TELECOM VoIP CONVERGENCE AND VoIP TELEPHONY
VoIP TELECOM STARTING PACKAGE

SOLUTION HIGHLIGHTS

- Firewall and NAT Friendly SIP Implementation
- VoIP Centrex/PBX Support
- Implementation Redundancy and Scalability
- 100% Compatibility with Tandem Telco Switching Infrastructures
- Complete H323/SIP/PSTN (ISDN/CAS/R2/SS7) Interoperability
- Complete Billing Integration
- Pre-Paid and Post-Paid Account Support
- Centralized VoiceMail Support
- Virtual PBX and Conference Services Support
- SIP/H323/PSTN Gateway
- SIP Registrar/Proxy and H323 Gatekeeper Routing Server
- Stand-Alone SQL Database

Problem

Existing Cable ISP provider wants to offer VoIP services to it subscribers to achieve the following business benefits:

- Reach subscribers that are outside the regional/geographical market.
- Provide low cost international calling plans.
- Increase customer loyalty.
- Provide local termination services for international carriers.

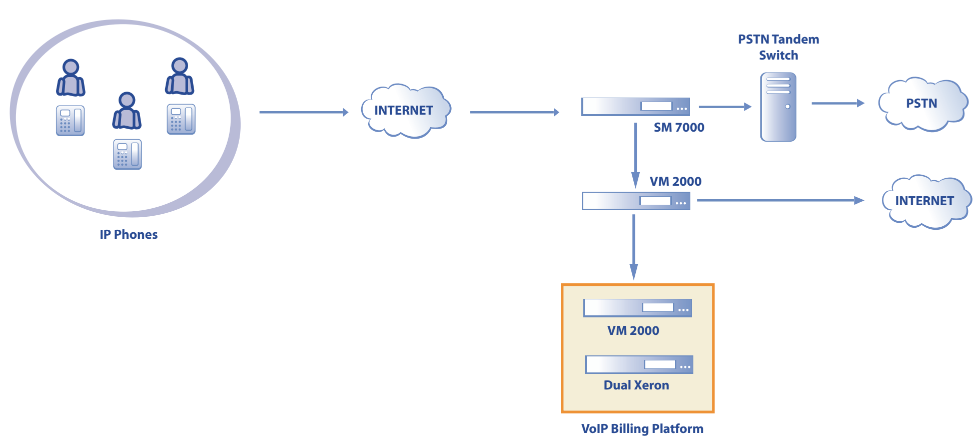
VoIP provides a lucrative business model that can be scaled up quickly in terms of service offerings and area coverage. The ISPs need to provision the complete solution form a single vendor to ensure simplified timely deployment.

Solution

SysMaster offers a Telco Starting Package to allow initial project launch in a short-time cycle. The package includes all necessary devices to provision for up to 5,000,000 VoIP customers with 120-1,200 concurrent calls. The solution consists from the following devices: VoIP Billing Platform, VoIP Gatekeeper/Registrar and Radius Server, VoIP H323/SIP/PSTN Gateway.

Solution Infrastructure

Telecom VoIP Convergence Infratsructure Setup



TELECOM VoIP CONVERGENCE

SOLUTION FEATURES

Firewall and NAT Friendly SIP Implementation

Telecom VoIP Convergence offers firewall and NAT friendly SIP implementation allowing seamless integration with existing Network Address Translation (NAT) devices and firewalls. All devices communicate through outbound connections, using protocols and ports that can transparently transit most home and business firewalls. The proxy-based solution architecture also ensures seamless work with NAT devices.

Pre-Paid and Post-Paid Account Support

SysMaster's solution supports both pre-paid and post-paid telephony subscribers. That functionality allows service providers and management companies to target various demographic groups and create flexible billing plans, tailored to meet individual customer needs. The system supports a rich set of rate plans for flexible calling plan offerings. It also features a Managed Service functionality to allow wholesale and reseller outsourcing and business partnerships as a cost effective way to increase revenues.

Centralized VoiceMail Support

SysMaster's Telecom VoIP Convergence solution is capable of storing over 5,000 voicemail accounts. It supports unified messaging for voicemail distribution via email and web with flexible configuration options to allow voicemail fine tuning. The server also allows Voicemail distribution, forwarding, and management via PSTN or Internet lines.

Complete H323/SIP/PSTN (ISDN/CAS/R2/SS7) Interoperability

SysMaster's Telecom VoIP Convergence solution supports H323, SIP, and TDM protocols. In addition the system supports IVR over IP for flexible user authentication and system interaction. Utilizing VoIP and PSTN based access methods the platform significantly increases the system performance and reduces the cost associated with local and long-distance call management.

Complete Billing Integration

SysMaster's solution allows flexible integration with VoiceMaster Billing Server or third-party billing solutions. The system communicates via open standard Radius protocols to allow flexible interoperability and data exchange. Additional functionality includes Call Detail Records (CDR) collection for batch processing and legacy billing.

100% Compatibility with Tandem Telco Switching Infrastructures

SysMaster Telecom VoIP Convergence architecture is fully compatable with existent tandem telco switching infrastructures. The solution allows you to combine traditional tandem switching features with IP-enabled services. Thus, you can take advantage of reduced IP network costs and at the same time provide unmatched connectivity to circuit switched and packet telephony networks.

Virtual PBX and Conference Services Support

SysMaster's PBX solution supports traditional call-on-hold, call-transfer, call-wait, caller id/name, caller-id for call-wait, call-park, private conference, call-pickup, and music on hold, as well as many advanced features such as follow-me service, session recording, call-screening and others.
The advanced Conference Server is capable of supporting over 380 simultaneous callers. The server offers a number of standard and advanced features, including scheduled conferences, conferences on demand, call-screening and pin authentication methods, conference recording and conference bridging (conference within the conference), as well as private and public conferences. It can be accessed via PSTN or Internet to allow flexible and low-cost conference attendance.

SIP Registrar/Proxy and H323 Gatekeeper Routing Server

SysMaster Telecom VoIP Convergence solutions comes with a fully compliant H.323 stand-alone gatekeeper.
SysMaster Gatekeeper supports proxy mode to allow complete call control. All call signals including H.225, H.245, and RTP (voice data) flow through the gatekeeper. This is the most bandwidth consuming mode for the gatekeeper, but it guarantees full network isolation and separation of the origination and termination provider networks. In this mode the approximate gatekeeper throughput is 600 concurrent calls.
SysMaster Gatekeeper also supports routed gatekeeper mode to allow wholesale Telco providers to take full advantage of their termination and origination routes and dynamically control calls. In addition to processing RAS (H.225) signals, the gatekeeper also handles H.245 signals (Q.931 and Call Setup) to allow dynamic call control. All calls are preauthorized to call credit time and can be disconnected by the gatekeeper at any time. This mode of operation is the most suitable for wholesale providers because RTP traffic (voice data) still flows between the origination and termination end points directly and not through the gatekeeper, thus saving bandwidth cost. The routed mode of operation guarantees complete call control. The gatekeeper throughput in this mode is approximately 900 concurrent calls.

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