Conferencing system solution

Conferencing system for operational and technological networks

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Contents

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1 Introduction

Present document describes different types of conferencing system which are used in OTC (operational and technological communications). More detail is described dedicated conferencing system for meeting organization within large enterprises which are spread along large geographical areas (svyaz sovesania). At the end of the document are described Iskratel products which are used for organization of different types of conferencing system in operational and technological networks.

2 Conferencing systems for operational and technological networks

In operational and technological telecommunications networks dedicated conferencing systems are used. Main goal of such systems is delivery of effective voice communication between personal involved in technological proses. Typically this type of conferencing system can be found in enterprises which have their working units spread among big geographical areas.
Examples of such enterprises are railways, which require voice communication system between stations arranged along tracks. The same examples are in energetic sector where power transmission utilities and oil&gas utilities has their production facilities, substations, compressor stations… arranged along their production, transmission and distribution infrastructure.
In all described cases high efficient and high availability voice communication system is essential to provide support for working process.

Main characteristic for voice conferencing system in operational and technological networks are:

- High availability
- High quality of voice and speech intelligibility
- High performance of the system (number of simultaneous conferences, number of participants…)
- Rich possibility of manipulation
- Openness to different technologies, depending on existing infrastructure (analog, digital, VoIP)
- Different approaches for integration in the conference (PTT, raise hand)
- Transfer of different information between participants (voice, video, content)
- Simple administration and moderation of conference calls (creation of the conference, call/destroy conference, manipulation of the participants, priorities...)
- Status information of conference participants (busy, active, active voice, participant of certain conference...)
- Different types of users (administrators, moderators, participant)
- The organizational structure of enterprise (moderator, leading participant)
- The ability to integrate with existing conference / dispatcher system
- Possibility of connecting to other systems (public announcement, video surveillance)

2.1 Types of conferencing systems in OTC networks

2.1.1 Dispatching communication system

Dispatching communication system should primarily assure voice connection between dispatchers, service workmen and other telephone subscribers within private and public communication networks. Mani task of dispatching system in operational and technological network of enterprise is managing of work process and assuring security and safety for personal working on site.

2.1.2 Meeting conference system, Selector conference system

It is intended for the organization of periodic meetings and the delegation of tasks between the geographically dislocated units within corporations. Participants have predetermined roles and rights within the conference. Moderator (using the dispatcher terminal) is responsible for establishing the conference calls, manipulation of the participants, assigning priorities, etc. If necessary, it can involve (voice participation) in the conference calls. Via dispatcher terminal, the moderator can easily moderate a multiple conference calls simultaneously.

The conference calls are voice guided by the leader of the conference, which can be any active participants of the conference. The conference leader has the highest priority in the conference, which means that at any moment his voice (speech) overdrive the other participants voice.
SSv conference is based on the OTC dispatcher circles - identical as for dispatching communication system.

2.2 Conference members and their role in conference

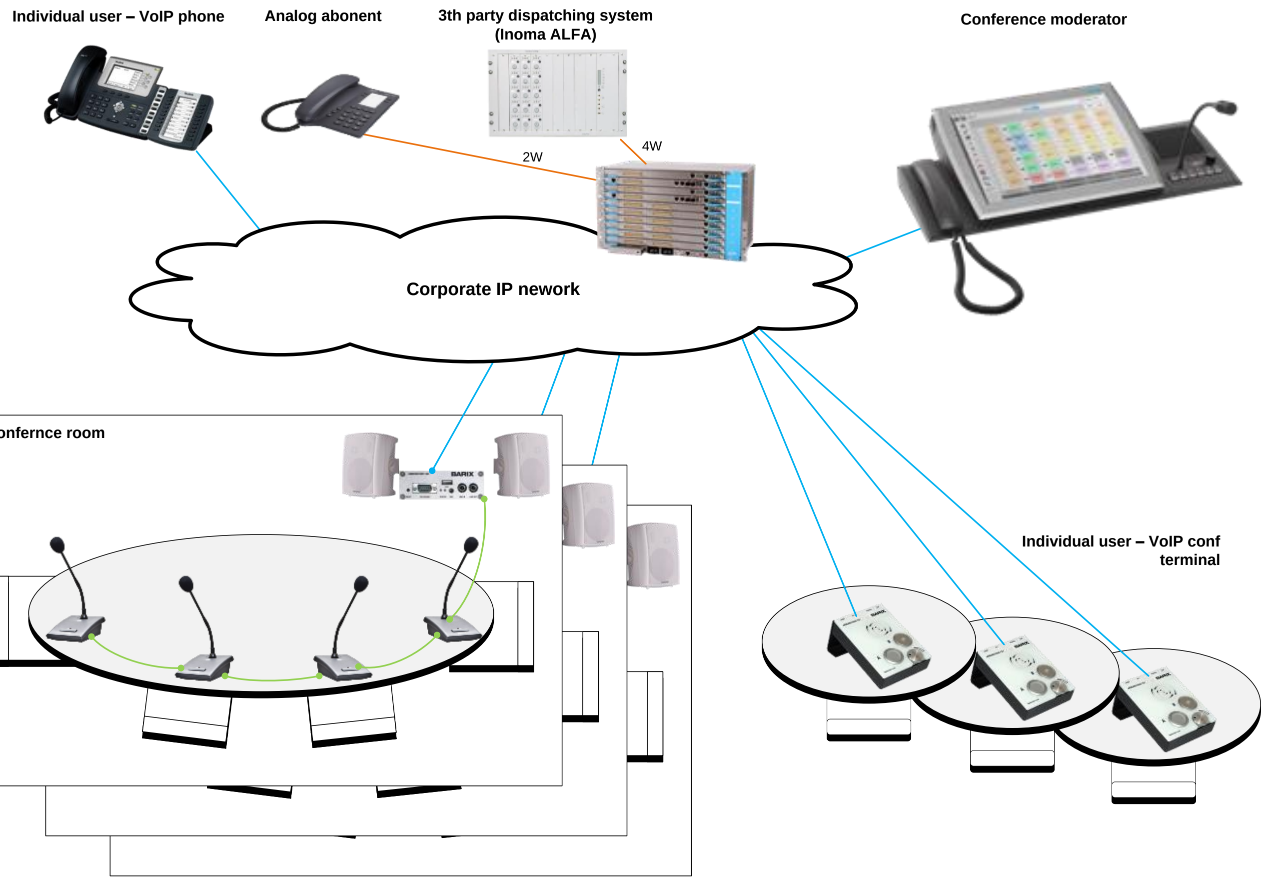

Figure 1 Basic setup of conferencing system in OTC network

2.2.1 Moderator

The moderator can create, call/start and configure conference calls via touch-screen interface on dispatcher terminal. He can determine members for several conferences before conference is started or during the conference. Before call/start of conference call, he can setup for each participant individually his initial status. When conference call will

be establish participant will become active member of conference with predefined settings. Typical settings that can be configure are: mute/unmute, speaker on/off, priority on/off and priority level.
The moderator can configure and moderate several conference calls at the same time. He can listen to voice conversations of several conference calls simultaneously on the speakerBox device. He can also become an active speaker in one of the active conference call. When conference is established moderator can add or remove participant in any active conference call and modify settings for each participant individually (mute/unmute, speaker on/off, priority…). When conversation in the particular conference is finish he can release all the participants.

2.2.2 Conference leader

The leader of conference is one of the active participants of the conference, which is responsible for the voice managing of conference call. The leader manages the participant via voice interaction with administrator. As a general rule, leader has the highest priority and the possibility of voice overdrive ("pereboj") upon other participants in conference call. The leader of conference call becomes an active speaker by pressing a Push-To-Talk button on his terminal. In this way the noise in active conference call is minimized. Beside PTT option it is also possible to use (configurable on conference server) VAD (Voice Activity Detection) feature for minimizing the noise in conference call.
The leader (participant) can connect into conference via a variety of terminal equipment's:

- Dedicated conferencing terminal for individual participant,
- Conventional telephone,
- Studio equipment.

2.2.3 Conference participant

Participant is always invited in to the conference by conference moderator. He doesn't have rights to moderate and organize the conference. Participant is typically joined in to conference in simplex mode so he could only listen to conversation. If required, moderator can enable microphone of participant and give him rights to participate in conference. Moderator can assign priority level to participant. Typical participant has the lowest priority level, so the leader or moderator can voice override him (level of his voice is reduced when higher priority member starts to speak).
Participant can use different types of terminal equipment. His terminal equipment usually supports push to talk option to reduce noise in conference when there are many conference members connected over different interfaces (analog, digital, IP). Types of conference terminal equipment are

- Single participant conferencing terminal
- Conferencing studio
- Telephone terminal

2.3 Conference system on basis of Iskratel equipment

Conference system is composed of the following products:

- SI3000 MPD - conference moderator working place
- SI3000 cCS - conference server
- SI3000 DRS - call recording system
- SI3000 MNS - management system
- Terminal equipment for the participants

2.3.1 SI3000 cCS - compact Call Server

SI3000 Compact Call Server (cCS) is multiservice infocommunication platform for array of subscribers offering a variety of interfaces both in TDM and in IP world. It offers wide range of enterprise services, signaling, regulatory functions (LI/SORM), conferencing, personnel notification system, transcoding and other functionalities.

SI3000 compact Call Server is suitable for enterprise environments and special networks (railways, energy, and government). It is engineered for capacities from up to 4096 subscribers (Analog, ISDN and IP). It supports up to 32 E1s in an optionally redundant configuration. Packed with the latest audio and video codec’s it enables true, wide band experience - a mandatory function in a today’s telephone and video communication.

The SI3000 cCS can be accommodated in 2-, 6-, 10- and 20-slot MED shelves.

MED housing supports the supply bus, TDM bus and a supply bus for the ringing generator. The basic connection diagram is a dual star of Ethernet connections. Each peripheral board is connected via two independent Ethernet connections to two central slots (CMJ boards). The system contains a dual IPMB bus (for IPMI control), a supply bus, a TDM bus, a supply bus for the ringing generator and a synchronization bus.

Figure 2 MED 2, 6, 10 and 20 shelf

Central blade of SI3000 cCS is CMJ blade. It can be acomodated in to 2-, 6-, 10- и 20-slot MED shelf. For connection to different type of terminal equipment variety peripherals boards are available.

CMJ	Central blade: 8xE1 or 16xE1
SAK	POTS ports. 64x analogue subscribers or 32x analogue subscribers at 1 Erl
SBK	ISDN ports 16x BRA
CLD	Carrier board for analog and command interfaces: The CLD blade can host up to two add-on boards of following type • **PBA**: Powerful ringing generator for LB lines, • **TAC**: LB, 2W/4W trunk, VF trunk, FXO line, 8x2w or 4x4w port, • **TAD**: I/O controller with 8x E&M wires, 8-ports, • **TAE**: 30kOhm trunk line, manual adjustment to the line impedance, 8 port, OTC, • **TAF**: PGS subscriber line, OTC, 2 ports. • **TAG**: I/O controller with 8 universal ports • **TAJ**: HD Voice trunk-line circuit: 7 ports

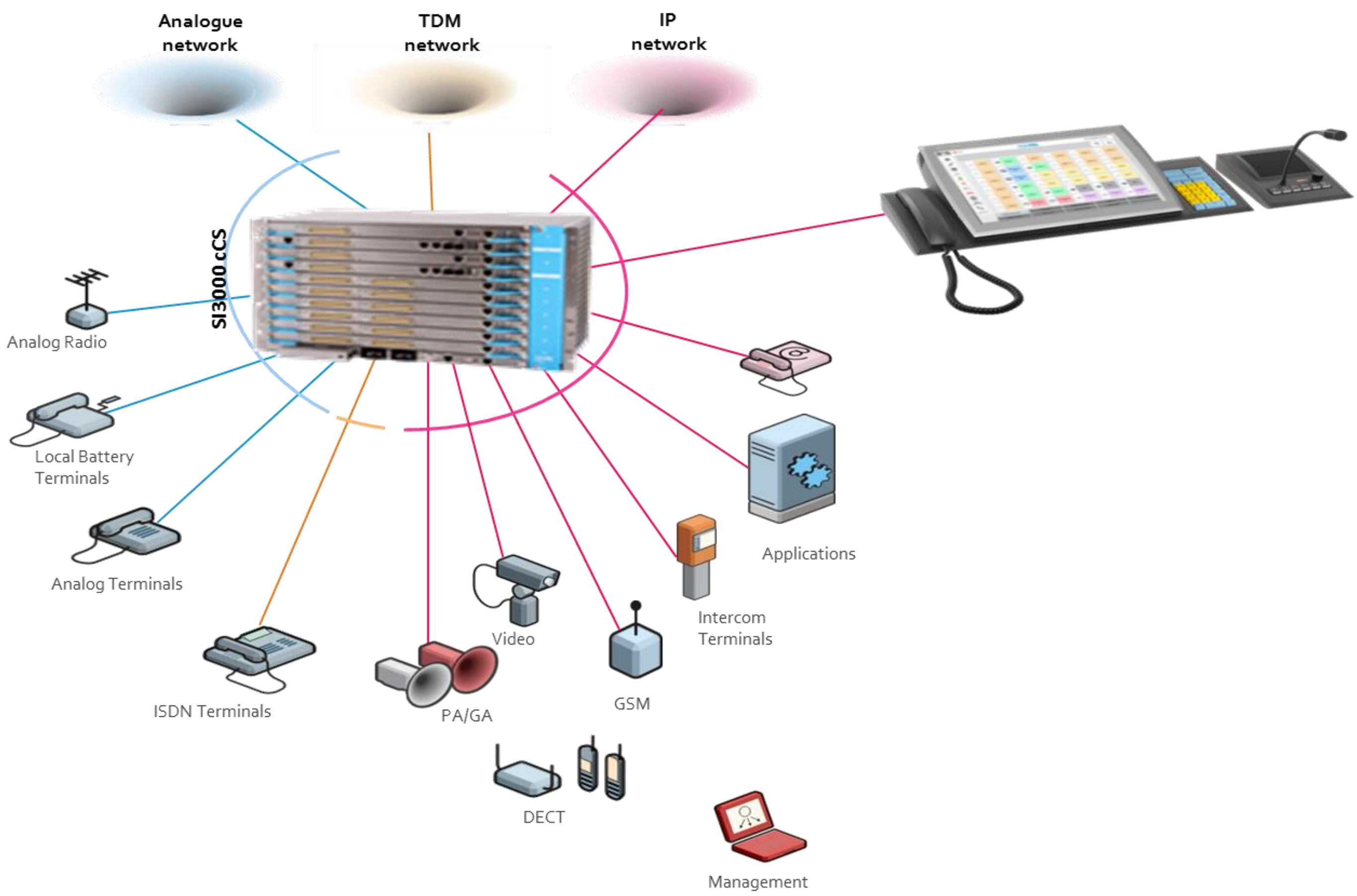

Figure 3 SI3000 cCS together with multifunctional dispatcher terminal MPD can control all types of voice communications in corporate network

2.3.2 MNS SI3000

The SI3000 Management System (SI3000 MNS) is the ultimate management solution for any configuration of Iskratel or third-party network elements in a multi-service network infrastructure.

Basic functionalities of SI3000 MSN:

- Integrated management solution for Iskratel & FMS for 3rd party NE (inventory, alarm monitoring)
- Multiuser centralized architecture
- Mass operations, wizards, …
- Powerful Alarm and Performance Monitoring platforms
- Northbound connectivity for OSS/BSS Integration
- High availability & Geo-redundant operation
- Virtualization

2.3.3 Цифровая запись SI3000

SI3000 Digital Recording Service is a solution which offers the user a possibility to record the calls to or from the switching system. The user of the service can be either public network or enterprise operator, or even an individual subscriber. Each of the authorized user can record calls according to his permissions. According to this, any call, regardless of the source, call direction and type of the used line can be recorded. Calls are recorded and stored in a way which enables secure storing and access to the authorized user.

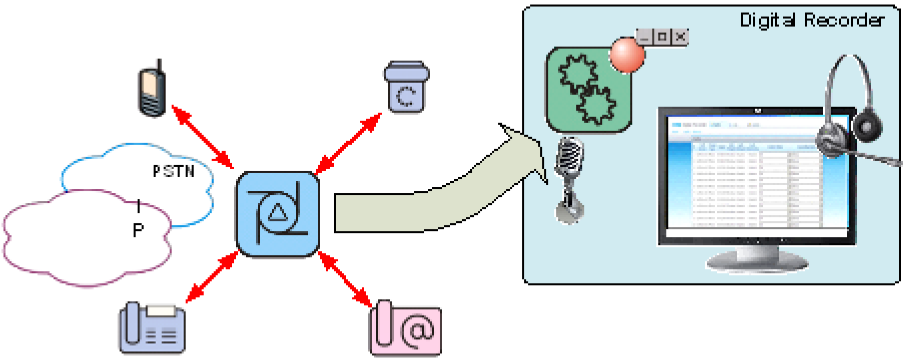

Figure 4 SI3000 DRS

SI3000 Digital Recorder is an application deployed on **SI3000 (AI) API** platform. It enables the user to fulfill a wide aspect of his demands for collecting voice recordings of any connection type, their searching by desired criteria and reproduction.

The Recording Service can be centralized and remote, this means that one Digital Recorder can take care of recording calls from a large number of remote switching nodes.

Main features of the SI3000 Digital Recorder service are:

- Simultaneous recording using up to 400 channels,
- Recording of calls regardless on the subscriber and line access type (analog, ISDN, VoIP, DECT),
- Recording of calls on permanent connected lines using VAD as a trigger,
- Administrative requested recording,
- No wiring of the recording objects, no additional HW probes,
- Centralized and remote recording,
- Web user’s access to recordings via GUI according to his permitions,
- Possibility of browsing, reading (listening), archiving/deleting of the recordings,
- Event history logging;

2.3.4 SI3000 MPD – Conference moderator working place

Si3000 MPD (Multipurpose Dispatcher Terminal - MPD terminal) is attendant managing an individual and conference calls between all types of telephone equipment in dispatcher networks and in operational technological communication networks - OTC.

Figure 5 Si3000 MPD - Multipurpose Dispatcher Terminal

MPD terminal is based on Intel Atom platform and is equipped with highly reliable 15’’ touch screen. Components of dispatcher terminal may differ depending on requirements. In order to achieve suitable configurations of MPD terminal, it can be combined with add-on or standalone modules such as handset, speakerbox, keyboards. Module speakerbox has integrated stereo speakers and a goose-neck microphone with three-color bar-graph display serving as VU meter, volume indicator and six programmable keys with LED indicators for mode control and volume adjustment.

As an additional option MPD also support to connect a PTT foot pedal and wireless headphones.

Figure 6 Wireless headset and PTT foot pedal

The connection toward call control server is made over two Ethernet interface with NIC teaming functionality to solved reliable redundant communication and long distance connectivity issues. It also supports up to 5 SIP accounts in order to achieve high availability in case of one call control server failure. For easy and comfortable managing of MPD terminal remote management is supported. MPD terminal support random alignment administration up to 192 large/432 medium/1024 small size of direct buttons within 12 tabs. Pictures as a background with in tabs are also supported for providing better user experience. The color profiles are supported on direct buttons for flexible configuration of visual calling indications (active call, hold call, priority call, type of phone,…).

MPD terminal supports different configuration of user interface which are tailed according to specific requirements of technological and operational networks. When MPD terminal is configured as conferencing moderator working place the basic GUI interface is as shown on below picture:

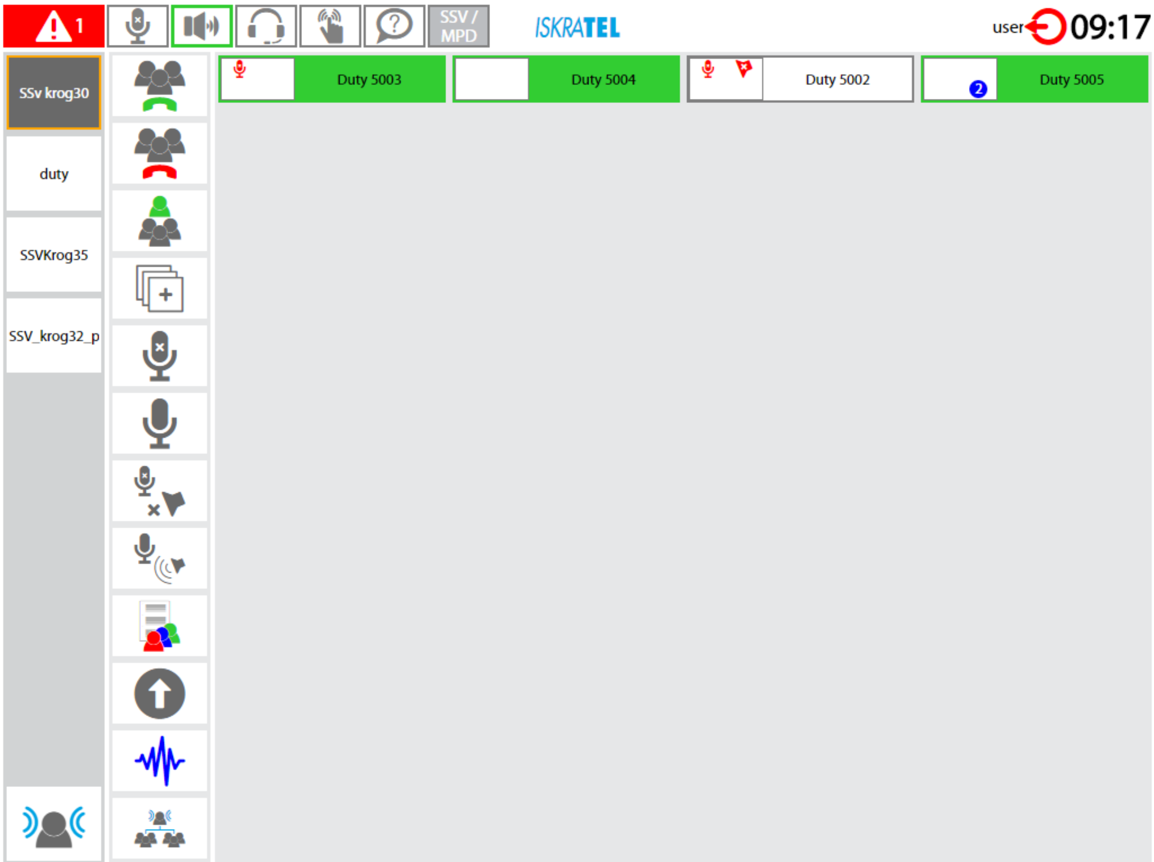

Figure 7 Conferencing user interface on SI3000 MPD

Basic functionalities of SI3000 MPD as moderator working place of conference calls:

- Creation of conferences,
- Creation of conference member list,
- Managing of conference members,
- Calling an IP based members to a conference,
- Calling an Analogue members to a conference,
- Adding a new members into active conference,
- Automatic connection of participants into a conference,
- Rejection of participants that are not authorize members to participate in conference,
- Removing of participants from conference,
- Monitoring a participant activity (status) in a conference,
- Activation of functionality "pereboj" upon participant (rasporjaditelni režim),
- Activation of mute functionality upon participant,
- Monitoring of selected active conference,
- Intrusion (speaking) into selected active conference,
- Monitoring activity of all other active conferences,
- Ability of calling/receiving a individual calls to/with other subscribers via dial pad, phonebook, call queue,
- Managing with max. 9 simultaneous conferences
- Managing of 56 participants in one conference via touch-screen approach
- Individual call from moderators working place (during the conference)
- Initial presets of member’s parameters (e.g.: mute/unmute, speaker On/Off, priority…),

- Presentation of priority status (enabled/disabled) and level of set priority (1 … 10),
- Simultaneous (speech activity) listening of several active conferences on speakerBox,
- Saving a setting of conferences (list of members, initial parameters of members).

SI3000 MPD Technical characteristics		
	Display	15” TFT LCD Multi Touch (1024*768)
	Processor	Intel Atom D525 1.8GHz 1MB L2 cache
	RAM / Disk	4GB DDR3 / 120 Gb SSD
	Interfaces	Ethernet, USB, COM, VGA, Audio Line
	LAN	2x 10/100/1000Base-TX
	USB 1.1 / 2.0 port	2 / 4
	COM port	2
	Audio	Integrated HD audio
	Operating input voltage	100–240 V, 50–60 Hz
	Consumption	100VA
	Power supply	12V / 80W AC/DC covertor includet
	Dimension	409 x 344 x 112 mm
	Weight	6.7 kg

2.3.5 Dedicated terminal equipment for conferencing

2.3.5.1 Studio

Conferencing studio represents one of possible types of conference participant. Studio is always joined in to the conference by moderator. From equipment point of view studio consist of room with audio equipment which enables user to listen and participate in the conference. Each user has gooseneck type microphone equipped with PTT button. To assure intelligibility of voice, high quality unidirectional electret microphones with AGC (Automatic Gain Control) are used. In parallel to PTT button optional external foot pedal can be connected to the microphone. In one studio could be up to 30 microphones.

There are following modes of operation possible:

- Mix mode: in a line made of 2 or more microphones, the audio signal of each microphone (if activated through the PTT button) is always mixed on the shared audio channel with all the others and set in to the conference over the IP SIP Gateway
- Interlocking mode: in a line made of 2 or more microphones, only one at a time can be activated (the first in chronological order). The ‘busy’ indication (LED: flashing red) will appear on all the other paging microphones, which will need to wait until the line is available again.

Properties of microphone:

- Gooseneck microphone with hi-quality unidirectional electret capsule
- Direct connection to power amplifiers input through CAT5 cable
- Adjustable line output with RJ45 connector
- Switchable automatic gain control (AGC)
- Up to microphones can be linked in a ‘daisy-chain’

- Selectable mixing, interlocking or priority mode among devices
- When activating a microphone, the announcement can be preceded by a chime

Studio can connect to conferencing server by following interfaces:

- VoIP studio … connection over SIP protocol (over SIP Gateway)
- Analog studio … connection to conference server over analog trunk

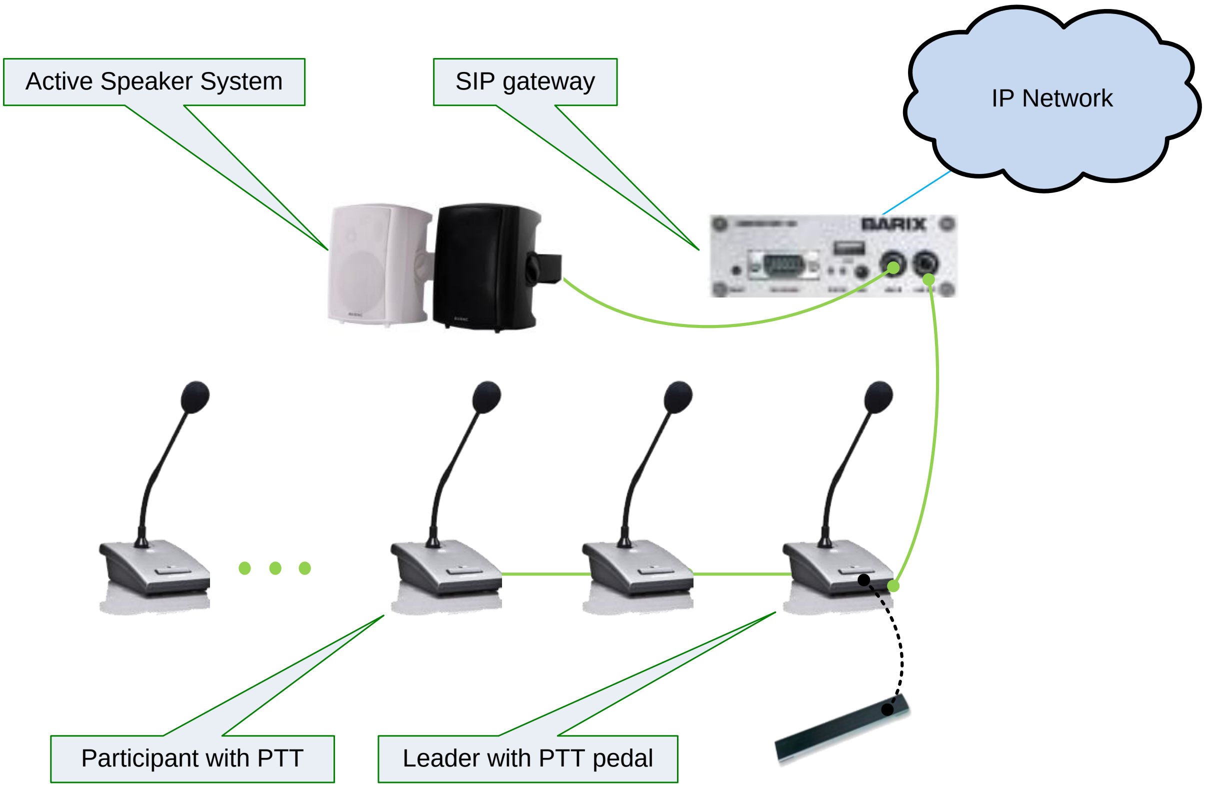

Figure 8 VoIP studio with SIP gateway

SIP gateway (Barix Announcicom 100)

Ethernet	RJ-45, 10/100Mbps auto,
Codex	G.711, G.722
Protocols:	TCP/IP, UDP, RTP, SIP, DHCP, Multicast
Power supply	24V DC, included adapter
Consumption	8W max
Mounting	Table top, rack and wall mountable using accessories

Gooseneck microphone (RFC BM3022)

Type	Desktop paging microphone with electret capsule
Signal / noise ratio	> 72 dB (AGC: ON), > 82 dB (AGC: OFF)
Distortion	< 0.65 % (AGC: ON)
Polar pattern	Cardioid
Sensitivity	– 65 dB ± 3 dB (0 dB = 1 µbar @ 1 kHz)
Impedance	2000Ohm ± 30% (@ 1 kHz)

ISKRATEL

Frequency response	50 Hz ÷ 16 kHz
Connector	RJ 45
Power supply	24 V dc, through included adapter, one adapter can power four microphones
Power consumption	100mA
Included accessories	Cable CAT5 FTP, 5 m
Dimensions	100 mm (l), 48 mm (h), 159 mm (d) (without gooseneck)

Active speakers (Audac)

RMS power 2 x 40 W	2 x 40 W
Sensitivity 1W / 1m	88 dB
SPLmax W/ 1m	101 dB
Frequency response	80 Hz - 20 kHz
Power	IEC Power connector, 230V AC

2.3.5.2 Conferencing terminal for individual participant – Barix PS1

For the needs of the individual conference participant dedicated terminal Barix PS1 is provided. Terminal is easy to use and provides high quality capture and reproductions of audio signal. Terminal is connected over the SIP protocol to conference server. To assure comfortable use of terminal external gooseneck microphone with PTT button is provided. The terminal has a built-in speaker. The volume can be controlled via an external button. Terminal contains two programmable buttons that you can use for joining the conference or receiving the calls. Incoming calls are signalized over sound and LED backlight of buttons. Terminal connects to the conference via the IP network using the SIP protocol.

Basic properties:

- G.711, G.722,
- IP Streaming via TCP, UDP, RTP, Multicast
- External gooseneck microphone
- Speaker
- Target/Function Buttons (2)
- Power over Ethernet (PoE)

PS1 – SIP terminal for conference participant		
	Ethernet	10/100Mbps auto, 802.3af (PoE)
	Protocols	TCP/IP, UDP, RTP, SIP, DHCP,
	Audio format	G.711, G.722
	Speaker power	1 Watt RMS max
	Frequency response)	300 Hz .. 20 kHz (-3dB)
	Sound pressure (1W, 1 meter)	77dB
	Button A	Programmable button with white status LED
	Button B	Programmable button with bicolor status LED (red/green)
	Power supply	802.3af (PoE) Ethernet, included PoE injector 230 B AC
	Mounting	Table top
External PTT microphone		

	Type	Condenser (back electret)
	Polar Pattern	Cardioid
	Frequency Response	50Hz - 18KHz
	Sensitivity	-40dB +/- 3dB at 1 KHz (0dB = 1 V/Pa)
	Impedance	1.8 K Ohms
	S/N Ratio	64 dB(A)

Figure 9 Barix PS1 Conferencing terminal for individual participant

Figure 10 External microphone with PTT

3 Explanation of abbreviations

OTC	Operational Technological Communications
PTT	Push To Talk
AGC	Automatic Gain Control

SIP	Session Initiation Protocol
VAD	Voice Activity Detection
NIC	Network interface controller
LED	Light emitting diode
MPD	Multipurpose Dispatcher Terminal
ISDN	Integrated Services Digital Network
VoIP	Voice over IP
DECT	Digital Enhanced Cordless Telecommunications