Decreasing Packet Loss of VoIP Calls by Optimising Transport Network

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Abstract

Availability of IP Phones in modern VoIP environment is lower than in traditional telephony and packet loss occurs more often. In this paper, we study packet loss and impact on quality of the call, when using different approaches in optimisation of transport network devices. Packet loss is studied during convergence; during failures and renewals of links in transport network. We compare results from unoptimised network with our two solution approaches. One approach provides ideal results for not only VoIP traffic. Second solution is applicable for most of the networks and devices and is presented as a cheap solution. Study is done for G.711 and G.729 codecs on a common network topology.

Categories and Subject Descriptors

K.3.1 [**Computer Uses in Education**]: Collaborative learning, Computer-assisted instruction (CAI); I.2.6 [**Learning**]: Knowledge acquisition

Keywords

Codec, Convergence, Packet loss

1. Introduction

IP phones and VoIP (Voice over Internet Protocol) are slowly replacing traditional telephony. Traditional telephony is characterised by its high availability and high quality. Convergence of traditional telephony and IP data networks raises new problems. In IP data networks are voice packets regular data packets as any others and QoS (Quality of Services) must be assured. However, packet loss is more common in these networks and assuring high availability should be as important goal as QoS.

In this article, we focus on different approaches of optimising IP transport network regarding to VoIP data traffic to ensure higher availability and satisfactory quality of VoIP calls. We introduce two different optimisation approaches and compare them with unoptimised data network. One approach uses the most modern techniques available in operating systems of network devices and is applicable for Ethernet networks. Another approach uses common techniques widely available also in low cost network devices. Goal of these two approaches is to lower packet loss or sustain satisfactory quality of the call during convergence in transport network. Convergence is a process and during this process, some of the packets may be lost or not routed optimally. Any failure or renewal of the link, node, or change in configuration files of network devices starts convergence. Convergence is never immediate and takes some time. During convergence, packet loss may occur. We study packet loss caused by convergence and not caused by congestion in networks. However, one of our solutions also solves this problem. Packet loss is at 50 % and 100 % level during our measurements.

The article is organised as follows: Section 2 describes the problem in details and provides existing solutions. Section 3 presents our test-bed network topology. Section 4 implements our proposed solutions, compare, and discuss results. Conclusion and ideas for future work are given in Section 5.

2. State of Art

Problem of packet loss in VoIP is well known and worldwide studied. This survey [1] describes VoIP briefly and describes problem of the packet loss besides others. Packet loss during a call should be under 1 - 3 %. Packet loss can be caused by many aspects, i.e. environment (Wireless networks), congestion in transport networks, capacity of the links, number of calls, jitter, etc. In addition, type of codec influences packet loss. One of the solutions can be an usage of FEC (Forward Error Correction) technique inside or within a VoIP codec. Delay and bandwidth demands are increased. FEC can solve problem up to 50 % packet loss, when every second voice packet is lost. When packet n is lost, packet n + 1 can recover previous one. Problem of packet loss can be inhibited by packet replica-

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tion or noise substitution, which can increase subjective quality of the call for the communicating people.

Article [2] studies impact of different factors affecting voice quality in packet networks. These factors include delay, jitter, packet loss, etc. We will focus on packet loss results, although other results are interesting too. They have studied, how delay and jitter impacts packet loss. Simulation was done on model topologies in Matlab. Codecs G.711A-law, iLBC and Speex were studied. MOS (Mean Opinion Score) was dropped by 0.2 - 0.9 from the 5 point scale, where higher value is better, depending on the codec. Packet loss was in range from 0 - 10 %. However, packet loss was due delay and jitter and not due convergence. We will compare our MOS values from convergence with these values.

Number of calls can influence packet loss. That is why it is important to have a good network design. Article [3] represents a case study with mathematical background. Authors have studied impact of several network parameters on a VoIP call. To decrease packet loss, low delay and jitter is recommended as specified in G.114 standard. In addition, packet loss concealment should be present.

Our solution is to optimise transport network. We will discuss our and previous mentioned solutions and results. One of our solutions uses IP/MPLS (Multiprotocol Label Switching) network, with MPLS VPN (Virtual Private Network), MPLS TE (Traffic Engineering) and routing protocol optimisation, which can relate to these studies [4, 5, 6, 7].

3. Test-bed Description

We test a common VoIP call using G.711μ-law codec and G.729 codec without VAD (Voice Activity Detection) on a network in Figure 1. Both codecs have sampling rate at 20 ms. Routers in oval represent service provider (SP) network. Routers that begin with letter ”C” are customer routers. Call flows from C2-1 site to C2-2 site. We use Cisco IP Phones 7940G series. As voice exchange, we use CUCME (Cisco Unified Communication Manager Express) solution, also known as Cisco CallManager Express. This voice exchange can register IP phones only one hop away. That means that IP phones must be on the same subnet as voice exchange. Voice exchange is implemented as part of IOS (Internetwork Operating System), which is operating system of Cisco devices. We can use any other voice exchange solution and it makes no difference. We use Cisco 2800 series routers and Cisco 3560 series switches capable of Power over Ethernet (PoE). PoE is used for powering phones.

SP network is configured as IP/MPLS with IS-IS (Intermediate System to Intermediate System) routing protocol and BGP (Border Gateway Protocol) protocol. Customer C2 has two sites - C2-1 and C2-2. They are interconnected via MPLS VPN technology, which is peer-to-peer type of VPN. Customer uses RIPv2 (Routing Information Protocol) as interior routing protocol. Both networks have IPv4 addresses. Customer configuration can be replaced by any other technologies and protocols and it makes no difference in measurements and results.

Customer sites are configured with basic configuration and SP network has basic configuration. We optimise SP network only. We make failures and renewals of links, causing packet loss and loss of availability, in SP network only. That is why we need to optimise SP network and do not need to optimise customer network.

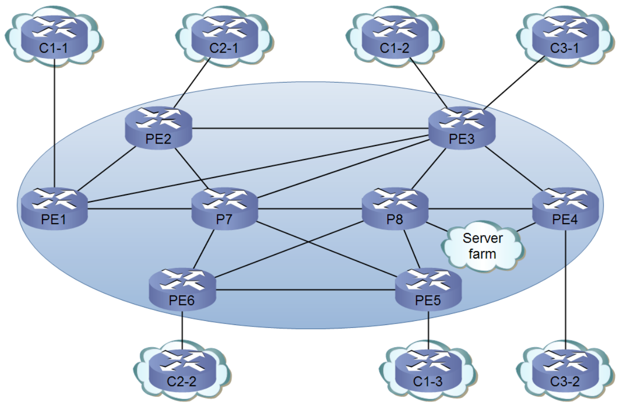

Figure 1: Test-bed topology.

Implementation is composed of three different approaches: unoptimised network, type A optimisation and type B optimisation. We measure network convergence and packet loss from our notebook placed in site C2-1 by using ping to the phone in C2-2 site. We also measure packet loss on IP phones, MOS, jitter and other network characteristics available by pressing twice question mark button on IP phones.

3.1 Unoptimised Network

In unoptimised network, we configure SP network with basic configuration steps. IS-IS is used as an interior routing protocol for IP protocol, LDP (Label Distribution Protocol) protocol is used for MPLS label exchange, BGP routing protocol is used for MPLS VPN information exchange, RSVP (ReSource reserVation Protocol) is used for resource reservation and also MPLS VPN. All of these are configured with least configuration steps possible.

3.2 Type A Optimisation

In type A optimisation, we optimise IS-IS and LDP protocols, configure MPLS TE tunnels for VoIP and other data traffic and we disable every unneeded negotiation protocol and enable interface dampening, which protects interface during interface flapping.

Protocol IS-IS has optimised hello mechanism, SPF, PRC and LSP generate timers. IS-IS has enabled ISPF, disabled hello padding, all interfaces are level 1 circuit type and metric type is set to wide. Interfaces are password protected. LDP has enabled loop-detection, graceful-restart and session protection. Targeted hellos are enabled and limited to three hops, timers are optimised and neighbours are password protected. IGP and LDP synchronisation are enabled in both directions and delayed. In addition, IS-IS hello mechanism can be replaced by BFD (Bidirectional Forwarding Detection). However, during our tests we shut down links administratively, so we skip detection phase - first phase of convergence. There is one TE tunnel for VoIP marked packets and one tunnel for other traffic. VoIP TE tunnel has higher priority and reserved bandwidth for VoIP calls. Bandwidth reservation is variable and automatically adjusted by auto bandwidth collection.

MPLS TE optimisation, i.e. SRLG (Shared Risk Link Group), Fast-reroute, etc. cannot be used, because we use Ethernet interfaces. These features should greatly speed up convergence.

3.3 Type B Optimisation

In type B, we configure two static routes with equal metrics to the destination for VoIP traffic. Traffic is load shared 1:1 over both static routes and we use per-packet sharing as load balancing algorithm. This ensures that one packet is sent via one path and second packet via second path. Static paths on router PE2 are PE2 → PE1 → P7 → PE6 and PE2 → PE3 → P8 → PE6 and vice versa on router PE6. This choice is made only because we want to have same network diameter and the same delay. Idea for this type of optimisation is based on [7].

4. Implementation

For all three configured networks, we use G.711μ-law and G.729 codec. Default MOS value for G.711 is 4.5 and for G.729 it is 3.9. IP phones are registered to their voice exchange and call can successfully flow from one customer site to another. Failures and renewals happen on link between PE2 and PE1 routers by logical shutdown, thus excluding detection phase of convergence. Best path from C2-1 to C2-2 is always via routers PE2 → PE1 → P7 → PE6.

When having unoptimised SP network and failure occurs on the PE2 - PE1 link, network convergence is in range from 5.6 seconds to 6.1 seconds. Lower values are for networks without congestion and higher values are for congested networks. Congestion is gradually up to 100 % saturation of links. Minimal network convergence must be 5.5 seconds, which is the default SPF timer in IS-IS protocol. SPF timer delays computation phase, which is third phase of convergence. Second phase - propagation is based on the network diameter and RTT between network devices. Last phase of convergence - updating routing tables is almost immediate, because we have very small routing tables. Renewal of the fallen link makes few packets lost, in range of 0 - 2 packets. There is 100 % packet loss during the whole convergence. Change in codec makes no difference.

Type A optimisation makes failures much shorter. Network convergence is less than 150 ms. However, there are 9 - 11 lost voice packets, which represent 180 - 230 ms convergence. Renewal of the fallen link makes no packet loss. There is 100 % packet loss during the whole convergence. Change in codec makes no difference.

Type B optimisation has failures in seconds. During the convergence, there is 50 % packet loss; every other packet is lost. However, when using G.711 codec, call is still understandable. MOS score lowers and personŠs voice sounds a little bit unnatural, more robotic or metal sound, but every word is clearly understandable. Speech rate or intelligibility does not make difference. When using G.729 codec, speech is not understandable at all and user experience is the same as in 100 % packet loss. Results apply for both failure and renewal of the link. Behaviour is based on the natural of codec. G.711 samples speech as is. If a voice chunk is lost and following is received, it does not influent resampling process on the end device. Quality is dropped, because every second packet is lost, but you lose only 20 ms of speech every second time, which is harder to notice for a user. Packet replication substitutes lost packet. G.729 codec uses compression and voice resample is dependent on the previous packet. That is why G.729 codec has by default wider dejitter buffers.

Problem with this optimisation type is that voice packets must be load balanced 1:1. If another traffic flows concurrently via the same static routes with these settings, it may happen that all voice packets flow via the fallen path. Second problem is, when there are two or more calls. It may happen that half of the calls go via the fallen path for the whole convergence. Enhancing type B by IP SLA and Object Tracking, as described in [7], can shorten convergence. Appropriate QoS mechanism can overcome problem, when having more than one call. Another solution is to create pair of two static routes to every destination IP phone, thus having 2 * n static routes, where n is number of phones in foreign site. This enhancement makes overall solution worse scalable, but we can easily overcome described problems.

Results are provided in next two tables. In Table 1, we present results from all three types of network implementation for failure of the link and Table 2 is dedicated to renewal of the fallen link. Results are an average from tens and hundreds of measurements. Delay between two packets is 20 ms. When comparing to other works described in Section 2, MOS was lowered more than in [2] for G.711, but we had 50 % packet loss instead of 10 % packet loss. Behaviour for codec G.711 was the same as described in [1, 2]. Packet replication or noise substitution took effect for G.711 and provided sufficient quality for end users. However, codec G.729 was not able to replicate previous lost packets and call quality was poor.

5. Conclusion and Future Work

In this paper, we have compared influence of convergence on VoIP traffic and proposed two approaches for increasing the availability. Type A has very short convergence times. There are 10 lost voice packets in average during link failures. If detection phase takes effect, there is no more than eight more lost packets (max. 160 ms). Renewal of the fallen link makes almost no packet loss for any data traffic. Type B approach has much more longer convergence, but only with 50 % packet loss. This packet loss impacts quality of voice in G.711 codec, but speech is still understandable. However, state is only temporarily until convergence ends. After convergence, MOS reaches previous higher values. G.729 codec and any other codec using compression or is dependent on previous voice packets are not satisfactory for this solution.

Type A solution is an ideal solution not only for voice traffic, but for any data traffic, i.e. IPTV, VoD, radio stream or FTP transfer. All of these applications and even more were tested with the similar results. Appropriate QoS configuration assures both availability and quality for voice calls. Disadvantage is that this solution is expensive. Expensive hardware and higher versions of IOS must be used. In addition, the newest IOS version is needed. Optimisation is very difficult from the administrator perspective. Not implementing the whole list can lead to much worse convergence values, i.e. not using IGP - LDP synchronisation makes renewal of the fallen link longer by 2 - 3 seconds.

Table 1: Comparison of evaluated network parameters during failure of the link regarding to VoIP.

Network type	Unoptimised		Type A		Type B	
Codec	G.711	G.729	G.711	G.729	G.711	G.729
Network convergence [s]	5.82		0.182		Seconds	
Packet loss	247.6		9.06		50 % packet loss during convergence	
MOS during convergence	1		1		3.5	1.2

Table 2: Comparison of evaluated network parameters during renewal of the fallen link regarding to VoIP.

Network type	Unoptimised		Type A		Type B	
Codec	G.711	G.729	G.711	G.729	G.711	G.729
Network convergence [s]	0.01		0		0.01	
Packet loss	0.09		0		0.09	
MOS during convergence	4.5	3.9	4.5	3.9	4.5	3.9

Type B solution uses only basic 1:1 load sharing, per-packet sharing algorithm for load balancing and static routing. Many of the network devices can satisfy these three demands. G.711 codec or similar must be used and there is 50 % packet loss for the whole convergence, which can take seconds or even more. When having an opportunity to speed up the process by using e.g. Object Tracking, or IP SLA tracking system, convergence can be shortened. Creation of pairs of static routes can overcome problem with inadequate packet sharing. At this point, we can propose a new algorithm, which will combine per-packet sharing algorithm with per-flow sharing, thus ensuring that one pair of static routes or MPLS TE tunnels will load balance packets evenly.

We personally recommend type A approach, but if network devices are not capable of every optimisation options and protocols, type B approach can be used. Type A cannot have load sharing as is presented in type B, because configuration of MPLS TE with MPLS QoS does not allow it. This article [8] tries to overcome this problem. We will further analyse other voice codecs in our optimised networks, i.e. iLBC and Speex. We want to work further on type A solution and find out anything else, which can even more speed up convergence and lead to less lost packets or to find out new approach, where low packet loss from type A and redundancy and load sharing from type B will be present.

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