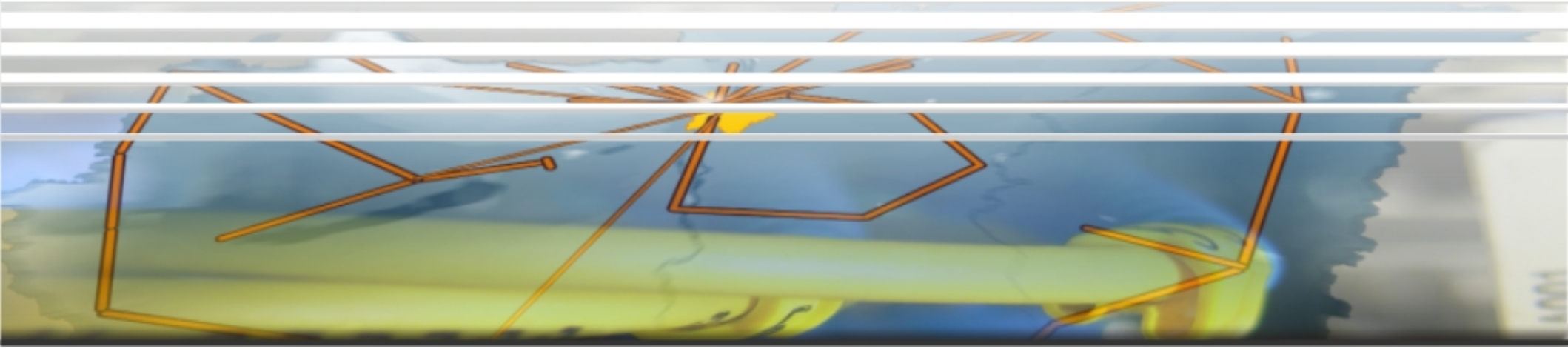
SIP Last but not Least!



05/12/10
Budapest / Hungary

MÉSZÁROS
Mihály

Environment (Network, Management, etc.)

- Network
  - Robust no Single Point of Failure
- Server Interface bonding
  - No downtime during switch or router software upgrade.
- QoS
  - over-provisioning
  - L3 DSCP painting packets
  - L2 CoS
- VLAN
  - Separate and segment LAN
  - security
- Rapid Spanning Tree
  - Fast l2 recovery old SPT is not good enough.
- Monitor everything!
  - Network
    - Netflow, SNMP
    - Nagios, Cricket, etc.
  - Service monitoring
    - Nagios, Munin, etc.
- Avoid single hard disk failure
  - Raid (mirroring)
- Backup
  - Backup data every night
    - Bacula, rsync etc.
  - Dump database recently
- Firewall
  - router ACL
  - netfilter/iptables (ferm)

Environment (Power / Emergency)

- Plain Old Telephony Service (POTS)
- Old features comparing to VoIP
  - reliability 99.9999%
  - local loop
  - Central Office/PBX as power source
- Voice/Video over IP
  - Power consumption
    - Green terminals (If N is big N x 30Watt can be huge)
    - Servers reliable, low power
  - PoE
    - switch (UPS),patch panel, terminal
  - Turned off terminals
    - (Wake on LAN?)
  - UPS
    - proxy, b2bua, terminal
- Emergency
  - PSAP
    - Public Safety Answering Point
  - Emergency case Location of PSAP and caller endpoint
    - POTS
      - Location of the caller is known exactly
    - Mobile
      - not 99.9999% solution
        - battery can be low
        - signal can be low
        - Network is complex, downtime, SP network can have outage
    - VoIP
      - complex
      - The location of the terminal currently mostly unknown
      - IETF ecrit WorkGroup

Accounting, CDR, Call log

- Proxy
  - Stateless
  - Stateful (only transaction statefull!)
- UA
  - dialog stateful
- Network border UA
  - B2BUA
  - SBC
- Database
  - SQL, Radius, Diameter
- Call Rating

QoS

- Per Hop Behavior (PHB)
  - DSCP(Differentiated Services Code Point)
    - 6bit
- Commonly defined Per Hop Behavior(s)
  - Expedited Forwarding(EF)
    dedicated, low-loss, low-latency
    - 101110
  - Assured Forwarding(AF)
    gives assurance of delivery under conditions
    - next silde
  - Class Selector (CS)
    backward compatibility with the IP Precedence
    - XXX000
    - CS1 = 001000, CS2 = 010000, CS3 = 011000, etc.
  - Default
    best effort
    - 000000

Assured Forwarding Classes

	Class 1	Class 2	Class 3	Class 4
Low Drop Prec	001010	010010	011010	100010
Medium Drop Prec	001100	010100	011100	100100
High Drop Prec	001110	010110	011110	100110

- Class1
  - AF11 = '001010'
  - AF12 = '001100'
  - AF13 = '001110'
- Class2
  - AF21 = '010010'
  - AF22 = '010100'
  - AF23 = '010110'
- Class3
  - AF31 = '011010'
  - AF32 = '011100',
  - AF33 = '011110'
- Class4
  - AF41 = '100010'
  - AF42 = '100100'
  - AF43 = '100110'

QoS recommendation (based RFC4594)

Application / Service	L3		L2	IETF
	PHB	DSCP	COS	RFC
Network Control	CS6	110000	6	RFC2474
VoIP Telephony	EF	101110	5	RFC3246
Call Signaling	CS5	101000	5	RFC2474
Multimedia Conferencing	AF41,AF42,AF43	100010,100100,100110	4	RFC2597
Real-Time Interactive	CS4	100000	4	RFC2474
Multimedia Streaming	AF31,AF32,AF33	011010,011100,011110	3	RFC2597
Broadcast Video	CS3	011000	3	RFC2474
Low-Latency Data	AF21,AF22,AF23	010010,010100,010110	3	RFC2597
OAM(Operations, Administration, Management)	CS2	010000	2	RFC2474
High-Throughput Data	AF11,AF12,AF13	001010,001100,001110	1	RFC2597
Best Effort	DF(CS0)	000000	0	RFC2474
Low-Priority Data	CS1	001000	1	RFC3662

- Cisco QoS recommendation differ! they commute two PHB
  - Cisco marking Call Signaling CS3
  - Cisco marking Broadcast Video CS5
- L2 COS values based on Cisco recommendation

Auto provisioning

- DHCP
  - Option 66 (Only one IP address or hostname can be set as value)
  - Option 150 (Can contain multiple TFTP server IP addresses for redundancy reason)
- Transport Protocols
  - Insecure
    - TFTP, HTTP, FTP
  - Secure
    - HTTPS
      - Client certificate authentication
- Config file encryption
- Update enpoint after config change
  - SIP notify
  - Periodic check
- Provisioning framework
  - A Framework for Session Initiation Protocol User Agent Profile Delivery
    - http://tools.ietf.org/html/draft-ietf-sipping-config-framework-16

Dialplan

- processing/validating dialed number
  - sip phone dialplan
  - proxy or b2bua dialplan
- corporate extensions
  - numeric/digits
  - 4 or 3 digit
  - ISN (ITAD Subscriber Number)
    - ITAD (Internet Administrative Domain)
    - 1234*356
    - 4.3.2.1.356.freenum.org
  - alphanumeric
    - URI
      - sip:bob@biloxi.test
      - h323:atlanta.test
      - tel:+36-1-4503084

Hungarian dialplan (General)

- Country code
  - +36
- geographical/fixed landline
  - 1[0-9]{7}
  - 2[23456789][0-9]{6}
  - 3[234567][0-9]{6}
  - 4[2456789][0-9]{6}
  - 5[234679][0-9]{6}
  - 6[23689][0-9]{6}
  - 7[23456789][0-9]{6}
  - 8[2345789][0-9]{6}
  - 9[234569][0-9]{6}
- unused prefix
  - 3[189]
  - 4[13]
  - 5[08]
  - 6[01457]
  - 8[16]
  - 9[78]
- mobil
  - [237]0[0-9]{7}
- freephone
  - 80[0-9]{6}
- shared cost
  - 40[0-9]{6}
- internet dial-up
  - 51[0-9]{6}
- corporate network
  - 71[0-9]{6}
- geographic test number
  - 55[0-9]{6}
- nomadic
  - 21[0-9]{7}
- premium
  - 90[0-9]{6} adult
  - 91[0-9]{6}

Hungarian dialplan (Short identifiers)

- emergency
  - 104
  - 105
  - 107
  - 112
- prefixes
  - 00
    - international call
  - 06
    - national call
  - 13[0-1]
    - caller id hiding
  - 15[0-9]{2}
    - alternative service provider prefix
- short numbers:
  - 116[0-9]{3}
  - 118[0-9]{2}
  - 139[0-9]
  - 14[01234][0-9]
  - 14[56789][0-9]{2}
  - 17[0-9]{1,3}
  - 18[056789]
  - 18[1234][0-9]
  - 19[0-9]

Directory / Phonebook

- Corporate Directory, Phonebook can be made using the SIP Terminal inbuilt LDAP or Integration API.
- LDAP (H.350?)
  - base DN
  - filter
  - mapping
- Integration API
  - XML micro browser
    - standard W3C XML browser
  - XML API
    - Vendor specific schema
    - XML RPC
- With terminal integration API very varied applications can be easily developed.
  - resource booking
  - temperature
  - etc.

Media, DTMF, FAX

- RTP
  - Real Time Transport Protocol
- RTCP
  - Real Time Transport Control Protocol
- Offer Answer model
  - Early Offer
  - Delayed offer
- SDP
  - Session Description Protocol

```
v=0
o=alice 2890844526 2890844526 IN IP4 host.biloxi.test
s=
c=IN IP4 host.biloxi.test
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
```

- DTMF
  - rfc2833(obsolated) New rfc4733, rfc4734
    - named telephony event in RTP stream
  - Inband rtp
  - INFO
    - in signaling out of band
  - NOTIFY
  - KPML (SUBSCRIBE - NOTIFY)
- Fax
  - Inband
  - T.38
    - udp transport
    - not rtp (QoS)
    - encryption (srtp?)

Codecs

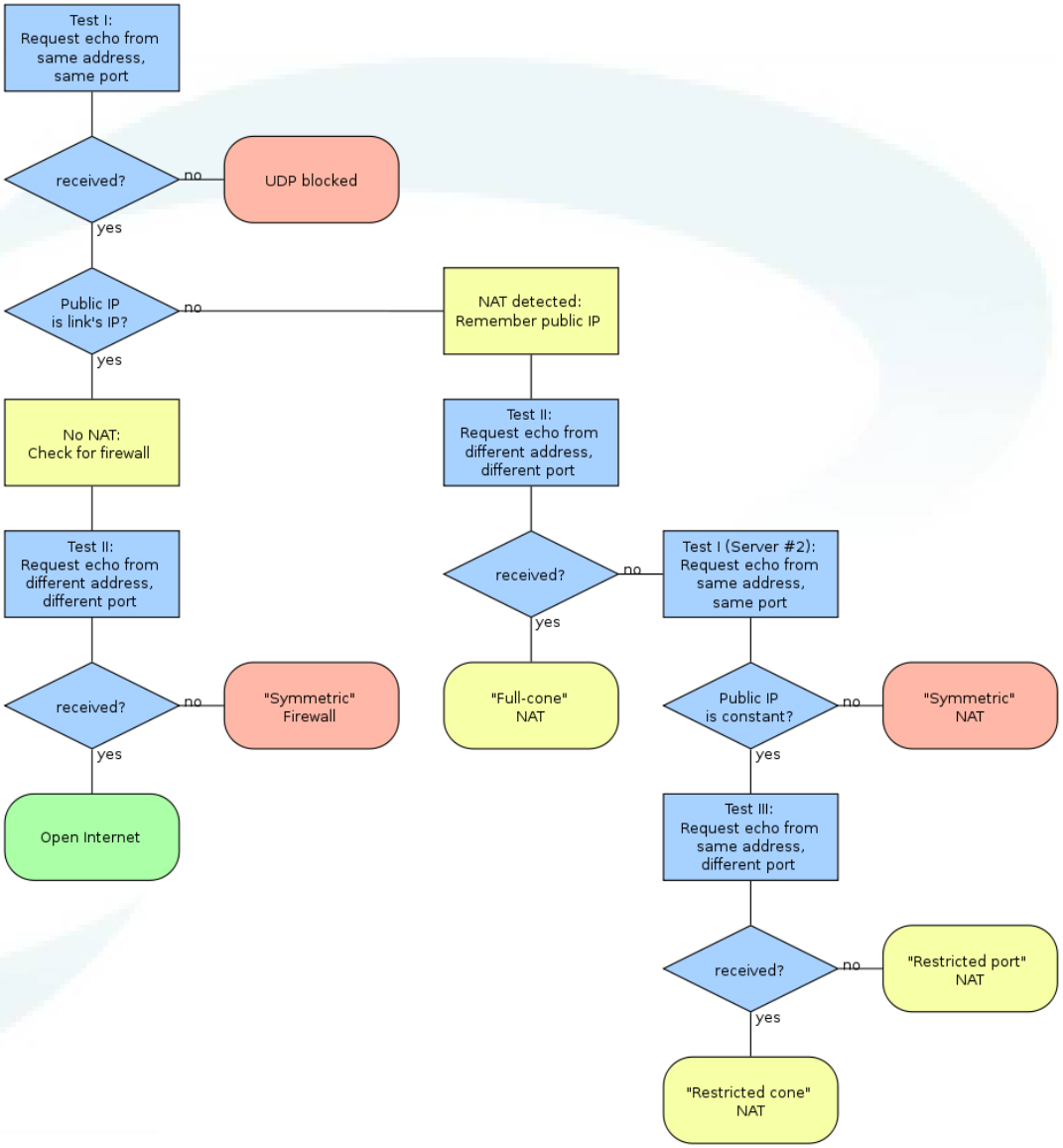
- Audio
  - G.711 alaw, ulaw
  - G.729A
  - G.722
  - G.722.1
  - G.722.1 Annex C
    - Polycom Siren 14
  - Polycom Siren 22
  - MPEG4 AAC-LD
- Ext.
  - Automatic Gain Control
  - Automatic Noise Suppression
    - keyboard, mobile, environment, etc.
  - Echo Cancel
- Video
  - H.261
  - H.263
    - H.263+(1998, v2)
    - H.263++(2000, v3)
  - H.264
    - AVC
    - SVC
  - Theora
    - based on VP3
    - open standard
  - Dirac
    - BBC

Security

- Problems
  - Software bugs
  - Careless administretos
  - Other mistakes
- Complex system
- VLAN L2 separation
  - data
  - voice
- IP Security
  - filtering, firewall, ACL
    - OS based
    - Router Based
- DoS,DDoS
- SPIT
- SIP
  - privacy
  - Asserted Identity
    - rfc4474
  - TLS (hop-by-hop)
  - SRTP
  - Digest
    - password strength
    - plaintext password
  - S/MIME (unused today)
- motivation
  - money (access to toll calls)
- Accounting, Log

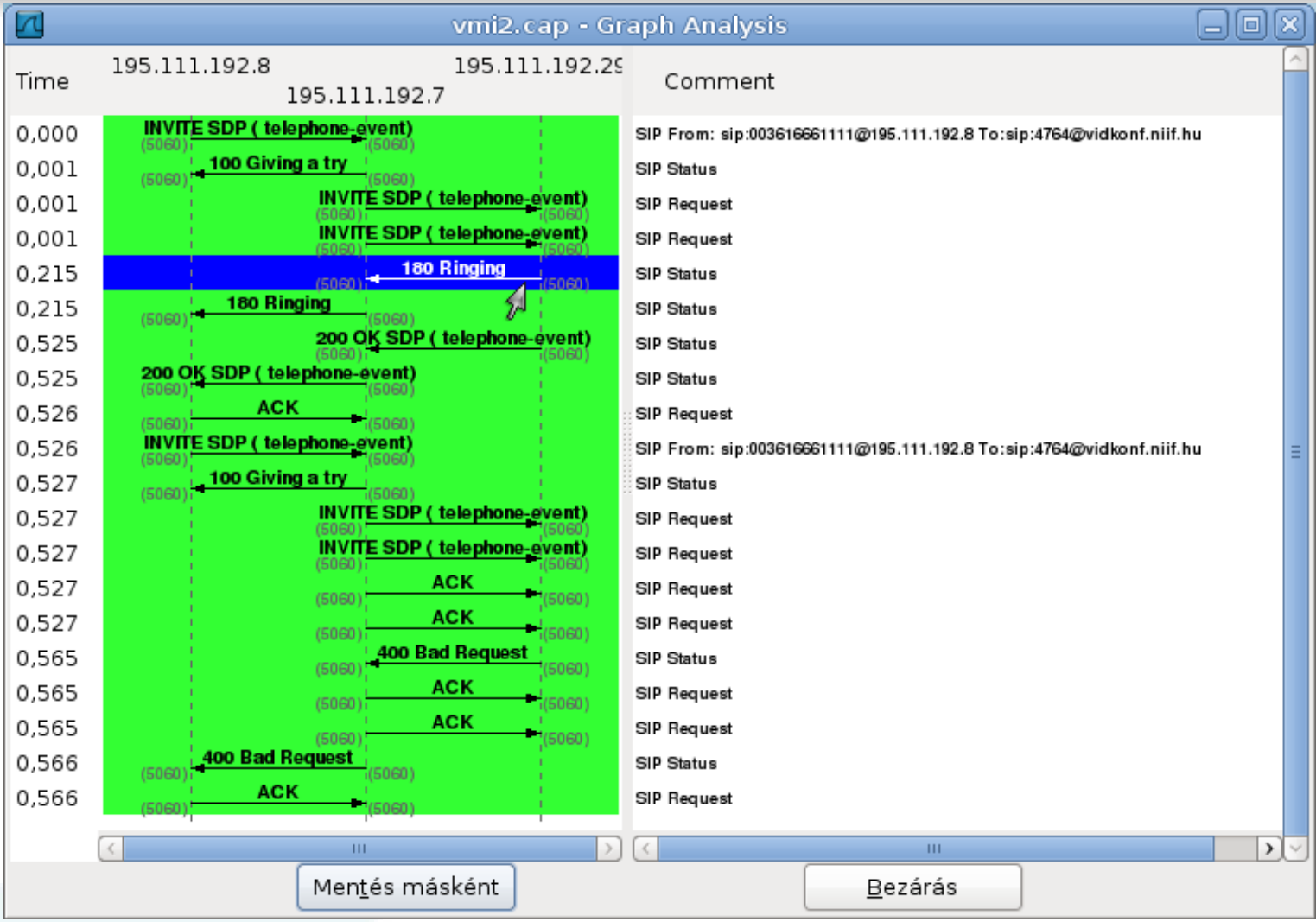
NAT

- STUN ("Classic STUN")
  - client, server modell
  - what is my outside ip and port?
  - 2 different IP address
  - 2 port (default 3478)
  - SRV
    - _stun._udp
    - _stun._tcp
  - NAT TYPE (obsolated)
    - Full Cone
    - Restricted Cone
    - Port Restricted Cone
    - Symmetric
- TURN (Traversal Using Relay NAT)
  - Relaying data
  - Symetric nat
- ICE (Interactive Connectivity Establishment)
  - ICE use STUN and TURN
  - offer-answer modell



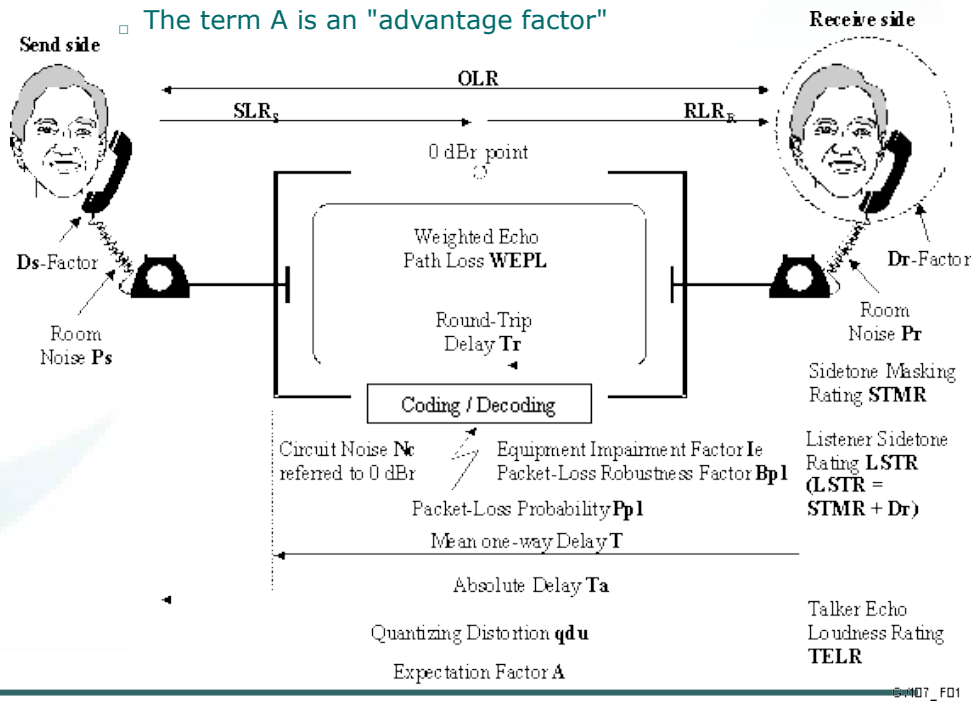
Test tools: debug, bechmark, load, torture

- Torture
  - RFC4475
  - RFC5118
- SIPp
  - sceanrios
  - XML
    - UAC
    - UAS
    - 3PCC
- SFTF
  - old
- WIRESHARK
  - voip calls
  - Graph analysis
  - RTP reassembly



Measuremnet

- MoS
  - measures subjective call quality for a call.
  - MOS scores range:
    from 1 for unacceptable
    to 5 for excellent.
- PESQ
  - Perceptual Evaluation of Speech Quality
  - reference test signal
  - requires a license from Psytechnics a spinoff from British Telecom
- ICPIF
  - Calculated Planning Impairment Factor
  - Icpif = Io + Iq + Idte + Idd + Ie – A
    - Io represents impairments caused by non-optimal loudness rating,
    - Iq represents impairments caused by PCM quantizing distortion,
    - Idte represents impairments caused by talker echo,
    - Idd represents impairments caused by one-way transmission times (one-way delay),
    - Ie represents impairments caused by equipment effects, such as the type of codec used for the call and packet loss, and
    - A represents an access Advantage Factor
- E-Modell (R factor)
  - R = Ro - Is - Id - Ie,eff + A
    - The term Ro expresses the basic signal-to-noise ratio (received speech level relative to circuit and acoustic noise).
    - The term Is represents all impairments that occur more or less simultaneously with the voice signal, such as: too loud speech level (non-optimum OLR), non-optimum sidetone (STMR), quantization noise (qdu), etc.
    - The term Id sums all impairments due to delay and echo effects.
    - The term Ie,eff is an "effective equipment impairment factor", which represents impairments caused by low bit-rate codecs. It also includes impairment due to packet-losses of random distribution.
    - The term A is an "advantage factor"



SIP Last but not Least!

References

- English
  - http://www.nhh.hu/dokumentum.php?cid=10780
  - http://www.nhh.hu/dokumentum.php?cid=10746&letolt
  - http://webold.nhh.hu/aga/common/setLanguageAction.do?lang=en
  - http://commons.wikimedia.org/wiki/File:STUN_algoritmus.svg
  - http://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm
  - http://www.cisco.com/en/US/docs/ios/12_4/ip_sla/configuration/guide/hsvoipj.html
  - QOS
    - http://en.wikipedia.org/wiki/Differentiated_services
    - http://tools.ietf.org/html/rfc2597
    - http://tools.ietf.org/html/rfc3246
    - http://tools.ietf.org/html/rfc2475
    - http://tools.ietf.org/html/rfc3260
    - http://tools.ietf.org/html/rfc2474
- Hungarian
  - http://www.nhh.hu/dokumentum.php?cid=8183
  - http://www.nhh.hu/dokumentum.php?cid=8880&letolt
  - http://webold.nhh.hu/aga/common/setLanguageAction.do?lang=hu
  - https://wiki.voip.niif.hu/index.php/Link

Thank You for Your time!

```
BYE sip:all@sipworkshop.hu SIP/2.0
Via: SIP/2.0/UDP pc.sipworkshop.hu:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
From: Mihály Mészáros <sip:misi@niif.hu>;tag=8321234356
To: All <sip:all@sipworkshop.hu>;tag=9fxced76sl
Call-ID: 3848276298220188511@sipworkshop.hu
CSeq: 1 BYE
Content-Length: 0
```

Thank You!