IoIP and VoIP

IoIP® – Intercom over IP

Security, high voice quality, additional control and display features, real-time protocol, reliable protocol – proven for more than 5 years, wide variety of existing Intercom terminals for all kinds of environments, markets and requirements.

Networking via IP

Often, various site locations, need to be linked together. Commend has developed networking solutions for IP, ISDN, E1 and HDSL platforms. Networking Intercom Servers allows the local Intercom system to act as one large system across different sites. All specified functions are available across the entire Intercom network and programming is conducted centrally from a single location.

IP Stations

IP-Intercom stations can be directly connected to the IP network. With the Commend IP-converter also every analogue or digital Intercom terminal can be IP enabled. Allowing the perfect solution on-site for every type of application. As most intercoms are equipped with local inputs and outputs, it is possible to create reporting and control functions alongside voice connections via the IP Network.

VoIP

Data networks allow transmission of manifold data. VoIP (Voice over IP) is the transmission of speech using Internet Protocol (IP), particularly in telephony.

Intercom in the VoIP World

Flexible VoIP interface providing SIP/IAX connectivity – to integrate standard SIP phones into the Intercom system or SIP trunks to 3rd party VoIP servers.

SIP

Today's standard for VoIP – Internet telephony, based on Internet protocols similar to http – web based.

- Mobility of phones (soft / hard phones)
- Easy interfacing
- Provides multimedia aspects (audio, data, video) in one platform

Learn more about the IoIP® protocol

IoIP® – Intercom over IP – is a protocol created by Commend especially for professional Intercom solutions. Instead of adapting various other protocols and taking into account that some definitions in those other protocols would not be used at all, we created our own protocol.

IoIP® was designed specificly for Intercom applications. It is a real-time protocol including all Intercom aspects, such as easy interfacing of third party auxiliary systems. High quality voice communication as well as security and safety aspects were included. On top of being extremely reliable, hassle resulting from a very common and thus not very secure protocol is excluded. Proven in countless installations all over the world, IoIP® has met professional user's expectations.

Understanding Protocols in general

All types of data communication between two devices are described by protocols – regardless if data, pictures or voice transmission. In fact, telephones or other devices used for VoIP communications are more or less computers, and before they can "talk" to each other, they have to "negotiate" between themselves how the voice communication will take place.

There are protocols for signalling – which only include the setup of a conversation, which is usually called a session. The most widely used protocol for this is [SIP or Session Initiation Protocol](). Then there are protocols describing how the data is transmitted during the session. VoIP very often uses RTP (Real Time Protocol). Other standards describe the en- and decoding of voice into data packets, which can be sent over a data network. Only when all parties taking part in a communication agree to use the same set of protocols, they will be able to communicate with each other. It is based on what needs to be transmitted, including multimedia aspects, that the appropriate set of protocols is selected, and manufacturers are free to use existing protocols, to adapt them or to create fully new protocols.

SIP and IAX

Connect your Control Desk to the SIP world
Commend's SIP/IAX Interface

One of today's mayor topics is the integration of various systems. Just like a decade ago standard telephones were to be integrated in professional Intercom solutions, today VoIP telephony systems need a gateway to Intercom systems.

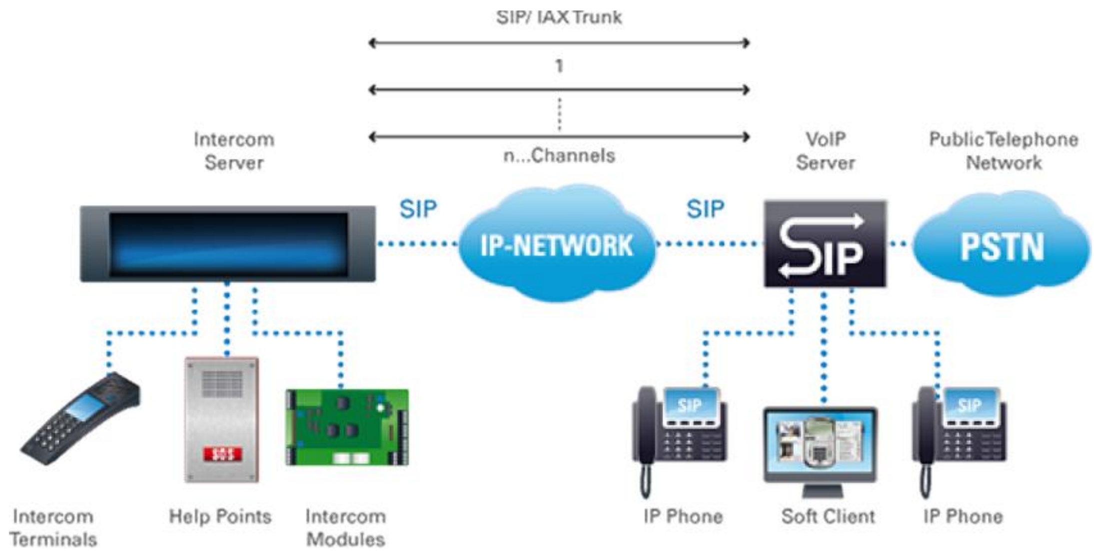
The idea is to use VoIP telephones or even a VoIP telephone exchange as an extension for Intercom systems. Telephones can even be used as sub- or master stations. The key to connect our Intercom Servers to a VoIP server is Commend's new SIP/IAX networking cards. The use of standard SIP/IAX protocol makes it easy to connect virtually any third party VoIP server and VoIP telephone to the Commend Intercom system.

Intercom SIP Phones

Also SIP telephones basically are simple computers – so called SIP clients. SIP clients (VoIP telephones) are connected with a SIP server (VoIP telephone exchange). Typically more than one SIP telephone need to be integrated into an Intercom solution, mostly in a control room situation. For this reason, the easiest connection is to connect the SIP clients via the SIP server on a SIP trunk with the Commend Intercom server.

Intercom SIP Trunks

SIP telephone exchanges or SIP servers are computers designed to operate as telephone exchange for VoIP telephones using SIP protocol. Any direct connection between two such computers is called a trunk line. Commend's new SIP/IAX interface connects a GE 800 or GE 300 Intercom Server to a SIP server via such a trunk line.

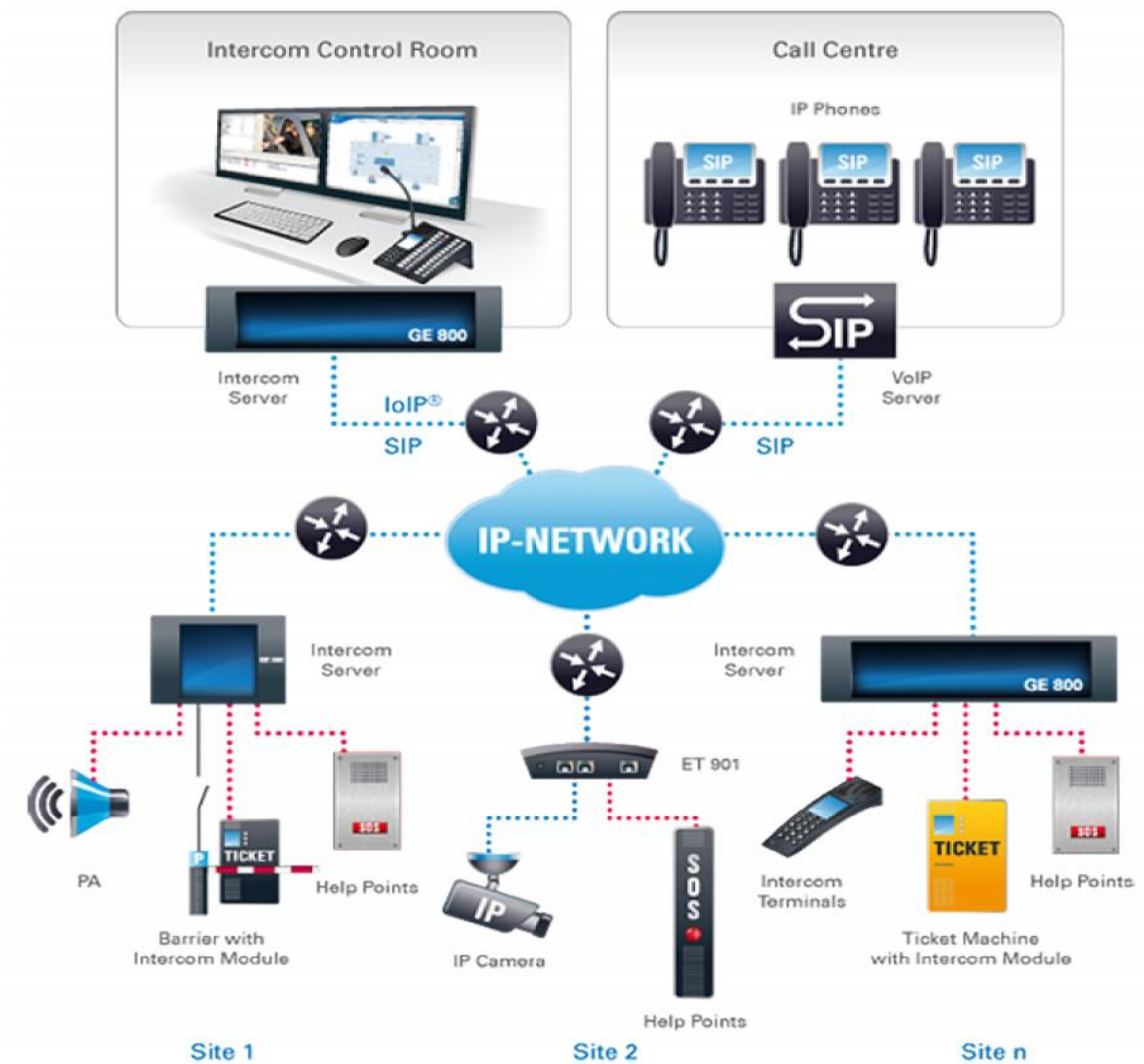


Practical Application

A classical control room is equipped with one or several master stations, which can also be a control desk on which buttons are assigned with signalling or control functions (like showing that a door is open, or activating a door opener, or just programming a direct call on a certain button). Using a graphic user interface (GUI) like ComWIN, the control desk is now on the screen of the computer as well.

By interfacing with SIP/IAX, it is now possible to use IP-telephones in a remote location (like a call center) as master stations. Here even more a GUI will be very helpful to assist the operators in identifying the location of problems.

All Intercom stations at various remote locations like parkings can be either master- or substations and are connected with the control room or call center via an IP network.



Commend SIP Series 3.0

Packed with amazing array of new functions and control options, the soon-to-be-released Version 3.0 of Commend's SIP Terminals is set to open the door to the Convenience Class in SIP (Session Initiation Protocol) telephony systems for simple, cost-efficient integration of sub-stations and security solutions.

Since the first series of Commend SIP Stations entered the Intercom dimension in 2011, the devices have grown up fast in terms of convenience and functional power. From the start, the SIP-capable terminals were in a class of their own – not just where hardware is concerned: their innermost software features, too, are quite unique, opening up new options of using simple, low-priced SIP systems with the help of intelligent, convenient functions from the world of Intercom.

SIP 3.0 – Three good reasons for upgrading to the top class in SIP telephony

With the new Firmware Release 3.0, Commend SIP Stations get their most powerful functionality boost so far, along with enhanced independence from proprietary, server-based SIP solutions. Here are three top highlights to look forward to:

1. **Provisioning** makes the solution ready for large-scale installations: a simple mouse click transfers directory entries, function settings, pre-recorded messages, firmware updates and lots more instantly to all (or selected individual) SIP Stations.
2. **Sequences** is the name of a powerful new function that enables easy programming of small, flexible "action scripts" for SIP Stations. A great example is high-convenience access for residential buildings, where punching in one's door access PIN not only opens the door, but triggers other actions as well, such as turning the light on and getting the elevator, for example. All these are functions that next-best SIP solutions either cannot provide or require great programming efforts to implement.
3. **HD Voice** now provides 7 kHz wide-bandwidth speech quality, which outperforms common VoIP by far in terms of clarity and intelligibility. Also, the new, improved 5-level **active noise suppression** feature makes Commend SIP Stations ready for application environments where things can get particularly noisy: tunnels or manufacturing facilities, for example.

As of **mid of March**, SIP 3.0 will be implemented on all SIP devices being shipped. It will than also be available as a **Firmware Update Package** for generation 1.0 and 2.0 SIP Stations. The package will be **free to download** at www.commend.com/sip.

SIP-Stations

Security for the SIP-world

Opening up many new possibilities

SIP Intercom Stations by Commend maintain connections to important areas, from entrances and vehicle gates to warehouses and adjoining buildings. Their durable, stylish design makes them suitable for any application environment.

We are constantly enhancing our SIP stations through further development and the addition of new, powerful functions. As a result, you can load local telephone directories, set up call chains, implement server redundancy and take advantage of lots of other useful features. We are also constantly expanding the various control options that can be carried out during calls.

However, it is still the acoustic dimension where the Commend SIP Intercom Station really comes into its own. In addition to its high audio quality, the station features a built-in memory for up to 60 seconds of pre-recorded messages. This can be used e.g. to provide reassurance or information messages at the local call point, supply automated acoustic location details to a receiving station, or simply to play custom ring tone.

- Easy to integrate into IP networks or IP telephone systems
- Fast and cost efficient installation, configuration and maintenance
- Built-in memory for up to 60 seconds of pre-recorded messages
- Well tested, e.g. in combination with SIP Servers of important manufacturers
- Easy to integrate into the product world of Commend's Intercom 2.0 range

Clear voice signals, economical and secure

Outstanding audio quality, reliable function monitoring, as well as mechanical and manufacturing quality are key factors of Commend's product development policy.

During the development of the SIP Series, a particular focus was also put on operational economy. For call stations that are required to be available around the clock every single day of the year, this is a factor that is easily noticeable not only economically, but ecologically as well.

- Higher sound volume than common SIP stations, thanks to Class ‚D' amplifier and two specially designed loudspeakers built into each device
- Active suppression of background noise ensures ultimate intelligibility
- Microphone and loudspeakers are individually adjustable on each station
- Call stations require only half the power of compareable devices

The open SIP (Session Initiation Protocol) standard stands for simplicity, cost savings and independence for today's business communication.

Obviously, this calls for the integration of substations as well, in addition to mere telephone sets. Stations, that require special features that only Intercom systems can provide: hands-free talking, high volume, function monitoring, wall mounting, and reliable protection against bad weather, vandalism and sabotage, to name but a few.

Keep your options open and get the best of both worlds:
SIP Series – powered by Commend.

SIP-Series P

Stations with polycarbonate housing, suitable for indoor and outdoor installation.

SIP-Series F

Membrane protected SIP stations with full keypad.

SIP-Series M

SIP modules suitable for installation into existing housings and panels to support the assembly of custom-built SIP stations.

SIP-Series V

Vandal-resistant stations with durable high-grade steel front panels, poke protection and special screws.

SIP-Series VE

Vandal-resistant emergency call stations with extra-large emergency call button.

SIP Series by Commend: simply powerful

- Very high volume
- Full Duplex capable SIP stations
- Full keypad support NEW!
- Handset support NEW!
- Local directory support NEW!
- Custom programmable call buttons for direct dialling
- Extended memory for audio files:
    - Playback at call initiation (e.g. for reassurance messages), programmable for each call button individually
    - Playback at call reception (e.g. individual ring tone)
    - Playback in case of error (e.g. special error warning sound)
- 2 relays for extra control functions or attendant contacts
- Three inputs for connecting add-on call button modules NEW!
- Line-Out ports on SIP stations NEW!
- Line-In ports on SIP modules NEW!
- Energy-saving operation, thanks to low power consumption
- SNMP for call station monitoring
- STUN support NEW!
- Communication via IP data networks
- Connects directly to compatible SIP servers
- Sever redundancy NEW!
- Can also be operated without a server
- Built-in switch with downlink function for direct connection of additional IP devices (e.g. IP-based camera)
- Robust enclosures with IP 65 rating
- Vandal-resistant stations with IK 07, IK 08 and IK 09 rating as per EN 62262
- Power supply either external or via PoE (Power over Ethernet)
- Remote controllable via HTTP NEW!

Well tested

To ensure full, reliable functionality of SIP Series stations in different system environments, the devices are constantly subjected to extensive tests in combination with SIP Servers of the followinfg manufacturers[1]:

- Digium Asterisk-Server
- Cisco Call Manager and Cisco UnifiedCommunication Manager
- Avaya Business Communications Manager
- Innovaphone Virtual Appliance (IPVA)
- Alcatel OmniPCX Entreprise (OXE)
- Siemens Hipath 4000
- [3CX for Windows]
- Aastra MX-ONE (formerly Ericsson MX-ONE)

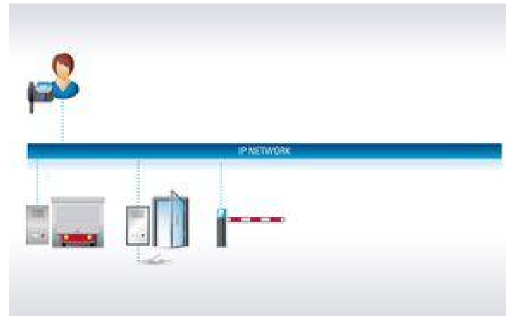
1) The listed products and company names are brand names or registered trademarks of their respctive owners.

A suitable connection for every need

When it comes to SIP solutions, Commend covers all application requirements – from simple linking of individual sub-stations with SIP-capable telephones to a comprehensive integration of SIP Server functionalities into Intercom 2.0 networks.

Serverless operation: One SIP telephone can serve multiple sub-stations

Simple, cost-efficient solutions:



- **Sub-stations can be remote-operated** via a SIP-capable telephone
- SIP addresses of the sub-stations are programmed into the device's memory for **direct-dialling** (up to 12 addresses, depending on manufacturer)
- Calls can be **initiated in both directions**
- **Suffix dialling** can be used e.g. for opening doors, shutters or gates
- Support of individual ring tones for each sub-station via**pre-recorded audio files**

Operation via SIP Server with support of Server's functions

Extended functionalities supported via SIP-based telephone systems:

- Sending of call signals to **multiple** sub-stations
- Support of **Call Queuing** (same function as used in control desks)
- Call forwarding to public telephone networks or mobile devices
- **Interfaces** to third-party systems and other subsections
- **Visual call indication** systems
- **Group calls** (if supported by telephone system)

Intercom Server Exchange: the Commend VOIPSERV card functions as a SIP Server

Full SIP integration into IP-based Intercom Servers:

- Up to **50 SIP call stations** per card
- **Line monitoring**
- SIP call stations can be **displayed** on Intercom control desk
- **Control desk visualisation** with ComWIN
- **Wide Area Networks**
- Full **integration of security systems**, including: CCTV, RF communication, Alarm systems, facility management
- Broad range of IoIP©-based terminals with high-quality **Intercom functionalities** and **DSP features**

Tisab d.o.o., Trećí bulevar 143, Novi Beograd,
tel: +381 11 3103 999,
http://www.tisab.rs/commend.html