ADMISSION CONTROL FOR
VOICE AND VIDEO OVER MPLS

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Biography

Bruce Davie works at Cisco Systems where he is a Cisco Fellow. He holds a B.E. from the University of Melbourne, Australia and a Ph.D. in computer science from the University of Edinburgh. Since 1995 he has been at Cisco, where he was part of the team that developed Tag Switching, a precursor of MPLS. He now leads a group working on the architecture and development of MPLS and quality of service capabilities for IP networks. He is the author of three books on networking, an active participant in the IETF, and a senior member of the IEEE.
Agenda

- Introduction and Motivation
- RSVP—Not Just for Traffic Engineering
- Relationship to MPLS-TE
- Comparison to Other CAC Approaches
- Conclusions
What Next for MPLS?

- MPLS has delivered on its promise to allow modular enhancements to control plane on a common forwarding plane.
- What are the next modules to come?
  - Multicast
  - Admission Control

<table>
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<tr>
<th>IP routing</th>
<th>Traffic Engineering</th>
<th>L3 VPNs</th>
<th>L2 VPNs</th>
<th>GMPLS</th>
<th>???</th>
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**Common Forwarding Machinery**
Voice and Video QoS Approaches

- **QoS degradation acceptable**
  
  E.g. free voice on the Internet

- **Overprovisioning of links for Aggregate load**
  
  E.g. corporate voice on switched gigabit campus

- **Overprovisioning of links + Diffserv for Voice/Video traffic**
  
  E.g. corporate voice/video on switched gigabit campus

- **Right-provisioning of links + Diffserv for Voice/Video traffic**

  Reject sessions which “don’t fit” (e.g. during failure scenarios or focused overload) to preserve QoS of other sessions

  E.g. corporate voice/video on inter-campus WAN links, Mobile Phone Trunking, PSTN Class 5 replacement

**THIS IS THE CALL ADMISSION SCENARIO**
Terminology

• **Call Control**: responsible for session establishment, maintenance and tear-down, authentication/authorization/accounting etc.
  
  E.g. SIP-based Call Control

• **Bearer Control**: responsible for establishing the necessary “transport pipe” with necessary QoS

  Connection-oriented networks: bearer control deals with connection setup

  Connectionless networks: bearer control only has to establish the QoS for the path—basic connectivity already available

  **THIS IS THE CAC FUNCTION**
Key Bearer Control/CAC Challenges

- Scalability: core may be carrying 100,000s of sessions simultaneously

- Synchronization between Bearer Control and Call Control:
  
  Check bearer before ringing or establishing session

  Tear-down session when bearer QoS is lost after session establishment (e.g. due to failure)

  Take appropriate action (e.g. fast busy, proceed with best effort service) if QoS establishment fails
IETF Components for Scalable CAC Solution

- **Clean separation between Bearer Control and Call Control**
  
  Call Control completely leaves it to Bearer Control to make the right QoS/CAC decisions and report

- **On-Path (“in-band”) CAC**
  
  Explicit CAC decision on actual path followed by sessions

- **Use of TE/DS-TE tunnel for Aggregate Bearer Control in Core**

- **Use of RSVP for finer Bearer Control on the edge**

- **RSVP Aggregation over MPLS TE Tunnels**

- **Synchronization between RSVP and Call Control (e.g. SIP) on end-device**

*draft-lefaucheur-rsvp-dste-02.txt*
Scalable Bearer Control in Core

- No per-session bearer in core
- Aggregate Bearer Control (e.g. one reservation per PoP pair)
- MPLS TE (or DS-TE) tunnel is ideal Aggregate Bearer:
  - Bandwidth Reservation
  - Aggregate CAC
  - Constraint Based Routing
  - Path engineered against many parameters (delay metric, max voice utilization, …)
  - Protection by MPLS Fast ReRoute
  - Operational experience
  - Dynamic resizing
  - Support for different classes of service via DS-TE
SIP and RSVP Synchronisation

- RFC3312: Integration of Resource Management and SIP
- Concept of “Preconditions” which need to be met before call can be setup
- QoS reservation via RSVP is one possible precondition
- Allows perfect synch between SIP and RSVP reservations on VoIP terminal or GW

```
|------------------(1) INVITE SDP1-----------------> |
|<---------2) 183 Session Progress SDP2-----------|
|---***---(3) PRACK------*R*-----------------> |
|-*E*--------------*E*-
|<--*S*---------200 OK (PRACK)----------*S*--|
|-*E*------*E*|
|-*R*------*R*|
|-*V*------*V*|
|-*A*------*A*|
|-*T*------*T*|
|-*I*------*I*|
|-*O*------*O*|
|-*N*------*N*|
|---***---***|
|------------------(5) UPDATE SDP3-----------------> |
|<--------(6) 200 OK (UPDATE) SDP4------------|
|<--------(7) 180 Ringing---------------------|
|------------------(8) PRACK----------------------|
|<--------(9) 200 OK (PRACK)------------------|
|<--------(10) 200 OK (INVITE)----------------|
|------------------(11) ACK---------------------|
```
RSVP Aggregation over MPLS TE Tunnels
RSVP for QoS?

- “I thought RSVP was…
  
  Dead
  Unscalable
  Only for TE”

- Scalability issues are all around per-flow reservations
  
  We avoid those, or push them to edges

- “Reports of RSVP’s death have been greatly exaggerated” —Mark Twain

- RSVP is undergoing resurgence due to
  
  Greater deployment of QoS-aware apps
  Need for policy-aware admission control (e.g. pre-emption of less important traffic during overload)
Example Application: CAC for Trunking of 2G/2.5G/3G Mobile Voice

One RSVP Reservation for All Calls Between a Given Pair of VoIP Trunk GW

RSVP can be Localized between TGW and PE

One TE Tunnel per Pair of PEs
CAC for Trunking of 2G/2.5G/3G Mobile Voice with Aggregate RSVP Reservation on GW

**GW1** | **PE1** | **Call Agent** | **PE2** | **GW2**
---|---|---|---|---
GW knows codec/bw | GW initiate RSVP for 100 calls | Path Resv | PE2 does CAC of 100 calls on PE2->PE1 tunnel | GW knows codec/bw

Call 1

GW knows codec/bw | PE1 does CAC of 100 calls on PE1->PE2 tunnel | GW knows codec/bw | GW knows this call fits in RSVP reservation

Call 2
RSVP Aggregation: Key Features

- No restrictions on which segments you do CAC on:
  GW→PE, PE→PE, PE→GW
- No assumption of symmetric bandwidth
- No restriction on TE Tunnel deployment model:
  PE→PE mesh, P→P mesh (i.e. GW not directly connected to TE Headend), any combination
- No restriction on granularity of GW-GW RSVP reservations
  1 GW→GW reservation per call, 1 GW→GW reservation for many calls, 1 GW→GW reservation for all calls GW→GW
- No restrictions on scope of RSVP signalling
  End-to-end RSVP signalling, RSVP signalling localised onto GW→Headend segment (while retaining CAC over TE Tunnel)
- Dynamic Adaptation to Topology change
  If a GW is suddenly reachable through a different tunnel, CAC adjusts immediately (and reservation is maintained if it fits)
Example Applications

- Consistent with growing interest and usage of RSVP for Voice/Video CAC in Enterprise, Government and Defence
  Can be applied to Voice/Video devices deployed and using RSVP + SIP/H323/SCCP synchronization

RSVP reservations for 
TGW → TGW

RSVP reservations for 
VPN_Site → VPN_Site
Voice and Video
Relationship to Other Technologies

- Consistent with operation over other Tunneling techniques
  - E.g. RSVP Aggregation over IPsec tunnels
  - E.g. RSVP Aggregation over Aggregate IP reservations (RFC3175)
- Inherits all work on Multi-Level Priority and Preemption (MLPP)
Conclusions

• CAC required in some environments for guaranteed QoS under all conditions (failure, focused overload,...)

• IETF developing all components for scalable CAC solution:
  - Separation of Call Control and Bearer Control, with synchronization by end-system (e.g. RSVP/SIP synchronization)
  - Aggregate Bearer Control in the core using MPLS TE Tunnels
  - CAC/Aggregation of RSVP reservations over TE Tunnels

• This CAC approach is flexible and consistent with rest of IETF work