

# Indication of support for keep- alive

draft-holmberg-sip-keep-01

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# PROBLEM

- draft-ietf-sip-outbound-15 defines two functionally separated mechanisms:
  - Multiple registration flow mechanism
  - SIP signalling keep-alive mechanism
- Possible to use only keep-alive mechanism between UA and edge proxy if registrar does not support Outbound
- However, NOT possible for SIP entity to indicate explicit support of the keep-alive mechanism
  - Only possible to indicate support of Outbound in general, which includes both mechanisms

# PREVIOUSLY IN IETF

- Consensus not to add explicit keep-alive indication (e.g. using the keep parameter which once was part of Outbound) to draft-ietf-sip-outbound.
- Statement that such indication should be specified in a separate draft

# NUTSHELL

- Allows SIP entities to indicate support of, and use, the NAT keep-alive mechanisms defined in draft-ietf-sip-outbound-15
- Hacks, e.g. based on frequent re-registrations (very common today), can be avoided if it is known that keep-alives will be sent.

# USE CASE #1:

## Outbound not supported

- Outbound is not – for whatever reason – supported/implemented, but still requirement to negotiate keep-alives
- Hack solution: indicate support of Outbound even if only the keep-alive mechanism is supported

# USE CASE #2:

## Non-registration emergency calls

- Emergency call made by non-registered user
- Keep-alive needed during the session
- Outbound cannot be used even if supported (Outbound requires registration)
- Requirement in 3GPP

# USE CASE #3:

## Intermediate-to-intermediate

- "Heartbeat" type-of function between specific intermediates during a session
  - Not necessarily for NAT keep-alive purpose
- Useful for SIP provider peering, proxy-to-PSTN-GW, etc.

# USE CASE #4: SIP IP-PBX Trunks

- Many Enterprises have fairly permanent PBX-to-Service-Provider connections
- IP-PBC and SP need to detect liveness of connections, even when no calls are being made
  - To choose alternate routing path
  - Don't want to wait for call request timeouts
  - Need some check to be able to revert when it's back up
- Today SIP OPTIONS is often used for this
  - Heavy, cumbersome, and has some interop issues



THANKS TO EVERYONE WHO HAS  
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