ELECTRONICS AND ELECTRICAL ENGINEERING
ISSN 1392 – 1215 2012. No. 1(117)
ELEKTRONIKA IR ELEKTROTECHNIKA

SYSTEM ENGINEERING, COMPUTER TECHNOLOGY
T 120
SISTEMŲ INŽINERIJA, KOMPIUTERINĖS TECHNOLOGIJOS

Comparative Analysis of the Performance of Different Codecs in a Live VoIP Network using SIP Protocol

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crossref http://dx.doi.org/10.5755/j01.eee.117.1.1050

Introduction

Voice over IP (VoIP) is the ability to transmit speech over packet-switched IP networks ant it is the common term for telephone service over the Internet. If the user has a satisfactory quality Internet connection voice signals can be delivered over an Internet connection instead of local phone company. Establishing a phone call using VoIP will create a digital signal from the analog input, place those digital signals into packets with source and destination network addresses and finally send the information over the Internet or internal company IP networks, thus bypassing the need for the PSTN lines. However, it is important to note that the source and destination terminals must support the codec for the correct encoding and decoding. Today, most popular VoIP standard is Session Initiation Protocol (SIP). This standard provides direct call between VoIP (SIP) terminals, or the use of media gateways, which can be used to negotiate connections between TDM and/or other endpoints. In VoIP, quality of service (QoS) simply means being able to listen and speak in a clear and continuous way, without unwanted noise or other distortions [1]. QoS depends on the technical parameters: delay, jitter and packet loss. In VoIP a delay of 150ms is acceptable, while a higher value results in degradation of voice quality, which becomes unacceptable at values higher than 300 ms [2].

As we know the term “service” can denote many things in networking. The VoIP service most commonly defines what an operator offers to the consumers to fulfill their expectations from the telephony communication. User’s satisfaction also consider as a big problem, if the users want to compare the quality of VoIP with circuit switch service, certainly they would complain from the inaudible conversation and from the delay of voice arriving specially if the calls come from long distance location. According to the speech quality they achieve, as measured by mean opinion score (MOS), codecs can be classified in two categories: toll quality and near toll quality [3]. MOS is subjective methods aim to find the average user perception of a system’s speech quality by asking a panel of human listeners and provide them a limited response choice. ITU-T in Recommendation P.800 introduced MOS based on user perception that is ranged from 1 (poor) to 5 (excellent) for subjective determination of voice quality.

The one of the most important roles in VoIP plays codec [4]. It determines QoS and how much bandwidth is needed for a call. The rule of thumb is the higher the voice quality, the more bandwidth is required for the call. The Table 1. [5] Shows the codecs that are commonly used in VoIP communications:

Table 1. Common VoIP Codecs

Codec/ Bandwidth (kbps)	Comments
G.711/ 64	Delivers precise speech transmission. Very low processor requirements. Needs at least 128 kbps for two-way.
G.723.1/ 5.3,6.3	High compression with high quality audio. Can use with dial-up. Lot of processor power.
G.729/8	Excellent bandwidth utilization. Error tolerant.

The paper [1] measured the performance of VoIP traffic over the public Internet in terms of packet loss and depending on the application of different codecs. Resulted in the loss of packages depending on the traffic load that is not acceptable for use in the public operator (the operators of public networks loss should be less than 3%). The objectives of this research are: analyzing the different techniques of coding/decoding in VoIP, measurements on the live network using the codecs G. 711, G. 723 and G. 729, evaluation and performance testing of selected codec for VoIP. This simple article is arranged through seven sections, which is organized as follows. The second section gives an introduction of VoIP network architecture, while in the third section are listed codecs commonly used in VoIP networks as well as their characteristics that affect to the quality of service and bandwidth calculation. In the fourth section we will illustrate a vary researches which have been compared chosen codecs depend of packet loss. The fifth section covers objectives of study, scope and scenario and gives explanation for the methodology which will be used to fulfill the objectives of this work, which were mentioned earlier in this section. The sixth section shows the results and evaluations which are come out from the measurements. Finally, section seven concludes and discuses all the finding.

Introduction to VoIP network architecture

The Fig. 1., shows the architecture of a VoIP network in which measurements were performed.

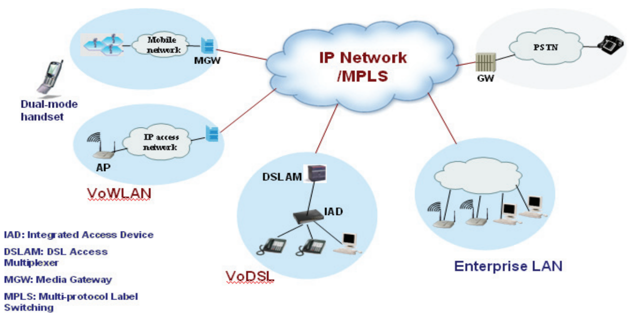

Fig. 1. VoIP network architecture

The VoIP network consists: the softswitch class 5, the softswitch class 4 to the respective media gateways (MGW) and IP MPLS Network [9]. As a signaling protocol used in softswitch is Session Initiation Protocol (SIP). SIP is a peer-peer signaling protocol for VoIP, developed by the IETF and defined in RFC 3261. It is a proposed standard for initiating, modifying, and terminating an interactive user session that involves multimedia elements such as video, voice, instant messaging, online games, and virtual reality. SIP requires a simple core network with intelligence embedded in endpoints; thus it is highly scalable [6].

Bandwidth calculation

The three variables involved in bandwidth calculation are busy hour traffic, blocking probability and bandwidth. Blocking probability is the probability of call failure due to an insufficient number of lines being available. Generally, a VoIP call will use between 10 kbps and 110 kbps in one direction. The exact amount depends on a number of factors, namely: Codec type used and sample length (how often the samples are transmitted), IP header, Transmission medium (i.e. DSL, Ethernet, Radio) and Silence suppression/echo cancellation.

The codec type determines the amount of bandwidth used for the actual voice information, and also defines the sample period (how often the samples are transmitted). The uncompressed IP/UDP/RTP header refers to the additional bandwidth needed for the various headers: IP (20bytes), UDP (8bytes), and RTP (12 bytes), in a total of 40 bytes. Sticking with a sample period of 20 ms, the IP header will require an additional 16 kbps to whatever codec is used above [7]. The main variation here is when the SRTP standard is used (between 2 and 4 bytes) [9]. It is geared toward mobile networks where bandwidth is at a premium. The transmission medium, like Ethernet adds its own header and checksums to the packet. For example, Ethernet adds another 38 bytes. When silence suppression is enabled, it can reduce the necessary bandwidth by as much as 50%. Silence suppression takes advantage of the inevitable gaps in speech during a call (e.g. when one party is talking, the other usually is silent). Putting it all together it is mean as a starting point, the G.711 codec itself requires 64 kbps. Adding in IP header overhead brings the total to 71.6 kbps.

Number of samples per package affects the size of the bandwidth and the packetization delay. By increasing the number of samples, we reduce the bandwidth required for VoIP connection, since we increase the effectiveness of packet transportation. Calculation of the VoIP packet ($VoIP_{psz}$) in the IP network is given by

$$VoIP_{psz} = \left(s_f * bps * p_t/8\right) + S_H\left(IP/UDP/RTP\right), \quad (1)$$

where is s_f sampling frequency, bps denotes bits per sample, p_t is packetization (framing) time and S_H is size of headers. In case transmission voice traffic over IP MPLS network, MPLS header size must be included in total packet length together with i.e. PPP header size (if user access or tunneling mechanism is PPP) [9].

Comparison and contrast in codecs

This section will illustrate a vary researches which have been compared chosen codecs depend of packet loss. Packet loss is an important parameter affecting the performance of VoIP and other researches have been studied and analyzed how the G.711, G.723.1 and G.729 were affected with packet loss. Generally, packet loss is related with the packet length, which is proportional to transmission time associated with each packet. VoIP system can tolerate packet loss to some extend as 1% or less is acceptable for toll quality while for business quality 3% or less is acceptable. Hence, more than 3% of packet loss degrades the speech quality as a listed in [3, 8]. This research have been applied the codecs within VoIP over IP MPLS, it has measured the packet loss according to the number of calls and for a different packet length.

Objectives of study, scope and scenario

The objectives of this study are stated in the in the following points:

- Analyzing the different techniques of coding / decoding in VoIP;
- Measurements in a live VoIP network using the codecs G. 711, G. 723 and G. 729;
- Evaluation and performance testing of selected codec for VoIP.

In order to achieve the objectives of this study, it is very important to indicate to the boundaries of investigation, which are stated in the following points:

- Measurements are going to be made on softswitch class 4;
- Only address the coding/decoding techniques;
- Re-coding techniques will not be addressed in this study.

The scenario of measurement is next:

1. The selected routes are the same capacity (2Mbps) that have been established through softswitch class 4. The first route (Route 1) is established across the optical system of transmission, while the other route (route 2) is established across the wireless transfer system. Therefore, carried out the measurement of traffic that goes through different media;
2. Defined four size of VoIP packets: 64,128,256 and 390 bytes, and of course defined is codecs to be used (G.711, G.723 and G.729);
3. Defined by the value of the interval measurements (the ten minutes at the selected codec);
4. Measurements were was started simultaneously on both routes, the first week of the month, at the hour of greatest traffic load (11:00 - 13:00);
5. The results from the measurements to be evaluated by comparing the Packet Loss parameter of each codec applied with different cases.

The second part of the scenario is a survey that was made to subscribers of both VoIP nodes. Specifically, subscribers to the questionnaire distributed to where they need to enroll in evaluating the quality of each connection they have made during the period of measurement traffic. The result of survey or MOS value is given in the section with the results.

Result and evaluation

In this section, the results which obtained from measurements is analyzed and discussed to evaluate the performance of VoIP codecs. One of the most affected parameters of QoS in VoIP is packet loss. We analyzed and studied packet loss for each codec with four values of packet size (64, 128, 256 and 390 byte). The term packet loss means the event that the package that was sent from the source did not reach its destination. The percentage of packet loss ratio is the percentage of packets lost in relation to the total number of packets transmitted. This percentage is calculated based on the results of measurements of codecs G.711, G723 and G.729, depending on the number of calls, and the previously mentioned values of packet length. Fig. 2 – Fig. 5, the results obtained by measurements on Route 1, while in Fig. 6 – Fig. 9 show the results obtained by measuring the route 2:

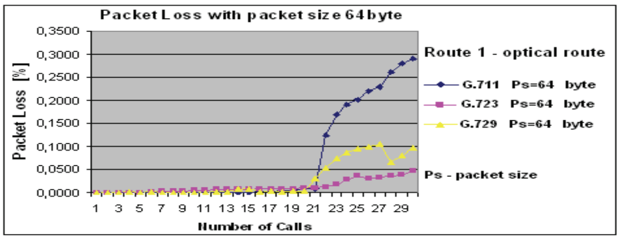

Fig. 2. Packet Loss for G.711, G.723 and G.729 with 64 byte of Packet Size – Route 1

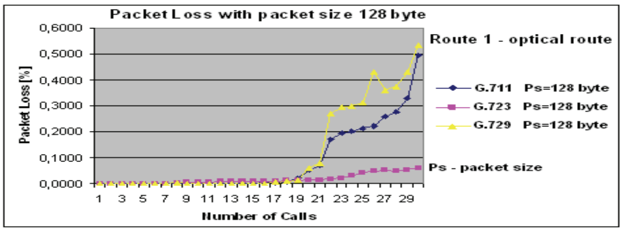

Fig. 3. Packet Loss for G.711, G.723 and G.729 with 128 byte of Packet Size – Route 1

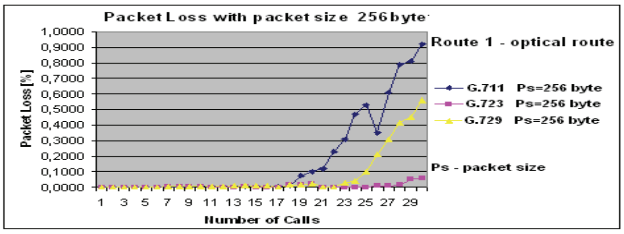

Fig. 4. Packet Loss for G.711, G.723 and G.729 with 256 byte of Packet Size – Route 1

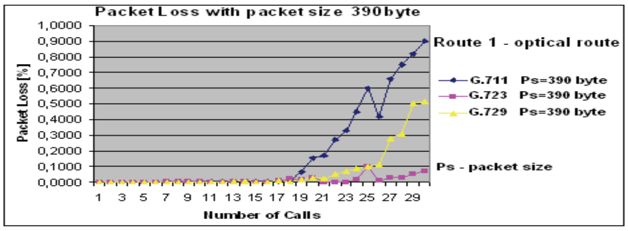

Fig. 5. Packet Loss for G.711, G.723 and G.729 with 390 byte of Packet Size – Route 1

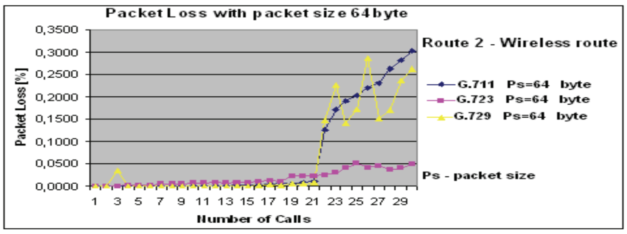

Fig. 6. Packet Loss for G.711, G.723 and G.729 with 64 byte of Packet Size – Route 2

Packet loss greatly affects the quality of VoIP service because it leads to the loss of some samples of the speech signal in the receiver. From the results of measurements made for the size of the package of 64, 128, 256 and 390 bytes, it is evident that the codecs G.711 present a significant packet loss compared to G.723 and G.729. The primary reason for the packet length and the time interval between two generated packets, which leads to the G.711 packet loss dramatically especially when the packet size 256 and 390 bytes with the number of calls from 17 or 18 to 30.

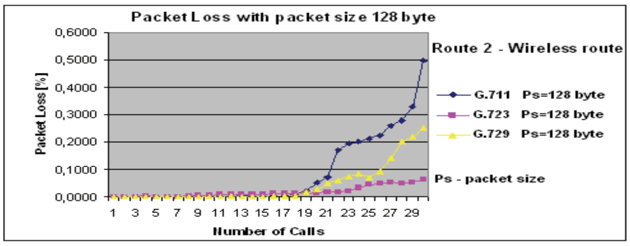

Fig. 7. Packet Loss for G.711, G.723 and G.729 with 128 byte of Packet Size – Route 2

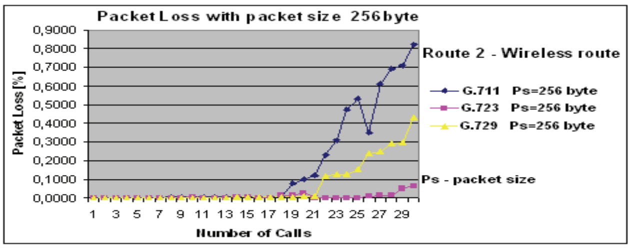

Fig. 8. Packet Loss for G.711, G.723 and G.729 with 256 byte of Packet Size – Route 2

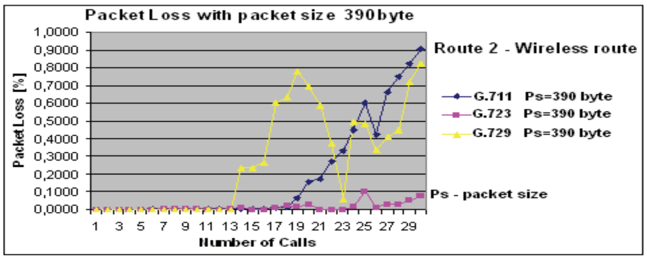

Fig. 9. Packet Loss for G.711, G.723 and G.729 with 390 byte of Packet Size – Route 2

G.723 codec has a small percentage of packet loss or even be said that there is no packet loss. The reason for this is the packet size and time interval that elapses between any two packets that follow one after another, and is 30 ms. In other words, it means that there is no overload of orders to send in routers. No matter what size package, using the G.723 codec can handle more users than it can on using the codec G.711 and G.729. Compared with them G.723 codec is almost no packet loss. Negligible packet loss is from 34 to 40 calls.

By using the codec G.729 packet loss occurs with increasing number of calls, but not as much as the codecs G.711 and G.723. The reason is that the .729 has the same intervals with G.711, G.729, but has a small packet size due to low data rates (high compression). On the other hand, is the same size packages with G.723, but the time interval of less than 10ms for G.723, which leads to packet loss when a large bandwidth.

Using the G.723 codec and still has the same distribution and can be done even with the loss of the package, though there are 6 calls where the packet loss 0%. Definitely, G.723 can operate with or without packet loss, due to the time interval of 30 ms.

The G.729 codec also has a small value of packet loss. Its packet size is small, the time interval is short, and the packages suffer algorithmic delay, so there is unlikely to come to load the package in terms of waiting to get in line to send the router and finally to reach a packet loss.

With all codecs, there is a high degradation of the capacity of calls with the packet size. With G.711 going from 64bytes to 390bytes reduce the capacity from 5 calls to 2, while with G.723 with 9 calls are possible at 64bytes. G.711 is preferred approach since it avoids both delay and data loss, hence have toll-quality voice. However, this may prove more expensive and use high capacity on channel especially when packet overhead is taken into account. Using G.729 as a higher compression speech codec are increasing the number of channels comparing with G.711. Unless a very high voice quality requirement precludes its use, G.729 as low bit rate codec is shown to allow a capacity greater than or equal to that when G.711 is used, for a given quality requirement. Unless a very high voice quality requirement precludes its use, G.729 as low bit rate codec is shown to allow a capacity greater than or equal to that when G.711 is used, for a given quality requirement. However, the low bit rate codecs such as G.723 and G.729 have lower throughput comparing with G.711 the high bit rate. Hence, G.723 has ability to provide the highest capacity of VoIP calls, due to its voice packets are little and frequently send which leads to low throughput in term of packet received. In addition, G.723 has an ability to deal with packet loss, therefore for a very busy network it will be a proper choice.

In the end, on the Fig. 10 and Fig. 11 are shown the results of survey for the subjective measurements of the quality of voice service.

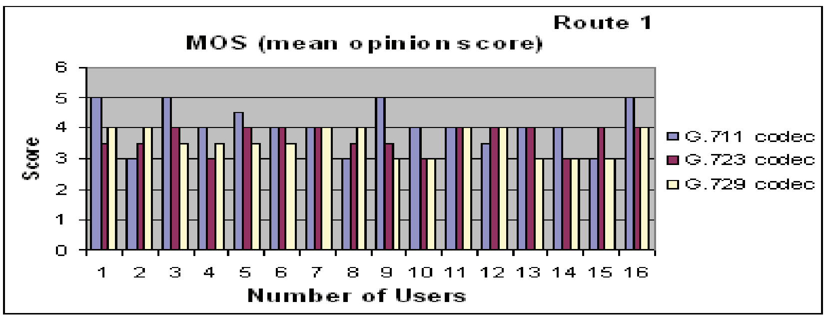

Fig. 10. Results for MOS -Route 1

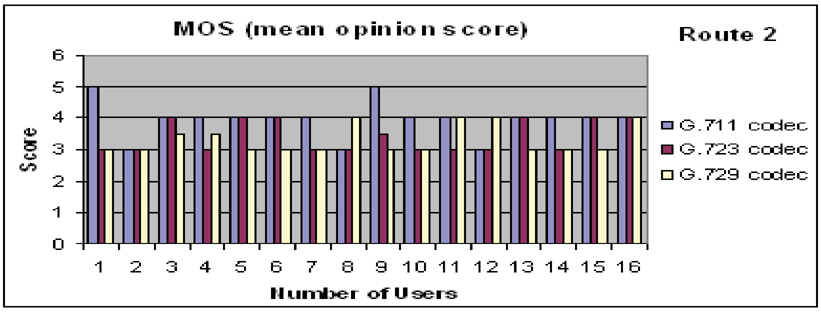

Fig. 11. Results for MOS -Route 2

Conclusions

In this research, the performances of VoIP networks were investigated by measuring quality parameters of the SIP traffic. The measurements are made for the route of the same capacity, which are carried over fiber optic and wireless transmission systems. It is evident that the number of calls and size of VoIP packets have significant impact on packet loss and bandwidth regardless of the codec used.

In conclusion, using the G.723 codec low bit rate can be processed a large number of calls with low-bandwidth, so it is G.723 the best option for managing traffic. Using the G.729 codec, which is characterized by low bit rate, but also a high degree of compression, also can process more calls than using G.711 codec. However, it must be stated with much less voice quality as G.729 has the highest value of the packetization delay. On the other hand, the G.711 codec that provides the desired high quality voice with no delay or with a small value of delay and no packet loss, except in the case of multiple calls when necessary to have a large bandwidth for a large number of packages.

Future work

We did not have the opportunity to have "extreme cases" in terms of measuring traffic on the wireless route, and thus make a better analysis of the codecs impact to the parameters of the quality of service for this type of transmission system. We will extend our work especially in the area of VoIP over wireless transmission systems and in the area of objective speech quality measurement methods for VoIP.

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Received 2011 01 26
Accepted after revision 2011 04 28

Z. Bojovic, Z. Peric, V. Delic, E. Secerov, M. Secujski, V. Senk. Comparative Analysis of the Performance of Different Codecs in a Live VoIP Network using SIP Protocol // Electronics and Electrical Engineering. – Kaunas: Technologija, 2012. – No. 1(117). – P. 37–42.

In this paper we analyzed the impact of different codecs on the quality of VoIP services. We analyzed the performance of VoIP networks to have made an assessment to what extent are fulfilled the expectations regarding the desired quality of service and the authors applied the following methodology: we selected according to certain criteria and the same codecs implemented in a VoIP network, then selected the appropriate traffic routes and measurements carried out on them, and finally at the end is based on measurement results obtained comparative analysis and the comparisons made packet loss parameter. In the paper are discussed the main problems regarding the quality of VoIP services: and the choice of coding techniques. Measurements were performed with common used codecs for VoIP: G.711, G.723 and G.729, and the variation values of packet size and number of calls. Also, measurements were performed with optical and wireless transmission system. Finally, the main objective of this study was to select an appropriate codec to get a high quality voice and tries to compromise between price/quality of service. In conclusion, G.711 is the preferred technique that provides high voice quality (MOS greater than 4) but with a flaw need high bandwidth, and G.723 provide relatively good voice quality (MOS between 3 and 4) with a much smaller bandwidth, which means it can handle multiple users. The G.729 has a high level of compression and proper technique for many users, but only when the voice quality is not taken into account. Ill. 11, bibl. 9, tabl. 1 (in English; abstracts in English and Lithuanian).

Ž. Bojovic, Z. Peric, V. Delic, E. Šecerov, M. Sečujski, V. Šenk. Įvairių kodekų naudojamų VoIP tinkle su SIP protokolu, lyginamoji analizė // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2012. – Nr. 1(117). – P. 37–42.

Analizuojama įvairių kodekų įtaka VoIP paslaugų kokybei. Pagal tam tikrą kriterijų parinkti kodekai buvo įdiegti į VoIP tinklą, buvo parinkti tam tikri maršrutai ir atlikti matavimai. Atlikta lyginamoji analizė ir paketų praradimo parametro palyginimas. Aptartos pagrindinės VoIP paslaugos kokybės užtikrinimo problemos. Matavimai buvo atlikti su dažniausiai naudojamais VoIP kodekais: G.711, G.723 ir G.729 keičiant paketų dydį ir skambučių skaičių. Matavimai buvo atlikti naudojant optinę ir bevielę perdavimo sistemas. Pagrindinis tyrimo tikslas buvo parinkti tinkamą kodeksą, užtikrinantį gerą balso kokybę, ir rasti kompromisą tarp kainos ir kokybės santykio. Il. 11, bibl. 9, lent. 1 (anglų kalba; santraukos anglų ir lietuvių k.).