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

International telephone connections and circuits – Apparatus associated with long-distance telephone circuits

Do-no-harm testing

Recommendation ITU-T G.161.1

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

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Recommendation ITU-T G.161.1

Do-no-harm testing

Summary

Recommendation ITU-T G.161.1 defines do-no-harm (DNH) tests for network-based and terminal-based voice quality enhancement (VQE) functions and non-VQE functions.

Conventionally, performance specifications are written for telecommunication equipment to specify their performance characteristics in their intended application. Do-no-harm testing is a concept that has been introduced to specify tests which may be carried out to evaluate situations where the equipment, although meeting its performance specifications in its intended application, may cause some other unexpected performance degradation. Specific DNH tests which relate to "fixed de-jitter buffer operation" and "long-term media path delay stability" are described in the annexes of this Recommendation. Where appropriate, these tests are backed up by case studies which are described in the appendices. It is anticipated that additional annexes covering other DNH scenarios will be added in the future.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T G.161.1	2014-01-13	16	11.1002/1000/12053-en

Keywords

De-jitter buffer; delay stability; do no harm; media path delay; testing; voiceband data; 2100 Hz answering tone.

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FOREWORD

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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/.

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Do-no-harm testing

1 Scope

This Recommendation defines do-no-harm (DNH) tests for network-based and terminal-based voice quality enhancement (VQE) functions and non-VQE functions. These tests are applicable to functions used in both fixed and mobile networks and are independent of transport technology (e.g., TDM, ATM and IP).

Where necessary, this Recommendation takes into account any similar tests that already exist in other Recommendations (e.g., [b-ITU-T G.160], [b-ITU-T G.168] and [b-ITU-T G.169]) and is applicable to both stand-alone VQE and non-VQE functions and VQE and non-VQE functions connected in tandem.

Where possible, the DNH tests defined in this Recommendation use standardized performance evaluation criteria (e.g., [ITU-T P.862] and [b-ITU-T P.863]) and cover all possible in-band audio signals including speech, voiceband data and network signalling tones. Each test has a well-defined test configuration and a set of requirements that allow repeatability and consistency of results between different test laboratories.

The following VQE and non-VQE functions are included in the scope for DNH:

– VQE functions:
  ○ Echo cancellers including comfort noise injection
  ○ Noise reduction/cancellation functions
  ○ Automatic gain/level control
– Non-VQE functions:
  ○ De-jitter buffers
  ○ Codecs
  ○ Voice activity detection
  ○ Silence suppression

Other functions which may be included are:

– Send or ingress (entering the network):
  ○ Microphone array processing (MAP)
  ○ High-frequency encoding (HFE)
  ○ Equalization (EQ)
  ○ Limiting (LM)
– Receive or egress (exiting the network):
  ○ Bandwidth extension (BWE)
  ○ Equalization (EQ)
  ○ Noise compensation (NC; also referred to as ALE)
  ○ Limiting (LM)

This Recommendation provides guidance on how to conduct DNH testing, including case studies, as specific examples.

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 P.501] Recommendation ITU-T P.501 (2012), *Test signals for use in telephonometry.*

[ITU-T P.862] Recommendation ITU-T P.862 (2001), *Perceptual evaluation of speech quality PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs.*

[ITU-T V.8] Recommendation ITU-T V.8 (2000), *Procedures for starting sessions of data transmission over the public switched telephone network.*

[ITU-T V.25] Recommendation ITU-T V.25 (1996), *Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls.*

[ETSI TS 102 929] ETSI TS 102 929 (V2.1.2) (2013-03), *Speech and multimedia Transmission Quality (STQ); Procedures for the identification and selection of common modes of de-jitter buffers and echo cancellers.*

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 answering tone [ITU-T V.25]: The tone transmitted from the called end.

NOTE 1 – The answering tone is an uninterrupted 2100 ± 15 Hz tone with a duration, except when truncated as described in clause 4.3 of [ITU-T V.25], of 3.3 ± 0.7 s.

NOTE 2 – Recommendation [ITU-T V.8] specifies, for the purpose of starting sessions of data transmission, an amplitude modulated answering tone, ANSam.

3.1.2 de-jitter buffer [b-ITU-T G.1020]: A buffer designed to remove the delay variation (i.e., jitter) in packet arrival times. Data is put into the de-jitter buffer at a variable rate (i.e., whenever they are received from the network) and taken out at a constant rate.

3.1.3 phase reversals [ITU-T V.25]: Reversals (180°) in the phase of the answering tone at intervals of 425 to 475 ms. The reversal in phase shall be accomplished such that the phase is within 180 ± 10 degrees in 1 ms, and that the amplitude of the answering tone is not more than 3 dB below its steady state value for more than 400 ms.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 do-no-harm (DNH): DNH is an attribute of a network-based or terminal-based function which states that if the function is included in the transmission path it does not degrade the performance of the media signal (e.g., speech or voiceband data) as measured using a stated metric. Note that depending on the metric, a tolerance may be necessary to allow some measureable degradation that is not significant for the user or application. For example, if the stated metric for voiceband data is the bit error rate (BER), then the requirement for DNH may be "zero increase in

BER when the specified function is enabled". The stated metrics and their associated DNH tolerances for different media types are for further study.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ALE	Automatic Listener Enhancement
ANS	Answering tone
ANSam	Answering tone with amplitude modulation
ATA	Analogue Terminal Adapter
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
BERT	Bit Error Rate Tester
BWE	Bandwidth Extension
DJB	De-Jitter Buffer
DNH	Do No Harm
EQ	Equalization
HFE	High Frequency Encoding
IP	Internet Protocol
LM	Limiting
MAP	Microphone Array Processing
MGW	Media Gateway
NC	Noise Compensation
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
TDM	Time Division Multiplex
VoIP	Voice over Internet Protocol
VQE	Voice Quality Enhancement

5 Conventions

None.

6 General considerations for DNH testing

In general, do-no-harm (DNH) testing may be carried out on equipment at several different times in the product life-cycle, and this may require different test equipment and procedures, although for a given test the underlying test methodology should not change.

1) During product development in the laboratory, including integration of sub-modules into the final product:

- ○ The main goal of DNH testing at this time is to evaluate certain DNH aspects of the performance of equipment, which should take into account anticipated worst-case operational scenarios, and modify the design as necessary. Typically at this stage, some form of network simulator will be used to simulate the expected range of real-life operational parameters.

2) During field-trials, including integration with other equipment in the field, prior to actual in-service operation:

   - ○ The main goal of DNH testing at this time is to confirm certain DNH aspects of the performance of the equipment in a real-world situation, particularly under practical performance conditions which are not always entirely under control, and when integrated with other equipment.

3) During in-service operation

   - ○ The main goal of DNH testing at this time is to diagnose the root cause of performance issues which have been reported on equipment which is in actual use, to help to determine where the fault lies. Testing at this stage may require a combination of testing the equipment under real-life network impairments, as well as in the laboratory under simulated impairments.

7 Structure of do-no-harm tests

Specific DNH tests are described in the annexes of this Recommendation. Where appropriate these tests are backed up by case studies which are described in the appendices. The case studies, which are included for information, describe problems and issues that have been encountered during the use of a particular function and are designed to assist in understanding the background to a particular test.

The arrangement of tests in the annexes allows them to be self-contained, with a clearly defined structure and a set of requirements that allow repeatability and consistency of results between different test laboratories. This set of requirements may or may not include detailed numerical targets.

7.1 Mandatory tests

Unless stated otherwise a test is considered to be mandatory and, as a minimum, an implementation must meet the requirements of a mandatory test to be considered compliant to a particular annex of this Recommendation.

7.2 Optional tests

A test (or a part of a test) can be indicated as "optional". An implementation does not need to meet the requirements of an optional test (or an optional part of a test) to be compliant to a particular annex of this Recommendation.

7.3 Interoperability

There are no interoperability requirements specified in this Recommendation.

Annex A

Fixed de-jitter buffer operation

(This annex forms an integral part of this Recommendation.)

A.1 Introduction

The fixed de-jitter buffer operation test is applicable to any network component or transmission path that contains a de-jitter buffer and is expected to handle voiceband data calls. An adaptive de-jitter buffer should change to a fixed mode on detection of answering tone (ANS) as defined in [ITU-T V.25] and [ITU-T V.8] (see [ETSI TS 102 929]). This tone consists of a 2100 Hz sine wave (with or without phase reversals) and is issued by the *answering* terminal to signal a voiceband data call. On the transmission paths that contain de-jitter buffers and are expected to handle voiceband data calls this tone should be detected and the adaptive de-jitter buffers should be changed to a fixed mode (in each direction of transmission) for the duration of the call.

A.2 Test purpose

A simple objective test is used to confirm that:

– the de-jitter buffer in the component under test operates correctly (i.e., moves to a fixed mode) on detection of the answering tone signal in each direction of transmission. Detection of the answering tone may occur in the component under test or elsewhere in the network;

– the de-jitter buffer remains in a fixed mode for the duration of the call.

These characteristics are checked indirectly by measuring the one-way delay through the component under test as a function of time, and looking for any irregularities.

A.3 Test method

The set-up for this test is shown in Figure A.1. The component under test is connected so that it is possible to set up a call through the component. The component under test contains a de-jitter buffer on the receive side of the A-to-B direction, which is the direction in which the test is performed.

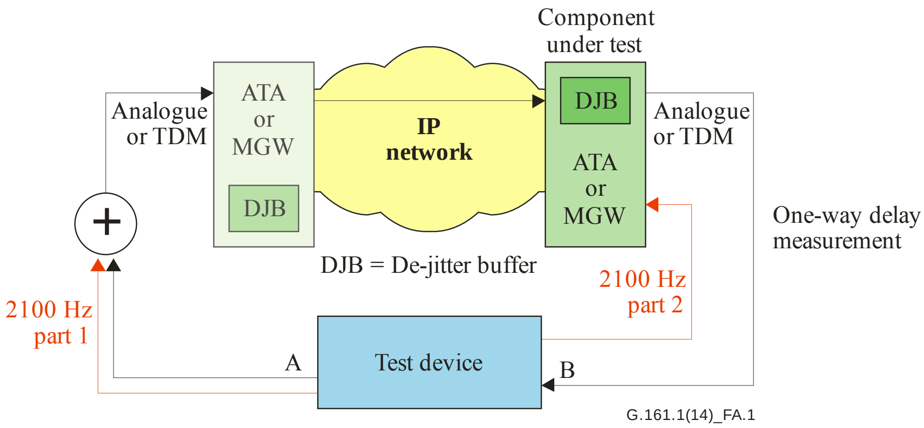

Figure A.1 – De-jitter buffer test configuration

The test device will typically have analogue interfaces. The component under test may have an analogue or time division multiplex (TDM) interface depending on whether it is an analogue terminal adapter (ATA) or media gateway (MGW). In the latter case, analogue-to-digital and digital-to-analogue converters are required between the test device and the component under test. Alternatively, test devices with TDM interfaces can be used to connect directly to the MGW.

A test device that injects a test signal into the network at End A and measures the resulting one-way delay from End A through the network to End B should be used for this test. The test device should also be capable of injecting a 2100 Hz signal (with and without phase reversals) to emulate answering tone in each direction. An example of such a device is one that measures speech quality according to [ITU-T P.862], where the one-way delay is calculated as part of the time alignment process for comparing the original and degraded signals (see clause 10.1.3.2 of [ITU-T P.862]). The test consists of two parts. In Part 1, a 2100 Hz tone is injected in the direction A-to-B and in part 2, a 2100 Hz tone is injected in the direction B-to-A. In both cases, measurements are made in the A-to-B direction only.

The test signal used for the one-way delay measurements should be long enough for a reliable measurement to be made. A typical test signal is a 0.5 seconds burst of White Gaussian Noise, band-limited 300 Hz – 3400 Hz, with a crest factor of 11 dB ± 1 dB, as described in [ITU-T P.501], followed by one second of silence. The silence period is necessary to accommodate calculation of the one-way delay before the next measurement is made. The level of the test signals (including the 2100 Hz tone) should be −20 dBm0.

A.3.1 Part 1 – De-jitter buffer operation with 2100 Hz injected in direction A-to-B

The test device should be configured to make a test call and, following call set-up, a repetitive sequence of test signals is used to make a series of one-way delay measurements in the direction A-to-B. Five measurements are considered to be sufficient to allow for any initial de-jitter buffer adjustments at the start of the call. After these five measurements, a 2100 Hz Answer Tone signal is injected. This should cause the de-jitter buffer to transition into fixed mode operation and a further sequence of 20 measurements are then made to ensure that the buffer remains fixed and the one-way delay remains stable over a longer period of time. After these 20 measurements, a 20-second period of silence is inserted to confirm the de-jitter buffer does not revert to adaptive mode operation. A further five measurements are performed to confirm this and the call is cleared down. This sequence of events is shown in Figure A.2.

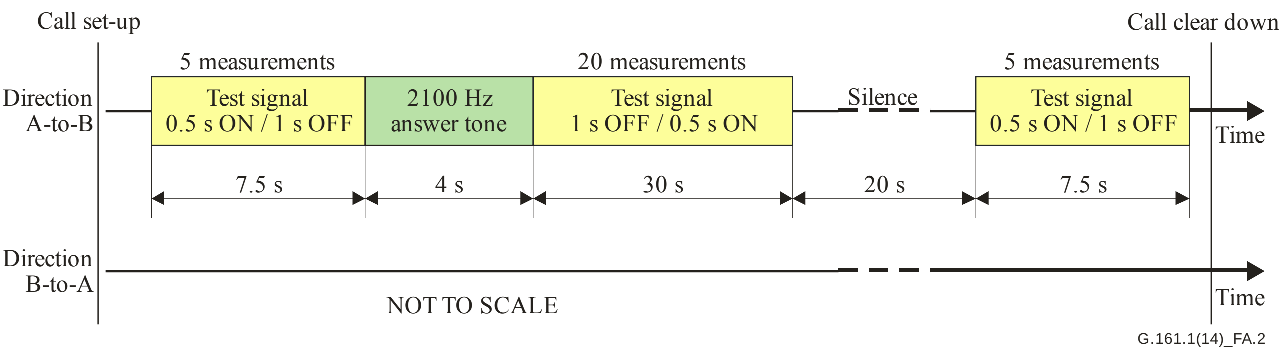

Figure A.2 – Part 1 test sequence

The one-way delay measurements, which include the delay through the de-jitter buffer in the network component under test, should be plotted against time.

A.3.2 Part 2 – De-jitter buffer operation with 2100 Hz injected in direction B-to-A

The test device should be configured to make a test call and, following call set-up, a repetitive sequence of test signals is used to make a series of one-way delay measurements in the direction A-to-B. Five measurements are considered to be sufficient to allow for any initial de-jitter buffer adjustments at the start of the call. After these five measurements, a 2100 Hz Answer Tone signal is injected in the direction B-to-A. This should cause the de-jitter buffer to transition into fixed mode operation and a further sequence of 20 measurements are then made in the A-to-B direction to ensure that the buffer remains fixed and the one-way delay remains stable over a longer period of time. After these 20 measurements, a 20-second period of silence is inserted to confirm the de-jitter buffer does not revert to adaptive mode operation. A further five measurements are performed to confirm this and the call is cleared down. This sequence of events is shown in Figure A.3.

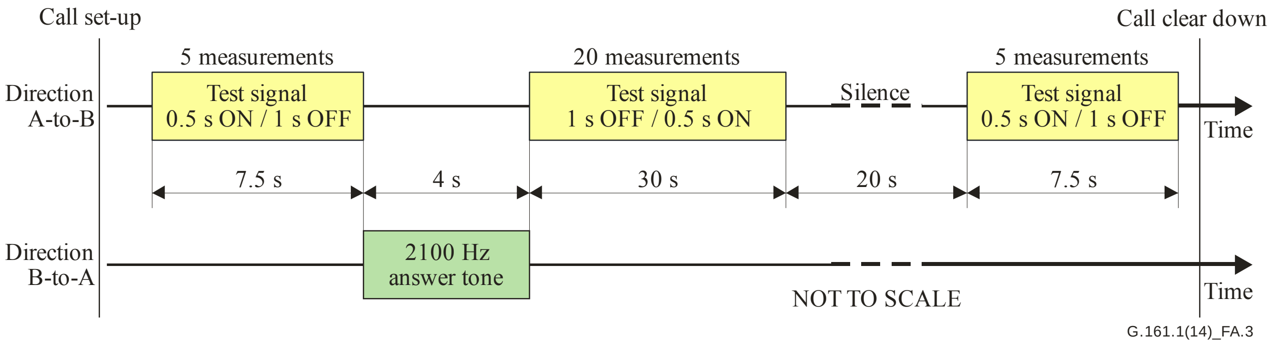

Figure A.3 – Part 2 test sequence

The one-way delay measurements, which include the delay through the de-jitter buffer in the network component under test, should be plotted against time. This test can be considered an end-to-end test and any variations in the one-way delay that are not due to fixing of the de-jitter buffer may be due to either the component under test or any other part of the end-to-end transmission path, examples of which are given in clause B.5.

A.4 Expected results

The measured one-way delay will depend on several factors:

a) the setting of the fixed de-jitter buffer

b) the delay through the network components, including any packetization time, processing time, transmission time, etc.

Taking these factors into account, it should be possible to estimate the expected value of the one-way delay with a fixed de-jitter buffer. This value should be used for the requirement below.

An example set of expected test results is shown in Figure A.4. The graph represents a plot of all delay measurements made for the duration of the test.

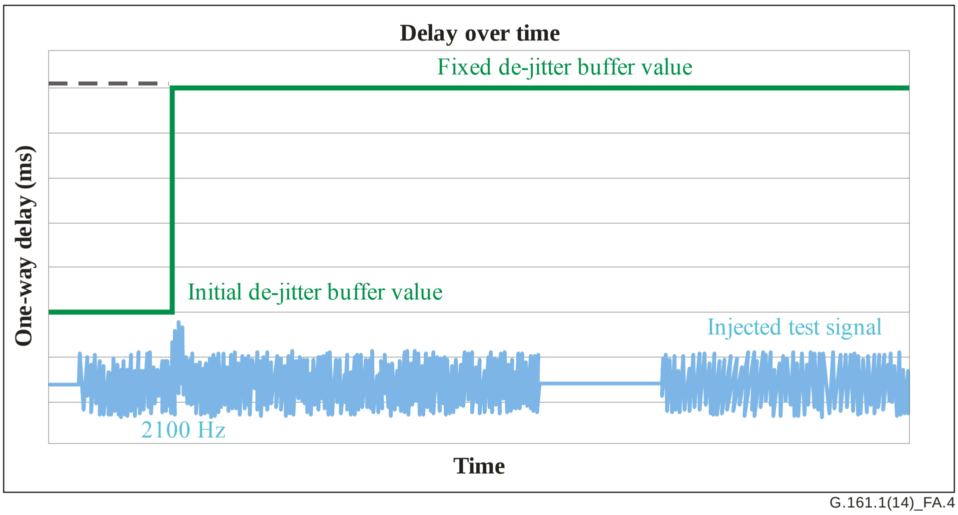

Figure A.4 – Expected test results

NOTE – The initial and fixed de-jitter buffer values are likely to be implementation dependent.

A.5 Requirements

1) Following injection of 2100 Hz Answer Tone in Part 1 and Part 2 of this test, the measured one-way delay should be the expected one-way delay for a fixed de-jitter buffer.

2) The measured one-way delay should remain constant (within the accuracy of the measurement device and any jitter associated with the receive sample clock) for the duration of the call.

Annex B

Long-term media path delay stability

(This annex forms an integral part of this Recommendation.)

B.1 Introduction

The long-term media path delay stability test is applicable to any network component or transmission path that contains a de-jitter buffer and is designed to carry voiceband data signals. It should be performed after successful completion of the "fixed de-jitter buffer operation" test in Annex A. Voiceband data applications are sensitive to fluctuations in the media path delay during a call. For this reason, network components that contain a de-jitter buffer move the buffer from an adaptive to a fixed mode on detection of answering tone (ANS) as defined in [ITU-T V.25] and [ITU-T V.8] (see [ETSI TS 102 929]). Whilst the function of the de-jitter buffer is to "absorb" end-to-end delay variation it is possible that certain processes within the component could introduce delay variation after the signal has egressed the de-jitter buffer.

B.2 Test purpose

The purpose of this test is to measure long term media path delay stability to verify that the overall quality of the media path is adequate to support voiceband data applications. In particular to ensure that processes within the component under test do not introduce delay variation to signals after egressing the de-jitter buffer output. A simple objective test is used to measure end-to-end media path delay as a function of time, and to demonstrate stability over time.

B.3 Test method

The component under test is connected so that it is possible to set up a call through the component as shown in Figure B.1. The de-jitter buffer is either configured for fixed mode operation or in the case of an adaptive de-jitter buffer, it is forced into fixed mode by injecting a 2100 Hz tone.

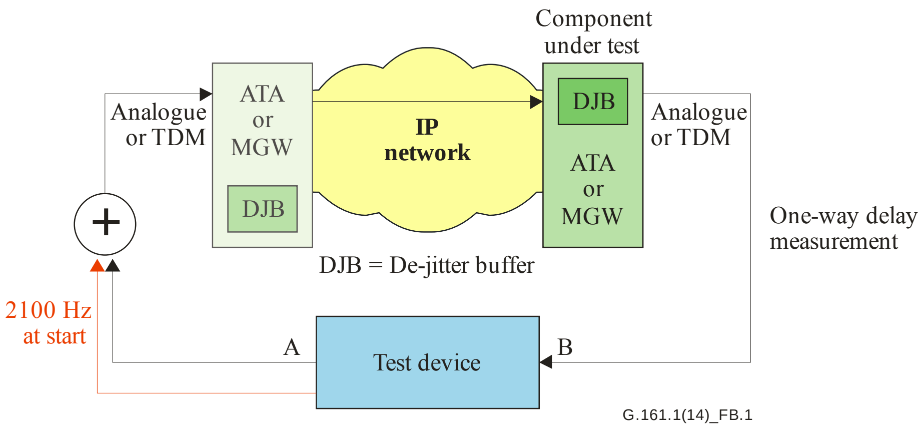

Figure B.1 – Long term media path delay stability test configuration

The test device will typically have analogue interfaces. The component under test may have an analogue or TDM interface depending on whether it is an ATA or MGW. In the latter case, analogue-to-digital and digital-to-analogue converters are required between the test device and the component under test. Alternatively, test devices with TDM interfaces can be used to connect directly to the MGW.

A test device that injects a test signal into the network at End A and measures the resulting one-way delay from End A through the network to End B should be used for this test. The test device should also be capable of injecting a 2100 Hz signal to emulate answering tone. An example of such a device is one that measures speech quality according to [ITU-T P.862], where the one-way delay is calculated as part of the time alignment process for comparing the original and degraded signals (see clause 10.1.3.2 of [ITU-T P.862]). The device should be configured to make a long duration test call (at least two hours is recommended). Once the call is established a 2100 Hz signal is injected in the A-to-B direction to cause the B-end de-jitter buffer to transition to fixed mode operation. A test signal is then used to make a repetitive series of end-to-end delay measurements in the direction A-to-B.

This test signal should be relatively short in duration so that a large number of measurements are made in quick succession throughout the call. However, the test signal also needs to be long enough for a reliable one-way delay measurement to be made. A typical test signal is a 0.5 seconds burst of White Gaussian Noise, band-limited 300 Hz – 3400 Hz, with a crest factor of 11 dB ± 1 dB, as described in [ITU-T P.501], followed by 1 second of silence. The silence period is necessary to accommodate calculation of the one-way delay before the next measurement is made. The level of the test signals (including the 2100 Hz tone) should be –20 dBm0.

The call is cleared down once the required test duration has been achieved. This sequence of events is shown in Figure B.2.

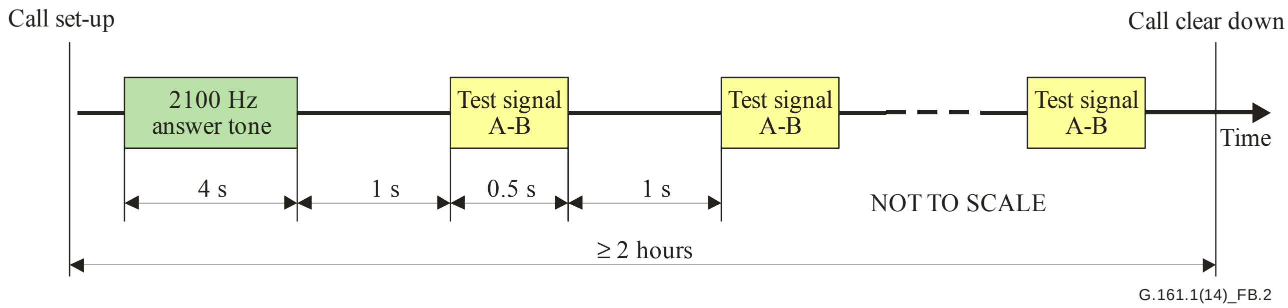

Figure B.2 – Test sequence

The one-way delay measurements, which include the delay through the de-jitter buffer in the network component under test, should be plotted against time. This test can be considered an end-to-end test and any variations in the one-way delay may be due to either the component under test or any other part of the end-to-end transmission path.

B.4 Expected results

This test is not concerned with the *absolute* delay, but with demonstrating that the media path delay remains stable over time, without any random or periodic fluctuations in delay.

An example set of expected test results, for a stable end-to-end media path is shown in Figure B.3. The graph represents a plot of all delay measurements made for the duration of the test.

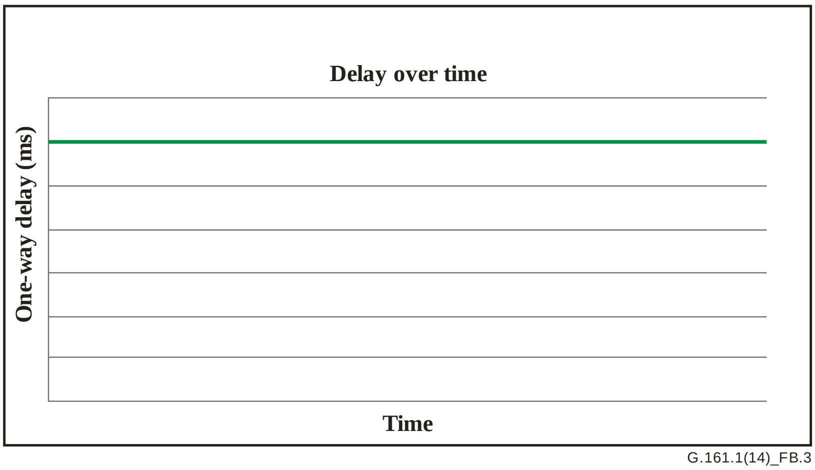

Figure B.3 – Expected test results

B.5 Unacceptable delay variation on the end-to-end path

Random or periodic fluctuations in media path delay can have many causes. The example results shown in Figure B.4 are indicative of an unacceptable test outcome. This shows a periodic fluctuation in media path delay (this particular example, typified by a saw-tooth profile, is indicative of a lack of end-to-end synchronisation).

NOTE – Assuming synchronised clocks are provided at each end, a lack of end-to-end synchronisation may be a result of the relevant components not accepting the synchronised clock source. In the case of a test device that has a digital interface (e.g., time division multiplex (TDM)), a synchronised clock source will also need to be provided to the test device itself.

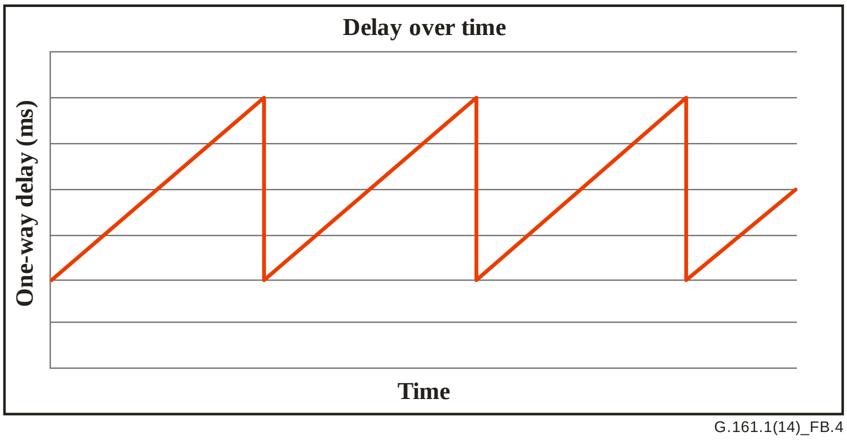

Figure B.4 – Unacceptable test results (1)

The example results shown in Figure B.5 are also indicative of an unacceptable test outcome. This shows very small (e.g., 100 s of microseconds) occasional fluctuations in media path delay (typical of de-jitter buffer play-out or the digital to analogue conversion process being disrupted by some higher priority process being executed by the component).

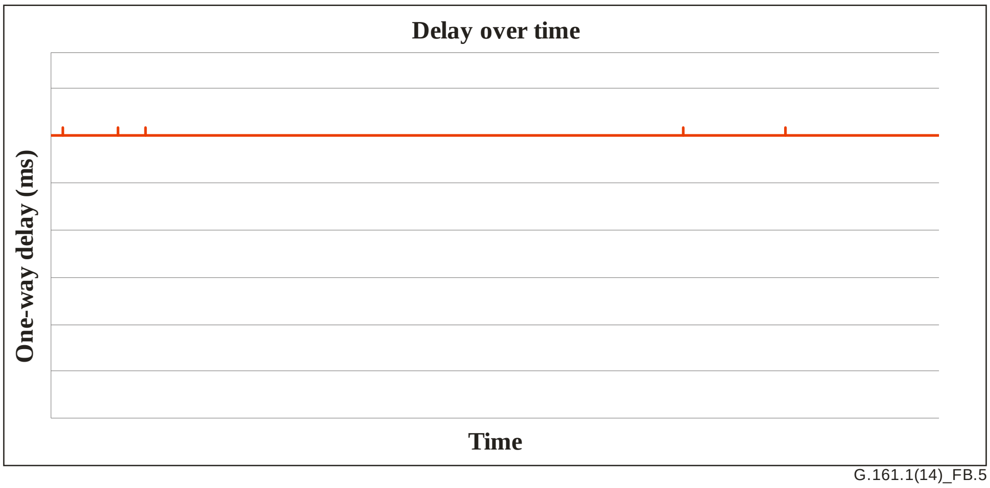

Figure B.5 – Unacceptable test results (2)

B.6 Requirements

1) The measured one-way delay should remain constant (within the accuracy of the measurement device and any jitter associated with the receive sample clock) for the duration of the call, with no random or periodic fluctuations.

Appendix I

Case study relating to the performance of voiceband data applications over a Voice over Internet Protocol (VoIP) service

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

I.1 Background

One organisation has conducted end-to-end tests as part of its roll-out of a Voice over Internet Protocol (VoIP) service. This is achieved by plugging a conventional telephone into an analogue telephony adapter (ATA). The voice signal is converted to IP by the ATA and carried over a QoS-enabled network to its destination, which could be either another ATA-hosted customer, or via a trunk media gateway to a customer connected to the conventional PSTN.

The voice service offered includes the possibility of customers connecting a modem, fax machine or text telephone into the analogue port of the ATA, and the expectation is that functionality and performance will be similar to that of the PSTN. Voiceband data is carried in-band. Figure I.1 shows an overall diagram of this scenario.

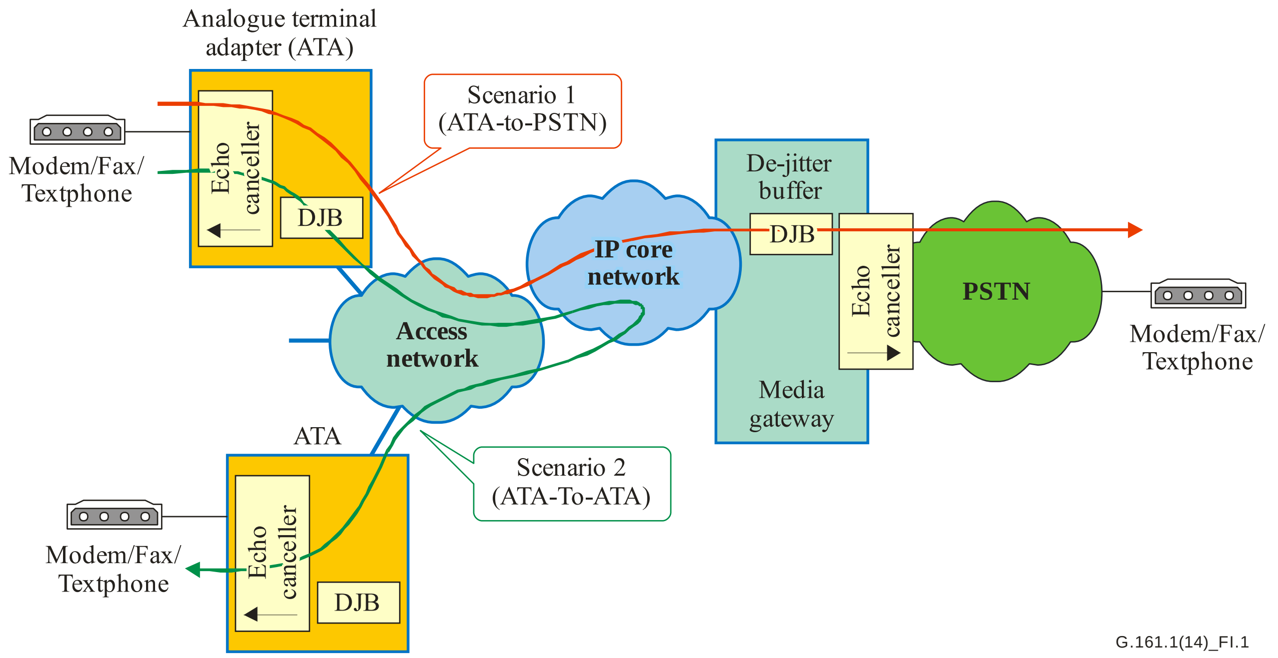

Figure I.1 – Diagram showing new voice service scenario

I.2 The problem

End-to-end tests were performed on Scenarios 1 and 2 in Figure I.1 to evaluate the performance of different types of modem, fax machine and text telephone. The result of these tests showed occasional bursts of errors for certain modems which did not occur on benchmark PSTN connections. A typical error report is given in Figure I.2, where the number of individual bit errors are plotted against the time they occur.

To investigate the cause of these problems, it was necessary to repeat the test in a controlled environment in the laboratory.

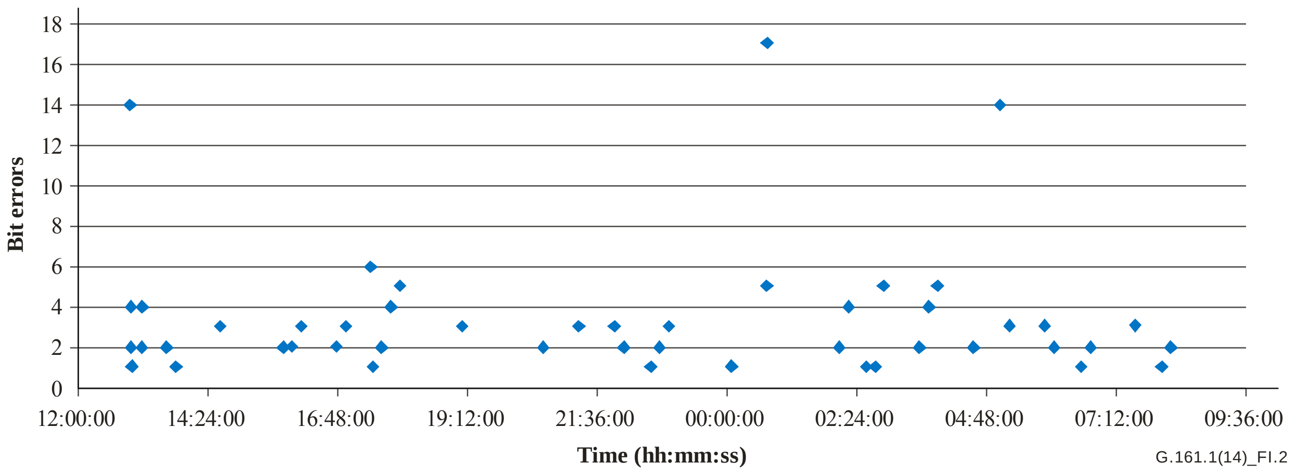

Figure I.2 – Modem error report for an ITU-T V.21 modem with no answer tone

I.3 Test set-up

The simplest set-up consists of two ATAs connected back-to-back as shown in Figure I.3. The Session Initiation Protocol (SIP) server allows calls to be made between the two ATAs and also acts as a layer 2 switch to provide IP connectivity and hence a media path.

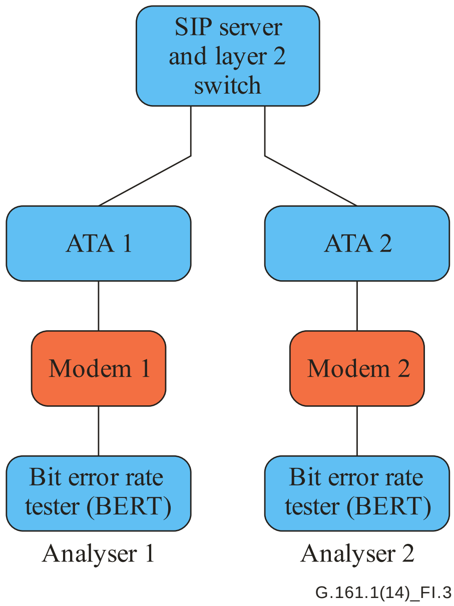

Figure I.3 – Test set-up using modems

Repeating the modem test using the test set-up of Figure I.3 gave similar results to those shown in Figure I.2. The modems were then replaced with a sine wave generator at one end and a waveform capture tool at the other, as shown in Figure I.4. A call was set up and a sine wave was injected for a long period of time.

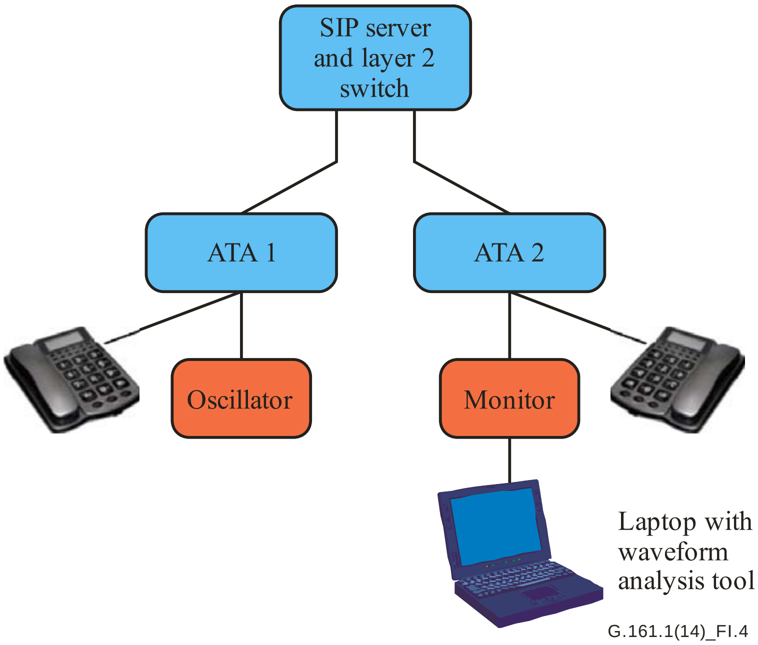

Figure I.4 – Test set-up using a signal generator

Analysis of the captured waveform at the receive end of the connection showed occasional corruption of the sine wave as shown in Figures I.5 and I.6. Such distortions are easy to spot on a sine wave, but may not be so easy to spot on a complex modem signal.

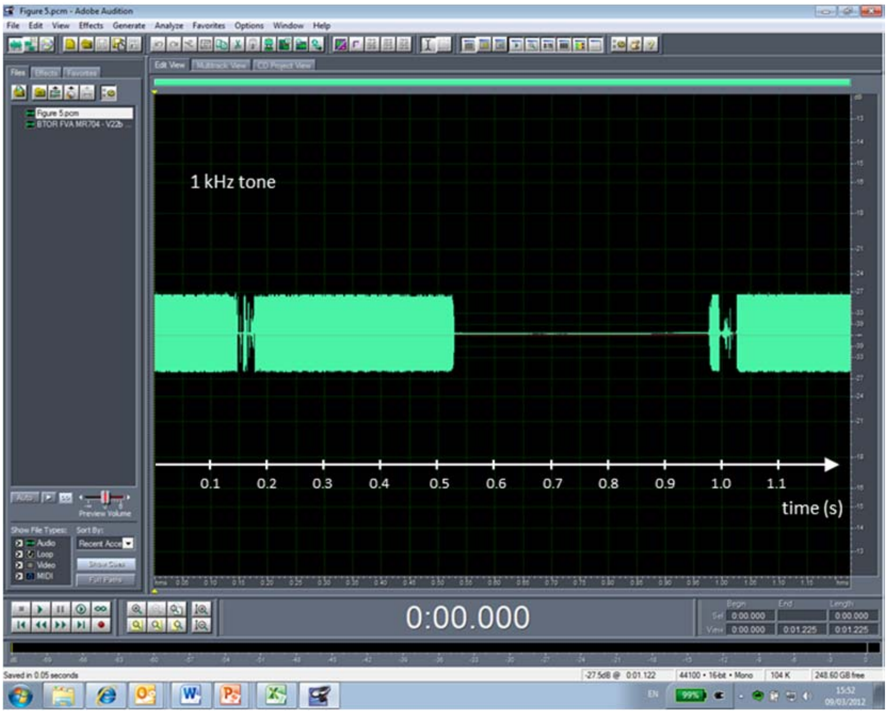

Figure I.5 – Received signal waveform showing distortion 1

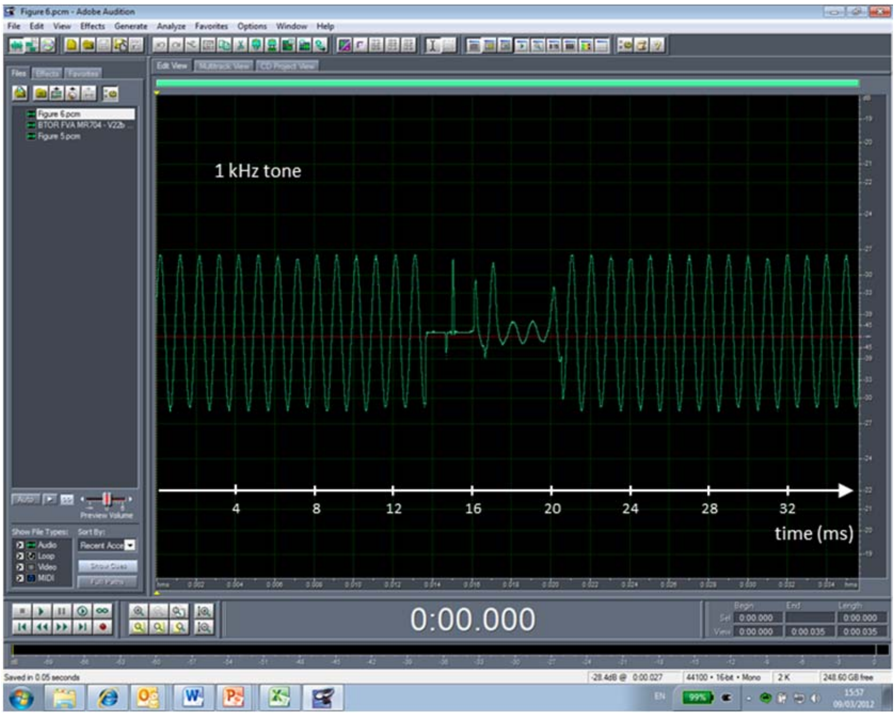

Figure I.6 – Received signal waveform showing distortion 2

From this, it was suspected that disturbances in the media path that led to corruptions to the sine wave test signal could also be responsible for the bit errors experienced by the modems.

I.4 The cause of the problem

These results were fed back to the ATA vendor who was able to improve performance, although the root cause of the problem is not clear. Possible causes could be:

– De-jitter buffer adaptations causing changes to the end-to-end delay

– Packet loss (packet loss concealment was not enabled)

– Timing problems (including synchronization within the new network)

– Interruptions to processing the RTP due to poor prioritization in the allocation of resources.

I.5 Summary and conclusions

This case study has shown that a modem signal traversing a network was suffering "harm" in the form of bit errors. These bit errors were replicated using a simple laboratory test set-up. The modems were then replaced with a simple sine wave generator and some waveform analysis software. This set-up was able to detect corruptions to the media path which were causing harm to the modem signals.

In summary:

– For the purpose of demonstrating "harm", a simple laboratory test set-up was able to replicate a more complex network scenario (modems showed similar bit errors on both)

– A simple sine wave generator, together with some waveform analysis software, replaced the need for more complex analysis of modem signals

– It was necessary to treat the ATA as a "black box" since the end-to-end network performance was of interest and it was not possible to separate out the individual signal processing functions (echo canceller, de-jitter buffer, etc.)

– Compliance to relevant standards (e.g., [b-ITU-T G.168]) is not necessarily sufficient to guarantee overall end-to-end performance.

Bibliography

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