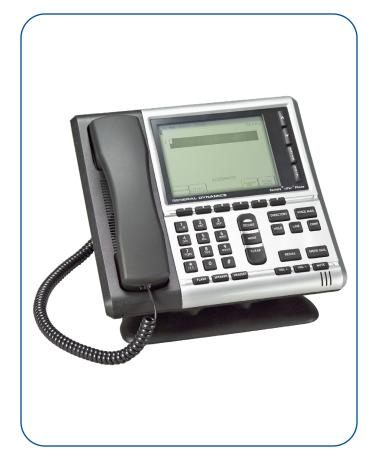
GENERAL DYNAMICS

Mission Systems

TalkSECURE™ vIPer™

Universal Secure Phone



Secure, Versatile, Reliable

Universal Secure Phone – VolP and Analog

In use with Cisco® Unified Communications Technology Networks

AES Encryption for SBU communications

Easy-to-use, cost-effective

Overview

The TalkSECURE™ vIPer™ phone provides the latest technology for secure, end-to-end Voice over IP and analog (PSTN) networks, using AES (Advanced Encryption Standard) encryption to protect SBU (Sensitive But Unclassified) communications. Built to provide seamless communications with legacy phones and encryption systems, the secure vIPer phone is software programmable with extensive memory to easily accommodate future upgrades and functionality.

Cost-Effective Solution for Today and Tomorrow

The TalkSECURE vIPer Phone provides high-quality, clear voice and data communications for VoIP and analog networks. The vIPer phone is a single desktop phone with integrated security and PIN-based access control.

TalkSECURE vIPer features include:

- Embedded fiber interface for direct connection to fiber networks.
- Analog (PSTN) and VoIP (Cisco Skinny Call Control Protocol [SCCP] or Session Initiation Protocol [SIP]) connectivity.
- Low satcom latency for satellite communications
- Secure data and fax transfer
- Low power
- Free software upgrades
- Precedence and preemption

Easy to Use and Manage

Simple to install, the vIPer phone can be set up out of the box within minutes. The large, easy-to-read display is intuitive and user-friendly, as is the web-based management tool.

TalkSECURE™ vIPer™

Benefits/Features

Easy-to-Use

- Switch between analog (PSTN) and VolP networks via the easy-to-use menu
- Fast secure call set-up
- Web-based GEM[™] One administration
- Fast, touch-free software upgrades for non-secure call features
- Supports DHCP for fast set-up

State-of-the-Art Technology

- Secure data transfer to other SCIP-capable devices (key material, secure fax)
- Integrated security no Fortezza card required
- Commercial open standards
- Powered over Ethernet or AC
- MIL-STD-810F for temperature, humidity, vibration, shock and altitude

Non-secure Call Features

Dialing

- Corporate Directory (Cisco CUCM)
- Directory (200 entries)
- Speed Dial (10 entries)
- Inbound Call List (50 entries)
- Outbound Call List (50 entries)
- Last Number Redial

Visual Display

- LCD display with backlight
- Time and Date (dependent on network)
- Footstand adjustment for display angle

■ Audio Control

- Selectable ringtone
- Speakerphone
- Headset capable
- Volume controls
- Mute
- Hearing aid compatible



Technical Specifications

Size

- Width: 10 in.
- Depth: 3 in. (without footstand)
- Length: 9.5 in.
- Weight: 4.5 lbs (with footstand)
- Volume: 285 cu in.

Powe

 Powered over Ethernet (802.3af) -or- AC power 100 to 240 VAC, 50-60 Hz 8 Watts maximum operating

Environment

- MIL-STD-810F (temperature, humidity, vibration, shock and altitude)
- Operational: 0°C to 50°C (32°F to 122°F)
- Storage: -30°C to 80°C (-22°F to 176°F
- Humidity: 95% (non-condensing
- Altitude: Sea level up to 40,000 ft. (non-operating);
 Sea level up to 10,000 ft. (operating)

Black Interfaces

- 10/100BaseT to LAN/WAN
- 10/100BaseT to Black Computer
- 100Base-FX Fiber Interface -1300/1310 nm wavelength LED
- -62.5/125 and 50/125 mm multimode LC type connector
- USB port

Red Interfaces

- RS-232 data port for DS-101 key fill and data transfer
- USB port

■ Secure Data Rate

■ 100+ kb/s

■ Speech Processing

- Non-secure: G.711¹, G.723.1², G.726², G.729AB
- Secure: G.729D, MELP

Approvals

- TEMPEST (approval pending)
- Safety: CB Scheme IEC 62368-1
- EMI/EMC: FCC Part 15 subpart B, Class B, EN-55032 (IP modes)
- TSG
 - VIPC Models: PSTN/VoIP approved
 - VIPF Models: PSTN/VoIP approved
- JITC:
 - VIPC Models: PSTN/SIP/SCCP approved
 - VIPF Models: PSTN/SIP/SCCP approved

Secure Dial

■ Transmit/Receive: Yes

■ VoIP Network Protocol Support

- Cisco SCCP (Skinny Call Control Protocol
- SIP (Session Initiation Protocol)
- IPv4, IPv6
- DHCP, DSCP, RTP, TLS/SRTP, LLDP, DNS, TFTP, HTTP, TCP, UDP, MoIP, E.164, SDP

SIP Info

Avaya

- Aura Application Server 5300
 - M800, MG3000 Gateways⁵
- Avaya Aura Communication Manager (minimum release Aura 6.3-FP4)
 - G450 Media Gateway (with MP 160 media module)⁴

RIBBON (formerly GENBAND)

- C20 Call Session Controller (min release SE17, EXPERIUS 11.2)
- Gateway platforms: VX900, VX1200, VX1800, (requires 4.7.4v17 or higher)

REDCOM

 SIP Server & Gateway platforms: High Density Exchange (HDX•C), SLICE® 2100™ (requires 4.0AR3P8 or higher)

NFC

- NEC Univerge 3C Release 8.6.1.14
 - MP112, 124, M800, MG3000⁵

SCCP Info

■ Cisco Call Manager³

■ 10.5.2 SU3

Cisco Routers

- 2811, 2821, 2851 (requires IOS: 12.4(20)T1⁴ or higher)
- 2911, 2921, 2951 (requires IOS: 15.1(4)M3 or higher)
- **3725, 3745**
- (IOS: 12.4(15)T14 or higher recommended)
- 3825, 3845 (IOS: 12.4(20)T1⁴ or higher recommended)
- 3925, 3945 (requires IOS: 15.1(4)M3 or higher)

Cisco Gateway Cards

- NM-HDv2-1 (T1/E1)
- NM-HDv2-2 (T1/E1)
- VWIC2-1MFT (T1/E1)VWIC2-2MFT (T1/E1)

1 Both G.711 a-law and G.711 µ-law are supported.

2 Not currently supported by Cisco Call Manager.

3 Use of this phone with a Cisco Call Manager System requires an

additional license from Cisco

4 Advanced Enterprise Services image required.
5 V.150.1 support for V.32 and V.34 modulations

GENERAL DYNAMICS

Mission Systems

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