Management and QoS for VoIP

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VoIP: Service or Application?

It depends who you ask:
• Carriers have a $500B voice business to defend
• The suppliers of carrier equipment as well
• The IT industry likes voice in their way:
  – CPE based, lowest cost and IT grade security
  – IT has fundamentally no need for carrier VoIP
• End users are happy with VoIP providers
  Skype, Vonage, Packet8, AHIS, beroNet, DENTEL, DialOst, etc.
• What the authors think:
  – All advanced SIP features can be invoked P2P
  – The GIPS Sound Package is another proof the value is in the application (SIP endpoint)
  – The “Internet Is The Service” (Jon Peterson/IETF)
Personal Observations

Most people “concerned” about quality of service for VoIP are actually not using it

Desire to sell expensive VoIP and QoS network infrastructure? Platforms to monitor voice quality?

VoIP service providers do not control the network e2e, thus proving:
- QoS is adequate or better when IP-IP
- No VoIP network infrastructure is required
- No VoIP network to manage, except some critical server based applications. This is applications management.
QoS does not create bandwidth

but

Bandwidth and avoiding congestion is the insurance for QoS

Conventional IP performance monitoring is just fine

• Switched 100 Mb/s Ethernet to the desktop is the rule
  it is hard to buy any other Ethernet switches
• Capacity glut on the Internet backbones is the real problem
• Congested access links to the Internet maybe the problem
Useful Technologies

• Avoid obsolete ITU-T G.7xx codecs and their license payments
• MPLS can invoke DiffServ but why not just use DiffServ wherever possible
• Use the iLBC free codec as default
• Use voice quality monitoring only where users perceives quality: In the SIP endpoints so as to avoid error due to the fractal behavior of IP traffic
• P2P self organizing networks and Internet resilience will provide the highest possible availability for VoIP.

Internet Facts for VoIP

• Packet loss on the Internet is in the 1%-0.1% range

• Delay on the Internet is close to that on fiber

• Internet codecs provide the better voice

• 2/3 of Internet traffic is P2P anyhow

• P2P communications dominate VoIP: Skype
Internet Packet Loss Trend: From 1% to 0.1%

Packet Loss Seen from ESnet

% Monthly Median loss

http://www.slac.stanford.edu/xorg/icfa/icfa-net-paper-jan05/
Internet Delay

Global Response Time (MS): Past 24 Hours

Global Response Time (MS): Past 30 Days

http://www.internettrafficreport.com/main.htm
Internet Codec Performance

http://www.ilbcfreeware.org/

RFC 3951: Internet Low Bit Rate Codec (iLBC)

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RFC 3952: RTP Payload for iLBC
Internet Traffic Load: Mostly P2P

Peer-to-Peer over the Last Mile

- Peer-to-Peer is inherently symmetrical in upload:download for consumer ISPs, where as most traditional protocols are download orientated (e.g. web).

- This makes Peer-to-Peer a huge problem for last mile providers, where it makes up 80% or more of the traffic on the network.

- Cable companies in particular encounter large problems with upstream capacity over the last mile.

Source: CacheLogic StreamSight S10 deployed in a Tier 1 ISP

http://www.cachelogic.com/research/index.php
SIP Models and Skype

1999: RFC2543

2002: RFC3261

2004: RFC2543

VoIP islands
Softswitch
IP PBX

2004 Commercial SIP:
VoIP islands
Softswitch
IP PBX

2005 Commercial SIP:
Interdomain
Session Border Controller
ENUM is still mostly discussed

Post 2005 Commercial SIP:
Interdomain
Session Border Controller
ENUM is still mostly discussed

2004: Columbia University
Just like in 1997
Key standards work is still required

Peer to Peer SIP
No or minimal infrastructure
No customer support
Global
Advanced services

* Note: Skype is the largest VoIP service by any metric

* Better than Skype?*