

- Functionality of private branch exchange, rural, city, trunk line, combined and international telephone exchanges
- Virtual PBX
- Call center functions
- Teleconference
- Geographical redundancy
- Active-active redundancy mode
- Scalability
- User-friendly management interface

Softswitch ECSS-10 is a hardware and software platform designed for integrated infocomm networks construction. The software and hardware components of ECSS-10 were developed and manufactured by ELTEX and have a high level of reliability.

Application

4/5 class Softswitch ECSS-10 is a flexible system for any level communication center construction: departmental networks, enterprise networks and provider networks (local, zonal, transit, intercity, international).

Key features

- More than 100 000 subscribers
- AutoProvision (AuP)
- Certified as private branch exchange, rural, city, trunk line, combined and international telephone exchange
- Virtual PBX (vPBX)
- Call center
- Teleconference
- Operation as a SaaS platform
- Support for Session border controller functions (SBC)
- Group notification
- Support for Ubuntu/Astra Linux
- Geographic redundancy
- Local redundancy
- Hot software update
- Load balancing
- Flexible graphical IVR builder
- Support for widely used CRM systems, integration with customer CRM
- Operation under KVM and VMa Ware
- Support for TTS (Text to speech) and ASR (Automatic speech recognition)
- Subscriber portal
- SSW node licensing with License Manager

Scalability

ECSS-10 modular architecture provides its scalability. It allows using the solution in small corporate communication centers as well as in international transit stations.

Virtual PBX

ECSS-10 supports Virtual PBX service. It allows connecting subscribers to dedicated Virtual PBX with private dial plans, modern services, billing reports, etc. Customers obtain up-to-date VoIP services without additional expenses for installing and maintaining of a standalone PBX.



Mobile VoIP

Mobile IP telephony provides telecommunication services to remote subscribers and helps to optimize roaming costs. The mobile VoIP service can be used anywhere **via the Internet** due to session border controllers (SBC). To access this service, just install a SIP application on your mobile device and activate an account.

With the FMC service, a subscriber mobile phone can be connected to a corporate network even without the local network.

Monitoring and management

All software and hardware components of ECSS-10 Softswitch are developed and produced by Eltex. That allows implementing a unified monitoring and control interface via the Eltex.EMS system. The system provides a customer with easy-to-use management tools: elements aggregation, centralized configuration and firmware version management, scheduled maintenance, main parameters monitoring in a single window.

AutoProvision (AuP)

AutoProvision subsystem is used for centralized configuration of phones and VoIP gateways. The subsystem allows uploading configuration files to end-user devices automatically. Equipment from a wide range of vendors is supported: Eltex, Yealink, Cisco, Grandstream, Snom, Siemens, Fanvil, etc.

AutoProvision subsystem provides automatic configuration for different subscriber accounts and transparent replacing of a phone (not only by another model, but by a device from another vendor as well). After replacing a phone, the configuration will be adapted automatically.

The subsystem is able to synchronize a list of subscribers with ECSS-10 to keep the AutoProvision user base up-to-date.

Fault tolerance

The cluster architecture of ECSS-10 Softswitch allows achieving 99.9999% reliability. Due to the local active-active redundancy and geographic redundancy, any single hardware failure won't be able to affect calls at any stage of their processing.

SIGTRAN

The SIGTRAN (M2UA, MTP3 and ISUP) subsystem is implemented on ECSS-10. The redundancy system for signal and media traffic is supported. H.248/MEGACO are used as gateway management protocols. Supported transport protocols are SCTP, UDP and TCP.

Features and capabilities

Supported protocols¹

- SIP 2.0 (RFC 3261)
- H.248/Megaco
- RADIUS AAA
- ISUP
- SIP-T/SIP-I
- T.38
- SNMP
- SIGTRAN

Protocols supported via gateways¹

- SS7
- CAS
- V5.1, V5.2
- R1.5, R2
- EDSS-1/Q.931
- Coral IPNET

Supported audio codecs¹

- G.729A/B
- G.711A/U
- G.726
- G.723.1 (5.3, 6.3 kbps)
- G.722, G.722.1, G.722.1c
- GSM FR
- iLBC
- Speex
- L16
- OPUS

Supported video codecs¹

- H.263-1998
- H.264

Management interfaces

The main means that provide efficient management and access rights delimitation:

- MML console (SSH) (CoCon)
- Web 2.0 interface HTTP(S) (ECSS 10 Configuration manager)
- Web interface for call center (ARM Call Center)
- Web interface for subscriber's VAS management (Subscriber portal)
- Customizable web interface for virtual PBX management
- Web interface for teleconference management

Redundancy

- Support for hardware redundancy (active-active)
- Hot-swappable software modules
- Support for geographic redundancy

Telephone routing¹

- Route selection based on the parameters:
 - Calling party number (CgPN)
 - Calling party category (CPC)
 - Called party number (CdPN)
 - Subscriber group ID
 - Nature of address (NoA)
 - Numbering Plan (NP)
 - Calling Party Address Presentation Restricted Indicator (Calling Party APRI)
 - Weekday
 - Time of day
 - Gateway/direction load levels

- Tag
- Redirecting number
- Original Called Party Number
- Numbers modification
- Flexible management of call processing by a graphical scenario
- Call center organization, flexible routing among queues
- Support for external routing via RADIUS and HTTP
- DisplayName search by phone number on an external service (2gis, Yandex and others) (InfoByPhone)

Teleconference

- Up to 50 participants in a conference
- Support for conference templates
- Support for VoIP SIP phones and extension panels
- Flexible system of access rights to conference templates and conferences
- Easy-to-use improved web interface
- Conference planning
- The number of available active teleconferences is determined by license

Call charging

- RADIUS Accounting
- CDR files

Call center

- Corporate phonebook
- Possibility for an agent to work with a phone only (without a PC)
- Web workstation with a wide function set for calls processing
- Supervisor web workstation for call center monitoring
- Call center settings management via the web interface
- Flexible configuration of agent's rights
- Wide range of call distribution algorithms
- Smart prediction of queuing time
- Call statistics collecting and reporting
- Call prioritization
- Color and sound indication of priority calls in the operator and supervisor workstations
- Call distribution according to agent's qualification
- Collection of subscribers' feedback on call center agents performance
- Queue hierarchy
- Call pickup from a queue
- Supervisor driven manual mode for calls distribution in a queue
- Support for Callback feature in a queue

Additional functions

- Support for different media resources formats
- SIP Registrar
- Authentication via LDAP and/or RADIUS
- Session border controller functionality
- Secure media streams using SRTP

¹ The list might be extended upon a request

Features and capabilities (continued)

Additional functions (continued)

- Connecting subscribers with incompatible codecs
- IPv6 for modern IP networks
- Text to speech/Automatic speech recognition engine for services
- Renewed subscriber web interface
- Email2Fax and Fax2Email services
- Reconfigurability – the opportunity to increase performance and functionality
- Integration with Microsoft Active Directory
- Text message transmission
- Trunk based black and white lists
- Trunk based CPS limiting
- Load balancing among several media servers
- Location based media traffic routing
- Phonebook synchronization
- Direct RTP forwarding mode
- Voice traffic recording
- Support for Distinctive ring and Distinctive picture
- Caller ID characters encoding conversion
- Multi-user video conference (up to 5 subscribers) through internal conference server
- Integration with CRM: AmoCRM/Megaplan
- Requesting external catalogues of subscribers/companies for substitution of displayed names/companies according to phone number
- Multicast IP Paging, Multicast IP Listen
- Forwarding of external and local calls to different phone numbers
- Semipermanent links
- Integration with Skype for business
- Integration with PhoneUP

Value Added Services (VAS)¹

Call group services

- Call Hunt
- Group call with optional hidden redirection (CGG)
- Boss group
- Zone Page
- Group Pickup
- Auto Attendant
- Direct Inward Dialing

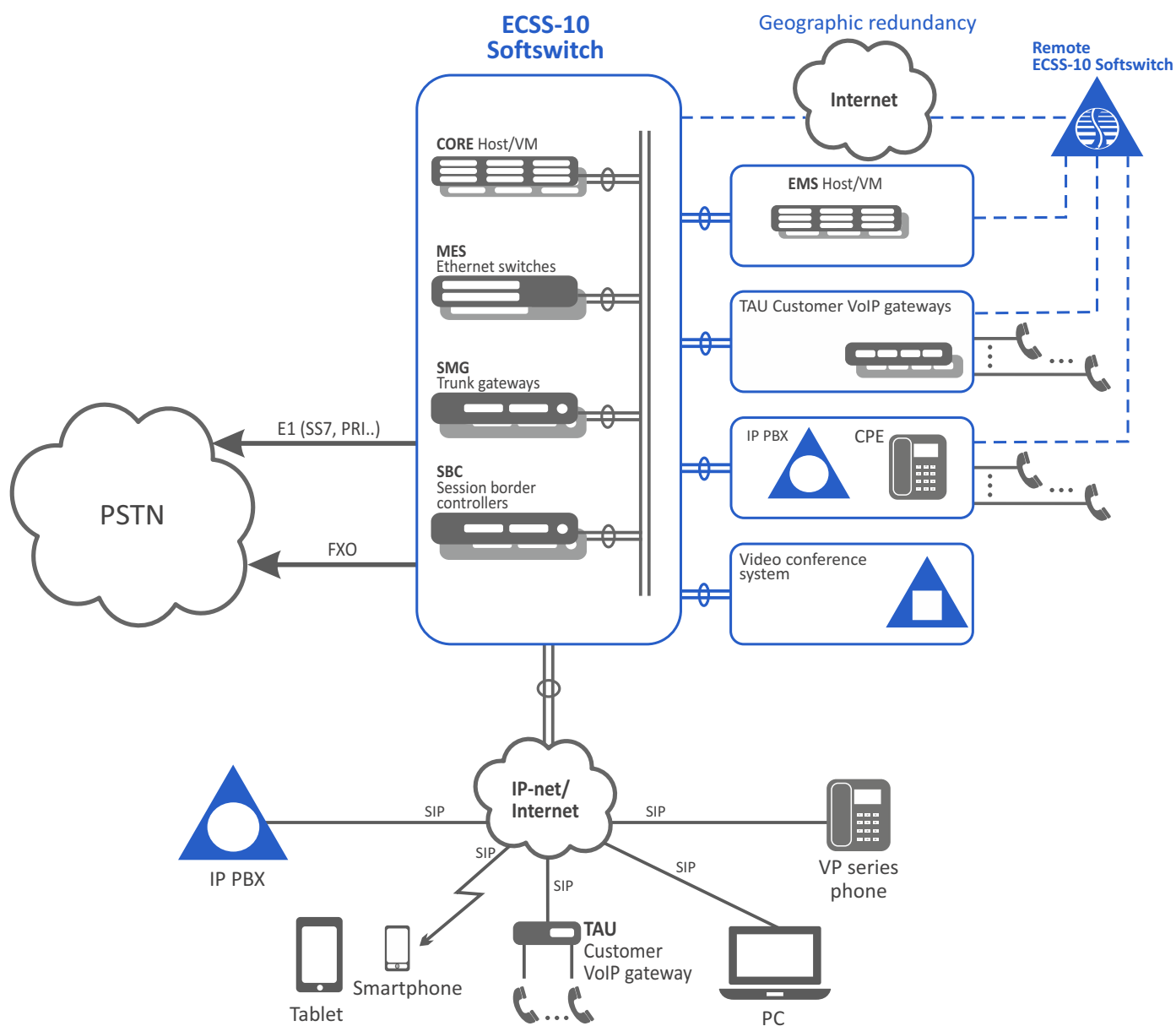
Subscriber services

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Calling Line Identification Restriction Override (CLIRO)
- Calling Name Identification Presentation (CNIP)
- Calling Picture Identification Presentation (CPIP)
- Customizing Ring Back tone
- Personal voice notification/intercom (Voice Page)
- Call intervention
- Distinctive Picture
- Distinctive Ring

- Anonymous call barring (ACB)
- Call forwarding: unconditional, busy, no response, out of service, time based, based on Caller ID (CFU, CFB, CFNR, CFOS, CFT, CFCID)
- Hidden call forwarding (HideCFName)
- Call forwarding barring: outgoing calls (FBC), incoming calls (RFC)
- FindMe (forwarding to a group of number), FindMe No Response
- Follow Me, Follow Me No Response
- Call forwarding through subscriber SIP terminal (CFSIP)
- Call Waiting service (CWAIT)
- Multiline
- Call Hold (CHOLD)
- Call Pickup
- Call Transfer (CTR)
- Call Park
- 3-way Conference
- Conference call, Add-on
- VIP calls
- Walkie-talkie mode
- Selective Call Acceptance, Incoming Whitelist
- Selective Call Rejection, Incoming Blacklist
- Selective Call Origination, Outgoing Whitelist
- Selective Call Origination, Outgoing Blacklist
- Missed call notification
- VAS management via a phone
- Call recording
- Video Call Recording
- Voice Mail
- Message Waiting Indication (MWI)
- SIP Presence and Busy Lamp Field
- Call back
- Auto Redial, Auto Redial with Call back
- Redial
- Speed Dial
- Do Not Disturb (DND)
- Alarm call
- Malicious call Identification (MCID)
- My Number
- Outgoing calls barring (PBR)
- Authorization on a remote phone (Remote phone)
- Duplicating incoming calls to additional internal or external number (FlexiCall)
- Introducing message for called subscriber (Introduce)
- Second handset
- Smart cancel
- Subscriber IVR script
- Privacy – intervention barring
- Adjustable remote ring tone
- Fax-to-Email, Email-to-Fax

¹ The list might be extended upon a request

VoIP node design scheme



ECSS-10 is a complete solution for integrated infocomm networks construction

All the software and hardware components (SBC, Ethernet switches, digital and customer gateways, access gateways, IP-phones) are designed and manufactured by Eltex:

- **ECSS-10 Softswitch** – a hardware and software system with capability of customer equipment autoconfiguration.
- **Eltex.EMS** – a centralized management system.
- **Server** – the servers are selected by performance requirements. You can use a virtual server or a physical server of any manufacturer.
- **MES series switches** – carrier grade Ethernet switches with stacking support and high throughput.
- **SMG** – signaling gateways SMG-1016M, SMG-2016 or SMG-3016 for interfacing of signal and media streams of TDM and VoIP networks.
- **TAU** – subscriber gateways TAU-XX.IP used for voice and fax data transmission via IP network. The gateways provide users with high-quality telephony service with support for an isolated office PBX mode and a basic set of value added services.
- **SBC** – session border controllers (SBC-1000, SBC-2000 or SBC-3000) designed to protect internal VoIP network against untrusted networks threats and to hide the network topology.
- **Eltex VP-XX IP phones** – modern IP phones with an integrated router which allows connecting a PC and IP phone via a single physical line. The models with «P» letter in the name support Power over Ethernet technology. The devices are perfect solutions for companies with high requirements to voice data transmission.

To obtain more information on the license policy, contact the managers of Eltex commercial department: voip@eltex-co.ru

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About Eltex

Eltex Enterprise is a leading Russian developer and manufacturer of communication equipment with 30 years of history. Complete solutions and their seamless integrability into the Customer's infrastructure are the priority growth areas of the company