SysMaster™

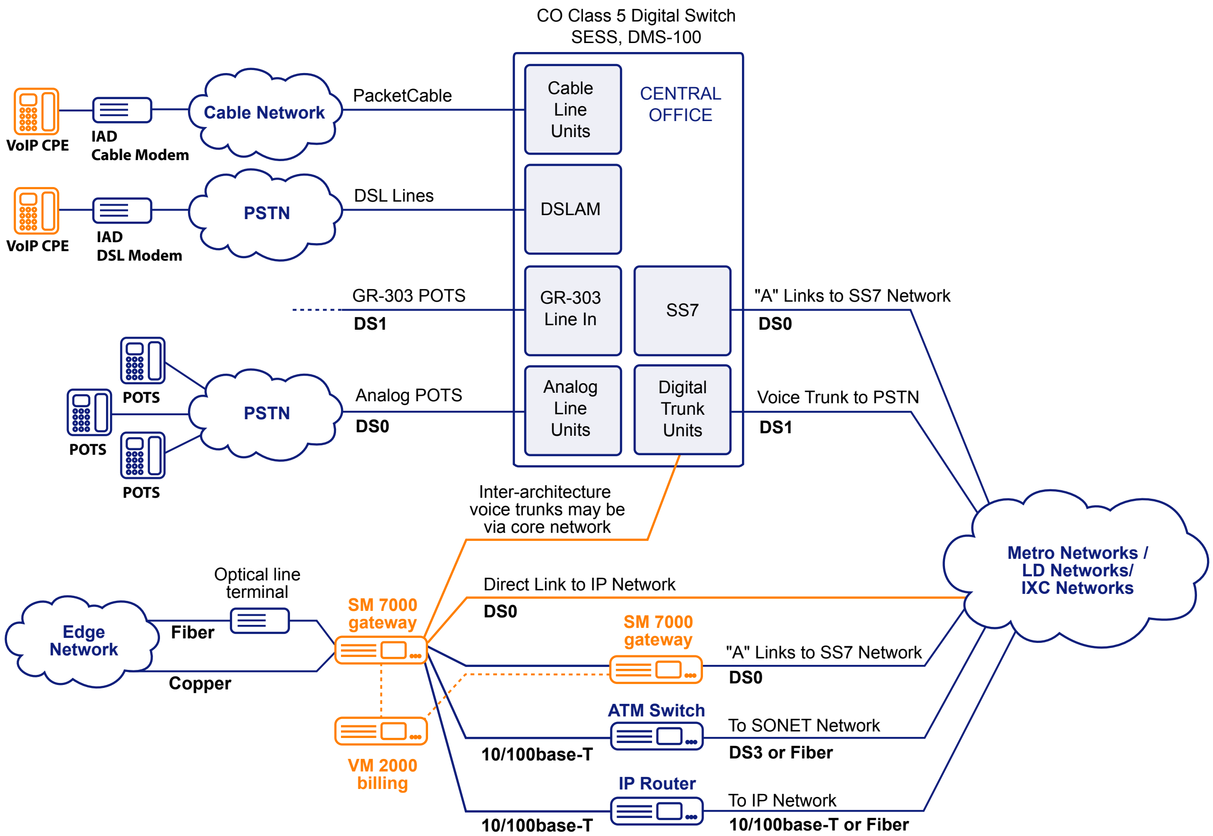
SysMaster SM 7000
Typical Implementation Scenarios

VoIP Solutions

CARRIER SERVICES WITH REAL-TIME INTEGRATED BILLING API

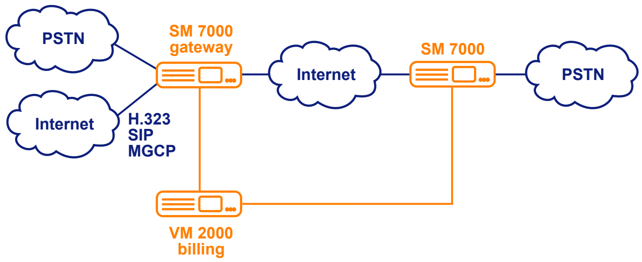
CLEC, RLEC, ILEC Integration Options

SM7000 can be fully integrated with any IP or PSTN/ISDN/SS7 infrastructure. The equipment will typically reside within the Central Office (CO) and will be directly connected to the Local Exchange Carrier (LEC) Switch or it will be located externally to the CO and connect to the LEC Switch via a Local Loop connection (utilizing ISDN/CAS/SS7). In both cases, SM7000 can make intelligent routing decisions and route based on its routing settings to termination devices using SIP, H323, MGCP or PSTN/SS7/ISDN/CAS/R2 connections.



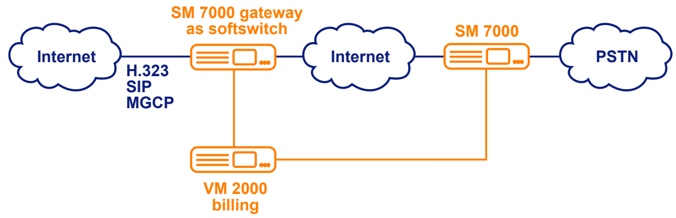
Pre-Paid/Post-Paid Calling Card Application

SM 7000 will accept a PSTN or IP call and authenticate the caller based on entered PIN number or Caller ID, and then ask for destination number. Once the destination number is received the Gateway will route the call via PSTN/SIP/H.323/MGCP to the termination provider. All calls are routed dynamically and controlled dynamically via the credit-time parameter. Multi-session options are fully supported. In addition, IVR and DTMF over IP are also supported.



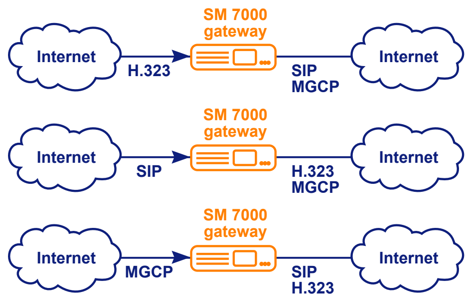
Pre-Paid/Post-Paid Wholesale Call Routing

SM 7000 will accept the first call leg via IP or PSTN and then authenticate the caller based on credit balance and authorize the call time. Once the call is authenticated and authorized, it is routed to the termination provider (call leg 2). Once the call time is exhausted the call is dynamically terminated.



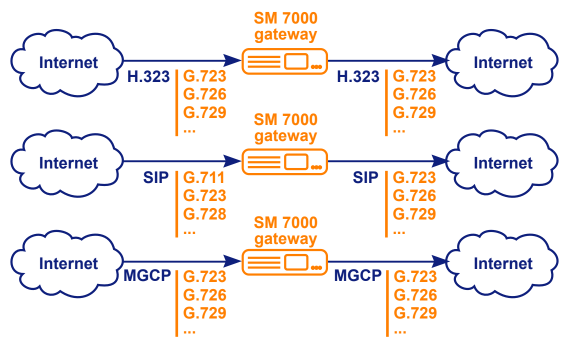
SIP to H.323/H.323 to SIP Conversion and SoftSwitch Routing

SM 7000 will accept H.323 or SIP call play IVR over IP (if required), recognize DTMF tones (if required) and convert the call signal into an alternative SIP of H.323 signal. The system can also terminate H.323 or SIP call into the PSTN/ISDN line or originate a PSTN/ISDN call and convert it to SIP or H.323 for termination purposes. Both call legs are controlled independently to allow flexibility in route failover and port overflow scenarios. The system can also convert from and to MGCP protocol. In this scenario SM 7000 will play the role of a traditional SoftSwitch and Proxy Server.



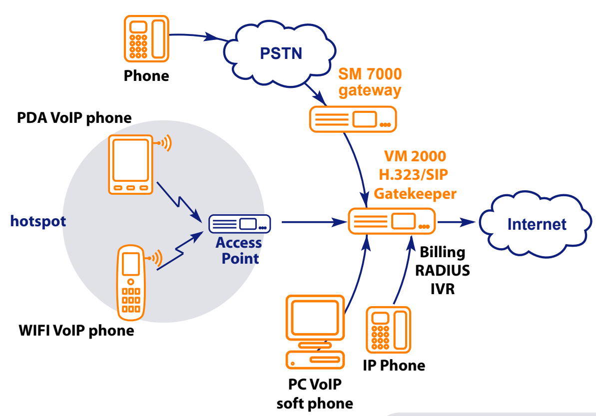
Codec Conversion and Transcoding

SM 7000 plays the role of a traditional SoftSwitch and Proxy Server. The system fully support Codec Transcoding - asymmetrical conversion of codecs to allow connectivity between endpoints with incompatible codecs. The system also supports Intelligent Codec Proxy - to negotiate common codec between the endpoints that it will connect so that no RTP (Voice Data) codec conversion is required to optimize voice quality and throughput.



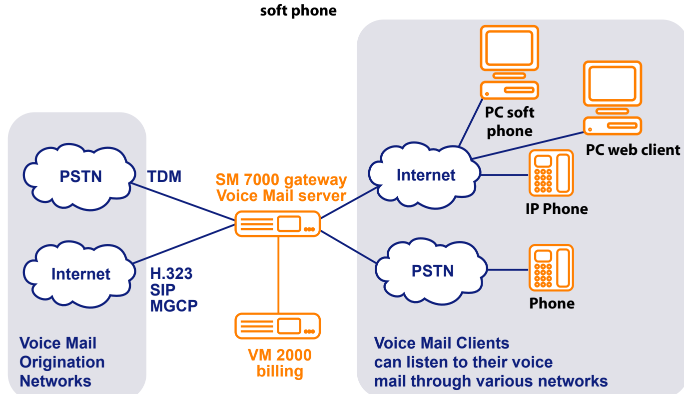
CPE (Customer Premise Equipment), IP Phone, Soft Phone and WiFi Phone Management

SM 7000 allows dynamic registration of SIP devices into its SIP Registrar/Proxy Server with full NAT Traversal functionality. It also supports registration of H.323 devices to its H.323 Gatekeeper. Both the SIP Registrar and the H.323 Gatekeeper provide support for dynamic terminal call processing even behind NAT and Firewalls. SM 7000 can route inbound PSTN calls to dynamically registered CPE devices by utilizing advanced call filtering and DNIS management.



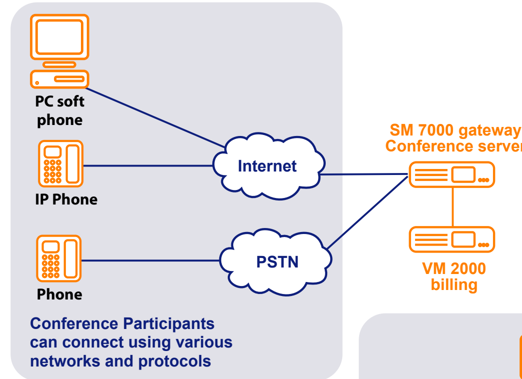
SIP, H.323, MGCP, and PSTN IP Voice Mail Services

SM7000 offers advances Voice Mail services over IP. The system supports basic Voice mail services as well as advanced services such as Unified Message interface, automated voicemail e-mail distribution and notification, one-click callback, subscriber paging, voicemail forwarding and broadcasting, and advanced voicemail surveying. The system also supports standard message notification for POTs and VoIP devices for new voicemail. Voicemail can be managed via IVR and Web interfaces.



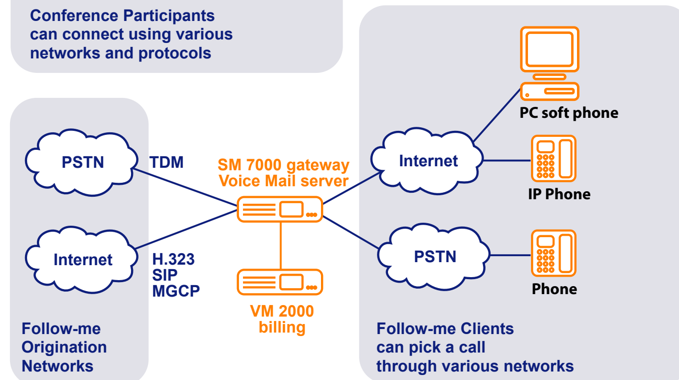
SIP, H323, MGCP, and PSTN IP Conference Services

SM 7000 allows advanced conference service support for up to 460 concurrent conference callers. The system supports private (PIN authenticated) and public conferences, scheduled conferences, lecture conferences, private bridges (virtual conferences within conferences), IVR and Web conference administration.



SIP, H323, MGCP, and PSTN IP Follow-Me Services

SM 7000 offers advanced follow me services for customers that require global roaming and number forward services. The follow-me services supports number hunting and location for improved roaming performance. The follow-me service can be managed via IVR or Web interfaces.



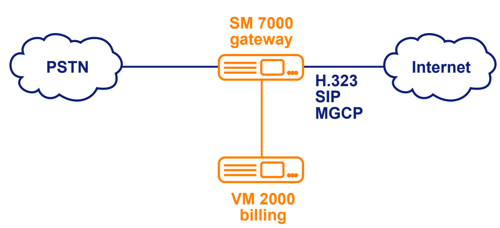
SIP, H323, MGCP, and PSTN Device Support

SM 7000 can be used to dynamically control CPE and POTS devices and route inbound and outbound call to such devices. The system offers advanced NAT Traversal functionality for SIP devices. Registered devices can utilize any of the licensed services including IP Centrex, Voice Mail, Conference Calling, and Follow-Me services.

CARRIER VOICEMASTER BILLING SERVER FOR SIP AND H323 CALL ROUTING

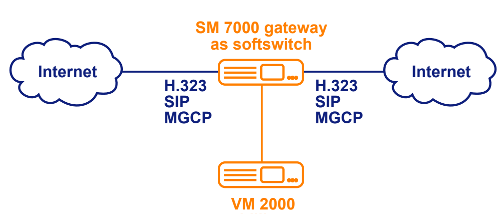
Pre-Paid/Post-Paid Calling Card Application

VM 2000 will communicate with SM 7000 or third-party gateways (including Cisco, Quintum) via RADIUS protocol to collect authentication (caller ID or PIN) and authorization information (destination number) and then route the call. VoiceMaster will then control the call dynamically via its gatekeeper/Proxy functionality and disconnect the call once the authorized time is reached or upon certain event. Speed dial and multi-sessions are fully supported. The system will also do call signal routing (via H.323/SIP protocol) or Trunk routing (via PSTN switching). Complete ASP Hosted Billing model (managed services) is supported.



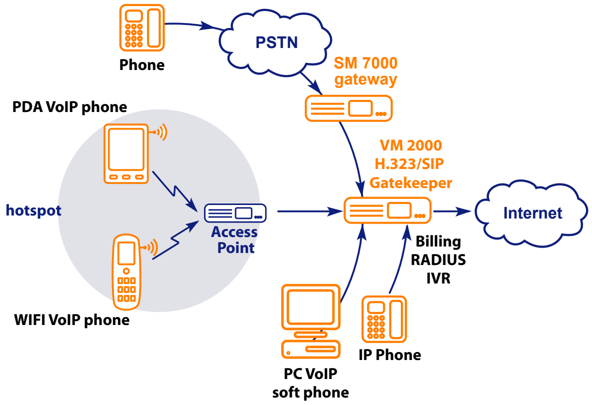
Pre-Paid/Post-Paid Wholesale Call Routing

VM 2000 supports complete softswitching functionality without codec conversion. It can accept any H.323 or SIP call and route it to the proper destination in a routed (RAS and Call Signal, without RTP) or proxy (with RTP) mode. The system can handle thousands of concurrent calls and fully support intelligent Least Cost Routing (LCR), Port-Overflow, Round-Robin routing and Route Failover. The system will authenticate and authorize every call and dynamically reduce balances for concurrent wholesale clients to provide real-time billing and routing functionality and disconnect calls dynamically. Complete ASP Hosted Billing and Routing model (managed services) is supported.



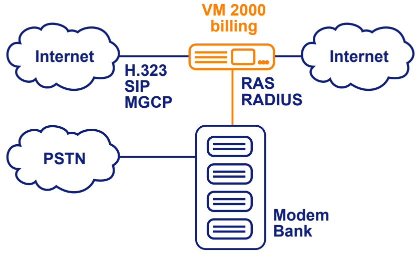
SIP and H323 CPE, IP Phone, SoftPhone, WiFi Phone Device Support

VM2000 will allow unlimited number of registered terminal SIP or H323 endpoints to be registered with its H323 Gatekeeper or SIP Registrar thus supporting dynamic terminal routing. The system will support complete NAT Traversal functionality for compatible SIP devices as well as H323 SoftPhones. The system can route inbound calls to and from the registered endpoints and provide failover services if required. VoiceMaster supports both pre-paid and post-paid consumer billing models as well as event-driven (intercepted calls and IP connections) and time-driven (flat monthly/weekly plans) based billing.



ISP Internet Based Billing

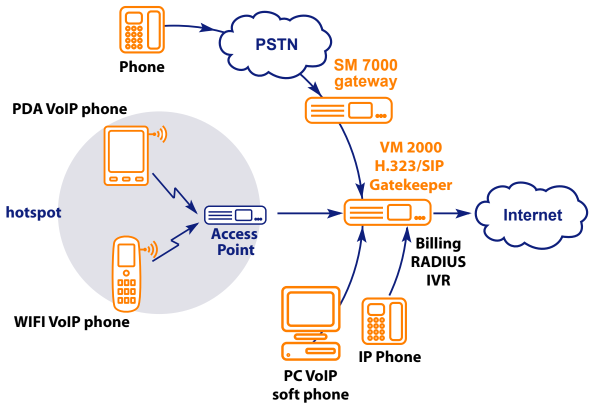
VM2000 offers advanced access billing based on bandwidth utilization, connection time, or time-periods. The system can act as an access controller with dynamic firewall support to provide authentication and authorization services in a post-paid and pre-paid billing formats. This is extremely useful for hot-spot based billing.



IP CENTREX AND PBX SERVICES WITH REAL-TIME INTEGRATED BILLING API

SIP, H323, MGCP, and PSTN IP Centrex Services

SM7000 supports various VoIP protocols to allow IP Centrex functionality. Full IVR PBX processing is supported that includes the basic IP Centrex features like call wait, caller id, call on hold, park call, private conference, 3-way calling, call transfer, music on hold, and others, as well as advanced services such as ACD (automated call distribution) management, agent management, follow-me services, conference services, call recording, and others. The PSTN POTs can be connected via traditional analog lines via channel banks. VoIP CPE devices are dynamically registered into the system to support complete PBX functionality and function as active PBX extensions. IP PBX can be managed via IVR and Web interfaces.



SysMaster
2700 Ygnacio Valley Rd, Suite 210
Walnut Creek, CA 94598
United States of America

Email: sales@sysmaster.com
Web site: www.sysmaster.com