INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.108

Amendment 2
(03/2004)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – General definitions

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Application of the E-model: A planning guide

Amendment 2: New Appendix II – Planning examples regarding delay in packet-based networks

ITU-T Recommendation G.108 (1999) – Amendment 2

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation G.108

Application of the E-model: A planning guide

Amendment 2

New Appendix II – Planning examples regarding delay in packet-based networks

Source

Amendment 2 to ITU-T Recommendation G.108 (1999) was agreed on 31 March 2004 by ITU-T Study Group 12 (2001-2004).

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

ITU-T Recommendation G.108

Application of the E-model: A planning guide

Amendment 2

New Appendix II – Planning examples regarding delay in packet-based networks

This appendix provides guidance for the transmission planner on how to deal with the delay occuring in packet-based networks in conjunction with VoIP terminals and gateways.

For the calculations quoted in this appendix, the algorithm of the E-model has been taken from ITU-T Rec. G.107 at the time of publication. In case a later revision of ITU-T Rec. G.107 does contain an improved version of the algorithm, this appendix will still provide valid guidance for tutorial purposes. However, for actual transmission planning tasks it should, in any case, be referred to the most recent version of ITU-T Rec. G.107.

For illustratory purposes, the following scenarios have been investigated:

1) two VoIP terminals interconnected via a packet-based network that complies with ITU-T Rec. Y.1541, Class 0;
2) two VoIP islands interconnected via the digital PSTN;
3) a mixed connection between a VoIP terminal (connected to a LAN) and an analogue phone (connected to the PSTN).

In addition, two codecs, G.711 and G.729A, have been considered for each scenario, while no other impairments have been considered (proper echo control assumed).

For the E-model calculations, all parameters which are not explicitly mentioned have been set to their default values as per Table 2/G.107.

The scenario in Figure II.1 describes a connection between two VoIP terminals via a packet-based network with a total network delay of 100 ms. The network delay is composed of the fixed delay and the value of the delay variation (jitter). It should be noted that, in this example, the packet-based network performs better than the maximum values set out in ITU-T Rec. Y.1541.

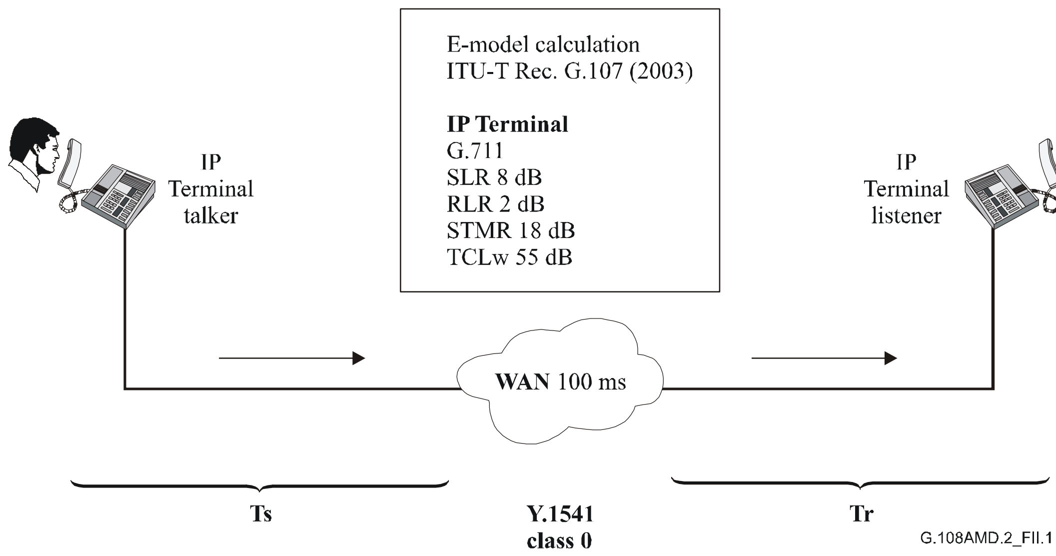

Figure II.1/G.108 – VoIP terminals with G.711 and WAN according to Y.1541 Class 0

For the E-model calculations, three different cases of terminal delay have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms
  total delay = Ts + Twan + Tr = (20 + 100 + 30) ms = 150 ms
  R = 90: Users very satisfied
- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms
  total delay = Ts + Twan + Tr = (35 + 100 + 65) ms = 200 ms
  R = 86: Users satisfied
- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms
  total delay = Ts + Twan + Tr = (50 + 100 + 100) ms = 250 ms
  R = 79: Some users dissatisfied

The quality as perceived by the user in these example calculations is identical for both sides, because the example is symmetrically composed.

The scenario in Figure II.2 describes a connection between two VoIP islands via the PSTN with a delay in the PSTN of 125 ms. The delays in the LANs are 5 ms each which are composed of the fixed delay and the value of the delay variation (jitter).

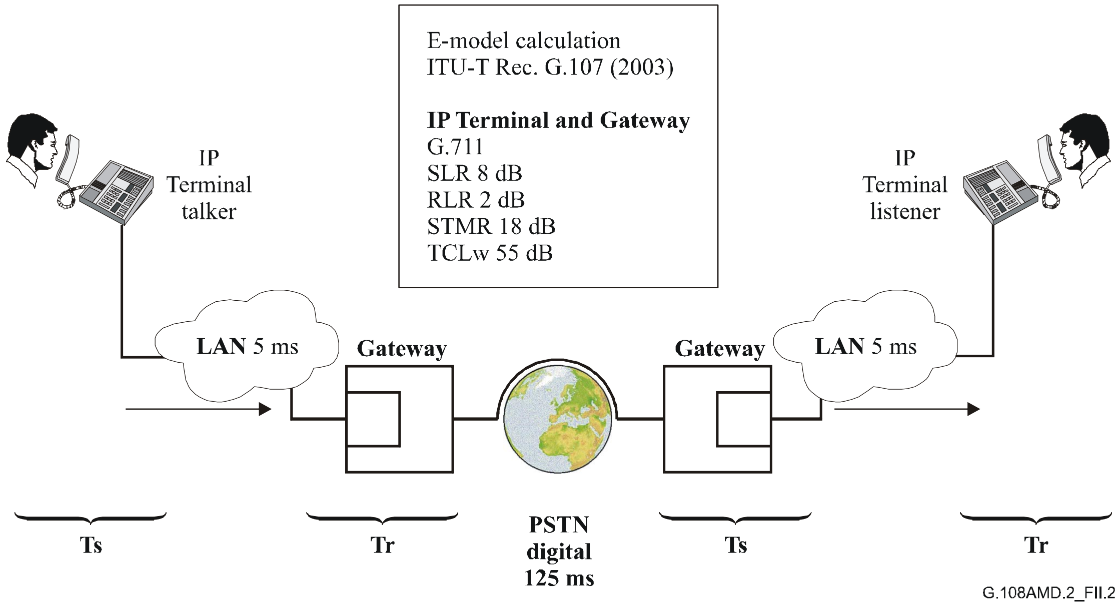

Figure II.2/G.108 – VoIP islands with G.711 and PSTN

For the E-model calculations, three different cases of terminal delay (and gateway delay) have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms

  VoIP gateway send delay = 20 ms, VoIP gateway receive delay = 30 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (20 + 5 + 30 + 125 + 20 + 5 + 30) ms = 235 ms

  R = 81: Users satisfied

- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms

  VoIP gateway send delay = 35 ms, VoIP gateway receive delay = 65 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (35 + 5 + 65 + 125 + 35 + 5 + 65) ms = 335 ms

  R = 69: Many users dissatisfied

- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms

  VoIP gateway send delay = 50 ms, VoIP gateway receive delay = 100 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (50 + 5 + 100 + 125 + 50 + 5 + 100) ms = 435 ms

  R = 59: Nearly all users dissatisfied

The quality as perceived by the user in these example calculations is identical for both sides, because the example is symmetrically composed.

The scenario in Figure II.3 describes a connection between one VoIP island and an analogue terminal connected to the PSTN with a delay in the PSTN of 25 ms. The delay in the LAN is 5 ms which is composed of the fixed delay and the value of the delay variation (jitter). In addition, the VoIP gateway provides an echo canceller according to ITU-T Rec. G.168 with sufficient tail delay covering the echo path via the local office.

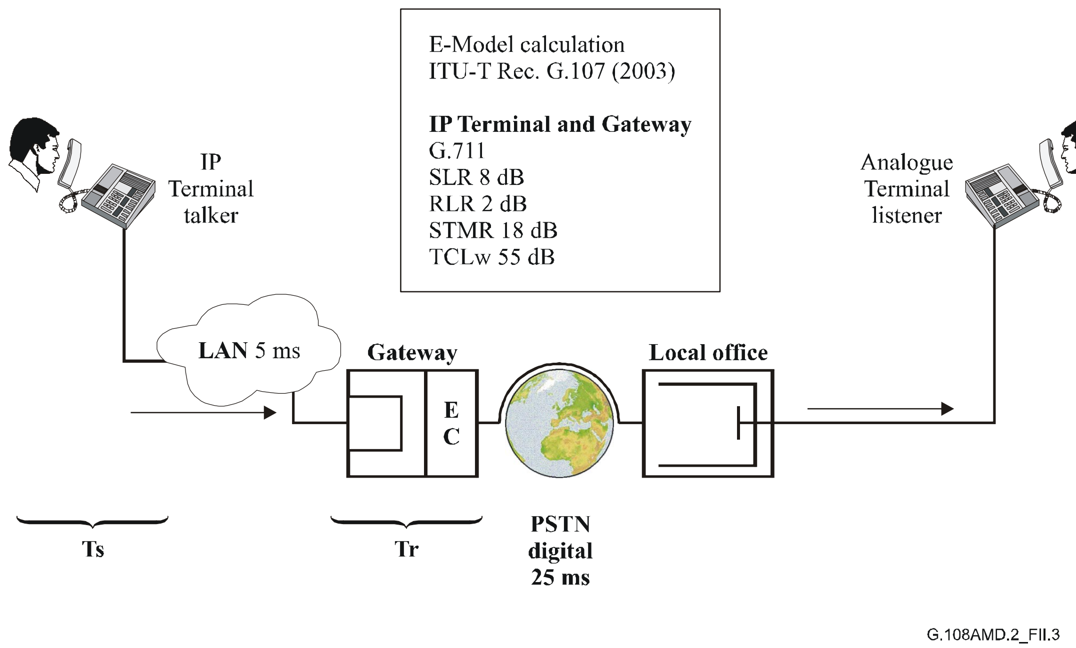

Figure II.3/G.108 – VoIP terminal with G.711 and PSTN with analogue terminal

For the E-model calculations, three different cases of terminal delay (and gateway delay) have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms
  VoIP gateway send delay = 20 ms, VoIP gateway receive delay = 30 ms
  total delay = Ts + Tlan + Tr + Tpstn = (20 + 5 + 30 + 25) ms = 80 ms
  R = 91: Users very satisfied
- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms
  VoIP gateway send delay = 35 ms, VoIP gateway receive delay = 65 ms
  total delay = Ts + Tlan + Tr + Tpstn = (35 + 5 + 65 + 25) ms = 130 ms
  R = 90: Users very satisfied
- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms
  VoIP gateway send delay = 50 ms, VoIP gateway receive delay = 100 ms
  total delay = Ts + Tlan + Tr + Tpstn = (50 + 5 + 100 + 25) ms = 180 ms
  R = 88: Users satisfied

The quality as perceived by the user in these example calculations is identical for both sides, because the example is symmetric in its behaviour.

The scenario in Figure II.4 describes a connection between two VoIP terminals via a packet-based network with a total network delay of 100 ms. The network delay is composed of the fixed delay and the value of the delay variation (jitter). It should be noted that, in this example, the packet-based network performs better than the maximum values set out in ITU-T Rec. Y.1541. In addition, low bit-rate coding as per ITU-T Rec. G.729A is deployed; thus, Ie = 11.

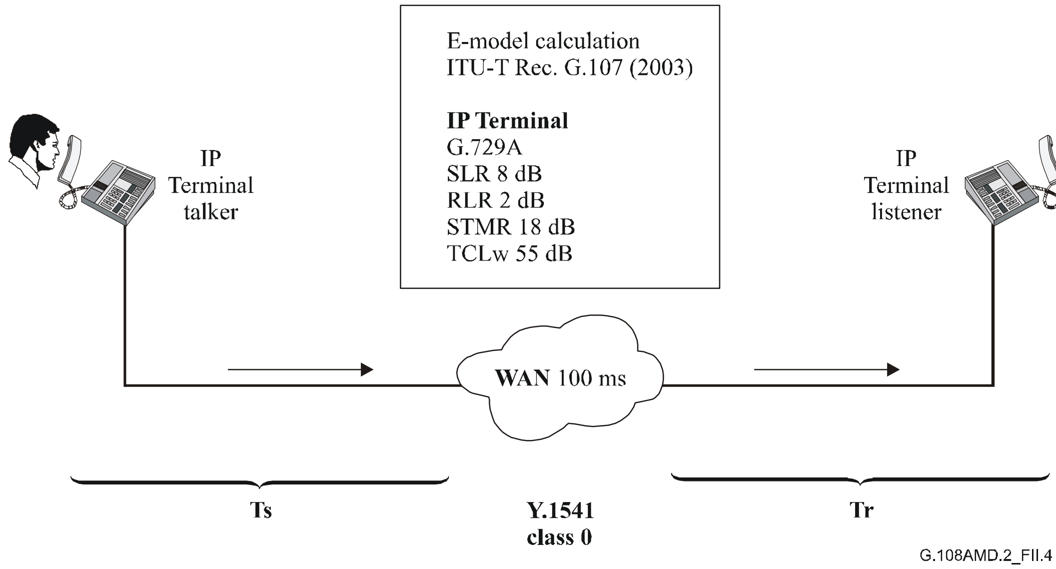

Figure II.4/G.108 – VoIP terminals with G.729A and WAN according to Y.1541 Class 0

For the E-model calculations, three different cases of terminal delay have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms
  total delay = Ts + Twan + Tr = (20 + 100 + 30) ms = 150 ms
  R = 79: Some users dissatisfied
- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms
  total delay = Ts + Twan + Tr = (35 + 100 + 65) ms = 200 ms
  R = 75: Some users dissatisfied
- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms
  total delay = Ts + Twan + Tr = (50 + 100 + 100) ms = 250 ms
  R = 68: Many users dissatisfied

The quality as perceived by the user in these example calculations is identical for both sides, because the example is symmetrically composed.

The scenario in Figure II.5 describes a connection between two VoIP islands via the PSTN with a delay in the PSTN of 125 ms. The delays in the LANs are 5 ms each which are composed of the fixed delay and the value of the delay variation (jitter). In addition, low bit-rate coding as per ITU-T Rec. G.729A is deployed in both VoIP islands; thus, Ie = 2 × 11 = 22.

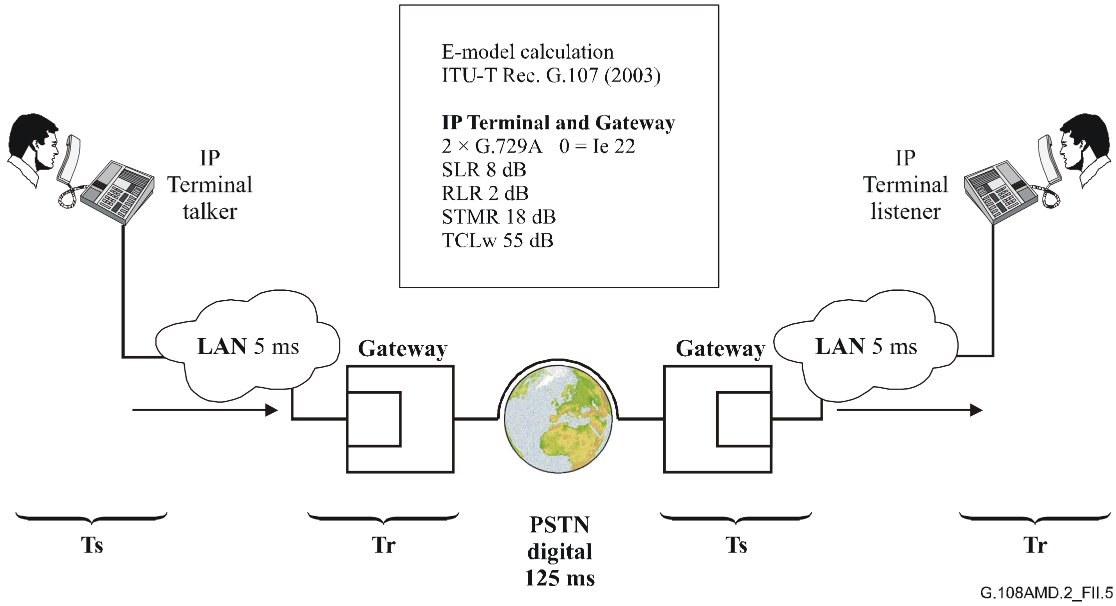

Figure II.5/G.108 – VoIP islands, each with G.729A, and PSTN

For the E-model calculations, three different cases of terminal delay (and gateway delay) have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms

  VoIP gateway send delay = 20 ms, VoIP gateway receive delay = 30 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (20 + 5 + 30 + 125 + 20 + 5 + 30) ms = 235 ms

  R = 59: Nearly all users dissatisfied
- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms

  VoIP gateway send delay = 35 ms, VoIP gateway receive delay = 65 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (35 + 5 + 65 + 125 + 35 + 5 + 65) ms = 335 ms

  R = 47: Not recommended
- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms

  VoIP gateway send delay = 50 ms, VoIP gateway receive delay = 100 ms

  total delay = Ts + Tlan + Tr + Tpstn + Ts + Tlan + Tr = (50 + 5 + 100 + 125 + 50 + 5 + 100) ms = 435 ms

  R = 37: Not recommended

The quality as perceived by the user in these example calculations is identical for both sides, because the example is symmetrically composed.

The scenario in Figure II.6 describes a connection between one VoIP island and an analogue terminal connected to the PSTN with a delay in the PSTN of 25 ms. The delay in the LAN is 5 ms which is composed of the fixed delay and the value of the delay variation (jitter). In addition, the VoIP gateway provides an echo canceller according to ITU-T Rec. G.168 with sufficient tail delay covering the echo path via the local office. Furthermore, low bit-rate coding as per ITU-T Rec. G.729A is deployed; thus, Ie = 11.

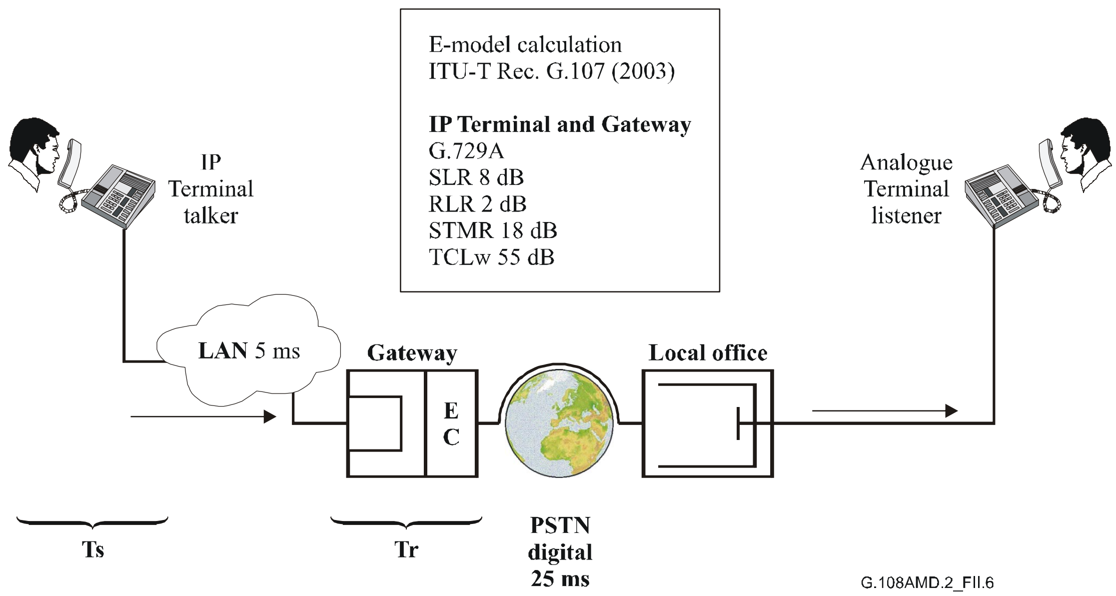

Figure II.6/G.108 – VoIP terminal with G.729A and PSTN with analogue terminal

For the E-model calculations, three different cases of terminal delay (and gateway delay) have been investigated:

- VoIP terminal send delay = 20 ms, VoIP terminal receive delay = 30 ms

  VoIP gateway send delay = 20 ms, VoIP gateway receive delay = 30 ms

  total delay = Ts + Tlan + Tr + Tpstn = (20 + 5 + 30 + 25) ms = 80 ms

  R = 80: Users satisfied
- VoIP terminal send delay = 35 ms, VoIP terminal receive delay = 65 ms

  VoIP gateway send delay = 35 ms, VoIP gateway receive delay = 65 ms

  total delay = Ts + Tlan + Tr + Tpstn = (35 + 5 + 65 + 25) ms = 130 ms

  R = 79: Some users dissatisfied
- VoIP terminal send delay = 50 ms, VoIP terminal receive delay = 100 ms

  VoIP gateway send delay = 50 ms, VoIP gateway receive delay = 100 ms

  total delay = Ts + Tlan + Tr + Tpstn = (50 + 5 + 100 + 25) ms = 180 ms

  R = 77: Some users dissatisfied

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