SIP for Telephony

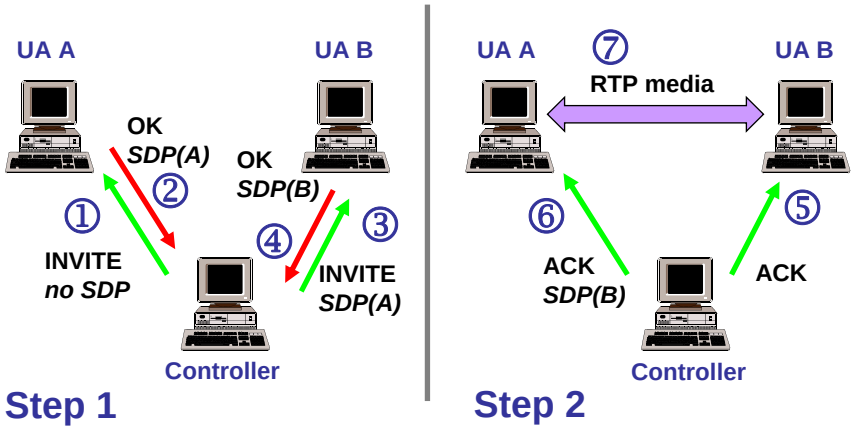
…yet another set of SIP services…

"Supplementary Services"

- The easy ones first
  - Call Diversion
    - Intrinsic support (301/302 redirection)
  - Call Forwarding (unconditional, busy, no answer)
    - Performed by proxy and indicated by means of 181 Call Is Being Forwarded response
  - Call screening (for incoming and outgoing calls) at proxy
    - Respond with 403 Screening Failure when needed
  - Call Waiting
    - Implemented in endpoints
- More sophisticated SIP signaling required for
  - 3rd party call control
  - Call transfer
  - Call park and pickup
  - Message waiting
  - PSTN (ISUP) and PBX (QSIG) interworking
  - Conferencing

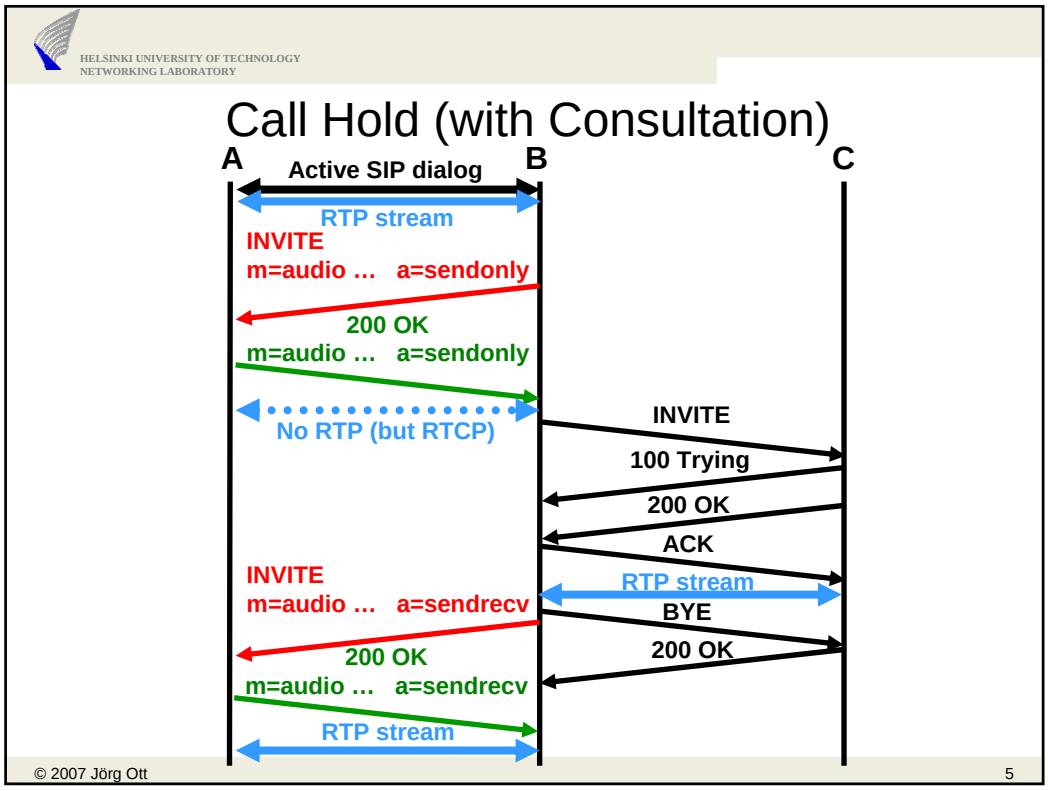
Third-Party Call Control (3PCC)

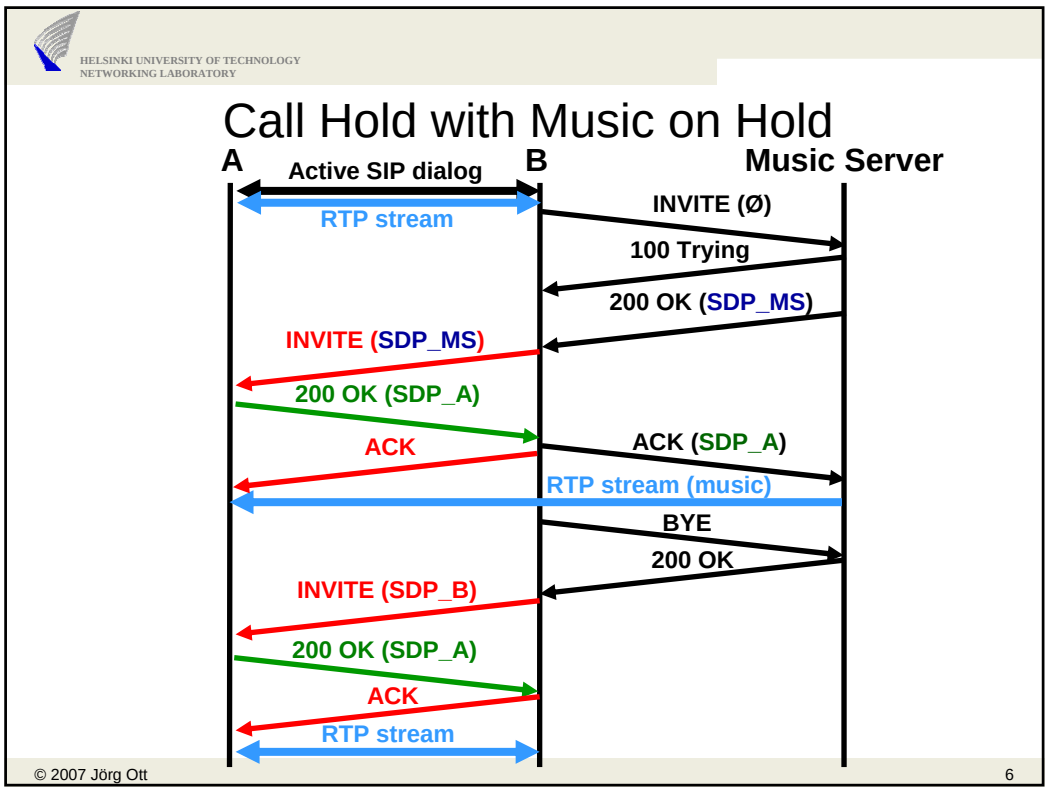
- Examples: “Click-to-dial”, conference bridge control, …
- Several approaches with different advantages / drawbacks
- Simplest call flow:



Call Hold and Retrieve

- Call Hold and Retrieve
  - Media-level operation only (does not affect SIP call state!)
  - User agent sends re-INVITE to mute other party (a=inactive)
    - Repeated offer/answer exchange
    - “Hold” condition may apply into one direction only
    - a=sendonly indicates that the initiator will not pay attention to incoming media
  - Another round of re-INVITEs to re-establish media (a=sendrecv)
- Call Hold with Music on Hold
  - Create a separate dialog with music server
  - Connect the peer to the music server by forwarding SDP
    - Direct music stream to the peer on hold
    - Actually: UA putting peer on hold acts as third-party call controller
  - When taking user off hold, close the dialog to music server
    - Redirect peer to talk to local UA again

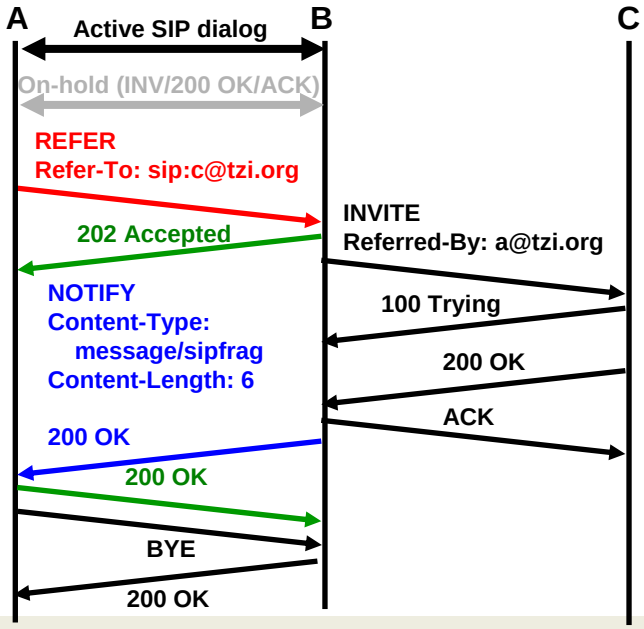




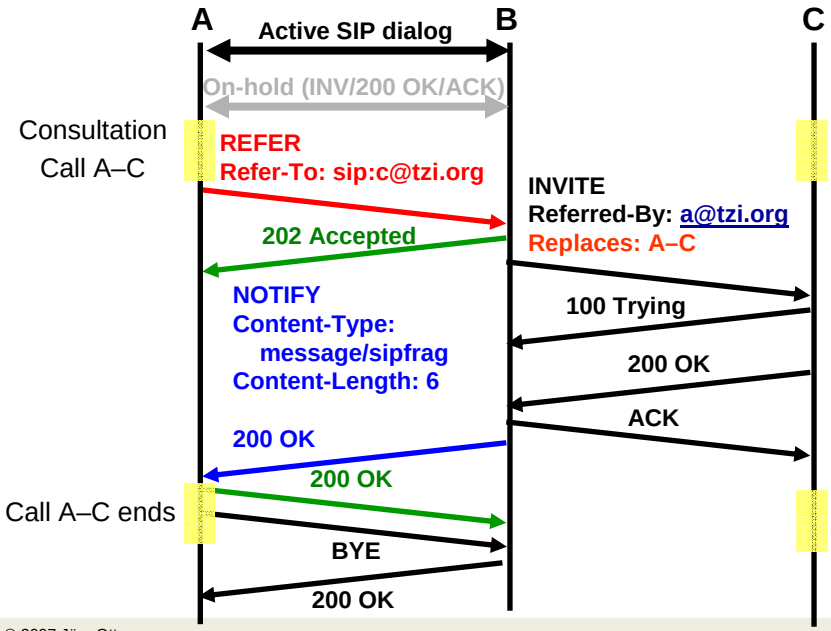
Call Transfer

- Part of Call Control Framework
- Uses basic SIP protocol features and extensions
  - REFER method to invoke another (INVITE) transaction
  - NOTIFY (with implicit subscription) to indicate success or failure
  - New Replaces: header to indicate substitution of an existing call
  - Refer-To: header to indicate to peer which target to contact
  - Referred-By: header to inform refer target about origin of call
- Supports numerous variants
  - Attended
  - Unattended
  - Intermediate three-way calling
  - Optional protection of transfer target

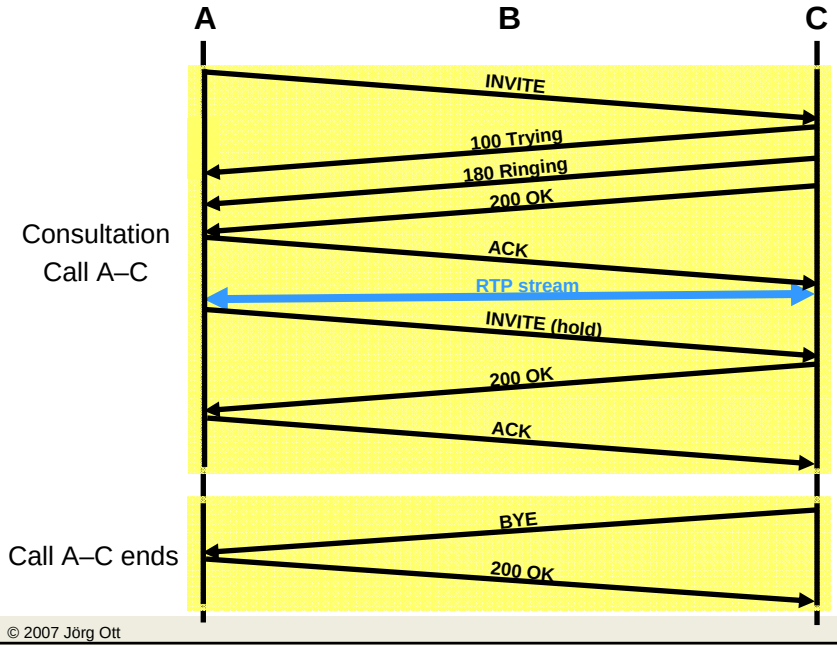
Simple Unattended Transfer



Attended Transfer



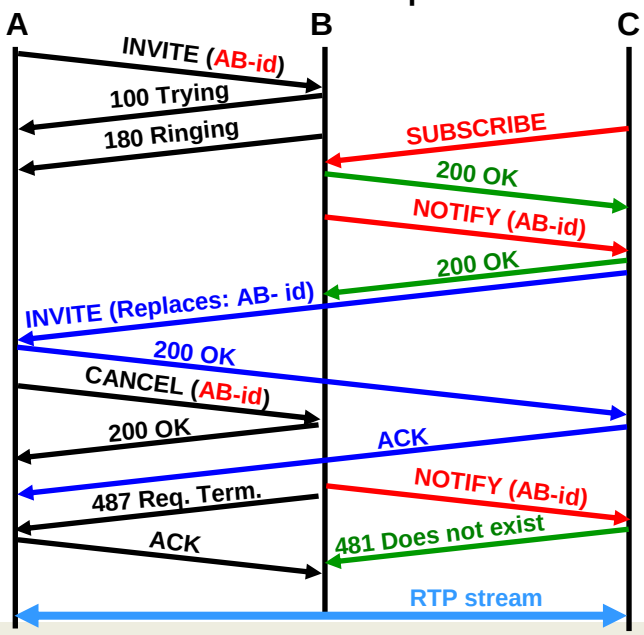
Attended Transfer



Call Park & Pickup

- Park a call from one endpoint, pick up from another
  - Requires remote URI and session identification
  - Park on the endpoint itself
  - Park elsewhere (refer to some “parking area”, e.g. a SIP server)
  - For pickup, endpoint contacts remote URI and provides session identification
    - Use Replace: header to substitute (early) dialog
  - Pickup UA needs to learn about dialog-specific information
    - Use SUBSCRIBE/NOTIFY with dialog event package
    - Other alternatives for shared state conceivable, too
- Pick up a call destined for another phone
  - Common PBX feature, e.g., in working groups

Call Pickup



Dialog Event Package

- Event package for INVITE-initiated dialogs (“dialog”) at endpoints
  - Entities authenticated with same AoR as the endpoint
- XML-based extensible format
- Parameters
  - call-id:, from-tag:, to-tag: Uniquely identifies a SIP dialog
  - with-sessd:
- Contained elements
  - State (“early”, “trying”, “proceeding”, “terminated”, “confirmed”)
    - possibly with event and reason parameter
  - Duration (since creation of the dialog state machine)
  - Referred-By and Replaces
  - Peers of the dialog (“local” and “remote”)
    - Identity, target (SIP URI), SDP session description
- NOTIFY message body MIME type: application/dialog-info+xml

Dialog Event Package

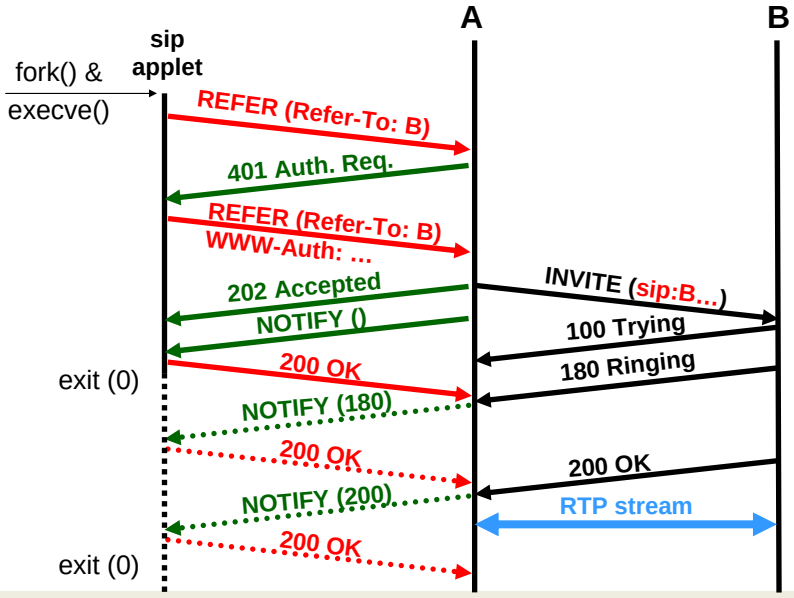
- Event p
  - Entiti
- XML-ba
- Parame
  - call-i
  - with-
- Contain
  - State
    - p
  - Dura
  - Refe
  - Peer
    - Id
- NOTIF

```
<?xml version="1.0" encoding="UTF-8"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
 xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:dialog-info"
  version="1" state="full">
  <dialog id="123456">
     <state>confirmed</state>
     <duration>274</duration>
     <local>
       <identity display="Alice">sip:alice@example.com</identity>
       <target uri="sip:alice@pc33.example.com">
         <param pname="isfocus" pval="true"/>
         <param pname="class" pval="personal"/>
       </target>
     </local>
     <remote>
       <identity display="Bob">sip:bob@example.org</identity>
       <target uri="sip:bobster@phone21.example.org"/>
     </remote>
  </dialog>
</dialog-info>
```

Further Supplementary Services

- Automatic redial ("call completion on busy subscriber")
  - Again: use dialog event package
  - Calling UA subscribes to dialog state of called party
  - Caller is notified when the call state at the called party changes
  - Calling UA can automatically re-invoke call setup (i.e., send INVITE)
- Click-to-dial
  - Server-side implementation
    - Client needs to provide its own URI and is then called back
    - E.g., by means of third party call control on server side
  - Client-side implementation
    - Plug-in for web browser (linked to "sip:" and "sips:" as URI schemes)
    - Configured with local IP phone (or 3PCC server)
    - Sends REFER message to IP phone with target SIP URI in Refer-To: header
    - May require explicit user confirmation (risk of malicious "dialer" pages otherwise)

Click to Dial

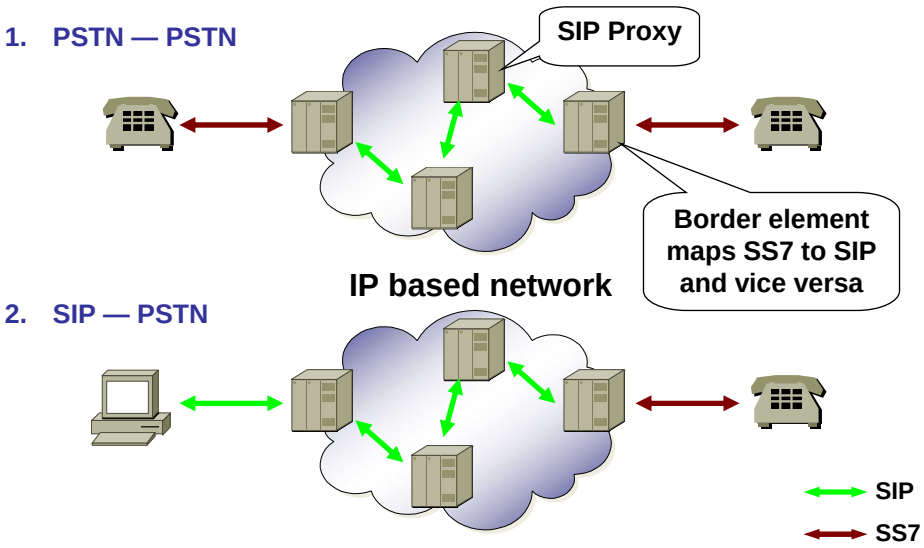


Message Waiting Indication

- Asynchronously notify endpoint(s) about messages
  - Voice, fax, pager, multimedia, text (per RFC 3458)
- Defines a new SIP event package
  - Subscribe to one or more mailboxes
  - Results from many sources may be merged
- Content-Type: application/simple-message-summary
  - General indicator for new messages
  - Message type followed by new/old and (new-urgent/old-urgent)
  - Encoding as plain text
  - May include selected RFC 2822 message headers
- Example

```
Messages-Waiting: yes
Message-Account: jo@tzi.org
Voice-Message: 4/8 (1/2)
Text-Message: 238/42116 (0/1)
```

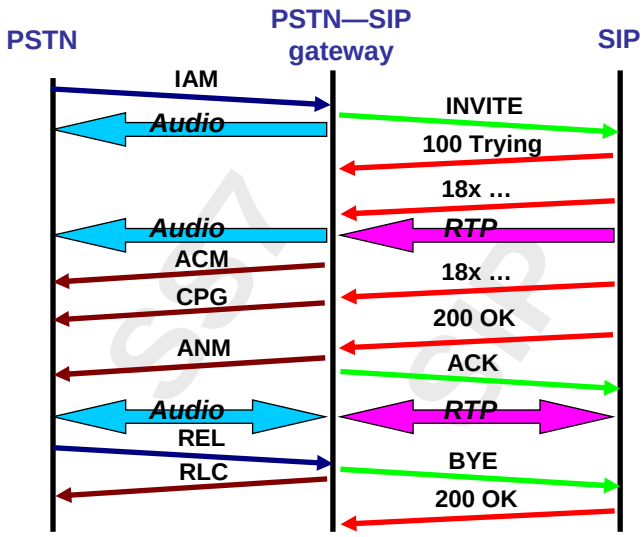

Interfacing to the PSTN



Interfacing to the PSTN

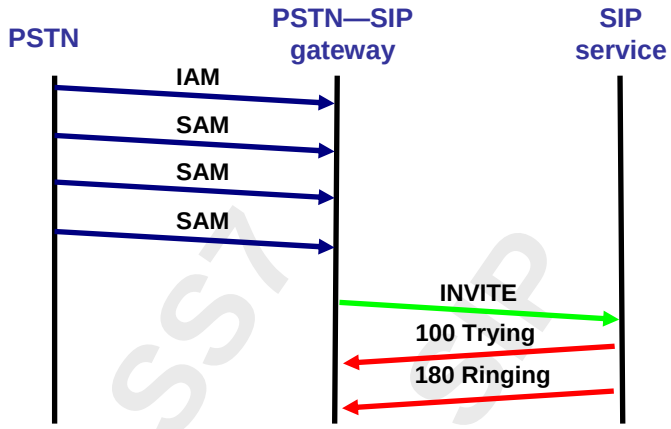
- Preserve feature transparency
  → transport SS7 information (ISUP MIME type)
  - Conversion between different ISUP versions to be done on gateways
- Provide enough routing information to find callee
  → (partially) translate ISUP to SIP
  - Support for tel:-URLs to indicate Called Party Number
    - Address resolution using ENUM or TRIP
- Enable early media (e.g., in-band alerting and announcements)
- Convey additional information during call
  - PSTN – SIP – PSTN case
  → INFO method (RFC 2976)
- SIP to PSTN mapping
  - Call flows for basic interoperation (RFC 3372, 3398, 3666)
  - RFC 3578: Support for *Overlap Sending*

Basic SIP-ISUP Mapping



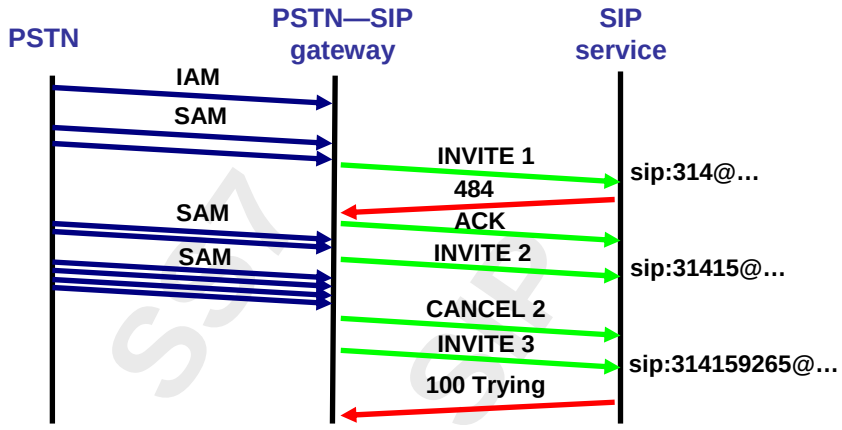
Example for SIP and Overlap Sending (1)

- Approach 1: Collect digits in gateway
  - Use timeout
  - Collect minimal number of digits



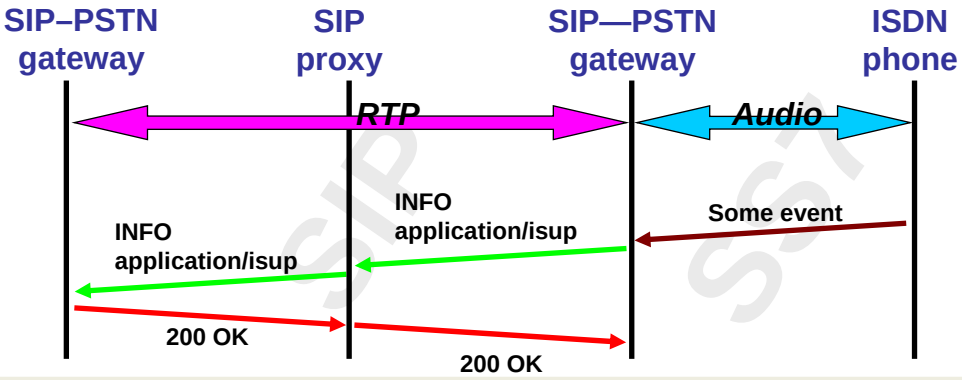
Example for SIP and Overlap Sending (2)

- Approach 2: Send multiple INVITEs
  - Use timeout, possibly collect minimal number of digits
  - Wait for feedback from the SIP network
  - CANCEL obsolete INVITEs



INFO Method

- Transmit application-layer information during call (RFC 2976)
  - Use SIP signaling path of current session
  - Information is carried in message headers or body
    - Specific MIME types defined for, e.g., ISUP (application/isup)
  - No change of (SIP-related) call state



183 Session Progress Message

- INFO not applicable *before* call is established
- ISUP mapping requires inband data prior to final response

→ provisional response 183

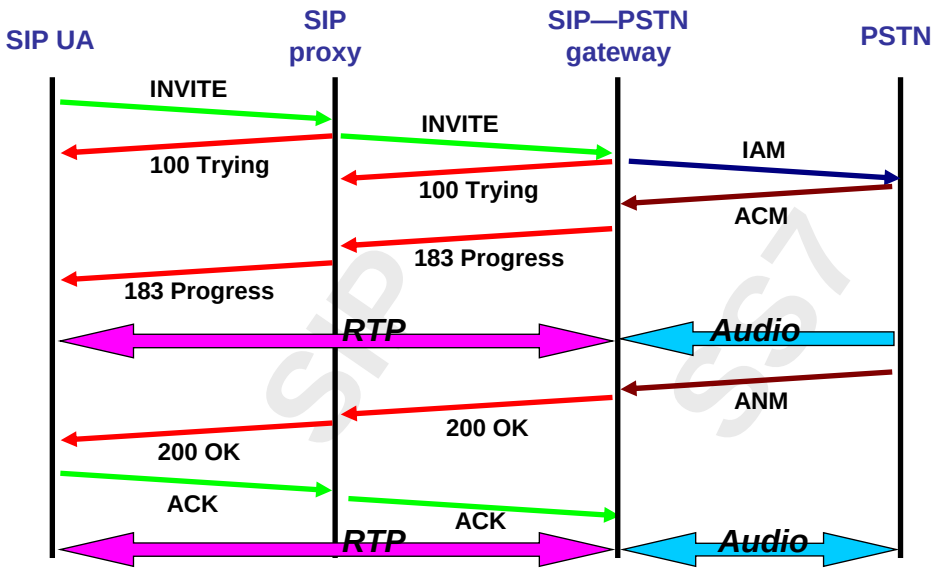
- Additional information in message body
  - SDP description of early media stream (e.g., for announcements)
- Message headers also conceivable

- May be made reliable (100rel)
  - So that sender can count on synchronized state
  - PRACK message to confirm receipt of 183

Early Media Support

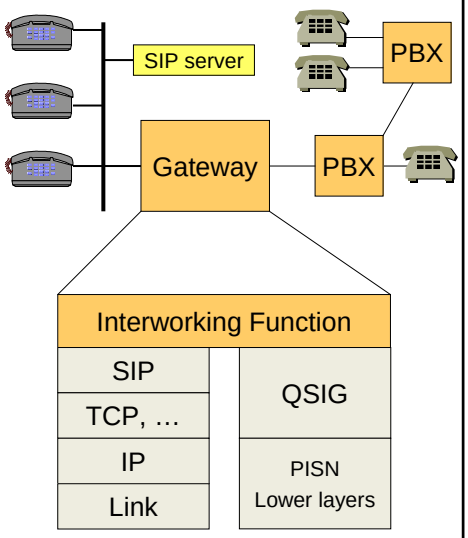
- Early media during call setup (SIP INVITE)
  - One-way transmission to report progress
  - Announcements, specific dial tones, …
  - No charge
  - Inhibit local alerting at calling user agent
- Problem: Media negotiation
  - Send SDP message in provisional response
    → *fast setup*
  - Create SDP from initial INVITE's capability set
    → What if not suitable for early media session?
  - Calling UA will establish *early* media session
    → Cannot decline session or change codecs
  - UPDATE may be used for modifying media session

SIP-PSTN Call With In-band Alerting

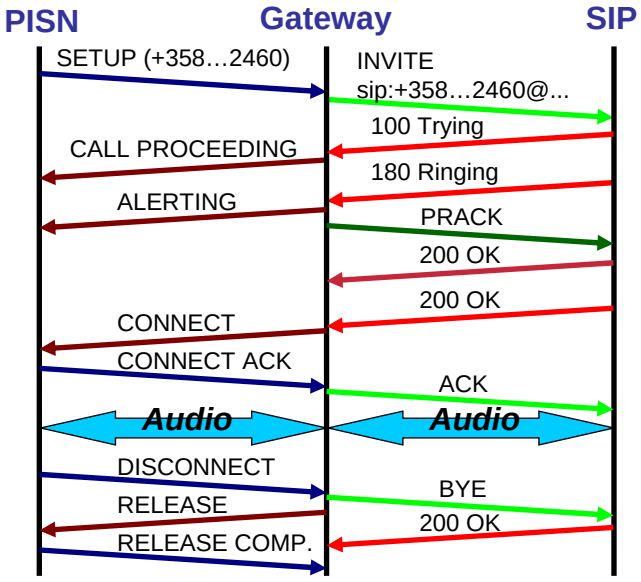


SIP Interworking with QSIG (ISDN PBXes)

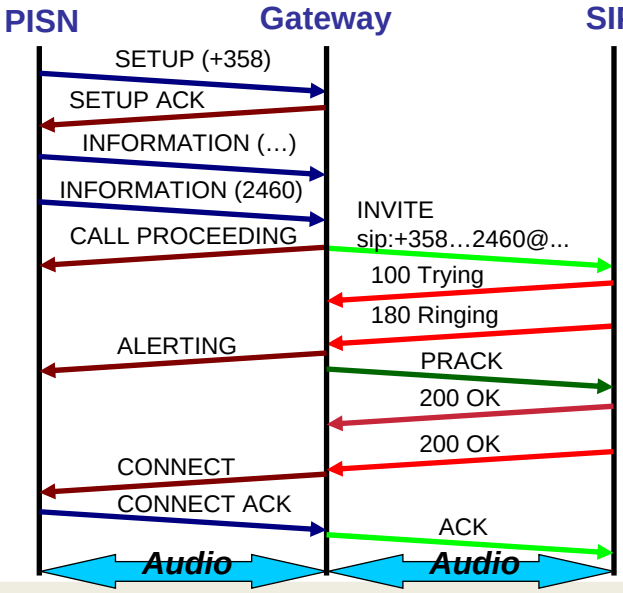
- Interconnecting a SIP-based signaling environment with a traditional (ISDN) PBX
  - Relevant for stepwise migration from traditional to IP telephony
  - First, create SIP islands connected to the PBX
    - Old PBX runs as “master”
  - Then, switch trunk from old PBX to (IP-based) SIP server
    - And run old PBXes as “slaves”
  - Finally, throw away PBX and old phones
- Gateway function needed for semantic mapping between SIP and QSIG
  - Call signaling (stateful operation)
  - Numbering plans
  - Support for “overlap sending”
- Naturally limits available functionality



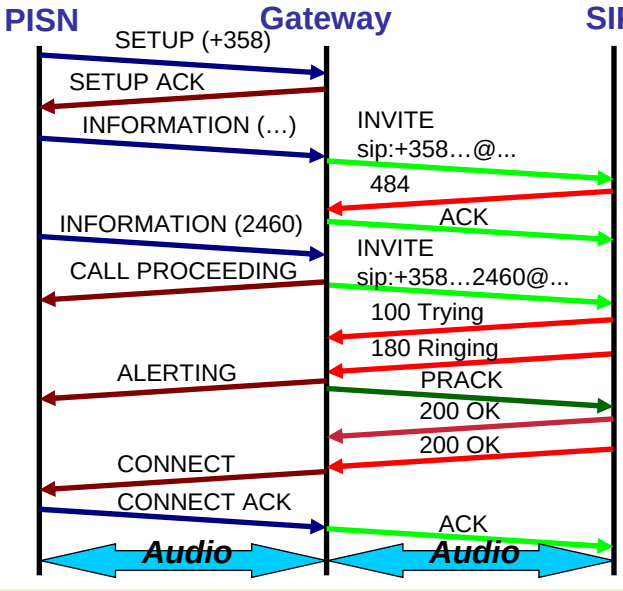
SIP-QSIG: Call Setup (en-bloc dialing)



SIP-QSIG: Call Setup (overlap sending 1)



SIP-QSIG: Call Setup (overlap sending 2)



Quality of Service

- Best-effort quality potentially too poor for IP telephony
  → need support for resource reservation in SIP
- Some applications need multi-phase call-setup
  - Resource reservations to assure certain QoS
  - Establish particular security relationships
    - Extend security services to media channels
    - Avoid fraud and theft-of-service

  → Establish call only if preconditions are met
    - No charge for defect calls
    - No inconsistent state of involved endpoints
      (alert user only if resource reservation succeeded, …)
    - Fallback options: e.g. use codecs with lower bandwith

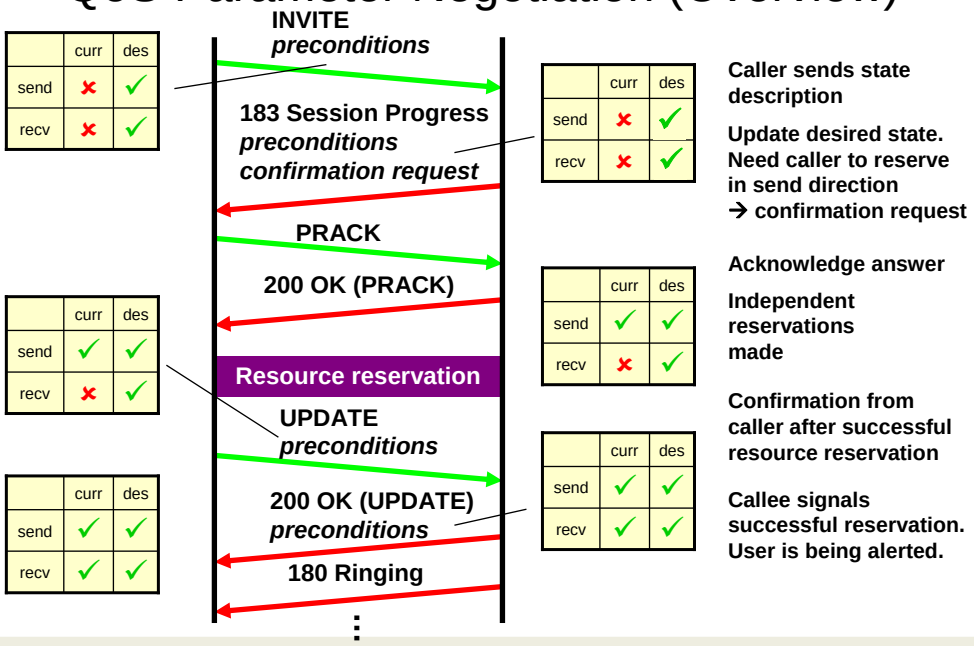
SIP Preconditions (RFC 3312)

- SDP extension handles negotiation of QoS parameters
  - SDP attributes describe preconditions to be met
  - Current status (curr:) vs. desired status (des:)
  - Either party may request confirmation on current state (conf:)

```
'a=curr:' type status direction
'a=des:'  type status strength direction
'a=conf:' type status direction
```

- Offer/answer to create common view on both UAs state
  - Typically INVITE, UPDATE, PRACK
  - Option tag precondition
  - New response code 580 Precondition Failure
  - Mark failed and unknown preconditions in SDP answer

QoS Parameter Negotiation (Overview)



Example Negotiation of QoS Parameters

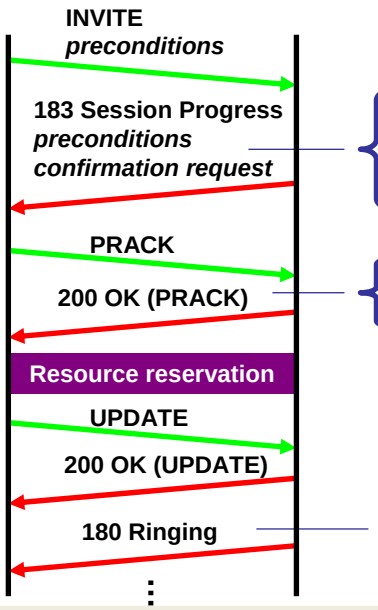
Confirmation request

```
v=0
o=jo 7849 2873246 IN IP4 ruin.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.1
m=audio 52392 RTP/AVP 98 99
a=rtpmap:98 L8/8000
a=rtpmap:99 L16/8000
a=curr:qos e2e none
a=des:qos mandatory e2e sendrecv
m=video 59485 RTP/AVP 31
a=rtpmap:31 H261/90000
a=curr:qos local send
a=curr:qos remote none
a=des:qos optional local sendrecv
a=des:qos optional remote sendrecv
```

Confirmation response

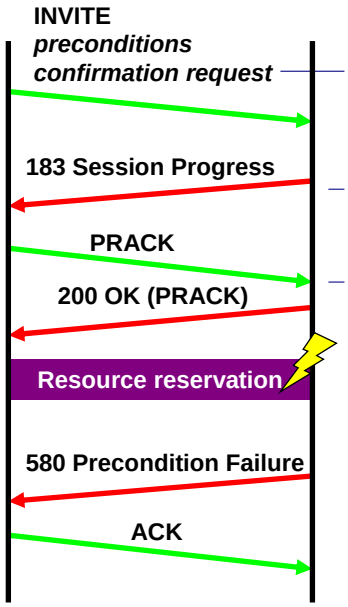
```
v=0
o=cabo 2552 892834 IN IP4 dmn.inf…
s=SIP call
t=0 0
c=IN IP4 134.102.218.46
m=audio 50239 RTP/AVP 98 99
a=rtpmap:98 L8/8000
a=rtpmap:99 L16/8000
a=curr:qos e2e none
a=des:qos mandatory e2e sendrecv
a=conf:qos e2e recv
m=video 56112 RTP/AVP 31
a=rtpmap:31 H261/90000
a=curr:qos local none
a=curr:qos remote send
a=des:qos failure local sendrecv
a=des:qos optional remote sendrecv
```

Successful Reservation



- INVITE indicates support for reliable provisional responses
- Turn off local user alerting
- Choose appropriate subset of preconditions offered by sender
- Include confirmation request
- Reliable provisional response must be acknowledged
- Initiator of confirmation request awaits UPDATE message or CANCEL
- Continue call setup

Failed Reservation



- INVITE contains preconditions with confirmation request
- Indicate support for reliable provisional responses
- Turn off local user alerting
- No reservation results
- Reliable provisional response must be acknowledged
- Callee's resource reservation fails
- Send error response to caller
- Failed call setup complete

Other Preconditions

- Security (“sec”)
  - Users do not want to send unprotected RTP over the wire
  - Negotiate a (set of) crypto algorithm(s) for the media streams and exchange keying material (e.g., using MIKEY)
  - DO NOT ring the phone before the security association is established and data can be encrypted and decrypted
    - Precondition helps to avoid clipping (if one party answers the phone too quickly)
- Connectivity preconditions (“conn”)
  - Endpoints want to ensure mutual reachability for media streams
  - UDP packets
    - DO NOT ring the phone before ICE connectivity checks have passed
  - TCP / SCTP connection establishment
    - DO NOT ring the phone before

SIP Conferencing

- Motivators: n-way calling, video conferencing, …
  - Tightly coupled conferences
- Different conferencing models preserved
  - Except for “fully meshed” conference: complexity just not worth it!
- Terminology
  - Trying to avoid already overloaded terms as much as possible
- Set of protocols
  - Definition of basic building blocks (SIP and other)
  - Sample combinations to implement conferencing services
- Focus on basic SIP model
  - Additional developments in XCON currently in progress

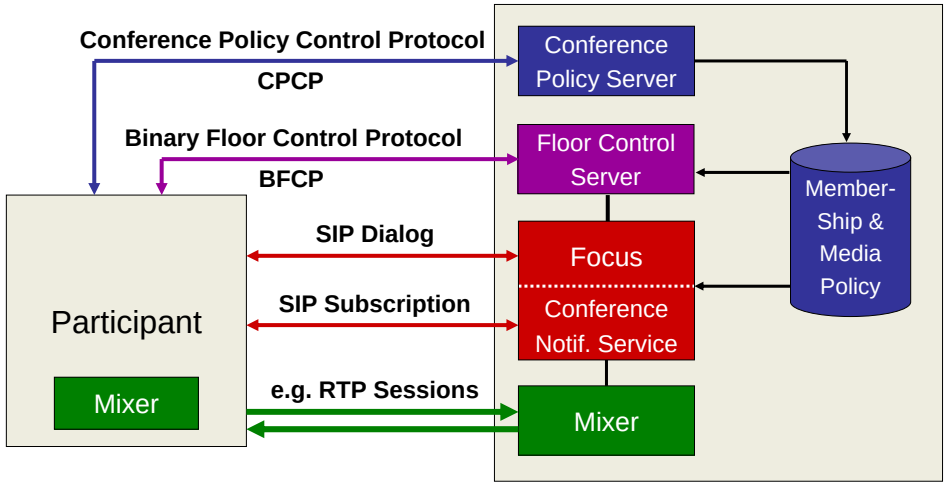
SIP Conferencing

- MUST work with basic SIP support only
  - No awareness of conference
  - Just a point-to-point call with minimal means for control
  - Possibly augmented by out-of-band control (e.g. HTTP)
- Member types (SIP UA)
  - Conference-unaware: plain-old SIP device
  - Conference-aware: supports conferencing features
  - Focus: (one) center of a conference
  - Anonymous: Visible but unidentified participant
  - Invisible: Participant whose presence is not known

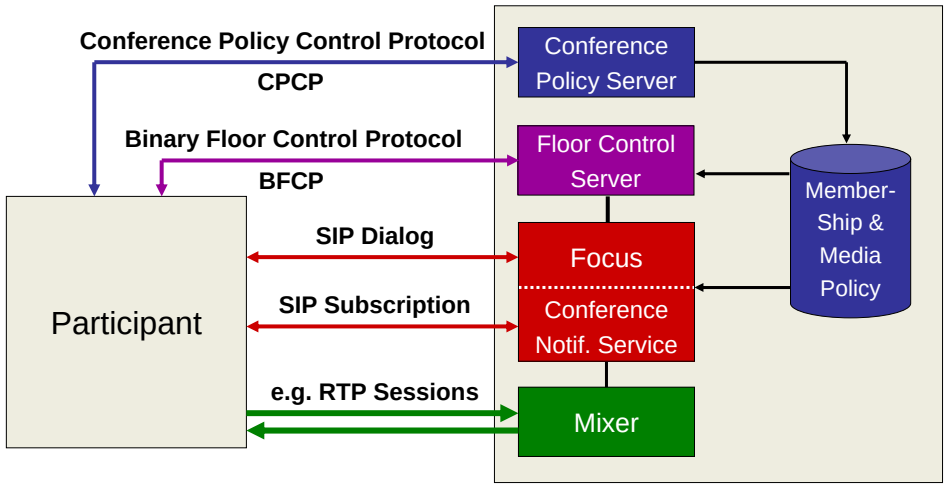
Conferences and URIs

- Conference types
  - Basic: just plain SIP, no further means for control
  - Complex: some conferencing features provided
  - Cascaded: several foci concatenated in a conference
  - Sidebar: conference as (logical) part of another
- Focus — Signaling center of a conference
- Conference URI — Identifies a focus (isFocus parameter may indicate this)
- Factory URI — for automated conference creation
  - Yields a dynamically generated conference URI in return

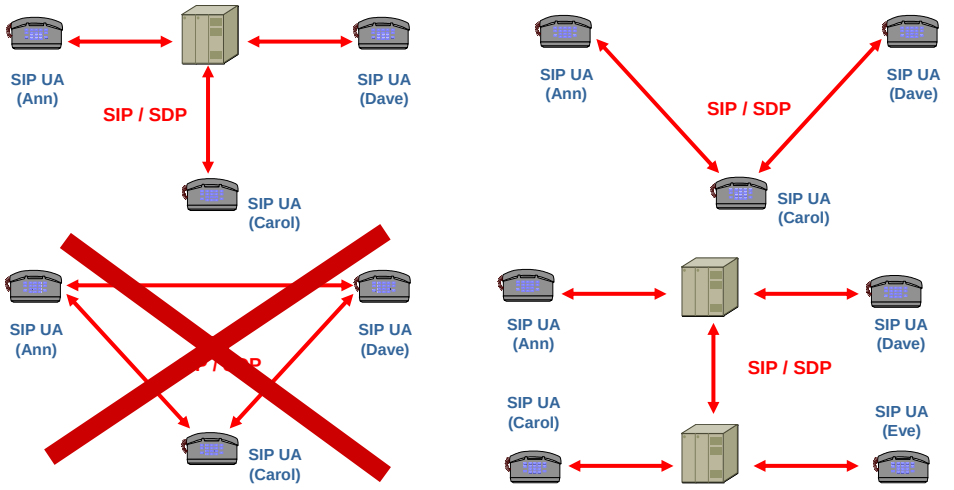
Terminology and Model



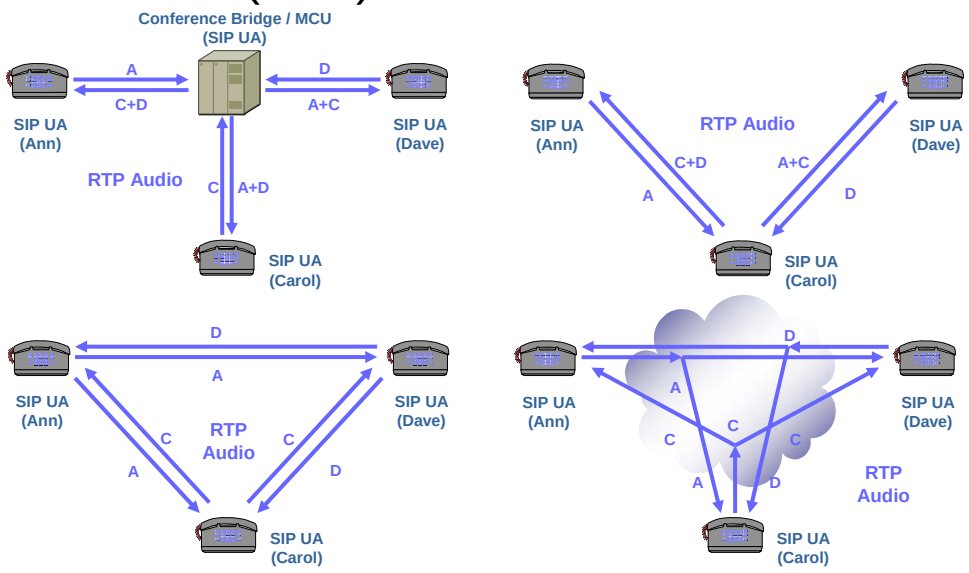
Terminology and Model



SIP Signaling Relationships



(RTP) Media Sessions



Conferencing Scenarios (1)

- Simple conferencing scenarios
  - Plain SIP only (RFC 3261, 3264)
- Extend point-to-point call
  - Works only with local focus; otherwise, new call required
- Ad-hoc conference
  - Automated creation at focus
  - IVR / DTMF for control
  - Audio for information about the conference and its members
- Reserved conference
  - Same as ad-hoc
  - Use external means for reservation and configuration (e.g. web)

Conferencing Scenarios (2)

- Advanced conferencing scenarios:
  - Support for Call Transfer
  - Support means to communicate information from focus to UA
  - Optional: means to manipulate conference and media policy
- Extend point-to-point call
- Join / create a conference based upon an existing dialog
- Ad-hoc conference
- Reserved conference
- Make use of additional conferencing features

Sample Conferencing Features

- Invite participants (dial-in, dial-out), expel participants
- Authenticate new participants by members
- Obtain conference and media policy information
- Manipulate membership/conference policy
  - Participant privileges, participant management (black list, white list)
  - Floor control
- Explicit media control (media policy)
  - Configure media distribution
  - Add / remove media sessions
- Floor control
  - Manage access to conference resources
  - Executed by floor control protocol, governed by floor control policy
- Create, control, and terminate sidebars
  - Separate conference vs. media policy

Membership Policy

Define, retrieve/notify, modify, and act upon…

- Formal rules for the conference
  - Conference creation, termination, (policy) modification
  - Access control: black list, white list, rules for authentication
  - Privileges of individual participants
  - Visibility of the conference and its members
  - Access to floor and media policy (defined separately)
- General conference attributes
- Participation management
  - Invite, expel
- …

Media Policy

- Mixer model
  - Input switch: collecting & selecting input streams from partcipants
    - Possibly transcoding and other per-media functions
  - Mixing topologies describing mixing policies
  - Output switch: selecting & distributing output streams to participants
    - Possibly transcoding and other per-media functions
- Mixer may be centralized or not
- Media policy defines how incoming streams are processed, combined, and then distributed
  - Individual mixing functions may be defined per participant
  - Common mixing functions may be defined for the conference
  - Mixing function may take into account “events” from other components

SIP Signaling Building Blocks

- Membership control
  - Initiation of conferences: INVITE
  - Inviting / adding to conferences: INVITE, REFER
  - Leaving a conference: BYE
  - Expelling from a conference: REFER (method="BYE")
- Conference control
  - State change notifications: SUBSCRIBE / NOTIFY
    - Dialog event package, conference event package, ...
  - Conference / media policy control
    - Might use XCAP data manipulation framework discussed in SIMPLE context
- Other
  - Determine focus URI OPTIONS
- Augmented by other protocols
  - Floor control: BFCP
  - Conference control: XCAP? SOAP? “CCCP”? BFCP-based? …?
  - HTTP/HTML access to a web page for human interaction
    - Conference control, reservation, group management, etc.

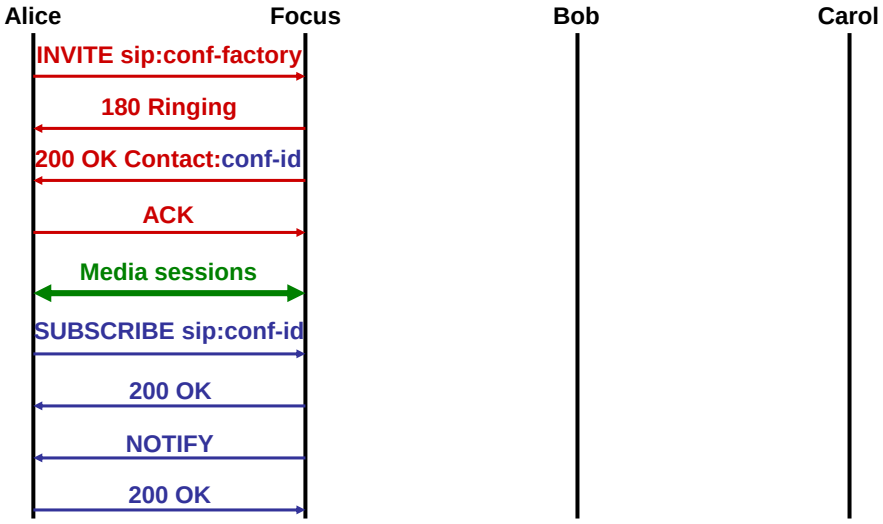
Conferencing Event Package (1)

- Event package for notifying UAs about conference state
  - XML-based encoding of conference status
  - Goes together with dialog state
- Allows minimally enhanced SIP UAs to learn about
  - General conference information (focus URI, state, version)
  - Description (display text, subject, free text, keywords)
    - Conference URIs (for participation and streaming)
    - Service URIs (web page, conference recording, event subscription)
    - Maximum user count
    - Available media (display text, type, status)
  - Host information (display text, lifetime, web page, URIs)
  - Conference state (user count, active, locked)
  - Users (entity == user URI, state)
    - Display text, Associated AoRs, roles, languages, cascaded focus vs. endpoint
    - Endpoint: further detailed status information
- Notification only: other mechanisms for conf. state manipulations

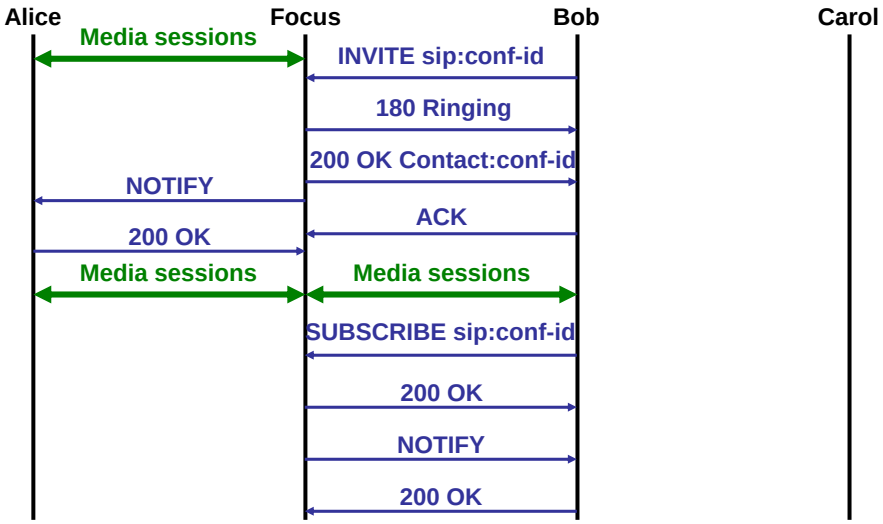
Conferencing Event Package (2)

```
<conference-info version="1" state="partial"
                 entity="sip:3402934234@conf.example.com">
  <users>
    <user entity="sip:carol@chicago.example.com" state="full">
      <display-text>Carol</display-text>
      <endpoint entity="sip:carol@client.chicago.example.com">
        <status>connected</status>
        <joining-method>dialed-out</joining-method>
        <media id="1">
          <display-text>Main Audio</display-text>
          <type>audio</type>
          <src-id>583398</src-id>
          <status>sendrecv</status>
        </media>
        <media id="2">
          <type>video</type>
          <src-id>345212</src-id>
          <status>sendrecv</status>
        </media>
      </endpoint>
    </user>
  </users>
</conference-info>
```

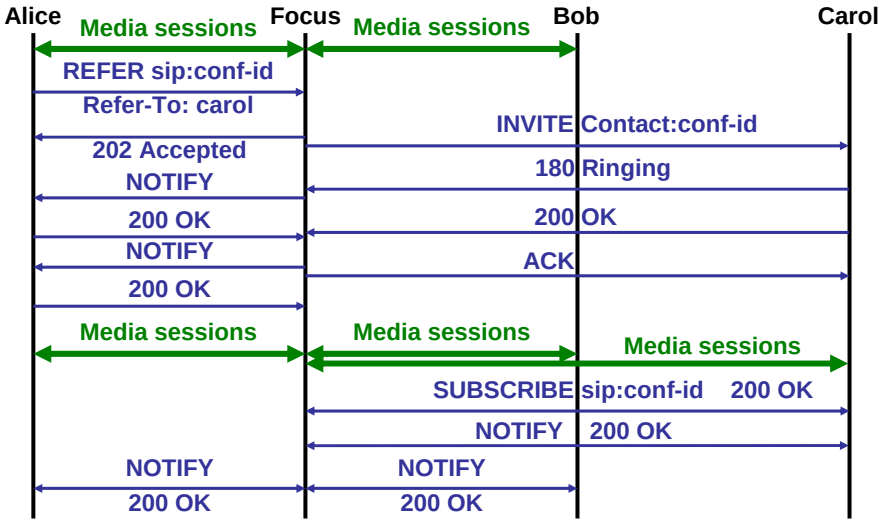
Call Flow Example: Conference Creation



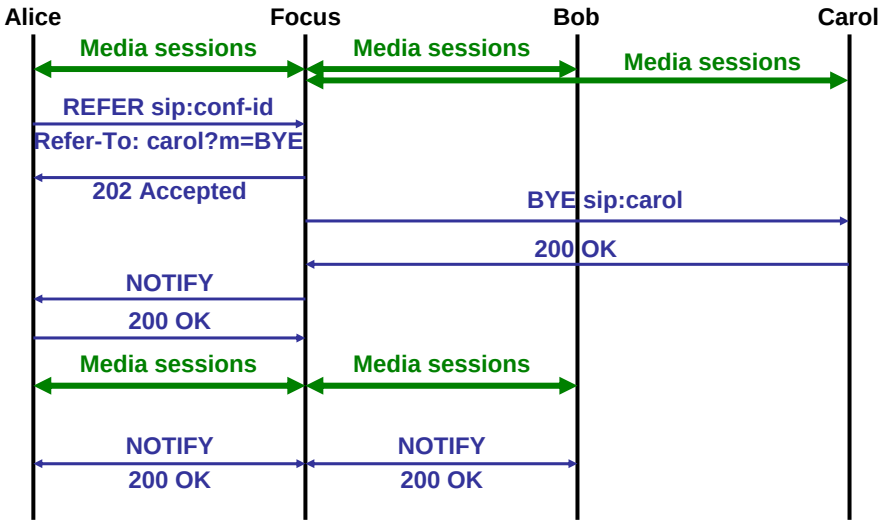
Call Flow Example: User Joining



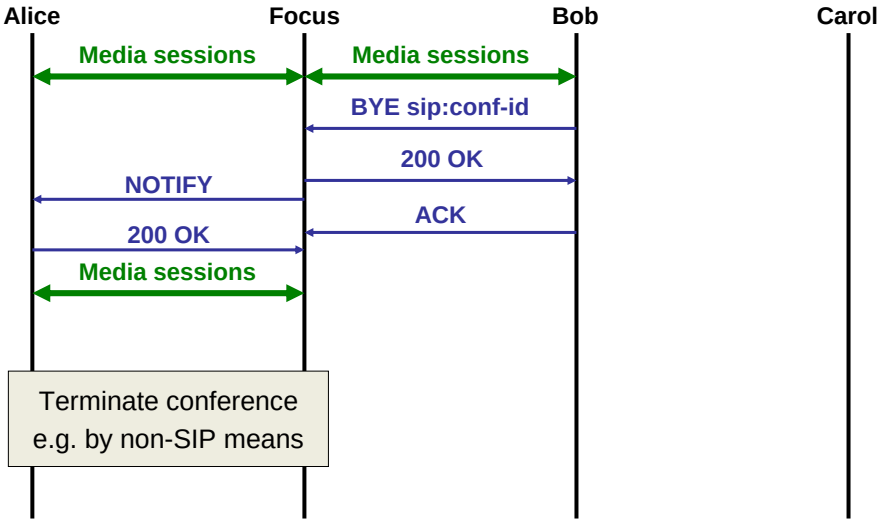
Call Flow Example: Adding a User



Call Flow Example: Removing a User



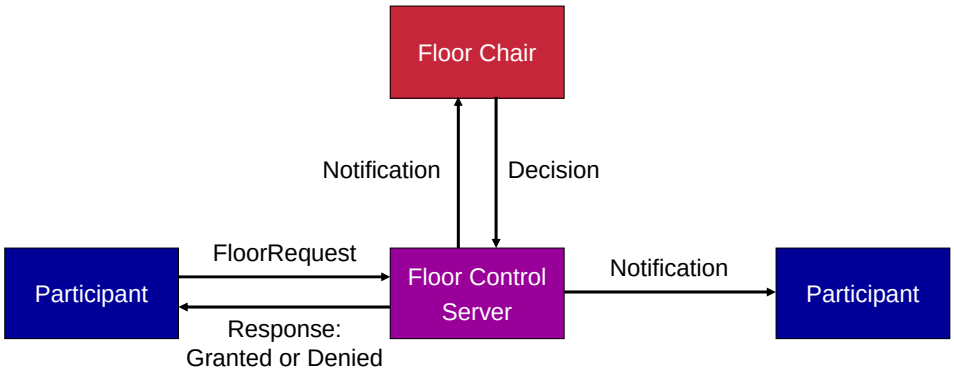
Call Flow Example: User Leaving



Binary Floor Control Protocol (BFCP)

- Support access control to “floors” aka. media sessions
  - Session identification via SDP attributes
  - Shared or exclusive access (equivalent to read or write lock)
    - Limited number of simultaneous floor holders
  - Clients (=participants) issue floor requests or release the floor
    - Requests identify floor(s) concerned and may include hint message
  - Server manages incoming requests and maintain floor state
    - Queuing of requests possible
    - Notifies clients about state changes (granting, revoking floor, floor holder)
    - Governed by some conference policy rules (beyond BFCP scope)
  - Floor chair(s)
    - Manage (grant, revoke, …) floors
    - One or more chairs per floor – one chair for all floors
- Protocol runs on top of TCP or TLS
  - Negotiated via SDP offer/answer (connection-oriented media)
  - Binary TLV encoding
  - Request–response protocol + asynchronous notifications
  - Transaction identifiers for individual operations

BFCP Overview



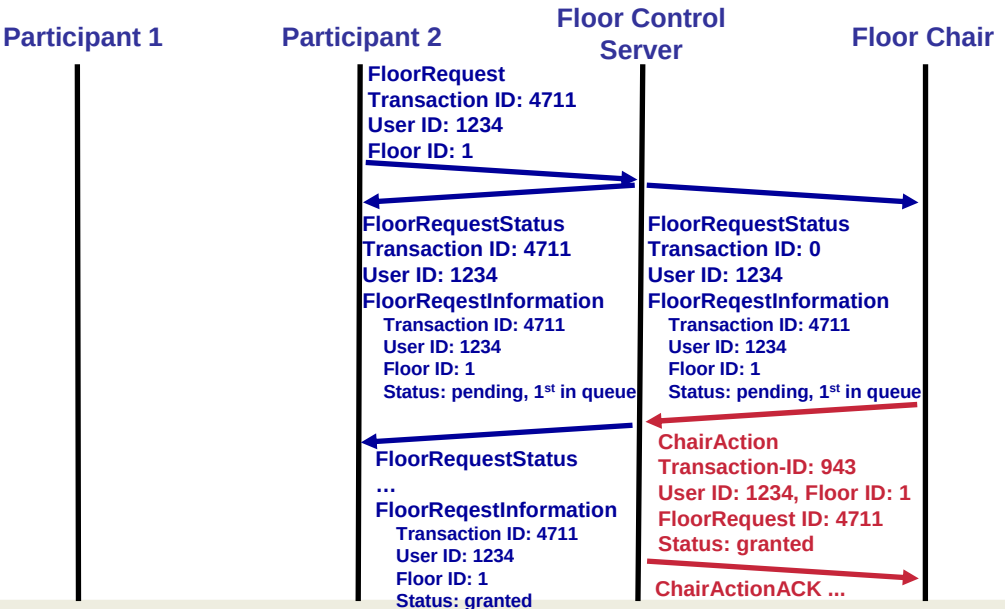
The floor chair may also be co-located with the floor control server.
The floor chair may be an automaton executing a predefined policy.

BFCP and SDP

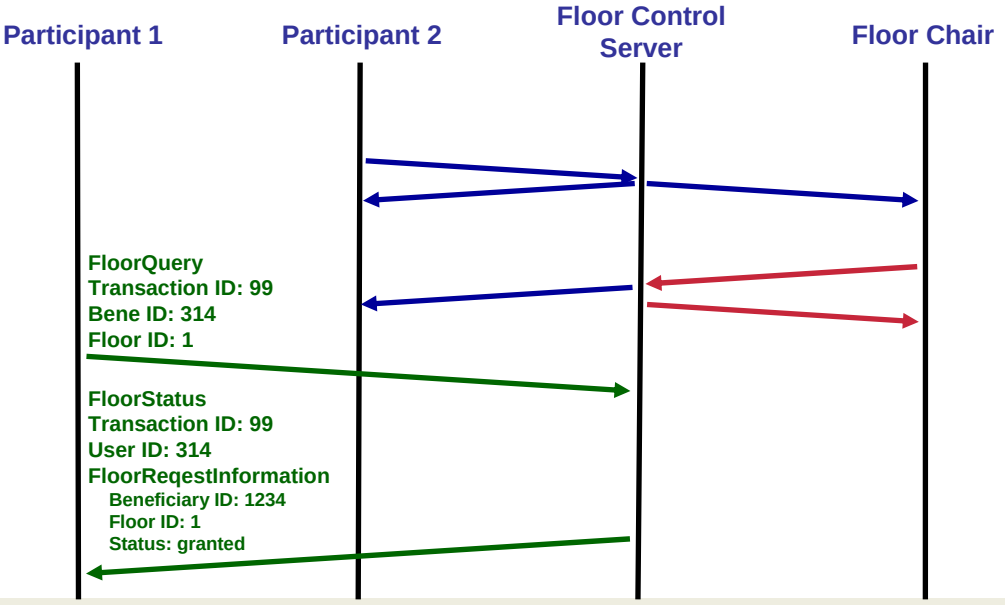
- BFCP negotiation: another media stream
  - Determine connection setup parameters and the floor control server
    - Uses m=application (a=control is not a top-level MIME type)
    - a=floorctrl:c-s (c-only, s-only)
- Media session identification for floor control
  - Media level attributes for floor control
  - Refer to SDP label attribute at media level

```
m=application 50000 TCP/TLS/BFCP *
a=setup:passive
a=connection:new
a=fingerprint:SHA-1 4A:AD:B9:B1:3F:82:18:3B:54:02:...
a=floorctrl:s-only
a=confid:4321
a=userid:1234
a=floorid:1 m-stream:10
a=floorid:2 m-stream:11
m=audio 50002 RTP/AVP 0
a=label:10
m=video 50004 RTP/AVP 31
a=label:11
```

BFCP Example (1)



BFCP Example (2)



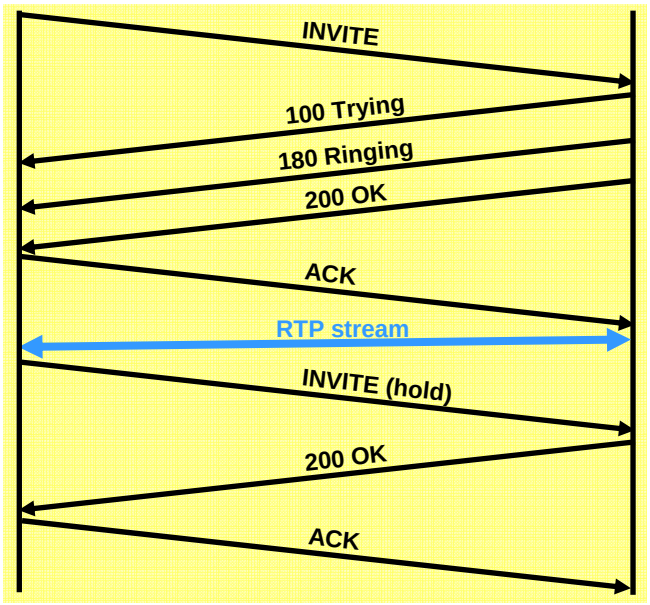
Conference Policy Control Protocol

- In progress…
- Needs to address different aspects
  - Data model for conference and media policy, etc.
  - Protocol for manipulating this data
    - Minimal set of semantics-independent operations on a data structure or
    - Semantic operations that are cause data manipulation (as a side effect)
- Several proposals around
  - Conference State Change Protocol (CSCP)
    - Realized as extensions to the Binary Floor Control Protocol (BFCP)
  - Centralized Conference Control Protocol (“C3P”)
    - XML-based protocol elements; may be carried over various transports (including SIP, SOAP, …)
  - COMP: Conference Object Manipulation Protocol
    - Object manipulation based upon web services + SIP events
  - CCMP: Centralized Conference Manipulation
    - SOAP-based
- Presently under evaluation

SIP & Multimedia

- Text-based conferencing
  - Supported by SDP media descriptions
  - Considerations exist for audio-to-text translation service (and vice versa)
- Video
  - SDP description readily available for negotiation
  - Conference and media policy gain weight (“who sees whom?”)
  - Open issue: time-critical video control “commands”
  - Other remote access features (e.g., camera control) unresolved
- Application sharing (in the very beginning at this point)
  - Old ITU-T T.120: shared whiteboard, file transfer, and generic app sharing
    - Could be launched from within a SIP session
    - Example: Microsoft NetMeeting
  - GUI “remoting”
    - Newly formed Widget Description Exchange (WIDEX) WG in the IETF
  - Some ideas around RTP-based application sharing
    - Immature at this point
  - Other
    - VNC? X Window System-based approaches?

Remaining old fragments



Passing Session State Information

- HERFP: inconsistent information about session status
  - Need to tell initiating UAC before dialog state update
  - "Tunnel" 4xx error response in reliable provisional response

Open issue!

→ 155 Please Update

- Inquires session status update from UAC
  - Reason header (RFC 3326) numeric reason code, SIP error code, SIP error phrase
  - UAC can send appropriate information in following request (e.g. update credentials, SDP capability set, ...)
  - Dialog state may not be affected
- Issue: when to send 155?
  - UAC/user interaction vital
  - Establishing security associations, early media sessions, ...

155 Example: Heterogeneous Error Responses