Lotus Sametime Unified Telephony

Telephony Integration

An IBM Proof of Technology

Agenda

- Telephony Integration Strategy : Two different approaches
  - Telephony Vendor plug-ins
  - Sametime Unified Telephony
- IBM Lotus Sametime Unified Telephony
  - Features and Functions
- Architectural Overview
  - Components
  - Call Flows

Objectives

- Explain the new component named Lotus Sametime Unified Telephony
- Understand the positioning and the capabilities of that component
- Visualize the new user experience in term of communications.
- Describe possible architecture pattern of Lotus Sametime Telephony
- ...

Terminology

- MCS – Media Convergence Server
- TCSPI – Telephony Conference Service Provider Interface
- SIP – Session Initialization Protocol
- TCS – Telephony Control Server
- TAS – Telephony Application Server

Positioning with IBM Business Partners

In single-vendor PBX environments, direct connection by means of APIs or SPIs to Sametime 8.0 client is the preferred model

- Business Partners should continue with their integration roadmaps
- New telephony presence options become available with Sametime 8.0
- Where there is no partner integration effort, Sametime Unified Telephony can be used to integrate with a single-vendor PBX environment

In heterogeneous PBX environments, Sametime Unified Telephony is the preferred choice from IBM

- Other heterogeneous approaches can use the same APIs as Sametime Unified Telephony if desired
- Aggregated presence can be done using Sametime Toolkits

ST native, ST + SUT or ST + Plug-in for telephony

Telephony features	Sametime	ST + SUT	ST + Plugin
Click-to-call using VOIP network	X	X	X
Click-to-conference	X	X	X
Connectivity to IP based telephony systems	X	X	X
Seamless integration with ST Connect Client	X	X	
Seamless integration with ST Web Client (v8.5)	X	X	
Integration with Notes or MS products	X	X	
Availability/telephony presence		X	X
Call management application		X	X
Embedded Softphone integrated into ST Connect client		X	
Context-based incoming call management (Status, location)		X	
Connectivity to multi-vendor telephony systems		X	
Connectivity to non-IP (TDM) telephony systems		X	
Incoming Call Notification and Control on the PC		X	
Device Management (e.g., move calls from one device to another)		X	
Advanced audio conferencing control:		X	X
2-way to n-way calls via the PC UI		X	
drag-and-drop additional users		X	
view invited user status (accepted, rejected, on mute, active speaker)		X	X
PC-based moderator controls within the Sametime UI		X	
Separate Softphone			X

Why Lotus Sametime Unified Telephony

- IBM's unique approach to Unified Communications and Collaboration (UC^2)
  - Start from Presence
  - Find – Reach - Collaborate
  - From a Unified Experience
- Leverage and simplify your collaboration and communication infrastructures
  - A middleware approach to Unified Communications
  - No rip-and-replace ... not waiting for upgrades to the entire infrastructure to get benefits
- Making Users and organizations more productive/responsive
  - Unified Experience for users
  - More effective communication and collaboration

Lotus Sametime Unified Telephony User Experience

- Social experience; not simply recreating a keypad-centric experience on the PC
  - Intimacy of participant pictures
  - Drag and drop
- Simple to use
  - Immediacy of call and call functions: all telephony functions a click away
  - Manage your phone(s), phone contacts and how you can expect to be contacted
  - Intuitive: minimize complexity of telephony infrastructure
- Leverage what Lotus Sametime users do naturally every day
  - Use presence to find people
  - Use status and location to enrich presence
  - Collaborate!

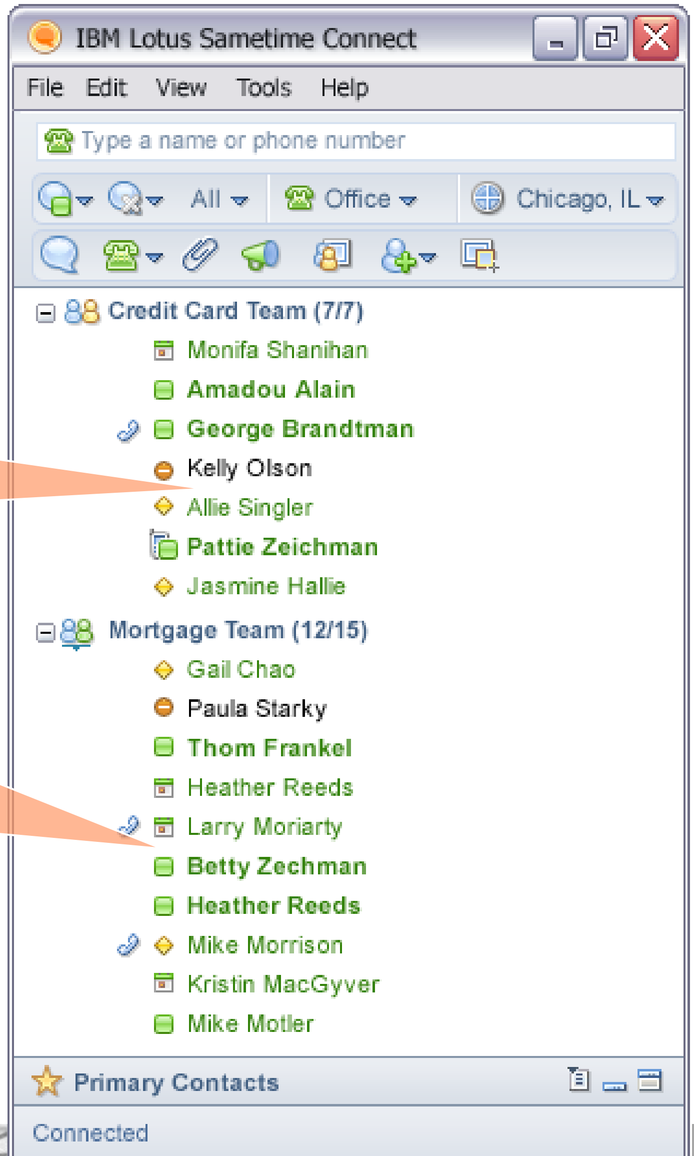
Major Features of Lotus Sametime Unified Telephony

- Telephony presence
- Click-to-call and click-to-conference
- Embedded Softphone
- Incoming call management
- Integrates with YOUR telephony environment

Telephony Presence

Users can see Sametime IM availability status

Users can see if contacts are 'On the Phone'

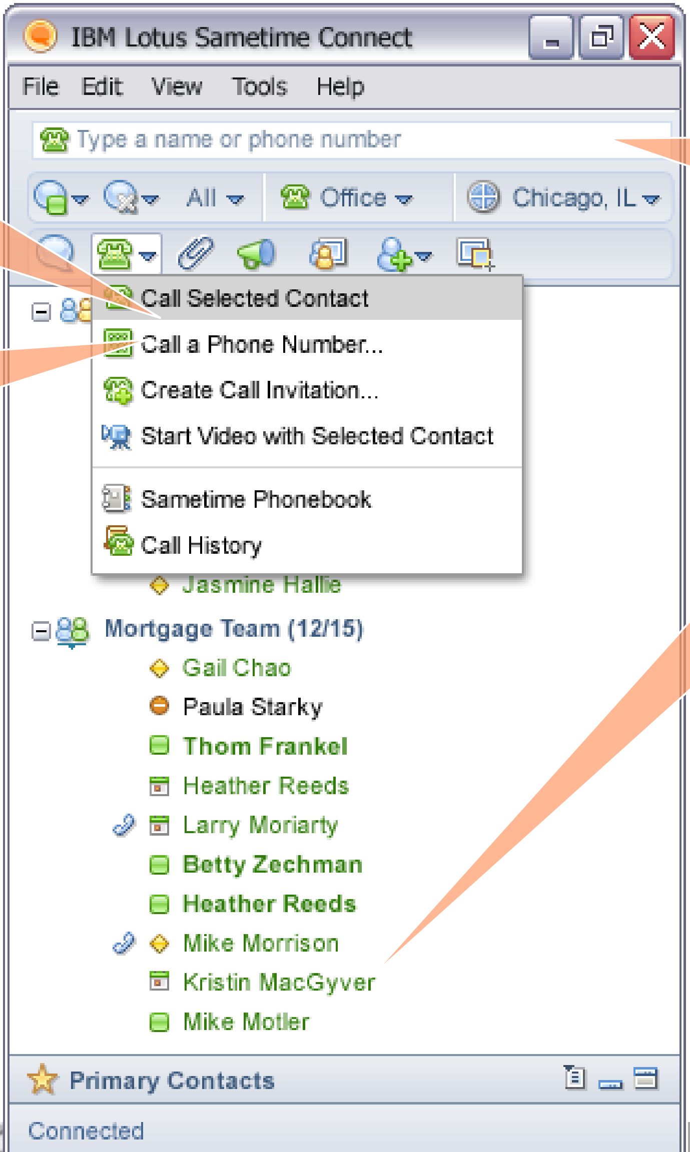


All UI depictions are not final and are subject to change

Click-to-call someone in your contact list

Select a name in the contact list and select 'Call Selected Contact' from the call options

Click 'call a Phone Number' to access the dialpad

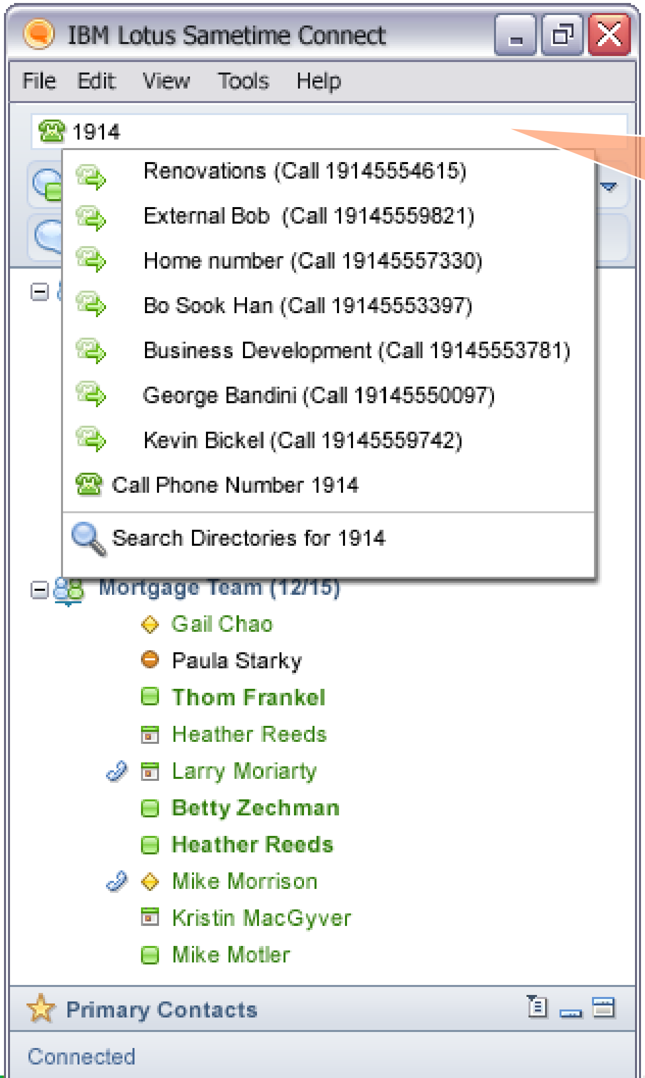


Enter a name or number in the QuickFind

Right-click on a name in the contact list and select 'Call' from the menu.

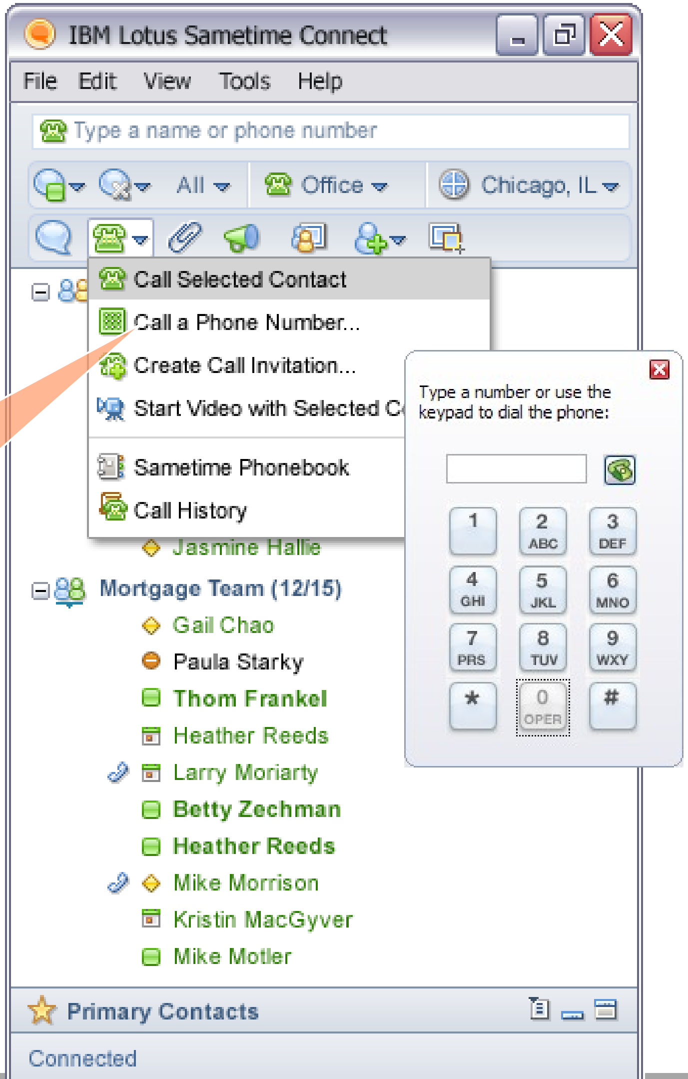
Start an IM session with a contact and click on the 'Call' icon from within the chat window

Making an external call



Start typing a name or number – select from results or 'Call Phone Number'

Select 'Call a Phone Number...' and type a number into the phone keypad



Embedded softphone

- Participant List
  - speaker notification
  - connection status
  - business card
  - context menu
- Participant Call Controls
  - mute/ unmute
  - adjust speaker and microphone volume
  - hold/ resume
  - disconnect
  - rejoin
  - call transfer
  - call forward
    - to another person / device/ number
  - call merge (consultation hold)
  - invite others
  - call-in number
- NOTE: Connects directly to SUT SIP proxy

In a two-way or multi-way call…with any device

Participant List

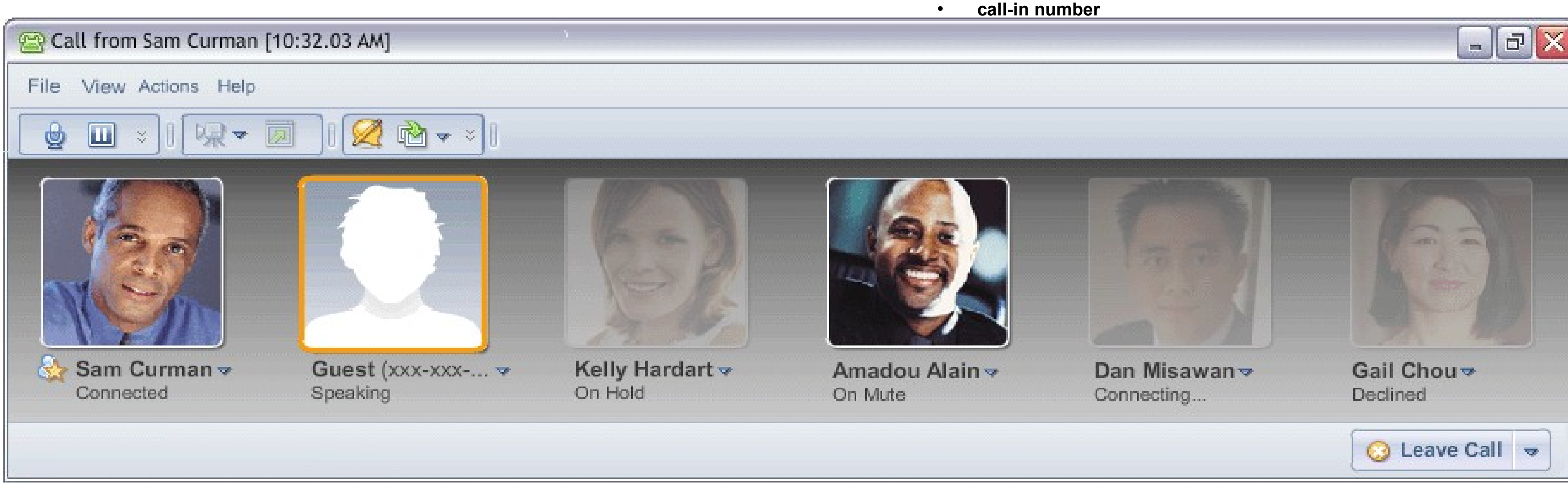
- speaker notification
- connection status
- business card
- context menu

Participant Call Controls

- **mute/unmute**
- **hold/resume**
- **disconnect**
- **rejoin**
- **call transfer**
- **call forward to another**
  - **person**
  - **device**
  - **number**
- **call merge**
  - **(consultation hold)**
- **invite others**
- **call-in number**

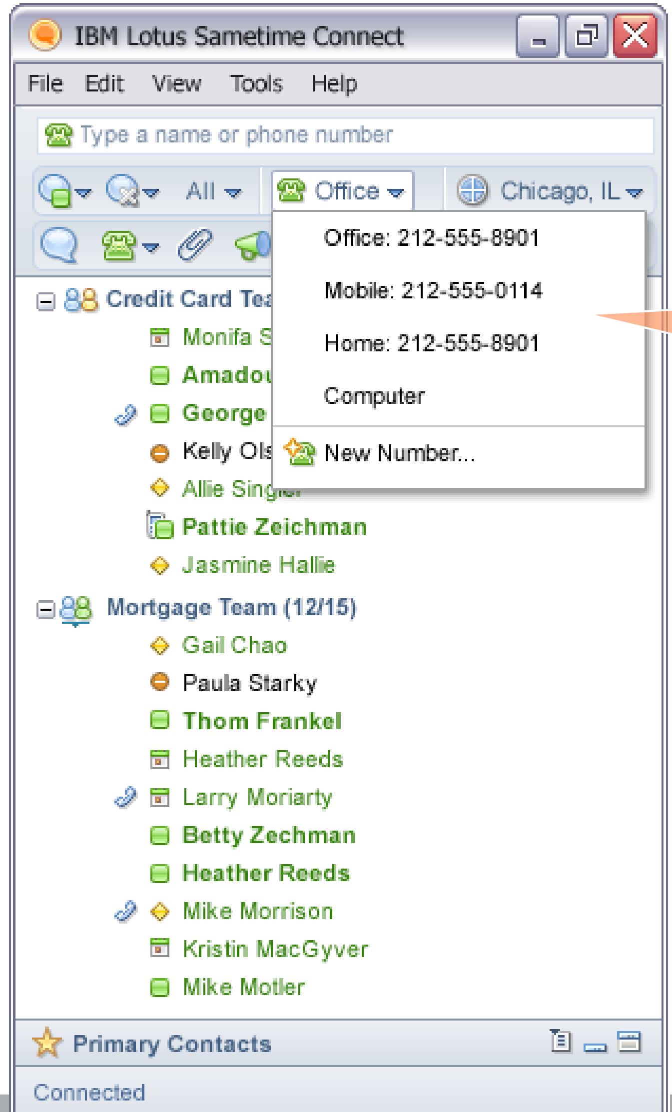
Moderator Call Controls

- mute one or all participants (with/ w/o lock)
- drop participant from call
- lock call
- end call for everyone
- adjust microphone volume for any participant



All UI depictions are not final and are subject to change

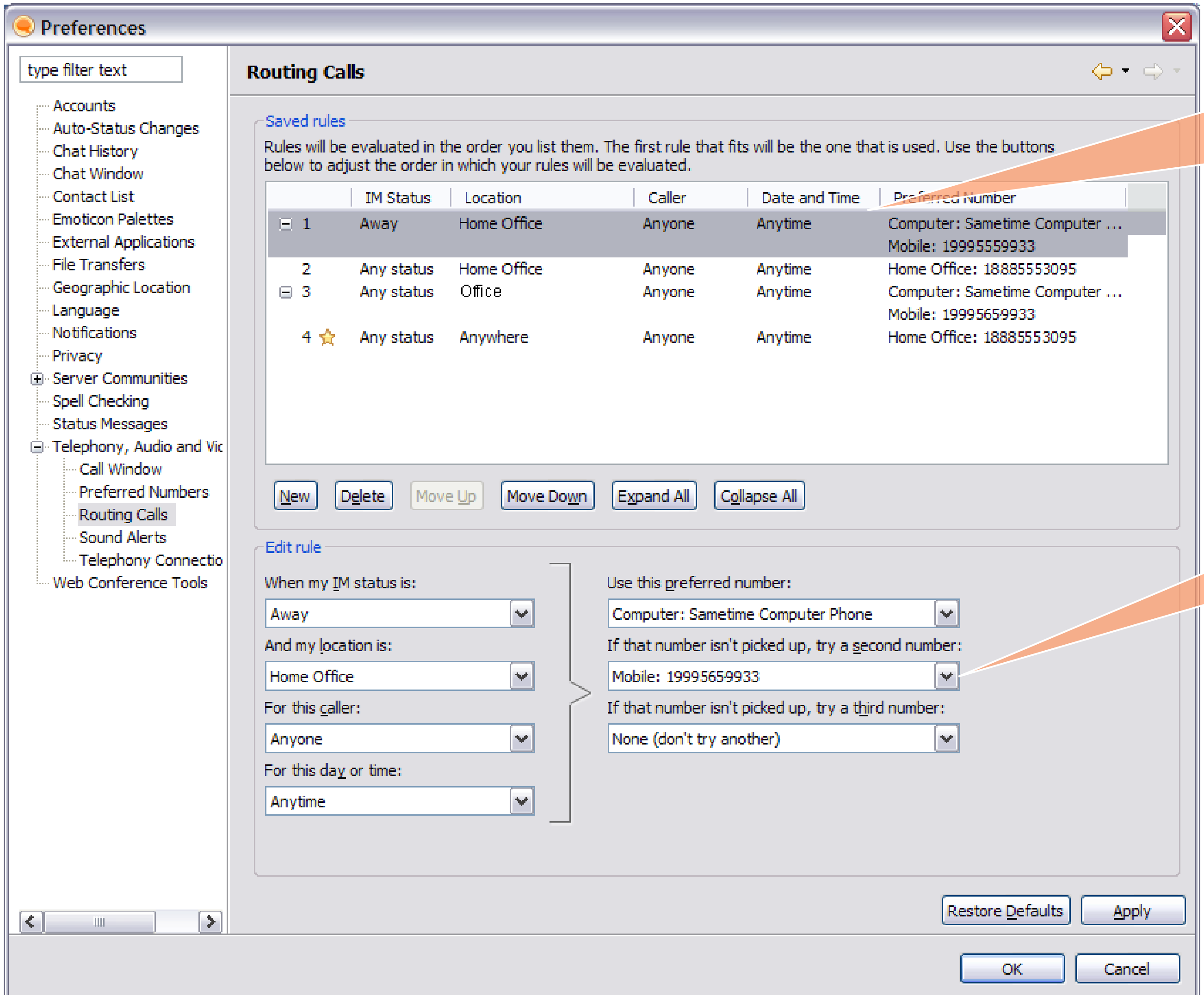
Setting your preferred device



Which phone do I want to use to make and receive calls?

All UI depictions are not final and are subject to change

Incoming call management rules



Rules can be set according to IBM Lotus Sametime IM status, dynamic location, and caller, date and time.

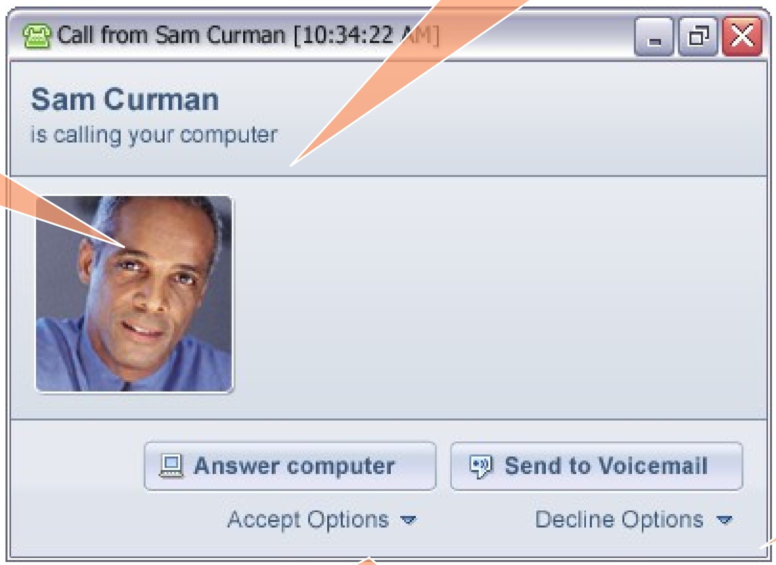
Calls can target a sequence of numbers.

All UI depictions are not final and are subject to change

Incoming call notification

Targeted device/number.

Incoming call notification shows who is calling.

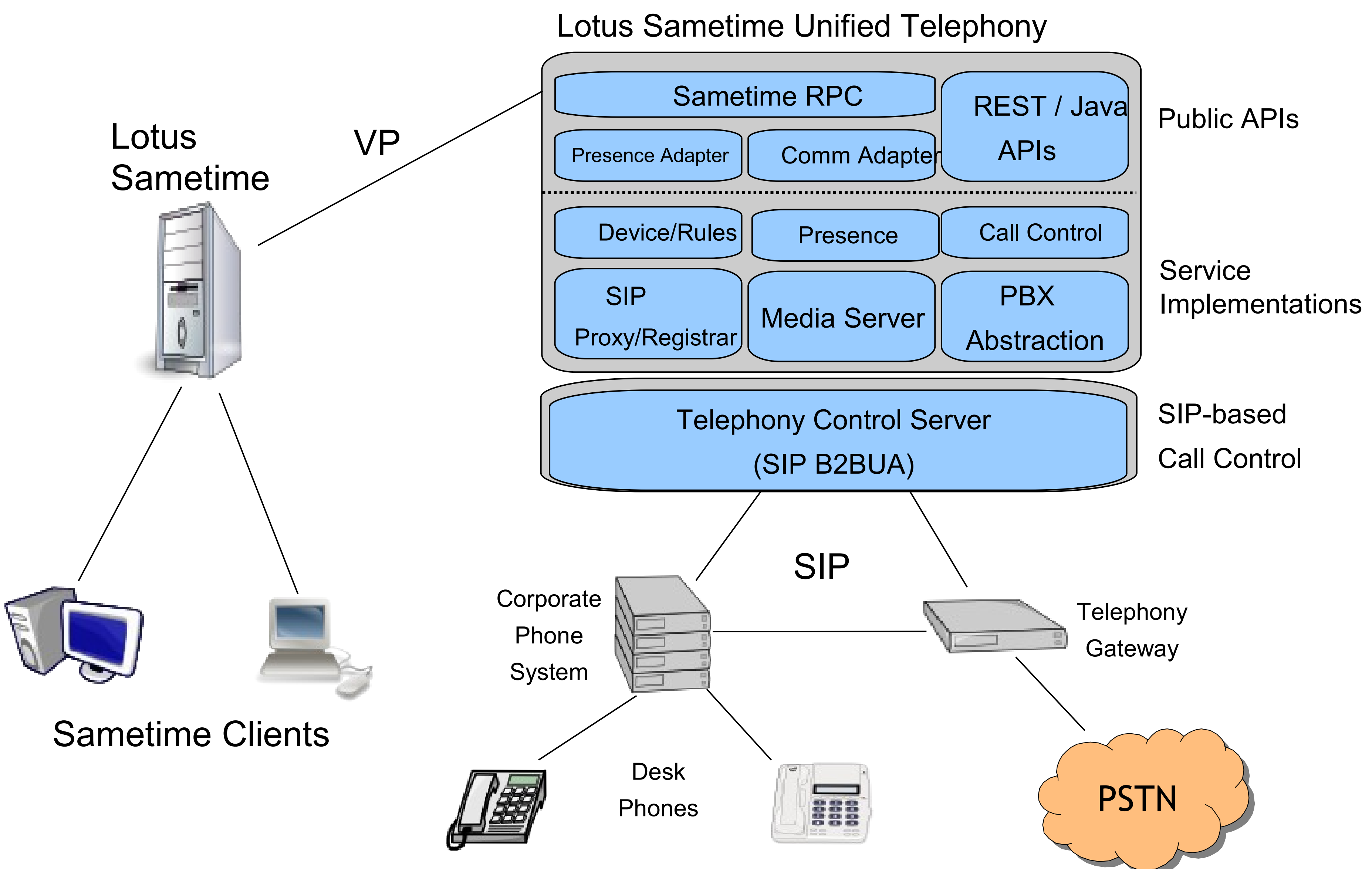


Send to office voicemail, convert to chat or forward to another person.

Accept incoming call to your current preferred number or device. If another number is preferred, click the dropdown arrow and select from list.

All UI depictions are not final and are subject to change

Lotus Sametime Unified Telephony Architecture

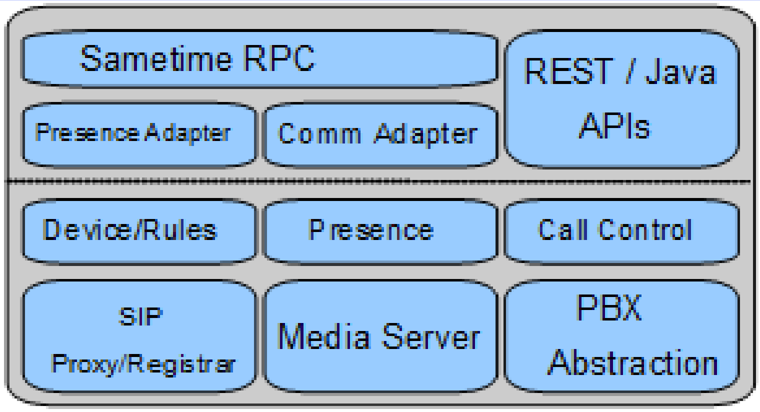

Lotus Sametime Unified Telephony
Sametime RPC
REST / Java APIs
Public APIs
Presence Adapter
Comm Adapter
Device/Rules
Presence
Call Control
Service Implementations
SIP Proxy/Registrar
Media Server
PBX Abstraction
Telephony Control Server (SIP B2BUA)
SIP-based Call Control
Lotus Sametime
VP
SIP
Corporate Phone System
Telephony Gateway
Sametime Clients
Desk Phones
PSTN

Telephony Control Server

Telephony Control Server
(SIP B2BUA)

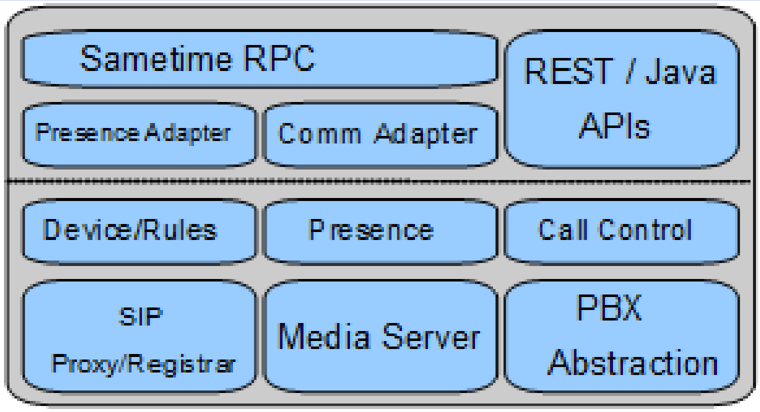
- B2BUA? English please ...
  - Back-to-Back User Agent (a term from RFC 3261)
- Logical entity that sits in the SIP call flow between two endpoints, maintaining state-full dialogs with each, allowing it to manipulate the call at any time to add value
- This allows Sametime Unified Telephony to provide sophisticated call handling with straightforward SIP signaling
- For example, call transfer is accomplished by creating a new call “leg” and then issuing a re-INVITE to the device being transferred
- The TCS can maintain “SIP trunks” to any number of PBXs and Gateways, based on the appropriate Internet Engineering Task Force (IETF) standards

Telephony Application Server



- Call Control / PBX Abstraction
  - Manages third-party call model
  - Permits sophisticated creation and manipulation of calls
- Presence Aggregation
  - Combines telephony presence from multiple devices, and publishes the result to the Sametime server
- Device / Rules Management
  - Keeps track of devices for each user (multiple provisioning options)
  - Stores call routing rules, and executes work flows for all calls

Telephony Application Server

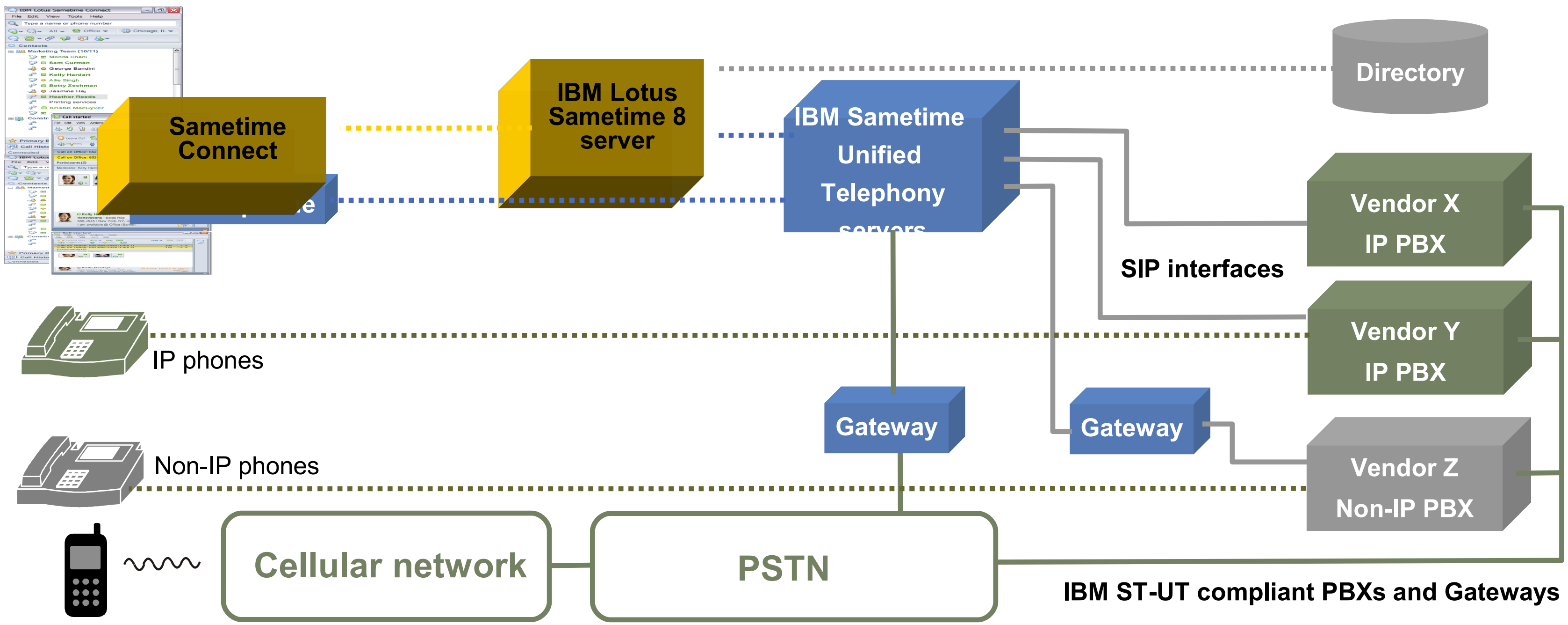


- Media Server
  - Announcements and ring back
  - Handles mixing for audio conferences
- SIP Proxy / Registrar
  - Manages registration and call routing for the Sametime soft phone
- Application Programming Interfaces
  - Remote by use of the Virtual Places protocol for client-side integration
  - REST / Java for server-side integration

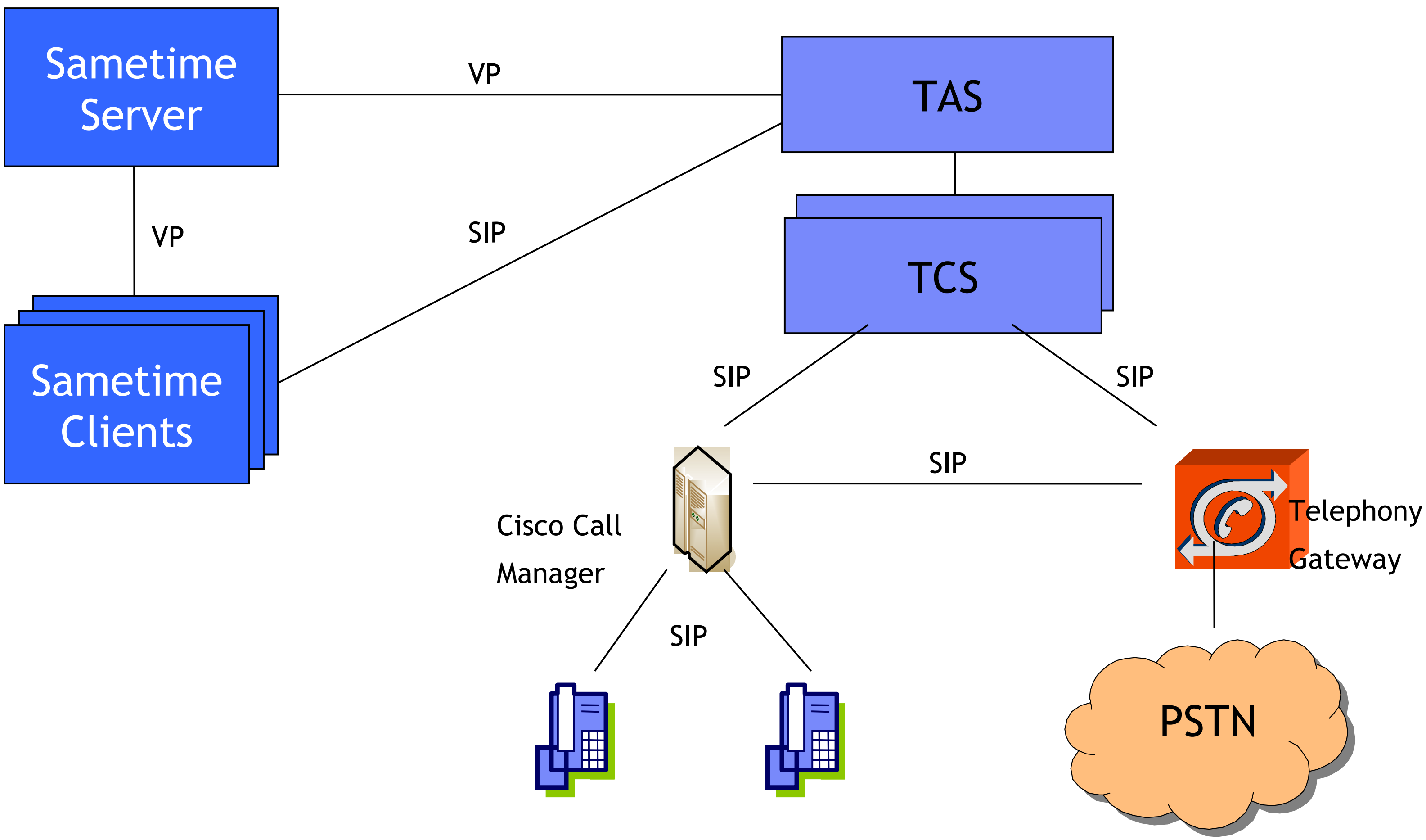
IBM Sametime Unified Telephony – Solution architecture

IBM Lotus Sametime Unified Telephony solution consists of:

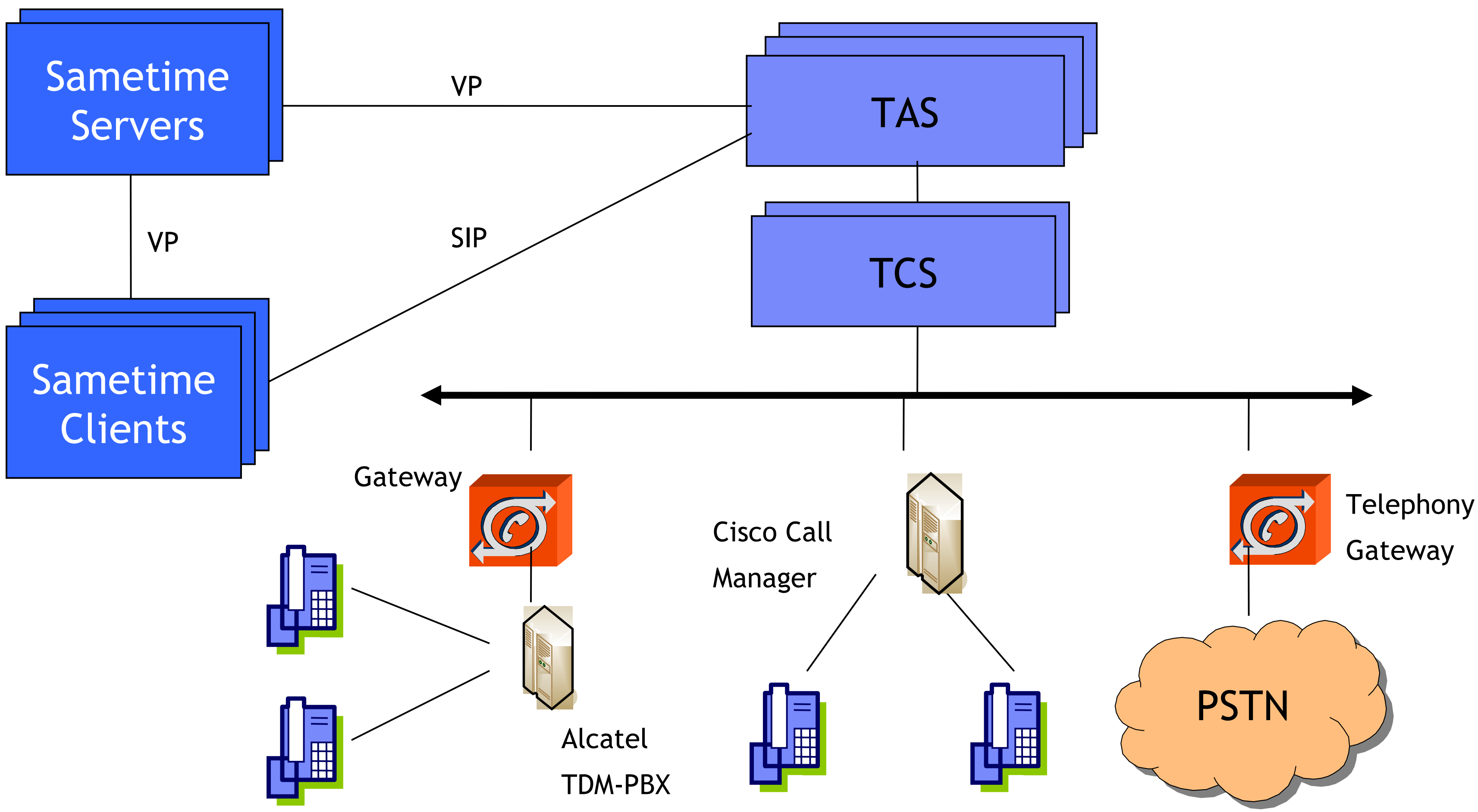
- IBM Sametime Standard or Advanced Deployment (pre-req)
- IBM Sametime Unified Telephony Software + softphone clients
- IBM Series x servers (minimum four: 2 x 3650T + 2 x 3455)
- OEM telephony gateways – PSTN and non-SIP PBXs (optional)
- IBM Global Technology Services and IBM Software Services for Lotus



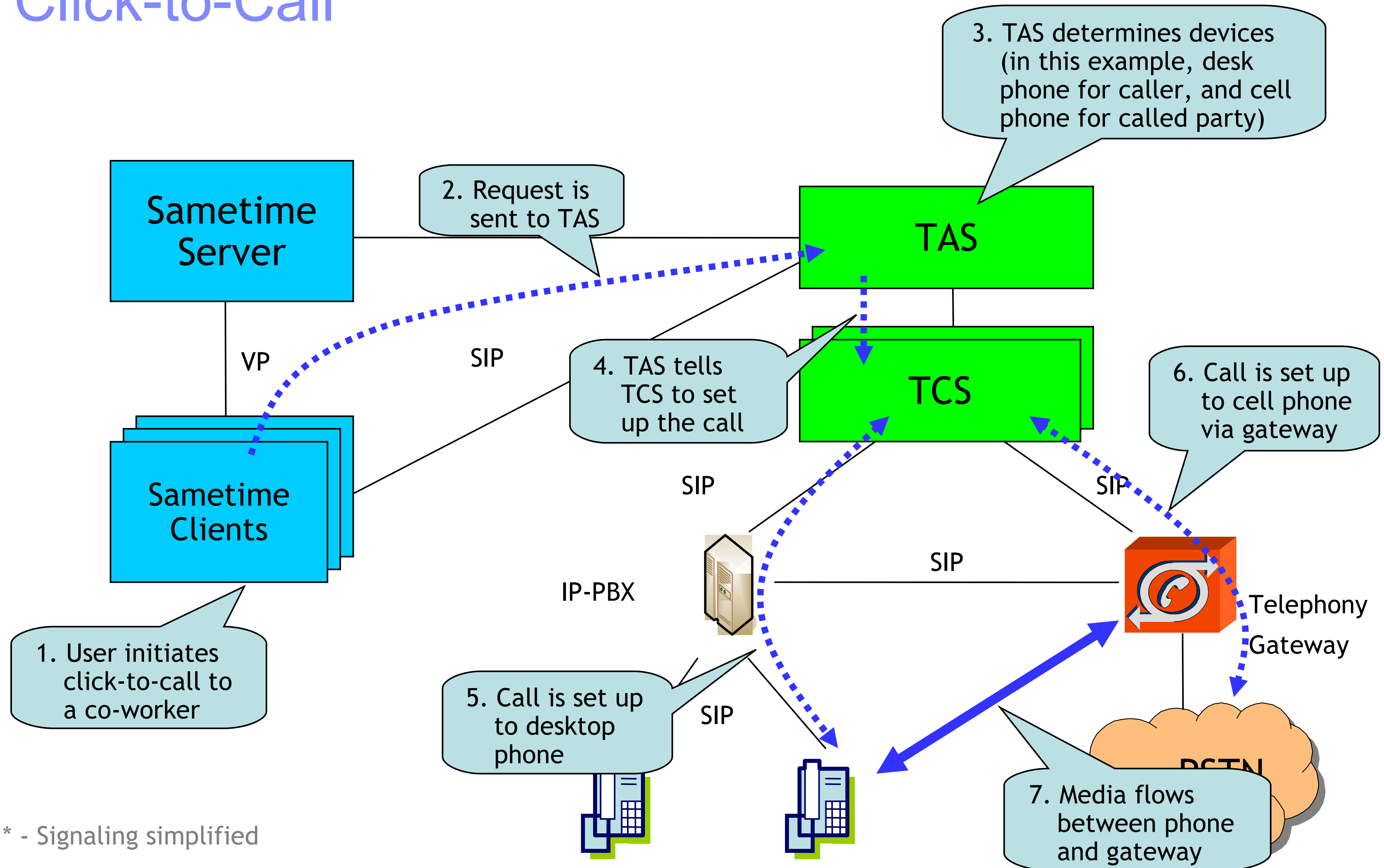
Minimum Deployment Architecture



Multi-Vendor Deployment Architecture

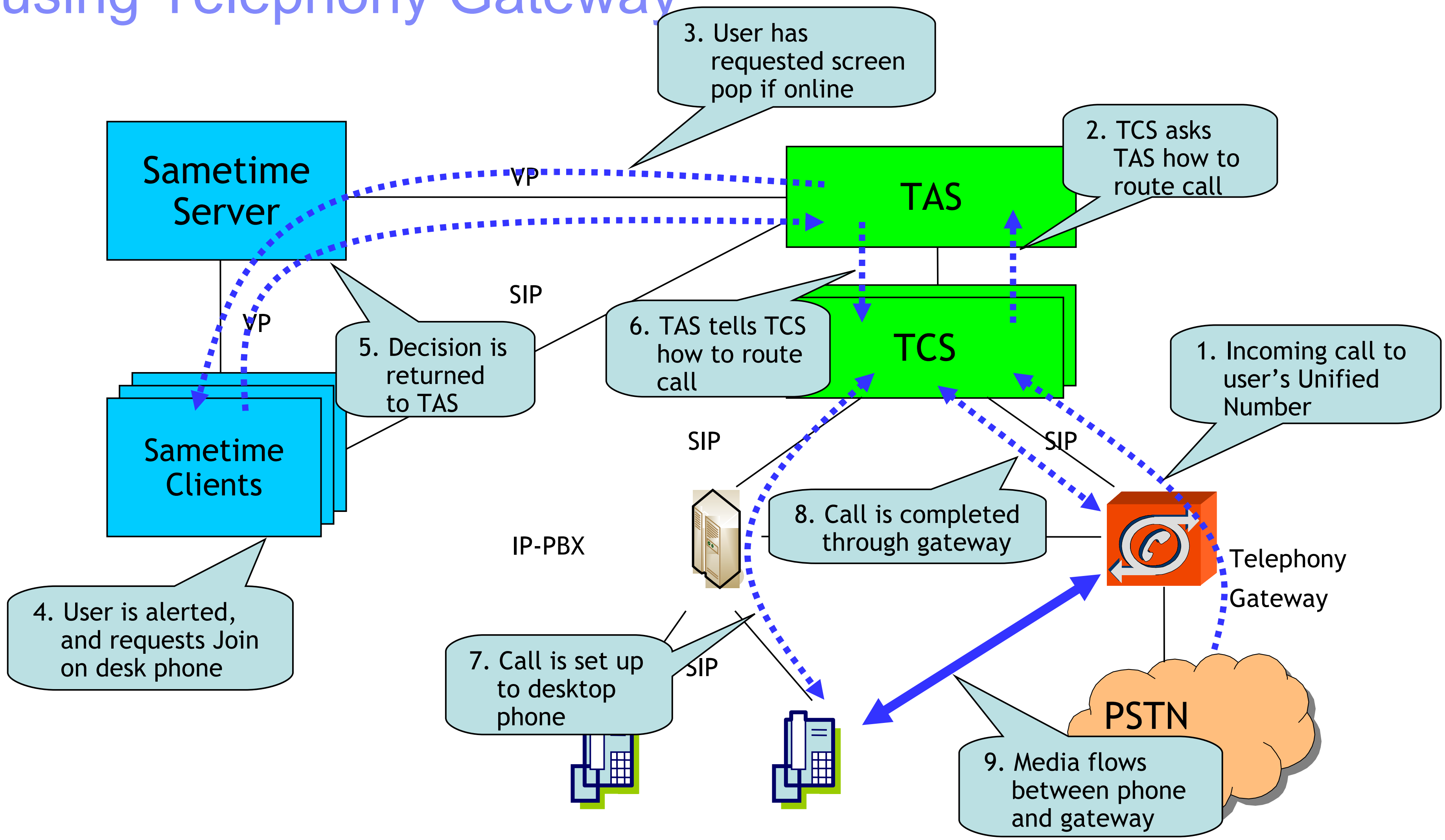


Click-to-Call



* - Signaling simplified

Sample Incoming Call Flow using Telephony Gateway



* - Signaling simplified

Performance

IBM System x™ servers

- **TAS - Telephony Application Servers**
  - Minimum 2 x IBM System x3550 servers
  - SuSE Linux® Enterprise Server (version 10)
  - one server needed for fail-over
  - one TAS server supports up to 15,000 users
  - 100,000 users would require 8 x TAS servers
- **TCS - Telephony Control Servers**
  - Minimum 2 x IBM System x3650T
  - SuSE Linux Enterprise Server (version 9)
  - additional TCS server needed for failover
  - supports up to 100,000 users
  - 200,000 users would require 4 x TCS servers
  - 3650T supports 5-9's of availability

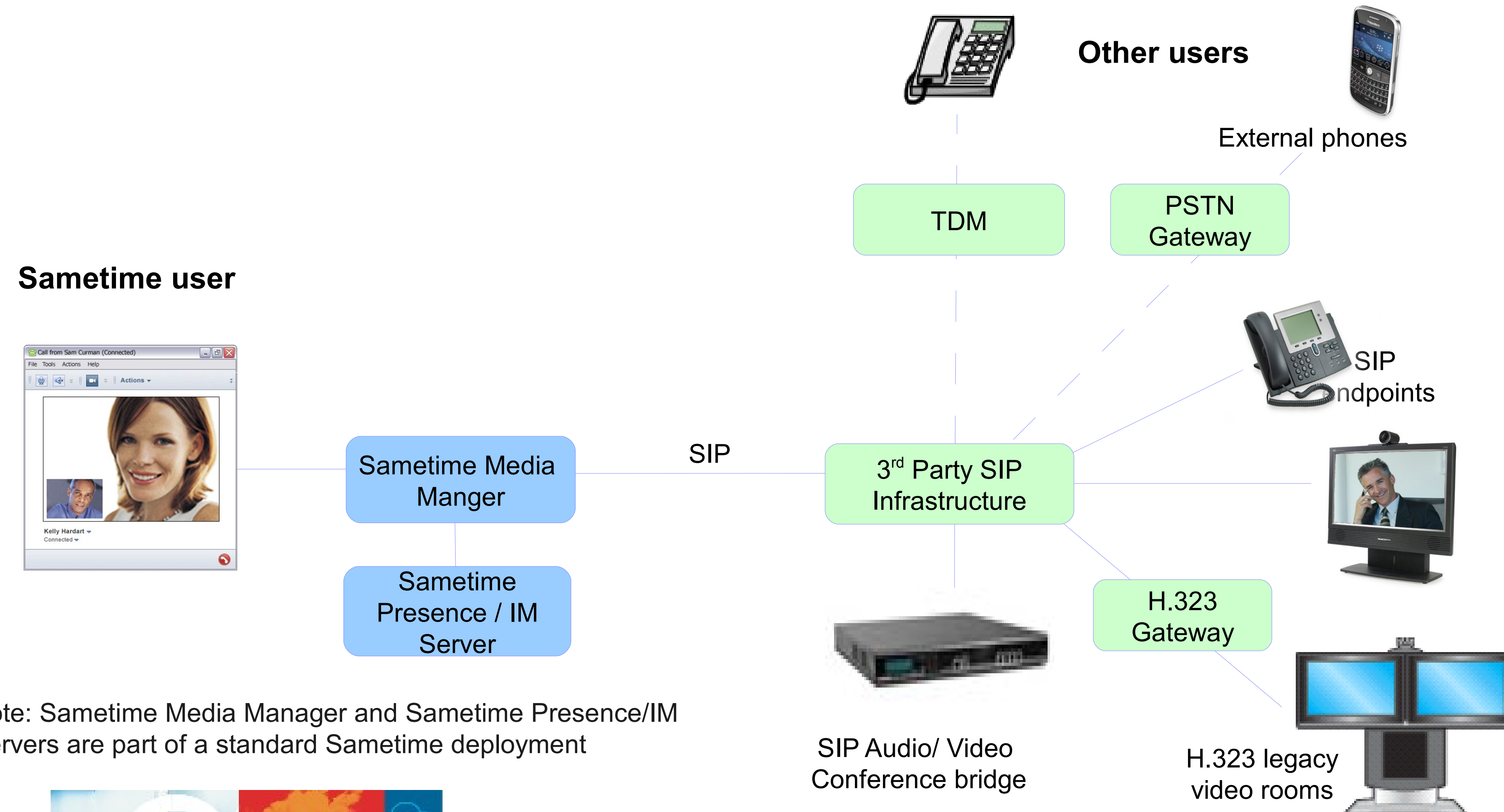
Summary

- Lotus Sametime Unified Telephony is unified communications middleware :
  - Integrating telephony across multivendor Private Branch Exchange (PBX) systems
  - providing a unified end user experience,
  - including integrated softphones; phone and IM presence awareness; and call management and call control across multiple communications systems.

Multiple integration options.....with SUT Lite

Plugins	SUT Lite Client	SUT
• Provided by Partner • Exposes unique partner features within Sametime • Best suited for single vendor solutions	• Provided by IBM • SIP standard calls to and from voice / video systems • Minimal infrastructure	• Provided by IBM • Unique, rich user experience – intelligent call routing, multiple devices, ad-hoc conferencing • Best suited for diverse telephony infrastructures
Examples • Cisco Unity Plugins • ALU telephony integration	**Example** • Tandberg video integration	

What does the SUT Lite architecture look like?



Note: Sametime Media Manager and Sametime Presence/IM Servers are part of a standard Sametime deployment

The difference between the SUT Lite Client License and full SUT

SUT Lite Client

- Place / receive calls from the Sametime 8.5.2 Connect client.
- Call video endpoints or video MCUs
- Call telephone numbers or conference bridges
- Within a call: mute/unmute, raise/lower volume, start/stop video, leave call
- Other features: Click to call, dial through Quickfind or Dial Pad, view call history

Sametime Unified Telephony

- Single number service
- “Off hook”” presence status
- Intelligent Incoming call rules & routing
- Multiple device support
- Transfer calls between devices
- Visual audio conferencing
- Moderator conference controls
- Hold, Transfer, merge calls
- Works with multiple PBXs to create a seamless UC environment

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Questions

Reference materials

- IBM Sametime 8.5 Infocenter
  http://publib.boulder.ibm.com/infocenter/sametime/v8r5/index.jsp