RTMP Gateway with SEMS

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RTMP Key facts

Real Time Messaging Protocol (RTMP):

- Developed by Adobe to support Media and data transport between the Flash player and Flash Media Server (FMS).
- Transport protocol supported:
  - TCP/IP (default port 1935)
  - HTTP (RTMPT)
  - HTTPS (RTMPS)
- Specs published for open use in 2009
  - Several open source projects supporting RTMP(T/S)
  - Various levels of feature supported, mostly because the specs are very fuzzy (reverse engineering still needed).

RTMP to SIP gateways

The idea is not really new. Here are some existing open source projects:

- Red5phone:
  - Based on the Red5 RTMP server.
  - Embedded in several other projects, as they were the first to offer a RTMP to SIP gateway (as far as I know...).
- siprtmp:
  - Code in Python and based on rtmplite, a python RTMP server.
  - URL: http://code.google.com/p/siprtmp/
- RTMP support in FreeSwitch:
  - URL: http://wiki.freeswitch.org/wiki/Mod_rtmp

Adobe also offers the Flash Media Gateway:

- Commercial product (rather from the expensive sort ;-))
- URL: http://labs.adobe.com/technologies/flashmedia_gateway/

Shopping list for an RTMP gateway

Obviously, you need an RTMP stack:

- Red5:
  - quite complex.
  - written in java.
- rtmpserver:
  - fully blown server software.
  - not very well supported.
- librtmp:
  - very small piece of software, however, somehow messy.
  - extracted from rtmpdump (useful tool when it comes to downloading videos from youtube).
  - this is what I decided to use because I like small & focused libraries.

Shopping list for an RTMP gateway (2)

Then, you need some codecs:

- For streams originated from the client's microphone, Flash supports:
  - Nellymoser Asao:
    - proprietary codec developed by Nellymoser Inc.
    - optimized for real-time and low-latency encoding of audio.
  - Speex:
    - here, 16kHz (Wideband) only.
    - several bit modes providing different quality levels depending on available bandwidth.

And at last, you'll need the SIP part, for which, luckily, I already had something ;-)

SEMS' RTMP Gateway

Current status:

- Full source code available:
  - rtmp plugin is now accessible on git (apps/rtmp)
  - Test flash app included (apps/rtmp/flash_phone)
- Calls can be made from the test flash client to any SIP destination.
- No headsets:
  - Handsfree talking supported by the test client.
  - Beware that if you want to compile the flash app by yourself, you will have to tweak the Flex SDK (probably temporary state until Adobe fixes the SDK distribution)

SEMS' RTMP Gateway

Roadmap:

- Short term:
  - Registrations from the flash client:
    - Automatic URI attribution if the flash app does not provide a registrar & credentials.
    - With registrar & credentials provided, registration against any registrar.
  - Flash client receiving calls while registered.
  - Invisible flash client controlled by javascript:
    - allows for full customization and easy integration into any website.
- Mid term:
  - End-to-end wideband quality.
  - Transcoder-free calls (when speex wideband is supported on the other end)

Flash API: methods

- Methods:
  - dial:
    - Parameters: URI
    - ActionScript: NetConnection.call('dial', null, uri);
  - hangup:
    - Parameters: none
    - ActionScript: NetConnection.call('hangup', null);
  - register (future):
    - Parameters: username, registrar, credentials (all optional)

Flash API: events

- All events are reported through NetStatusEvent:
  - Event name: “Sono.Call.Status”
  - Property: status_code

- Status codes:
  - RTMP_CALL_NOT_CONNECTED (0)
  - RTMP_CALL_IN_PROGRESS (1)
  - RTMP_CALL_CONNECTED (2)
  - RTMP_CALL_DISCONNECTING (3)

  - RTMP_CALL_CONNECT_STREAMS (4):
    - special status code that instructs the flash app to connect the audio streams

Questions?

Thank you for your attention

* further discussions: sems@lists.iptel.org *