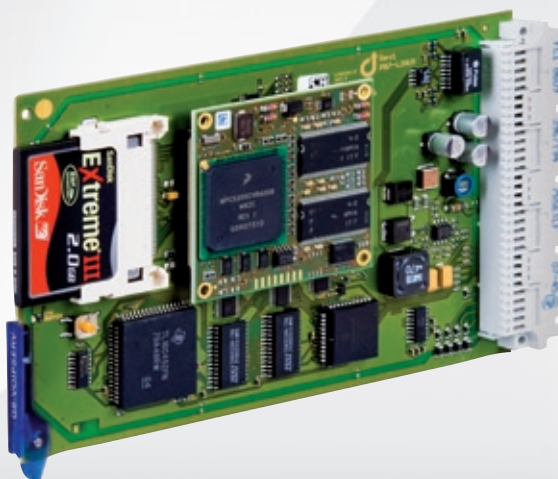


## G8-VOIPSERV



## VoIP Server Card G8-VOIPSERV

The multifunctional, high-performance server card based on Linux and Asterisk® technology enables integration of various VoIP standards such as SIP or IAX2 directly in the Intercom

Server. Via the integrated web interface, easy remote configuration is possible.

### TYPICAL APPLICATIONS

#### Trunk Links to VoIP PBX Systems

The interconnection of an Intercom System with Telephony combines both systems to one platform for the user. Calls can be established in both directions including display information. One physical link provides several simultaneous speech channels using state of the art codecs. Trunks to various VoIP servers like Cisco Unified Communications Manager, Alcatel, Siemens HiPath, Aastra, Brekeke and many more are available.

#### SIP Trunks to PSTN

The Intercom Server gets direct access to the public telephone network (via a SIP trunk to a VoIP provider). No additional equipment for ISDN or analog ports is necessary. Up to 8 simultaneous conversations via one network link are possible. Incoming calls can dial directly to Intercom stations.

#### VoIP Gateways

VoIP Gateways are often used for migration of existing equipment to a modern VoIP system like an Intercom Server. With ATAs (analog terminal adapters) and ISDN gateways it is possible to convert analog and digital phone lines to SIP and vice versa. Existing phones, DECT phones and dialers can be connected via SIP to the Intercom Server. In addition VoIP gateways can be used for trunking to existing telephone systems, which do not provide SIP connectivity.

#### VoIP Phones & Intercom Servers

The range of applications start with simply using a 3rd party SIP phone instead of an Intercom station up to using a VOIPSERV card as VoIP PBX for small sized business applications. Asterisk® provides all features required in a modern PBX including voice mail, transfer features and extensive possibilities for customizing.

#### Interfaces and Applications

The VOIPSERV card uses Linux as operating system and provides a data interface to the Intercom Server. Rapid development of Interfaces to 3rd party systems by using existing protocols and libraries. Pre-installed tools like a DHCP server, TFTP server and many others can be easily activated and reduce the amount of additional hardware in your network environment.

## Technical Data – Benefits

### TECHNICAL DATA

|                              |                                                                    |
|------------------------------|--------------------------------------------------------------------|
| Operating temperature range: | 0° C to +50° C<br>(32° F to 122° F)                                |
| Storage temperature range:   | –20° C to +60° C<br>(–4° F to 140° F)                              |
| Relative humidity:           | 20% to 80%                                                         |
| Plug:                        | 96-way PC board connector according DIN 41612                      |
| Network connection:          | 1 x RJ 45<br>(G8-NET or G3-GEM onboard)                            |
| Serial Connection:           | 1 x RS 232<br>(optional – not in extent of supply)                 |
| Power supply:                | from Intercom Server                                               |
| Emergency power consumption: | minimum: 140 mAh (idle)<br>maximum: 180 mAh (8 channels)           |
| Cabling:                     | min. Cat. 5                                                        |
| VoIP Protocols:              | SIP (RFC3261), IAX2 (RFC5456)                                      |
| Operating System:            | Linux, Kernel 2.6.23.1, Asterisk: 1.4                              |
| Processor:                   | PowerPC: 400 mHz, 64 MB SRAM,<br>16 MB onboard Flash, CF card slot |
| Dialling method:             | in band / out of band (RFC2833)                                    |
| Storage:                     | 2 GB compact flash                                                 |
| Audio codec:                 | preferred G.711 a-law                                              |
| Audio bandwidth:             | 3.4 kHz (G.711)<br>7 kHz (G.722)                                   |
| Measurements:                | 100 x 167 mm (3.94 x 6.58 in)                                      |
| Weight including package:    | 250 g (0.55 lbs)                                                   |

### BENEFITS

- Asterisk® support onboard for direct connection of SIP phones to the Intercom System
- Up to 10 SIP trunk lines with a total of 8 simultaneous speech channels
- Up to 2 IAX trunk lines with a total of 8 simultaneous speech channels
- Up to 50 SIP users (i.e. SIP phones to be registered on the "SIP Registration Server" of the card)
- Voicemail is available for each SIP phone
- Web based Asterisk® GUI for maintenance and remote configuration
- SSH (secure shell) for maintenance and remote configuration
- Supported audio codecs additionally to G.711 a-law:
  - G.711 u-law, gsm, G.722, adpcm, slin, lpc10, G.726, G.726aal2, iLBC
  - "pass-through only" (no recoding):  
speex, G.723.1, G.729A
- Simultaneous speech channels:
  - G.711 a-law: 8 simultaneous conversations
  - G.711 u-law: 8 simultaneous conversations
- Adaptive jitter buffer
- Supported protocols:  
SIP, IAX2, RTP, UDP, TCP, NTP, SMTP, HTTP, HTTPS, DNS, DHCP, TLS/SSL, SSH, SFTP, ICMP and more ...
- One-click backup over web interface
- Automatic transmission of the caller-ID as text indicator at SIP phones
- Direct dialling to Intercom stations possible
- Pre-recorded audio e.g. for waiting information

### SYSTEM REQUIREMENTS

#### GE 800:

- Intercom Server software min. PRO 800 1.0B
- G8-IAX (*internal trunk to Intercom Server*)
- Configuration software min. CCT 800 1.0B

#### GE 300:

- Intercom Server software min. PRO 800 1.0B
- G3-IAX (*internal trunk to Intercom Server*)
- Interface housing GEI 300
- Configuration software min. CCT 800 1.0B

### EXTENT OF SUPPLY

Server card G8-VOIPSERV including:

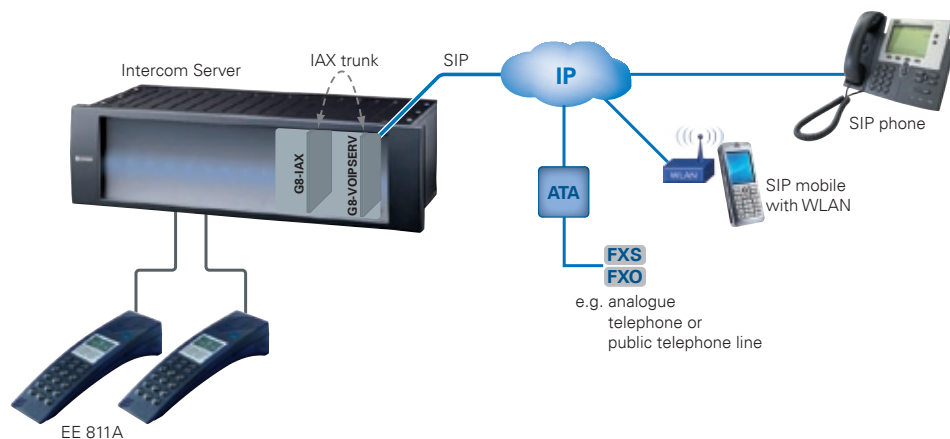
- Short reference
- GPL license note

## System Overview – Applications

### SIP REGISTRATION SERVER

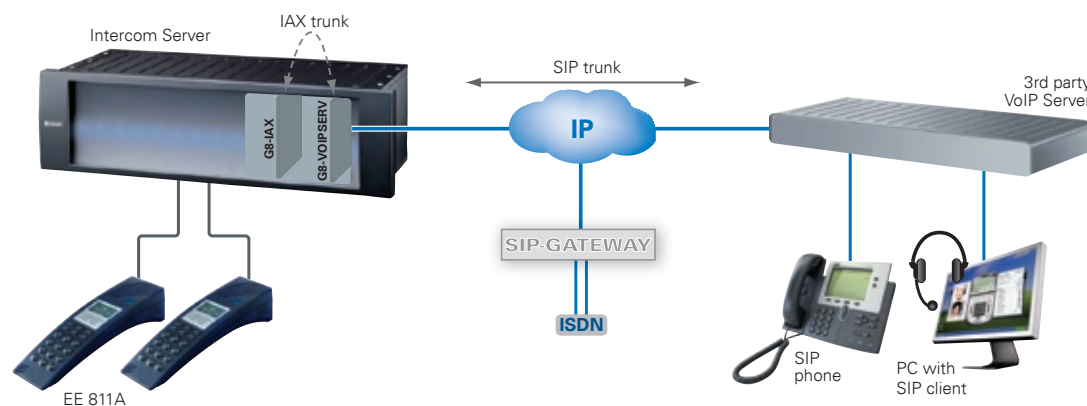
The server card G8-VOIPSERV provides a SIP registration Server, with which it is possible to integrate SIP phones in the same network into the Intercom system. I.e it is possible to integrate SIP phones without the requirement of an intermediate trunk line to a 3rd party VoIP Server.

- Intercom calls (call requests, input messages) can be transferred to SIP phones.
- Calls can be set up from a SIP phone to an Intercom station.



### SIP TRUNK CONNECTIONS TO 3<sup>RD</sup> PARTY VOIP SERVERS

Via the G8-VOIPSERV trunk lines to 3rd party VoIP Servers or VoIP gateways – in the same network or to public VoIP Servers – are possible (e.g. SIP trunk connection to "Cisco Unified Communications Manager").



# Installation

## PRECAUTIONS

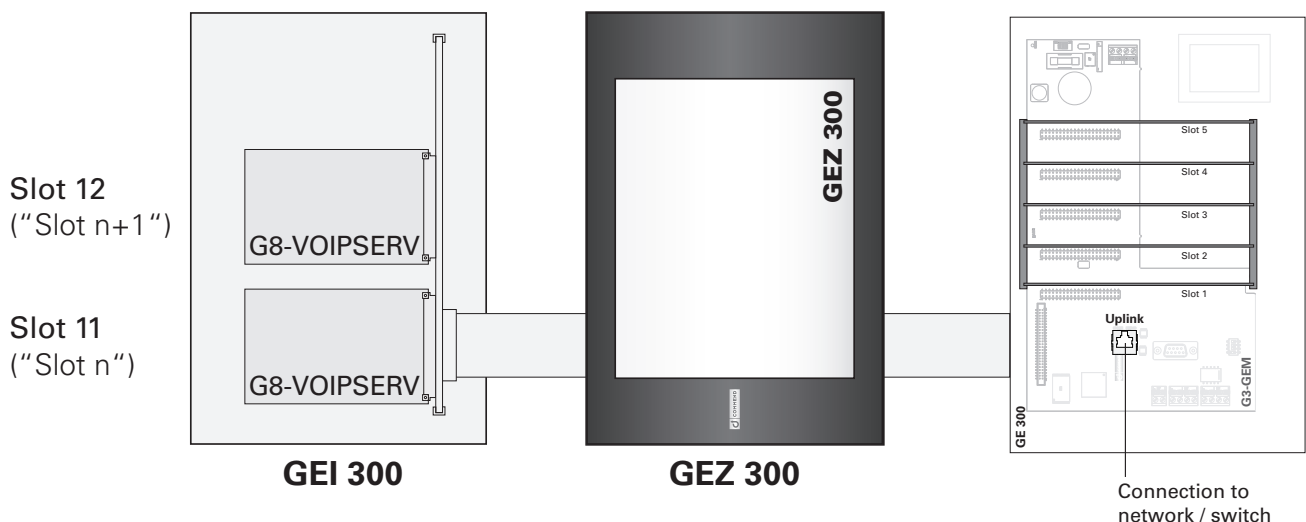
- ESD precautions must be observed.
- Cards may be inserted or removed only with voltage switched off.
- Mounting and installation of the Intercom Servers and of the equipment may be carried out by authorised service personnel only.

## CONNECTION GE 800

- In an Intercom Server GE 800 the server card can be installed in the slots 1 – 14 (max. six G8-VOIPSERV per housing).
- The G8-VOIPSERV is connected to the network via the installation board G8A-NET of the G8-NET card.

## CONNECTION GE 300

- The G8-VOIPSERV can be inserted in slots 11 – 12 of the interface housing GEI 300 for the Intercom Server GE 300 (GEI 300 offers place for 2 cards).
- The G8-VOIPSERV is connected to the network via the ethernet port of the GE 300.



## TRADEMARK NOTICE

- Cisco Unified Communications Manager is a registered trademark of Cisco Systems Inc.
- Siemens HiPath is a registered trademark of Siemens AG

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The development and manufacturing processes are certified in accordance with **EN ISO 9001:2008**.

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