

# Multimedia Conferencing

A cura di:

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Corso di ***Applicazioni Telematiche***  
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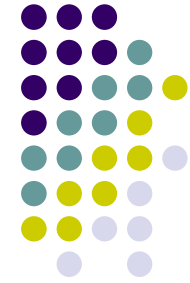
Università degli Studi di Napoli Federico II  
Facoltà di Ingegneria

# Roadmap



- Part I:
  - History, background and state of the art
    - Conferencing as a service
    - Standardization approaches
    - Related topics
      - Media control
- Part II:
  - Hands-on conferencing
    - Ongoing activities at the University of Naples
      - *CONFIANCE* & *DCON* projects
    - Contribution to standards
    - Implementation efforts
    - Open issues

# Conference

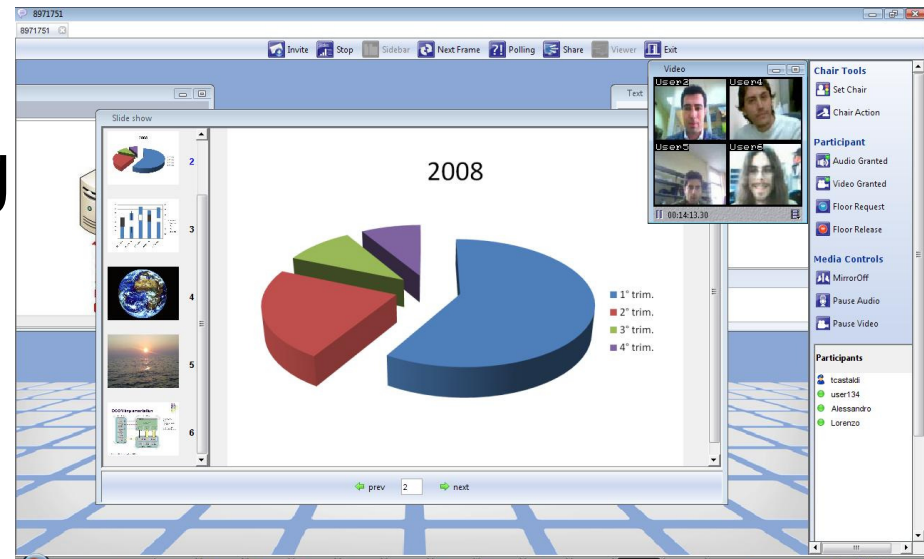
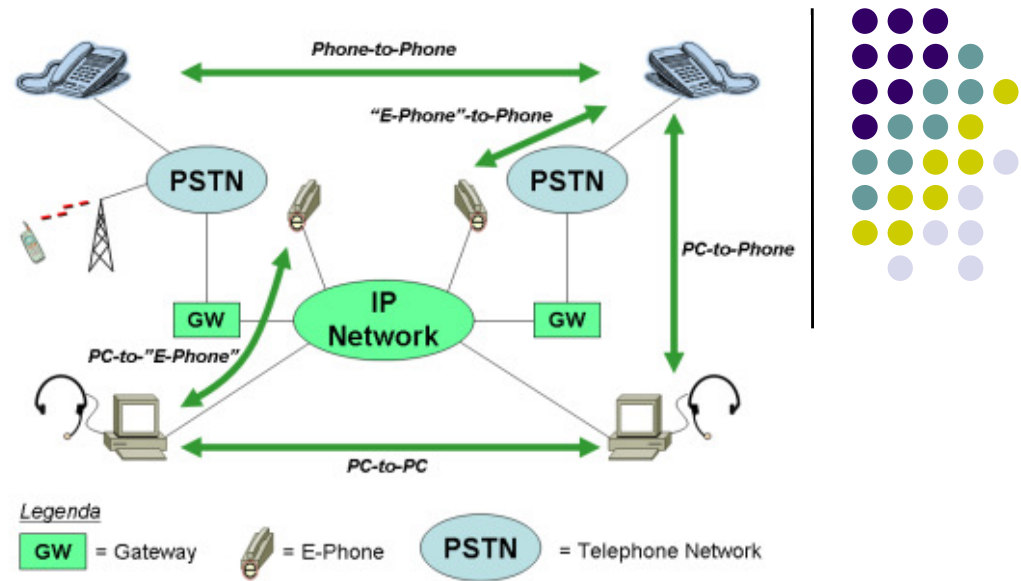


- The term “Conference” can be used to describe any meeting of people that “confer” about a certain topic.
- Web Conferencing is used to conduct live meetings or presentations over the Internet.



# Features

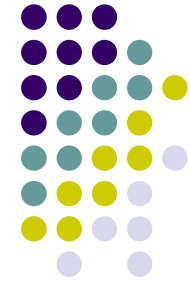
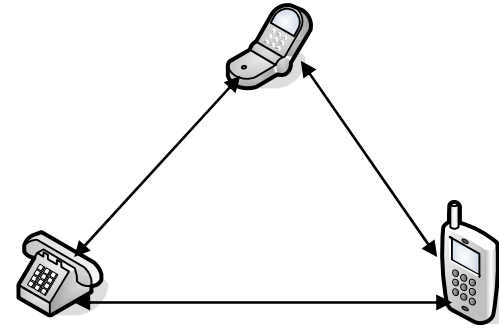
- Voice over IP
- Live video
- Text chat
- Slide presentations
- Whiteboard with annotation
- Screen/desktop sharing
- Application sharing
- Recording
- Polls and surveys



# History

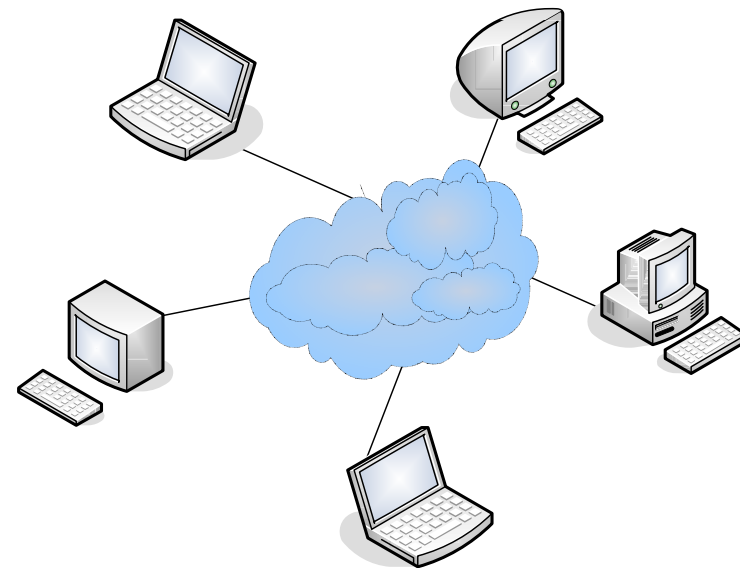
- Tele-Conferencing

- Conference calls (Audio Tele-Conferencing)
- Video conferences (Video Tele-Conferencing)



- Web Conferencing

- Text Conferencing
- Audio/Video Conferencing
- Data Conferencing

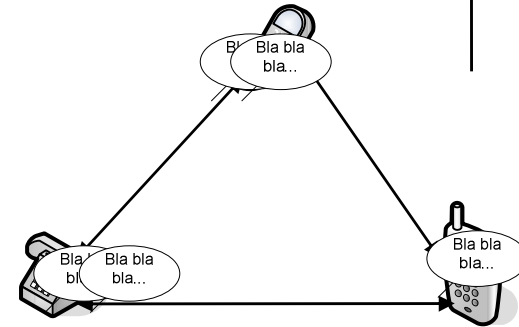


# Audio Tele-Conferencing (ATC)



- Analog Phone Lines (PSTN)

- Conference calls
  - Three-way calling
  - Conference bridges



- Digital Telephony (ISDN)

- ITU-T H.320 umbrella recommendation

- IP-based Tele-Conferencing

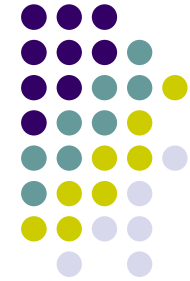
- Real-time Transfer Protocol (RTP)
- Voice over IP (VoIP)

# Video Tele-Conferencing (VTC)

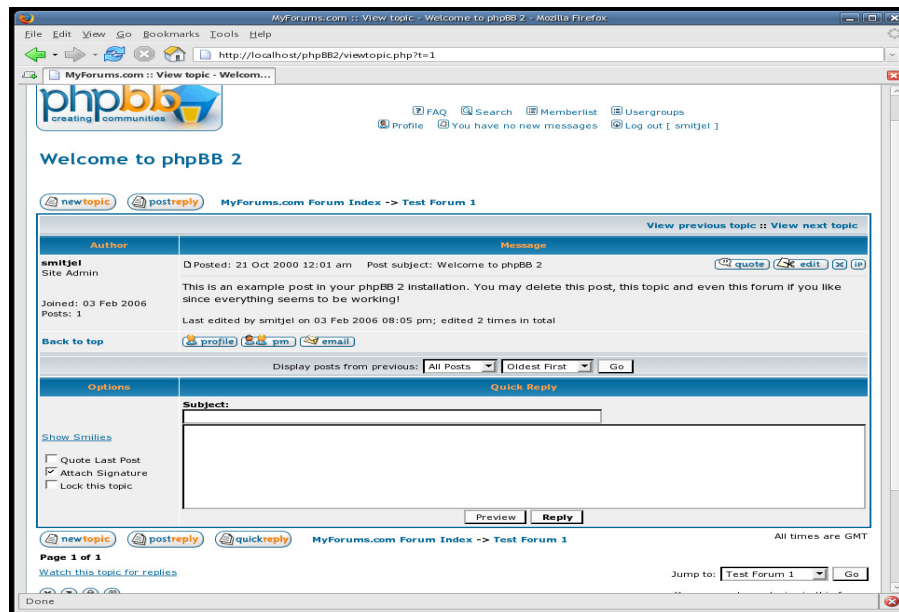


- Closed-circuit television systems
- Radiofrequency (UHF or VHF) links
- Mobile links to satellites
- Analog phone lines (PSTN)
  - Videotelephony (AT&T PicturePhone)
- Digital Telephony (ISDN)
  - ITU-T H.320 Umbrella Recommendation
  - Multipoint Videoconferencing (MCU)
- IP-based Videoconferencing
  - Better video-compressing technologies

# Text Conferencing



- Asynchronous Meetings
  - Posted text messages (not live)
    - Message/Bulletin Boards
    - Fora/Forums
    - Network news groups/Mailing lists

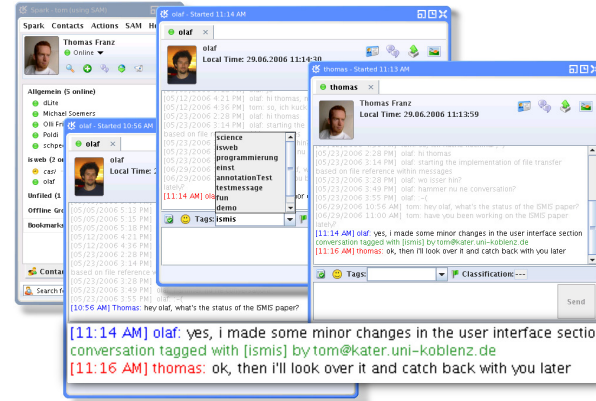




# Text Conferencing



```
mIRC - [#koleje [14] [-nrt]: Kanal zostal poddany reanimacji, zyje i zyc bedzie! :]
File Tools DCC Commands Window Help
wroclaw.ko.pl... #koleje
@Q_EN57_ ja nie nan @G1oon_
<+o1o_2002> ja tez nie nan @Hys2or
<mpau> a o1o nasz? @P1oUarOFF
<mpau> szkoda @Q_EN57_
@Q_EN57_ zapytaj na pmk +AFra
<mpau> (en, daj woica :P) +cysio
<+o1o_2002> ale jak wan zaley na zielonej E006 +Kovalek
<mpau> no, chyba tak zrobie... +Harcin
@Q_EN57_ > aa, fakt, zapomnialen +mpau
<+o1o_2002> to moze pomoc +o1o_2002
<mpau> no MCR[g]
*** _EN57_ sets mode: +u mpau piotr[d]aw
<+mpau> chodzi mi o zielone czolo glownie Sec1o
<+mpau> EN: dzieki sk-uebn
@Q_EN57_ o1o: 1) jak mozesz pomoc? 2) jechales do pracy
N1R czy z N1R?
<+o1o_2002> no taki elektrowoz to jezdzi co drugi dzien
przez ilawe do olsztyna i krakowa
<+o1o_2002> z zielonym czolem
<+mpau> ale do Szczecina :P
<+o1o_2002> nie pomalowanym na zolto
<+mpau> I need info about green czolo to Szczecin !!!!!
*** cysio[jaw is now known as cysio
```

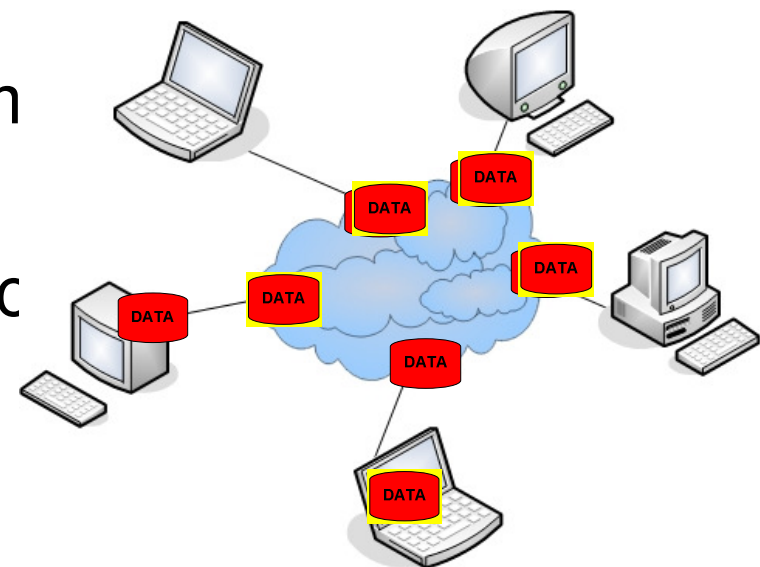


- Synchronous (Live) Meetings
  - Live text communication
    - talk/ntalk/ytalk (Unix)
    - Internet Relay Chat (IRC)
    - Web-based Chat (CGI/Java)
    - Instant Messaging (Skype/MSN/ICQ/XMPP/SIMPLE/etc.)

# Data Conferencing

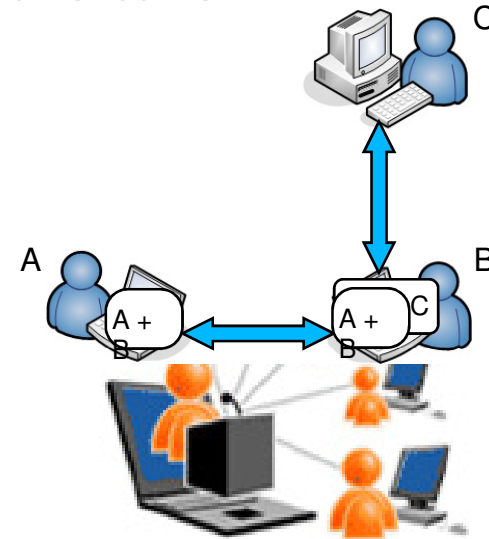


- Participants sharing computer data in real time
  - Text (Instant Messaging)
  - Audio/Video
  - Screen/Documents/Graphics/Applications
- Desktop Systems
  - Placeware/ProShare/Databeam
  - Netmeeting/Gnomemeeting
  - Skype/AIM/ICQ/MSN/Yahoo/etc



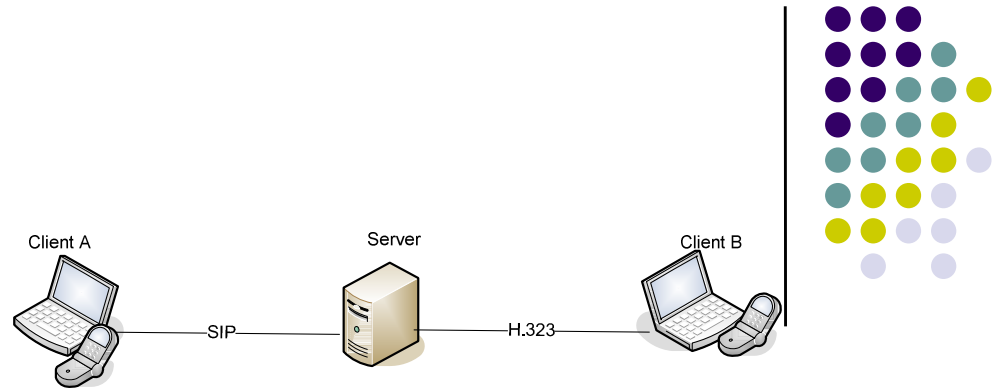
# Typical Scenarios

- Point-to-Point Calls to Multipoint Calls
  - Three-way calling
  - Coaching scenario
- Lecture-mode Conferences
  - Presentation
  - Question & Answers session
- Ad-hoc and Reserved Conferences
  - Conference-aware/-unaware participants
    - Manage conference/users/media/policies
    - Sidebars/Whispers

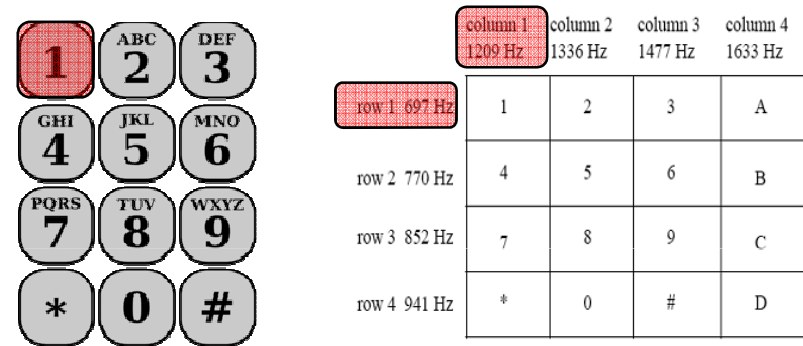


# Issues

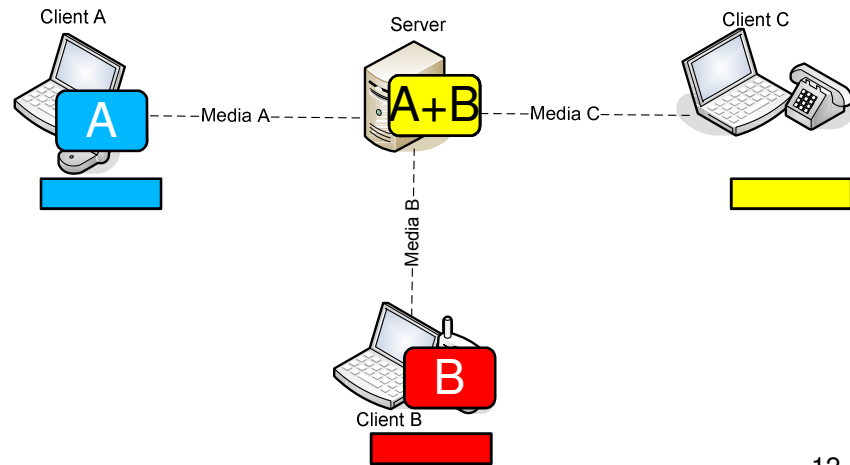
- Call Signaling
  - Gateway functionality



- Control and Management
  - Tone detection (DTMF)
  - Dedicated protocols



- Mixing and Transcoding
  - Terminal capabilities
  - User media profiling
    - Coaching scenario
    - Videoswitching



# Standardization Efforts



- No standardization for many years
  - Lack of interoperability
  - Platform dependency
  - Security issues
  - Cost
  - Market segmentation
- Standardization Bodies
  - ITU (International Telecommunication Union)
  - IETF (Internet Engineering Task Force)
  - 3GPP (3rd Generation Partnership Project)

# Standardization Efforts: ITU



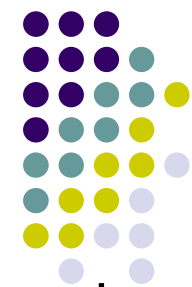
- Established to standardize and regulate international radio and telecommunications
- International Standards referred to as Recommendations”
- ITU-T: Telecommunication Sector
  - G: Transmission Systems and Media
    - G.71x (Audio compression, mu-law and a-law)
    - G.72x (Audio compression, ADPCM)
  - H: Audiovisual and Multimedia Systems
    - H.320 (PSTN/ISDN, Telephone Systems)
    - H.323 (IP, Packet-based Communication Systems)
  - T: Terminals for Telematic Services
    - T.120 (Data Sharing Protocols)
    - T.140 (RTP Interactive Text)

# Standardization Efforts: IETF



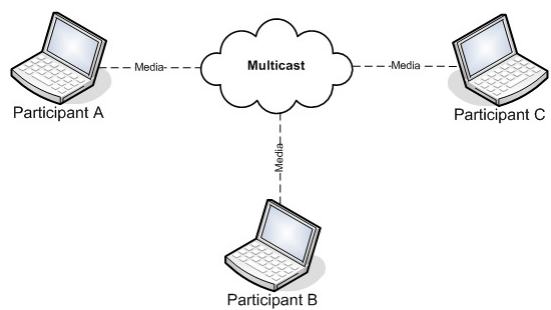
- Under the umbrella of the Internet Society
- Develops and promotes Internet Standards
- Deals in particular with standards of the TCP/IP suite
- Organization
  - Working Groups (WG)
  - Internet Drafts
  - Requests for Comments (RFC)
  - “Rough consensus, running code”

# SIPPING Working Group

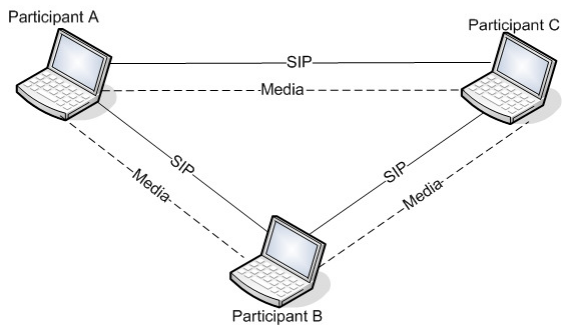


- Session Initiation Proposal Investigation
- Documents the use of SIP for several applications related to telephony and multimedia
- SIP Conferencing

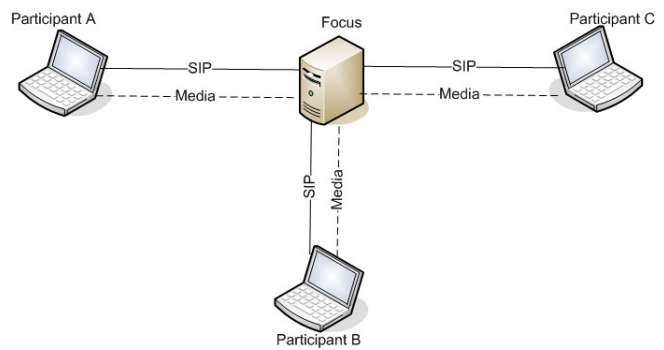
Loosely-Coupled Conference



Fully Distributed Multiparty Conference



Tightly-Coupled Conference



## SIP Conferencing Framework (RFC 4353): fundamental elements

- Focus
- Policy Server
- Mixer
- Notification Service (Event Package, RFC 4575)
- Participants

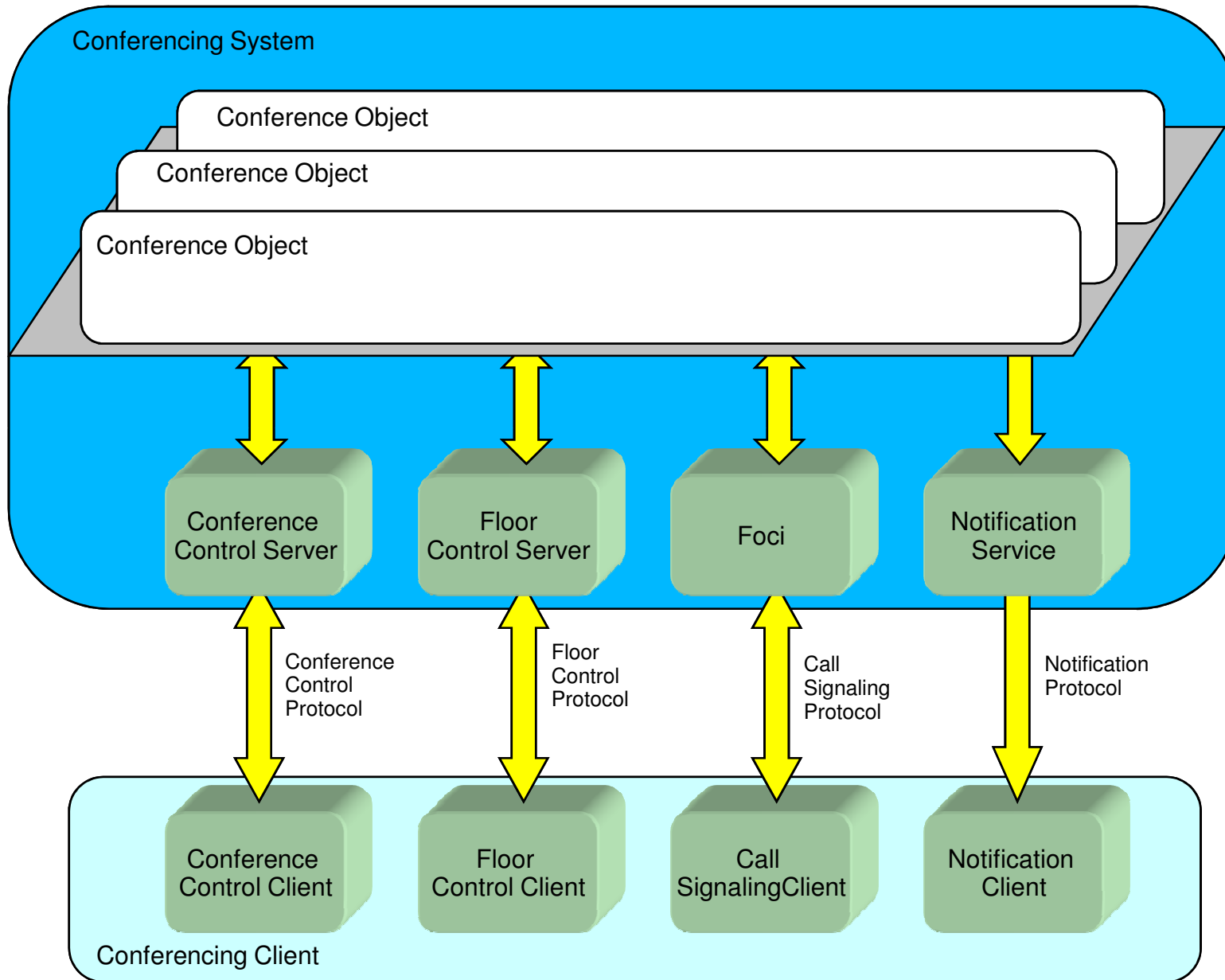


# XCON Working Group

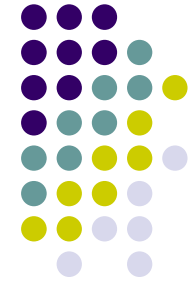


- Centralized Conferencing (XCON)
- Extends RFC 4353
  - Protocol-agnostic (not only SIP)
  - Data Sharing (not only audio/video)
- Suite of Protocols
  - Conference Control (CCMP)
  - Floor Control (BFCP)
  - Call Signaling (SIP/H.323/IAX/etc.)
  - Notification (Event Package?)

# XCON Framework

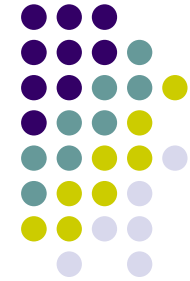


# Conference Control Protocol

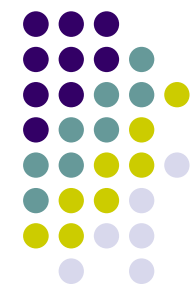


- Create/Manage/Schedule/etc. Conferences
- Several candidates in the past, all rejected
- New proposal
  - Centralized Conferencing Manipulation Protocol (CCMP)
    - State-less client-server protocol
    - Based on a request/response model
      - Uses HTTP as the protocol to transfer messages
- University of Naples (COMICS research group):
  - Highly active in this field
    - Running code 😊 and... eventually rough consensus 😊

# Floor Control Protocol

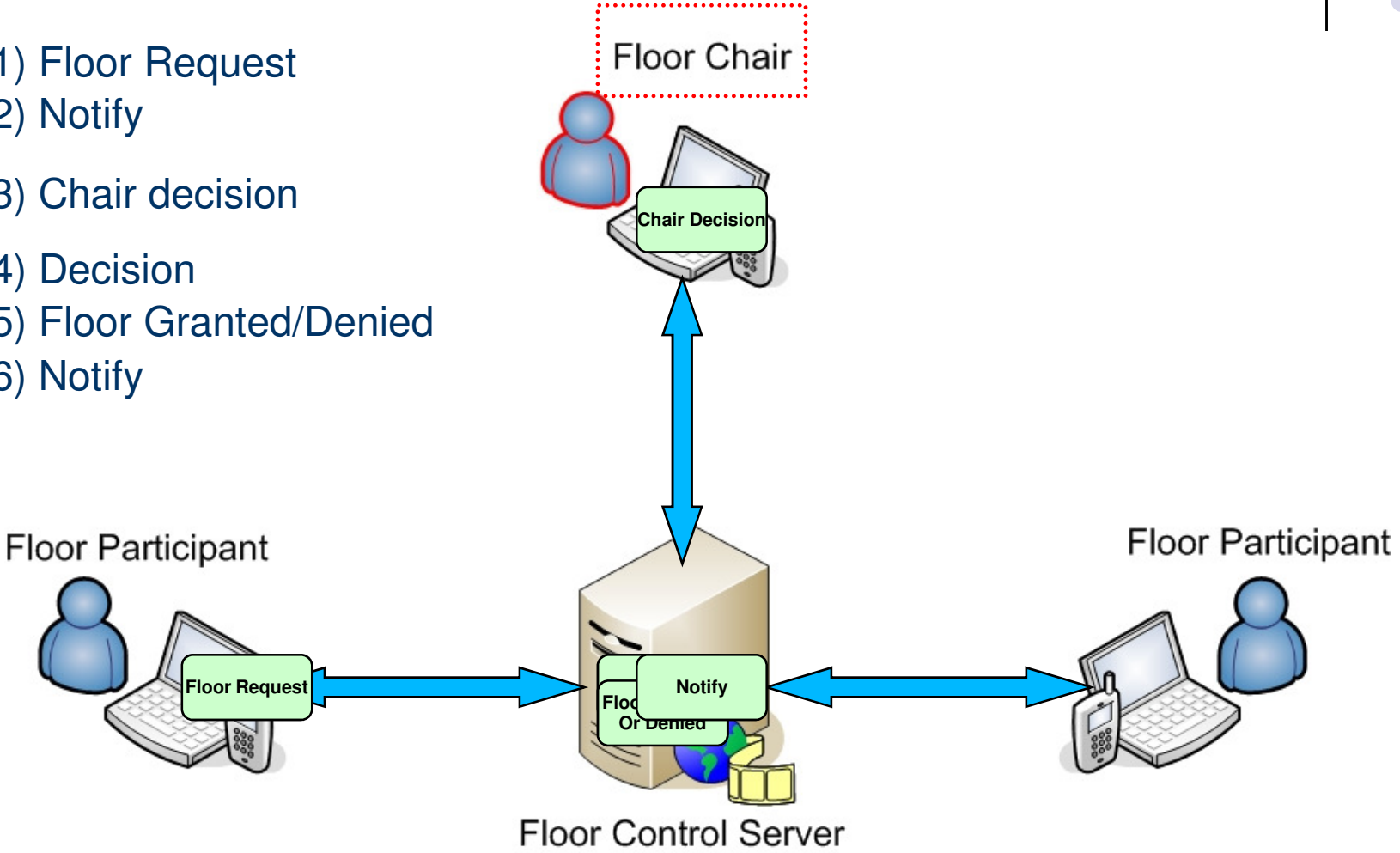


- Coordinates access to set of shared resources
  - A “Floor” is a token, a temporary permission to access or manipulate a specific shared resource or set of resources
- Binary Floor Control Protocol (BFCP)
  - Standardized in RFC 4582
    - Identifiers (Conferences/Floors/Users)
    - Floor Control Server
    - Floor Control Participant
      - Floor Chair
    - Only existing implementation to date: COMICS/Ericsson
  - Negotiation of BFCP connections within SIP/SDP standardized in RFC 4583



# BFCP

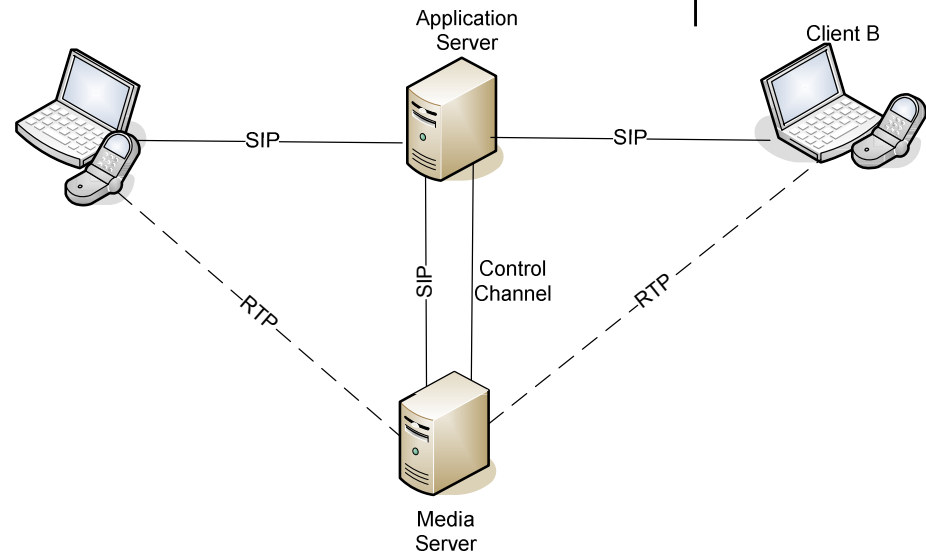
- 1) Floor Request
- 2) Notify
- 3) Chair decision
- 4) Decision
- 5) Floor Granted/Denied
- 6) Notify



# MEDIACTRL Working Group



- Media Server Control
  - Media Processing
    - Mixing/Transcoding
    - Playing/Recording
    - Storing/Retrieving
    - Detecting Tones (DTMF)
    - Interactive Voice Response (IVR)/VoiceXML
    - Text-to-Speech/Speech Recognition
  - RTP Streams Manipulation
- Of great interest to the XCON working group
- MRFC/MRFP (interface/container) in IMS



# CONFIANCE

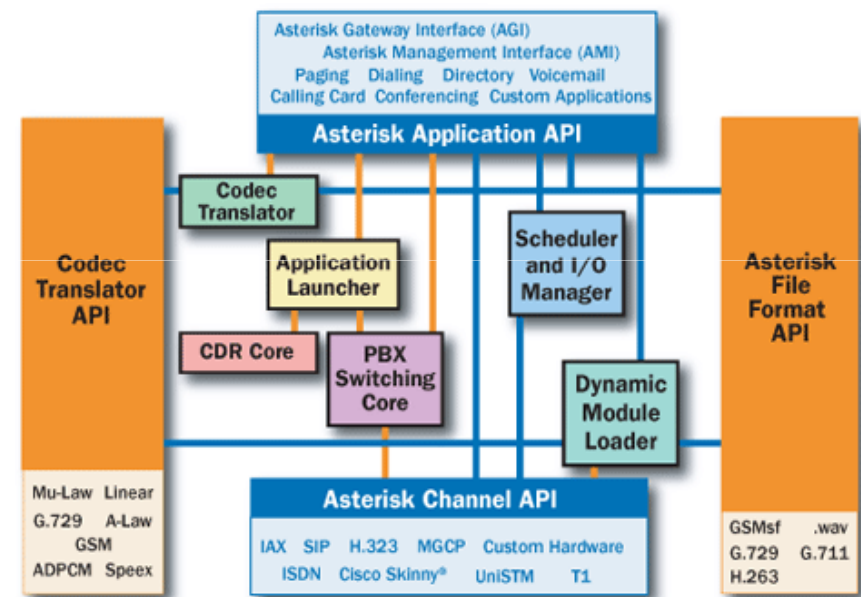


- CONFerencing IMS-enabled Architecture for Next-generation Communication Experience
- Open source prototype implementation of the XCON Framework, compliant with the IMS specification
- Extends the Asterisk PBX functionality
  - Enhanced “MeetMe” application
  - Support for Conference Management (CCMP)
  - Support for Floor Control (BFCP)
    - BFCP-guided audiomixing
    - BFCP-guided videomixing

# Asterisk PBX



- Open source Private Branch eXchange (PBX)
- Advanced features
  - Highly configurable dialplan
  - Modular architecture
    - Channel API
      - SIP channel driver
    - Application API
      - MeetMe conference bridge
    - Codec and File Format API
      - Audio transcoding
      - Video passthrough
  - Remote Manager Interface





# Asterisk dialplan: `extensions.conf`



## Definiton of a single extension with name "123".

```
exten => 123,1,Answer
exten => 123,2,Playback(tt-weasels)
exten => 123,3,Voicemail(44)
exten => 123,4,Hangup
```

When a call is made to extension 123, Asterisk will answer the call itself, play a sound file called "tt-weasels", give the user an opportunity to leave a voicemail message for mailbox 44, and then hangup.

## Extension Patterns

A single extension can also match *patterns*. In the `extensions.conf` file, an extension name is a pattern if it starts with the underscore symbol (`_`).

```
exten => _123.,1,Answer
exten => _123.,2,Playback(tt-weasels)
exten => _123.,3,Voicemail(${EXTEN})
exten => _123.,4,Hangup
```

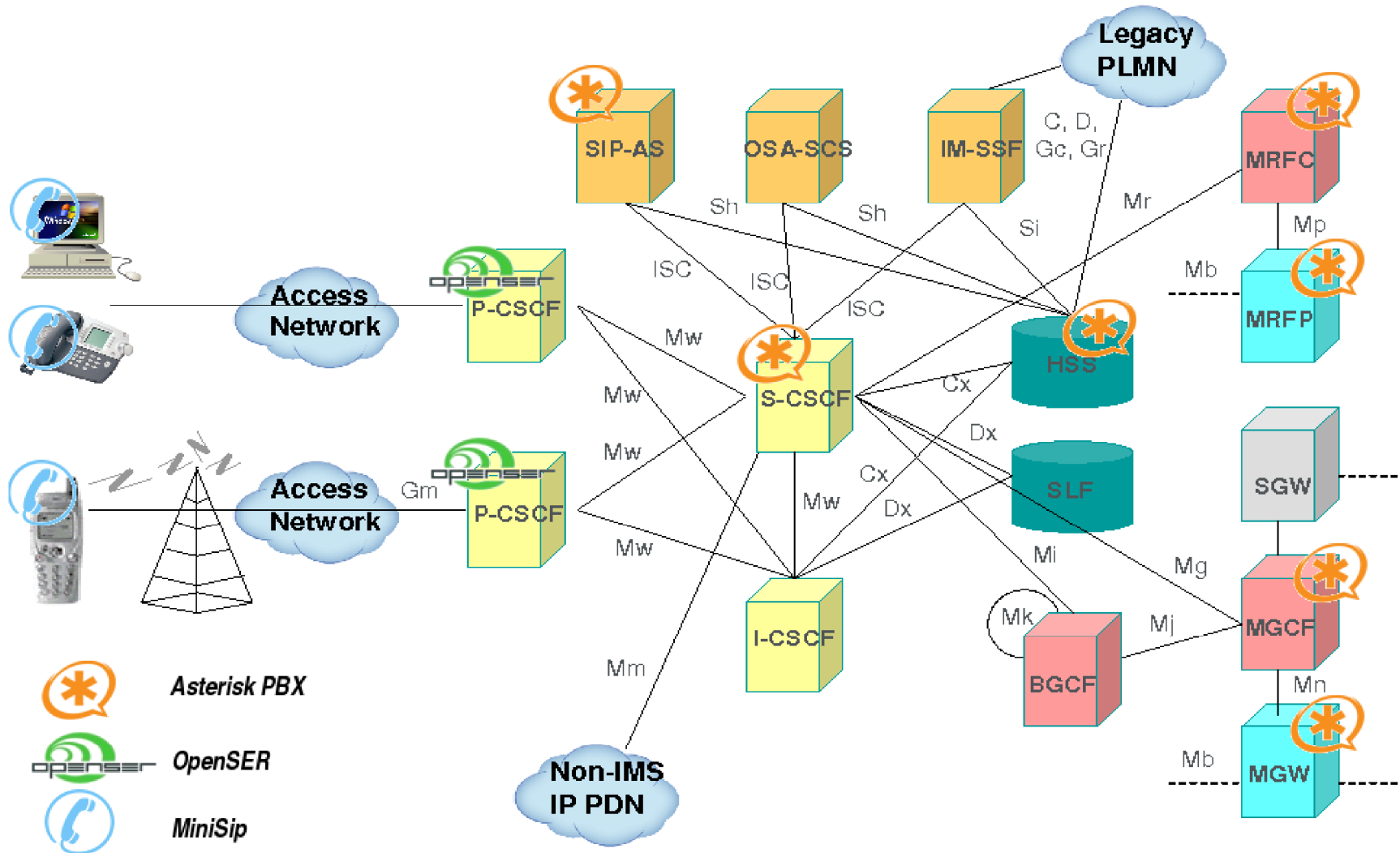
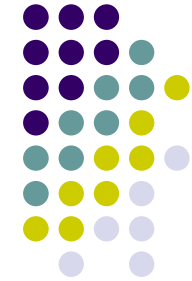
# XCON through MeetMe



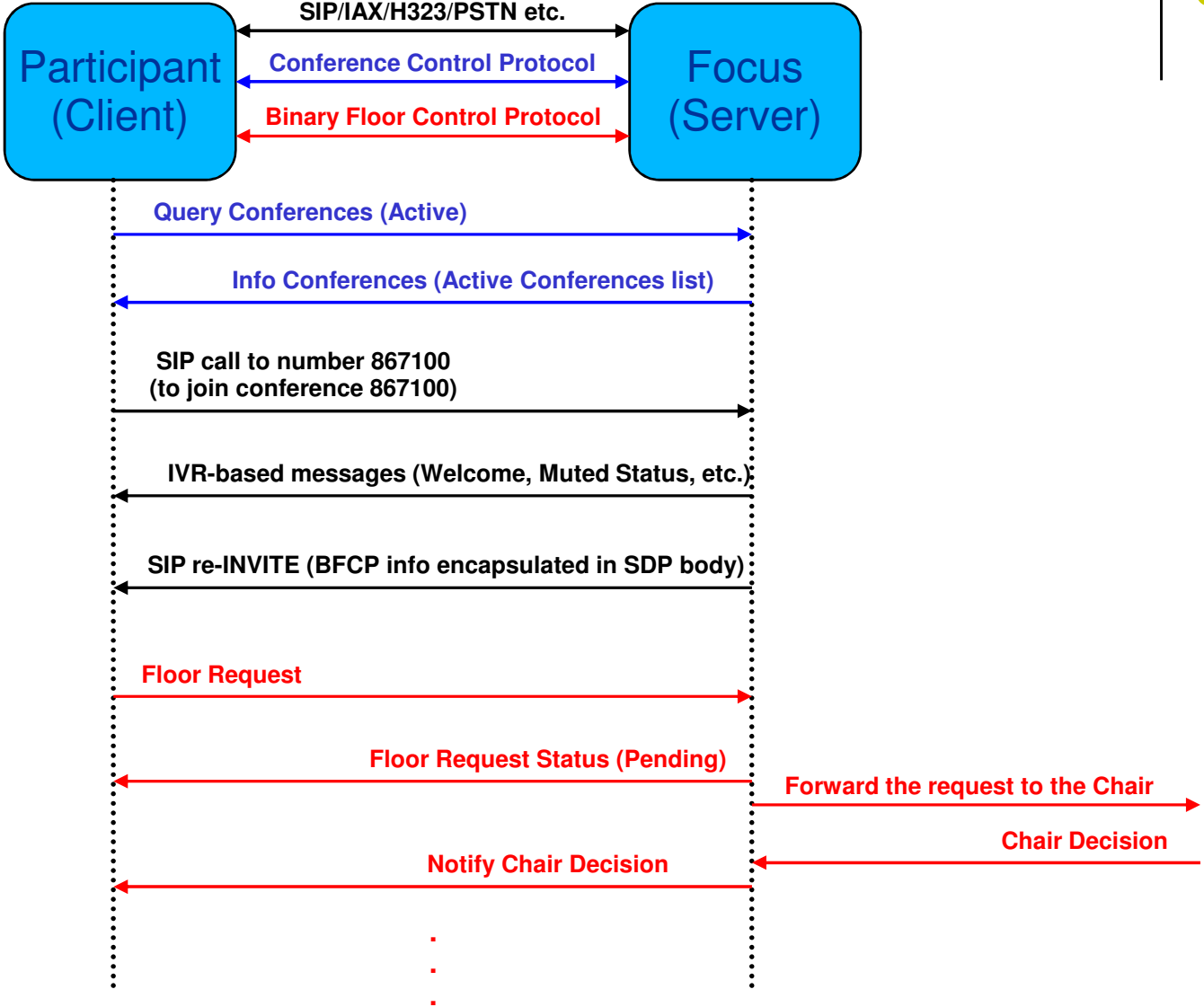
`extensions.conf`

```
[...]  
  
; XCON through MeetMe: example of wildcards to add flexibility  
; - First 7 numbers = conference  
; - Next (1-4) numbers = PIN (Phone PIN, not Admin's password)  
;  
; the 'B' flag tells MeetMe this is an XCON conference (B => BFCP)  
;  
exten => _857.,1,Meetme(${EXTEN:0:7}|B|${EXTEN:7})  
exten => _857.,2,Hangup  
  
[...]
```

# CONFIANCE in IMS



# CONFIANCE Use Case





# Coffee break?

# Distributed Conferencing



- Centralized Conferencing being standardized
  - Poorly scalable
  - Limited capabilities
  - Single point of failure

- Distributed Conferencing

- Cascaded Conferencing

- Each focus is seen as a participant by the others
    - Only affects mixers' distribution
    - Centralized protocols like BFCP don't work

- P2PSIP Working Group

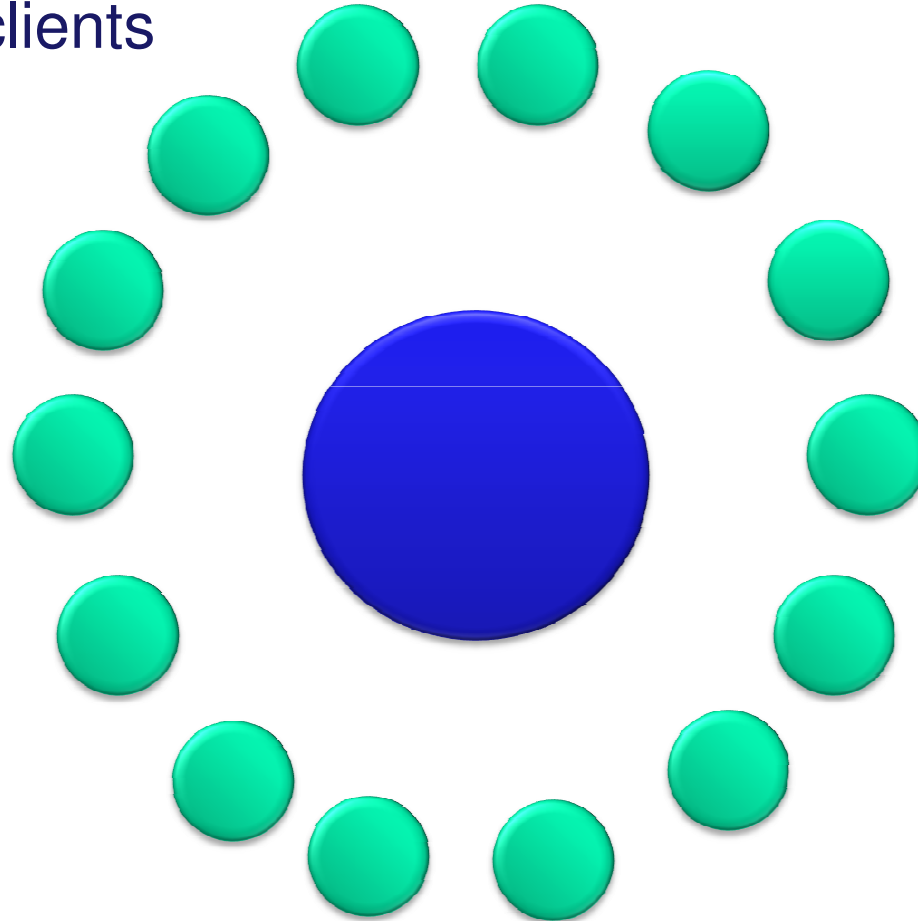
- Has not dealt with conferencing yet





# Centralized conferencing

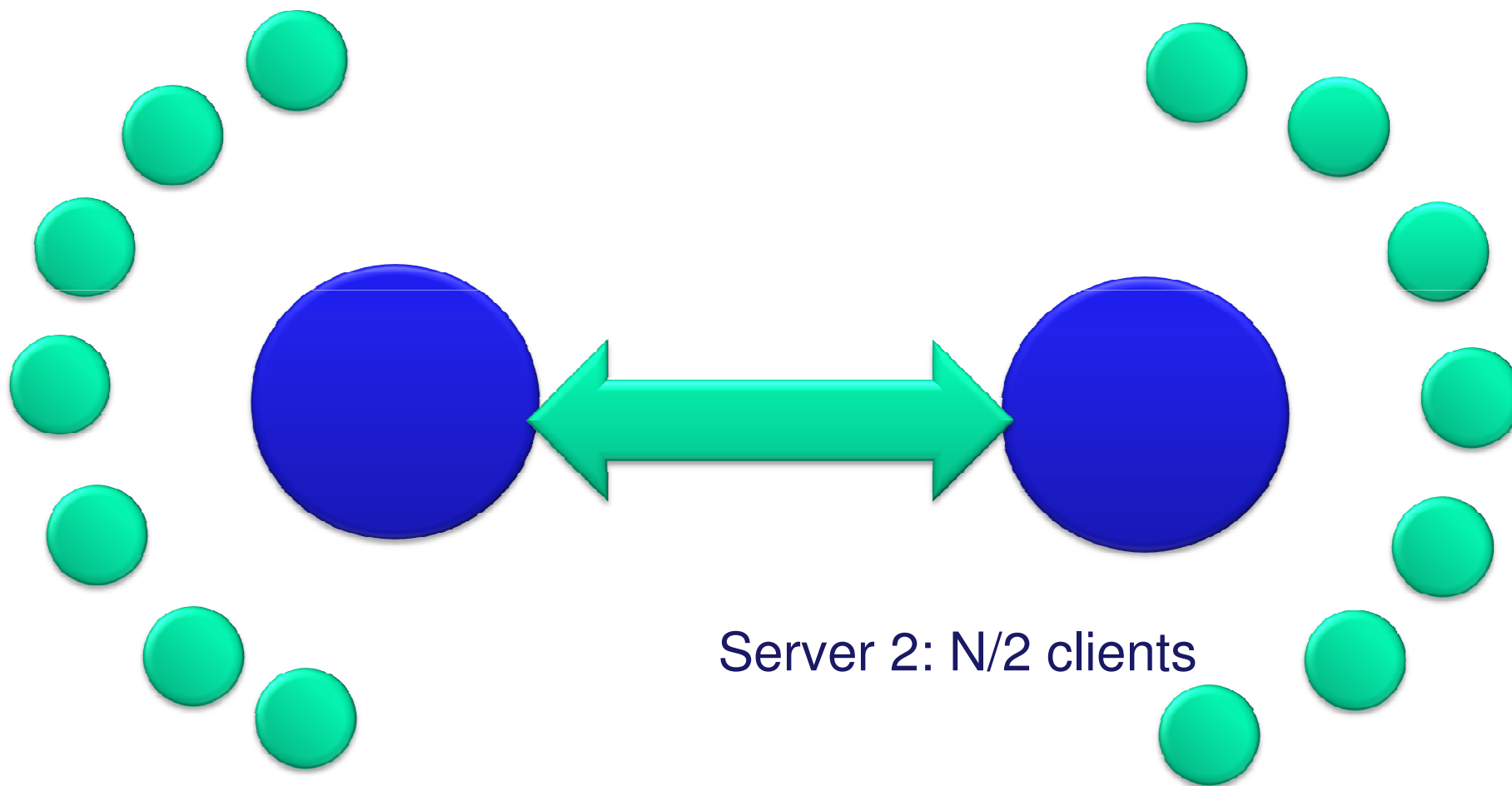
Server 1: N clients





# Cascaded Conferencing

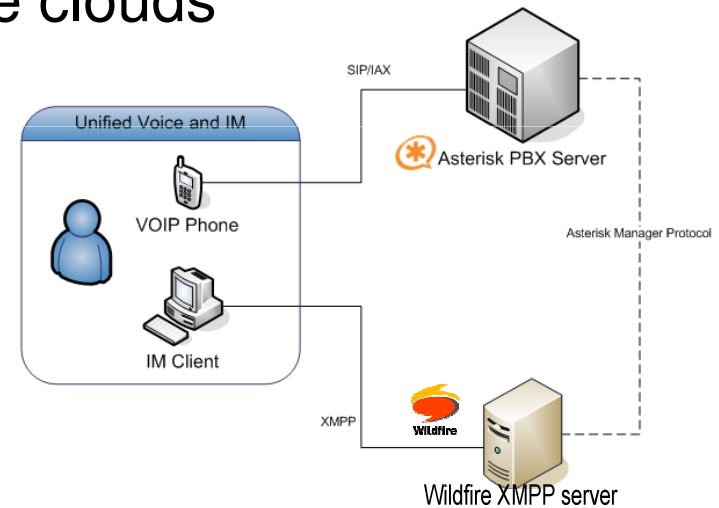
Server 1:  $N/2$  clients



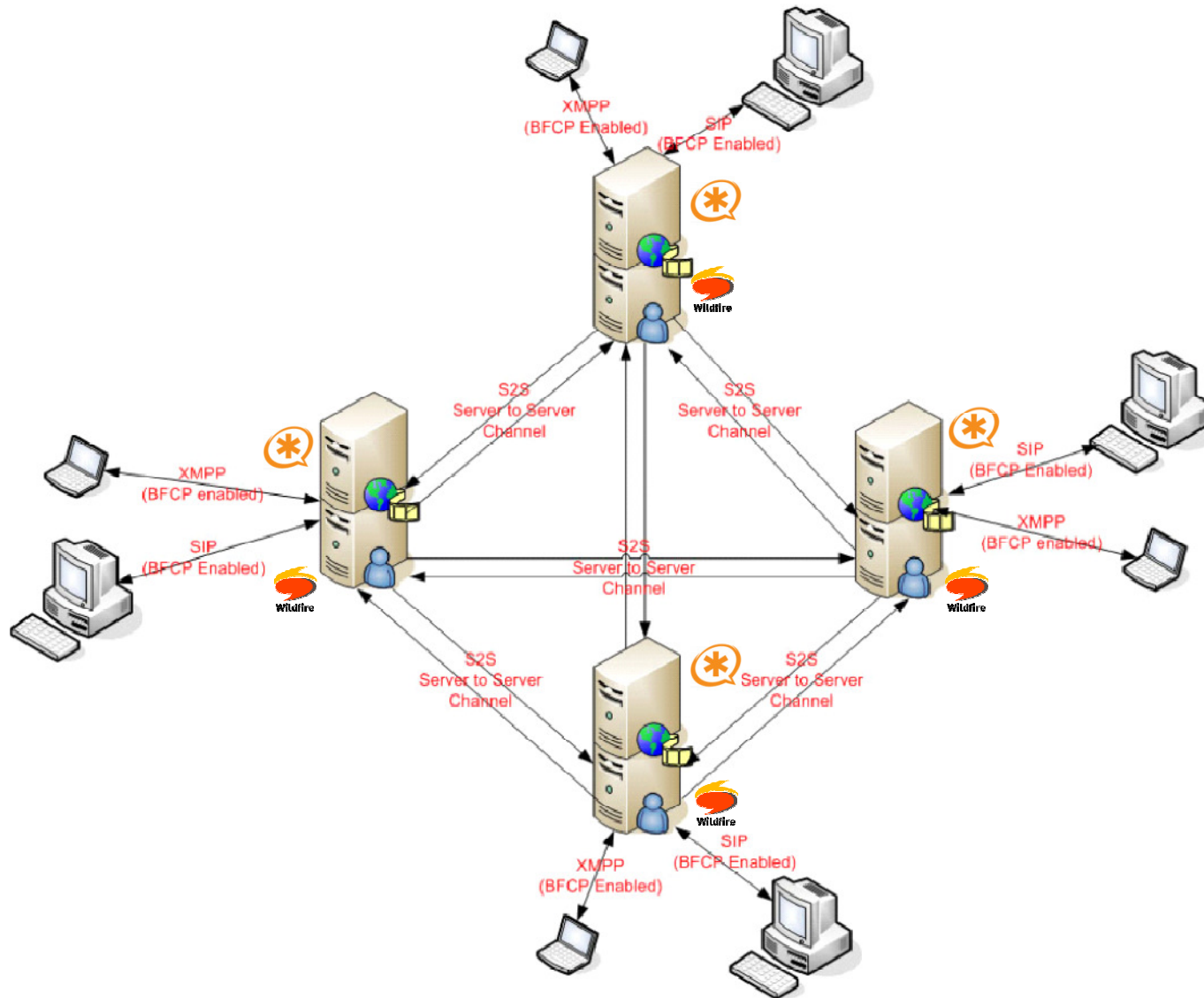


# DCON Proposal

- Distributed Conferencing (DCON)
  - Explicitly recalls XCON
    - Orchestrates the operation of a set of XCON focus elements, called “clouds”
    - Overlay network interconnecting the clouds
    - Intra-focus communication
      - Still based on XCON protocols
    - Inter-focus communication
      - Exploits Server-to-Server (XMPP)



# DCON architecture

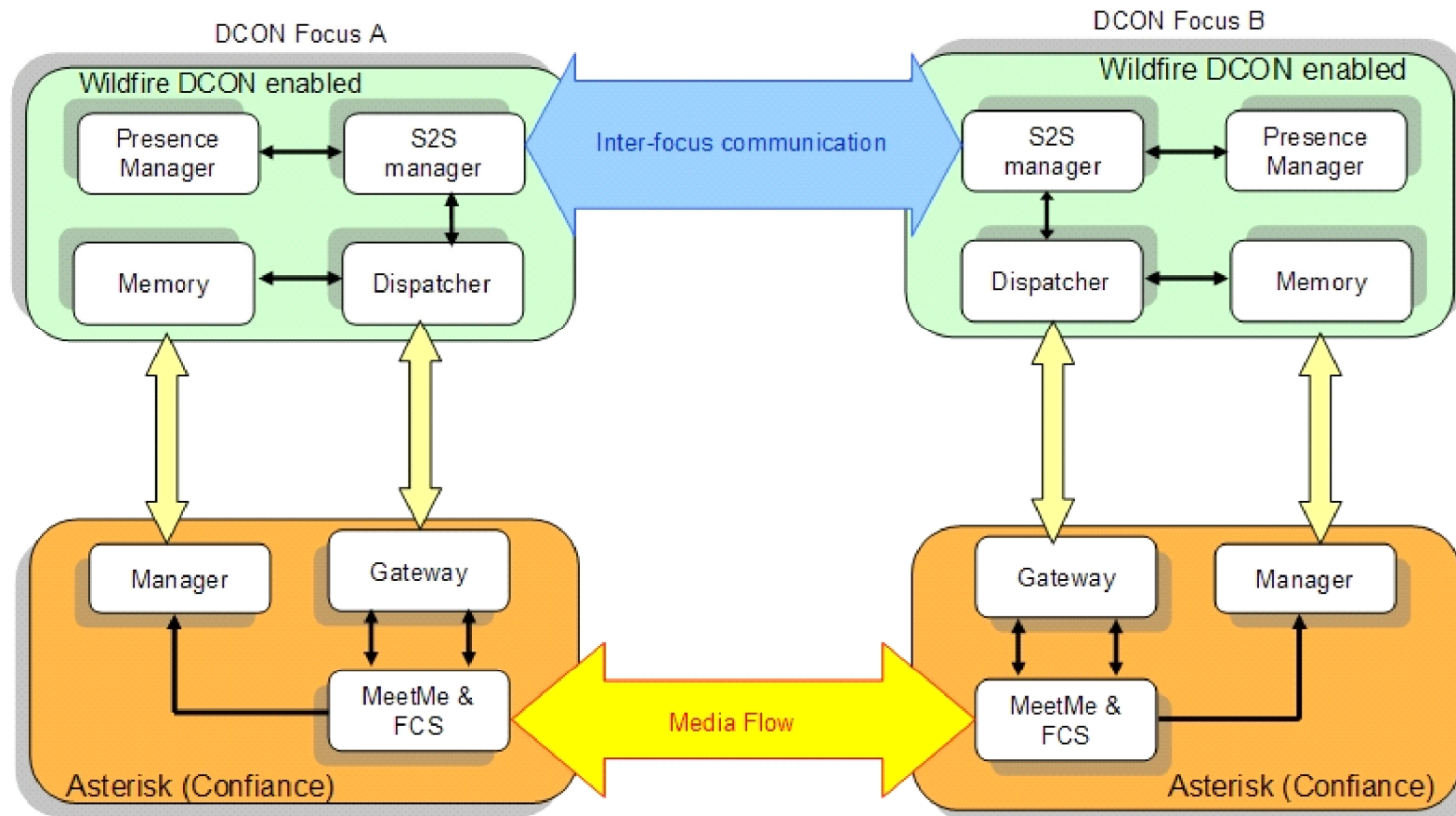


# Requirements

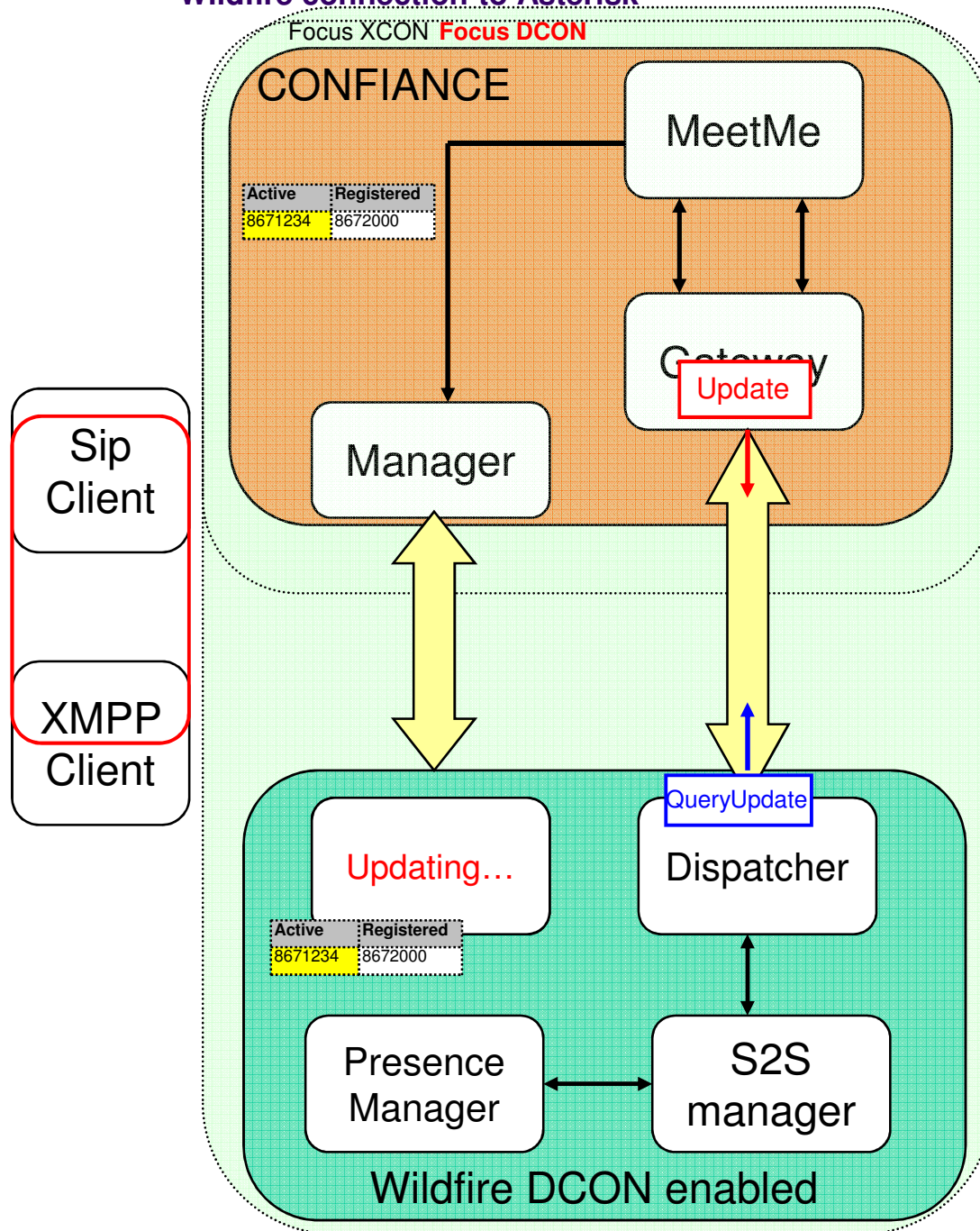


- Focus discovery
- Initialization information & spreading of conference events
- Setup and managing of distributed conferences
- Transparent dispatching of natively centralized protocols among the involved conferencing clouds

# DCON Implementation



## Wildfire connection to Asterisk



We suppose CONFIANCE is working

When the DCON component starts, 3 main events happen:

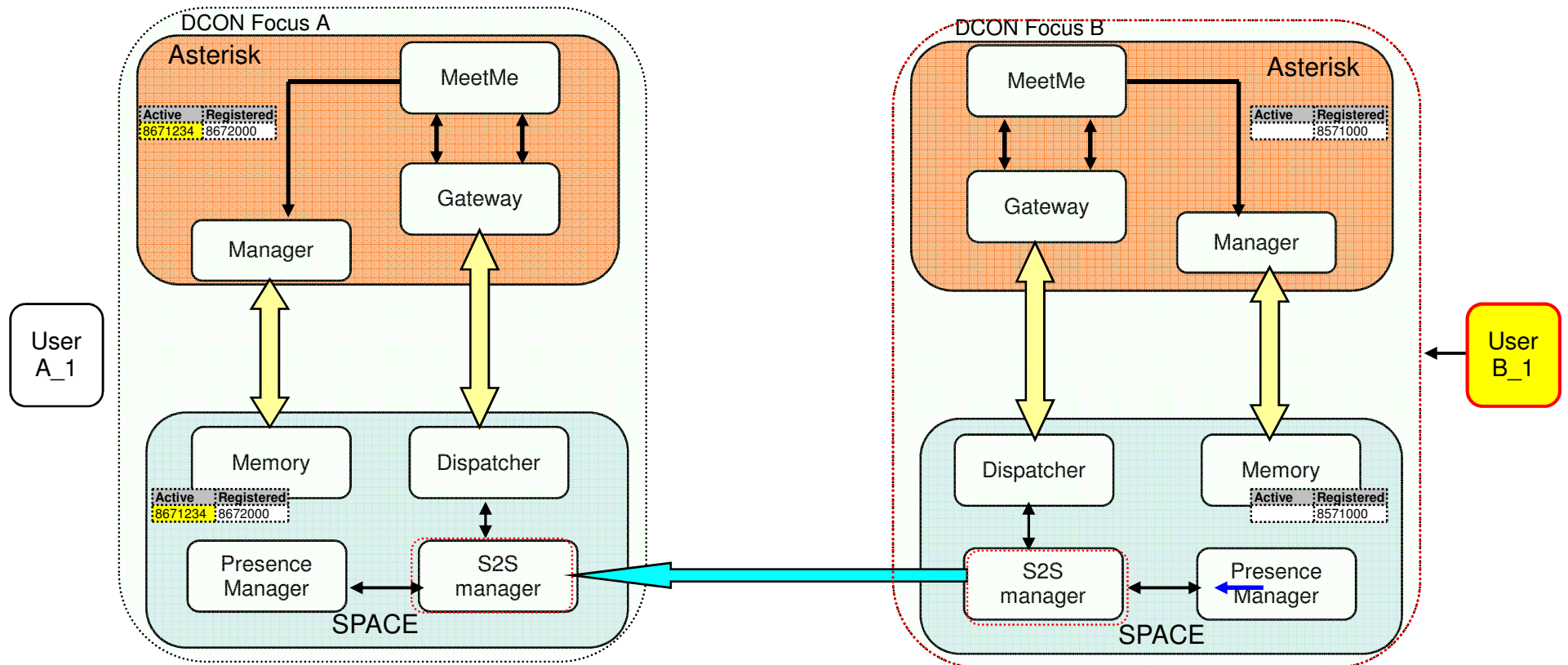
- 1) Connection to the Asterisk Manager interface
- 2) Connection to the Gateway interface
- 3) Request for initialization information

Now the focus cloud involves also the Wildfire server and SPACE component which has in charge:

- 1) Discovery of other foci
- 2) Managing of DCON information and BFCP packets.

**DCON focus discovery**

## Discovery of active Foci



We suppose that the roster of user A\_1 (belonging to focus A) contains user B\_1 (belonging to focus B) and viceversa .

Once an user (we suppose B\_1) joins the focus, the Presence Manager enforces the S2S Manager to try to contact all the foci in the B\_1's roster.

Two cases are possible...





## Asterisk dialplan: remote prefix

[...]

```
exten => _857.,1,Meetme(${EXTEN:0:7}|B|${EXTEN:7})
```

```
exten => _857.,2,Hangup
```

```
exten => _867.,1,Meetme(${EXTEN:0:7}|G|${EXTEN:7})
```

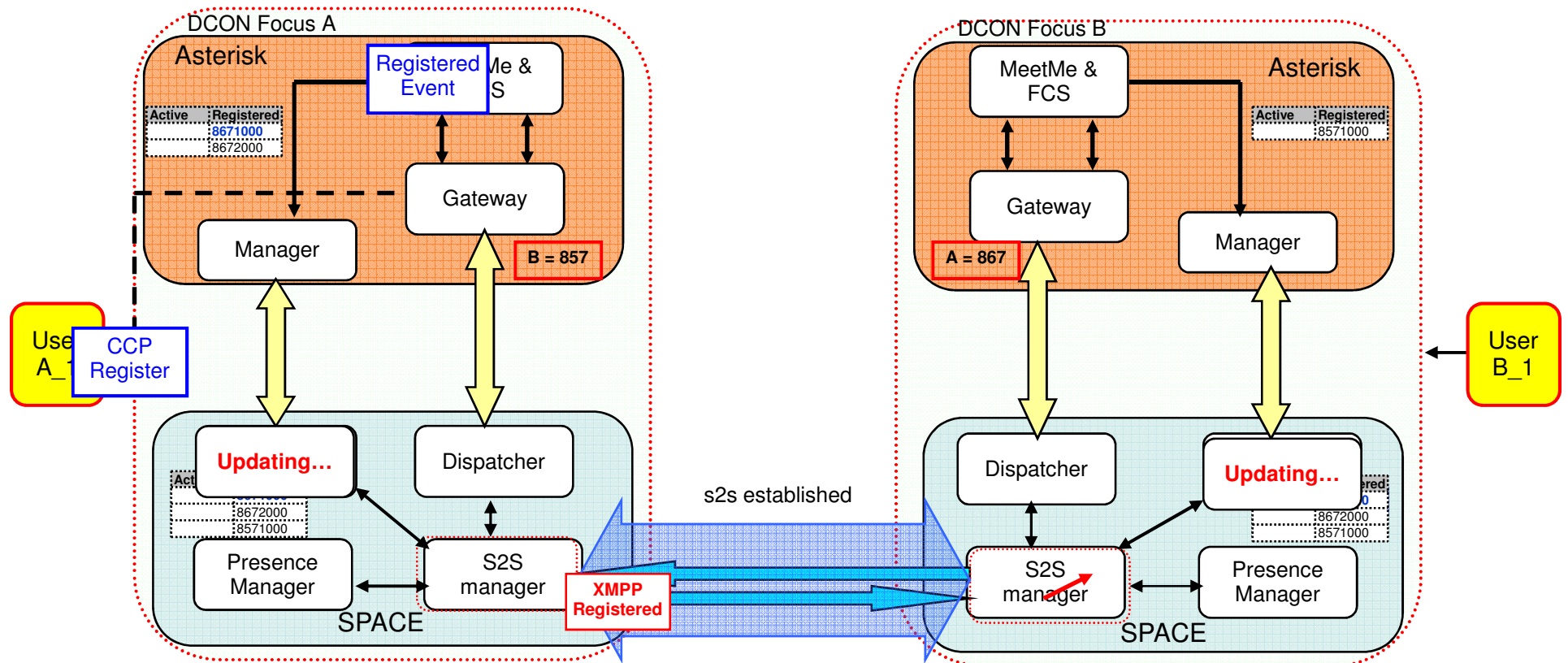
```
exten => _867.,2,Hangup
```

[...]



**DCON: spreading of conference events**

## Spreading of Conference Events

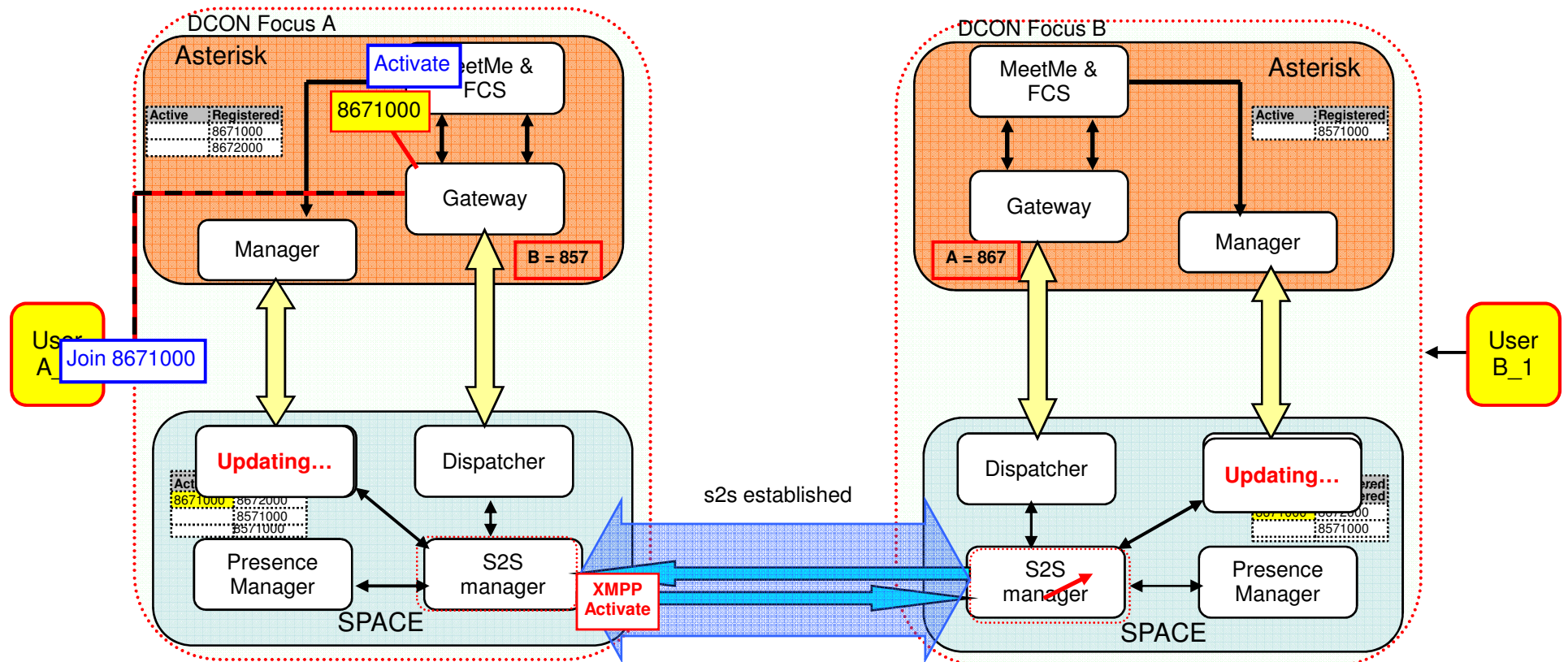


When the s2s connection has been established and the prefixes have been exchanged every local event is spread to the remote connected foci .

We suppose A\_1 registers the new local conference 8671000 by means of the CCP: a “RegisteredEvent” will be sent to SPACE by means of the Manager Interface.

SPACE will then spread it to all the active foci which will update their information.

## Spreading of Conference Events



If A\_1 joins the local conference 8671000 an "ActivateEvent" will be sent to SPACE by means of the Manager Interface.

SPACE will then spread it to all the active foci which will update their information.

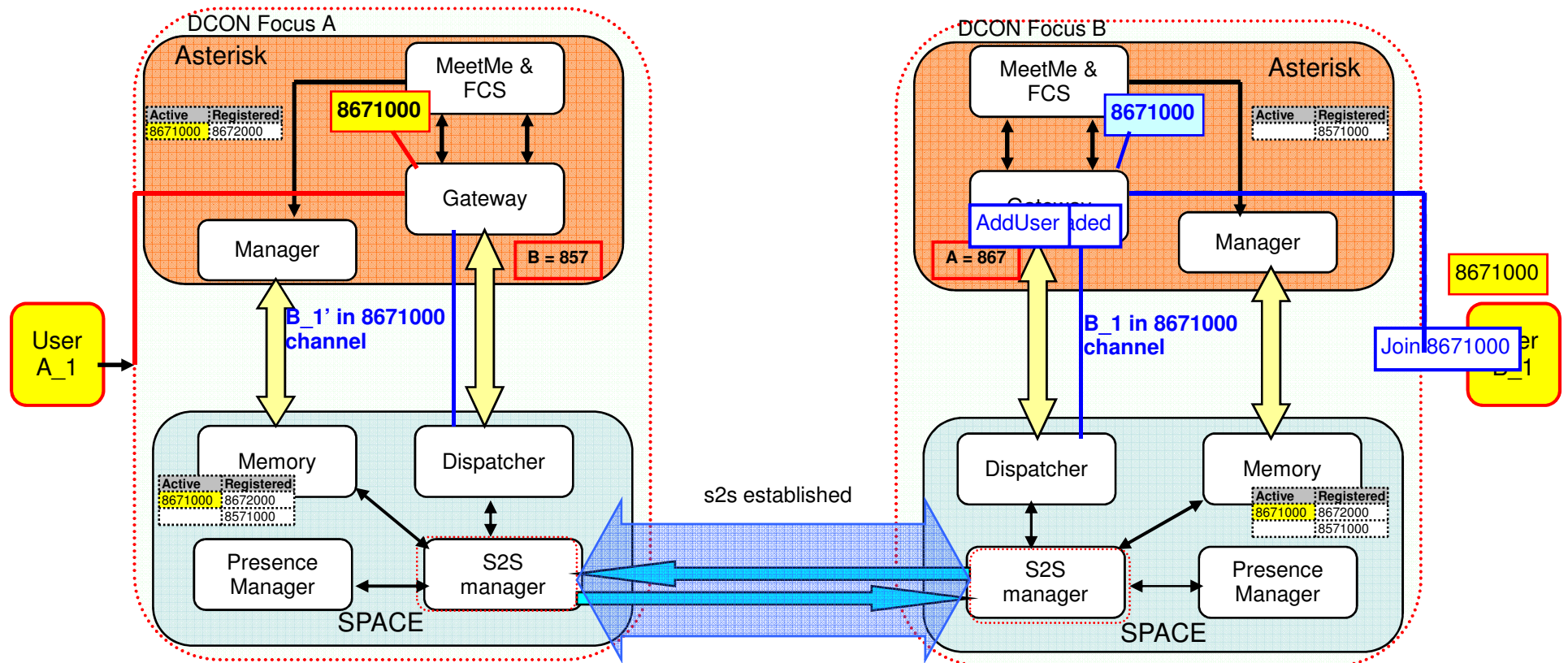
## **DCON and CCP**



**DCON: remote conference join**



## JOIN to remote conference: AddCascaded and AddUser Messages

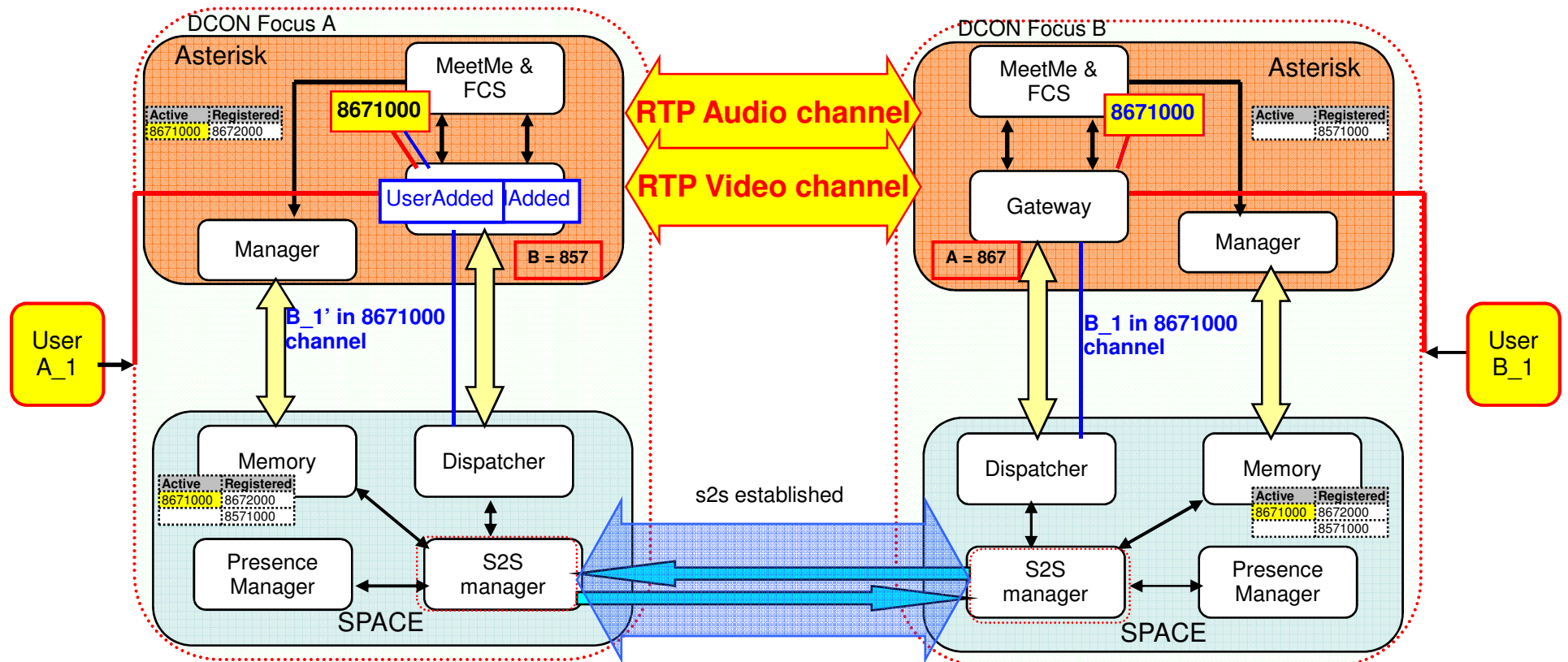


If now the user B\_1 wants to join the remote conference 8671000, he/she simply calls this conference number.

The Gateway checks the prefix and understands this is a remote conference so:

- 1) Triggers the creation of the Local Stub Conference 8671000
- 2) Sends the AddCascaded and AddUser messages to the remote focus by means of the dispatcher

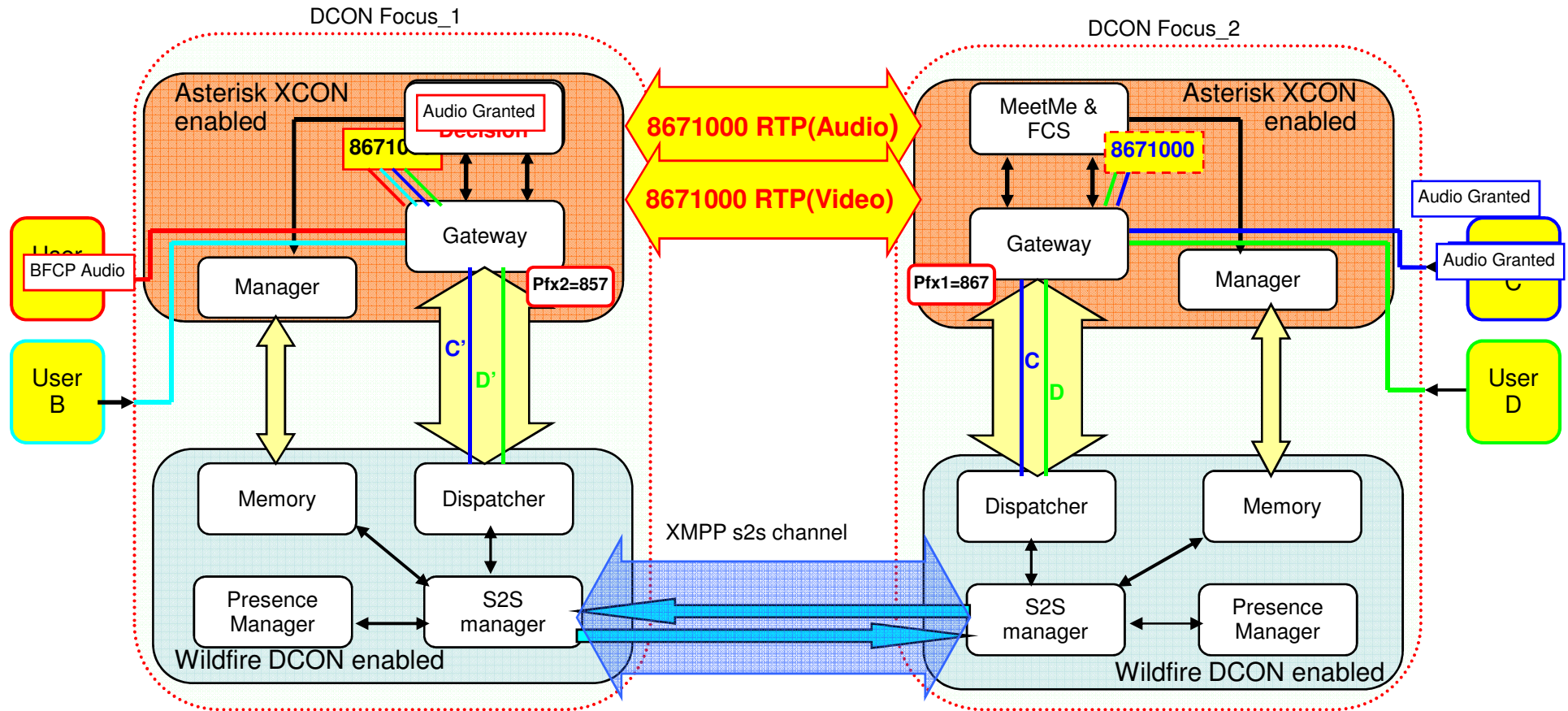
## JOIN to remote conference: CascadedAdded and UserAdded Messages



The main focus adds the Cascaded conference, sends a CascadedAdded message. Two RTP channels (Audio and Video) are opened between the foci and the Stub Conference is activated.

Then the main focus adds the remote user B\_1 to the conference and sends the new assigned "userID" encapsuled into a UserAdded Message by means of the established B\_1' channel. User B\_1 is now in the conference.

**DCON:**  
**protocol dispatching and local mixing**



If local user A sends a BFCP request, the Gateway directly forwards it to the main FCS. After the chair's decision the FCS sends the response.

If remote user C sends a BFCP request, the Gateway forwards it to the main FCS by means of the dispatcher through the C-s2s-C' channels.



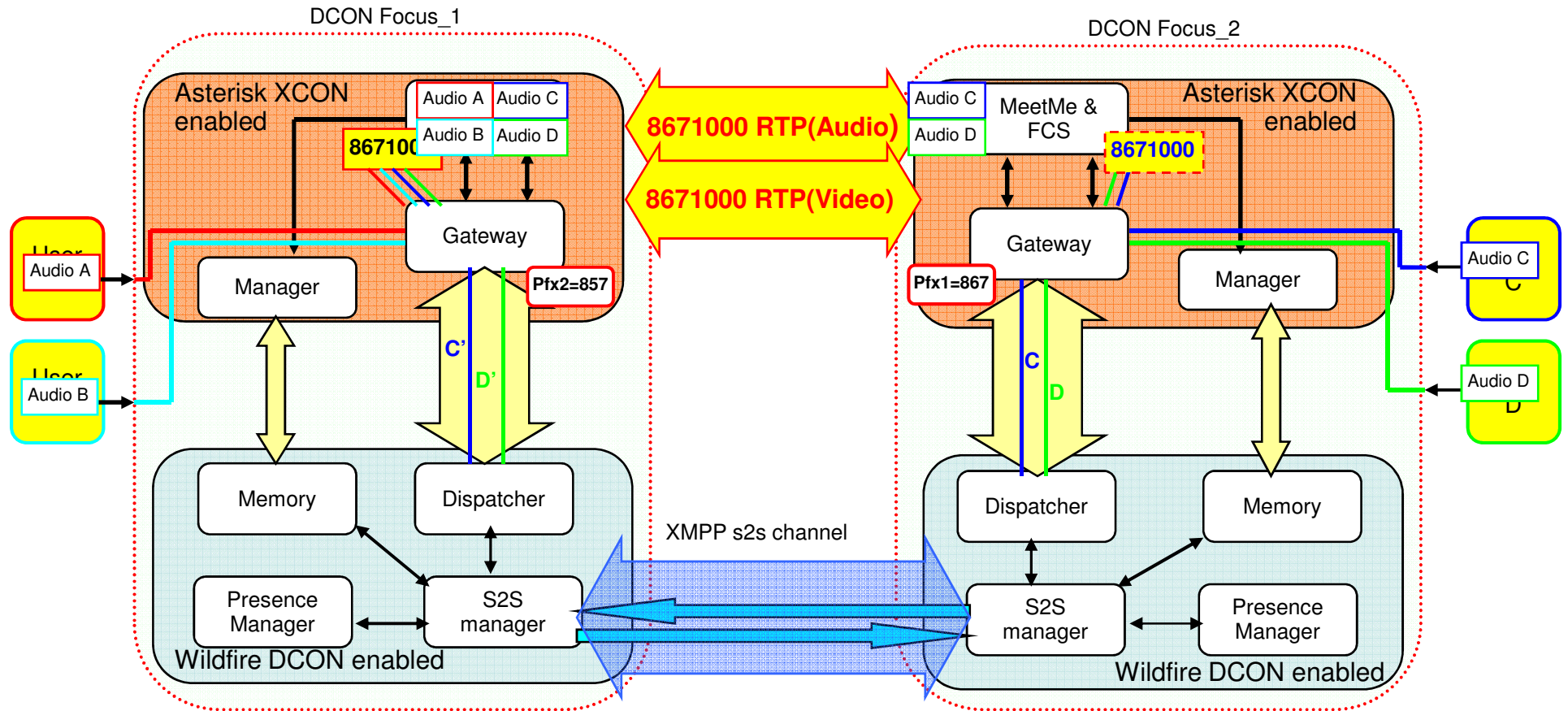
Signalling flow



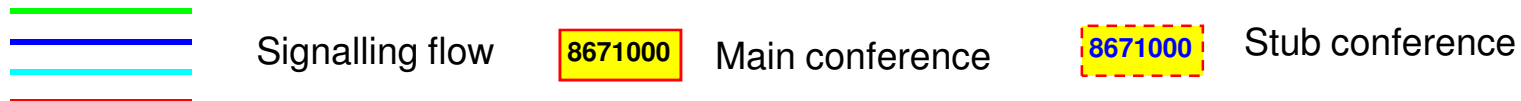
Main conference

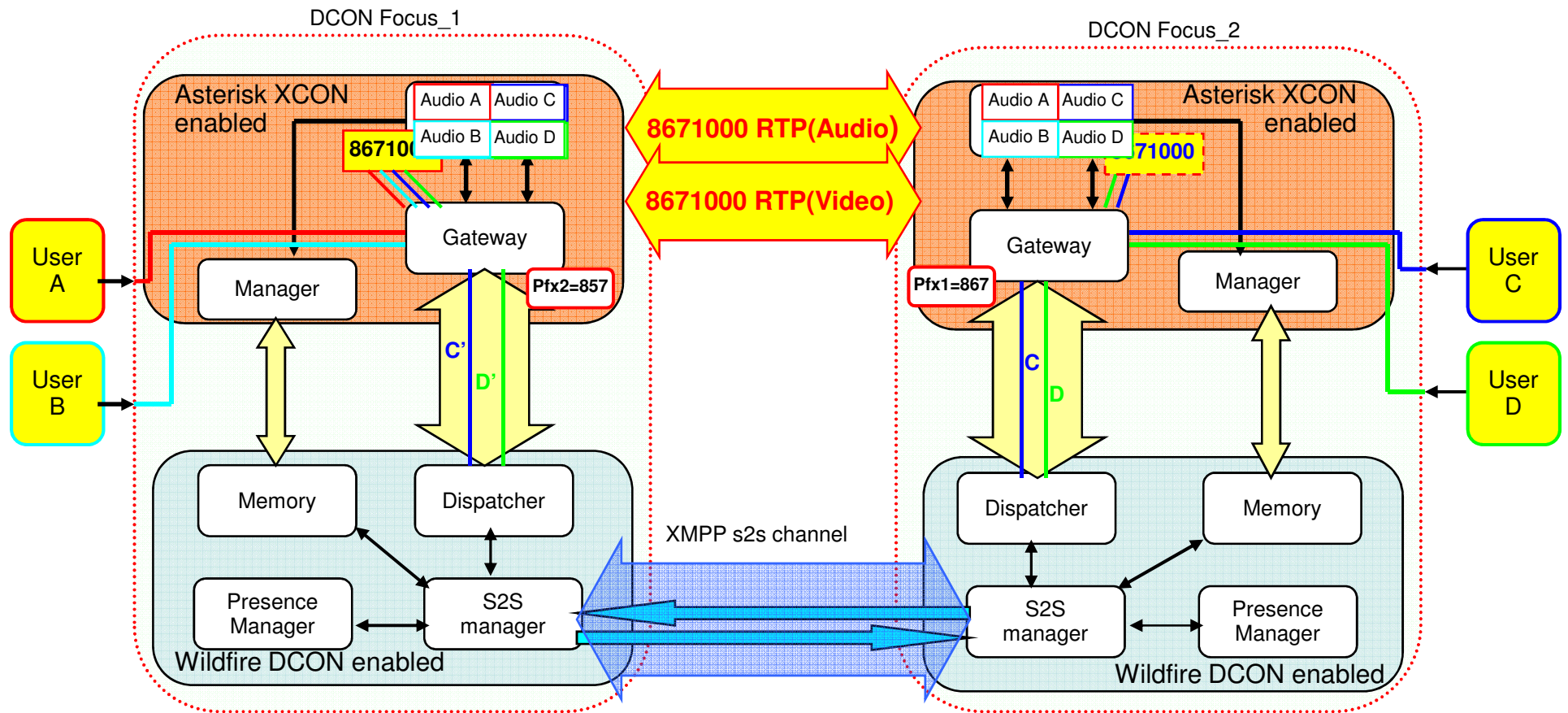


Stub conference

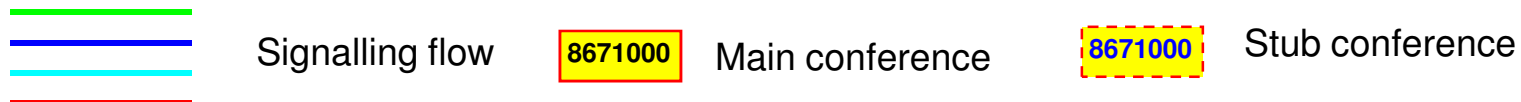


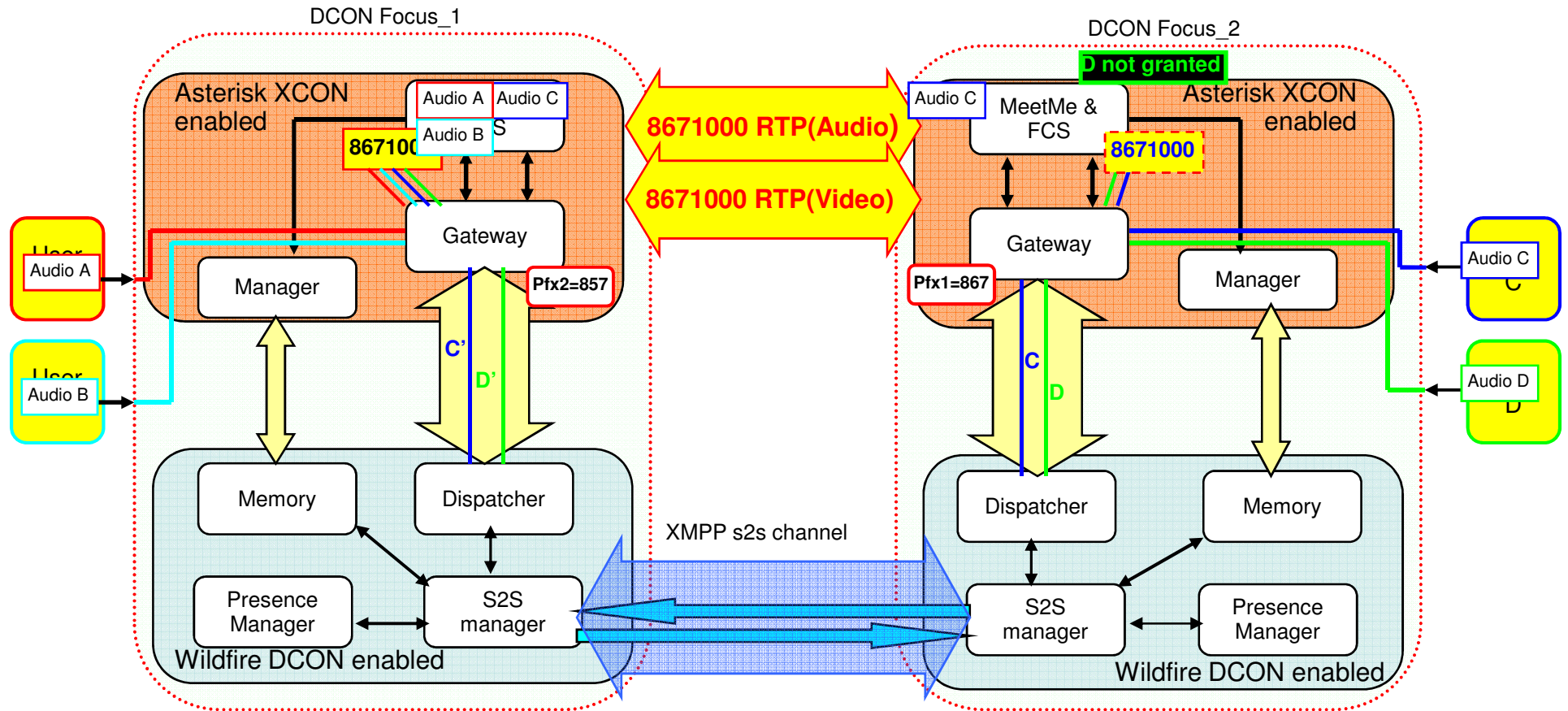
We suppose audio floor is granted to all participants. C and D will send their audio flows to local focus that, after FCS controls, will mix and forward them through the RTP channel.



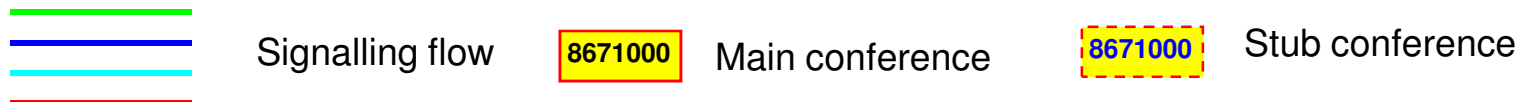


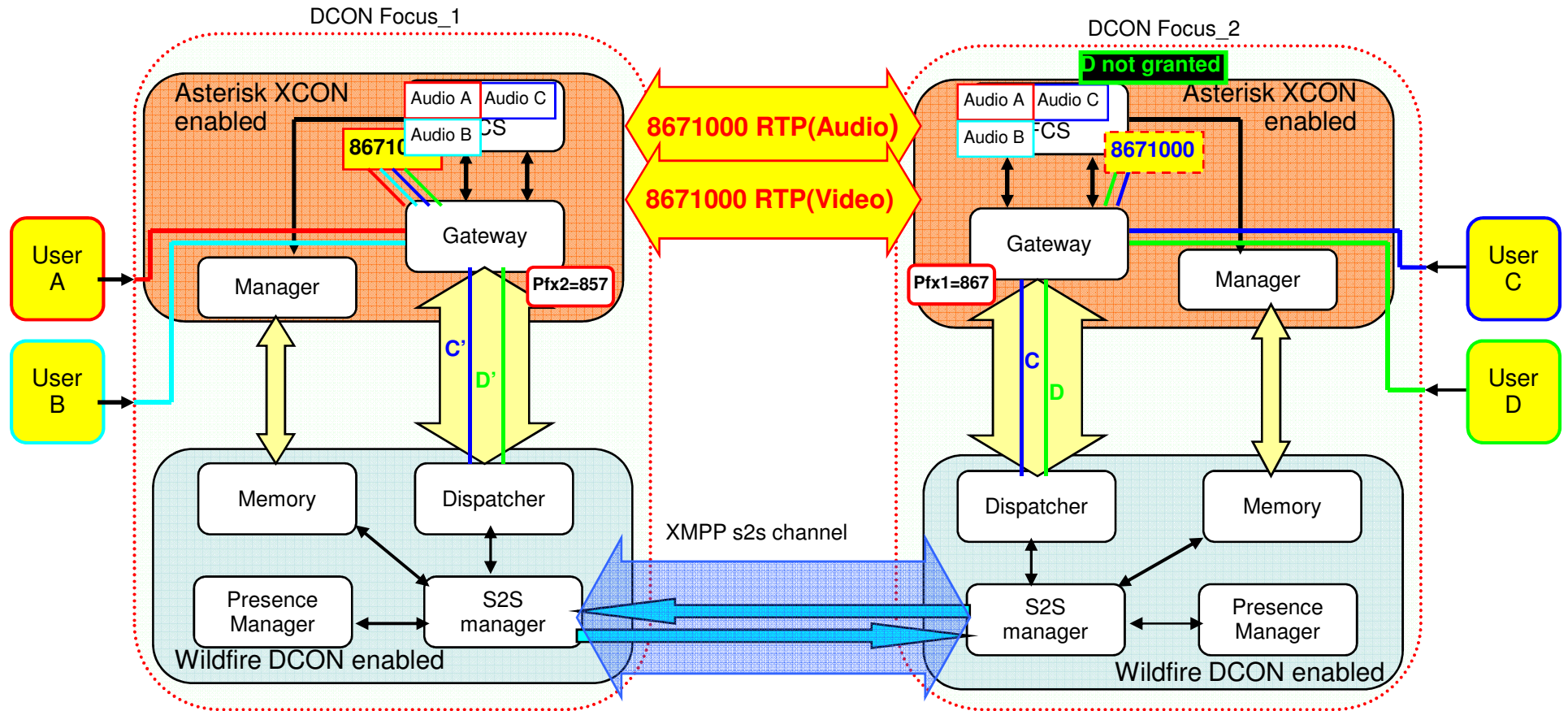
The main focus will mix the received mixed flow (C-D) with the local user's flows and will send back the resulting mixed flow to the local users and to the remote focus (through the RTP channel). The remote focus then will spread the received mixed flow (A-B-C-D) to its users.





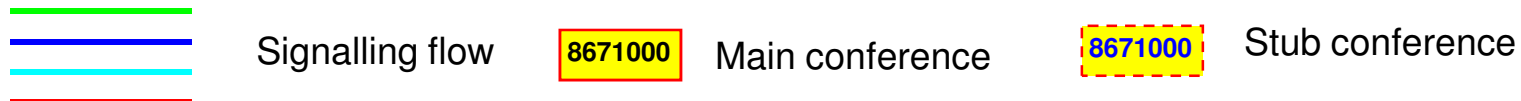
We suppose audio floor isn't granted to user D but her/his softphone is BFCP unaware so it however sends the audio flow. C and D will send their audio flows to local focus that, after FCS controls, will forward only the granted user flow.





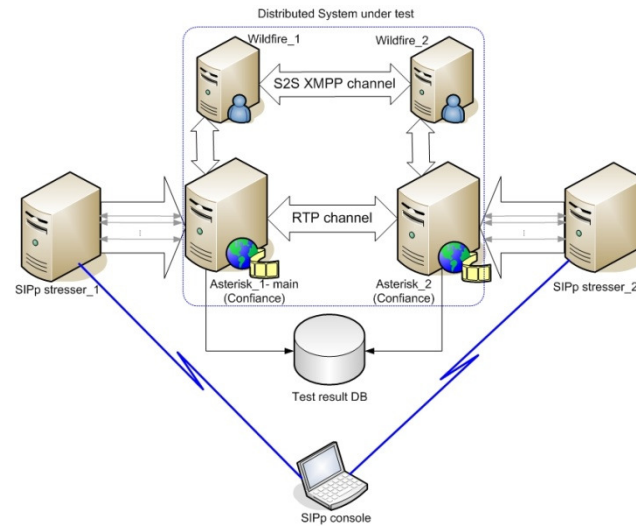
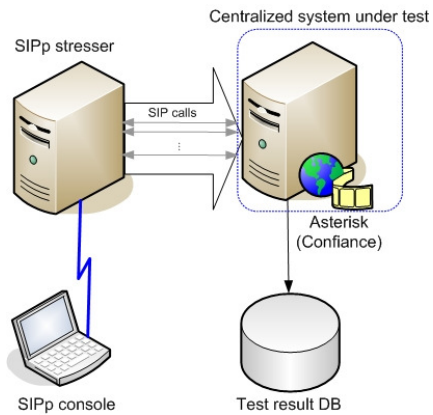
The main focus will mix the received flow (C) with the local user's flows and will send back the resulting mixed flow to the local users and to the remote focus (through the RTP channel).

The remote focus then will spread the received mixed flow (A-B-C) to its users.

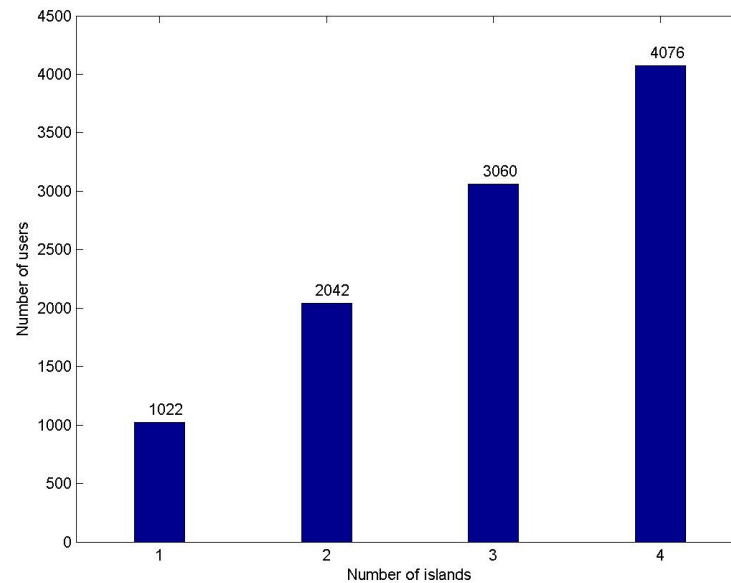




# Testing DCON: Scalability



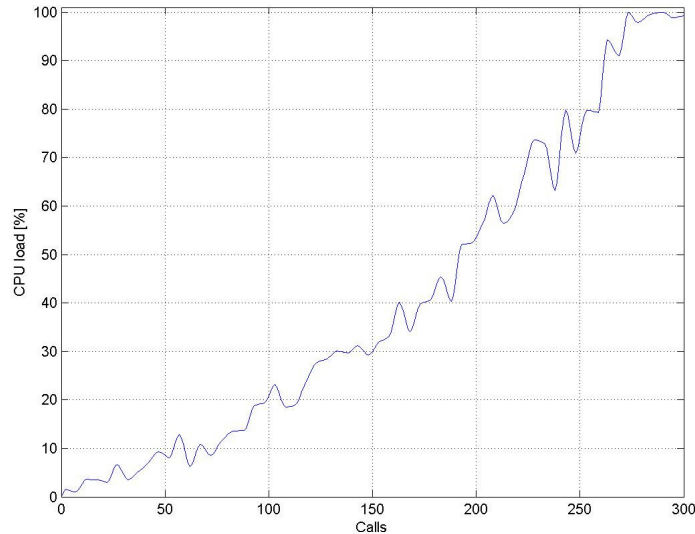
- The maximum number of participants linearly grows with the number of DCON islands



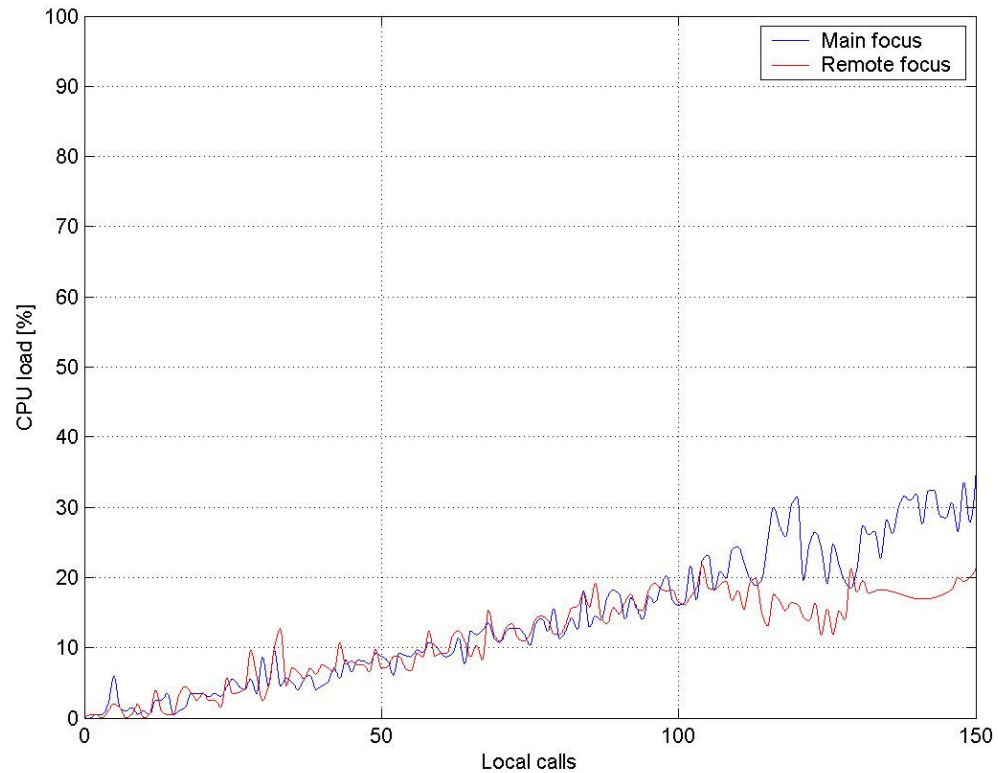
# Testing DCON: Performance



- 2 islands



Focus	calls	CPU load (%)
Main	300	99,4

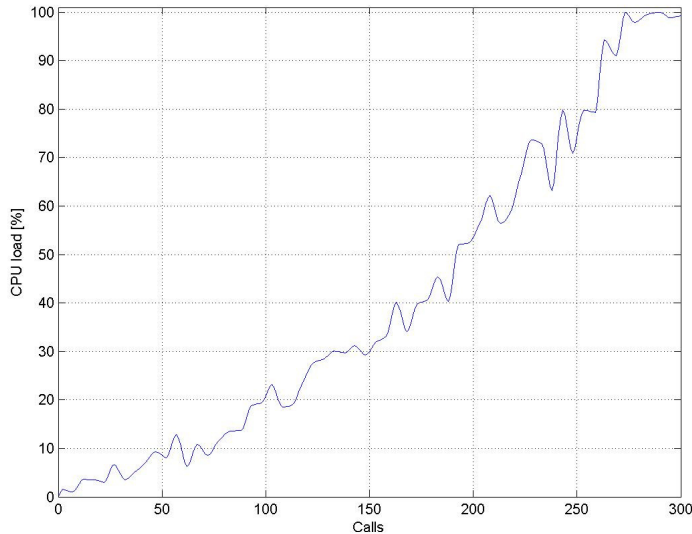


Focus	Number of calls	CPU load (%)
Main	150	30,04
Remote	150	20,19

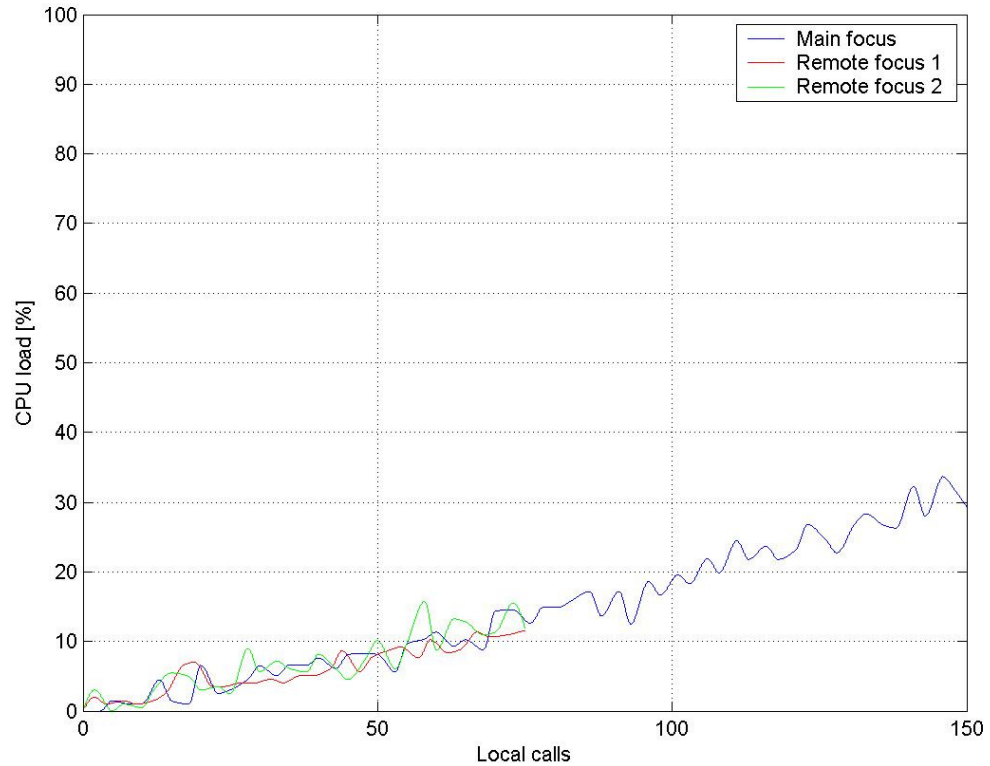
# Testing DCON: Performances



- 3 islands

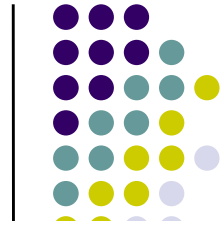


Focus	calls	CPU load (%)
Main	300	99,4

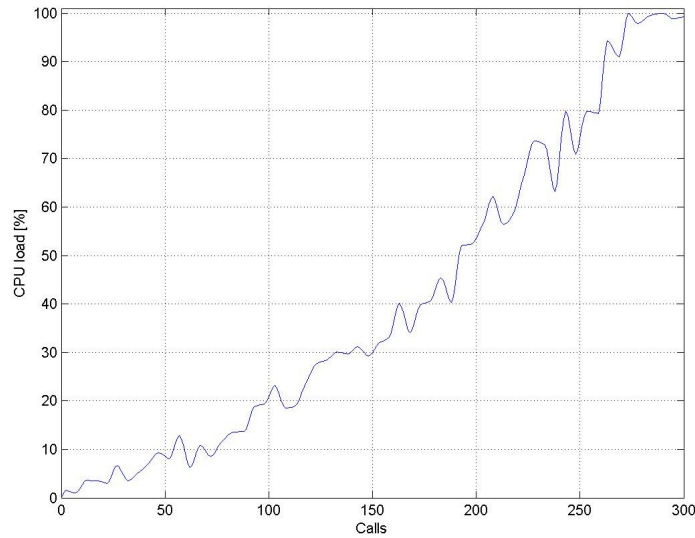


Focus	Number of calls	CPU load (%)
Main	150	31,05
Remote_1	75	12
Remote_2	75	12

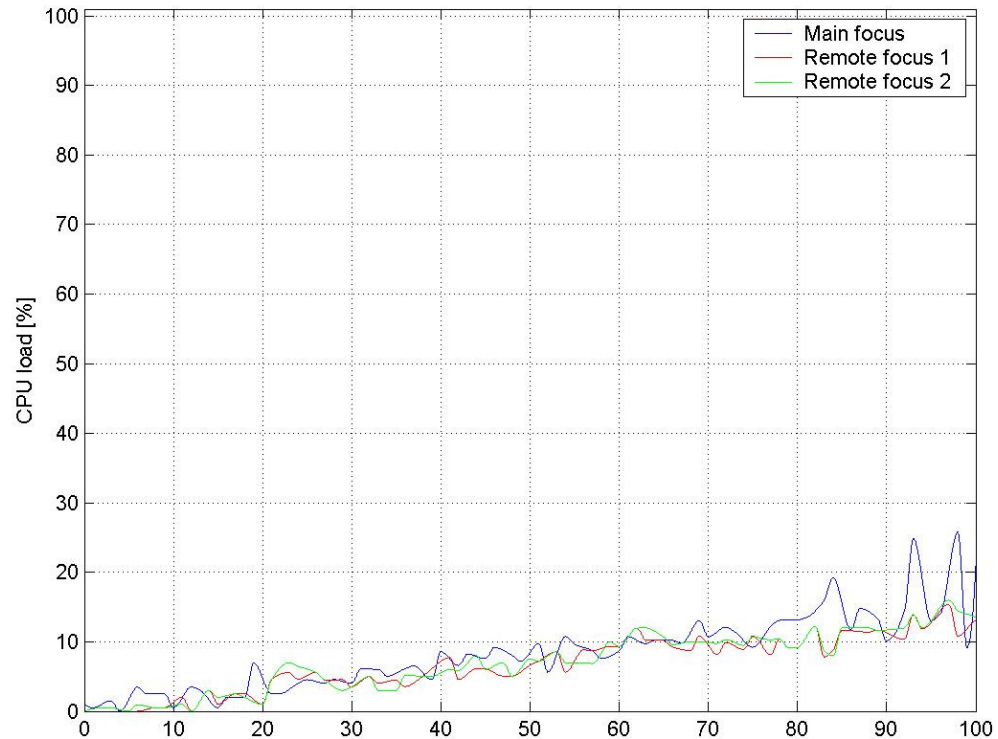
# Testing DCON: Performance



- 3 islands

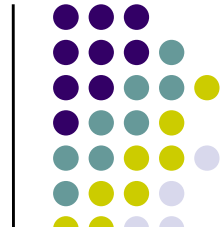


Focus	calls	CPU load (%)
Main	300	99,4

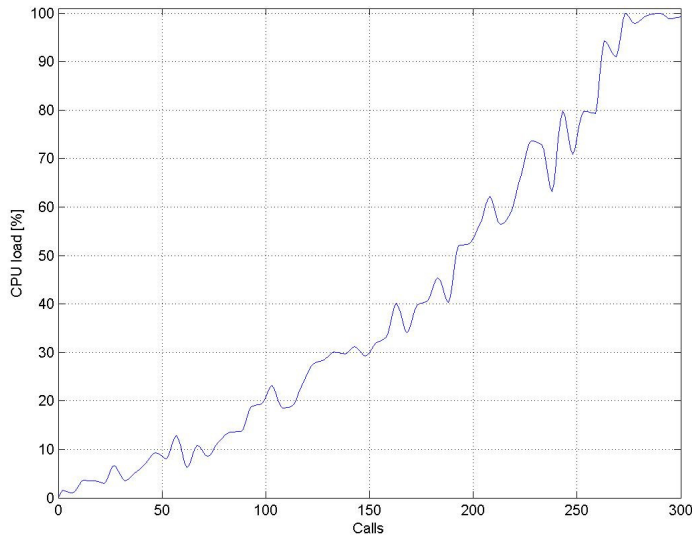


Focus	calls	CPU load (%)
Main	100	20
Remote_1	100	18
Remote_2	100	18

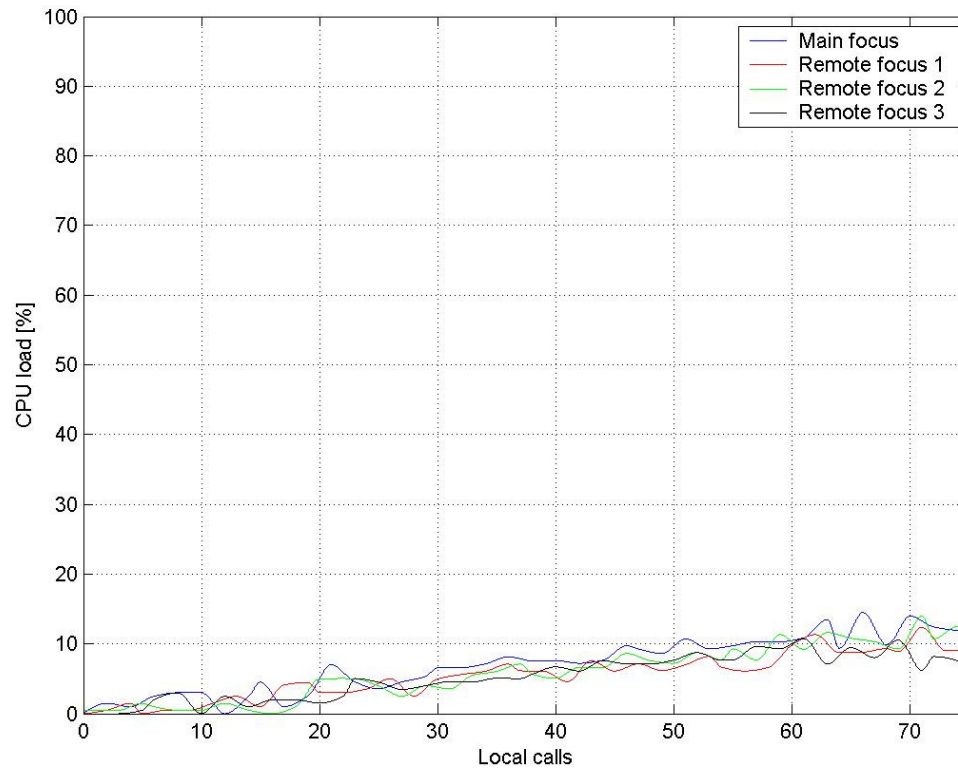
# Testing DCON: Performance



- 4 islands

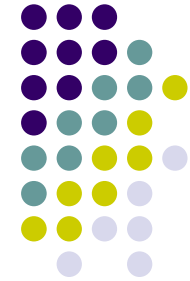


Focus	calls	CPU load (%)
Main	300	99,4



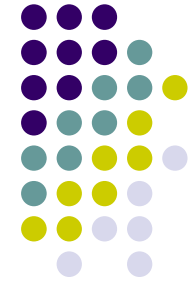
Focus	Number of calls	CPU load (%)
Main	75	12,66
Remote_1	75	12
Remote_2	75	12
Remote_3	75	12

# Testing DCON: Performance



Number of islands	Number of local users	Number of remote users	Main focus CPU load	Remote focus 1 CPU load	Remote focus 2 CPU load	Remote focus 3 CPU load
1	300	-	99.4%	-	-	-
2	150	150	30.04%	20.19%	-	-
3	100	200 (100/100)	20%	18%	18%	-
3	150	150 (75/75)	31.05%	12%	12%	-
4	75	225 (75/75/75)	12.66%	12%	12%	12%
4	150	150 (50/50/50)	32.4%	7.8%	7.8%	7.8%

# Market overview



- From the lab to the real world: research becomes progress
- Gartner group prediction:
  - The market for *Web Conferencing* and *Collaboration Tools* will grow at a compound annual rate of 23% through 2011
- Main cost benefits
  - Savings on business travels
  - Efficient enterprise communications
    - Improving and simplifying collaboration activities
- Other valuable benefits
  - Environmental concerns and initiative for “Green IT”
    - CO<sub>2</sub> emission reduction
  - Travel stress decrease



# Market needs

- MUST features:
  - Presentation delivery
  - Desktop/screen sharing
  - Text chat
  - Shared whiteboard
  - Basic security
  - Remote control
- More advanced features:
  - Integrated PSTN audio
  - Integrated Voice over IP audio
  - Live video
  - File sharing
  - Application /document sharing
  - Advanced security
  - Archiving
  - Feedback
  - Polls and surveys
  - E-learning
  - Mobility support





# Meetecho spin off

- Open-source, Java-based, multiplatform client
- Features:
  - ☑ **Presentation delivery**
  - ☑ **Desktop/screen sharing**
  - ☑ **Text chat**
  - ☑ **Shared whiteboard**
  - ☑ **Basic security**
  - ☑ **Remote control**
- More advanced features:
  - ☑ **Integrated PSTN audio**
  - ☑ **Integrated Voice over IP audio**
  - ☑ **Live video**
  - ☑ **File sharing**
  - Application /document sharing
  - Advanced security
  - Archiving
  - Feedback
  - ☑ **Polls and surveys**
  - E-learning
  - ☑ **Mobility support**

# Meetecho in action



The screenshot displays the Meetecho web interface. At the top, a toolbar includes buttons for Invite, Stop, Sidebar, Next Frame, Polling, Share, Viewer, and Exit. The main content area features a slide show titled "3rd Party Call Control" with a sequence diagram. The diagram shows the interaction between UAC, AS, and MS:

- UAC sends INVITE (x) to AS.
- AS sends 180 (Ringing) to UAC.
- AS sends INVITE (x) as 3PCC to MS.
- MS sends 100 (Trying) to AS.
- AS sends 200 OK to UAC.
- MS sends 200 OK to AS.
- UAC sends ACK to AS.
- AS sends ACK to MS.
- A large blue arrow at the bottom indicates "RTP Media Stream(s) flowing" between UAC and MS.

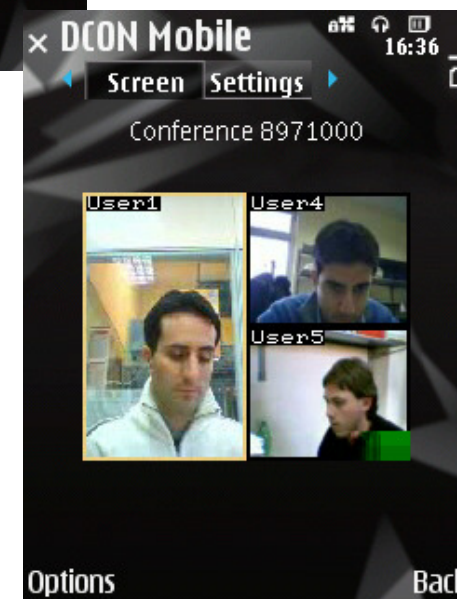
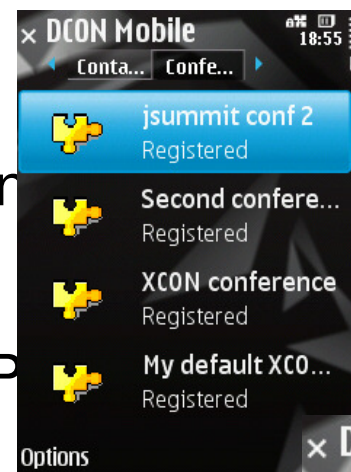
Below the diagram, the text reads: "NEW244, St. Petersburg, 04 September 2008".

On the right side, there is a "Video" window showing three users (User1, User2, User3) and a "Chair Tools" panel with options like Set Chair, Chair Action, Audio Granted, Video Granted, Floor Request, and Floor Release. Below that is a "Media Controls" panel with options like MirrorOff, Pause Audio, and Send Video. At the bottom right, a "Participants" list shows "tcastaldi", "134user", and "Alessandro".



# Meetecho: mobile access

- Client for Symbian S60 fp2
- Protocols
  - eXtensible Messaging and Presence Protocol (XMPP)
  - Session Initiation Protocol (SIP)
  - Real-time Transport Protocol (RTP)
- Features
  - Retrieves DCON conference information and events
  - Audio
    - Sends and receives
  - Video
    - Receives mixed flows



# References



- IETF
  - <http://www.ietf.org>
- XCON
  - <http://www.ietf.org/html.charters/xcon-charter.html>
- CONFIANCE web site
  - <http://confiance.sourceforge.net/>
- DCON web site
  - <http://dcon.sourceforge.net/>
- Meetecho web site
  - <http://www.meetecho.com>