Evaluating the Use of SIP for Streaming Media Applications

draft-whitehead-mmusic-SIP-for-streaming-media

Marie-José Montpetit - presenter
Xavier Marjou
Steven Whitehead

IETF 65 - Dallas
Rationale for Exploring the Use of SIP

• Create value-added ‘blended’ services based on the use of a common signaling platform.
• Build on SIP instead of duplicating its capabilities in other protocols when SIP capabilities are needed.
• Unify and re-use common service architectures.
• Use SIP Rendezvous mechanism to access NATed resources.
• Use SIP extensions for conveying charging/settlement information.
• Use SIP extensions for pre-conditions and for signaling QOS transport.
• Other?
Use Cases

Service Convergence

- **Scenario: “Streaming Media and More”**:  
  - Sarah receives her communications and entertainment services from a single service provider.  
  - Services include: multi-media communications & messaging (text, voice, video); broadcast and on-demand streaming media (music and video).  
  - Services are delivered to a wide variety of end-user devices (fixed & mobile).

- **Common service elements are used to implement common functions for multiple services:**  
  - Service signaling, subscriber management, AAA & billing, service-based QOS  
  - Use of common (SIP-based) features facilitates:  
    - Development of ‘blended services’ by coordinating service interactions.  
    - Avoiding duplicated functions.  
    - New service deployment by building on existing mechanisms.
Use Cases

Access to Content on Private Networks

• **Scenario 1: Remote Monitoring:**
  – Susan has contracted with a video surveillance company to watch her house while she’s away on vacation.
  – Using a SIP-based streaming media application, employees at Acme Security can remotely control and view content from video cameras strategically located around Susan’s home.

• **Scenario 2: Sharing Personal Videos:**
  – Bob has just uploaded the latest video of his newborn daughter Jessica on the family’s personal media server. He sends an email to his parents that contains a URL to the movie. The grandparents click on the link, which initiates a SIP-based ‘click-to-view’ session with Bob’s personal media server.

• **Use SIP rendezvous to locate and initiate a session with a server located behind a NAT/FW on private IP-network.**
Use Cases

Multi-provider Service Delivery

• **Scenario: Multiple Affiliated Content Providers:**
  – As part of Jane’s IPTV service, her VOD entertainment package provides her with access to thousands of titles offered by dozens of content providers.
  – When she selects title to view, her service provider transparently coordinates subscriber identification, authorization, accounting, signaling and settlements with the affiliated content provider.
  – Moreover, QOS-enabled delivery of the video stream may require coordination across multiple transport providers, this too is transparently managed by Jane’s service provider via a SIP-based signaling network.

• **Use SIP-based mechanisms to coordinate service delivery amongst multiple providers:**
  – Use a single-sign on mechanism to support multiple services offered by multiple providers.
  – Establish trusted signaling channels between providers.
  – Exchange transaction identifiers to facilitate charging and settlements between providers.
Use Cases

Local QoS Settings

• **Scenario: wireless customer watching a streaming video:**
  – When the server is not located in the same area as the client it prevents the VOD server from realistically acting as the Application Function (AF) to the regional/local Resource and Admission Control Function (RACF).
  – In the case of wireless video this is a realistic scenario since it might be reasonable to deliver a “wireless” video stream long-distance across the Internet.

• **Use SIP-preconditions to establish local QOS.**
Solution Space

• **SIP-only solutions:**
  – Use SIP to signal streaming media application sessions.
  – Use SIP-based extensions to support application control signaling.
    • Extension Techniques: (supported/signaled via ‘required/option headers’):
      – New SIP headers (e.g., SIP-MEX proposed extensions to Invite/Update).
      – New SIP bodies (e.g., control messages carried in INFO messages).
      – New SIP methods (e.g., new SIP methods for ‘trick-plays’).

• **Dual-protocol solutions:**
  – Use SIP to signal sessions for both media AND control flows.
  – Use a second protocol for application control signaling (e.g., RTSP or other).
  – Options:
    • SIP & RTSP2 Interworking (via synchronization extensions)
    • SIP & RTSP-lite
    • SIP & MRCP2
    • SIP & ‘Boulton application control framework’

• **All RTSP**
Draft Status

- **V0:** Individual contribution.
- Informational.
- Collects inputs from authors and contributions from the mailing list.
- Provides use cases, defines requirements and refines the solution space.
Next Steps

• **Evaluate WG interest in this topic:**
  – Are more people interested in developing this topic?
  – Under which working group Charter if any?
    • MMUSIC, SIP, SIPPING, Other, None?
  – Current I-D: Use Cases & Requirements:
    • New version after meeting based on comments?

• **Prototypes and working implementations:**
  – Work in progress on implementations POC.
## Contacts

<table>
<thead>
<tr>
<th>Name</th>
<th>Company</th>
<th>Email</th>
</tr>
</thead>
<tbody>
<tr>
<td>Marie-José Montpetit</td>
<td>Motorola</td>
<td><a href="mailto:mmontpetit@motorola.com">mmontpetit@motorola.com</a></td>
</tr>
<tr>
<td>Xavier Marjou</td>
<td>France Telecom</td>
<td><a href="mailto:xavier.marjou@francetelecom.com">xavier.marjou@francetelecom.com</a></td>
</tr>
<tr>
<td>Steven Whitehead</td>
<td>Verizon</td>
<td><a href="mailto:steven.d.whitehead@verizon.com">steven.d.whitehead@verizon.com</a></td>
</tr>
</tbody>
</table>

Mailing List: sip-rtsp@external.cisco.com  
Send email to authors to be added (no external procedure as of yet)