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.

**Overview**

Secure, Versatile, Reliable

**Universal Secure Phone**

In use with Cisco® Unified Communications Technology Networks

AES Encryption for SBU communications

Easy-to-use, cost-effective
Benefits/Features

Easy-to-Use
- Switch between analog (PSTN) and VoIP networks via the easy-to-use menu
- Fast secure call set-up
- Web-based GEMX 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
- 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.

Power
- 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.711, G.723.1, G.728, G.729AB
- Secure: G.729d, MELP

Approvals
- TEMPEST (approval pending)
- Safety: CB Scheme - IEC 62368-1
- TSG:
  - VPC Models: PSTN/VoIP approved
  - VPV Models: PSTN/VPV approved
  - JITC:
  - VPC Models: PSTN/VPIL approved; SCCP approval pending
  - VPV Models: PSTN/SIP/SCCP approval pending

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
  - Avaya Aura Communication Manager (minimum release Aura 6.2 FP2)
    - G450 Media Gateway (with MP 160 media module)®
- GENBAND
  - C20 Call Session Controller (min release SE17, EXPERIUS 11.2)
- NET
  - SIP Server & 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)

SCCP Info
- Cisco Call Manager
  - 4.2(S)/R3 and higher recommended
- 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)
  - VWIC2-2MFT (T1/E1)
  - VWIC2-2MFT (T1/E1)

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1 Both G.711 a-law and G.711 u-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 Supported for V.32 and V.34 modulations