



Product Name: Sonus Quintum Tenor AFG200 2FXS VoIP Gateway

Manufacturer: -

Model Number: AFG-200

Please Note: This product has been discontinued. Please see the Ribbon Communications Session Border Controllers range for an alternative.

Sonus Quintum Tenor AFG200 Gateway (AFG-200) Entry-Level Gateway
The Sonus Quintum Tenor AFG200 gateway allows small businesses, SOHOs, and branch offices
with analog telephony infrastructures the ability to utilise the VoIP network to reduce costs.
Key Features

� Telephony Interfaces: 2 x FXS

� Auxiliary Ports: 10/100 Ethernet, RS-232, DIAG

� Concurrent Calls: up to 2

าั¿½ Protocol Support: DHCP, DNS, H.323, SIP, SNMP, TCP, UDP

ī¿½ SIP Endpoints: up to 40 ī¿½ Chassis Type: 1U form factor ī¿½ Echo Cancellation: yes

The Sonus Quintum Tenor AFG200 Gateway is the simplest path to smarter, better communications for small offices and branch offices: superior Voice over IP sound quality, reliable security, cost-saving features and full integration with your existing PBXs and analog phones. Sonus Quintum Tenor AFG200 - Technical Specifications System Capabilities (per Switch)

ï¿⅓ Sessions

� Up to eight simultaneous VoIP calls � Maximum call rate: 900 calls per hour

រ៉េះ Business Continuity/Survivability(for AFM and AFT Series only)

ï¿⅓ SIP outbound proxy

ï¿1/2 Supports up to 40 SIP endpoints

ï¿1/2 AFG-200 = 2 VoIP calls = 2 FXS

Management Capabilities

12/2 Operations, Administration & Samp; Management

� Auto-provisionable

រ៉េ្ស GUI-based Configuration Manager for multiple, remote switches

īస్ట్ GUI-based alarm monitoring, call monitoring and CDR (call detail record) monitoring

ï¿1/2 Remote management via external

17.1/2 Remote Management Session Server

ï¿1/2 SNMPv2 Agent

រ៉េ្តែ Command Line Interface (CLI) configuration via Telnet or RS-232 connection

� Call Detail Record (CDR) generation

ï¿⅓ Authentication



آذًا IVR/RADIUS server support for AAA with integrated multilingual IVR+

� ANI authentication (Types 1 and 2)

### Media Services

- าั¿½ Auto codec negotiation: G.711, G.723.1, G.726, G.729a/b
- ï¿⅓ Modem support with G.711
- าั¿½ G.168 Echo Cancellation with standard 128 ms tail length
- � Adaptive Voice Activity Detection (VAD)
- آذِ 1/2 Comfort Noise Generation (CNG)
- ī¿½ Fax support: Group III at 2.4, 4.8, 7.2, 9.6 and 14.4 Kbps using industry standard T.38 (Super G3 compatible up to 33.6 Kbps)
- تزير Automatic call type detection voice, fax or modem

### Signaling

� Any-to-any connectivity: SIP<-&gt;SIP,SIP&lt;-&gt;H.323, H.323&lt;-&gt;H.323,

TDM<-&gt;TDM,SIP&lt;-&gt;TDM, H.323&lt;-&gt;TDM

- ï¿1/2 Tandem/TDM switching
- i¿1/2 Support for multiple SIP User Agents(RFC 3261)
- ï¿1/2 SIP supplementary services
- าั¿½ DTMF signaling via RFC 2833 or SIP Info
- าัง. SIP header manipulation: automatic appending and stripping of digits to dialed numbers
- � Loop Start, Reverse Battery and Battery Disconnect
- � Coding: A law, I law

## Protocol Support

- ï¿⅓ DHCP
- ï¿⅓ DNS
- H.323 ½ کرت
- SIP ½'5ï
- ï¿⅓ SNMP
- TCP ½ TCP
- UDP ½٪

## Routing/Policy

- 17.1/2 Transparent MultiPath Call Routing
- ī¿½ Least Cost Routing (via Sonus PSX™ Centralized Routing and Policy Server)
- � Forced IP routing and IP port mapping
- ī¿½ Four types of configurable routing databases: Bypass Directory Numbers, Hunt Local Directory Numbers, Hop-Off Directory Numbers and Static Routes
- تزرير Multiple dial plan options: public/private dial plan, user-programmable dial plan
- � Answer and disconnect supervision
- � Pass-through support for calls to tollfree, local and special service numbers (e.g., emergency services)
- آذِيًا Type 1 Caller ID/Name Support(Telcordia, ETSI, NTT and DTMF)
- ï¿⅓ SIP REFER method

## Security

ī¿½ NATAccess™ network address translation firewall protection



- � IP packet filtering
- ï¿1/2 G.168 Echo Cancellation with standard 128 ms tail length
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## Quality of Service

- � Adaptive jitter buffer
- � Packet loss concealment
- آذِر IP Type of Service (TOS) and DiffServ support
- ī¿½ PacketSaver™ multiplexing technology: Reduces bandwidth consumption up to 57% by combining voice packets from multiple calls into a single packet

## Package Features

- � Qty/CTN: 5 PCS � N.W/CTN: 6.3kg � G.W/CTN: 7.0kg
- � Giftbox size: 246mm\*223mm\*120mm � Carton Meas: 627mm\*256mm\*235mm

### Hardware SpecificationsFront Panel

- ï¿1/2 Status Indicators Front Panel LEDs
- ï¿⅓ Operational Status
- ï¿⅓ LAN Link
- � LAN Activity
- � Port(s) In Use

## Rear Panel

- ī¿½ Standard RJ-45 LAN Interface (IEEE 802.3) for 10 BASE-T or 100 BASE-TX connections
- تزايد Two, four, six or eight standard RJ-11 connectors for analog line/trunk interfaces
- � RS-232 connector for PC console
- � DIAG connector for software diagnostics

### Chassis

- آزٰ 1U, rack mount
- � Inches: 8.25 Wide x 2.0 High x 7.0 Deep
- าั¿1/2 Centimeters: 21.0 Wide x 5.1 High x 18.73 Deep

## AC Power Option

� 100-240 VAC, 50-60 Hz � Max power consumption: 22W

### Operating Altitude



� 10,000 ft. (3,000 m.)

Weight Maximum Fully Populated

� 1.3 lbs. (0.6 kg)

Environmental

� 5˚ to 40˚ C operating � -10˚ to 60˚ storage � 20 to 80% non-condensing operating humidity

**Industry Standards Support** 

آزا Telco: FCC Part 68, AS/ACIF S003,CS03, JATE AS/ACIF S002:2001

� EMC: FCC Part 15 Class B, EN55022,EN55024, EN61000-2-3, EN61000-3-3,AS/NZS3260

ï¿1/2 Safety: UL60950, EN60950, AS/NZS60950

Price: £480.00