



# **ITU Asia-Pacific CoE Training**

## **Broadband Quality of Service - End User Experience**

### **Session 3: Broadband Quality of Service and Quality of Experience Framework and ITU initiatives**

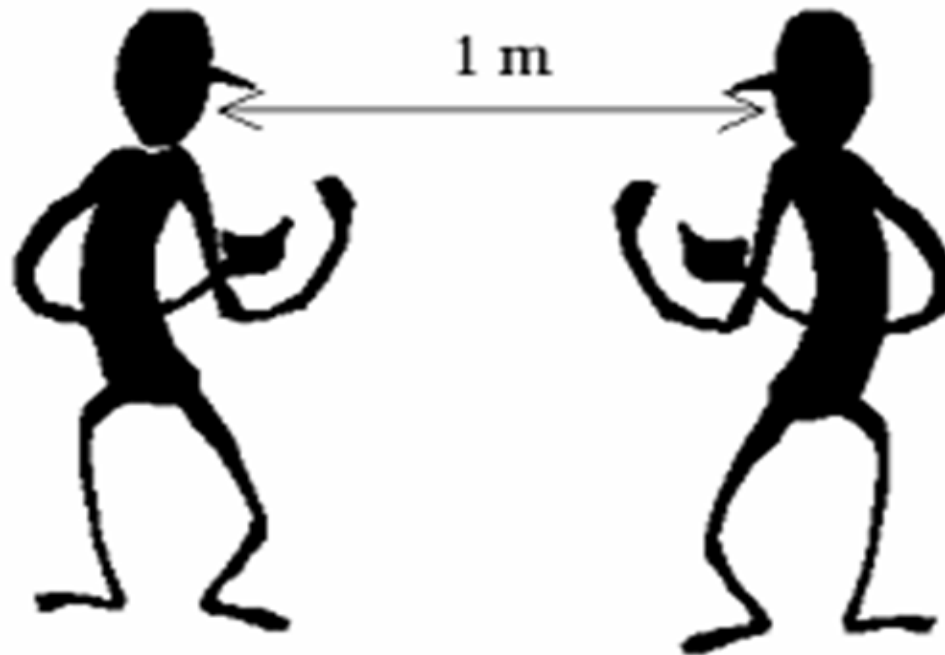
**27-30 October, 2015**

**TOT Academy, Thailand**

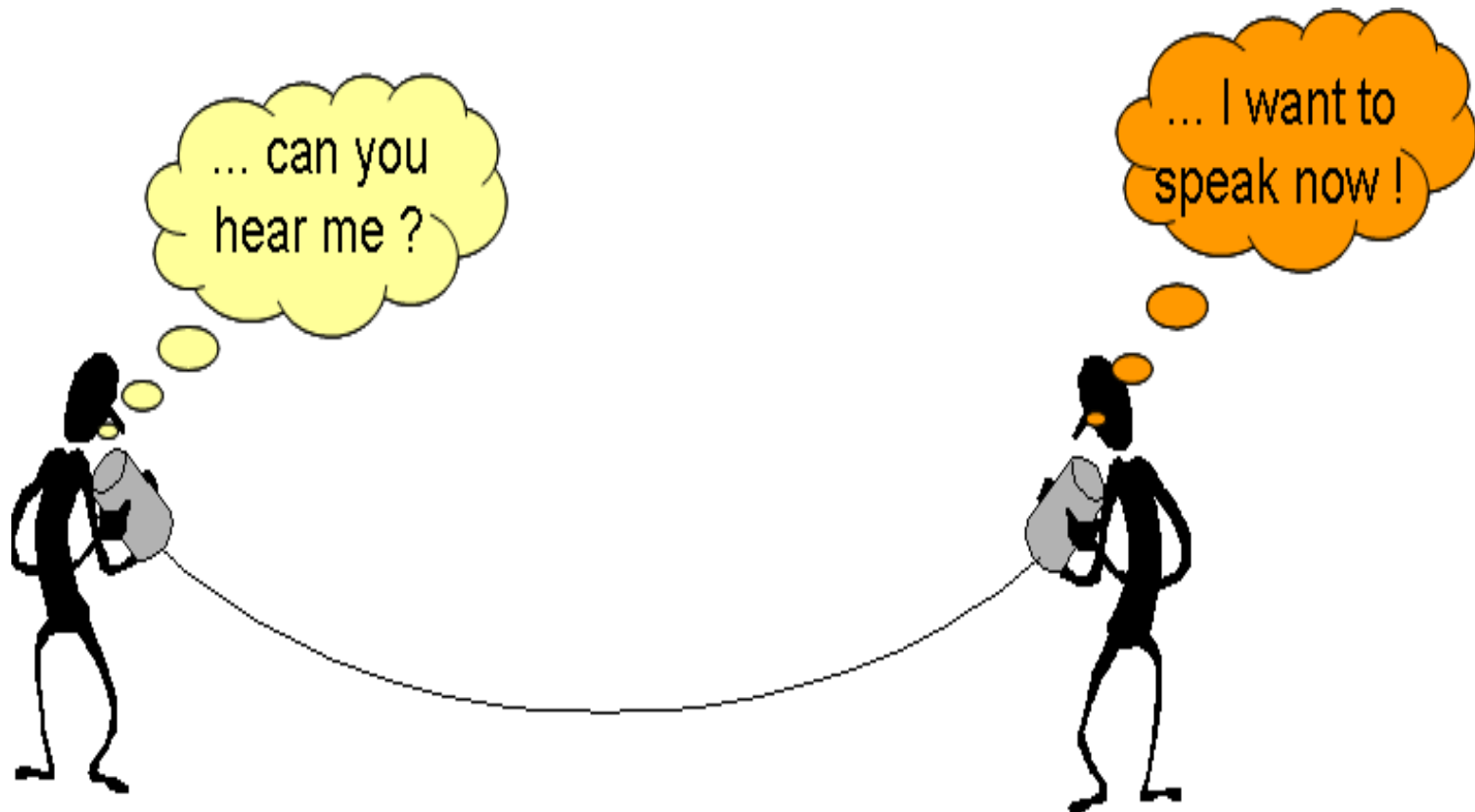
ITU Regional Office for Asia and the Pacific

- Objective:
  - To understand the concepts of network performance, quality of service & quality of experience and the relevant standards
  - The session also introduces approaches to develop a QoS/QoE framework from an end user perspective
  - The session also details the ITU standards and activities in this regard

## Where it All Begins: Real Communication Situation



## ... and where Technology comes to Play: Employing a Telecommunication System



# Talking Quality

- Describes the perceived quality by the talker while he is talking and the other user is listening only
- Includes Parameters like:
  - Talker Echo, Delay
  - Sidetone
  - Receive Noise
  - Required Voice Level of the Talker
    - In order to be heard by the Listener at the other End
- Does not include parameters like
  - Listener Echo
  - Absolute Delay

# Listening Quality

- Describes the perceived quality by the listener if one user is talking and the other user is listening, only
- Includes Parameters like:
  - Codec Distortions
  - Listener Echo
  - Receive Noise
  - Received Voice Level
- Does not include parameters like
  - Sidetone
  - Talker Echo
  - Delay

# Conversational Quality

- In a Conversation a Talker is also a Listener at the same time
- Describes the perceived quality by the user if all conversational situations of the user are considered
  - Talking situation
  - Listening situation
  - Conversational situation
  - Doubletalk situation
  - Acoustical environment of the user
    - Background noise
    - Reflection characteristics
- Includes all Talker & Listener Parameters that potentially impact Voice Quality

# MOS = Mean Opinion Score

- The mean of opinion scores, i.e., of the values on a predefined scale that subjects assign to their opinion of the performance of the telephone transmission system used either for conversation or for listening to spoken material
- True MOS values can only be derived from subjective tests
- Usefulness of MOS values outside the original subjective test depends on statistical exercises:
  - Selection of subjects
  - Compilation of speech samples
  - Normalization of results
  - Language Dependency



# Subjective Tests

- Require large group of people
- Very costly and time-consuming
- Cannot be done in real-time
- But it is the Reference for the other methods:
  - Objective models
  - Estimation models
- Subjective tests are not further addressed in detail in this presentation

# Objective Models

- Reproducing human perception as accurate as possible
- Real-time Recording or Monitoring of Waveform Signals
- Use of an algorithm to predict the results of a subjective test
- Faster and cheaper but correlation with subjective test may vary
- Current Models include P.862 (PESQ), P.563 and the new P.863 “POLQA”
- Obsolete Models include P.861 (PSQM, for Codec Validation only) and a variety of vendors' proprietary Models

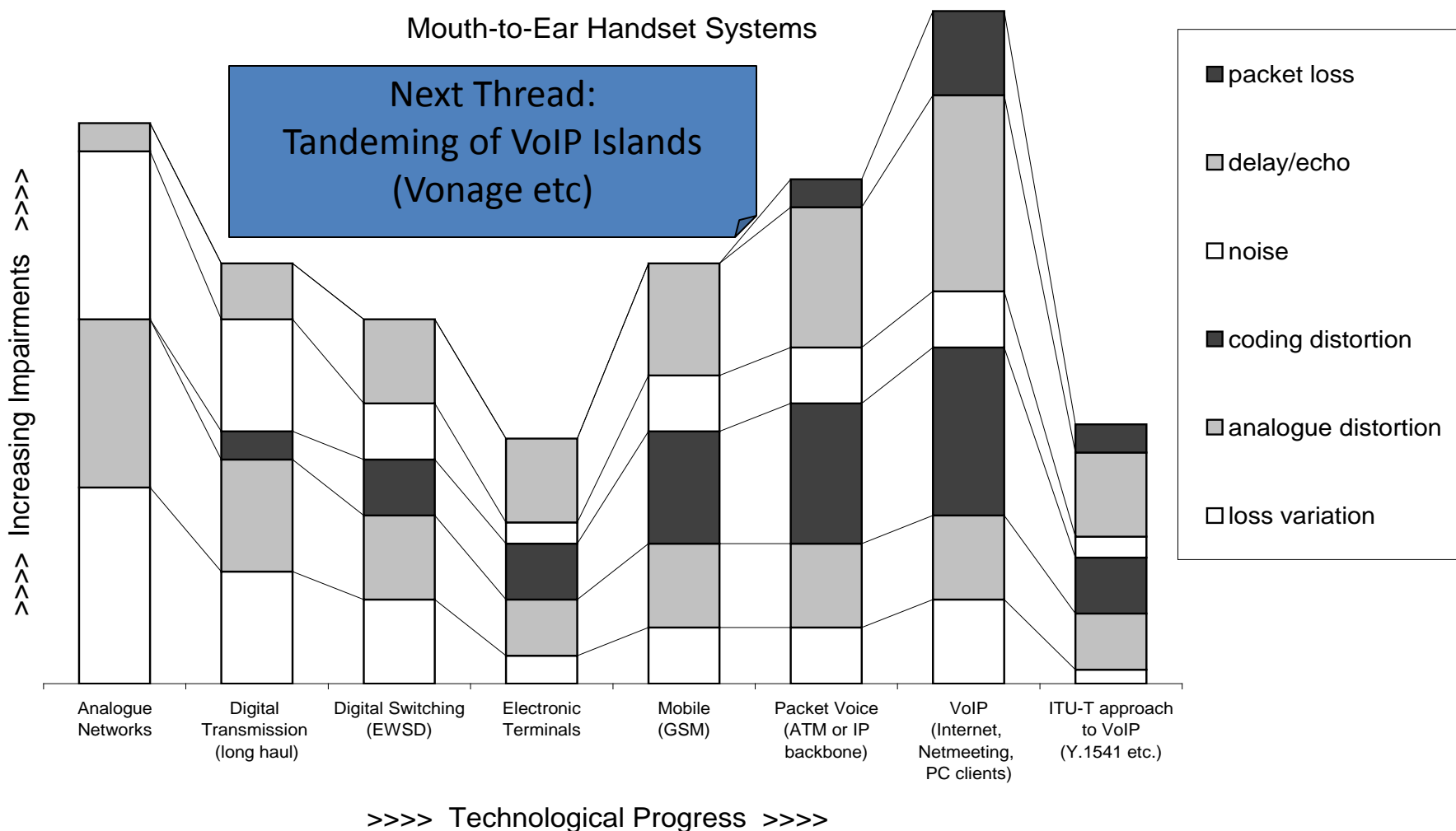
# Estimation Models

- Estimation or Parametric Models are based on combination of parameters that again can be measured, estimated or pre-determined by separate means
- Use the actually measured parameter values such as echo, delay, noise, levels etc.
- Probably has an even lower correlation with subjective testing than objective testing
- Requires lower computational effort than objective testing
- Basically two different Approaches
  - E-Model for offline transmission planning
    - Several tools available, E-Model
  - Real-time collection of transmission parameters with RTCP-XR
    - Several vendors' products use this to estimate Voice Quality

# MOS Test & Models Terminology

- Subjective Tests
  - MOS-LQS = for Listening Quality from a Subjective Test
  - MOS-CQS = for Conversation Quality from a Subjective Test
- Objective Models
  - MOS-LQO = for Listening Quality from an Objective Model
  - MOS-CQO = for Conversation Quality from an Objective Model
- Estimation Models
  - MOS-LQE = for Listening Quality from an Estimation Model
  - MOS-CQE = for Conversation Quality from an Estimation Model
- Important to understand
  - MOS-xxx is NOT comparable with MOS-yyy

# Evolution of Voice Quality Impairments



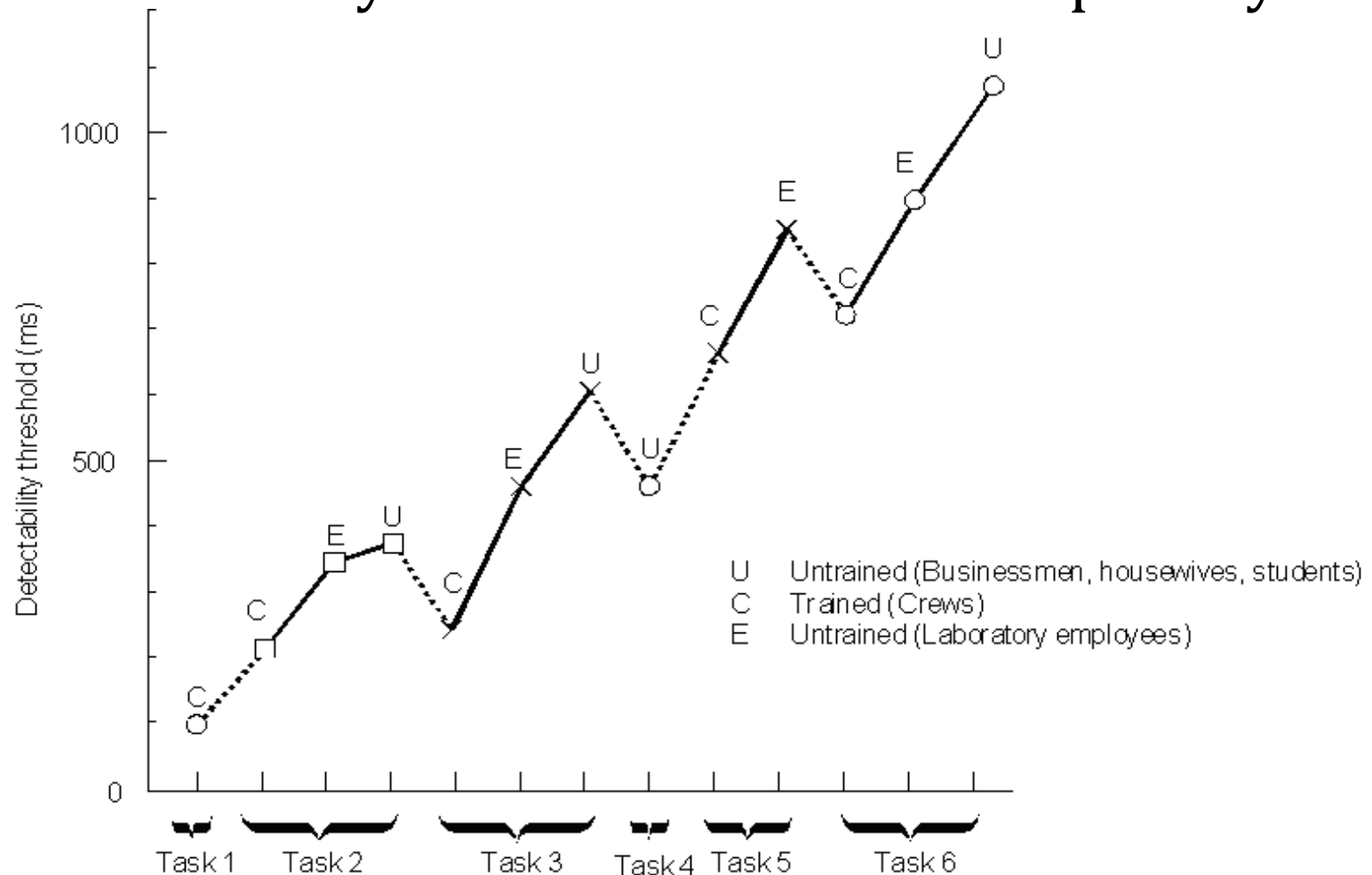
# Types of Impairments

- Traditional Networks & Terminals
  - Short Delay
  - Circuit & Ambient Noise
  - Too High or Too Low Voice Levels
  - Distortion & Impairments Due to Transcoding
- Perception of Delay strongly depends on Conversational Task
  - Additional Impairments Due to IP Transport
    - Longer Delay (Latency)
  - Jitter
  - Packet Loss

# Delay Affects the Flow of Conversation

- Perception of Delay strongly depends on Conversational Task
  - Task 1 – Read out random numbers as quickly as possible in turn,
  - Task 2 – Verify random numbers as quickly as possible in turn,
  - Task 3 – Complete words with lost letters as quickly as possible by exchanging information,
  - Task 4 – Verify city names as quickly as possible in turn,
  - Task 5 – Determine the shape of a figure by receiving oral information,
  - Task 6 – Free conversation.
- Results of a Study are shown on the next slide

# Detectability Threshold of Round-Trip Delay





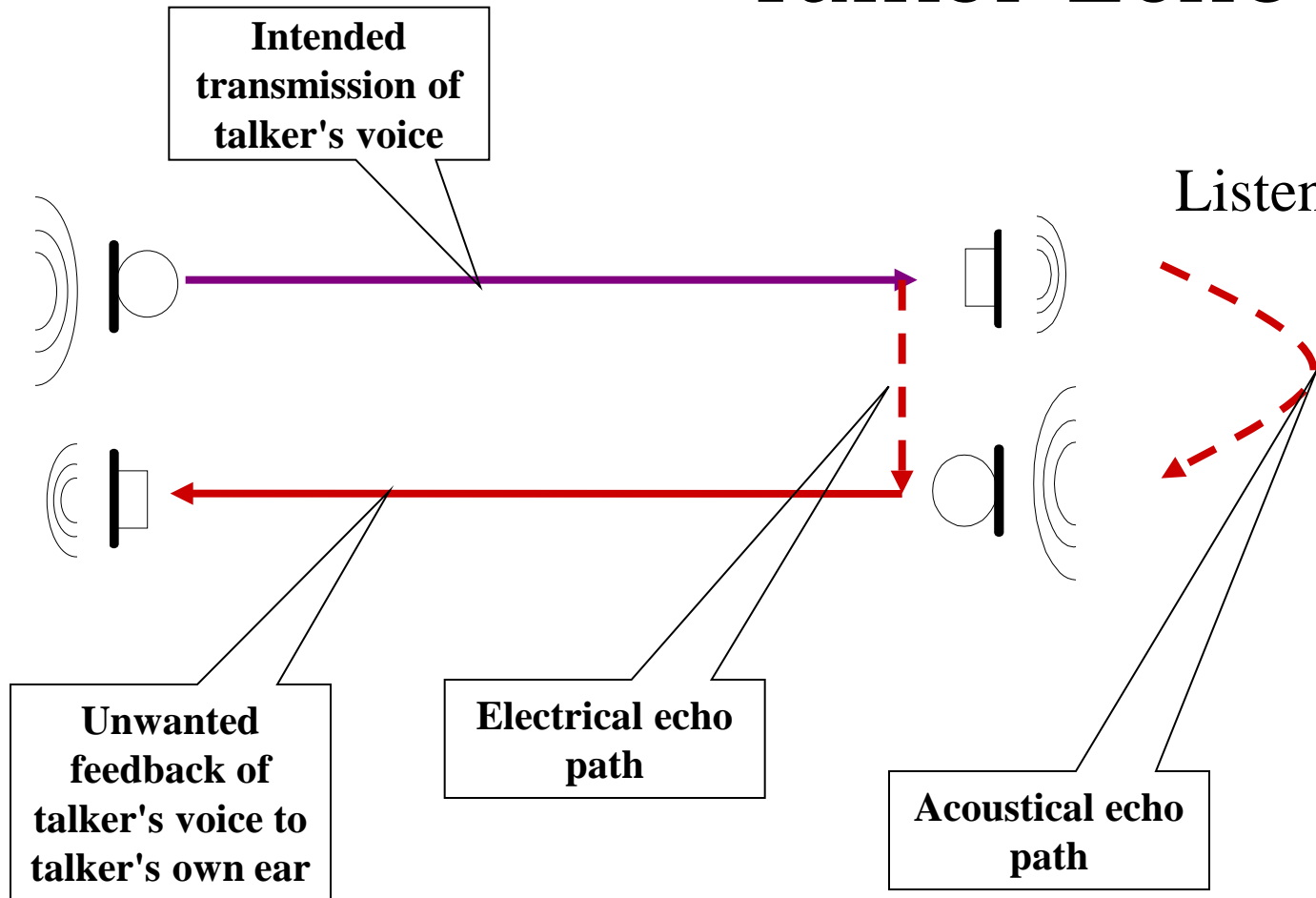
# Effects of Echo

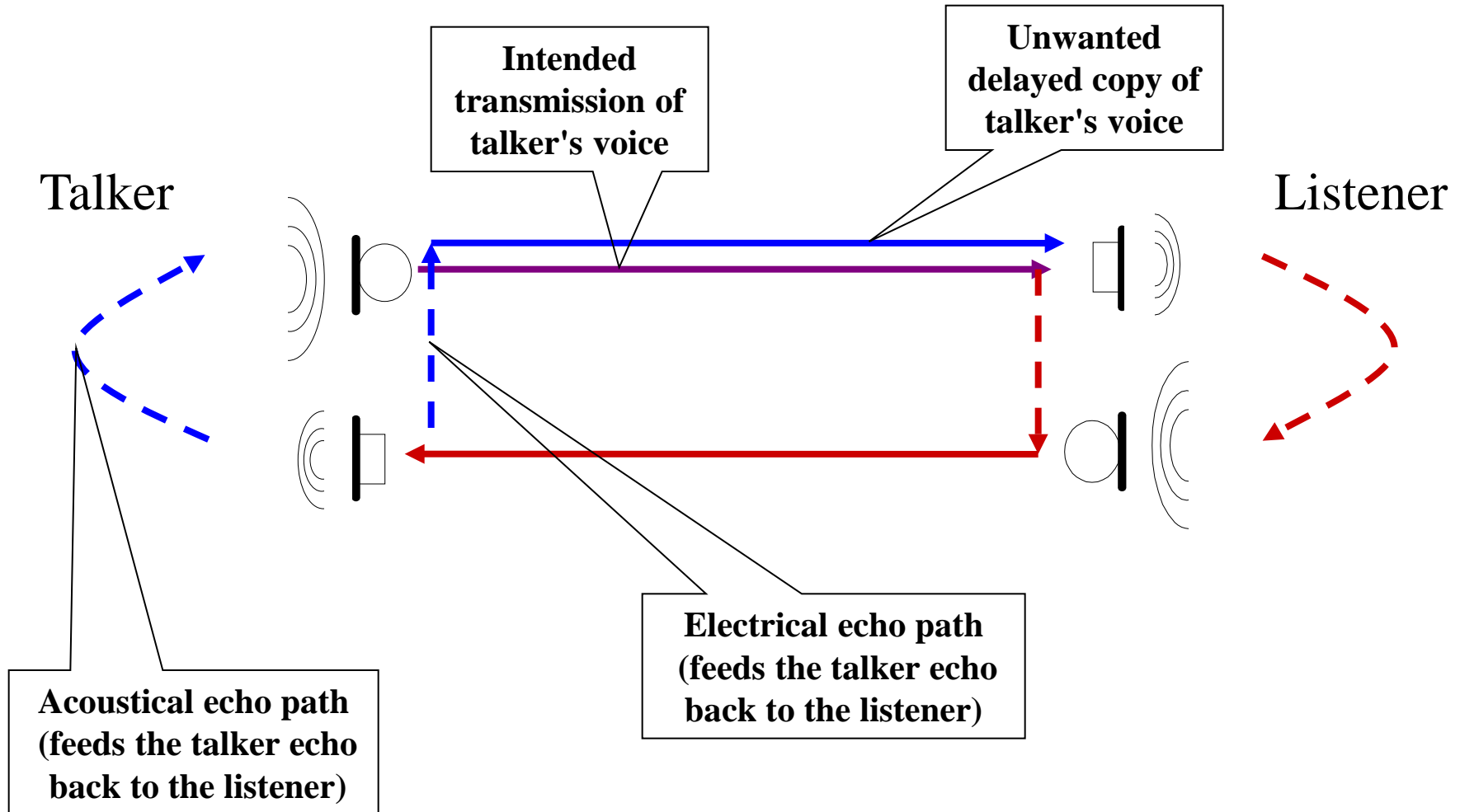
- In Telecom World Echo describes delayed & unwanted feedback of send signal into receive path
- Echo Source = Reflection Point:
  - Hybrid Circuits (multiple reflections possible)
  - Coupling in Handset Cords
  - Structure Borne Coupling in Handsets
  - Acoustical Coupling between Earpiece and Microphone
- Two Types of Echo
  - Talker echo
  - Listener echo

# Talker Echo

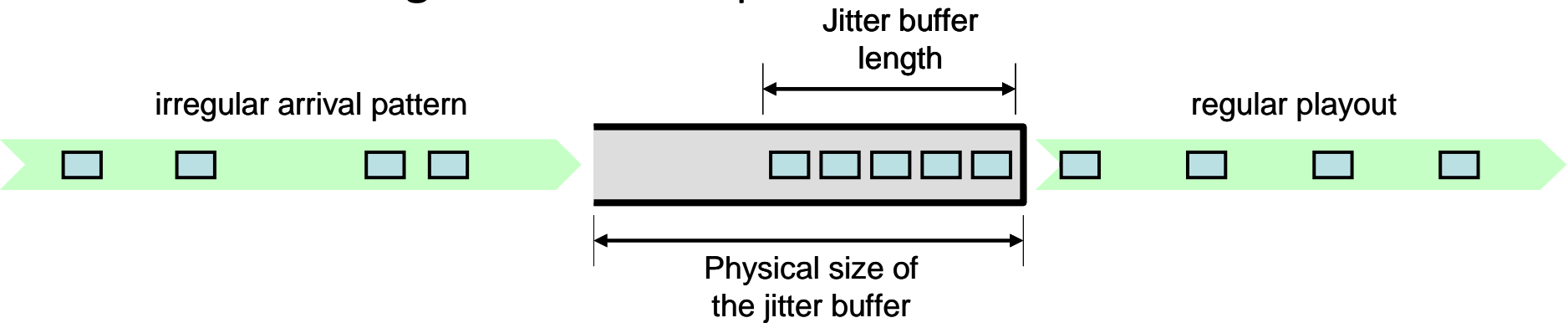
Talker

Listener

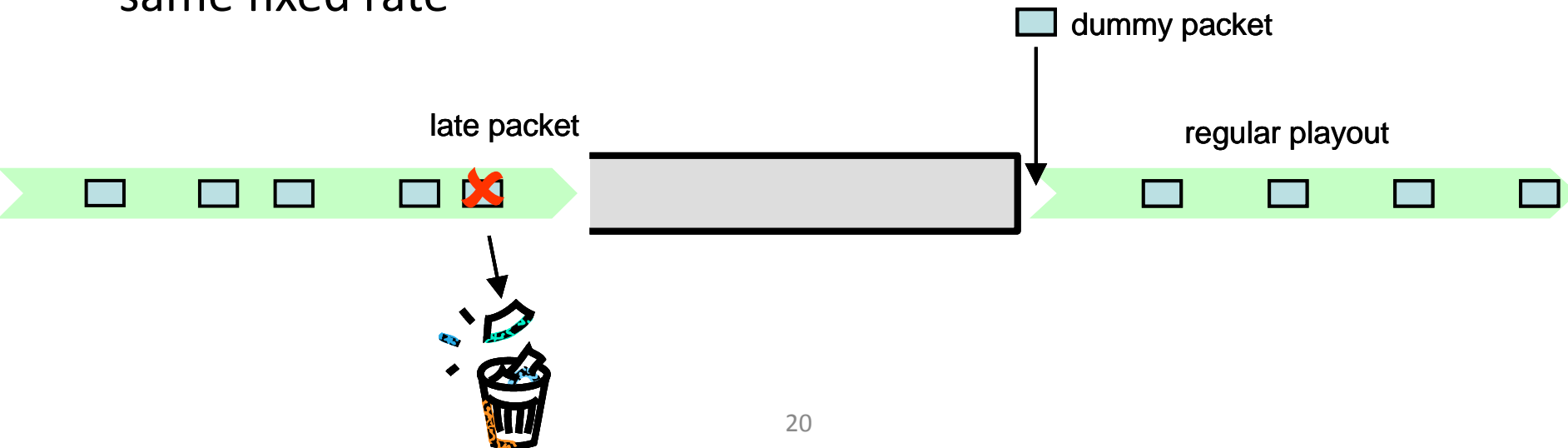




- Voice coders generate voice packets at a fixed rate



- Receiving Decoder expects to be fed with voice packets at the same fixed rate

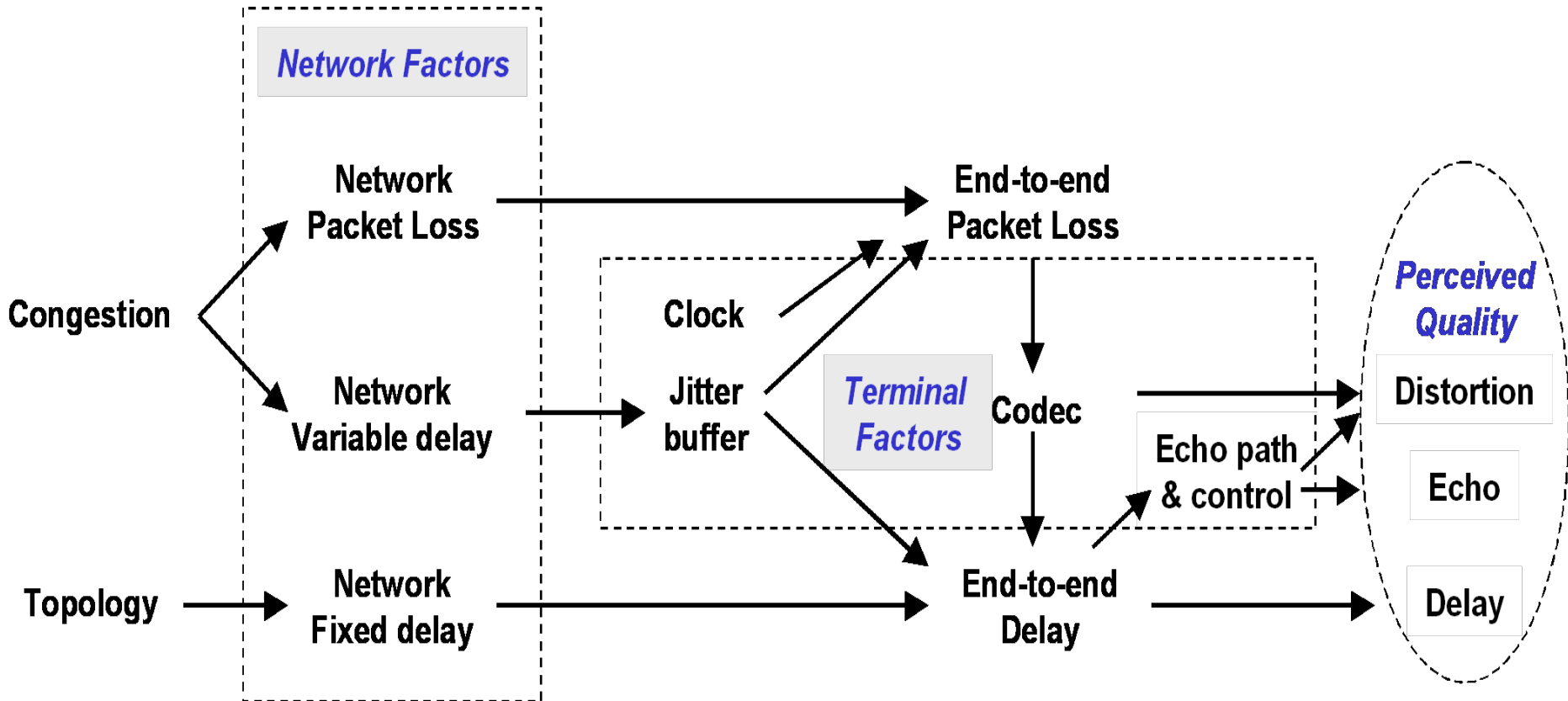


## Packet Loss is Loss of Information

- Packet Loss caused by
  - Network conditions are poor and packets become damaged in transit due to transmission errors
  - The packet was deliberately dropped at a switch or router due to congestion.
  - The packet was dropped due to late arrival
- Packet Loss Concealment
  - Minimizing Perceptual Impact due to Lost Packets
    - Different Methods
    - Different Complexity
    - Different Quality

# Impairments in packet networks

- Distinction between Effects
  - that occur in the Network and
  - Mechanisms in the Terminals that are affected
- Terminals can be used to correct for the Effects in the Network
- Remaining Issues are:
  - End-to-End Delay is increased when compensating for other Effects
  - Loss of Information can be Concealed but Not Recovered



# QoS of IP Voice Terminals

- ETSI Standards for VoIP Terminals
  - On E2E Voice Quality as perceived by the User
  - ITU-T does not have directly comparable standards
  - 4 Standards
    - ES 202 737 – Narrowband, Handset & Headset
    - ES 202 738 – Narrowband, Loudspeaking & Handsfree
    - ES 202 739 – Wideband, Handset & Headset
    - ES 202 740 – Wideband, Loudspeaking & Handsfree



# Example #1: Send Delay

- 7.3.1 Send Delay
- For a VoIP terminal, send delay is defined as the one-way delay from the acoustical input (mouthpiece) of this VoIP terminal to its interface to the packet based network. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figures 2 and A.1 in ITU-T Recommendation G.1020 [16], respectively.
- The sending delay  $T(s)$  is defined as follows:
  - $T(s) = T(ps) + T(la) + T(aif) + T(asp)$  (formula 1)
- Where:
  - $T(ps)$  = packet size =  $N * T(fs)$
  - $N$  = number of frames per packet
  - $T(fs)$  = frame size of encoder
  - $T(la)$  = look-ahead of encoder
  - $T(aif)$  = air interface framing
  - $T(asp)$  = allowance for signal processing
- The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.
- NOTE 1: The size of  $T(aif)$  is for further study.
- Requirement
- The allowance for signal processing shall be  $T(asp) < 10$  ms.

## Example #2: Receive Delay

- 7.3.2 Receive delay
- For a VoIP terminal, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its acoustical output (earpiece). The total receive delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figures 3 and A.2 of ITU-T Recommendation G.1020 [16], respectively.
- The receiving delay  $T(r)$  is defined as follows:
  - $$T(r) = T(fs) + T(aif) + T(jb) + T(plc) + T(asp) \quad (\text{formula 2})$$
  - Where:
  - $T(fs)$  = frame size of encoder
  - $T(aif)$  = air interface framing
  - $T(jb)$  = jitter buffer size
  - $T(plc)$  = PLC buffer size
  - $T(asp)$  = allowance for signal processing
- The additional delay required for IP packet dis-assembly and presentation from the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.
- NOTE 1: The size of  $T(aif)$  is for further study.
- Requirements
- The allowance for signal processing shall be  $T(asp) < 10$  ms.
- The additional delay introduced by the jitter buffer shall be  $T(jb) \leq 10$  ms.
- For Coders without integrated PLC the additional PLC buffer size shall be  $T(plc) < 10$  ms.
- For Coders with integrated PLC the additional PLC buffer size shall be  $T(plc) = 0$  ms.

# Multimedia Quality

- For Media other than Voice New Parameters
- Distinction QoS / QoE of more Importance
- Extension of OSI 7 Layer model
  - According to Bauer and Patrick\*
  - Layer 1 to 7 = QoS
  - Layer 8 – 10 = QoS Extension
  - Advantage is a homogeneous view on QoS and QoE

# The complete 10-layer OSI+HCI model

HCI	10.	Human Needs (communication, education, acquisition, security, entertainment...)
	9.	Human Performance (perception, cognition, memory, motor control, social...)
	8.	Display (keyboard, GUI/CLI, vocal, Bpp, Ppi, Ppm...)
OSI	7.	Application (HTTP, FTP, NFS, POP...)
	6.	Presentation (PS, LZ, ISO-PP...)
	5.	Session (DNS, RPC, PAP...)
	4.	Transport (TCP, UDP, RTP...)
	3.	Network (IP, DHCP, ICMP, AEP...)
	2.	Data Link (ARP, PPP...)
	1.	Physical (10bt, xDSL, V.42 [I.48]...)

## New P.1200-series of Recommendations

- Recommendation ITU-T P.1201 (2012) Parametric non-intrusive assessment of audiovisual media streaming quality
- Recommendation ITU-T P.1201.1 (2012), Parametric non-intrusive assessment of audiovisual media streaming quality – lower resolution application area
- Recommendation ITU-T P.1201.2 (2012), Parametric non-intrusive assessment of audiovisual media streaming quality – higher resolution application area
- Recommendation ITU-T P.1202 (2012) Parametric non-intrusive bitstream assessment of video media streaming quality
- Recommendation ITU-T P.1202.1 (2012), Parametric non-intrusive bitstream assessment of video media streaming quality – lower resolution application area

# Video Conferencing (1/2)

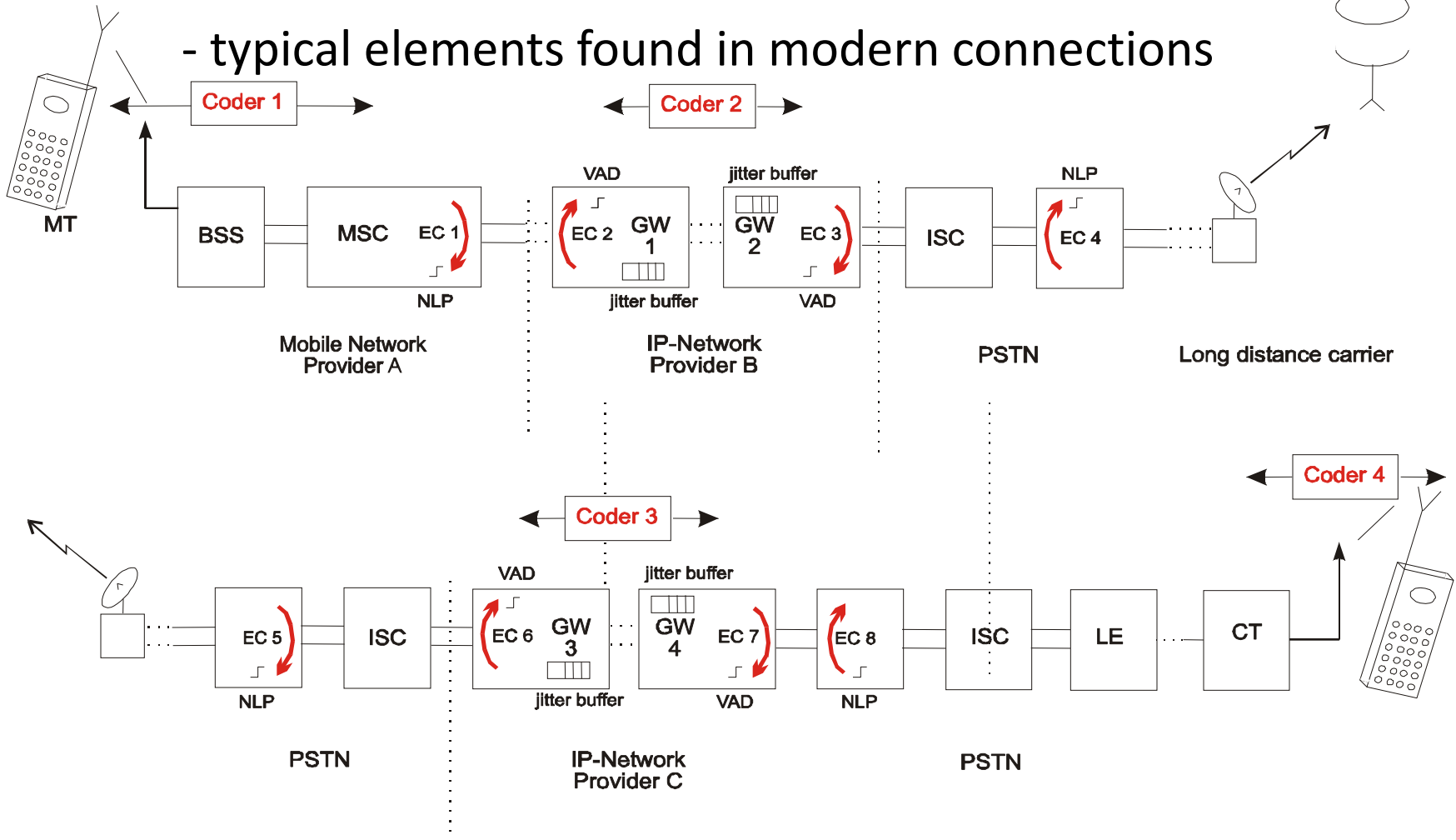
- Draft Rec. H.TPS-AV “Audio/Video Parameters for Telepresence systems”
  - defines the audio/video/environment parameters for Telepresence systems
  - gives the specific format definition for these parameters
  - contains several clauses related to audio parameters such as
    - audio level
    - reverberation time
    - background noise
    - sound insulation.

# Video Conferencing (2/2)

- New Recommendation P.1301
  - Subjective quality evaluation of audio and audiovisual multiparty telemeetings
- New Supplement 26 to the P-series
  - Scenarios for the subjective quality evaluation of audio and audiovisual multiparty telemeetings

# Terminals & Network in Reality

- typical elements found in modern connections

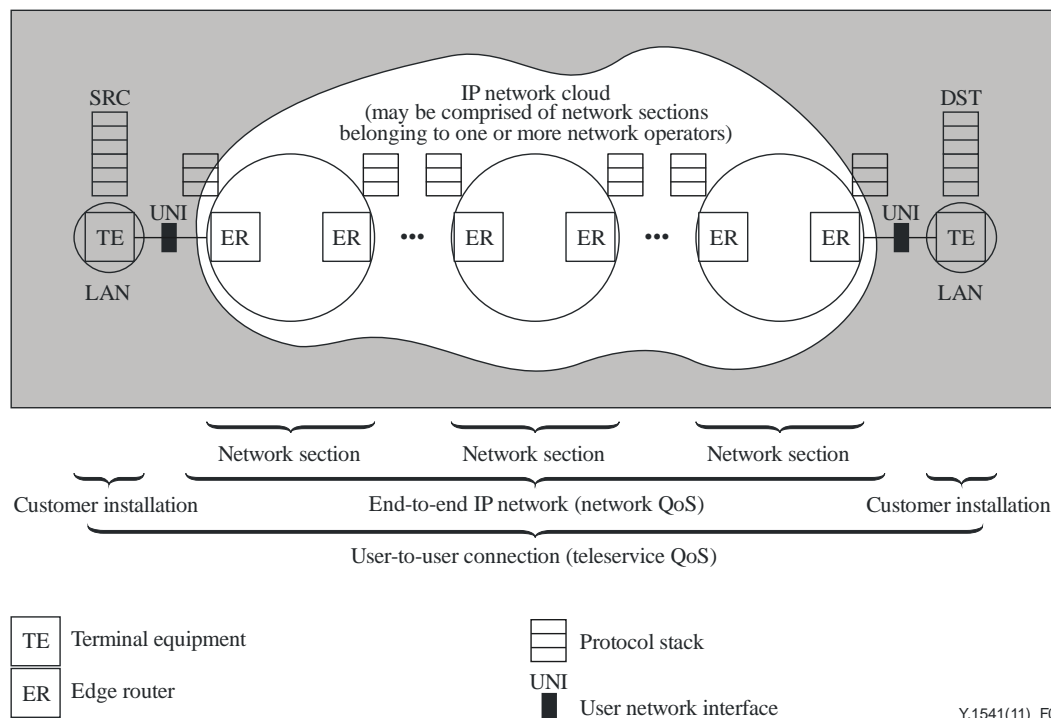




# Recommendation ITU-T Y.1541

- “Network performance objectives for IP-based services”
  - Specifies network IP performance values between User Network Interfaces (UNI)
  - For each of the performance parameters defined in ITU-T Y.1540
    - Performance values vary, depending on the network QoS class
    - Defines eight network QoS classes
  - QoS classes intended to be the basis of
    - Agreements between end-users and network service providers
    - Between service providers
  - Rec. Y.1541 does only apply to fixed networks

# UNI-to-UNI reference path for network QoS objectives



NOTE – Customer Installation equipment (shaded area) is for illustrative purposes only.

# QoS classes

Network performance parameter	Nature of network performance objective	QoS Classes						provisional	
		Class 0	Class 1	Class 2	Class 3	Class 4	Class 5 Unspecified	Class 6	Class 7
IPTD	Upper bound on the mean IPTD	100 ms	400 ms	100 ms	400 ms	1 s	U	100 ms	400 ms
IPDV	Upper bound on the $1 - 10^{-3}$ quantile of IPTD minus the minimum IPTD	50 ms	50 ms	U	U	U	U	50 ms	
IPLR	Upper bound on the packet loss probability	$1 \times 10^{-3}$	$1 \times 10^{-3}$	$1 \times 10^{-3}$	$1 \times 10^{-3}$	$1 \times 10^{-3}$	U	$1 \times 10^{-5}$	
IPER	Upper bound	$1 \times 10^{-4}$ (Note 5)						U	$1 \times 10^{-6}$
IPRR	Upper bound								$1 \times 10^{-6}$

# Applications

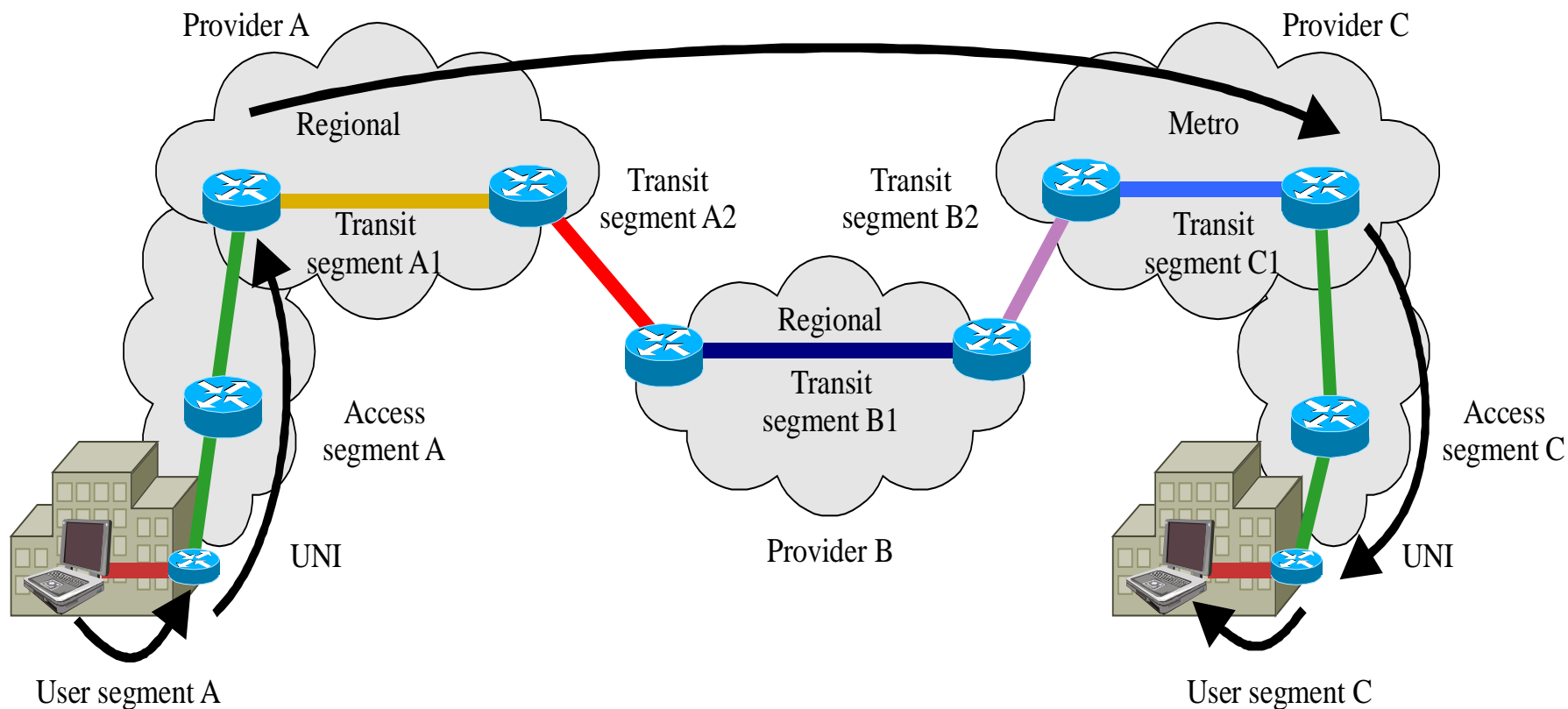
QoS class	Applications (examples)
0	Real-time, jitter sensitive, high interaction (VoIP, VTC)
1	Real-time, jitter sensitive, interactive (VoIP, VTC).
2	Transaction data, highly interactive (Signalling)
3	Transaction data, interactive
4	Low loss only (short transactions, bulk data, video streaming)
5	Traditional applications of default IP networks

# Y.1541 is evolving

- Classes 0 to 5 are no longer sufficient
  - For new applications, e.g. IPTV
  - Therefore, classe 6 and 7 have provisionally been defined
  - Introducing a new parameter
  - IPRR = IP Packet Reordering Ratio

# Example topology for impairment allocation see Rec. ITU-T Y.1542

Total transit segment



## Example of accumulating and estimating UNI-UNI performance (1)

	Requested	Network 1	Network 2	Network 3	Estimated UNI-UNI
QoS Class	Class 0				Class 0
Mean transfer delay (IPTD)	100 ms	22.4 ms	10.6 ms	32.4 ms	65.4 ms
99.9% – min delay var. (IPDV)	50 ms	25 ms	2 ms	25 ms	47.5 ms
Minimum transfer delay	–	10 ms	10 ms	20 ms	–
Variance of transfer delay	–	52.4 ms	0.23 ms	55.1 ms	–
Loss (IPLR)	$10^{-3}$	$10^{-4}$	$10^{-4}$	$10^{-4}$	$3 \times 10^{-4}$
Errored packets (IPER)	$10^{-4}$	$3 \times 10^{-5}$	$3 \times 10^{-5}$	$3 \times 10^{-5}$	$9 \times 10^{-5}$

- In this example, the user has requested QoS class 0, and the network sections above are determined to compose the UNI-UNI path. The next step is to request the performance (impairment) levels from each segment of the path.

## Example of accumulating and estimating UNI-UNI performance (2)

	Requested	Network 1	Network 2	Network 3	Estimated UNI-UNI
QoS Class	Class 0				Class 1
Mean transfer delay (IPTD)	100 ms	42.4 ms	20.6 ms	42.4 ms	105.4 ms
99.9% – min delay var. (IPDV)	50 ms	25 ms	2 ms	25 ms	47.5 ms
Minimum transfer delay	–	30 ms	20 ms	30 ms	–
Variance of transfer delay	–	52.4 ms	0.23 ms	55.1 ms	–
Loss (IPLR)	$10^{-3}$	$10^{-4}$	$10^{-4}$	$10^{-4}$	$3 \times 10^{-4}$
Errored packets (IPER)	$10^{-4}$	$3 \times 10^{-5}$	$3 \times 10^{-5}$	$3 \times 10^{-5}$	$9 \times 10^{-5}$

- In this example, the estimated delay exceeds the limit for Class 0.



# 3GPP TS 23107

- UMTS QoS classes are also referred to as traffic classes
- Restrictions and limitations of the air interface have to be taken into account
- QoS mechanisms provided in the cellular network have to be robust and capable of providing reasonable QoS resolution. There are four different QoS classes:
  - Conversational class
  - Streaming class
  - Interactive class
  - Background class
- Main distinguishing factor delay sensitivity of traffic

# Conversational class

- Typical applications
  - Telephony speech (e.g. GSM)
  - Voice over IP
  - And video conferencing
- Real time conversation is always performed between peers (or groups) of live (human) end-users
- Transfer delay requirement of the conversational class
  - Significantly lower and more stringent compared to interactive traffic case.
- Real time conversation - fundamental characteristics for QoS: Preserve
  - Time relation (variation) between information entities of the stream
  - Conversational pattern (stringent and low delay)

# Streaming class

- Typical applications
  - Real time audio and video streaming
- Real time data flow is always aiming at a live (human) destination
- Delay variation of the end-to-end flow shall be limited
- Stream normally is time aligned at the receiving end
  - Highest acceptable delay variation given by the capability of the time alignment function of the application
- Acceptable delay variation much greater than in conversational class

# Interactive class

- Typical applications
  - End-user, that is either a machine or a human, is online requesting data from remote equipment (e.g. a server)
    - Web browsing
    - Data base retrieval
    - Server access
    - Polling for measurement records
    - Automatic data base enquiries
- Round trip delay is one of the key attributes.
- Another characteristic is that the content of the packets shall be transparently transferred (with low bit error rate)
- Interactive traffic - fundamental characteristics for QoS:
  - Request response pattern
  - Preserve payload content

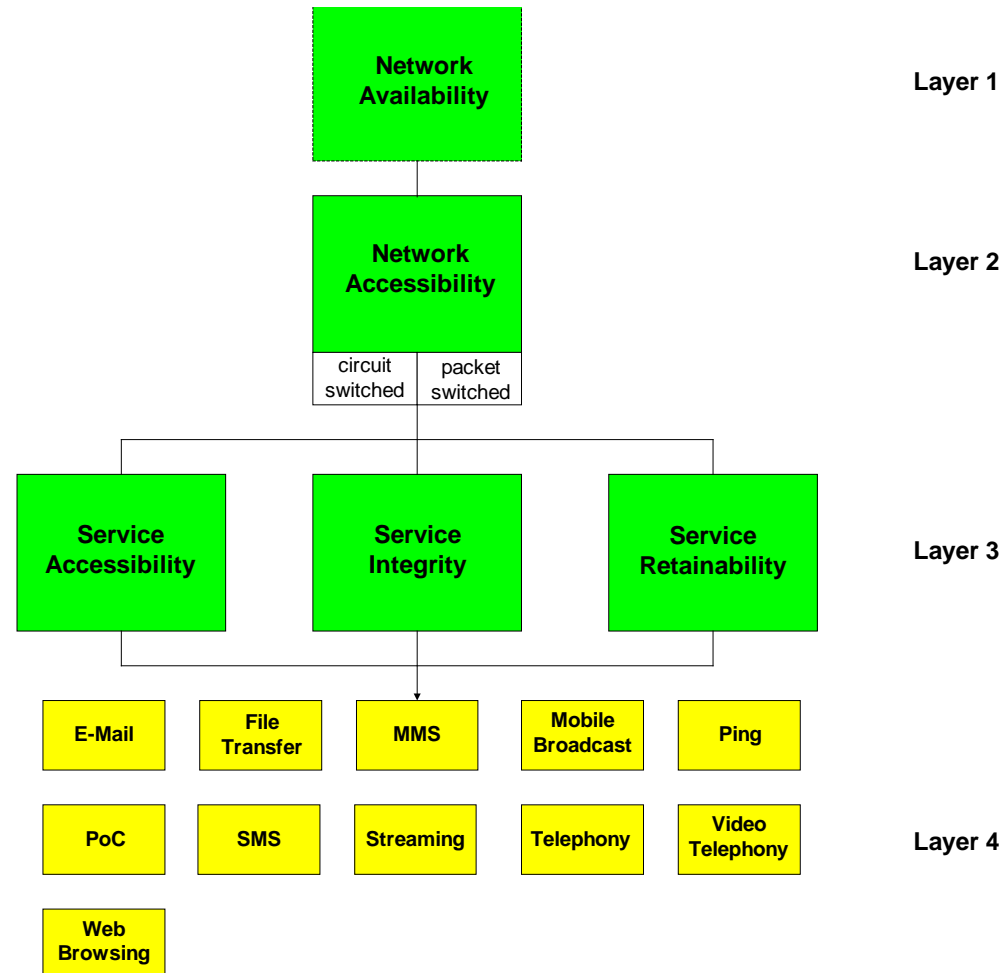
# Background class

- Typical applications
  - End-user, that typically is a computer, sends and receives data-files in the background
    - Delivery of E-mails or SMS
    - Download of databases
    - Reception of measurement records
- Background traffic is one of the classical data communication types
  - Destination is not expecting the data within a certain time
  - More or less delivery time insensitive
  - Content of the packets shall be transparently transferred (with low bit error rate).

# QoS Layers in Mobile

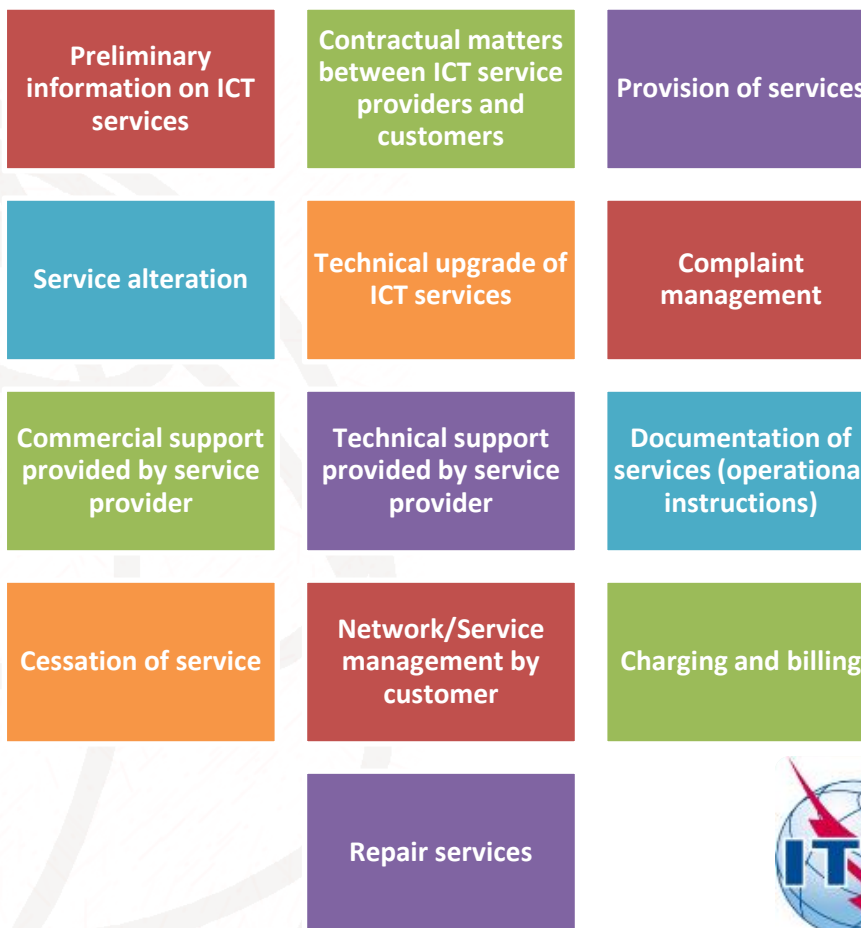
- QoS model for mobile has four layers.
- First layer is the Network Availability
  - defines QoS rather from the viewpoint of the service provider than the service user
- Second layer is the Network Access
  - from user's point of view basic requirement for all the other QoS aspects and parameters
- Third layer contains other QoS aspects
  - Service Access, Service Integrity & Service Retainability
- Different services are located in the fourth layer
  - Their outcome are the QoS parameters as perceived by the user
- See Recommendation ITU-T E.804  
„Quality of service aspects for popular services in mobile networks” for mobile KPIs etc.

# QoS aspects of Mobile



# Potential QoS Parameters Classified under Categories

ITU-T Recommendation E.803



*For details of QoS parameters that have been identified as being potentially useful for comparison of SPs' performance levels , please read ITU-T Recommendations E.803*



# Any questions ?



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# IMPROVING QUALITY OF LIFE..



Emergency



Education



Health



Agriculture



Investment



Applications



Policy & Regulation



Governance



Transport



Sensor Networks



Universal Broadband



Green ICT & E-Waste



Capacity Building



Measurements



Electricity



SMART  
SUSTAINABLE  
CITIES



Infrastructure Security



Privacy & Security



Water



Digital Inclusion



Spectrum Management



Standards, Conformity &  
Interoperability



Teleworking