PCP for IPv6-enabled SIP Deployments

draft-boucadair-pcp-sip-ipv6

IETF 87-Berlin, July 2013

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Goal & Scope

- Goals
  - Discuss how PCP can be used in IPv6-enabled SIP deployments
  - List PCP advanced features which are useful for SIP-bases services
- Scope
  - Target SIP deployments in managed networks
  - For deployments where ICE is required, PCP can be of great help as discussed in [I-D.penno-rtcweb-pcp]

Why this effort is needed?

- PCP is still in its infancy stage
- No so that popular in other communities
- A companion dissemination effort is needed to be conducted by the WG so as
  - PCP to be considered as a viable option in some other context
  - to see PCP adopted and widely deployed
- SIP-based services are typical usage which can benefit from PCP

PCP Benefits for SIP-based services

- **Simplified middleboxes**
  - Avoids embedding an ALG in the middleboxes
  - Avoids overloading the middleboxes with keepalive messages
- **Simplified SIP Servers**
  - Does not require any Hosted NAT Traversal function to be embedded in the SIP server
  - Does not require symmetric RTP/RTCP to work
  - Does not Not require symmetric SIP to work (i.e., rport)
  - Avoids overloading the server with keepalive messages
- Easily support unidirectional sessions
- No issue with early media
- The combination of PCP and RFC6947 allows to avoid NAT64, DS-Lite and any other IPv4-IPv6 interwoking resources
- Because there is no need for connectivity checks, session establishment delay is not impacted (pairs of ports can be pre-reserved)

PCP Features: Some Examples

- Unidirectional media flows (e.g., announcement server) can be forwarded without any issue
- Preserve port parity
- Preserve port contiguity
- Learn PREFIX64
- Compliant with "a=rtcp" attribute
- Discover the DSCP value to be used when sending real-time flows

Next Step

- This draft contributes to the PCP dissemination effort
- Should this dissemination effort be endorsed by the WG?
  - Thoughts?