SysMaster™

TYPICAL CABLE AND DSL PROVIDER
VoIP Implementation Infrastructure

Cable/DSL Solution

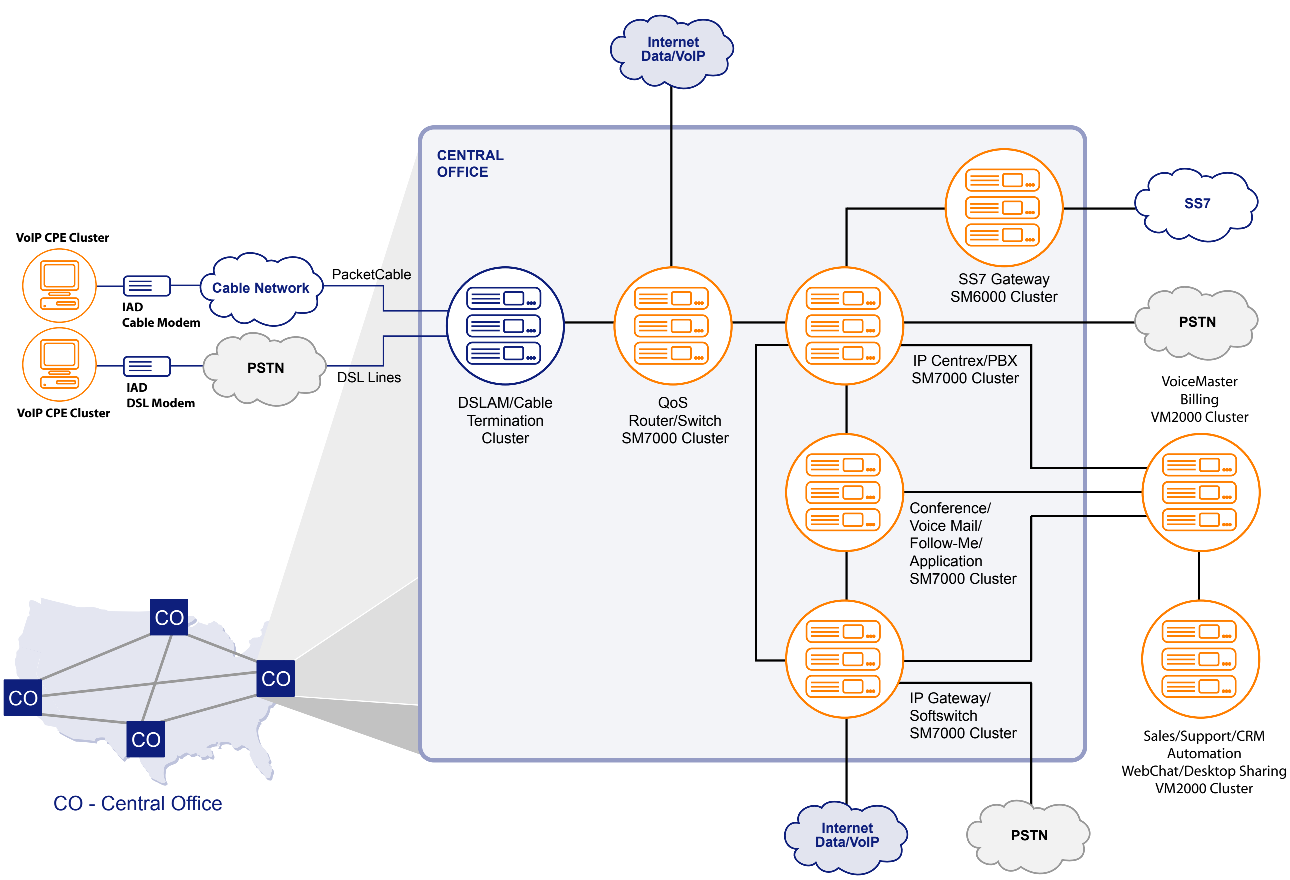
Problem

Existing DSL and Cable Operators want to implement VoIP telephony offering for business and residential customers and utilize the existing infrastructure. The offering should support traditional services over IP such as IP Centrex, Voicemail, Conferencing, Follow-me/Find-me in an outbound and inbound formats (calls must be made from the subscribers to outside lines as well as from outside lines to the network). In addition, the subscribers will use CPE devices (customer premise equipment) that are hardware based (such as CPE adapters, IP Phones, and Cable/DSL modes with integrated IP Phones) as well as software based VoIP clients (such as SoftPhones, and USB Phones). Operators require dynamic billing for pre-paid and post-paid formats to allow easy entry for customers without the need for credit verification and contract signup.

Solution

SysMaster Corporation provides a full scale, end-to-end solution to address all implementation requirements for DSL and Cable Operators. The solution offers advanced IP Centrex/PBX services, Conferencing, Follow-me, and Voicemail. It also provides dynamic softswitch routing from outside lines to network CPE devices and from CPE devices to outside lines. SysMaster supports a variety of CPE adapters and IP Phones, as well as SIP VoIP SoftPhones. SysMaster also offers an advanced real-time billing and call routing engine to ensure proper subscriber call management. The system supports both pre-paid and post-paid billing options with large scale dynamic authentication and traffic management capability.

Infrastructure Layout for DSL and Cable Operators



NOTE: All devices in orange are originally manufactured by SysMaster Corporation.

CPE/IAD Device Cluster

(these are the customer premise equipment devices):

- CPE Adapter - usually this is a hardware adapter that is attached to a physical phone and to the Ethernet network. They are used to convert analog voice signals into VoIP signals. The commonly supported codecs are g711, g729, g723, GSM.
- IP Phone - similar to the CPE adapter, however it has a built-in phone device. The device also converts analog voice into data.
- SoftPhone - this is software that runs on a PC computer and allows the subscribers to make calls utilizing the computer to convert the analog voice signal coming from the computer headset (microphone and speaker) into VoIP data signal. Supported codecs include g711, g729, g723, GSM.
- USB Phone - this is a hardware unit that acts as a computer headset and utilizes the SoftPhone software to convert the voice into data. This unit can not be used without the computer running the SoftPhone client.
- IAD Devices - these are third-party devices that have an integrated DSL/Cable modem with an optionally integrated VoIP CPE adapter.

DSLAM/Cable Termination Equipment Cluster

This is equipment that is already installed and in production to terminate DSL and Cable network connections from the IAD subscriber units. This equipment is provided by third-parties.

QoS Router/Network Switch Cluster

Description

This is made up of one or multiple units that will allow VoIP prioritization over traditional IP traffic. VoIP prioritization is required specifically for uplinks (traffic flow from the IAD to the DSLAM/Cable Multiplexor) due to the limited bandwidth resources. This equipment will also guarantee the Quality of Service for all inbound and outbound VoIP telephony phone calls. The QoS routers are manufactured by SysMaster Corporation and allow up to 2GB pass-thru connection bandwidth per unit. Full network redundancy and multi-gateway switching is supported.

Features

- Service level prioritization.
- Dynamic QoS and TOS of voice traffic - fully redundant.

Proposed Setup

Level1 SM7000 per 1GB network throughput.

VoIP Gateway/IP Centrex Cluster

Description

These units are used to terminate VoIP traffic coming from CPE devices and allow for IP Centrex/PBX services. In addition, these units are located in the CO (central office) and interface with the outside PSTN network via ISDN/PRI/CAS/MFC interfaces. They are responsible not only for routing outbound calls from the CPE equipment to outside lines, but also for the conversion of DNIS/DID numbers into equivalent E.164 terminal numbers (this allows traditional number assignment to VoIP terminals with dynamic registration).

Features

- H323/SIP/MGCP support and protocol conversion
- PSTN support for ISDN/PRI, CAS, and MFC/R2 protocols
- Virtual Number Support and unlimited DNIS/DID translations
- IP Centrex/PBX advanced support including all basic and advanced features.
- IVR over IP support for Advanced Auto Attendant services
- Programmable IVR Script Logic
- Hunt Groups and ACD (automated call distribution) support
- E.164, ENUM, E.911, CALEA support
- DND, Call Transfer, Call-on-Hold, 3-way Conference, Call Park, Call Wait, Caller ID
- Whitelist/Blacklist support
- Directory Service and Extension Dialing with Voicemail or Number failover (Busy/No Answer)
- Last Number Redial
- Wireless Voice over IP
- Dynamic Terminal H323 Gatekeeper/ SIP Registrar Registration and Provisioning
- DOCSIS PacketCable Support
- TFTP APS (Automated Provisioning Service)
- NAT Traversal for SIP terminals

Proposed Setup

Level3 SM7000 for each 4,000 customers assuming 1/10 usage ratio. A single Level 3 SM7000 unit can concurrently support up to 1920 IP calls and 480 ISDN/PRI calls.

VoIP Gateway/Conferencing/ Voicemail/Application Cluster

Description

These units host the VoIP applications that provide Conferencing, Voicemail, Follow-me and other services. These units can operate utilizing VoIP (H323/SIP/MGCP protocols) or can interface with PSTN lines (ISDN/CAS/MFC) protocols. Each unit can support multiple applications and provide advanced voice services.

Features

- 365 Participant Advanced Conferencing
- Unlimited Voicemail Boxes (100GB storage with 1MB/minute allocation)
- Fax to Email
- Voicemail to Email
- Unified Messaging
- SMS Notifications
- Last Voicemail Redial
- Video Conferencing
- Desktop Sharing and Collaboration

Proposed Setup

Level3 SM7000 for each 4,000 customers assuming 1/10 user ratio. A single SM7000 Voicemail server that can accommodate up to 100,000 minutes of recorded voice mail messages.

VoIP Termination/Origination Gateway/SoftSwitch Cluster

Description

These units allow direct PSTN termination into existing DS0 lines or IP-to-IP softswitching for Long Distance and IXC interfacing. The gateways offer protocol conversion (H323/SIP/MGCP) as well as codec transcoding and intelligent proxying (unique application) for RTP management. Full LCR, Route-overflow and Failover are supported with the capability for pass-thru Radius authentication and CDR collection.

Features

- H323/SIP/MGCP support
- ISDN/PRI, CAS, MFC/R2 support
- LCR and Advanced Route Management
- Advanced proxy mode and protocol conversion
- Advanced codec transcoding and intelligent proxy
- Multi-gatekeeper registration support
- IP-to-IP origination and termination services
- IVR over IP for alternative applications
- Origination POP setup and DID filtering
- IVR profiling and ANI/DNIS mapping and conversion

Proposed Setup

Dual Level2 SM7000 for PSTN termination and origination with the capacity of 480 calls each (total 960) and with direct PSTN DS0 termination. Dual Level2 SM7000 for IP termination and origination with the capacity of 960 channels each (total 1920).

VoIP Termination/Origination SS7 Signaling Gateway Cluster

These devices will allow direct access to SS7 networks for origination and termination purposes. SS7 Signal processing will allow a single unit to manage redundant SS7 links in a softswitch format.

Features

- SS7 signal conversion
- Redundant link management and call routing
- LCR and advanced route management
- Ethernet or HDLC implementation
- Advanced management of remote gateways without SS7 capability

Proposed Setup

Dual SM6000 SS7 Signaling Gateway to manage alternative PSTN origination and termination gateways over IP.

VoiceMaster Billing Server Cluster

This device will authenticate, authorize, and bill calls in real-time for pre-paid and post-paid customers. The system has a Radius billing interface, and a real-time routing engine (based on SIP proxy and H323 gatekeeper) that can support implementations with up to 10,000 concurrent calls per unit. CDR collection is also supported for legacy systems. The system will also provide CRM (customer relationship management) interface to allow easy service management for customers via Web.

Features

- Post-Paid and Pre-Paid billing
- Radius Interface
- Authentication, Authorization, Accounting for PSTN Switches and VoIP Gateways
- Real-time call routing for H323 and SIP calls
- LCR, Port-overflow and Endpoint-Failover support
- Time-of-day billing
- Unlimited Billing and Routing Plans support
- Progressive and FlagFall billing
- Advanced customer CRM
- Advanced IP Centrex and Application management via Web
- Custom Service Plan billing
- Specialized Services billing
- Managed Services and Hosting Solution Support

Proposed Setup

One Level5 VoiceMaster Billing server and one Level2 VoiceMaster billing server with real-time data replication for database redundancy and scalability. The system will handle up to 10,000 concurrent calls and over 5 million user accounts. Additional servers are required for traffic over 10,000 concurrent calls. All calls are controlled dynamically in real-time.

VoiceMaster Automated Sales and Support Cluster

This server hosts the Automated Sales and Support application, to allow efficient CRM (customer relationship management) and ticket provisioning. The system accepts emails, voice, and web requests and aggregates them into one CRM data stream that is distributed to service agents based on rotation, skill, or service role. The CRM system also supports advanced sales lead management and lead provisioning, customer contact management, and quote management.

Features

- Web Chat functionality with dynamic guest interface
- Email Ticket Management
- Desktop Sharing and Collaboration
- PBX IVR scripting and advanced IVR interface
- Lead management and distribution
- Customer Management
- Quote Management
- Inventory and Unit Provisioning and Management

Proposed Setup

Dual Level1 VoiceMaster Sales and Support Automation servers with capacity for up to 200 events per second processing.

Contact Info

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