for Hidden Tables Removed from ini File

For configuration tables that are not displayed / "hidden" (due to security) in the ini file when they are modified, the message in the ini file indicating that the table has been modified will no longer be displayed.

Applicable Application: SBC and Gateway.

Applicable Products: All.

2.48.1.23 Device Uptime Display Format Changed

The format of the device uptime that is displayed in the 'Device Up Time' field on the Web interface's Device Information page (Monitor menu > Monitor tab > Summary folder > Device Information) has changed. The hundredth of a second is now displayed after a dot ("th" removed).

Applicable Application: All.

Applicable Products: All.

2.48.1.24 Temperature Indication for Media Components

For the Media Cluster feature, the 'Temperature' field has been added to the Media Components table. The field displays the Media Component's (i.e., Mediant 4000) CPU and DSP temperatures.

Applicable Application: SBC.

Applicable Products: Mediant 9000 (MC); Mediant Software (MC).

2.48.1.25 SIP Local and Remote Tags for CDRs

Gateway and SBC CDRs that are generated by the device can be customized to add the SIP 'tag' parameter values, which may be present in the From and To headers, for example:

```
To: Bob@company.com; tag = 1930394343437322
```

The tags are included in the CDR using the new CDR fields—SIP Local Tag (445) and SIP Remote Tag (446)—in the existing SBC CDR Format table and Gateway (Test) CDR Format table. The "local" tag is generated by the device in the outgoing SIP message while the "remote" tag is received in the incoming message. These fields are applicable to all types of CDRs, except Syslog SBC media. By default, these fields are not included in the CDR.

Note: The SIP tags are not always included in all CDR report types. For example, both tags are not included in Call-End CDR report types, while for Call-Connect CDR report types they are typically included.

Applicable Application: All.

Applicable Products: All.

2.48.1.26 Hostname for HA Network Monitoring

Monitored network entities (destinations) for the device's HA Network Monitor feature can now be configured as hostnames or FQDNs (in addition to IP addresses in dotted-decimal notation, as supported previously). This is configured in the existing HA Network Monitor table (HaNetworkMonitor).

Applicable Application: SBC & Gateway.

Applicable Products: Mediant 500; Mediant 800; Mediant 2600/4000; Mediant 9000; Mediant Software.

2.48.1.27 Minor Severity for acProxyConnectionLost SNMP Alarm

The existing SNMP alarm, acProxyConnectionLost has been enhanced to indicate a Minor severity alarm. This severity is raised upon any one of the following scenarios:

- All proxy servers were online and now at least one proxy server in the Proxy Set is offline (and at least one proxy server is still online)
- All proxy servers were offline and now at least one proxy server in the Proxy Set is online (and at least one proxy server is still offline)

Applicable Application: SBC & Gateway.

Applicable Products: All.

2.48.1.28 Device Authentication for the Automatic Update Feature

Device authentication with an HTTP/S or FTP server for the Automatic Update feature has changed. If the username and password (for basic authentication) is not included in the existing parameter AutoCmpFileUrl (which is used to configure the URL to the .cmp file), as supported in earlier versions, the device uses the username and password configured by the new parameter, AUPDUserPassword (configure system > automatic-update > credentials).

The syntax value is 'username:password'. This new parameter applies to basic and digest authentication (MD5).

The AUPDDigestUsername and AUPDDigestPassword parameters are now obsolete. If these parameters were configured and the device is subsequently upgraded to this version, the username and password configured for these parameters are automatically configured for the new AUPDUserPassword parameter.

Applicable Application: All.

Applicable Products: All. -

2.48.1.29 Debug Recording Packets Filtered by SIP Messages

Debug recording packets generated by the device can now be filtered to include only SIP messages (without Syslog messages). This is supported by configuring the 'Log Type' parameter in the Logging Filters table to the new optional value, **SIP Only**.

Applicable Application: All.

Applicable Products: All.

2.48.1.30 Improved Log Filtering

The device's log filtering feature (configured in the Logging Filters table) has been improved. Up until now, the device sent unfiltered syslog messages in the early call processing stages (e.g., Classification) and only once the source and destination IP Groups were determined did it start sending filtered Syslog messages according to configuration. Now, the initial syslog messages are not sent; only the filtered syslog mesages.

Applicable Application: All.

Applicable Products: All.

2.48.1.31 CLI Command Update for PSTN Debug Recording and Trace Level

PSTN debugging through CLI has been updated as follows:

- The command **debug voip interface trace-level** is now obsolete. The trace level can now only be configured per interface, using the existing command, **configure voip** > **interface** > **trace-level**.
- The command **debug pstn** is now obsolete. To send PSTN traces to Syslog, the following command has been added: **configure troubleshoot > pstn-debug**.
- To start a PSTN trace:
  - Per trunk (trace level is configured by the **debug voip interface trace level** command – see above): **configure troubleshoot** > **logging logging-filters** (existing command)
  - All trunks (whose trace level have been configured by the **debug voip interface trace level** command - see above): **debug debug-recording <IP Address> pstn-trace** (existing command)

Applicable Application: Gateway.

Applicable Products: Mediant 5xx; Mediant 8xx; Mediant 1000B.

2.48.1.32 Ping and Traceroute CLI Enhancements

The following enhancements have been introduced to the ping and traceroute CLI commands:

- Ping: QoS (ToS or Class) of the pinged packets can now be configured:

```
ping … [tos|traffic-class <0-254>]
```

- Traceroute: Protocol type (UDP or ICMP) of the traceroute can now be configured:

```
traceroute … proto [udp|icmp]
```

Applicable Application: All.

Applicable Products: All.

2.48.2 Known Constraints

This section lists known constraints.

Table 2-63: Known Constraints in Version 7.20A.250.003

Incident	Description
SBC-175	When downgrading the software version, Hitless Upgrade is not supported. **Applicable Products:** HA.
SBC-10527	For the Gateway application, when the SrtpOfferedSuites is configured to All, the device sends the SDP Offer with only four crypto lines, excluding AES 256. To offer AES 256, the parameter must be configured to the AES 256 option. **Applicable Products:** Gateway.
SBC-9523	Sometimes the Web browser doesn't display the 'IP Network', 'Signaling & Media' and 'Administration' tabs in the Web interface. Pressing the F5 key resolves this display problem. **Applicable Products:** All.
SBC-11172	After editing the description of a BRI port on the Web interface's Monitor page, the port's status, displayed on the Trunks & Channels Status page, is incorrect. **Applicable Products:** Mediant 500.
SBC-11343	The WebRTC feature will not function with the new Google Chrome version that will be released in a few weeks. **Applicable Products:** All.
SBC-11197	Call forking performed by the device does not function. **Applicable Products:** Mediant CE.
SBC-11399	The Test Call feature cannot accept incoming calls. A workaround is to first configure a Test Call rule for an outgoing call ('Call Party' parameter configured to Caller) and activate it (by clicking Dial), and then edit the same rule by changing the 'Call Party' parameter to Called. **Applicable Products:** All.
SBC-11356	For Azure platform, when set to use instance type (size) "Standard_D4_v2", the device (Media Components - MCs) crash (reset). **Applicable Products:** Mediant CE.
SBC-11568 / SBC-11603	If the manipulation tables of the Gateway application are configured with a pattern that exceeds 51 characters, the device crashes (resets). A workaround is to split the pattern into new patterns that are shorter than 51 characters). **Applicable Products:** Gateway.

2.48.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-64: Resolved Constraints in Version 7.20A.250.003

Incident	Description
VI-155352	The "sip" special string value for the 'Protocol' field in the Firewall table generates an error. This bug has been resolved and the string is now no longer required for specifying SIP ports. Instead, specific udp/tcp rules must be configured for SIP traffic according to the SIP Interfaces table. Therefore, before upgrading to this version, replace the rules containing the "sip" string. **Applicable Products:** All.
SBC-10597	If the user's password expires (configured by the Password Age parameter), upon a login attempt the user is prompted to change the password. However, after the user changes the password and logs in to the Web interface with the new password, no indication occurs that the user must click the Save button (i.e., not encircled by red). If it is not clicked, the user will be prompted again on the next login attempt (or for HA systems, when a switchover occurs) to change the expired password. **Applicable Products:** All.
SBC-8541 (VI-153895)	The device doesn't send the SIP Privacy header to ARM, which may result in the display of the Calling Party Number, which is not allowed. **Applicable Products:** All.
SBC-8720 (VI-154205)	When the device contains the DATA feature key, it resets when running the SNMP walk command. **Applicable Products:** All.
SBC-8843 (VI-154386)	The device doesn't send calls that were not established (due to an ARM routing error, for example) to the SIP call flow ladder in OVOC. **Applicable Products:** All.
SBC-8857 (VI-154409)	The device repeatedly generates syslog messages with "VQMON_DIVIDE". **Applicable Products:** All.
SBC-9168 (VI-154869)	The device displays CPU overload alarms for the NWST task. **Applicable Products:** All.
SBC-9203 (VI-154921)	The device has the incorrect timer calculation in Session-Expires when operating in Transparent mode **Applicable Products:** All.
SBC-10171 (VI-154923)	The parameter 'Digital Out Of Service Behavior Per Trunk' has a mismatch between the Web interface and CLI. **Applicable Products:** All.
SBC-9312 (VI-155074)	The device resets due to the HW Watchdog with exception Signal 904, Task IDLE **Applicable Products:** All.
SBC-9318 (VI-155081)	The device generates syslog messages with "SYS_HA: KA_MONITOR_POLL_TIMEOUT" **Applicable Products:** HA Devices.
SBC-9369 (VI-155167)	The device resets with the exception information of TASK DSPD. **Applicable Products:** All.

Incident	Description
SBC-9386 (VI-155189)	The device resets upon the receipt of an invalid RTCP-XR packet. **Applicable Products:** All.
SBC-9428 (VI-155253)	When the Proxy Set is associated with two SIP Interfaces, it gives precedence to the GW SIP Interface, which may cause issues with unsupported transport types on this SIP Interface. As a result, connectivity fails. **Applicable Products:** All.
SBC-9489 (VI-155352)	The device's firewall prevents the option to configure an access list rule with protocol 'SIP'. **Applicable Products:** All.
SBC-9509 (VI-155395)	If the ini file is downloaded from the device, attempting to upload it to the device fails if the parameter OSNInternalVLAN is configured to 1. **Applicable Products:** All.
SBC-9515 (VI-155407)	The CLI command TraceRoute doen't function. **Applicable Products:** All.
SBC-9535 (VI-155448)	Configuring the IP-to-IP Routing table with alternative routing rules based on a condition fails. **Applicable Products:** All.
SBC-9592 (VI-155532)	When trying to use the Rand.Number search key in the Call Setup Rule table, an error is generated: "CallSetupRules-CrossValidation: Rand.number.1.10 MATRIX CallSetupRules; Unable to Activate Line(29) since it is Invalid". **Applicable Products:** All.
SBC-9642 (VI-155619)	The device should use default of "4" for the UDP Port Spacing parameter (currently, the default is "5"). **Applicable Products:** Mediant 4000; Mediant Software.
SBC-9691 (VI-155696)	When performing a debug recording, SIP packets are not marked with the correct DSCP QoS. **Applicable Products:** All.
SBC-9694 (VI-155701)	The device resets due to configuration of a Message Manipulation variable that exceeds its range (0-13). **Applicable Products:** All.
SBC-9728 (VI-155751)	When the device sends an HTTP GET message, it uses an IP address in the Host instead of an FQDN, as required by the HTTP 1.1 RFC. **Applicable Products:** All.
SBC-9792	The UTC offset is limited to +/-12 hours and therefore, it's impossible to set it to +13 (for example, for New Zealand). **Applicable Products:** All.
SBC-9893	The CDR sorting in the Web interface doesn't function. **Applicable Products:** All.
SBC-9942	When the Web interface's session timeout expires, it is not possible to reconnect to the device (only by refreshing the browser page). **Applicable Products:** All.
SBC-9972	The device truncates the HTTP GET path and limits it to 125 characters, causing an incorrect HTTP GET query. **Applicable Products:** All.

Incident	Description
SBC-10576	The IpProfile_SBCRemoveCryptoLifetimeInSDP parameter in the IP Profiles table doesn't remove the crypto lifetime from SIP 18x. **Applicable Products:** All.
SBC-10538	The **default-window-height** CLI command (window resize) doesn't function. **Applicable Products:** All.
SBC-10320	The **show dsp status** CLI command displays incorrect units ('sessions' instead of 'DSP resources'). **Applicable Products:** All.
SBC-10556	The performance monitoring SNMP MIB, acPMSBCMediaLegsVal shows incorrect number of active media channels when some calls fail upon a switchover. **Applicable Products:** All.
SBC-10587	The device's SIPREC metadata format doesn't comply with RFC 7865 - UUID for each metadata should be 128 bits (16 bytes) and not 64 bits (8 bytes). **Applicable Products:** All.
SBC-10717	When the **set-default snapshot** CLI command is run from its old, hidden (but supported) directory, the device resets. **Applicable Products:** Mediant 9000; Mediant Software.
SBC-10812	The **show voip subscribe list** CLI command resets the device. **Applicable Products:** All.
SBC-10846	The device doesn't save the CLI Privilege password in the backed-up ini file. **Applicable Products:** All.
SBC-109050	The performance monitoring SNMP MIB, acPMSIPIPGroupInDialogs has incorrect counter calculation. **Applicable Products:** All.
SBC-10320	The number of channels displayed in the 'Num DSP Channels' field in the ini file is incorrect. As the number of channels depends on many factors (features and coders), this field has been removed from the ini file. **Applicable Products:** All.

2.49 Patch Version 7.20A.250.256

This patch version includes new features and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.49.1 New Features

This section describes the new features introduced in this version.

2.49.1.1 Maximum DNS-Resolved IP Addresses per Proxy Set

The maximum number of DNS-resolved IP addresses per Proxy Set has been increased from 15 to 50 for Mediant Software 8-16GB memory.

Applicable Application: SBC.

Applicable Products: Mediant Software (8-16GB).

2.49.1.2 Call Forking with ICE in Microsoft Teams Environment

The device now supports working in a Microsoft Teams environment that implements call forking with Interactive Connectivity Establishment (ICE). The only required device configuration for this feature is enabling ICE (STUN message handling) for the IP Group representing Microsoft Teams (existing 'ICE Mode' parameter set to **Lite**).

Applicable Application: All.

Applicable Products: MP-1288; Mediant 500; Mediant 500L; Mediant 800; Mediant 2600; Mediant 4000; Mediant 90xx; Mediant SW.

2.49.2 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-65: Resolved Constraints in Version 7.20A.250.256

Incident	Description
SBC-9664 (VI-155567)	Modifications on the Web interface's Trunk Group page doesn't affect the registration\un-registration of the Trunk Group. As a result, registration fails. **Applicable Products:** Gateway.
SBC-9886 / SBC-10049	When the device receives STUN responses, it doesn't latch the RTP to the correct STUN (should latch RTP according to the STUN with the highest priority). As a result, no voice is experienced. **Applicable Products:** SBC.
SBC-10424	When upgrading the License Key for an HA system using the Hitless method, after the upgrade, the active unit shows an alarm about a License Key mismatch. **Applicable Products:** HA.
SBC-10459	The device sends incorrect packet loss report values to OVOC **Applicable Products:** SBC.
SBC-10871	For a direct media session, when the device receives an SDP offer with 'a=sendonly', it answers with 'a=sendonly' (instead of 'a=recvonly'). As a result, direct media ends. **Applicable Products:** SBC.
SBC-10939	When the Proxy Set is configured with a DNS, for every DNS-resolved IP address received, the device sends the trap message "TRAP: E_acProxyConnectionLost[2] CLEAR Proxy found" to OVOC (flooding OVOC with these messages). **Applicable Products:** SBC.
SBC-11252	The RAI alarm is not functioning correctly (sometimes false alarms). **Applicable Products:** Gateway.
SBC-11269	When the device receives a SIP 200 OK with a large Allow header, it crashes (resets). **Applicable Products:** SBC.
SBC-11392	The device experiences DSP utilization errors ("not enough utilization for resource 14"). As a result, no voice is experienced. **Applicable Products:** SBC.
SBC-11496	When a SIP Connect user registers with the device on two different SIP Interfaces, the device fails to handle responses from the user. As a result, calls fail. **Applicable Products:** SBC.
SBC-11548	The device crashes (resets) with exception information of "TASK:SPMR" **Applicable Products:** SBC.
SBC-11568 / SBC-11603	The device crashes (resets) when the 'Destination Phone Pattern' parameter in the Tel-to-IP Manipulation Table is longer than 50 characters. **Applicable Products:** Gateway.

Incident	Description
SBC-11572	The device sends a SIP CANCEL message with the wrong Reason header value. Instead of "Reason: SIP ;cause=200 ;text="Call completed elsewhere"" it sends "Reason: SIP ;cause=400 ;text="local"". **Applicable Products:** SBC.
SBC-11589	The device crashes (resets) due to several DSP errors such as “Max number of failures (type=200) was reached for Dsp 52. Dsp is refreshed” and “Restart reason of DSP #52 is: Keep_Alive_Failure (RDC=1653)”. **Applicable Products:** SBC.
SBC-11747	The device fails to process SIP PRACK correctly for specific scenarios (received 18x while PRACK in process). As a result, the call fails. **Applicable Products:** SBC.
SBC-11808	When the device is configured with 1+1 port redundancy and the cable is later disconnected from the active Ethernet port, a device reset is required. **Applicable Products:** SBC.
SBC-11867	The device's CLI menu “voip > application” was removed (it should be hidden). As a result, CLI Script file fails when it includes an enable application setting associated with this command path. **Applicable Products:** SBC.
SBC-11913	In certain scenarios, the device fails to remove users (doesn't unregister them) from its users' database, resulting in a database overflow. **Applicable Products:** SBC.
SBC-12100	The device's optimized syslog contains “^^” prefix on each line (which should not appear). **Applicable Products:** SBC.

2.50 Patch Version 7.20A.250.273

This patch version includes resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.164 and EMS/SEM Version 7.2.3113.

2.50.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-66: Resolved Constraints in Version 7.20A.250.273

Incident	Description
SBC-12224	The device experiences a problem with stack creation in standalone mode (not HA), which produces a network interface error. **Applicable Products:** Mediant CE

2.51 Patch Version 7.20A.250.413

This patch version includes new features and known constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1116 and EMS/SEM Version 7.2.3113.

2.51.1 New Features

This section describes the new features introduced in this version.

2.51.1.1 Periodic CDR Transfer to Remote SFTP Server

CDRs that are locally stored on the device can be transferred periodically to remote SFTP servers. The CDRs are sent in the same file format as stored on the device - compressed (GZIP), comma-separated value (CSV). The servers (currently, up to two) are configured in the new SBC CDR Remote Servers table (Troubleshoot menu > Troubleshoot tab > Call detail Record > SBC CDR Remote Servers). In addition, the following new standalone parameters have been added to the Call Detail Record Settings page:

- 'CDR Servers Send Period' - defines the periodic interval (in seconds) at which the device checks if a CDR file is available for sending to the remote server
- 'CDR Servers Bulk Size' – defines the maximum number of files that the device can send to the remote server per transfer operation (i.e., batch of files)
- 'Pending CDR Files' (read-only) - displays the number of CDR files that are waiting to be sent to the remote server

The CDR servers (including number of pending files) are also shown in the new CLI command, `show cdr-servers`.

If the device fails to send CDRs to **all** the configured servers, the new SNMP alarm, acCDRServerAlarm (OID 1.3.6.1.4.1.5003.9.10.1.21.2.0.142) is sent.

Applicable Application: SBC.

Applicable Products: Mediant 90xx; Mediant Software.

2.51.1.2 New HA Network Monitor Status for Unresolved Hostnames

For the HA Network Monitor feature, if the destination host is configured with a hostname and it cannot be resolved into an IP address, the device displays a status message in the HA Network Monitor table indicating this ("Host not resolved").

Applicable Application: SBC.

Applicable Products: Mediant 500; Mediant 800; Mediant 2600; Mediant 4000; Mediant 90xx; Mediant Software.

2.51.2 Known Constraints

This section lists known constraints.

Table 2-67: Resolved Constraints in Version 7.20A.250.413

Incident	Description
SBC-13833	When sending CDR files to a remote SFTP server during high traffic load (CPS), in rare cases the server may receive an unzipped file with the .zip filename extension, which is invalid (even though its contents are valid). **Applicable Products:** Mediant 90xx; Mediant Software.

2.52 Patch Version 7.20A.252.011

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1116 and EMS/SEM Version 7.2.3113.

2.52.1 New Features

This section describes the new features introduced in this version.

2.52.1.1 Call Setup Rules for HTTP POST Requests

Call Setup rules can be configured for HTTP POST methods (in addition to the already supported HTTP GET method). Such Call Setup rules allows the device to send HTTP POST requests to an HTTP server. This is configured in the Call Setup Rules table by the following new optional values of the 'Request Type' parameter:

- **HTTP POST Notification:** This option is used to send an HTTP POST request that is for notification purposes only (i.e., device does not expect a response from the server). For example, a Call Setup Rule can be configured to send an HTTP POST request whenever the device receives a 911 (emergency) call.
- **HTTP POST Query:** This option is used to send an HTTP POST request that requires a response from the server (Http.Result keyword).

In addition, a new optional value—**None**—has been added to the 'Action Type' parameter. This is used when no action is needed by the device, for example, when sending an HTTP POST request to notify the HTTP server.

Unlike HTTP GET requests which include all required data in the URL, HTTP POST requests include a URL and a message body. Call Setup Rules can now be used to customize (manipulate) the HTTP Content-Type header (e.g., Content-Type: application/x-www-form-urlencoded) and message body of POST requests, using the following new keywords:

- Http.Request.Body: Customizes the message body and is used in the 'Action Subject' parameter to add or modify the value of the body.
- Http.Request.Content-Type: Customizes the Content-Type header and is used in the 'Action Subject' parameter to add or modify the value. For POST requests, the header is omitted by default; for GET requests, it is set to "html/text".

Due to this feature, the parameters in the Call Setup Rules table have been renamed:

- 'Query Type' has been renamed 'Request Type'
- 'Query Target' has been renamed 'Request Target'
- 'Search Key' has been renamed 'Request Key'

Applicable Application: All.

Applicable Products: All.

2.52.1.2 Customization of SNMP Alarm Severity Levels

SNMP trap alarms (not events) can now be customized. This includes the following:

- Changing an alarm's severity level (e.g., from Minor to Major)
- Suppressing an alarm's severity level

- Suppressing an alarm

This feature is configured in the following new table:

- Web: Alarms Customization (Setup menu > Administration tab > SNMP folder > Alarms Customization)
- CLI: configure system > snmp alarm-customization
- ini: AlarmSeverity

Applicable Application: All.

Applicable Products: All.

2.52.1.3 Customization of User Access Privileges per Web Page

Read-write and read-only access privileges of user levels (Monitor, Administrator, or Security Administrator) per Web page in the device's Web interface can be customized (overriding default access privileges).

This feature is configured in the following new table:

- Web Interface: Customize Access Level table (Setup > Administration > Web & CLI > Customize Access Level)
- ini File: WebPagesAccessLevel

Applicable Application: All.

Applicable Products: All.

2.52.1.4 User-Defined Performance Monitoring SNMP MIBs

Up to 26 user-defined Performance Monitoring (PM) MIB groups can be configured to count specified SIP failure responses (e.g., SIP 408) or responses generated internally by the device (e.g., CAC limit reached or NER threshold crossed). The PMs can be configured to count the responses due to sent SIP INVITE or REGISTER messages.

Each user-defined PM group includes the following PM MIBs:

- acPMSBCInUserDefinedFailures<1-26>Table: Counts the total incoming responses
- acPMSBCOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses
- acPMSBCSRDInUserDefinedFailures<1-26>Table: Counts the total incoming responses per SRD
- acPMSBCSRDOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses per SRD
- acPMSBCIPGroupInUserDefinedFailures<1-26>Table: Counts the total incoming responses per IP Group
- acPMSBCIPGroupOutUserDefinedFailures<1-26>Table: Counts the total outgoing responses per IP Group

The feature is configured in the new table, User Defined Failure PM (Monitor menu > Monitor tab > Performance Monitoring folder > User Defined Failure PM).

Applicable Application: SBC.

Applicable Products: All.

2.52.1.5 New SBC Performance Monitoring SNMP MIBs

The following new Performance Monitoring (PM) SNMP MIBs (counters) have been added for the SBC application:

- Attempted calls:
  - acPMSBCInAttemptedCallsTable
  - acPMSBCSRDInAttemptedCallsTable
  - acPMSBCOutAttemptedCallsTable

- acPMSBCSRDOutAttemptedCallsTable

- Established calls:
  - acPMSBCInEstablishedCallsTable
  - acPMSBCSRDInEstablishedCallsTable
  - acPMSBCOutEstablishedCallsTable
  - acPMSBCSRDOutEstablishedCallsTable
- Broken connection:
  - acPMSBCMediaBrokenConnectionCallsTable
  - acPMSBCSRDMediaBrokenConnectionCallsTable
  - acPMSBCIPGroupMediaBrokenConnectionCallsTable
- Short call duration: Calls whose duration are less than a configurable duration in seconds, using the new ini file parameter, ShortCallSeconds.
  - acPMSBCInShortCallsTable
  - acPMSBCOutShortCallsTable
  - acPMSBCSRDInShortCallsTable
  - acPMSBCSRDOutShortCallsTable
  - acPMSBCIPGroupInShortCallsTable
  - acPMSBCIPGroupOutShortCallsTable
- Attempted registrations:
  - acPMSBCInAttemptedRegistrationsTable
  - acPMSBCOutAttemptedRegistrationsTable
  - acPMSBCSRDInAttemptedRegistrationsTable
  - acPMSBCSRDOutAttemptedRegistrationsTable
  - acPMSBCIPGroupInAttemptedRegistrationsTable
  - acPMSBCIPGroupOutAttemptedRegistrationsTable
- Successful registrations:
  - acPMSBCInSuccessfulRegistrationsTable
  - acPMSBCOutSuccessfulRegistrationsTable
  - acPMSBCSRDInSuccessfulRegistrationsTable
  - acPMSBCSRDOutSuccessfulRegistrationsTable
  - acPMSBCIPGroupInSuccessfulRegistrationsTable
  - acPMSBCIPGroupOutSuccessfulRegistrationsTable
- Calls per second (CPS):
  - acPMSBCInCapsTable
  - acPMSBCOutCapsTable
  - acPMSBCSrdInCapsTable
  - acPMSBCSrdOutCapsTable
  - acPMSBCIPGroupInCapsTable
  - acPMSBCIPGroupOutCapsTable

Note: To free up memory for these new PM MIBs, the maximum number of SRDs, IP Groups and Routing Policies that can be configured has been reduced. For configuration table capacity, see Section Configuration Tables Capacity on page 27

Applicable Application: SBC.

Applicable Products: All.

2.52.1.6 FXS Phone Number Configuration via Phone Keypad

Phone numbers of analog phones connected to the device's FXS ports can now be configured using the phones' keypad. The feature is configured by the new parameter, KeyPortConfigure (configure voip > gateway analog keypad-features key-port-configure). The parameter configures the key sequence that needs to be dialed on the phone's keypad to access this configuration mode. Once accessed, the phone number can be configured using the keypad. The number sign key (#) indicates the end of the number. For example, if the parameter is configured to "*81", the key sequence to configure the phone number is "*81<phone number>#". To delete the phone number, the key sequence is dialed without the phone number (e.g., "*81#").

The FXS port must be assigned to a Trunk Group ID (in the Trunk Group table) that is dedicated only to this port. It can be configured with or without a phone number, which can be changed or deleted by pressing the special key sequence.

Applicable Application: Gateway (FXS).

Applicable Products: MP-1288; Mediant 500L; Mediant 800; Mediant 1000.

2.52.1.7 IDS Count for WebSocket Connection Failures

The device's Intrusion Detection System (IDS) feature now also counts WebSocket connection (establishment) failures.

In addition, IDS for TLS handshake failures is counted only for incoming connections (instead of both incoming and outgoing).

Applicable Application: All.

Applicable Products: All.

2.52.1.8 IP Subnet Conditions for Message Manipulation Rules

Message Manipulation rules can now be configured with conditions that check if an IP address (IPv4 or IPv6) in a SIP header (e.g., From and To) belongs to a specific subnet. The feature is configured using the new operand, "insubnet" (or "!insubnet" for not in subnet) in the 'Condition' field. The subnet is expressed in CIDR (Classless Inter-Domain Routing) notation. A few examples are shown below:

```
Header.From.URL.Host insubnet '10.8.0.0/8'
Header.To.URL.Host !insubnet ' 172.0.0.0/10'
Header.From.URL.Host insubnet 'ffff:a08:705:0:0::/32'
```

Applicable Application: All.

Applicable Products: All.

2.52.1.9 Enhanced Test Call Feature

The Test Call feature, configured in the Test Call Rules table, has been enhanced with the following new features:

- Test Calls can now be configured to play a specific tone from the installed PRT file to the called party when the call is answered. This feature is configured by the new parameter, 'Play Tone Index' (Test_Call_PlayToneIndex), which specifies the index of the tone as defined in the file. Up until now, a default tone (Index 22) was played from the PRT file.
- Test Calls can now be configured with the Stream Control Transmission Protocol (SCTP) transport type. This is configured by the new optional value, **SCTP** for the 'Destination Transport Type' parameter.
- The 'Route By' parameter's optional value **Tel-to-IP** is now obsolete (the remote endpoint is now defined as an IP Group or IP address).
- The following parameters concerned with SDP Offer-Answer negotiations and which are typically configured in the IP Profiles table have been added to the Test Call Rules

table:

- 'Offered Coders Group': Assigns a Coders Group, whose coders are added to the SDP Offer
- 'Allowed Coders Group': Assigns an Allowed Audio Coders Group, which lists permitted coders
- 'Allowed Coders Mode': Configures the mode of operation for Allowed Coders - only allowed coders (restriction) or according to listed priority (preference)
- 'Media Security Mode': Configures media security (SRTP and/or RTP)
- 'Play DTMF Method': Configures the method for sending DTMF digits (RFC 2833 or In-band)

The values of these parameters override the values of the corresponding parameters in the IP Profile of the IP Group that is associated with the Test Call.

- Support for basic NetAnn parameters in the Request-URI of incoming INVITE messages to play specific tones from the installed PRT file:
  - *early=yes*: The device sends a SIP 183 with SDP instead of connecting the call (no 200 OK).
  - *play=<Tone Index in PRT File>*: Defines the tone (prompt) to play from the PRT file
  - *repeat=<Integer>*: Defines how many times the prompt is played before disconnecting the call
  - *delay=<Time in msec>*: Defines the delay between each played prompt

  For example: INVITE sip:200@1.1.1.1;early=yes;play=15;repeat=3

Note: Some of these new features are not backward compatible. For more information, see the known constraint SBC-10191 in this document.

Applicable Application: All.

Applicable Products: All.

2.52.1.10 Increase in Number Ranges for Dial Plan Rules

The maximum set of numbers (consisting of single numbers and/or range of numbers) that can be configured for prefixes and suffixes of dial plan rules (in the Dial Plan Rule table) has been increased significantly. Up until now, each dial plan rule could only include up to two sets of numbers (on average) for the prefix/suffix, for example, [101-103,911].

The following dial plan rule example represents number prefixes and is configured with six sets of numbers (each separated by a comma) consisting of ranges and single numbers: [120-125,150,160-164,170,200,210-215]

The maximum set of numbers can be calculated by multiplying the maximum number of supported dial plan rules by six. For example, if the maximum number of supported dial plan rules is 100,000, then the maximum set of numbers is 600,000 (6*100,000).

Applicable Application: All.

Applicable Products: All.

2.52.1.11 Modification and Deletion of IDS Default Policies

The default IDS policies in the IDS Policies table can now be modified and deleted (previously, they were read-only). In addition, if the table is empty (i.e., all policies have been deleted), the default policies can be returned by disabling IDS, and then enabling it again.

Applicable Application: All.

Applicable Products: All.

2.52.1.12 Dedicated TCP Socket for FXS Channel Signaling

The device can be configured to use a dedicated TCP socket for SIP traffic (REGISTER, re-REGISTER, SUBSCRIBE, and INVITE messages) for each FXS analog channel (endpoint). The dedicated TCP socket is the socket from which the endpoint successfully registered to the registrar server.

Up until now, multiple endpoints used the same TCP socket. If SIP authentication failed for one endpoint with the server and the server "blacklisted" the TCP socket, it meant that the server blocked traffic from all the other endpoints that used this same socket.

The dedicated socket is used **only** for SIP requests from the Trunk Group (to which the endpoints belong) whose destination is the same as the destination where the endpoint registered (i.e., same Proxy Set and "Serving" IP Group). If the endpoint is not registered to the Serving IP Group over a TCP connection, calls from the endpoint to the Serving IP Group are rejected (and trigger an immediate registration attempt).

The feature is configured by the new parameter—'Dedicated Connection Mode' (**Reuse Connection**/0 and **Connection per Endpoint**/1)—in the Trunk Group Settings table. When enabled, the table's 'Serving IP Group' must be configured and the 'Registration Mode' parameter must be configured to **Per Endpoint**.

Applicable Application: Gateway (FXS).

Applicable Products: MP-1288.

2.52.1.13 Call Forking by Third-Party Routing Server

The device supports call forking that is handled by a third-party Routing server. When an IP-to-IP Routing rule is matched and its 'Destination Type' parameter is configured to **Routing Server**, the device sends an HTTP getRoute request to the Routing server. When it receives a successful response from the server, the device sends an INVITE message to a destination based on the response. If the routingMethod from the server is "fork", the device sends another getRoute request and upon a successful response, sends another INVITE to another destination based on the response, and so on. This call forking process continues until no routingMethod is received or it is set to “seq”, or there is a failed response from the server. If all the contacts fail (4xx), the device falls back to an alternative route, if exists, from the server. If 3xx is received for any of the forked destinations, the device handles it after all the forked INVITEs have been terminated.

Applicable Application: All.

Applicable Products: All.

2.52.1.14 Standalone OVOC QoE Parameters in Table Format

The standalone OVOC parameters for QoE have been replaced by a table to facilitate configuration (especially, the option to select an IP Interface). As such, the OVOC page (Setup > Signaling & Media > Media > Quality of Experience > OVOC) has been replaced by the following new table:

- Web: Quality of Experience Settings (Setup > Signaling & Media > Media > Quality of Experience > Quality of Experience Settings)
- ini File: QOESettings
- CLI: configure voip > qoe qoe-settings

Note: Due to this feature, the following standalone parameters are now obsolete: QOEServerIP, QOESecondaryServerIp, QOEInterfaceName, QOEEnableTLS, QoETLSContextName, and QoeReportMode.

Applicable Application: All.

Applicable Products: All.

2.52.1.15 Configurable Keep-Alive Time with OVOC

The keep-alive interval (in seconds) between every keep-alive message sent by the device to OVOC is now configurable. Increasing the rate of keep-alive messages can be useful in keeping the communication link between the device and OVOC open when there is no traffic flow between them.

The feature is configured by the new 'Keep Alive Time Interval' parameter in the Quality of Experience Settings table (Setup > Signaling & Media > Media > Quality of Experience > Quality of Experience Settings).

Applicable Application: All.

Applicable Products: All.

2.52.1.16 32-bit Prefix Length for IPv4 Network Interfaces

IPv4 network interfaces in the IP Interfaces table can now be configured with 32-bit prefix length. In addition, the interface's Default Gateway no longer needs to be in the same subnet as the interface.

Applicable Application: SBC.

Applicable Products: Mediant 9000; Mediant Software.

2.52.1.17 DiffServ for HA Maintenance Traffic

Differentiated Services (DiffServ) can now be configured for HA Maintenance traffic. This is traffic that flows on the HA Maintenance interface between the two devices participating in a High-Availability (HA) system. Therefore, if needed, higher priority can be given to this traffic type. This feature is configured by the new ini file parameter, HAMaintenanceIFDiffServValue (device reset required).

Applicable Application: All.

Applicable Products: Mediant 500; Mediant 800; Mediant 2600; Mediant 4000; Mediant 9000; Mediant Software.

2.52.1.18 Delayed Transition to HA Operational State

HA systems can now be configured to delay (in seconds) their transition from HA non-operational state, which occurs during HA synchronization between active and redundant devices, to HA operational state. This feature may be useful, for example, to delay HA switchover when using switches with spanning tree protocol (STP) that take a long time until their ports (to which the redundant device is connected) is ready. In such scenarios, if this feature were not enabled, after synchronization there would be no connectivity between the redundant device's network interface and the switch.

The feature is configured by the following new parameter:

- CLI: configure network > high-availability settings > operational-state-delay
- ini file: HAOperationalStateDelayInSec

Applicable Application: All.

Applicable Products: HA.

2.52.1.19 Idle Timeout for CLI Sessions through RS-232 Serial Interface

The device now automatically activates a session timeout counter when a CLI session becomes idle. If the session is idle for up to five minutes (not configurable), the user is automatically logged out of the CLI. This applies only to CLI sessions that are established (successfully logged in) through an RS-232 serial connection (i.e., not Telnet or SSH).

Applicable Application: All.

Applicable Products: All.

2.52.1.20 Reset Confirmation Message for File Loads via Web

When loading files (ini file, CLI Startup Script file, and Configuration Package file) using the Web interface's Configuration File page, a confirmation message box informs the user that the device will reset after the file is loaded. The user can accept or cancel the operation.

Applicable Application: All.

Applicable Products: All.

2.52.1.21 Configurable Hostname for SBCs and Gateways

The device can now be configured with a hostname (FQDN) in addition to the already supported IP address. This is configured by the following new parameter:

- Web interface: 'Host Name' (Setup > IP Network > Advanced > Network Settings)
- Cli: configure network > network-settings > hostname
- ini File: Hostname

When configured, the hostname affects the following:

- The device's Web interface and CLI (remotely via Telnet/SSH) can be accessed using its' IP address or hostname.
- The CLI prompt displays the hostname instead of the device type.
- The Web interface displays the hostname (first 16 characters only) on the toolbar instead of the device type.
- The SNMP interface's SysName object (under MIB-2) is set to this hostname.
- For certificate signing requests (CSR) to a Certification Authority (CA), the hostname can be used as the Common Name (CN or Subject Name) and Subject Alternative Name (SAN)
- For CDR local storage (supported only by Mediant Software and Mediant 9000 SBCs), the name of the CDR file can include the device's hostname, by using the newly supported format specifier (placeholder) "%<hostname>" when configuring the name with the 'CDR File Name' (CDRLocalFileName) parameter.
- In HA systems, the device-pair share the same hostname

Applicable Application: All.

Applicable Products: All.

2.52.1.22 FQDN Address for OVOC Server for QoE Reporting

The address of the OVOC server (primary and redundant) to where the device sends QoE reports can now be configured as an FQDN (in the Quality of Experience Settings table). If only the primary server is configured with an FQDN, the device accepts up to two DNS-resolved IP addresses, which are used as primary and redundant IP addresses. If both the primary and redundant server are configured with an FQDN, only the first DNS-resolved IP address from each FQDN is used.

Applicable Application: All.

Applicable Products: All.

2.52.1.23 TLS Certificate Verification and FQDN

The device now can verify certificates based on hostname:

- TLS certificates used by the device for HTTPS-based communication with OVOC now supports a hostname (FQDN). Up until now, certificates were issued only for IP addresses.
- The following new certificate-related parameters have been added to the Quality of

Experience Settings table:

- 'Verify Certificate': validates the server's certificate
- 'Verify Certificate Subject Name': validates the server's certificate subject name (CN/SAN), which contains the hostname or IP address of the server

- The 'Verify Certificate Subject Name' has been added to the Remote Web Services table, which validates the certificate's subject name, which contains the hostname or IP address of the server.
- Auto-Update mechanism: verify-ssl-subject-name has been renamed to verify-cert-subject-name. If the server's URL contains a hostname, the device validates the server's certificate subject name (CN/SAN) against this hostname (and not IP address); otherwise, the device validates the server's certificate subject name against the server's IP address.

Applicable Application: All.

Applicable Products: All.

2.52.1.24 Default Cipher Suite Changed for TLS Context

The default cipher suite for TLS clients and servers (configured in the TLS Contexts table) has changed to "DEFAULT". This is an OpenSSL keyword for the recommended configuration for the default cipher list, which is determined at compile time and is normally ALL:!EXPORT:!LOW:!aNULL:!eNULL:!SSLv2.

Applicable Application: All.

Applicable Products: All.

2.52.1.25 Automatic Re-Generation of Default Self-Signed TLS Certificate

If the device's default self-signed certificate (Index 0) is about to expire (less than a day), the device automatically re-generates a new self-signed certificate.

Applicable Application: All.

Applicable Products: All.

2.52.1.26 Password Display Obscured (Encrypted) in CLI

The existing CLI command, **password-obscurity** (configure system > cli-settings), which enables the display of passwords in the CLI as encrypted (obscured), now applies to all parameters –standalone and tables—that configure passwords.

Below shows two examples of a password for a Remote Web Service that is displayed in obscured and plain text format in the output of the **show running-config** command:

- Password displayed as encrypted:

```
rest-password 8ZybmJHExMTM obscured
```

- Password displayed in plain text:

```
rest-password john1234
```

Note: Due to this feature, the names of the following parameters have been modified to reflect that they configure passwords:

- SNMPReadOnlyCommunityString has been renamed SNMPReadOnlyCommunityStringsPassword
- SNMPReadWriteCommunityString has been renamed SNMPReadWriteCommunityStringsPassword
- SNMPTrapCommunityString has been renamed SNMPTrapCommunityStringPassword

- ntpAuthMd5Key has been renamed ntpAuthMd5KeyPassword
- CLIPrivPass has been renamed CLIEnableModePassword

Applicable Application: All.

Applicable Products: All.

2.52.1.27 SNMP Alarm for Off-hooked Phone

The device can be configured to send the new SNMP alarm, acAnalogLineLeftOffhookAlarm, to indicate that an FXS phone connected to one of the device's FXS ports has been in off-hook state for more than a user-defined time (in seconds). The tImeout is configured by the new parameter, FXSOffhookTimeoutAlarm (configure voip > gateway analog fxs-setting fxs-offhook-timeout-alarm). The timer starts once the reorder tone begins playing when the phone goes off hook. The alarm is cleared when the hook-flash button is pressed or the phone returns to on-hook state.

Applicable Application: Gateway (FXS).

Applicable Products: MP-1288; Mediant 500L; Mediant 800; Mediant 1000.

2.52.1.28 Test Call CDR Customization and Display Changes

CDR customization and historical display of Test Calls has been updated as follows:

- Test Call CDRs are now customized in the existing SBC CDR Format table instead of in the Test Call CDR Format table (Troubleshoot menu > Troubleshoot tab > Call Detail Record folder > Test CDR Format), which is now obsolete.
- Historical Test Call CDRs are now displayed in the existing SBC CDR History table instead of in the Test Call CDR History table (Monitor menu > Monitor tab > VoIP Status folder > Test Call CDR History), which is now obsolete. To differentiate between SBC and Test calls, the 'Endpoint Type' parameter has been added to the table.

Applicable Application: All.

Applicable Products: All.

2.52.1.29 New Customizable CDR Field for Call Success

CDRs generated and sent by the device can be customized to include a new optional field, "Call Success" [447], which indicates whether the call succeeded ("yes") or failed ("no"). Customization is done in the existing SBC CDR Format table or Gateway CDR Format table.

Applicable Application: All.

Applicable Products: All.

2.52.1.30 New Customizable CDR Field for Multiple Media Types

CDRs generated and sent by the device can be customized to include a new optional field, "Media List" [819], which lists all the media types (e.g., "audio", "video" and "text") that were used during the call. This applies only to SBC signaling CDRs and for "CALL_END" Report Types (sent after a SIP BYE).

Applicable Application: SBC.

Applicable Products: All.

2.52.1.31 Call-End CDR Features

The following new features have been introduced for CDRs that are sent at the end of a call (SIP BYE):

- The device can be enabled to not send Call-End CDRs for calls of zero (0) duration:
  - Web interface: Call-End CDR Zero Duration Filter
  - ini: CallEndCDRZeroDurationFilter

- CLI: call-end-cdr-zero-duration-filter

- The device can be configured to not send Call-End CDRs if a specific SIP release cause(s) is received (comma-separated list from 300 through to 699; supports "xx" to denote decimal range such as 3xx):
  - Web interface: Call-End CDR SIP Release Reasons Filter
  - ini: CallEndCDRSIPReasonsFilter
  - CLI: call-end-cdr-sip-reasons-filter

Applicable Application: All.

Applicable Products: All.

2.52.1.32 Minimum Severity Level in Syslog Messages

Syslog messages generated by the device can now be configured to include only messages from a specific severity level (minimum) and higher. Severity levels include (from highest to lowest severity): Fatal, Alert, Critical, Error, Warning, Notice, Informational, and Debug. The feature is configured by the following new parameter:

- Web: 'Log level' (Logging Settings page)
- CLI: log-level
- ini File: SyslogLogLevel

Applicable Application: All.

Applicable Products: All.

2.52.1.33 Enhanced CPU Overload Details in Syslog Messages

Syslog messages sent when the device detects a CPU overload now includes more detailed information (processes and tasks), which can help identify the cause of the overload. When the device detects a CPU overload, it sends a Syslog message every 10 seconds until the device returns to normal state.

Applicable Application: SBC.

Applicable Products: All.

2.52.1.34 Consolidation of Log-Related Parameters

The Syslog Settings page (Troubleshoot menu > Troubleshoot tab > Logging folder > Syslog Settings) has been removed and all the parameters that were on this page have been moved to the Logging Settings page (Troubleshoot menu > Troubleshoot tab > Logging folder > Logging Settings).

Applicable Application: All.

Applicable Products: All.

2.52.2 Known Constraints

This section lists known constraints.

Table 2-68: Known Constraints in Version 7.20A.252.011

Incident	Description
SBC-11649	SCTP transport type is not supported in this version. **Applicable Products:** All.
SBC-12469	If the device sends an INVITE without an SDP body, it cannot handle the receipt of multiple SIP 18x responses for call forking. **Applicable Products:** All.
SBC-9958	Due to improved adaptive memory configuration (SPD), the device performs a double reset when the device is software upgraded for the first time. **Applicable Products:** Mediant 500C; Mediant 800C.

	Upgrading to Version 7.20A.252 from any version earlier than 7.20A.202 fails. This is due to the increased software version file (.cmp) size of Ver. 7.20A.252 due to various new features.
	To upgrade from a version earlier than 7.20A.252, the device must be loaded with an ini file (through the Auxiliary Files page) containing the following, which enables the device to handle the large .cmp file size:
	`BSPMAXCMPFILESIZE = 180`
	`initialshellcommand = 'HideAndNotBurn; RunOnTheFly; ForceRunAll; */ResetWebServer RESET'`
	Applicable Products: Mediant 90xx; Mediant Software.

SBC-10191	For the Test Call feature, there is backward incompatibility when upgrading to Version 7.20A.252: ▪ Routing of test calls according to the Tel-to-IP Routing table is no longer a configurable option (**Tel-to-IP** value has been removed from the 'Route By' parameter). After upgrading the device, this field needs to be re-configured to one of the remaining options (**IP Group** or **Dest Address**). ▪ Coder choice, played DTMF method (RFC 2833 or In Band), and SRTP are now configurable for Test Calls. After upgrading the device, these need to be configured. **Applicable Products:** All.
SBC-13226	When many SRDs are configured, the colors of the SRDs (as well as the #IDs in the SRD names) are no longer displayed in the Web interface. **Applicable Products:** All.
SBC-13306	For the Media Transcoding Cluster (MTC) feature, despite HA synchronization because of a switchover, operations on the MTC can still be performed through the Web interface (instead of being blocked). For example, software upgrade (.cmp) can still be initiated during this phase (although the system may fail). **Applicable Products:** Mediant 90xx; Mediant Software.
SBC-13319	If the redundant device in an HA system fails (e.g., due to invalid configuration) and you load an ini file to it containing the original HA settings to configure it as the redundant device, it may result in an active-active scenario without network communication between the devices. A workaround to this issue: prior to loading the ini file, access the redundant device (Web interface or CLI) and configure the 'HA Remote Address' parameter to the Maintenance IP address of the active device. **Applicable Products:** HA.

Incident	Description
-	The Media Transcoding Cluster (MTC) feature is not supported in this release. **Applicable Products:** Mediant 90xx; Mediant Software.

2.52.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-69: Resolved Constraints in Version 7.20A.252.011

Incident	Description
152338 /	The Intelligent Platform Management (IPMI) chassis indicators (i.e., status of fans, chassis temperature and power supply) are currently unavailable from the device's management interfaces. However, these indicators can be viewed directly from the Integrated Lights Out (iLO 5) interface (Web, SNMP or REST). For more information on the support, see the feature description for Mediant 9000 in Section 2.26.1.18 on page 167. **Applicable Products:** Mediant 9000 Rev. B; Mediant 9030; Mediant 9080.
SBC-8631 (VI#154056)	Loading a CLI Script file through the Web interface automatically resets the device. (Now it can be done with or without a reset.) **Applicable Products:** All.
SBC-9329 (VI#155107)	If the device receives early media (18x + SDP) for an SBC call that doesn't support video (port in the 'm=video' line is '0'), then upon a 200 OK, the device erroneously sends a 200 OK with the same port for media and video. **Applicable Products:** All.
SBC-9696 (VI#155708)	The Web interface's SBC Configuration Wizard doesn’t save the WAN NAT IP address. **Applicable Products:** All.
SBC-9925	The device's management interfaces display the wrong speed for the Ethernet ports when the NIC is 20 Gbps. **Applicable Products:** All.
SBC-10459	The device reports incorrect packet loss (PL) statistics to OVOC in a specific scenario (upon receiving incorrect report or RTCP from remote side). **Applicable Products:** All.
SBC-10676	The device sends alarms relating to a high CPU overload after upgrading to Version 7.20A.204 **Applicable Products:** All.
SBC-10719	In response to PRACK, the device sends a SIP 200 OK with a Contact header (but according to RFC 3262, it should not add the Contact header). **Applicable Products:** All.
SBC-10861	The device sends an alarm indicating ARM connection failure if it reconnects successfully to ARM. **Applicable Products:** All.
SBC-10882	The HTTP proxy settings in the downloaded CLI script is listed in the wrong order and therefore, the file cannot be loaded to the device. **Applicable Products:** All.

Incident	Description
SBC-11317	Downloading the device's Configuration Package file through SFTP (with keys) fails. **Applicable Products:** All.
SBC-11610 / SBC-12135	When the device experiences a CPU overload, it resets. **Applicable Products:** All.
SBC-11744	The device's TCP port 2424 is opened by default, creating a security breach. **Applicable Products:** All.
SBC-11933	The device tries allocating DSPs for call transfer (which it shouldn't) and the transfer fails. **Applicable Products:** All.
SBC-12011	The Web interface erroneously displays the error message "Error getting peer address" for the HA Network Monitor feature **Applicable Products:** All.
SBC-12109	The device's Pre-Parsing Message Manipulation causes a corrupted header (race condition that creates memory overrun). **Applicable Products:** All.
SBC-12265	The device's Web interface doesn't display the User Information table when the License Key is obtained from the Floating License. **Applicable Products:** All.
SBC-12283	The output of the CLI command **show voip cpu-stats** always displays 0%. **Applicable Products:** All.
SBC-12415	Debug Recording causes some syslog error messages "drPollIpFilterSocket: Socket recv failed with error". **Applicable Products:** All.
SBC-12448	The device doesn't trigger the session timer on specific call flows (refresher sends re-INVITE \ UPDATE without Session-Expires header). As a result, the call is disconnected. **Applicable Products:** All.
SBC-12572	The device stops sending SIP REGISTER messages to the IP side when registration is by Trunk Group and the Trunk Group Settings is modified. As a result, registration fails. **Applicable Products:** Gateway.
SBC-12591	When the device applies a configured Message Manipulation rule that removes the Request-URI header to a call, it resets. **Applicable Products:** All.
SBC-12745	Fax transcoding fails (T.38 negotiation) for SBC calls. **Applicable Products:** All.

Incident	Description
SBC-13021	Modifying the name of the Maintenance network interface in the IP Interfaces table causes HA to fail. **Applicable Products:** HA.
SBC-13190	When running both Debug Recording and 30,000 concurrent calls, the device runs out of memory buffers and as a result, an HA switchover occurs. **Applicable Products:** Mediant 9000 HA.
SBC-13229	When the Admin user creates a Call Admission Control Profile, the same Admin user cannot edit this profile after a device reset. **Applicable Products:** All.

2.53 Patch Version 7.20A.252.023

This patch version includes new features.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1116.

2.53.1 New Features

This section describes the new features introduced in this version.

2.53.1.1 HA Support for Mediant CE for Microsoft Azure Deployments

Mediant Cloud Edition (CE) can now be deployed as a High Availability (HA) system on the Microsoft Azure cloud platform.

Applicable Application: SBC.

Applicable Products: Mediant CE.

2.54 Patch Version 7.20A.252.261

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1132 and EMS/SEM Version 7.2.3113.

2.54.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-70: Resolved Constraints in Version 7.20A.252.261

Incident	Description
SBC-11142	The device's Linux kernel is vulnerable to security attacks (CVE-2018-5391). **Applicable Products:** All.
SBC-12989	When RTP Transparent and DTMF RFC 2833 use the same payload type, the device doesn't transfer the RTP packets. **Applicable Products:** Gateway.
SBC-13168	For WebRTC-to-WebRTC calls, the device handles the SDP's 'ssrc' attributes incorrectly (should send all 'ssrc' attributes between sides, while SIP-to-WebRTC calls should remove all 'ssrc' attributes). **Applicable Products:** All.
SBC-13251	Access to the device's Web Interface from another subnet is not possible. **Applicable Products:** All.
SBC-13267	An SBC call for Microsoft Teams Direct Routing with media bypass results in no media (no voice). **Applicable Products:** All.
SBC-13268	Modifying any row in the Ethernet Devices table (such as the 'Tagging' field) in an HA system causes an error ("SYS_HA: Offline Parameter PhysicalPortsTable was changed, in case HA is lost, HA reestablishment must be after system reboot"). As a result, HA mode fails. **Applicable Products:** HA.
SBC-13372 / SBC-13421	When the device handles QoE over T.38 (T.38 fax session while VQMON is enabled), in certain scenarios the device resets. **Applicable Products:** All.
SBC-13385	Tel-to-IP routing fails when configuring IP Group Index 0 and Proxy Set Index 0. **Applicable Products:** Gateway.
SBC-13420	Call forwarding fails for FXS interfaces (in Ver. 7.20.250). **Applicable Products:** Gateway.
SBC-13478	When using Call Setup Rules (CSR) with ENUM queries and the query target is defined, the device doesn't cache the resulting information and CSR fails. **Applicable Products:** All.

Incident	Description
SBC-13511 / SBC-13783	The EnableSingleDspTranscoding parameter is applicable only to Mediant 3000, however, it also appears in the Web interface of other products. **Applicable Products:** All.
SBC-13534	The LoginNeeded parameter restores to default after the device resets or a failover occurs. **Applicable Products:** All.
SBC-13617	For IP-to-Tel calls, the device doesn't send the redirect number received in the SIP History-Info header to the PSTN. **Applicable Products:** Gateway.
SBC-13621	DNS resolution for the HA Network Monitor feature is incorrect when the destination is configured as a hostname. **Applicable Products:** HA.
SBC-13623 / SBC-13678	The device's NGINX HTTP Proxy doesn't start after the device resets. **Applicable Products:** All.
SBC-13734	WebRTC client forking fails when ARM replies (to the device's GetRoute request) with a User-type IP Group – the call is not forked to all the users (Client Forking Mode configured to Parallel). **Applicable Products:** All.
SBC-13754 / SBC-14030	Graceful Lock doesn't function in Ver. 7.2.252. **Applicable Products:** All.
SBC-13756	The CPU Overload (in SPLB) alarm isn't cleared when the device's CPU utilization returns to normal. **Applicable Products:** All.
SBC-13772	Some passwords configured on the device are not masked (displayed in plain text) in the Web interface, exposing it to security vulnerability. **Applicable Products:** All.
SBC-13775	The device allows the loading of any file type through the Auxiliary Files page (Setup > Administration > Maintenance > Auxiliary Files), exposing it to security vulnerability. **Applicable Products:** All.
SBC-13150	The device resets upon receipt of a SIP INVITE message containing the Replaces header in specific scenarios: 1) device sends call to A, 2) A sends a REFER to connect with B, 3) after connection, C sends INVITE with Replaces to replace B. **Applicable Products:** All.

2.55 Patch Version 7.20A.252.269

This patch version includes resolved constraints only.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1133 and EMS/SEM Version 7.2.3113.

2.55.1 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-71: Resolved Constraints in Version 7.20A.252.269

Incident	Description
SBC-13125	During an SBC SIP session with voice and video, after repeatedly turning the video off and on, the video is transmitted with a delay of about 2-3 seconds. **Applicable Products:** All.
SBC-13139	For an outgoing call that has an established connection, when a re-NVITE is sent with a different DTLS key, the connection results in one-way voice and cannot be restored. **Applicable Products:** All.
SBC-13616	In a WebRTC-to-WebRTC SBC call, after a re-INVITE the following Syslog is generated "DTLSContext(#242)::StartHandshake" and voice is lost. **Applicable Products:** All.
SBC-14262	The device resets due to memory overrun in the following scenario: when a high SIP call traffic load occurs, and the GWDebugLevel parameter is configured to 5, and DR captures all Syslog packets. **Applicable Products:** All.

2.56 Patch Version 7.20A.254.202

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.1133 and EMS/SEM Version 7.2.3113.

2.56.1 New Features

This section describes the new features introduced in this version.

2.56.1.1 SIPRec for Audio-Video Calls

The device's SIPRec feature now also supports the recording of video streams for audio-video calls. Up until now, the SIPRec feature recorded only audio streams. This feature can be used when the device (Session Recording Client or SRC) interacts with third-party Session Recording Servers (SRS) that support video streams recording.

The new parameter 'Video Recording Sync Timeout' (VideoRecordingSyncTimeout) has been introduced for video recording synchronization (media port negotiation). If the SRS doesn't send the SIP 200 OK within this timeout period, the device connects the video stream between the UAs, and then starts sending recorded video and audio streams to the SRS.

Note: For audio-video calls, video recording requires additional SBC media channel resources and therefore, the purchased License Key needs to accommodate for this.

Applicable Application: SBC.

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 90xx; Mediant Software.

2.56.1.2 SIP Signaling over SCTP Transport

The device now supports the Stream Control Transmission Protocol (RFC 4960 "Stream Control Transmission Protocol") transport protocol, which provides multi-homing capabilities, preventing single points of failure. This is used for SIP signaling (SIP over SCTP) in accordance with RFC 4168 (The Stream Control Transfer Protocol (SCTP) as a Transport for SIP).

SCTP multi-homing provides redundancy capabilities which maintains all SIP sessions in case of network failure between the device and the remote SIP proxy. The device can support up to two local IP addresses (IP Interfaces) per SIP Interface and multiple IP addresses on the remote proxy. Each SIP Interface creates its own SCTP association, which can include multiple paths between the device and proxy.

For this feature, the following new configuration parameters and optional values have been added:

- SIP Interfaces table:
  - 'SCTP Port' parameter (currently, Web only) - defines the port on which the device listens for inbound SCTP connections.
  - SCTP Secondary Network Interface' parameter (currently, Web only) - assigns an additional local address or IP Interface (from the IP Interfaces table) for multi-homing. The primary IP Interface for multi-homing is configured for the SIP Interface by the existing 'Network Interface' parameter.
- Proxy Address table (child of Proxy Sets table):
  - **SCTP** optional value for the 'Transport Type' parameter.

- For multi-homing support, multiple remote proxy IP addresses can be configured for the same Proxy Set. When at least one IP address in a Proxy Set is configured to use the SCTP transport type, the device assumes that all the IP addresses in the same Proxy Set are a set of multi-homing remote addresses for a single proxy. In this case, all the IP addresses in the Proxy Set must be configured with the SCTP transport type and with the same remote SCTP port number.

- SCTP has been added as an optional value for the following parameters:
  - Classification table: 'Source Transport Type' (Classification_SrcTransportType)
  - IP-to-IP Routing table: 'Destination Transport Type' (IP2IPRouting_DestTransportType)
  - Test Call Rules table: 'Destination Transport Type' (Test_Call_DestTransportType)
- SCTP parameters (Web and CLI) according to the SCTP RFC:
  - sctpHeartbeatInterval (heartbeat-interval): Defines the SCTP heartbeat interval
  - sctpInitialRTO (initial-rto): Defines the Initial retransmission timeout (RTO)
  - sctpMinimumRTO (minimum-rto): Defines the Minimum Retransmission Timeout (RTO)
  - sctpMaximumRTO (maximum-rto): Defines the maximum retransmission timeout (RTO)
  - sctpMaxPathRetransmit (max-path-retransmit): Defines the maximum path retransmissions per address
  - sctpMaxAssociationRetransmit (max-association-retransmit): Defines the maximum association retransmit
  - sctpMaxDataTxBurst (max-data-tx-burst): Defines Maximum number of DATA chunks that can be transmitted at one time – default 4 packets
  - sctpMaxDataChunksBeforeSACK (max-data-chunks-before-sack): Defines SACK sent after number of received packets
  - sctpTimeoutBeforeSACK (timeout-before-sack): Defines SACK sent after timeout since the received packet
- (CLI) show sctp connections – displays local SCTP endpoint and SCTP associations
- (CLI) show sctp statistics – displays statistics of all SCTP associations

Applicable Application: SBC.

Applicable Products: Mediant 90xx; Mediant Software.

2.56.1.3 Message Session Relay Protocol Support (MSRP)

The device now supports Message Session Relay Protocol (MSRP), which is a text-based protocol for exchanging a series of related instant messages (IM) across an IP network (TCP/TLS only) in the context of a session. The protocol can also be used to transfer large files or images or sharing remote desktops or whiteboards. MSRP is typically used for Next Generation 911 (NG911) services, allowing 911 callers to not only access 911 services through voice calls, but also through text messages with Public Safety Answering Points (PSAPs). MSRP support is in accordance with RFC 4975 (The Message Session Relay Protocol (MSRP)) and RFC 6135 (An Alternative Connection Model for the Message Session Relay Protocol (MSRP)).

The device establishes MSRP sessions using the SDP offer/answer negotiation model over SIP. The MSRP session starts with a SIP INVITE and ends with a SIP BYE message. As a B2BUA, the device interoperates between the MSRP endpoints, terminating the incoming MSRP message on the inbound leg and then generating a new MSRP message on the outbound leg. Before sending the INVITE, the device manipulates the SDP body (e.g., 'a=path', 'c=', 'm=', 'a=setup' and 'a=fingerprint' lines). The device can perform optional

message manipulation and other translations such as resolving NAT traversal issues when the endpoints or device are located behind NAT. The device also supports secure MSRP sessions (MSRPS), using TLS certificates (TLS Context).

Below shows an example of an MSRP message (*SEND* request):

```
MSRP a786hjs2 SEND
To-Path: msrp://biloxi.example.com:12763/kjhd37s2s20w2a;tcp
From-Path: msrp://atlanta.example.com:7654/jshA7weztas;tcp
Message-ID: 87652491
Byte-Range: 1-25/25
Content-Type: text/plain

Hey Bob, are you there?
-------a786hjs2$
```

This feature provides the following configuration updates:

- IP Profiles table:
  - 'MSRP Offer Setup Role' parameter (new): Configures the device's MSRP role mode in SDP negotiations ('a=setup' line).
  - 'Data DiffServ' parameter (new): Configures DiffServ marking of MSRP traffic and media of TCP traffic in the IP header's DSCP field.
  - 'MSRP re-INVITE/UPDATE' parameter (new): Configures if the destination UA participating in the MSRP session supports the receipt of re-INVITE requests and UPDATE messages.
  - 'SBC Media Security Mode' parameter (existing): Configures the transport protocol for the outgoing leg (**Secured** and **Both** for MSRPS; **Not Secured** for MSRP). The optional values of the parameter have been renamed -- **SRTP** to **Secured**; **RTP** to **Not Secured** -- to reflect that it's also for MSRP (not only SRTP).
  - 'MSRP Empty Message Format' parameter (new): Configures if the device adds a Content-Type header to the first empty (no body) MSRP message that is used in the connection.
- IP address and port of the 'path' attribute is determined by the Media Realm associated with the IP Group. The port is configured by the new Media Realm parameters 'TCP Port Range Start' (CpMediaRealm_TCPPortRangeStart) and 'TCP Port Range End' (CpMediaRealm_TCPPortRangeEnd), which enables a single Media Realm to serve both RTP and MSRP. As a result of these new parameters, the existing Media Realm parameters 'Port Range Start' and 'Port Range End' have been renamed ('UDP Port Range Start' and 'UDP Port Range End').
- For NAT traversal, the existing NAT Translation table is used (and the new IP address:port is used in the 'a=path' field).
- The IP Group's existing 'DTLS Context' parameter has been renamed (Web only) to 'Media TLS Context' to reflect that the parameter can also be used for MSRP sessions (not only DTLS).

CDRs generated by the device for MSRP calls include the value "msrp" for the CDR field, Media List (existing).

Applicable Application: SBC.

Applicable Products: All.

2.56.1.4 Automatic Topology Hiding of URI Host Part by IP Group's SIP Group Name

The device now provides an easy-to-configure method to replace the host part of the URI in outgoing SIP messages with the destination IP Group's 'SIP Group Name' parameter value. This feature enhances the device's capability for hiding the incoming network topology (i.e.,

URI) from the outbound IP Group. Up until now, Message Manipulation rules for each header were required to implement such topology hiding.

The SIP headers to which this feature can be applied is specified by the new IP Groups table parameter 'SIP Topology Hiding Headers List' (IPGroup_TopologyHidingHeaderList).

Note:

- If the 'SIP Group Name' field is not configured and the 'SIP Topology Hiding Headers List' parameter is configured, the device replaces the host part of the URI with the local IP address (IP Interface) associated with the IP Group.
- If Outbound Message Manipulation is also configured for the IP Group, it's applied only after this topology hiding feature is applied.
- The 'SIP Topology Hiding Headers List' parameter doesn't change the default behavior of the 'SIP Group Name' (i.e., replaces the host part of the Request-URI and To headers in outbound messages and replaces the host part in the From header in inbound messages).

Applicable Application: SBC.

Applicable Products: All.

2.56.1.5 Enhanced SIP PRACK Handling

The device's SIP PRACK handling, which is configured by the existing IP Profile parameter 'PRACK Mode' (IpProfile_SbcPrackMode) has been enhanced with the following additional optional values:

- [0] **Disabled**: Depending on scenario, the device either disables PRACK with the SIP User Agent (UA) or rejects the call.
- [4] **Optional With Adaptations**: Optimized PRACK handling, which is based on the presence of PRACK-related SIP headers and parameters ('Require:100rel' or 'Supported: 100rel') as well as the presence of SIP message bodies (e.g., SDP) in 18x responses. This option may be useful, for example, to avoid PRACK congestion due to the device being flooded with 18x messages without a body.

For a detailed description, refer to the device's *User's Manual*.

Applicable Application: SBC.

Applicable Products: All.

2.56.1.6 Customizing CDR Call Success Indication Based on Responses

Call success indication in CDRs (using the optional 'Call Success' field - "yes" or "no"), which is based on call release (termination) reason (i.e., SIP response code or internal response generated by device) can be changed from default (customized). For example, by default, calls released with SIP 486 (Busy Here) responses are indicated in CDRs as call failure. By employing this feature, these SIP responses can be reported in CDRs as call success.

To support this feature, the following new parameters have been added (under Troubleshoot-> Call Detail Record -> Call Detail Record Settings) to override the device's default behavior for determining call success:

- SIP response codes:
  - 'Call Success SIP Reasons' [CallSuccessSIPReasons] - defines SIP responses for call success
  - 'Call Failure SIP Reasons' [CallFailureSIPReasons] - defines SIP responses for call failure
- Internal response codes:
  - 'Call Success Internal Reasons' [CallSuccessInternalReasons] - defines internal responses for call success

- 'Call Failure Internal Reasons' [CallFailureInternalReasons] - defines internal responses for call failure

This feature also allows customization of call status (success or failure) for the following special internal responses before or after call connect (SIP 200 OK):

- "GWAPP_NO_USER_RESPONDING" (18):
  - 'No User Response before Connect' [NoUserResponseBeforeConnectSuccess] - defines call status when the response is received before call connect
  - 'No User Response after Connect' [NoUserResponseAfterConnectSuccess]- defines call status when the response is received after call connect
- "RELEASE_BECAUSE_CALL_TRANSFERRED" (807):
  - 'Call Transferred before Connect' [CallTransferredBeforeConnectSuccess] - defines call status when the response is received before call connect
  - 'Call Transferred after Connect' [CallTransferredAfterConnectSuccess] - defines call status when the response is received after call connect

The valid value of these parameters is the SIP response code number (e.g., 486). Multiple responses can be configured, whereby each code is separated by a comma (e.g., 486,408,406). A range of responses can also be configured using the "xx" wildcard (e.g., 4xx).

Applicable Application: All.

Applicable Products: All.

2.56.1.7 Alternative Routing Based on SIP Responses per IP Group

Alternative call routing based on SIP response codes can now be configured per IP Group. Up until now, this was configured globally (for all calls), using the Alternative Routing Reasons table. Now, multiple SIP response codes for alternative routing can be configured and grouped under an Alternative Reasons Set and then assigned to a specific IP Group.

To support the feature, configuration has been updated as follows:

- The Alternative Routing Reasons table has been replaced with the following parent-child table:
  - Alternative Reasons Set (parent) – defines a name for the group of SIP response codes
  - Alternative Reasons Rules (child) – defines the SIP response codes for the Alternative Reasons Set to trigger alternative routing
- The new parameter 'SBC Alternative Routing Reasons Set' has been added to the IP Groups table to assign the Alternative Reasons Set to an IP Group.

Applicable Application: SBC.

Applicable Products: All.

2.56.1.8 Enhanced IPMI Indication for Fan and CPU Temperature Alarms

The device's Intelligent Platform Management (IPMI or iLO) chassis indicators for fan status and CPU temperature has been enhanced as follows:

- Fan status: Up until now, the SNMP alarm acFanTrayAlarm indicated a failure for the entire Fan Tray module (Major severity). Now, it's also sent to indicate failures per fan (removed or faulty). For example, if a failure occurs in fan 3, the alarm is sent ("Fan-Tray Alarm. Fan 3 is faulty"). If a failure then occurs in fan 4 as well, the first alarm is cleared and a new alarm is sent indicating failures in fans 3 and 4 ("Fan-Tray Alarm. Fans 3,4 are faulty"). If fans 3 and 4 return to normal operation, the alarm is cleared.
- Temperature status: Up until now, the SNMP alarm acBoardTemperatureAlarm was sent only when the overall temperature of the CPU exceeded a specific threshold (configured by the HighTemperatureThreshold parameter). Now, it's sent per temperature sensor. For example, if the temperature threshold exceeds at sensor 1,

the alarm is sent ("Board Temperature Alarm: Sensor #1 is 88 degrees celsius. Exceeded threshold of 70"). If the temperature threshold at sensor 2 then exceeds as well, the first alarm is cleared and a new alarm is sent indicating exceeded temperature thresholds at both sensors ("Board Temperature Alarm: Sensors #1,#2 are 88,90 degrees celsius. Exceeded threshold of 70").

Applicable Application: SBC.

Applicable Products: Mediant 9000 Rev. B; Mediant 9030; Mediant 9080.

2.56.1.9 VMware Tools Version Update

VMware Tools is deployed as part of the SBC image for the VMware virtualization platform. VMware Tools has been updated from Version 9.4.10 to Version 10.3.5.

Applicable Application: SBC.

Applicable Products: Mediant VE; Mediant CE.

2.56.1.10 Enhanced System Snapshot Features

The device's System Snapshot feature has been enhanced as follows:

- For HA systems, renaming a System Snapshot on the active device synchronizes with the redundant device and renames the corresponding System Snapshot on the redundant device. (Note that if the new name is not unique on the active and redundant devices, the renaming operation fails).
- The device generates a Syslog error message when the user tries to create a System Snapshot and there is insufficient memory on the device to store it ("SSM: CreateSystemSnapshot(): Failed to create snapshot (name=<Snapshot Name>, rc=<ERR_CODE>)"). Up until now, the System Snapshot was created even though it was corrupted.

Applicable Application: SBC.

Applicable Products: Mediant 90xx; Mediant Software.

2.56.1.11 Additional User Activity Details in Activity Log and Syslog

Additional details regarding management user activity through the device's management interfaces have been added to the Activity Log and Syslog. These additional details improve security and tracing capabilities regarding the actions taken by users. Note that the format of these messages in the Activity Log and Syslog were changed accordingly. These additional activities that are reported include the following:

- User attempts to log in with incorrect username or password.
- Blocked or inactive (new or inactive user) user attempts to log in.
- Session limit exists when user attempts to log in.
- User's access level is changed (e.g., Monitor to Administrator).
- Added or deleted user (in the Local Users table).

Applicable Application: All.

Applicable Products: All.

2.56.1.12 DNS Rebinding Protection

The device now provides protection against DNS rebinding attacks. This may occur when management users access the device using its hostname (configured by the existing parameter HostName) instead of the IP address. The feature is enabled by the new parameter 'DNS Rebinding Protection Enabled'.

Applicable Application: All.

Applicable Products: All.

2.56.1.13 IPv6 Addresses for IP Traces in Logging Filters Table

Logging filter configuration for IP traces now supports IPv6 source and destination addresses. Up until now, only IPv4 addresses were supported.

The IPv6 addresses are configured by the existing 'Value' parameter in the Logging Filters table, when the 'Filter Type' parameter is configured to **IP Trace**. The addresses are configured using the new keywords "ipv6.src", "ipv6.dst" and "ipv6.addr". Examples are shown below:

- "ipv6.addr==2001:0db8:85a3:0000:0000:8a2e:0370:7334"
- "ipv6.src==2001:db8:abcd:0012::0/64" (where /64 is the prefix length)

For IPv4, the existing keywords are used ("ip.src", "ip.dst" and "ip.addr").

Applicable Application: All.

Applicable Products: All.

2.56.1.14 Hidden Password when Configuring Users through CLI

When configuring a management user for the device in the Local Users table through CLI, the user's password can be concealed (hidden) when typing it in. The feature is supported by pressing the Enter key intermediately after typing the existing `password` command:

```
(config-system)# user john
Configure new user john
(user-john)# password
Please enter hidden password (press CTRL+C to exit):
```

Applicable Application: All.

Applicable Products: All.

2.56.1.15 Direct Media Calls Automatically Disabled for SIPRec

When the device needs to record calls that are configured for direct media or media bypass (i.e., media stream doesn't traverse the device), it automatically disables direct media for these calls (during their setup). This ensures that the media passes through the device so that it can be recorded and sent to the SRS.

Note: This feature doesn't apply if direct media is enabled using the global parameter SBCDirectMedia (i.e., for all calls). In this scenario, direct media is maintained and SIP recording is not done on these calls. The feature is applicable only if direct media is enabled per specific calls using, for example, IP Profiles ('Direct Media Tag' parameter) or SIP Interfaces ('Direct Media' parameter).

Applicable Application: SBC.

Applicable Products: Mediant 500; Mediant 500L; Mediant 800; Mediant 2600; Mediant 4000; Mediant 90xx; Mediant Software.

2.56.1.16 Max. RADIUS-Accounting Attributes for CDR Customization

The maximum number of RADIUS Accounting attributes that can be customized (and sent) in the CDR generated by the device has been increased from 70 to 128.

Applicable Application: SBC.

Applicable Products: Mediant 2600; Mediant 4000.

2.56.1.17 Configured Hostname Exposed to Hypervisor

If the device is configured with a hostname (using the existing 'Host Name' parameter), the hostname is now also "exposed" to the VMware vSphere hypervisor on which the SBC is deployed.

Applicable Application: SBC.

Applicable Products: Mediant 9xx; Mediant Software.

2.56.2 Known Constraints

This section lists known constraints.

Table 2-72: Known Constraints in Version 7.20A.254.202

Incident	Description
SBC-15055	On the Mediant CE, only one snapshot can be added (in addition to the default). New files can be uploaded only with one snapshot, in addition to the default. **Applicable Products:** All Mediant CE
-	SCTP is currently not supported by the products listed below. Please contact your AudioCodes' representative for more information. **Applicable Products:** MP-1288; Mediant 500; Mediant 500L; Mediant 800; Mediant 1000; Mediant 2600; Mediant 4000
-	Even though the products listed below don't support SCTP, the **SCTP** optional value appears for the following parameters: Classification_SrcTransportType; IP2IPRouting_DestTransportType; Test_Call_DestTransportType. **Applicable Products:** All (except Mediant 90xx; Mediant Software)
-	Even though the products listed below don't support SCTP, the **SCTP** optional value appears for the 'Transport Type' parameter in the Proxy Address table. **Applicable Products:** All (except Mediant 90xx; Mediant Software)
-	Even though the products listed below don't support SCTP, the CLI `sctp-port` ('SCTP Port') parameter appears in the SIP Interfaces table. **Applicable Products:** All (except Mediant 90xx; Mediant Software)
-	Even though the products listed below don't support SCTP, the CLI `sctp-second-network-interface` ('Secondary Network Interface') parameter appears in the SIP Interfaces table. **Applicable Products:** All (except Mediant 90xx; Mediant Software)

2.56.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-73: Resolved Constraints in Version 7.20A.254.202

Incident	Description
SBC-10026	The Message Conditions table configuration is lost after a device reset (or HA switchover) because of the comma (,) symbol in its name. **Applicable Products:** All
SBC-11727	A warning message on the virtual host occurs. The configured guest OS (Red Hat Enterprise Linux 6 (64-bit)) for this virtual machine does not match the guest that is currently running (CentOS 4/5/6/7 (64-bit)). **Applicable Products:** Mediant Software
SBC-12526	The device's TLS Contexts don't offer the chain of trust in Server Hello responses. **Applicable Products:** All

Incident	Description
SBC-12758	For SBC calls, the device offers an "extended" coder with the same dynamic payload type as the existing coder, resulting in a faulty SDP offer. **Applicable Products:** All
SBC-13043	The device sends a DNS query as an A-Record instead of an SRV. **Applicable Products:** All
SBC-13384	The ini file that is extracted from the configuration_package.tar.gz file through SFTP is corrupt. **Applicable Products:** All
SBC-13437	The device is not able to configure the virtual machine host name (default is localhost.localdomain). **Applicable Products:** Mediant VE (VMware Sphere)
SBC-13622	The HA system experiences a mismatch in Proxy IP addresses between active and redundant devices after an HA switchover. **Applicable Products:** HA
SBC-13638	The device's Syslog warning message "invalid payload type" doesn't include the source IP address. **Applicable Products:** All
SBC-13642	When using DTLS without STUN, the device sends DTLS Client Hello packets with the destination MAC containing only zeros, causing a DTLS delay. **Applicable Products:** All
SBC-13757	The device sends invalid SRTP packets after upgrade to Ver. 7.2.252. **Applicable Products:** All.
SBC-13770	The device is exposed to security vulnerability (XML Entity Expansion Injection), as it accepts document type definition from untrusted sources. **Applicable Products:** All.
SBC-13771	The device is exposed to security vulnerability (unrestricted file upload), as it loads any type of file from the Auxiliary Files page. **Applicable Products:** All.
SBC-13773	The device is exposed to a security vulnerability (Page after login has been cached locally), as its Web interface can be accessed after login without authentication. **Applicable Products:** All.
SBC-13774	The device is exposed to a security vulnerability (form validation has been turned off), as its missing HTML5. **Applicable Products:** All.
SBC-13776	The device is exposed to a security vulnerability (X-Content-Type-Options is not specified), as the X-Content-Type-Options is missing. **Applicable Products:** All.
SBC-13812	The device limits CDR types that are for RADIUS SBC to 70 (instead of 128). **Applicable Products:** Mediant 2600.
SBC-13899	The device provides only 100 resources for RTCP-XR PUBLISH messages, which is insufficient for high traffic loads. **Applicable Products:** All.

Incident	Description
SBC-13942 / SBC-14546	If the device receives an SDP offer with ICE and the IP Profile parameter SBCIceMode is disabled, it doesn't update the remote IP address, causing no voice. **Applicable Products:** All.
SBC-13951	The device sends CDR reports (Media_End) to the CDR server even if it's configured to 0.0.0.0. **Applicable Products:** All.
SBC-13967 / SBC-14221 / SBC-14541	When the Proxy Set is modified, the device resets, which is caused by a memory overrun. **Applicable Products:** All.
SBC-13982	The device shows Syslog messages in debug recording (DR) even though it is configured to include SIP only (Logging Filters table 'Log Type' parameter configured to SIP Only). **Applicable Products:** All.
SBC-13991	The device's Web interface displays the wrong number of active calls, due to incorrect calculation of alternative routing. **Applicable Products:** Gateway.
SBC-14024	The device experiences no voice when the IP address of the IP Interface is changed. **Applicable Products:** All
SBC-14057	The device resets with the exception info of TASK PMON, which is caused by a dereference of a null pointer. **Applicable Products:** All
SBC-14112	One-way voice occurs due to a specific call transfer scenario **Applicable Products:** All
SBC-14142 / SBC-14478	The device resets when receiving a SIP 200 OK with 'a=inactive' for WebRTC and SIPRec calls. **Applicable Products:** All
SBC-14223	The device doesn't show the Caller ID name, when Caller ID is enabled for IP-to-Tel calls. **Applicable Products:** MP-1288
SBC-14234	A TLSSocket issue causes the device to be in a state of "connection not established", stopping the device from reporting the Floating License Pool to OVOC. **Applicable Products:** All
SBC-14300 / SBC-14501	If the device receives an SDP answer with ICE lite or no ICE, the channel mode isn't updated and thus, remains in ICE mode. According to RFC, in such a scenario it should be in no-ICE mode. As a result, no voice occurs. **Applicable Products:** All
SBC-14313	The device rejects STUN with high priority after a re-INVITE, generating the error message "STUN_ATTRIBUTE_USERNAME failed". As a result, no voice occurs. **Applicable Products:** All
SBC-14323	The device forwards a second identical 183 with the wrong SDP. **Applicable Products:** All

Incident	Description
SBC-14401	The device tries to extend coders even though it has no transcoding capabilities. As result, the call fails. **Applicable Products:** All
SBC-14417	Logging into the device's CLI (USB, SSH, Telnet, Web) triggers a buffer overflow, which causes a reset (i.e. denial of service attack) and potentially allows a break-in into the underlying operating system and a full takeover of the CPU. **Applicable Products:** All
SBC-14426	The Gateway CDR Format table doesn't appear in the Web interface (appears as Test Call CDR Format table). **Applicable Products:** Gateway
SBC-14499	The device disconnects from ARM upon the receipt of a 504 reply from OVOC. **Applicable Products:** All
SBC-14552	The device's Call Forward activation and deactivation fails for calls, as the device doesn’t release the Gateway call towards the IP. **Applicable Products:** Gateway
SBC-14650	The device's WebRTC drops the video stream from calls due to different SSRCs. **Applicable Products:** All
SBC-14664	SIP parsing errors occur due to record route. **Applicable Products:** All
SBC-14689	Prefix length 120 is not supported for IPv6 network interfaces. **Applicable Products:** All

2.57 Patch Version 7.20A.254.375

This patch version includes new features, known constraints and resolved constraints.

Note: This patch version is compatible with AudioCodes One Voice Operations Center (OVOC) Version 7.6.2125 and EMS/SEM Version 7.2.3113.

2.57.1 New Features

This section describes the new features introduced in this version.

2.57.1.1 Mediant CE Deployable on Google Cloud Platform

Mediant Cloud Edition (CE) SBC can now be deployed on the Google Cloud platform.

Applicable Application: SBC.

Applicable Products: Mediant CE.

2.57.1.2 Microsoft Teams License Included in Evaluation License Key

The device's evaluation License Key (three free SBC sessions) now also includes the license for Microsoft Teams Direct Routing ("TEAMS").

Note: For post-evaluation deployment, this evaluation License Key cannot be used. Instead, a new License Key must be purchased with the required SBC sessions and features, including the "TEAMS" license for proper operation with Teams Direct Routing.

Applicable Application: SBC.

Applicable Products: Mediant Software.

2.57.1.3 License Keys for Microsoft Teams

When operating in a Microsoft Teams environment (for example, Phone System Direct Routing), both the Teams license ("TEAMS") and the general Microsoft license ("MSFT") must be present in the installed License Key. Up until now, only the "TEAMS" license was required.

Note: For all products not listed below under Applicable Products, the "MSFT" license is provided by default in the License Key.

Applicable Application: SBC.

Applicable Products: Mediant 500L; Mediant 500.

2.57.1.4 Registration Status Updates with ARM and Third-party Routing Server

The device can notify AudioCodes ARM or third-party Routing servers of all the SIP user agents (endpoints) that are registered with the device. It does this by periodically synchronizing its registration database with the Routing server to keep it up to date, enabling the Routing server to use this information to perform correct and optimal routing decisions based on user registration.

The feature is enabled by the new parameter 'Routing Server Registration Status'. AudioCodes REST API also supports GET, PUT and POST actions for this parameter from a REST client, using the following new REST URL path: /api/v1/rmConfig/globals/routingServerRegistrationStatus

In addition to enabling the feature, the 'Type' parameter of the Remote Web Service (i.e., Routing server) in the Remote Web Services table must be configured to the new optional value **Registration Status**.

Applicable Application: All.

Applicable Products: All.

2.57.2 Known Constraints

This section lists known constraints.

Table 2-74: Known Constraints in Version 7.20A.254.375

Incident	Description
SBC-15254	For Mediant CE on Google Cloud, use of the Internal Network Load Balancer is not supported. **Applicable Products:** Mediant CE
SBC-15397	For devices in HA mode, log messages from the redundant device that are longer than 254 bytes, which are sent by the active device to syslog, are truncated to 254 bytes. **Applicable Products:** HA
SBC-15526	For recording (SIPRec) audio-video calls, the order of the 'm=' lines in the SDP body that the device sends to the SRS might be incorrect for specific call scenarios (m=video, m=audio, m=audio, and then m=video). As a result, some third-party SRS's may reject the SIPRec INVITE message. A workaround is to use the device's Message Manipulation feature to reorder the lines in the SDP. **Applicable Products:** All

2.57.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved.

Table 2-75: Resolved Constraints in Version 7.20A.254.375

Incident	Description
SBC-14561	Direct media (media bypass) calls disconnect when the SBC remains without any active Media Components (MCs). (Now, such calls don't get disconnected upon this scenario, as long as they don’t change to non-direct media calls.) **Applicable Products:** Mediant CE
SBC-14626	Upon an HA switchover, the device's STWR task gets stuck in the read Linux file, causing a CPU overload. **Applicable Products:** HA
SBC-14647	Changing the Web interface's Admin user password fails, and it can be changed only 24 hours after the previous password change. (Now, the default of the WebPassChangeInterval has been changed to 0.) **Applicable Products:** All
SBC-14682	The performance monitoring counter UnAvailable Seconds (UAS) displays (show voip interface e1-t1) incorrect values. **Applicable Products:** Gateway
SBC-14683	The device resets upon configuring debug syslog messages using the HTTPProxySyslogDebugLevel parameter. **Applicable Products:** All
SBC-14707	The device fails to authenticate re-INVITE requests of a rerouted call (alternative routing or REFER). **Applicable Products:** All

Incident	Description
SBC-14902	The device removes the SIP header, P-Asserted-Identity after the call is rerouted by ARM. **Applicable Products:** All
SBC-15131	The device doesn't increase the SDP's session ID upon a re-INVITE after switching from a direct-media call to non-direct media call. As a result, no voice occurs. **Applicable Products:** All
SBC-15198	When the device does a second HA switchover, the call that was established before the first switchover, disconnects upon hold/unhold scenario. **Applicable Products:** HA
SBC-15225	The device resets when the Web interface session opens, because of a memory issue. **Applicable Products:** All
SBC-15226	The device's CDR displays the wrong IP address of the remote side. **Applicable Products:** All
SBC-15227	The device replies with a SIP 488 to a 200 OK of a forked call if the original INVITE contains a Record-Route header. As a result, the call fails. **Applicable Products:** All
SBC-15240	The device fails to play the hold tone from the PRT file during an on-hold scenario. As a result, the call disconnects. **Applicable Products:** All
SBC-15244	The device is exposed to the CSFR (Cross-Site Request Forgery) security vulnerability. **Applicable Products:** All
SBC-15250	The device's ELIN number is limited to 12 digits. (Now, it has been increased to 16.) **Applicable Products:** All
SBC-15251	The device fails to ping its own IP address. **Applicable Products:** Mediant 5xx; Mediant 8xx; Mediant 1000B; MP-1288
SBC-15351	The device resets upon connecting to the Web interface. **Applicable Products:** All
SBC-15361	The device runs out of MDML resources on high SBC traffic when call duration is long (more than 30s). As a result, calls fail. **Applicable Products:** All
SBC-15401	The CLI command `hotline-dia-ltone-duration` has a typo (should be `hotline-dial-tone-duration`). **Applicable Products:** All UpdatedCLI-UM
SBC-15496	The device resets upon a call attempt to a full Trunk\Trunk Group when operating with ARM. **Applicable Products:** All (Gateway)

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3 MSBR Series

This chapter describes new features, known constraints and resolved constraints relating to data-router functionality of the Mediant MSBR product series.

3.1 Patch Version 7.20A.150.004

This is the initial version of the 7.2 Software Release for the MSBR product series.

Note:

- This version is based on MSBR 6.8 Version **6.80A.335.005**, released in March 2017. In other words, all data capabilities of this MSBR 7.2 version are fully aligned to the above mentioned 6.8 version (with a few exceptions, as listed in the document *MSBR Data Feature Additions from 6.8 to 7.2*).
- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ VDSL ISDN
  - √ VDSL POTS
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.
- This patch version is compatible with AudioCodes EMS/SEM Version 7.2.3083.

3.1.1 New Features

Please see the note in Section 3.1.

3.1.2 Known Constraints

This section lists known constraints.

Table 3-1: Known Constraints in Version 7.20A.150.004

Incident	Description
-	TR-181 is not supported. **Applicable Products:** MSBR.
143283	DHCPv6 NTP "Current Dynamic NTP Server" information is not displayed in the CLI when running the CLI command **show system ntp-status**. **Applicable Products:** MSBR.
143295	The CL command **debug reset-history** saves only the last three reset reasons. **Applicable Products:** MSBR.
144181	The device does not support 802.1X. **Applicable Products:** MSBR.
144214	The CLI command **debug capture data physical clear** is not supported. **Applicable Products:** MSBR.
144076	The CLI command **show data interfaces cellular 0/0** fails. **Applicable Products:** MSBR.
141108	Running speed tests through TR-069 is not supported.

3.1.3 Resolved Constraints

Please see the note in Section 3.1.

3.2 Patch Version 7.20A.154.025

This patch version includes new features and resolved constraints.

Note:

- This version is based on MSBR Version 6.80A.347.001 (released in July 2017). In other words, all data capabilities of this MSBR 7.2 version are fully aligned to the above mentioned 6.8 version (with a few exceptions, as listed in the document *MSBR Data Feature Additions from 6.8 to 7.2*).
- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ VDSL ISDN
  - √ VDSL POTS
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.2.1 New Features

New features introduced in this version include the following:

- Bidirectional Forwarding Detection (BFD) support for Open Shortest Path First (OSPF). The new command to enable BFD for an OSPF interface is as follows:

```
(config-if)# ip ospf bfd interval <Value> min_rx <Value>
multiplier <Value>
```

  where:

  - *interval*: Interval (in msec) for outgoing BFD messages. The interval is increased if required by remote system.
  - *min_rx*: Minimal interval (in msec) between BFD messages in milliseconds. The remote system uses this interval for sending messages if its interval is lower.
  - *multiplier*: Maximum number of packets that can be missed before the session status is considered down.
- Bidirectional Forwarding Detection (BFD) support for static routes. The new command

to enable BFD for a static route is as follows:

```
(config-data)# bfd neighbor <Neighbor ID> <IP Address>
<Interface ID> interval <Value> min_rx <Value> multiplier
<Value> [multihop]
```

where:

- *neighbor id:* (1-20) Neighbor identifier.
- *ip address:* Address of the remote BFD device.
- *interface id:* Name and number of the outgoing interface.
- *interval*: Interval (in msec) for outgoing BFD messages. The interval is increased if required by remote system.
- *min_rx*: Minimal interval (in msec) between BFD messages in milliseconds. The remote system uses this interval for sending messages if its interval is lower.
- *multiplier*: Maximum number of packets that can be missed before the session status is considered down.
- *multihop:* Set the neighbor to multihop mode in case the remote device is not on the local LAN
  The parameter **bfd-neighbor <neighbor ID>** was added to the **ip route** command:

```
(config-data)# ip route <Ip Address> <Ip Destination Mask>
[next-hop IP address] <Interface> <Interface ID> [<Metric
Value>] [track <Track Id>] [bfd-neighbor <Neighbor ID>]
[output-vrf <VRF ID>] [description <String>]
```

where:

- *bfd-neighbor:* Defines the ID of a BFD neighbor to attach the route to.

- Management ACL for TR-069 can now be configured, using the new command:

```
(config-system)# cwmp
(cwmp-tr069)# cwmp-acl <ACL name>
```

- Auto-detect mode (ADSL or VDSL) feature has been added for A/VDSL. For more information, refer to *Mediant MSBR LAN-WAN Access CLI Configuration Guide*.
- Triggering DNS entries of all types (A, AAAA, NAPTR, etc.) is now supported. For more information, refer to *Mediant MSBR IP Networking CLI Configuration Guide*.
- Hostnames can now be configured for the management ACL.
- The CLI terminal window height can now be locked. The feature can be configured through CLI using the command **default-window-height <value>** or through the Web interface using the new parameter 'Default terminal window height' (System > Management > Telnet/SSH Settings > General).
- ACL can now be applied to NAT port forwarding rules, by using the new option "match" for the **ip nat inside source** command. For example:
  - Access list rule called “PF-ACL”:

```
(config-data)# access-list PF-ACL permit ip host 4.4.4.4 any
```

  - Access list “PF-ACL” used in NAT port forwarding:

```
(config-data)# ip nat inside source static tcp 192.168.0.16
same gigabitethernet 0/0 8080 match PF-ACL
```

- Vendor-specific TR-069 log string can now be configured, using the DeviceLog parameter (InternetGatewayDevice.DeviceInfo.DeviceLog).
- Sending TR-069 connection request (send-connection-request) is now also available in unprivileged CLI mode, using the new command **debug cwmp send-connection-request**.

- Auto assign self IPv6 address has been added to **ipv6 dhcp-server dns-server address** when using a DHCP server.
- The status of all interfaces (**show data interfaces atm/bvi** ...) is now also available in unprivileged CLI mode.

3.2.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-2: Resolved Constraints for Patch Version 7.20A.154.025

Incident	Description
139561	New command has been added to view DDNS status (**show data ddns**).
142487	Configuration of GRE tunnel without a source interface is not allowed in order to prevent a mismatch with the other side of the tunnel.
145580	InternetGatewayDevice.LANDevice.1.LANEthernetInterfaceConfig.4 is not displayed in the TR-069 ACS.
142488	The configuration **qos match-map** input "NAME" VLAN 4001 appears as **qos match-map** input "NAME" internal-LAN when the **show** command is run.
142981	The RTP port is different than that advertised in the SDP body of the SIP 200 OK.
145066	The OSPF **max-metric router-lsa** command has no effect when the OSPF process is closed.
145636	The cellular, dynamic option driver is not saved after a device reset.
138373	In some cases, the Huawei 4G USB stick does not receive an IP address after a device reset.
145342	TR-069 provisioning code is lost after device reset and reverts to default ("VOIP.DATA").
142257	No option to configure dynamic learning of IPv6 NTP addresses on a PPPoE interface.
146575	QoS calibration on VDSL/EFM lines.
141714	Unable to display L2 hosts in the TR hosts table (but now possible (InternetGatewayDevice.LANDevice.{i}.Hosts.Host.{i}).
142836	LAN-based host feature doesn't show all hosts.
146255	Statically configured IPv6 route does not function when a dynamic IP address is configured.
146430	PPPoE interface cannot be underlying to an ATM interface.
144064	Single Network Mode - no RTP between local extensions (FXS and IP Phone) when using VRF.
145463	Single Network Mode – ringback tone from PRT file is not played.
144063	Single Network Mode - no RTP between local extensions (FXS and IP Phone / FXS and FXS) when using the loopback interface.
143933	Upload of files through TR-069 via HTTPS fails.
144486	TR-069 change of PPPoE credentials terminates too early and causes transaction error.

Incident	Description
144197	Configuring "cellular-backup" in the backup-group when IPSec crypto map is configured, causes the cellular interface to remain in non-operational mode.
144974	The **show run** command does not display IPSec, PFS or metric parameters under the crypto map if the crypto map is not associated with the interface.
146327	IPv6 addresses on the PPPoE interface does not function with IPv4 addresses.
143282	For DHCPv6 NTP, the **ipv6 dhcp-client ntp-server** command is not displayed by the **show run** command under the PPPoE interface.

3.3 Patch Version 7.20A.154.061

This patch version includes new features and resolved constraints.

Note:

- This version is based on MSBR Version 6.80A.347.001 (released in July 2017). In other words, all data capabilities of this MSBR 7.2 version are fully aligned to the above mentioned 6.8 version (with a few exceptions, as listed in the document *MSBR Data Feature Additions from 6.8 to 7.2*).
- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ VDSL ISDN
  - √ VDSL POTS
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.3.1 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-3: Resolved Constraints for Patch Version 7.20A.154.061

Incident	Description
147776	The device's DHCPv4 server now supports fast revival after reset, using DHCPREQUEST messages.
147781	TR-181 operations cause the device's CLI to freeze.
147706	AAA TACACS configuration is not saved to configuration.
147732	Issue with saving configuration of Access List with SNMP community.
146955	Device crashes on rare occasions when SNMP is used to GET QoS information.

3.4 Patch Version 7.20A.154.078

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.4.1 New Features

New features introduced in this version include the following:

- Support for DNS with VRRP.
- Support for disabling the DHCP "dynamic" mode. When the command **no ip dhcp-server dynamic** is run, the DHCP server only answers to statically configured hosts.
- Support for the ZTE MF833V cellular dongle.

3.4.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-4: Resolved Constraints for Patch Version 7.20A.154.078

Incident	Description
146911	IPSec does not function when **ipsec access list destination** is set to "any".
147776	DHCP client does not renew its DHCP lease if the device undergoes an unplanned reset. DHCP lease renewal is possible only if the device is restarted during DHCP client lease time.
148218	When VRRP backup becomes operational, it erases dynamic leases. To prevent this, the VRRP backup device uses ARP to keep the lease of active IPs.
147955	Under some conditions, the ini file cannot be loaded using the Automatic Update mechanism (IniFileURL parameter).

3.5 Patch Version 7.20A.200.038

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.5.1 New Features

New features introduced in this version include the following:

- DNS lookup queries for a specific VRF. To support this feature, the following new command has been added:

```
nslookup [Hostname] source data vrf [VRF Name]
```

- Configuration of a specific protocol bind (snmp|http|https|telnet|ssh) per management server. :

```
bind source-address interface [Interface] management-servers
[http|https|snmp|ssh|telnet]
```

- Support for multicast in VRFs, using the new 'pim' command:

```
ip vrf <VRF Name> enable pim
```

- Configuration of Gratuitous ARP (GARP) per interface with timer, using the following new commands:

```
(config-data)# garp timer <Seconds 1-3600, Default 60>
(conf-if-GE 0/0)# garp enable | no garp enable
```

  The feature is applicable only to Gigabit and fiber WAN interface types (VLAN 1 only).

- Support for Y.1731.

- Loading License Key file through CLI (from HTTP, HTTPS, FTP, TFTP, or NFS server), using the following new command:

```
# copy feature-key from [URL]
```

- Web-based management interface (Web End-User) for end users, allowing basic configuration, for example, LAN ports settings, WAN ports settings, Wi-Fi settings, and port forwarding settings. For more information, refer to the *Mediant MSBR Basic System Setup CLI Configuration Guide.*
- Configuration of maximum path for BGP, using the following new command:

```
(config-data)# router bgp [AS Number] maximum-paths [Number]
```

3.5.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-5: Resolved Constraints for Patch Version 7.20A.200.038

Incident	Description
147996	Incorrect order of SNMP configuration through CLI prevents configuration to be applied.
149101	When the WAN interface is configured on VRF, the Auto-Update and copy features do not function if DNS resolution is required.
148592	For TR-069 management, digest authentication messages are sent in the wrong format.

3.6 Patch Version 7.20A.202.112

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.6.1 New Features

New features introduced in this version include the following:

- Static WAN IP address configuration through the End User Web interface.
- The InternetGatewayDevice.Time. object is now supported for TR-098.
- The InternetGatewayDevice.Time. object is now supported for TR-069, allowing NTP to be enabled or disabled.
- The Device.Time object is now supported for TR-069.
- The Debug Capture feature now allows the naming of the captured files and sending them to specific folders on TFTP servers.
- Support for DHCPv4 Option 66 to obtain the TFTP server name and Option 67 to obtain the configuration file name, when the MSBR is a DHCP client.
- The output of the CLI commands **show system alarms** and **show system alarms-history** can now be displayed in JSON format, using the following new CLI command:

```
output-format json
```

The output is returned to plain text format using the following command:

```
output-format plain
```

3.6.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-6: Resolved Constraints for Patch Version 7.20A.202.112

Incident	Description
141040	The settings of the **web-restrict** CLI command does not save after a device reset.
142368	Rate counters don't show fast-path and non fast-path traffic.
148992	Access list IPv6 rule has no 'precedence' option.
148811	WiFi "Krack" vulnerability: PTK rekeying to generate a new ANonce.
148909	Hostname resolution for VRFs is not supported. Access list is not binding to a single VRF. Hostname resolution for IPv6 is not supported. Access list FQDN does not support multiple VRFs.
148983	Automatic switching from EFM to ATM for SHDSL interfaces does not function.
150110	Bind to WAN - the internal LAN IP address appears in the **show run data** output (and should not).
150432	The **show data interface pppoe** CLI command does not show the subnet mask in its output.
150518	IPv6 cannot be enabled or disabled through TR-181.
150755	The NTP server IP address cannot be obtained from DHCP.
150763	The device doesn't renew its' IP address after a DHCP server configuration change.
150904	The device doesn't bind VLAN 100 and higher in Single Networking Mode.
150931	The device doesn't accept BGP configuration of peers in peer-group.
150932	CLI **show** commands for the Wi-Fi interface do not display "connected" status.
150963	Multiple VRRP IDs (per interface) doesn't support DHCPv4.
151017	The **copy** command fails first time with the error "(6) Could not resolve host".
152270	The configuration by the **coders-and-profiles** command is displayed in the wrong location in the CLI script, which causes an error when applying the script to the device.
152974	The license for the Zero Configuration feature is not retained after a hardware (button) reset.
153155	Debug Recording cannot be sent to the WAN over a non-default VRF.
153545	The **show data interface <Interface Name>** command does not show the subnet mask when the IP address is obtained through DHCP.
152984	After clicking the Edit button in the SIP Interfaces table, the network interface doesn't appear even though it is configured.
149652	The Auto Provisioning process doesn't complete configuration file load.

Incident	Description
153235	For the configuration of **sip-definition account**, no space appears before "obscured" in the obscured password (e.g., password /cnHyzQwOzo9NjY/OA==obscured").
152930	When DHCPv4 is configured with a static IPv6 address, the **show data ipv6 interface brief** command does not display IPv6 status properly. DHCP client configuration on WAN copper disables IPv6 protocol.
152875	A WAN IPv6 cannot be assigned to a Media Realm in the Web interface.
152395	The **show data ip igmp proxy groups** command displays an error.
150707	In TR-069, no support for MTU fragmentation.

3.7 Patch Version 7.20A.202.307

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.7.1 New Features

New features introduced in this version include the following:

- Support for an integrated LTE modem.
- New debug log commands:
  - Displays the device's syslog of exceptions:
    ```
    debug exception-syslog-history
    ```
  - Displays the device's syslog of resets:
    ```
    debug reset-syslog-history
    ```
- End-User Web Interface:
  - DHCP Settings has been moved to the LAN Interface page.
  - Configuration of multiple PPPoE interfaces (configure system > end-user > wan-if pppoe auto).

- Display of the connected (active) PPPoE interface, even when multiple PPPoE interfaces have been configured.
- New configuration table has been added "Multiple Subscriber Number" (Voice folder > Multiple Subscriber Number) for FXS and BRI interfaces.

3.7.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-7: Resolved Constraints for Patch Version 7.20A.202.307

Incident	Description
155106 / 154404 / 154463	Uploading the CLI Script file through the Web interface fails and only part of the script is applied. This occurs when some of the configuration is not accepted.
152228	In the End User Web interface, the pages under the LAN Guest folder display incorrect DHCP information.
153838	A CLI command has been added (**ipv6 dhcp-client prefix-len-128**), which changes the prefix length of a received IPv6 address through DHCP to 128 bit (instead of the default 64). This has been done to comply with RFC 5942.
153921	In the End-User Web interface, the login password is not saved after a device reset.
153977	DNS resolution with NAPTR is not functioning.
153996	In the End-User Web interface, an error occurs when assigning a static IP address for the WAN Backup interface.
154240	When the WAN cable is unplugged and then plugged in again, no connection is experienced to the management interfaces (HTTP, telnet, etc..).
154292	After a Media Realm is configured, it cannot be edited.
154296	In the End-User Web interface, the Apply button doesn't function for LAN Guest interfaces.
154769	TLS configuration is not saved to the device's CLI.
154859	In IPv6 PD configuration, the default route to the LAN subnet appears in the Routing table to its own address.
154912	The **show run** command displays an unwanted DynDNS configuration after each device reset.
155104	After upgrading from Version 6.8 to 7.20A.202, the device reports a different hardware version through TR-069.

3.8 Patch Version 7.20A.250.028

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.8.1 New Features

This section describes the new features introduced in this version.

3.8.1.1 WAN Status and Performance Monitoring Display

The device's Web interface now displays the following WAN interface information:

- WAN status (Monitor menu > Monitor tab > Data Status folder > Network Status)
- WAN statistics (Monitor menu > Monitor tab > Data Status folder > Network Statistics)
- WAN performance statistics (Monitor menu > Monitor tab > Data Status folder > Network Performance Monitors)

Applicable Products: All.

3.8.1.2 Copper WAN through SFP

The small form-factor pluggable (SFP) optical module, typically used for WAN fiber interface, can also be used for WAN copper interface. This new feature provides support by the device's management interfaces to display the duplex mode (full or half) of this WAN copper connection type.

Applicable Products: All.

3.8.1.3 Display of DSL Transmission Statistics

A new CLI command has been introduced that displays historical statistics of upstream and downstream transmission properties (speed, power, SNR margin and attenuation) of the DSL interface:

```
show data interface dsl <Slot>/<Port> history
```

For example:

```
# sh d in dsl 0/2 history
Time: 03/01/2018 11:11:03
Downstream: Actual speed 112636000, power 13.9, SNR margin 26.2,
Attenuation 0.1
Upstream: Actual speed 83680000, power 8.1, SNR margin 5.3,
Attenuation 1.6
Time: 03/01/2018 11:09:53
Downstream: Actual speed 112636000, power 13.9, SNR margin 25.9,
Attenuation 0.1
Upstream: Actual speed 83680000, power 8.1, SNR margin 5.2,
Attenuation 1.6
```

Applicable Products: All.

3.8.1.4 DHCPv4 Option 82 Support

The device supports DHCP Option 82. When this feature is enabled and a DHCP relay agent forwards client-originated DHCP packets containing Option 82 to the device (acting as a DHCP server), the device "echos" the information of Option 82 back to the DHCP client. The feature is enabled for the interface on which the DHCPv4 server is configured, using the following new CLI command:

```
ip dhcp-server option82
```

Applicable Products: All.

3.8.1.5 LTE WWAN Support

The device supports Long-Term Evolution (LTE) wireless WAN (WWAN). This is supported by an integrated 4G LTE cellular modem, two cellular antennas, and a slot for inserting a Subscriber Identity Module (SIM) card to connect with the 4G cellular network.

Applicable Products: Mediant 500L MSBR.

3.8.1.6 QoS on L2TP Interfaces

The device supports the configuration of Quality of Service (QoS) on Layer 2 Tunneling Protocol (L2TP) interfaces.

Applicable Products: All.

3.8.1.7 TR-069 Annex F

The device supports TR-069 Annex F. Annex F is relevant when the Gateway and the Device (CPE) are managed by the same ACS. According to Annex F, the Device and the Gateway (to which the Device is connected) pass their private information to one another, and the ACS identifies the Device as being under the Gateway. The MSBR can be the Device or the Gateway. When the MSBR uses TR-181 Data Model, it functions as the Device; when the MSBR uses TR-098 Internet Gateway Device, it functions as the Gateway.

As a result of this feature, the following new TR-069 objects and parameters are now supported:

- Device.GatewayInfo object (TR-181 Device Data Model)
- ManageableDeviceNumberOfEntries parameter to InternetGatewayDevice.ManagementServer. object (TR-098 Device Data Model)
- nternetGatewayDevice.ManagementServer.ManageableDevice object and its parameters (TR-098 Device Data Model)

Applicable Products: All.

3.8.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-8: Resolved Constraints for Patch Version 7.20A.250.028

Incident	Description
MSBR-8212	When an SNMP trap destination is configured, it appears twice in the CLI script.
MSBR-8141	Configuration cannot be backed up in OVOC when VRF is configured on the WAN interface.
MSBR-8100	The device crashes upon a JSON output of configuration through TR-069.
MSBR-8072	The device doesn't create a default route when using DHCP IPv6.
MSBR-7947	Configuration is not saved to flash under MSN configuration in the End User Web GUI.
MSBR-1447	In Single Network Mode, Debug Recording packets show the internal WAN IP address instead of the WAN interface IP address.
MSBR-1434	The subnet musk is not displayed in the **show data ip interface** command when the IP address mode is DHCP.
MSBR-8248	The device reboots after running the **show run** command through TR-069.
SBC-9536 (VI-155449)	When the device operates with the CRP application, it has invalid default configuration, which prevents the CRP from being configured correctly (IP Group and IP-to-IP Routing tables).

3.9 Patch Version 7.20A.252.062

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.9.1 New Features

This section describes the new features introduced in this version.

3.9.1.1 Read-Only for LAN Guest-LAN Interface Page for Web End-Users

For Web End Users, the parameters on the LAN Interface page in the LAN Guest folder can be made read-only. In other words, this applies to the parameters under the LAN Interfaces Settings group and DHCP Settings group.

The feature is configured by the Administrator using the following new parameter:

- ini: EndUserAllowLanGuestSettings
- CLI: configure system > end-user > allow-lan-guest-settings enable|disable

Applicable Products: MSBR.

3.9.1.2 Hide and Read-Only for Multiple Subscriber Number Table for Web End-Users

For Web End-Users, the Voice folder (Monitor > Voice) can be hidden by the following new command:

```
(config-system)# end-user > allow-voice-settings enable|disable
```

This folder contains the Multiple Subscriber Number table and therefore, if the folder is hidden, the table will also be hidden.

In addition, when the folder is shown (enabled), the Administrator can apply the following security features to the table:

- 'User ID' parameter value is shown read-only
- The 'Password' parameter and value are hidden

This is configured by the following new parameter:

- ini: EndUserMsnSettings
- CLI: configure system > end user > allow-msn-authentication-settings enable|disable

Applicable Products: MSBR.

3.9.1.3 BFD for IPv6 BGP

The device supports Bidirectional Forwarding Detection (BFD) for a Border Gateway Protocol (BGP).

The feature includes the following new commands:

- BFD for a BGP AS (autonomous system) is enabled by a new command:

```
(config-data)# router bgp <as-id>
(bgp-router) # neighbor < neighbor ip> fall-over bfd interval
<value> min_rx <value> multiplier <value>
```

  Where:

  - *interval*: interval (in msec) for outgoing BFD messages. The interval is increased if the remote system requires it.
  - *min_rx*: minimum interval (in msec) between BFD messages. The remote system uses this interval for sending messages in case its interval is lower.
  - *multiplier*: maximum number of packets that can be missed before the session status is considered down.

Applicable Products: MSBR.

3.9.1.4 MD5 Password for IPv6 BGP Sessions

An MD5 password can now be configured for IPv6 BGP network interfaces, using the following existing command:

```
(config-data)# router bgp 1
(conf-router)# neighbor 2010:18::200:200 password 0101010101
```

Applicable Products: MSBR.

3.9.1.5 OVOC Floating License Support via VRF

The device supports the Floating License application when communication with OVOC is through one of the device's VRF.

Applicable Products: MSBR.

3.9.2 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-9: Resolved Constraints for Patch Version 7.20A.252.062

Incident	Description
MSBR-8446	For DHCPv6, NTP servers advertised on DHCP Option 31 instead of Option 56.
MSBR-8445	When the device uses IPv6, it connects LAN media streams to an internal IPv6 address, which causes incorrect SBC signaling on the device.
MSBR-8438	For DHCPv6, there is no option to configure a DHCPv6 server on the LAN to advertise IPv6 NTP servers dynamically learned on the WAN,
MSBR-8409	The Save button in the Web interface is erroneously displayed in red after a Startup Script file is loaded successfully.
MSBR-8398	The device changes the TCP port and resets the socket connection.
MSBR-8380	The device crashes (resets) when changing MTU on the cellular interface.
MSBR-8356	IPSec via LTE is not functioning on the cellular interface.
MSBR-8347	The login password cannot be changed by the End User.
MSBR-8341	Console accessible via Telnet (exposing security risk).
MSBR-8340	Cross scripting vulnerability on JSON pages in Web interface.
MSBR-8275	Vulnerability in shell command injection in sysupgrade.sh in .cmp firmware file.
MSBR-8218	WAN SIP address gets wrong loopback IP address as source.
MSBR-1486	IPv6 PD (Prefix Delegation) doesn't function with Stateful DHCPv6 mode.
MSBR-8328	In some scenarios, a problem in configuration is experienced when it is loaded using the Auto-Update mechanism.
MSBR-8390	High data CPU experienced during 200 concurrent voice calls.

3.10 Patch Version 7.20A.252.078

This patch version includes new features and resolved constraints.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- This version is not recommended for migrating existing MSBR fleets from Version 6.8, but rather for new customers, and for existing customers for evaluation and preparation for future upgrades from Version 6.8.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
  - √ SHDSL
  - √ T1 WAN
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-181 customers.

3.10.1 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-10: Resolved Constraints for Patch Version 7.20A.252.078

Incident	Description
SBC-13728	The device locks (AdminState = 0) after a reset.
MSBR-8590	The device tries to establish a TCP session using an internal IP address instead of the associated SIP Interface.
MSBR-8579	When the device connects to a third-party gateway, it doesn't receive GW Info from it.
MSBR-8553	Some voice parameters are in the ini file, but not in the CLI. The following CLI commands were added under configure voip > media voice: ▪ **mf-transport-type** (MFTransportType)

Incident	Description
	▪ **mfr1-detector-enable** (MFR1DetectorEnable) ▪ **dtmf-detector-enable** (DTMFDetectorEnable)
MSBR-8540	When both copper WAN and fiber are connected, DHCP lease renewal from the copper WAN is not sent.
MSBR-8535	The BGP password is not encrypted in the CLI **show run** output.
MSBR-8499	When the device operates in Single Networking mode, it cannot send SIP messages to its own IP interface.

3.11 Patch Version 7.20A.254.026

This patch version includes resolved constraints only.

Note:

- This version is introduced as a controlled release and is not backward compatible with Version 6.8 concerning some of the management interfaces and mainly the command-line interface. For more details, please refer to the relevant documents released together with this document.
- If the MSBR device is running a software version that is earlier than 6.80A.286.002, to upgrade the device to Version 7.2, the device must first be upgraded to Version 6.80A.286.002 and only then upgraded to Version 7.2.
- This version is applicable only to the following MSBR devices:
  - √ Mediant 500 MSBR
  - √ Mediant 500L MSBR
  - √ Mediant 800B MSBR
  - √ Mediant 800C MSBR
- This version is applicable only to the following physical WAN interfaces:
  - √ Copper
  - √ Cellular
  - √ Fiber
  - √ A/VDSL ISDN
  - √ A/VDSL POTS
- This version is not released for customers with T1 and SHDSL WAN flavors.
- This version is not released for TR-069 customers.

3.11.1 Resolved Constraints

Constraints from previous versions that have now been resolved include the following:

Table 3-11: Resolved Constraints for Patch Version 7.20A.254.026

Incident	Description
MSBR-8356	IPSec (PSK and RSA) on the device's LTE interface doesn't function in branch office scenarios. **Applicable Products:** Mediant 500L
MSBR-8376	The NTP server is unreachable when SNMP is disabled.
MSBR-8398	The device uses a different TCP port (resets socket connection) from the one used for registering with the SIP proxy, for subsequent INVITE messages. As a result, calls fail. **Applicable Products:** All
MSBR-8423	In some scenarios, the device doesn't periodically send the ACS inform messages. **Applicable Products:** All
MSBR-8550	No data transmission possible due to short connection timeout.

Incident	Description
	Applicable Products: All
MSBR-8568	The login password interval (WebPassChangeInterval) can now be configured through CLI using the new command web-password-change-interval. **Applicable Products:** All
MSBR-8599	The device doesn't save port mirroring configuration after device reset. (Now, it does if configured by the new command configure data > port-monitor-save-after-reset.) **Applicable Products:** All
MSBR-8611	The device doesn't support the receipt of TR-069 redirect links that contain many characters. (Now, it supports up to 256 characters.) **Applicable Products:** All
MSBR-8693	A new CLI command has been added -- configure voip > gateway digital settings > pstn-compatibility-profile (PstnCompatibilityProfile). **Applicable Products:** All
MSBR-8755	The device supports up to 10 SRDs. (Now, it supports up to 15). **Applicable Products:** Mediant 800 MSBR

4 Capacity for Gateways & SBCs

This section provides capacity for the Gateway and SBC products.

4.1 SIP Signaling and Media Capacity

The following table lists the maximum, concurrent SIP signaling sessions, concurrent media sessions, and registered users per product.

Table 4-1: SIP Signaling and Media Capacity per SBC and Gateway Product

Product		Signaling Capacity		Media Sessions			
		SIP Sessions	Registered Users	Session Type	RTP	SRTP	Detailed Media Capabilities
Mediant 500		250	1,500	Hybrid	250	200	Transcoding: n/a GW: Table 4-5
				GW-Only	30	30	
Mediant 500L		60	200	Hybrid	60	60	Transcoding: n/a GW: Table 4-7
				GW-Only	8	8	
Mediant 800A		60	200	Hybrid	60	60	GW & Transcoding: Table 4-9 SBC Only: Table 4-8
Mediant 800B		250	1,500	Hybrid	250	250	GW & Transcoding: Table 4-9 SBC Only: Table 4-8
				GW-Only	64	64	
Mediant 800C		400	2,000	Hybrid	400	300	GW & Transcoding: Table 4-11
				GW-Only	124	124	
Mediant 1000B		150	600	Hybrid	150	120	Transcoding: Table 4-15 GW: Tables Table 4-12, Table 4-13, Table 4-14
				GW-Only	192	140	
MP-1288		588	350	Hybrid	588	438	Transcoding: n/a GW: Table 4-16
				SBC-Only	300	300	
				GW-Only	288	288	
Mediant 2600		600	8,000	SBC-Only	600	600	Table 4-17
Mediant 4000		5,000	20,000	SBC-Only	5,000	3,000	Table 4-18
Mediant 4000B		5,000	20,000	SBC-Only	5,000	5,000	Table 4-19
Mediant 9000	Hyper-Threading (HT) Disabled	24,000	180,000	SBC-Only	16,000	16,000	Table 4-20
		24,000	0	SBC-Only	24,000	16,000	Table 4-20
	SIP Performance Profile (HT Enabled)	30,000	300,000	SBC-Only	30,000	16,000	-
		55,000	0	SBC-Only	55,000	18,000	-
	DSP Performance Profile (HT Enabled)	50,000	0	SBC-Only	50,000	18,000	Table 4-20
	SRTP Performance Profile (HT Enabled)	50,000	0	SBC-Only	50,000	40,000	-
Mediant 9000 Rev. B	SIP Performance Profile	50,000	500,000	SBC-Only	50,000	30,000	-
		70,000	0	SBC-Only	70,000	30,000	-
	DSP Performance Profile	50,000	0	SBC-Only	50,000	28,000	Table 4-20

Product			Signaling Capacity		Media Sessions			
			SIP Sessions	Registered Users	Session Type	RTP	SRTP	Detailed Media Capabilities
	SRTP Performance Profile		70,000	0	SBC-Only	70,000	40,000	-
Mediant 9030	SIP Performance Profile		30,000	200,000	SBC-Only	30,000	30,000	-
	DSP Performance Profile		30,000	200,000	SBC-Only	30,000	15,000	Table 4-23
Mediant 9080	SIP Performance Profile		50,000	500,000	SBC-Only	50,000	30,000	-
			70,000	0	SBC-Only	70,000	30,000	-
	DSP Performance Profile		50,000	0	SBC-Only	50,000	28,000	Table 4-20
	SRTP Performance Profile		70,000	0	SBC-Only	70,000	40,000	-
Mediant 9000 with Media Transcoders (MT-type)			24,000	180,000	SBC-Only	24,000	16,000	Table 4-22
Mediant 9000 Rev. B with Media Transcoders (MT-type)			60,000	200,000	SBC-Only	60,000	40,000	Table 4-22
Mediant 9080 with Media Transcoders (MT-type)			60,000	200,000	SBC-Only	60,000	40,000	Table 4-22
Mediant CE	AWS / EC2		40,000	0	SBC-Only	40,000	40,000	Forwarding: Table 4-25 Transcoding: Table 4-26
			20,000	100,000	SBC-Only	20,000	20,000	
	Azure		10,000	50,000	SBC-Only	10,000	10,000	Table 4-28
Mediant VE	VMware	1 vCPU, 2-GB RAM	250	1,000	SBC-Only	250	250	-
		1/2/4 vCPU, 8-GB RAM	3,000	15,000	SBC-Only	3,000	2,000	1 vCPU (n/a) 2 vCPU (Table 4-31) 4 vCPU (Table 4-33)
		4/8 vCPU 16-GB RAM	9,000	75,000	SBC-Only	6,000	5,000	4 vCPU (n/a) 8 vCPU (Table 4-35)
	OpenStack KVM	1 vCPU 2-GB RAM	250	1,000	SBC-Only	250	250	-
		1/2/4 vCPU 8-GB RAM	1,800	9,000	SBC-Only	1,800	1,400	1 vCPU (n/a) 2 vCPU (Table 4-31) 4 vCPU (Table 4-33)
		4/8 vCPU 16-GB RAM	4,000	75,000	SBC-Only	2,700	2,700	Table 4-35
		8 vCPU 32-GB RAM SR-IOV Intel NICs	24,000	75,000	SBC-Only	24,000	10,000	-
	Hyper-V	1 vCPU 2-GB RAM	250	1,000	SBC-Only	250	250	-
		1/2/4 vCPU 4-GB RAM	900	10,000	SBC-Only	600	600	1 vCPU (n/a) 2 vCPU (Table 4-41) 4 vCPU (Table 4-43)
	Azure	DS1_v2	400	1,000	SBC-Only	400	400	Table 4-40
		DS2_v2	500	15,000	SBC-Only	500	500	Table 4-40
		DS3_v2	600	50,000	SBC-Only	600	600	Table 4-40

Product			Signaling Capacity		Media Sessions			
			SIP Sessions	Registered Users	Session Type	RTP	SRTP	Detailed Media Capabilities
	AWS / EC2	r4.large	3,200	20,000	SBC-Only	3,200	3,200	-
		c4.2xlarge	2,000	75,000	SBC-Only	2,000	2,000	Table 4-37
		c4.8xlarge	3,200	75,000	SBC-Only	3,200	3,200	Table 4-38
Mediant VE with Media Transcoders	OpenStack KVM	8 vCPU 64-GB RAM SR-IOV Intel NICs	24,000	75,000	SBC-Only	24,000	12,000	MT-type (Table 4-45) vMT-type (Table 4-46)
Mediant SE	DL360p Gen8 or DL360 Gen9		24,000	120,000	SBC-Only	16,000	14,000	-
			24,000	0	SBC-Only	24,000	14,000	-
	DL360 Gen10	SIP Performance Profile	50,000	500,000	SBC-Only	50,000	30,000	-
			70,000	0	SBC-Only	70,000	30,000	-
		DSP Performance Profile	50,000	0	SBC-Only	50,000	28,000	Table 4-29
		SRTP Performance Profile	70,000	0	SBC-Only	70,000	40,000	-

Notes:

- The figures listed in the table are accurate at the time of publication of this document. However, these figures may change due to a later software update. For the latest figures, please contact your AudioCodes sales representative.
- "GW" refers to Gateway functionality.
- "SIP Sessions" refers to the maximum concurrent signaling sessions for both SBC and Gateway (when applicable). Whenever signaling sessions is above the maximum media sessions, the rest of the signaling sessions can be used for Direct Media.
- "Session Type" refers to Gateway-only sessions, SBC-only sessions, or Hybrid sessions which is any mixture of SBC and Gateway sessions under the limitations of Gateway-only or SBC-only maximum values.
- "RTP Sessions" refers to the maximum concurrent RTP sessions when all sessions are RTP-RTP (for SBC sessions) or TDM-RTP (for Gateway sessions).
- "SRTP Sessions" refers to the maximum concurrent SRTP sessions when all sessions are RTP-SRTP (for SBC sessions) or TDM-SRTP (for Gateway sessions).
- "Registered Users" refers to the maximum number of users that can be registered with the device. This applies to the supported application (SBC or CRP).

- Regarding signaling, media, and transcoding session resources:
  - √ A signaling session is a SIP dialog session between two SIP entities, traversing the SBC and using one signaling session resource.
  - √ A media session is an audio (RTP or SRTP), fax (T.38), or video session between two SIP entities, traversing the SBC and using one media session resource.
  - √ A gateway session (i.e. TDM-RTP or TDM-SRTP) is also considered as a media session for the calculation of media sessions. In other words, the maximum Media Sessions specified in the table refer to the sum of Gateway and SBC sessions.
  - √ In case of direct media (i.e., Anti-tromboning / Non-Media Anchoring), where only SIP signaling traverses the SBC and media flows directly between the SIP entities, only a signaling session resource is used. Thus, for products with a greater signaling session capacity than media, even when media session resources have been exhausted, additional signaling sessions can still be handled for direct-media calls.
  - √ For call sessions requiring transcoding, one transcoding session resource is also used. For example, for a non-direct media call in which one leg uses G.711 and the other leg G.729, one signaling resource, one media session resource, and one transcoding session resource is used.
- Capacity of the Cloud Resilience Package (CRP) application is listed under "Registered Users".
- Capacity of the Lync Analog Device (LAD) application is listed under "Media Sessions".
- **MP-1288:** The maximum number of media and signaling sessions is the summation of the maximum 300 RTP-to-RTP (SBC) sessions and the maximum 288 TDM-RTP (Gateway) sessions. The maximum number of SRTP sessions is the summation of the maximum 150 RTP-to-SRTP (SBC) sessions and the maximum 288 TDM-SRTP (Gateway) sessions.
- Hyper-Threading (HT) is disabled by default on Mediant 9000 with 1G ports only. To enable HT, please refer to the *Mediant 9000 SBC Installation Manual*.
- Media Transcoding Cluster (MTC) feature is not supported by Mediant 9030 SBC.
- **Mediant 90xx SBC and Mediant VE SBC with Media Transcoders limitations:**
  * To allow DSP capabilities (such as transcoding), the Performance Profile parameter must be configured to the DSP profile.
  Each transcoding session is weighted as two RTP-RTP sessions without transcoding. Therefore, the number of sessions without transcoding plus the doubled number of sessions with transcoding must be less than the maximum RTP-RTP figure specified in the table. As result, if all sessions involve transcoding, the maximum number of sessions is half the maximum RTP-RTP sessions without transcoding specified in the table.
  ** The maximum SRTP-RTP sessions is also effected by the above limitations. For example, if sessions involve transcoding, the maximum number of SRTP-RTP sessions is also limited by half of the maximum SRTP-RTP sessions without transcoding.
- **Mediant 9030:** The SRTP Performance Profile is recommended for this product.
- **Mediant VE SBC with vMT-type Media Transcoder:** The host running the vMT virtual machine requires the following configuration:
  - √ At least 2.8 GHz CPU with Intel® AVX support
  - √ SR-IOV enabled NICs
  - √ KVM environment
  - √ 8 hyper-threaded vCPUs should be allocated to the vMT virtual machine (4 physical cores)
  - √ 4-GB RAM should be allocated to the vMT virtual machine

- **Mediant VE SBC and vMT-type Media Transcoder:** Codec-transcoding functionality is supported only on Intel CPUs with AVX enhancement. In addition, AVX support must be reflected on the vCPU of the SBC virtual machine.
- **Mediant VE SBC with Media Transcoder Cluster** is currently supported only on the OpenStack KVM hypervisor.
- **Mediant CE:** This is based on the following AWS instances:
  - √ Signaling Components (SC): r4.2xlarge
  - √ Media Components (MC) - forwarding only: r4.large
  - √ Media Components (MC) - forwarding and transcoding: c4.4xlarge
- **Mediant SE:** For new deployments, it's highly recommended to use the DL360 G10 server. For exact specifications and BIOS settings, please contact your AudioCodes sales representative.

4.2 Session Capacity per Feature

The table below lists capacity per feature, per product.

Table 4-2: Capacity per Feature for Gateways and SBCs

Product	Max. Concurrent WebRTC Sessions	Maximum One-Voice Resiliency (OVR) Users	Max. Concurrent SIPRec Sessions
MP-1288	-	-	-
Mediant 500	-	-	200
Mediant 500L	-	-	30
Mediant 800B	100	100	200
Mediant 800C	100	150	200
Mediant 1000B	-	50	-
Mediant 2600	600	-	300
Mediant 4000B	1,000	-	2,500
Mediant 9000	5,000	-	20,000 (16,000 without HyperThreading)
Mediant 9030	5,000	-	20,000
Mediant 9080	8,000	-	20,000
Mediant VE	5,000	2,000	12,000*
Mediant SE	5,000	-	12,000*
Mediant CE	5,000	-	20,000

Note:

- The figures in the table above for SIPRec capacity assume that there are no other concurrent, regular (non-SIPRec) voice sessions.
- For Mediant VE SBC, SIPRec capacity depends on instance size.

4.3 Configuration Tables Capacity

The maximum rows (indices) that can be configured per configuration table is listed in the table below.

Table 4-3: Configuration Table Capacity for SBC and Gateway Products

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
SRDs	20	280	600	▪ 2 GB: 20 ▪ 3.5 GB: 70 ▪ 4 GB: 100 ▪ 8 GB: 200

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
				▪ 16 GB: 400 ▪ 32-64 GB: 600
SIP Interfaces	82	500	1,200	▪ 2 GB: 600 ▪ 3.5-64 GB: 1,200
Media Realms	12	1,024	1,024	1,024
Media Realm Extension	2 x Max. Media Realms (MP-1288, Mediant 500, Mediant 500L, Mediant 800 Only)	2 x Max. Media Realms (Mediant 2600) 5 x Max. Media Realms (Mediant 4000B)	5 x Max. Media Realms	5 x Max. Media Realms
Remote Media Subnet	5	5	5	5
Proxy Sets	102	625	5,000	▪ 2 GB: 80 ▪ 3.5 GB: 1,000 ▪ 4-16 GB: 1,500 ▪ 32-64 GB: 5,000
Proxy Address (and DNS-resolved IP addresses) per Proxy Set	10 (15 DNS-resolved IP addresses)	10 (15 DNS-resolved IP addresses)	50 (50 DNS-resolved IP addresses)	▪ 2 GB: 10 (15 DNS-resolved IP addresses) ▪ 3.5 GB: 1,000 ▪ 8-16 GB: 10 (50 DNS-resolved IP addresses) ▪ 32-64 GB: 50 (50 DNS-resolved IP addresses)
IP Groups	80	700	5,000	▪ 2 GB: 80 ▪ 3.5 GB: 1,000 ▪ 4-16 GB: 1,500 ▪ 32-64 GB: 5,000
IP Profiles	20 (MP-1288 / Mediant 500/L / Mediant 800); 40 (Mediant 1000)	125	300	▪ 2 GB: 150 ▪ 3.5-64 GB: 300
Tel Profiles	9	-	-	-
Coder Groups	11	21	21	21

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
Allowed Audio Coders Groups	10	20	20	20
Allowed Video Coders Groups	4	4	4	4
Classification	102	625	1,500	▪ 2 GB: 750 ▪ 3.5-64 GB: 1,500
Routing Policies	20 (SBC)	280	600	▪ 2 GB: 20 ▪ 3.5 GB: 70 ▪ 4 GB: 100 ▪ 8 GB: 200 ▪ 16 GB: 400 ▪ 32-64 GB: 600
IP-to-IP Routing	615	3,750	9,000	▪ 2 GB: 4500 ▪ 3.5-64 GB: 9,000
Alternative Routing Reasons	20	20	20	20
IP Group Set	51	312	750	750
Inbound Manipulations	205	1,250	3,000	3,000
Outbound Manipulations	205	1,250	3,000	3,000
Call Admission Control Profile	102	625	1,500	1,500
Call Admission Control Rule (per Profile)	8	8	8	8
Malicious Signature	30	30	30	30
External Media Source	1	1	1	1
Trunk Group	288 (MP-1288); 24 (Mediant 500/L; Mediant 800); 240 (Mediant 1000)	-	-	-
Trunk Group Settings	289 (MP-1288); 101 (Mediant 500/L; Mediant 800); 241 (Mediant 1000)	-	-	-
Tel-to-IP Routing	180	-	-	-
IP-to-Tel Routing	120	-	-	-
Forward On Busy Trunk Destination		-	-	-
Routing Policies	1 (Gateway)	-	-	-
Charge Codes	25	-	-	-
Reasons for IP-to-Tel Alternative Routing	10	-	-	-

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
Reasons for Tel-to-IP Alternative Routing	10	-	-	-
Destination Phone Number Manipulation for IP-to-Tel Calls	120	-	-	-
Destination Phone Number Manipulation for Tel-to-IP Calls	120	-	-	-
Calling Name Manipulation for IP-to-Tel Calls	120	-	-	-
Calling Name Manipulation for Tel-to-IP Calls	120	-	-	-
Source Phone Number Manipulation for IP-to-Tel Calls	120	-	-	-
Source Phone Number Manipulation for Tel-to-IP Calls	120	-	-	-
Redirect Number IP-to-Tel	20	-	-	-
Redirect Number Tel-to-IP	20	-	-	-
Phone Contexts	20	-	-	-
Release Cause Mapping from SIP to ISDN	12	-	-	-
Release Cause Mapping from ISDN to SIP	12	-	-	-
Release Cause ISDN->ISDN	10	-	-	-
Char Conversion	40	-	-	-
Supplementary Services	100	-	-	-
Tone Index	50	-	-	-
Accounts	102	625	1,500	1,500
Call Setup Rules	64	64	64	64
Cost Groups	10	10	10	10
Time Band	70 (21 per Cost Group)	70 (21 per Cost Group)	70 (21 per Cost Group)	70 (21 per Cost Group)
Dial Plan	10	25	50	50

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
Dial Plan Rule	2,000	10,000	100,000	▪ < 16 GB: 2,000 ▪ > 16 GB: 100,000
Message Manipulations	100 (MP-1288 / Mediant 500/L / Mediant 800); 200 (Mediant 1000)	500	500	500
Message Conditions	82	500	1,200	1,200
Message Policies	20	20	20	20
Pre-Parsing Manipulation Sets	10	10	10	10
Pre-Parsing Manipulation Rules	10 (per Set)	10 (per Set)	10 (per Set)	10 (per Set)
Quality of Experience Profile	256	256	256	256
Quality of Experience Color Rules	256	256	256	256
Bandwidth Profile	486	1,009	1,884	1,884
Quality Of Service Rules	510	3,125	7,500	7,500
IDS Policies	20	20	20	20
IDS Rule	100 (20 per Policy)	100 (20 per Policy)	100 (20 per Policy)	100 (20 per Policy)
IDS Matches	20	20	20	20
SIP Recording Rules	30	30	30	30
IP Interfaces	12	1,024	1,024	1,024
Ethernet Devices	16	1,024	1,024	1,024
Static Routes	30	30	30	30
HA Network Monitor	10	10	10	10
NAT Translation	32	32	32	32
TLS Contexts	12 (15 for Mediant 1000)	100	100	100
Firewall	50	500	500	500
QoS Mapping	64	64	64	64
Internal DNS	20	20	20	20
Internal SRV	10	10	10	10
Remote Web Services	7	7	7	7
HTTP Remote Hosts	10 (per Remote Web Service)	10 (per Remote Web Service)	10 (per Remote Web Service)	10 (per Remote Web Service)

Configuration Table	MP-1288 / Mediant 500 / Mediant 500L / Mediant 800 / Mediant 1000B	Mediant 2600 / Mediant 4000B	Mediant 90xx / Mediant SE	Mediant VE / Mediant CE
HTTP Proxy Servers	10	10	10	10
HTTP Locations	40	40	40	40
TCP/UDP Proxy Servers	10	10	10	10
Upstream Groups	10	10	10	10
Upstream Hosts	50 (5 per Upstream Group)	50 (5 per Upstream Group)	50 (5 per Upstream Group)	50 (5 per Upstream Group)
HTTP Directive Sets	30	30	30	30
HTTP Directives	500	500	500	500
OVOC Services	1	1	1	1
RADIUS Servers	3	3	3	3
LDAP Server Groups	41	250	600	600
LDAP Servers	82	500	1,200	1,200
DHCP Servers	1	1	1	1
Local Users	20	20	20	20
Access List	10	10	10	10
Additional Management Interfaces	16	64	64	64
SNMP Trap Destinations	5	5	5	5
SNMP Trusted Managers	5	5	5	5
SNMPv3 Users	10	10	10	10
Logging Filters	60	60	60	60
SBC CDR Format	128 (Syslog); 40 (RADIUS); 64 (Locally Stored & JSON)	128 (Syslog); 40 (RADIUS); 64 (Locally Stored & JSON)	128 (Syslog); 128 (RADIUS); 64 (Locally Stored & JSON)	128 (Syslog); 128 (RADIUS); 64 (Locally Stored & JSON)
Gateway CDR Format	128 (Syslog); 40 (RADIUS); 64 (Locally Stored & JSON)	-	-	-
Test Call Rules	5 (default)	5 (default)	5 (default)	5 (default)

4.4 Detailed Capacity

This section provides detailed capacity figures.

4.4.1 Mediant 500 E-SBC

The SBC session capacity and DSP channel capacity for Mediant 500 E-SBC are shown in the tables below.

Table 4-4: Mediant 500 E-SBC (Non-Hybrid) SBC Capacity

Hardware Configuration	TDM-RTP Sessions				Max. SBC Sessions (RTP-RTP)
	DSP Channels Allocated for PSTN	Wideband Coders			
		G.722	AMR-WB (G.722.2)	SILK-WB	
SBC	n/a	n/a	n/a	n/a	250

Table 4-5: Mediant 500 Hybrid E-SBC (with Gateway) Media & SBC Capacity

Hardware Configuration	TDM-RTP Sessions				Max. SBC Sessions (RTP-RTP)
	DSP Channels Allocated for PSTN	Wideband Coders			
		G.722	AMR-WB (G.722.2)	SILK-WB	
1 x E1/T1	30/24	√	-	-	220/226
	26/24	√	√	-	224/226
	26/24	√	√	√	224/226

4.4.2 Mediant 500L Gateway and E-SBC

The SBC session capacity and DSP channel capacity for Mediant 500L Gateway and E-SBC is shown in the tables below.

Table 4-6: Mediant 500L E-SBC (Non-Hybrid) SBC Capacity

Hardware Configuration	TDM-RTP Sessions			Max. SBC Sessions (RTP-RTP)
	DSP Channels Allocated for PSTN	Wideband Coders		
		G.722	AMR-WB (G.722.2)	
SBC	n/a	n/a	n/a	60

Table 4-7: Mediant 500L Hybrid E-SBC (with Gateway) Media & SBC Capacity

Hardware Configuration	DSP Channels Allocated for PSTN	Additional Coders				Max. SBC Sessions
		Narrowband	Wideband			
		Opus-NB	G.722	AMR-WB (G.722.2)	Opus-WB	
2 x BRI / 4 x BRI	4/8	-	-	-	-	56/52
	4/8	-	√	-	-	56/52
	4/6	√	-	√	-	56/54
	4	-	-	-	√	56

4.4.3 Mediant 800 Gateway & E-SBC

This section describes capacity for Mediant 800 Gateway & E-SBC.

4.4.3.1 Mediant 800A/B Gateway & E-SBC

The DSP channel capacity and SBC session capacity for Mediant 800A/B Gateway & E-SBC are shown in the tables below.

Table 4-8: Mediant 800A/B Gateway & E-SBC SBC Session Capacity per Capabilities (SBC Only)

H/W Configuration	DSP Channels for PSTN	SBC Transcoding Sessions								Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities						To Profile 1	To Profile 2	Mediant 800A	Mediant 800B
		Opus-NB	Opus-WB	AMR-NB / G.722	AMR-WB (G.722.2)	SILK-NB / iLBC	SILK-WB				
SBC	n/a	-	-	-	-	-	-	57	48	60	250
	n/a	-	-	√	-	-	-	51	42	60	250
	n/a	-	-	-	-	√	-	39	33	60	250
	n/a	-	-	-	√	-	-	36	30	60	250
	n/a	-	-	-	-	-	√	27	24	60	250
	n/a	√	-	-	-	-	-	27	24	60	250
	n/a	-	√	-	-	-	-	21	21	60	250

Note: "Max. SBC Sessions" for Mediant 800B applies to scenarios without registered users. When registered users are used, "Max. SBC Sessions" is reduced according to the main capacity table (see Section 4.1).

Table 4-9: Mediant 800A/B Gateway & E-SBC Channel Capacity per Capabilities (with Gateway)

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions									Conf. Participants	Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities							To Profile 1	To Profile 2		Mediant 800A	Mediant 800B
		AMR-NB / G.722	AMR-WB (G.722.2)	SILK-NB	SILK-WB	Opus-NB	Opus-WB	V.150.1					
2 x E1/T1	60/48	-	-	-	-	-	-	-	3/15	2/13	-	0/12	190/202
2 x T1	48	-	-	-	-	-	-	√	11	9	-	12	202
1 x E1/T1 &	38/32	-	-	-	-	-	-	-	22/28	18/22	-	22/28	212/218

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions									Conf. Participants	Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities							To Profile 1	To Profile 2		Mediant 800A	Mediant 800B
		AMR-NB / G.722	AMR-WB (G.722.2)	SILK-NB	SILK-WB	Opus-NB	Opus-WB	V.150.1					
8 x FXS/FXO Mix	38/32	-	-	√	-	-	-	-	8/12	7/11	-	22/28	212/218
1 x E1/T1	30/24	-	-	√	-	-		√	14/18	12/16	-	30/36	220/226
1 x E1 & 4 x BRI	38	-	-	-	-	-	-	-	22	18	-	22	212
1 x E1 & 4 x FXS	34	-	-	-	-	-	-	-	26	21	-	26	216
2 x E1 & 4 x FXS	64	-	-	-	-	-	-	-	0	0	-	0	186
4 x BRI & 4 x FXS & 4 x FXO	16	-	-	-	-	-	-	-	5	4	-	44	234
8 x BRI & 4 x FXS	20	-	-	-	-	-	-	-	1	1	-	40	230
8 x BRI	16	-	-	-	-	-	-	-	5	4	-	44	234
12 x FXS	12	-	-	√	-	-	-	√	3	3	-	48	238
4 x FXS & 8 x FXO	12	-	-	√	-	-	-	-	3	3	-	48	238
8 x FXS & 4 x FXO	12	-	-	√	-	-	-	-	3	3	-	48	238
4 x BRI & 4 x FXS	12	-	-	√	-	-	-	-	3	3	-	48	238
4 x FXS & 4 x FXO	8	-	-	-	-	-	-	-	7	5	6	52	242
	8	-	-	√	-	-	-	-	6	6	-	52	242
4 x BRI	8	-	-	-	-	-	-	-	7	5	6	52	242
	8	-	-	√	-	-	-	-	6	6	-	52	242
1/2/3 x BRI	2/4/6	-	-	-	-	-	-	-	17/15/14	14/13/11	-	58/56/54	248/246/244

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions									Conf. Participants	Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities							To Profile 1	To Profile 2		Mediant 800A	Mediant 800B
		AMR-NB / G.722	AMR-WB (G.722.2)	SILK-NB	SILK-WB	Opus-NB	Opus-WB	V.150.1					
	2/4/6	-	-	√	-	-	-	-	11/10/8	10/8/7	-	58/56/54	248/246/244
4 x FXS or 4 x FXO	4	-	-	√	-	-	-	√	10	8	-	56	246
	4	√	-	-	-	-	-	-	12	10	4	56	246
	4	-	-	√	-	-	-	-	6	6	4	56	246
	4	-	√	√	-	-	-	-	4	4	4	56	246
	4	-	√	√	√	-	-	-	3	3	4	56	246
	4	-	-	-	-	√	-	-	1	0	4	56	246
	4	-	-	-	-	-	√	-	0	0	3	56	246
FXS, FXO, and/or BRI, but not in use	0	-	-	-	-	-	-	-	19	16	-	60	250

Notes:

- "Max. SBC Sessions" for Mediant 800B applies to scenarios without registered users. When registered users are used, "Max. SBC Sessions" is reduced according to the main capacity table (see Section 4.1).
- *Profile 1*: G.711 at 20ms only, with In-band signaling (in voice channel) and Silence Suppression (no fax detection or T.38 support).
- *Profile 2*: G.711, G.726, G.729 (A / AB), and G.723.1, T.38 with fax detection, In-band signaling (in voice channel), and Silence Compression.
- All hardware assemblies also support the following DSP channel capabilities: echo cancellation (EC), CID (caller ID), RTCP XR reporting, and SRTP.
- SBC enhancements (e.g. Acoustic Echo Suppressor, Noise Reduction) are also available for these configurations. For more information, please contact your AudioCodes sales representative.
- Automatic Gain Control (AGC) and Answer Detector / Answer Machine Detector (AD/AMD) are also available for these configurations. For more information, please contact your AudioCodes sales representative.
- V.150.1 is supported only for the US Department of Defense (DoD).
- *Transcoding Sessions* represents part of the total SBC sessions.
- Conference Participants represents the number of concurrent analog ports in a three-way conference call.
- For availability of the telephony assemblies listed in the table above, please contact your AudioCodes sales representative.

4.4.3.2 Mediant 800C Gateway & E-SBC

The DSP channel capacity and SBC session capacity for Mediant 800C Gateway & E-SBC are shown in the tables below.

Table 4-10: Mediant 800C Gateway & E-SBC SBC Session Capacity per Capabilities (SBC Only)

H/W Configuration	SBC Transcoding Sessions								Max. SBC Sessions
	From Profile 2 with Additional Advanced DSP Capabilities						To Profile 1	To Profile 2	
	Opus-NB	Opus-WB	AMR-NB / G.722	AMR-WB (G.722.2)	SILK-NB / iLBC	SILK-WB			
SBC	-	-	-	-	-	-	114	96	400
	-	-	√	-	-	-	102	84	400
	-	-	-	-	√	-	78	66	400
	-	-	-	√	-	-	72	60	400
	-	-	-	-	-	√	54	48	400
	√	-	-	-	-	-	54	48	400
	-	√	-	-	-	-	42	42	400

Note: "Max. SBC Sessions" applies to scenarios without registered users. When registered users are used, "Max. SBC Sessions" is reduced according to the main capacity table (see Section 4.1).

Table 4-11: Mediant 800C Gateway & E-SBC SBC Session Capacity per Capabilities with Gateway

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions				Max SBC Sessions
		From Profile 2	From Profile 2 with SILK NB / iLBC	To Profile 1	To Profile 2	
4 x E1/T1 + 4 x FXS	124/100	√	-	2/23	2/18	276/300
2 x E1/T1 + 4 x FXS	64/52	√	-	0/10	0/8	336/348
Not in use		√	-	114	96	400
		-	√	78	66	400

Notes:

- "Max. SBC Sessions" applies to scenarios without registered users. When registered users are used, "Max. SBC Sessions" is reduced according to the main capacity table (see Section 4.1).
- *Profile 1*: G.711 at 20ms only, with In-band signaling (in voice channel) and Silence Suppression (no fax detection or T.38 support).
- *Profile 2*: G.711, G.726, G.729 (A / AB), and G.723.1, T.38 with fax detection, In-band signaling (in voice channel), and Silence Compression.
- All hardware assemblies also support the following DSP channel capabilities: echo cancellation (EC), CID (caller ID), RTCP XR reporting, and SRTP.
- SBC enhancements (e.g. Acoustic Echo Suppressor, Noise Reduction) are also available for these configurations. For more information, please contact your AudioCodes sales representative.
- Automatic Gain Control (AGC) and Answer Detector / Answer Machine Detector (AD/AMD) are also available for these configurations. For more information, please contact your AudioCodes sales representative.
- V.150.1 is supported only for the US Department of Defense (DoD).
- *Transcoding Sessions* represents part of the total SBC sessions.
- Conference Participants represents the number of concurrent analog ports in a three-way conference call.
- For availability of the telephony assemblies listed in the table above, please contact your AudioCodes sales representative.

4.4.4 Mediant 1000B Gateway & E-SBC

This section lists the channel capacity and DSP templates for Mediant 1000B Gateway & E-SBC DSP.

Notes:

- The maximum number of channels on any form of analog, digital, and MPM module assembly is 192. When the device handles both SBC and Gateway call sessions, the maximum number of total sessions is 150. When the device handles SRTP, the maximum capacity is reduced to 120.
- Installation and use of voice coders is subject to obtaining the appropriate license and royalty payments.
- For additional DSP templates, contact your AudioCodes sales representative.

4.4.4.1 Analog (FXS/FXO) Interfaces

The channel capacity per DSP firmware template for analog interfaces is shown in the table below.

Table 4-12: Channel Capacity per DSP Firmware Template for Mediant 1000B Analog Series

	DSP Template	
	0, 1, 2, 4, 5, 6	10, 11, 12, 14, 15, 16
	Number of Channels	
	4	3
Voice Coder		
G.711 A/Mu-law PCM	√	√
G.726 ADPCM	√	√
G.723.1	√	√
G.729 (A / AB)	√	√
G.722	-	√

4.4.4.2 BRI Interfaces

The channel capacity per DSP firmware template for BRI interfaces is shown in the table below.

Table 4-13: Channel Capacity per DSP Firmware Template for Mediant 1000B BRI Series

	DSP Template					
	0, 1, 2, 4, 5, 6			10, 11, 12, 14, 15, 16		
	Number of BRI Spans					
	4	8	20	4	8	20
	Number of Channels					
	8	16	40	6	12	30
Voice Coder						
G.711 A/Mu-law PCM	√			√		
G.726 ADPCM	√			√		
G.723.1	√			√		
G.729 (A / AB)	√			√		
G.722	-			√		

4.4.4.3 E1/T1 Interfaces

The channel capacity per DSP firmware template for E1/T1 interfaces is shown in the table below.

Table 4-14: Channel Capacity per DSP Firmware Templates for Mediant 1000B E1/T1 Series

	DSP Template																						
	0 or 10					1 or 11					2 or 12					5 or 15					6 or 16				
	Number of Spans																						
	1	2	4	6	8	1	2	4	6	8	1	2	4	6	8	1	2	4	6	8	1	2	4	6	8
	Number of Channels																						
Default Settings	31	62	120	182	192	31	48	80	128	160	24	36	60	96	120	24	36	60	96	120	31	60	100	160	192
With 128-ms Echo Cancellation	31	60	100	160	192	31	48	80	128	160	24	36	60	96	120	24	36	60	96	120	31	60	100	160	192
With IPM Features	31	60	100	160	192	-	-	-	-	-	-	-	-	-	-	-	-	-	-	-	31	60	100	160	192
Voice Coder																									
G.711 A-Law/M-Law PCM	✓					✓					✓					✓					✓				
G.726 ADPCM	✓					✓					✓					✓					-				
G.723.1	✓					-					-					-					-				
G.729 (A / AB)	✓					✓					✓					✓					✓				
GSM FR	✓					✓					-					-					-				
MS GSM	✓					✓					-					-					-				
iLBC	-					-					-					✓					-				
EVRC	-					-					✓					-					-				
QCELP	-					-					✓					-					-				
AMR	-					✓					-					-					-				
GSM EFR	-					✓					-					-					-				
G.722	-					-					-					-					✓				
Transparent	✓					✓					✓					✓					✓				

Note: "IPM Features" refers to Automatic Gain Control (AGC), Answer Machine Detection (AMD) and Answer Detection (AD).

4.4.4.4 Media Processing Interfaces

The transcoding session capacity according to DSP firmware template (per MPM module) is shown in the table below.

Notes:

- The device can be housed with up to four MPM modules.
- The MPM modules can only be housed in slots 1 through 5.

Table 4-15: Transcoding Sessions Capacity per MPM According to DSP Firmware Template for Mediant 1000B

	DSP Template				
	0 or 10	1 or 11	2 or 12	5 or 15	6 or 16
IPM Detectors Automatic Gain Control (AGC), Answer Machine Detection (AMD) and Answer Detection (AD)	**Number of Transcoding Sessions per MPM Module**				
-	24	16	12	12	20
✓	20	-	-	-	20
Voice Coder					
G.711 A-law / Mμ-law PCM	✓	✓	✓	✓	✓
G.726 ADPCM	✓	✓	✓	✓	-
G.723.1	✓	-	-	-	-
G.729 (A / AB)	✓	✓	✓	✓	✓
GSM FR	✓	✓	-	-	-
MS GSM	✓	✓	-	-	-
iLBC	-	-	-	✓	-
EVRC	-	-	✓	-	-
QCELP	-	-	✓	-	-
AMR	-	✓	-	-	-
GSM EFR	-	✓	-	-	-
G.722	-	-	-	-	✓
Transparent	✓	✓	✓	✓	✓

4.4.5 MP-1288 Analog Gateway & E-SBC

Session capacity includes Gateway sessions as well as SBC sessions without transcoding capabilities. The maximum capacity of Gateway sessions for MP-1288 Gateway & E-SBC is shown in the table below.

Table 4-16: MP-1288 Gateway Sessions Capacity

Coder	Gateway Sessions Capacity	
	Single FXS Blade	Fully Populated (4 x FXS Blades)
Basic: G.711, G.729 (A / AB), G.723.1, G.726 / G.727 ADPCM	72	288
G.722	72	288
AMR-NB	72	288
Opus-NB	60	240

Note:

- Quality Monitoring and Noise Reduction are not supported.
- SRTP is supported on all configurations.

4.4.6 Mediant 2600 E-SBC

The maximum number of supported SBC sessions is shown in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below:

Table 4-17: Transcoding Capacity per Coder-Capability Profile for Mediant 2600 E-SBC

Session Coders		Max. Sessions	
From Coder Profile	**To Coder Profile**	**Without MPM4**	**With MPM4**
Profile 1	Profile 1	400	600
Profile 2	Profile 1	300	600
Profile 2	Profile 2	250	600
Profile 1	Profile 2 + AMR-NB / G.722	275	600
Profile 2	Profile 2 + AMR-NB / G.722	225	600
Profile 1	Profile 2 + iLBC	175	575
Profile 2	Profile 2 + iLBC	150	500
Profile 1	Profile 2 + AMR-WB (G.722.2)	200	600
Profile 2	Profile 2 + AMR-WB (G.722.2)	175	525
Profile 1	Profile 2 + SILK-NB	200	600
Profile 2	Profile 2 + SILK-NB	175	525
Profile 1	Profile 2 + SILK-WB	100	350
Profile 2	Profile 2 + SILK-WB	100	350
Profile 1	Profile 2 + Opus-NB	125	425
Profile 2	Profile 2 + Opus-NB	125	375
Profile 1	Profile 2 + Opus-WB	100	300
Profile 2	Profile 2 + Opus-WB	75	275

Notes:

- *Profile 1:* G.711 at 20ms only, with in-band signaling (in voice channel), DTMF transcoding (RFC 2833 to in-band signaling), and Silence Suppression (no fax detection or T.38 support).
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38 with fax detection, in-band signaling (in voice channel), and Silence Compression.
- Acoustic Echo Suppressor reduces performance. For more information, contact your AudioCodes sales representative.
- MPM is the optional, Media Processing Module that provides additional DSPs, allowing greater capacity.

4.4.7 Mediant 4000 SBC

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below.

Table 4-18: Transcoding Capacity per Coder-Capability Profile for Mediant 4000 SBC

Session Coders		Max. Sessions	
From Coder Profile	**To Coder Profile**	**Without MPM8**	**With MPM8**
Profile 1	Profile 1	800	2,400
Profile 2	Profile 1	600	1,850
Profile 2	Profile 2	500	1,550
Profile 1	Profile 2 + AMR-NB / G.722	550	1,650
Profile 2	Profile 2 + AMR-NB / G.722	450	1,350
Profile 1	Profile 2 + iLBC	350	1,150
Profile 2	Profile 2 + iLBC	300	1,000
Profile 1	Profile 2 + AMR-WB (G.722.2)	400	1,200
Profile 2	Profile 2 + AMR-WB (G.722.2)	350	1,050
Profile 1	Profile 2 + SILK-NB	400	1,200
Profile 2	Profile 2 + SILK-NB	350	1,050
Profile 1	Profile 2 + SILK-WB	200	700
Profile 2	Profile 2 + SILK-WB	200	700
Profile 1	Profile 2 + Opus-NB	250	850
Profile 2	Profile 2 + Opus-NB	250	750
Profile 1	Profile 2 + Opus-WB	200	600
Profile 2	Profile 2 + Opus-WB	150	550

Notes:

- *Profile 1:* G.711 at 20ms only, with in-band signaling (in voice channel), DTMF transcoding (RFC 2833 to in-band signaling), and Silence Suppression (no fax detection or T.38 support).
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38 with fax detection, in-band signaling (in voice channel), and Silence Compression.
- Acoustic Echo Suppressor reduces performance. For more information, contact your AudioCodes sales representative.
- MPM is the optional, Media Processing Module that provides additional DSPs, allowing greater capacity.

The device can handle up to 5,000 fax detections, answer detections (AD), answering machine detections (AMD), beep detections, and Call Progress Tone detections, with the following assumptions:

- Timeout for fax detection is 10 seconds (default), after which fax detection is turned off and the call is resumed without the fax detector.
- Fax detection is applied on both legs of the SBC call.
- Minimum call duration is 100 seconds.
- AD, AMD, beep detection, and Call Progress Tone detection is only on one leg of the SBC call (should this not be the case, capacity figures will be reduced)

4.4.8 Mediant 4000B SBC

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below.

Table 4-19: Transcoding Capacity per Coder-Capability Profile for Mediant 4000B SBC

Session Coders		Number of Sessions				
From Coder Profile	To Coder Profile	Without MPM	1 x MPM8B	1 x MPM12B	2 x MPM12B	3 x MPM12B
Profile 1	Profile 1	800	2,400	3,250	5,000	5,000
Profile 2	Profile 1	600	1,850	2,450	4,350	5,000
Profile 2	Profile 2	500	1,550	2,100	3,650	5,000
Profile 1	Profile 2 + AMR-NB / G.722	550	1,650	2,200	3,850	5,000
Profile 2	Profile 2 + AMR-NB / G.722	450	1,350	1,800	3,150	4,550
Profile 1	Profile 2 + iLBC	400	1,200	1,600	2,850	4,050
Profile 2	Profile 2 + iLBC	350	1,050	1,400	2,500	3,600
Profile 1	Profile 2 + AMR-WB (G.722.2)	400	1,200	1,600	2,850	4,050
Profile 2	Profile 2 + AMR-WB (G.722.2)	350	1,050	1,400	2,500	3,600
Profile 1	Profile 2 + SILK-NB	400	1,200	1,600	2,850	4,050
Profile 2	Profile 2 + SILK-NB	350	1,050	1,400	2,500	3,600
Profile 1	Profile 2 + SILK-WB	200	700	950	1,650	2,400
Profile 2	Profile 2 + SILK-WB	200	700	950	1,650	2,400
Profile 1	Profile 2 + Opus-NB	250	850	1,150	2,000	2,850
Profile 2	Profile 2 + Opus-NB	250	750	1,050	1,800	2,600
Profile 1	Profile 2 + Opus-WB	200	600	850	1,500	2,150
Profile 2	Profile 2 + Opus-WB	150	550	750	1,300	1,900

Notes:

- *Profile 1:* G.711 at 20ms only, with In-band signaling (in voice channel), DTMF transcoding (RFC 2833 to in-band signaling), and Silence Suppression (no fax detection or T.38 support).
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, AMR-NB, T.38 with fax detection, In-band signaling (in voice channel), and Silence Compression.
- Acoustic Echo Suppressor reduces performance by about 30%. For more information, contact your AudioCodes sales representative.
- MPMB is the optional, Media Processing Module that provides additional DSPs, allowing greater capacity.

The device can handle up to 5,000 fax detections, answer detections (AD), answering machine detections (AMD), beep detections, and Call Progress Tone detections, with the following assumptions:

- Timeout for fax detection is 10 seconds (default), after which fax detection is turned off and the call is resumed without the fax detector.
- Fax detection is applied on both legs of the SBC call.
- Minimum call duration is 100 seconds.
- AD, AMD, beep detection, and Call Progress Tone detection is only on one leg of the SBC call (should this not be the case, capacity figures will be reduced)

4.4.9 Mediant 9000, Mediant 9000 Rev. B and Mediant 9080 SBC

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below.

Table 4-20: Transcoding Capacity per Coder-Capability Profile for Mediant 9000, Mediant 9000 Rev. B and Mediant 9080 SBC

Session Coders		Number of Sessions			
From Coder Profile	To Coder Profile	Without Hyper-Threading (Only Mediant 9000 SBC)		With Hyper-Threading	
		Extended	Basic	Extended	Basic
Profile 1	Profile 1	2,525	3,025	3,875	6,575
Profile 2	Profile 1	1,325	1,500	1,700	2,125
Profile 2	Profile 2	900	1,000	1,100	1,275
Profile 1	Profile 2 + AMR-NB / G.722	1,300	1,500	1,625	2,075
Profile 2	Profile 2 + AMR-NB / G.722	900	1,000	1,050	1,225
Profile 1	Profile 2 + AMR-WB (G.722.2)	475	500	575	600
Profile 2	Profile 2 + AMR-WB	400	425	475	500
Profile 1	Profile 2 + SILK-NB	1,175	1,300	1,450	1,700
Profile 2	Profile 2 + SILK-NB	825	900	975	1,100
Profile 1	Profile 2 + SILK-WB	750	775	950	1,000
Profile 2	Profile 2 + SILK-WB	600	625	725	750
Profile 1	Profile 2 + Opus-NB	750	825	900	1,050
Profile 2	Profile 2 + Opus-NB	600	650	700	775
Profile 1	Profile 2 + Opus-WB	575	625	700	800
Profile 2	Profile 2 + Opus-WB	475	525	575	625

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-21: Channel Capacity per Detection Feature for Mediant 9000, Mediant 9000 Rev. B and Mediant 9080 SBC

Detection Features	Maximum Sessions		
	Mediant 9000 SBC		Mediant 9000 Rev. B & Mediant 9080 SBC
	Without Hyper-Threading	With Hyper-Threading	
Fax Detection	24,000	40,000	45,000
AD/AMD/Beep Detection	24,000	39,000	45,000
CP Detection	24,000	44,000	45,000
Jitter Buffer	2,225	5,000	6,000

4.4.10 Mediant 9000, Mediant 9000 Rev. B, and Mediant 9080 SBC with Media Transcoders

Mediant 9000, Mediant 9000 Rev. B, or Mediant 9080 SBC with Media Transcoders allows increasing the number of transcoding sessions by using Media Transcoders. The maximum number of transcoding sessions depends on the following:

- Number of Media Transcoders in the media transcoding cluster.
- Cluster operation mode (Best-Effort or Full-HA mode).
- Maximum transcoding sessions. Each transcoding session is weighted as two RTP-RTP sessions without transcoding. Therefore, the number of sessions without transcoding plus the doubled number of sessions with transcoding must be less than the maximum RTP-RTP value specified in the table. As a result, if all sessions are with transcoding, the maximum number of sessions is half the maximum RTP-RTP sessions without transcoding as specified in Table 4-1.

The following table lists maximum transcoding sessions capacity of a single Media Transcoder.

Table 4-22: Transcoding Capacity per Profile for a Single Media Transcoder

Session Coders		Number of Sessions		
From Coder Profile	To Coder Profile	1 x MPM12B	2 x MPM12B	3 x MPM12B
Profile 1	Profile 1	2875	5000	5000
Profile 2	Profile 1	2300	4025	5000
Profile 2	Profile 2	1800	3175	4550
Profile 1	Profile 2 + AMR-NB / G.722	2000	3525	5000
Profile 2	Profile 2 + AMR-NB / G.722	1625	2850	4075
Profile 1	Profile 2 + AMR-WB (G.722.2)	1425	2500	3600
Profile 2	Profile 2 + AMR-WB (G.722.2)	1225	2175	3100
Profile 1	Profile 2 + SILK-NB	1425	2500	3600
Profile 2	Profile 2 + SILK-NB	1225	2175	3100
Profile 1	Profile 2 + SILK-WB	850	1500	2150
Profile 2	Profile 2 + SILK-WB	850	1500	2150
Profile 1	Profile 2 + Opus-NB	1050	1825	2625
Profile 2	Profile 2 + Opus-NB	950	1675	2400
Profile 1	Profile 2 + Opus-WB	750	1325	1900
Profile 2	Profile 2 + Opus-WB	650	1175	1675

Notes:

- *Profile 1:* G.711 at 20ms only, with In-band signaling (in voice channel) and Silence Suppression (no fax detection or T.38 support).
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, AMR-NB, T.38 with fax detection, In-band signaling (in voice channel), and Silence Compression.
- Acoustic Echo Suppressor reduces performance by about 30%. For more information, contact your AudioCodes sales representative.
- MPM12B is a Media Processing Module in the Media Transcoder that provides additional DSPs, allowing higher capacity.
- For best cluster efficiency, all Media Transcoders in the Cluster should populate the same number of MPM12Bs.
- The SBC employs load balancing of transcoding sessions among all Media Transcoders in the Cluster. Each Media Transcoder can handle up to 200 calls (transcoded sessions) per second (CPS).

4.4.11 Mediant 9030 SBC

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below.

Table 4-23: Transcoding Capacity per Coder-Capability Profile for Mediant 9030SBC

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	1,950	3,300
Profile 2	Profile 1	850	1,075
Profile 2	Profile 2	550	650
Profile 1	Profile 2 + AMR-NB / G.722	850	1,050
Profile 2	Profile 2 + AMR-NB / G.722	525	625
Profile 1	Profile 2 + AMR-WB (G.722.2)	300	300
Profile 2	Profile 2 + AMR-WB	250	250
Profile 1	Profile 2 + SILK-NB	725	850
Profile 2	Profile 2 + SILK-NB	500	550
Profile 1	Profile 2 + SILK-WB	475	500
Profile 2	Profile 2 + SILK-WB	375	375
Profile 1	Profile 2 + Opus-NB	450	525
Profile 2	Profile 2 + Opus-NB	350	400
Profile 1	Profile 2 + Opus-WB	350	400
Profile 2	Profile 2 + Opus-WB	300	325

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)

- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-24: Channel Capacity per Detection Feature for Mediant 9030 SBC

Detection Features	Maximum Sessions
Fax Detection	23,000
AD/AMD/Beep Detection	23,000
CP Detection	23,000
Jitter Buffer	3,000

4.4.12 Mediant Cloud Edition SBC

The Media Components (MC) in the media cluster of the Mediant CE must all be of the same instance type: either forwarding-only, or forwarding and transcoding. A maximum of 21 MCs can be used.

4.4.12.1 Mediant CE SBC for AWS EC2

4.4.12.1.1 Forwarding Sessions

The number of concurrent forwarding sessions per MC is shown in the following table.

Table 4-25: Forwarding Capacity per MC Instance Type

MC Instance Type	Number of Forwarding Sessions
r4.large	3,200
c4.4xlarge	3,200

Note: Forwarding performance was tested in AWS Ireland Region.

4.4.12.1.2 Transcoding Sessions

For transcoding capabilities, the Media Component (MC) must be of the AWS instance type c4.4xlarge. The number of supported transcoding sessions per MC is shown in the following table.

Table 4-26: Transcoding Capacity per c4.4xlarge MC

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	2,425	3,200
Profile 2	Profile 1	1,050	1,325
Profile 2	Profile 2	675	800
Profile 1	Profile 2 + AMR-NB / G.722	1,000	1,300
Profile 2	Profile 2 + AMR-NB / G.722	650	750
Profile 1	Profile 2 + AMR-WB (G.722.2)	350	375
Profile 2	Profile 2 + AMR-WB	275	300
Profile 1	Profile 2 + SILK-NB	900	1,050
Profile 2	Profile 2 + SILK-NB	600	675
Profile 1	Profile 2 + SILK-WB	575	625

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 2	Profile 2 + SILK-WB	450	450
Profile 1	Profile 2 + Opus-NB	550	650
Profile 2	Profile 2 + Opus-NB	425	475
Profile 1	Profile 2 + Opus-WB	425	500
Profile 2	Profile 2 + Opus-WB	350	375

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

4.4.12.2 Mediant CE SBC for Azure

4.4.12.2.1 Forwarding Sessions

The number of concurrent forwarding sessions per MC is shown in the following table.

Table 4-27: Forwarding Capacity per MC

MC VM Size	Number of Forwarding Sessions
DS3_v2	475

4.4.12.2.2 Transcoding Sessions

For transcoding capabilities, the Media Component (MC) must be of the Azure DS3_v2 virtual machine size. The number of supported transcoding sessions per MC is shown in the following table.

Table 4-28: Transcoding Capacity per DS3_v2 MC

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	475	475
Profile 2	Profile 1	275	350
Profile 2	Profile 2	175	225

Session Coders		Number of Sessions	
From Coder Profile	**To Coder Profile**	**Extended**	**Basic**
Profile 1	Profile 2 + AMR-NB / G.722	325	400
Profile 2	Profile 2 + AMR-NB / G.722	200	250
Profile 1	Profile 2 + AMR-WB (G.722.2)	100	125
Profile 2	Profile 2 + AMR-WB	75	100
Profile 1	Profile 2 + SILK-NB	275	300
Profile 2	Profile 2 + SILK-NB	175	200
Profile 1	Profile 2 + SILK-WB	150	175
Profile 2	Profile 2 + SILK-WB	125	125
Profile 1	Profile 2 + Opus-NB	150	200
Profile 2	Profile 2 + Opus-NB	125	125
Profile 1	Profile 2 + Opus-WB	125	150
Profile 2	Profile 2 + Opus-WB	100	100

4.4.13 Mediant Server Edition SBC

Note: Digital signal processing (DSP) is supported only on Mediant SE SBC based on DL360 G10.

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required, the number of sessions that can use DSP capabilities is reduced, as shown in the table below.

Table 4-29: Transcoding Capacity per Coder-Capability Profile for Mediant SE SBC Based on DL360 G10

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	3,875	6,575
Profile 2	Profile 1	1,700	2,125
Profile 2	Profile 2	1,100	1,275
Profile 1	Profile 2 + AMR-NB / G.722	1,625	2,075
Profile 2	Profile 2 + AMR-NB / G.722	1,050	1,225
Profile 1	Profile 2 + AMR-WB (G.722.2)	575	600
Profile 2	Profile 2 + AMR-WB	475	500
Profile 1	Profile 2 + SILK-NB	1,450	1,700
Profile 2	Profile 2 + SILK-NB	975	1,100
Profile 1	Profile 2 + SILK-WB	950	1,000
Profile 2	Profile 2 + SILK-WB	725	750
Profile 1	Profile 2 + Opus-NB	900	1,050
Profile 2	Profile 2 + Opus-NB	700	775
Profile 1	Profile 2 + Opus-WB	700	800
Profile 2	Profile 2 + Opus-WB	575	625

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-30: Channel Capacity per Detection Feature for Mediant SE SBC Based on DL360 G10

Detection Features	Maximum Sessions
Fax Detection	45,000
AD/AMD/Beep Detection	45,000
CP Detection	45,000
Jitter Buffer	6,000

4.4.14 Mediant Virtual Edition SBC

The maximum number of supported SBC sessions is listed in Section 4.1 on page 319. These SBC sessions also support SRTP and RTCP XR. When DSP capabilities are required (DSP Performance Profile), the number of sessions that can use DSP capabilities is reduced, as shown in the tables in this section.

4.4.14.1 Mediant VE SBC for OpenStack and VMware Hypervisors

The following tables list maximum channel capacity for Mediant VE SBC 2.8 GHz running on OpenStack or VMware hypervisors.

4.4.14.1.1 Two-vCPU Mediant VE SBC

The following table lists maximum channel capacity for the 2-vCPU (1 vCPU reserved for DSP) Mediant VE SBC.

Table 4-31: Transcoding Capacity for 2-vCPU Mediant VE SBC on OpenStack/VMware

Session Coders		Number of Sessions	
From Coder Profile	**To Coder Profile**	**Extended**	**Basic**
Profile 1	Profile 1	250	300
Profile 2	Profile 1	125	150
Profile 2	Profile 2	75	100
Profile 1	Profile 2 + AMR-NB / G.722	125	150
Profile 2	Profile 2 + AMR-NB / G.722	75	100
Profile 1	Profile 2 + AMR-WB (G.722.2)	25	50
Profile 2	Profile 2 + AMR-WB (G.722.2)	25	25
Profile 1	Profile 2 + SILK-NB	100	125
Profile 2	Profile 2 + SILK-NB	75	75
Profile 1	Profile 2 + SILK-WB	75	75
Profile 2	Profile 2 + SILK-WB	50	50
Profile 1	Profile 2 + Opus-NB	75	75
Profile 2	Profile 2 + Opus-NB	50	50
Profile 1	Profile 2 + Opus-WB	50	50
Profile 2	Profile 2 + Opus-WB	25	50

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-32: Channel Capacity per Detection Feature for 2-vCPU Mediant VE SBC on OpenStack/VMware

Special Detection Features	Number of Sessions
Fax Detection	2,400
AD/AMD/Beep Detection	2,400
CP Detection	2,400
Jitter Buffer	200

4.4.14.1.2 Four-vCPU Mediant VE SBC

The following table lists maximum channel capacity for the 4-vCPU (3 vCPUs reserved for DSP) Mediant VE SBC.

Table 4-33: Transcoding Capacity for 4-vCPU Mediant VE SBC on OpenStack/VMware

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	750	900
Profile 2	Profile 1	375	450
Profile 2	Profile 2	250	300
Profile 1	Profile 2 + AMR-NB / G.722	375	450
Profile 2	Profile 2 + AMR-NB / G.722	250	300
Profile 1	Profile 2 + AMR-WB	125	150
Profile 2	Profile 2 + AMR-WB	100	125
Profile 1	Profile 2 + SILK-NB	350	375
Profile 2	Profile 2 + SILK-NB	225	250
Profile 1	Profile 2 + SILK-WB	225	225
Profile 2	Profile 2 + SILK-WB	175	175
Profile 1	Profile 2 + Opus-NB	225	250
Profile 2	Profile 2 + Opus-NB	175	175
Profile 1	Profile 2 + Opus-WB	175	175
Profile 2	Profile 2 + Opus-WB	125	150

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-34: Channel Capacity per Detection Feature for 4-vCPU Mediant VE SBC on OpenStack/VMware

Special Detection Features	Number of Sessions
Fax Detection	7,200
AD/AMD/Beep Detection	7,200
CP Detection	7,200
Jitter Buffer	650

4.4.14.1.3 Eight-vCPU Mediant VE SBC

The following table lists maximum channel capacity for the 8-vCPU (4 vCPUs reserved for DSP) Mediant VE SBC.

Table 4-35: Transcoding Capacity for 8-vCPU Mediant VE SBC on OpenStack/VMware

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	1,000	1,200
Profile 2	Profile 1	525	600
Profile 2	Profile 2	350	400
Profile 1	Profile 2 + AMR-NB / G.722	525	600
Profile 2	Profile 2 + AMR-NB / G.722	350	400
Profile 1	Profile 2 + AMR-WB	175	200
Profile 2	Profile 2 + AMR-WB	150	150

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 2 + SILK-NB	475	500
Profile 2	Profile 2 + SILK-NB	325	350
Profile 1	Profile 2 + SILK-WB	300	300
Profile 2	Profile 2 + SILK-WB	225	250
Profile 1	Profile 2 + Opus-NB	300	325
Profile 2	Profile 2 + Opus-NB	225	250
Profile 1	Profile 2 + Opus-WB	225	250
Profile 2	Profile 2 + Opus-WB	175	200

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-36: Channel Capacity per Detection Feature for 8-vCPU Mediant VE SBC on OpenStack/VMware

Special Detection Features	Number of Sessions
Fax Detection	9,600
AD/AMD/Beep Detection	9,600
CP Detection	9,600
Jitter Buffer	875

4.4.14.2 Mediant VE SBC for Amazon AWS EC2

The following tables list maximum channel capacity for Mediant VE SBC on the Amazon EC2 platform.

Table 4-37: Transcoding Capacity for Mediant VE SBC on c4.2xlarge

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	1,164	1,524
Profile 2	Profile 1	618	750
Profile 2	Profile 2	420	498
Profile 1	Profile 2 + AMR-NB / G.722	492	570
Profile 2	Profile 2 + AMR-NB / G.722	354	408
Profile 1	Profile 2 + AMR-WB	174	180
Profile 2	Profile 2 + AMR-WB	156	162
Profile 1	Profile 2 + SILK-NB	438	486
Profile 2	Profile 2 + SILK-NB	324	366
Profile 1	Profile 2 + SILK-WB	270	288
Profile 2	Profile 2 + SILK-WB	222	240
Profile 1	Profile 2 + Opus-NB	276	312
Profile 2	Profile 2 + Opus-NB	228	258
Profile 1	Profile 2 + Opus-WB	216	228
Profile 2	Profile 2 + Opus-WB	186	198

Table 4-38: Transcoding Capacity for Mediant VE SBC on c4.8xlarge

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	3,200	3,200
Profile 2	Profile 1	3,200	3,200
Profile 2	Profile 2	2,225	2,650
Profile 1	Profile 2 + AMR-NB / G.722	2,600	3,025
Profile 2	Profile 2 + AMR-NB / G.722	1,875	2,175
Profile 1	Profile 2 + AMR-WB	925	950
Profile 2	Profile 2 + AMR-WB	825	850
Profile 1	Profile 2 + SILK-NB	2,325	2,575
Profile 2	Profile 2 + SILK-NB	1,725	1,950
Profile 1	Profile 2 + SILK-WB	1,425	1,525

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 2	Profile 2 + SILK-WB	1,175	1,275
Profile 1	Profile 2 + Opus-NB	1,450	1,650
Profile 2	Profile 2 + Opus-NB	1,200	1,375
Profile 1	Profile 2 + Opus-WB	1,150	1,200
Profile 2	Profile 2 + Opus-WB	975	1,050

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-39: Channel Capacity per Detection Feature for Mediant VE SBC on Amazon EC2

Special Detection Features	Number of Sessions	
	c4.2xlarge	c4.8xlarge
Fax Detection	2,000	3,200
AD/AMD/Beep Detection	2,000	3,200
CP Detection	2,000	3,200
Jitter Buffer	650	3,200

4.4.14.3 Mediant VE SBC for Azure

The following tables list maximum channel capacity for Mediant VE SBC on the Azure platform.

Table 4-40: Transcoding Capacity for Mediant VE SBC on DS1_v1, DS2_v2 and DS3_v2

Session Coders		Number of Sessions DS1_v2 and DS2_v2		Number of Sessions DS3_v2	
From Coder Profile	To Coder Profile	Extended	Basic	Extended	Basic
Profile 1	Profile 1	200	275	600	600
Profile 2	Profile 1	75	125	275	350
Profile 2	Profile 2	50	75	175	225
Profile 1	Profile 2 + AMR-NB / G.722	100	125	325	400
Profile 2	Profile 2 + AMR-NB / G.722	50	75	200	250
Profile 1	Profile 2 + AMR-WB (G.722.2)	25	25	100	125
Profile 2	Profile 2 + AMR-WB	25	25	75	100
Profile 1	Profile 2 + SILK-NB	75	100	275	300
Profile 2	Profile 2 + SILK-NB	50	50	175	200
Profile 1	Profile 2 + SILK-WB	50	50	150	175
Profile 2	Profile 2 + SILK-WB	50	50	125	125
Profile 1	Profile 2 + Opus-NB	50	75	150	200
Profile 2	Profile 2 + Opus-NB	25	50	125	125
Profile 1	Profile 2 + Opus-WB	25	50	125	150
Profile 2	Profile 2 + Opus-WB	25	25	100	100

4.4.14.4 Mediant VE SBC for Hyper-V Hypervisor

The following tables lists maximum channel capacity for Mediant VE SBC 2.1 GHz running on Hyper-V hypervisor.

4.4.14.4.1 Two-vCPU Mediant VE SBC

The following table lists maximum channel capacity for the 2-vCPU (1 vCPU reserved for DSP) Mediant VE SBC.

Table 4-41: Transcoding Capacity for 2-vCPU Mediant VE SBC on Hyper-V

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	175	225
Profile 2	Profile 1	100	100
Profile 2	Profile 2	50	75
Profile 1	Profile 2 + AMR-NB / G.722	75	100
Profile 2	Profile 2 + AMR-NB / G.722	50	75
Profile 1	Profile 2 + AMR-WB	25	25
Profile 2	Profile 2 + AMR-WB	25	25
Profile 1	Profile 2 + SILK-NB	75	75
Profile 2	Profile 2 + SILK-NB	50	50
Profile 1	Profile 2 + SILK-WB	50	50
Profile 2	Profile 2 + SILK-WB	25	25
Profile 1	Profile 2 + Opus-NB	50	50
Profile 2	Profile 2 + Opus-NB	25	25
Profile 1	Profile 2 + Opus-WB	25	25
Profile 2	Profile 2 + Opus-WB	25	25

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)

- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-42: Channel Capacity per Detection Feature for 2-vCPU Mediant VE SBC on Hyper-V

Special Detection Features	Number of Sessions
Fax Detection	1,800
AD/AMD/Beep Detection	1,800
CP Detection	1,800
Jitter Buffer	150

4.4.14.4.2 Four-vCPU Mediant VE SBC

The following table lists maximum channel capacity for the 4-vCPU (3 vCPUs reserved for DSP) Mediant VE SBC.

Table 4-43: Transcoding Capacity for 4-vCPU Mediant VE SBC on Hyper-V

Session Coders		Number of Sessions	
From Coder Profile	To Coder Profile	Extended	Basic
Profile 1	Profile 1	550	600
Profile 2	Profile 1	300	325
Profile 2	Profile 2	200	225
Profile 1	Profile 2 + AMR-NB / G.722	275	325
Profile 2	Profile 2 + AMR-NB / G.722	200	225
Profile 1	Profile 2 + AMR-WB	100	100
Profile 2	Profile 2 + AMR-WB	75	75
Profile 1	Profile 2 + SILK-NB	250	275
Profile 2	Profile 2 + SILK-NB	175	200
Profile 1	Profile 2 + SILK-WB	150	175
Profile 2	Profile 2 + SILK-WB	125	125
Profile 1	Profile 2 + Opus-NB	150	175
Profile 2	Profile 2 + Opus-NB	125	125
Profile 1	Profile 2 + Opus-WB	125	125
Profile 2	Profile 2 + Opus-WB	100	100

Notes:

- *Profile 1:* G.711 at 20ms only, without T.38 support.
- *Profile 2:* G.711, G.726, G.729 (A / AB), G.723.1, T.38.
- *Basic*: Excludes in-band signaling (in voice channel), VAD, Silence Suppression and fax detection.
- *Extended*: Includes DTMF transcoding (RFC 2833 to in-band signaling), VAD, Silence Suppression and fax detection
- Acoustic Echo Suppressor may reduce capacity. For more information, contact your AudioCodes sales representative.

The table below lists the maximum number of concurrent sessions per detection feature. The figures are based on the following assumptions:

- Timeout for fax detection is 10 seconds (default)
- Call duration is 100 seconds
- Fax detection is required on both legs of the SBC call
- Answer Detection (AD), Answering Machine Detection (AMD), Beep Detection, and Call Progress Tone Detection (CP) is only on one leg of the SBC call (should this not be the case, figures will be reduced)
- Sessions are only for forwarding sessions (i.e., no transcoding)

Table 4-44: Channel Capacity per Detection Feature for 4-vCPU Mediant VE SBC on Hyper-V

Special Detection Features	Number of Sessions
Fax Detection	5,400
AD/AMD/Beep Detection	5,400
CP Detection	5,400
Jitter Buffer	500

4.4.14.5 Mediant VE SBC with Media Transcoders

Mediant VE SBC with Virtual Media Transcoders allows increasing the number of transcoding sessions by using Media Transcoders. The maximum number of transcoding sessions depends on the following:

- The number of Media Transcoders in the media transcoding cluster.
- The cluster operation mode (Best-Effort or Full-HA mode).
- The maximum transcoding sessions that the Mediant VE SBC is capable of performing. Each transcoding session is weighted as two RTP-RTP sessions without transcoding. Therefore, the number of sessions without transcoding plus the doubled number of sessions with transcoding must be less than the maximum RTP-RTP value specified in the table. As a result, if all sessions are with transcoding, the maximum number of sessions is half the maximum RTP-RTP sessions without transcoding as specified in Table 4-1.

The following table lists maximum transcoding session capacity of a single MT-type Media Transcoder:

Table 4-45: Transcoding Capacity per Profile for Mediant VE SBC with Single MT

Session Coders		Number of Sessions		
From Coder Profile	**To Coder Profile**	**1 x MPM12B**	**2 x MPM12B**	**3 x MPM12B**
Profile 1	Profile 1	2,875	5,000	5,000
Profile 2	Profile 1	2,300	4,025	5,000
Profile 2	Profile 2	1,800	3,175	4,550
Profile 1	Profile 2 + AMR-NB / G.722	2,000	3,525	5,000
Profile 2	Profile 2 + AMR-NB / G.722	1,625	2,850	4,075
Profile 1	Profile 2 + AMR-WB (G.722.2)	1,425	2,500	3,600
Profile 2	Profile 2 + AMR-WB (G.722.2)	1,225	2,175	3,100
Profile 1	Profile 2 + SILK-NB	1,425	2,500	3,600
Profile 2	Profile 2 + SILK-NB	1,225	2,175	3,100
Profile 1	Profile 2 + SILK-WB	850	1,500	2,150
Profile 2	Profile 2 + SILK-WB	850	1,500	2,150
Profile 1	Profile 2 + Opus-NB	1,050	1,825	2,625
Profile 2	Profile 2 + Opus-NB	950	1,675	2,400
Profile 1	Profile 2 + Opus-WB	750	1325	1900
Profile 2	Profile 2 + Opus-WB	650	1175	1675

The following table lists maximum transcoding session capacity of a single vMT-type Media Transcoder:

Table 4-46: Transcoding Capacity per Profile for a Single vMT

Session Coders		Number of Sessions	
From Coder Profile	**To Coder Profile**	**Extended**	**Basic**
Profile 1	Profile 1	1,225	1,600
Profile 2	Profile 1	650	775
Profile 2	Profile 2	425	525
Profile 1	Profile 2 + AMR-NB / G.722	500	575
Profile 2	Profile 2 + AMR-NB / G.722	350	425
Profile 1	Profile 2 + AMR-WB	175	175
Profile 2	Profile 2 + AMR-WB	150	150
Profile 1	Profile 2 + SILK-NB	450	500
Profile 2	Profile 2 + SILK-NB	325	375
Profile 1	Profile 2 + SILK-WB	275	300
Profile 2	Profile 2 + SILK-WB	225	250
Profile 1	Profile 2 + Opus-NB	275	300
Profile 2	Profile 2 + Opus-NB	225	250
Profile 1	Profile 2 + Opus-WB	200	225
Profile 2	Profile 2 + Opus-WB	175	200

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5 Capacity for MSBRs

This section provides capacity figures per product.

5.1 SIP Signaling and Media Capacity

The following below lists maximum, concurrent SIP signaling sessions, concurrent media sessions, and registered users per product.

Table 5-1: SIP Signaling and Media Capacity per MSBR Product

Product	Signaling Capacity		Media Sessions			
	SIP Sessions	Registered Users	Session Type	RTP Sessions	SRTP Sessions	Detailed Media Capabilities
Mediant 500 MSBR	60	500	Hybrid	60	60	Table 5-3 Transcoding: n/a
			GW-Only	30	30	
Mediant 500L MSBR	60	200	Hybrid	60	60	Table 5-4 Transcoding: n/a
			GW-Only	8	8	
Mediant 800A MSBR	60	200	Hybrid	60	60	GW & Transcoding: Table 5-5
Mediant 800B MSBR	60	500	Hybrid	60	60	GW & Transcoding: Table 5-5
Mediant 800C MSBR	200	600	Hybrid	200	200	GW & Transcoding: Table 5-5

Notes:

- The figures listed in the table are accurate at the time of publication of this document. However, these figures may change due to a later software update. For the latest figures, please contact your AudioCodes sales representative.
- "GW" refers to Gateway functionality.
- The "SIP Sessions" column displays the maximum concurrent signaling sessions for both SBC and Gateway (when applicable). Whenever signaling sessions is above the maximum media sessions, the rest of the signaling sessions can be used for Direct Media.
- The "Session Type" column refers to Gateway-only sessions, SBC-only sessions, or Hybrid sessions which is any mixture of SBC and Gateway sessions under the limitations of Gateway-only or SBC-only maximum values.
- The "RTP Sessions" column displays the maximum concurrent RTP sessions when all sessions are RTP-RTP (for SBC sessions) or TDM-RTP (for Gateway sessions).
- The "SRTP Sessions" column displays the maximum concurrent SRTP sessions when all sessions are RTP-SRTP (for SBC sessions) or TDM-SRTP (for Gateway sessions).
- The "Registered Users" column displays the maximum number of users that can be registered with the device. This applies to the supported application (SBC or CRP).
- Regarding signaling, media, and transcoding session resources:
  - √ A signaling session is a SIP dialog session between two SIP entities, traversing the SBC and using one signaling session resource.
  - √ A media session is an audio (RTP or SRTP), fax (T.38), or video session between two SIP entities, traversing the SBC and using one media session resource.
  - √ A gateway session (i.e. TDM-RTP or TDM-SRTP) is also considered as a media session for the calculation of media sessions. In other words, the maximum Media Sessions specified in the table refer to the sum of Gateway and SBC sessions.
  - √ In case of direct media (i.e., Anti-tromboning / Non-Media Anchoring), where only SIP signaling traverses the SBC and media flows directly between the SIP entities, only a signaling session resource is used. Thus, for products with a greater signaling session capacity than media, even when media session resources have been exhausted, additional signaling sessions can still be handled for direct-media calls.
  - √ For call sessions requiring transcoding, one transcoding session resource is also used. For example, for a non-direct media call in which one leg uses G.711 and the other leg G.729, one signaling resource, one media session resource, and one transcoding session resource is used.

5.2 Session Capacity per Feature

The table below lists capacity per feature, per product:

Table 5-2: Capacity per Feature for MSBRs

Product	WebRTC Sessions	One-Voice Resiliency (OVR) Users	SIPRec Sessions
Mediant 500	-	-	30
Mediant 500L	-	-	30
Mediant 800B	-	-	30

Note: The figures in the table above for SIPRec capacity assume that there are no other concurrent, regular (non-SIPRec) voice sessions.

5.3 Detailed Capacity

This section provides detailed capacity figures.

5.3.1 Mediant 500 MSBR

The channel capacity and SBC session capacity for Mediant 500 MSBR are shown in the table below.

Table 5-3: Mediant 500 MSBR Capacity per PSTN Assembly and Capabilities

Telephony Interface Assembly	DSP Channels Allocated for PSTN	Max. SBC Sessions
1 x E1/T1	30/24	30/36
4 x BRI	8	52
1/2/3 x BRI	2/4/6	58/56/54
4 x FXS or 4 x FXO	4	56
FXS, FXO, and/or BRI, but none in use	0	60

Notes:

- All hardware assemblies also support the following DSP channel capabilities: echo cancellation (EC), CID (caller ID), RTCP XR reporting, and SRTP.
- Three-way conferencing is supported by the analog ports.
- For availability of the telephony assemblies listed in the table above, please contact your AudioCodes sales representative.

5.3.2 Mediant 500L MSBR

The channel capacity and SBC session capacity for Mediant 500L MSBR are shown in the table below.

Table 5-4: Mediant 500L MSBR Capacity per PSTN Assembly and Capabilities

Telephony Interface Assembly	DSP Channels Allocated for PSTN	Max. SBC Sessions
4 x FXS & 4 x FXO	8	52
2 x BRI & 2 x FXS	6	54
2 x BRI	4	56
4 x FXS	4	56
FXS, FXO, and/or BRI, but not in use	0	60

Notes:

- All hardware assemblies also support the following DSP channel capabilities: echo cancellation (EC), CID (caller ID), RTCP XR reporting, and SRTP.
- Three-way conferencing is supported by the analog ports.
- For availability of the telephony assemblies listed in the table above, please contact your AudioCodes sales representative.

5.3.3 Mediant 800 MSBR

The DSP channel capacity and SBC session capacity for Mediant 800 MSBR are shown in the table below.

Table 5-5: Mediant 800/B MSBR Channel Capacity per PSTN and Capabilities

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions								Conf. Participants	Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities						To Profile 1	To Profile 2		Mediant 800A	Mediant 800B
		IPM Detectors	G.722	AMR WB	SILK NB / iLBC	SILK WB	V.150.1					
2 x E1/T1	60/48	-	-	-	-	-	-	4/12	3/11	-	0/12	0/12
2 x T1	48	√	-	-	-	-	√	9	7	-	12	12
1 x E1/T1 & 8 x FXS/FXO Mix	38/32	√	-	-	-	-	-	16/21	14/18	-	22/28	22/28
	38/32	√	-	-	√	-	-	3/7	2/6	-	22/28	22/28
1 x E1/T1	30/24	√	-	-	√	-	√	9/14	7/11	-	30/36	30/36
1 x E1 & 4 x BRI	38	√	-	-	-	-	-	16	14	-	22	25
1 x E1 & 4 x FXS	34	√	-	-	-	-	-	19	16	-	26	26
2 x E1 & 4 x FXS	60	-	-	-	-	-	-	0	0	-	0	0
4 x BRI & 4 x FXS & 4 x FXO	16	√	-	-	-	-	-	3	2	-	44	44
8 x BRI & 4 x FXS	20	√	-	-	-	-	-	1	1	-	40	40
8 x BRI	16	√	-	-	-	-	-	3	2	-	44	44
12 x FXS	12	√	-	-	√	-	√	1	1	-	48	48
4 x FXS & 8 x FXO	12	√	-	-	√	-	-	1	1	-	48	48
8 x FXS & 4 x FXO	12	√	-	-	√	-	-	1	1	-	48	48
4 x BRI & 4 x FXS	12	√	-	-	√	-	-	1	1	-	48	48
4 x FXS & 4 x FXO	8	-	-	-	-	-	-	9	8	6	52	52
	8	√	-	-	√	-	-	4	3	-	52	52
4 x BRI	8	-	-	-	-	-	-	9	8	6	52	52

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions								Conf. Participants	Max. SBC Sessions	
		From Profile 2 with Additional Advanced DSP Capabilities						To Profile 1	To Profile 2		Mediant 800A	Mediant 800B
		IPM Detectors	G.722	AMR WB	SILK NB / iLBC	SILK WB	V.150.1					
	8	√	-	-	√	-	-	4	3	-	52	52
1/2/3 x BRI	2/4/6	-	-	-	-	-	-	13/12/10	12/11/10	-	56/52/48	56/52/48
	2/4/6	√	-	-	√	-	-	9/7/6	7/6/5	-	56/52/48	56/52/48
4 x FXS or 4 x FXO	4	-	-	-	√	-	√	7	6	-	56	56
	4	-	√	-	-	-	-	12	10	8	56	56
	4	-	-	-	√	-	-	8	7	7	56	56
	4	√	-	√	√	-	-	7	6	4	56	56
	4	√	-	√	√	√	-	5	4	4	56	56
FXS, FXO, and/or BRI, but not in use	0	-	-	-	-	-	-	15	14	-	60	60

Table 5-6: Mediant 800C MSBR Channel Capacity per PSTN and Capabilities

Telephony Interface Assembly	DSP Channels Allocated for PSTN	SBC Transcoding Sessions				Max SBC Sessions
		From Profile 2	From Profile 2 with SILK NB / iLBC	To Profile 1	To Profile 2	
4 x E1/T1 + 4 x FXS	124/100	√	-	2/23	2/18	76/100
2 x E1/T1 + 4 x FXS	64/52	√	-	0/10	0/8	136/148
Not in use		√	-	114	96	200
		-	√	78	66	200

Notes:

- *Profile 1*: G.711 at 20ms only, with In-band signaling (in voice channel) and Silence Suppression (no fax detection or T.38 support).
- *Profile 2*: G.711, G.726, G.729 (A / AB), G.723.1, and AMR-NB, T.38 with fax detection, In-band signaling (in voice channel), and Silence Compression.
- All hardware assemblies also support the following DSP channel capabilities: echo cancellation (EC), CID (caller ID), RTCP XR reporting, and SRTP.
- *IPM Detectors* includes Automatic Gain Control (AGC) and Answer Detector (AD).
- V.150.1 is supported only for the US Department of Defense (DoD).
- *Transcoding Sessions* represents part of the total SBC Sessions.
- *Conference Participants* represents the number of participants in one or more conference (bridge), where each conference may include three or more participants. Conferences are supported on all above configurations. For figures on the maximum number of participants per configuration, please contact your AudioCodes sales representative.
- For availability of the telephony assemblies listed in the table above, please contact your AudioCodes sales representative.

6 Supported SIP Standards

This section lists SIP RFCs and standards supported by the device.

6.1 Supported SIP RFCs

The table below lists the supported RFCs.

Table 6-1: Supported RFCs

RFC	Description	Gateway	SBC
draft-choudhuri-sip-info-digit-00	SIP INFO method for DTMF digit transport and collection	√	√
draft-ietf-bfcpbis-rfc4583bis-12	Session Description Protocol (SDP) Format for Binary Floor Control Protocol (BFCP) Streams	×	√ (forwarded transparently)
draft-ietf-sip-connect-reuse-06	Connection Reuse in SIP	√	√
draft-ietf-sipping-cc-transfer-05	Call Transfer	√	√
draft-ietf-sipping-realtimefax-01	SIP Support for Real-time Fax: Call Flow Examples	√	√ (forwarded transparently)
draft-ietf-sip-privacy-04.txt	SIP Extensions for Network-Asserted Caller Identity using Remote-Party-ID header	√	√
draft-johnston-sipping-cc-uui-04	Transporting User to User Information for Call Centers using SIP	√	√ (forwarded transparently)
draft-levy-sip-diversion-08	Diversion Indication in SIP	√	√
draft-mahy-iptel-cpc-06	The Calling Party's Category tel URI Parameter	√	√ (forwarded transparently)
draft-mahy-sipping-signaled-digits-01	Signaled Telephony Events in the Session Initiation Protocol	√	√
draft-sandbakken-dispatch-bfcp-udp-03	Revision of the Binary Floor Control Protocol (BFCP) for use over an unreliable transport	×	√ (forwarded transparently)
ECMA-355, ISO/IEC 22535	QSIG tunneling	√	√ (forwarded transparently)
RFC 2327	SDP	√	√
RFC 2617	HTTP Authentication: Basic and Digest Access Authentication	√	√
RFC 2782	A DNS RR for specifying the location of services	√	√
RFC 2833	Telephone event	√	√
RFC 2976	SIP INFO Method	√	√
RFC 3261	SIP	√	√

RFC	Description	Gateway	SBC
RFC 3262	Reliability of Provisional Responses	√	√
RFC 3263	Locating SIP Servers	√	√
RFC 3264	Offer/Answer Model	√	√
RFC 3265	(SIP)-Specific Event Notification	√	√
RFC 3310	Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)	√	×
RFC 3311	UPDATE Method	√	√
RFC 3323	Privacy Mechanism	√	√
RFC 3325	Private Extensions to the SIP for Asserted Identity within Trusted Networks	√	√
RFC 3326	Reason header	√	√ (forwarded transparently)
RFC 3327	Extension Header Field for Registering Non-Adjacent Contacts	√	×
RFC 3361	DHCP Option for SIP Servers	√	×
RFC 3362	Real-time Facsimile (T.38) - image/t38 MIME Sub-type Registration	√	√
RFC 3372	SIP-T	√	√ (forwarded transparently)
RFC 3389	RTP Payload for Comfort Noise	√	√ (forwarded transparently)
RFC 3420	Internet Media Type message/sipfrag	√	√
RFC 3455	P-Associated-URI	√	√ (using user info \ account)
RFC 3489	STUN - Simple Traversal of UDP	√	√
RFC 3515	Refer Method	√	√
RFC 3550	RTP: A Transport Protocol for Real-Time Applications	√	√
RFC 3578	Interworking of ISDN overlap signalling to SIP	√	×
RFC 3581	Symmetric Response Routing - rport	√	√
RFC 3605	RTCP attribute in SDP	√	√ (forwarded transparently)
RFC 3608	SIP Extension Header Field for Service Route Discovery During Registration	√	×
RFC 3611	RTCP-XR	√	√
RFC 3665	SIP Basic Call Flow Examples	√	√
RFC 3666	SIP to PSTN Call Flows	√	√ (forwarded transparently)
RFC 3680	A SIP Event Package for Registration (IMS)	√	×

RFC	Description	Gateway	SBC
RFC 3711	The Secure Real-time Transport Protocol (SRTP)	√	√
RFC 3725	Third Party Call Control	√	√
RFC 3824	Using E.164 numbers with SIP (ENUM)	√	√
RFC 3842	MWI	√	√
RFC 3891	"Replaces" Header	√	√
RFC 3892	The SIP Referred-By Mechanism	√	√
RFC 3903	SIP Extension for Event State Publication	√	√
RFC 3911	The SIP Join Header	Partial	×
RFC 3960	Early Media and Ringing Tone Generation in SIP	Partial	√
RFC 3966	The tel URI for Telephone Numbers	√	√
RFC 4028	Session Timers in the Session Initiation Protocol	√	√
RFC 4040	RTP payload format for a 64 kbit/s transparent call - Clearmode	√	√ (forwarded transparently)
RFC 4117	Transcoding Services Invocation	√	×
RFC 4168	The Stream Control Transfer Protocol (SCTP) as a Transport for SIP	×	√
RFC 4235	Dialog Event Package	Partial	Partial
RFC 4240	Basic Network Media Services with SIP - NetAnn	√	√ (forwarded transparently)
RFC 4244	An Extension to SIP for Request History Information	√	√
RFC 4320	Actions Addressing Identified Issues with SIP Non-INVITE Transaction	√	√
RFC 4321	Problems Identified Associated with SIP Non-INVITE Transaction	√	√
RFC 4411	Extending SIP Reason Header for Preemption Events	√	√ (forwarded transparently)
RFC 4412	Communications Resource Priority for SIP	√	√ (forwarded transparently)
RFC 4458	SIP URIs for Applications such as Voicemail and Interactive Voice Response	√	√ (forwarded transparently)
RFC 4475	SIP Torture Test Messages	√	√
RFC 4497 or ISO/IEC 17343	Interworking between SIP and QSIG	√	√ (forwarded transparently)
RFC 4566	Session Description Protocol	√	√
RFC 4568	SDP Security Descriptions for Media Streams for SRTP	√	√
RFC 4582	The Binary Floor Control Protocol (BFCP)	×	√ (forwarded transparently)

RFC	Description	Gateway	SBC
RFC 4715	Interworking of ISDN Sub Address to sip isub parameter	√	√ (forwarded transparently)
RFC 4730	A SIP Event Package for Key Press Stimulus (KPML)	Partial	×
RFC 4733	RTP Payload for DTMF Digits	√	√
RFC 4904	Representing trunk groups in tel/sip URIs	√	√ (forwarded transparently)
RFC 4960	Stream Control Transmission Protocol	×	√
RFC 4961	Symmetric RTP and RTCP for NAT	√	√
RFC 4975	The Message Session Relay Protocol (MSRP)	×	√
RFC 5022	Media Server Control Markup Language (MSCML)	√	×
RFC 5079	Rejecting Anonymous Requests in SIP	√	√
RFC 5627	Obtaining and Using Globally Routable User Agent (UA) URIs (GRUU) in SIP	√	√ (forwarded transparently)
RFC 5628	Registration Event Package Extension for GRUU	√	×
RFC 5806	Diversion Header, same as draft-levy-sip-diversion-08	√	√
RFC 5853	Requirements from SIP / SBC Deployments	-	√
RFC 6035	SIP Package for Voice Quality Reporting Event, using sip PUBLISH	√	√
RFC 6135	An Alternative Connection Model for the Message Session Relay Protocol (MSRP)	×	√
RFC 6140	Registration for Multiple Phone Numbers in the Session Initiation Protocol (SIP)	√	√
RFC 6337	Session Initiation Protocol (SIP) Usage of the Offer/Answer Model	-	√
RFC 6341	Use Cases and Requirements for SIP-Based Media Recording (Session Recording Protocol - draft-ietf-siprec-protocol-02, and Architecture - draft-ietf-siprec-architecture-03)	√	√
RFC 6442	Location Conveyance for the Session Initiation Protocol	-	√
RFC 7245	An Architecture for Media Recording Using the Session Initiation Protocol	√	√
RFC 7261	Offer/Answer Considerations for G723 Annex A and G729 Annex B	√	√
RFC 7865	Session Initiation Protocol (SIP) Recording Metadata	√	√
RFC 7866	Session Recording Protocol	√	√
RFC 8068	Session Initiation Protocol (SIP) Recording Call Flows	√	√

6.2 SIP Message Compliancy

The SIP device complies with RFC 3261, as shown in the following subsections.

6.2.1 SIP Functions

The device supports the following SIP Functions:

Table 6-2: Supported SIP Functions

Function	Comments
User Agent Client (UAC)	-
User Agent Server (UAS)	-
Proxy Server	The device supports working with third-party Proxy Servers such as Nortel CS1K/CS2K, Avaya, Microsoft OCS, Alcatel, 3Com, BroadSoft, Snom, Cisco and many others
Redirect Server	The device supports working with third-party Redirection servers
Registrar Server	The device supports working with third-party Registration servers

6.2.2 SIP Methods

The device supports the following SIP Methods:

Table 6-3: Supported SIP Methods

Method	Comments
INVITE	-
ACK	-
BYE	-
CANCEL	-
REGISTER	Send only for Gateway/IP-to-IP application; send and receive for SBC application
REFER	Inside and outside of a dialog
NOTIFY	-
INFO	-
OPTIONS	-
PRACK	-
UPDATE	-
PUBLISH	Send only
SUBSCRIBE	-

6.2.3 SIP Headers

The device supports the following SIP headers:

Table 6-4: Supported SIP Headers

SIP Header	SIP Header
Accept	Proxy- Authenticate
Accept–Encoding	Proxy- Authorization
Alert-Info	Proxy- Require
Allow	Prack
Also	Reason
Asserted-Identity	Record- Route
Authorization	Refer-To
Call-ID	Referred-By
Call-Info	Replaces
Contact	Require
Content-Disposition	Remote-Party-ID
Content-Encoding	Response- Key
Content-Length	Retry-After
Content-Type	Route
Cseq	Rseq
Date	Session-Expires
Diversion	Server
Expires	Service-Route
Fax	SIP-If-Match
From	Subject
History-Info	Supported
Join	Target-Dialog
Max-Forwards	Timestamp
Messages-Waiting	To
MIN-SE	Unsupported
P-Associated-URI	User- Agent
P-Asserted-Identity	Via
P-Charging-Vector	Voicemail
P-Preferred-Identity	Warning
Priority	WWW- Authenticate
Privacy	-

Note: The following SIP headers are not supported:

- Encryption
- Organization

6.2.4 SDP Fields

The device supports the following SDP fields:

Table 6-5: Supported SDP Fields

SDP Field	Name
v=	Protocol version number
o=	Owner/creator and session identifier
a=	Attribute information
c=	Connection information
d=	Digit
m=	Media name and transport address
s=	Session information
t=	Time alive header
b=	Bandwidth header
u=	URI description header
e=	Email address header
i=	Session info header
p=	Phone number header
y=	Year

6.2.5 SIP Responses

The device supports the following SIP responses:

Table 6-6: Supported SIP Responses

Response Type		Comments
1xx Response (Information Responses)		
100	Trying	The device generates this response upon receiving a Proceeding message from ISDN or immediately after placing a call for CAS signaling.
180	Ringing	The device generates this response for an incoming INVITE message. Upon receiving this response, the device waits for a 200 OK response.
181	Call is Being Forwarded	The device doesn't generate these responses. However, the device does receive them. The device processes these responses the same way that it processes the 100 Trying response.

Response Type		Comments
182	Queued	The device generates this response in Call Waiting service. When the SIP device receives a 182 response, it plays a special waiting Ringback tone to the telephone side.
183	Session Progress	The device generates this response if the Early Media feature is enabled and if the device plays a Ringback tone to IP
2xx Response (Successful Responses)		
200	OK	
202	Accepted	
3xx Response (Redirection Responses)		
300	Multiple Choice	The device responds with an ACK, and then resends the request to the first new address in the contact list.
301	Moved Permanently	The device responds with an ACK, and then resends the request to the new address.
302	Moved Temporarily	The device generates this response when call forward is used to redirect the call to another destination. If such a response is received, the calling device initiates an INVITE message to the new destination.
305	Use Proxy	The device responds with an ACK, and then resends the request to a new address.
380	Alternate Service	The device responds with an ACK, and then resends the request to a new address.
4xx Response (Client Failure Responses)		
400	Bad Request	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
401	Unauthorized	Authentication support for Basic and Digest. Upon receiving this message, the device issues a new request according to the scheme received on this response.
402	Payment Required	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
403	Forbidden	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
404	Not Found	The device generates this response if it is unable to locate the callee. Upon receiving this response, the device notifies the User with a Reorder Tone.
405	Method Not Allowed	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
406	Not Acceptable	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.

Response Type		Comments
407	Proxy Authentication Required	Authentication support for Basic and Digest. Upon receiving this message, the device issues a new request according to the scheme received on this response.
408	Request Timeout	The device generates this response if the no-answer timer expires. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
409	Conflict	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
410	Gone	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
411	Length Required	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
413	Request Entity Too Large	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
415	Unsupported Media	If the device receives a 415 Unsupported Media response, it notifies the User with a Reorder Tone. The device generates this response in case of SDP mismatch.
420	Bad Extension	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
423	Interval Too Brief	The device does not generate this response. On reception of this message the device uses the value received in the Min-Expires header as the registration time.
433	Anonymity Disallowed	If the device receives a 433 Anonymity Disallowed, it sends a DISCONNECT message to the PSTN with a cause value of 21 (Call Rejected). In addition, the device can be configured, using the Release Reason Mapping, to generate a 433 response when any cause is received from the PSTN side.
480	Temporarily Unavailable	If the device receives a 480 Temporarily Unavailable response, it notifies the User with a Reorder Tone. This response is issued if there is no response from remote.
481	Call Leg/Transaction Does Not Exist	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
482	Loop Detected	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
483	Too Many Hops	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
484	Address Incomplete	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.

Response Type		Comments
485	Ambiguous	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
486	Busy Here	The SIP device generates this response if the called party is off-hook and the call cannot be presented as a call waiting call. Upon receipt of this response, the device notifies the User and generates a busy tone.
487	Request Canceled	This response indicates that the initial request is terminated with a BYE or CANCEL request.
488	Not Acceptable	The device doesn't generate this response. Upon receipt of this message and before a 200 OK has been received, the device responds with an ACK and disconnects the call.
491	Request Pending	When acting as a UAS: the device sent a re-INVITE on an established session and is still in progress. If it receives a re-INVITE on the same dialog, it returns a 491 response to the received INVITE. When acting as a UAC: If the device receives a 491 response to a re-INVITE, it starts a timer. After the timer expires, the UAC tries to send the re-INVITE again.
5xx Response (Server Failure Responses)		
500	Internal Server Error	Upon receipt of any of these responses, the device releases the call, sending an appropriate release cause to the PSTN side. The device generates a 5xx response according to the PSTN release cause coming from the PSTN.
501	Not Implemented	
502	Bad gateway	
503	Service Unavailable	
504	Gateway Timeout	
505	Version Not Supported	
6xx Response (Global Responses)		
600	Busy Everywhere	Upon receipt of any of these responses, the device releases the call, sending an appropriate release cause to the PSTN side.
603	Decline	
604	Does Not Exist Anywhere	
606	Not Acceptable	

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International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
200 Cottontail Lane,
Suite A101E,
Somerset NJ 08873
Tel: +1-732-469-0880
Fax: +1-732-469-2298

Contact us: https://www.audiocodes.com/corporate/offices-worldwide
Website: https://www.audiocodes.com/

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