

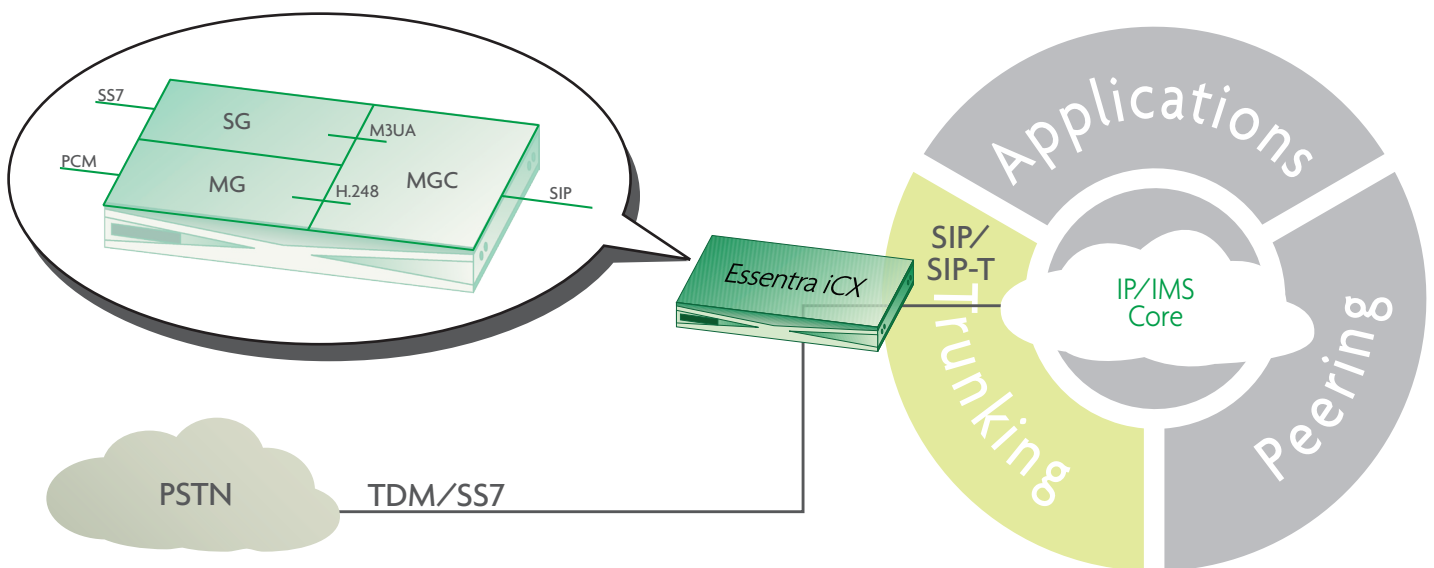
Essentra™ iCX: Integrated SIP-to-SS7 Solution

Operators of national long distance and international wholesale and retail long distance networks are deploying VoIP-based next-generation networks (NGNs) to reduce operating costs and enhance service flexibility. These service providers are looking for cost-effective solutions to connect their VoIP networks to the PSTN.

The Essentra iCX is a compact SIP-to-SS7 solution specifically designed to meet service providers' needs. It features a standards based solution incorporating VocalTec's Essentra iCX - media gateway controller together with Audiocodes' Mediant™ trunk gateway. Its embedded SIGTRAN (M3UA) support and Megaco/H.248 control provides an integrated IMS/TISPAN SIP-to-SS7 solution.

Essentra iCX incorporates VocalTec's media gateway controller, a scalable carrier grade console switch that offers high-quality voice services, carrier grade reliability and maximum service flexibility. With its open interfaces to non-proprietary media devices and application servers, Essentra iCX facilitates the creation of best-of-breed network solutions.

The Essentra iCX SIP-to-SS7 solution enables fast and simple deployment of NGNs, while maintaining seamless connectivity to PSTN/SS7 services. In addition to controlling media gateways, Essentra iCX provides intelligent call control, advanced routing, and PSTN signaling support, providing comprehensive control and interworking functionality.



Features & Benefits

- Media gateway controller, media gateway and signaling gateway all in a 1U chassis
- Intelligent and flexible call routing maximizes business opportunities and network efficiency
- Unmatched SS7 capabilities, including full SS7 backhauling via SIGTRAN (M3UA) and support for over 60 SS7 variants enabling large-scale global deployments
- Vendor neutrality facilitating integration within multi-vendor environments
- Carrier-grade reliability based on NEBS certified hardware ensuring 99.999% availability
- Multi-protocol interworking, supporting SIP/SIP-I/SIP-T/Q.1912.5, Megaco/H.248, MGCP, SS7/ISUP and SIGTRAN, allowing for quick and easy deployment
- Enhanced scalability enabling cost-effective incremental growth of ports, call capacity and services

Specifications:

VoIP Protocols

- SIP (RFC 3261, 3262, 3264, 2976, 3323, 3325, 3326, 3455), SIP-T (RFC 3372, 3204), SIP-I (Q.1912.5)
- Megaco (RFC 3525, H.248.1), fax packages (H.248.2), error codes and service change reasons (H.248.8), generic announcements (H.248.7)
- MGCP (RFC 3435)
- SDP (RFC 2327)
- Fax support: T.38 fax relay
- DTMF handling: RFC 2833, in-band and out-of-band

PSTN Signaling

- Support for over 60 SS7 ISUP variants, including ITU-T, ETSI, ANSI
- Sigtran/M3UA interface to signaling gateway
- Glare resolution (Q.764)

Carrier Grade

- 1+1 high availability
- 99.999% availability
- No single point of failure
- No downtime and no call loss
- NEBS (level 3) compliant

Call Control Functionality

- Channel Selection Order: ascending/descending, random, odd/even, least recently used idle, cyclic
- Overlap sending and receiving
- CLIP/CLIR support
- Call progress tones and recorded announcements

Routing

- Policies: source, time, random, weighted random, locality, prefix, codec
- Least Cost Routing (LCR), for near-real-time optimization of routing policies
- Alternate endpoints for improved call completion
- Calling party and called party number analysis
- Calling party and called party number manipulation
- Routing-prefixes

Security

- Pinhole firewall
- SIP authentication

Interoperability

- Tuning of SIP protocol options using 'Interop toolkit'
- Manipulation of ISUP parameters using scripts

Accounting

- Successful and unsuccessful calls
- CDR files for off-line billing (through FTP)
- Radius accounting for real-time billing

Capacity

- Up to 16 E1/T1 spans
- 8 signaling links

Management

- Standards based: HTTPS, XML, SNMPv3
- Built in EMS

Hardware Specifications

- Single universal 90-260 V AC dual redundant AC / single -48 V DC
- 1U high, 19-inch wide rack mount

Operating System

- Red Hat Enterprise Linux 4
- VxWorks

About VocalTec

VocalTec Communications (Nasdaq: VOCL) is a global provider of carrier-class multimedia and voice-over-IP solutions for communication service providers. A pioneer in VoIP technology since 1994, VocalTec provides proven trunking, peering and residential/enterprise VoIP application solutions that enable the flexible deployment of next-generation networks (NGNs). Partnering with prominent system integrators and equipment manufacturers, VocalTec serves an installed base of dozens of leading carriers including Deutsche Telekom and Telecom Italia San Marino. VocalTec is led by a management team comprised of respected industry veterans.

www.vocaltec.com