Advanced Services Using SIP

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Agenda

• Voicemail and Unified Messaging
• Instant Messaging
• Presence
• SIP Mobility
• Click to Dial
Voicemail -Deposit- How it works?

1. Caller calls called party
2. Switch tries to connect to called party
3. Called party no answers – switch
   Connects the call to VM Server
4. Caller calls leaves VM Message. VM server stores the message in mailbox
5. VM server sends notification to switch
6. Switch lights the lamp on phone
Voicemail - Access – How it works?

1. Caller dials VM Number
2. Switch connects the call to VM server
3. Caller interacts with VM system and Listens to VM. Deletes VM after listening
4. VM System sends notification to switch
5. Switch puts the phone lamp off
What is inside VM Server?

- VM Application
- Message Store
- Media Server (IVR)
- Notification Server

Voice Mail Server

Protocols:
- http
- IMAP4/POP3
- SMDI
- SMPP
- SMTP
- SIP MWI
Access to VM through Web

Web mail

Voicemail System

VM Application

Message Store

Media Server (IVR)

Notification Server

http

IMAP4/POP3

Browser

Outlook

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Unified Messaging

- Same mailbox for
  - Voice Mail
  - E-mail
  - Fax
- UM System has additional functionality to send and receive fax – either through the media server or separate Fax server
How it works in SIP?

1. INVITE from Calling Party - A to VoIP Network
2. INVITE from VoIP Network to Called Party - B
3. INVITE from VoIP Network to B2BUA
4. INVITE from B2BUA to Media Server
5. NOTIFY from B2BUA to UM App
6. NOTIFY from UM App to B2BUA

VoIP Network:
- INVITE
- NOTIFY

B2BUA:
- INVITE
- NOTIFY

Media Server:
- RTP

UM App:
- Message Store
- Notification

Voice files:
- http://file://
VM Deposit detailed call flow (1/4)

Phone A: INVITE, SDP A  
Phone B: INVITE, SDP A

Switch: 100 Trying

VM Server: INVITE, SDP A  
Diversion: ‘B’, reason="no answer"

Media Server: 100 Trying

Phone A: 200 OK, SDP MS  
Phone B: 200 OK, SDP MS

Switch: ACK

VM Server: ACK

Media Server: ACK

RTP
Caller interacts with MS which is controlled by VM Server.

Caller records message. Sets sending options e.g. priority etc.
VM Deposit detailed call flow (3/4)

Phone A | Phone B | Switch | VM Server | Media Server

Caller hangs up | BYE | BYE | 200 OK | BYE | 200 OK

200 OK | | | VM Server sends the notification to user phone .. (and other devices) | | Call disconnected

200 OK | | | |
Usually switch subscribes on behalf of all the end points (phones) connected to it.

VM server sends notification to the switch when a new voice mail is deposited in the mail box.
VM – Notification

• User has to be notified when a new message arrives in the user mailbox

• Different ways of sending notifications
  – Stutter dial tone
  – Message Waiting Indicator (lamp glows on phone)
  – VM icon on the cell phone (via SMPP)
  – Message to pager (via paging protocol or SMTP)
  – E-mail to use on alternate e-mail box
  – Message to client running on PC
VM – Notification

- Advanced VM systems allow user to define the devices where they would like to get notifications
  - User may select to get notification on more than one device
  - User may select to get notification on specific device based on message priority
    - If message priority is ‘Urgent’ send SMS to cell phone
- Notifications may also be combined with presence
  - If I am logged in on ‘IM’ and message is ‘Urgent’ send notification to IM
Instant Messaging

• Transfer of messages between users in near real time
• Messages are usually short
• Two types
  – Page-mode messaging
  – Session-mode messaging
Instant Messaging using SIP

- SIP extensions for Instant messaging
  - MESSAGE method
  - Does not initiate a SIP dialog just for IM
  - Could be sent in existing SIP dialog
  - Content is carried by the MIME body parts
  - Defined by RFC 3428
How does it happen?

End point indicate the support of MESSAGE method when registering
How does it happen?

MESSAGE sip:usera@abc.com SIP/2.0
Via: SIP/2.0/UDP compa.abc.com; branch=y52y3hjga65
From: sip:usera@abc.com; tag=56789
To: sip:userb@abc.com
Call-ID: 56uiuweh@2.4.6.8
Cseq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 18
Hello, how are you

MESSAGE sip:usera@abc.com SIP/2.0
Via: SIP/2.0/UDP compb.abc.com; branch=63h4gsytd
From: sip:userb@abc.com; tag=67682
To: sip:userb@abc.com
Call-ID: jsdhkjf4@5.6.7.9
Cseq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 9
I am fine
Instant Messaging – Session Mode

• Instant messages are formally associated with a session
• Good when more messages are to be exchanged or conferencing is required
• SIP session establishment (INVITE/ACK) and tear down (BYE) required
• Being defined in drafts under SIMPLE WG
Presence

• What is presence?
  – Awareness of other user’s status, location and availability etc.

• Examples
  – Buddy list – online/offline status of buddy, ‘I am on SMS’
  – Availability status – Away, busy, idle etc.
Presence Model

Presence Service

Accepts, stores and distributes PRESENCE INFORMATION

Presence Entity (Presentity)

Provides PRESENCE INFORMATION

Watcher

Requests PRESENCE INFORMATION from presence service

Ref: RFC 2778
Presence Model

Presence Service

Presence Entity (Presentity)
- Requests current value of the presence information

Fetcher (Poller)

Watcher
- Requests notifications for the change in presence information

Subscriber

Ref: RFC 2778
Presence Model

- Presence Service
- Presence Entity (Presentity)
  - Presence UA
- Watcher
  - Watcher UA
- Presence Protocol
Presence using SIP

SIP entities subscribes for getting the presence state of other entities by sending

**SUBSCRIBE**

Presence service sends the presence state information by sending

**NOTIFY**
How does it happen?

Event: presence
Expires: 2400
Accept: test/plain

Event: presence
Expires: 1200

Subscription is accepted

Event: presence
Subscription-State: active; expires=1099

Update presence

Presence information changes. State is active now
How does it happen?

User A

Presence Server

SUBSCRIBE

202 Accepted

200 OK

NOTIFY

Event: presence
Expires: 2400
Accept: test/plain

Message received but the subscription is still not accepted

Event: presence
Subscription-State: pending

Event: presence
Subscription-State: active; expires=1099
Mobility

- Terminal Mobility
- Session Mobility
- Personal Mobility
- Service Mobility
Terminal Mobility

- Allows a device to move between IP subnets
  - Changing IP address of the terminal
- SIP Registration – binds user identifier with the IP address
- (re)INVITE can change session parameters during a session
Session Mobility

- Allows to maintain a media session even while changing terminals
- Transfer of session from one device to another device
- REFER is used for session transfer
Personal Mobility

• Access to the user on multiple devices using the same logical address and addressing user terminal by multiple addresses
  – Many addresses -> one terminal
  – One address     -> Many terminals

• Forking
Service Mobility

• Maintaining access to services even while moving or changing devices and network service providers
  – Speed dial list, media preferences, incoming call handling preferences, buddy list etc.
Click to dial

- Initiation of a telephone call by clicking on a web link or button
- Application running on the server does a 3pcc and connects the call to both caller and called phone.
Click to Dial – SIP Call flow

Phone A → Server → Phone - B → Web browser

1. INVITE no SDP
2. 200 SDP - A
3. ACK SDP -B
4. INVITE SDP-A
5. 200 OK SDP-B
6. ACK
7. RTP
Questions please!