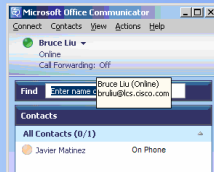# Cisco Unified Communications

Aleksandar Vulović

System Engineer

alvulovi@cisco.com

Cisco Unified Communications System

3RD Party Applications

Microsoft outlook

Vertical apps.

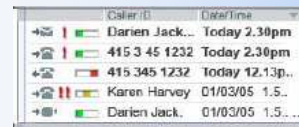**IBM Lotus Notes**

Applications

Mobility Manager

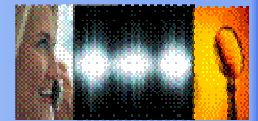**Unity Unified Messaging**

Unified Meeting Place conferencing

Unified Presence Server

Unified Personal Communicator

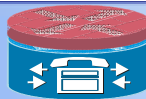**Unified Contact Centre**

Endpoints

Unified IP phones

Unified IP Communicator

Nokia Dual-Mode

Media Control

Cisco Unified Communications Manager Express

Cisco Unified Communications Manager

Cisco Unified Application Environment

Cisco IPICS

Infrastructure

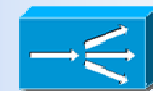**Routing**   **Switching**   **Availability**   **Management**   **QoS**   **Security**   **Administration**
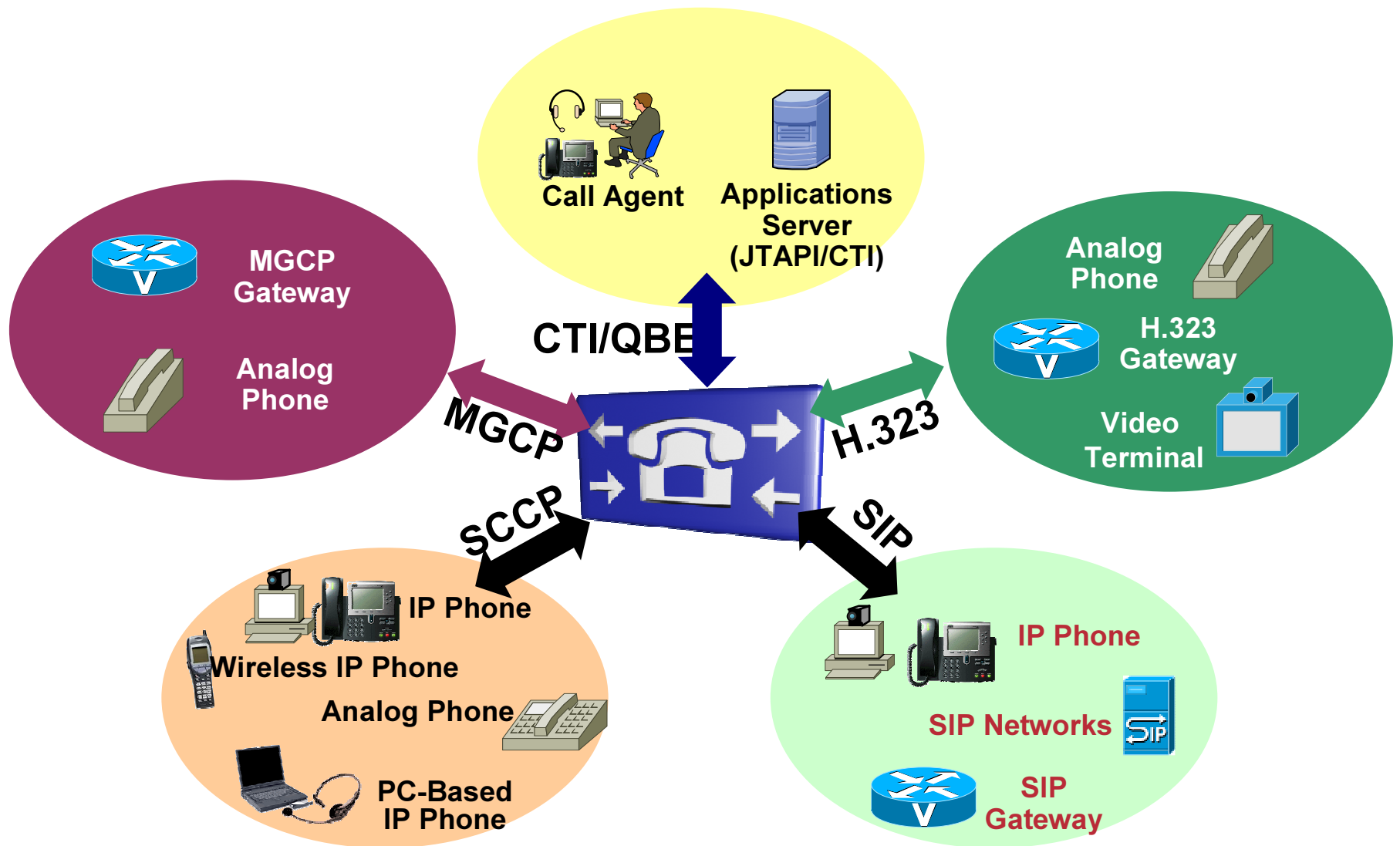
Agenda

1. Cisco Unified Communications Manager Overview and Update

2. Endpoints

3. New Secure UC Network Infrastructure Capabilities

4. Mobility

5. Collaboration

6. Webex Connect

7. Future

Cisco Unified Communications Manager Overview and Update

Signaling Protocols
CUCM as a "Protocol Translator"

Call Agent

Applications Server (JTAPI/CTI)

CTI/QBE

MGCP Gateway

Analog Phone

MGCP

H.323

Analog Phone

H.323 Gateway

Video Terminal

SCCP

SIP

IP Phone

Wireless IP Phone

Analog Phone

PC-Based IP Phone

IP Phone

SIP Networks

**SIP Gateway**

Cisco IP Telepony End-to-end
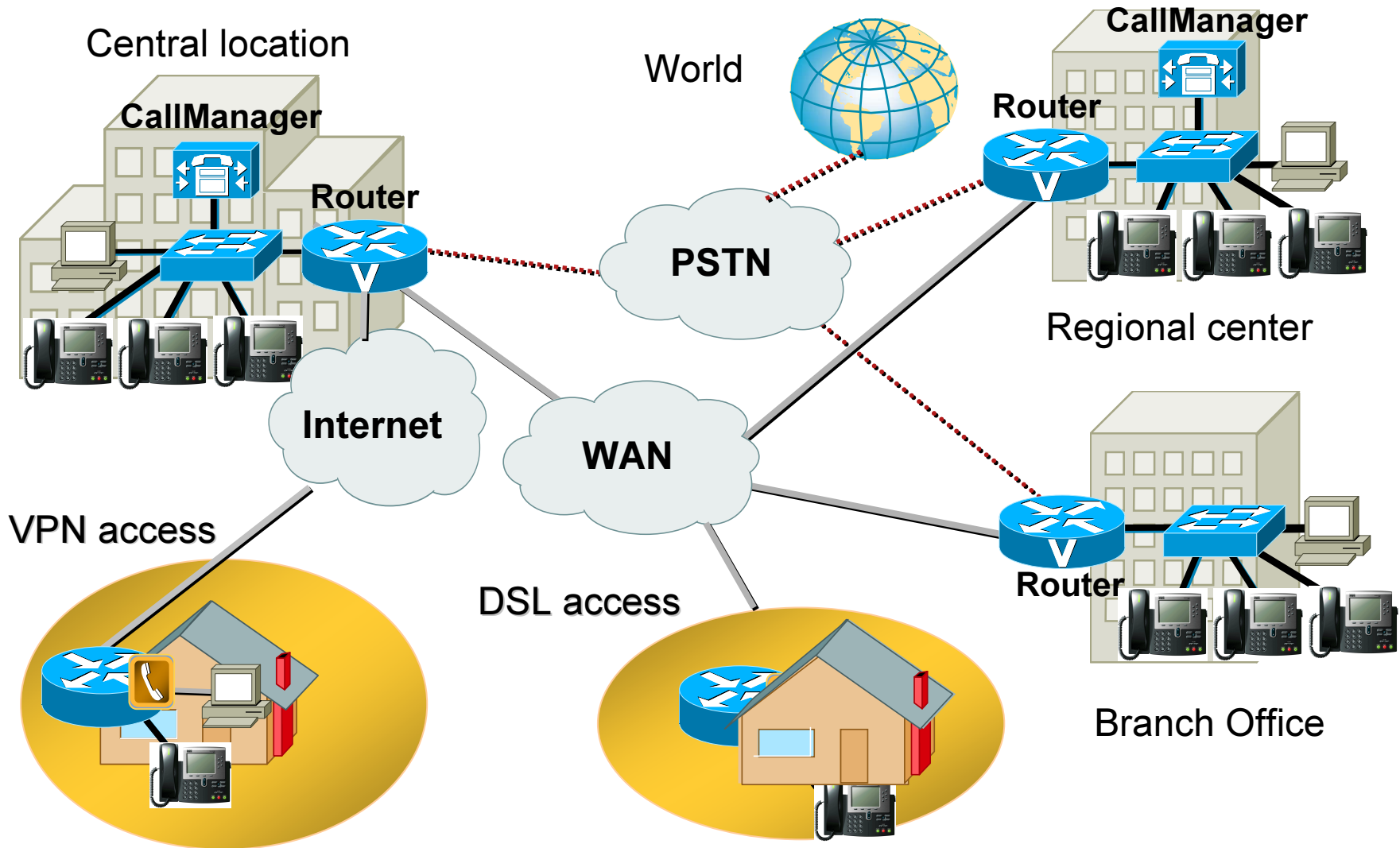
Survivable Remote Site Telephony

1. SRST provides redundancy in the case of WAN failure

2. Ideal for enterprise customers utilizing Centralized Cisco CallManager Deployment for IP telephony at their branch offices

3. Provides near feature parity with Cisco Call Manager

4. Configure it and forget it! SRST inherits its configuration from Cisco Call Manager and requires no configuration or intervention in the event of a WAN failure

5. Supported across all Voice platforms, up to 720 Phones per Router
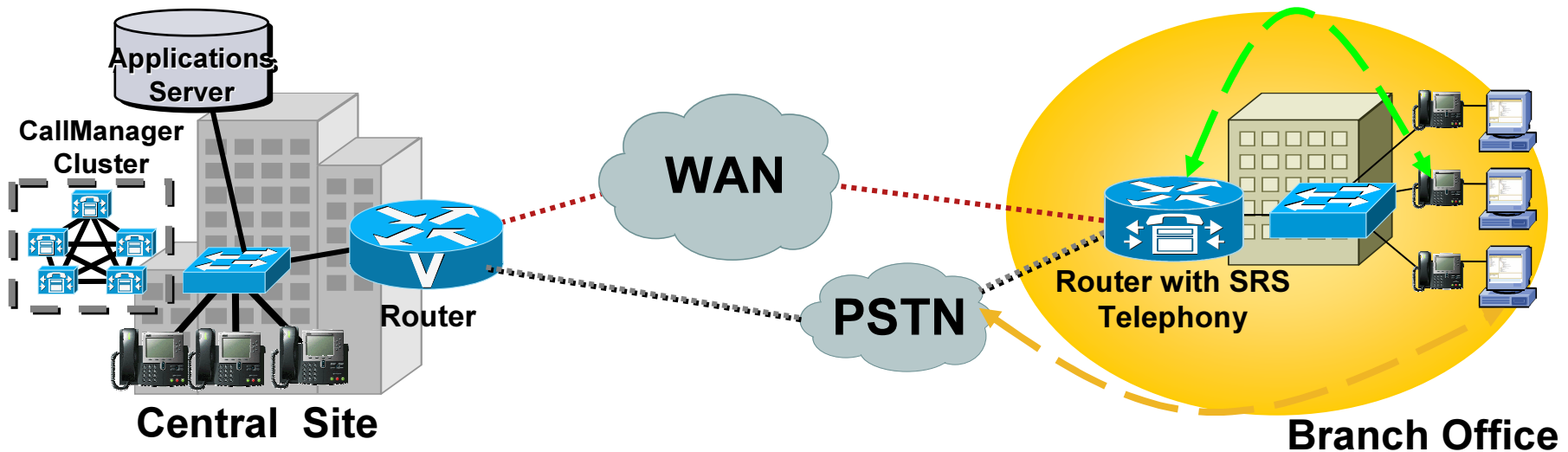
Interoperability with 3rd party PBX

E1 QSIG
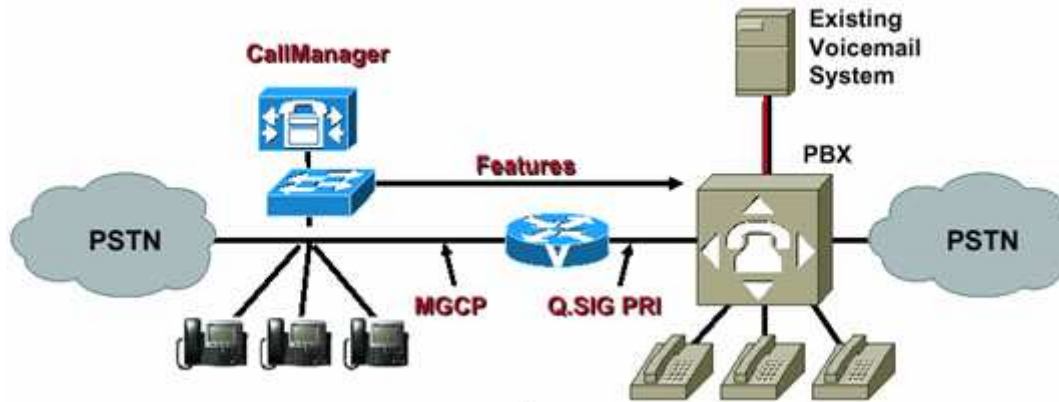
SIP trunk
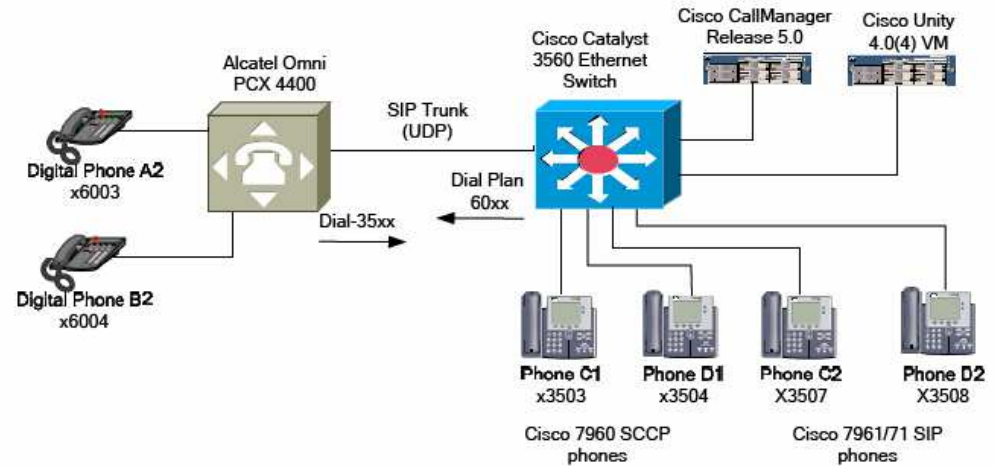
Cisco Interoperability Portal:

http://www.cisco.com/en/US/netsol/ns728/networking_solutions_program_category_home.html

Presence-Enabled Speed Dials, Call History Lists and Directories
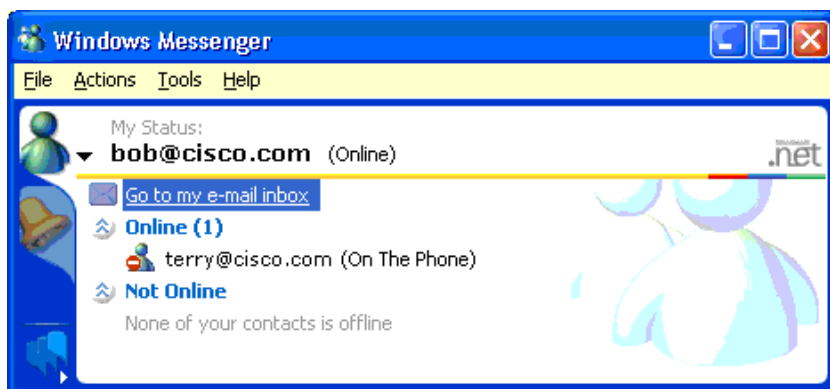
- **Presence status enabled on Speed Dial buttons, Call History Lists and Directories**

- **SIMPLE used to provide presence status to external SIP-based applications (requires Cisco Unified Presence Server)**

CUCM 7.0+ OS – Appliance only !

CUCM 3.3-4.2 **CUCM 4.3**

Win2000 **Win2003**

CUCM 5.0-6.1

Appliance

CUCM 7.0+

Appliance

Informix IDS
Tomcat
Endpoints 8.4(1) firmware release
RedHat Enterprise Linux Advanced Server 4

New in CUCM 7.0

1. **Called Party Normalization**

   ➤ Allows for easier callbacks, one touch dial from logs, no 'edit dial'

   ➤ Preserves the identity information of the caller. Adds flexibility to the presentation of international, national, or local subscriber numbers on the phone

2. **Support for "+" dialing**

   ➤ Dual mode phones (our existing phones don't have a +, but will support phones in SpeedDials and logs)

   ➤ "+" is displayed in Call logs (Missed Calls, etc.) and directories

   ➤ Facilitates global dialing plans that use escape codes

New in CUCM 7.0: Trusted Relay Point

UC Trusted QoS Enforcement
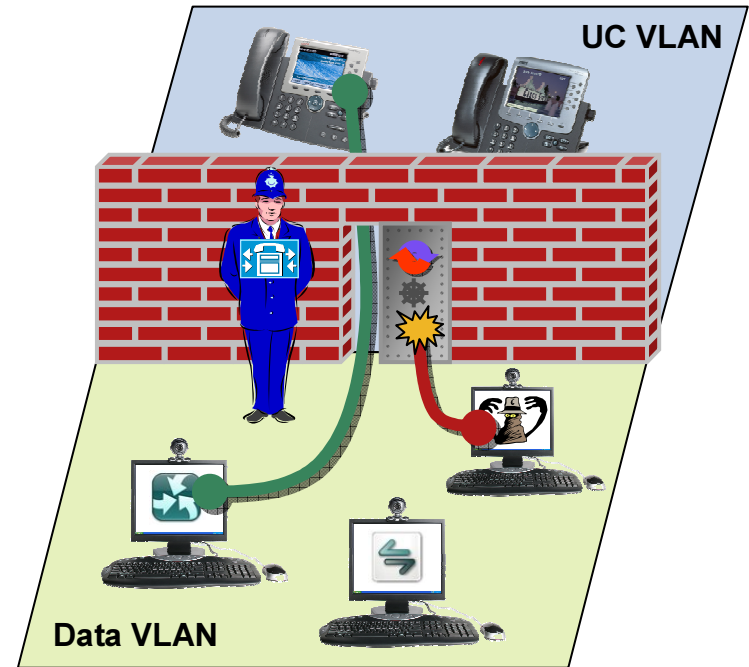

TRP

Fast
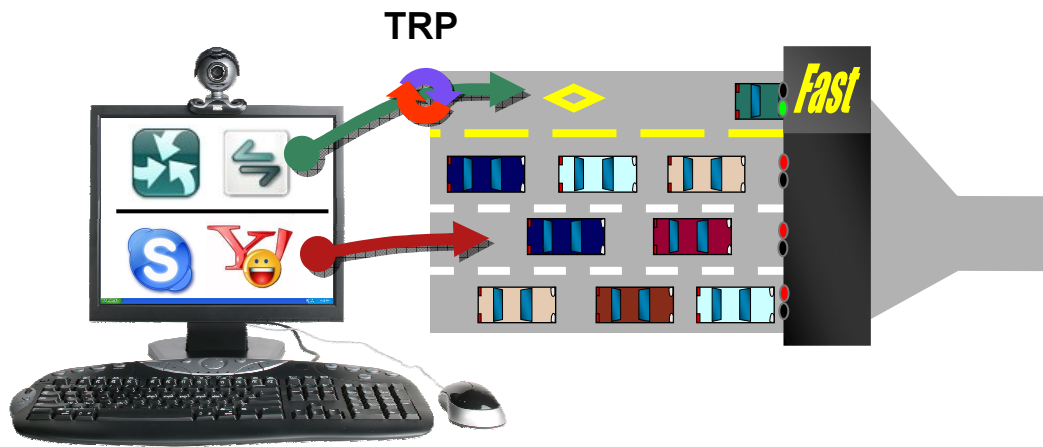
Cisco UC collaborates with the network to enable QoS for software UC clients

Simple solution applicable to any "untrusted" device that registers with CUCM
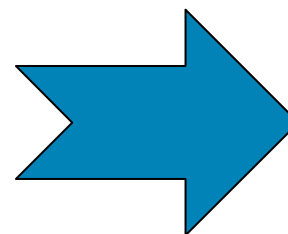
UC Trusted VLAN Traversal

UC VLAN

Data VLAN

Allows customers to effectively control access of UC VLAN from within data VLAN

8.4(1) Firmware phone features
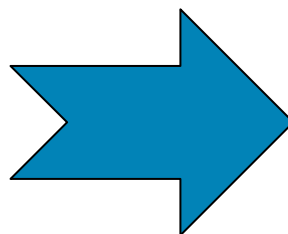
Transparent Line Label

Dynamic Call Window

Endpoints

Cisco Endpoints

Executive

Cisco Unified IP Phone
7970G / 7971G-GE

Manager

Cisco Unified IP Phone
7960G / 7961G / 7961G-GE

Business

Cisco Unified IP Phone
7940G / 7941G / 7941G-GE

Multibutton

Cisco Unified IP Phone
7931G

Basic

Cisco Unified IP Phone 7906G /
7911G

Video

Cisco Unified IP Phone
7985G

Cisco Unified Video
Advantage

Cisco Unified
Personal Communicator

Mobility

Cisco IP Communicator

Cisco Unified Wireless
IP Phone 7921/7925

Cisco Unified
Personal Communicator

Analog

Cisco ATA 186

Cisco VG202/204/VG224/VG248
Analog Phone Gateway

The Evolution of the Cisco Unified IP Phone Portfolio

Enhanced Power & Data Throughput Options

Enhanced Applications Capability

Unicode Support / Asian Localization

User Experience:

- Designed for Wideband/hi-fidelity voice, adds iLBC
- Choice of models with Gigabit Ethernet and large backlit color displays

Classic IP Phones

Manager

Cisco Unified IP Phone 7960G

Business

Cisco Unified IP Phone 7940G

Enhanced IP Phones

Executive

Cisco Unified IP Phone 7970G/71GE

Enhanced Manager

Cisco Unified IP Phone 7961G/GE

Enhanced Business

Cisco Unified IP Phone 7941G/GE

Advanced Media IP Phones

Cisco Unified IP Phone 7975G

Cisco Unified IP Phone 7945G/65G

Cisco Unified IP Phone 7942G/62G

Java-based '3rd-gen' platforms

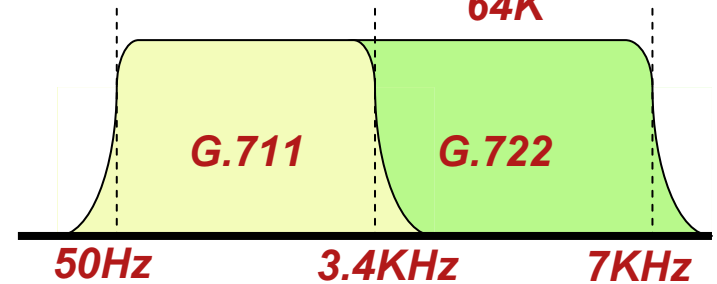# Enhancing the User Experience

Large (5"+) 16-bit TFT color displays on 3 new phone models

Wider Audio range – same bandwidth 64K

G.711   G.722

50Hz   3.4KHz   7KHz

Native wideband speaker and handset for optimal voice quality

Navigation and UI enhancements

Two New Specialty Endpoints

Cisco Unified IP Conference Station 7937

- New Industrial Design

- Enhanced audio (incl. G.722 wideband)

- Supports XML services

- 802.3af PoE power

- Support for wireless microphone kit

CUCM 4.1, 4.2, 4.3, 5.1, 6.0+

Cisco SIP Phone 3911

- Cost-effective, entry level endpoint

- Basic feature set only

- Single line

- SIP only

CUCM 5.1, 6.0+

Cisco Unified IP Phone 7915 / 7916 Expansion Modules

New

1. **Grayscale (7915) and Color (7916) modules for Cisco Unified IP Phones 7962G, 7965G, & 7975G**
2. **Large displays with an effective pixel resolution of 272x480 for numbers and names**
3. **12 physical keys with access to 12 additional keys for a total of 24 keys per KEM via 2 "shift/page" keys**
4. **Each 12 illuminated key is programmable as a line appearance (DN) or feature key.**
5. **Each of the 2 "shift/page" keys is illuminated to highlight the currently active page and to alert for activity on other page, i.e., incoming calls.**
6. **A maximum of two modules per device may be configured for 48 additional keys in future Unified Communications Manager Releases (Max of 42 additional keys in previous CUCM 6.1(1) and earlier)**
7. **Uses same footstands and power cubes as 7914**
8. **Supports all languages available on base IP phone**

Enhancing Productivity With Cisco Unified Communications Widgets

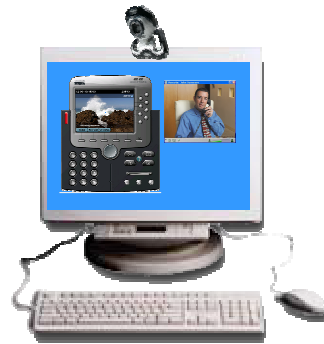**Cisco Unified Communications Widgets:** speedier collaboration to applications and team-members

- Ringtones/wallpaper

- Visual voicemail on Cisco

    Unified IP phones

- Click to dial from Outlook

Cisco Unified Communication Clients

Unified Personal Communicator
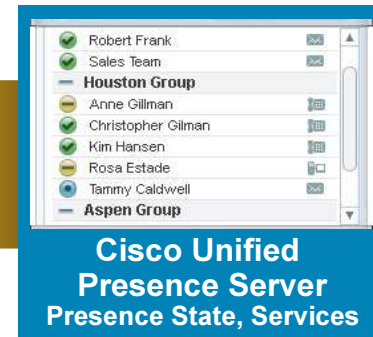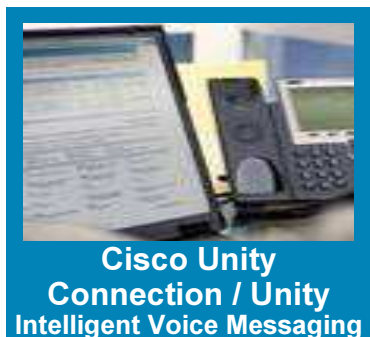
IP Communicator + VT Advantage

IP Communicator

Telephony focused　　　　　　　**Rich Media/ Unified Interface**

Cisco Unified Personal Communicator

Powerful communication tools in a single, easy-to-use desktop software application

Cisco Unified CallManager
IP Communications System

Cisco Unity Connection / Unity
Intelligent Voice Messaging

Cisco Unified Presence Server
Presence State, Services

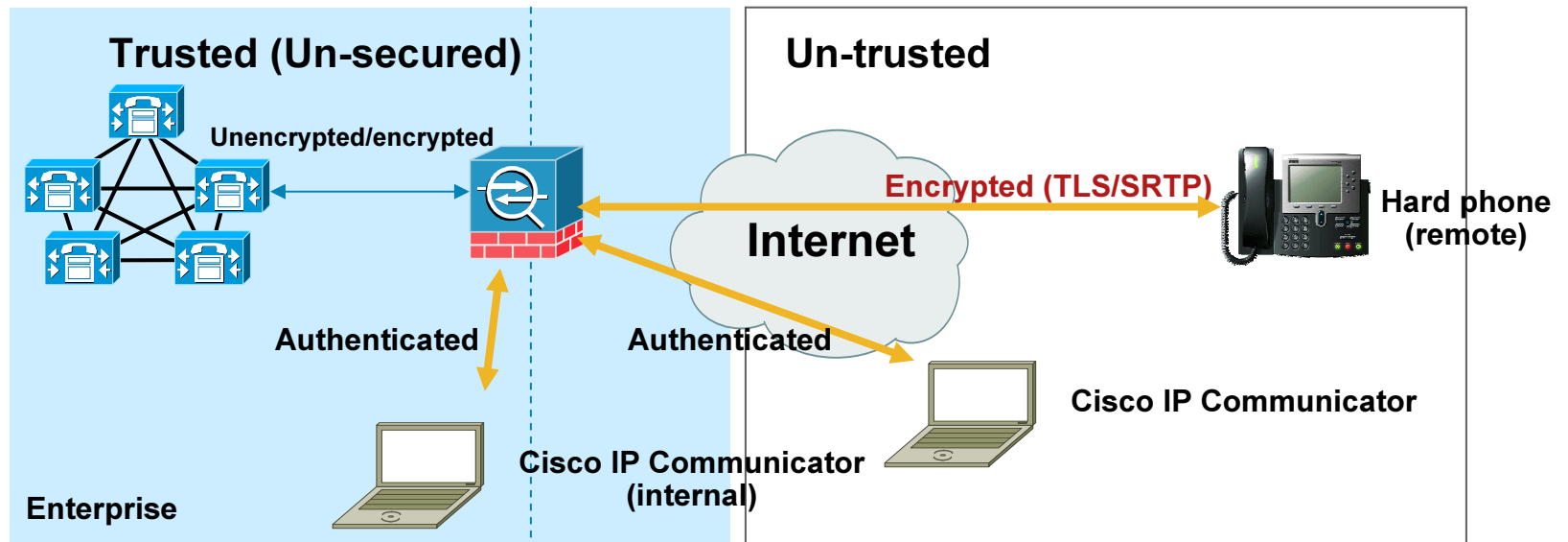**Cisco Unified MeetingPlace Express**
Web Conferencing

- Adaptive User Interface
- Presence-enabled
- Call, Collaborate, Escalate
- Desktop Video Calling
- IM
- Contact Lists, Directory Search
- Document Sharing

New Secure UC Network Infrastructure Capabilities

Cisco ASA Phone Proxy
Remote Access and Voice/Data Segmentation

Deployment Scenarios:

1. **Secure Remote Access:**
   - Terminates encrypted endpoints for secure remote access

2. **Voice/data VLAN segmentation for Softphone Applications:**
   - Authenticating soft clients before they reach CUCM
   - All communicator traffic originating from soft clients must be "proxied"
   - Soft client communication is restricted to specific VLAN on ASA
   - Cisco ASA performs inspection on traffic and opens media port dynamically for soft clients

Large Enterprise Customer Challenge
Solving the Firewall & Encryption Integration Problem

Customer Security policy mandates
- All Servers, including CUCM, must be firewalled for their protection
- Key end users must have all phone calls encrypted

Firewalls need to inspect the signaling traffic to
- Open Media pinholes
- Apply Protocol Conformance
- Apply Application Inspection and Control (AIC)

Encrypted calls must encrypt the signaling (TLS) because phones have the media encryption keys sent to them by CUCM via the signaling

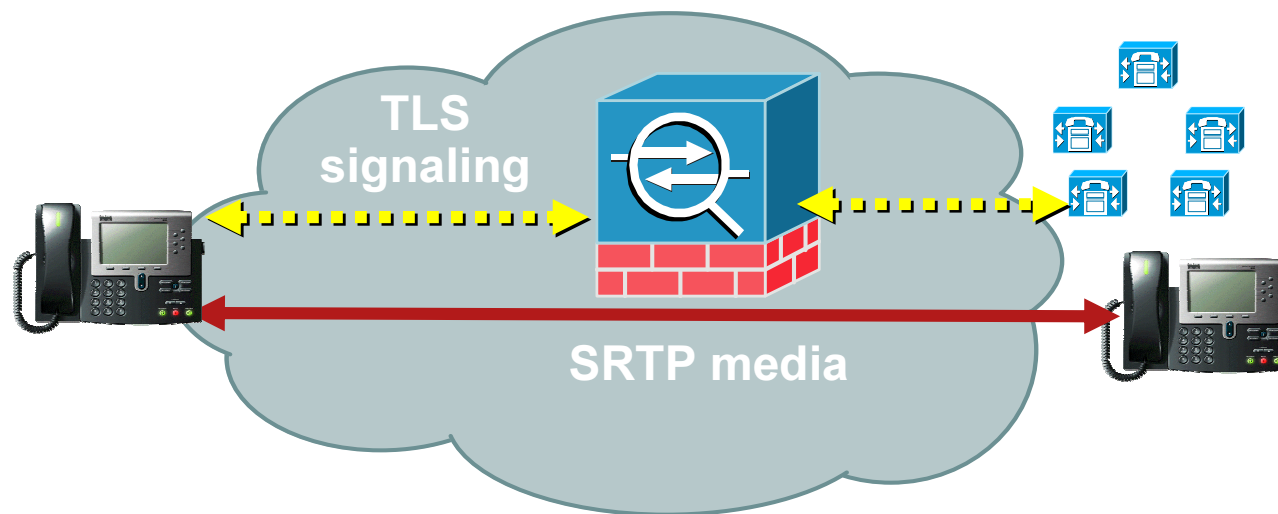| Problem	Customer Options	Cisco Solution
Two key security functions cannot co-exist or integrate	Choose Encryption or Firewalling, but not Both	The ASA TLS Proxy (ASA 8.0)

Cisco ASA TLS Proxy
Security – UC – Network Integration

CUCM encrypted calls with SRTP/TLS can now be inspected by Cisco ASA 5500 Adaptive Security Appliances:

- Maintains integrity and confidentiality of call while enforcing security policy through advanced SIP/SCCP firewall services

- TLS signaling is terminated and inspected, then re-encrypted for connection to destination (HW Based encryption)

- Dynamic port is opened for SRTP encrypted media stream, and automatically closed when call ends

NOTE: Not the same as Phone Proxy – NOT designed for Remote Access phones

Mobility

Cisco Mobile Unified Communications Solution Portfolio

Wireless IP Phones	Single Business Number & Voice Mail	Seamless Mobile Collaboration

Cisco Unified Wireless IP Phone (792x)

Nokia dual mode phones

Cisco Unified Mobility

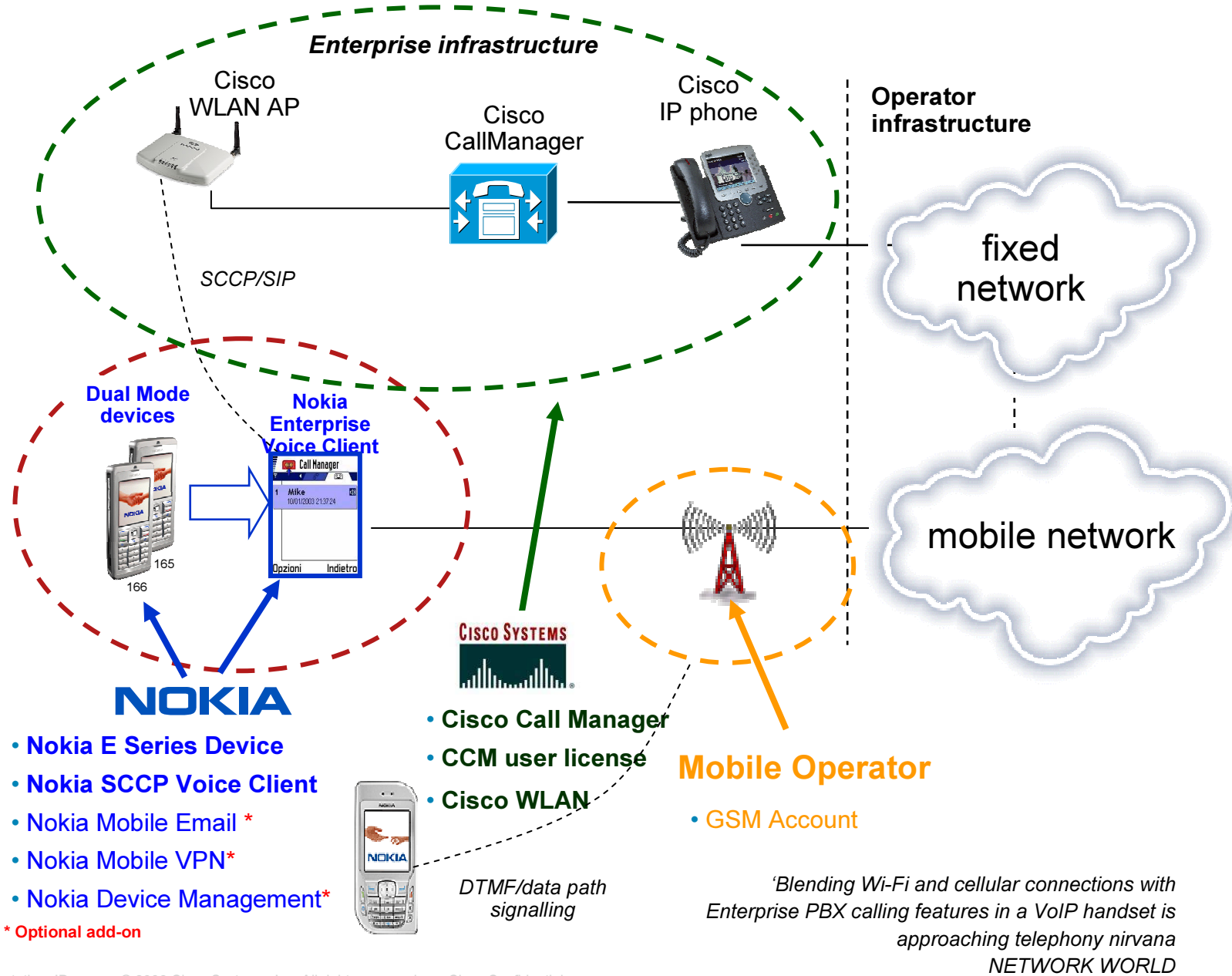**Cisco Unified Mobile Communicator**

Cisco Unified Wireless IP Phone
WLAN Phone: 7925G

New

1. New 802.11 a/b/g SCCP phone

2. Rugged and compact handset designed for users across wide range of vertical markets

3. Ideal for demanding environments such as Healthcare, Retail, Manufacturing

4. Hermetically sealed - Resistance to dust, liquid splashes, and moist wipes

5. Rugged - More resistant to drops, shocks, and vibration

6. Color display (same size as 7921G)

7. Bluetooth 2.0 - Bluetooth and WLAN co-existence

8. Built-in speakerphone for hands-free operation

9. Standard battery

   – Up to 150 hrs standby OR up to 10 hrs talk time *(7921G is 70 hrs standby/6 hrs talk)*

10. Extended battery

   – Up to 200 hrs standby OR up to 13 hrs talk time *(7921G is 90 hrs standby/8 hrs talk)*

Nokia Dual Mode phone

Enterprise infrastructure

Cisco
WLAN AP

Cisco
CallManager

Cisco
IP phone

Operator infrastructure

SCCP/SIP

fixed network

Dual Mode devices

Nokia Enterprise Voice Client

Call Manager

1 Mike
10/01/2003 21:37:24

Opzioni   Indietro

165

166

mobile network

CISCO SYSTEMS

NOKIA

- **Nokia E Series Device**
- **Nokia SCCP Voice Client**
- Nokia Mobile Email *
- Nokia Mobile VPN*
- Nokia Device Management*

* Optional add-on

- **Cisco Call Manager**
- **CCM user license**
- **Cisco WLAN**

NOKIA

DTMF/data path signalling

Mobile Operator

- GSM Account

'Blending Wi-Fi and cellular connections with Enterprise PBX calling features in a VoIP handset is approaching telephony nirvana
NETWORK WORLD

30

Nokia Dual-Mode Phone

- **Nokia Intellisync Call Connect**
- **Phone handsets: Nokia E60, E61, E61i,E65, E51, E66 and E71**
- **SCCP-only Client**
- **Switch to cellural: supported on E51, E66 and E71**
- **Nokia Mobile VPN**
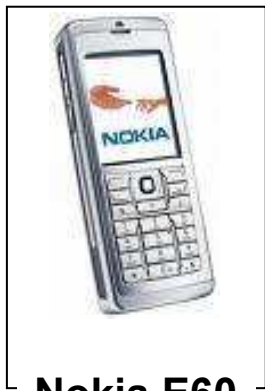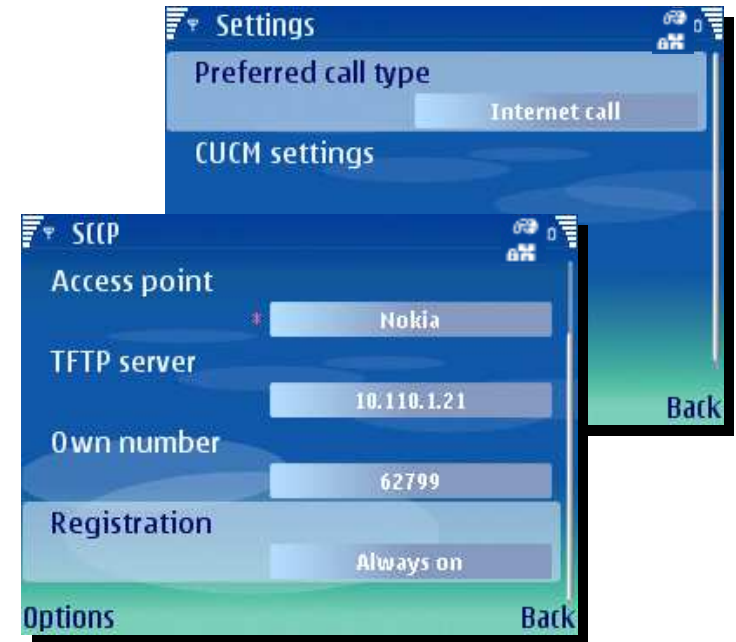
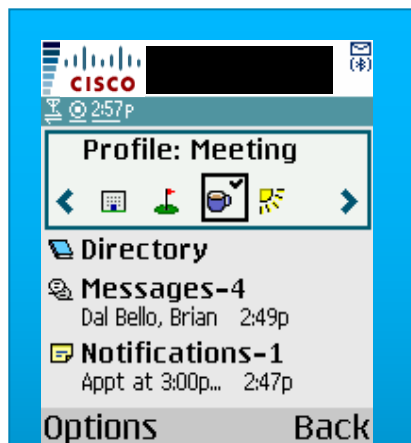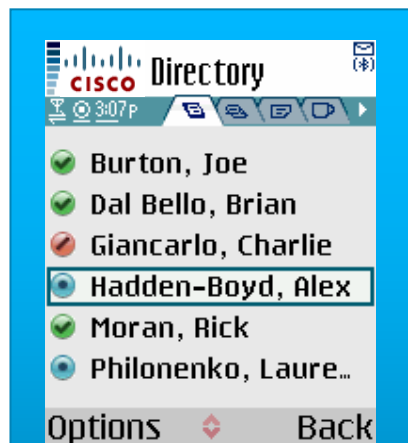Nokia E60   Nokia E61   Nokia E61i   Nokia E65   **Nokia E66**   **Nokia E71**
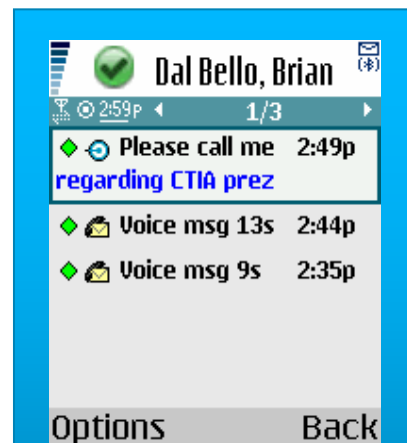
Cisco Unified Mobile Communicator

Cisco Unified Mobile Communicator

- Client available on multiple Smart and Feature Phones: Symbian, BREW, Windows Mobile and Blackberry
- Provides Notifications from MeetingPlace via Outlook

Presence-enabled Enterprise Directory

- Provides presence status
- Same user experience as Cisco Unified Personal Communicator

Visual Indication of Unity Voice-Mail

- Matches caller ID with Directory to give name display
- Provides Presence information of Caller
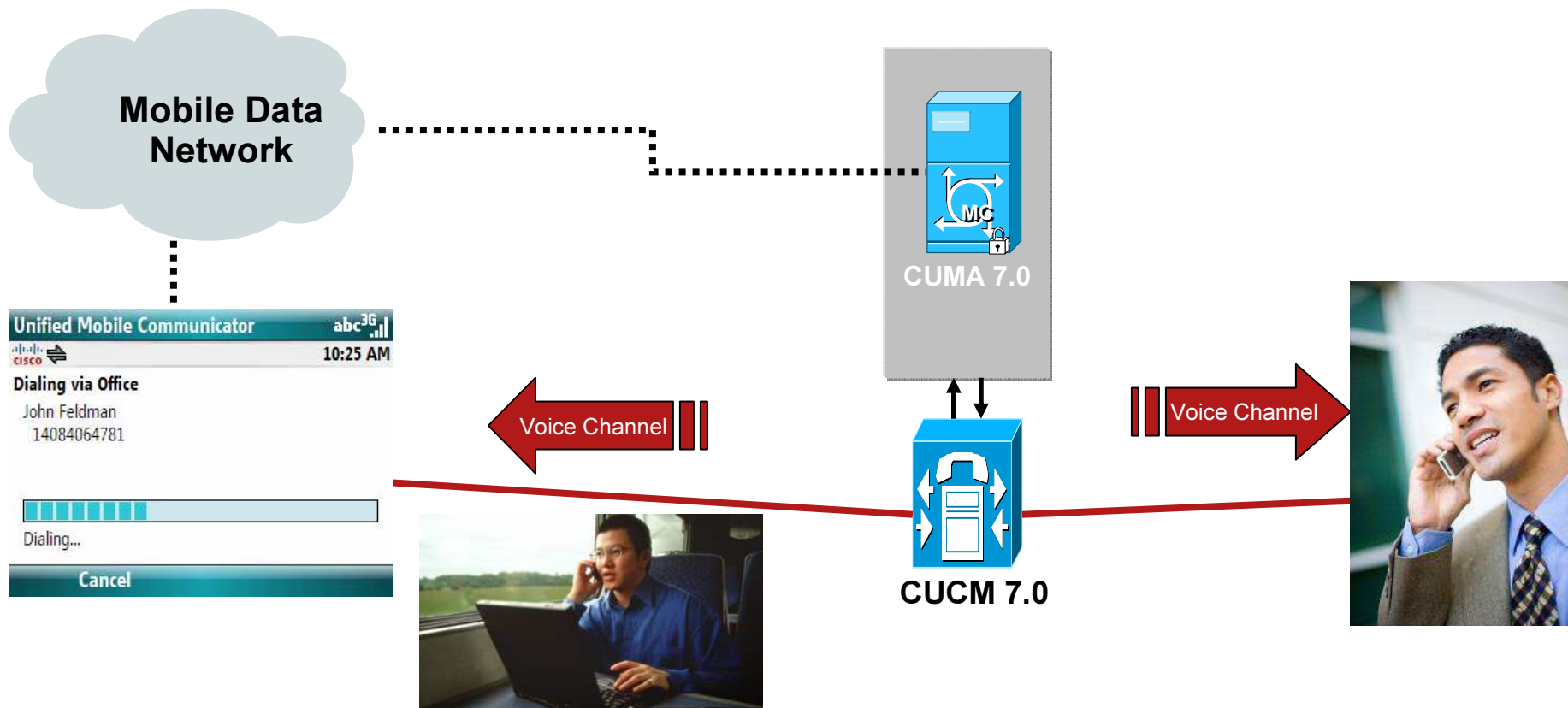- Click to call Unity and playback specific message
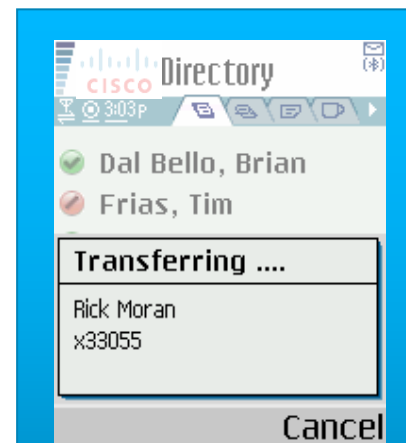
Unified CallManager Call Control

- Click to call – routed through Cisco Unified CallManager leverages Least Cost Routing

- While on Call
- Transfer
- Conference

Dial Via Office

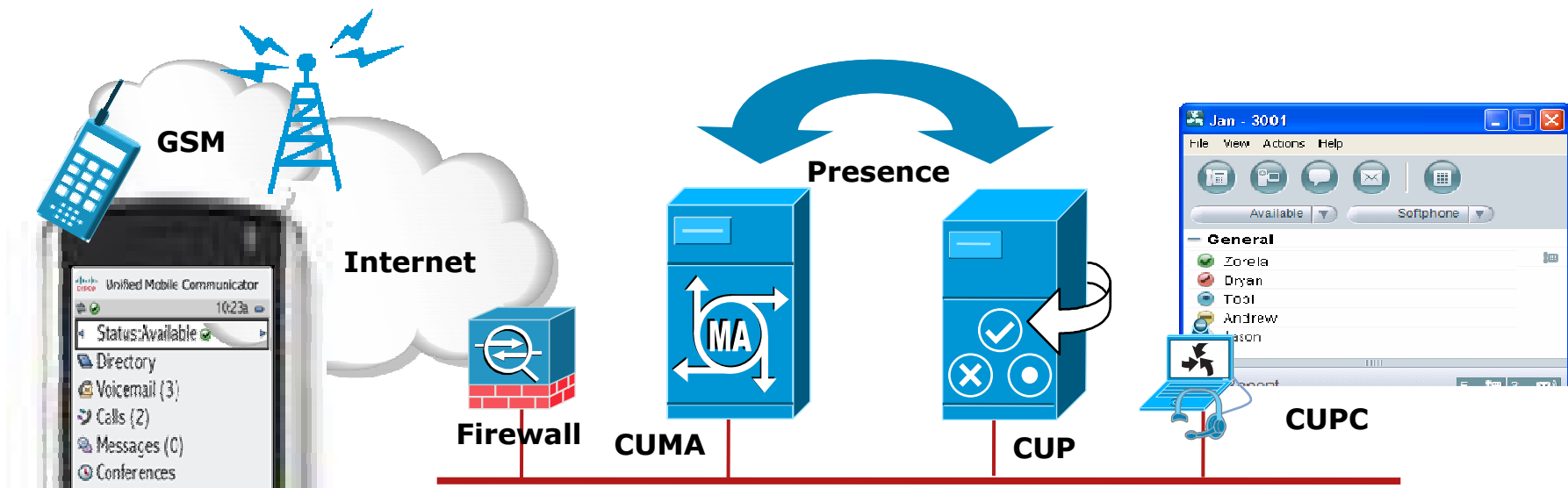**Make calls from mobile phone leveraging enterprise telephony infrastructure**

Demo: http://10.89.242.18/%5CUsing_DVO.htm

Cisco Unified Presence 7.0
Mobility: Diagram and Key Points

CUP 7.0 introduces common presence with CUMA/MC

| Presence	Contact Lists	Instant Messaging
•CUMC/CUMA will coordinate presence with CUP so it is in synch with other Cisco enterprise clients	CUMC/ CUMA will coordinate its buddy list with CUP so it is in synch with other Cisco enterprise clients	IM is between CUPC and CUMC is NOT supported in this release.

34

ASA as a Mobility Proxy

Adaptive Security Appliance (ASA) provides improved security and replaces CUMA Proxy server

Multi-purpose use appliance provides strong TCO

- **ASA provides deep packet inspection and hardware encryption/de-encryption for MMP packets.**

- **Requires ASA proxy session licenses**

Cisco Unified Mobile Communicator

Enterprise

MMP over SSL

DMZ

CUMA

ASA mobility Proxy

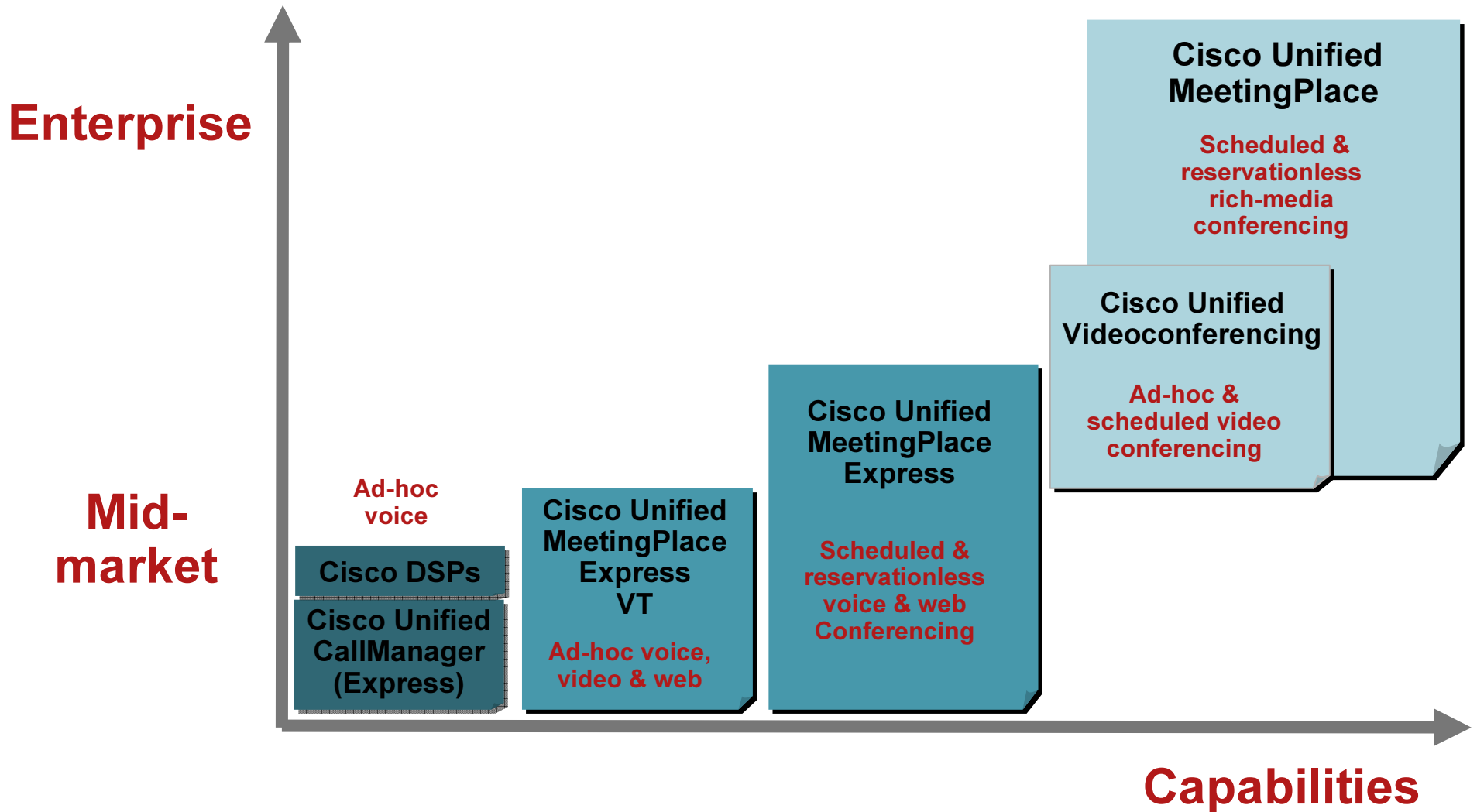Cisco Unified Mobility Advantage Enterprise Server
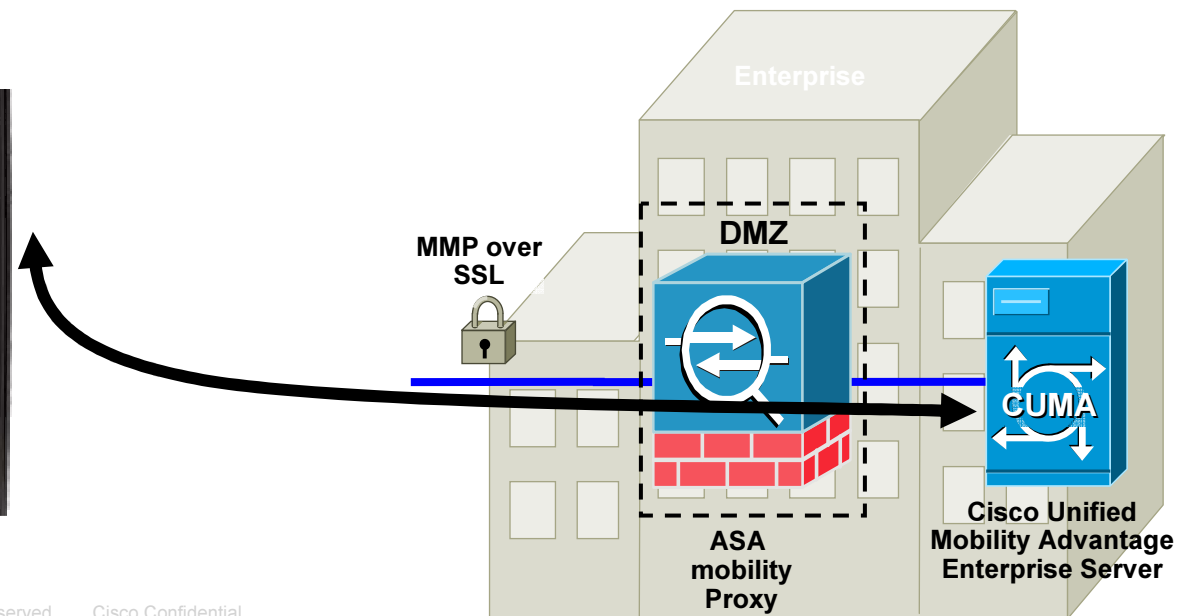
Collaboration

Introducing IP Video Telephony

Enterprise-class IP telephony and video is delivered over a single Cisco IP Communications infrastructure

Cisco Call Manager 4.1+

Cisco IP Communications Network

Professional

Color

Business

Wireless

Executive Desktop

IP Phone with PC Video

Standards Based H.323 System

Small Group System

Cisco Conferencing Solutions

Enterprise

Mid-market

Cisco Unified MeetingPlace

Scheduled & reservationless rich-media conferencing

Cisco Unified Videoconferencing

Ad-hoc & scheduled video conferencing

Ad-hoc voice

Cisco DSPs

Cisco Unified CallManager (Express)

Cisco Unified MeetingPlace Express VT

Ad-hoc voice, video & web

Cisco Unified MeetingPlace Express

Scheduled & reservationless voice & web Conferencing
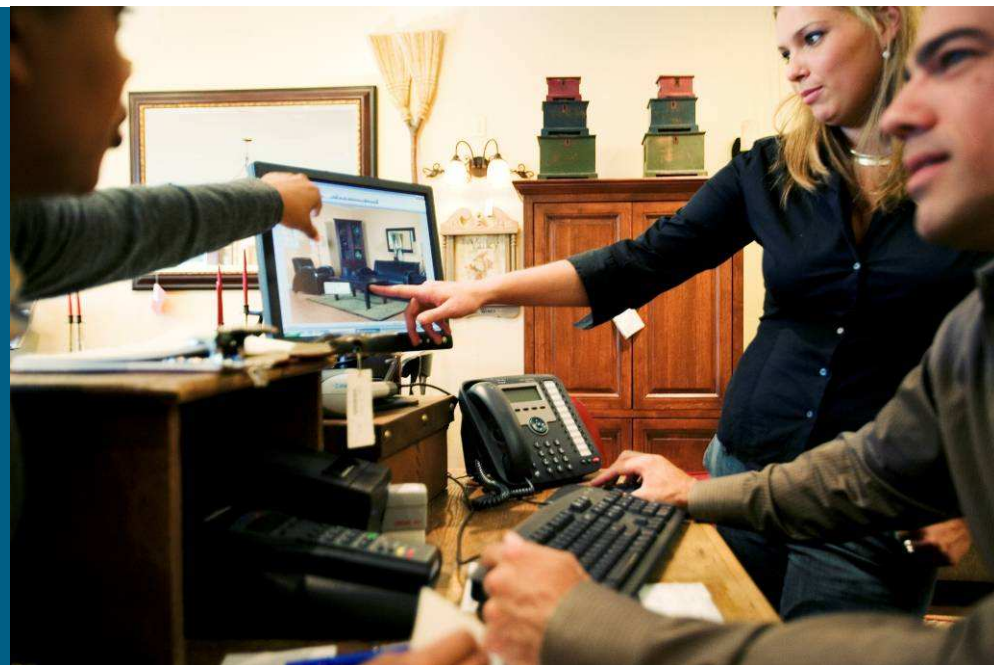
Capabilities

Can you describe me the taste of "Gravče na tavče"?

Can I describe you Cisco Telepresence?

Webex Connect

WebEx Snapshot

1. **10+ year history of innovation, growth, and market leadership**

2. **Pioneer in cross-company collaboration**

3. **Pioneer in SaaS (Software as a Service) model**

4. **Acquired by Cisco in 2007**

5. **>70,000 paying customers**

6. **2.8 million registered users**

7. **Over 125,000 meetings per day, > 7M people in WebEx sessions each month**

8. **MediaTone Network: Intelligent, Secure, Reliable, and Scalable Global Delivery. 99.99%+ availability with full hot-site redundancy**

Growth in Customers
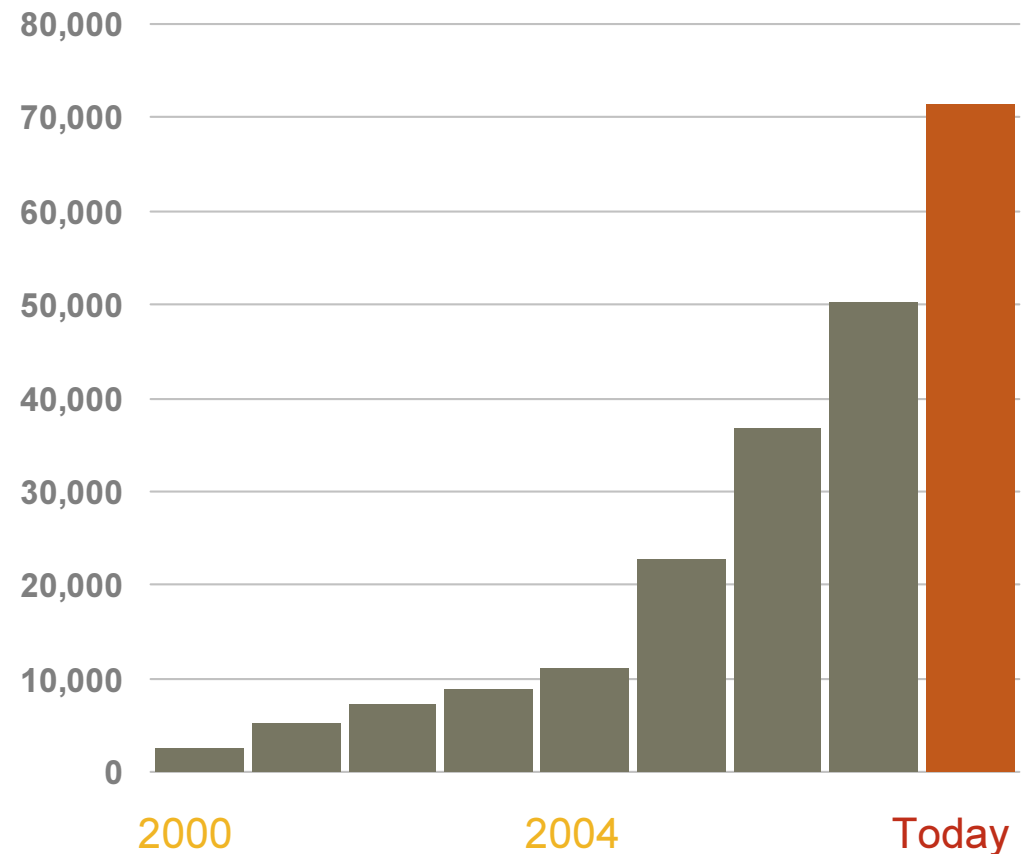
Includes WebOffice, MeetMeNow, and PCNow Customers Starting in 2006

WebEx Connect

Email

Presence

Instant Messaging
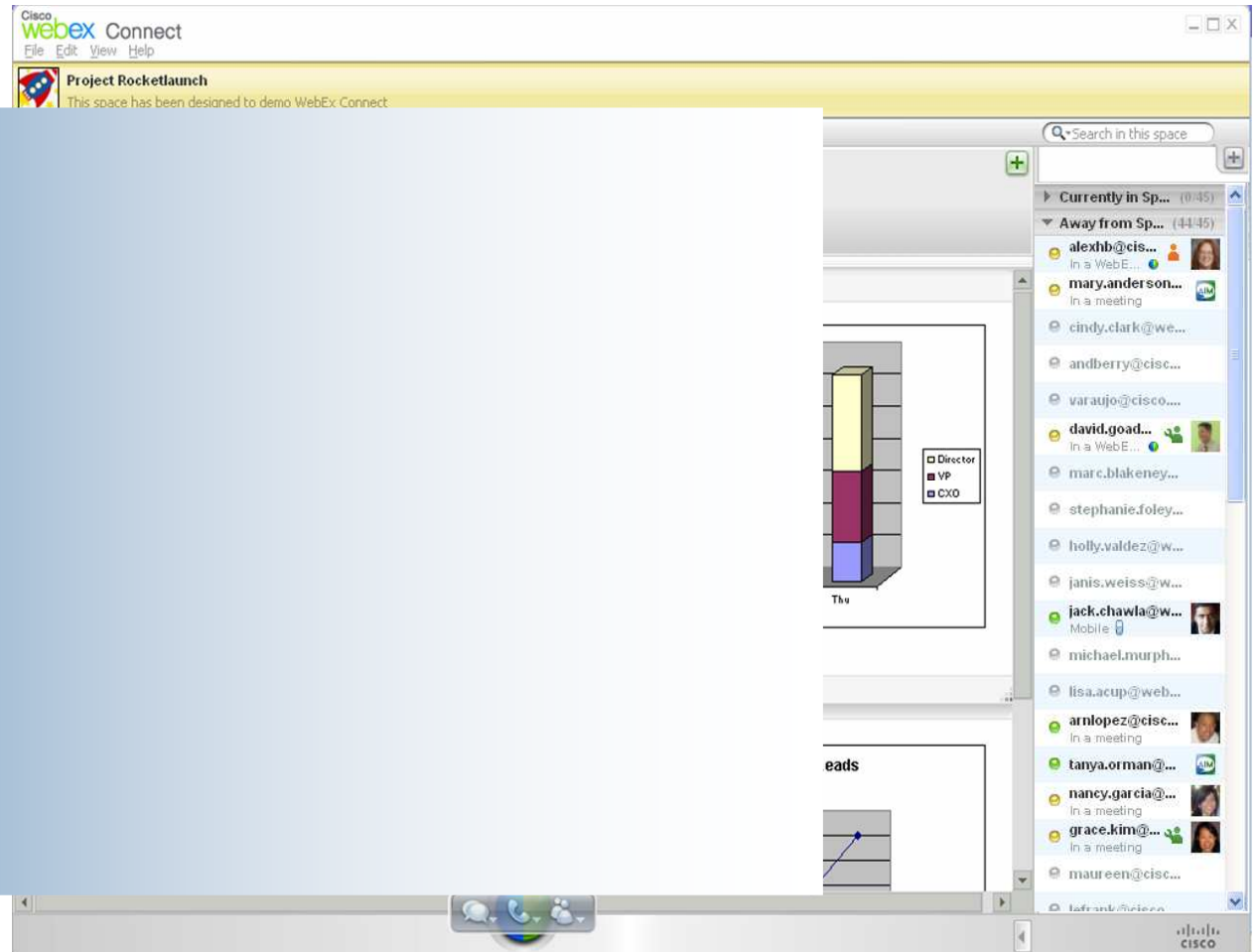
Shared Workspaces

Voice

Video

Web Meetings

Team Spaces

Framework to Create Custom, Collaborative Workspaces With Group Discussions, Persistent Chat, Document Sharing, and Widgets
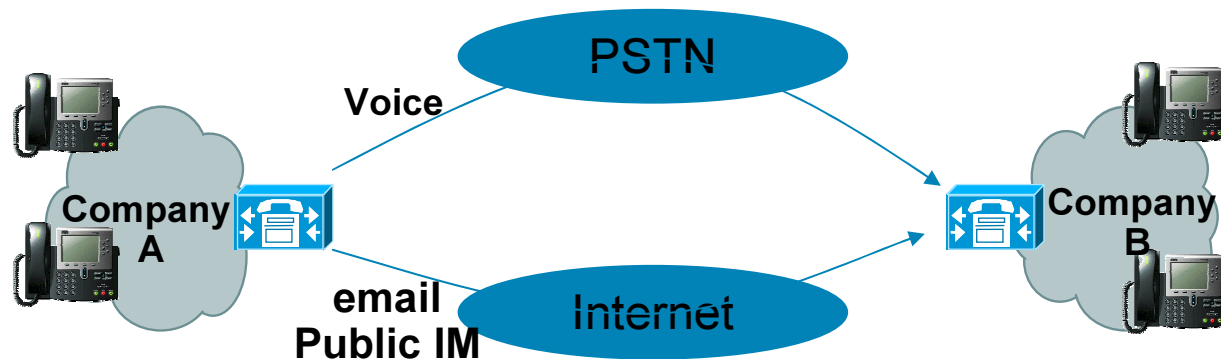
Widgets

Many Business Widgets like Calendar, Task, Bulletin Board, etc. for you to Customize each Team Space

Future

Today's Story on Business to Business

1. Voice is entirely over the PSTN – basic functionality and features (caller ID, basic narrowband voice)

2. Email between enterprises

3. IM, video entirely through public providers (Yahoo, Skype, Google) when it exists

4. The power of Target's robust IP network is not leveraged outside the company

Cisco's Vision and Strategy for the Future

1. Voice, video, IM, email and group collaboration all over IP

2. Connect to any other business in the world using existing phone numbers, phones and even call control infrastructure

3. Secure media and signaling across the Internet

4. Quality guaranteed through PSTN fallback

5. Easy to use – no configuration of dial plans – just like email

6. Huge cost savings by bypassing PSTN

7. Enterprise communications feature set transparently bridged between enterprises – wideband, video, etc.

Geo Location
"Find the Expert" Application

Airport example:

1. Access application via XML phone service on desktop IP phone

2. Enter "target location" (or let network derive it)—e.g., "gate 70"

3. Display map of "experts" (carrying 802.11 wireless phones)

4. Allow user to select one from the map or click "Closest"

5. Extend call to that expert's wireless phone