Key Features

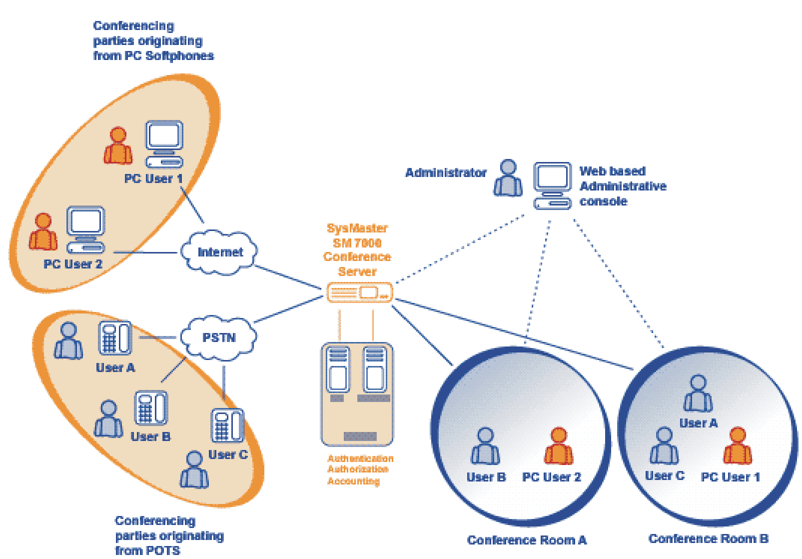
- Concurrent VoIP and PSTN Conference Call Support
- Up to 360 Concurrent Callers
- One or Two-Stage Conference Dialing Options
- Custom Conference Announcements
- Conference and IVR over IP
- H323, SIP, and PSTN Protocol Support
- Conference PIN Authentication
- Advanced Conference Administration
- Conference Recording and Caller Management
- Inbound and Outbound Conference Calling
- Radius Billing Interface
- Managed Services and Virtual Platform Partitioning Support
- Private Subscriber Bridging (Virtual Conference)
- Conference Call Out
- Time Scheduled Conferences
- VoIP - H323/SIP/MGCP Support
- PSTN - ISDN/SS7/CAS/R2 Support
- Inbound Call Filtering
- Least Cost Routing
- Protocol Switching
- Codec Transcoding
- IVR Feature Management
- Dynamic Call Control and Disconnect
- Private Conferences
- Public Conferences
- Admin PIN Authentication
- Auto Conference Recording
- Music On Hold
- Subscriber PIN Authentication
- Subscriber Role Support (Admin/User/Listen-Only User)
- Music On Hold
- Subscriber PIN Authentication
- Subscriber Role Support (Admin/User/Listen-Only User)
- Advanced Rule Management
- Admin Triggered Conferences
- Subscriber Screening
- Unlimited Language Support
- Radius Authentication Interface
- Admin Web Management Interface - Enterprise
- User Web Management Interface - Enterprise
- User CRM Interface - Consumer/Hosted Services
- User CRM Management Interface - Consumer/Hosted Services
- Up to 360 Participants
- Call Management (Mute, Disconnect)
- Lecture-Mode Support

The Problem

ISP provider needs to expand the types of services to allow corporate clients to conduct local and long-distance conference calls. The required equipment must support callers that connect using traditional phone lines, as well as callers utilizing IP Phones and Soft Phones.

The Solution

SysMaster VoIP Conference server allows advanced conference server setup where callers coming via PSTN and IP are connected into secure conference rooms. The application support advanced administrative options to allow conference room managers to records sessions, control room access, and manage the status of the conference room.



CONFERENCE SERVER

SOLUTION FEATURES

H.323, SIP and PSTN Support

SM7000 Conference Server supports H323, SIP, and TDM protocols to allow all types of callers to connect to the conference server. In addition it 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.

Conference PIN Authentication

SM7000 supports PIN based authentication for all callers as well as the conference room administrator. The callers may be asked based on the conference profile settings, to provide PIN in order to connect to the desired virtual conference room.

Flexible Conference Profile Administration

SM7000 allows platform managers to setup unlimited number of conference profiles to allow flexibility and high system throughput. In addition, the profiles allow Managed Services support for virtual platform partitioning.

RADIUS Billing Interface

SM7000 supports Radius Billing Interface to SysMaster VM2000 Billing Platform. This allows the conference server to support real time billing procedures where as all outbound and inbound conference calls are accounted and billed for. All call billing and routing is done in real-time.

Custom Announcement Procedure

SM7000 supports flexible Language Server setup to allow easy prompt and IVR management in multiple languages. Each customer can define a custom prompt on all conference call levels.

One and Two-Stage Dialing

SM7000 supports one and two-stage conference profiles. The one-stage profile allows the callers to enter a conference room number associated with a dedicated DID number. The one-stage conferencing reduces the number of steps to enter a virtual conference room number. The two-stage profile allows the callers to enter a virtual conference room number once they connect to the system via a universal DID number. This procedure allows the platform manager to support a single DID access number and allow the callers to dial a virtual conference room number once they are connected.

Conference Call Flow

One-Stage Voice Mail Calling

1. Caller dial an 800 number (DID) or connects to SM7000 via VoIP
2. SM 7000 accepts the call
3. SM7000 plays Welcome Message
4. SM7000 asks for PIN Number
5. Caller Enters PIN Number
6. Caller is entered into a virtual conference room associated with the DID
7. All callers are now in conference
8. SM7000 sends Radius signals to VM2000 for Billing purposes
9. Conference administrators manage callers and record sessions

Two-Stage Voice Mail Calling

1. Caller dial an 800 number (DID) or connects to SM7000 via VoIP
2. SM7000 accepts the call
3. SM7000 plays Welcome Message
4. SM7000 asks for a Virtual Conference Room Number
5. SM7000 asks for PIN Number
6. Caller Enters PIN Number
7. Caller is entered into a virtual conference room
8. All callers are now in conference
9. SM7000 sends Radius signals to VM2000 for Billing purposes
10. Conference administrators manage callers and record sessions

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