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

- <u>Capacity glut</u> on the Internet backbones is the real problem
- Congested access links to the Internet maybe the problem

Useful Technologies

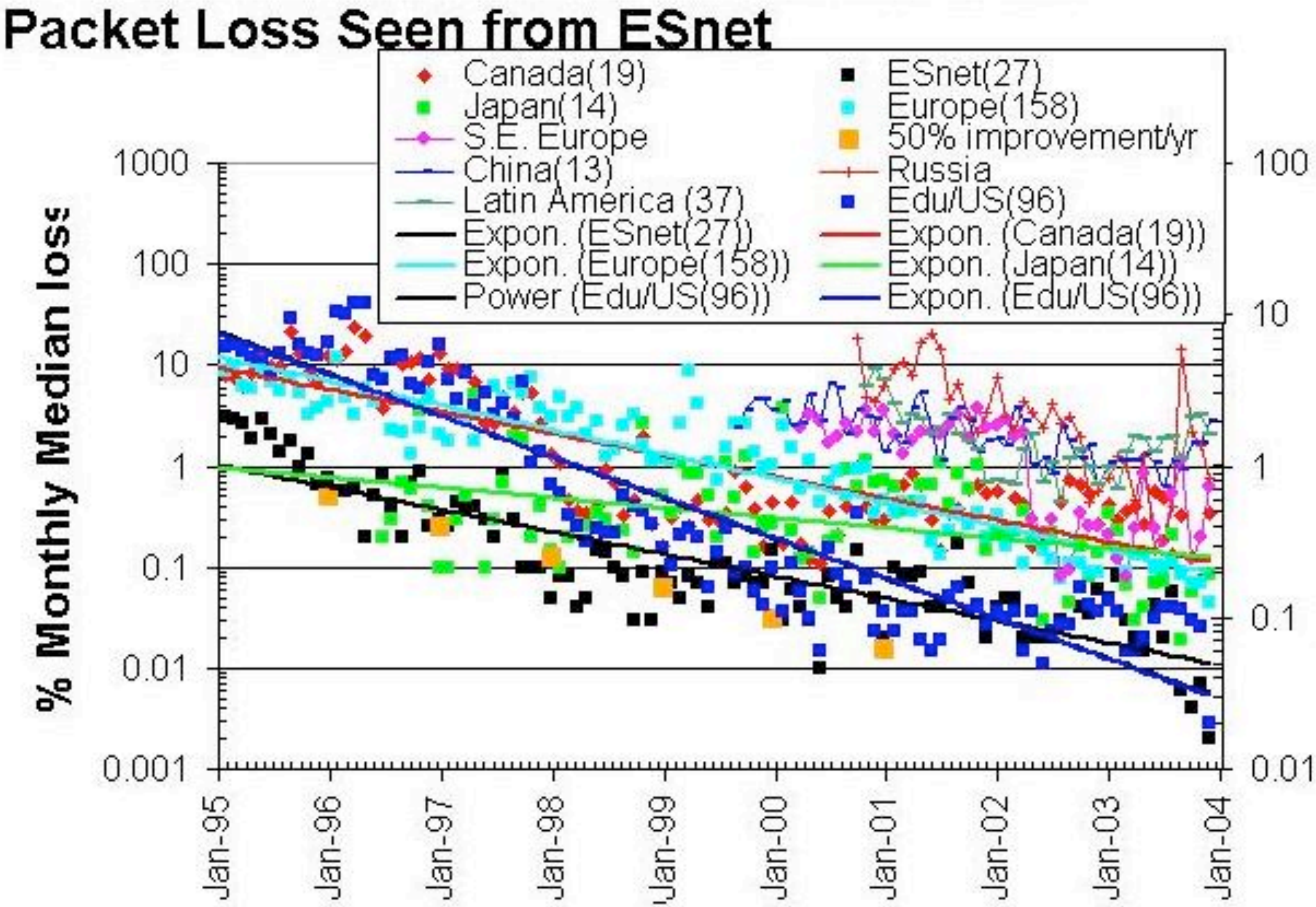
- Avoid obsolete ITU-T G.7xx codecs and their license payments
- MPLS can invoke DiffServ but why not just use DiffServ wherever possible
- Use the iLBC free codec as default
- Use voice quality monitoring only where users perceives quality: In the SIP endpoints so as to avoid error due to the fractal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

* http://www.ietf.org/internet-drafts/draft-sinnreich-sipdev-req-07.txt

Internet Facts for VoIP

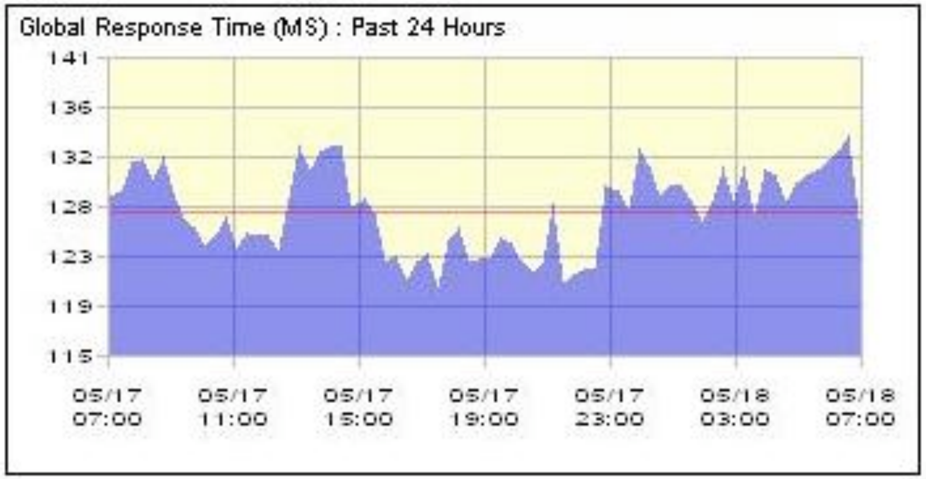
- Packet loss on the Internet is in the 1%-0.1% range
- Delay on the Internet is close to that on fiber
- Internet codecs provide the better voice
- 2/3 of Internet traffic is P2P anyhow
- P2P communications dominate VoIP: Skype

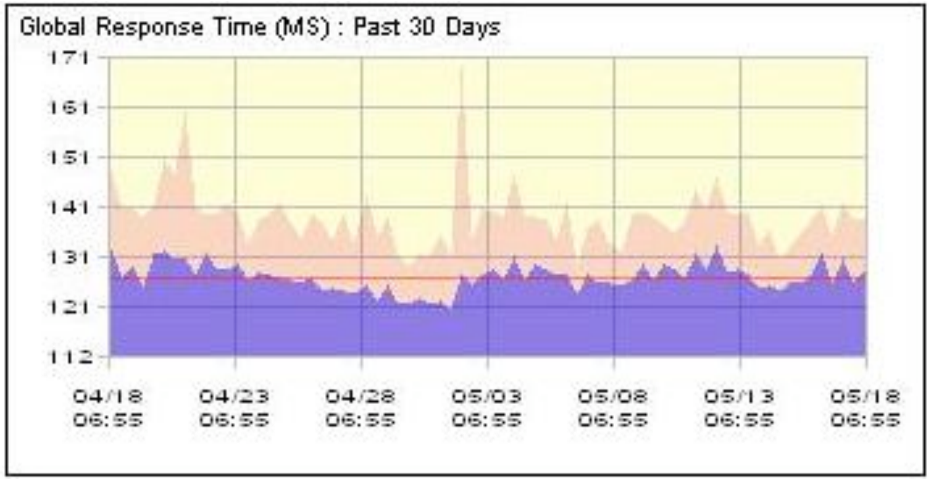
Internet Packet Loss Trend: From 1% to 0.1%



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

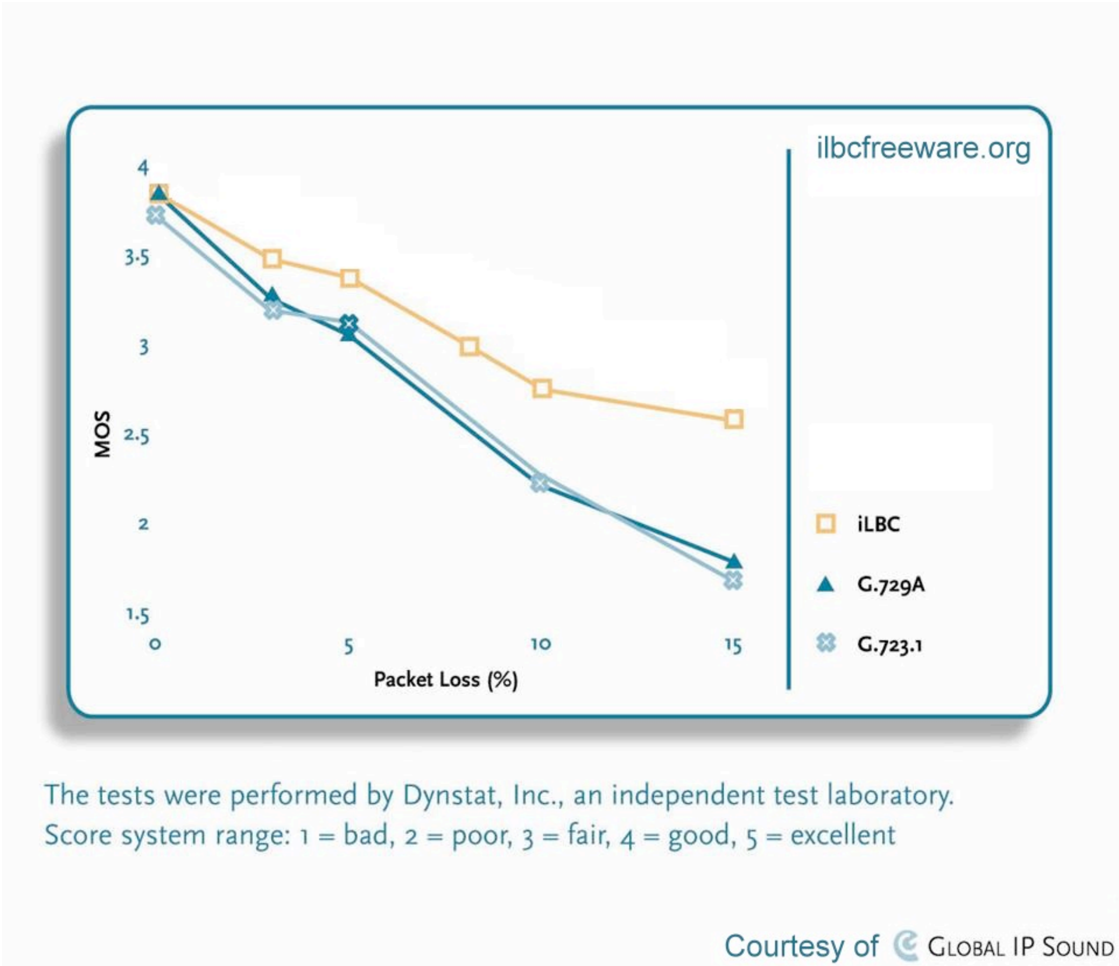
Internet Delay





http://www.internettrafficreport.com/main.htm

Internet Codec Performance

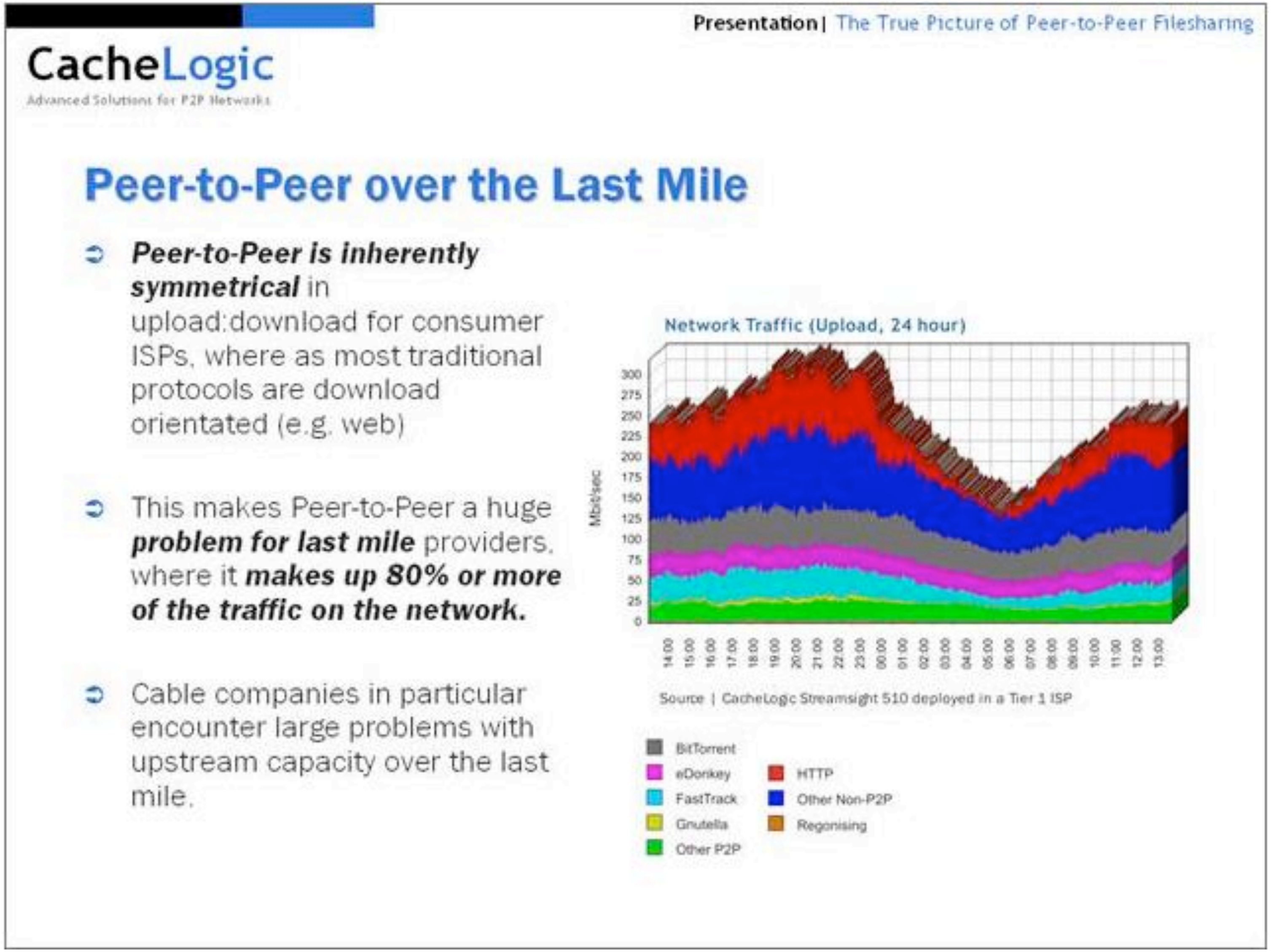


http://www.ilbcfreeware.org/

RFC 3951: Internet Low Bit Rate Codec (iLBC)

RFC 3952: RTP Payload for iLBC

Internet Traffic Load: Mostly P2P



http://www.cachelogic.com/research/index.php

SIP Models and Skype

Better than Skype?*

Client Server SIP

Peer to Peer SIP

No or minimal infrastructure

No customer support

Global

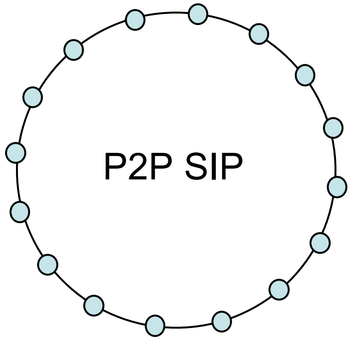
Advanced services

Post 2005 Commercial SIP:

Interdomain

Session Border Controller

ENUM is still mostly discussed



2004: Columbia University

Just like in 1997

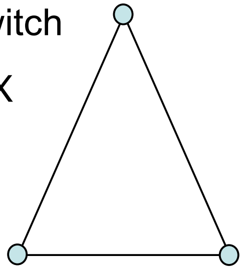
Key standards work is still required

2004 Commercial SIP:

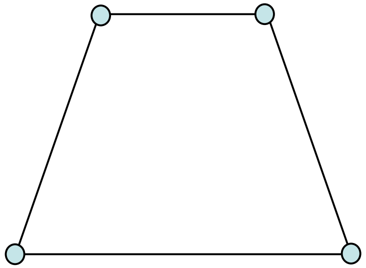
VoIP islands

Softswitch

IP PBX



1999: RFC2543



2002: RFC3261

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