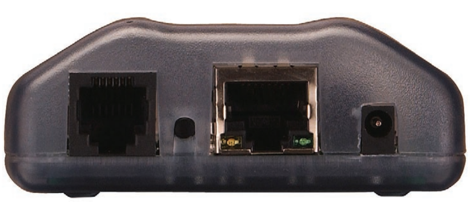
digium data sheet

S101I ("IAXy")

Taking Asterisk to the Customer Premises

The S101I, affectionately known as the IAXy, takes Asterisk from the PC to the CPE. The IAXy provides a single, fully featured FXS interface, with an Ethernet back-end, speaking the Asterisk-native IAX protocol, at a highly competitive volume price. The IAXy is aimed at Voice-Over Broadband and Internet Telephone Service Providers. IAXy provisioning, firmware updating, and feature activation can be performed entirely within Asterisk.

The IAX protocol provides complete NAT transparency, enabling full operation behind NAT and PAT firewalls. This includes the ability to robustly transfer calls between endpoints, allowing on-net calls to be moved off of a service provider's network for better quality and lower cost.



Target Applications

Internet Telephony Service Provider
Remote PBX Extensions
Wireless Phone Service with External Bridge

Features

Auto Upgrade
Remote Reprovisioning
Caller ID
Call Waiting
Cancel Call Waiting (*70)
Caller ID on Call Waiting
Caller ID Disable, Enable (*67, *82)
Three-way Calling
Call Transfer
Blind Transfer
Call Parking
VMWI (Voice Mail Waiting Indicator)
Mute Rx on-Hook
Pulse Dial
Call Hold

VoIP Codecs

µlaw (G.711)
ADPCM

VoIP Protocol

Inter-Asterisk eXchange (IAX)

Telephone

Connector: RJ11
Ringer Equivalence Number (REN): 5 at 1500 ft.

Power Requirements

6V DC, 1000mA Regulated Switching
Tip Positive 3–3.8mm outer-diameter, 1–1.3mm inner-diameter connector, locking tip, 11.5mm length

Environment Conditions

Operation Range: 0° to 50° C, 32° to 122° F
Storage Range: -20° to 65° C, 4° to 149° F
Humidity: 10-90% non-condensing

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About Digium

Based in high-tech Huntsville, Alabama, Digium is the creator and primary developer of Asterisk, the industry's first Open Source PBX. Used in combination with Digium's PCI telephony interface cards, Asterisk offers a strategic, highly cost-effective approach to voice and data transport over TDM, switched, IP, and Ethernet architectures.

Digium solutions reduce the costs of traditional TDM and VoIP implementations through Open Source, standards-based software and innovative hardware solutions, including legacy PBX, IVR, Auto-attendant, and next-generation gateways, media servers, and application servers. Digium hardware supports traditional voice protocols, including PRI, RBS, FXS, FXO, E&M, Feature Group D, Groundstart, and Loopstart. Data protocols include PPP, Cisco HDLC, and Frame Relay. For packet voice, Asterisk supports IAX (Inter-Asterisk eXchange), SIP, MGCP, Skinny, and H.323 VoIP protocols.

Digium provides a highly refined selection of quality hardware and software products, developed and implemented using innovative engineering techniques (primarily Open Source development). A full range of professional services complement these product lines, including consulting, technical support, and custom software development services.

The Open Source communications revolution is here, and Digium is leading the way.

Digium, Inc.