International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.1040
Amendment 1
(10/2007)

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

Quality of service and performance – Generic and user-related aspects

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Network contribution to transaction time

Amendment 1: New Appendix I – Network contribution to SIP set-up time

ITU-T Recommendation G.1040 (2006) – Amendment 1

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ITU-T Recommendation G.1040

Network contribution to transaction time

Amendment 1

New Appendix I – Network contribution to SIP set-up time

Summary

ITU-T Recommendation G.1040 defines a performance metric for typical data transactions, called the network contribution to transaction time (NCTT). Session initiation protocol (SIP) produces data transactions that resemble a typical call signalling flow between an originating and terminating station, and sometimes several other entities located within the service provider's network. Appendix I to ITU-T Recommendation G.1040 describes how NCTT can be adapted for use with SIP session flows to estimate the network contribution to session set-up time.

Source

Amendment 1 to ITU-T Recommendation G.1040 (2006) was agreed on 11 October 2007 by ITU-T Study Group 12 (2005-2008).

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). 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, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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I.1 Introduction

Session initiation protocol (SIP) packet exchanges are data transactions that resemble a typical call signalling flow between an originating and terminating station. Sometimes these exchanges include several other entities located within the service provider's network. This appendix describes how NCTT calculations can be adapted for use with SIP session flows to estimate the network contribution to session set-up time.

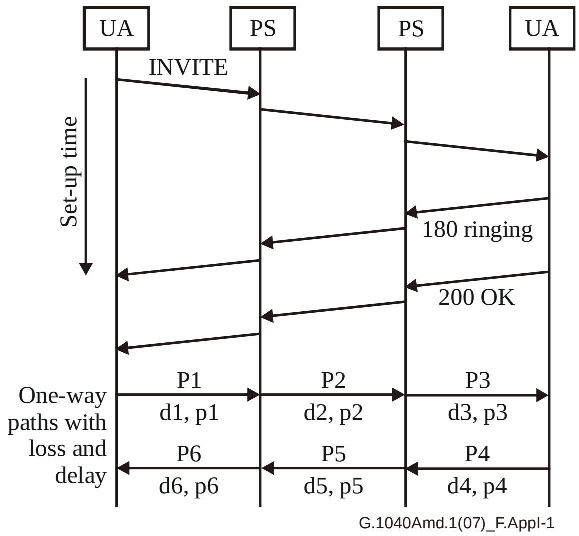

Figure I.1 – Typical SIP session flow with proxy servers (PS)

I.2 SIP set-up compared with a data transaction

One-way packet transfers: A typical and generic data transaction is illustrated in Figure 2/G.1040, where a client is communicating with a host across one or more IP networks. The main difference between a SIP set-up transaction and Figure 2/G.1040 is that the INVITE and other SIP messages proceed to their intended destination and the "desired reply" cannot arrive until the INVITE (and its corresponding reply) has proceeded successfully to all devices in the signalling path. There may be some provisional responses conveyed, but these are not the "desired reply" of Ringing or other state information. The figure above illustrates a SIP set-up between two user agents (UA) with two proxy servers (PS) in the signalling path.

Links in the signalling path have different characteristics: The typical transaction in ITU-T Rec. G.1040 assumes point-to-point communication between client and host over a single stable path with the same delay and loss characteristics for every exchange of the transaction. SIP messages may use several different paths as they travel from user agent to proxy server, proxy to proxy, and proxy to another UA (paths P1, P2, …, P6 in Figure I.1). It is likely that both the average one-way delay and loss will be different on the path between each signalling entity.

Time-out assessed at the originating agent: The UA that originates the INVITE waits to receive some definitive reply, and then re-starts the entire set-up process by sending another INVITE. In the typical data transaction illustrated in Figure 3/G.1040, any exchanges completed prior to a time-out are effective in progressing the transaction. With SIP however, failure at some point in the set-up results in the progress of the INVITE being discounted, and the set-up must be attempted again from the start. Note however that the time for the set-up (contributed by the network) remains the sum of time-outs plus the time for the INVITE and reply to traverse the network. In other words, equations for NCTT still hold for either case.

Variable time-out with exponential back-off: We have assumed a constant time-out value (RTO) for our typical data transaction in ITU-T Rec. G.1040. SIP conforms to the IETF's mantra that all protocol designs be congestion-aware. The default initial time-out value is 500 ms, and this value is doubled on every successive attempt to set up a session.

I.3 Revised equations for SIP set-up

One approach to adapting the NCTT calculations for SIP is to treat the multiple link exchange as a single transaction.

In the example UA-UA signalling path with two proxy servers as shown above, the network contribution to the average loss-less set-up time would be:

$$\text{Average NCTT}_{\text{no loss}} = \text{d1} + \text{d2} + \text{d3} + \text{d4} + \text{d5} + \text{d6} = \text{RTT}$$

where d1 through d6 are the average delays for the six one-way paths. This is the same quantity as round-trip time (RTT) calculated in clause 4.3/G.1040.

The overall set-up loss probability can be calculated from the individual loss probabilities (p1, p2, …, p6) for each path between signalling entities:

$$p = 1 - \{(1 - p1) \times (1 - p2) \times (1 - p3) \times (1 - p4) \times (1 - p5) \times (1 - p6)\}$$

With the number of exchanges, E = 1, the Expected number of losses, L, reduces to:

$$E\{L\} = \frac{p}{(1 - p)}$$

A weighted average time-out can be computed by calculating the probabilities of L = 1, 2, 3, etc., and construct various ratios that result in different weights for each of the different time-out values. Several schemes to determine a set of weights have been examined. However, for any reasonably small value of p, such as p = 0.001, the weight of the default time-out is much larger than all others. Therefore, a good approximation is to use the default initial time-out value as the RTO under this condition.

With all components now known, it is possible to calculate the average network contribution to SIP set-up time, as:

$$\text{Average}(NCSIP_{set-up}) = \text{Average}(NCTT) = (1 \times RTT) + (E\{L\} \times RTO)$$

I.4 Additional measurement considerations for SIP set-up

Clause 5/G.1040 gives general measurement considerations for data transactions. Additional considerations for measuring network contribution to SIP set-up include:

- One-way measurements are assumed above, but it may be possible to use round-trip measurements of each path, as long as both directions of transmission follow the same path that the signalling packets will take.
- Packet size: Some SIP INVITEs may be large enough that they are fragmented at the IP layer. Measurement packet size should take this into account.

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