SIP Extensions for Supporting Distributed Call State

<draft-dcsgroup-mmusic-state-00.txt>

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IETF Presentation

State Header

- Motivation:
  - Call state stored at endpoints by their SIP-Proxies during the initial INVITE exchange. This allows Proxies to be stateless during the call.
  - Endpoint passes state information to Proxies when call characteristics require change.
  - State information includes, but is not limited to: participating endpoint information, billing information.
  - State information cannot be altered undetectably by endpoints.
- Syntax of the State Header

```
State     = "State" ":" private
private   = alpha *alphanum
```

- Usage:
  - “State” header encrypted and signed by Proxy and sent to called endpoint in an INVITE message.
  - ‘State” header encrypted and signed by Proxy and sent to the calling endpoint in the response to the INVITE.

SIP URL Extensions

- Host and user-param extensions to include keyword "private"
  - Motivation:
    - During call-setup, proxy encrypts and signs call state information and stores this at endpoints using the State header.
    - To invoke calling features, the endpoint sends the stored state information as the Request-URI tagged with user=private.
    - A SIP-URL with user=private implies that the contents can be usefully interpreted by the proxy that created the contents (usually encrypted and signed by the proxy).
  - Format of extensions
    - `host = hostname | IPv4address|telephone-subscriber` **`| private`**
    - **`private = alpha *alphanum`**
    - `user-param = "user=" ( "ip" | "phone"| "lnp-phone" | "`**`private`**`" )`

SIP URL Extensions

- User = phone
  - RFC 2543 ***requires*** SIP URL of the form:

    phone-number@gateway; user = phone
  - In the DCS architecture, endpoints collect dialed digits and are unaware of the "gateway" associated with the number.
  - We relax the requirement on phone-number@gateway when user=phone is present. The host parameter preceding user=phone considered as a "hint" to the receiver.
    - » **sip:212-555-1212; user=phone is considered a valid phone URL.**

OSPS Header
(Operator Services Positioning System)

- Motivation:
  - PSTN based services like Busy Line Verify and Emergency Interrupt require special treatment.
  - PSTN operator is unaware that the call is to a destination on the IP network.
  - PSTN gateway initiates SIP INVITE to endpoint. This includes the OSPS header.
  - An active endpoint receiving an INVITE containing this header does not return “Busy”.
- Header Format

OSPS = “OSPS” “:” OSPS-Tag

OSPS-Tag = “BLV” | “EI”

Also and Replaces Headers

- Also and Replaces Headers
  - Included in the draft to support call transfers.
    - » The interpretation of Also and Replaces headers is the same as that described in the recently submitted draft - "draft-ietf-mmusic-sip-cc-01.txt"
    - » At the time of submission of the dcsgroup drafts, there was no current I-D to reference Also and Replaces headers (draft-ietf-mmusic-sip-cc-00.txt expired in March 1998).