Central Scientific Research Telecommunication Institute

International Telecommunication Union

Setting up a National Transmission Plan & QoS for International Connections

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Workshop

IMPLEMENTATION EXPERIENCE OF NETWORK PERFORMANCE PARAMETERS CONTROL SYSTEMS AND GRANTING REQUIRED LEVEL OF SERVICES QUALITY ON THE OPERATOR NETWORKS. SENSOR NETWORKS – AS OPTIMIZATION TOOL FOR VEHICULAR TRAFFIC FLOW

Moscow

27-29 April 2011

Outline

- Setting up a National Transmission Plan
- QoS for International Connections
- Multi-Vendor Environment
- IP Telephony Aspects
- IP Terminal Issues

Pre-Requisites for a National Transmission Plan

- Definition of Network Structure & Topology
- Definition of Desired End-to-End QoS
  - Absolute Minimum
  - Average over Usage/Regions/Calendar Time
- Specification of QoS & NP Parameters
  - End-to-End
  - For each Network Element & Terminals
- Selection of Reference Connections
  - Typical Connections (High # of Occurence)
  - Complex Connections (QoS Problems expected)
- Legal Regulatory Framework
  - To enforce Requirements
  - To strengthen Customers‘ Rights
  - To solve issues in Cases of not achieved QoS

Transmission Planning in Europe (1)

- Historically, separate national transmission plan have been enforced and utilized in European countries:
  - National transmission plans based on ITU-T (CCITT) Recommendations
  - Inter-Country, intra-European telephony connections ruled by
  - The International transmission plan as per ITU-T Recommendations G.101, G.111 and G.121
- Regulatory Treatment of a Telephony Connection in Europe consists of two parts:
  - Regulation of the public network (through the Directives on an Open Network Provision) and
  - Regulation of the terminal market (through a “Terminal Directive")
  - Both of these regulations are undergoing changes with the effect that national regulatory authorities do not intervene where quality is ensured through effective competition
- The new directive for Radio equipment and Telecommunications Terminal Equipment (the "R&TTE" directive) includes a possibility for the Commission to issue regulation regarding voice performance.

Transmission Planning in Europe (2)

- As long as the market actors behave in a responsible manner, there will be no EU regulation of voice performance of customer premises equipment connected to a public network
- For the telecommunications industry it is however of value to arrive at a common transmission plan for future networks, to ensure successful global communications
- Pan-European Loss Plan has been developed
  - ETSI ES202020 harmonized with TIA
  - To assist manufacturers in achieving satisfactory voice performance
  - Not a regulatory requirement

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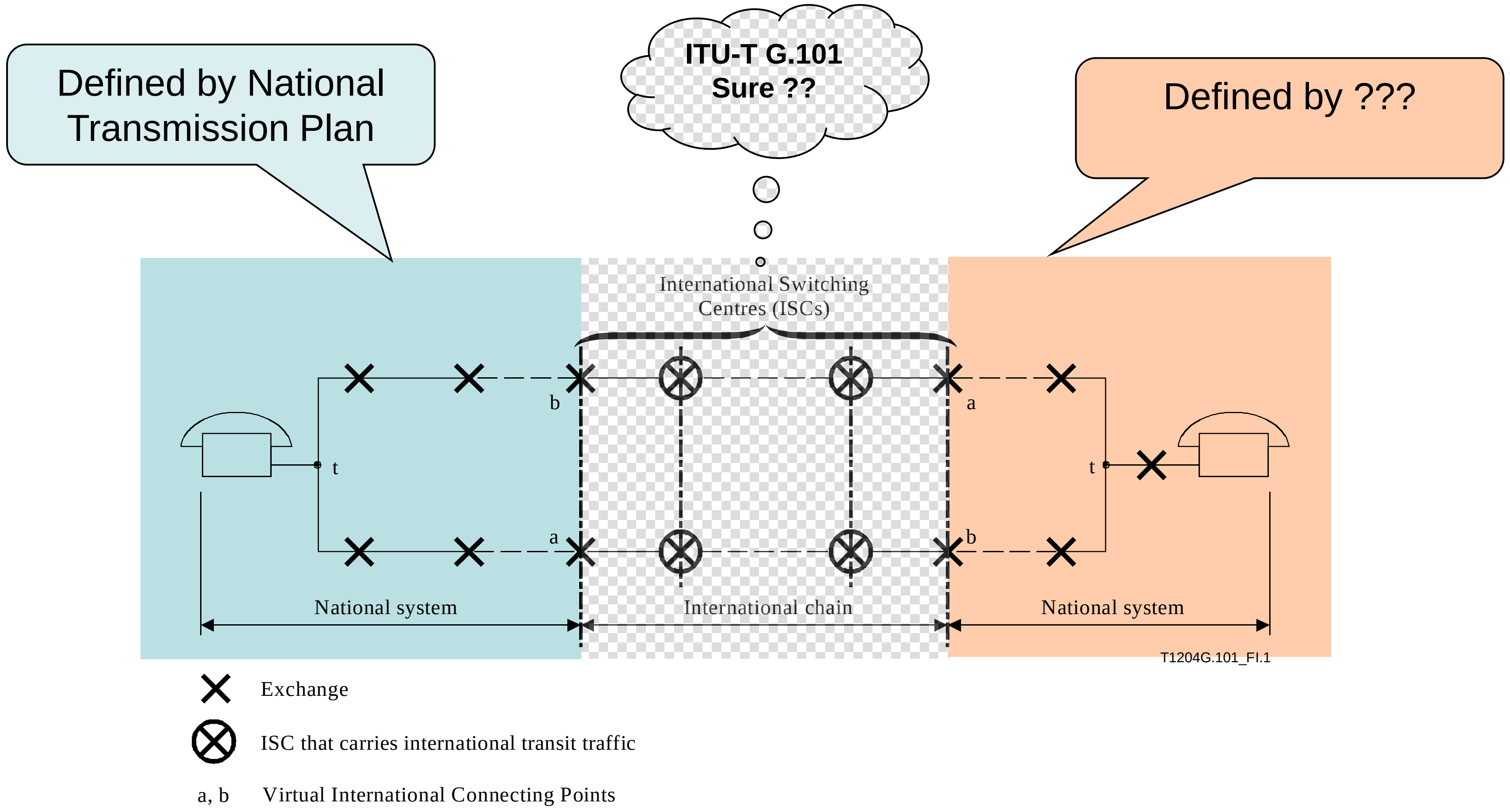
- **IP Terminal Issues**

National T-Plan as Part of Int‘l Connection



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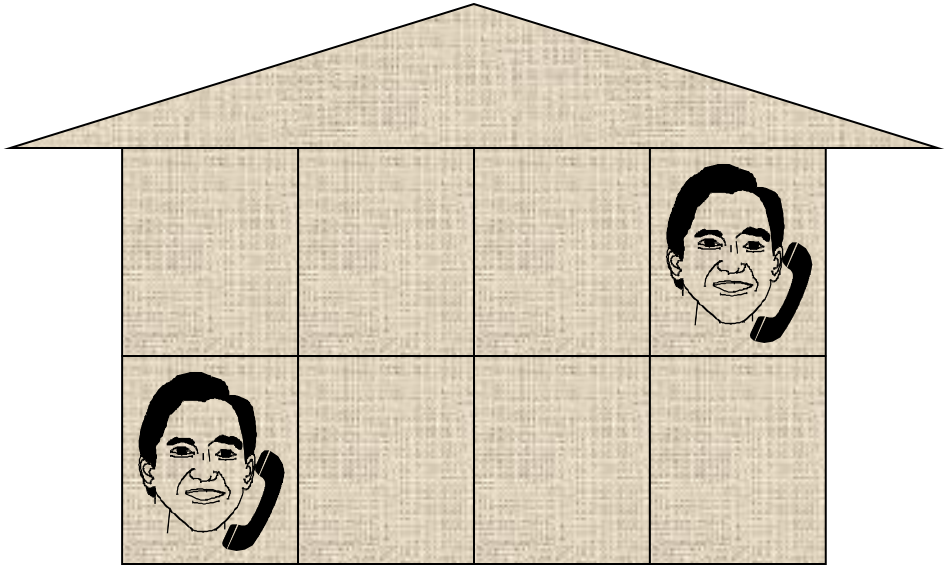
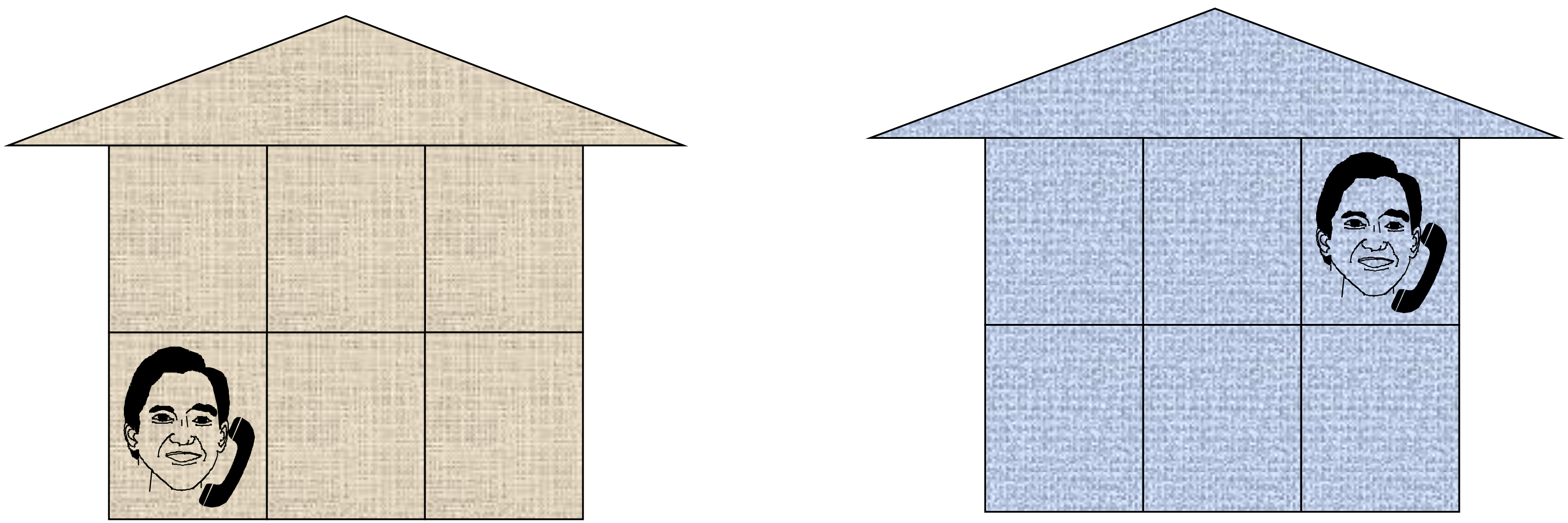
National Transmission Plan
On-Net Connections

- Single Vendor's Technology Implementation
  - Has Potential to deliver homogeneous QoS
  - Tends to be easy controllable



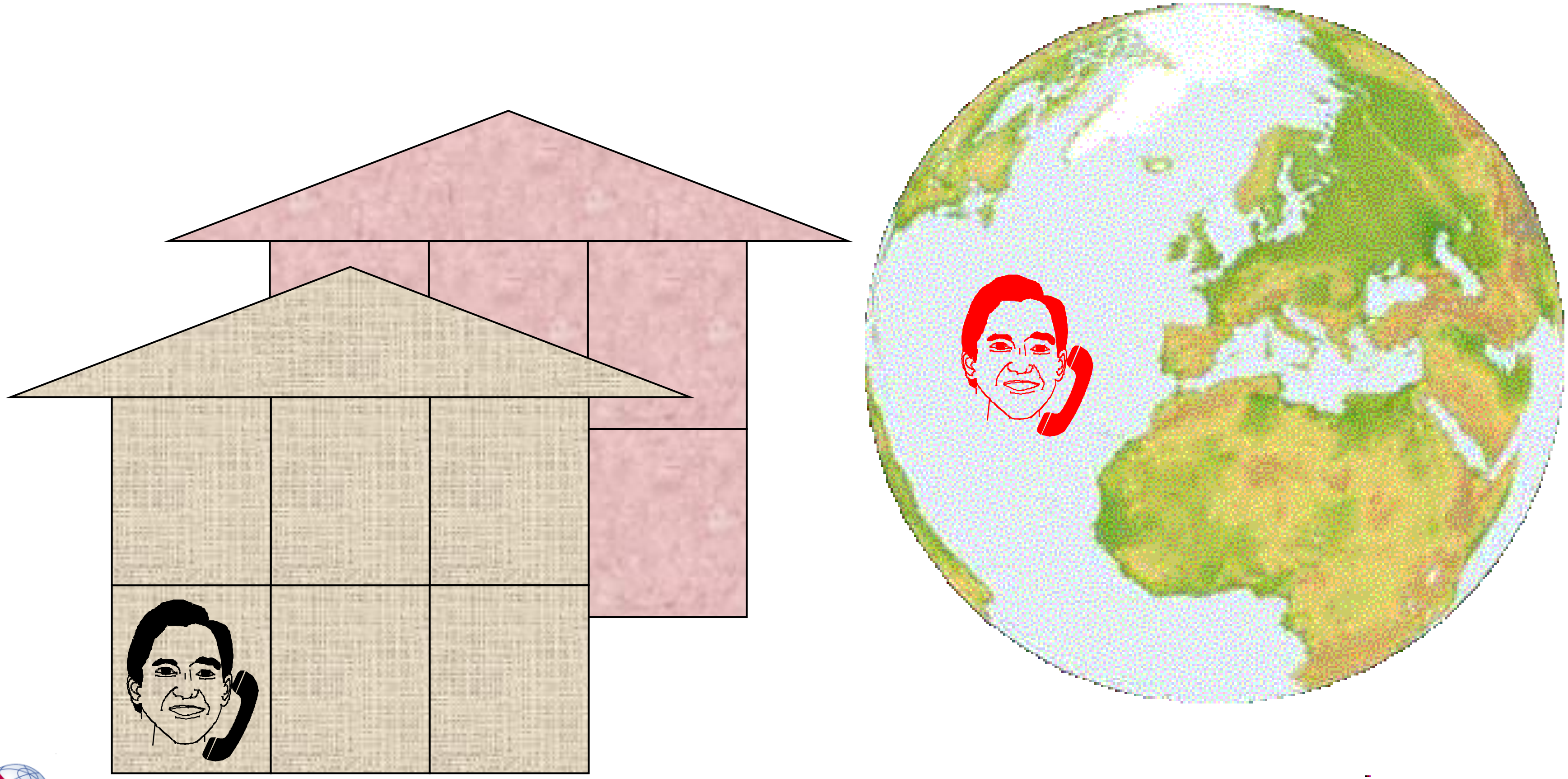
National Transmission Plan
Multi-Vendor Inter-Connections

- Multiple Vendor’s Technology Implementations
  - Has Potential to deliver QoS-Problems
  - Tends to be less controllable



National Transmission Plan
International Connections

- Connection to other Jurisdictions
- Plus Multiple Vendors in National Network



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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 <u>subjective</u> 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 the human vocal generation, reception at the ear and perception at the brain as accurate as possible
- Real-time Recording or Monitoring of Waveform Signals
- Use an algorithm to predict the results of a subjective test
- Faster but correlation with subjective test may vary
- May have a large computational footprint
- Current Models include P.862 (PESQ), P.563 (both between electrical interfaces), TOSQA (also suitable for acoustical interfaces) and of course P.863 “POLQA”
- Obsolete Models include P.861 (PSQM, for Codec Validation only) and a variety of vendors' proprietary Models

Estimation Models

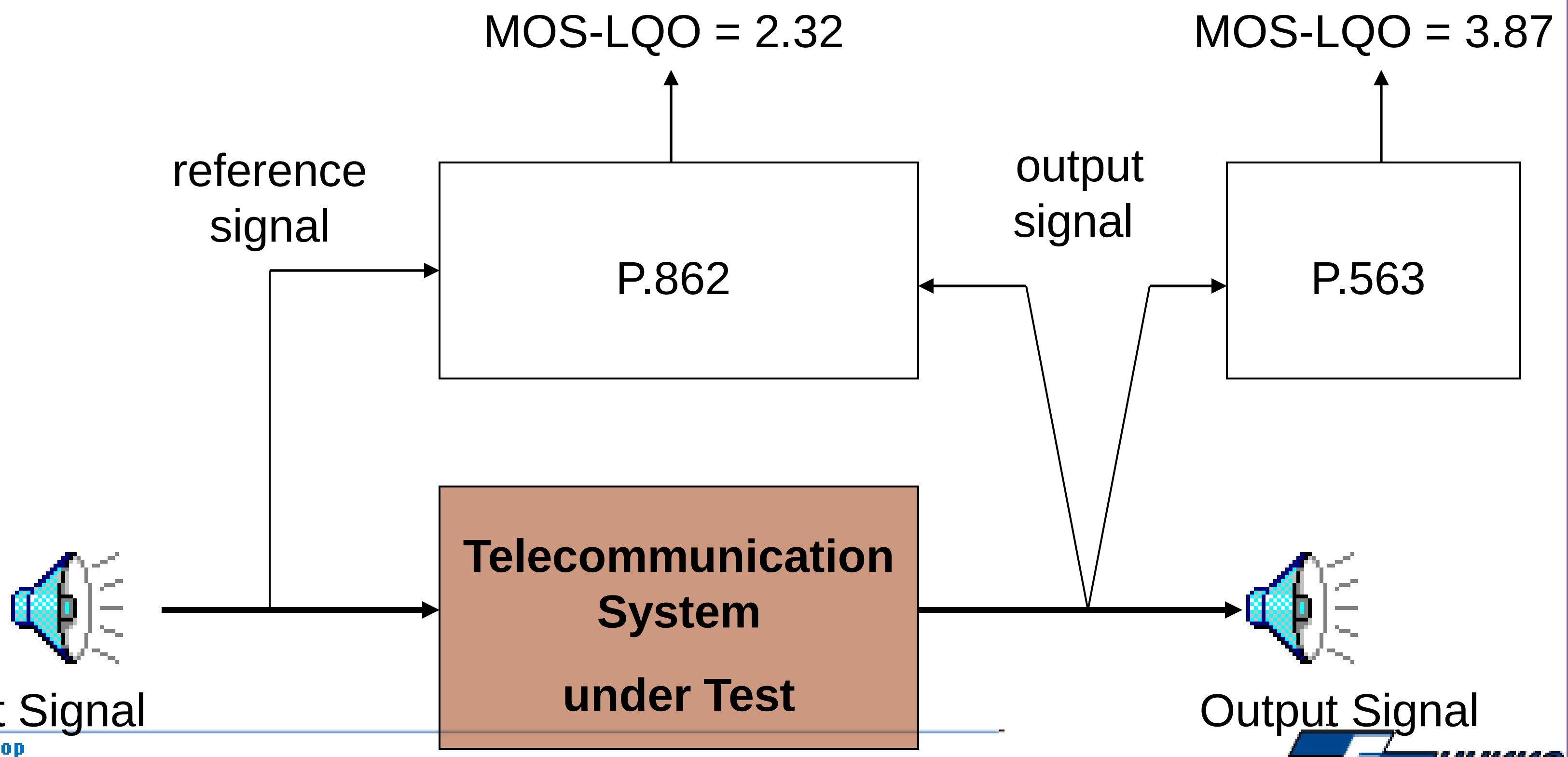
- Estimation or Parametric Models are based on combination of parameters that again can be measured, estimated or pre-determined by separate means
- Use the actually measured parameter values such as echo, delay, noise, levels etc.
- Probably has an even lower correlation with subjective testing than 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

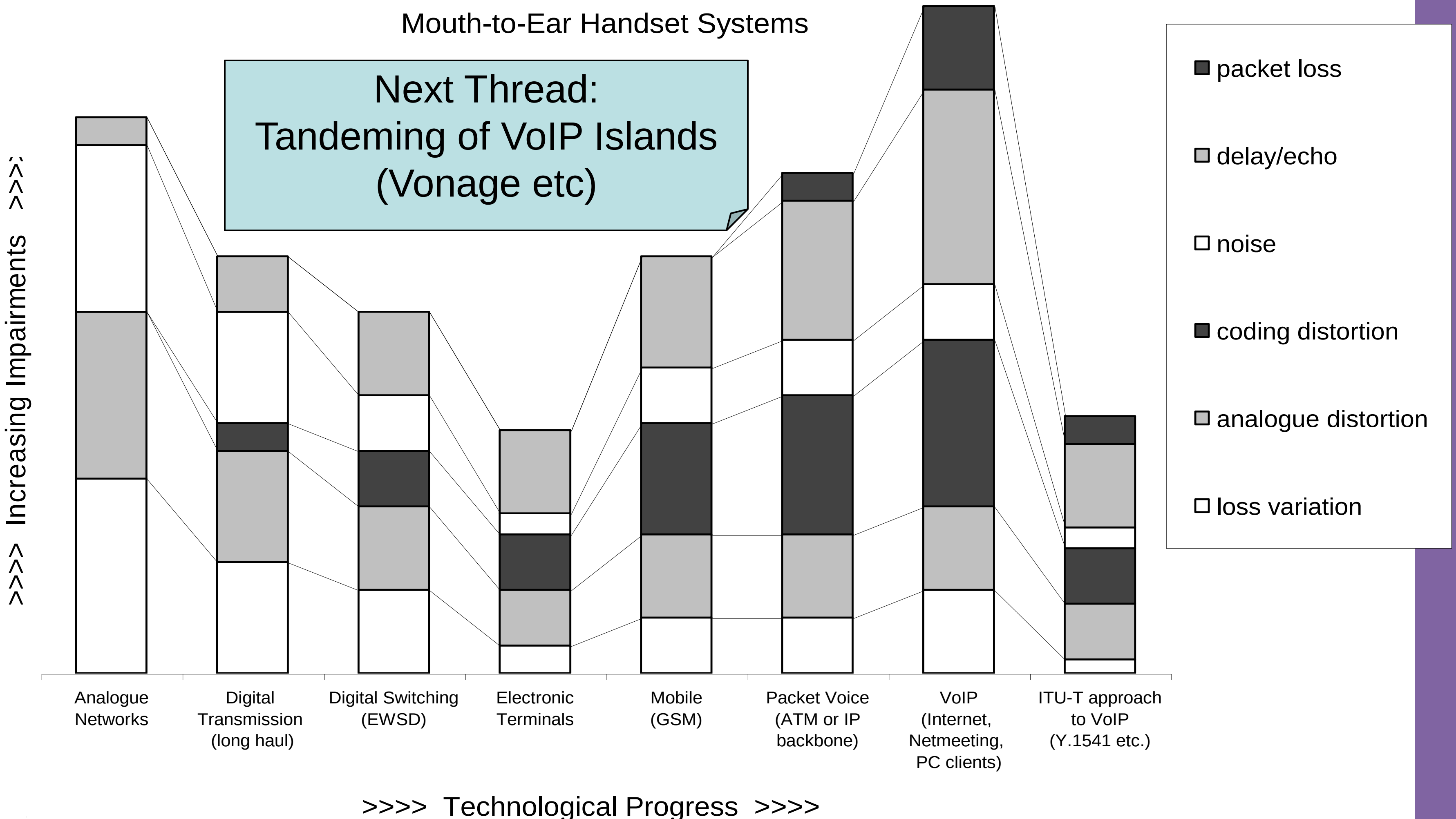
- Subjective Tests
  - MOS-LQS = for Listening Quality from a Subjective Test
  - MOS-CQS = for Conversation Quality from a Subjective Test
- Objective Models
  - MOS-LQO = for Listening Quality from an Objective Model
  - MOS-CQO = for Conversation Quality from an Objective Model
- Estimation Models
  - 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

MOS-LQO Assessment Comparison

- With proper operating Telecom System P.862 and P.563 produce Similar Results
- With Certain Impairments (e.g. packet drop-outs) in the Telecom System P.862 produces Correct Results while P.563 cannot produce proper Results



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

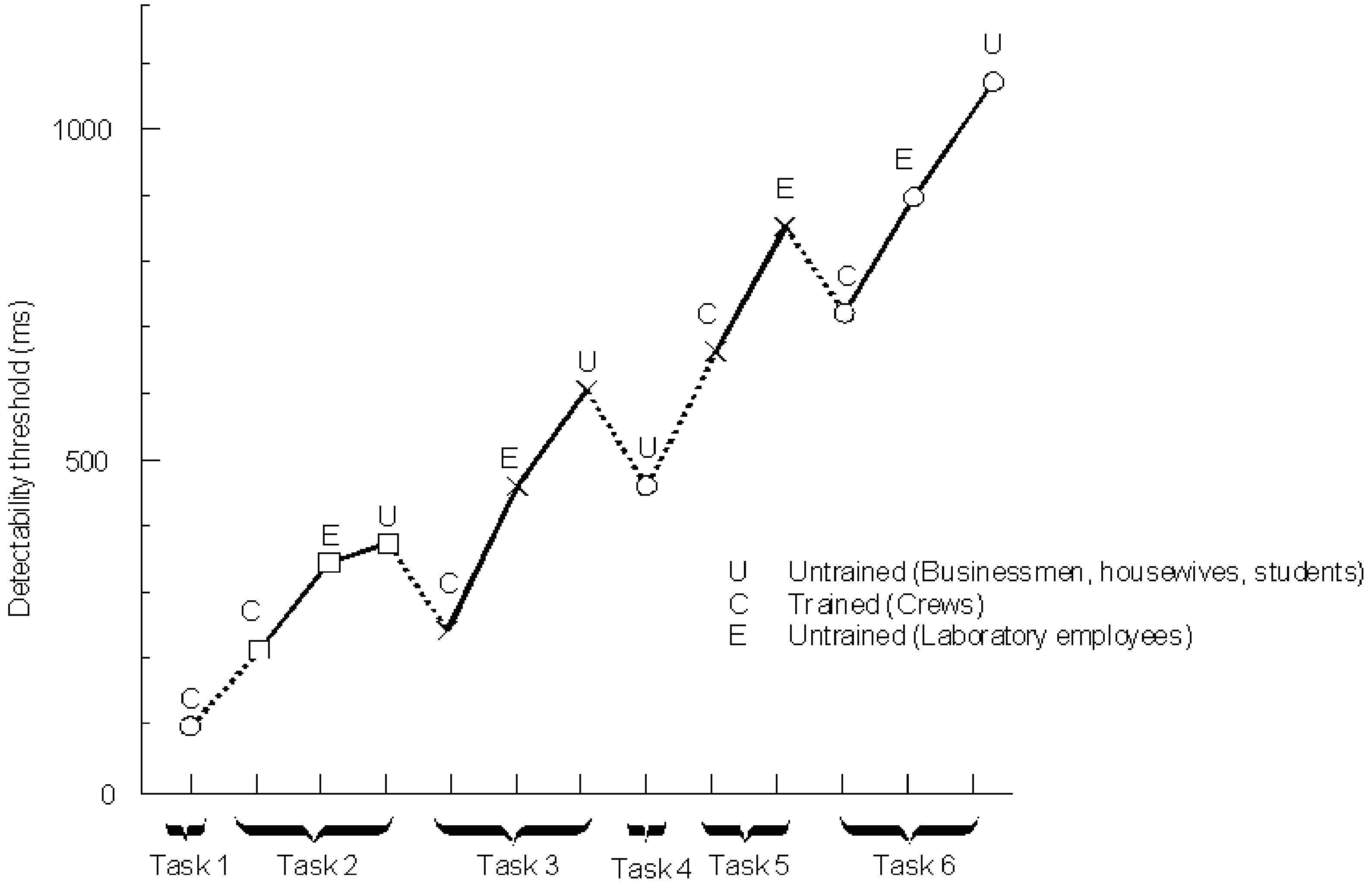
- Additional Impairments Due to IP Transport
  - Longer Delay (Latency)
  - Jitter
  - Packet Loss

- Perception of Delay strongly depends on Conversational Task

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

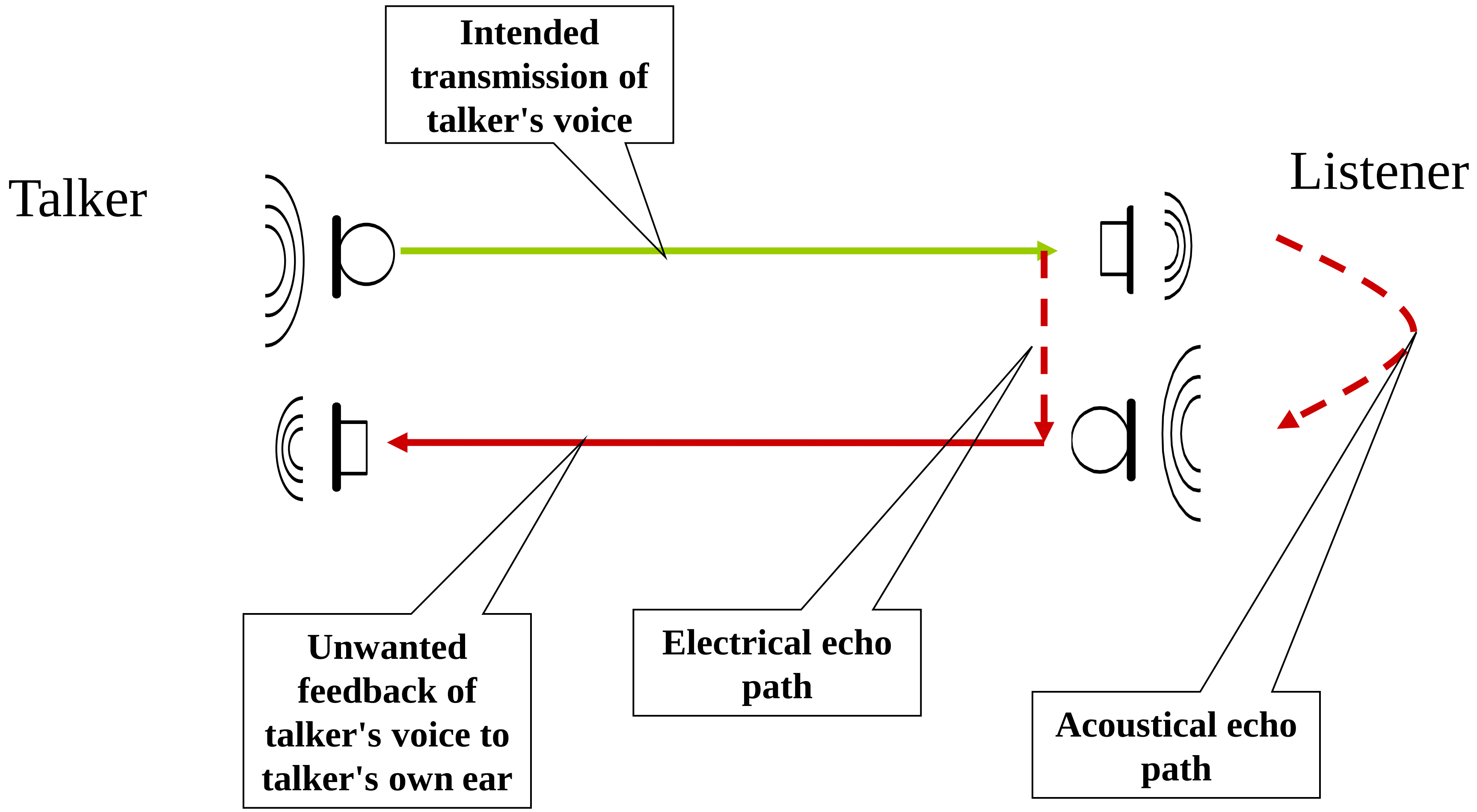
Detectability Threshold of Round-Trip Delay



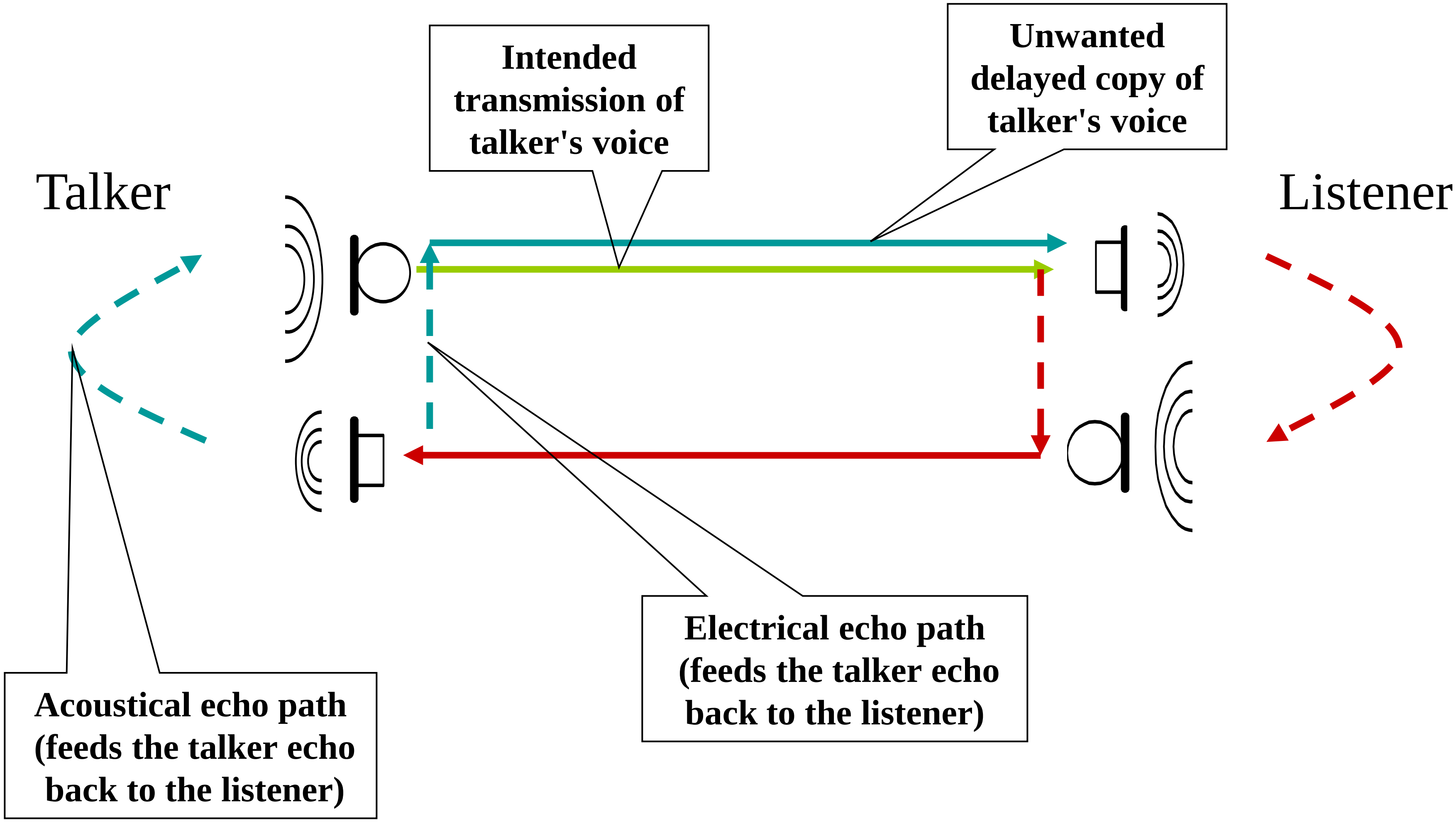
Effects of Echo

- 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

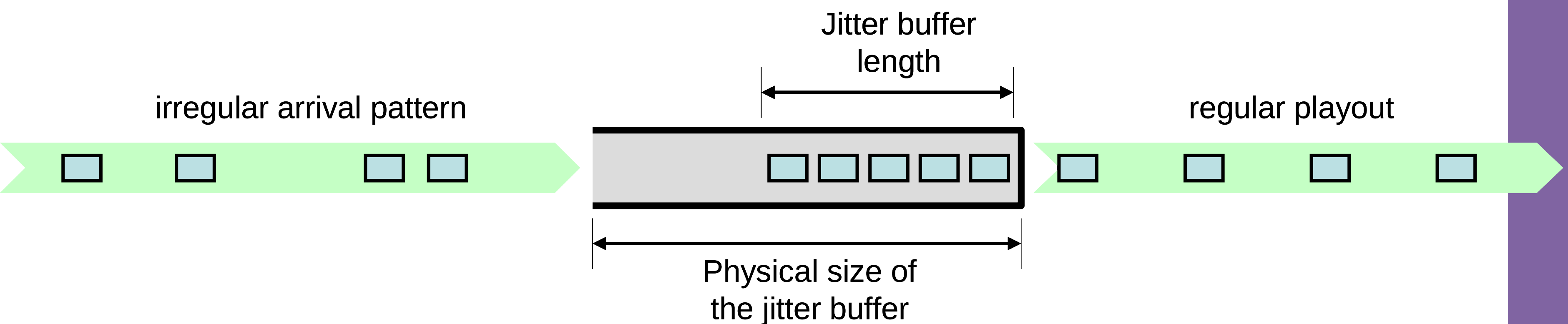

Intended transmission of talker's voice
Talker
Listener
Unwanted feedback of talker's voice to talker's own ear
Electrical echo path
Acoustical echo path

Listener Echo

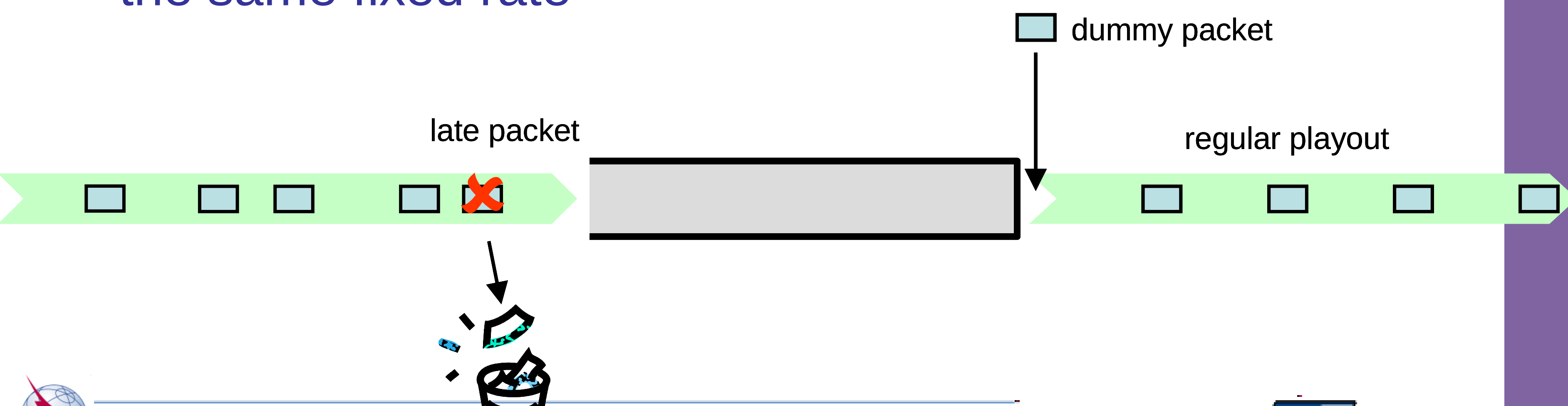


Jitter Buffer converts Jitter to Delay

- 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 Perceptional Impact due to Lost Packets
    - Different Methods
    - Different Complexity
    - Different Quality

Voice Quality Recommendations for IP Telephony **(1)**

- *Delay Rec. #1:* **Use G.711 end-to-end because it has the lowest Ie-value and therefore it allows more delay for a given voice quality level.**
- *Delay Rec. #2:* **Minimize the speech frame size and the number of speech frames per packet.**
- *Delay Rec. #3:* **Actively minimize jitter buffer delay.**
- *Delay Rec. #4:* **Actively minimize one-way delay.**
- *Delay Rec. #5:* **Accept the E-Model results, which permit longer delays for low Ie-value codecs, like G.711, for a given R-value.**

Voice Quality Recommendations for IP Telephony **(2)**

- *Delay Rec. #6:* **Use priority scheduling for voice-class traffic, as well as RTP header compression and data packet fragmentation on slow-speed links to minimize the contribution of this variable delay source.**
- *Delay Rec. #7:* **Avoid using slow serial links.**
- *Speech Compression Rec. #1:* **Use G.711 unless the link speed demands compression.**
- *Speech Compression Rec. #2:* **Speech compression codecs for wireless networks and packet networks should be rationalized to minimize transcoding issues.**

Voice Quality Recommendations for IP Telephony **(3)**

- *Packet Loss Rec. #1:* **Keep (random) packet loss well below 1%.**
- *Packet Loss Rec. #2:* **Use packet loss concealment with G.711.**
- *Packet Loss Rec. #3:* **If other codecs are used, then use codecs that have built-in or add-on PLCs.**
- *Packet Loss Rec. #4:* **New PLCs should be optimized for less than 1% of (random) packet loss.**
- *Transcoding Rec. #1:* **Avoid transcoding where possible. Adds Ie and delay impairment.**

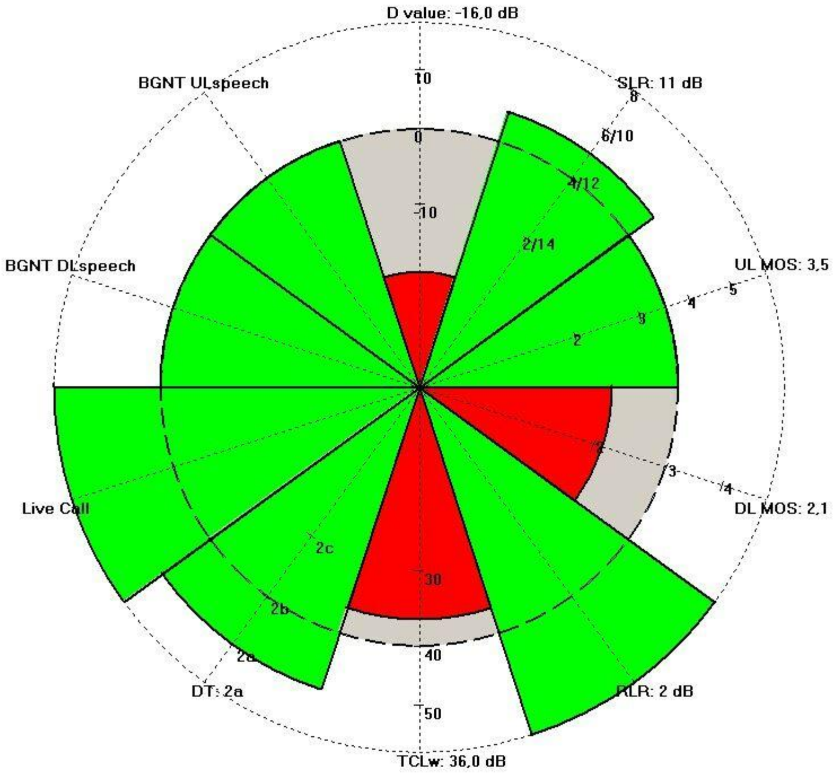
Voice Quality Recommendations for IP Telephony **(4)**

- *Transcoding Rec. #2:* **For interoperability, IP gateways should support wireless codecs or IP should implement unified Transcoder Free Operation with wireless.**
- *Tandeming Rec. #1:* **Avoid asynchronous tandeming if possible. Adds Ie and delay impairment.**
- *Tandeming Rec. #2:* **Synchronous tandeming of G.726 is generally permissible. Impairment is delay dependent, so long delay DCME equipment should be avoided.**

One View Visualization (OVV)

- Models reduce Voice Quality to Single Number
- Alternative is OVV Methodology of ITU-T Rec. P.505

- Easy to use & to understand for Non-experts
- Can serve as a Basis for Commercial Decisions on a Management or Marketing level



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QoS of IP Voice Terminals

- ETSI Standards for VoIP Terminals
  - On E2E Voice Quality as perceived by the User
  - ITU-T is missing 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

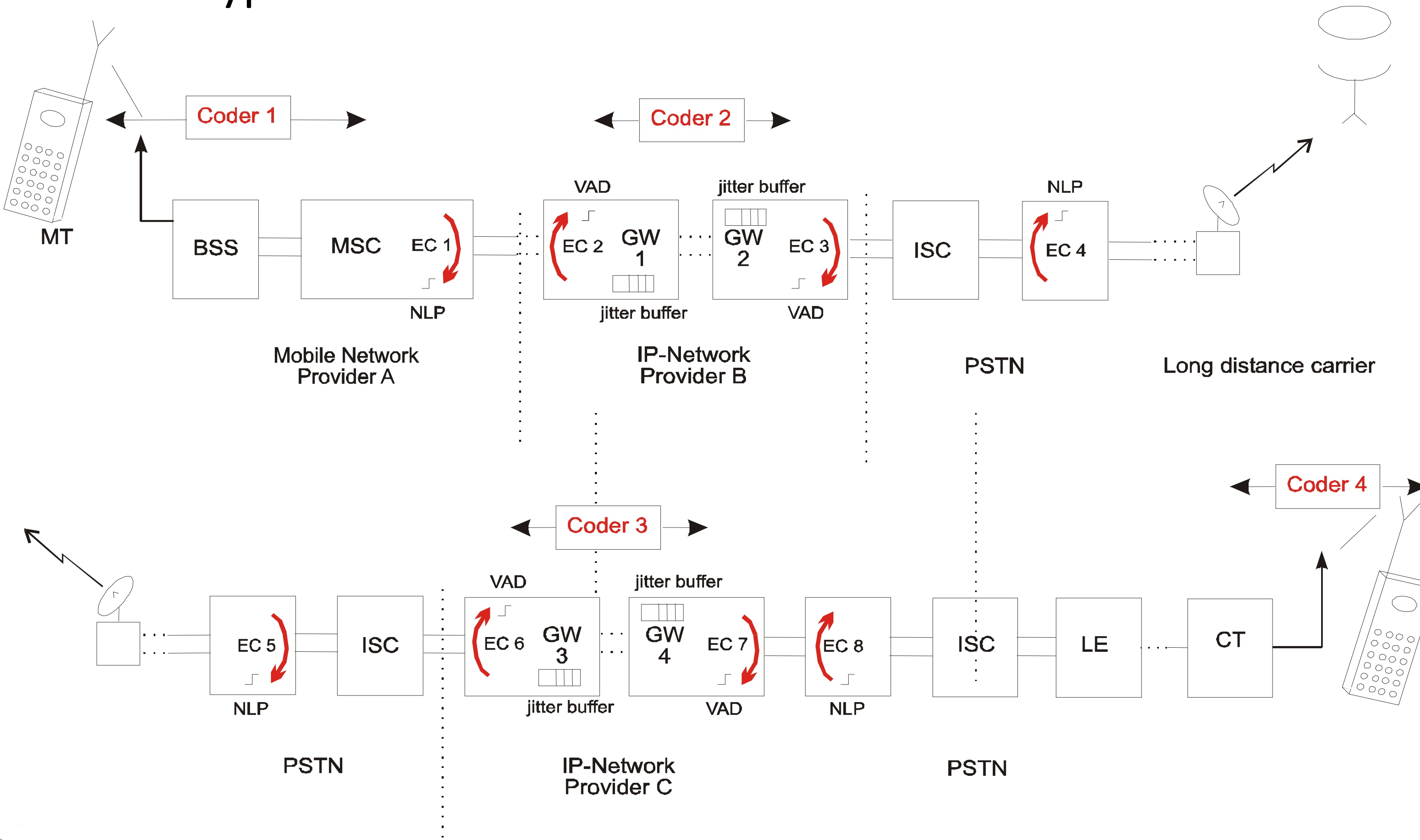
- 7.3.1 Send Delay
- For a VoIP terminal, send delay is defined as the one-way delay from the acoustical input (mouthpiece) of this VoIP terminal to its interface to the packet based network. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all 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) (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) ≤ 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.

Terminals & Network in Reality

- typical elements found in modern connections



Any Questions

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