Your carrier grade ip telephony platform

sip:provider

The fastest and most cost-effective way to start Next Generation VoIP Services

Product Positioning

sip:provider is a turnkey platform targeting global markets with small/medium size deployments in the range of up to 50.000 subscribers for fixed, converged and wireless service providers supporting a variety of access technologies like Cable, xDSL, FTTx, WiFi and WiMAX.

Platform Design

The design of the system is focused on easy manageability while still offering cutting edge next generation communication features. The sip:provider platform is designed to provide an all-in-one system for various IP Telephony and Next Generation Communication services. All components of the sip:provider appliance are fully SIP standard compliant.

Services

sip:provider delivers a comprehensive suite of voice and multimedia services for residential and business subscribers.

Modules

The platform integrates Call Routing Services, Session Border Controller functionality, Pre- and Post-Paid Billing, Monitoring and Reporting Systems and Customer Web-Interfaces designed for implementations in mixed VoIP/PSTN and pure VoIP telecom infrastructures.

Performance

The sip:provider platform supports 50 call attempts per second (CAPS) resulting in 180.000 busy hour call attempts (BHCA) for deployments in the range of up to 50.000 subscribers.

Hardware

The full system comes as a pre-installed rack mountable 1U appliance on IBM Hardware.

High Availability

High availability is provided by deployment of two sip:provider appliances in active/backup mode.

Deployment

In an NGN architecture, the sip:provider is a full-feature platform that can be connected directly to voice carriers using SIP-Trunks and to softswitches / media gateways with SS7 routing, passing calls from and to PSTN.

Investment Protection

Once the customer base exceeds 50.000 subscribers, the installed base can easily be migrated to the powerful sip:carrier platform, supporting millions of subscribers with full investment protection.

Services

The sip:provider platform is designed to provide Next Generation Services for residential and business customers. Its features include:

- Voice and Video Telephony
- Instant Messaging (IM)
- Presence and Location Based Service
- Voice/IM conferencing

Architecture

The open architecture, solely based on standardized protocols, allows to target new business models by providing open APIs (Application Programming Interfaces) for integrating well-established services like Voice/Video Telephony and emerging technologies like Presence/IM into existing environments and systems. The open APIs and Protocols not only provide typical end-to-end services (like buddy lists, messaging, telephony etc.), but allow for innovative push-services (subscriptions to online information, which is delivered on demand to softphones or mobiles) and find-me/follow-me solutions.

Administrator Web Interface

The Administrator Web Interface provides system administrators and customer support agents with access to the platform's configuration and troubleshooting applications through a standard Web browser.

Billing and Rating Engine

The centralized approach of the NGC Platform utilizing the SIP routing engine as the data source simplifies the accounting process and provides a reliable and consistent way for the creation of call detail records. With the usage of a real-time rating engine, an up-to-date balance can be provided to consumers at any time with both prepaid and postpaid billing models.

Customer Web Interfaces

The appearance of the user interfaces can be customized supporting specific corporate look and feel. Multiple languages are supported through the use of language packs enabling the display of sites in a given language, including resource files, localized images and voice prompts. Packs for English, French, German and Spanish are pre-installed, additional language packs can be added easily.

Provisioning

Strict isolation combined with well defined and standardized interfaces are an important part of every successful IT framework. As systems evolve, they force their surrounding systems to adapt. A mature and deliberate interface implementation keeps these adaptations at a minimum, or eliminates them at all.

Using standardized transport protocols like HTTPS and description languages like SOAP simplifies integration with third party applications and ensures a minimum time to market. For security reasons, all requests to the provisioning system are encrypted and authenticated.

Monitoring

Efficient system monitoring and alerting is critical when providing highly available services to customers. The NGC Platform provides an SNMP interface to poll the status of every single element in the system to monitor the health of the platform and the availability of its services.

Beside polling low-level metrics like CPU utilization, memory- and disk-usage and the status of all applications running on the platform, also high-level black-box tests are performed by routing SIP messages via the different subsystems to detect lock-ups and other service failures. To ensure real-time alerting, SNMP traps can be sent to 3rd party monitoring and alerting systems like Nagios.

TECHNICAL SPECIFICATIONS

Softswitch Hardware

- IBM or Dell System Hardware
- Rack-mount chassis 1U
- 350 Watt PSU

Software Management

- Transparent software versioning
- Seamless and non-interruptive software updates and upgrades

Capacity

- Up to 50.000 subscribers
- 180.000 Busy Hour Call Attempts
- Unlimited number of Class4 peerings
- Unlimited number of SIP peerings

Provisioning Interface

- Accessible via SOAP and XMLRPC
- Encrypted via HTTPS
- Authenticated via HTTP digest methods
- Authorized via fine-grained

Access Control Lists

- Integrated Customer Care and Administrator web interface
- Integrated Customer Self Care web interface
- Extensive set of Batch Provisioning tools

Signaling Protocols

- SIP
- Audio
  - → fully transparent RTP pass-through G.711a/u, G.723, G.726, G.729, iLBC
  - → license-free RTP transcoding between G.711a/u, G.726, iLBC
- Video
  - → fully transparent pass-through H.261, H.264
- Fax
  - → T.38 Fax-over-IP support

Monitoring and Alerting

- Passive monitoring via SNMP requests
- Active alerting via SNMP traps
- Compatible to Nagios and various 3rd Party Vendors
- Continuous health checks on multiple levels
  - → Low level component checks
  - → High level black box checks

System Management

- Accessible via Serial Line, Ethernet or KVM
- Web/Telnet/SSH interface for hardware management
- Web/SSH interface for subscriber-, peering- and billing management
- Backup/restore procedures
- Regulatory Services
- Local Number Portability
- Lawful Intercept
- Emergency Number Routing
- Numbering Plan
- E.164 compliant
- Flexible number format schemes for peerings

Class 5 subscriber features

- Click-to-Dial
- Simultaneous Ring
- Multi-line Appearance
- Selective Outgoing Call Blocking
  - → Call Restrictions – International
  - → Call Restrictions Long Distance
  - → Call Restrictions – Premium
- Selective Incomming Call Blocking
  - → Call Restrictions – Contact List
  - → Selective Call Acceptance
  - → Selective Call Rejection
  - → Anonymous Call Rejection
- Call Forwarding:
  - → Call Forwarding Busy
  - → Call Forwarding No Answer
  - → Call Forwarding After Time
  - → Call Forwarding Unconditional
  - → Call Forwarding Voicemail
  - → Call Waiting / Cancel Call Waiting
  - → Call Hold
- CLIP (Calling Line Identification Presentation)
- CLIR Overwrite
- CLIR (Calling Line Identification Presentation Restriction)
- Caller ID Block per Call
- Caller ID Block per Line
- Direct Inward Dialing (DID)
- Direct Outward Dialing (DOD)
- Extension Dialing
- Network URL Dialing
- 3-way Conferencing

Local Exchange Services

- Authentication
- Real-time accounting
- Prepaid/Postpaid billing support
- Call forwarding (unconditional, busy, no answer, unavailable)
- Call blocking (incoming, outgoing, black- and white-listing)
- Vertical service codes
- Voice-box
- Voice-to-Mail
- Announcements
- Multi-Conferencing
- Voicemail Features
- Password protection
- Default or custom greetings
- Web interface for checking of voicemail
- E-mail notification of voicemail with audio file attachment
- Voicemail forwarding
- Visual message waiting indicator (MWI)
- Message waiting stutter dialtone
- Transit Services
- Static peerings via SIP
- Dynamic peerings via SIP and ENUM
- Least cost routing
- International routing
- Peering fail-over support
- Advanced Communication Services
- Instant Messaging via SIP
- Rich Presence via SIP/SIMPLE
- Location based services (GEOPRIV, AJAX API for Web2.0 integration)
- Billing and Rating Engine
- Prepaid and postpaid billing profiles
- Realtime rating engine
- Fraud protection
- CDR export as CSV, XML
- Invoice generation and PDF export

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