

mSwitch[®] Call Server Processor (CS-P)

NEXT GENERATION CALL CONTROL FOR TRUNKING, ACCESS AND SERVICES

A SCALABLE, ROBUST, SOFTSWITCH CALL SERVER PROVIDING CALL CONTROL AND USER ACCESS FOR BROADBAND BASED TELEPHONY SERVICE



- **COST EFFECTIVE**
- **MODULAR**
- **CARRIER CLASS**
- **SCALABLE**
- **HIGH CAPACITY**
- **STANDARDS - COMPLIANT**
- **CONVERGED WIRELINE, WIRELESS AND BROADBAND ACCESS**

UTStarcom Inc.'s mSwitch[®] CS-P enables service providers to offer both the PSTN and the PLMN telephony services to residential and business customers over broadband-based IP networks. This next generation platform provides seamless integration with existing access networks and providers. Subscribers access their voice services via packet telephony voice devices such as IP phones, IAD and VoIP access gateways.

When used in conjunction with our Trunk Gateway and Signaling Gateway, the CS-P provides call routing and enables connectivity across multiple network domains: PSTN, PLMN, SS7, PHS, third-party VoIP and others. The mSwitch CS-P gives you the power of wireline/wireless convergence with its IMS-enabled architecture. CS-P enables multimedia communication services for fixed and mobile subscribers, including support for major services such as calling supplementary services, IN, CENTREX services, and value-added services (VAS).

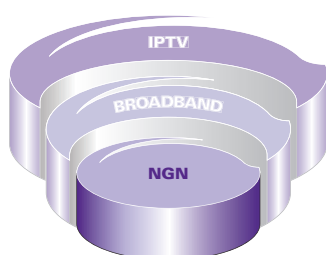
The mSwitch CS-P delivers a high capacity switching platform, that scales up to millions of subscribers. This IP infrastructure helps maximize your return on investment by providing highly profitable next generation services while protecting your investment. This is accomplished by investment using a scalable architecture that can be integrated with legacy PSTN/PLMN infrastructure for seamless call service.

KEY FEATURES AND BENEFITS

Carrier-Class Communications– The CS-P incorporates carrier-class design elements in their hardware design, switching design, PSTN/PLMN signaling, multi-VoIP protocol implementations, and load sharing power supplies are a few of the ways in which we can provide outstanding telephony services over IP-based networks.

Cost Effective High Technology– A design approach that rides both the technology cost and performance curve, delivering field proven platforms and applications, with hardware that has the lowest cost, highest scalability and best performance.

High Capacity– Enables a modularized, cost-effective highly scalable solution.



Standards Compliant – IETF MEGACO ITU H.248 : SIP, SIP-T, MGCP, ITU H.323 : SIGTRAN and SNMP standards compliant.

Field Proven– Built upon an architecture currently deployed worldwide.

Rich Set of Supplementary Features– Our set of supplementary features enable service providers to deliver the latest revenue generating features and suite of services to their customers.

Wireline/Wireless Convergence– Integrates with the both the PSTN and the PLMN to deliver toll quality voice and support for a suite of IN and Value Added Service (VAS) capabilities.

SYSTEM CHASSIS

Dimensions: 622mm (24.5")(14U) H x 482mm (19") W x 432mm (17")D (Rack Mountable)

Temperature: -5° to 55°C (23° to 131°F) (Operating)

Humidity: 5% to 95% non-condensing

Power: -40v to -72v DC

Rack capacity: Up to six redundant CS-P module pairs per chassis and three chassis per standard 7 ft rack

Compliance: FCC Part 15 Class A, UL 1950, Canadian-UL, NEBS Level 3, CE Mark, CB Mark, VCCI, RoHS

INTERFACES

- IP: Redundant 10/100BaseT and GE

STANDARDS / PROTOCOL SUPPORT

- Megaco v1 ITU H.248
- SIP, SIP-T, SIP-I with RFC3261 and related SIP standards
- ITU H.323
- MGCP V1
- SIGTRAN, IUA, M2UA, M3UA
- ITU and ANSI ISDN user-network interface layer 3 specification for basic call control standard Q.931
- ITU and ANSI ISUP
- ITU and ANSI TCAP
- SNMP

RELIABILITY

- Redundant Modules
- Geographic Redundancy
- Hot Swap capability
- Redundant DC feeds
- N+1 Hot Swap/Load Sharing Power Supplies
- NEBS Level 3 Compliant

MANAGEMENT

- Graphical User Interface
- User Self provisioning of all applicable features
- Password Modification for user self service
- SNMP for configuration, monitoring and troubleshooting
- In-service software upgrades
- Enhanced call data records

CAPACITY (Fully Redundant Configuration)

- Up to 200K BHCA or 4000 Erlang or 20K SIP VoBB subscribers per redundant CS-P module pair
- Up to 800K BHCA or 16K Erlangs or 80K SIP VoBB subscribers per chassis
- Scalable with multiple chassis

Please note the information contained herein is for informational purposes only. Technical claims listed depend on a series of technical assumptions. Your experience with these products may differ if you operate the products in an environment, which is different from the technical assumptions. UTStarcom reserves the right to modify these specifications without prior notice. UTStarcom makes no warranties, express or implied, on the information contained in this document.

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About UTStarcom, Inc.

UTStarcom is a global leader in IP-based, end-to-end networking solutions and international service and support. The company sells its broadband, wireless, and handset solutions to operators in both emerging and established telecommunications markets around the world. UTStarcom enables its customers to rapidly deploy revenue-generating access services using their existing infrastructure, while providing a migration path to cost-efficient, end-to-end IP networks. Founded in 1991 and headquartered in Alameda, California, the company has research and design operations in the United States, China, Korea and India. UTStarcom is a FORTUNE 1000 company. For more information about UTStarcom, visit the company's Web site at www.utstar.com

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CALL CONTROL CAPABILITY

- On-net to on-net calling over different protocols
- On-net to PSTN and PLMN calling
- Time based/least cost routing to PSTN
- ETSI Lawful intercept
- Emergency Calls †
- LNP †
- Toll Free Calls †
- SIP Session Timer
- Special (HEX) Digit Analysis
- Invalid Number Handling

SUPPLEMENTARY FEATURES†

- Centrex Feature Set
- Fixed Mobile Convergence Feature Set
- Parlay Based Audio/Video Conference
- PBX Support
- Call Forwarding (Universal, Ring no answer, Busy, Operator Activated)
- Call Waiting
- Conference Calling (Three-way and six-way)
- Call Transfer
- Speed Dialing
- Coin phone support
- Anonymous Call Reject
- Automatic Callback
- Automatic Recall
- Caller ID and Caller ID on Call Waiting
- Caller ID suppression (per-call and all-calls)
- Outgoing Call Block
- Incoming call screen
- Voice mail integration
- Voice mail message waiting indicator
- Distinctive Ring

† Feature localization may be required