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P. Fan
L. Li
China Mobile
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Requirements for IP/MPLS network transmission interruption duration
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Abstract

The transmission performance of IP/MPLS network affects upper layer services and networks, but there is no consensus in the industry on transmission interruption for IP/MPLS network up to now. This memo studies requirements for the interruption duration criteria in several service scenarios.

Status of This Memo

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1. Introduction

Today's IP/MPLS network is widely used as a bearer network to carry diversified packet switched services. The transmission qualities of these services are closely related to the performance of bearer layers, as network failure, delay, congestion and other abnormities will inevitably bring about service interruption and user perception degradation. However, there is no consensus in the industry on transmission interruption for IP/MPLS network up to now. This memo studies relationships between service performance and transmission interruption duration in several scenarios, and is intended to reach a list of requirements for these interruption duration criteria.

For a long time the industry has been aspiring for the so-called golden standard for network resilience, that is the 50-millisecond recovery threshold. [HeavyReading] gives us a basic introduction to the origin of this fast protection legacy which can date back to 1980s. The 50ms threshold was established informally in the early 1980s, and then formally through standardization of [G.841] recommendation on SDH network protection architects. The specific requirement shows a maximum threshold for detecting and restoring a fault of 60ms, which adds up fault detection duration of less than 10ms and protection switching time of less than 50ms. The report also mentions original concerns that the threshold results from. The voice channel banks deployed in early 1980s had limited fault

tolerance. Failures that lasted longer than 200ms would generate a Carrier Group Alarm (CGA) which caused the channel bank to terminate all connections over that given TDM line. So an outage budget was developed by carriers and the 50ms standard was employed to protect voice services. However newer channel banks at that time had started

to implement a CGA timer of 2s, so the 50ms protection was adopted to protect a small and diminishing fraction of digital network.

Historically this 50ms fast protection speed has been achieved by SDH network. Using various fast convergence technics, IP/MPLS is also able to react within 50ms. As for network applications that are carried by optical or packet core, changes have been made through the past decades, accompanied by the continuing questions about needs for 50ms protection. Here we list three basic considerations about services and their requirement for IP/MPLS: for services like TDM over IP/MPLS, the traditional 50ms guarantee should be kept and met; for current IP services (e.g. voice, internet), experiences or experiments are to be provided for guidance; for services in future, we are supposed to propose requirement early and give consideration to IP/MPLS.

2. Services and Performance Criteria

Services delivered by IP/MPLS network have different transmission quality requirements, thus introduce different performance criteria for the bearing IP/MPLS network. We believe there are two principles that need to be considered during network and service design, configuration and operation. The IP/MPLS bearer should satisfy quality requirements of upper level services and applications, while services and applications should also take into account the intrinsic IP capabilities. In this section we will describe concerns on IP/ MPLS and service mutual adaptation from aspects of several kinds of service scenarios.

2.1. Softswitch

From the softswitch point of view, the IP carrying nature imposes certain influence to the service quality. Especially when speech is delivered by IP, the communication quality of voice is impaired, and in turn makes higher requirements for the transmission performance of IP. The following table gives a list of criteria regarding

transmission quality of a typical GSM network as well as impacting factors brought by IP bearer.

Criteria of GSM Transmission Quality		Impacting Factors Brought by IP Bearer
Call Loss	Call loss of wireless channel	None
	Call loss between switches (typical value: <=1%)	Failure of Nc/Mc interface carried by IP
	Call loss between switch and BSC (typical value: <=0.5%)	None
Call Cut-off	Call cut-off rate (typical value: <1%)	Failure of Nc/Mc interface carried by IP
Connection Delay	Service providing delay	None
	Calling party connection delay (typical value: <=4s)	IP carried signaling delay
	Called party connection delay (typical value: <=4s)	None

If voice is carried by IP, communication quality criteria of call loss, call cut-off and connection delay are likely to be influenced. This subsection focuses on the three criteria and their impacting factors to give requirements for softswitch and IP bearer networks, with detailed analysis described in the appendix. Note that the current discussion on softswitch is focused on quality of transmission while not on quality of voice. In another word, the scope of discussion is limited to network related QoS aspect, while subjective QoE criteria such as PESQ (Perceptual Evaluation of Speech Quality) and MOS (Mean Opinion Score) are left to later revisions.

Call loss related requirement: The duration of SCTP interface association timer should be shorter than that of the state machine

message timer of upper layer protocols, and this duration is further recommended to be no longer than 6 seconds in order to maintain detection sensitivity; the interruption duration of IP bearer network should be as short as possible to avoid call loss, and this duration is further recommended to be no longer than 5 seconds.

Call cut-off related requirement: The SCTP association should be guaranteed during IP layer interruption to avoid interface breakoff alert. The requirements are the same as those related to call loss.

Connection delay related requirement: The IP convergence time should be no longer than 3 seconds to ensure that connection delay is shorter than 4 seconds.

The overall requirement for IP/MPLS interruption duration is no longer than 3 seconds.

2.2. SS7 transport

The Signaling System No. 7 (SS7/C7) network is one of the examples of the principle that services should take into account the ability of IP. The bearer of SS7 protocol stack has been experiencing evolution from TDM to IP. Traditionally the user parts of SS7 (including MAP, CAP, BSSAP+, ISUP, etc.) are carried by MTP layers, but the bearer has gradually been evolved into a packetized form with SIGTRAN (including M2PA, M2UA, M3UA, etc.) using SCTP associations over IP. The change requires transport layer to take mechanisms to meet demand of SCN signaling, and more importantly it requires protocols to make adaption to the "best effort" fact of IP.

The SIGTRAN uses an architecture that can be described as standard IP plus unified transport plus diversified adaption units. It introduces SCTP to realize reliable signaling transport over IP. The SCTP itself provides reliable transmission mechanisms, such as path selection and monitoring, validation and acknowledgment mechanisms, and retransmission timing management.

The unreliable nature of IP makes it necessary for the upper-level protocols to be more tolerable to the possible instability of bearer.

Once a service request from a UE is accepted, the system allocates resources and establishes paths for the user. A breakoff caused by IP will result in signaling disconnection or rerouting. Signaling transmission path may also be switched back after IP layer restores. Frequent switchovers and disconnections lead to unnecessary system cost and service interruption, so parameters should be configured a little bit "insensitive" to try to sustain connections on control plane.

One of the examples of parameter configuration is the timer value. The following gives two cases about SCTP on transport layer and M2PA on adaption layer. The values should not be set very small to prevent unnecessary disconnection caused by IP instability. However, because upper services of SS7 may also have timeout rules, values should not be set very large too to avoid violating the rules.

1) SCTP

SCTP uses RTO to manage timeout duration for retransmission in case of feedback missing. The RTO is given an initial, a max and a min value, and is calculated instantaneously with a set of management rules. Many other parameters are used for fault detection in SCTP. Association.Max.Retrans is used to indicate the upper limit of number of possible retransmission without considering endpoint down. Path.Max.Retrans is a similar value to detect path failure. The parameters together characterize the ability of SCTP to tolerate

bearer downwards and provide reliable SS7 transport upwards. The typical values of the parameters are RTO.Initial = 0.5 sec, RTO.MIN = 0.5 sec, RTO.MAX = 1.5 sec, Path.Max.Retrans = 5, Assoc.Max.Retrans = 10.

2) M2PA

Although protocols like H.248 and BICC can be carried directly upon SCTP, the user part protocols of SS7 usually have to be carried by SCTP/IP with the help of different adaption layers. In this case, the attributes of adaption layers, e.g. M2PA used between STPs, are more important to SS7. M2PA uses a T7 timer to indicate the maximum delay of acknowledgement and start T7 at the time of data transmission. If no message is acknowledged after the maximum waiting time, T7 expires and M2PA sends a message of out of service

to the peer end. Because propagation delays in IP networks are more variable than in traditional SS7 networks, the value of T7 should be set considering IP propagation delays, as well as acknowledgement time, SCTP slow-start algorithms, upper service timers and other factors. Typical value of T7 is 7~10 sec.

Parameter configuration induced tolerance to bearer may have some influence on service, but it avoids service cut-off or severe user perception degradation. For services like SMS or route lookup, possible latency may be introduced, but operations can still be completed after short delay. Because SMS has no strict requirement for instantaneity, impact on service is limited. If route lookup takes more time due to IP interruption and convergence, user may experience longer setup delay when dialing. For service of location update, even if operation fails because bearer is interrupted for too long, UE has the mechanism to initiate request again.

2.3. LTE Backhaul

To be further analyzed.

2.4. Ethernet VPN

Ethernet VPNs (e.g. VPLS) are used to provide transparent Ethernet type layer 2 connections for customers. Ethernet frames are treated as service payload and encapsulated and transported in providers MPLS network. The interruption criteria of IP/MPLS bearer should guarantee continuity of Ethernet service, and IP/MPLS failover is not supposed to generate outage of Ethernet service.

[Y.1731] and [IEEE802.1ag] describe in detail OAM functions and mechanisms for Ethernet, with specific recommendation on connectivity fault management. Ethernet uses continuity check function to detect

loss of continuity between any pair of MEPs in a MEG, and this function is realized by sending CCMs (connectivity check messages) between peer MEPs. When a MEP does not receive CCM from a peer MEP within a certain interval, it detects loss of continuity to that peer MEP. The threshold interval is specified as 3.5 times the CCM transmission period, which corresponds to a loss of three consecutive CCMs from the peer MEP, and the CCM transmission period is recommended to be the default value of 1 second. So the interruption

duration of IP/MPLS for Ethernet VPN services should be less than 3 seconds.

2.5. IPTV

To be further analyzed.

3. Other considerations

So far this document has focused on use cases and their requirement for IP/MPLS, and other practical issues are not included in this version. For example, an IP/MPLS packet core is expected to carry a variety of services, so the requirement for IP/MPLS may have to include additional concerns on this multi-service co-existence scenario. A simple and straight-forward way may be to satisfy the most critical need for protection time required by the services. Another issue is related to service awareness. Whether service type is or can be known by IP/MPLS would influence the ability of IP/MPLS to provide reliability guarantee accordingly. It seems to be easier to perform service identification on edge devices than network core. We believe these kinds of issues need to be taken into account, and currently we will just leave them to be updated in future revisions.

4. Security Considerations

TBD

5. IANA Considerations

This memo includes no request to IANA.

6. Appendix: Impact Analysis on Transmission Quality of IP Carried Softswitch Voice

This section describes impact on transmission quality of softswitch voice when carried by IP and requirements for IP bearer convergence time.

1) Call Loss

Call loss is used to describe the circumstance where a phone call

fails to establish after initiated by a subscriber due to network faults. In the practical network, the call loss rate is mainly associated by the factors as follows:

1. Interfaces, including Nc, Mc and interface between MSS and SG.

2. State machine message timer. If a timeout takes place, the state machine releases signaling messages, producing a call loss. Typical value of BICC timer is 10~15 seconds and value of DTAP timer about 15 seconds.

3. Interface association timer. Associations breaks off at the expiration of timer.

4. Bearer network convergence time.

If the configured timer duration of a state machine is shorter than the timer duration of interface association, then although interface association may not be broken off, call loss is still possible to occur due to message timer expiration. If the association timer duration is shorter than IP routing convergence time, the association is considered broken off by SCTP, hence message loss at interface between MSS and SG as well as interface Nc results in massive call loss, and new calling request cannot be satisfied because of interface Mc breakoff. In this case, the call loss rate can be calculated as

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Call Loss Rate = ( IP Convergence Time + Association Restoration
                 Time ) * CAPS / BHCA.
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However, if the association timer duration is longer than IP routing convergence time, then the association is considered normal by SCTP, and data will be retransmitted. Although this may cause buffer overflow leading to call loss, the call loss rate is possible to achieve approximately zero if buffer is big enough.

From the analysis above and practical operation experience, the requirements for softswitch and IP bearer are as follows: the duration of SCTP interface association timer should be shorter than that of the state machine message timer, and this duration is further recommended to be no longer than 6 seconds in order to maintain detection sensitivity; the interruption duration of IP bearer network should be as short as possible to avoid call loss during the IP layer interruption period, and this duration is further recommended to be no longer than 5 seconds.

2) Call Cut-off

Call cut-off is referred to the abnormal release during a phone call due to reasons other than intentional release by any of the parties involved in the call. The call cut-off rate is related with:

1. Interfaces, including Nc and interface between MSS and SG.

2. Interface association timer.

3. Bearer network convergence time.

If the association timer duration is shorter than IP routing convergence time, established phone calls will be released once interruption of interface Nc or interface connecting MSS and SG is detected. In the case of association breakoff, call cut-off rate can be calculated as

Call Cut-off Rate = (CAPS * Call Duration) * Busy Hour Association Breakoffs / BHCA.

While if the association is not interrupted, the call cut-off rate can be approximately zero.

In conclusion, the SCTP association should be guaranteed during IP layer interruption to avoid interface breakoff alert. The requirements for softswitch and IP bearer are the same as those related to call loss.

3) Connection Delay

The connection delay from a call initiation by a calling party to PLMN should be no longer than 4 seconds. This delay is affected by factors below:

1. RRC connection setup delay (irrelevant to whether service is carried by IP or not).

2. Core network signaling interaction delay. The message number at interface Nc/Nb is 6, and is 8 (calling side) or 16 (called side, in case of IP-IP) at interface Mc. Each message is with a delay of no longer than 50 milliseconds. Calling message delay at interface Nc is no longer than 300 milliseconds. If long distance call is made though CMN, the message delay is to be increased by transmission delay of 5 msec/km and CMN process

delay. So the message delay is likely to be 400 milliseconds.

3. IP bearer network QoS and load.

The connection delay is influenced by the delay criterion defined in the IP bearer network QoS, and is raised by delay, jitter, packet loss caused by network overload. In addition, if the configured timer duration of interface association is too long, the SCTP sensitivity to the retransmitted messages after packet loss will be decreased, which increases connection delay.

Connection delay is generally expressed as

Connection Delay = IP convergence time + RRC connection setup delay
+ Signaling Interaction Delay,

and is no longer than 4 seconds. So the IP network in normal working state should be constrained within a certain range of load to ensure that delay is shorter than 50 milliseconds, while in interruption state the IP convergence time should be no longer than 3 seconds to ensure that connection delay is shorter than 4 seconds.

From the analysis of IP/MPLS performance according to the three criteria above, we suggest the transmission interruption duration of IP/MPLS network for softswitch service should be no longer than 3 seconds.

7. Acknowledgements

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Authors' Addresses

Peng Fan
China Mobile
32 Xuanwumen West Street, Xicheng District
Beijing 100053
P.R. China

Email: fanpeng@chinamobile.com

Lianyuan Li
China Mobile
32 Xuanwumen West Street, Xicheng District
Beijing 100053
P.R. China

Email: lilianyuan@chinamobile.com