Testing the effect of load on delay and voice quality in VoIP networks

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Abstract: - An implementation of a VoIP network brings a lot of advantages, mainly lower purchase and operating costs for the operator and advanced phone services at very reasonable prices for network users. In addition, the new type of network that is based on convergence of telephony and data network on IP principles offers to users unrestricted access to different service providers and supports generalized mobility which will allow consistent and ubiquitous provision of services to users. Despite all the updates the voice quality remains the main parameter for assessing the network quality. Nowadays, we can find a series of software tools working in accordance with the ITU-T recommendations and laws of statistics. The tools are designed to assess the current state of ongoing VoIP communication. We chose Alcatel VoIP Assessment Tool 3.0 and widely known CommView SW. Firstly we monitored and assessed a live network in normal working hours. Then the influence of load on delay and consequently on voice quality was tested. These results are obtained from the measurement done on systems which are normally performed in real networks of the Armed Forces of the Czech Republic (the ACR).

Key-Words: - VoIP, quality, testing, Alcatel VoIP Assessment Tool

1 Introduction

VoIP platform is a new modern sort of communication network that refers to transportation of voice, video and data communication over IP network. Nowadays, the concept of VoIP cannot describe all possible kinds of services of next-generation communication systems. Thus, other concepts such as NGN (Next Generation Network) or IPMC (IP Multimedia Communications) start to be widely used. NGN has been defined according to ITU-T Y.2001 Recommendation as a packet-based network able to provide a wide number of multimedia services, including VoIP. IPMC has been introduced to be more descriptive than the VoIP term because next generation system will provide much more than simple audio or video capabilities in a truly converged platform, for example instant messaging, email, and all other kinds of packet-switched communication services.

On one hand implementation of the VoIP network brings a lot of advantages for both providers and network users, especially cost savings. On the other hand reliability and quality of transmitted speech can never be better than in conventional networks. Also security of VoIP systems is questionable. The VoIP networks work on well-known TCP/IP protocols, so they are more open to common types of attacks well known from data networks. So network safety becomes a big challenge for network operators.

Regardless of the number of advanced voice services that are offered by the new VoIP environment, a key criterion for assessment of the service quality still remains speech quality. With regard to the method of voice transmission which VoIP networks operate we have to use new approaches and new parameters for measurement of a VoIP transmission channel [5]. Nowadays, we can find a series of software tools working in accordance with the ITU-T recommendations and laws of statistics. The tools are designed to assess the current state of ongoing VoIP communication. We chose Alcatel VoIP Assessment Tool 3.0 and widely known CommView SW.

In the previous part of the project we tested chosen parameters such as delay, jitter and packet loss and their various combinations that most affect the quality of the transmitted voice. The results
obtained were surprising because they did not confirm substantially greater impact of large delay on speech quality. The load with a large jitter and batch mode of operation had a more significant influence on the voice quality than packet delay [2]. So in this part of the project, at first we monitored and assessed a live network in normal working hours. Then we focused on the detection of limit values for the delay which already significantly degraded the voice quality. Finally impact the ratio of payload and overhead on the quality of the speech was verified.

2 Description of software platform used
As mentioned above, nowadays, we can find a series of software tools that are designed to assess the current state of ongoing VoIP communication. In order to verify the quality of speech signal, Alcatel VoIP Assessment Tool 3.0 in the IP environment is utilized for measurement experiments. To supplement and verified obtained results, we consequently used the ComView software platform. Series of tests was carried out gradually in a live network in normal working hours and in laboratory conditions in LAN and WAN built from Cisco and Alcatel – Lucent products [3].

2.1 Alcatel VoIP Assessment tool
Alcatel VoIP Assessment Tool is a PC application designed by Alcatel for:
- VoIP Network Assessment in order to qualify Data Network for transport of VoIP flows. In this case, the tool has to be used in combination with tools like Alcatel VoIP Simulation Tool or Smartbit for example, in order to stress the Data Network with VoIP flows.
- Pro-active management of VoIP networks in order to anticipate degradation of Quality of service.
- Troubleshooting VoIP audio quality problems by collecting Quality of Service (QoS) informations provided by dedicated IP-Phones.
- VoIP demonstrations through “Immediate Mode” which allows establishment of VoIP calls between IP-Phones in a real environment.

The VoIP Assessment Tool application is based on a range of e-reflexes IP-Phones (IP-Phone v2) or IP-Touch phones which were designed to provide Quality of Service information through “QoS tickets” at the end of each communication. The application offers following possibilities:
- Setup an arbitrary number of calls between connected IP Phones.
- Launch VoIP calls on a predefined schedule.
- Choice of the compression algorithm (G711, G723.1 at 6.4Kb/s, G729).
- Level 3 QoS ToS/Diffserv tagging for each individual call.
- Level 2 QoS 802.1p/Q tagging for each individual call.
- Easy to use “Immediate Mode” for demonstration purposes: Possibility to launch up to 3 calls manually and analyze resulting QoS tickets.
- Extraction and analysis of IP Phone Quality of Service (QoS) tickets.

Limitations of VoIP Assessment Tool:
- Incompatibility with Network Address Translation (NAT) / Port Address Translation (PAT) as it is not supported by IP-Phones.
- Static IP addressing mandatory for IP-Phones: DHCP not supported.

VoIP Assessment Tool QoS evaluates the quality of communication with tickets received from IP phones after ending the current session. Analysis of the results obtained by monitoring call quality in real network will be presented in the next chapter. Therefore, here is only a brief definition of the parameters as they are understood by software developers.

Each ticket is represented on a single line, with the following information for each ticket:
- Time: The time the ticket was received in VoIP Assessment Tool.
- Local device: The user label or the IP address of the IP phone sending the ticket.
- Duration: The duration of the call in seconds, as reported by the local IP phone.
- DSP samples: The number of DSP samples played by the local IP phone during the call.
- RTP Packets (Sent / Received / Lost): The number of RTP packets (on the IP link) sent and received by the local IP phone. The number of lost packets is the number of packets lost between remote and local IP phone.
- Codec: The codec used by IP phones during the call.
- MOS: Estimated Mean Opinion Score attributed to the call.
- Packets Lost: The percentage of RTP lost packets in the direction from remote IP phone to local IP phone compared to the number of
packets that should have been received in that direction.

- **Roundtrip Delay (good, acceptable, fair, poor)** - Four columns giving the number of measurements of the network roundtrip delay time that occurred in the Good, Acceptable, Fair and Poor quality ranges.

- **Jitter (good, acceptable, fair, poor)** - Four columns giving the number of measurements of the jitter buffer depth that occurred in the Good, Acceptable, Fair and Poor quality ranges.

- **Voice Continuity (good, acceptable, fair, poor)** - We measure for every 100ms of the conversation how the voice packets are received at the digital signal processor (DSP) level. The values in the columns are the number of 100ms voice packets that have been played back normally or have been discarded in more and more quality-unfavorable ("bursty") conditions.

- **Max RT Delay (ms)** - Maximum value of roundtrip delay measurements. The maximum roundtrip delay is the highest value measured from the RTCP roundtrip delay measurements. This value must be checked only when implementing transparent modem/data services over IP, and can be ignored when analysing VoIP call quality. The quality criteria applies to transparent modem/data over IP services only. They present the reliability thresholds for which transparent modem/data operation is possible.

On each line and for each of the quality parameters, the field with the lowest quality is shown in the colour corresponding to the quality level.

### 3 Measurements and results

For testing the voice quality in VoIP networks several tools can be used. One of them is the above mentioned Assessment Tool software developed by Alcatel – Lucent company. In the first task, we used this tool to test the effect of the real network behavior on voice quality.

#### 3.1 Testing call quality in a real network

The measurement was carried out in the university network by using a control station (a PC with Assessment Tool) and two Alcatel IP Touch 4038 phones. One phone is connected to the same switch as the control station, and the second one to a laptop, working in bridge mode, uses the Eduroam Wi-Fi network. An example of workplace and a network topology is shown in Figure 1.

![Fig. 1.: Workplace for the measurement of voice quality in a real network](image)

The control station also operates as a Call Server to which IP phones are connected. The phones exchange short messages KEEP A LIVE with the control station in a period of three seconds (even during a call). Then, the Assessment Tool allows remote building of a standard phone call between the selected phones. Apart from the RTP stream, the phones exchange RTCP packets, carrying information about the quality of the ongoing call, every five seconds. After ending a phone call, the phones send tickets containing information about call quality to a control station. The control station stores tickets and allows displaying the results in a graphical form. It is important to realize that the phones transmit an actual real RTP stream which enables subjective evaluation of voice quality using the call / listening over headphones.

In case of this measurement, we set the repetition of the call in a loop with a 30 seconds repeating period. The measurement took one day. One segment of the results obtained is shown in Figure 2. Based on information received from tickets the software evaluates the quality of speech, according to the MOS scale. The end phones get their information from RTCP messages.

The RTCP messages include information about:

- lost packets;
- round trip delay;
- jitter.
The phones have also evaluated a system parameter called voice continuity (the number of 100 ms voice segment that have been played back normally).

![Fig. 2: The results of measurements of the voice quality in a real network](image)

The results obtained show that during testing random sporadic packet losses occurred. These losses were most likely caused by using the transmission WiFi technology, but they had practically no effect on the call quality. Among the parameters monitored, jitter varied the most. It corresponds to a random batch character of data loading in the monitored network (the university network does not use VoIP). Despite the fluctuation of jitter, the resulting voice quality ranged generally from 4.2 to 3.8, according to the MOS scale.

### 3.2 The effect of delay on the voice quality

The Assessment Tool also allows the simulation of delay, jitter and loss rate of the transmitted RTP stream. In such a case, the call does not proceed directly between phones, but it is executed between the phone and the control station that delays and drops packets according to selected parameters and forwards the modified RTP stream to the second telephone device.

Information about the quality of a phone call is exchanged also indirectly through the control station using the RTCP protocol. The phones send tickets to the control station after ending of the call. We used these properties in another task, in which we measured the influence of constant delay on the voice quality. When measuring, the control station and end phones were connected to the same switch.

In Figure 3 is shown the result of the measurement of a five-minute call when we simulated constant delay of 100 ms (200 ms in the loop). The voice quality was assessed by value of 4.3 or 4.2, according to the MOS scale. In both directions of communication is a discrepancy between the number of packets sent and received, but the loss is 0%. When repeating the test, the situation was always the same. In addition, for short calls (up to 30 s) the software tool evaluated the quality of voice between 2 to 3 on the MOS scale. A more detailed analysis of signaling reports and RTP streams discovered an error of SW.

![Fig. 3: Simulation results of a constant delay of 100 ms](image)

The measurements were repeated for constant delays of 150 and 300 ms. The results are shown in Table 1.

### Table 1: Influence of constant delays on the voice quality

<table>
<thead>
<tr>
<th>Size of delay</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>4.2</td>
</tr>
<tr>
<td>150</td>
<td>4.2</td>
</tr>
<tr>
<td>300</td>
<td>3.8</td>
</tr>
</tbody>
</table>
From the results shown in the table it is evident that the constant delay has not a significant impact on the voice quality. Up to 500 ms in the loop it is completely acceptable.

3.3 Effect of payload on the voice quality

In this task, we measured the influence of load (or number of simultaneous calls and the size of their payload) on the voice quality. There we had two Cisco 2811 routers with Cisco Unified Communications Manager Express (CUCME), version 7.1. According to the documentation, these routers allow the processing of up to 42 phones. Because we do not have a sufficient number of IP phones, we used the properties of routing of telephone calls (dialed-peers) and the translation of telephone numbers. Routers are configured in such a way that when dialing a phone number x9900, it was gradually translated to up to x4414 value and the resulting RTP stream was forwarded between routers. After reaching the number x9914, the call was switched to number 20x. (Note: The CUCME, version 8.6 it is not longer possible, the software detects a looping call routing.) So one telephone device simulated the load of one line with 15 current RTP streams. The connection of the workplace and the principle of routing is shown in Figure 4.

![Fig. 4.: An example of workplace and call routing rules](image)

We collected the communication between the routers using the CommView and Wireshark programs. The disadvantage of CommView SW compared to Wireshark CommView is a limitation of maximum storage and evaluation of maximum of 20,000 packets. The advantage is a more sophisticated graphical analysis of VoIP traffic. For communication we selected the G729 codec with 20 B payload size. It corresponds to processing 20 ms segment of the voice.

Dialing of 5 phone numbers (it corresponded to transfer of 75 concurrent calls between routers), already caused an overly large delay of packets (at 15 passages through the active elements the delay in one direction reached several 100th ms), so it was impossible to communicate at all. The situation is shows in Figure 5.

![Fig. 5.: The quality of voice of 75 calls and 20 B payload size](image)

We repeated the same experiment but for the 100 B payload size (it means processing and transmission of 100 ms voice segments). We need to transfer the same amount of useful information, but we will use 5 times less packets for it. This step reduces the computational load of routers and bandwidth required, but at the expense of increasing packet delay [1]. The measured results are shown in Figure 6.

![Fig. 6.: The quality of voice of 75 calls and 100 B payload size](image)
4 Conclusion

Based on the obtained results, quite surprising conclusions can be drawn. The packet delay does not significantly affect the voice quality (in terms of readability) in VoIP networks. Even a greater delay than stated in the standard ITU-T G.114 is acceptable [4]. Jitter and packet loss have more significant influence on comprehensibility. We also showed the influence of the size of the RTP packet payload on the voice quality. Increasing payload will increase a packet delay, but the number of transmitted packets drops significantly (in our case 5 times). This will lead to a lower load of routers, more efficient bandwidth utilization, and, ultimately, to a significantly smaller delay in routers (in our example there are 15 routers in the transmission path), which has a greater importance than the extension of packet delay with a large number of calls. We want to continue to work on determining the optimal payload size for the G729 codec that is often used in VoIP networks with regard to a special character of the military network.

References: