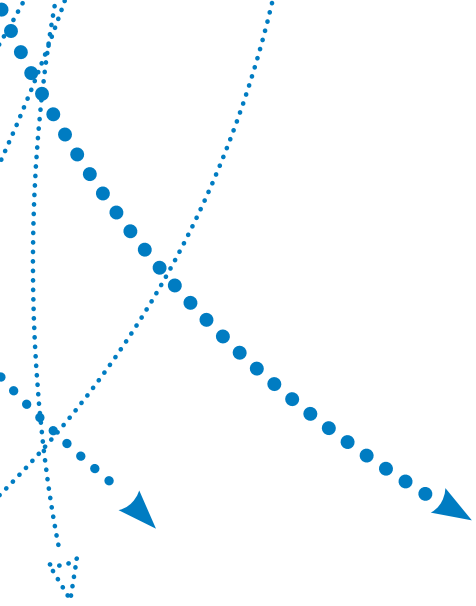




IoIP[®] Technology Guideline

Version 1.0/1109



COMHEND
SECURITY AND COMMUNICATION

» IoIP® Technology Guideline

This Guideline is intended for network administrators and other engineers, who are implementing/integrating a Commend Intercom System in a network environment. It outlines the basic operation, system requirements and technologies used by the Commend Intercom Servers.

1 IP Intercom System

The Intercom over IP protocol (IoIP®) is used to network the Commend Intercom System via any kind of IP network. Generally there are two kinds of network links possible: Trunk connections between Intercom Servers and IP-Intercom Stations connected to an Intercom Server. Both use the IoIP® protocol, which is optimized for real-time operation, high availability and perfect audio quality up to 16kHz HD Audio.

In addition to IoIP® networking, the Commend system provides interfaces for standard VoIP protocols, like SIP or IAX2. These can be used for trunk links to 3rd party VoIP Servers, or as SIP Registrar Server to integrate standard SIP Phones to the Intercom System.

The Intercom Exchange Protocol (ICX) provides bi-directional data exchange between 3rd party applications and the Commend Intercom system via standard TCP sockets (client or server). ASCII based datagrams can be used to monitor the Intercom system, or control Intercom calls and I/Os. The same protocol is used for remote configuration via the CCT 800 tool.

2 Intercom over IP Protocol

2.1 Protocol Specifications

The IoIP® protocol is based on UDP using a single port for audio transport and signalling, which makes it NAT- and firewall-friendly. The audio transport is similar to the standard RTP protocol using time-stamps for re-ordering and unique identifiers as channel information. Any kind of signalling data is transmitted using a higher-level protocol similar to TCP, but optimized for low latency and guaranteed transmission.

The default port number for trunks between Intercom Servers (LAN/WAN cards) is:	UDP 16384
The default port number for IP Intercom Stations (ET 901, EF963, ET 908) is:	UDP 16400
The default port number for the ICX/TCP data interface is:	TCP 17000

All port numbers can be changed via the configuration tool CCT 800.

The Intercom Servers and interfaces run on static IP addresses. All IP-Intercom Stations can be configured to use static IP addresses or DHCP. In case of using dynamic IP-addresses, the IP-Stations are authenticated at the Intercom Server for security reason.

IoIP® protocol does not use any broadcast addresses and therefore can be routed across any router or layer 3 device.

2.2 Audio Technology

Depending on configuration, the Commend Intercom system can be operated up to 16kHz audio quality. Telephony interfaces start with 3.4kHz audio using G.711 codec. The previous Intercom Server family GE 200/GE 700 operates with 7kHz audio quality using a 16 kHz sampled G.711 codec or standard G.722 compression. The current Intercom Server generation GE 300/GE 800 uses a G.722 based codec sampled with 32 kHz for resulting 16 kHz crystal-clear audio.

Intercom Stations can be operated in different audio modes:

- Simplex operation – PTT (push to talk) for manual control of speech direction
- Switched duplex operation – automatic switching of speech direction
- OpenDuplex® operation – hands-free full-duplex conversations

OpenDuplex® from Commend uses an echo-cancellation algorithm designed for high volume levels, even with 16 kHz audio quality.

2.3 Bandwidth Requirements

Depending on which audio codec is configured, the Commend Intercom System requires the following bandwidth per audio channel in the IP-network (including IP and IoIP® header):

a.	3.4 kHz audio (G.711/8 codec):	82	kBps
b.	7 kHz audio (G.711/16 codec):	146	kBps
c.	7 kHz audio (G.722/16 codec):	82	kBps
d.	16kHz audio (G.722/32 codec):	146	kBps

All IoIP® audio packets are transmitted in the network following a constant frame rate of one packet per 16 ms. Packet length (including IP and IoIP® header) can be either 164 bytes or 292 bytes. Signalling packets are not longer than 96 bytes.

In addition to audio transmission, signalling requires a certain amount of bandwidth. A single IP-Intercom Station should have at least 14 kbps reserved for signalling. Trunk links between Intercom Servers should provide at least a total bandwidth of 128 kBps (one audio channel plus signalling).

All network interface cards of the Commend System provide a built-in traffic shaping algorithm, to avoid packet bursts in case of broadcasts applications like alarm features or paging (group calls).

Every IoIP® link is monitored by exchanging idle packets. The time frame can be configured, but is typically set to 60 seconds. The idle-packet length of 36 bytes results in a traffic volume for one IP Station during one day of 104 kBytes.

2.4 Quality of Service

Whenever running VoIP applications over shared IP networks, it will be necessary to guarantee low latency transport for audio packets without packet loss. If the available bandwidth of the network links and components is high enough, may not be necessary to configure QoS. But especially WAN links normally provide asymmetric or low bandwidth links shared by many applications. In that case, a simply priority queuing for VoIP packets in the routers can ensure perfect audio packet transport.

The Commend Intercom System provides several possibilities to design a QoS compliant network environment. IoIP® packets can be marked by setting the DSCP values for a Diffserv network environment. In local area networks, QoS is typically done by frame-tagging following the IEEE 802.1Q and 802.1P standard. Frame-tagging and VLANs can be configured for all Commend IP-interfaces.

Quality requirements for IoIP® links:

	Recommended network performance parameters	Limits
One-Way-Delay	< 100ms	< 350ms
Jitter (IPDV)	< 50ms	up to 320ms
Packet loss	0 %	5 - 10%

Every Commend IoIP® network interface provides automatic jitter buffers, which can be configured to recover even bad network conditions.

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2.5 Security and Availability

The IoIP® protocol is a binary and slim protocol, which cannot be easily decoded, compared to standard SIP signalling. All audio codecs are based on international standards, but optimized for usage in the Commend Intercom System, and therefore cannot be decoded. The IoIP® protocol is prepared to use encryption for audio and data transport.

Security in the Commend Intercom system starts with the physical separation of two Ethernet ports in the Intercom Server GE 800, to run a LAN and WAN link in a certain network infrastructure. On Ethernet level, applications and networks can be logically separated using VLANs (IEEE 802.1q). Ethernet frames can be tagged in the Intercom System for each interface. Using static IP addresses, every IoIP® link does IP source and port filtering for incoming packets. Dynamic addressing uses authentication methods for registration in the IP-Intercom Server. A built-in firewall in the Intercom Server GE 800 prevents unwanted incoming traffic on certain ports or interfaces.

The ICX/TCP data interface can be operated using MD5 authentication plus additional user passwords. The system can be locked for unauthorized 3rd party users to control the Intercom System.

Commends Intercom Server architecture splits several server tasks to independent IP interface cards within a server housing. If one service or link goes down, the other interfaces will still remain in operation. That results in high availability, and extremely low down-times of Intercom Stations and links. Every trunk link can be configured with a fall-back route through another network (IP) or network technology like HDSL or E1. IP Intercom Stations can be configured with fall-back server addresses to automatically re-connect in case of link loss to a fall-back Intercom Server.

3 Intercom VoIP Interfaces

The Commend Intercom System provides standardized VoIP interfaces to establish trunk links to 3rd party VoIP servers, or directly connect VoIP phones to the Intercom Server.

3.1 IAX2 Interface Card

The IAX interface card from Commend provides InterAsteriskExchange protocol version 2, standardized in RFC 5456. It can be used to set up IAX trunks to 3rd party VoIP servers. The protocol is similar to the Commend IoIP® protocol and runs on a single UDP port 4569. Commend provides G.711 a-law codec for audio transport. Link authentication can be set up via plain text or MD5.

3.2 SIP Interface Card

An embedded Asterisk® based interface card in the Commend Intercom System provides SIP and IAX2 protocol. The interface can be used to set up trunks to 3rd party VoIP servers via SIP or IAX2 protocol or can be operated as SIP/IAX registrar server to connect standard VoIP phones to the Intercom system.

The extensive features of the well-known Asterisk® technology provide high flexibility by integrating almost any kind of VoIP server or phone application. The interface card runs on Linux operating system, which can be easily extended with applications like DHCP server or TFTP for provisioning applications.

The SIP interface card provides SIP transport over UDP port 5060 (can be configured) following RFC 3261 and RFC 3265. DTMF support via SIP Info, Inband or RFC 2833. IAX2 support following RFC 5456. RTP port ranges for audio or video transport can be configured. Preferred audio codecs are G.711 a/u-law. Other Codecs like G.722, Speex, GSM, iLBC, aso. are supported or can be transcoded.