Cost-effective solution for delivering advanced IP communications by enabling interoperability between a converged network and legacy equipment

OVERVIEW

3Com® VoIP gateways help enterprises migrate to IP communications using an economical overlay strategy that aligns financial considerations with performance needs and lays the foundation for a successful deployment of Session Initiation Protocol (SIP)-based applications. The gateways enable interoperability between legacy equipment and applications—including PBXs, the public switched telephone network (PSTN), analog phones and fax machines—and a converged IP network. 3Com digital gateways provide all necessary functions and protocol support for handling voice calls between traditional circuit-switched phone networks and an IP network. 3Com analog gateways, including both FXS and FXO models, let analog phones and other legacy analog devices use 3Com SIP-based IP telephony features.

KEY BENEFITS

MAINTAIN UNIVERSAL CONNECTIVITY
With 3Com VoIP gateways, organizations can retain PBXs and analog devices for operational or budgetary reasons while they implement IP telephony in a phased overlay. Employees can continue to use familiar equipment and interfaces, including dialing plans, email systems, fax machines and modems. The SIP-compliant gateways may be positioned between the IP network and any brand of PBX to deliver calls or applications such as the SIP-based conferencing and call center capabilities of the 3Com Convergence Applications Suite.

IMPLEMENT REMOTE SITE SURVIVABILITY
In environments where remote branch sites are dependent on a single wide-area IP network connection to the 3Com IP Telephony Module, a local analog gateway enables incoming and outgoing calls to the PSTN. This capability provides telephony survivability and business continuity for the remote site even if its IP connection has been compromised.

REDUCE TELEPHONY COSTS
As part of a converged communications solution, the gateways let enterprises reduce PSTN costs by implementing least-cost routing features delivered by the 3Com IP telephony platform. To minimize long distance calling charges, calls can be routed to the most cost-effective gateway. Plus, remote sites can retain PSTN connections for supporting local Direct Inward Dialing (DID or DDI) numbers, and for receiving the lowest local call tariffs from their communications service provider.
FEATURE HIGHLIGHTS

Enable voice and fax calls from legacy PBX systems and PSTN services to be connected to SIP-based phones and applications.

Support consolidation of PSTN connections and optimize use of toll-bypass.

Use SIP signaling, allowing communications with 3Com VoIP applications.

Dynamically support the latest compression algorithms to optimize bandwidth management.

Implement echo control and a dynamic programmable jitter buffer for high voice quality.

Allow analog phones or fax machines to connect to 3Com VoIP systems [FXS analog gateways].

Allow calls from the 3Com VoIP systems to analog PBXs or PSTN lines [FXO analog gateways].

Scale to handle up to 480 simultaneous voice calls in a single gateway, modular and fixed digital gateway models support from one to 16 T1/E1 lines and are able to use a variety of PRI and CAS protocols.

Easily expand traffic capacity via modular digital gateway.

SPECIFICATIONS

ENCLOSURES

Analog Gateways

Desktop or half-width 19-inch rack-mount unit for 4-port and 8-port gateways, full-width 19-inch rack-mount unit for 24-port gateways; all 1U high.

Digital Gateways

19-inch rack-mount unit; 1U high.

INTERFACES

FXS Analog Gateways

Support for 4, 8, or 24 analog phone loop-start FXS ports; 4- and 8-port gateways use RJ1 connectors, 24-port gateways use single Telco connector (50-pin Life Line connected to the unused pins on port 4 with relay to an analog line, even if the gateway is powered off).

FXO Analog Gateways

Support for 4 or 8 analog FXO ports using RJ1 connectors.

Modular Digital Gateways

Support for 4/2/4 T1/E1 digital span modules; to enable support of 3Com VoIP applications, upgradeable with CPU module.

Fixed Configuration Digital Gateways

Support for 4/8/16 T1/E1 digital spans (for LAN redundancy, 10/100BASE-T connections to the IP network).

SIP FUNCTIONALITY

SIP (RFC 3261) support by performing a SIP User Agent (UA) role.

VOICE ENCODING


FAX SUPPORT

T.38 with round-trip delay up to 9 seconds.

SIGNALING PROTOCOLS (DIGITAL GATEWAYS)

PRI or CAS including MFC/R2, E&M immediate start, E&M delay dial/start, loop start, ground start.

ORDERING INFORMATION

PRODUCT DESCRIPTION

Analog Gateways

V7111 Analog Media Gateway [FXS] 2 Channels
V7111 Analog Media Gateway [FXS] 4 Channels
V7111 Analog Media Gateway [FXS] 8 Channels
V7111 Analog Media Gateway [FXS] 16 Channels
V7111 Analog Media Gateway [FXO] 2 Channels
V7111 Analog Media Gateway [FXO] 4 Channels
V7111 Analog Media Gateway [FXO] 8 Channels
V7111 Analog Media Gateway [FXO] 16 Channels
V7111 Analog [2 FXS, 2 FXO] Media Gateways
V7111 Analog [4 FXS, 4 FXO] Media Gateways

Digital Gateways and Power Module

V6100 Digital Gateway Chassis [maximum four spans]*
V6100 Digital Single-Span Module
V6100 Digital Two-Span Module
V6100 Digital Four-Span Module
V6100 Redundant Power Supply Module (optional)
VoiP Gateway - Single Span
VoiP Gateway - Two Span
VoiP Gateway - Four Span
VoiP Gateway - Eight Span
VoiP Gateway - Sixteen Span

*Upgradeable to the V6100 Integrated Branch Communications Platform.

3COM SKU

3CRV711105-07
3CRV711107-07
3CRV711111-07
3CRV711113-07
3CRV711114-07
3CRV711115-07
3CRV711116-07
3CRV711117-07
3CRV71225-07
3CRV71226-07
3CRV71227-07
3CRV71228-07
3CRV60005A-06
3CRV71220-07
3CRV71221-07
3CRV71222-07
3CRV71223-07
3CRV71224-07

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