# Mediant<sup>™</sup> 1000

#### **SPECIFICATIONS**

Capacities			
Max. Signaling/Media Sessions	150	Max. SRTP/RTP Sessions	120
Max. Transcoding Sessions	96	Max. Registered Users	600
Telephony Interfaces			
Modularity and Capacity	6 slots for hosting voice processing and F	PSTN termination modules (up to 19	92 channels)
Digital Module	Up to 6 E1 or 8 T1/J1 spans provided on trunk modules. Each module supports 1, 2, or 4 E1/T1/J1 spans, with an option of PSTN Fallback		
Digital PSTN Protocols	Supporting various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. It also supports different variants of CAS protocols, including MFC R2, E&M immediate start, E&M delay dial / start and others.		
BRI Module	Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN Fallback. Providing S/T interfaces; NT or TE termination; 2W per port (power supplied)		
Analog Module	Up to 24 FXS/FX0 interfaces, provided on 4 ports FX0 / FXS modules, ground / loop start		
Media Processing Module	Up to 4 Media Processing modules (MPM	1), providing additional DSP resource	es
Network Interfaces			
Ethernet	Up to 6 GE interfaces configured in 1+1 r	redundancy or as individual ports	
Security			
Access Control	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting		
VolP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching		
Encryption/Authentication	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest		
Privacy	Topology hiding, user privacy		
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces		
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access		
Interoperability			
SIP B2BUA	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode		
SIP interworking	3xx redirect, REFER, PRACK, session time	er, early media, call hold, delayed of	fer
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users		
Transport Mediation	SIP over UDP/TCP/TLS, IPv4 / IPv6, RTP / SRTP (SDES)		
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)		
URI and Number Manipulations	URI user and host name manipulations, ingress and egress digit manipulation		
Transcoding and Vocoders	Order normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB/WB, G.727, I.BC, QCELP, GSM EFR		
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, V.34, packet-time conversion		
NAT	Local and far-end NAT traversal for support of remote workers		
Voice Quality and SLA			
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Call Admission Control	Based on bandwidth, session establishm	nent rate, number of connections/re	gistrations
•	Based on bandwidth, session establishm 802.1p/Q VLAN tagging, DiffServ, TOS	nent rate, number of connections/re	gistrations
Call Admission Control	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN		
Call Admission Control Packet marking	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911)	failure. Outbound calls can use PS1	
Call Admission Control Packet marking Standalone Survivability	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progi redundancy, broken connection detection	failure. Outbound calls can use PST rammable Jitter Buffer, Silence Sup	TN fallback for external connectivity
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc	failure. Outbound calls can use PS1 grammable Jitter Buffer, Silence Sup cellation, replacing voice profile due	I'N fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo cand voice gain control	failure. Outbound calls can use PS1 rammable Jitter Buffer, Silence Sup n cellation, replacing voice profile due	I'N fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring)	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece	failure. Outbound calls can use PS1 rammable Jitter Buffer, Silence Sup cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM)	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup to cellation, replacing voice profile due essary media delays and bandwidth to Manager (SEM) cements based on QoE and bandwidth	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhance	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup to cellation, replacing voice profile due essary media delays and bandwidth to Manager (SEM) cements based on QoE and bandwidth	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption
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Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo cand voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhance Ability to remotely verify connectivity, voice	failure. Outbound calls can use PSI rammable Jitter Buffer, Silence Sup bellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) between the based on QoE and bandwid be quality and SIP message flow between the based on LDAP, third-party routing of advanced LDAP, third-party routing of	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption  Ith utilization ween SIP UAs
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progi redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo candiolocie gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhance Ability to remotely verify connectivity, voice	failure. Outbound calls can use PSI prammable Jitter Buffer, Silence Sup a cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) perments based on QoE and bandwid per quality and SIP message flow between the party routing country of the party routing country to the party routing country of the party routing country to the party routing country of the party routing country routing country of the party routing coun	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption  Ith utilization ween SIP UAs control through REST API eters
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Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progredundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhanc Ability to remotely verify connectivity, voic Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque: Least-cost routing, call forking, load balar IETF standard SIP recording interface Browser-based GUI, CLI, SNMP, INI Config Embedded, Open Network Solution Platfo Up to 8GB RAM	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup 1 cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) cements based on QoE and bandwide equality and SIP message flow between the based on QoE and bandwide equality and SIP message flow between the company of th	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption  Ith utilization ween SIP UAs control through REST API eters
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory Storage Physical / Environmental	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progredundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhanc. Ability to remotely verify connectivity, voic  Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque: Least-cost routing, call forking, load balar IETF standard SIP recording interface  Browser-based GUI, CLI, SNMP, INI Config Embedded, Open Network Solution Platfo Up to 8GB RAM HDD or SSD	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup 1 cellation, replacing voice profile due essary media delays and bandwidth to Manager (SEM) cements based on QoE and bandwide equality and SIP message flow between the based on QoE and bandwide equality and SIP message flow between the company of t	IN fallback for external connectivity  pression/Comfort Noise Generation, RTP  to impairment detection, Fixed & dynamic  consumption  Ith utilization  ween SIP UAS  control through REST API  eters  gency call detection and prioritization
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Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory Storage Physical / Environmental Dimensions Mounting	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Prog- redundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc- voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhance Ability to remotely verify connectivity, voic  Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque- Least-cost routing, call forking, load balar IETF standard SIP recording interface  Browser-based GUI, CLI, SNMP, INI Config  Embedded, Open Network Solution Platfo Up to 8GB RAM HDD or SSD  1U x 320mm x 345mm (HxWxD)  Desktop or 19" mount  Operational: 0 to 40° C (32 to 104°F); St	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup n cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) cements based on QoE and bandwid ce quality and SIP message flow between the second	IN fallback for external connectivity pression/Comfort Noise Generation, RTP to impairment detection, Fixed & dynamic consumption  Ith utilization ween SIP UAs control through REST API sters gency call detection and prioritization  Approx. 9.7lb (4.4kg) Single power supply 100-240V, 50-60
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory Storage Physical / Environmental Dimensions Mounting Environmental	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progredundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo candivolde gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhance Ability to remotely verify connectivity, voic Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque) Least-cost routing, call forking, load balar IETF standard SIP recording interface Browser-based GUI, CLI, SNMP, INI Config Embedded, Open Network Solution Platfo Up to 8GB RAM HDD or SSD  1U x 320mm x 345mm (HxWxD) Desktop or 19" mount	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup n cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) cements based on QoE and bandwid ce quality and SIP message flow between the second	IN fallback for external connectivity  pression/Comfort Noise Generation, RTP  to impairment detection, Fixed & dynamic  consumption  Ith utilization  ween SIP UAs  control through REST API  eters  gency call detection and prioritization  Approx. 9.7lb (4.4kg)  Single power supply 100-240V, 50-60 Hz, 1.5M max. optional redundant
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory Storage Physical / Environmental Dimensions Mounting Environmental Regulatory Compliance	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progredundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhanc Ability to remotely verify connectivity, voic Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque: Least-cost routing, call forking, load balar IETF standard SIP recording interface Browser-based GUI, CLI, SNMP, INI Config Embedded, Open Network Solution Platfo Up to 8GB RAM HDD or SSD  1U x 320mm x 345mm (HxWxD) Desktop or 19" mount Operational: 0 to 40° C (32 to 104°F); SI Relative Humidity: 10 to 85% non-conder	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup 1 cellation, replacing voice profile due essary media delays and bandwidth the Manager (SEM) cements based on QoE and bandwid the quality and SIP message flow beth advanced LDAP, third-party routing c set, coder type, etc.), Layer-3 parame incing, E911 gateway support, emerging guration file, REST API, EMS form for third-party services  Weight Power torage: -20 to 70 °C (-4 to 158 °F) to residue of the control	IN fallback for external connectivity  pression/Comfort Noise Generation, RTP  to impairment detection, Fixed & dynamic  consumption  Ith utilization  ween SIP UAs  control through REST API  eters  gency call detection and prioritization  Approx. 9.7lb (4.4kg)  Single power supply 100-240V, 50-60 Hz, 1.5M max. optional redundant
Call Admission Control Packet marking Standalone Survivability Impairment Mitigation Voice Enhancement Direct Media (No Media Anchoring) Voice Quality Monitoring Quality of Experience Test agent SIP Routing Routing Methods Advanced Routing Criteria Routing Features SIPRec Management OAM&P OSN Server Platform (Optional) Single Chassis Integration Memory Storage Physical / Environmental Dimensions Mounting Environmental	802.1p/Q VLAN tagging, DiffServ, TOS Maintains local calls in the event of WAN (including E911) Packet Loss Concealment, Dynamic Progredundancy, broken connection detection Transrating, RTCP-XR, Acoustic echo canc voice gain control Hair-pinning of local calls to avoid unnece RTCP-XR, AudioCodes Session Experience Access control and media quality enhanc. Ability to remotely verify connectivity, voic Request URL, IP address, FQDN, ENUM, a QoE, bandwidth, SIP message (SIP reque: Least-cost routing, call forking, load balar IETF standard SIP recording interface  Browser-based GUI, CLI, SNMP, INI Config Embedded, Open Network Solution Platfo Up to 8GB RAM HDD or SSD  1U x 320mm x 345mm (HxWxD) Desktop or 19" mount Operational: 0 to 40° C (32 to 104°F); SI Relative Humidity: 10 to 85% non-conder	failure. Outbound calls can use PST grammable Jitter Buffer, Silence Sup 1 cellation, replacing voice profile due essary media delays and bandwidth e Manager (SEM) cements based on QoE and bandwid ce quality and SIP message flow beto educated LDAP, third-party routing c est, coder type, etc.), Layer-3 parame noing, E911 gateway support, emerg guration file, REST API, EMS comm for third-party services  Weight Power torage: -20 to 70°C (-4 to 158°F) sing	IN fallback for external connectivity  pression/Comfort Noise Generation, RTP  to impairment detection, Fixed & dynamic  consumption  Ith utilization  ween SIP UAs  control through REST API  eters  gency call detection and prioritization  Approx. 9.7 lb (4.4kg)  Single power supply 100-240V, 50-60  Hz, 1.54 max. optional redundant
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#### ABOUT AUDIOCODES

AudioCodes Ltd. (NasdagGS: AUDC) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Enterprise networks and Cable. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfect HDTM, relies on AudioCodes' leadership in DSP, voice coding and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.

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