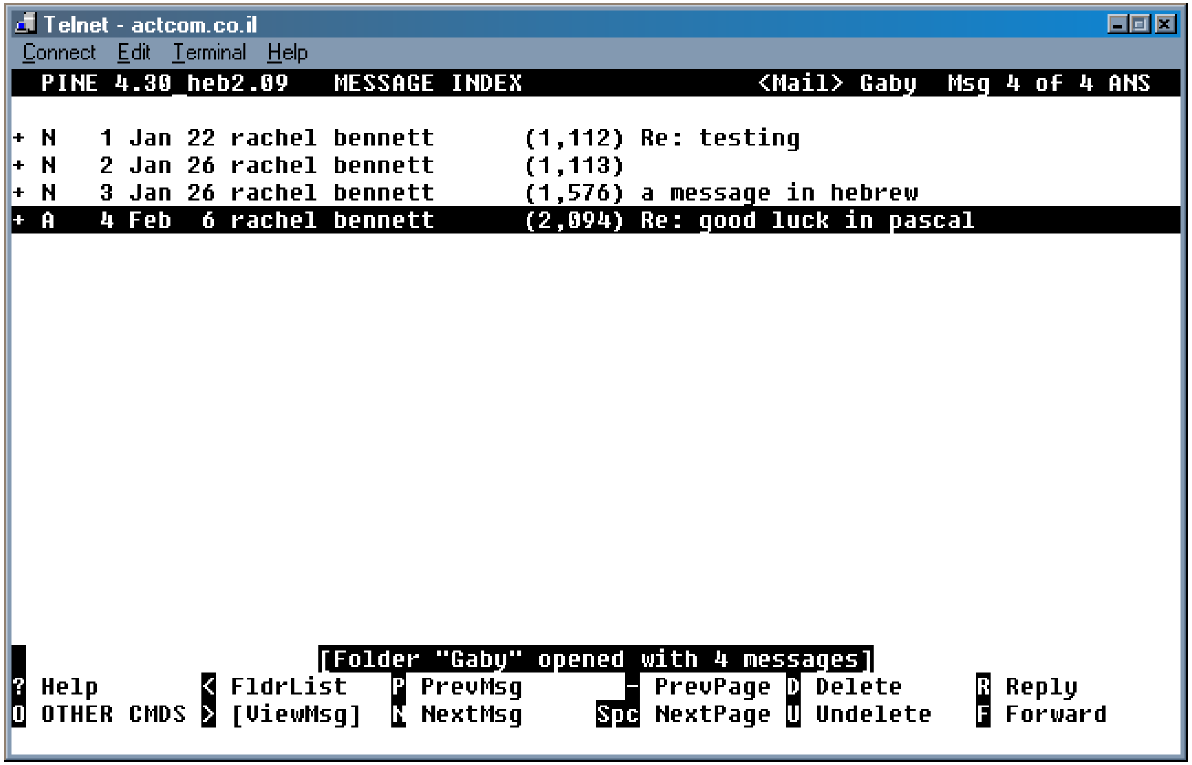
פרק 10 – אפליקציות בתקשורת IP

- Applications
  - Telnet
  - FTP
  - Email
  - Media Applications
    - VoIP
- Media Quality Measurement methods

Telnet – Remote Terminal

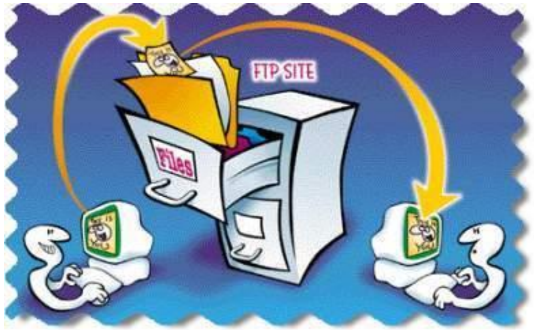
- Text based Terminal
- Allows users to remotely access a command shell on another computer
- Runs over a TCP connection
- Telnet flow:
  - Client opens a telnet request to the server (to port 23)
  - Server responds and opens a session (back to source port >1023).
- Used for system maintenance and accessing basic terminal based applications



FTP – File Transfer Protocol

- **Two way file transfer (copy) between client and server hosts**

- **Two** TCP Ports
  - **Port** 21 - control information
  - **Port** 20 - data files between the client and the server
- Features
  - Uses reliable TCP connections
  - Follows Client-Server model
  - Use two port/socket connections – control and data
  - Provides ability to perform some file related operations (list dir) in addition to file transfer
  - Client can send or receive files
  - Some character translation supported
  - Some basic file formats defined
  - Authentication – user, anonymous
  - Data Type and Transmission Mode (RFC959)



Internet Mail Protocols

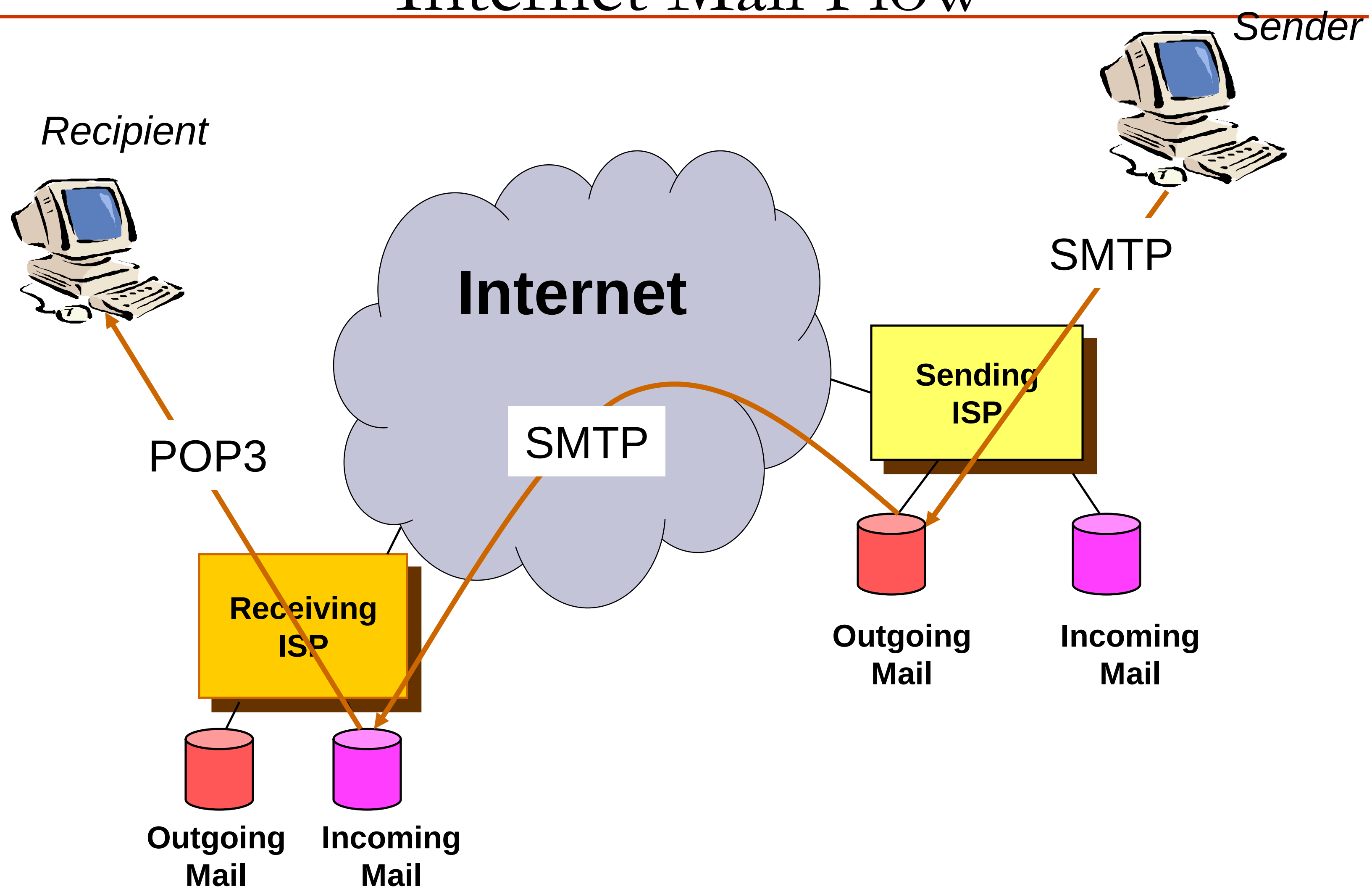
- **SMTP - Simple Message Transfer Protocol**
  - Interactive Command Based Mail Protocol for direct IP Connection
- **Mime - Multipurpose Internet Mail Extensions**
  - Extension to send non-ASCII
- **Addressing**
  - bill.gates@microsoft.com

▪Local mailbox

▪Mail Server Address
▪ - mail.microsoft.com

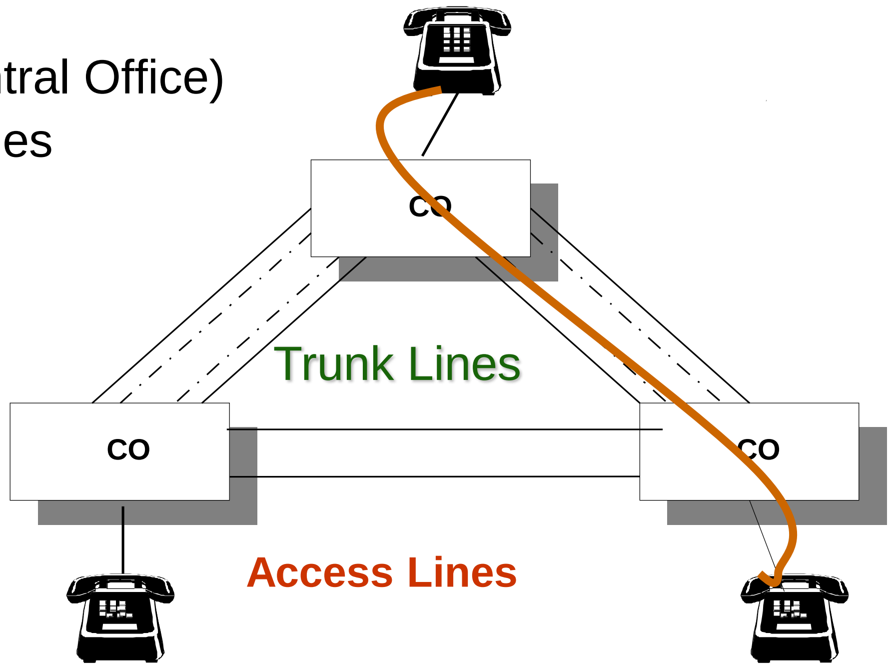
- POP3 - Post Office Protocol version 3
  - Offline Mail Client
  - Server sends/receives mail using SMTP
  - Receiver picks up mail from Post box using POP3
- IMAP – Internet Message Access Protocol
  - More Power Client/Server Mail System
  - Allows mail to be handled on Mail server
- Webmail
  - Online server through browser such as Hotmail

Internet Mail Flow



The Public Switched Telephone Network

- PSTN provides telephony service
  - QoS
  - E2E BW=64Kbps
  - Connection-Oriented
- The Network is made up of:
  - POTs
  - Class 5 Switches (CO - Central Office)
  - Class 4 and Tandem Switches



VoIP and NGN

- Voice over Internet Protocol-Internet Telephony
  - Digitizing voice
  - Encapsulating the digitized voice into packets
  - Transmitting those packets over a packet switched IP network

- VoIP technology is the foundation for the
- **N**ext **G**eneration **N**etwork concept and implementation

Why IP Telephony?

- **Cost reduction**
  - Lowered Infrastructure Costs
- **Operational Improvement**
  - LAN/Campus Integration
- **Add-on services**
  - Unified Messaging
  - Mobility
  - Distributed Contact Center
  - FMC (Fixed Mobile Convergence)
  - Unified Communication

3Cs: “Convergence” & “Costs” & “Competition”

PSTN versus IP Telephony

	Traditional PSTN	IP Telephony
Underlying Technology	TDM circuit switching	Packet switching
QoS guarantees	Yes	No
Network resource reserved at call setup	Yes	No
Network elements	Class 4, Class 5 switching systems	Gateways, Softswitch, SBC, App. Servers
Call processing intelligence	Mostly integrated in switching system	In separate gateway controllers
Bandwidth per call	64 kb/s	Variable 5.3 – 64 kb/s
Signaling	CAS, CCS (MFC, PRI, SS7)	SIP, H.323, MGCP, Megaco, Sigtran
How reliability achieved	Redundancy within each network element	Redundant IP elements through network

IP Telephony Protocols

MGCP

RTP/RTCP

SIP

SDP

Megaco

"The nice thing about standards is that you have so many to choose from; if you do not like any of them, you can just wait for next year's model." [Tanenbaum]

H.450

Sigtran

H.245

H.323

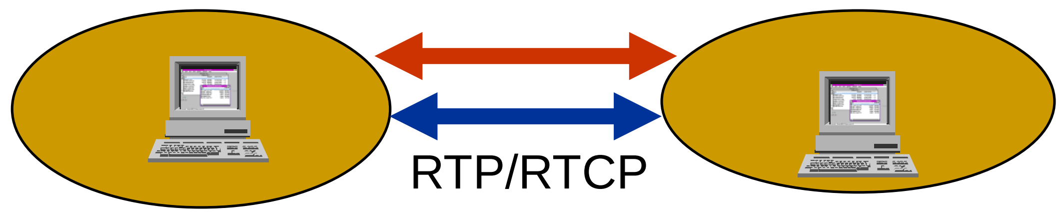
IP Telephony Standards

- ITU – International Telecommunications Union
  - H.323
  - H.248/Megaco
- IETF – Internet Engineering Task Force
  - RTP, RTCP
  - SIP
  - MGCP
  - Megaco

IP Telephony Network Architecture

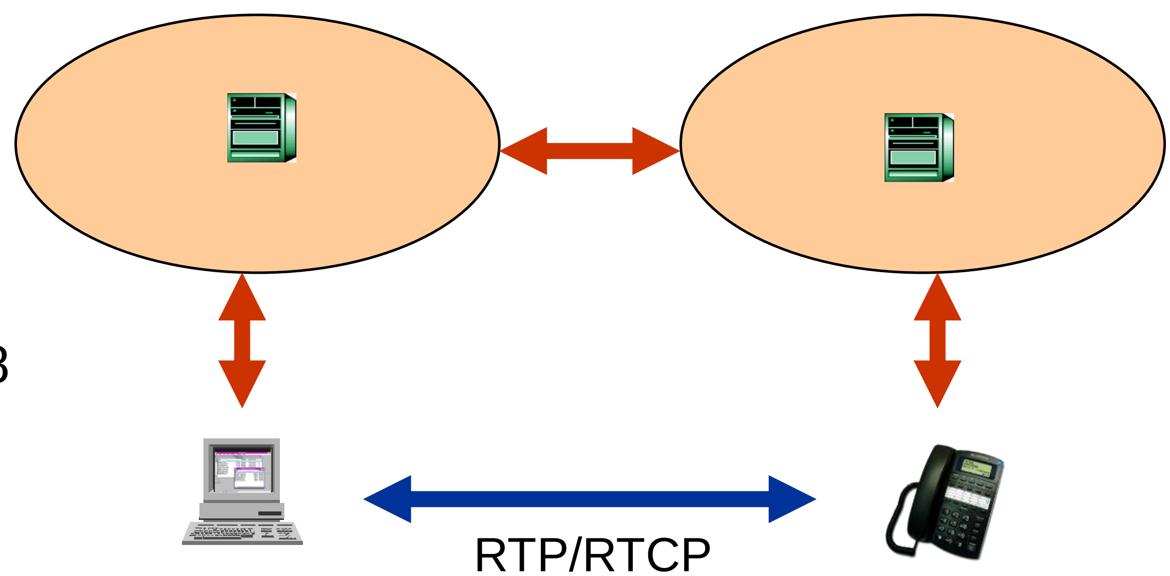
Peer To Peer

- H.323
- SIP

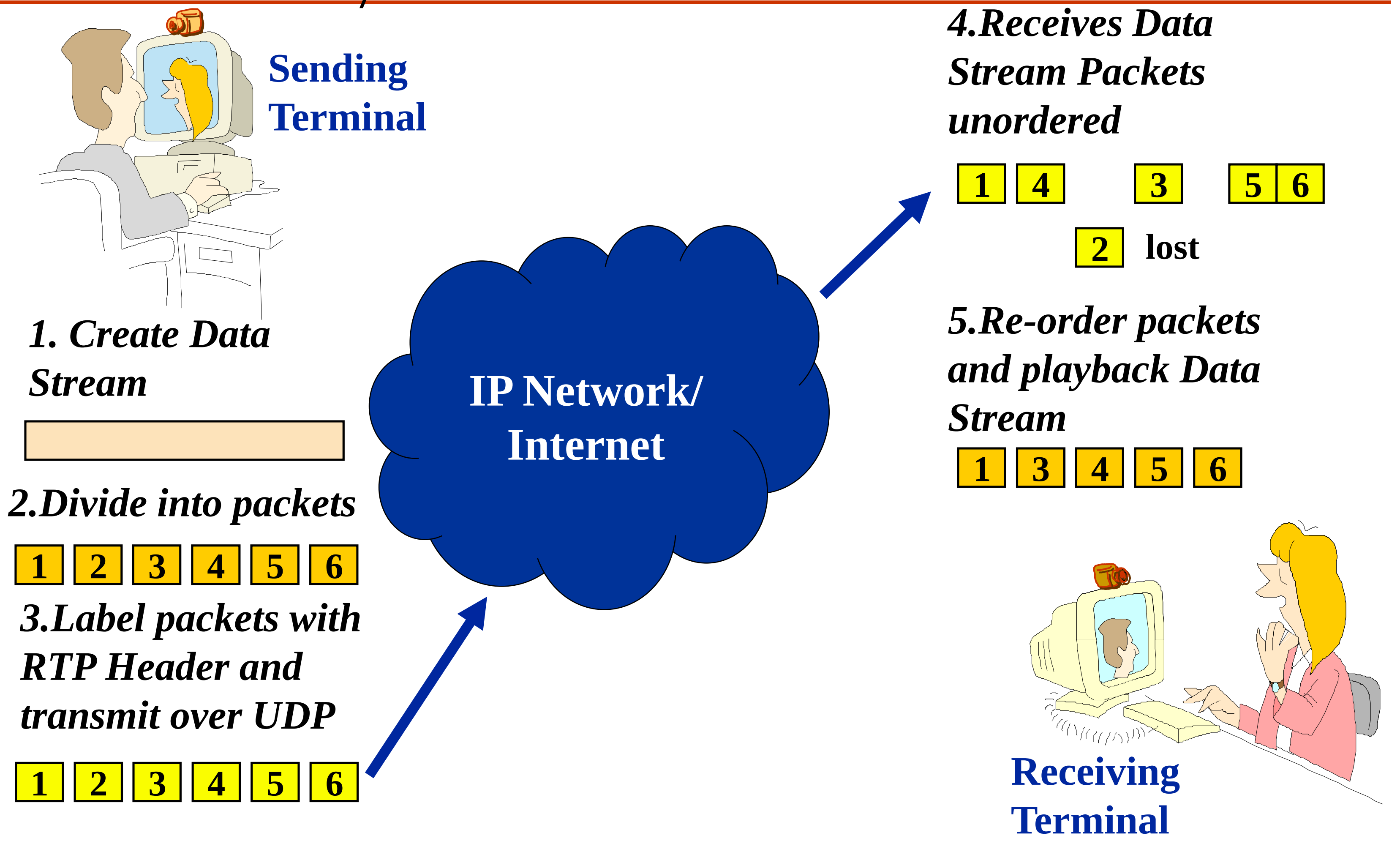


Master/Slave

- NCS
- MGCP
- Megaco/H.248

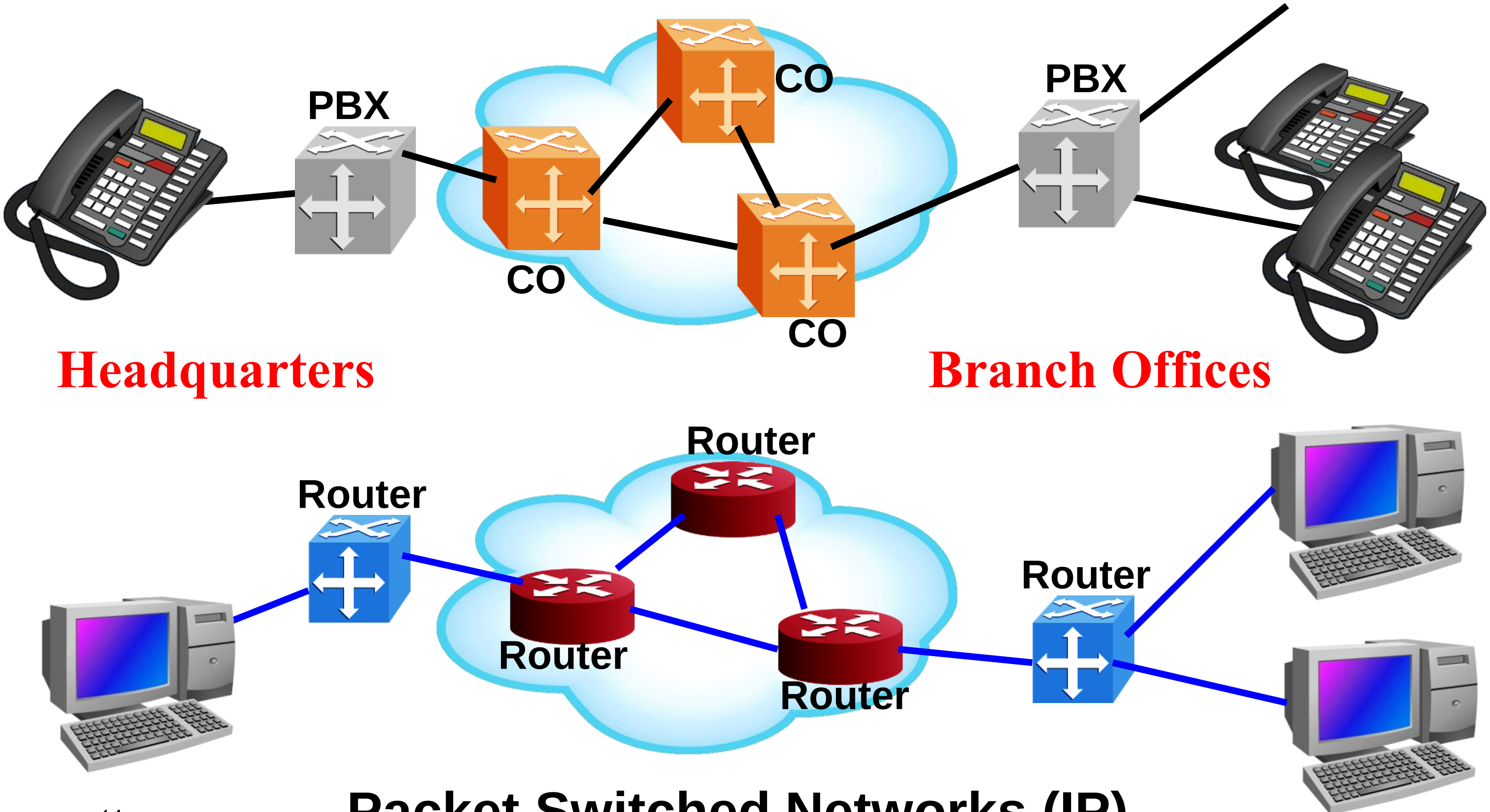


RTP/RTCP- for Media Transfer



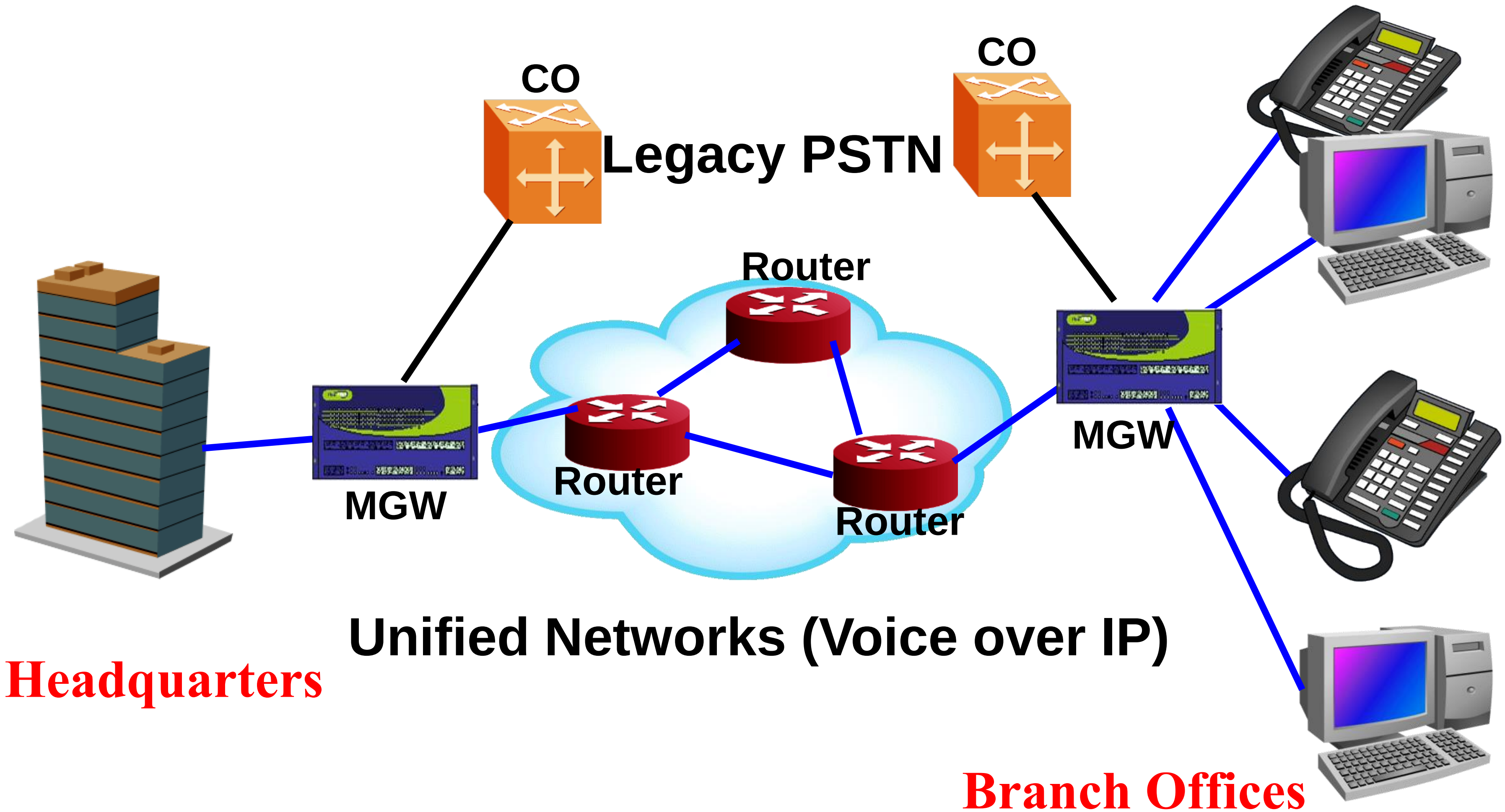
Enterprise: Yesterday's networks

Circuit Switched Networks (Voice)

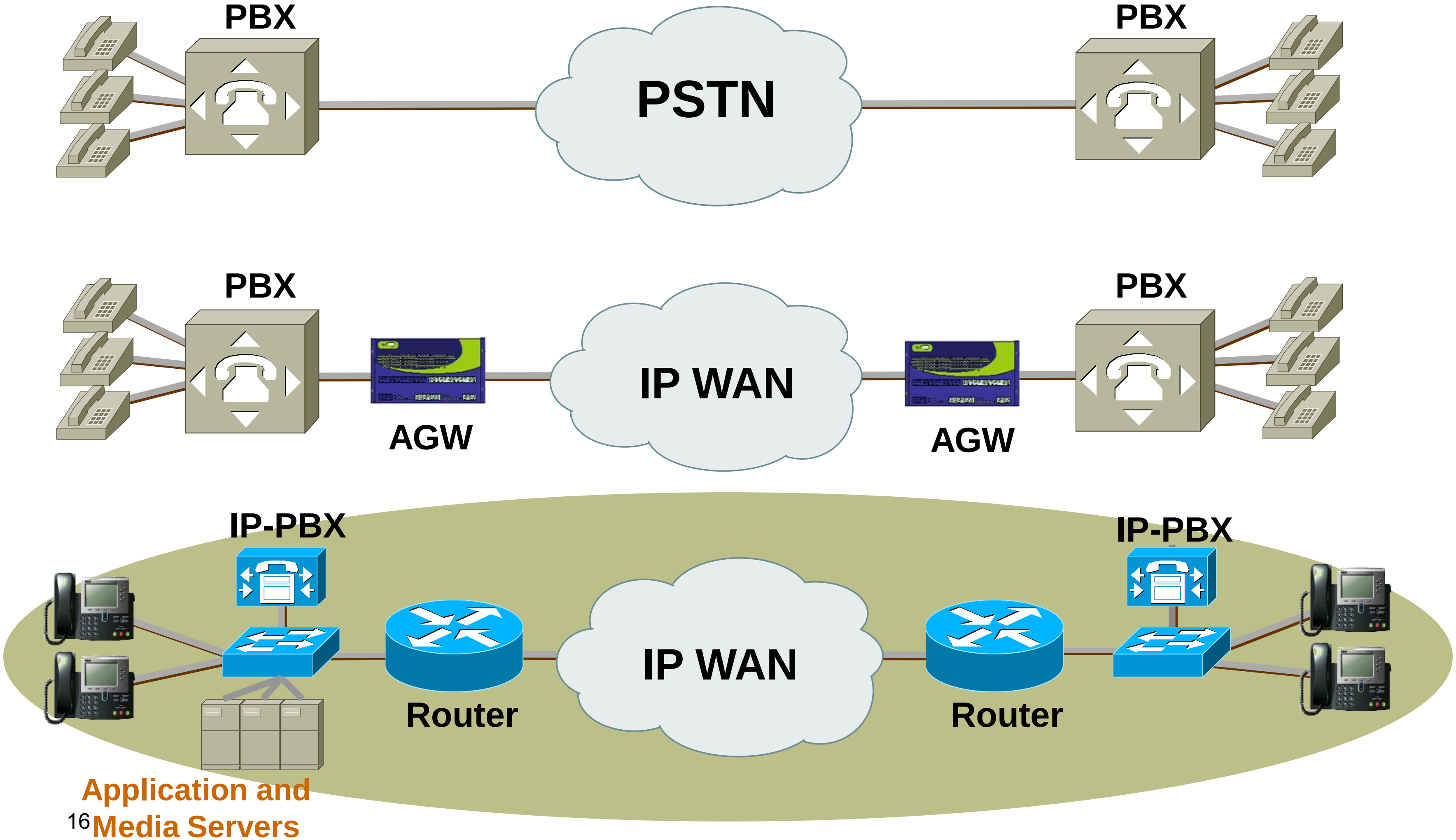


Packet Switched Networks (IP)

Unified/Converged Networks

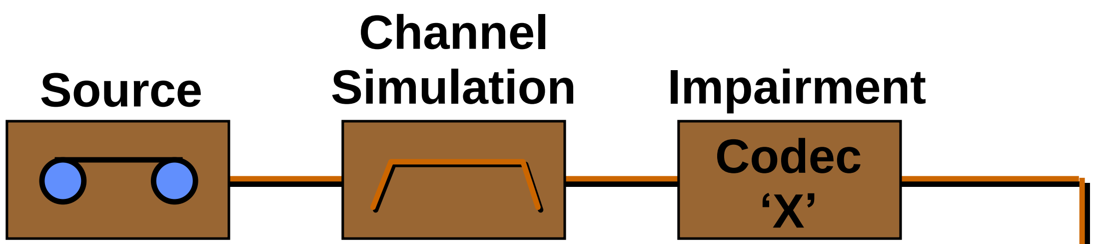


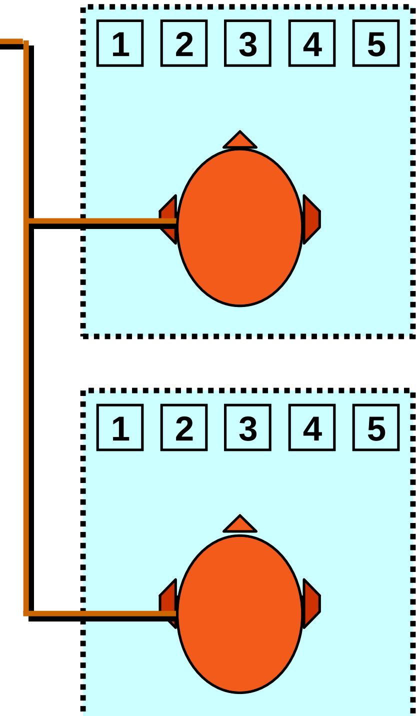
Enterprise Migration process to VoIP



Speech Quality measurement

MOS Scale





Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible, not annoying
3	Fair	Perceptible, slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying, objectionable

MOS of 4.0 = Toll Quality

Objective Speech Measurement

- Subjective tests- expensive, slow and unusable in applications like service monitoring
- Objective models - based on human perception for predicting the results of subjective tests
- Complex algorithms were developed
  - Emphasize only those parts of the input speech signal of interest to human listeners

Well-Known Objective methods

- PSQM, PSQM+: Perceptual Speech Quality
  - Developed by KPN ITU-T P.861

- PAMS: Perceptual Analysis Measurement System
  - Developed by British Telecom

- PESQ: Perceptual Evaluation of Speech Quality
  - ITU-T P.862

- **Intrusive** listening speech quality assessment tools
  - Compare original, unprocessed signals with degraded output from the communications system

Well-Known Objective Standards

- Tests use speech samples from both male & female speakers, or from artificial speech-like test signals

- PSQM output scores range from 0-6.5
  - The lower the score, the better the voice quality

- PAMS output scores range from 1-5
- PESQ output score range from 1-4.5
  - The higher the score, the better the voice quality

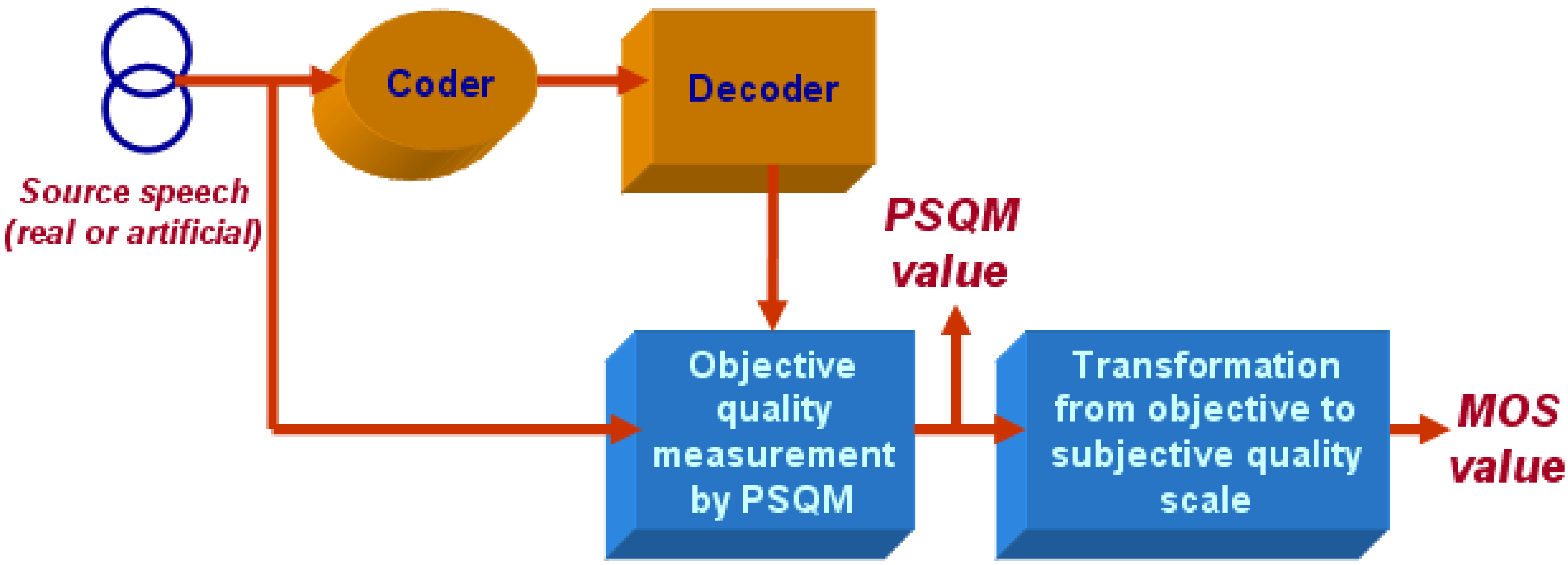
- E-Model is non-intrusive method
  - R-Factor in the range of 0-100

PSQM Standard

- Defined by ITU-T Recommendation P.861
- Algorithm provides method for speech within the 300-3400 Hz voice bandwidth
- Evaluates whether a particular voice signal is distorted in such a way that will be annoying/distracting to human listeners

PSQM Standard (cont.)

- Algorithm compares the signal received after the coder/decoder process with the original signal



PSQM+ Standard

- Enhanced version of PSQM
  - Correlates more to MOS scores in the presence of network degradation
- Algorithm performs some post-processing on the PSQM value based on PSQM and signal power
- Accounts for different perceptions of volume and loud distortions

PSQM+ Standard (cont.)

- As speech signals vary greatly in time, correlation errors can occur
  - Jitter of 20 mSec can cause → drop in quality equivalent to approx. 1 MOS

  - PSQM/PSQM+ models show very low correlation with subjective opinion

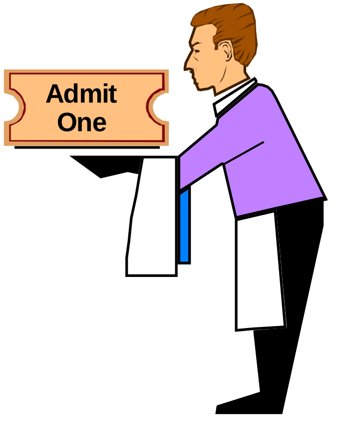
PAMS Method

- Designed to provide a repeatable, objective means for measuring perceived voice quality
- Uses different, effective signal processing model to PSQM which produces different types of scores
- Provides a "Listening Quality" and "Listening Effort" scores
- Signals in the model are divided into sections known as utterances

PAMS Method (cont.)

- Delay packet-base transmission such as IP telephony are identified
- Computes two quality ACR (absolute category rating method) measures:

  - Listening quality opinion scale 5-1
  - Listening effort opinion scale 5-1



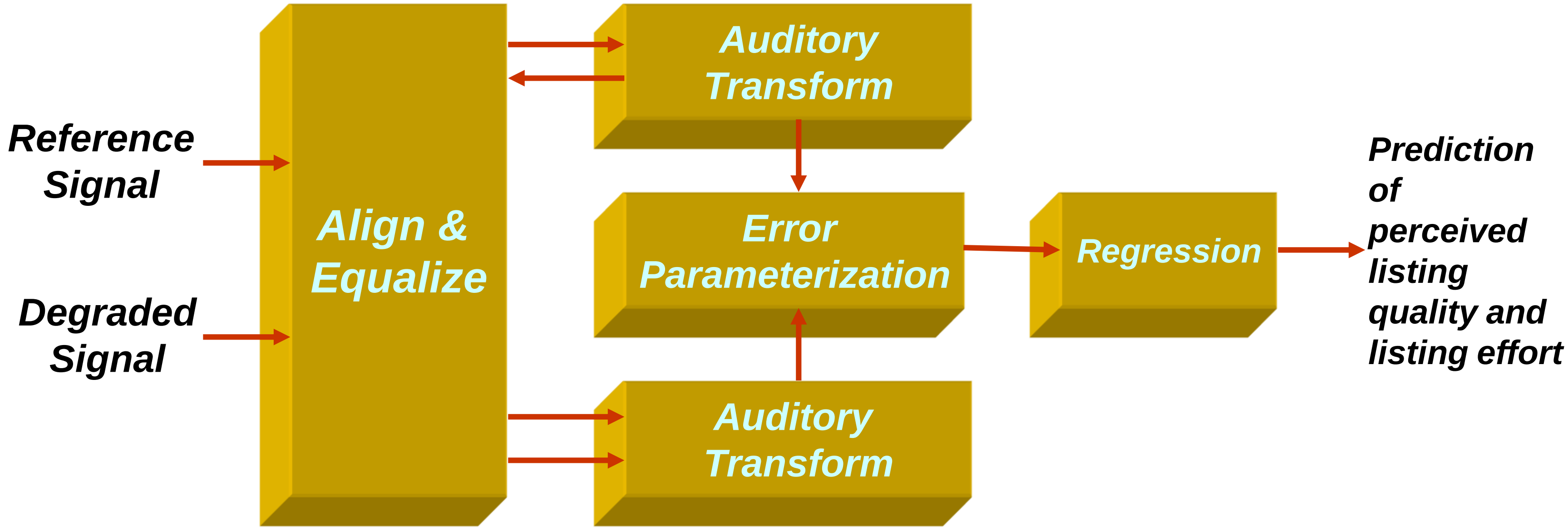
PAMS Method

- Measures listening quality in intrusive applications
- Accurate at predicting subjecting quality (MOS) at wide range applications
- Provide additional information to identify quality problems
- Superior performance to PSQM (ITU-T P.861), PSQM+
- Listening Quality and Listening Effort are MOS scaled which is defined in ITU standard P.800

PAMS Method (cont.)

- Algorithm performs variable delay time alignment and transfer function equalization for reliable subjective assessment

PAMS Structure



Source: BT

PESQ Standard (ITU-T P.862)

- PESQ: Perceptual Evaluation of Speech Quality
- An objective speech quality measure applicable to both speech codecs and end-to-end measurements
- Listening quality rating: 1-4.5
- Combines the strongest parts of PAMS and PSQM+ algorithms

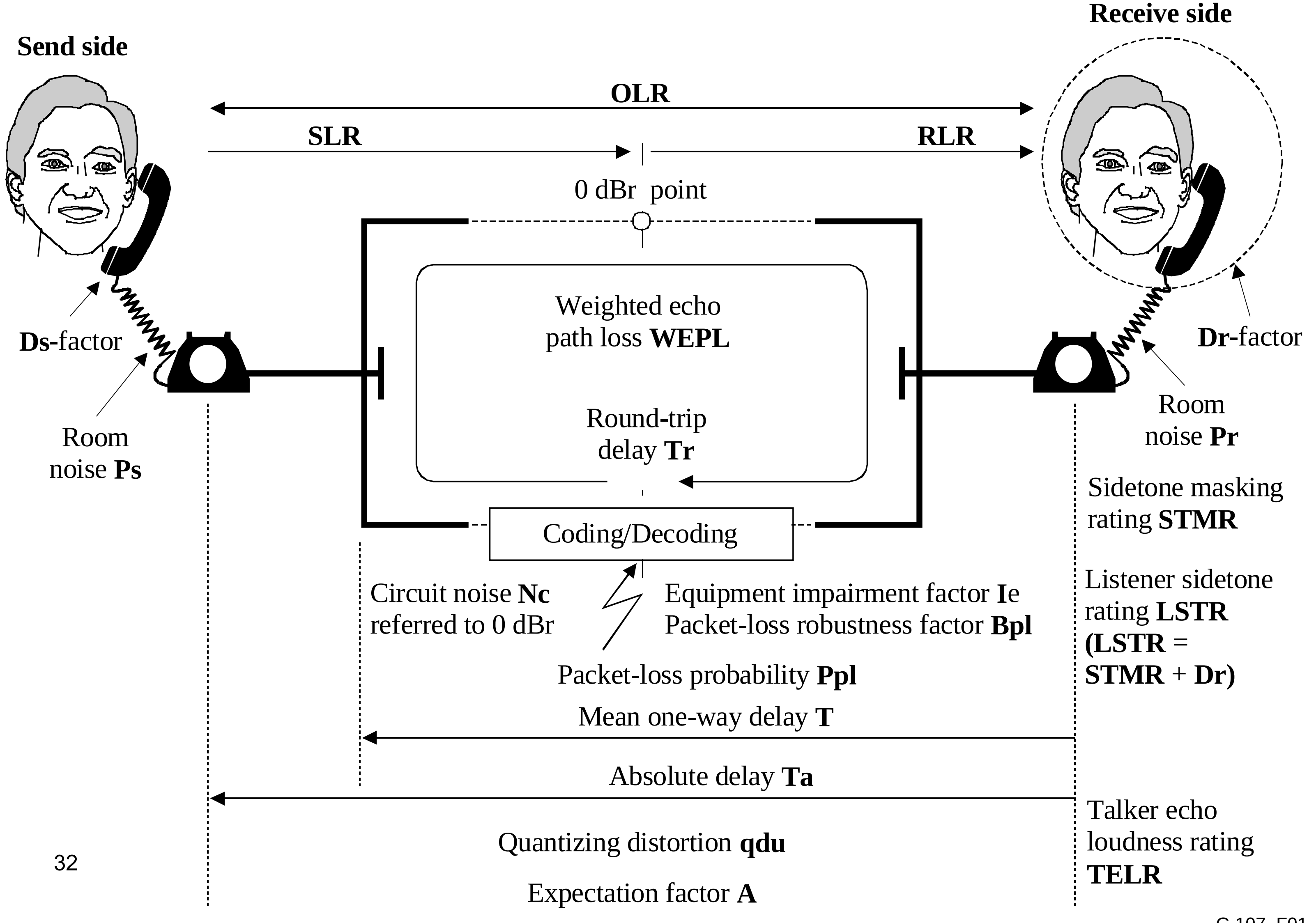
  - Combines PAMS time alignment routine with PSQM+ perceptual model

- Designed to take variable delay into account, showing much higher correlation than PSQM+

E-Model (G.107)

- Non Intrusive method for estimating the user satisfaction
- It is not intended for predicting absolute user satisfaction
- The E-Model has proven to be versatile tool that has adapted well to the impairments of IP telephony
- Main VoIP impairments
  - Delay, including delay variation and jitter
  - Echo
  - Speech compression
  - Packet loss

E-Model: non Intrusive method



Transmission rating factor R

$$R = Ro - Is - Id - Ie + A$$

Ro - signal-to-noise ratio based on send and receive loudness ratings and the circuit and room noise

Is - the sum of real-time or simultaneous speech transmission impairments

- loudness levels, side- tone and PCM quantization distortion

Id - the sum of delayed impairments relative to the speech signal

- talker echo, listener echo and absolute delay

Transmission rating factor R (cont.)

Ie - the Equipment Impairment factor for special equipment

- low bit-rate coding (determined subjectively for each codec and for each % packet loss

A - the Advantage factor adds to the total and improves the R-value for new services

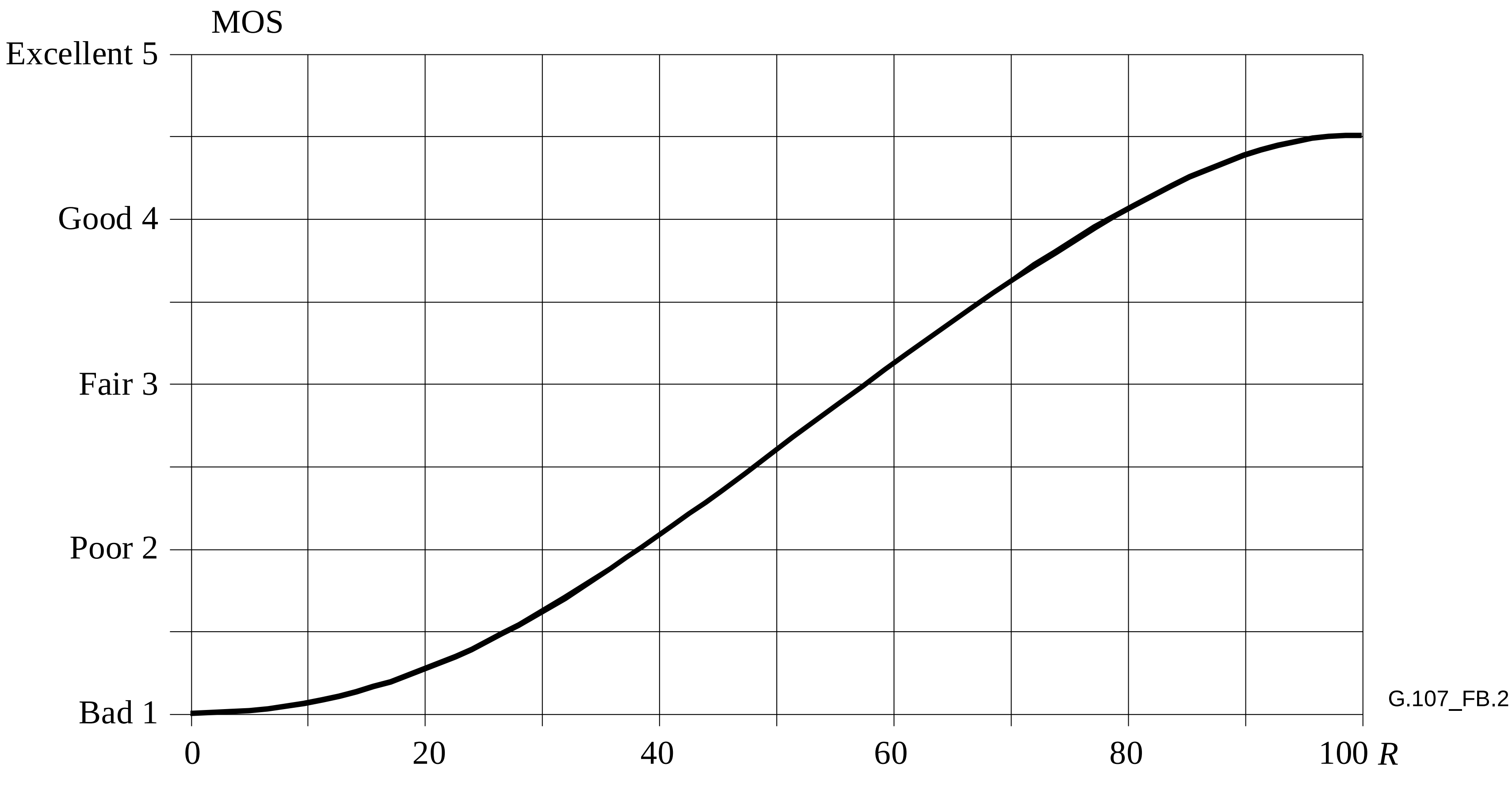
- like satellite phones, to take into account the advantage of using a new service and to reflect acceptance of lower quality by users for such services

Transmission rating factor R (cont.)

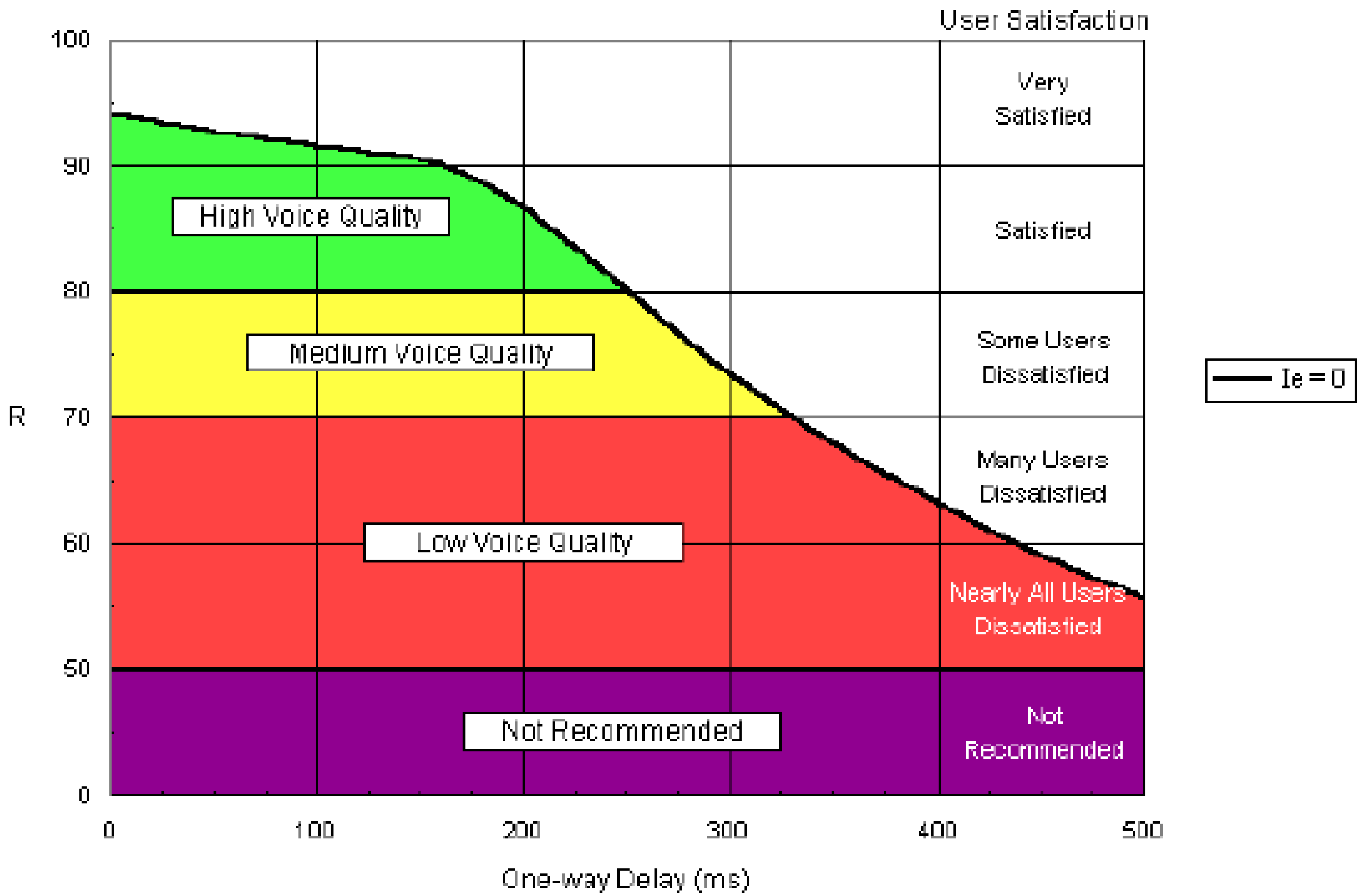
- The Equipment Impairment factor and the Advantage factor are unique to the E-Model

- The Equipment Impairment factor makes the E-Model a powerful tool for estimating the relative user satisfaction of IP Telephony conversations

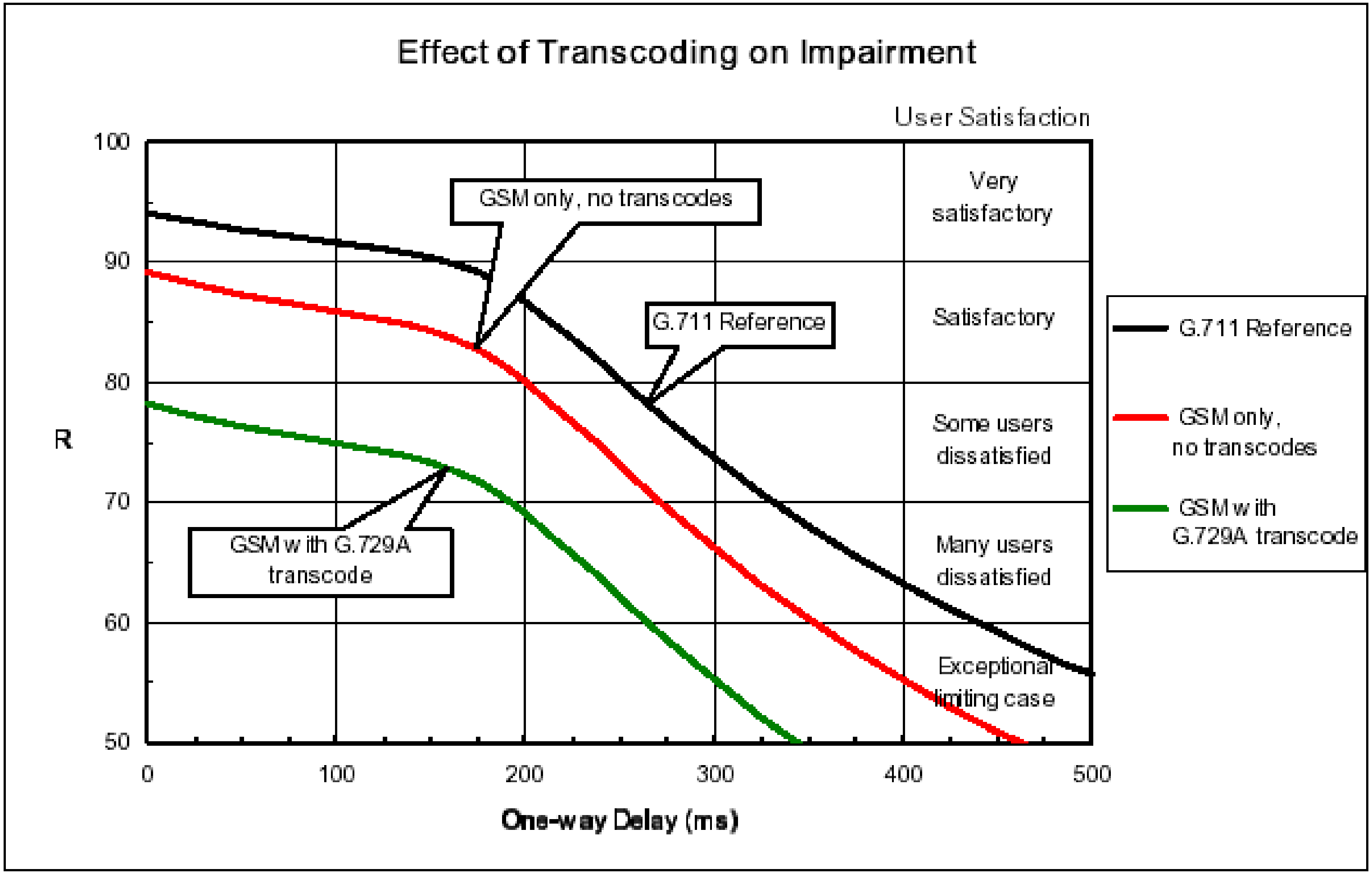
MOS as function of rating factor R



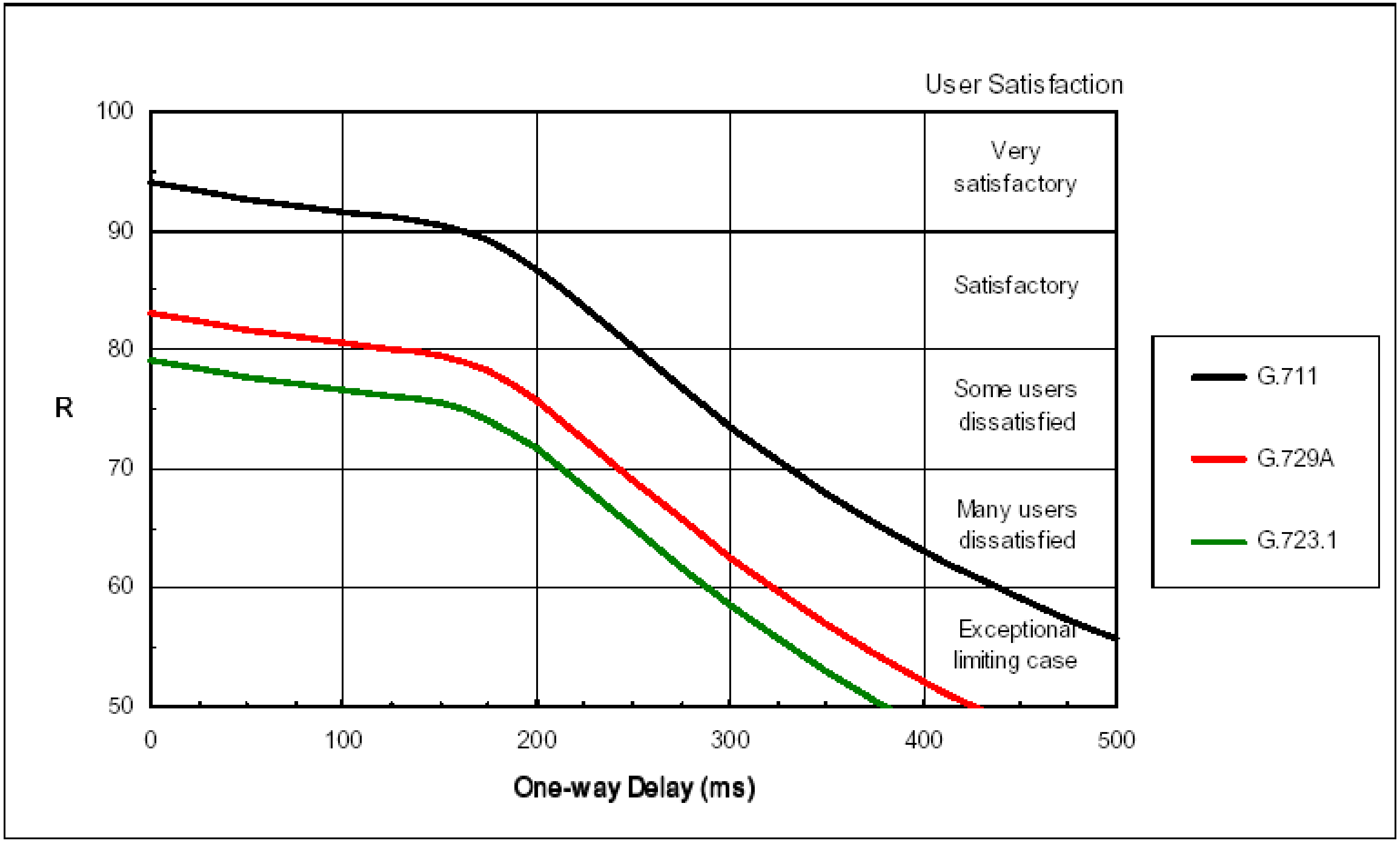
Recommended IP Telephony Voice Quality Categories



One way delay versus Quality



Speech Compression Impairment



Compression method versus Quality

Effects Of Different Codecs					
Codec	**Default Data Rate**	**Default Datagram Size**	**Packetization Delay**	**Default Jitter Buffer Delay**	**Theoretical Maximum MOS ***
G.711u	64 kbps	20 ms	1.0 ms	2 datagrams (40 ms)	4.4
G.711a	64 kbps	20 ms	1.0 ms	2 datagrams (40 ms)	4.4
G.729	8 kbps	20 ms	25.0 ms	2 datagrams (40 ms)	4.07
G.723.1 MPMLQ	6.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87
G.723.1 ACELP	5.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.69

Correlation coefficient for 38 subjective tests

Number of Tests	Type	Correlation Coefficient	PESQ	PAMS	PSQM	PSQM+
19	Mobile Network	Average	0.962	0.954	0.924	0.935
		Worst-case	0.905	0.895	0.843	0.859
9	Fixed network	Average	0.942	0.936	0.881	0.897
		Worst-case	0.902	0.805	0.657	0.652
10	VoIP Multi-type	Average	0.918	0.916	0.674	0.726
		Worst-case	0.810	0.758	0.260	0.469

סיכום

- Vanilla Applications
  - Telnet – basic terminal
  - FTP – file transfer and download
  - Mail Applications
- Voice over IP
  - far more than the delivery of the information packets
    - to deliver signaling information between the different networks
    - to set up, manage and tear down calls
    - to provide the necessary databases to manage customers and services
    - to deliver the information in the appropriate media format
- Media quality measurement methods
- Non objective
- Objective
  - Intrusion methods
  - Non-Intrusion methods