



# Evolution of SIP

Second JANET Voice Briefing

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# Session Initiation Protocol (SIP)

- I normally say something like this...



- SIP:
  - Defined in RFC 2543
    - Further clarified in RFC 3261
  - Lightweight signalling protocol, can signal
    - Video
    - Voice
    - Instant messaging
  - Bearer for SDP (session description protocol)
  - Audio typically carried over RTP
  - Similar (in messages) to http (request-based)



- But then go onto talk about voice only, and a lot of open source systems
- So is SIP still a niche protocol, only used by systems because it is open and free?



# Increased adoption of SIP

- Previously, telecoms manufacturers stuck to proprietary protocols to provide a rich feature set
- Recently, more telecoms manufacturers adopting SIP as the signalling protocol because of its open architecture
- Now, additional features being added back into the SIP protocol stack to add these additional features but in an open way
- With so many manufacturers all developing extensions to the SIP protocol the trick is to ensure we don't end up with several forks of the same protocol none of which interoperate
  - “ah yes, but which flavour of SIP do you want?”

# Cisco Call Manager 5.0

Find and List Phones - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Address <https://10.89.242.9:8443/ccmadmin/phoneFindList.do>

Navigation CCM Administration

Cisco CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCMAdministrator

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Find and List Phones Related Links: [CAPF Report in File](#)

Status  
8 records found

Search Options  
Find Phone where Device Name begins with Find  Search Within Results  
Select item or enter search text

(device.name begins with any)

Search Results

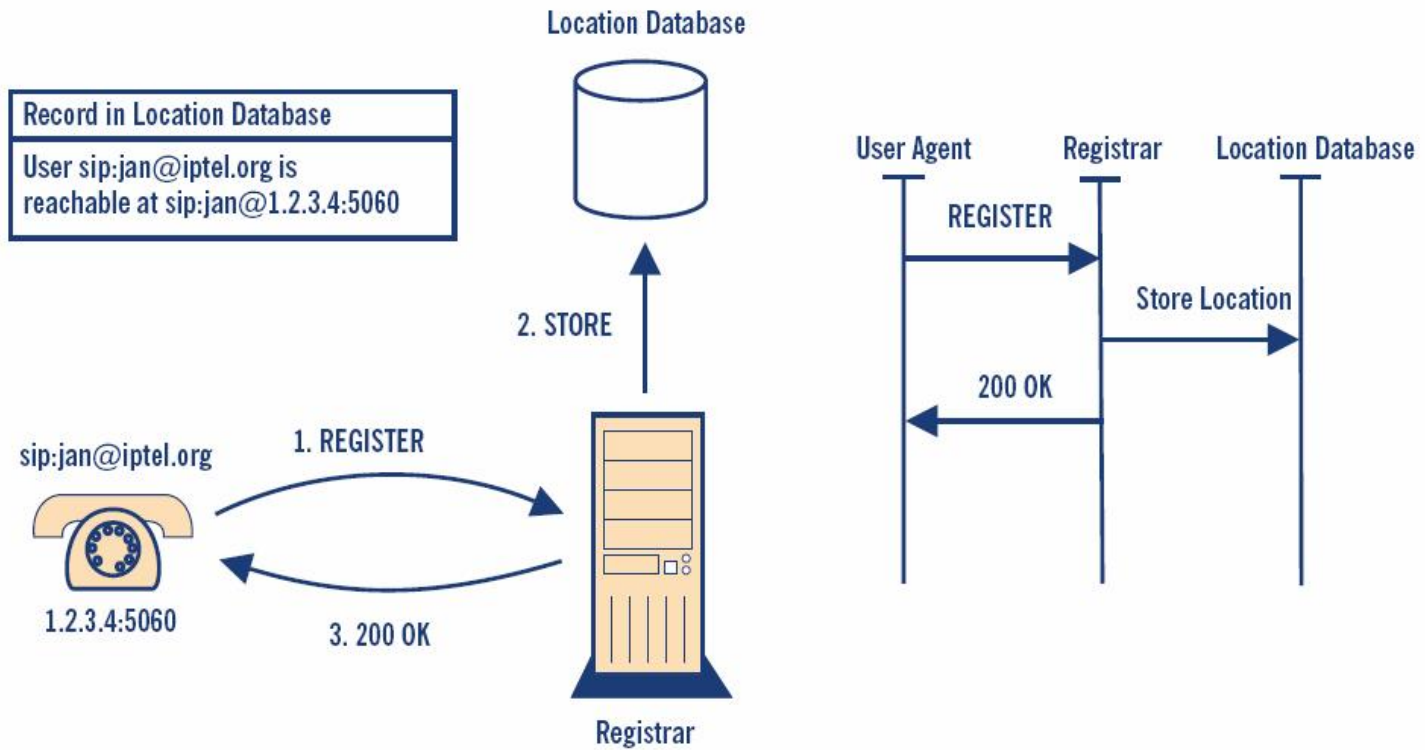
	Device Name(Line)	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Copy w/Lines
<input type="checkbox"/>	<a href="#">SEP00036BE7B991</a>	SEP00036BE7B991	<a href="#">Default</a>	SIP	Registered with 10.89.242.9	<a href="#">10.16.20.11</a>		
<input type="checkbox"/>	<a href="#">SEP0003E32A2AEE</a>	Auto 1000	<a href="#">Default</a>	SCCP	Registered with 10.89.242.9	<a href="#">10.16.20.2</a>		
<input type="checkbox"/>	<a href="#">SEP000F23F9890A</a>	SEP000F23F9890A	<a href="#">Default</a>	SCCP	Registered with 10.89.242.9	<a href="#">10.88.15.217</a>		
<input type="checkbox"/>	<a href="#">SEP00112045E4F7</a>	SEP00112045E4F7	<a href="#">Default</a>	SIP	Unknown	Unknown		
<input type="checkbox"/>	<a href="#">SEP0011215A1B44</a>	SEP0011215A1B44	<a href="#">Default</a>	SCCP	Unknown	Unknown		
<input type="checkbox"/>	<a href="#">SEP003094C3AFF6</a>	SEP003094C3AFF6	<a href="#">Default</a>	SIP	Registered with 10.89.242.9	<a href="#">10.88.15.159</a>		
<input type="checkbox"/>	<a href="#">SEP003094C3BDAC</a>	Auto 1004	<a href="#">Default</a>	SCCP	Registered with 10.89.242.9	<a href="#">10.88.15.223</a>		
<input type="checkbox"/>	<a href="#">SEP003094C3D532</a>	Auto 1002	<a href="#">Default</a>	SCCP	Registered with 10.89.242.9	<a href="#">10.88.15.224</a>		

Add New Select All Clear All Delete Selected Reset Selected Rows per Page 50

UKERNA

Internet

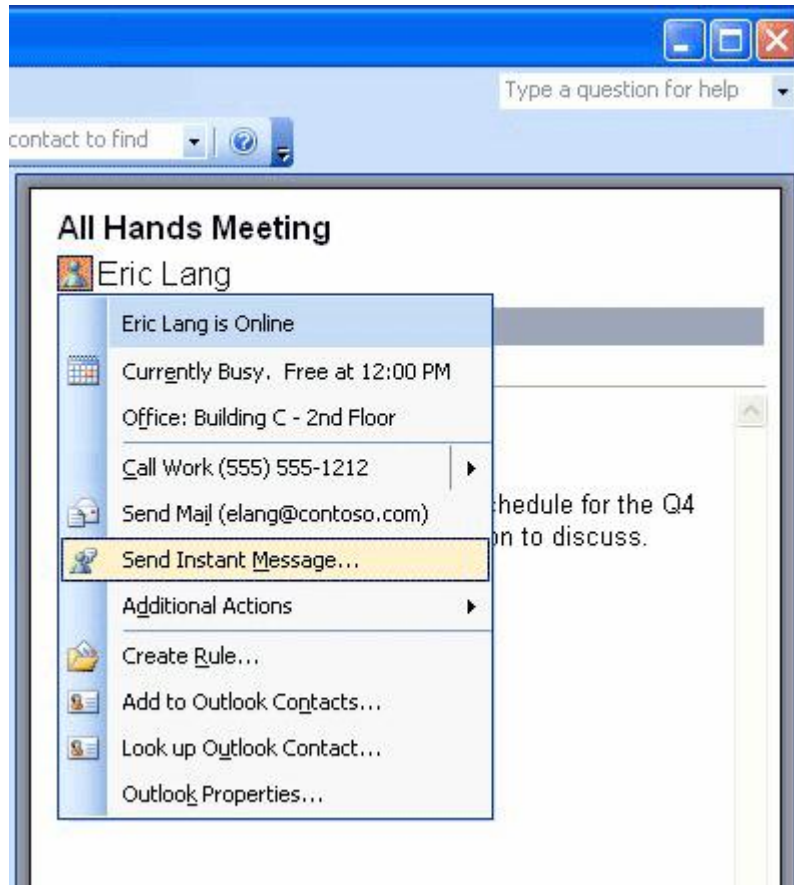
# SIP Registration



# SIP & Presence

- The IETF has created extensions to the SIP protocol to support presence functionality
  - RFC 2778: A Model for Presence and Instant Messaging
  - RFC 2779: Instant Messaging/Presence Protocol Requirements
  - RFC 3261 SIP: Session Initiation Protocol
  - RFC 3856: A Presence Event Package for the Session Initiation Protocol
  - RFC 3859: Common Profile for Presence
- Extensions can use a presence agent (PA)
- The PA accepts and stores information about the extension or target user
  - It can also send notifications to subscribers when a user changes their status
- PA is collocated with the SIP proxy server (can sometimes be standalone)
- Presence User Agent (PUA) queries and updates the PA as to presence status

# SIP Presence



## SIP Presence Object

```
<?xml version="1.0">
<presence entity=\u201dsip:bob@domain.com\u201d
xmlns="urn:ietf:params:xml:ns:pidf"
xmlns:ep="urn:ietf:params:xml:ns:pidf:
status:rpidd-status"
xmlns:et="urn:ietf:params:xml:ns:pidf:
rpidd-tuple">
<tuple id="x765">
<status>
<basic>open</basic>
<ep:activity>meeting</ep:activity>
<ep:placetype>office</ep:placetype>
<ep:privacy>private</ep:privacy>
<et:contact-type>device</et:contact-type>
<ep:idle>2004-06-27T10:43:00Z</ep:idle>
</status>
<contact>sip:bob@bobs-comp.com</contact>
</tuple>
</presence>
```



# SIMPLE

- SIMPLE (Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions) is an instant messaging (IM) and presence protocol suite based on Session Initiation Protocol (SIP)
- SIMPLE applies the SIP protocol to the problems of:
  - Registering for presence information and receiving notifications when such events occur, for example when a user logs-in or comes back from lunch.
  - Sending short messages, analogous to SMS or two-way paging.
  - Managing a session of real-time messages between two or more participants.
- Currently IETF work in progress
  - Some parts already standardised (RFC3428)
  - Others (IM Sessions) still under discussion
  - Many early implementations now appearing (Microsoft LCS/Windows Messenger)
- Not very simple however...
  - RFCs weigh in at 30 documents, more than 1000 pages
    - 7 times more than http 1.1
    - 15 times more than SMTP



# Microsoft LCS

- Live Communication Server
- Integrated into most MS apps
- Supports presence, IM, voice, video, and collaboration in one application.
- “Closed” SIP implementation
  - Ensuring user experience
    - I suspect (read hope) this will change

# Microsoft LCS



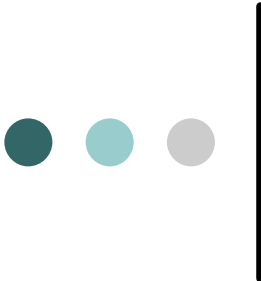
The screenshot displays a Microsoft Office Live Collaboration Suite (LCS) environment. It features several overlapping windows:

- Microsoft Excel - Sales Figures...:** A spreadsheet with columns A, B, C and rows 1, 2. Row 1 contains 'Sales figures' and 'Q1'. Row 2 contains 'Adult Vitamins' and '\$50,324,823.00'. A search bar at the top shows '343534'.
- Microsoft Office Word 2003:** A document titled 'QuarterlySalesReport.doc' with a 'Shared Workspace' pane. The main content includes a 'CONFIDENTIAL' watermark and an 'Executive Summary' section. The text reads: 'This quarter marked the... drugs which resulted in... sales in the first six weeks... Overall, sales were over... over quarter, and even... were more than 18% higher... the same quarter last year'.
- Microsoft Internet Explorer:** A browser window showing a conversation between Maria Hammond and Chris Preston. The participants list includes Anders Madsen, Maria Hammond, and Chris Preston. The chat history shows messages about a proposal.
- Microsoft Outlook:** A window showing a conversation with Craig Dewar. The participants list includes Anders Madsen and Craig Dewar. The chat history shows a message from Anders Madsen and a response from Craig Dewar. A large handwritten 'ok!' is visible in the chat area. The status bar at the bottom indicates 'Last message received on 2/17/2005 at 6:40 PM'.
- Outlook Alert:** A pop-up alert titled 'Alert: Daily summary - 7 results' for Aidan Delaney [aidan@contoso.com]. It lists various actions such as 'Send Mail', 'Send Instant Message', and 'Remove Member from Workspace'.



# SIP Videoconferencing

- H.323 is the dominant protocol in the IP Video Conferencing environment
- But whilst H.323 has many advantages for large, room-based systems. SIP has the potential to make desktop video conferencing even better
  - Presence functionality
  - Instant messaging, then signalling in video or voice as required



# Asterisk & Video Calling

- Asterisk can carry video as well as voice
  - Codecs supported
    - H.261
    - H.263
    - H.263p
    - H264 (in bleeding-edge CVS tree)
- If you are using asterisk voicemail then it will leave video in the message as well



# Asterisk & Video

- A couple of simple additions to the existing configuration

## **sip\_additional.conf**

```
[12501]
type=friend
callerid="Ian Anderson"
host=dynamic
dtmfmode=inband
username=12501
secret=abc123
nat=no
disallow=all
allow=gsm
allow=ulaw
allow=alaw
allow=h263
```

## **sip.conf**

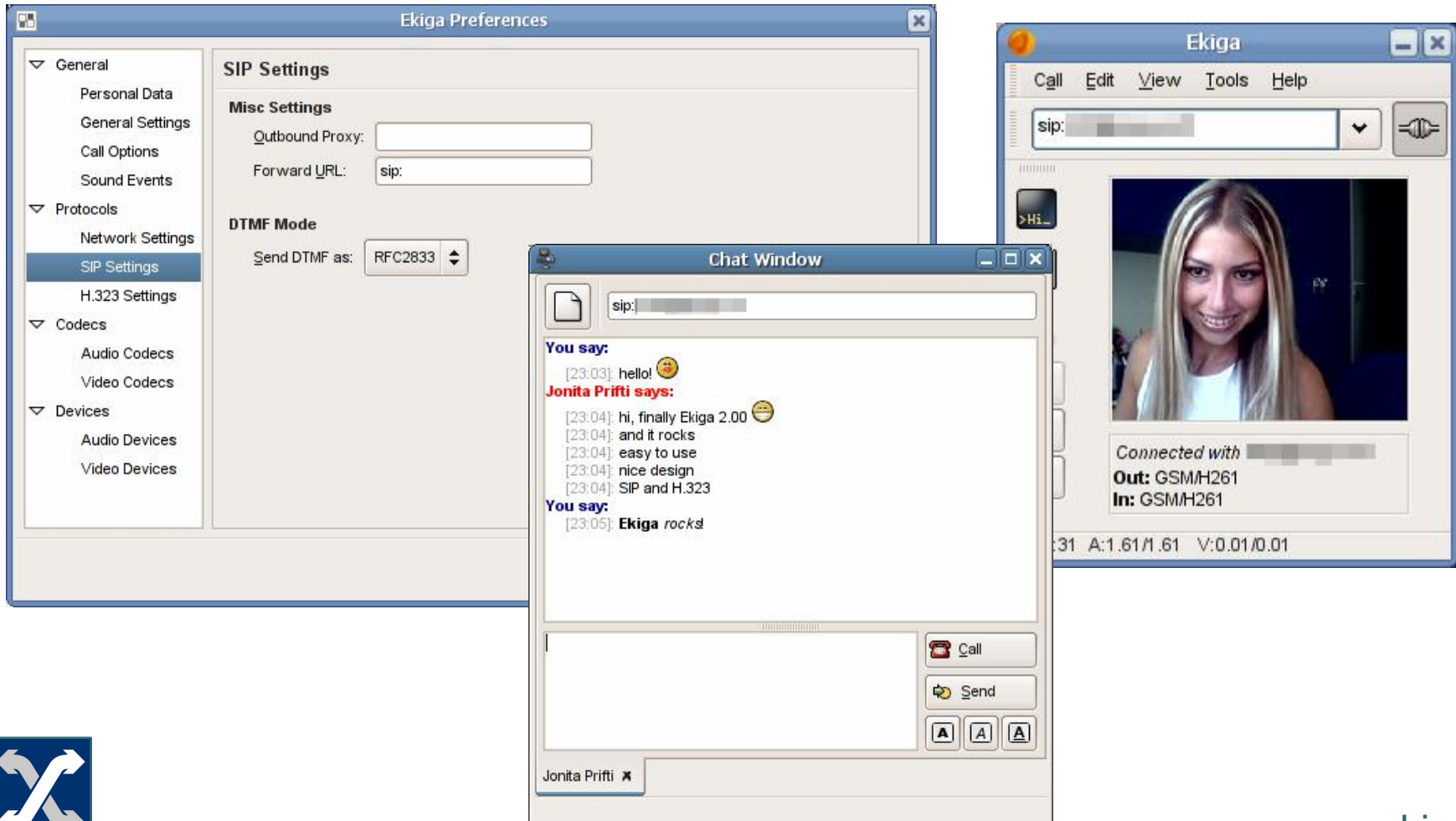
```
[general]
videosupport=yes
```

# SIP Videophones



Needs H264

# SIP Video Softphone (Ekiga)





# Nortel MCS

- SIP-enabled communication solution
  - Video
  - Voice
  - IM
  - Presence
- Is this the nerd-vana of SIP-based communication? (well best screenshot I could find...)
  - Start with an IM session
  - Then start a voice call
  - Launch some video
  - Share some files
    - All from the one interface
    - All signalled using SIP

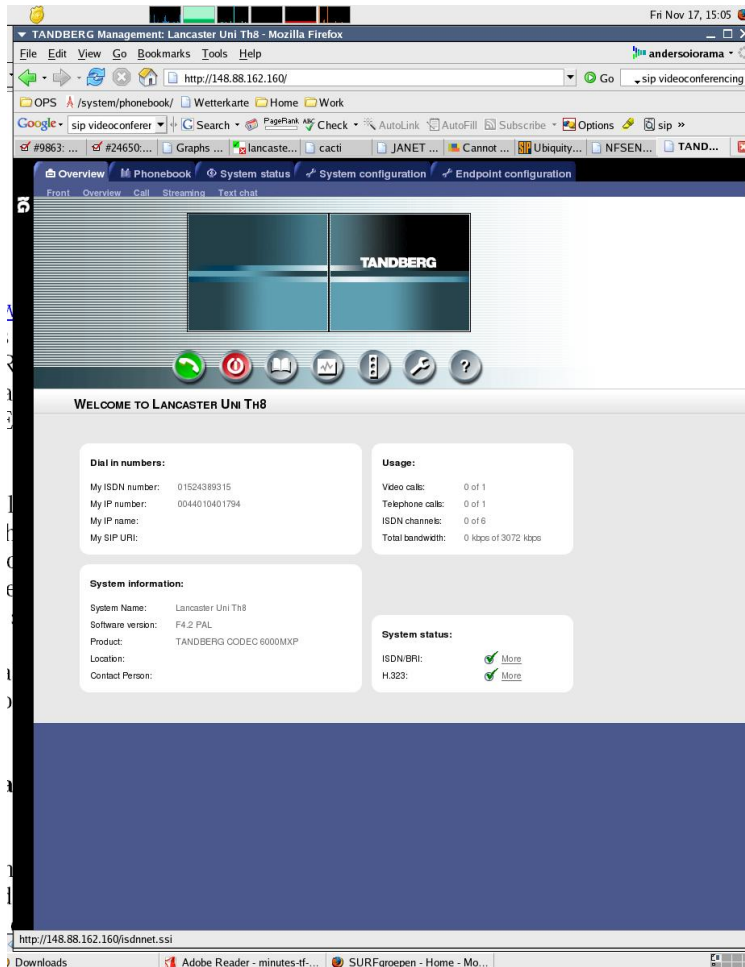


# Nokia E61

- GSM
- GPRS/EDGE/3G
- WiFi
  - SIP Client
  - Call Manager Client
  - Avaya Client
  - Alcatel Client

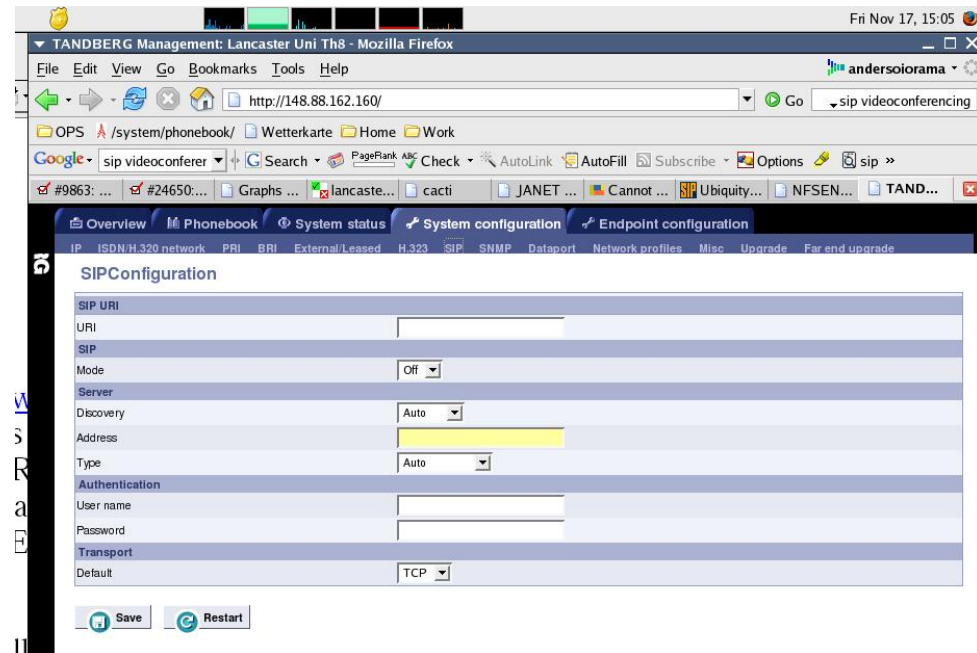


# Tandberg MXP



The screenshot shows the Tandberg MXP management interface in a Mozilla Firefox browser. The page title is "TANDBERG Management: Lancaster Uni Th8". The main content area displays a "WELCOME TO LANCASTER UNI TH8" message and several information boxes:

- Dial in numbers:**
  - My ISDN number: 01524389315
  - My IP number: 0044010401794
  - My IP name:
  - My SIP URI:
- Usage:**
  - Video calls: 0 of 1
  - Telephone calls: 0 of 1
  - ISDN channels: 0 of 6
  - Total bandwidth: 0 kbps of 3072 kbps
- System information:**
  - System Name: Lancaster Uni Th8
  - Software version: F4.2 PAL
  - Product: TANDBERG CODEC 6000MXP
  - Location:
  - Contact Person:
- System status:**
  - ISDN/BRI:  More
  - H.323:  More



The screenshot shows the "SIPConfiguration" page in the Tandberg MXP management interface. The page title is "TANDBERG Management: Lancaster Uni Th8". The configuration fields are as follows:

- SIP URI:**
  - URI:
  - SIP Mode:  Off
  - Server:
  - Discovery:  Auto
  - Address:
  - Type:  Auto
- Authentication:**
  - User name:
  - Password:
- Transport:**
  - Default:  TCP

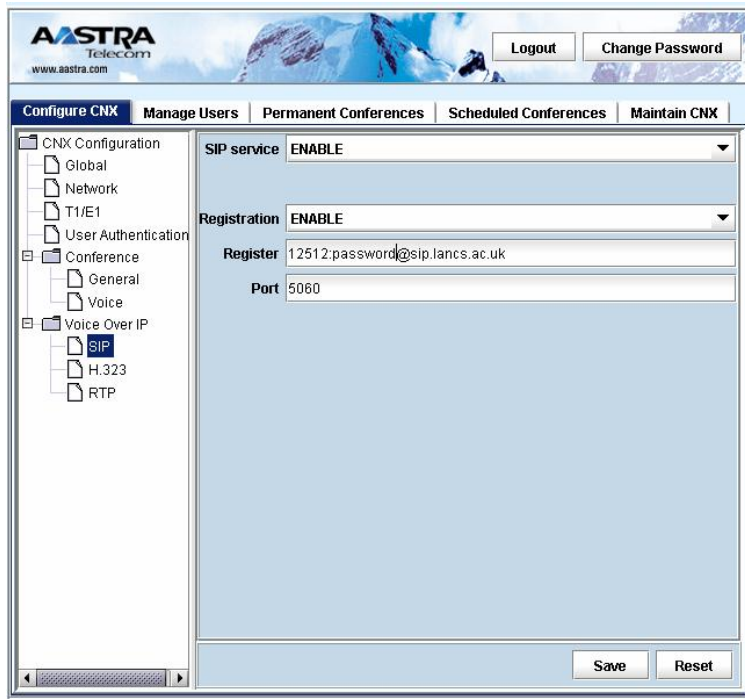
At the bottom of the configuration page, there are "Save" and "Restart" buttons.

- Tandberg MXP Room-based system supports
  - H.323
  - SIP
  - ISDN
  - Cisco Call Manager

# CNX Conference Bridge



- 4 x T1/E1 interfaces for connecting to a PBX or the PSTN
- 2 x 10/100 Ethernet ports for connecting to VoIP networks using SIP or H.323
- Registers to a SIP server as a simple extension

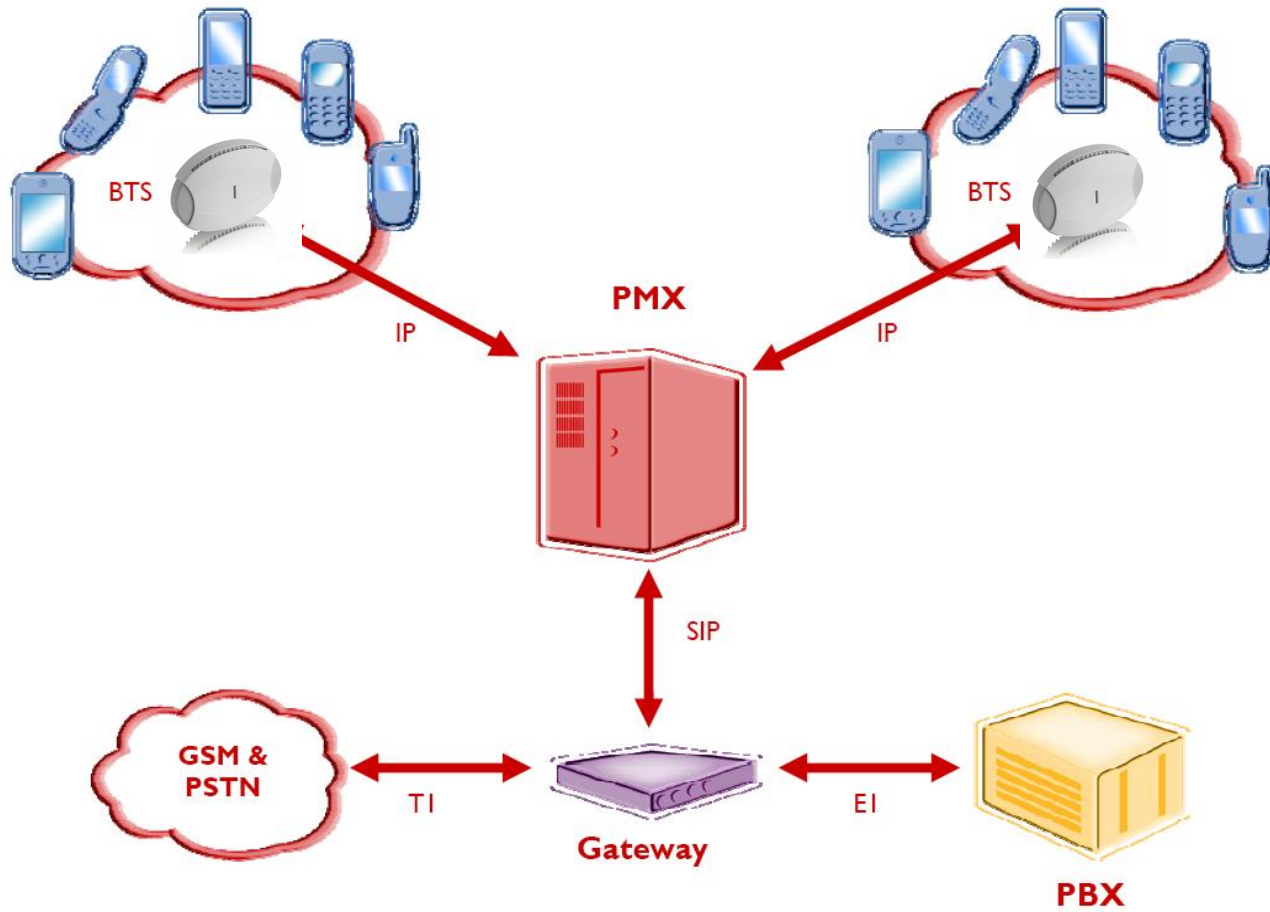


# GSM SIP Integration

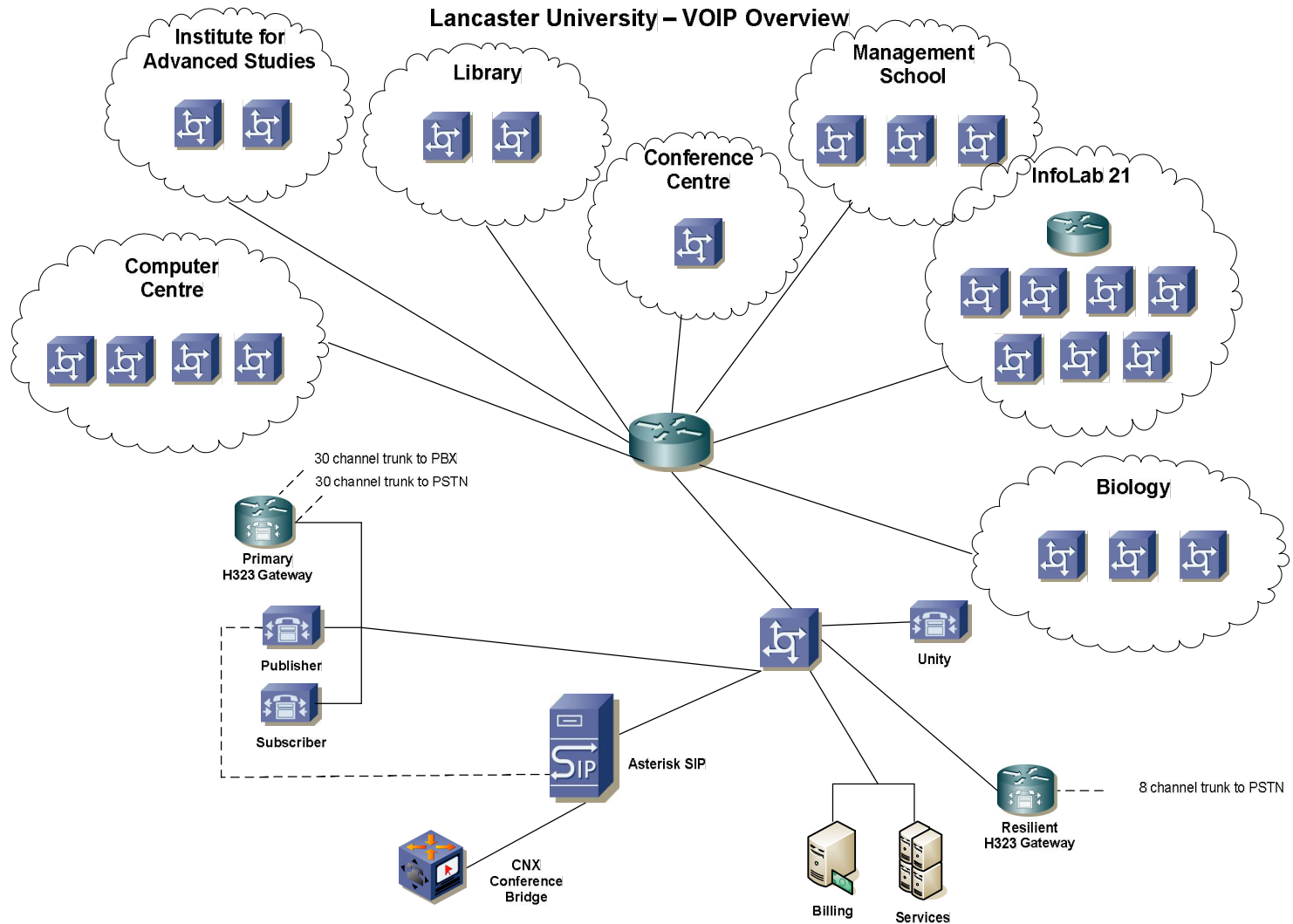
- WiFi phones beginning to appear
- WiFi coverage still not quite ubiquitous
- Handsets still fairly specialist and expensive
- How about a method of trunking between the GSM and IP worlds?
- GSM Picocells becoming smaller and cheaper
- Use IP/SIP at the back-end to link these together



# GSM SIP Integration



# Lancaster SIP Evolution





# Questions?