Bearer compatibility & voice quality in packet networks

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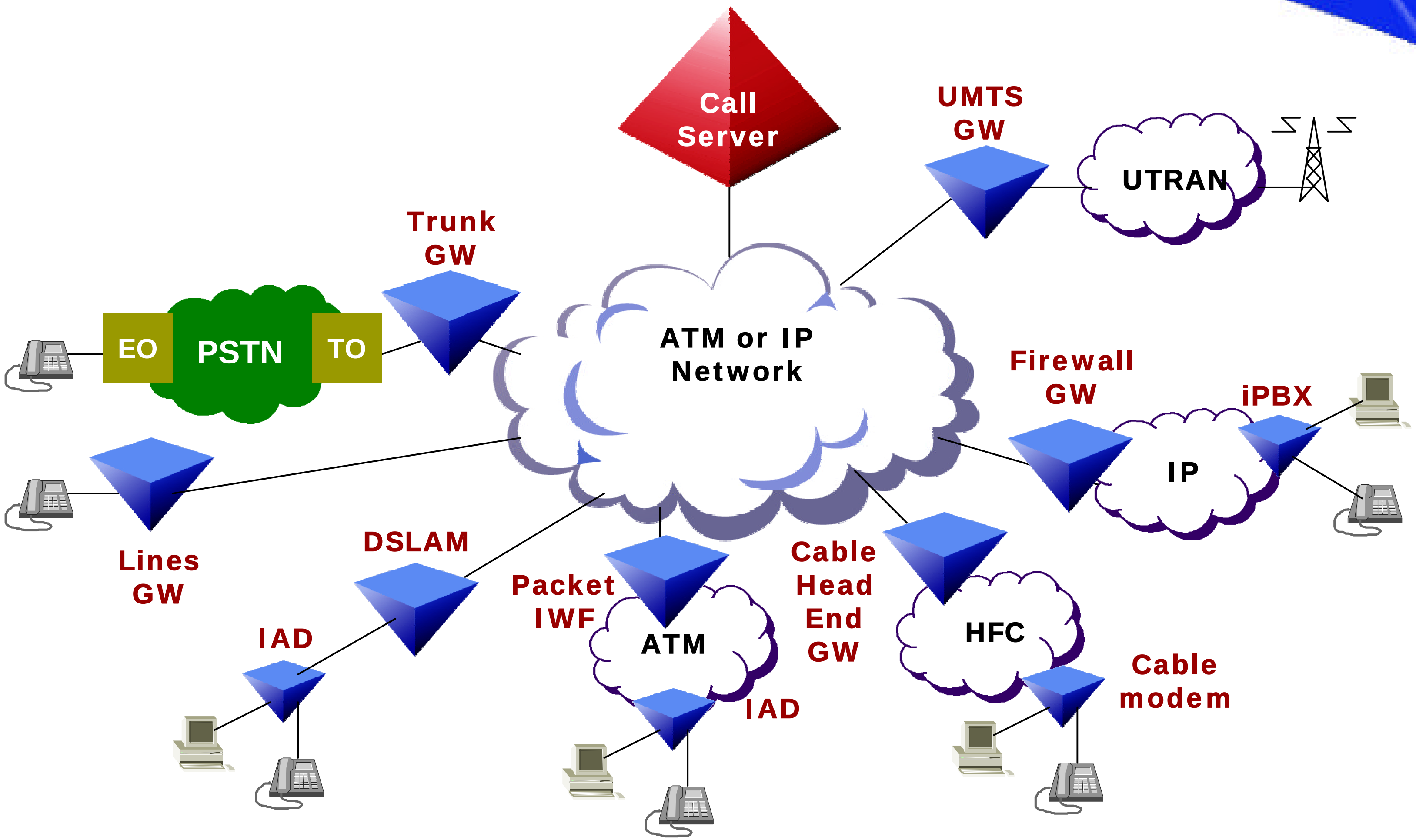
Packet networks



Packet networks

- Packet network encompass a variety of access scenarios that will continue to broaden in the future, as will the scope and roles of the gateways.
- Interoperability needs to be ensured for end-to-end solutions, and wherever possible this needs to be standards compliant to allow interworking with multiple vendor's product.
- There are many new aspects of bearer capability to be defined in packet networks when compared to the PSTN.
- Packet networks introduce additional delay and distortion artifacts that impair voice quality, although most are controllable they cannot be eliminated only minimised.
- Many new applications use voice compression which also affects quality.

Planning packet networks

What planning rules and constraints should be applied to packet networks?

- Bearer compatibility
  - Addressed by standards - although these are open to interpretation and somewhat incomplete
  - Network architecture plays a large role in the possible treatment of the bearer
  - Myriad proposals create a new legacy
- Control of voice quality
  - The PSTN and mobile networks are carefully engineered for controlled voice quality
  - No work has yet standardised packet network reference connections.
- Inter-operator hand-off
  - Communications is naturally multi-operator
  - Addressing the above should cater for this requirement.

Bearer capabilities (1)

There are several bearer capabilities affecting interworking compatibility:

green is end-to-end voice quality affecting

- **Packet protocol stack** - the transport, data-link and networking layers (and there could be several nested levels), the use of optional trunk group multiplexing and address/header compression.
- **Coding algorithm** - the range of possible codecs, the acceptable bit rates, the use of and algorithm for Voice Activity (VAD) and Comfort Noise (CNG), and any encoding variants (A/u).
- **Packet length and format** - the number of codec frames per packet and specification of optional features such as sequence numbers and other controls.
- **Ancilliary bearer control** - support for controlling quality such as bit rate changes, packet loss, end-to-end jitter.
- **Packet loss tolerance and packet loss concealment algorithms** - tolerance of receiver, any use of special encoding at the transmitter.
- **Facsimile and modem treatments** - clear channel or demodulation method and particular encoding.

Bearer capabilties (2)

green is end-to-end voice quality affecting

- **Access signalling treatment** - CAS encoding mechanism & method for tone transparency in voiceband or messaging.
- **Voice-band tone treatment** (DTMF & call progress) - messages in-band/out-band, voice squelching strategy.
- **Echo cancellation** - availability thereof and echo signal return loss quality, adjustments to loudness rating.
- **Connection continuity**, performance measurements and fault isolation detection method.
- **Circuit mode data** - support therefor and method.
- **Receiver dejitter tolerance** - source based jitter in low speed access networks can easily dominate high speed core networks.

• **Architectural definition and specification, and limited interworking capability can contain the problem**

• **Careful and considered selection of codecs, packet sizes, packet loss rate and jitter targets can ensure voice quality to a defined level of acceptability.**

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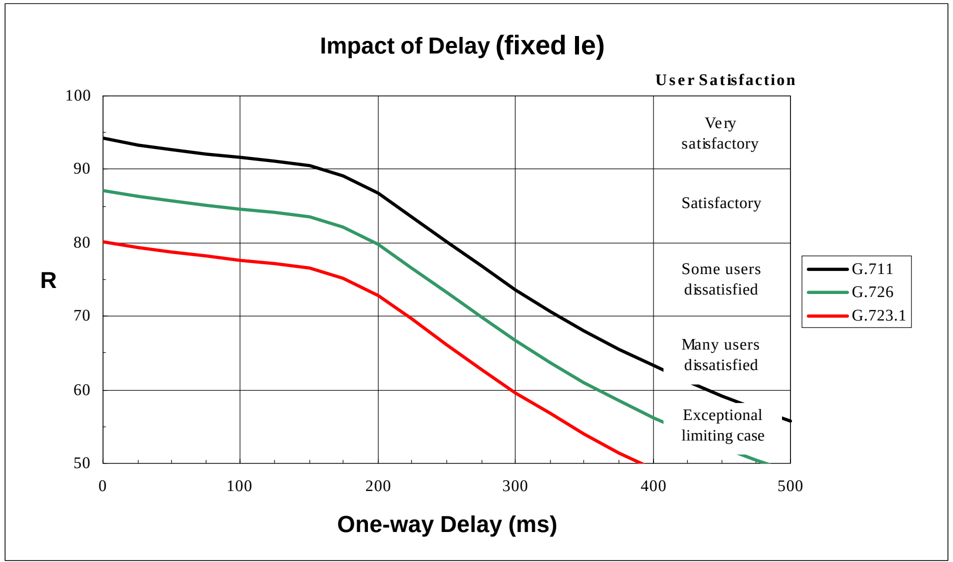
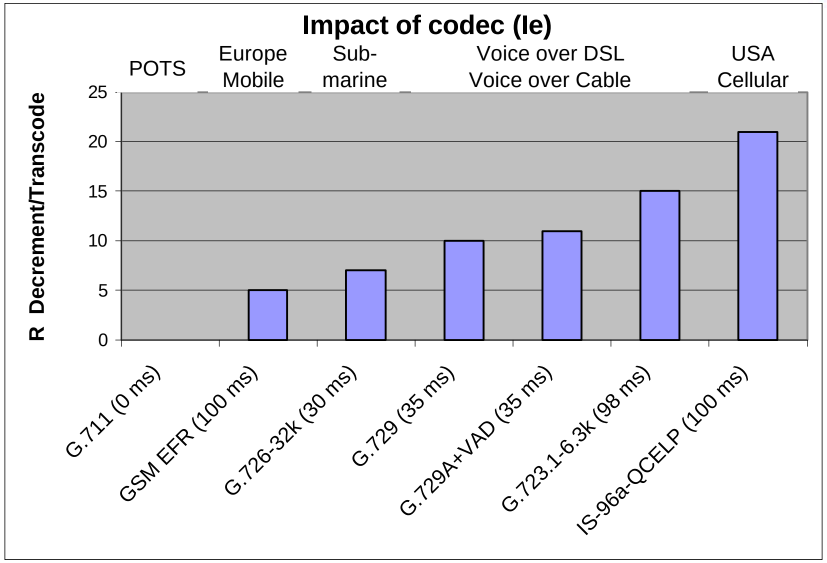
Protocol stacks

- It is simplest to consider two broadly distinct packet protocol stacks:
    - ATM
    - IP
- Voice over ATM protocol stack is straightforward:
    - firstly select the adaptation layer
    - secondly select the codec that will give acceptable performance or service compatibility.
    - thirdly fix the packet sizes that give acceptable performance, where ATM transport provides a well-defined packet-loss rate and a bounded jitter.
- Voice over IP (defined as having IP in the stack) is more difficult:
    - the format of voice packets is as much driven by RFCs as it may be by proprietary applications (codec and packet size). User v's Operator.
    - the supporting layers are manifold as are the mechanisms for congestion control (affecting jitter and packet loss rates).
    - the four parameters can be fixed to tolerable amounts as per ATM, but far from some current capabilities.

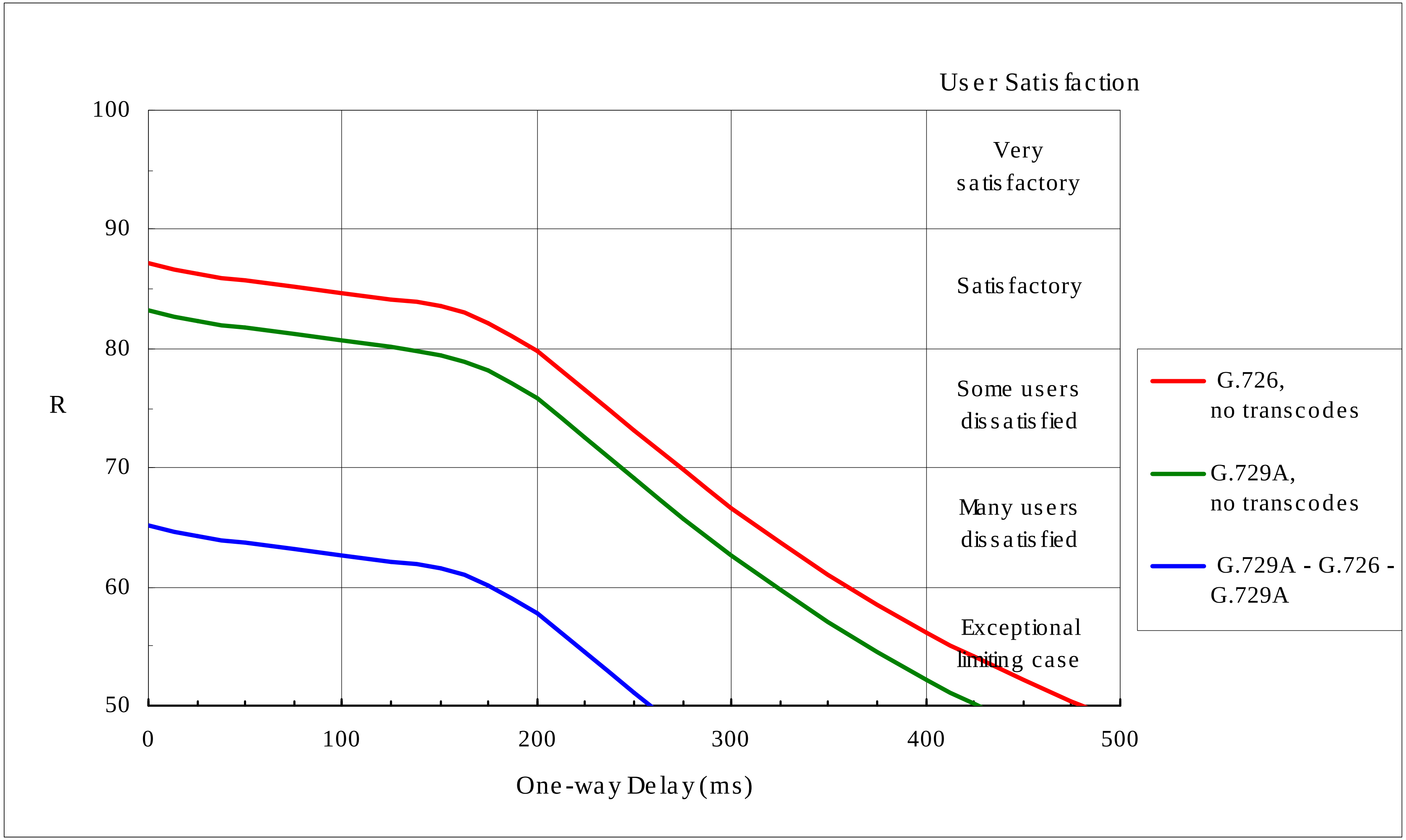
Voice quality
E-model metric - "R" value



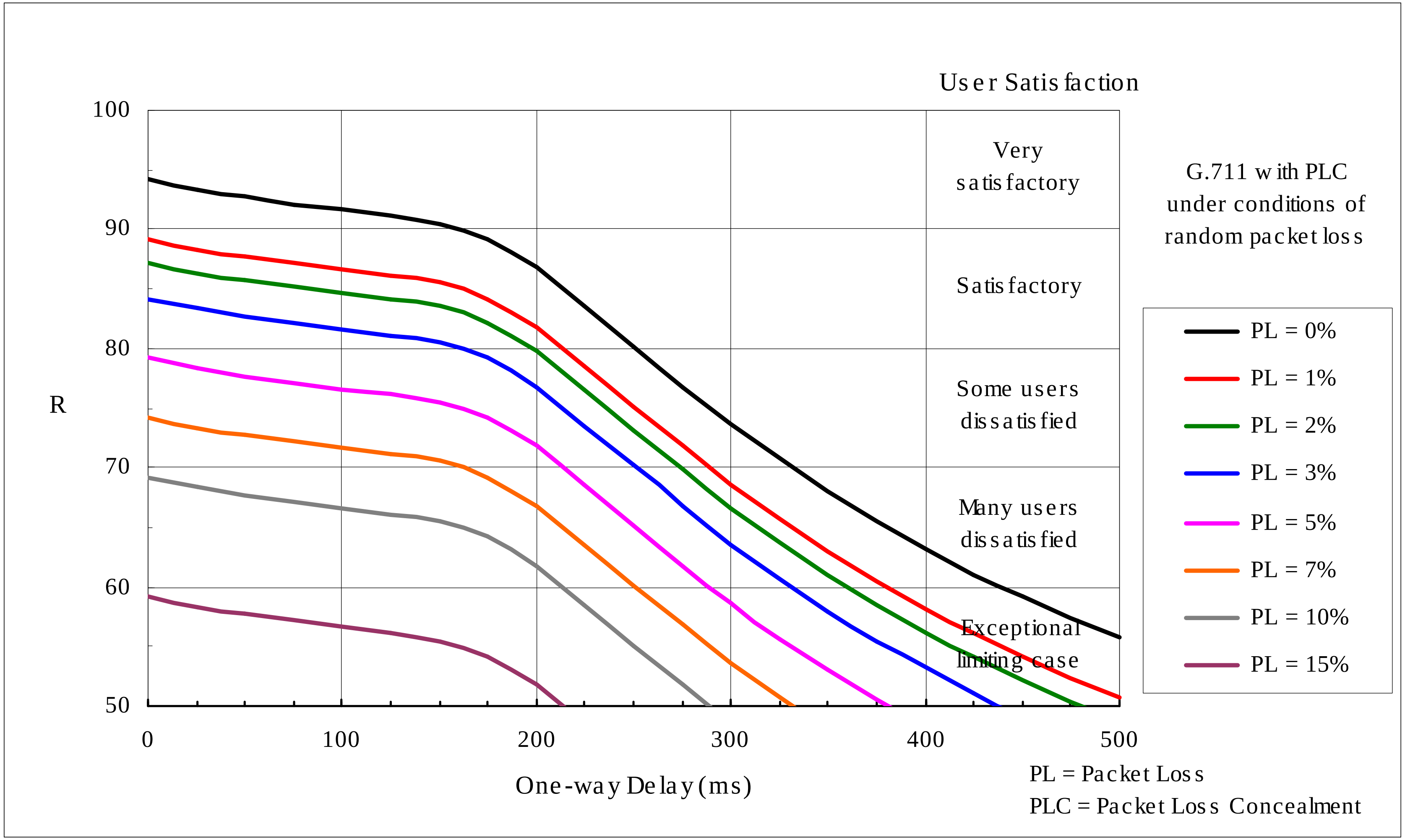


- R value describes the perceived quality of speech and forms a calculated objective measure calibrated by subjective tests.
- Delay begins to impact R value significantly above 200ms, independent of codec used, delay is introduced by packetisation, codec look-ahead, jitter in source and network.
- Quality of codec impacts the R value directly, as does packet loss based on codec.
- Direct transcoding to/from certain codecs can improve on the 'via G.711' value.

E-model: Transcoding impairment



E-model: G.711 codec
Impairment from packet loss



R value and PoW/GoB

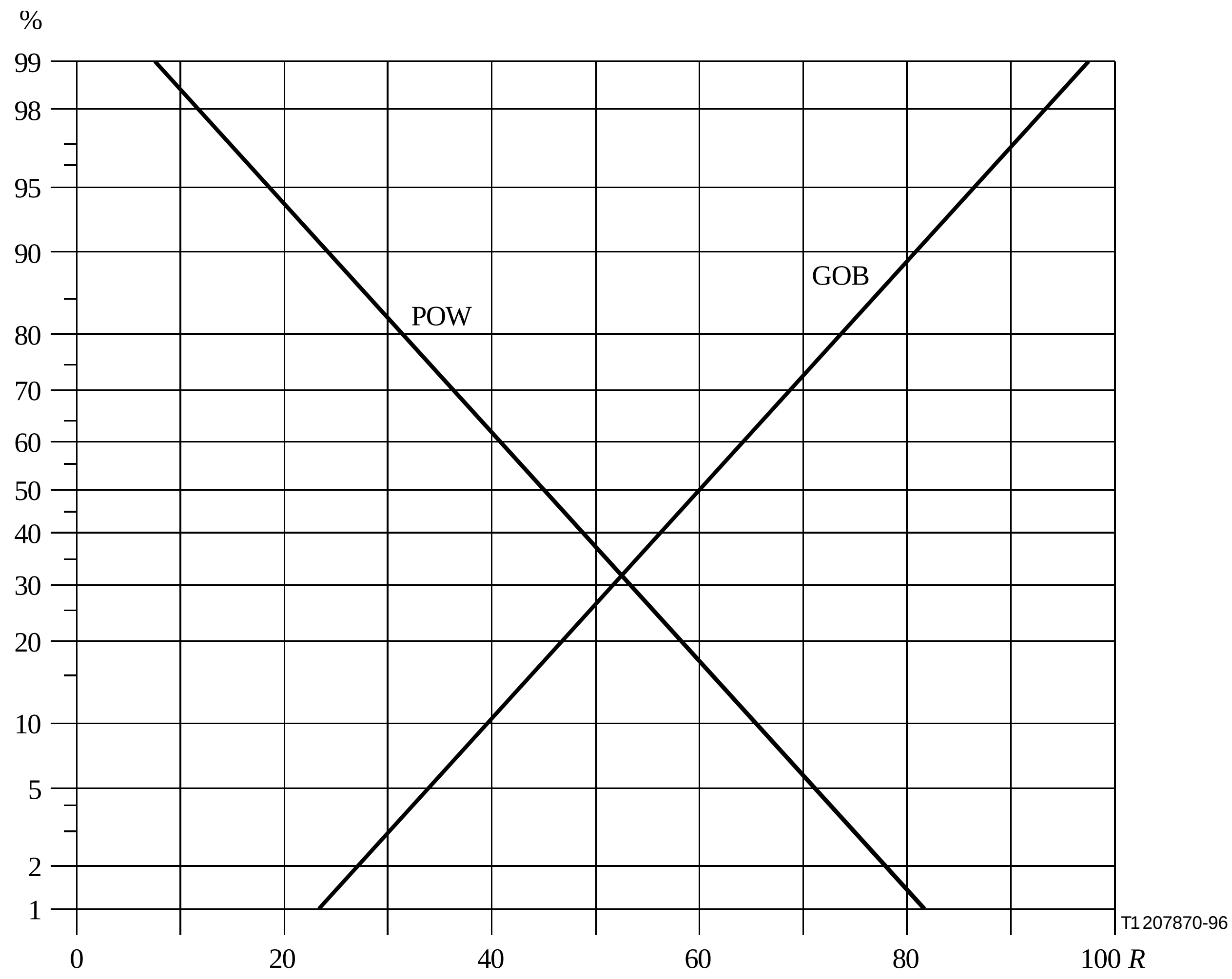


Figure B.1/G.107 - GOB (Good or Better) and POW (Poor or Worse) as functions of rating factor R

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R value and MOS

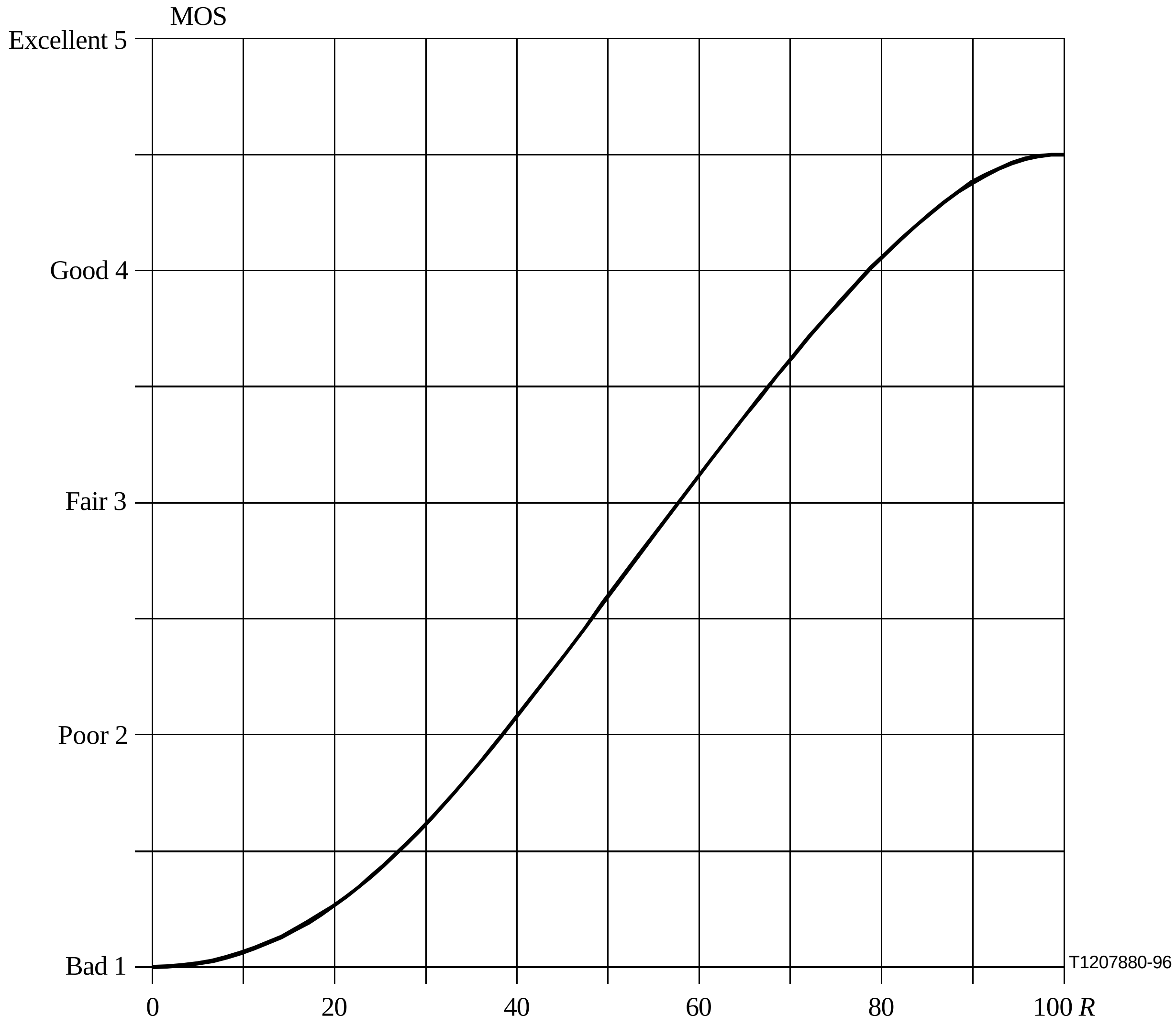
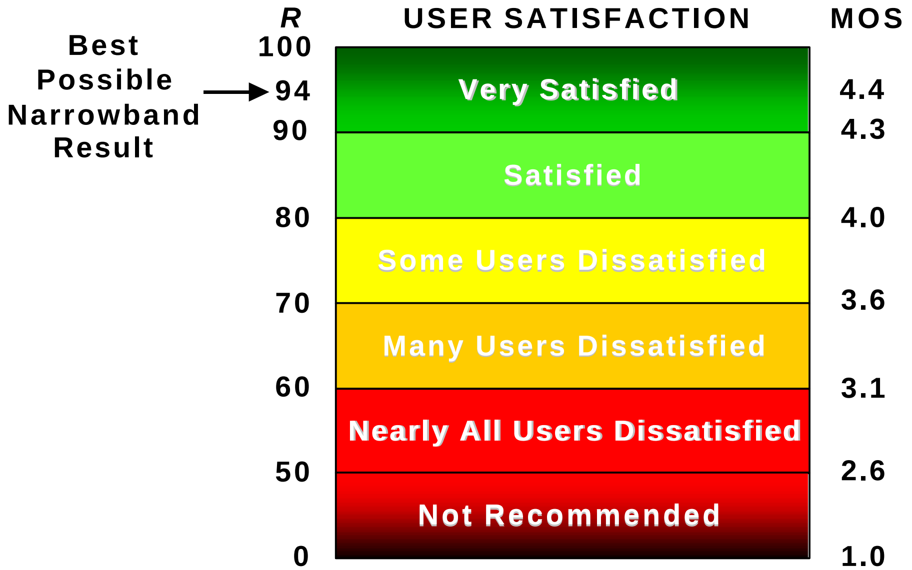


Figure B.2/G.107 – MOS as function of rating factor *R*

Succession voice quality



- **ITU-T G.109 draws decile thresholds for user satisfaction**
- **BUT the R value can and should NOT be used in absolute terms:**
  - R is a continuum - what's the difference between R = 69 & R=71?
  - the E-model is still being developed and its accuracy enhanced
  - its relationship to other subjective measures is highly non-linear, most significantly % population satisfied/dissatisfied
  - There is ongoing debate regarding the representative talker model & hence R
- **HOWEVER, relative comparisons between R can be drawn, as can be done by using a method of selecting benchmarks of known user experience.**

Voice quality benchmarks

Q. What benchmarks should voice quality meet?

A. Depends on the customer to a degree…

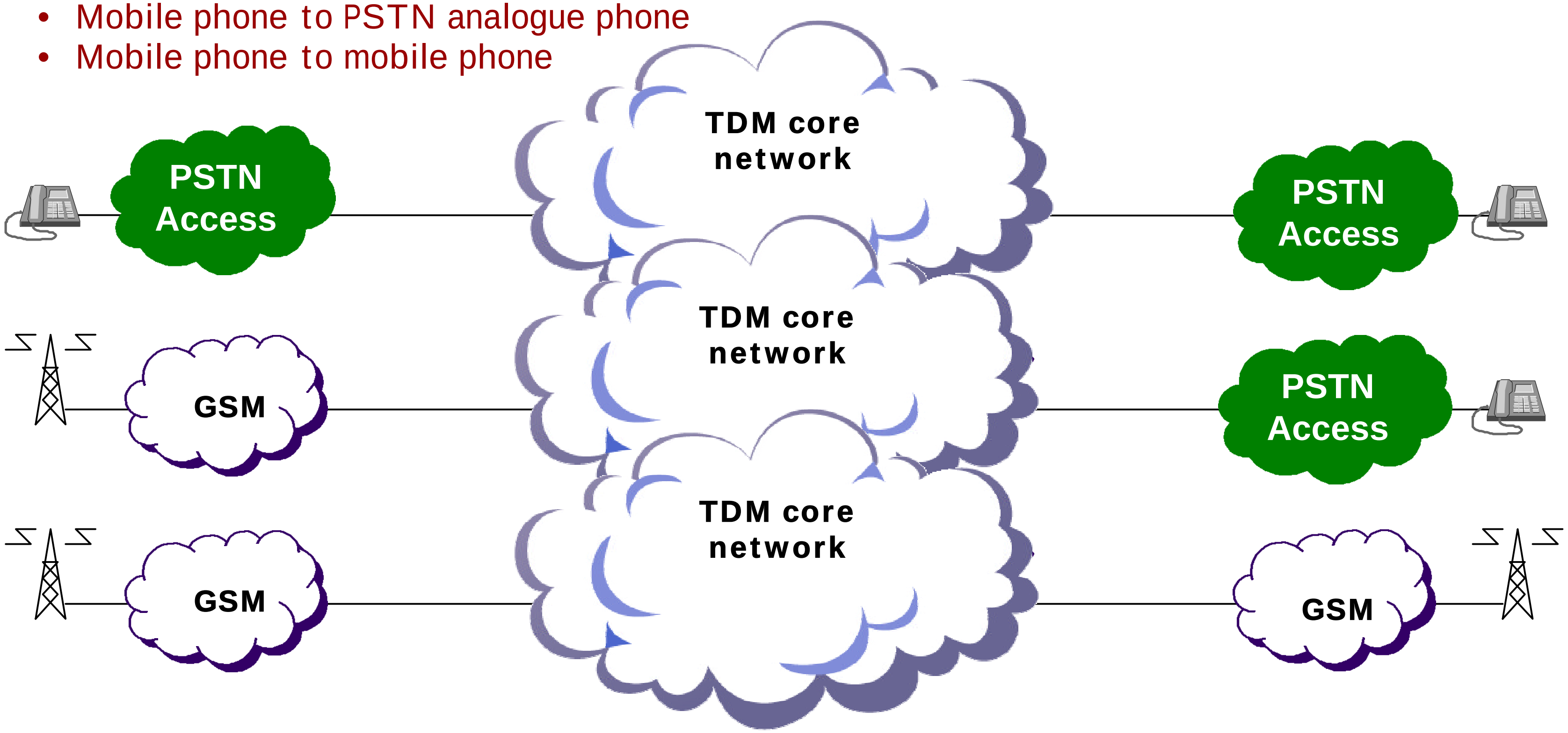
Voice quality benchmarks

- **Clarity = Quality first, low £££ or other advantage second**
  - comparable and equivalent to existing networks
  - carriers wishing to extend or replace their PSTN and mobile networks with packet core and access technologies.
  - user expectation is firmly established in using PSTN and mobile.
  - there is not necessarily any user incentive to alter that perception.
  - carrier has full control over the network technologies and their deployment.
  - margin is very small for packet network artifacts
- **Carrot = Quality, and low £££ or other advantage traded**
  - lower than existing networks by a *controlled* degree for some incentive
  - a conventional or alternative operator that provides distinct advantages to the user to lower expectation; operator gains from a wider technology selection.
  - control of terminal equipment may or may not be under the operator control.
- **Cut Price = Low £££/variety first, quality second**
  - a just serviceable quality at a rock-bottom competitive price
  - control of terminal equipment may reside mostly with the user.

User experience

What user experience is a known quantity to operators?

- For Clarity it is sensible to measure against the PSTN in accord with the most appropriate access scenario (apples for apples):
  - PSTN analogue phone to PSTN analogue phone
  - Mobile phone to PSTN analogue phone
  - Mobile phone to mobile phone



User experience

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  - PSTN analogue phone to PSTN analogue phone
  - Mobile phone to PSTN analogue phone
  - Mobile phone to mobile phone
- Newer *wired* access technologies (cable, VoDSL) can be measured against the PSTN access they replace.
- For Carrot & Cut - lower quality user experience than the PSTN can be chosen, for example to provide 'significant' (~100ms) and 'very significant' (~150ms) delay and distortion margins when compared with Clarity.
- The Carrot 'significant' delay is equivalent to the difference between a wired and mobile network.
- The Cut 'very significant' delay is equivalent to the difference between a wired and mobile used internationally.
- Delay and distortion can be traded to a certain degree, so the margins can be used in a budget allocation process to select codecs and packet sizes, and jitter and packet loss bounds that meet the individual quality criteria.

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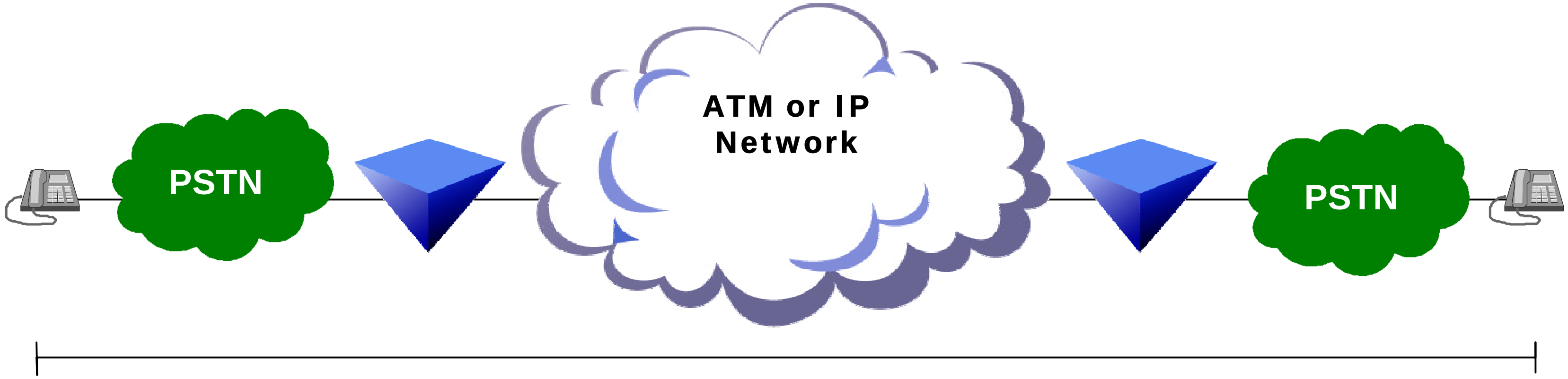
Reference calls
National and International

What reference calls will be the most demanding quality measure?

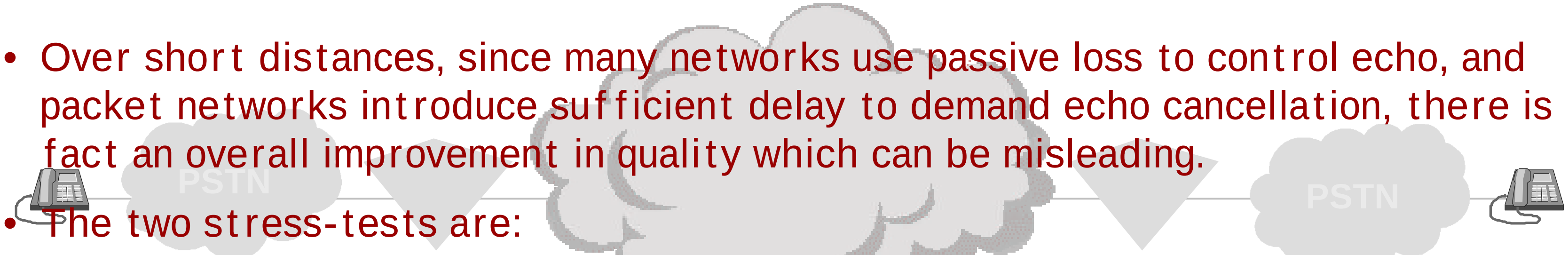


2,000km local call?
8,000km longest national call?
16,000km international adjacent country call?
27,500km longest international call?

Reference calls
National and International

What reference calls will be the most demanding quality measure?

- Over short distances, since many networks use passive loss to control echo, and packet networks introduce sufficient delay to demand echo cancellation, there is in fact an overall improvement in quality which can be misleading.



- The two stress-tests are:
  - *longest national* voice (8,000km) in a network in which echo cancellation is already present - this is also a high volume high revenue scenario.
  - the *longest international connection* (27,500), in which any semblance of quality has to contend with extremes of delay and typical use of compression on submarine links.
- There is argument that compression is no longer necessary on international connections, but:
  - submarine links do not share the same fibre technological criteria as terrestrial links
  - bandwidth is not necessarily cheap as the facilities are often leased from consortia that install them.

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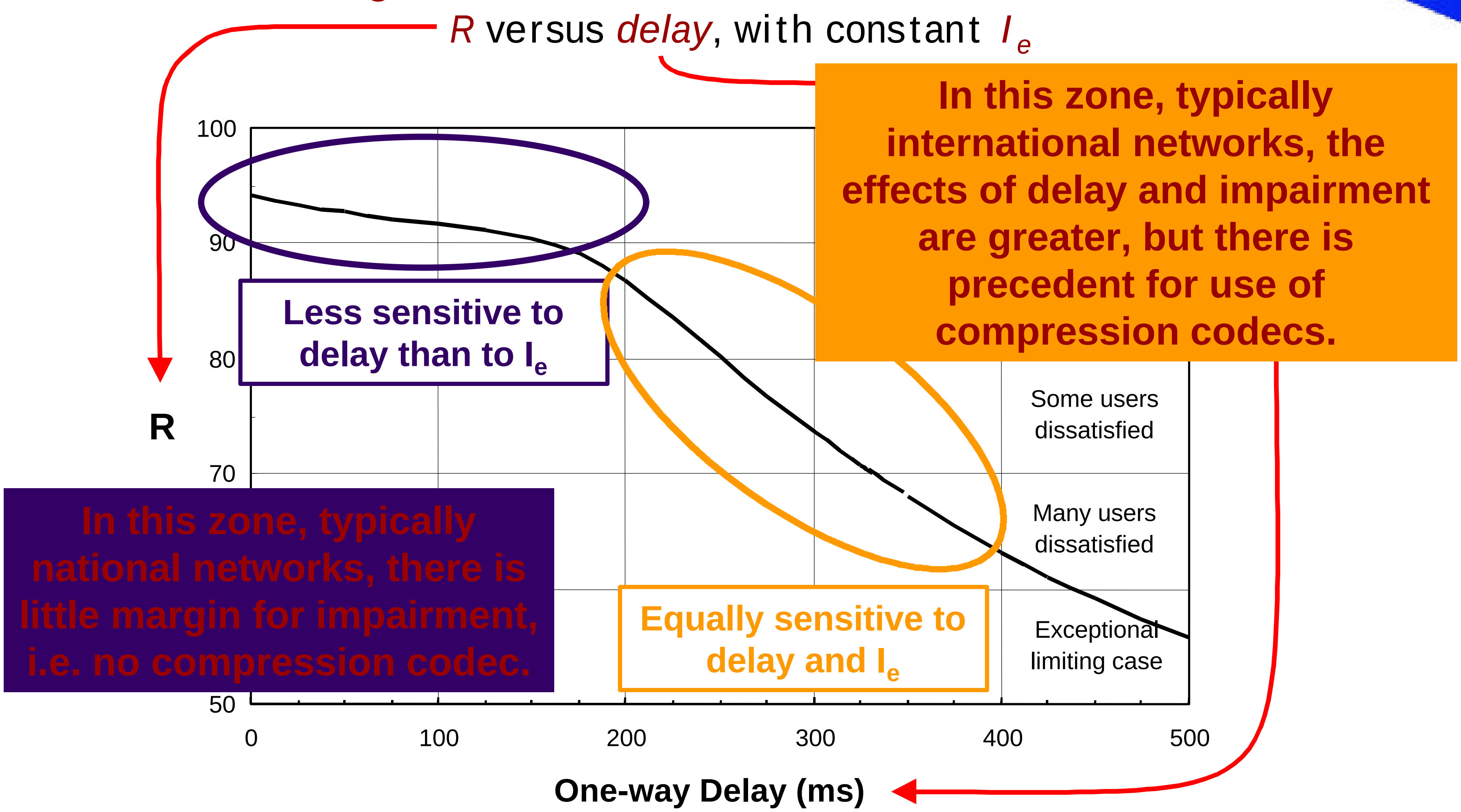
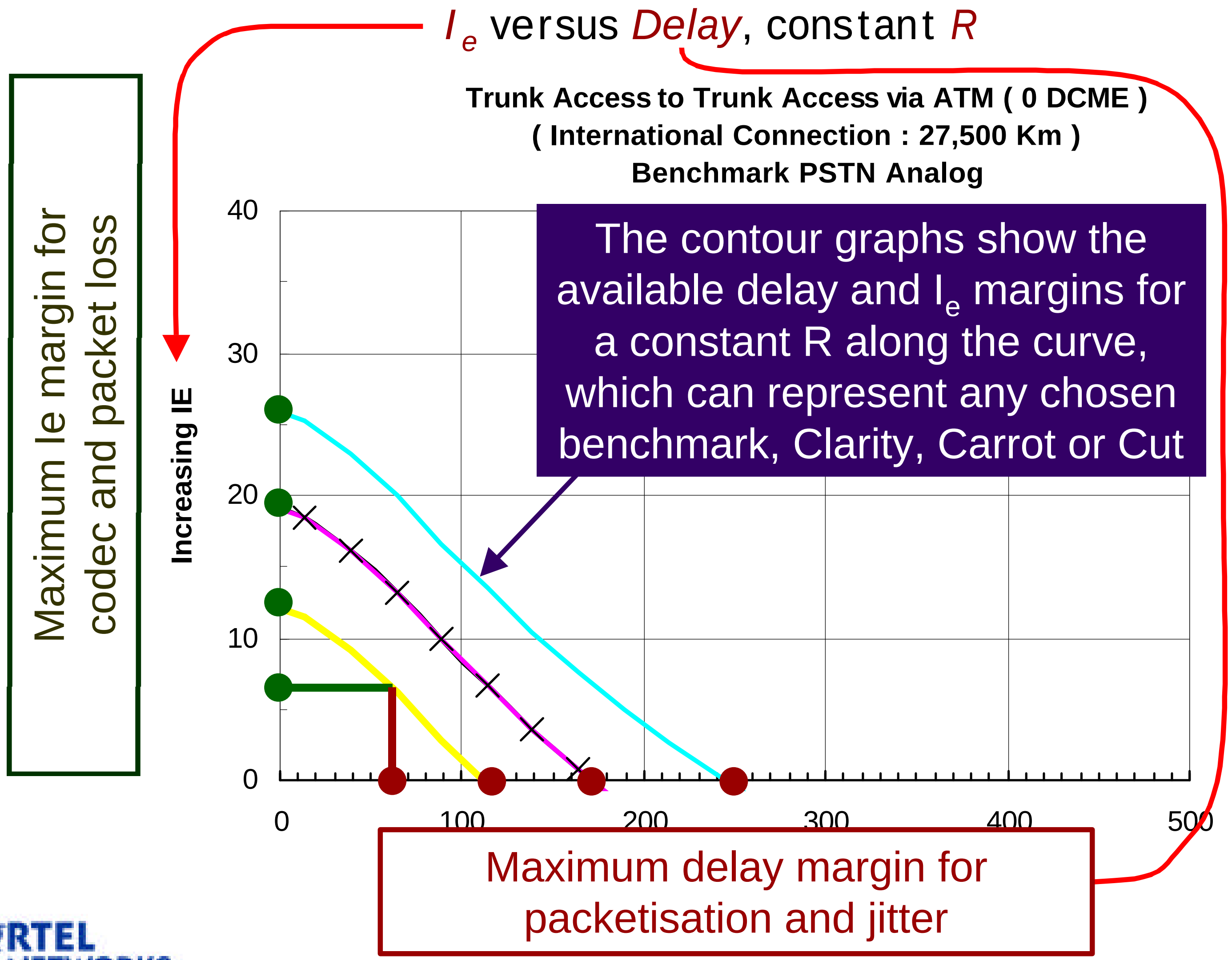
E-model:
Constant I_e contour



E-model:
Constant R contour



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When to use compression?

Voice compression has been selected for one or more of:

- a business-economic decision (mobile, submarine) where an operator profits by squeezing more calls out of a given size of pipe.
- a necessity – when the pipe is just too small because of physics.
- Historically voice compression has only been used on isolated point to point links and multiple access technologies (mobile).
  - the 64kb/s switches prevented transparency.
  - certain direct links are expensive but there is an aggregated and predictable community of interest, so the equation is simple.
  - in mobile capped available radio bandwidth forces repeated spectrum, so the equation is a complex one of mast location, radio performance and size of community.
- In the future, packet networks allow compression to be used end-to-end. Considerations now become:
  - a compatibility requirement, where it has been standardised (mobile).
  - a business-economic decision based on criteria of the *end to end* model, since there is little margin to entertain transcoding.
  - a necessity - when the pipe is just too small because of physics.
- This is much more complex than the previously isolated point-to-point decisions, as community size, traffic aggregation, blocking probability, network architecture, transport protocol all have to be taken into account.

Conclusions

To deploy packet networks:

- Voice quality can only be controlled by careful planning rules
- Derivation of the rules and bearer capabilities relies on there being a pre-planned budget allocation process - similar to the hypothetical reference connections of the PSTN.
- There is no absolute voice quality that can be considered acceptable to all, other than today's PSTN benchmark.
- Other voice quality levels are best assessed from the perspective of a known user experience, to adjudge market acceptability.
- Voice quality lower than the PSTN can be traded for convenience, price or other advantage, c.f. mobile and the ultimate trade - internet telephony.
- Congestion must be tightly controlled as there is little available margin for jitter and packet loss impairments, under conditions of:
  - Mass calling events (emergencies)
  - Focussed overload (phone-ins, emergency numbers)
  - Limited bandwidth facilities (satellite, submarine, mobile, low-speed access).
- Scalable, distributed and robust control of congestion is impossible unless bandwidth is reserved, and there is admission and traffic placement control, such as ATM and MPLS

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Questions?

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