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Signalling requirements and protocols for the NGN – Testing for NGN networks

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Parameters for monitoring voice services in NGN

Recommendation ITU-T Q.3911

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Recommendation ITU-T Q.3911

Parameters for monitoring voice services in NGN

Summary

Recommendation ITU-T Q.3911 defines the set of monitoring parameters that will impact the quality of voice services in NGN. These parameters are generated by network elements, such as terminal elements, connection elements, transmission elements, etc. The definitions provided here are dependent on NGN, which uses the session initiation protocol (SIP) as call control protocol. How to monitor these parameters is out of the scope of this Recommendation.

History

Edition	Recommendation	Approval	Study Group
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Monitoring parameters, NGN, voice services.

FOREWORD

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The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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Recommendation ITU-T Q.3911

Parameters for monitoring voice services in NGN

1 Scope

This Recommendation defines the set of parameters that will impact the quality of voice services in NGN. These parameters are generated by network elements, such as terminal elements, connection elements, transmission elements, etc. The definitions provided here are dependent on NGN, which uses SIP as call control protocol. How to monitor these parameters is out of the scope of this recommendation.

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 references to a document within this Recommendation do not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T G.711] Recommendation ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[ITU-T G.722] Recommendation ITU-T G.722 (1988), *7 kHz audio-coding within 64 kbit/s.*

[ITU-T G.722.2] Recommendation ITU-T G.722.2 (2003), *Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)*.

[ITU-T G.729] Recommendation ITU-T G.729 (2007), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*.

[ITU-T G.729.1] Recommendation ITU-T G.729.1 (2006), *G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729*.

[ITU-T P.564] Recommendation ITU-T P.564 (2007), *Conformance testing for voice over IP transmission quality assessment models*.

[ITU-T P.800] Recommendation ITU-T P.800 (1996), *Methods for subjective determination of transmission quality*.

[ITU-T P.800.1] Recommendation ITU-T P.800.1 (2006), *Mean Opinion Score (MOS) terminology*.

[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*.

[3GPP TS 26.071] 3GPP TS 26.071 V6.0.0 (2004), *Mandatory speech CODEC speech processing functions; AMR speech Codec; General description (Release 6)*.

[TIA-127-C] TIA-127-C (2007), *Enhanced Variable Rate Codec, Speech Service Options 3, 68, and 70 for Wideband Spread Spectrum Digital Systems*.

3 Abbreviations

This Recommendation uses the following abbreviations and acronyms:

IP	Internet Protocol
MOS	Mean Opinion Score
NGN	Next Generation Networks
PESQ	Perceptual Evaluation of Speech Quality
RTCP	Real-time Transfer Control Protocol
RTP	Real-time Transfer Protocol
SCTP	Session Control Transfer Protocol
SIP	Session Initiation Protocol
UAC	User Agent Client
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol

4 Definitions

None.

5 Conventions

None.

6 Measurement metrics

6.1 Active measurement

Active measurements are used to obtain the value of measurement parameters by injecting a test stream or signals into the network. In this case, the test stream or signals are sent from one endpoint of the network, and received by the remote answering machine located at another endpoint. The analysis of the measurement parameters value can help to determine the performance or behaviour of the system.

6.2 Passive measurement

In the case of passive measurements, a sniffer is used to detect call control signalling and VoIP flows, obtain the call leg RTP/RTCP, and then capture the statistics generated by the end-user. The sniffer can be deployed near the "gateway" or equipped in the end-user.

The hybrid usage of the two measurement metrics is usually preferred.

7 Monitoring parameters

The following gives the most probable failure or poor performance reasons. In some cases, other reasons might apply. The comparison of several measurements might give additional information.

7.1 Registration

This clause introduces the monitoring parameters for the registration phase in voice services, such as successful register rate, register delay, failed register rate, etc. The parameters are based on end-to-end measurements.

7.1.1 Successful register rate

Successful register rate is defined as the rate of successful registration attempts in the voice service. This parameter can evaluate the performance and quality of voice service during the registration phase. Successful register rate is measured at the element node in the SIP link, for example, at the UAC. The parameter is calculated by monitoring the element node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The successful register rate is calculated using the following formula:

$$\text{Successful register rate} = \frac{Rn_{suc}}{Rn} \times 100\%$$

Rn is the total number of REGISTER messages that are received and sent in the monitoring interval time, such as 30 minutes. Rn_{suc} is the total number of successful and completed registrations, for example, the node receives or sends 200 OK responses.

7.1.2 Register delay

Register delay is defined as the average time from the start of registration to the successful completion of registration in the voice service. This parameter is measured at the element node in the SIP link, for example, at the UAC. The parameter is averaged over a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be stated in units of milliseconds. The register delay parameter is calculated using the following formula:

$$\text{Register delay} = \frac{\sum_{i=1}^{n}(Rtt_i - Rt_i)}{Rn}$$

Rt_i is the time when the ith registration's REGISTER Request message starts to be sent by the originating node. Rtt_i is the time when the ith registration's final Response, e.g., 200 OK, is completely received by the originating node. Rn is the total number of successful and completed registrations in the monitoring interval, such as 30 minutes.

7.1.3 Failed register rate

Failed register rate is defined as the rate of failed registration attempts in the voice service. Failed register rate can evaluate the performance of the voice service in the network and indicates the inability of the network elements. For example, it may indicate a registrar has become overloaded and is unable to respond to the request. This parameter is measured at the element node in the SIP link, for example, at the UAC. The parameter is calculated by monitoring the element node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The failed register rate parameter is calculated using the following formula:

$$\text{Failed register rate} = \frac{Rn_{failed}}{Rn} \times 100\%$$

Rn is the total number of REGISTER messages that are received and sent in the monitoring interval time, such as 30 minutes. Rn_{failed} is the total number of failed registrations; for example, the node receives or sends message 401 as a response to REGISTER messages.

7.2 Call establishment

This clause provides the monitoring parameters for the call establishment phases in the voice service, such as successful call establishment rate, pre-release rate, "route unavailable" rate, etc. The parameters are based on the end-to-end communication mode.

7.2.1 Successful call establishment rate

Successful call establishment rate is defined as the rate of the successful call attempts in the voice service. This parameter can evaluate the performance and quality of the voice service during the call establishment phase. Successful call establishment rate is measured at the element node in the SIP link, for example, the UAC or terminals. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Successful call establishment rate} = \frac{Cn_{suc}}{Cn} \times 100\%$$

Cn is the total number of INVITE messages that are received and sent by the node in the monitoring interval time, such as 30 minutes. Cn_{suc} is the number of successful and completed call establishment, for example, 200 OK responses for the INVITE requests received or sent.

7.2.2 Pre-release rate

Pre-release means that the CANCEL request is received or sent before the call is established. The pre-release rate can evaluate the performance of the voice service and indicates the inability of the network. The parameter is measured at the UAC or end-users' terminals. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Pre-release rate} = \frac{Cn_{prere}}{Cn} \times 100\%$$

Cn is the total number of INVITE messages that are received and sent by the node in the monitoring interval time, such as 30 minutes. Cn_{prere} is the number of CANCEL messages that are received and sent before the ACK messages for the final 200 OK responses.

7.2.3 Failed call establishment rate

Failed call establishment rate is defined as the rate at which failure indication responses are received at the monitoring node in the voice service. A failure indication response is described as a 4XX (excluding 401, 402, and 407 non-failure challenge response codes), 5XX, or possible 6XX messages. This parameter can evaluate the performance of the voice service and indicates the inability of the network. The parameter is measured at the UAC or end-users' terminals. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Failed call establishment rate} = \frac{Cn_{failed}}{Cn} \times 100\%$$

Cn is the total number of INVITE messages that are received and sent by the node in the monitoring interval time, e.g., 60 minutes. Cn_{failed} is the number of the calls that receive failure indication responses to INVITE messages. Usually, Cn_{failed} is calculated by counting 4XX messages (excluding 401, 402, and 407 non-failure challenge response codes), 5XX and 6XX messages that are received and sent by the node during the monitoring interval time.

7.2.4 No response rate

No response rate is defined as the rate of no answers for the call in the voice service. This parameter is measured during the call establishment phase. The parameter is measured at the UAC or end-users' terminals. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 60 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{No response rate} = \frac{Cn_{noresp}}{Cn} \times 100\%$$

Cn is the total number of INVITE messages that are received and sent by the node in the monitoring interval time. Cn_{noresp} is the number of the calls which have no answer for a long time. Normally, Cn_{noresp} is calculated by counting 480 messages received and sent by the node during the monitoring interval time.

7.2.5 Call establishment delay

Call establishment delay is defined as the average time from the start of INVITE messages to the successful completion of the call establishment in the voice service. This parameter is measured at the element node in the SIP link; for example, the UAC or terminals. The parameter is averaged over a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be stated in units of milliseconds. The call establishment delay parameter is calculated using the following formula:

$$\text{Call establishment delay} = \frac{\sum_{i=1}^{n}(Ctt_i - Ct_i)}{Cn}$$

Ct_i is the time when the ith call establishment's INVITE Request message starts to be sent by the originating node. Ctt_i is the time when the ith call establishment's final Response, e.g., 200 OK, is completely received by the originating node. Cn is the total number of successful and completed call establishment at the monitoring interval, such as 30 minutes.

7.3 Quality of voice service

This clause refers to clauses 8 and 9 of [ITU-T P.564], *Conformance testing for voice over IP transmission quality assessment models*. [ITU-T P.564] specifies minimum criteria for objective speech quality assessment models that predict the impact of observed IP network impairments on the one-way listening quality experienced by the end-user in IP/UDP/RTP-based 3.1 kHz narrowband telephony applications. The subjective assessment of quality is also considered. This part can refer to [ITU-T P.862]. The mapping of objective assessment to subjective assessment can refer to [b-ITU-T P.862.1].

7.3.1 Delay

The delay parameter is defined as the average value of the RTP packets delay. This parameter is measured at the element nodes that receive and send RTP packets, for example, the terminals. The parameter is averaged over a predefined time interval, e.g., 5 minutes. The output value of this parameter is numerical and should be stated in units of milliseconds. The parameter is calculated using the following formula:

$$\text{Delay} = \frac{\sum_{i=1}^{n}(Ptt_i - Pt_i)}{Pn}$$

where Pt_i is the time when the ith RTP packet starts to be sent by the originating node, Ptt_i is the time when the ith RTP packet is completely received by the terminal node, and Pn is the total number of RTP packets sent or received at the monitoring interval, such as 5 minutes.

7.3.2 Packet loss rate

The packet loss rate parameter is defined as the rate of lost RTP data packets in the voice service. This parameter is measured at the element nodes that receive and send RTP packets, for example, the terminals. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Packet loss rate} = \frac{Pn_{exp} - Pn_{act}}{Pn_{exp}} \times 100\%$$

Pn_{exp} is the number of RTP data packets expected to be received. Pn_{act} is the cumulative number of RTP data packets actually received. Pn_{act} includes any packets that are late or duplicates. Thus, packets that arrive late are not counted as lost, and the packet loss rate may be negative if there are duplicates.

7.3.3 Jitter

The jitter parameter is defined as the average value of the RTP packet jitter. This parameter is measured at the element nodes that receive and send RTP packets, for example, the terminals. The parameter is averaged over a predefined time interval, e.g., 5 minutes. The output value of this parameter is numerical and should be stated in units of milliseconds. The parameter is calculated using the following formula:

$$\text{Jitter} = \frac{\sum_{i=1}^{Pn-1} \left| (Ptt_i - Pt_i) - (Ptt_{i+1} - Pt_{i+1}) \right|}{Pn - 1}$$

where Pt_i is the time when the ith RTP packet starts to be sent by the originating node, Ptt_i is the time when the ith RTP packet is completely received by the terminal node, and the value of Pn is the total number of RTP packets sent or received during the monitoring interval, such as 5 minutes.

7.3.4 MOS

In the monitoring interval, the MOS parameter is generated and reported in order to monitor the quality of the voice service in NGN. In this Recommendation, the MOS parameter refers to [ITU-T P.800.1] and [ITU-T P.800]. As defined in [ITU-T P.800.1], MOS is the mean of opinion scores, i.e., of the values on a predefined scale that subjects assign to their opinion of the performance of the telephone transmission system used either for conversation or for listening to spoken material.

7.3.5 PESQ

In the monitoring interval, the PESQ parameter is generated and reported in order to monitor the quality of voice service in NGN. PESQ is the end-to-end speech quality assessment of voice service by objective measurement. PESQ is defined in [ITU-T P.862]. In this Recommendation, the PESQ parameter is the same as that of [ITU-T P.862].

7.4 Codecs

Speech codecs certainly have a great impact on the voice service performance. The parameters that indicate the usage rate of various codecs are given as follows. These parameters are intended to present the proportion of the different codecs used in the network.

7.4.1 ITU-T G.711 codec used rate

Parameter ITU-T G.711 codec used rate is defined as the rate at which the ITU-T G.711 codec is used during the monitoring interval in voice services. This parameter is measured at the element nodes that are involved in the SIP session, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{ITU-T G.711 codec used rate} = \frac{Dn_{711}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time. Dn_{711} is the number of sessions that use ITU-T G.711 as the audio codec.

7.4.2 ITU-T G.729 codec used rate

Parameter ITU-T G.729 codec used rate is defined as the rate at which the ITU-T G.729 codec is used during the monitoring interval in voice services. This parameter is measured at the element nodes that are involved in the SIP session, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{ITU-T G.729 codec used rate} = \frac{Dn_{729}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time. Dn_{729} is the number of sessions that use ITU-T G.729 as the audio codec.

7.4.3 ITU-T G.722 codec used rate

Parameter ITU-T G.722 codec used rate is defined as the rate at which the ITU-T G.722 codec is used during the monitoring interval in voice services. This parameter is measured at the element nodes that are involved in the SIP session, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{ITU-T G.722 codec used rate} = \frac{Dn_{722}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time. Dn_{722} is the number of sessions that use ITU-T G.722 as the audio codec.

7.4.4 ITU-T G.729.1 codec used rate

Parameter ITU-T G.729.1 codec used rate is defined as the rate at which the ITU-T G.729.1 codec is used during the monitoring interval in voice services. This parameter is measured at the element nodes that are involved in the SIP session, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{ITU-T G.729.1 codec used rate} = \frac{Dn_{729.1}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time. $Dn_{729.1}$ is the number of sessions that use ITU-T G.729.1 as audio codec.

7.4.5 Mobile codec rate

Mobile codecs are the audio codec for 3GPP terminals and 3GPP2 terminals. Mobile codecs include AMR [3GPP TS 26.071], AMR-WB/G.722.2 [ITU-T G.722.2], EVRC/EVRC-B [TIA-127-C] and EVRC-WB [TIA-127-C].

Parameter mobile codec used rate is defined as the rate at which mobile codecs are used during the monitoring interval in voice services. This parameter is measured at the element nodes that are involved in the SIP session, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Mobile codec used rate} = \frac{Dn_{AMR} + Dn_{AMR\text{-}WB} + Dn_{EVRC/EVRC\text{-}B} + Dn_{EVRC\text{-}WB}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time.

Dn_{AMR} is the number of sessions that use AMR as the audio codec.

$Dn_{AMR\text{-}WB}$ is the number of sessions that use AMR-WB/G.722.2 as the audio codec.

$Dn_{EVRC/EVRC\text{-}B}$ is the number of sessions that use EVRC/EVRC-B as the audio codec.

$Dn_{EVRC\text{-}WB}$ is the number of sessions that use EVRC-WB as the audio codec.

7.4.6 Conversion rate between different codecs

Conversion rate between different codecs is defined as the rate of transcoding events that happen at the monitoring node. This parameter is measured at the element nodes that are responsible for codec conversion, for example, at the UAC. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Conversion rate between different codecs} = \frac{Dn_{convers}}{Dn} \times 100\%$$

Dn is the total number of SIP audio sessions that take place at the node in the monitoring interval time. $Dn_{convers}$ is the number of sessions that need audio codec conversion.

7.5 Call completion

This clause provides the monitoring parameters for the call completion phase in voice services, such as successful call completion rate, call completion delay, failed call completion rate, etc. The parameters are based on the end-to-end communication mode.

7.5.1 Successful call completion rate

Successful call completion rate is defined as the rate of successful call disconnects in the voice service. This parameter can evaluate the performance and quality of the voice service during the call completion phase. Successful call completion rate is measured at the element node, for example, at the UAC or the terminals. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Successful call completion rate} = \frac{En_{suc}}{En} \times 100\%$$

En is the total number of BYE messages that are received and sent by the node in the monitoring interval time, such as 30 minutes. En_{suc} is the number of successful and completed call completions, for example, 200 OK responses for the BYE requests received or sent.

7.5.2 Call completion delay

Call completion delay is defined as the average time from the start of a BYE message to the successful completion of the call in the voice service. This parameter is measured at the element node, for example, at the UAC or the terminals. The parameter is averaged over a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be stated in units of milliseconds. The call completion delay parameter is calculated using the following formula:

$$\text{Call completion delay} = \frac{\sum_{i=1}^{En}(Et_i - Ett_i)}{En}$$

Et_i is the time when the ith call completion's BYE Request message starts to be sent. Ett_i is the time when the ith call completion's final Response message, e.g., 200 OK, is completely received by the node. En is the total number of successful and completed call completions in the monitoring interval, such as 30 minutes.

7.5.3 Failed call completion rate

In some cases, no response is received after a session completion message is sent and potentially retried. Failed call completion rate is defined as the rate of no responses received for BYE messages in the voice service. The parameter is measured at the element node, for example, at the UAC or the terminals. The parameter is calculated by monitoring the node for a predefined time interval, e.g., 30 minutes. The output value of this parameter is numerical and should be reported as a percentage. The parameter is calculated using the following formula:

$$\text{Failed call completion rate} = \frac{En_{failed}}{En} \times 100\%$$

En is the total number of BYE messages that are received and sent by the node in the monitoring interval time, such as 30 minutes. En_{failed} is the number of BYE messages that do not receive responses in a predefined time, for example, timer F expired.

Bibliography

[b-ITU-T P.862.1] Recommendation ITU-T P.862.1 (2003), *Mapping function for transforming P.862 raw result scores to MOS-LQO*.

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