Trunking

Essentra™ CX: Media Gateway Controller/SS7-Enabled SIP Gateway

Operators of wholesale and retail long-distance networks are migrating their legacy Trunking networks to VoIP-based next-generation networks (NGNs) to reduce operating costs and enhance service flexibility. At the same time, VoIP service providers are looking for a cost-effective solution to connect their networks to the PSTN.

Essentra CX media gateway controller is a scalable, carrier-grade SIP-based MGC 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 CX facilitates the creation of best-of-breed network solutions.

Essentra CX is offered in two configuration options:

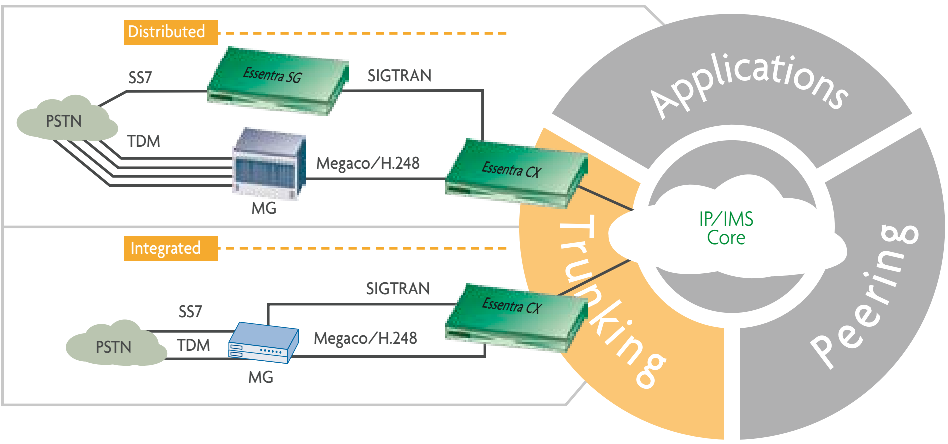
1. An integrated media gateway and signaling gateway controlled by the Essentra CX; a solution well suited for deployments requiring fewer E1/T1 spans and signaling links.
2. The Essentra SG signaling gateway and a separate media gateway both controlled by the Essentra CX; this solution enables increased E1/T1 capacity and signaling links.

Essentra CX enables smooth migration to NGNs, while maintaining seamless connectivity to PSTN services.
In addition to controlling the media gateway, Essentra CX features intelligent call control, an advanced routing engine, and SS7/PRI signaling support. A flexible and scalable solution, Essentra CX facilitates the deployment and migration of existing networks to NGNs.

Taking advantage of a large SS7 protocol library and supporting industry-standard control and signaling protocols, including Megaco/H.248, MGCP, SS7/ISUP, ISDN PRI and SIGTRAN (M3UA and IUA), Essentra CX provides service providers with the capability to seamlessly route voice and data calls between the PSTN and SIP-based packet networks. Supporting both traditional and converged services, Essentra CX is the ideal SIP-to-PSTN solution for today's carriers.

IMS-TISPAN pronto: MGCF, BGCF, SGF, T-MGF

HD VoIP



Features & Benefits

- Intelligent and flexible call routing maximizes business opportunities and network efficiency
- Unmatched SS7 capabilities, including full SS7 tunneling via SIGTRAN and support for over 60 SS7 country variants, enable large-scale global deployments
- Vendor neutrality facilitates integration within multi-vendor network environments
- Multi-protocol interworking supporting SIP/SIP-T/Q.1912.5, Megaco/H.248, SS7/ISUP, ISDN PRI and SIGTRAN (M3UA/IUA), allows for quick and easy deployment
- Carrier-grade reliability based on NEBS certified hardware and full component redundancy ensures 99.999% availability
- Enhanced scalability enables cost-effective incremental growth of ports, call capacity and services

VocalTec

Expanding the Borders of VoIP

Specifications:

VoIP Protocols
- SIP (RFC 3261, 3262, 3264, 2976, 3323, 3325, 3326, 3455, 3515, 3891, 3892), SIP-T (RFC 3372, 3204), SIP-I (Q.1912.5), SIP I-Ds (draft-levy-sip-diversion-10, draft-mahy-iptel-cpc-06)
- 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, V.152, fallback to high bitrate codecs
- 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/IUA interface to signaling gateway
- EuroISDN (DSS-1) PRI, network side

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

Routing
- Policies: source, time, random, weighted random, locality, prefix, ASR, ACD ,calling party category, bearer capabilities, online
- 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

Call Control Functionality
- Channel Selection Order: ascending/descending, random, odd/even, least recently used, cyclic
- Overlap Sending and Receiving
- CLIP/CLIR support
- Call progress tones and recorded announcements

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
Capacity per single server
- 480 E1/T1 spans
- 100 Call Attempts Per Second (CAPS)
- 32 redundant SS7 Signaling Gateways
- 256 local SS7 Point Codes
- 50 Media Gateways

Management
- Standards based: HTTPS, XML, SNMPv3
- Built-in EMS, or optional Essentra OSS
- Real-time call monitoring

Hardware Specifications
- IBM BladeCenter-T/HT
- IBM xSeries 3550

Operating System
- Red Hat Enterprise Linux 4

About VocalTec

VocalTec Communications (NasdaqCM: 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. VocalTec is led by a management team comprised of respected industry veterans.

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