Mediant™ 800

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 800 enterprise session border controller (E-SBC)** and media gateway offers a complete connectivity solution for small-to-medium sized enterprises.





In addition, the Mediant 800 supports up to 124 voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN.

400 SBC Sessions | 124 TDM Sessions | 1+1 High Availability | Certified SBC for Teams Direct Routing | Supports OPUS and SILK



Comprehensive interoperability

Proven interoperability with SIP trunks, SIP platforms and IP cloud services



Hybrid functionality

True hybrid SBC and gateway platform for gradual migration, low CAPEX and reduced space and power footprints



Enhanced security

Robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft



Superior voice quality

Advanced capabilities for optimizing and monitoring voice service quality



High resiliency

High availability using 1+1 redundancy, local branch survivability and PSTN fallback



Mediant™ 800

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Approx. 5.9510 (2.7kg) loaded Wi	<u> </u>	111 v 245mm v 220mm (Ll. MALE)) Wai-l-t		Approx 5 95lb (2 7kg) loaded with OSN
Agusting Dockton or 10" rack mount			,	Tomporaturo	
	nounting	Desktop or 19" rack mount Operating Temperature 5°-40° C Internal AC power supply rated: 100-240 VAC ~50- 60Hz 1.5A maximum			



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