SPEECH QUALITY MONITORING IN CZECH NATIONAL RESEARCH NETWORK

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Abstract. *This paper deals with techniques of measuring and assessment of the voice transmitted in IP networks and describes design of quality measurement, which can be used for Cisco Gateways. Cisco gateways send Calculated Planning Impairment Factor in every CDR (Call Detail Record). Our design is based on collection of CDR's, their storing into SQL database and their visualization through web page. This design was implemented and successfully tested in CESNET network.*

Keywords

QoS, R-factor, E-model, Speech quality, Monitoring.

1. Introduction

Generally we can divide the measurement of the voice quality into two basic techniques: subjective and objective. In the subjective measuring techniques we use the assessment on the basis of the human perception and assessment of several listeners.

In the objective techniques we use the latest piece of knowledge from the discipline the measurement of the quality. The aim of these techniques is to reach the qualitative estimation with results which are being on-coming the best to MOS value, which we would get from subjective techniques. ITU-T suggested the objective, automatic and reproducible experimental techniques, which include the influence of the personal character and human perception. We can divide these techniques into three categories:

- comparative - based on the knowledge of the transmitted and figurative voice sample, where we evaluate the quality with the mutual comparison of suitable algoritmus.
- absolute estimation - based on the absolute estimation of the quality, which is determined from parameters, acquired from the passive monitoring of the conversations which are taking place.
- measurement based on the objective parameters - deducing the quality estimation of the network acquaintance.

In the next we are going to deal only with the objective non-intrusive techniques, because we can come out of the parameters of the transmitted network and the total acquaintance with the behavior of the voice gateways.

2. E-Model

The idea of "E-model" comes from Nils-Olof Johannesson, who was the Swedish expert working for ITU-T and later for ETSI. E-model was chosen by ITU-T, to become a new international recommendation, which is different from another objective measuring techniques, in that point, because it deals with a tool of planning, which can help to ensure the satisfaction of users with the quality of connection (end-to-end). The first developed E-model was simple and inaccurate, but the following expansion and improvement changed it to a model, suitable for planning such comprehensive systems, as VoIP. The first standardized version of E- model was released by ITU-T at the end of year 1998 under trademark G.107. It was revised in 2000 to include the effect of the noise in the room of the transmitting side and the influence of the signal distortion, which was caused by quantization. Consecutively, in 2002, it was enhanced to cover the influence of the irregular loos of the packets during transmission for various types of codecs. Nowadays the most actual version G.107 from March 2005 and another modification is prepared [1].

Tab. 1. Assigning R-factor on the base of the quality**.**

R-factor	MOS	Speech quality	Users' satisfaction
R > 90	>4.3	Best	very satisfied
80-90	4.0 – 4.3	High	satisfied
70-80	3.6 – 4.0	Medium	some are satisfied
60-70	3.1 – 3.6	Low	many are satisfied
50-60	2.6 – 3.1	Poor	all are not satisfied

The result of E-model is R-factor, which takes on values from 0 to 100 and quantifies the influence of noise, signal level, distortion of quantization, type from coding,

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echo, delay and a number of other effects. Table 1 presents assigning values of R-factor to the qualitative categories, where it is not recommended to run connections with value lower than 50. R-factor is set for the whole transmission path between the network users. It takes into account the telephone channel and the termination equipment. The transmission path evaluated by R-factor, with the most important parameters for the quality of transmission of speech signal. The result of the E-model calculation is R-factor, which combines all transmitting parameters important for specific connection. The R-factor can be represented by the equation (1).

$$R = R_0 - I_S - I_D - I_{E-EF} + A \qquad (1)$$

The parameter A in equation (1) represents Expectation Factor, the next one I_D includes impairments caused by delay, I_{E-EF} represents equipment effective factor, I_S all impairments which occur simultaneously with the speech and R_0 represents Signal-to-Noice ratio. The values of MOS, R-factor and MIPS for the chosen codecs are presented in the Table 2. The detailed information and description of MOS and R-factor is possible to find in references [1], [2] and [3].

Tab. 2. Overview of the codec´s quality**.**

Codec	Algorithm	bit rate [kbps]	MIPS	MOS	R
G.711	PCM	64	0.34	4.1	82.6
G.726	ADPCM	32	14	3.85	75.3
G.728	LD-CELP	16	30	3.61	69.4
GSM 06.10	RPE-LTP	13	10	3.5	67
G.729A	CS-ACELP	8	10.5	3.7	71.5
G.729	CS-ACELP	8	20	3.92	77.2
G.723.1	MP-MLQ	6.3	16	3.9	76.6
G.723.1	ACELP	5.3	16	3.65	70.3

3. Factor of the planning impairment ICPIF

$I_{CPIF}$is Calculated Planning Impairment Factor. It is used by Cisco for the assessment of the quality I_{CPIF}, according to the recommendation G.113. I_{CPIF} is defined by the following equation:

$$I_{CPIF} = I_{TOT} - A \qquad (2)$$

where I_{CPIF} represents Factor of the planning impairment and I_{TOT} includes the total factor of the impairment, defined as

$$I_{TOT} = I_0 + I_q + I_{DTE} + I_{DD} + I_E \qquad (3)$$

I_{DD} represents impairments caused by too long delay and I_{DTE} caused by talker echo. Parameter I_E includes impairments caused by low bit rate codecs and I_q quantizing distortion. After substitution of the relationships we get

$$I_{CPIF} = I_0 + I_q + I_{DTE} + I_{DD} + I_E - A \qquad (5)$$

We used mathematical application in the description and the calculation of the individual quantities. I_{CPIF} can take values in the range from 5 to 55, where the lower number represents the better quality, as we can see in Table below.

Tab. 3. Scale I_{CPIF} from the point of view of the speech quality

The top limit ICPIF	Speech quality
5	very good
10	good
20	adequate
30	bad (only for limited usage)
45	very bad (only for limited usage)
55	very bad (unusable)

The relationship between scales MOS and I_{CPIF} is shown the table 4.

Tab. 4. Relationship between MOS and I_{CPIF}

ICPIF	MOS	Quality Category
0-3	5	Excellent
4-13	4	Good
14-23	3	Fair
24-33	2	Poor
34-43	1	Bad

4. Application indicating speech quality and its implementation in Czech national research network

To collect data from individual voice gateways we can use the syslog file or RADIUS records, generated by individual gateways. In the case that we want to use this data, we have to ensure that detailed CDR records. The generations of the detailed records are being sent by the requested the following command:

```
radius-server vsa send accounting.
```

The realized tests proved that it is possible to use only data at termination of the call (no at start). The termination records contain all necessary data to produce statistics. The example of the record is shown below and is interpreted the quality I_{CPIF} with value 15.

```
Acct-Session-Id = "00003B79"
 Calling-Station-Id = "731928xxx"
 Called-Station-Id = "950073074"
 Cisco-AVPair = "call-id=C7296BC6-6AB411DB-
9419ADB1-C07B1984@195.113.222.10"
 h323-setup-time = "21:57:40.560 MET Fri Nov
3 2006"
 h323-gw-id = "R102.cesnet.cz"
 h323-conf-id = "C7283376 6AB411DB 8643000C
30E39800"
 h323-call-origin = "originate"
 h323-call-type = "VoIP"
 Cisco-AVPair = "h323-incoming-conf-
id=C7283376 6AB411DB 8643000C 30E39800"
 Cisco-AVPair = "subscriber=RegularLine"
 Cisco-AVPair = "session-protocol=sipv2"
 Cisco-AVPair = "gw-rxd-
cdn=ton:2,npi:1,#:950073074"
 h323-connect-time = "21:57:42.723 MET Fri
Nov 3 2006"
 Acct-Input-Octets = 128640
 Acct-Output-Octets = 129280
 Acct-Input-Packets = 804
 Acct-Output-Packets = 808
 Acct-Session-Time = 16
 h323-disconnect-time = "21:57:58.903 MET
Fri Nov 3 2006"
 h323-disconnect-cause = "10"
 h323-remote-address = "195.113.144.245"
 Cisco-AVPair = "release-source=4"
 h323-voice-quality = "15"
 Cisco-AVPair = "alert-
timepoint=21:57:42.073 MET Fri Nov 3 2006"
 Cisco-AVPair = "remote-media-
address=194.212.192.202"
 Cisco-AVPair = "gw-rxd-
cgn=ton:0,npi:0,pi:0,si:3,#:731928xxx"
 Cisco-AVPair = "gw-final-xlated-
cdn=ton:2,npi:1,#:950073074"
 Cisco-AVPair = "gw-final-xlated-
cgn=ton:0,npi:0,pi:0,si:3,#:731928xxx"
 User-Name = "731928xxx"
 Acct-Status-Type = Stop
 Service-Type = Login-User
 NAS-IP-Address = 195.113.144.4
 Acct-Delay-Time = 0
 call-id = "C7296BC6-6AB411DB-9419ADB1-
C07B1984@195.113.222.10"
 h323-incoming-conf-id = "C7283376 6AB411DB
8643000C 30E39800"
 subscriber = "RegularLine"
 session-protocol = "sipv2"
 gw-rxd-cdn = "ton:2,npi:1,#:950073074"
 release-source = "4"
 remote-media-address = "194.212.192.202"
 gw-rxd-cgn =
"ton:0,npi:0,pi:0,si:3,#:731928xxx"
 gw-final-xlated-cdn =
"ton:2,npi:1,#:950073074"
 gw-final-xlated-cgn =
"ton:0,npi:0,pi:0,si:3,#:731928xxx"
 Client-IP-Address = 195.113.144.4
 Acct-Unique-Session-Id = "704a0272350ca40e"
 Timestamp = 1162587478
```

We created our own application for the evaluation of voice quality in CESNET2 VoIP network, which evaluates individual phone calls. The application provides the assessment of the calls, which are terminated to PSTN (Public Switched Telephone Network). The application is based on the collection of data from voice gateways, which send information about individual calls through RADIUS protocol. This data is stored in text files in server, which is running the Linux operation system. The data is processed with help of the created script. This data is then imported into SQL database, which serves as the data resource for our own evaluating web interface.

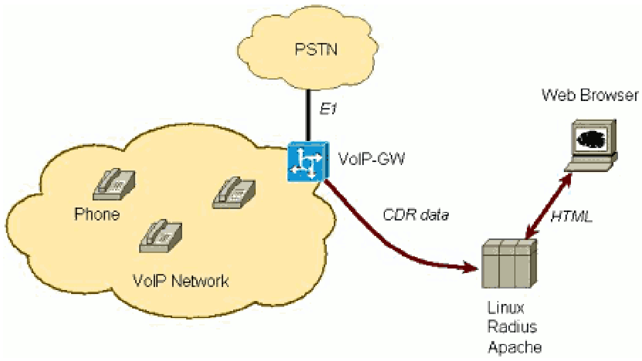

Fig. 1. CDR transmitted from VoGW to database

The individual calls are stored in the database created by processing of the individual records. These calls can be further evaluated. The database stores all calls, as well as unrealized connections. The records include also the reasons why the individual calls were disconnected. The subsequent items can be obtained from the database:

h323connecttime;acctsessiontime;callingstationid;calledstationid;h323remoteaddress; nasipaddress;h323voicequality;h323disconnectcause;acctinputoctets;acctoutputoctets

The own verification of the results, which we got from CDR reports we compared with the measurement in the common operation of the CESNET2 network.

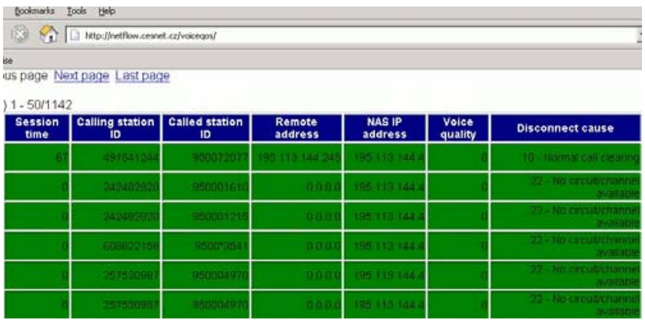

Fig. 2. The list of calls with disconnect causes

The data traffic was duplicated with the help of a SPAN port. This traffic was coming to the output gateway of the public networks and we used the application Surveryor to find R-factor of the individual calls [5]. In most cases we got the $I_{CPIF} = 0$ quality of the call. We reached higher values only when we called into mobile networks and from the networks of local providers. In the CESNET2 or PASNET networks I_{CPIF} was 0 or 1. The voice quality in Fig. 3 is represents by I_{CPIF} explained in chapter 3.

Session time	Calling station ID	Called station ID	Remote address	NAS IP address	Voice quality	Disconnect cause	Input octects	Output octects
39	420950072077	420603483660	195.113.144.245	195.113.144.4	23	10 - Normal call clearing	386400	397488
53	777668484	950073073	195.113.144.245	195.113.144.4	18	10 - Normal call clearing	423040	229592
22	481120044	950073073	195.113.144.245	195.113.144.4	18	10 - Normal call clearing	176640	95744
20	380120256	950074095	195.113.144.245	195.113.144.4	18	10 - Normal call clearing	162400	167832
45	220515402	950073074	195.113.144.245	195.113.144.4	17	10 - Normal call clearing	362560	340536
24	731928102	950073074	195.113.144.245	195.113.144.4	17	10 - Normal call clearing	194080	184968
20	234	420271750040	147.32.240.25	195.113.144.4	16	10 - Normal call clearing	217316	165816

Fig. 3. The calls sorted by quality (the worst is listed first)

5. Conclusion

The developed application is used for the resolving of problems, which originate in the common operation of voice services in the CESNET network. Thanks to the universality of this application, which does not have the aim to ensure any other functions, it is possible to import any data obtained from the defined CDR records. This application makes it easier to solve problems and allows to get the statistics about connected and also unrealized calls. For the own statistics of the individual transmission paths, we will use in the future, the function in IOS of the particular gateways SLA, which has been implemented since the version 12.4. Finally we can note, that the application fulfilled our expectations and simplifies and accelerates the management of the voice gateways, which are used in the CESNET network.

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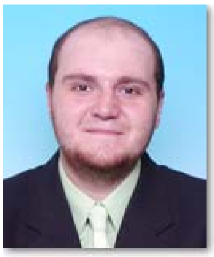
Miroslav VOZNAK holds a position as an associate professor at Department of telecommunications in Ostrava, Czech Republic. He received his M.S. and Ph.D. degrees in telecommunications from the VSB – Technical University of Ostrava, in 1995 and 2002, respectively. He is author or co-author more than 130 publications indexed in research database RIV (Research and Development IS, Czech Republic) and he solved overall 13 grants.. Topics of his research interests are Next Generation Networks, IP telephony, speech quality and network security.



Filip REZAC received his M.S. degree in telecommunications from VSB - Technical University of Ostrava, Czech Republic, in 2009 and now he continues with Department of Telecommunication as assistant and also continues in studying Ph.D. degree on the same department. His research is focused on Voice over IP Security, Networks and Speech Quality.

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