Digium® VoIP Gateways

Robust features and effortless set-up at a great price

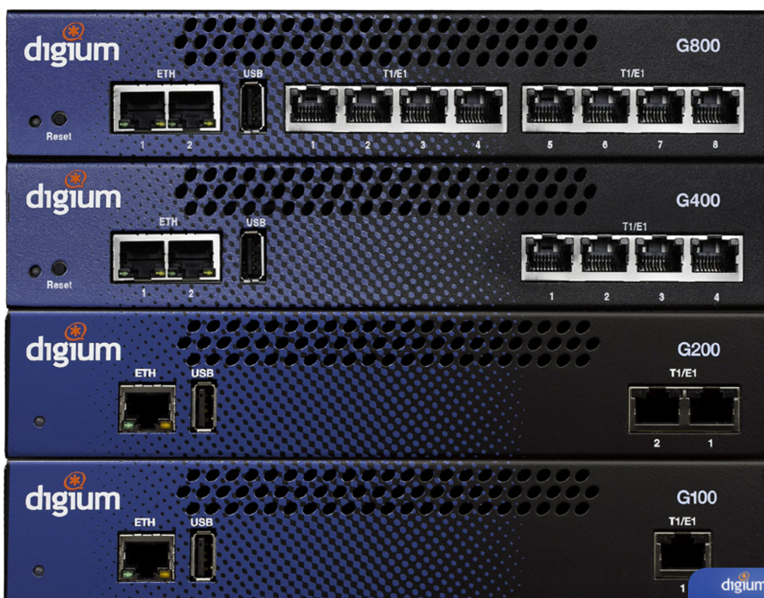
digium®
The Asterisk Company

Built on a powerful combination of the Asterisk Open Source communications engine and a state-of-the-art embedded platform, Digium VoIP Gateways provide the best value for connecting disparate topologies of traditional telephony (T1/E1/PRI) to IP (SIP).

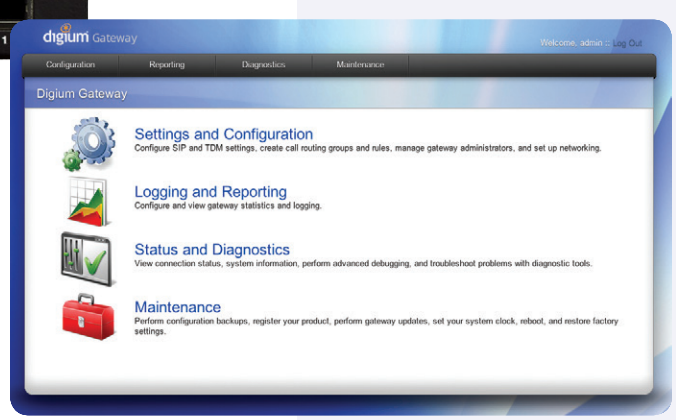
The gateway software is based on Asterisk and is managed through Digium's intuitive point-and-click graphical user interface (GUI), which allows for easy navigation and effortless setup. The gateways feature a power-saving embedded design with a highly efficient digital signal processor (DSP) handling all media-related operations. With no moving parts and built to last, they can be deployed in the toughest environments.

The Digium Gateways robust feature set includes the ability to configure calling rules for connecting many combinations of telephony providers (traditional and VoIP) and PBX's (legacy and VoIP), failover routing to ensure calls won't fail, codec and fax licensing for the maximum number calls each appliance supports, software-selectable T1/E1/PRI interfaces, and VLAN tagging.

Deployed in any application, Digium gateways will provide proven reliability for a fraction of the cost of other gateway platforms on the market. The easy to setup gateways and API will reduce administration and troubleshooting costs for any organization, from a small business to a large enterprise.



The Digium family of gateways



Digium's intuitive point-and-click GUI allows for easy navigation and effortless setup.

Features:

- Available in 1-, 2-, 4- or 8-port T1/E1/PRI
- Easy-to-navigate GUI
- Intelligent call routing
- Fax and modem support
- No moving parts
- Remote configuration and software download
- Cost-effective
- Low power consumption
- Octasic® DSP processor

Sample applications:

- Connect legacy PBX systems to low-cost VoIP services
- Connect legacy PBX systems to remote sites over private VoIP links
- Connect IP PBX systems to legacy TDM services
- Phased transition from legacy PBX to IP PBX
- Connect virtualized systems to legacy TDM services
- Transcoding by connecting systems using varying codecs
- Lync connectivity to SIP or legacy TDM providers and SIP or Legacy PBX

Models:

G100 Single T1/E1/PRI appliance:
Supports up to 30 concurrent calls

G200 Dual T1/E1/PRI appliance:
Supports up to 60 concurrent calls

G400 Quad T1/E1/PRI appliance:
Supports up to 120 concurrent calls

G800 Octal T1/E1/PRI:
Supports up to 240 concurrent calls

Digium VoIP Gateways
Technical Specifications

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Digium is the creator, sponsor, and innovative force behind Asterisk®, the industry's first and world's most popular open source telephony software. Additionally, Digium provides a variety of VoIP communication solutions that fit the needs of small, medium, and large businesses. Digium's product lines include commercial business phone systems, as well as software, hardware, and other components needed to create powerful custom communications solutions.

Custom Communications Solutions

Digium empowers users, developers and integrators to build custom telephony solutions by offering a variety of software, hardware, and third-party components. From basic voice applications to sophisticated phone systems, Digium makes it possible for the world to communicate at a fraction of the cost of proprietary solutions.

At the heart of these offerings is Asterisk, the powerful open source telephony engine. Asterisk is free software that turns an ordinary computer into a feature-rich voice communications platform. Its flexible architecture lets you configure it as an IP PBX, a voicemail server, IVR server, VoIP gateway, call recorder, automatic call distributor or virtually any other voice-enabled application that you can imagine.

Business Communications Systems

Digium's line of award-winning Switchvox IP PBX phone systems are built on a strong foundation of our open source Asterisk software. Switchvox solutions are designed to be extraordinarily easy to use and provide features that most small and medium businesses have previously considered out of their reach.

Call Management Features:

Automatic Call Type Detection:
  Voice/Modem/Fax
Answer and Disconnect Supervision
Trunk Group Support
Dial Plan Support
- Call Routing Rules
- Call Routing Groups

Pass Through Support for calls to toll free, local and emergency services numbers
Automatic appending and stripping of digits to dialed numbers
Caller ID name and number support
Fax and Modem support

Physical Interfaces:

Single (G100 & G200) or Dual (G400 & G800) RJ45 Ethernet connectors
Single, Dual, Quad or Octal T1/E1/PRI (RJ45 connectors)
Internal Universal Power Supply for 100 – 240 VAC
USB Port for System Reload/Recovery

IP Networking

- 10/100/1000Base-T Ethernet
- 802.1Q VLAN Tagging

IP Telephony:

SIP (Support for multiple SIP endpoints)
Audio Codecs
- G.711 (Coding Support for A-law and µ-law)
- G.722
- G.729
- GSM-FR
- G.726

Auto Codec Negotiation:

Fax and Modem Support (T.38 and G.711)

T1 Signaling:

PRI Signaling protocols:
- National ISDN-1
- National ISDN-2
- 4ESS (AT&T)
- 5ESS (Lucent)
- DMS100 (Nortel)
- Q.SIG

T1 CAS:

- E&M
- E&M Wink
- Feature-Group-D (DTMF, MF)
- FXS Loop Start with Forward Battery Disconnect
- FXS Loop Start
- FXS Ground Start
- FXO Loop Start with Battery Disconnect Supervision
- FXO Loop Start
- FXO Ground Start

E1:

- PRI (EuroISDN or Q.SIG)

DTMF Signaling via RFC 2833

Echo Cancellation (G.168):

- 128ms tail length
- 1024 taps

Configuration/Management:

Admin setup options:
- Web server

Remote setup:
- HTTP
- HTTPS
- DHCP w/ Option 66

Configuration Backup and Restore

Security Protocols:

- HTTPS
- Access Control List (ACL)

Troubleshooting Tools:

- Reporting Tools
- Advanced Debugging Tools (SIP, PRI, RTP)
- Diagnostics (System, Connections, Digium Support Lifeline)
- T1 Loopback
- Syslog

Specifications:

Size: 8.6" x 9.0" x 1.72" (21.84 x 22.86 x 4.37cm)
Weight: 2lb 12oz (1.7kg)
Style: Wall and rack mount

Environment:

- Temperature: 0 to 50° C (32 to 122° F) operation
  -20 to 70° C (4 to 158° F) storage
- Humidity: 0 to 90% non-condensing

Power Requirements:

- Voltage: 100-240v AC
- Frequency: 47-63 Hz
- Current: 65mA @ 120V, 33mA @ 240V

Compliance Certification and Agency Approvals:

Safety/Telecom:
- US/CSA 60950
- FCC Part 68, ANSI/TIA-968-A
- IC CS03
- CE Mark (European Union)
- IEC 60950
- EN 60950
- AS/NZ 60950

EMC:
- FCC Part 15 Class A
- EN55022/CISPR22 Class A
- EN55024
- EN61000

Environmental:

- 2002/95/EC Restrictions on Hazardous Substances (RoHS), 2005/747/EC
- Lead free exemption (Annex C)

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