

Effect of Network factors on VoIP

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Abstract— “Internet’ is desired services platform at home. The PSTN telephone is replacing to the VoIP (Voice over IP) telephone. Call cost of the VoIP telephone is inexpensive. But, it is not quickly wide-spread device.

There are various reasons. One of reasons is a phone voice quality.

The VoIP phone is using network. Network has many obstacles (Jitter, Drop, Delay, and etc.). They are reasons of bad voice quality. But, we don’t know about what is important network factor for VoIP.

In this paper, we operated testing for standard and quality basis.

Keywords— VoIP, VoIP Phone, Delay, Drop, Jitter

I. INTRODUCTION

PSTN Telephone was constructed during 100 years. Internet was used by telephone (Modem). Now, Voice over IP technology is able to using telephone by Internet.

Quality and reliability were the two darkest spots on VoIP's reputation for the past years. Now, in many cases, gone are the days when using VoIP was like testing walkie-talkies! There has been much improvement. But still, people are very finicky about voice quality in VoIP because they are used for years to the impeccable quality of landline phones. Here are the main things that affect voice quality in VoIP and what can be done to maximize quality [2].

VoIP Phone's voice quality becomes different by network condition. But, we don't know that voice quality is varied according to network factor changing.

II. RELATIVE WORKS

2.1 Measuring Voice quality

Speech quality is only one of the different factors ultimately determining the acceptability of a telecommunication service [3].

Test methods are a lot of kinds.

2.1.1 Subjective Testing

Subjective testing is the most "authentic" method of measuring voice quality however is a specialized and costly process. This approach is typically used by CODEC designers and equipment manufacturers to validate VoIP

technology before deployment. A MOS score ranges from 1 for an unacceptable call to 5 for an excellent call. A typical MOS range for Voice over IP would be from 3.5 to 4.2.

2.1.2 P.861 (PSQM)/ P.862 Testing

PSQM and the newer P.862 are used to analyse the distortion that has occurred on test voice signals that have been transmitted through a VoIP network, and to produce an estimated MOS score. These algorithms are implemented in test equipment available from a number of companies. The advantage of this approach is that it is measuring the effects of much different impairment and their interactions - the disadvantage is that it requires a call to be set up through the network for each test.

2.1.3 Non-Intrusive Monitoring

Non-Intrusive or Passive Monitoring examines a stream of voice traffic and produces a transmission quality metric that can be used to estimate a MOS score. This has the advantage that all calls in a network can be monitored without any additional network overhead but the disadvantage is that effects of some impairment are not incorporated.

2.1.4 The E Model

The E Model was originally developed within ETSI as a transmission planning tool, described in ETSI technical report ETR 250, and then standardized by the ITU as G.107. The objective of the model was to determine a quality rating that incorporated the "mouth to ear" characteristics of a speech path. The range of the R factor is nominally from 0-100 however values of below 50 are generally unacceptable and typical telephone connections don't get above 94 giving a typical range of 50-94. For wideband CODECs the R factor may increase above 100, typically 110 for an unimpaired connection.

The basic model is: $R = R_o - I_s - I_d - I_e + A + W$

Where R_o is a base factor determined from noise levels, loudness and the others. I_s represents impairments occurring simultaneously with speech, I_d represents impairments that are delayed with respect to speech, I_e represents the so-called "equipment impairment factor" and A the "advantage factor" and W is a wideband correction factor.

The Equipment Impairment factor, I_e , is typically used to represent the effects of Voice over IP. For example, assuming default values for everything else then the R Factor for an "ideal" G.729A connection with no loss, jitter or delay would be $R = R_o - I_e = 94 - 11 = 83$

The Advantage factor, A , is used to represent the convenience to the user of being able to make the phone call, i.e. a phone is convenient to use therefore people are more forgiving on quality[4].

Table 1. Comparison of R Value and MOS Scores[1].

R-Value	Characterization	MOS
90-100	Very satisfied	4.3+
80-90	Satisfied	4.0-4.3
70-80	Some Users Dissatisfied	3.6-4.0
60-70	Many Users Dissatisfied	3.1-3.6
50-60	Nearly All Users Dissatisfied	2.6-3.1
0-60	Not Recommended	1.0-2.6

We selected PSQM MOS and E-Model. PSQM MOS is only voice quality measuring score. But E-Model (R Score) is voice quality measuring include delay.

Table 2. VoIP Phone test result in Best Condition

VoIP Phone	G.711		G.723.1		G.726		G.729	
	One way Delay(ms)	MOS (R)	One way Delay(ms)	MOS (R)	One way Delay(ms)	MOS (R)	One way Delay(ms)	MOS (R)
D VoIP Phone	119.03	3.65 (71.1)	-	-	-	-	150.9	3.38 (65.51)
C VoIP Phone	190.44	3.72 (72.7)	182.77	3.67 (71.58)	-	-	-	-
W VoIP Phone	227.86	4.15 (83.5)	165.95	3.28 (63.5)	-	-	262.95	3.24 (62.7)
A VoIP Phone	56.3	4.02 (81.5)	94.0	3.55 (71.5)	55.9	3.98 (79.8)	58.3	3.72 (73.4)
K VoIP Phone	50.3	4.48 (91.8)	92.3	3.85 (74.5)	54.8	4.29 (91.6)	59.9	3.94 (78.4)

※ - : VoIP Phone has not audio codec.

III.EXPERIMENTATION

3.1 Inside of the country, business VoIP Phone field Test

3.1.1 Test in Best condition

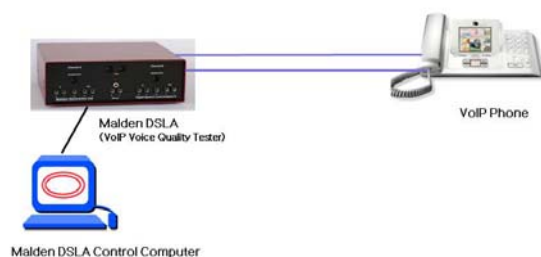


Figure 1. Test Configuration(Best Condition)

In Figure 1, VoIP Phone is tested in best conditions. Because, we need index value for comparison.

3.1.2 Test in Internet service provider's service

We joined VoIP Phone Service Provider's service, and measured telephone service's voice quality.

- Network : S networks 2 subscribers (Daejeon 2)

- Phone : S Phone, D Phone

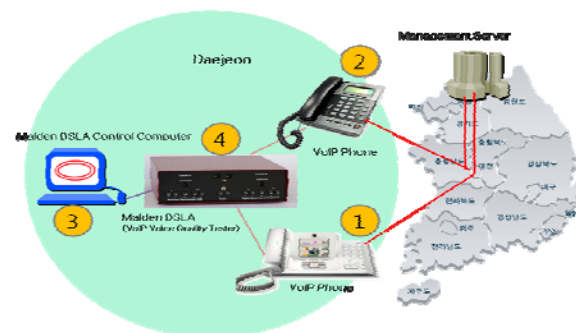


Figure 2. Field Test Configuration(Daejeon ↔ Seoul ↔ Daejeon)

Table 3. Business VoIP Phone field quality standard

Company	Codec	Measure	Measure Frequency	Measure Time
S	G.711-A/u, G.729A/B	PESQ MOS	20times per codec	3min/call
D	G.711-A/u, G.729A/B	E-R value Delay		

※ MOS : PESQ algorithm Voice Quality Mean Opinion Score

※ One way Delay: Sender/Receiver incoding/decoding processing time between two terminals and network delay time

※ Measure Time : PSTN 1 Call

Testing Process is below.

- ① In daejeon, S Terminal, D terminal's sender & receiver connect to DSLA II(Voice Quality tester)
- ② Selection of VoIP Phone's codec
- ③ Checking of VoIP Phone call condition, keeping call by taking up the receiver
- ④ Select Standard Voice Source(American, British 8k voice), setting on bi-direction(Send/receive)
- ⑤ Measure Voice quality by DSLA II

※ Test repetition times are 10times per direction. (All 20times)

Table 4. Result of VoIP Phone Test result

Time	Voice Codec	E-Model R Score		One-way Delay (ms)
		D VoIP Phone	S VoIP Phone	
9:00	711u	65.9	90.3	192.26
	729a	61.7	66.9	203.43
11:00	711u	63.3	86.3	165.25
	729a	42	66.9	224.93
13:00	711u	51.6	87.4	187.53
	729a	51	70.6	260.38

15:00	711u	64.2	73.3	152.30
	729a	50.2	63.9	178.46
18:00	711u	79.4	86.9	158.64
	729a	65	68.6	208.08
20:00	711u	77.5	78	193.99
	729a	49.4	54	216.16
22:00	711u	80.6	80.6	107.99
	729a	70.5	71.6	168.09
0:00	711u	90.4	87.9	121.11
	729a	76.5	75	126.94
Average	711u	71.62	83.84	159.88
	729a	58.29	67.19	198.31

VoIP Phone test in real network.

o VoIP Phone test's result is that it needs network situation and terminal quality standard.

- S VoIP Phone G.711u's R score is average 84. Call quality is usual thing, but Delay is little slow (Delay average is 159.88ms). D VoIP Phone's R score is average 72. Call quality is low, and delay is slow (Delay average is 198.31ms).

- It have to improve VoIP Phone's quality (Good quality of Audio module, High performance of Processor)

- It needs quality guarantee of Network.

- Network using is increasing in afternoon, audio quality is bad.

3.1.3 Test in plural Internet service providers's services

This test is different Internet service providers(ISP). Because, we needs result between ISPs.

We tested 4 networks.

First, We tested in Testbed network(Figure 3). This network is for experimetation. So, this is slow network.

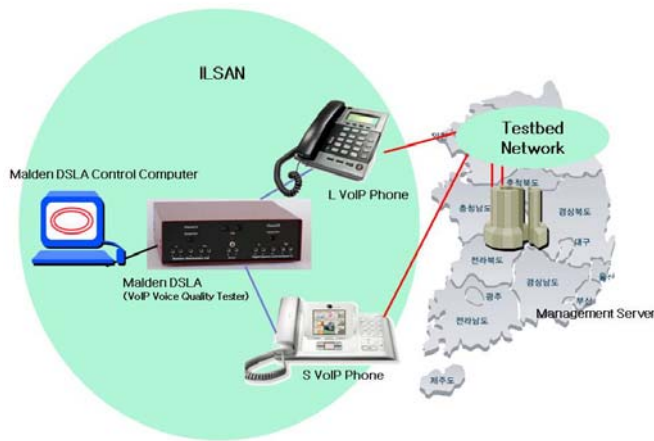


Figure 3. Test Configuration(Testbed network)

Second, Commercial A network is ISP A's Internet Services(Figure 4). This is Metro Ethernet and Fast Ethernet.(up to 100Mbps)

This network is very fast and stable network.



Figure 4. Test Configuration(Commercial A network)

Third, Commercial B network is ISP B's Internet Services(Figure 5). This is Optical Network.(up to 1Gbps)

This network is very fast network.

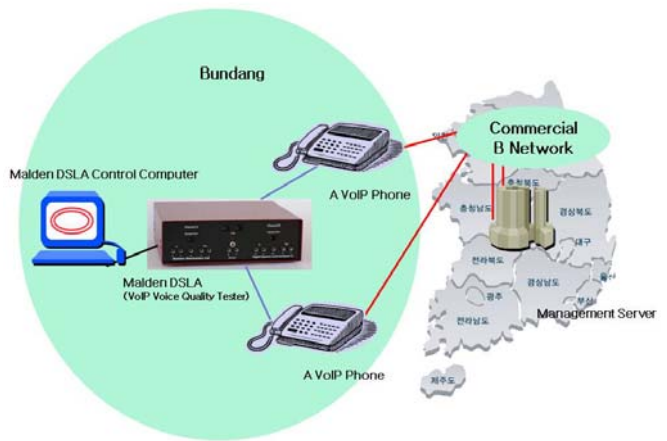


Figure 5. Test Configuration(Commercial B network)

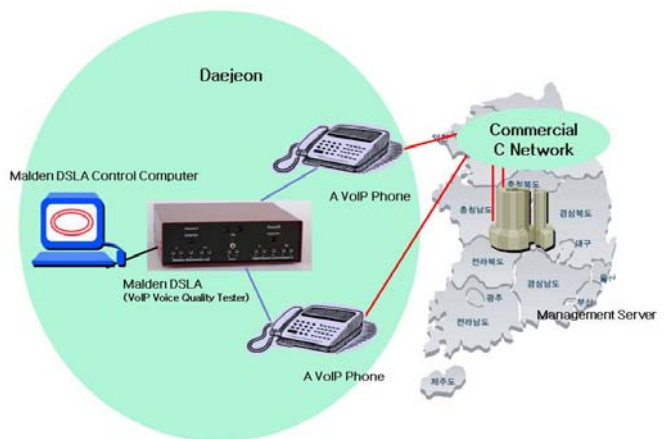


Figure 6. Test Configuration(Commercial C network)

Fourth, Commercial C network is ISP C's Internet Services(Figure 6). This is Optical-Fast Ethernet Network(100Mbps).

This network is very fast and stable network.

Table 5. Test result in different ISPs.

	Testbed Network	Commercial A Network	Commercial B Network	Commercial C Network
PESQ MOS	3.80	4.29	3.90	3.95
R-Score	85.1	93.2	88.6	91.5

In Table 5, network's status is very important for VoIP's voice quality. B Network and C network used same VoIP Phones.

According to this result, VoIP Phone's Audio device is best important, and network is important factor, too.

3.2. Local Test

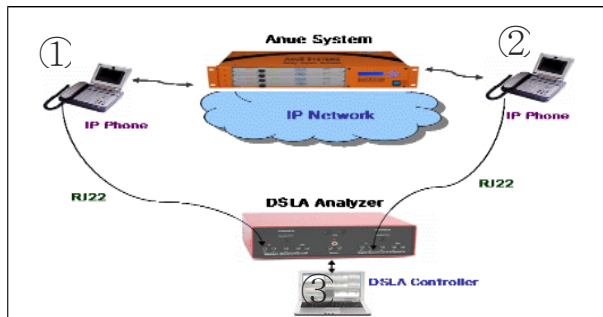


Figure 7. Local Test configuration

We added Network Impairment Equipment(ANUE). We change network situation(Delay, Packet Loss, Jitter), and measure voice quality.

Testing Process is below

- (1) Select Audio Codec between IP Phones
- (2) Check call status between IP Phones, and call busy
- (3) Setting Standard Audio Source(American, British's 8k Sampling Source) on DSLA Analyzer, and setting Sender (①)/Receiver(②) Bi-direction Test.
- ※ Test Repetition is 20 times per direction, all 40 times.
- (4) Set up One way Delay, Jitter, Packet Loss and Test by ANUE

※ In First test, all network indexes are set '0', and it puts to the Internet Phone's quality test on best network condition.

- Delay measured range : 40ms ~ 100ms
- Jitter measured range : 20ms ~ 60ms
- Loss measured range : 10-2, 5x10-2, 10-3, 5x10-3, 10-4, 5x10-4

(5) Received Traffic data, Measure quality indexes(R score, One way Delay, Jitter).

We choiced K VoIP Phone. Because It has gestic quality voice devices.

3.2.1 Test result.

This test is same phone and different network conditions.

We tested delay, jitter, drop.

Drop is packet drop. So, 10^{-4} means it drops 1packet per 10,000 packets sending. Delay and Jitter's unit is millisecond.

Figure 8 is DSLA's test result screen. We selected complex network configuration. Because, we need result closed real network. But, network factor is changed by degrees.

Table 6. Test result in different network conditions

Network Impairment			Codec								
			G.711u-law			G.723.1			G.729A		
Delay	Jitter	Drop	MOS	Delay	R	MOS	Delay	R	MOS	Delay	R
0ms	0	0	4.48	50.3	91.8	3.85	92.3	74.5	3.94	59.9	78.4
10ms	Δ 10ms	10^{-4}	4.48	79.9	91.3	3.84	122.4	74.7	3.93	79.8	78.4
20ms	Δ 20ms	10^{-4}	4.48	99.9	90.7	3.81	162.4	73.2	3.93	109.8	77.7
20ms	Δ 25ms	10^{-4}	4.48	109.9	90.5	3.82	137.4	74.2	3.92	119.8	77.3
30ms	Δ 20ms	10^{-4}	4.48	114.9	90.2	3.84	167.2	73.5	3.91	114.8	77.2
40ms	Δ 25ms	10^{-4}	4.48	124.8	90.2	3.84	182.4	72.5	3.93	134.8	77.2
20ms	Δ 20ms	10^{-3}	4.43	99.9	90.7	3.79	162.4	72.8	3.89	104.8	76.8
20ms	Δ 25ms	10^{-3}	4.44	104.9	90.6	3.74	137.4	72.3	3.91	104.8	77.0
40ms	Δ 25ms	10^{-3}	4.42	125.7	90.2	3.81	182.4	71.7	3.92	134.8	76.8
40ms	Δ 30ms	10^{-3}	4.43	139.9	89.9	3.82	217.4	68.0	3.87	144.8	75.4
40ms	Δ 40ms	10^{-3}	4.43	159.9	89.4	3.77	182.4	70.9	3.93	169.8	75.1
40ms	Δ 50ms	10^{-3}	4.38	164.9	88.9	3.84	217.4	70.2	3.87	174.8	73.9
40ms	Δ 25ms	10^{-2}	4.14	134.9	85.2	3.69	182.4	68.8	3.68	134.8	71.0
40ms	Δ 25ms	2×10^{-2}	3.39	124.9	66.4	3.27	202.4	61.1	3.15	144.8	61.2
40ms	Δ 25ms	3×10^{-2}	3.27	150.3	63.7	2.92	202.4	52.7	3.16	149.9	60.8

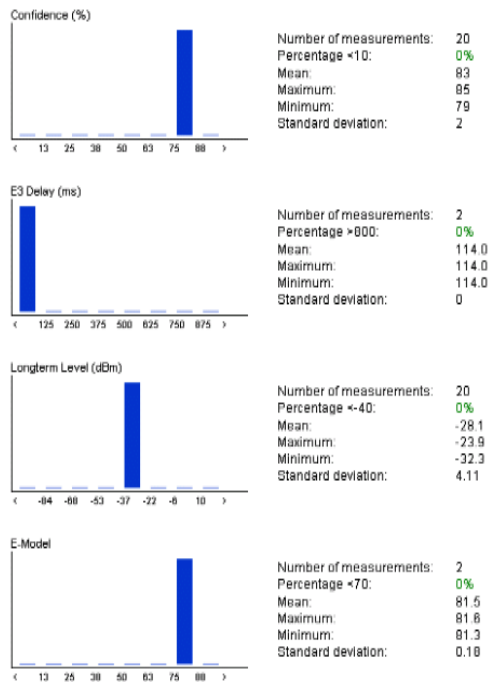


Figure 8. Test Result Screen

Summing up above results, best important network factor is packet drop for voice quality. VoIP Phone is able to revise voice, because it has buffer. But, VoIP phone must

have good quality audio devices, and good processing time, because of delay.

The voice Quality measuring is depends upon voice waveform. The waveform is very sensitive to drop of packet, but it is insensitive to delay. If a voice starting point moving to delayed starting point, waveform should be same. In this case voice quality is good. But, receiver is uncomfortable of calling.

IV. CONCLUSIONS

VoIP became popular because of low call cost. But people have preconceived Idea about voice quality and etc. We prove that it reduces drop of packets, and raises VoIP Phone's processing speed for VoIP's quality.

In the future, this paper's data is useful for VoIP call quality standard and basis. And, we need study of VoIP emergency call.

ACKNOWLEDGMENT

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