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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

Voice enhancement devices

Recommendation ITU-T G.160

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Recommendation ITU-T G.160

Voice enhancement devices

Summary

Recommendation ITU-T G.160 applies to the characteristics, performance and testing of voice enhancement devices (VEDs) for use in digital network-based equipment. Such devices are commonly found in mobile networks and this version of this Recommendation is only applicable to mobile networks.

VEDs are defined as certain signal processing network functions (SPNFs) (such as noise reduction and acoustic echo control) in the digital transmission path that perform voice enhancement functions on voiceband signals that traverse mobile networks.

Voice enhancement functions include the control of acoustic echo generated by wireless handsets, noise reduction, and the recognition and accommodation of tandem free operation (TFO) and interworking function (IWF) signals.

The current version of this Recommendation does not include certain tests that are still under study. In particular, it does not contain any objective tests covering voice quality performance of noise reduction functions, although a placeholder for such a test is given. In the meantime, the reader should be aware that voice quality performance of noise reduction functions is not addressed by this version of this Recommendation.

Appendix II presents an objective methodology for characterizing the basic effect of noise reduction (NR) methods and is accompanied by a reference ANSI-C source code. Three objective measures are specified for characterizing NR solutions. Recommendation target value ranges in specified conditions are given for the measures to serve as basic guidelines for the proper functioning of NR methods.

This Recommendation includes an electronic attachment containing:

- Reference ANSI-C source code and test vectors for implementing Appendix II for noise suppression performance measurement.

History

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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.

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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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Electronic attachment: Reference ANSI-C source code and test vectors for implementing Appendix II for noise suppression performance measurement.

Recommendation ITU-T G.160

Voice enhancement devices

1 Scope

This Recommendation applies to the characteristics, performance and testing of voice enhancement devices (VEDs) for use in digital network-based equipment. Such devices are commonly found in mobile networks, and this version of the Recommendation is only applicable for mobile networks.

VEDs are defined as certain signal processing network functions (SPNFs) (such as noise reduction and acoustic echo control) in the digital transmission path that perform voice enhancement functions on voiceband signals that traverse mobile networks. Possible locations for VEDs located in mobile networks are shown in Appendix I.

Voice enhancement functions covered by this Recommendation include the control of acoustic echo generated by wireless handsets, noise reduction, and the recognition and accommodation of tandem free operation (TFO) and interworking function (IWF) signals.

The purpose of this Recommendation is to define certain performance constraints for VEDs and, where appropriate, to define laboratory tests that may be performed on a VED to verify that these constraints are met.

A VED that satisfies all the constraints and passes all of the tests in this Recommendation should ensure (but will not guarantee) that the pre-existing standard of the overall network performance (e.g., the transmission of speech, voiceband data and other voiceband signals, ISDN, etc.) is not degraded when the VED is installed in the network.

This Recommendation also describes the general characteristics of VEDs and indicates what characteristics are important to provide acceptable performance in the network. Issues relating to the interaction of VEDs with other network and subscriber equipment are also discussed in this Recommendation.

This Recommendation does not define a standard algorithm for voice enhancement functions. Nor does it apply to either electrical echo control (governed by [ITU-T G.168]) or automatic level control (governed by [b-ITU-T G.169]).

Note that VEDs may be incorporated into other network equipment, such as the transcoder or electrical echo canceller. In this case, the performance requirements of this Recommendation refer to the performance of the VED only.

Note that some terminal devices (e.g., mobile stations) incorporate voice enhancement functions (acoustic echo control, noise reduction, etc.) that are similar to those performed by the VED. This Recommendation does not apply to functions contained in these terminal devices.

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.164] Recommendation ITU-T G.164 (1988), *Echo suppressors*.
- [ITU-T G.168] Recommendation ITU-T G.168 (2012), *Digital network echo cancellers*.
- [ITU-T P.50] Recommendation ITU-T P.50 (1999), *Artificial voices*.
- [ITU-T P.501] Recommendation ITU-T P.501 (2000), *Test signals for use in telephony*.
- [ITU-T P.800] Recommendation ITU-T P.800 (1996), *Methods for subjective determination of transmission quality*.
- [ITU-T P.830] Recommendation ITU-T P.830 (1996), *Subjective performance assessment of telephone-band and wideband digital codecs*.
- [ITU-T P.835] Recommendation ITU-T P.835 (2003), *Subjective test methodology for evaluating speech communication systems that include noise suppression algorithm*.
- [ITU-T Q.24] Recommendation ITU-T Q.24 (1988), *Multifrequency push-button signal reception*.
- [ITU-T Q.143] Recommendation ITU-T Q.143 (1988), *Specifications of signalling system No. 5 – Line signal sender*.
- [ITU-T Q.144] Recommendation ITU-T Q.144 (1993), *Specifications of signalling system No. 5 – Line signal receiver*.
- [ITU-T Q.271] Recommendation ITU-T Q.271 (1988), *Specifications of signalling system No. 6 – Section 5: Continuity check of the speech path*.
- [ITU-T Q.724] Recommendation ITU-T Q.724 (1988), *Specifications of signalling system No. 7 – Signalling procedures*.
- [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.18] Recommendation ITU-T V.18 (2000), *Operational and interworking requirements for DCEs operating in the text telephone mode*.
- [ITU-T V.21] Recommendation ITU-T V.21 (1988), *300 bits per second duplex modem standardized for use in the general 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*.
- [ITU-T V.27 ter] Recommendation ITU-T V.27 ter (1988), *4800/2400 bits per second modem standardized for use in the general switched telephone network*.
- [ITU-T V.32] Recommendation ITU-T V.32 (1993), *A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits*.

- [ETSI TS 128 062] ETSI TS 128 062 V7.0.0 (2007), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Inband Tandem Free Operation (TFO) of speech codecs; Service description; Stage 3*.
<http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=26667>
- [IEC 61260] IEC 61260:1995, *Electroacoustics – Octave-band and fractional-octave-band filters*. <<http://webstore.iec.ch/webstore/webstore.nsf/artnum/019426>>
- [TIA-825-A] ANSI/TIA-825-A (2003), *A Frequency Shift Keyed Modem for Use on the Public Switched Telephone Network*.
< <http://tia.nufu.eu/std/TIA-825-A>>
- [TIA-895-A] ANSI/TIA-895-A (10/2002), *CDMA Tandem Free Operation*.
<<http://tia.nufu.eu/std/TIA-895-A>>

3 Terms and definitions

This Recommendation defines the following terms:

- 3.1 acoustic echo:** Acoustic echo is the reflected signal resulting from the acoustic path between the earphone/loudspeaker and microphone of a hand-held or hands-free mobile station.
- 3.2 disabled VED:** A VED that has been forced into a 64 kbit/s bit-sequence integrity mode of operation, by whatever means.
- 3.3 echo canceller (EC):** A device placed in the 4-wire portion of a circuit and used for reducing cancelled-end echo present on the send path by subtracting an estimation of that echo from the cancelled-end echo.
- 3.4 encoded acoustic echo:** Encoded acoustic echo is the acoustic echo signal that has passed through the codecs and any other signal processing in the mobile network.
- 3.5 laboratory tests:** Tests which may be performed in a laboratory or any environment where the VED is not installed in the network and is therefore not 'in service'.
- 3.6 noise:** For the purposes of this Recommendation, noise is defined as a slowly varying stochastic process appearing additive to the desired speech signal. Specifically, the variations in the characteristics of the noise process are such that it can be considered approximately stationary over much longer time intervals than a typical speech signal.
- 3.7 signal processing network device (SPND):** A physical entity (e.g., chipset) that contains one or more signal processing network functions (SPNFs).
- 3.8 signal processing network equipment/element (SPNE):** A stand-alone physical entity that supports one or more signal processing network functions (SPNFs). It may also be controlled internally and/or externally and managed through a network management system which contains managed objects, a management communications function and a management application function. SPNEs include, e.g., echo cancellers, voice enhancement devices, circuit multiplication equipment, voice gateways.

NOTE 1 – Examples of managed objects are: receive/transmit port, power supply, plug-in cards, SPNFs. The communications function provides facilities for the transport of telecommunications management network (TMN) messages to and from the management application function, as well as facilities for the transit of messages. The message communications function does not originate or terminate messages. A management application function is the origin and termination for all TMN messages.

NOTE 2 – An SPNE in this context is a combination of hardware and software that performs SPNFs.

- 3.9 signal processing network function (SPNF):** A function within a physical entity (e.g., SPNE, SPND) that performs signal processing to provide support or services to the transport network and/or to the users. Examples include electric or acoustic echo control, noise reduction, automatic level control, digital speech interpolation, low-rate encoding, transcoding.



Documents

Rec. ITU-T G.160 (06/2012)

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3.10 voice enhancement device (VED): VEDs are defined as certain SPNFs (such as noise reduction and acoustic echo control) in the digital transmission path that perform voice enhancement functions on voiceband signals that traverse communication networks.

3.11 voice enhancement: To be defined.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations:

A/D	Analogue-to-Digital converter
AEC	Acoustic Echo Control
ALC	Automatic Level Control
AMR	Adaptive Multi-Rate
BER	Bit Error Rate
BSC	Base Station Controller
BSS	Base Station Subsystem
BTS	Base Transceiver Station
CER	Character Error Rate
CN	Core Network
CSS	Composite Source Signal
CTM	Cellular Text Modem
D/A	Digital-to-Analogue converter
DCME	Digital Circuit Multiplication Equipment
DTMF	Dual-Tone Multi-Frequency
EC	Echo Canceller
ERL	Echo Return Loss
GSM	Global System for Mobile communication
ISDN	Integrated Services Digital Network
Iub	Interface between a radio network subsystem and a Node B
Iu-cs	The physical instance of Iu (Interconnection point between the Radio Network Subsystem and Core Network) towards the Circuit-Switched Service Domain of the core network
IWF	Interworking Function
LBR	Low Bit Rate
MG	Media Gateway (CDMA-2000)
MGW	Media Gateway (3GPP)
MS	2G Mobile Station
MSC	2G Mobile Switching Centre
MSIN	Mobile Station In
Nb	3G internal mobile network interface

Node-B	3G network component which serves one cell
NPLR	Noise Power Level Reduction
NR	Noise Reduction
NSS	Network and Switching Subsystem
OMC	Operation and Maintenance Control
PSTN	Public Switched Telephone Network
RAN	Radio Access Network
RMS	Root Mean Square
RNC	Radio Network Controller
SNR	Signal-to-Noise Ratio
SNRI	Signal-to-Noise Ratio Improvement
SPND	Signal Processing Network Device
SPNE	Signal Processing Network Equipment
SPNF	Signal Processing Network Function
TC	Transcoder (3GPP)
TCE	Transcoder Equipment
TDM/PCM	Time Division Multiplex Interface
TFO	Tandem Free Operation
TNLR	Total Noise Level Reduction
TRAU	2G/3G Transcoder Unit
TrFO	Transcoder Free Operation
UE	3G User Equipment
UMSC	3G Mobile Switching Centre
UTRAN	3G Radio Access Network
VBD	Voiceband data
VED	Voice Enhancement Device

5 Characteristics of VEDs

5.1 General

VEDs are defined as certain SPNFs (such as noise reduction and acoustic echo control) in the digital transmission path that perform voice enhancement functions on voiceband signals that traverse mobile networks. Voice enhancement functions include the control of acoustic echo (AEC) generated by wireless handsets, noise reduction (NR), and the recognition and accommodation of tandem free operation (TFO) and interworking function (IWF) signals.

If a VED is provisioned OFF, the delay in the processing path (excluding any network interface delays) should not exceed 0.5 ms.

When a VED is provisioned ON, the delay in the processing path should be kept to a minimum.

A VED may incorporate processing in both directions of transmission. However, if there is no processing present in a certain direction, then the delay in that path (excluding any network interface delays) should not exceed 0.25 ms.

NOTE 1 – VED AEC or NR functions occurring together or with other SPNFs may be algorithmically coupled. In this case, the OFF processing delay requirement applies when all such coupled SPNFs are OFF.

NOTE 2 – A VED is expected to be provisioned OFF by action of the network administration or by a network coordination mechanism at call set-up, and should be performed before the answer side goes off-hook to prevent possible step delay changes during an active call.

NOTE 3 – [b-ITU-T G.161] gives advice on the interaction of signal processing functions, such as AEC and NR and recommends their relative placement in networks.

5.2 Acoustic echo control (AEC)

Acoustic echo control is a signal processing function located in the digital transmission path, which has the objective of improving subjective speech quality by reducing encoded acoustic echo. A handset may inject acoustic echo into a digital mobile telephony system which is transmitted together with signals originating from mobile handsets to the location of the VED through the air interface, speech codecs and other signal processing and transport functions of the mobile network.

An acoustic echo control function has the following general characteristics:

- The ability to reduce acoustic echo generated in the mobile handsets.
- The ability to maintain quality of the signals originating from mobile handsets.
- The ability not to corrupt voice-band data or facsimile signals.
- The ability not to interfere with in-band network signalling tones.
- Exhibit low throughput delay.
- Provide 64 kbit/s bit-sequence integrity when disabled.

5.2.1 External disabling/enabling

The AEC function should be disabled during voice-band data and facsimile transmissions. The AEC function may be disabled by an external digital signal. If external disabling is supported, the AEC function should provide 64 kbit/s bit-sequence integrity in the externally disabled state.

5.2.2 In-band disabling

AEC functions covered by this Recommendation should be equipped with a tone disabler that conforms to clause 5 of [ITU-T G.164]. Additionally, the tone disabler should meet the requirements of [ITU-T G.164] when the disable tone is phase and/or amplitude modulated in accordance with [ITU-T V.25] and [ITU-T V.8]. When disabled, the AEC function should provide 64 kbit/s bit-sequence integrity.

5.3 Noise reduction (NR)

Noise reduction is a signal processing function located in the digital transmission path, which has the objective of reducing the noise level, thereby aiming to improve subjective speech quality as perceived by humans.

Noise reduction functions have the following general characteristics:

- The ability to modify a noise-corrupted voice signal, so as to improve the subjective quality.
- The ability to maintain the quality of the voice signal when it is not corrupted by noise.
- The ability not to corrupt voice-band data or facsimile signals.
- The ability not to interfere with in-band network signalling and information tones.

- Exhibit low throughput delay.
- Provide 64 kbit/s bit-sequence integrity when disabled.

5.3.1 External disabling/enabling

The noise reduction function should be disabled during voice-band data and facsimile transmissions. The NR function may be disabled by an external digital signal. If external disabling is supported, the noise reduction function should provide 64 kbit/s bit-sequence integrity in the externally disabled state.

5.3.2 In-band disabling

Noise reduction functions covered by this Recommendation should be equipped with a tone disabler that conforms to clause 5 of [ITU-T G.164]. Additionally, the tone disabler should meet the requirements of [ITU-T G.164] when the disable tone is phase and/or amplitude modulated in accordance with [ITU-T V.25] and [ITU-T V.8]. When disabled, the noise reduction function should provide 64 kbit/s bit-sequence integrity.

5.4 Tandem free operation (TFO)

A VED that encounters TFO signals should support tandem free operation according to [ETSI TS 128 062] and [TIA-895-A]. Once in TFO mode, a VED may optionally choose to invoke speech enhancement functions, causing the termination of TFO. The decision should be based on whether TFO or speech enhancement techniques will provide better overall speech quality. The evaluation of overall speech quality on which this decision is based is for further study. The following factors should also be taken into consideration:

- The decision to terminate TFO should take into account the apparent noise generated by the TFO signalling in the LSBs;
- VEDs should not terminate TFO for AMR wideband calls; and therefore
- VEDs need to detect TFO signalling messages in order to determine the codec type;
- VEDs should take care to avoid frequent switching between TFO and non-TFO.

NOTE – There is a need to have interoperability of different TFO standard Recommendations. The interoperability of TFO techniques described in [ETSI TS 128 062] and [TIA-895-A] is for further study.

5.5 Interworking functions (IWF)

Under study.

6 Test signals

6.1 General

For noise reduction devices, it is imperative that the speech and noise models used for evaluation represent the real-world conditions in which the noise reduction device is expected to operate. For example, in a wireless environment, the additive noise can be highly non-stationary. In [b-Budagavi] it was found that simple additive white noise is turned into harmonic-related signals by mobile-radio speech coders. Additionally, harmonically rich background sounds can cause wavering or warbling in the output speech signal.

6.2 Measuring input and output signal levels

Unless otherwise noted, for the tests in this Recommendation, the input and output signals are measured using an RMS method. The power of the signal, $S(k)$, is measured in dBm0 using the following equations for A-law and μ -law encoding:

$$S(k) = \begin{cases} 3.14 + 20 \log \left[\frac{\sqrt{\frac{2}{n} \sum_{i=k}^{k-n+1} x_i^2}}{4096} \right], & \text{for A-law encoding} \\ 3.17 + 20 \log \left[\frac{\sqrt{\frac{2}{n} \sum_{i=k}^{k-n+1} x_i^2}}{8159} \right], & \text{for } \mu\text{-law encoding} \end{cases}$$

Where x_i is the linear value of the encoded signal, k is the discrete time index, and n is the number of samples over which the measurement is made.

6.3 Gaussian white noise

Where Gaussian white noise is specified as the test signal in this Recommendation, it shall have the following characteristics, unless otherwise stated: Gaussian white noise, band-limited 300 Hz – 3400 Hz, with a crest factor of 11 dB $\pm$ 1 dB.

7 Tests

Each of the tests described in this clause is designed to evaluate different aspects of VED performance. Some tests should be performed on the overall VED, whereas others are designed to test specific functions (e.g., NR and AEC).

7.1 Overall VED tests

These tests should be performed on the overall VED, including all of the functions defined in clause 5.

7.1.1 Test 1.1 – Signalling tones test (optional)

This test applies to VEDs that are not externally disabled by the switch and which are located on the side of a switch where Signalling System No. 5, 6 and 7 tones are present.

VEDs should not distort or corrupt signalling tones in the network. This test ensures that the VED will not adversely affect signalling or continuity check tones.

Test procedure

With the VED in any initial state, the signalling tones of Table 1 are applied to the input of the VED. The level of the output from the VED is monitored.

Table 1 – Applicable signalling tones

System 5	System 6	System 7
2400 $\pm$ 15 Hz	2000 $\pm$ 30 Hz	2000 $\pm$ 30 Hz
2600 $\pm$ 15 Hz		
2400 $\pm$ 15 Hz & 2600 $\pm$ 15 Hz		

The power level M , of each signal applied, should be within the following limits:

System 5: $-16 \leq M \leq -2$ dBm0

NOTE 1 – This range applies to the single frequency signals f_1 and f_2 . The level of the individual signals in the compound signal may differ from each other by not more than 5 dB. Note that when the levels of the individual signals approach -2 dBm0, the compound signal may be clipped. The nominal transmit level is specified as -9 ± 1 dBm0.

See [ITU-T Q.143] and [ITU-T Q.144].

System 6: $-18 \leq M \leq -6$ dBm0

NOTE 2 – The nominal transmit level is -12 ± 1 dBm0.

See [ITU-T Q.271].

System 7: $-18 \leq M \leq -6$ dBm0

NOTE 3 – The nominal transmit level is -12 ± 1 dBm0.

See clause 7 in [ITU-T Q.724].

The above levels are designed to ensure that the VED will operate with signals that occupy the entire range of levels given in the appropriate signalling ITU-T Recommendations.

The VED detection time should be sufficiently long to provide immunity from false operation due to voice signals. Conversely, the VED detection time should not be so long as to needlessly extend the time for the signal to appear on the network.

Requirement

The level of the signalling tones measured at the output of the VED should not vary more than ± 2 dB from the level of the input after an allowed 1 second detection time.

7.1.2 Test 1.2 – DTMF tones test

The object of this test is to ensure that the VED will not interfere with the detection of DTMF tones. This test is not required if a VED will never encounter DTMF tones when it is deployed in a network. Examples of such cases are a VED integrated into other network equipment that has the ability to generate and detect DTMF tones, and a VED applied at the A-interface of a GSM wireless network.

In any test to check DTMF detection accuracy in a connection with a VED, the choice of DTMF detector may influence the percentage of detection failures. To circumvent this problem, the following test is based on a comparison of DTMF detection performance with and without the VED in-circuit. It is therefore unnecessary to specify a standard DTMF detector for the test.

The test requires a means of inputting a sequence of DTMF characters to the test circuit (e.g., sequence stored on digital audio tape, or PC file). A suitable DTMF detector (which may be PC-based for convenience) is also required, together with a means of recording the percentage of character detection failures in the replayed DTMF test sequence. A noise source is needed for the tests, and the basic test configuration is shown in Figure 1.

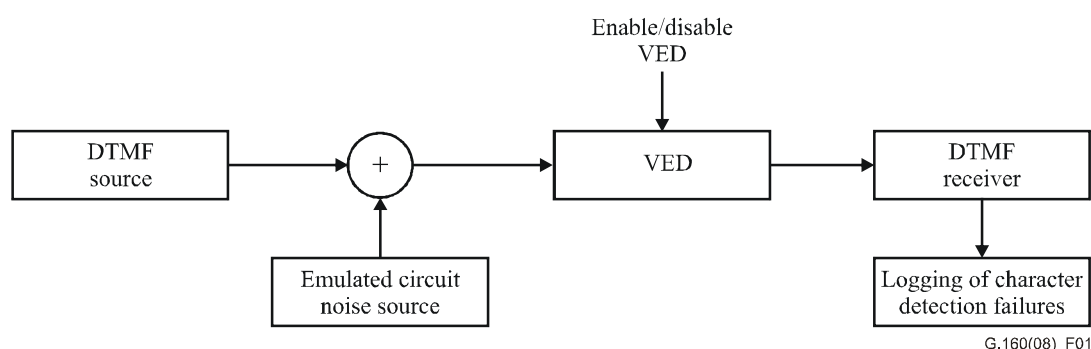


Figure 1 – Circuit configuration to check effect of VED on DTMF detection performance

The DTMF test sequence should consist of 256 different DTMF signals, arranged as 16 sets of the 16 DTMF characters (0..9, #, *, A, B, C, D). Each set has the frequency offset specified in Table 2. The replayed levels of the discrete frequencies making up the tone pairs should be as specified in Table 2. Each DTMF character should have a duration of 50 ms, with an interval between characters of 100 ms. The interval between each set should be 1 second, and the 16 sets of DTMF characters should be replayed as a continuous sequence of 256 characters. The timing relationships for this test are shown in Figure 2.

Table 2 – DTMF test sequences

Set No.	Freq. offset (%) (low freq.)	Freq. offset (%) (high freq.)	Level (dBm0) (low freq.)	Level (dBm0) (high freq.)
1	0	0	0	0
2	0	0	−18	−18
3	1.5	1.5	−10	−10
4	−1.5	−1.5	−10	−10
5	0	0	−12	−18
6	0	0	−14	−10
7	1.5	1.5	−14	−20
8	−1.5	1.5	−14	−20
9	1.5	−1.5	−6	−12
10	−1.5	−1.5	−6	−12
11	0	0	−12	−18
12	0	0	−14	−10
13	1.5	1.5	−10	−6
14	−1.5	1.5	−10	−6
15	1.5	−1.5	−18	−14
16	−1.5	−1.5	−18	−14

Annex A of [ITU-T Q.24] indicates that some Administrations require DTMF receivers to tolerate power levels per frequency up to 0 dBm.

Electrical circuit noise should be emulated by a Gaussian, white noise signal, band-limited to the frequency range 0-3.4 kHz at source. References to the level of this noise test signal relate to the mean power of the signal, measured flat (i.e., without psophometric weighting). The level of the noise should not exceed -38 dBm0.

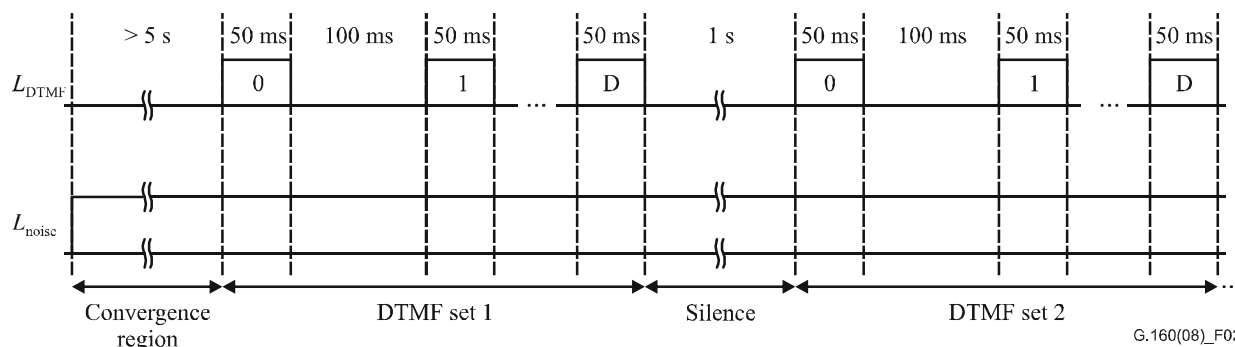


Figure 2 – Timing relationships for the DTMF tones test (two sets out of 16)

Test procedure

- 1) With the VED disabled, replay the test signal comprising the sum of DTMF test sequence and emulated circuit noise over the test circuit. The test signal should start with the noise-only segment of at least 5 seconds. This will allow the VED to converge during step 3. Record the percentage of detection failures and detection errors.
- 2) Repeat step 1 a sufficient number of times to satisfactorily establish the standard of DTMF detection performance. Detection with VED disabled should be reliable. If this is not the case, lower the noise level and repeat steps 1 and 2.
- 3) Repeat steps 1 and 2, but with the VED enabled. If DTMF detection performance is noticeably worse with the VED enabled, it must be assumed that the VED degrades the transmission of DTMF signals.

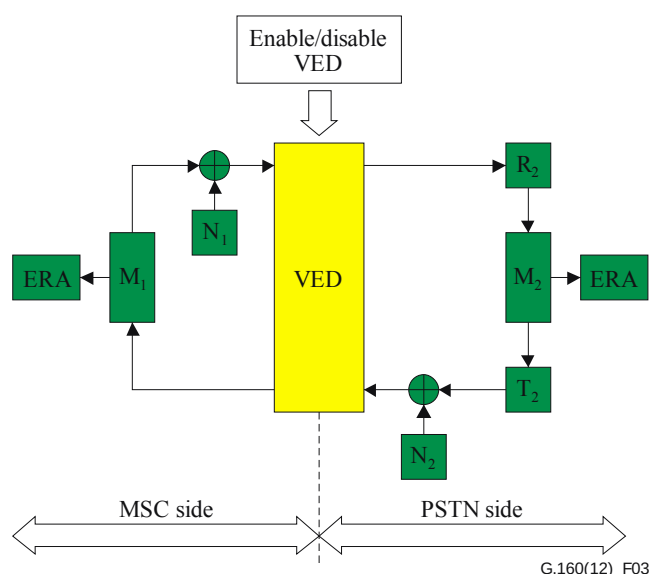
7.1.3 Test 1.3 – Text telephone, data modem and facsimile test

This test will ensure that a VED will not degrade voice-band data signals (text telephone, data modem and facsimile) used in mobile networks. Table 3 is a practical representation of modems most commonly used in GSM networks that should be used for testing. In a GSM network, all modems except cellular text modems (CTMs) are located on the PSTN side of the mobile switching centre (MSC). CTM modems are located in the voice path of the GSM network, e.g., at the A-interface as shown in Figure I.18. The intent of this test is to allow a VED vendor to demonstrate acceptable performance with a range of modem types used in GSM networks, and for network operators to gain a level of confidence that existing modem traffic will not be impaired. The performance criteria for this test depend on the type of modem being tested. For [ITU-T V.32] and [ITU-T V.21] data modems, bit error rate (BER) is used to measure performance (measured synchronously where possible). For textphone modems, however, character error rate (CER) is more appropriate. The performance criteria combined with minimum volume of data to be transmitted for each type of modem are given in Table 3.

This test is not required if a VED will never encounter text telephone, data modem and facsimile signals when it is deployed in a network. An example of such a case is a VED integrated into another network equipment that has the ability to bypass the VED for the treatment of text telephone, data modem and facsimile signals (e.g., using [b-ITU-T V.150.1] procedures).

**Table 3 – Minimum set of modems to be used in Test 1.3
(relevant to GSM and 3G mobile networks only)**

Application	Modem type	Data rate	Answer tone	Performance criteria	Minimum volume of data
GSM Circuit Switched Data	V.32	9600 bit/s 2400 bit/s	ANS	BER	10000 bytes
GSM Circuit Switched Data	V.21	300 bit/s		BER	10000 bytes
GSM Text Telephony Transcoding CTM/V.18 and V.18/CTM	V.18	300 bit/s	ANS	CER	1000 characters
	TIA 825-A	45.45, 50	None		
	Baudot	baud			
	V.21	300 bit/s	ANS (optional)		
	V.23	75/1200 bit/s	ANS (optional)		
	EDT	110 bit/s	ANS (optional)		
GSM Voice-band text data	ETSI Cellular Text Modem (CTM)	400 bit/s	None	CER	1000 characters
GSM Facsimile	V.27 <i>ter</i>	4800 bit/s	ANS		2 pages
ANS 2 100 Hz Answer Tone as defined in [ITU-T V.25] BER Bit Error Rate CER Character Error Rate NOTE – The CTM is described in [b-ETSI TS 126 226].					



ERA Error Rate Analyser
VED Voice Enhancement Device
M₁, M₂ Modems listed in Table 3
N₁, N₂ Emulated Circuit Noise
R₂ Receive Attenuator
T₂ Transmit Attenuator

NOTE 1 – Modem M1 represents a 4-wire modem located in the MSC that is controlled using AT commands from the MS or UE.

NOTE 2 – Modem M2 represents a 2-wire text telephone, data modem or facsimile machine connected to the PSTN.

NOTE 3 – Noise N1 represents background noise that may be present when using a CTM adapter for a text telephone. N1 is not required for testing data modem and facsimile.

Figure 3 – Test set-up for Test 1.3 – Text telephone, data modem and facsimile test

The values of the settings for all modems should be as follows:

R₂ = 6 dB to simulate access loss.

T₂ = 3 dB to 9 dB (3 dB is the nominal level, 9 dB simulates a 6 dB level offset).

M₁, M₂ = modem data transmission levels should be between –8 dBm and –20 dBm.

N₁, N₂ = set to produce signal-to-noise ratios of not less than 25 dB, and no noise.

Test procedure

The sequence for this test is to measure the performance of each modem type with the VED disabled and compare this performance with the VED enabled. The philosophy is that when enabled, the VED should not degrade performance compared to that when disabled. The VED is placed in the test configuration of Figure 3. The test procedure is to first disable the VED and allow the modems to train. The modems are then operated for the time it takes to send the minimum volume of test data as defined for each modem type given in Table 3 (e.g., bit error rate, character error rate, etc.). This performance is noted and the test is then repeated with the VED enabled.

Requirement

With the VED enabled, for the conditions specified above, the percentage of data errors should not increase compared with when the VED is disabled, when data is exchanged between the two terminals for the given minimum volume of data transmitted.

7.2 Noise reduction tests

These tests should be performed on the noise reduction function only, with all other functions disabled. Parameters Q_m (required) and L_{C_min} (optional) are specific for a particular noise reduction function. The parameter Q_m is the specified level of noise reduction in dB. The optional parameter L_{C_min} specifies a lower bound on input noise power levels, below which the function is not required to achieve the specified Q_m .

7.2.1 Test 2.1 – Noise reduction signal level test

This test is intended to ensure that a NR function included in a VED produces the stated amount of noise reduction and does not modify speech signal levels beyond acceptable limits.

Test procedure

The test set-up is shown in Figure 4 and the sequence of events is shown in Figure 5. The test procedure is to apply a noise signal during the entire test, starting at time t_0 . During the time period t_0 to t_1 , the NR function is allowed to converge and reach the specified amount of noise reduction (Q_m). Note that the level of the noise signal should be high enough to allow a reduction by Q_m dB to be observable in the PCM signal. In Figure 4, the levels in points A, B, C and D are measured in the unit of dBm0.

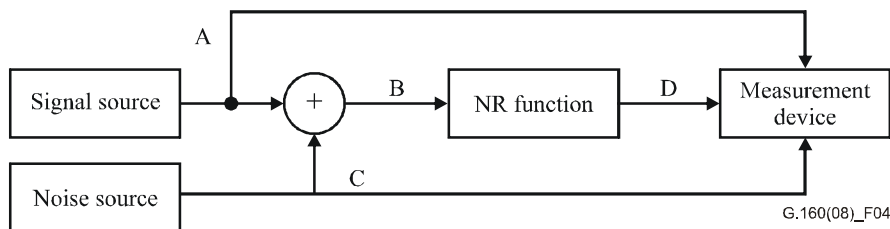


Figure 4 – Test set-up

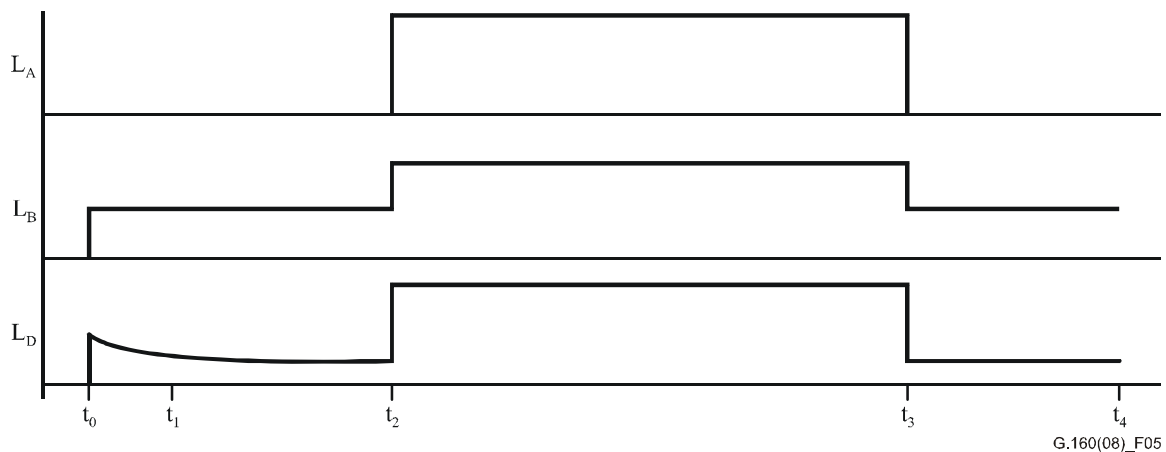


Figure 5 – Signal description

During the period t_1 to t_2 and t_3 to t_4 , the levels at the output of the NR function (point D) and at the output of the noise generator (point C) are measured and noise attenuation factors are computed as $Q_n = L_C - L_D$. At time t_2 , a measurement signal is added and kept for a period of at least 22 seconds from t_2 to t_3 . In order to make the contribution from the noise signal to the signal level measured in point D negligible, the level of the measurement signal should be at least 20 dB above the level of the reduced noise. During the period t_2 to t_3 , the output level of the NR function (point D) and at the measurement signal level (point A) are measured. The level difference is computed as $Q_s = L_A - L_D$.

Note that noise reduction functions may introduce a delay into the signal path. This delay needs to be compensated for before comparing input and output levels of the function.

White Gaussian noise should be used as the noise signal and the male and female artificial voice passages from [ITU-T P.50] as the measurement signals in this test. Concatenated, the artificial voice passages are approximately 22 seconds in length. The RMS method specified in clause 6.2 should be used for level measurements. Other measurement methods are for further study.

The time instants t_0 to t_4 are defined as follows: $t_0 = 0$, $t_1 = 3$ s, $t_2 = 10$ s, $t_3 = 32$ s, $t_4 = 39$ s.

Requirements

For all RMS signal levels $-6 \text{ dBm0} \geq L_A \geq -30 \text{ dBm0}$, noise levels $\max[L_{C_min}, -60 \text{ dBm0} + Q_m] < L_C < L_A - SNR$ and SNR values given by:

$$SNR \geq 12 \text{ dB} \quad \text{if } Q_m \geq 8 \text{ dB}$$

$$SNR \geq 20 \text{ dB} - Q_m \quad \text{if } Q_m < 8 \text{ dB.}$$

- 1) During the period t_1 to t_2 and t_3 to t_4 , the noise should be attenuated by the amount specified for the particular setting of the NR function under the test ± 3 dB, i.e., $Q_m - 3 < Q_n < Q_m + 3$. To verify this, the RMS signal level measurement is computed twice according to the definition of clause 6.2. First across the signal period from t_1 to t_2 ($k = t_2$, $n = t_2 - t_1$) and then from t_3 to t_4 ($k = t_4$, $n = t_4 - t_3$).
- 2) During the period from t_2 to t_3 , the signal levels measured at the output of the NR device (point D) and at the output of the measurement signal generator (point A) should satisfy $-3 < Q_s < 2$. To verify this, the RMS signal level is computed according to the definition of clause 6.2 across the signal period, from t_2 to t_3 ($k = t_3$, $n = t_3 - t_2$).

NOTE 1 – Three level measurements using the RMS method as defined in clause 6.2 are made in this test. Short-term signal level comparisons are likely not to meet Requirement 2 above in all cases and are not required to do so.

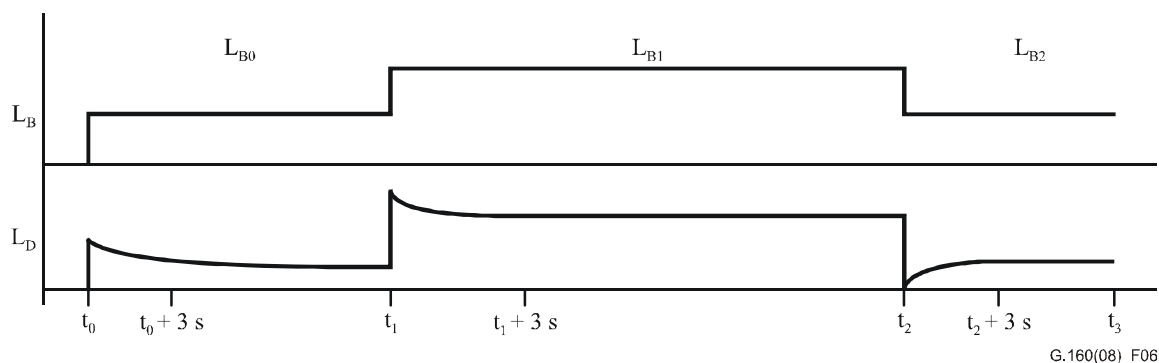
NOTE 2 – Noise attenuation for noise sources, other than white noise, may vary from these test results.

7.2.2 Test 2.2 – Noise reduction convergence/reconvergence test

This test is intended to ensure that a NR function included in a VED produces the stated amount of noise reduction in response to a step change in noise level after a maximum allowed convergence time.

Test procedure

The test set-up is the same as shown in Figure 4 except that the signal source is not applicable. The test procedure is to apply a white Gaussian noise signal abruptly changed in level as shown in Figure 6.



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Figure 6 – Signal levels

The L_B RMS noise powers for the indicated time periods are as follows:

For $t < t_0$, $L_B = \text{quiet}$

For t_0 to t_1 , $\max[L_{C_min}, -60 \text{ dBm0} + Q_m] < L_{B0} < -30 \text{ dBm0}$ where $t_1 = t_0 + 6 \text{ seconds}$

For t_1 to t_2 , $L_{B1} = L_{B0} + 12 \text{ (range) dBm0}$ where $t_2 = t_0 + 12 \text{ seconds}$

For t_2 to t_3 , $L_{B2} = L_{B0} \text{ (range) dBm0}$ where $t_3 = t_0 + 18 \text{ seconds}$

Optionally, other level combinations may be tested.

Requirements

3.0 seconds after the input noise level, L_B , changes as indicated in Figure 6, the NR output noise level, L_D , should be attenuated by the amount specified for the particular setting of the NR function under the test $\pm 3 \text{ dB}$, i.e., $Q_m - 3 < Q_n < Q_m + 3$.

7.2.3 Test 2.3 – Noise reduction voice quality test (under study)

This test is meant to assess perceived listening quality of a noise reduction algorithm. For this purpose, it is based on the use of objective methods (to be determined), which will predict the subjective quality as specified in [ITU-T P.835].

This test is split into 4 parts.

7.2.3.1 Test 2.3a

This test evaluates the ability of the NR function to maintain the quality of the voice signal at very high SNR.

Test method: TBD

Very high SNR values: $SNR \geq \text{TBD dB}$

7.2.3.2 Test 2.3b

This test evaluates the ability of the NR function to maintain the quality of the voice signal at high SNR.

Test method: TBD

High SNR values: TBD (same TBD as in Test 2.3a) $\text{dB} > SNR \geq 18 \text{ dB}$

7.2.3.3 Test 2.3c

This test evaluates the ability of the NR function to improve the quality of the voice signal at moderate SNR.

Test method: TBD

Moderate SNR values: $18 \text{ dB} > SNR \geq 12 \text{ dB}$

7.2.3.4 Test 2.3d

This test evaluates the ability of the NR function to maintain or improve the quality of the voice signal at low SNR.

Test method: TBD

Low SNR values: $SNR < 12 \text{ dB}$

NOTE – Future development of this test should take into account the optional provision for the parameter L_{C_min} as suitable test and requirement details are developed.