CommuniGate SYSTEMS


# CommuniGate Pro
# Real-Time Features

# CommuniGate Pro for VoIP Administrators

- Audience: Server Administrators and Developers
- Focus: CommuniGate Pro as the Signaling platform
- Method: Understanding CommuniGate Pro operation for Real-Time Signaling services. “How it works”
- Goal: Learn what CommuniGate Pro can do for you.

# Who is CommuniGate Systems

- Founded
- Communications Software
- Focus on electronic mail and collaboration
- Standards
- Carrier Grade
- Real-Time Communications
- CommuniGate Pro

# What is CommuniGate Pro

- Self-contained single package
- Multithreaded
- Multiplatform
- Flexible
- Extensible
- Not just a product – it's a platform

# Data Storage

- Hierarchical
  - Logically
  - Physically
- Efficient
- Settings, Mail, Metadata, Templates, Middleware, Software

# System Kernel

- Multithreading
- Disk I/O
- Network I/O
- OS interfaces
- Everything Else

# Standard Protocols

- SMTP
- POP3
- IMAP
- ACAP
- LDAP
- HTTP
- FTP
- POPPWD
- RADIUS
- TFTP
- SNMP
- SYSLOG

# Real-Time Protocols

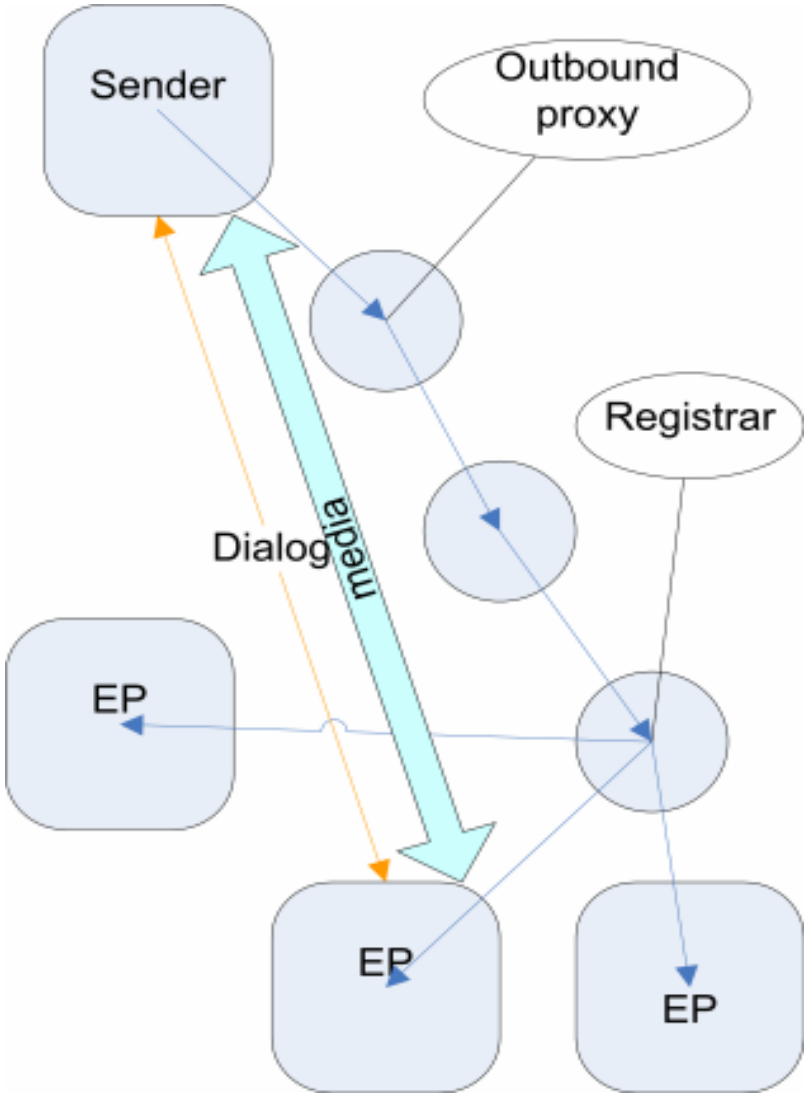
- SIP
- XMPP
- XIMSS
- RTP

# Additional protocols

- CLI
- Helper protocols
- Extended IMAP
- XIMSS

# Real-Time Messaging

- Synchronous
- Messages
  - Requests
  - Responses
- Addresses
- Payloads
  - Short messages
  - Set up direct data stream
  - Signaling separated from media



# Address

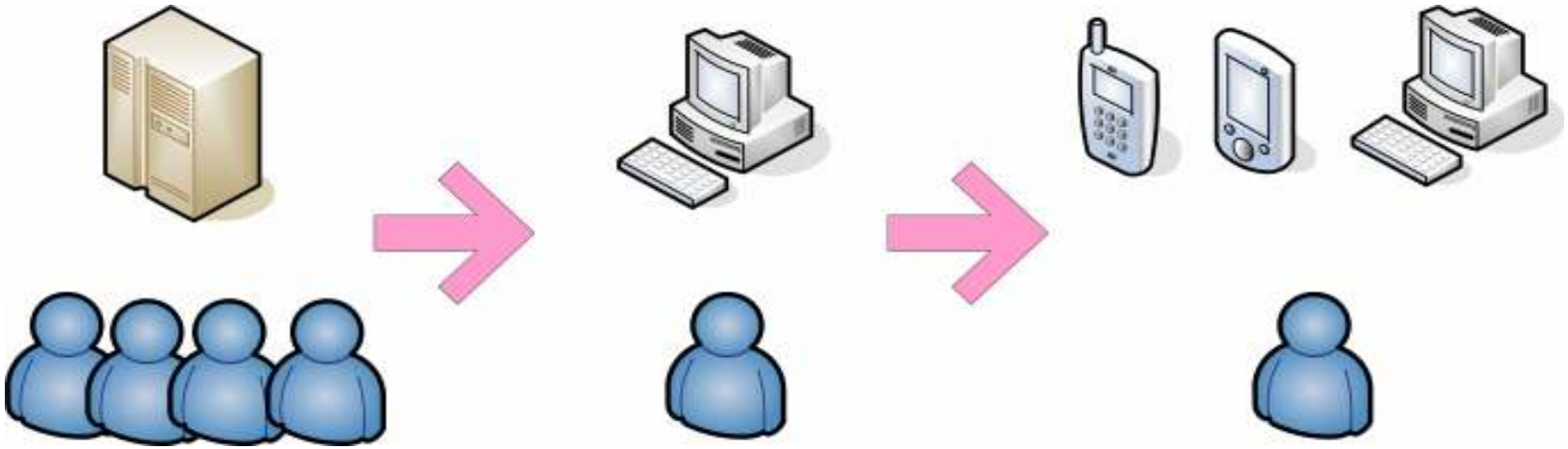
**jsmith@domain.com**

**program#jsmith@domain.com**

**jsmith%domain.com@relay.com.xmppq**

- Local Part
  - Account Name
  - Program Name
  - Special Addresses
- Domain Part
  - A/CNAME-records
  - SRV-records
  - Suffixes

# Address



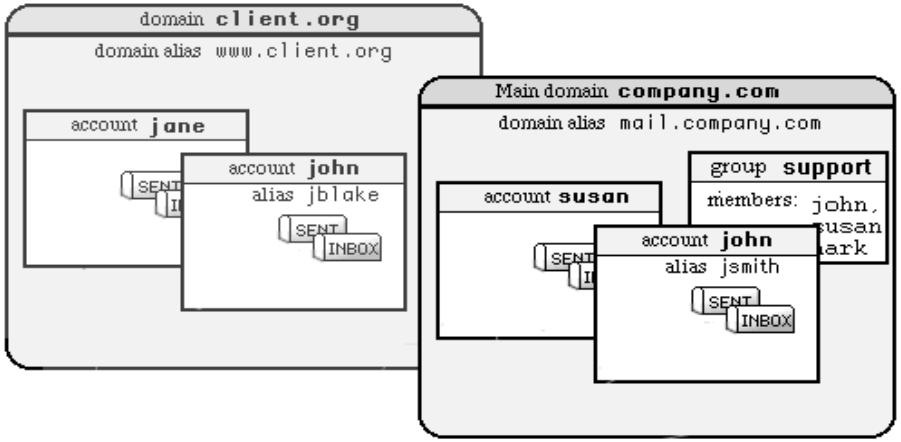
- Ubiquitous computing with real time messaging requires address type separation:
  - AOR
  - End-Point (User Agent)

# Registration

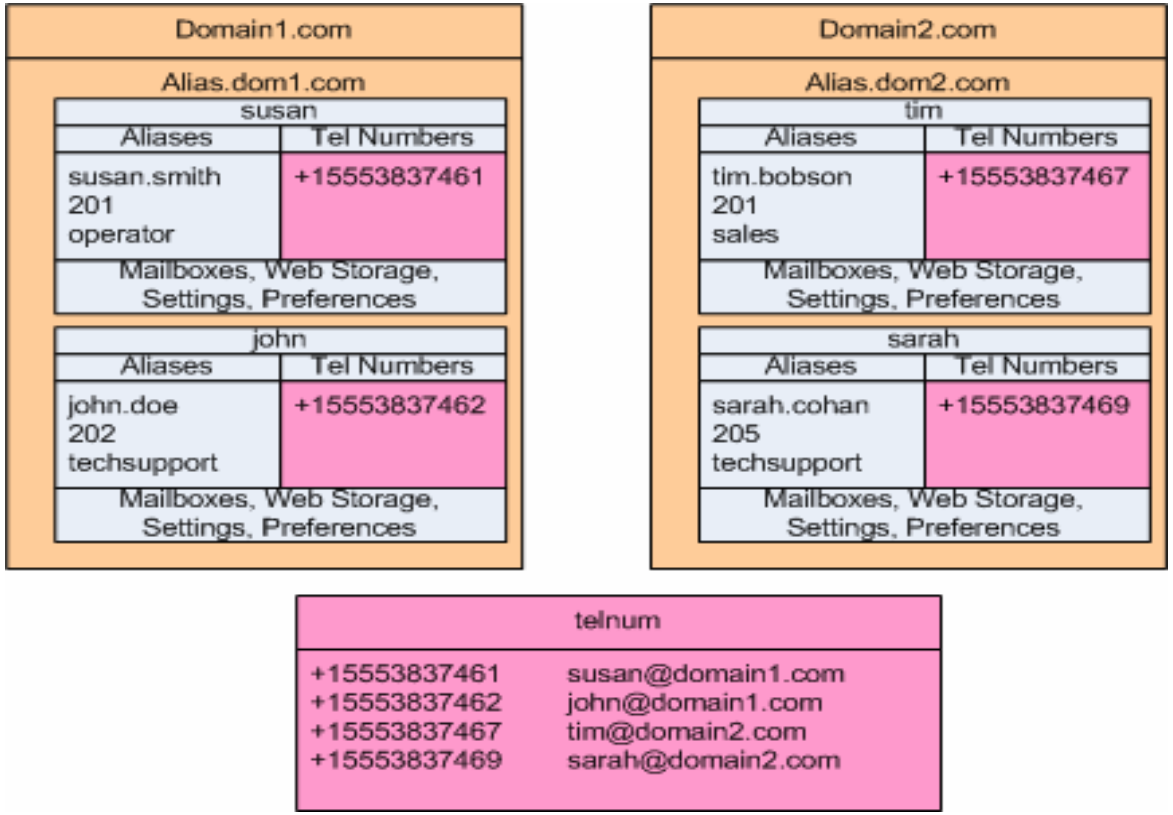
- Necessary only for mapping “Address of Record” to the physical address of available end-points (user agents)
- To receive signals on an EP that should be registered to some AOR.
- Some user agents require registration for making calls.

# Main Data Objects

- Domains and Aliases
- Accounts and Aliases
- Forwarders
- Groups
- Telephone Numbers
- DNS ENUM



# Telnum: System-wide Namespace




- 201
  - 201@dialer.dom
- +15553837461
- 3837461
  - +15553837461
- 03637111298
  - +13637111298

# Address Routing

- Destination – as “Signal”
- Originating – as “Access”
- Authentication – as “Access”
- Globally Unique – through “telnum”

# SIP Server Objects (SIPS)

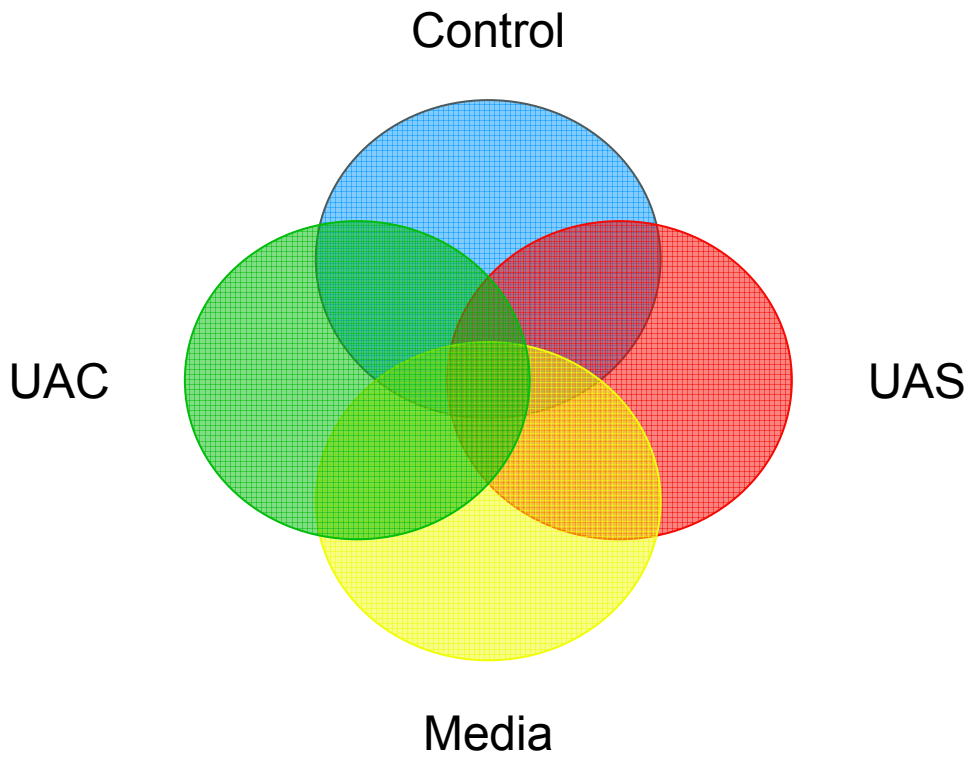
- Parses incoming requests
- Generates intermediate responses
- Creates SIGNAL object
- Sends Provisional responses
- Sends Final responses

# Signal Object

- Real-Time Message
  - Destination Address(es)
  - Attributes (From address, Expiration)
  - Payload
- Has a finite lifetime
- Processed as FSM

# SIP Nodes

- UAC
  - Sends requests
  - Reads responses
- UAS
  - Reads requests
  - Sends responses
- Media
  - Two-way
- Interface



# SIP Client Objects (SIPC)

- Receives SIGNAL objects
- Generates SIP requests
- Processes SIP Transactions
- Processes SIP Responses
- Updates SIGNAL objects

# Common for all Real-Time Objects

- Finite State Machines
- FSM Processors
- FSM Queue
- Events Queue

# SIP Request structure

- Request
- URI
- Headers
  - From/To
  - Call-ID
  - CSeq
  - Contact
  - Authorization
- Payload

- REGISTER
- OPTIONS
- INVITE
- ACK
- CANCEL
- PRACK
- REFER
- UPDATE
- SUBSCRIBE
- NOTIFY
- PUBLISH

# Body/Payload

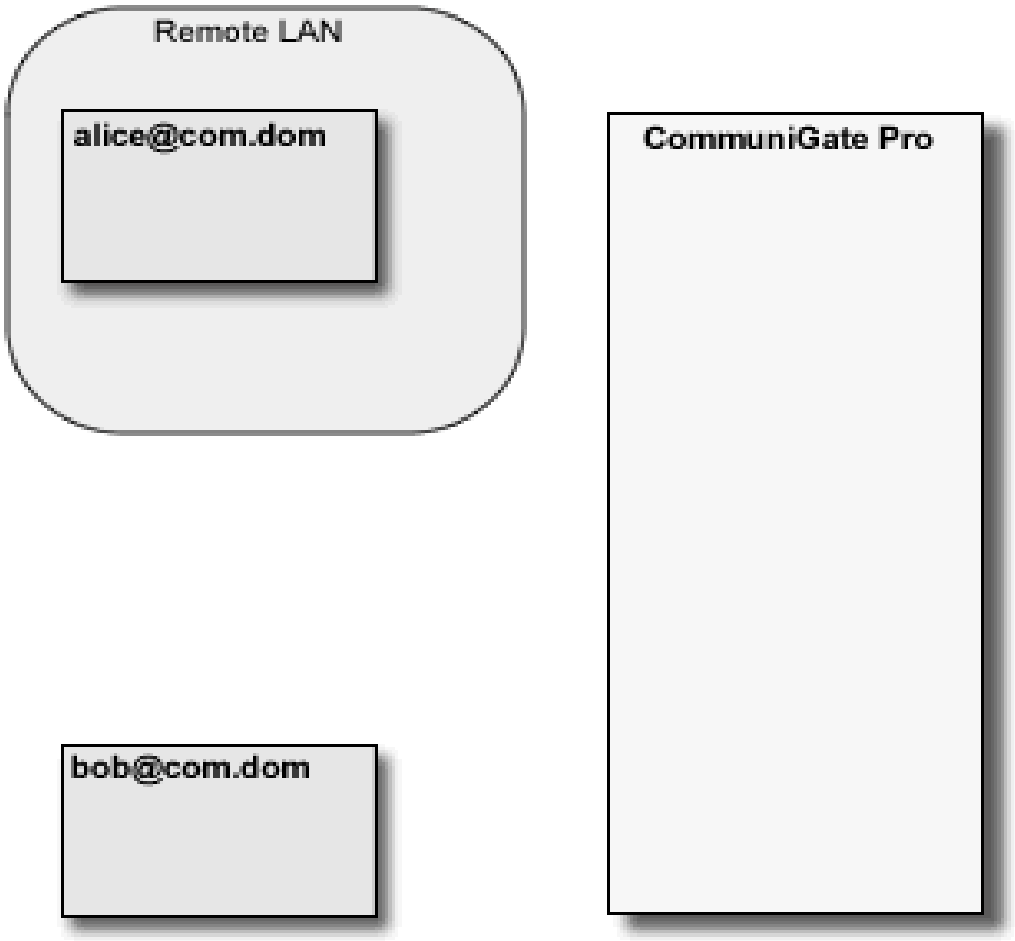
- SDP
  - Audio
  - Video
  - Image
- Presence
- MWI
- Dialog Info
- Anything
  - DTMF
  - ISUP

- Media description
- State aggregation
- Account status
- Dialogs
- Any custom data

# SDP: problems with NAT firewalls

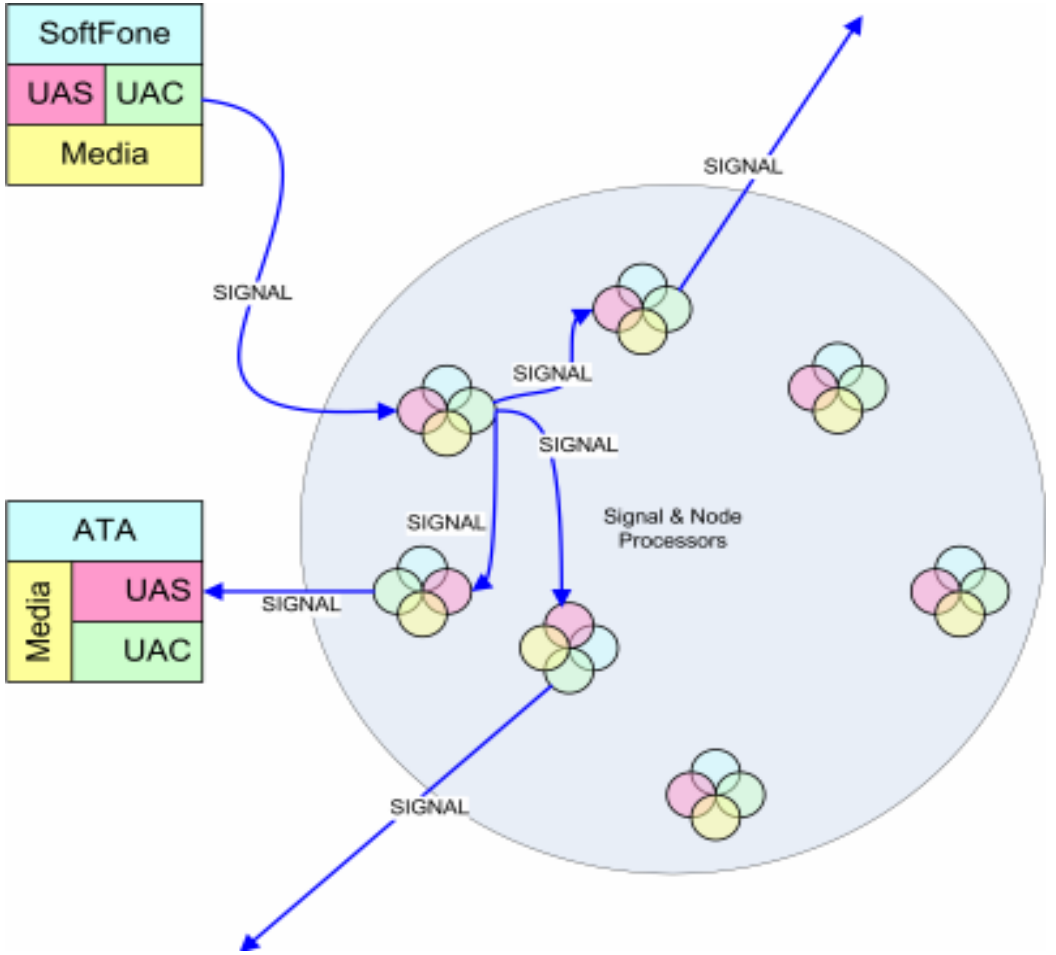
- NAT detection
  - "Via" address mismatch
  - "Gray" IPs in SDP
- Media Proxy
- NAT "Pingers"

# NAT Traversal



# Submitters

- SIP
- XMPP
- XIMSS
- CG/PL
- Core



# SIP implementation

- Stateful SIP Proxy
- SIP Registrar
- SIP Presence server with Presence Aggregation
- Roster management
- B2B User Agents
- SIP is used as internal signaling model

# XMPP Implementation

- Basic XMPP with few Jabber extensions
  - More to come
- Presence
- Roster
- Instant Messages
- Client-Server
- Server-Server

# XIMSS

- Client-Server
- Instant messaging
- Roster
- Presence
- VoIP
- Direct communications with NODEs

# CG/PL

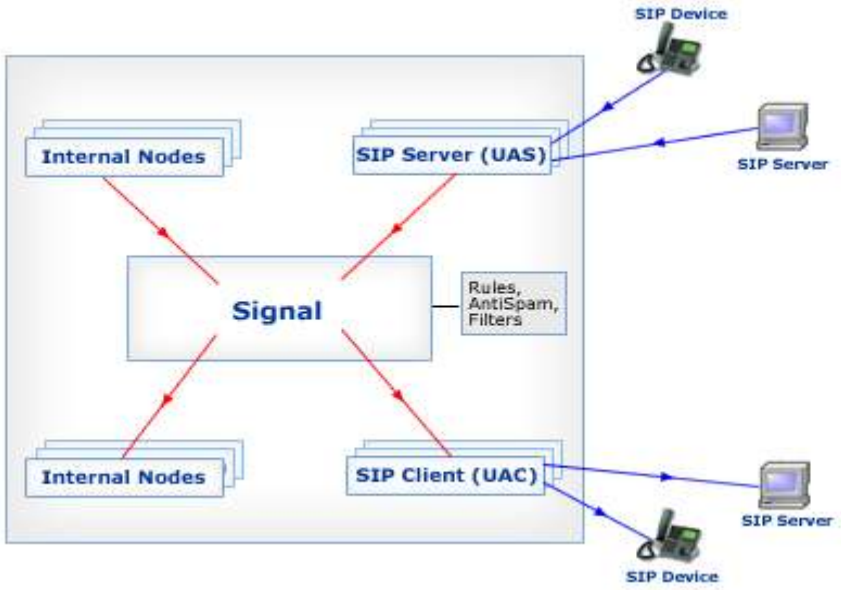
- Signal processing NODEs
- VoIP
- Instant Messaging
- B2BUA
- Mixer - Conferences

# Signal Rules

- Act on out-of-dialog requests only
- Conditions
  - Request Type
  - Submitter Address
- Payload
- Actions
  - Fork
  - Redirect

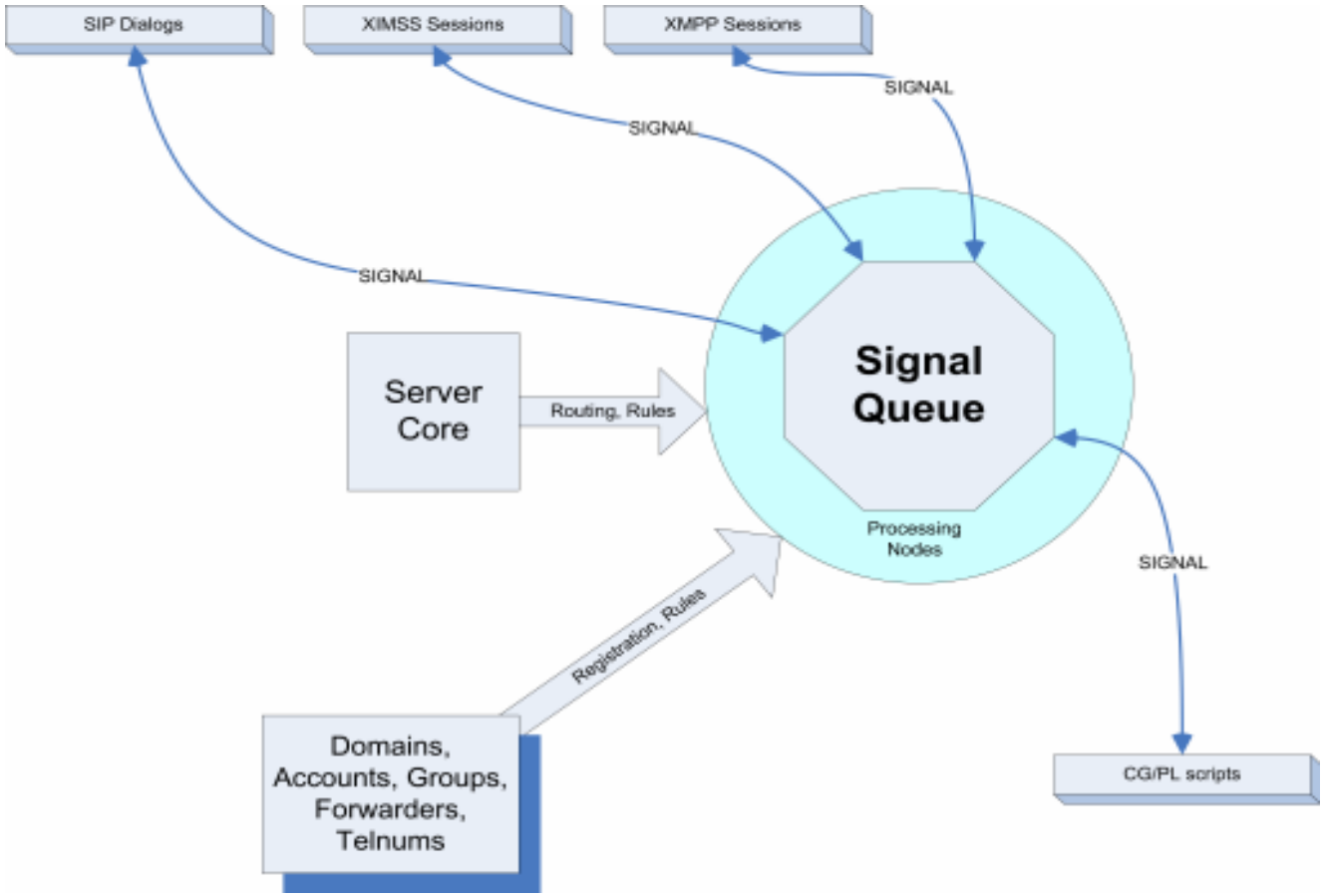
# Signal Flow

- SIP Server components receives requests and creates Signal object
- Signal object is processed by the core
- Signal results in either response sent back or request relayed



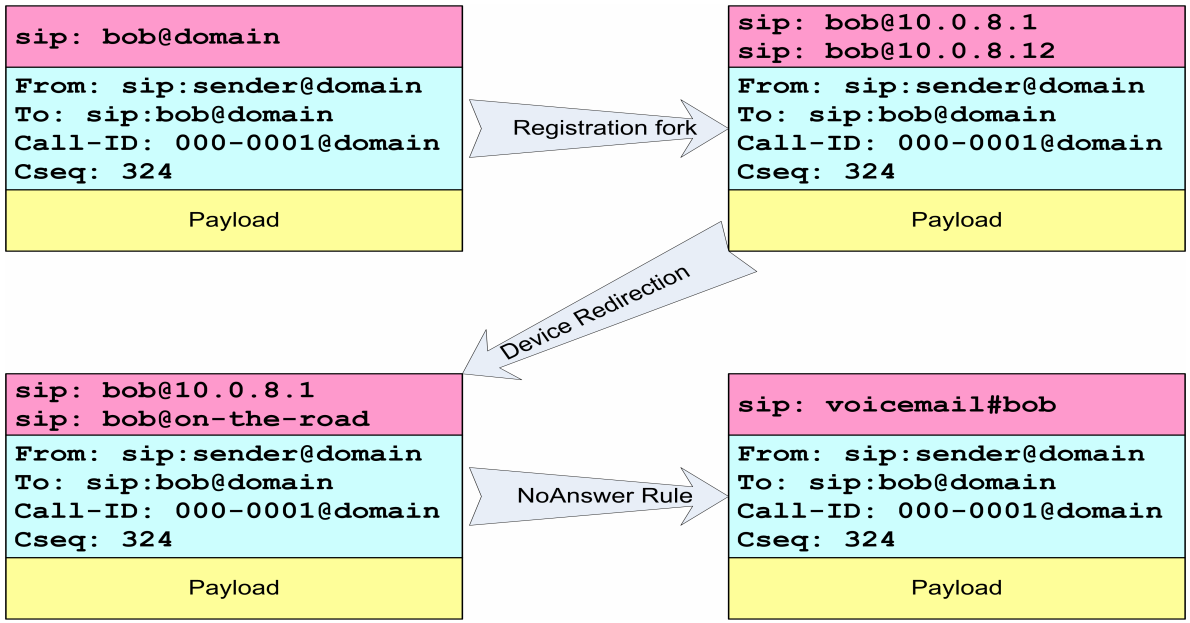
# Signal Flow

- Submitted
- Routed
- Modified
- Processed
- Responded

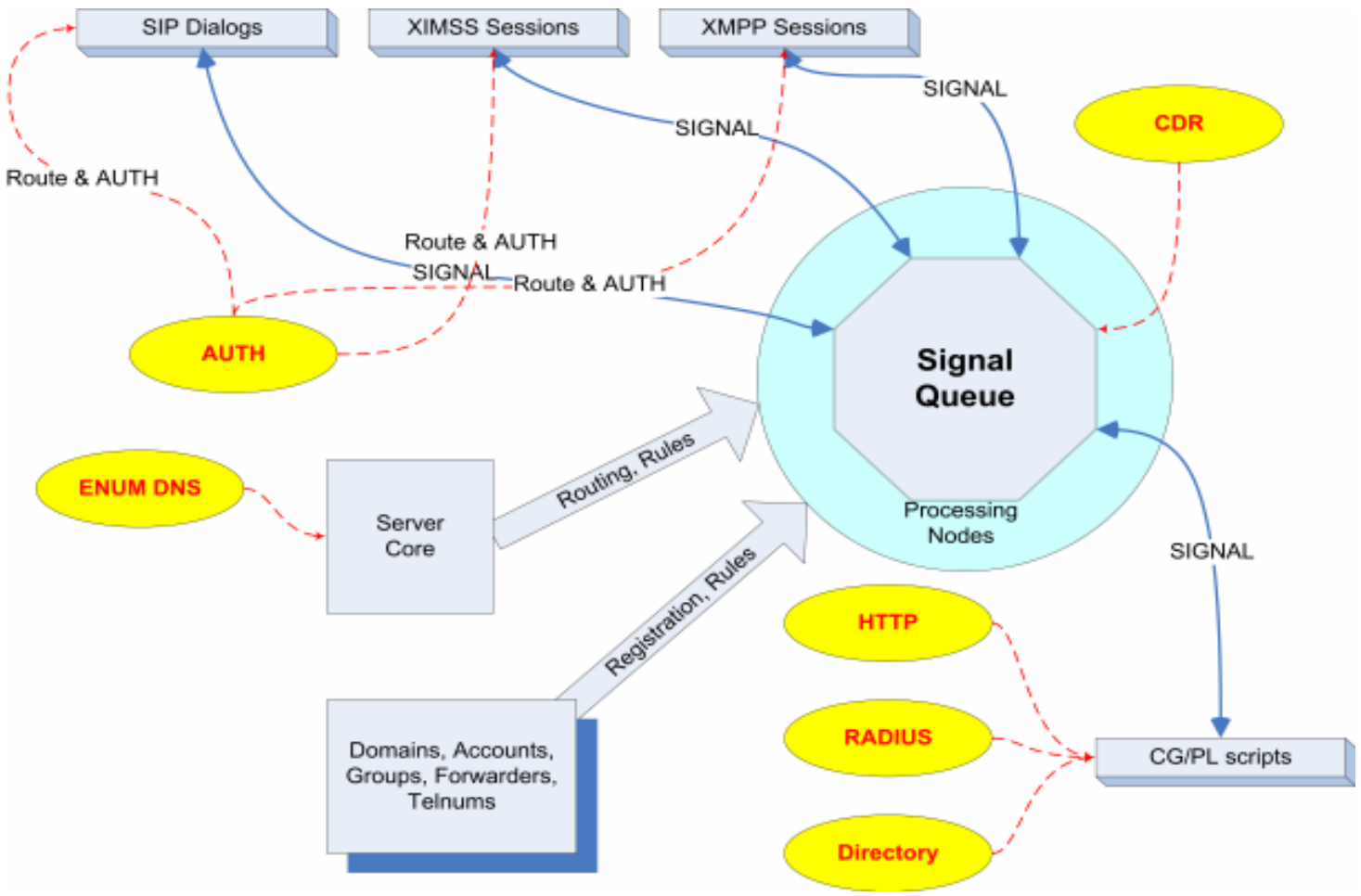


# Signal Transformations

- Registration Forking
  - (AOR -> EPs)
- Rule Forking
  - Add more URIs
- EP Redirection
  - Replace one with another
- Rule Redirection
  - Replace all with other(s)



# Available Hooks



# Signal Flow in a Cluster

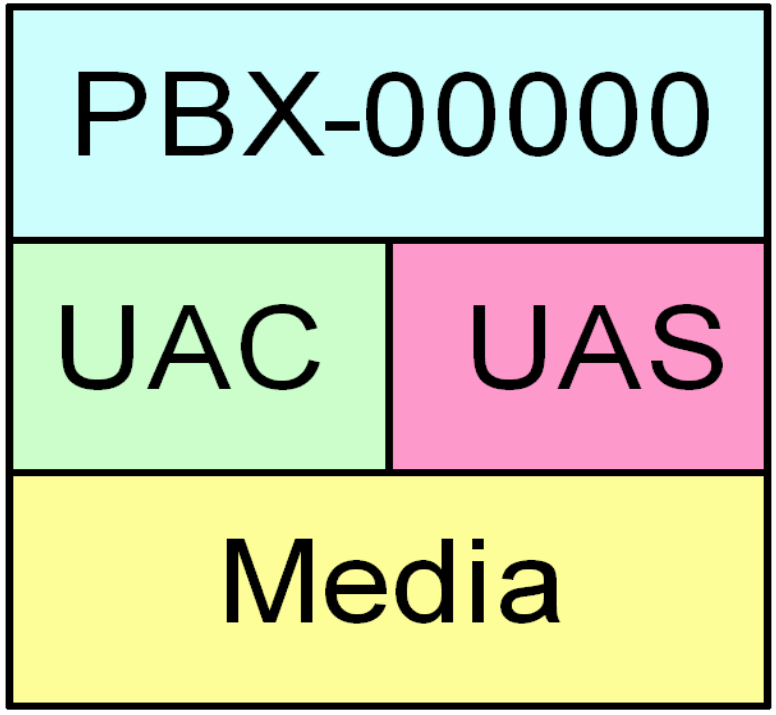
- Calculate hashes for transactions
- Maintain dialogs to running Nodes
- Account for NAT traversal

# Summary on Signal Flow

- Body can be changed for NAT traversal
- Envelope addresses are changed, added or removed
- Headers are not changed, new are added
- To external clients a cluster behaves as a single image

# Application (PBX Task)

- Node
  - Can accept signals
  - Can generate signals
- Media
  - Can send media
  - Can receive media
- Interfaces
  - Account



# States

- Disconnected (Incoming)
- Half-Connected (Provisioned)
- Connecting
- Connected
- Bridged

# Receiving a Call

- A task in the Disconnected state
- Always a new Task
- Can do
  - AcceptCall
  - ProvisionCall
  - RejectCall
  - RedirectCall
  - ForkCall
  - StartBridgedCall
- Should Expect
  - isCallCompletedEvent

# Placing a Call

- A task in the Disconnected state
- Same task can be reused for multiple calls
- Can do
  - StartCall
- Should Expect
  - isCallProvisionEvent, multiple
  - isCallCompletedEvent, one

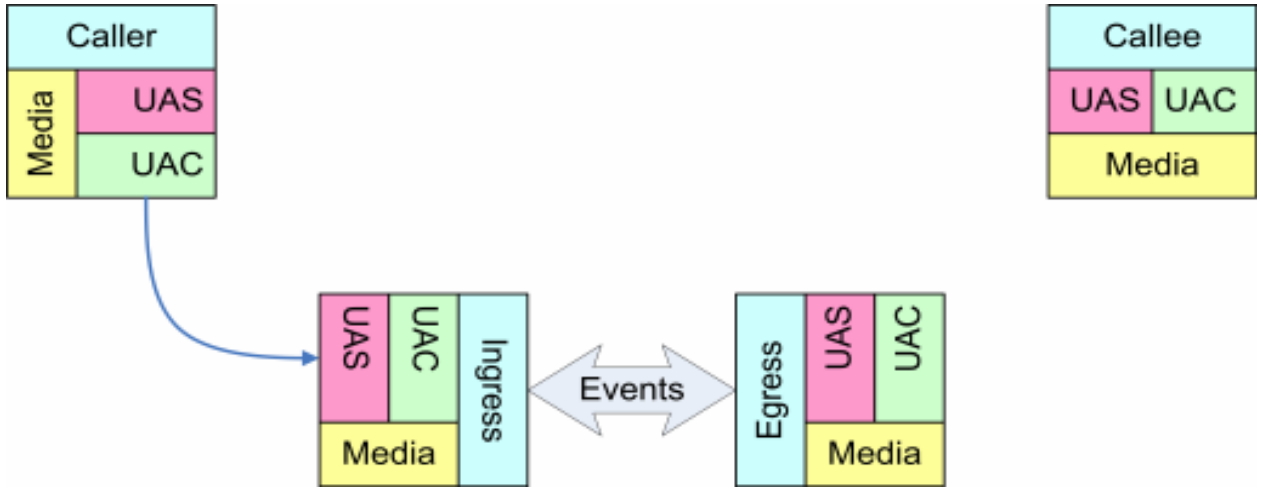
# Bridging Two Calls

- Two tasks in the Connected state
- One of the tasks starts the bridge
- Active
  - StartBridge (blocks)
  - Wait for the result
  - Reality: other events
- Passive
  - IsStartBridgeEvent
  - AcceptBridge
  - Reality: other events

# B2BUA

- Two tasks, communicating to each other through internal interfaces
- Used to isolate signaling paths between ingress and egress call legs

# B2BUA Simple Bridging



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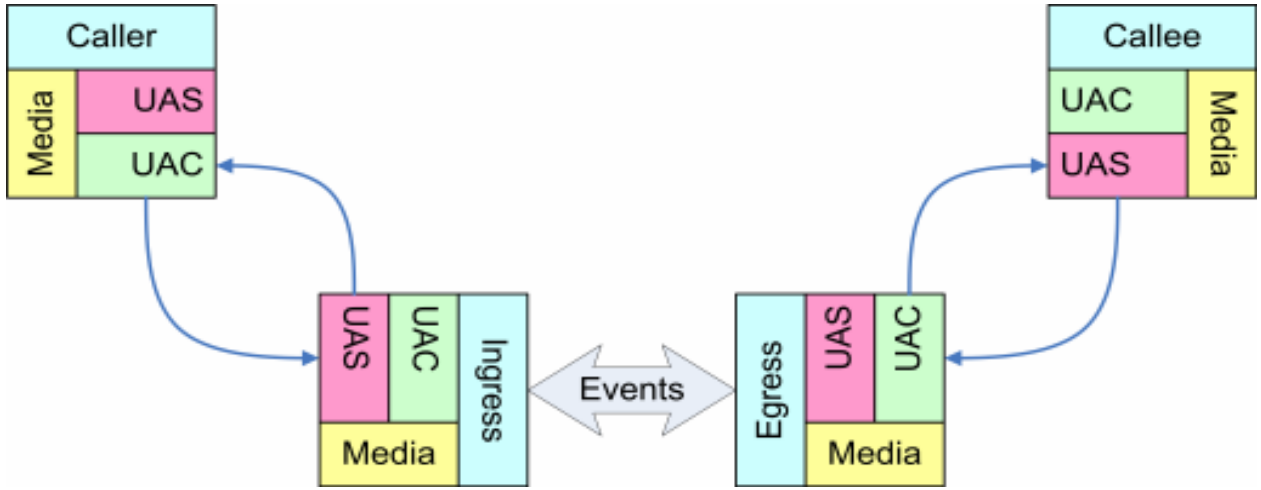

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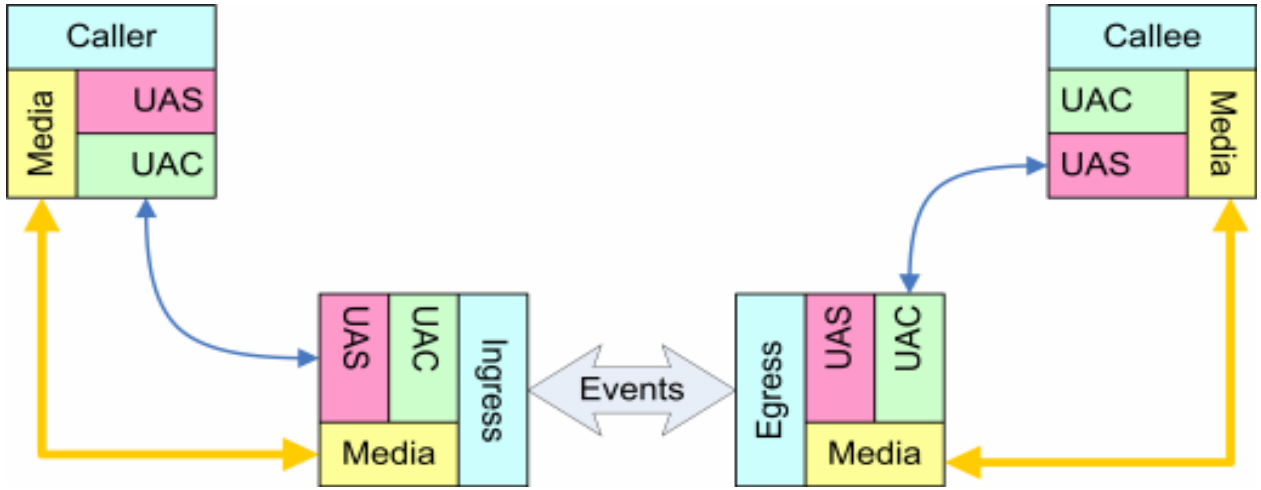

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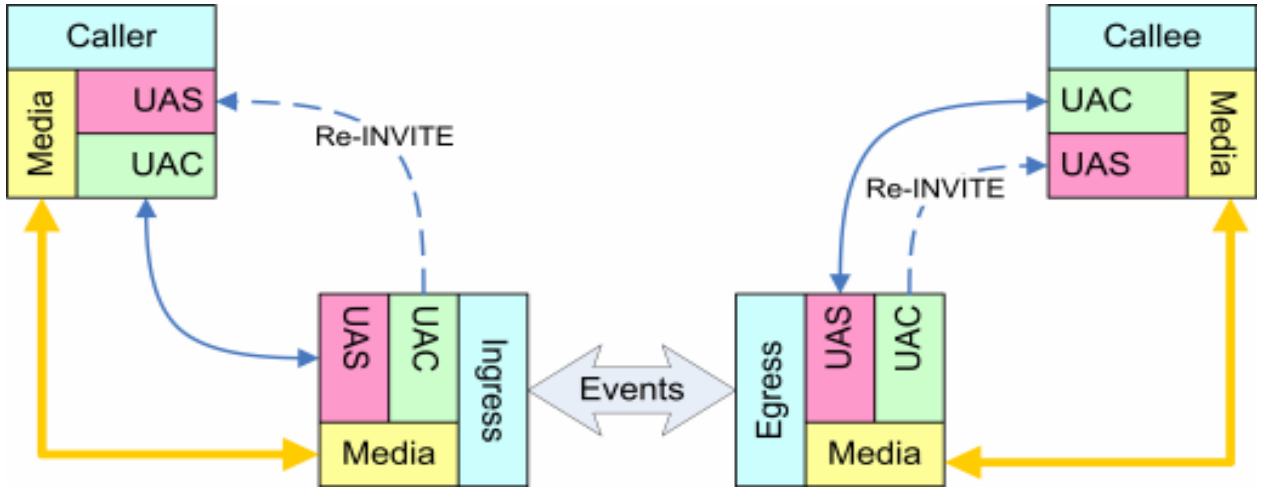

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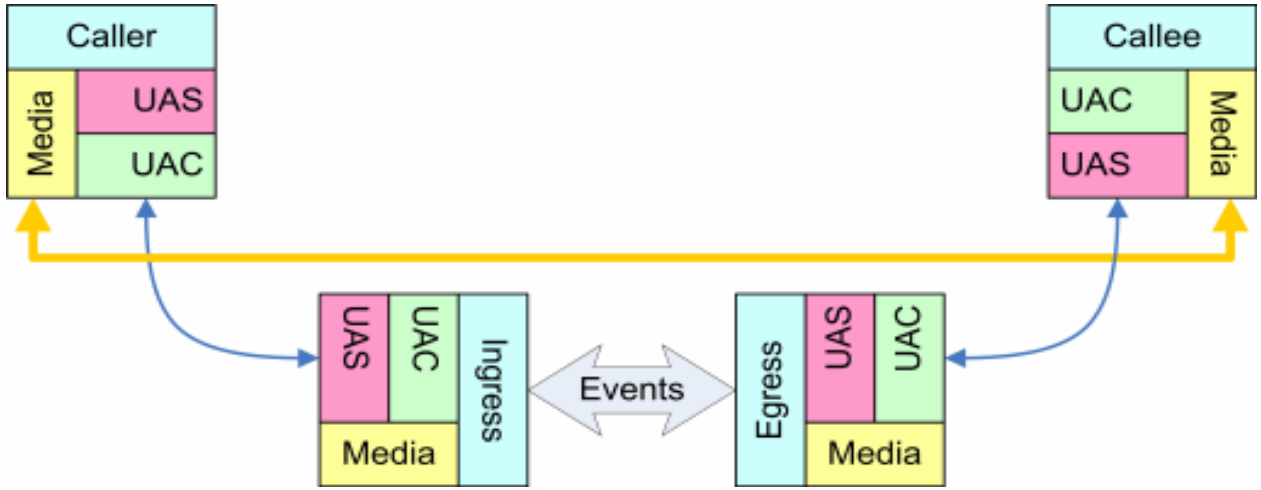
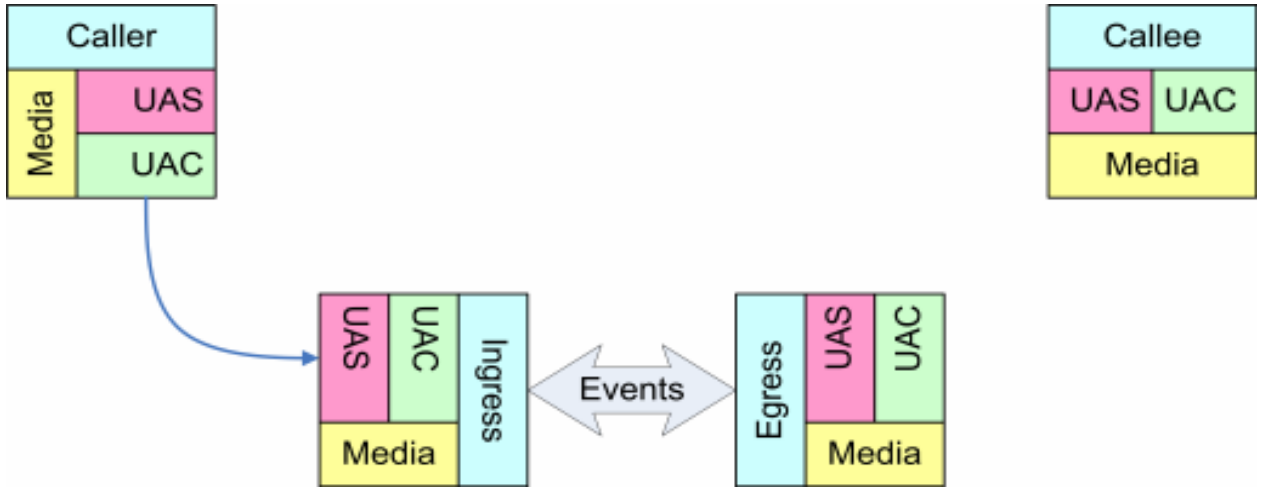


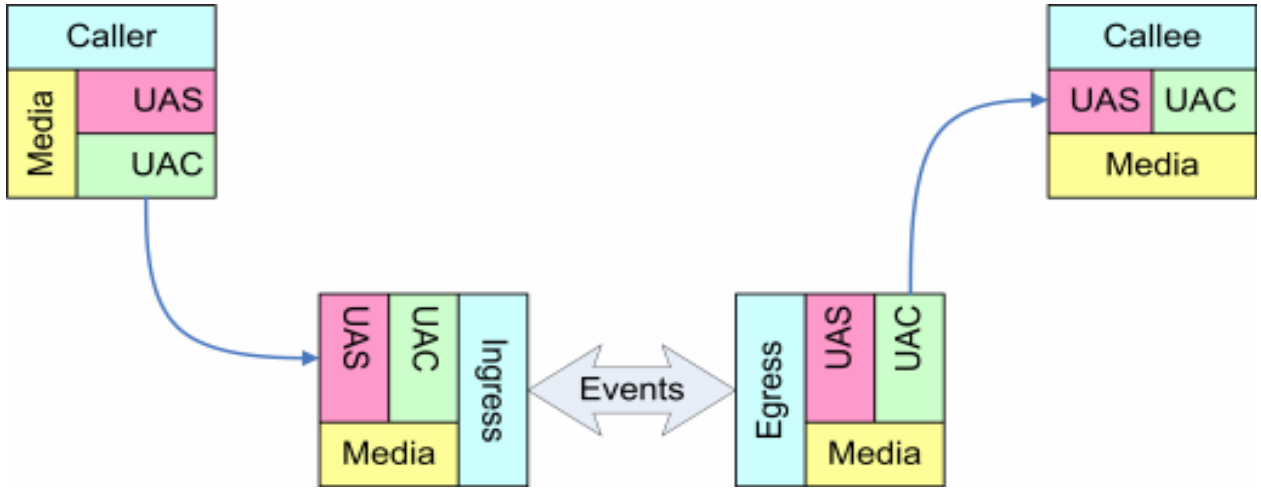
# B2BUA Media issues

- Media format negotiated by Applications and their peers.
- All participants have to support common codecs.
- Application media is initialized, though may be left unused
- Transparent Bridging

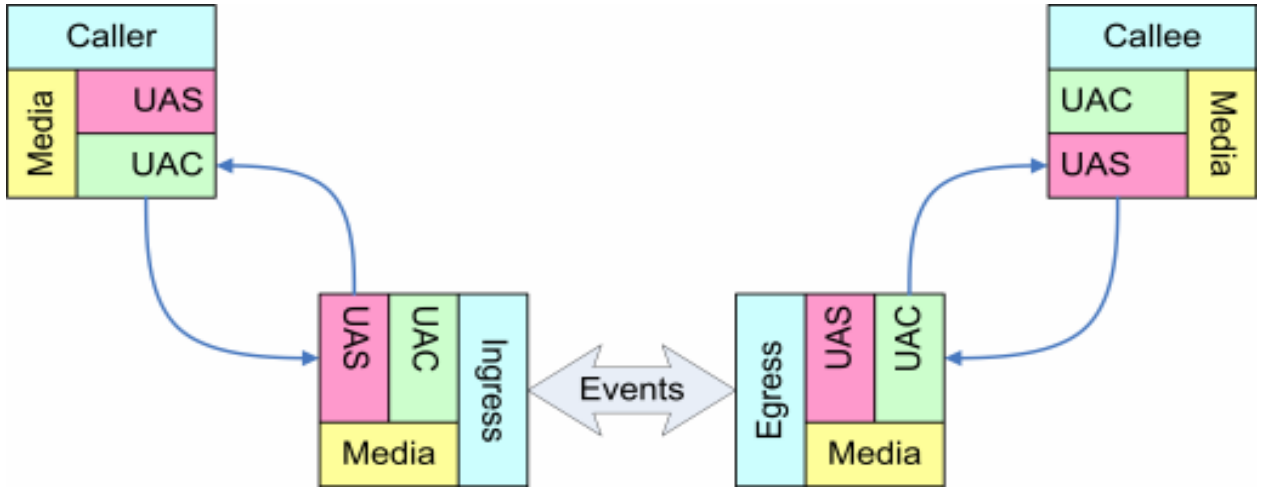
# B2BUA Transparent Bridging



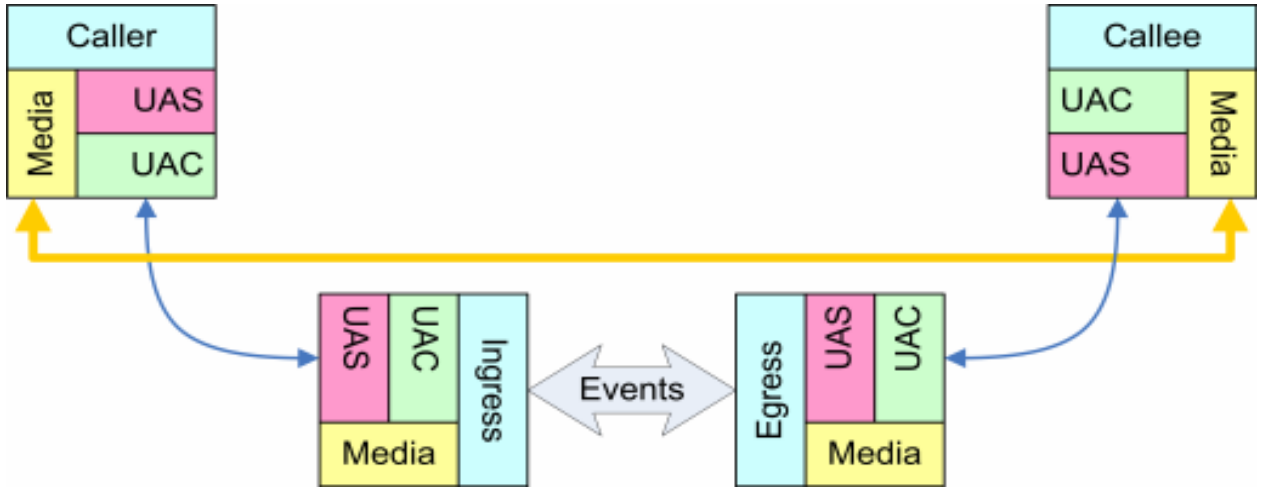
# B2BUA Transparent Bridging



# B2BUA Transparent Bridging



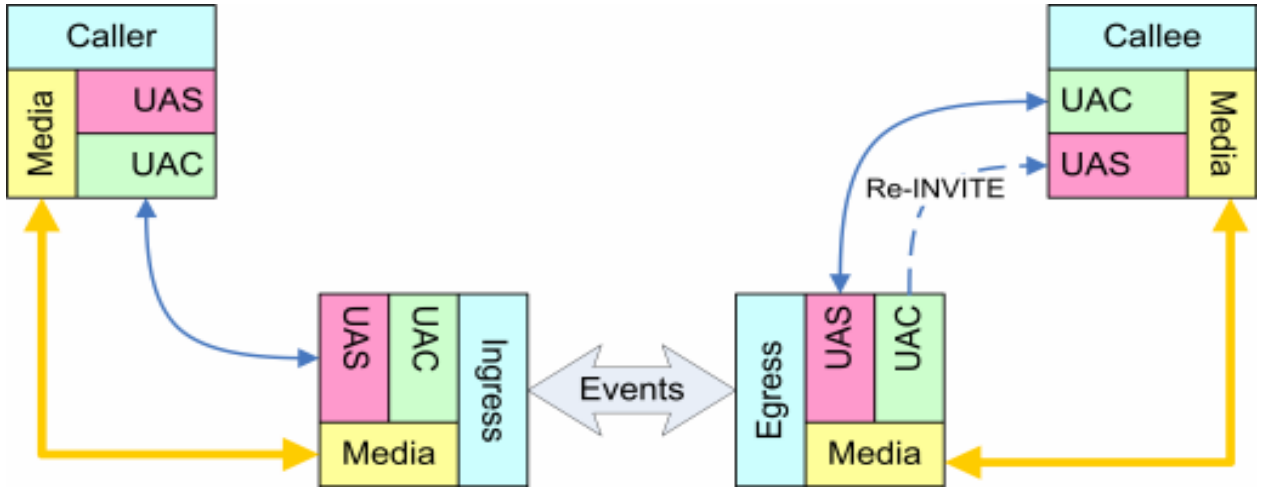
# B2BUA Transparent Bridging



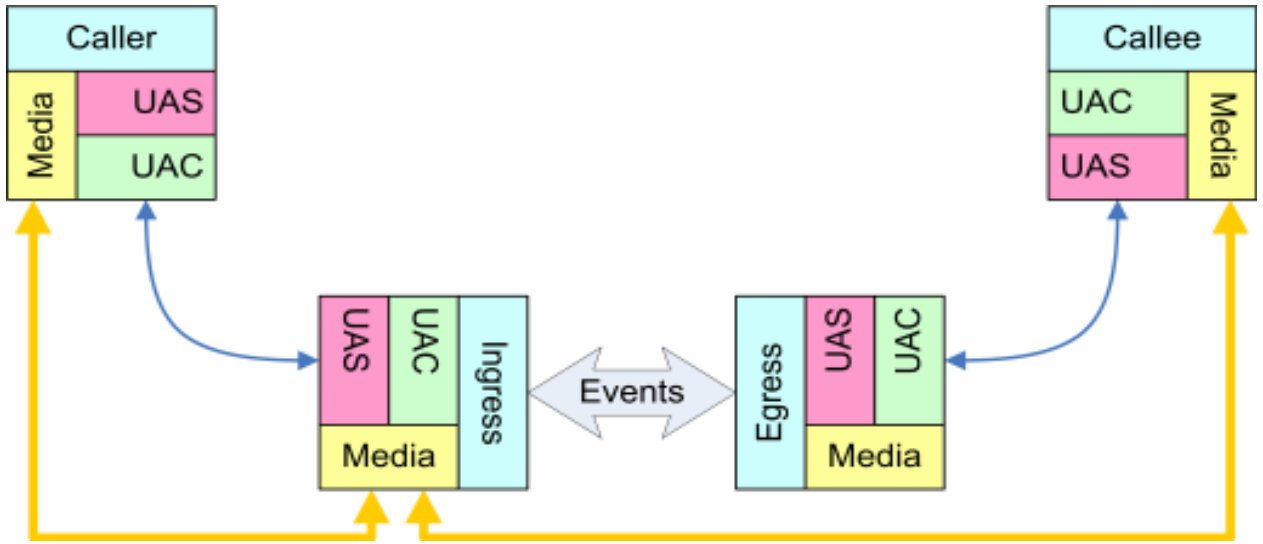
# B2BUA & Mixer

- Used for even more isolation
- Used to set up calls with multiple parties (conferences)

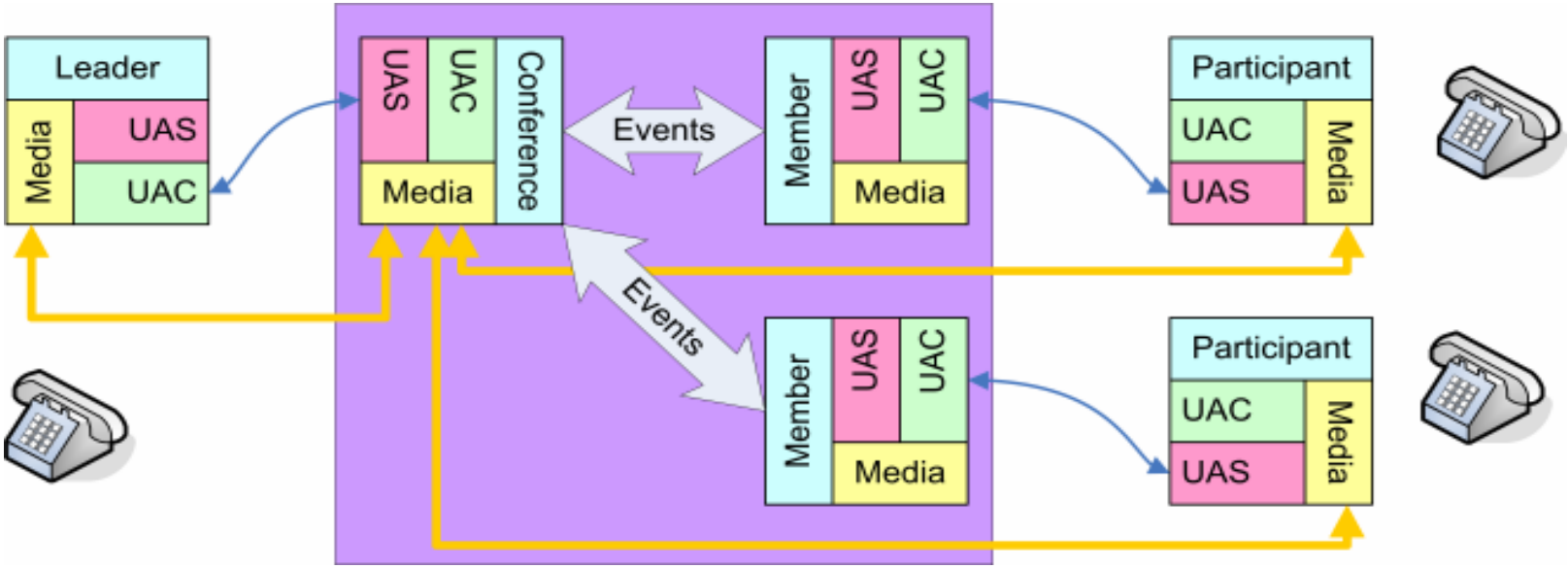
# B2BUA Mixer


Caller
Media
UAS
UAC
Callee
UAC
Media
UAS
Re-INVITE
UAS
UAC
Ingress
Media
Events
Egress
UAS
UAC
Media

# B2BUA Mixer


Caller
Media
UAS
UAC
Callee
UAC
UAS
Media
UAS
UAC
Ingress
Media
Events
Egress
UAS
UAC
Media

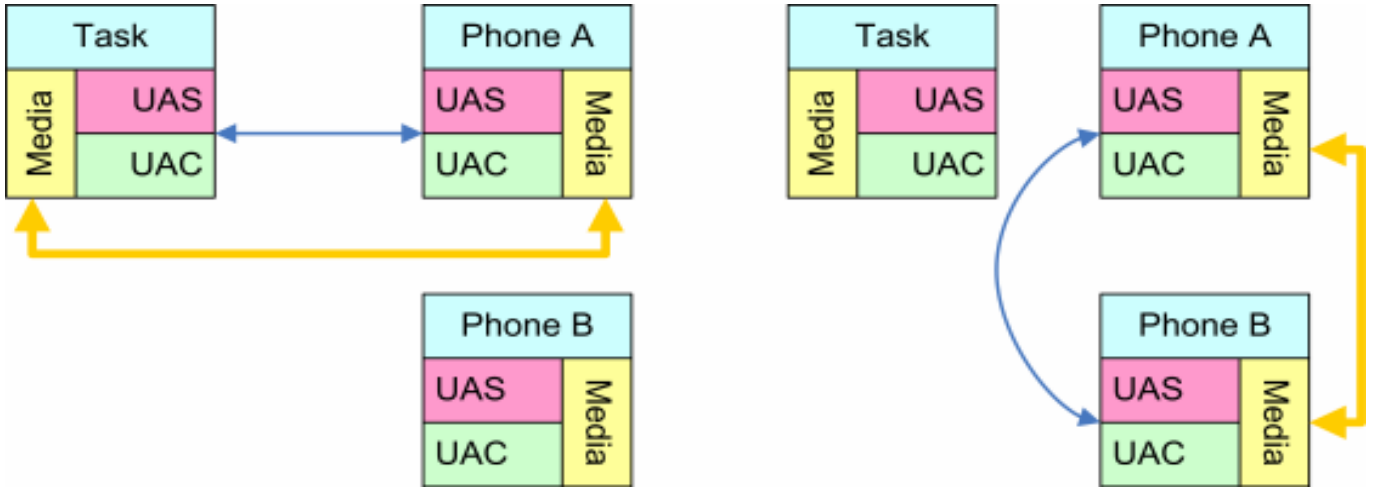
# Conferencing through a Mixer



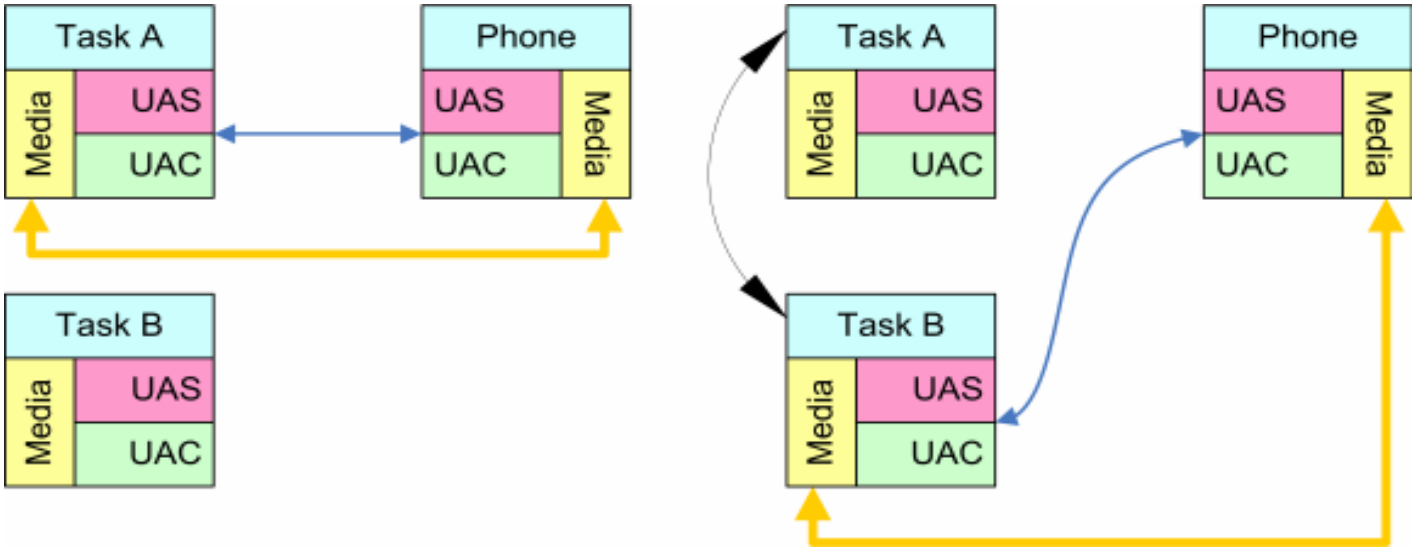
# Call Transfers

- Blind Transfer
  - Very basic, may be unreliable
  - From a Task to another Peer
- Attended Transfer
  - More complex, control while transferring
  - From a Task to another Task
  - The second Task may bridge to another Peer

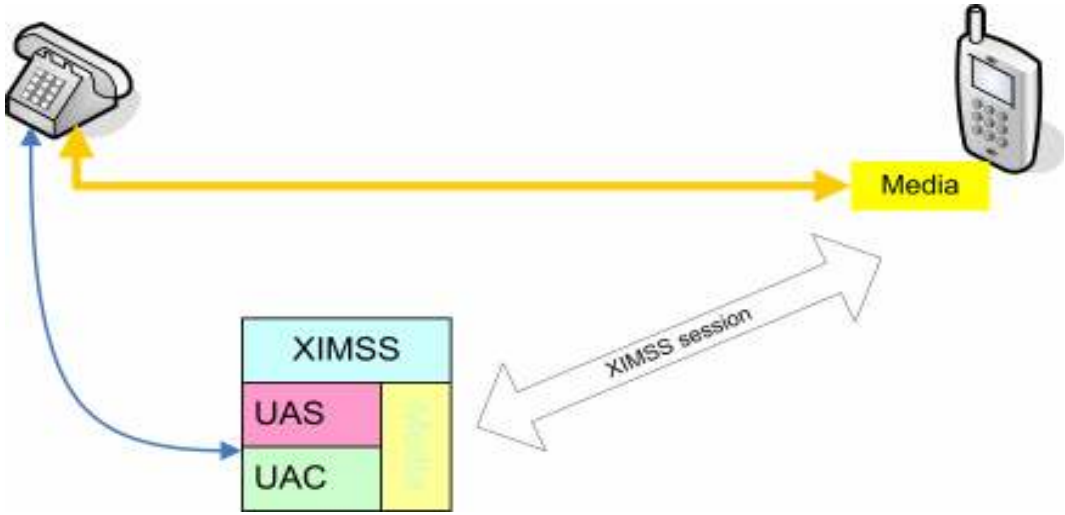
# Blind Transfer – TransferCall()



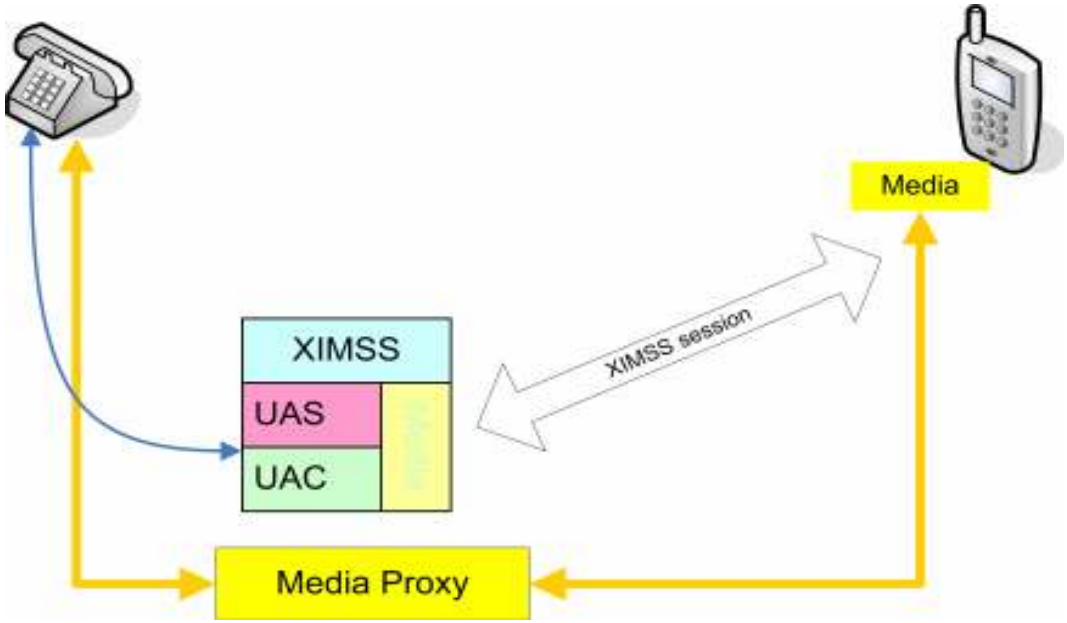
# Attended Transfer – StartTransfer()



# XIMSS Media



# XIMSS Media



# Registering to external PSTN Gateways

- Settings -> Real-Time -> SIP -> Gateways
- Only for Registration
- Routing out through .sipgw is deprecated
- Contact: the URI to receive incoming calls
- Best practices
  - `sip:incomingGW1@wan.ip:port`
  - `S:<incoming*@wan.ip> = gatewayincoming{*,bridge}#postmaster@localhost`

# Processing calls coming in from PSTN

- B2BUA – gatewayincoming
  - bridge
  - media
  - mixer
- incoming.*hostname*
  - `S:<*@incoming.hostname> = gatewayincoming{+*,bridge,media}#pbx@localhost`

# Processing calls going out to PSTN

- B2BUA – gatewaycaller
- Multiple gateways
- Bridge, mixer, forced media relaying
- Best Practices
  - `S:<+*@pstn> = gatewaycaller{00*}#postmaster@localhost`

# VoIP routing: best practices

- Convert numbers to E.164
  - `<+(6-15d)@*> = +*@telnum`
  - `S:<(7d)@*> = localAreaCall{*}#pbx`
  - `<00(6-15d)@*>=+*@telnum`
- Search through locally assigned E.164
- Optionally try ENUM routing
- Route everything not found locally to some fake domain
  - `S:<+36*@telnum> = 0*@pstn`
  - `S:<+*@telnum> = 00*@pstn`
- Send to gatewaycaller
  - `S:<*@pstn> = gatewaycaller{*}#postmaster`

# Data Hierarchy

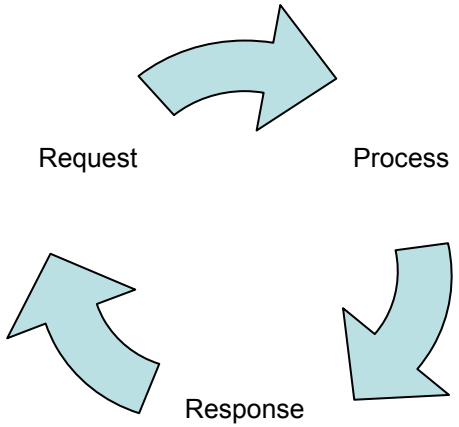
- Server-wide/Cluster-wide Account Defaults
- Domain-wide Account Defaults
- Account-specific Settings

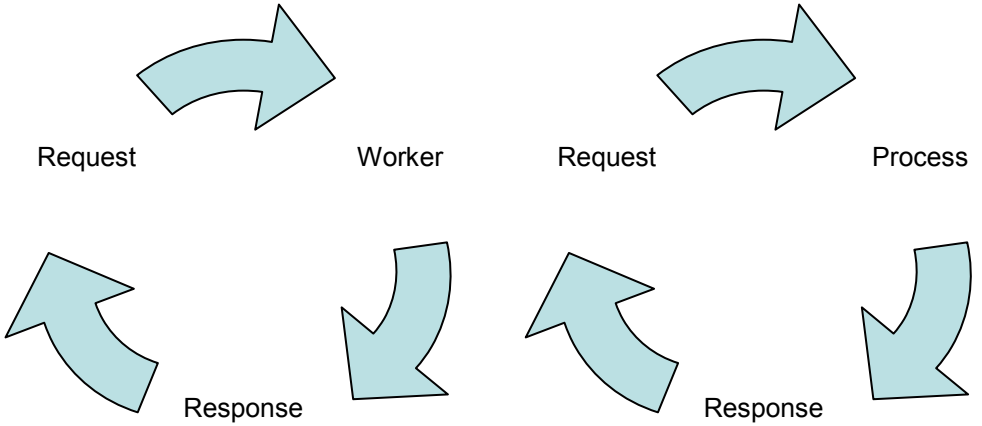
# PSTN settings

- “Custom” account settings – can be extended for custom versions of gatewaycaller
- Can be set as server-wide, domain-wide or account-specific
- “Bad” settings disable PSTN dial-out
- “Dictionary” format supported:
  - `{gw1="11.22.33.44:5061";gw2=pstn.org;}`

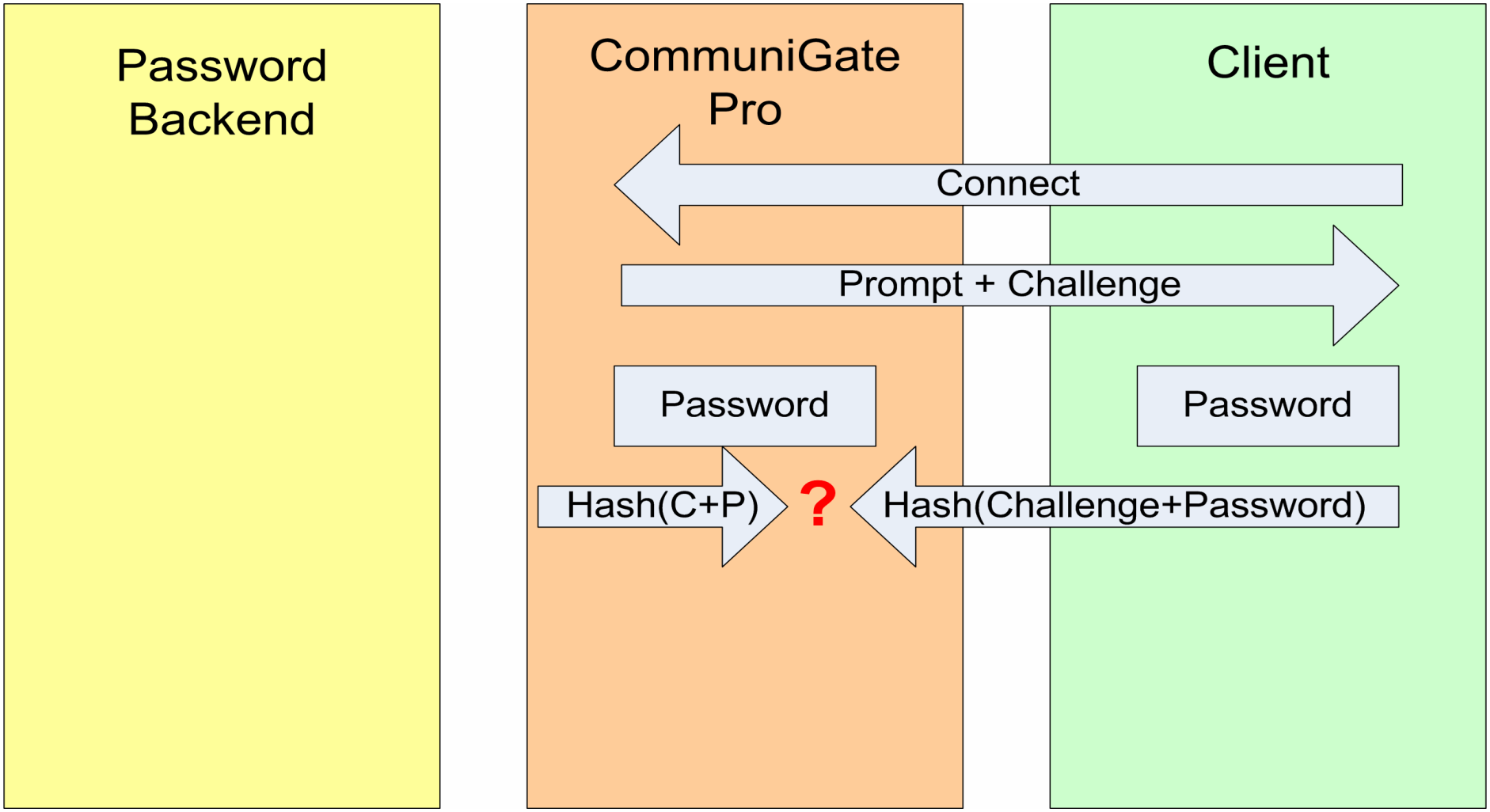
# Helpers

- Ever running programs
- Synchronous and asynchronous model
- Main thread
- Workers

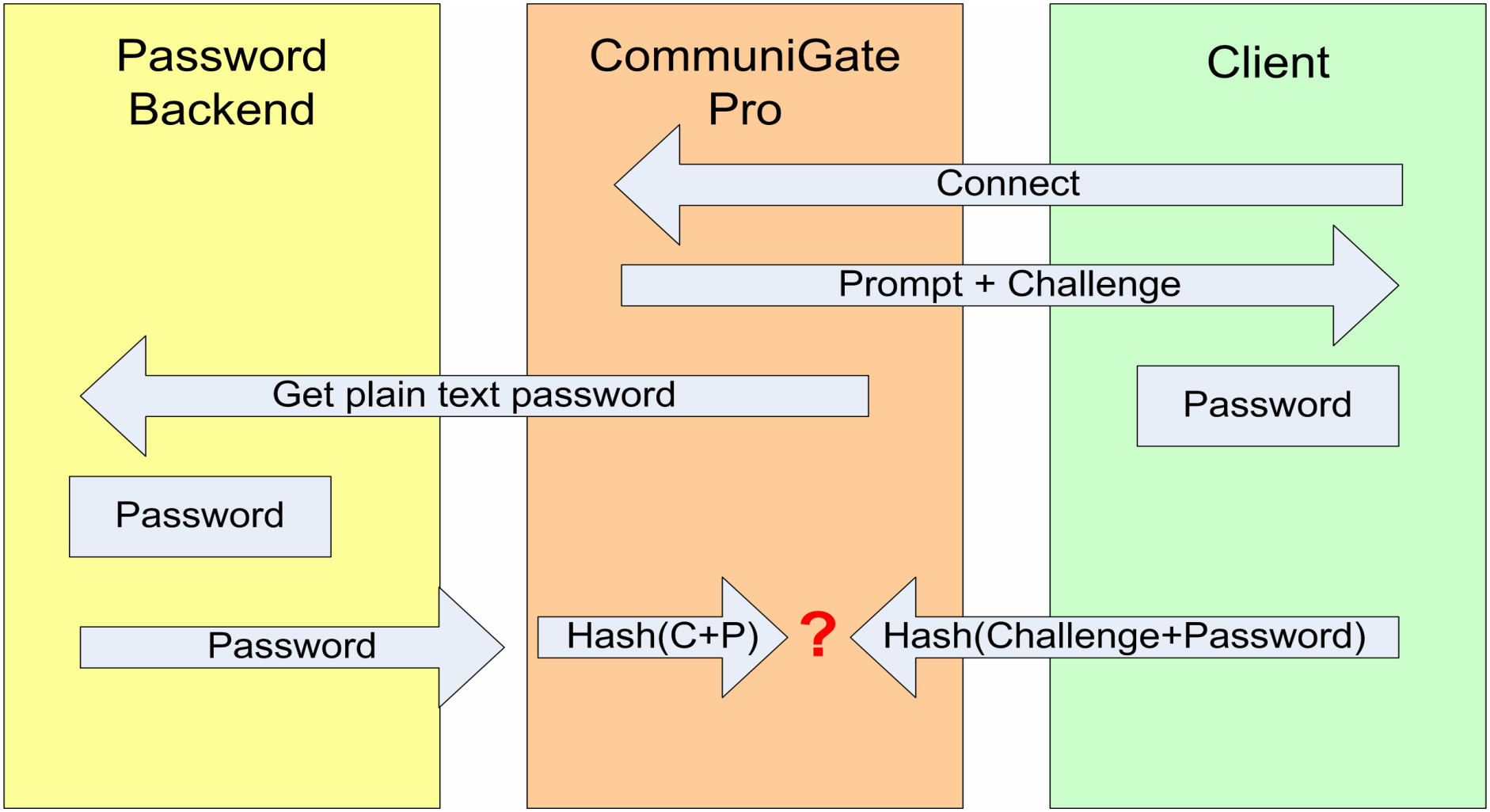


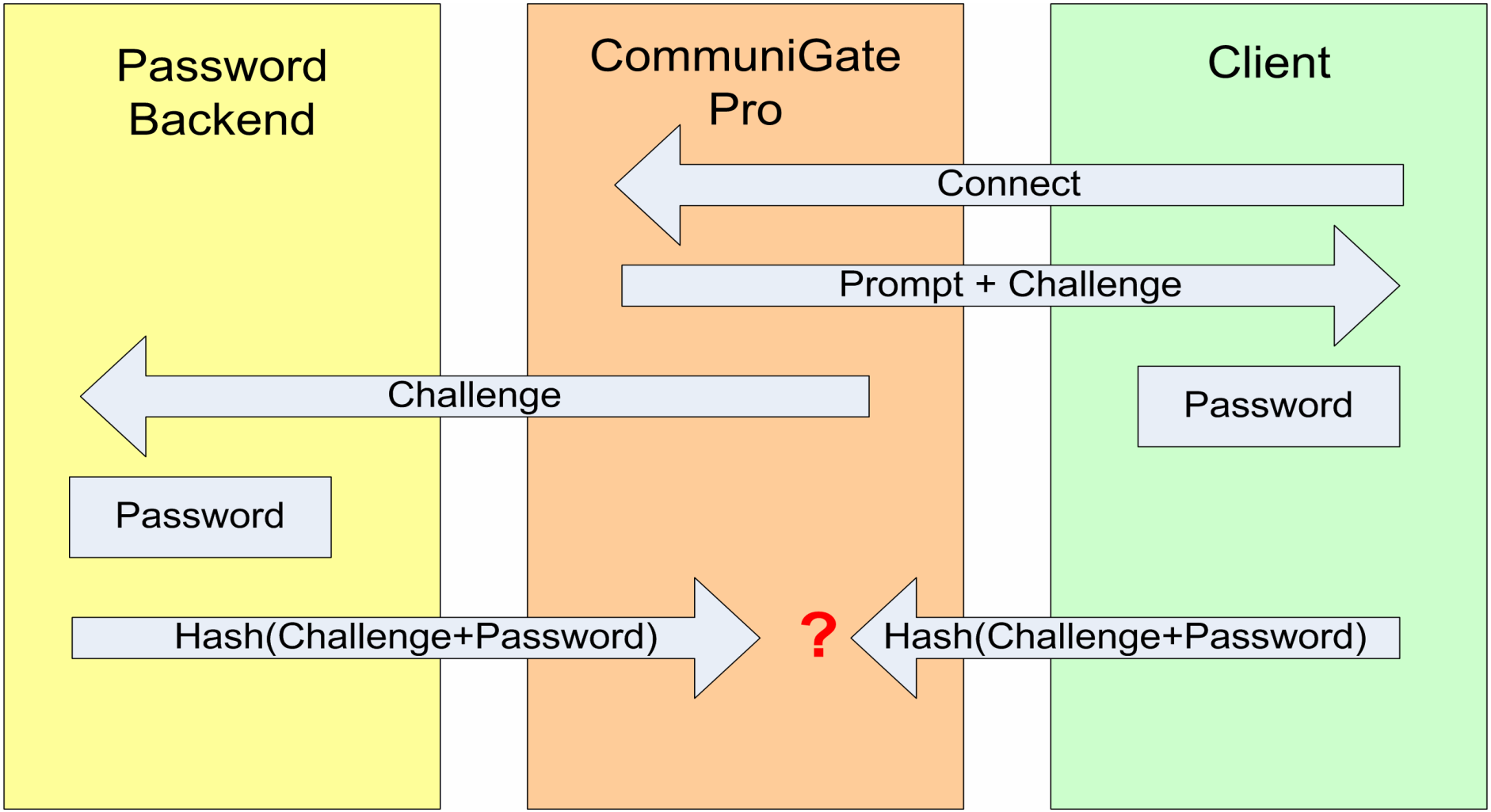
# SASL Auth


Password Backend
CommuniGate Pro
Client
Connect
Prompt + Challenge
Password
Password
Hash(C+P)
?
Hash(Challenge+Password)

# SASL External Auth



# SASL External Auth 2



# WebAdmin: Real-Time Settings

- SIGNAL
- SIP
  - Sending
    - Size limits
  - Receiving
    - Sockets
- Nodes
- Media

# WebAdmin: Router Settings

- Numbers matching
- E.164 conversion
- Telnum routing
- Extensions routing
- ENUM routing
- Routing to applications

# WebAdmin: Network settings

- WAN IP
- LAN IP
- Client IPs
- NATed IPs
- Debug IPs

# WebAdmin: Account Settings

- Rules
- Advanced Real-Time for auto-attendant accounts
- PSTN settings
- Preferences

# WebAdmin: Monitoring Real-Time

- Statistics
- Monitors

# WebAdmin: Working with SIP logs

- INVITE sip:number
- SIPDATA
- Call-ID
- Keyed filtering