ELTEX Integrated Networking Solutions

– 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

ECSS-10

Softswitch ECSS-10

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
– Certified as private branch exchange, rural, city, trunk line, combined and international telephone exchange
– Virtual PBX
– Call center
– Teleconference
– Operation as a SaaS platform
– Support for Session border controller functions
– Support for a wide range of VAS (Value Added Services)
– Group notification
– Support for Astra Linux
– Geographic redundancy
– Local redundancy
– Hot software update
– Load balancing
– Flexible 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

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 Internet. In this case, all the data will be transmitted via GSM channels.

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

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

Firmware version 3.14.6

ELTEX
Integrated Networking Solutions

Features and capabilities

Supported protocols*

– SIP 2.0 (RFC 3261)
– H.248/Megaco
– RADIUS AAA
– ISUP
– MTP3
– SIP-T/SIP-I
– T.38
– SNMP
– M2UA

Protocols supported via gateways*

– SS7
– CAS
– V5.1, V.5.2
– R1.5, R2
– EDSS-1/Q.931
– Coral IPNET

Supported audio codecs*

– 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
– AMR
– OPUS

Supported video codecs*

– H.263-1998
– H.264

Management

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

– MML console (SSH)
– Web 2.0 interface HTTP(S)
– Call center web interface
– Subscriber portal for subscriber's VAS management
– 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 routing*

– Routing by mask
– 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
  - Presence of a number in a number list
– 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 identification by phone number from an external database (2gis, Yandex and others)

Teleconference

– More than 200 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
– Wide range of call distribution algorithms
– Smart prediction of queuing time
– Call statistics collecting and reporting
– Call prioritization
– 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

** The list might be extended upon a request*

ELTEX Integrated Networking Solutions

Features and capabilities

Additional functions (continuation)

– 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
– Transscrambling — speech to text conversion engine
– 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

Value Added Services (VAS)*

Call group services

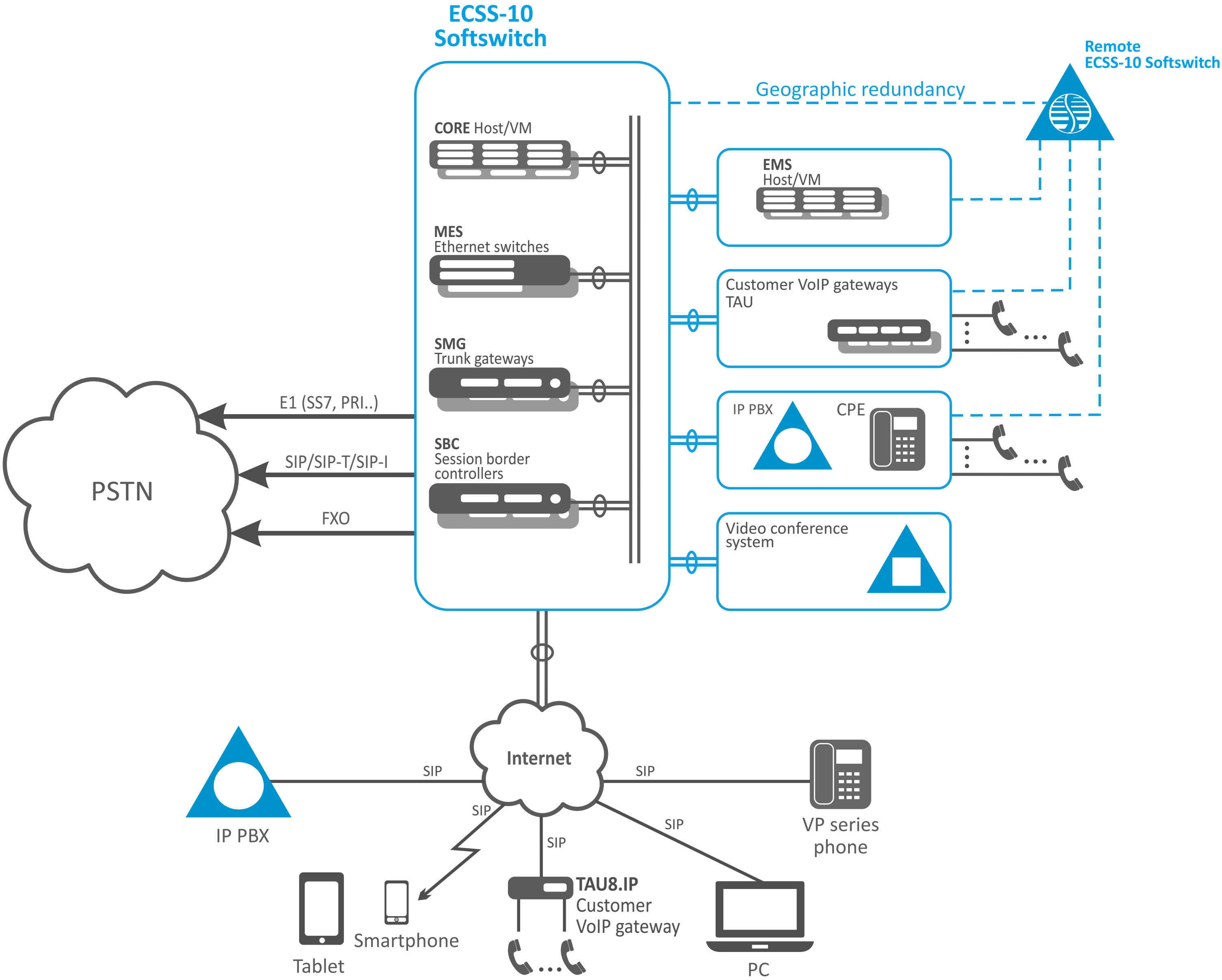
– Call Hunt
– Group call with optional hidden redirection (CGG)
– Boss group
– Zone Page
– Group Pickup
– Auto Attendant
– Direct Inward Dialing

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)
– Find me (forwarding to a group of number), Find me on No Response
– Follow me, Follow me on 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 (RBP)
– Hotline/Warmline
– 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

** The list might be extended upon a request*

VoIP node design scheme



Integrated Networking Solutions

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 or SBC-2000) 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

Contact us

+7 (383) 274 10 01
+7 (383) 274 48 48

eltex@eltex-co.ru

www.eltex-co.com

About ELTEX

Eltex company is a leading Russian developer and manufacturer of telecommunication equipment with more than 25 years of history. Integrity of solutions and seamless integration capability into Customer infrastructure is a priority area of company development.