

Centralized Conferencing using SIP

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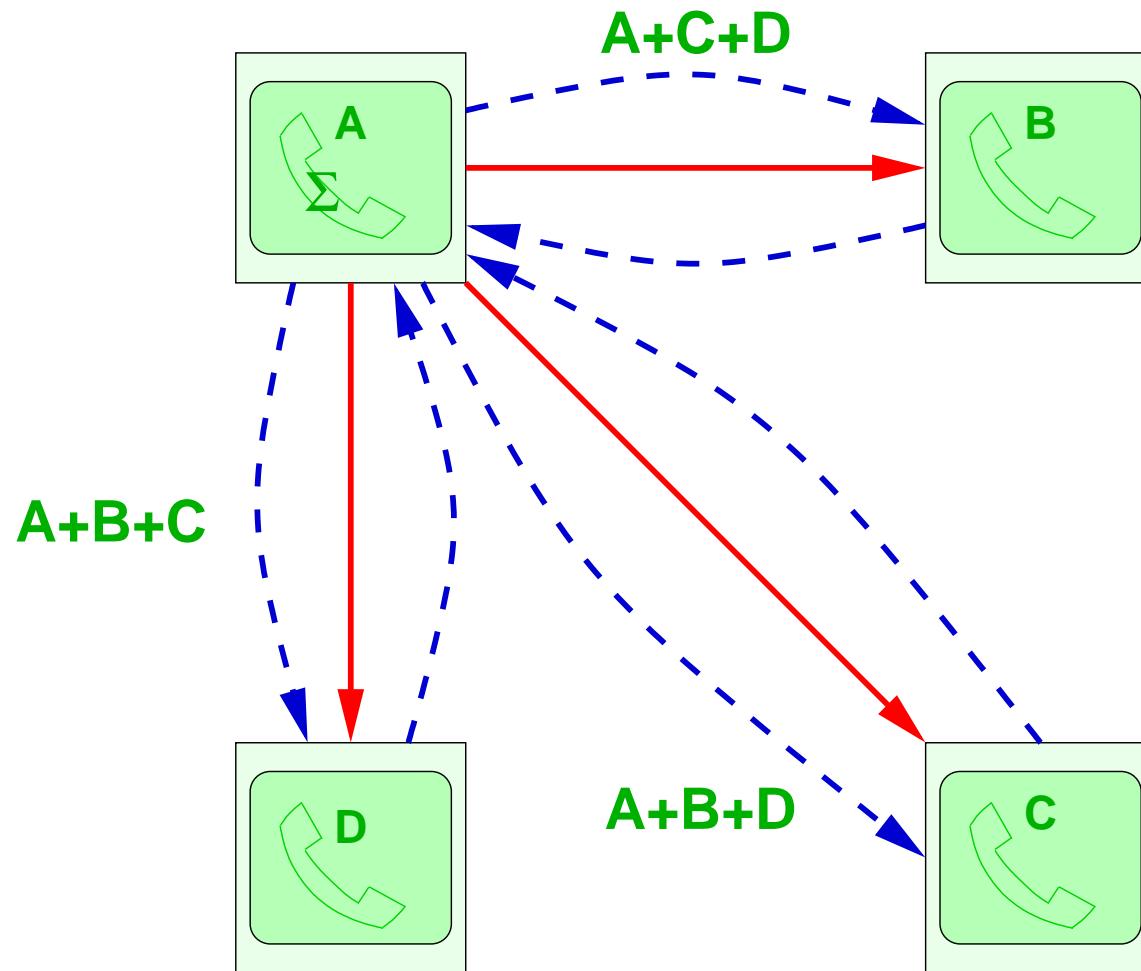
IP telephony workshop – Columbia University, New York

April 2, 2001

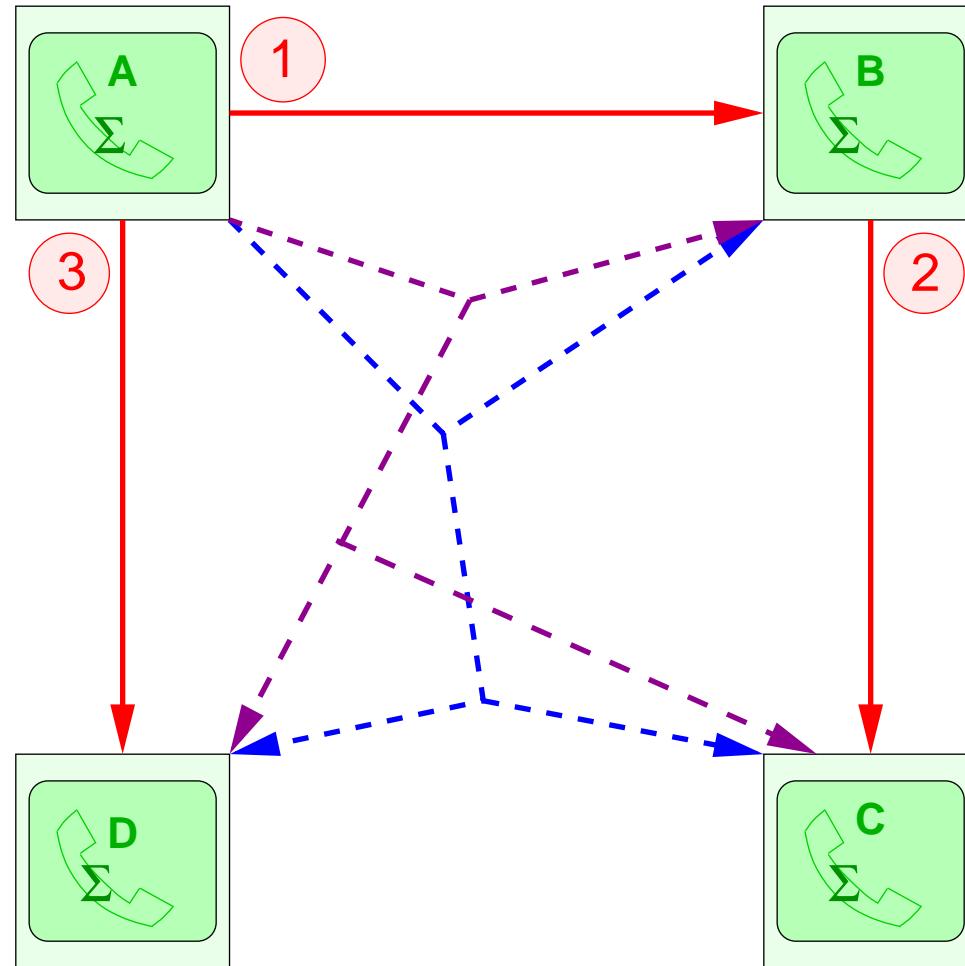
Overview

- conferencing models
- centralized conferencing server
- design issues
- measurement results

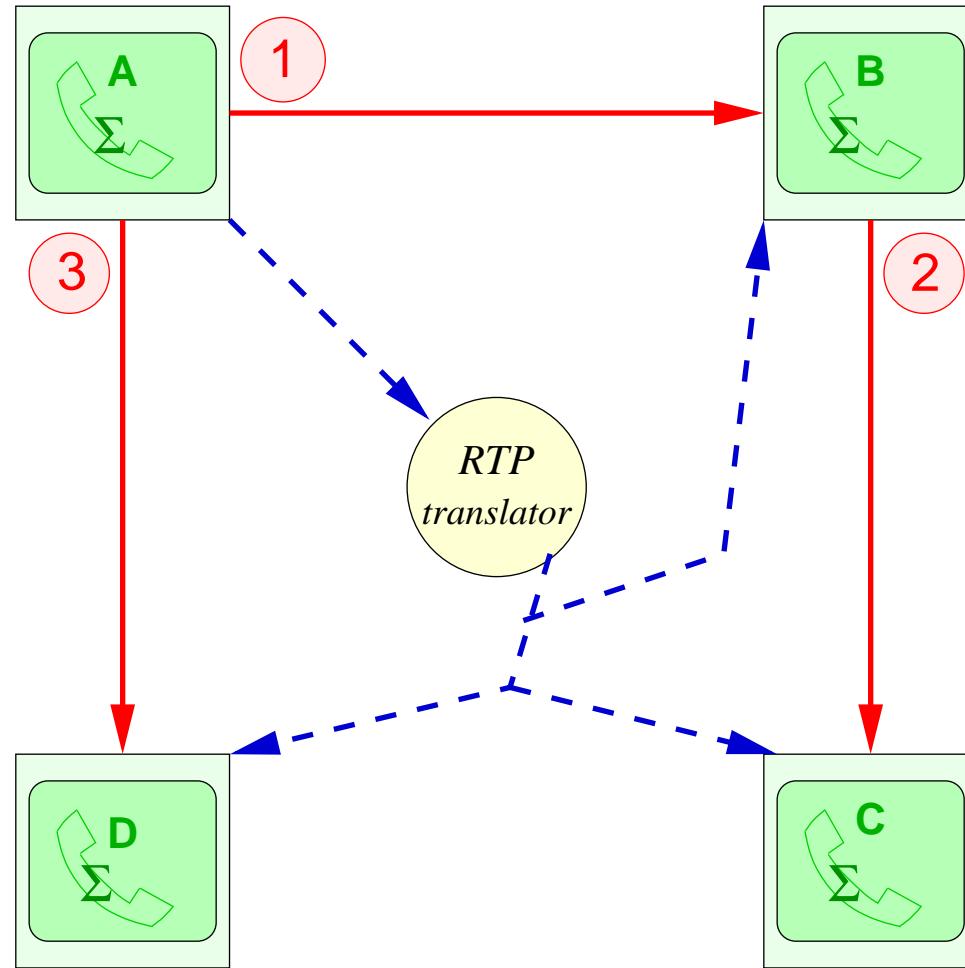
Conference models: end system mixing



Conference models: multicast transmit & receive

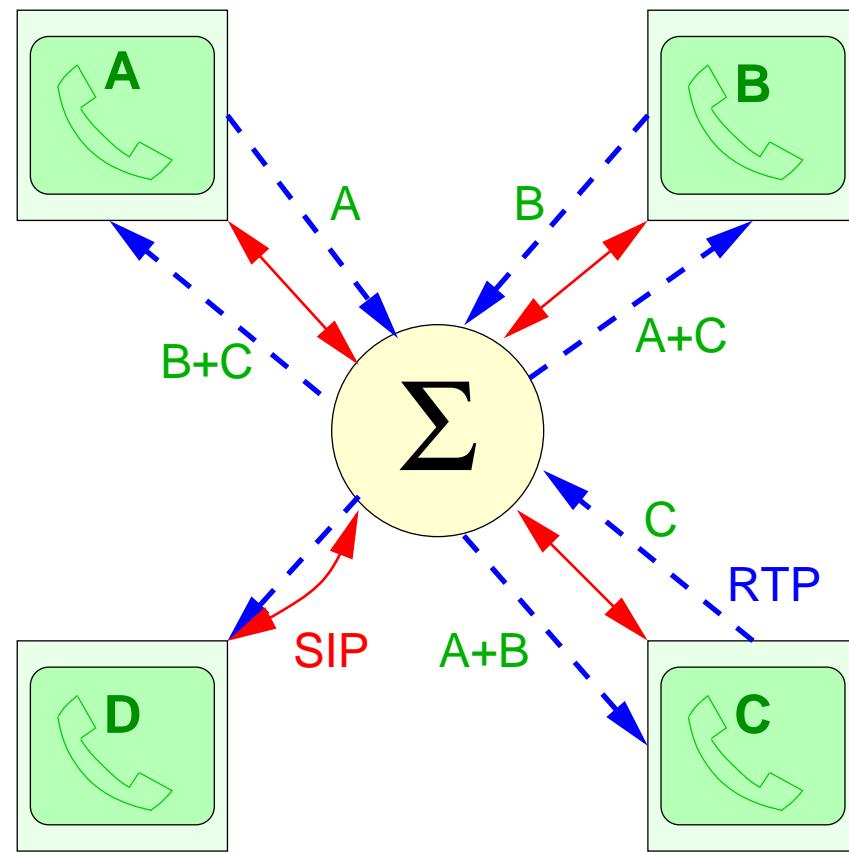


Conference models: multicast receive, unicast transmit

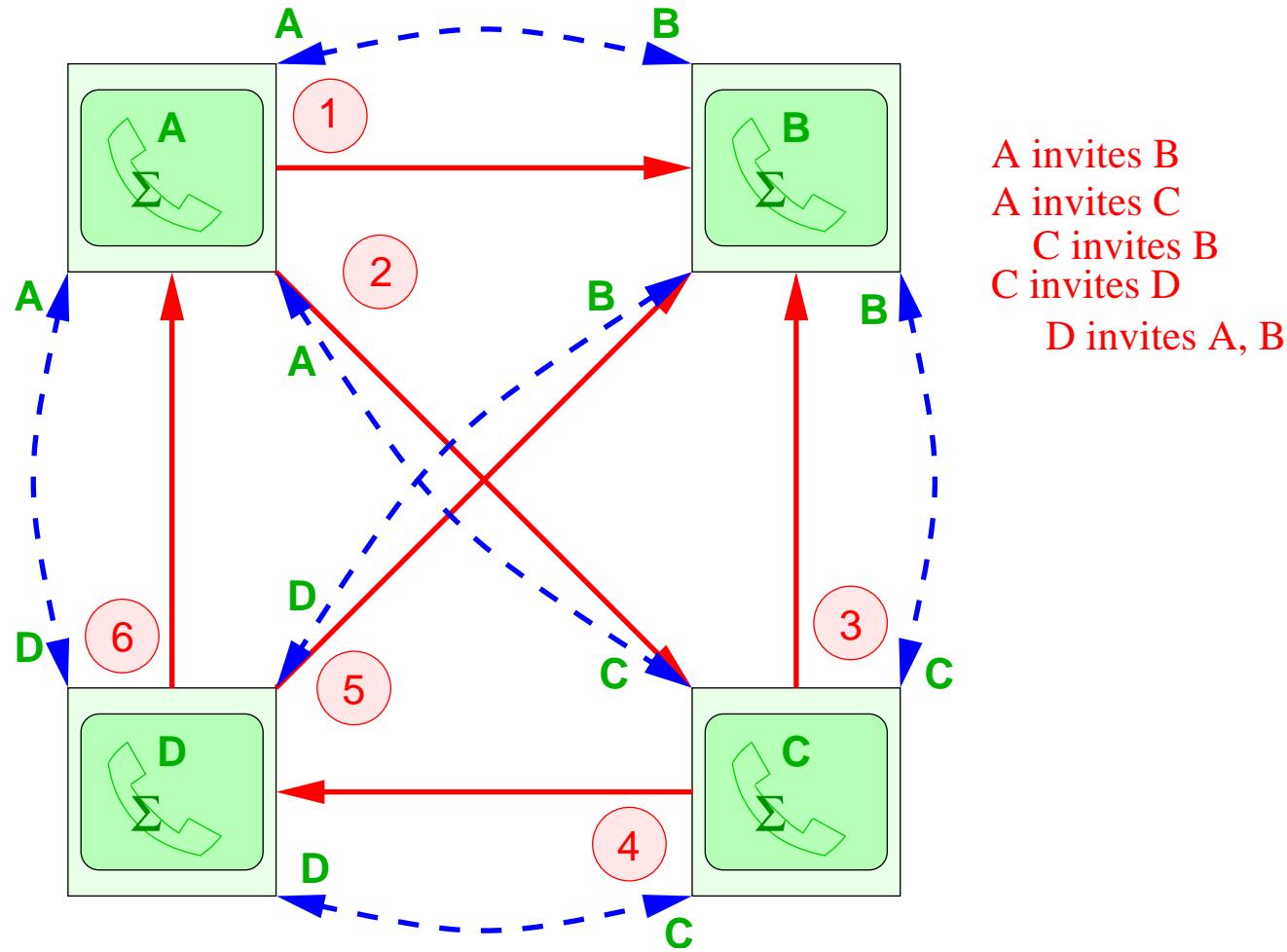


Conference models: central server

can be call-out or dial-in



Conference models: full mesh



Conference models – complexity

I active senders, N participants

Properties	central	full mesh	mcast	uni rx, mcast tx	end mixing
Topology	Star	full mesh	mcast tree	star+mcast tree	ad-hoc
Server proc.	$O(M + N)$	n/a	n/a	$O(M + N)$	n/a
Endpoint proc.	$O(1)$	$O(M)$	$O(M)$	$O(1)$	variable
Server bw	$O(M + N)$	n/a	n/a	$O(M)$	n/a
Endpoint bw	$O(1)$	$O(M)$	$O(1)$	$O(1)$	variable
Scaling	medium	medium	large	large	medium
Heterogen. UA	yes	yes	no	no	yes (partially)
Own media?	no	no	no	yes	no

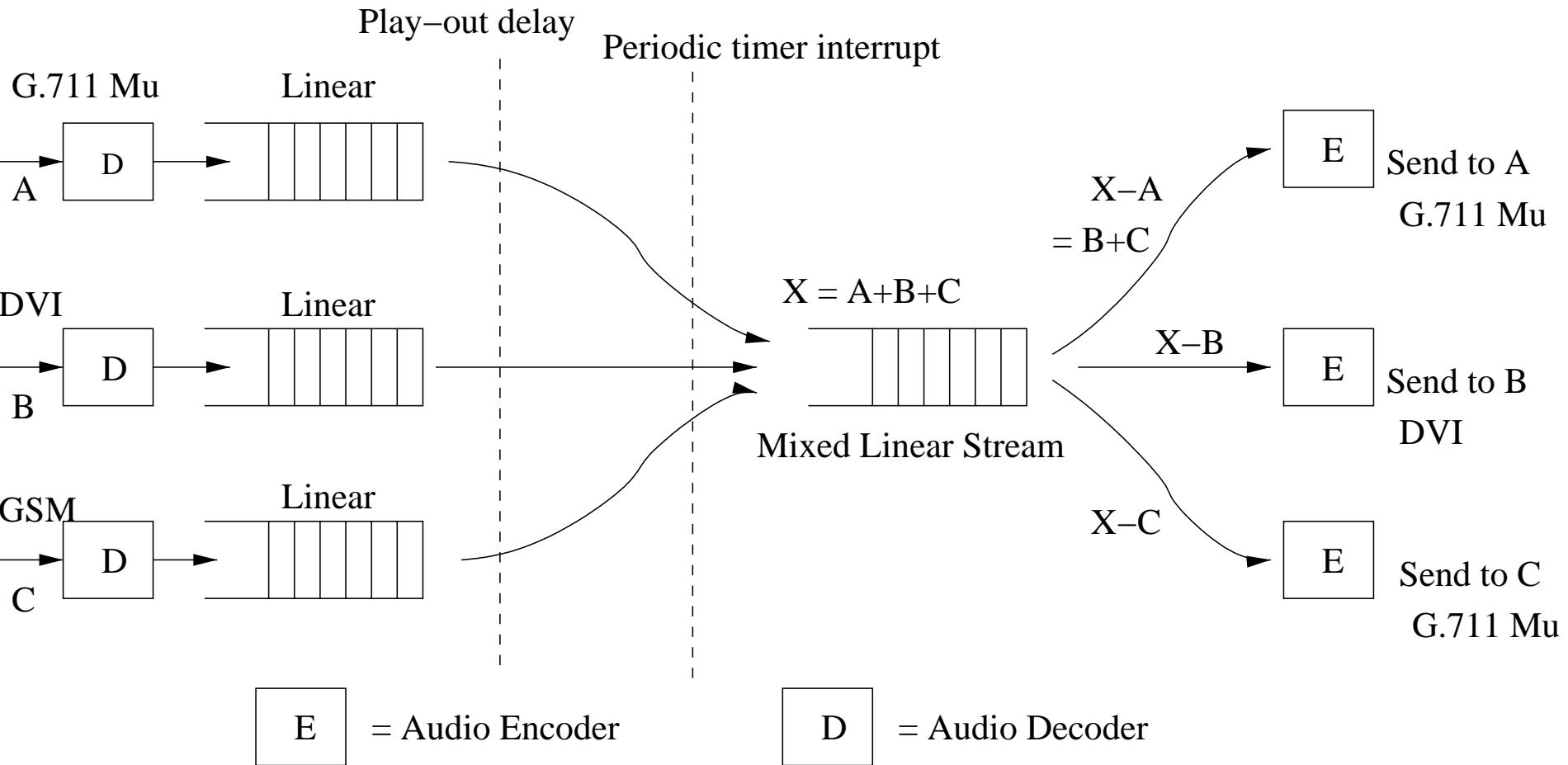
Central server

- conference identified by SIP URL, e.g., `staffmeet@cs.columbia.edu`
- simple end points
- centralized control
- ad-hoc conferences useful for three-party calls
- can create ad-hoc conference, e.g., `sip:letsmeet-adhoc@a.servers.com`, use **REFER** to get others to add themselves to that conference
- for ad-hoc, conference lasts until last one leaves

Columbia sipconf conferencing server

- central-server model for mixing
- mixes audio streams, replicates RTP (and UDP) streams
- audio (G.711, DVI, GSM) - others can be easily added
- works for video and text chat - packet replication

Mixing heterogeneous streams



Example: Columbia software conference server

Conference List

Conference URL (Click to edit)	Description	Duration	Participants		Media
demo	Demonstration of sipconf audio conference server.	always on	anyone can join	No restrictions on number of participants	audio
test	Testing	always on	restricted conference digest authentication required	Maximum 10 participants allowed	audio video

Click [here](#) to setup a new conference.

Example: Columbia software conference server

Conference URL :
(name or name@domain)

Password:

Description:

Start time:
YYYY-MM-DD HH:MM

End time:
9999-12-31 00:00 for never ending

Authentication: none

Conference Type: public

Participant List Type: public Click here for [participant list](#)

Max number of participants:

Supported media type: audio video chat

[Create New](#), [Update Existing](#), or [Delete Existing](#).

Example: Columbia software conference server

Conference Participants List

To add a new entry, update the last row To delete an entry simply clear the participant edit box and "Update"

Participants can be of the form
 wild-card for any address -- *
 hostname or domain or IP address -- @cs.columbia.edu
 userid of the participant -- kns10@
 userid and domain (preferred to userid) -- kns10@cs.columbia.edu
 a SIP URI as needed in From header -- sip:kns10@domain.net

Conference URL (Click to edit)	Participant	Privileges	Type	Status	Audio	Video	command
test	kns10@cs.columbia.edu	<input type="checkbox"/> send <input type="checkbox"/> receive <input type="checkbox"/> admin	dialin <input type="button" value=""/>	notconnected <input type="button" value=""/>	send-receive <input type="button" value=""/>	send-receive <input type="button" value=""/>	Update <input type="button" value=""/>
test	@cs.columbia.edu	<input type="checkbox"/> send <input type="checkbox"/> receive <input type="checkbox"/> admin	dialin <input type="button" value=""/>	notconnected <input type="button" value=""/>	send-receive <input type="button" value=""/>	send-receive <input type="button" value=""/>	Update <input type="button" value=""/>
test	#	<input type="checkbox"/> send <input type="checkbox"/> receive <input type="checkbox"/> admin	dialin <input type="button" value=""/>	notconnected <input type="button" value=""/>	send-receive <input type="button" value=""/>	send-receive <input type="button" value=""/>	Update <input type="button" value=""/>
test	~	<input type="checkbox"/> send <input type="checkbox"/> receive <input type="checkbox"/> admin	dialin <input type="button" value=""/>	notconnected <input type="button" value=""/>	send-receive <input type="button" value=""/>	send-receive <input type="button" value=""/>	Update <input type="button" value=""/>

Design and implementation issues

- Packetization time
- Scalability
 - server farm
 - multi-stage servers
 - dedicated hardware
- inactivity detection
- multi-protocol server

Performance for single conference

PARC Ultra 10 with 350 MHz CPU:

Participants	CPU (%)	memory (MB)	bandwidth (Mb/s)	
			inbound	outbound
2	< 0.1	2.7	0.08	0.07
20	< 1	6.0	0.08	1.37
40	2-3	9.6	0.08	2.81
60	5	13	0.08	4.25
80	10-15	17	0.08	5.69
100	35-50	22	0.08	7.13
120	50-70	26	0.08	8.59

Performance for three-party conferences

Good quality up to 15 conferences

Confer- ences	partici- pants	CPU (%)	memory (MB)	bandwidth (Mb/s)	
				inbound	outbound
3	9	< 0.4	4.1	0.72	0.65
6	18	< 2.0	5.7	1.44	1.30
9	27	7-13	7.3	2.16	1.94
12	36	15-20	9	2.88	2.60
15	45	25	10	3.60	3.24
18	54	30	12	4.32	3.89

Centralized conferencing - conclusion

- need many different models of conferencing
- as long as no multicast, central server is good for medium to large scale
- trade-off infrastructure vs. complexity vs. scaling
- can be handled by existing SIP mechanisms. Works with H.323 also.