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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 model for evaluating multimedia transmission performance over Internet Protocol

ITU-T Recommendation G.1050

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# ITU-T Recommendation G.1050

## Network model for evaluating multimedia transmission performance over Internet Protocol


### Summary

This Recommendation describes a model for evaluating multimedia transmission performance over an IP network. It is a statistical model in which likelihood of occurrence values are assigned to all network elements and impairments. Test results using these statistical models are expressed in terms of network model coverage. These results are unconditional – not dependent on the *a priori* specification of any network elements or impairments. Testing to a comprehensive statistical model suggests how communications devices may perform over an IP network in terms of network model coverage. This Recommendation focuses on the impact of impairments on layer 3 performance. IP streams from any type of network device can be evaluated using this model.

Emphasis is given to the fact that manufacturers of communications equipment and service providers are interested in a specification that accurately models the IP network characteristics that determine performance. Evaluators desire a definitive set of simple tests that properly measure the performance of communications devices from various manufacturers. Therefore, the objective of this Recommendation is to define a technology-independent model that is representative of the IP network, that can be simulated at reasonable complexity and that facilitates practical evaluation times.



### Source

ITU-T Recommendation G.1050 was approved on 29 November 2005 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

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## CONTENTS

## Introduction

Previous network transmission model standards for evaluating modem performance (see bibliography) have been statistical models in which likelihood of occurrence (LOO) values were assigned to all network elements and impairments. Test results using these statistical models were expressed in terms of network model coverage (NMC). These NMC results were unconditional – not dependent on the *a priori* specification of any network elements or impairments. This is an example of a statistical model. Testing to a comprehensive statistical model suggests how communications devices may perform over an IP network in terms of network model coverage.

Unlike the previous models, which focused on physical-layer impairments, this Recommendation focuses on the impact of impairments on layer 3 performance. IP streams from any type of network device can be evaluated using this model.

Emphasis is given to the fact that manufacturers of communications equipment and service providers are interested in a specification that accurately models the IP network characteristics that determine performance. Evaluators desire a definitive set of simple tests that properly measure the performance of communications devices from various manufacturers. Therefore, the objective of this standard is to define a technology-independent model that is representative of the IP network, that can be simulated at reasonable complexity and that facilitates practical evaluation times. The IP network model presented herein represents a snapshot of actual network data provided by anonymous IP service providers and IP network equipment manufacturers in the 2005 time-frame and will continue to evolve as more statistical information becomes available and as the IP network evolves.

In developing this model, certain assumptions have been made based on the best available statistical information. These assumptions are given in Appendix I.

The model described in this Recommendation is technology-independent and accommodates various IP communications networks and devices.

The following are parameters and impairments that affect quality of service and IP network performance:

- Network architecture;
- Types of access links;
- QoS controlled edge routing;
- Coding algorithms;
- A/D and D/A conversion;
- MTU size;
- Signalling protocol mismatches;
- Network faults;
- Link failure;
- Time drift;
- Route flapping;
- Reordered packets;
- Packet loss (frame loss);
- One-way delay (latency);
- Variable delays (jitter); and
- Background traffic (congestion, bandwidth, utilization, network load, load sharing).

# ITU-T Recommendation G.1050

## Network model for evaluating multimedia transmission performance over Internet Protocol

### 1 Scope

This Recommendation specifies an IP network model and scenarios for evaluating and comparing communications equipment connected over a converged wide-area network. The IP network model consists of many impairment combinations that are scenario-based and time-varying. IP streams from any type of network device can be evaluated using this model. The test scenarios combine LAN, access, and core network elements in a realistic way to create layer 3 IP network impairments that cause packets to experience varying delay or loss. These scenarios are based on actual network data provided by anonymous IP service providers and IP network equipment manufacturers.

Examples of the types of equipment that can be evaluated using this model include:

- IP-connected endpoints:
  - IP network devices (such as: user agents, call agents, media servers, media gateway controllers, gatekeepers, application servers, edge routers, etc.);
  - IP video;
  - IP phones;
  - IAF (Internet-aware fax).
- PSTN-connected devices through IP gateways:
  - POTS through voice-over-IP (VoIP) gateways;
  - T.38 facsimile devices and gateways;
  - V.150.1 and V.152 (voiceband data, VBD) modem-over-IP gateways;
  - V.151 textphone-over-IP gateways.

Figure 1 illustrates these devices and their interconnection through an IP network.

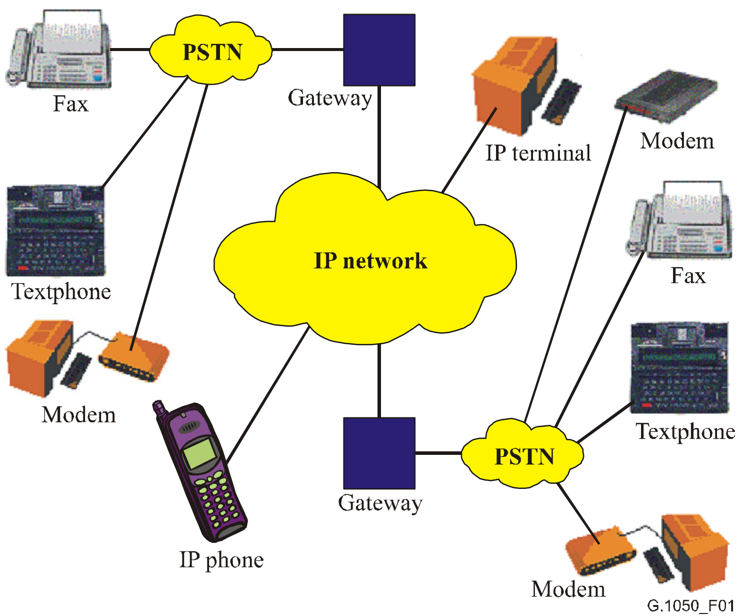


**Figure 1/G.1050 – Network model for evaluating multimedia transmission performances**

The models include parameters that can be used to configure and set up suitable emulator equipment.

This Recommendation includes mandatory requirements, recommendations and options; these are designated by the words "shall", "should", and "may", respectively.

**Limitations of this model**:

- This IP network model is not intended to represent any specific IP network. Rather, it provides a range of test scenarios that could represent a wide range of IP network characteristics, such as those experienced in well-managed (QoS-managed), partially-managed (non-QoS) and unmanaged (Internet) networks.
- Some networks may utilize PSTN at one or both ends of the connection through a media gateway. This model only addresses the IP portion of the network and does not address the PSTN portion of the end-to-end connection.
- The network models represented in this Recommendation do not model all possible connections that can be encountered between devices.
- The IP network model presented herein is based on an informal survey of anonymous IP service providers and IP network equipment manufacturers in the 2005 time-frame and will continue to evolve as more statistical information becomes available and as the IP network evolves.

## 2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.


– ITU-T Recommendation G.107 (2005), *The E-model, a computational model for use in transmission planning*.

– ITU-T Recommendation G.108 (1999), *Application of the E-model: A planning guide*.

– ITU-T Recommendation G.114 (2003), *One-way transmission time*.

– ITU-T Recommendation T.38 (2005), *Procedures for real-time Group 3 facsimile communication over IP networks*.

– ITU-T Recommendation V.150.0 (2003), *Modem-over-IP networks: Foundation*.

– ITU-T Recommendation V.150.1 (2003), *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.

– ITU-T Recommendation V.152 (2005), *Procedures for supporting voice-band data over IP networks*.

– ITU-T Recommendation Y.1541 (2002), *Network performance objectives for IP-based services*.

## 3 Terms and definitions

This Recommendation defines the following terms:

**3.1 burst loss**: A high density of packet loss over time, or loss of consecutive packets, due to congestion, bandwidth limitation or re-routing (delay translated into loss due to implementation) on the network.

**3.2 codec**: An acronym for coder/decoder that combines analogue-to-digital conversion and digital-to-analogue conversion functions.

**3.3 delay**: The time required for a packet to traverse the network or a segment of the network. See latency.

**3.4 downstream**: A transmission from a service provider toward an end user.

**3.5 end-to-end network**: Pertaining to an entire path from one endpoint to another. Metrics may refer to a single segment (example: core delay) or to the entire path (example: end-to-end network delay).

**3.6 E-model**: A standard-based (ITU-T Recs G.107 and G.108) model for planning the transmission quality of telephone networks. The output of the E-model is a transmission rating factor called the R-Factor. The scale for the R-Factor is between 0 and 100, where 0 is low and 100 is high transmission quality.

**3.7 gateway**: A network device that acts as an entrance to another network. One function is to convert media provided in one type of network to the format required in another type of network. For example, a gateway could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network).

**3.8 IP network**: A network based on the Internet Protocol, a connectionless protocol.

**3.9 jitter**: Variation in packet delay.

**3.10 jitter buffer**: A shared data area where packets can be collected, stored, and sent to the processor in evenly spaced intervals to improve the end-user experience.

**3.11 latency**: An expression of how much time it takes for a packet of data to get from one designated point to another. See delay.

**3.12 layer 3**: The third layer of the International Organization for Standardization (ISO) open systems interconnection (OSI) model, known as the network layer. IP is a layer 3 protocol.

**3.13 link failure**: A period of consecutive packet loss that can last for several seconds or, sometimes, minutes. The network model simulates the effect of link failure in the core segment by dropping consecutive packets for the duration of the link failure.

**3.14 link failure interval**: The interval between link failures.

**3.15 likelihood of occurrence (LOO)**: A normalized estimated probability, expressed in percent, that a particular impairment combination occurs in the IP network.

**3.16 mean opinion score conversational quality (MOS-CQ)**: A measure of the quality of a connection that characterizes how users rate the overall quality of a call based on listening quality and their ability to converse during a call. This includes any echo- or delay-related difficulties that may affect the conversation. Parameters are in the range from one to five.

**3.17 mean opinion score listening quality (MOS-LQ)**: A measure of the quality of a connection that characterizes how users rate what they "hear" during a call. Parameters are in the range from one to five.

**3.18 MTU size**: The largest size packet or frame, specified in octets, that can be sent in a packet- or frame-based network such as the Internet.

**3.19 NMC score**: A value used in the NMC curve. A score is calculated by multiplying the LOO for the LAN rate combinations by the LOO for the severity. The total score adds up to 100% for each severity (A, B, C). Score = $LOO_{LAN/access} \times LOO_{Severity}$.

**3.20 occupancy**: Background traffic on a LAN, including congestion from collisions, that is not part of the user signal being evaluated.

**3.21 packet loss**: The failure of a packet to traverse the network to its destination. (This model does not take into account discards due to buffer overflow).

**3.22 packet loss concealment**: A method of hiding the fact that media packets were lost by generating synthetic packets.

**3.23 peak jitter**: The maximum variation of delay from the mean delay.

**3.24 peak-to-peak jitter**: The full range of packet delay from the maximum amount to the minimum amount.

**3.25 QoS edge routing**: Routing between the customer premises network and the service provider network based on quality of service classification values.

**3.26 R-factor call quality (R-CQ)**: An R-factor measurement that characterizes how users rate the overall quality of a call based on listening quality and their ability to converse during a call. This includes any echo- or delay-related difficulties that may affect the conversation.

**3.27 R-factor**: An objective measure of transmission quality of telephone networks based on the E-model described in ITU-T Recs G.107 and G.108. The scale for the R-factor is between 0 and 100, where 0 is low and 100 is high transmission quality.

**3.28 R-factor listening quality (R-LQ)**: An R-factor measurement that characterizes how users rate what they "hear" during a call.

**3.29 reordered packets**: A packet that arrives at the destination with a packet sequence number that is smaller than the previous packet.

**3.30 route flap**: Repeated changes in a path due to updates to a routing table. The network model simulates the effect of route flaps by making incremental changes in the delay values of the core segment.

**3.31 total delay**: The cumulative delay for all segments in a connection.

**3.32 upstream**: A transmission from an end user toward a service provider.

## 4 Abbreviations

This Recommendation uses the following abbreviations:

| | |
|---|---|
| ADSL | Asymmetric Digital Subscriber Line |
| CSMA/CD | Carrier Sense Multiple Access/Collision Detection |
| IP | Internet Protocol |
| ISDN | Integrated Services Digital Network |
| LAN | Local Area Network |
| LOO | Likelihood Of Occurrence |
| MOS | Mean Opinion Score |
| MTU | Maximum Transmission Unit |
| NMC | Network Model Coverage |

| | |
|---|---|
| OSI | Open Systems Interconnection |
| PESQ | Perceptual Evaluation of Speech Quality |
| PLC | Packet Loss Concealment |
| POTS | Plain Old Telephone Service |
| PSQM | Perceptual Speech Quality Measurement |
| PSTN | Public Switched Telephone Network |
| QoS | Quality of Service |
| RF | Radio Frequency |
| SDSL | Symmetric Digital Subscriber Line |
| SLA | Service Level Agreement |
| VoIP | Voice over Internet Protocol |
| VTC | Video Teleconferencing |

## 5 Description of the model

The IP network model consists of many impairment combinations that are scenario-based, time varying IP network impairments that provide a significant sample of impairment conditions. Tests using this model may be unidirectional or bidirectional. Impairments occur in both directions. Since the access links may be asymmetrical in nature and packets travelling in one direction will encounter sections of the model in a different order than packets travelling in the other direction, the impairments in each direction may differ. Figure 2 shows the network parameters and impairments that apply to each section of the model:

A-side parameters:

LAN A rate and type, LAN A occupancy, local access A rates in each direction, access A occupancy, MTU size.

Core parameters:

Route flapping interval, route flapping delay change, link failure interval, link failure duration, one-way delay, jitter, reordered packets, and packet loss.

B-side parameters:

LAN B rate and type, LAN B occupancy, local access B rates in each direction, access B occupancy, MTU size.

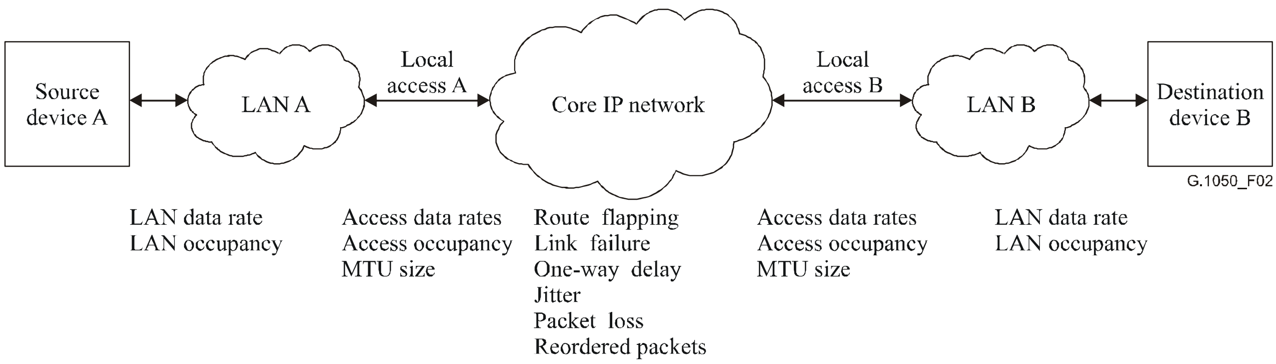


**Figure 2/G.1050 – IP network impairment model**

Appendix I provides the rationale for the network parameters and impairments for the IP network model.

Appendix II specifies algorithms for computing the delay, packet reordering and packet loss as a result of the network parameters and impairments at each section of the model.

IP streams from any type of network device can be evaluated using the IP network model and will yield results corresponding to the type of device or application under evaluation.

The tests are intended to allow completion of a full set of testing within 36 hours (with a test run of 2 minutes per test case) or less depending on the type of test that is being run. The test methodology easily lends itself to automation. The unit under test (UUT) is run over each impairment combination. This approach can be viewed as running over many individual IP nodes with a wide range of impairments.

Items outside the model that affect the end-to-end delay, jitter, and application quality include:

- Packet size.
- Source packet generation rate – assumed isochronous stream.
- Compression algorithms.
- Packet loss concealment algorithms.
- Jitter buffer type and size.
- Forward error correction.
- QoS edge routing.
- Voice activity detection.

## 6 IP impairment level set-up

### 6.1 Service test profiles

Table 1 describes service test profiles and the applications, node mechanisms and network techniques associated with them. ITU-T Rec. Y.1541 uses a similar approach, but a one-to-one mapping to these service profiles may not be possible.

**Table 1/G.1050 – Service test profiles**

| Service test profiles | Applications (examples) | Node mechanisms | Network techniques |
|---|---|---|---|
| Well-managed IP network (profile A) | High quality video and VoIP, VTC (real-time applications, loss sensitive, jitter sensitive, high interaction) | Strict QoS, guaranteed no over-subscription on links | Constrained routing and distance |
| Partially-managed IP network (profile B) | VoIP, VTC (real-time applications, jitter sensitive, interactive) | Separate queue with preferential servicing, traffic grooming | Less constrained routing and distances |

**Table 1/G.1050 – Service test profiles**

| Service test profiles | Applications (examples) | Node mechanisms | Network techniques |
|---|---|---|---|
| Unmanaged IP network, Internet (profile C) | Lower quality video and VoIP, signalling, transaction data (highly interactive) | Separate queue (drop priority) | Constrained routing and distance |
| | Transaction data, interactive | | Less constrained routing and distances |
| | Short transactions, bulk data (low loss) | Long queue (drop priority) | Any route/path |
| | Traditional Internet applications (default IP networks) | Separate queue (lowest priority) | Any route/path |

### 6.2 Network impairments

#### 6.2.1 Service test profiles

The following three test profiles are used in this IP network model that can be associated with service level agreements (SLA):

- Well-managed network (profile A) – a network with no over-committed links that employs QoS edge routing.
- Partially-managed network (profile B) – a network that minimizes over-committed links and has one or more links without QoS edge routing.
- Unmanaged network (profile C) – an unmanaged network such as the Internet that includes over-committed links and has one or more links without QoS edge routing.

These tables represent end-to-end impairment levels, including LAN and access. In Tables 2, 3 and 4, the total packet loss is the sum of the sequential packet loss and random packet loss. Note that service provider SLAs only guarantee characteristics of the core section of the network.

**Table 2/G.1050 – Impairment ranges for well-managed network (profile A)**

| Impairment type | Units | Range (min to max) |
|---|---|---|
| One-way latency | ms | 20 to 100 (regional)<br>90 to 300 (intercontinental) |
| Jitter (peak-to-peak) | ms | 0 to 50 |
| Sequential packet loss | ms | Random loss only (except when link failure occurs) |
| Rate of sequential loss | $sec^{-1}$ | Random loss only (except when link failure occurs) |
| Random packet loss | % | 0 to 0.05 |
| Reordered packets | % | 0 to 0.001 |

**Table 3/G.1050 – Impairment ranges for partially-managed network (profile B)**

| Impairment type | Units | Range (min to max) |
|---|---|---|
| One-way latency | ms | 50 to 100 (regional)<br>90 to 400 (intercontinental) |
| Jitter (peak-to-peak) | ms | 0 to 150 |
| Sequential packet loss | ms | 40 to 200 |
| Rate of sequential loss | $sec^{-1}$ | $\leq 10^{-3}$ (Note) |
| Random packet loss | % | 0 to 2 |
| Reordered packets | % | 0 to 0.01 |
| NOTE – Sequential packet loss occurs once every 1000 seconds. | | |

**Table 4/G.1050 – Impairment ranges for unmanaged network (profile C)** (Note 1)

| Impairment type | Units | Range (min to max) |
|---|---|---|
| One-way latency | ms | 50 to 500 |
| Jitter (peak-to-peak) | ms | 0 to 500 |
| Sequential packet loss | ms | 40 to 10 000 |
| Rate of sequential loss | $sec^{-1}$ | $\leq 10^{-1}$ (Note 2) |
| Random packet loss | % | 0 to 20 |
| Reordered packets | % | 0 to 0.1 |
| NOTE 1 – This table represents levels for a normally operating unmanaged network. Impairment levels for impairment condition H may exceed the ranges in this table to account for disaster conditions.<br>NOTE 2 – Sequential packet loss occurs once every 10 seconds. | | |

## 6.3 Test set-up

(See Figure 3.)

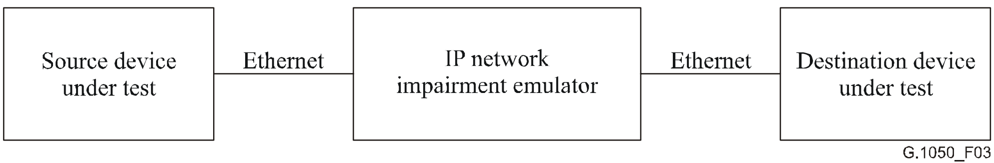


**Figure 3/G.1050 – Simulator set-up block diagram**

## 6.4 Impairment combination tables

Each test case consists of a complete set of parameters and impairments. The LAN and access rates at each end of the connection comprise the first of these parameters. These rates indicate effective rates and vary depending on multiple factors including distance from the central office, over-subscription, service offerings, number of concurrent users, physical plant impairments and other factors.

Tables 5 through 8 list typical rates for home and business locations rather than service offering rates.

The following location combinations are possible:

- Home to home
- Home to business
- Business to home
- Business to business

**Table 5/G.1050 – LAN rates for home locations**

| Effective LAN rate (Mbit/s) | LOO (%) | Represents |
|---|---|---|
| 4 | 75 | 802.11b, 10BaseT hub |
| 20 | 25 | 802.11g, 100BaseT hub |

**Table 6/G.1050 – LAN rates for business locations**

| Effective LAN rate (Mbit/s) | LOO (%) | Represents |
|---|---|---|
| 4 | 20 | 802.11b, 10BaseT |
| 20 | 20 | 802.11g, 100BaseT hub |
| 100 | 60 | 100BaseT switched, Gbit Ethernet |

**Table 7/G.1050 – Access rates for home locations**

| Access rate | | LOO (%) | Represents |
|---|---|---|---|
| Toward core (kbit/s) | From core (kbit/s) | | |
| 128 | 768 | 40 | ADSL |
| 384 | 1536 | 50 | Cable, ADSL |
| 384 | 3000 | 10 | Cable, ADSL |

**Table 8/G.1050 – Access rates for business locations**

| Access rate | | LOO (%) | Represents |
|---|---|---|---|
| Toward core (kbit/s) | From core (kbit/s) | | |
| 384 | 1536 | 40 | ADSL entry |
| 384 | 3000 | 15 | ADSL premium |
| 1 536 | 1 536 | 40 | T1 |
| 43 000 | 43 000 | 5 | T3 |

This model weights the prevalence of home and business LAN and access rates equally.

Table 9, the overall rate combination table, consists of all possible combinations of locations, LAN rates and access rates. The LOO column for each combination is calculated by multiplying the LOO for the LAN rate by the LOO for the access rate. Where duplicate rate combinations (or mirror image combinations) occur, they are combined and their associated LOO added together. This results in 133 unique rate combinations with LOO values that add to 100%.

**Table 9/G.1050 – LAN and access rate combinations**

| **Test Case #** | **LAN A rate** (Mbit/s) | **A->B access rate at A** (kbit/s) | **A->B access rate at B** (kbit/s) | **LAN B rate** (Mbit/s) | **B->A access rate at B** (kbit/s) | **B->A access rate at A** (kbit/s) | **LOO** (%) |
|---|---|---|---|---|---|---|---|
| 1 | 4 | 128 | 768 | 4 | 128 | 768 | 2.2500 |
| 2 | 4 | 128 | 768 | 20 | 128 | 768 | 1.5000 |
| 3 | 20 | 128 | 768 | 20 | 128 | 768 | 0.2500 |
| 4 | 4 | 128 | 1 536 | 4 | 384 | 768 | 3.4125 |
| 5 | 4 | 128 | 1 536 | 20 | 384 | 768 | 2.6750 |
| 6 | 20 | 128 | 1 536 | 20 | 384 | 768 | 0.5125 |
| 7 | 4 | 128 | 3 000 | 4 | 384 | 768 | 0.7875 |
| 8 | 4 | 128 | 3 000 | 20 | 384 | 768 | 0.6750 |
| 9 | 20 | 128 | 3 000 | 20 | 384 | 768 | 0.1375 |
| 10 | 4 | 384 | 768 | 4 | 128 | 1 536 | 3.4125 |
| 11 | 4 | 384 | 768 | 20 | 128 | 1 536 | 2.6750 |
| 12 | 20 | 384 | 768 | 20 | 128 | 1 536 | 0.5125 |
| 13 | 4 | 384 | 1 536 | 4 | 384 | 1 536 | 5.1756 |
| 14 | 4 | 384 | 1 536 | 20 | 384 | 1 536 | 4.6638 |
| 15 | 20 | 384 | 1 536 | 20 | 384 | 1 536 | 1.0506 |
| 16 | 4 | 384 | 3 000 | 4 | 384 | 1 536 | 1.1944 |
| 17 | 4 | 384 | 3 000 | 20 | 384 | 1 536 | 1.1638 |
| 18 | 20 | 384 | 3 000 | 20 | 384 | 1 536 | 0.2819 |
| 19 | 4 | 384 | 768 | 4 | 128 | 3 000 | 0.7875 |
| 20 | 4 | 384 | 768 | 20 | 128 | 3 000 | 0.6750 |
| 21 | 20 | 384 | 768 | 20 | 128 | 3 000 | 0.1375 |
| 22 | 4 | 384 | 1 536 | 4 | 384 | 3 000 | 1.1944 |
| 23 | 4 | 384 | 1 536 | 20 | 384 | 3 000 | 1.1638 |
| 24 | 20 | 384 | 1 536 | 20 | 384 | 3 000 | 0.2819 |
| 25 | 4 | 384 | 3 000 | 4 | 384 | 3 000 | 0.2756 |
| 26 | 4 | 384 | 3 000 | 20 | 384 | 3 000 | 0.2888 |
| 27 | 20 | 384 | 3 000 | 20 | 384 | 3 000 | 0.0756 |
| 28 | 4 | 128 | 1 536 | 100 | 384 | 768 | 1.8000 |
| 29 | 20 | 128 | 1 536 | 100 | 384 | 768 | 0.6000 |
| 30 | 4 | 128 | 3 000 | 100 | 384 | 768 | 0.6750 |
| 31 | 20 | 128 | 3 000 | 100 | 384 | 768 | 0.2250 |
| 32 | 4 | 128 | 1 536 | 4 | 768 | 1 536 | 0.6000 |

**Table 9/G.1050 – LAN and access rate combinations**

| Test Case # | LAN A rate (Mbit/s) | A->B access rate at A (kbit/s) | A->B access rate at B (kbit/s) | LAN B rate (Mbit/s) | B->A access rate at B (kbit/s) | B->A access rate at A (kbit/s) | LOO (%) |
|---|---|---|---|---|---|---|---|
| 33 | 4 | 128 | 1 536 | 20 | 768 | 1 536 | 0.8000 |
| 34 | 4 | 128 | 1 536 | 100 | 768 | 1 536 | 1.8000 |
| 35 | 20 | 128 | 1 536 | 20 | 768 | 1 536 | 0.2000 |
| 36 | 20 | 128 | 1 536 | 100 | 768 | 1 536 | 0.6000 |
| 37 | 4 | 128 | 43 000 | 4 | 768 | 43 000 | 0.0750 |
| 38 | 4 | 128 | 43 000 | 20 | 768 | 43 000 | 0.1000 |
| 39 | 4 | 128 | 43 000 | 100 | 768 | 43 000 | 0.2250 |
| 40 | 20 | 128 | 43 000 | 20 | 768 | 43 000 | 0.0250 |
| 41 | 20 | 128 | 43 000 | 100 | 768 | 43 000 | 0.0750 |
| 42 | 4 | 384 | 1 536 | 100 | 384 | 1 536 | 5.4600 |
| 43 | 20 | 384 | 1 536 | 100 | 384 | 1 536 | 2.4600 |
| 44 | 4 | 384 | 3 000 | 100 | 384 | 1 536 | 1.6538 |
| 45 | 20 | 384 | 3 000 | 100 | 384 | 1 536 | 0.7913 |
| 46 | 4 | 384 | 1 536 | 4 | 1 536 | 1 536 | 0.9100 |
| 47 | 4 | 384 | 1 536 | 20 | 1 536 | 1 536 | 1.3200 |
| 48 | 4 | 384 | 1 536 | 100 | 1 536 | 1 536 | 3.2100 |
| 49 | 20 | 384 | 1 536 | 20 | 1 536 | 1 536 | 0.4100 |
| 50 | 20 | 384 | 1 536 | 100 | 1 536 | 1 536 | 1.7100 |
| 51 | 4 | 384 | 43 000 | 4 | 1 536 | 43 000 | 0.1138 |
| 52 | 4 | 384 | 43 000 | 20 | 1 536 | 43 000 | 0.1650 |
| 53 | 4 | 384 | 43 000 | 100 | 1 536 | 43 000 | 0.4013 |
| 54 | 20 | 384 | 43 000 | 20 | 1 536 | 43 000 | 0.0513 |
| 55 | 20 | 384 | 43 000 | 100 | 1 536 | 43 000 | 0.2138 |
| 56 | 4 | 384 | 1 536 | 100 | 384 | 3 000 | 1.6538 |
| 57 | 20 | 384 | 1 536 | 100 | 384 | 3 000 | 0.7913 |
| 58 | 4 | 384 | 3 000 | 100 | 384 | 3 000 | 0.4725 |
| 59 | 20 | 384 | 3 000 | 100 | 384 | 3 000 | 0.2475 |
| 60 | 4 | 384 | 1 536 | 4 | 1 536 | 3 000 | 0.2100 |
| 61 | 4 | 384 | 1 536 | 20 | 1 536 | 3 000 | 0.3200 |
| 62 | 4 | 384 | 1 536 | 100 | 1 536 | 3 000 | 0.8100 |
| 63 | 20 | 384 | 1 536 | 20 | 1 536 | 3 000 | 0.1100 |
| 64 | 20 | 384 | 1 536 | 100 | 1 536 | 3 000 | 0.5100 |
| 65 | 4 | 384 | 43 000 | 4 | 3 000 | 43 000 | 0.0263 |
| 66 | 4 | 384 | 43 000 | 20 | 3 000 | 43 000 | 0.0400 |
| 67 | 4 | 384 | 43 000 | 100 | 3 000 | 43 000 | 0.1013 |
| 68 | 20 | 384 | 43 000 | 20 | 3 000 | 43 000 | 0.0138 |

**Table 9/G.1050 – LAN and access rate combinations**

| **Test Case** # | **LAN A rate** (Mbit/s) | **A->B access rate at A** (kbit/s) | **A->B access rate at B** (kbit/s) | **LAN B rate** (Mbit/s) | **B->A access rate at B** (kbit/s) | **B->A access rate at A** (kbit/s) | **LOO** (%) |
|---|---|---|---|---|---|---|---|
| 69 | 20 | 384 | 43 000 | 100 | 3 000 | 43 000 | 0.0638 |
| 70 | 4 | 384 | 768 | 100 | 128 | 1 536 | 1.8000 |
| 71 | 20 | 384 | 768 | 100 | 128 | 1 536 | 0.6000 |
| 72 | 4 | 384 | 768 | 100 | 128 | 3 000 | 0.6750 |
| 73 | 20 | 384 | 768 | 100 | 128 | 3 000 | 0.2250 |
| 74 | 4 | 768 | 1 536 | 4 | 128 | 1 536 | 0.6000 |
| 75 | 4 | 768 | 1 536 | 20 | 128 | 1 536 | 0.8000 |
| 76 | 20 | 768 | 1 536 | 20 | 128 | 1 536 | 0.2000 |
| 77 | 4 | 768 | 1 536 | 100 | 128 | 1 536 | 1.8000 |
| 78 | 20 | 768 | 1 536 | 100 | 128 | 1 536 | 0.6000 |
| 79 | 4 | 1 536 | 1 536 | 4 | 384 | 1 536 | 0.9100 |
| 80 | 4 | 1 536 | 1 536 | 20 | 384 | 1 536 | 1.3200 |
| 81 | 20 | 1 536 | 1 536 | 20 | 384 | 1 536 | 0.4100 |
| 82 | 4 | 1 536 | 1 536 | 100 | 384 | 1 536 | 3.2100 |
| 83 | 20 | 1 536 | 1 536 | 100 | 384 | 1 536 | 1.7100 |
| 84 | 4 | 1 536 | 3 000 | 4 | 384 | 1 536 | 0.2100 |
| 85 | 4 | 1 536 | 3 000 | 20 | 384 | 1 536 | 0.3200 |
| 86 | 20 | 1 536 | 3 000 | 20 | 384 | 1 536 | 0.1100 |
| 87 | 4 | 1 536 | 3 000 | 100 | 384 | 1 536 | 0.8100 |
| 88 | 20 | 1 536 | 3 000 | 100 | 384 | 1 536 | 0.5100 |
| 89 | 4 | 768 | 43 000 | 4 | 128 | 43 000 | 0.0750 |
| 90 | 4 | 768 | 43 000 | 20 | 128 | 43 000 | 0.1000 |
| 91 | 20 | 768 | 43 000 | 20 | 128 | 43 000 | 0.0250 |
| 92 | 4 | 768 | 43 000 | 100 | 128 | 43 000 | 0.2250 |
| 93 | 20 | 768 | 43 000 | 100 | 128 | 43 000 | 0.0750 |
| 94 | 4 | 1 536 | 43 000 | 4 | 384 | 43 000 | 0.1138 |
| 95 | 4 | 1 536 | 43 000 | 20 | 384 | 43 000 | 0.1650 |
| 96 | 20 | 1 536 | 43 000 | 20 | 384 | 43 000 | 0.0513 |
| 97 | 4 | 1 536 | 43 000 | 100 | 384 | 43 000 | 0.4013 |
| 98 | 20 | 1 536 | 43 000 | 100 | 384 | 43 000 | 0.2138 |
| 99 | 4 | 3 000 | 43 000 | 4 | 384 | 43 000 | 0.0263 |
| 100 | 4 | 3 000 | 43 000 | 20 | 384 | 43 000 | 0.0400 |
| 101 | 20 | 3 000 | 43 000 | 20 | 384 | 43 000 | 0.0138 |
| 102 | 4 | 3 000 | 43 000 | 100 | 384 | 43 000 | 0.1013 |
| 103 | 20 | 3 000 | 43 000 | 100 | 384 | 43 000 | 0.0638 |
| 104 | 100 | 384 | 1 536 | 100 | 384 | 1 536 | 1.4400 |

**Table 9/G.1050 – LAN and access rate combinations**

| Test Case # | LAN A rate (Mbit/s) | A->B access rate at A (kbit/s) | A->B access rate at B (kbit/s) | LAN B rate (Mbit/s) | B->A access rate at B (kbit/s) | B->A access rate at A (kbit/s) | LOO (%) |
|---|---|---|---|---|---|---|---|
| 105 | 100 | 384 | 3 000 | 100 | 384 | 1 536 | 0.5400 |
| 106 | 100 | 384 | 1 536 | 100 | 1 536 | 1 536 | 1.4400 |
| 107 | 100 | 384 | 43 000 | 100 | 1 536 | 43 000 | 0.1800 |
| 108 | 100 | 384 | 1 536 | 100 | 384 | 3 000 | 0.5400 |
| 109 | 100 | 384 | 3 000 | 100 | 384 | 3 000 | 0.2025 |
| 110 | 100 | 384 | 1 536 | 100 | 1 536 | 3 000 | 0.5400 |
| 111 | 100 | 384 | 43 000 | 100 | 3 000 | 43 000 | 0.0675 |
| 112 | 100 | 1 536 | 1 536 | 100 | 384 | 1 536 | 1.4400 |
| 113 | 100 | 1 536 | 3 000 | 100 | 384 | 1 536 | 0.5400 |
| 114 | 4 | 1 536 | 1 536 | 4 | 1 536 | 1 536 | 0.1600 |
| 115 | 4 | 1 536 | 1 536 | 20 | 1 536 | 1 536 | 0.3200 |
| 116 | 4 | 1 536 | 1 536 | 100 | 1 536 | 1 536 | 0.9600 |
| 117 | 20 | 1 536 | 1 536 | 20 | 1 536 | 1 536 | 0.1600 |
| 118 | 20 | 1 536 | 1 536 | 100 | 1 536 | 1 536 | 0.9600 |
| 119 | 100 | 1 536 | 1 536 | 100 | 1 536 | 1 536 | 1.4400 |
| 120 | 4 | 1 536 | 43 000 | 4 | 1 536 | 43 000 | 0.0400 |
| 121 | 4 | 1 536 | 43 000 | 20 | 1 536 | 43 000 | 0.0800 |
| 122 | 4 | 1 536 | 43 000 | 100 | 1 536 | 43 000 | 0.2400 |
| 123 | 20 | 1 536 | 43 000 | 20 | 1 536 | 43 000 | 0.0400 |
| 124 | 20 | 1 536 | 43 000 | 100 | 1 536 | 43 000 | 0.2400 |
| 125 | 100 | 1 536 | 43 000 | 100 | 1 536 | 43 000 | 0.3600 |
| 126 | 100 | 1 536 | 43 000 | 100 | 384 | 43 000 | 0.1800 |
| 127 | 100 | 3 000 | 43 000 | 100 | 384 | 43 000 | 0.0675 |
| 128 | 4 | 43 000 | 43 000 | 4 | 43 000 | 43 000 | 0.0025 |
| 129 | 4 | 43 000 | 43 000 | 20 | 43 000 | 43 000 | 0.0050 |
| 130 | 4 | 43 000 | 43 000 | 100 | 43 000 | 43 000 | 0.0150 |
| 131 | 20 | 43 000 | 43 000 | 20 | 43 000 | 43 000 | 0.0025 |
| 132 | 20 | 43 000 | 43 000 | 100 | 43 000 | 43 000 | 0.0150 |
| 133 | 100 | 43 000 | 43 000 | 100 | 43 000 | 43 000 | 0.0225 |

Table 10 lists eight severity levels (A through H). Each severity level consists of a combination of impairments from the source location, core network, and destination location. To minimize test time, the tester may choose to run a set of test cases associated with a particular SLA (profile A, B, or C) as described in 6.2.1. Refer to Appendix II for the precise usage of these parameters in the impairment algorithms.

**Table 10/G.1050 – Impairment severity combinations**

| Impairment | Severity=> Units | A | B | C | D | E | F | G | H* |
|---|---|---|---|---|---|---|---|---|---|
| **Profile A LOO** | % | 50 | 30 | 15 | 5 | 0 | 0 | 0 | 0 |
| **Profile B LOO** | % | 5 | 25 | 30 | 25 | 10 | 5 | 0 | 0 |
| **Profile C LOO** | % | 5 | 5 | 10 | 15 | 20 | 25 | 15 | 5 |
| **Source location (A) parameters** | | | | | | | | | |
| **LAN A occupancy** | % | 1 | 2 | 3 | 5 | 8 | 12 | 16 | 20 |
| **Access A occupancy** | % | 0 | 1 | 2 | 4 | 8 | 15 | 30 | 50 |
| **MTU A** | bytes | 512 | 512 | 1508 | 1508 | 1508 | 1508 | 1508 | 1508 |
| **Core network impairments** | | | | | | | | | |
| **Route flap interval** | seconds | 0 | 3600 | 1800 | 900 | 480 | 240 | 120 | 60 |
| **Route flap delay** | ms | 0 | 2 | 4 | 8 | 16 | 32 | 64 | 128 |
| **Delay (regional)** | ms | 4 | 8 | 16 | 32 | 64 | 128 | 256 | 512 |
| **Delay (intercontinental)** | ms | 16 | 32 | 64 | 128 | 196 | 256 | 512 | 768 |
| **Jitter (peak-to-peak)** | ms | 5 | 10 | 24 | 40 | 70 | 100 | 150 | 500 |
| **Link fail interval** | seconds | 0 | 3600 | 1800 | 900 | 480 | 240 | 120 | 60 |
| **Link fail duration** | ms | 0 | 64 | 128 | 256 | 400 | 800 | 1600 | 3000 |
| **Packet loss** | % | 0 | 0.01 | 0.02 | 0.04 | 0.1 | 0.2 | 0.5 | 1 |
| **Reordered packets** | % | 0 | 0.00025 | 0.0005 | 0.001 | 0.005 | 0.01 | 0.05 | 0.1 |
| **Destination location (B) parameters** | | | | | | | | | |
| **Access B occupancy** | % | 0 | 1 | 2 | 4 | 8 | 15 | 30 | 50 |
| **MTU B** | bytes | 512 | 512 | 1508 | 1508 | 1508 | 1508 | 1508 | 1508 |
| **LAN B occupancy** | % | 1 | 2 | 3 | 5 | 8 | 12 | 16 | 20 |
| * Condition H may exceed the ranges in Table 4 to account for disaster conditions. | | | | | | | | | |

The full list of the 1064 test cases are labelled as follows:

- 1A, 1B, 1C…1H combine rate combination 1 with severity levels A, B, C…H.
- 2A, 2B, 2C…2H combine rate combination 1 with the same severity levels A, B, C…H.
- and so on, until…
- 133H completes the 133 × 8 = 1064 total test cases.

## 6.5 Network model coverage

Figures 4 through 7 show examples of NMC curves for a VoIP connection based on example results statistics. In the sample curves, the Y axes show a desired quality parameter and the X axes show the percent of the network model coverage. NMC Curves are created by the following procedure:

1) Run each test case (has an associated NMC score).
2) Measure desired parameter(s) (e.g., PESQ, PEAQ, PSQM, MOS, throughput, connect rate, video quality measurement, etc.).
3) Sort measured parameter(s) along with associated NMC Scores in a descending order using a spreadsheet or similar mechanism.
4) Plot the measured parameter(s) on the Y axis and the associated NMC Score on the X axis.
5) The resulting curve shows the performance (in terms of the measured parameter) as a percentage of the network model.

The resulting graph is used to compare the performance/quality of service for different SLAs or devices. ITU-T Rec. G.107 assigns user satisfaction levels for R-factor and mean opinion score (MOS) – listening quality values. The point where NMC coverage (X axis) crosses a score (Y axis) indicates the percentage of users who will experience that level of user satisfaction or higher.

These example graphs illustrate the comparison of voice quality scores for a specific device across SLA profiles. However, any performance or quality metric can be used on the Y axis to evaluate NMC coverage across service level profiles or multiple devices.

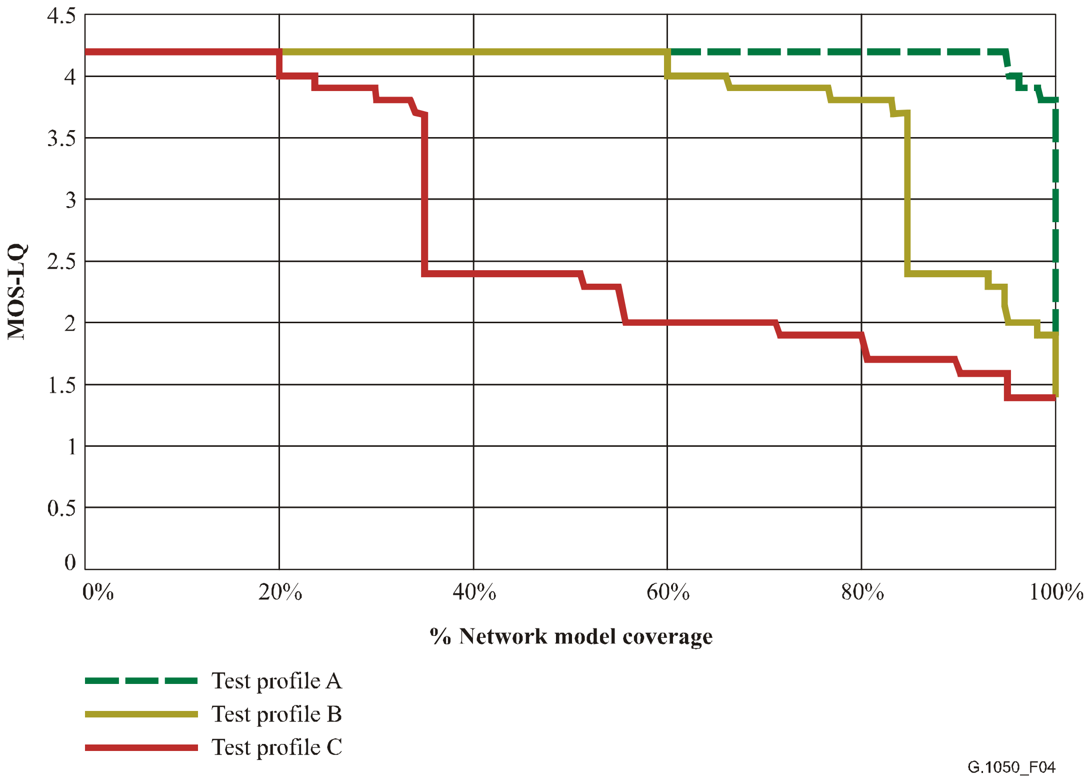


**Figure 4/G.1050 – Sample network coverage curves using MOS – listening quality (40 ms jitter buffer)**

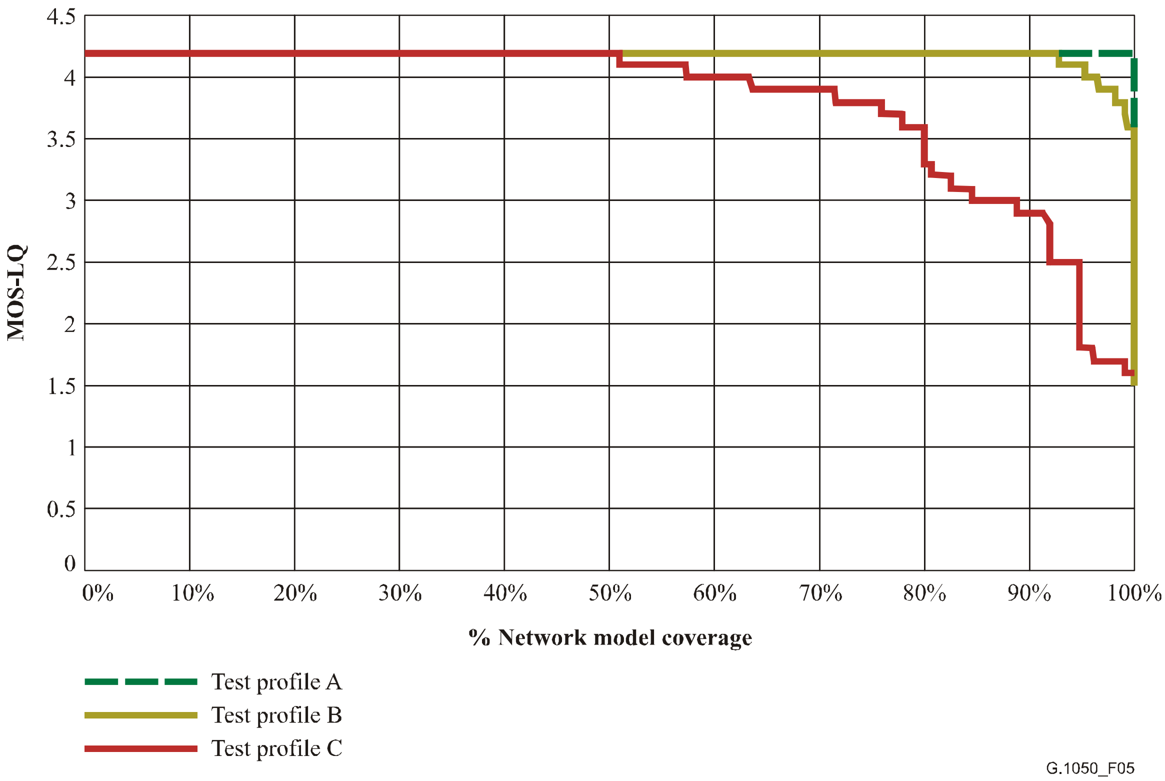


**Figure 5/G.1050 – Sample network coverage curves using MOS – listening quality (100 ms jitter buffer)**

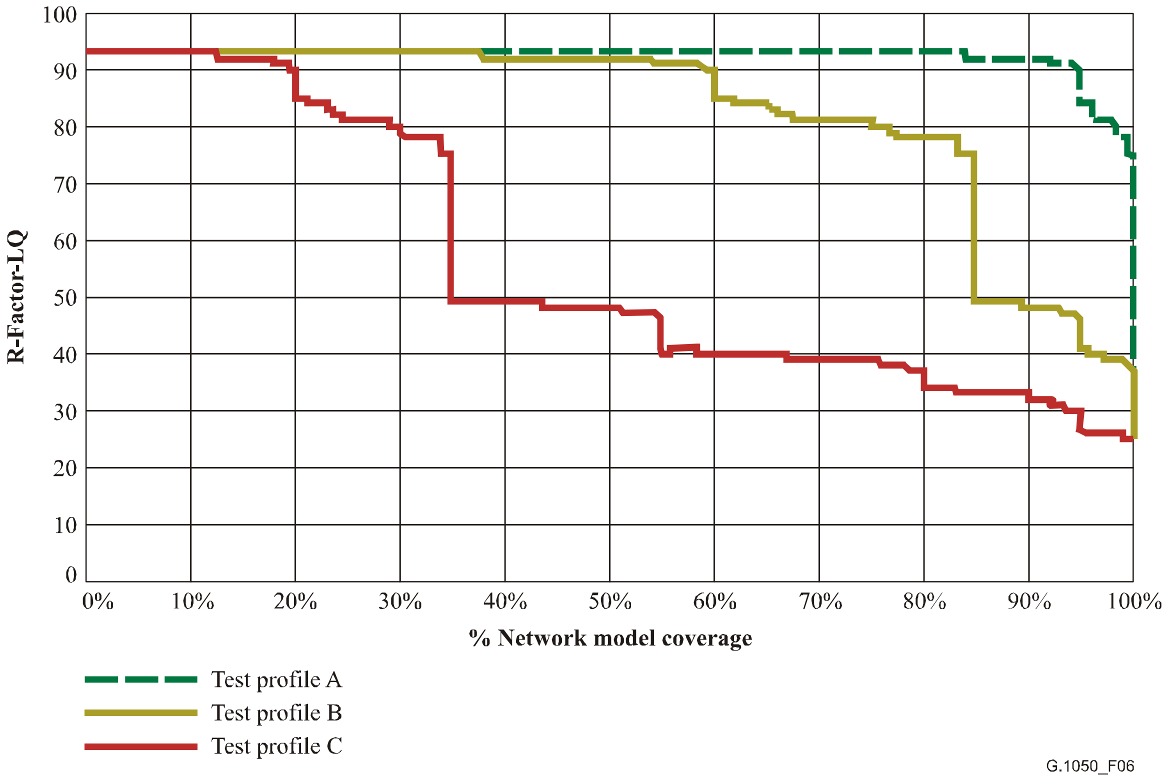


**Figure 6/G.1050 – Sample network coverage curves using R-factor – listening quality (40 ms jitter buffer)**

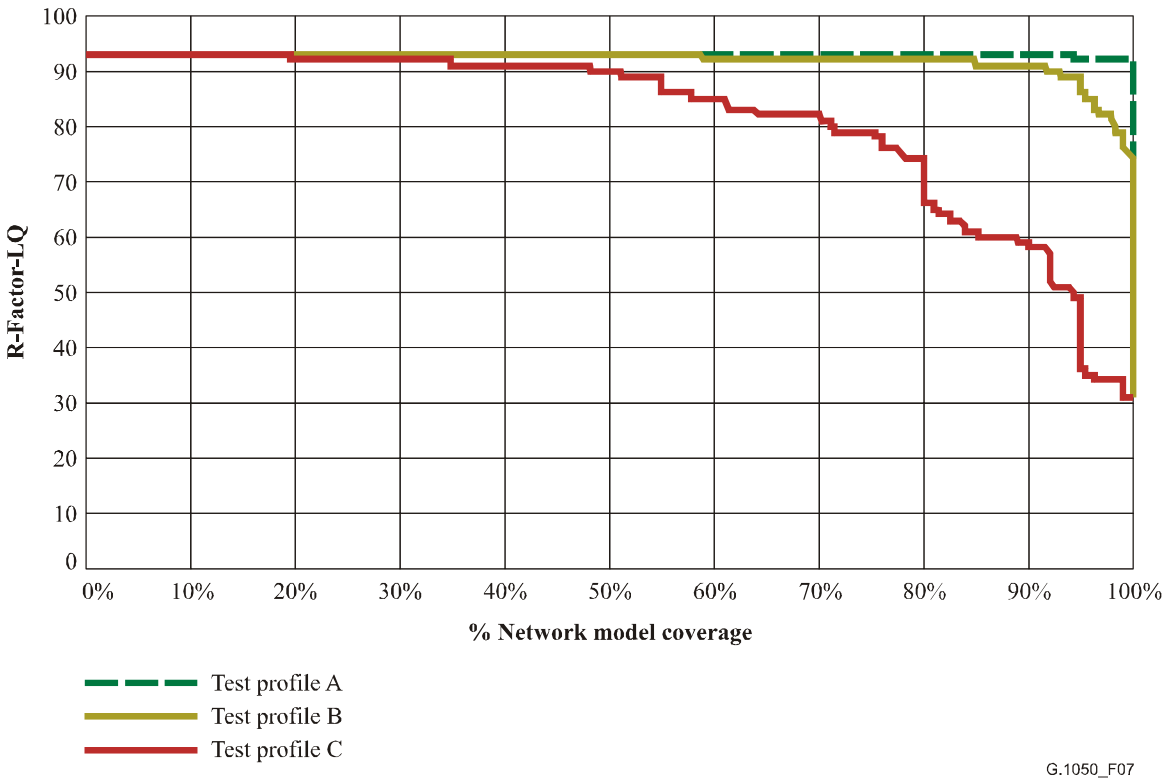


**Figure 7/G.1050 – Sample network coverage curves using R-factor – listening quality (100 ms jitter buffer)**

The values in Tables 11 and 12 represent the percentage of users who are at this level of user satisfaction or higher. The percentages approximately correlate to the values on the graphs of Figures 4 through 7. From these tables and graphs you can easily compare the effect of implementing a 40 ms jitter buffer versus a 100 ms jitter buffer.

**Table 11/G.1050 – Sample network model coverage and MOS listening quality scores**

| MOS – listening quality | Figure 4 (40 ms) | | | | Figure 5 (100 ms) | | | G.107 user satisfaction |
|---|---|---|---|---|---|---|---|---|
| | NMC A | NMC B | NMC C | | NMC A | NMC B | NMC C | |
| 4.3 | 95% | 60% | 20% | | 100% | 93% | 51% | very satisfied |
| 4.0 | 96% | 66% | 24% | | 100% | 97% | 64% | satisfied |
| 3.6 | 100% | 85% | 35% | | 100% | 100% | 80% | some users dissatisfied |
| 3.1 | 100% | 85% | 35% | | 100% | 100% | 85% | many users dissatisfied |
| 2.6 | 100% | 92% | 51% | | 100% | 100% | 92% | nearly all users dissatisfied |
| 1.0 | 100% | 100% | 100% | | 100% | 100% | 100% | not recommended |

**Table 12/G.1050 – Sample network model coverage and R-factor listening quality scores**

| R-factor – listening quality | Figure 6 (40 ms) | | | | Figure 7 (100 ms) | | | G.107 user satisfaction |
|---|---|---|---|---|---|---|---|---|
| | NMC A | NMC B | NMC C | | NMC A | NMC B | NMC C | |
| 90+ | 95% | 60% | 20% | | 100% | 93% | 51% | very satisfied |
| 80 | 98% | 77% | 30% | | 100% | 98% | 72% | satisfied |
| 70 | 100% | 85% | 35% | | 100% | 100% | 80% | some users dissatisfied |
| 60 | 100% | 85% | 35% | | 100% | 100% | 89% | many users dissatisfied |
| 50 | 100% | 90% | 44% | | 100% | 100% | 94% | nearly all users dissatisfied |
| <50 | 100% | 100% | 100% | | 100% | 100% | 100% | not recommended |

# Appendix I

# Rationale for IP network model

## I.1 Wireless LANs

Wireless LANs based on the IEEE 802.11 series standards are the most widely deployed LANs in the home. This is primarily due to the simplicity of networking computers when connected to broadband access with DSL or cable modem.

Wireless LAN rates are primarily determined by physical layer technology and operating conditions. Currently the most widely deployed LAN in the home is based on the legacy IEEE 802.11b standard. The typical user experience of throughput after considerations for overhead and for limitations due to RF noise from other 2.4 GHz unlicensed devices and also the distance between the access point and wireless modem is approximately 4 Mbit/s. Therefore, 4 Mbit/s was used in Table 5 (LAN rates for home locations). The next higher speed LAN is based on the IEEE 802.11g and also the IEEE 802.11a standards. Until recently, higher costs for these systems have limited early deployments of these units to early adopters; consequently, they are not as widely deployed. Typical user throughput after considerations for OSI layer 1-3 overhead and for limitations due to RF noise and distance between the access point and wireless modem is 20 Mbit/s. Therefore, 20 Mbit/s was used in Table 5 (LAN rates for home locations).

## I.2 Structured wiring

Wired Ethernet in the premises nearly always provides higher data rates than their wireless counterparts. This is primarily the result of significantly lower overhead in the Ethernet technology and also due to the CAT5/CAT6 transport medium's resistance to ingress of RF noise. Structured wiring rates include 10BT (10 Mbit/s) and 100BT (100 Mbit/s) in hub and switched arrangements, and the more recent gigabit Ethernet for the early adopters. Homes most often use 10/100 Ethernet connections while businesses typically use 100 Mbit/s or 1 Gbit/s Ethernet connections. Therefore, 10 Mbit/s was used in Table 5 (LAN rates for home locations) and 100 Mbit/s was used in Table 6 (LAN rates for business locations).

## I.3 Hubs versus switches

Hubs, when compared to switches, are a limiting factor on network speeds. Occupancy levels are higher on hub arrangements due to collisions among the traffic. Also, many hubs limit user data transport to half duplex. Conversely, switches are not encumbered with collisions and always operate in full duplex mode.

While reduced costs are encouraging use of switches over hubs in premises networks, there are still a significant number of legacy hub arrangements in use today. In order to reduce the number of data rate variables, the user throughput for a 10 Mbit/s wired LAN when used with a hub is assumed to be 4 Mbit/s, the same rate as an IEEE 802.11b wireless LAN using a switch.

## I.4 Access rates

Access technologies consist mostly of ADSL, cable modems, SDSL, ISDN, T1, T3/E3, or fibre. The most widely deployed of these technologies are ADSL and cable modems. Cable and telecom providers are very competitive and offer similar rates. More recently, service providers have been offering 3 Mbit/s downstream and 384-512 kbit/s upstream. After consideration for OSI layer 1-2 overheads and for reduced rates due to distances served and for impairments in the infrastructure, typical user throughputs are estimated to be 1536 kbit/s downstream and 384 kbit/s upstream. In order to reduce the number of variables in the model, the throughput numbers are also aligned with the throughputs of T1 and SDSL. Furthermore, 384 kbit/s SDSL is also included as a significant deployment. This rate is most useful as an extension of broadband on wired loops beyond the reach of ADSL. Unlike the ADSL technologies, SDSL and the other symmetric technologies permit guaranteed rates with service level agreements, increasing their popularity with businesses. T3 is assumed to have an effective throughput of 43 Mbit/s after considering overhead. These assumptions were used to derive the rates in Tables 7 and 8.

## I.5 Router delays

See Table I.1

**Table I.1/G.1050 – Example of typical delays of contribution by router roles**

| Role | Average total delay (sum of queuing and processing) | Delay variation |
|---|---|---|
| Access gateway | 10 ms | 16 ms |
| Internetworking gateway | 3 ms | 3 ms |
| Distribution | 3 ms | 3 ms |
| Core | 2 ms | 3 ms |

## I.6 Impairment data from anonymous IP network service providers

The end-to-end characteristics in Tables 2 through 4 are derived from network impairment data from anonymous service providers and network equipment manufacturers and include the contribution of LAN and access sections.

# Appendix II

## Packet delay and loss algorithms

### II.1 General IP network model

The IP network is modelled as a concatenation of five segments: local LAN segment, local access link segment, core IP network segment, remote access link segment, remote LAN segment. Each segment introduces packet loss with some probability and a time-varying delay. The input to the model is a set of segment parameters (LAN and access rates, occupancy and a set of core network metrics), packet size(s), packet rate and the total number of packets to be passed end to end. Time slices of 1 ms are assigned a delay value and loss probability using the model parameters. When a packet arrives, it is assigned the delay value and loss probability of the millisecond in which it arrives. The output is the total delay value for each packet and an indication of whether or not a packet was lost.

### II.2 Packet loss model

#### II.2.1 Bursty packet loss

It is well known that packet loss in IP networks is bursty in nature. Within the context of this model the definition of "burst" is a period of time bounded by lost packets during which packet loss is high. This is distinguished from a "consecutive loss period", which is a period of time bounded by lost packets during which all packets are lost.

Bursty packet loss is modelled with a two-state model, a Gilbert-Elliott model, which switches between a high-loss-rate state (HIGH_LOSS state) and a low-loss-rate state (LOW_LOSS state). The Gilbert-Elliott model has four parameters per segment: loss probability in the HIGH_LOSS state, loss probability in the LOW_LOSS state, the probability of transitioning from the HIGH_LOSS to the LOW_LOSS state and the probability of transitioning from the LOW_LOSS to the HIGH_LOSS state. Loss rates of the core network are given parameters. Loss rates of LAN and access links depend on LAN and access link occupancy parameters. Pseudo-code for such a model is shown below:

```
if rand() < loss_probability[LOSS_STATE]
     loss = TRUE
else
     loss = FALSE
endif
if rand() < transition_probability[LOSS_STATE]
     if LOSS_STATE == HIGH_LOSS
          LOSS_STATE = LOW_LOSS
     else
          LOSS_STATE = HIGH_LOSS
     endif
endif
```

#### II.2.2 Consecutive packet loss

Link failure is another source of loss in the core network. This leads to consecutive packet loss for some period of time. This is modelled with two parameters, a periodic link failure rate together with a duration of link outage once it occurs.

### II.3 Delay variation model

Time series models are used to represent the characteristics of sequences that have some properties that vary in time. They typically comprise one or more filter functions driven by a combination of noise and some underlying signal or periodic element.

The "spiky" nature of delay traces suggests that jitter can be modelled using an impulse noise sequence. The delay encountered by a packet at some specific stage in the network should be a function of the serialization delay of interfering traffic and the volume of traffic. The height of the impulses should therefore be a function of serialization delay and the frequency a function of congestion level. LAN congestion tends to occur in short bursts – with Ethernet's CSMA/CD algorithm one packet may be delayed, however the next may gain access to the LAN immediately; this suggests a short filter response time. Access link congestion tends to be associated with short-term delay variations due to the queue in the edge router filling; this suggests a longer filter response time. Pseudo-code for delay variation is shown below:

```
if rand() < impulse_probability
     i = impulse_height
else
     i = 0
endif
d(n) = d(n-1) * (TC) + i * (1-TC)
```

where d(n) = delay of packet n, and TC represents the filter time constant.

### II.3.1 LAN and access link jitter

Jitter in the LAN and access links is modelled with per-millisecond delay values created by passing impulses through a one-pole filter. Within each segment, for each millisecond an impulse or a zero is input to the filter based on some probability. The filter output is then computed and the result becomes the delay value for that millisecond. Delay values are applied to packets based on the current values in the millisecond during which the packet arrives, but arrival packet order is maintained. The amplitude of the impulses is proportional to the serialization delay of that segment. The probability of occurrence of an impulse is proportional to the congestion level for that segment. For the LAN segments no filter is used; delay comes directly from the impulses. For the access link segments a filter with a time constant is used to scale the values for 1 ms intervals.

### II.3.2 Core network jitter

The core network jitter is modelled differently. For each packet a random delay is added. This delay is uniformly distributed from 0 to the core network jitter parameter value.

### II.3.3 Core network base delay and route flapping

A base delay parameter is associated with the core network. Another source of delay variation is route flapping in the core network. This is modelled by change in the base delay of the core network. A periodic route flap rate is a given parameter. When a route flap occurs the model will add or subtract the route flap delay hit to or from the core network delay. For each route flap, the model toggles between adding and subtracting the route flap delay.

## II.4 Core packet reordering

In the model, only the core is allowed to reorder packets based on delays. Each time-slice has a delay value. When a packet arrives, the current delay value is applied to that packet. The core segment is the only segment that allows reordering. In the other segments packets are transmitted in the order they arrive, regardless of the delay values assigned.

## II.5 Model output

If a packet is marked as lost in any segment, then it is lost.

The total delay added to a packet is the sum of the delay from each segment. There may be out of order packets due to delay variations. LAN and access links should not cause packet reordering. Therefore, delay due to LAN and access links is summed together first and delays are adjusted to keep packets in order. Then delay due to the core network is added. This may result in out of order packets.

## II.6 Model input parameters

The following is a list of model input parameters and how those parameters are used.

### II.6.1 Local and remote LAN segment parameters

Input parameters from Tables 5 and 6:

1) LAN speed. This speed is used to compute LAN segment delay.
2) LAN percent occupancy.

Derived parameters:

1) LAN loss probability. One value for each loss state. Current values: for the low loss state the probability is 0. For the high loss state it is 0.004 × percent occupancy.
2) LAN loss state transition probability. One value for each loss state. Current values: the probability of transitioning from the low loss to the high loss state is 0.004 × percent occupancy. The probability of the reverse transition is 0.1.
3) LAN jitter filter impulse height. One value for each loss state. Current values: max impulse height = (MTU-size bit times) × (1 + (percent occupancy/40)). The value for the low loss state is a random variable uniformly distributed from 0 to max impulse height. The value for the high loss state is max impulse height.
4) LAN jitter filter impulse probability. Current values: the value for the low loss state is 0. The value for the high loss state is 0.5.
5) LAN jitter filter coefficients. The filter output is the delay value for the current packet. This delay is A × (impulse height) + (1 – A) × (previous delay). Current values: A = 1 (i.e., no filtering).

### II.6.2 Local and remote link segment parameters

1) Link speed. This speed is used to compute LAN segment delay.
2) Link percent occupancy.
3) Link MTU size.
4) Link loss state transition probability. One value for each loss state. Current values: the probability of transitioning from the low loss to the high loss state is 0.0003 × (percent occupancy). The probability of the reverse transition is 0.2/(1 + (percent occupancy)).
5) Link jitter filter impulse height. One value for each loss state. Current values: max impulse height = (MTU-size bit times) × (1 + (percent occupancy/40)). The value for the low loss state is a random variable uniformly distributed from 0 to max impulse height. The value for the high loss state is max impulse height.
6) Link jitter filter impulse probability. Current values: the value for the low loss state is 0.001 + (percent occupancy)/2000. The value for the high loss state is 0.3 + 0.4 × (percent occupancy)/100.
7) Link jitter filter coefficients. The filter output is the delay value for the current packet. This delay is A × (impulse height) + (1 – A) × (previous delay). Current values: A = 0.25.
8) Link loss probability. One value for each loss state. Current values: for the low loss state the probability is 0. For the high loss state it is 0.0005 × percent occupancy.
9) Link base delay. This is packet-size bit times. It is assumed that the packet size is fixed based on the application.

### II.6.3 Core IP network segment parameters

1) Delay.
2) Packet loss. There is only one loss state. The loss probability is just the given core network loss probability parameter.
3) Jitter. The jitter in the core network is modelled as added delay uniformly distributed between 0 and the core network jitter parameter value.
4) Route flap interval.
5) Route flap delay.
6) Link failure interval.
7) Link failure duration.
8) Reorder percentage.

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