International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.1050

(11/2007)

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

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

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

ITU-T Recommendation G.1050 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. 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 13 November 2007 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

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.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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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/.

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): the percentage of network connections over which a particular level of performance is achieved. 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 (layer 1) impairments, this Recommendation focuses on the impact of impairments on Internet Protocol (IP) 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 an application-independent model (e.g., data, voice, voiceband data, video) 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 2007 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 following are parameters and impairments that affect quality of service and IP network performance:

- Network architecture;
- Types of access links;
- QoS controlled edge routing;
- MTU size;
- Network faults;
- Link failure;
- 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 (IPTV, video conferencing, telepresence, etc.);
  - IP phones (including soft 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.

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 VoIP 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 2007 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 and other references 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 G.107] ITU-T Recommendation G.107 (2005), *The E-model, a computational model for use in transmission planning*.

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

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

[ITU-T T.38] ITU-T Recommendation T.38 (2007), *Procedures for real-time Group 3 facsimile communication over IP networks*.

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

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

[ITU-T V.151] ITU-T Recommendation V.151 (2006), *Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay*.

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

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

3 Definitions

This Recommendation defines the following terms:

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

3.2 delay: The time required for a packet to traverse the network or a segment of the network (see latency).

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

3.4 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).

3.5 E-model: A standard-based ([ITU-T G.107], [ITU-T 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.

3.6 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).

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

3.8 jitter: Variation in packet delay.

3.9 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.

3.10 latency: An expression of how much time it takes for a packet of data to get from one designated point to another (see delay).

3.11 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.

3.12 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.

3.13 link failure interval: The interval between link failures.

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

3.15 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 1-5.

3.16 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 1-5.

3.17 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.

3.18 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 = $LOO_{LAN/access} \times LOO_{Severity}$.

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

3.20 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).

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

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

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

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

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

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

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

3.29 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.

3.30 sequential packet loss: Two or more consecutive lost packets.

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 and acronyms

This Recommendation uses the following abbreviations and acronyms:

ADSL	Asymmetric Digital Subscriber Line
CATV	Cable Television
CSMA/CD	Carrier Sense Multiple Access/Collision Detection
IP	Internet Protocol
IPTV	Internet Protocol Television
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
PEAQ	Perceived Audio Quality
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
UUT	Unit Under Test

VDSL	Very high speed Digital Subscriber Line
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 1 (LAN-to-LAN) illustrates an end-to-end network with LAN and access links on each side of the core as would occur in a client-to-client application such as VoIP. Figure 2 (Core-to-LAN) illustrates an end-to-end network with LAN and access on the destination side, but not on the source side as would occur in a server-to-client application such as IPTV or web server access over the Internet.

Figures 1 and 2 show 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.

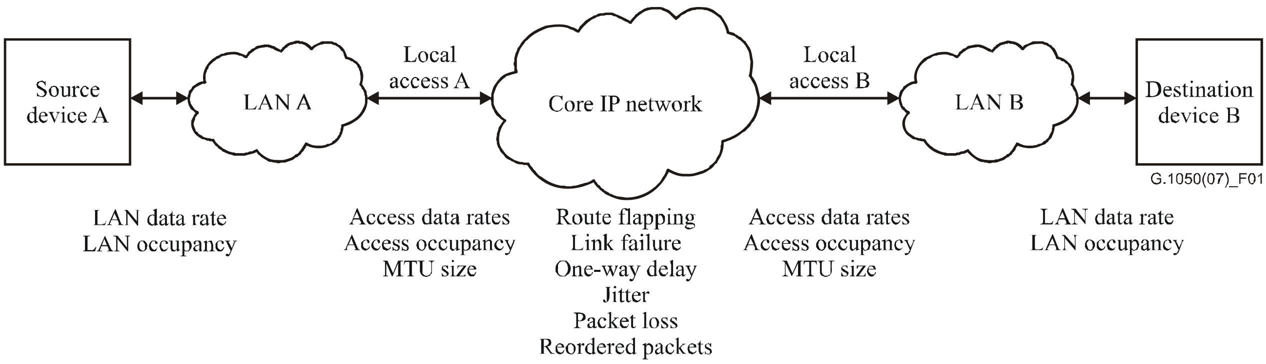

Figure 1 – IP network impairment model: LAN-to-LAN

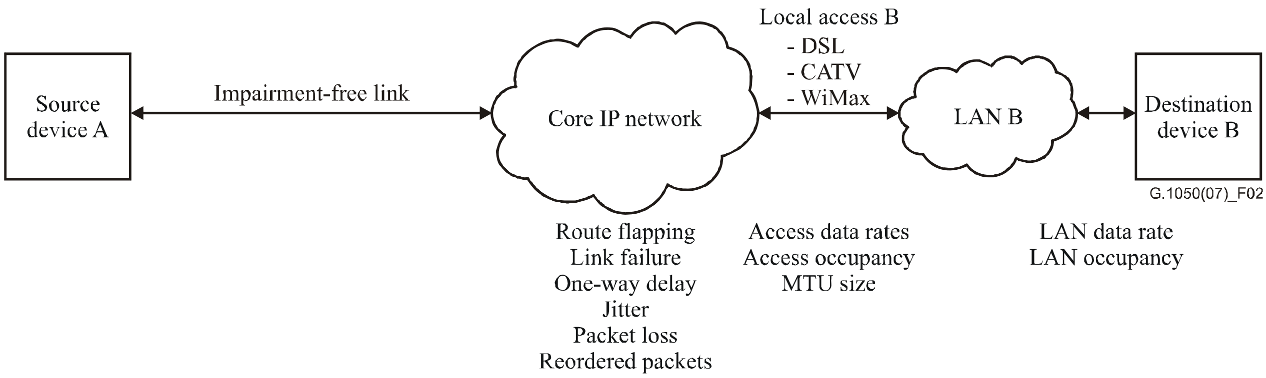

Figure 2 – IP network impairment model: Core-to-LAN

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 setup

6.1 Service test profiles

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

Table 1 – 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
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 (SLAs):

- 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 – Impairment ranges for well-managed network (profile A)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 100 (regional) 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 – Impairment ranges for partially-managed network (profile B)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 100 (regional) 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 – Impairment ranges for unmanaged network (profile C) (Note 1)

Impairment type	Units	Range (min to max)
One-way latency	ms	20 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. NOTE 2 – Sequential packet loss occurs 1 every 10 seconds.		

6.3 Test set-up

(See Figure 3)

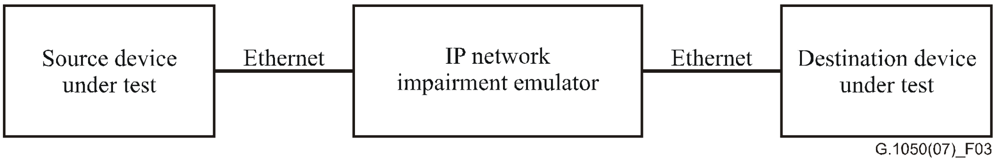

Figure 3 – Simulator set-up block diagram

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 10 list typical rates for home and business locations rather than service offering rates. To reduce the number of test cases, similar effective rates are combined.

Table 5 – LAN rates for home locations, applications ≤3 Mbit/s

Effective LAN rate Mbit/s	**LOO** %	**Represents**
4	40	802.11b, 10BaseT hub
20	40	802.11g, 100BaseT hub
100	20	100 BaseT switched, Gbit Ethernet

Table 6 – LAN rates for home locations, applications >3 Mbit/s

Effective LAN rate Mbit/s	**LOO** %	**Represents**
20	40	802.11g, 100BaseT hub
100	60	100BaseT switched, Gbit Ethernet

Table 7 – LAN rates for business locations

Effective LAN rate Mbit/s	**LOO** %	**Represents**
20	20	802.11g, 100BaseT hub
100	80	100BaseT switched, Gbit Ethernet

Table 8 – Access rates for home locations, applications ≤3 Mbit/s

Access rate		**LOO** %	**Represents**
Toward core kbit/s	**From core** kbit/s		
128	768	30	CATV, ADSL
384	1536	60	CATV, ADSL
384	3000	10	CATV, ADSL

Table 9 – Access rates for home locations, applications >3 Mbit/s

Access rate		LOO %	Represents
Toward core kbit/s	**From core** kbit/s		
768	7000	50	CATV, ADSL, VDSL
1024	20'000	35	Bonded ADSL2, VDSL
3072	30'000	15	CATV, Bonded ADSL2, VDSL, PON

Table 10 – Access rates for business locations

Access rate		LOO %	Represents
Toward core kbit/s	**From core** kbit/s		
384	1536	5	ADSL entry
384	3000	30	ADSL premium
1536	1536	15	T1, WiMax
768	7000	45	ADSL2, VDSL, multipoint WiMax
13'000	13'000	5	Point-to-point WiMax, PON

The LAN and access rates are combined in three different ways for three different application scenarios:

1) LAN-to-LAN: These rates reflect applications as pictured in Figure 1. The rate combinations are listed in Table 11. This table is relevant to applications that work between one LAN and another, such as a VoIP call. Both home and business locations are included in the LAN-to-LAN scenario.

2) Core-to-LAN, excluding IPTV (≤3 Mbit/s): These rates reflect applications as pictured in Figure 2. The rate combinations are listed in Table 12. This table is relevant to applications that work between a server in the core to a LAN, such as a co-located web server or a storage solution. Both home and business locations are included in the core-to-LAN scenario.

3) IPTV core-to-LAN (>3 Mbit/s): This scenario is also shown in Figure 2, but includes only those LAN and access rates that are appropriate for IPTV solutions, listed in Table 6 and Table 9. The rate combinations are listed in Table 13. Business locations are not included in the IPTV core-to-LAN scenario.

These three models are independent. The sum of the likelihoods of occurrence (LOO) in each of the three scenarios is 100%.

6.4.1 LAN-to-LAN scenario rate combination table

Table 11 consists of all possible combinations of locations, LAN rates and access rates for home applications ≤3 Mbit/s and all business applications. These combinations are taken from Tables 5, 7, 8 and 10. The LOO column for each combination is calculated by multiplying the LOO for the LAN rate by the LOO for the access rate. Home and business locations are weighted equally. Where duplicate rate combinations (or mirror image combinations) occur, they are combined and their associated LOO added together. This results in 168 unique rate combinations with LOO values that add to 100%. The LAN-to-LAN rate combinations in Table 11 apply to the network impairment model of Figure 1.

Table 11 – LAN-to-LAN scenario rate combinations

Rate comb #	LAN A rate	A->B access rate at A	A->B access rate at B	LAN B rate	B->A access rate at B	B->A access rate at A	LOO
Units=>	Mbit/s	kbit/s	kbit/s	Mbit/s	kbit/s	kbit/s	%
1	4	128	768	4	128	768	0.360
2	4	128	768	20	128	768	0.720
3	4	128	768	100	128	768	0.360
4	20	128	768	20	128	768	0.360
5	20	128	768	100	128	768	0.360
6	100	128	768	100	128	768	0.090
7	4	128	1536	4	384	768	0.720
8	4	128	1536	20	384	768	1.470
9	4	128	1536	100	384	768	0.840
10	20	128	1536	20	384	768	0.750
11	20	128	1536	100	384	768	0.855
12	100	128	1536	100	384	768	0.240
13	4	128	3000	4	384	768	0.120
14	4	128	3000	20	384	768	0.420
15	4	128	3000	100	384	768	0.840
16	20	128	3000	20	384	768	0.300
17	20	128	3000	100	384	768	0.930
18	100	128	3000	100	384	768	0.390
19	4	384	768	4	128	1536	0.720
20	4	384	768	20	128	1536	1.470
21	4	384	768	100	128	1536	0.840
22	20	384	768	20	128	1536	0.750
23	20	384	768	100	128	1536	0.855
24	100	384	768	100	128	1536	0.240
25	4	384	1536	4	384	1536	1.440
26	4	384	1536	20	384	1536	3.000
27	4	384	1536	100	384	1536	1.920
28	20	384	1536	20	384	1536	1.563
29	20	384	1536	100	384	1536	2.000
30	100	384	1536	100	384	1536	0.640
31	4	384	3000	4	384	1536	0.240
32	4	384	3000	20	384	1536	0.850
33	4	384	3000	100	384	1536	1.720
34	20	384	3000	20	384	1536	0.625
35	20	384	3000	100	384	1536	2.025
36	100	384	3000	100	384	1536	1.040

Table 11 – LAN-to-LAN scenario rate combinations

Rate comb #	LAN A rate	A->B access rate at A	A->B access rate at B	LAN B rate	B->A access rate at B	B->A access rate at A	LOO
Units=>	Mbit/s	kbit/s	kbit/s	Mbit/s	kbit/s	kbit/s	%
37	4	384	768	4	128	3000	0.120
38	4	384	768	20	128	3000	0.420
39	4	384	768	100	128	3000	0.840
40	20	384	768	20	128	3000	0.300
41	20	384	768	100	128	3000	0.930
42	100	384	768	100	128	3000	0.390
43	4	384	1536	4	384	3000	0.240
44	4	384	1536	20	384	3000	0.850
45	4	384	1536	100	384	3000	1.720
46	20	384	1536	20	384	3000	0.625
47	20	384	1536	100	384	3000	2.025
48	100	384	1536	100	384	3000	1.040
49	4	384	3000	4	384	3000	0.040
50	4	384	3000	20	384	3000	0.200
51	4	384	3000	100	384	3000	0.520
52	20	384	3000	20	384	3000	0.250
53	20	384	3000	100	384	3000	1.300
54	100	384	3000	100	384	3000	1.690
55	4	128	1536	20	768	1536	0.090
56	4	128	1536	100	768	1536	0.360
57	20	128	1536	20	768	1536	0.090
58	20	128	1536	100	768	1536	0.405
59	100	128	1536	100	768	1536	0.180
60	4	128	7000	20	768	768	0.270
61	4	128	7000	100	768	768	1.080
62	20	128	7000	20	768	768	0.270
63	20	128	7000	100	768	768	1.215
64	100	128	7000	100	768	768	0.540
65	4	128	13000	20	768	13000	0.030
66	4	128	13000	100	768	13000	0.120
67	20	128	13000	20	768	13000	0.030
68	20	128	13000	100	768	13000	0.135
69	100	128	13000	100	768	13000	0.060
70	4	384	1536	20	1536	1536	0.180
71	4	384	1536	100	1536	1536	0.720
72	20	384	1536	20	1536	1536	0.188

Table 11 – LAN-to-LAN scenario rate combinations

Rate comb #	LAN A rate	A->B access rate at A	A->B access rate at B	LAN B rate	B->A access rate at B	B->A access rate at A	LOO
Units=>	Mbit/s	kbit/s	kbit/s	Mbit/s	kbit/s	kbit/s	%
73	20	384	1536	100	1536	1536	0.870
74	100	384	1536	100	1536	1536	0.480
75	4	384	7000	20	768	1536	0.540
76	4	384	7000	100	768	1536	2.160
77	20	384	7000	20	768	1536	0.563
78	20	384	7000	100	768	1536	2.610
79	100	384	7000	100	768	1536	1.440
80	4	384	13000	20	1536	13000	0.060
81	4	384	13000	100	1536	13000	0.240
82	20	384	13000	20	1536	13000	0.063
83	20	384	13000	100	1536	13000	0.290
84	100	384	13000	100	1536	13000	0.160
85	4	384	1536	20	1536	3000	0.030
86	4	384	1536	100	1536	3000	0.120
87	20	384	1536	20	1536	3000	0.075
88	20	384	1536	100	1536	3000	0.495
89	100	384	1536	100	1536	3000	0.780
90	4	384	7000	20	768	3000	0.090
91	4	384	7000	100	768	3000	0.360
92	20	384	7000	20	768	3000	0.225
93	20	384	7000	100	768	3000	1.485
94	100	384	7000	100	768	3000	2.340
95	4	384	13000	20	3000	13000	0.010
96	4	384	13000	100	3000	13000	0.040
97	20	384	13000	20	3000	13000	0.025
98	20	384	13000	100	3000	13000	0.165
99	100	384	13000	100	3000	13000	0.260
100	4	768	1536	20	128	1536	0.090
101	20	768	1536	20	128	1536	0.090
102	20	768	1536	100	128	1536	0.405
103	4	768	1536	100	128	1536	0.360
104	100	768	1536	100	128	1536	0.180
105	4	1536	1536	20	384	1536	0.180
106	20	1536	1536	20	384	1536	0.188
107	20	1536	1536	100	384	1536	0.870
108	4	1536	1536	100	384	1536	0.720

Table 11 – LAN-to-LAN scenario rate combinations

Rate comb #	LAN A rate	A->B access rate at A	A->B access rate at B	LAN B rate	B->A access rate at B	B->A access rate at A	LOO
Units=>	Mbit/s	kbit/s	kbit/s	Mbit/s	kbit/s	kbit/s	%
109	100	1536	1536	100	384	1536	0.480
110	4	1536	3000	20	384	1536	0.030
111	20	1536	3000	20	384	1536	0.075
112	20	1536	3000	100	384	1536	0.495
113	4	1536	3000	100	384	1536	0.120
114	100	1536	3000	100	384	1536	0.780
115	4	768	768	20	128	7000	0.270
116	20	768	768	20	128	7000	0.270
117	20	768	768	100	128	7000	1.215
118	4	768	768	100	128	7000	1.080
119	100	768	768	100	128	7000	0.540
120	4	768	1536	20	384	7000	0.540
121	20	768	1536	20	384	7000	0.563
122	20	768	1536	100	384	7000	2.610
123	4	768	1536	100	384	7000	2.160
124	100	768	1536	100	384	7000	1.440
125	4	768	3000	20	384	7000	0.090
126	20	768	3000	20	384	7000	0.225
127	20	768	3000	100	384	7000	1.485
128	4	768	3000	100	384	7000	0.360
129	100	768	3000	100	384	7000	2.340
130	4	768	13000	20	128	13000	0.030
131	20	768	13000	20	128	13000	0.030
132	20	768	13000	100	128	13000	0.135
133	4	768	13000	100	128	13000	0.120
134	100	768	13000	100	128	13000	0.060
135	4	1536	13000	20	384	13000	0.060
136	20	1536	13000	20	384	13000	0.063
137	20	1536	13000	100	384	13000	0.290
138	4	1536	13000	100	384	13000	0.240
139	100	1536	13000	100	384	13000	0.160
140	4	3000	13000	20	384	13000	0.010
141	20	3000	13000	20	384	13000	0.025
142	20	3000	13000	100	384	13000	0.165
143	4	3000	13000	100	384	13000	0.040
144	100	3000	13000	100	384	13000	0.260

Table 11 – LAN-to-LAN scenario rate combinations

Rate comb #	LAN A rate	A->B access rate at A	A->B access rate at B	LAN B rate	B->A access rate at B	B->A access rate at A	LOO
Units=>	**Mbit/s**	**kbit/s**	**kbit/s**	**Mbit/s**	**kbit/s**	**kbit/s**	**%**
145	20	1536	1536	20	1536	1536	0.023
146	20	1536	1536	100	1536	1536	0.180
147	100	1536	1536	100	1536	1536	0.360
148	20	1536	7000	20	768	1536	0.068
149	20	1536	7000	100	768	1536	0.540
150	100	1536	7000	100	768	1536	1.080
151	20	1536	13000	20	1536	13000	0.015
152	20	1536	13000	100	1536	13000	0.120
153	100	1536	13000	100	1536	13000	0.240
154	20	768	1536	20	1536	7000	0.068
155	20	768	1536	100	1536	7000	0.540
156	100	768	1536	100	1536	7000	1.080
157	20	768	7000	20	768	7000	0.203
158	20	768	7000	100	768	7000	1.620
159	100	768	7000	100	768	7000	3.240
160	20	768	13000	20	7000	13000	0.023
161	20	768	13000	100	7000	13000	0.180
162	100	768	13000	100	7000	13000	0.360
163	20	7000	13000	20	768	13000	0.023
164	20	7000	13000	100	768	13000	0.180
165	100	7000	13000	100	768	13000	0.360
166	20	13000	13000	20	13000	13000	0.003
167	20	13000	13000	100	13000	13000	0.020
168	100	13000	13000	100	13000	13000	0.040

6.4.2 Core-to-LAN (≤3 Mbit/s excluding IPTV) scenario rate combination table

Table 12 lists the LAN and access rate combinations for applications excluding IPTV, including home and business locations. Home and business locations are weighted equally in computing these LOO percentages. The core-to-LAN rate combinations in Table 12 apply to the network impairment model of Figure 2.

Table 12 – Core-to-LAN (excluding IPTV ≤3 Mbit/s) rate combinations

Rate comb #	**LAN** (Mbit/s)	**Access** (kbit/s)		**LOO** (%)
		To core	**From core**	
169	4	128	768	6.0
170	20	128	768	6.0
171	100	128	768	3.0
172	4	384	1536	12.0
173	20	384	1536	12.5
174	100	384	1536	8.0
175	4	384	3000	2.0
176	20	384	3000	5.0
177	100	384	3000	13.0
178	20	1536	1536	1.5
179	100	1536	1536	6.0
180	20	768	7000	4.5
181	100	768	7000	18.0
182	20	13000	13000	0.5
183	100	13000	13000	2.0

6.4.3 IPTV core-to-LAN scenario rate combination table

Table 13 lists the LAN and access rate combinations typically used for IPTV in home locations. The core-to-LAN rate combinations in Table 13 apply to the network impairment model of Figure 2.

Table 13 – IPTV core-to-LAN rate combinations

Rate comb #	**LAN** (Mbit/s)	**Access** (kbit/s)		**LOO** (%)
		To core	**From core**	
184	20	768	7000	20.0
185	100	768	7000	30.0
186	20	1024	20000	14.0
187	100	1024	20000	21.0
188	20	3072	30000	6.0
189	100	3072	30000	9.0

6.4.4 Impairment severities

Table 14 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 clause 6.2.1. Refer to Appendix II for the precise usage of these parameters in the impairment algorithms.

Table 14 – Impairment severity combinations

Impairment	Severity=> Units	A	B	C	D	E	F	G	H (Note)
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
NOTE – Condition H may exceed the ranges in Table 4 to account for disaster conditions.									

The test cases for the LAN-to-LAN scenario 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…
- 168H completes the 168 × 8 = 1344 test cases for this scenario.

The core-to-LAN (excluding IPTV) test cases are labelled as follows:

- 169A, 169B, 169C…169H combine rate combination 169 with severity levels A, B, C…H;
- 170A, 170B, 170C…170H combine rate combination 170 with the same severity levels A B, C…H;
- and so on, until…
- 183H completes the 15 × 8 = 120 test cases for this scenario.

The IPTV core-to-LAN test cases are labelled as follows:

- 184A, 184B, 184C…184H combine rate combination 184 with severity levels A, B, C…H;
- 185A, 185B, 185C…185H combine rate combination 185 with the same severity levels A, B, C…H;
- and so on, until…
- 189H completes the $6 \times 8 = 48$ test cases for this scenario.

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 for the model of interest (LAN-to-LAN, core-to-LAN ≤3 Mbit/s, core-to-LAN >3 Mbit/s).
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 G.107] assigns user satisfaction levels for R-factor and mean opinion score listening quality (MOS-LQ) 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.

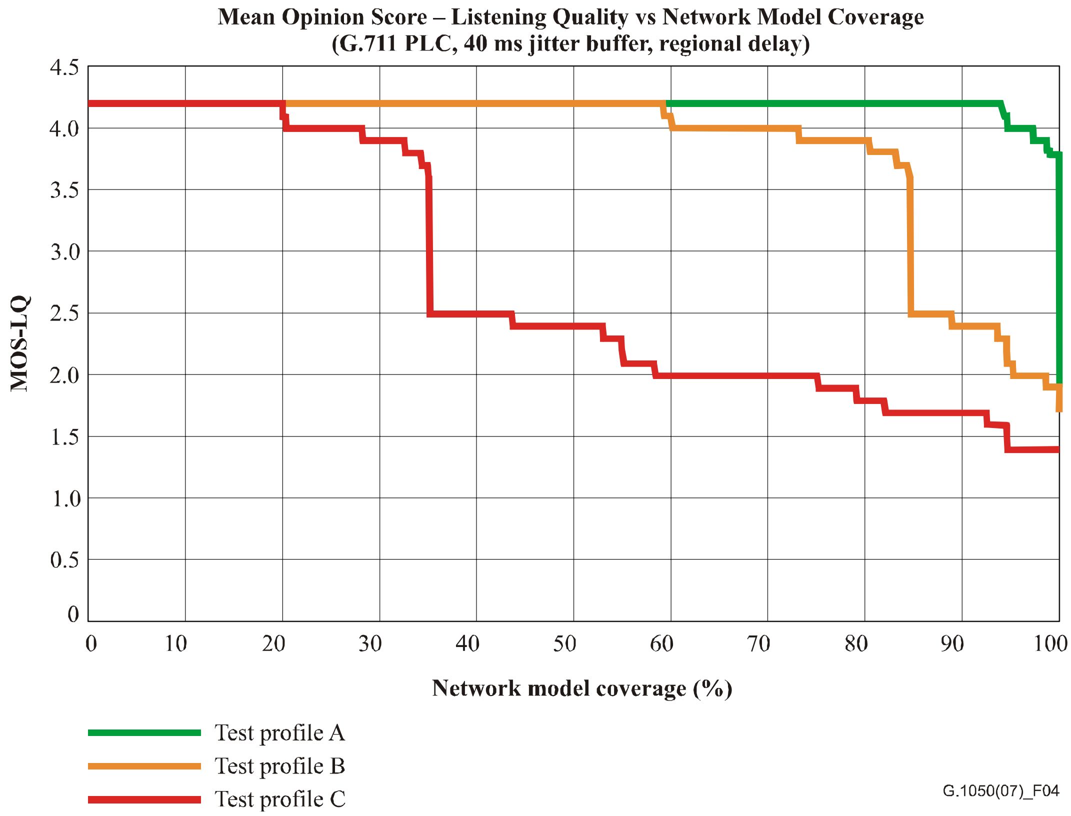

Figure 4 – Sample network coverage curves using MOS-LQ (40 ms jitter buffer)

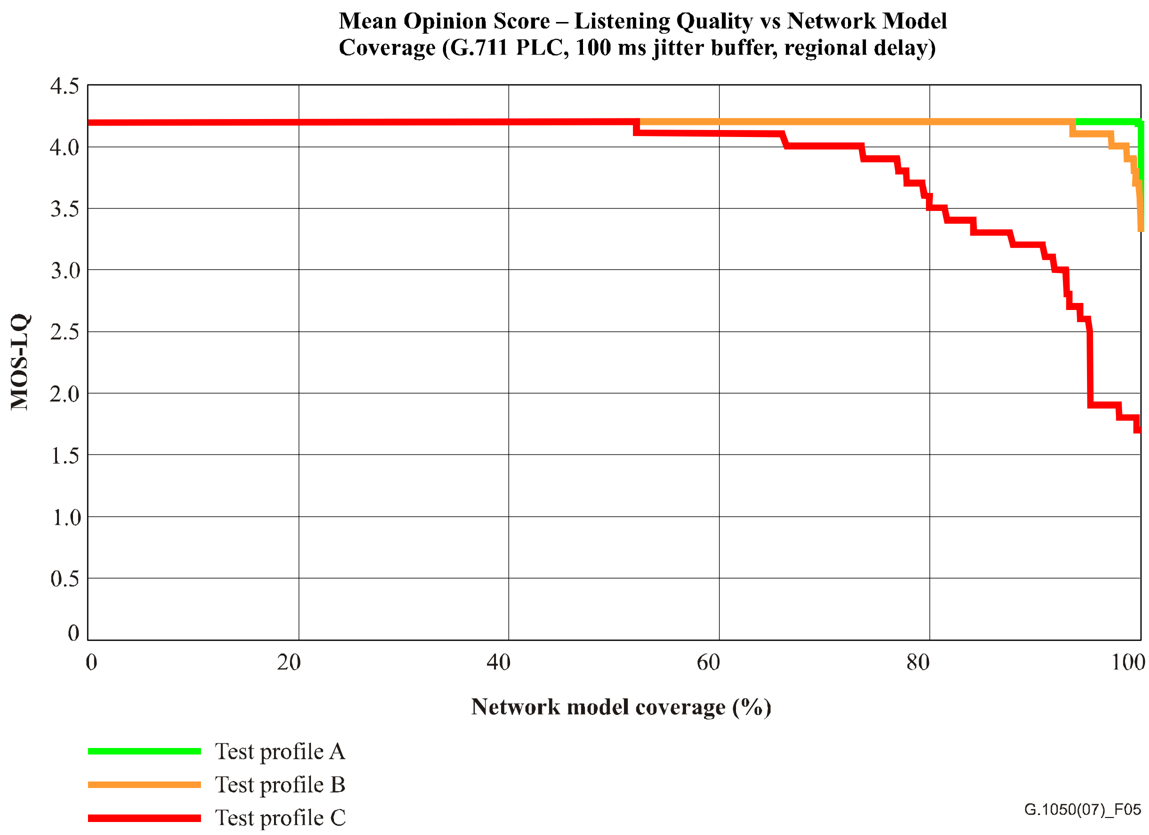

Figure 5 – Sample network coverage curves using MOS-LQ (100 ms jitter buffer)

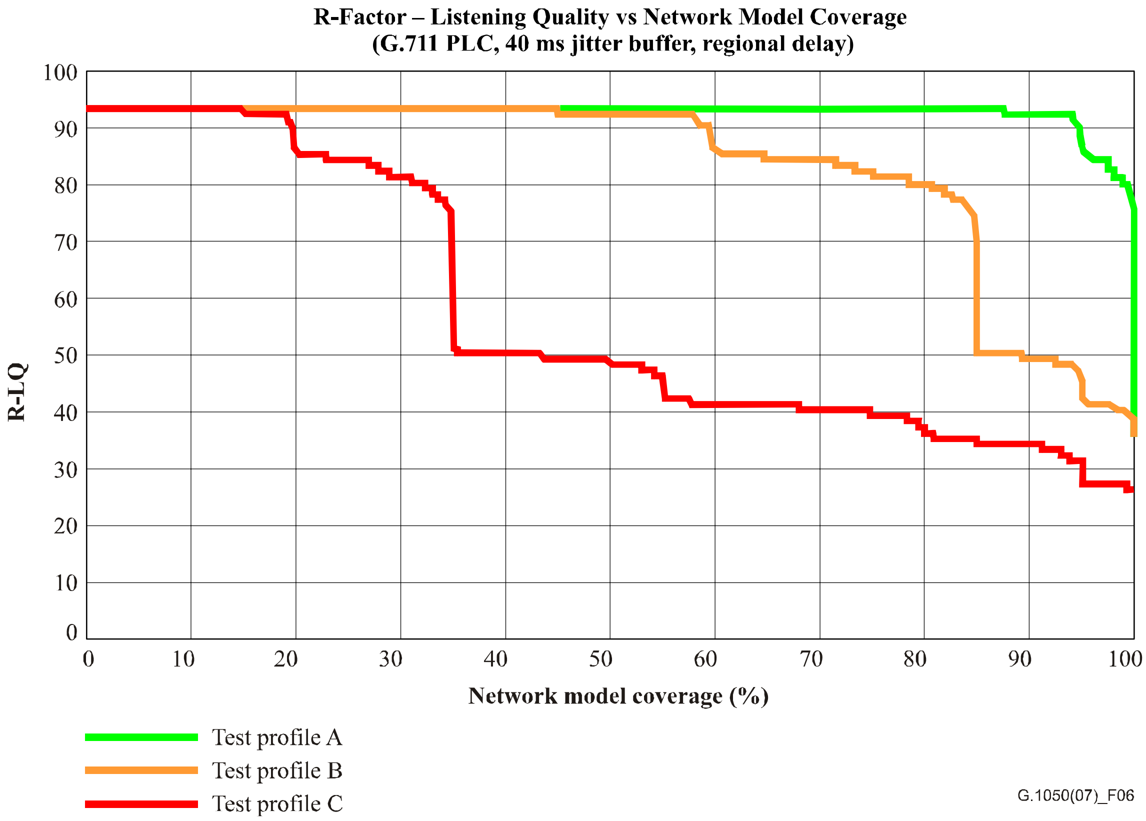

Figure 6 – Sample network coverage curves using R-LQ (40 ms jitter buffer)

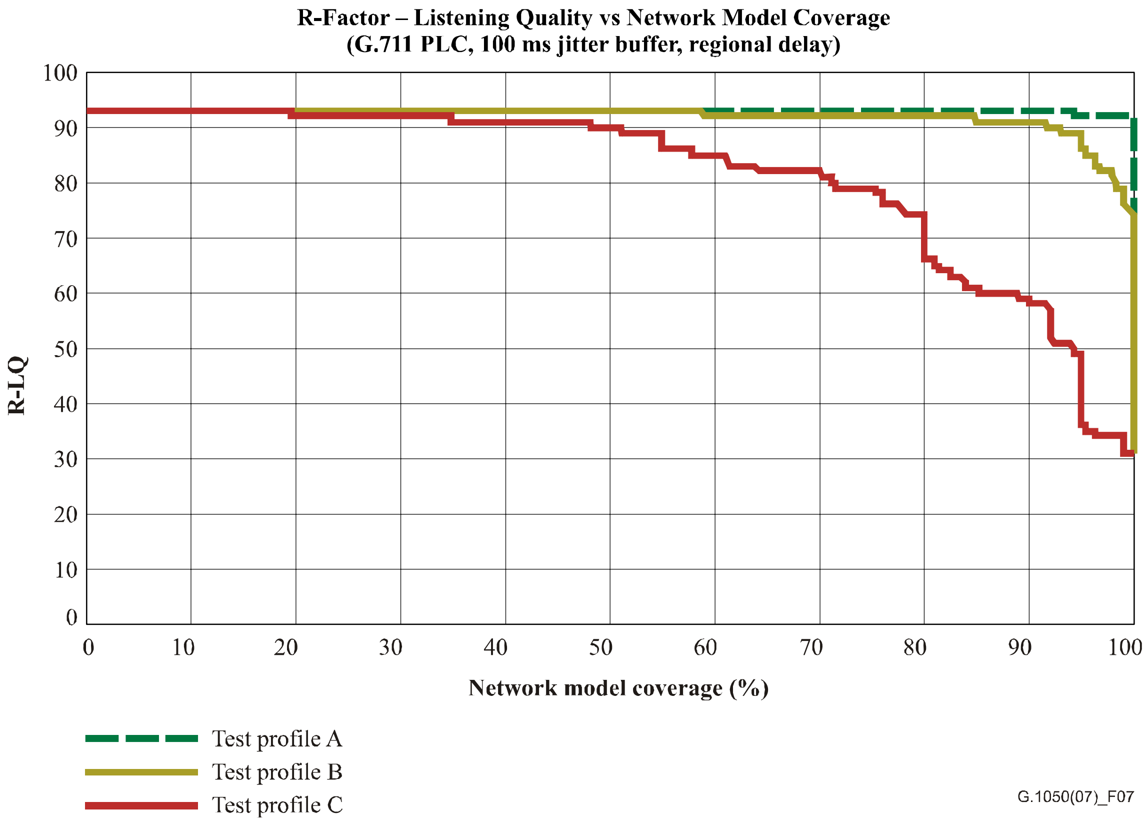

Figure 7 – Sample network coverage curves using R-LQ (100 ms jitter buffer)

The values in Tables 15 and 16 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 15 – Sample network model coverage and MOS-LQ scores

MOS-LQ	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	98%	73%	29%		100%	99%	74%	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%	85%	35%		100%	100%	92%	Nearly all users dissatisfied
1.0	100%	100%	100%		100%	100%	100%	Not recommended

Table 16 – Sample network model coverage and R-LQ scores

R-LQ	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+	94%	58%	20%		100%	92%	50%	Very satisfied
80	99%	80%	31%		100%	99%	75%	Satisfied
70	100%	85%	35%		100%	100%	81%	Some users dissatisfied
60	100%	85%	35%		100%	100%	93%	Many users dissatisfied
50	100%	89%	44%		100%	100%	95%	Nearly all users dissatisfied
<50	100%	100%	100%		100%	100%	100%	Not recommended

Appendix I

Rationale for IP network model

(This appendix does not form an integral part of this Recommendation)

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 LANs are now installed in most businesses.

Wireless LAN rates are primarily determined by physical layer technology and operating conditions. The first wireless LAN deployed in the home was based on the 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. 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, 20 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. 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. As fibre to the residence becomes more widely deployed, even higher rates will become commonplace, with downstream rates as high as 30 Mbit/s and upstream rates of 3-5 Mbit/s. In a competing architecture, fibre is extended to the edge of a neighbourhood and bonded ADSL or VDSL is used to extend high data rates over copper from the edge to the residence. Rates offered in these hybrid architectures are in the range of 20 Mbit/s to 30 Mbit/s downstream and 1 Mbit/s upstream rates.

I.5 Router delays

(See Table I.1)

Table I.1 – Example of typical delays of contribution by router roles

Role	Average total delay (sum of queuing and processing)	Delay variation
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

(This appendix does not form an integral part of this Recommendation)

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 from 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.

Some IP applications (e.g., IPTV, Internet access) do not involve the full generality of all five segments. These "core-to-LAN" models include the core IP network segment, one access link segment, and one LAN segment.

II.2 Packet loss model

II.2.1 Bursty packet loss model

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 the packet loss rate is high. Such a burst may include sequential lost packets.

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 Link failure model

Link failure is another source of loss in the core network. This leads to sequential 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. Also, for LAN segments a random delay value between 0 and 1.5 ms is added.

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 parameters

Figure II.1 represents the five components of the end-to-end network and the associated modules in the simulation/emulation algorithm. The values from Table 5 through Table 10 and Table 14 provide the inputs to the modules. The outputs of the algorithm are the impairments to be emulated.

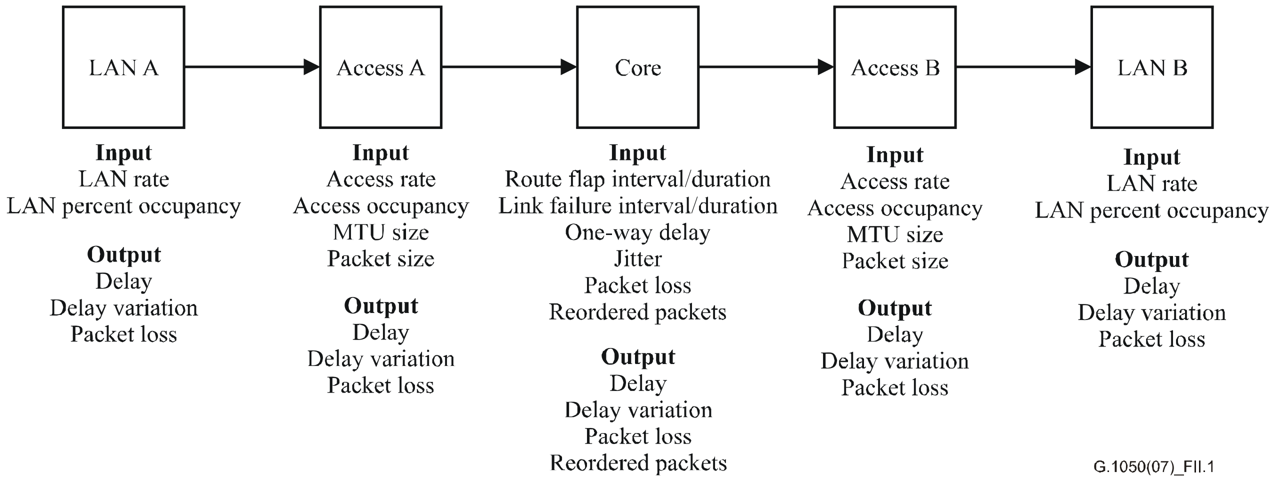

Figure II.1 – Algorithm components

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, 6 and 7:

1) LAN rate. 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 0.000025 × 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 0.0001 × 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 0, and for the high loss state is a random variable uniformly distributed from 0 to 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 value: A = 1 (i.e., no filtering).

II.6.2 Local and remote link segment parameters

1) Link rate. This rate 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 0.0002 × (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 = A × (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. Current value: A = 0.25.
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 value: A = 0.25 (same A as in item 5).
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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Printed in Switzerland
Geneva, 2008