White Paper of Smart Speed Control and Super Error Correction Technology

Abstract: With the Network technology development, the existing circuit-switched network's safety, quality and stability are guaranteed, but there is high cost and inconvenience of network deployment; new IP packet switched network costs are lower and it's flexible to be deployed. It has become the development trend of network technology. It can ensure the safety, quality and stability conditions to improve the utilization of the network to achieve multi-purpose, but inevitably there are network jitter, packet loss and other problems. For video conferencing system demand on the network, if there is jitter, packet loss, it will result in the decline in the quality of audio and video. So there are higher requirements of network adaptability for real-time video conferencing systems. This article focuses on the network adaptive technologies for IP networking, video conferencing systems, and Huawei's audio and video QOS solutions is given in the last. Practice has proved that Huawei's video conferencing systems, audio and video QoS solution can automatically adapt to problems in IP networks, providing high-quality audio and video effects experience.

Key words: Network quality, QOS, video conference, smart speed control, super error correction, anti-package loss

1. Introduction

Audio and video conferencing is a set of video communication, audio communication and data communication in the new generation of interactive multimedia communications system. It is based on a value-added service in the communications network to meet the multimedia communication demand between two or more users at the same time.

Traditional audio and video conferencing is using the circuit-switched networks. ISDN, E1, V35, the DDN and other networking technologies are mature and dedicated private network. Safety, quality and stability are guaranteed. It is the first choice for security-sensitive industries. But they are difficult to make full use of network resources, and the construction and use are fairly expensive.

In recent years, with the rapid development of IP technology as the core of the Internet, the IP packet switched network can achieve multi-purpose network. It has a greater advantage in terms of operation, interaction, and price. So a variety of IP networks new applications are rapidly becoming popular. In this context, the audio and video conferencing is developing from the circuit switched networks to packet-switched network.

However, because of IP network's "do my best" (Best Effort) strategy, it does not pay attention to the quality of service. It's very suitable for data services, but for high real-time requirements of audio and video conferencing, you must solve delay, jitter and packet loss during transmission of video and audio data in real time. The guarantee of audio and video QoS is the focus and difficult technology to be solved for audio and video conferencing based on the IP network.

2. Main problems of IP network

In IP networks, the main factors affecting the quality of audio and video conferencing include:

1. Bandwidth: It refers to the average rate of special application flow between two network nodes. In general, the higher the bandwidth, the more data transmission, the better audio and video QoS. IP networks carry a variety of video, voice and data services, including VOIP, IPTV, instant messaging, file transfer, online games, BT, WEB, E-Mail, a variety of application mode. Different modes of application have diverse data traffic and burst, which results in unstable bandwidth of the audio and video conferencing services.
2. Delay: It refers to the average round trip time of packet transmission between two nodes of the network. Although audio and video conferencing equipment codec and synchronization also have a delay, this delay is relatively fixed; network delay depends on the network topology complexity and processing delay of the network equipment and other factors, affecting the overall end-to-end delay. Audio and video conferencing services that require high real-time standard are very sensitive to the delay. It is generally believed that if the delay exceeds 300ms, the interaction of both sides can clearly feel the pause, which impact of subjective experience.
3. Jitter: it refers to the change of the delay. The IP network jitter depends on the dynamic routing, and factors of network equipment due to congestion. Audio and video decoder suits for a stable stream. If the stream jitter after line transmission is beyond the tolerance range of the decoder, the decoder will discard (or cache a lot), the final expression is package loss (or delay), which ultimately affect the quality of the audio and video.
4. Packet loss: It refers to the percentage of lost packets in the network transmission process. The actual IP network environment leads to loss of packets due to congestion of network equipment. It will affect the quality of audio and video, such as: mess images or mosaic, the sound intermittent, etc. It can even interrupt the meeting seriously.

3. Common audio and video QoS guarantee

QoS stands for Quality of Service. It shows the guaranteed performance to pass information in the data communication system. Audio and video QoS guarantee refers to the ability of the audio and video conferencing systems that considering audio and video quality factors on the IP network provide the required services for the protection of audio and video services. In the case of network conditions remain unchanged, the rate and error control processing in the audio and video conferencing endpoints and MCU improve the audio and video effect to a certain extent. QoS security technologies include IP priority, rate adjusting, lost packet retransmission (ARQ), forward error correction (FEC), backward error correction (PLC) and other QoS policies. It can control the packet congestion, eliminate errors in the transmission and improve the quality of audio and video.

1. IP priority

IP priority uses related part of the IP packet to prioritize audio, video and RTCP data streams. There is a special byte in the IP packet header, called the domain of service type. The first three bits of this byte is used to define the datagram priority that described with 0-7, eight different priority levels. In audio and video conferencing systems, when network bandwidth is below a certain level, it can adjust the packet priority level to help routers select the priority for IP packets sending and receiving. Generally speaking, the audio package is the most sensitive to the time delay. When network uses IP priority for traffic matching, equipments can modify the IP priority field in packets to ensure priority of audio and video conferencing delivery.

2. Rate adjustment

In harsh network environment, audio and video conferencing data transfer rate causes saturation of the network, resulting in packet loss and severe network jitter. It’s better to reduce the data transfer rate to eliminate packet loss and network jitter. In this case, low down the rate of meeting will help to improve the coherence of the video and audio effects. If the device supports dynamic rate adjustment technology, it can make the terminal and the MCU automatically adapt to network capacity and performance through detecting favorable and unfavorable factors in the network to provide end users with the best video quality.

Adaptive bandwidth adjustment of audio and video conferencing equipment is realized by detecting packet loss rate. If the device detects packet loss rate exceeds a specified threshold, it will automatically reduce the bit rate of the conferencing and provide one rate which has the best effects.

3. Packet loss retransmission（ARQ）

When there is severe network congestion, network equipment (such as a router) discards a number of video packets with the relevant mechanism based on the size of cache. In audio and video conferencing systems, packets are using the UDP protocol for transmission, while UDP does not have retransmission mechanisms. So the receiver finds the image frames lost or mosaic phenomenon. Video equipments that support packet loss retransmission ensure the continuity of the conference image by adding the packet loss detection and retransmission mechanism.

ARQ is the lost packet retransmitted. The receiver requires buffering and sorting received packets. IP network delay, jitter and other factors have a significant impact on the retransmission performance. ARQ is difficult to meet real-time requirements of audio and video conferencing services when there is too much delay.

4. FEC（Forward Error Correction）

FEC algorithm is to add redundant information in the audio data issued by the sender. The receiver can do error detection and correction with the redundant information. It does not need to wait for the retransmission of lost information. So it is suitable to solve the audio and video conferencing adaptability of the network in such real-time business. FEC algorithm generates inspection packet by the data XOR method, and designs buffer strategy to reduce delay and jitter on the network to reduce the emergence of packet loss.

When use FEC coding to ensure reliability of data transmission, it should be noted that the strategy of selecting the encoding scheme. In general, the greater redundancy of the FEC coding, the stronger capability of error correction. But greater redundancy means more bandwidth and less utilization.

5. PLC（Packet Loss Concealment）

Audio packet loss in IP network causes voice distortion. In order to reduce packet loss affect on the quality of speech perception, the PLC algorithm uses the previous packet or adjacent packet (in case a later packet is available) to forecast loss of packets, and restore the original voice message as much as possible. Handle Most of the PLC algorithm is based on the receiving end, which does not require the sender to participate in.

The above technique is commonly used audio and video QOS support technology in video conferencing systems. There are different effects for different network conditions, such as ARQ retransmission can achieve the desired results in the case of small packet loss and lower network latency network. So the key issues is how we can integrate these technologies to improve the experience of audio and video, while each technology can play to the best effect.

4. Key technologies of Huawei audio and video conferencing systems

1、Super error correction（SEC）

Huawei super error correction (SEC) adopts association calibration and parity packet alternative to further ensure packet loss recovery, while maintaining complete compatibility with existing network equipments, point to point or multipoint conference of anti-packet loss that does not affect any business.

（1）Association calibration：

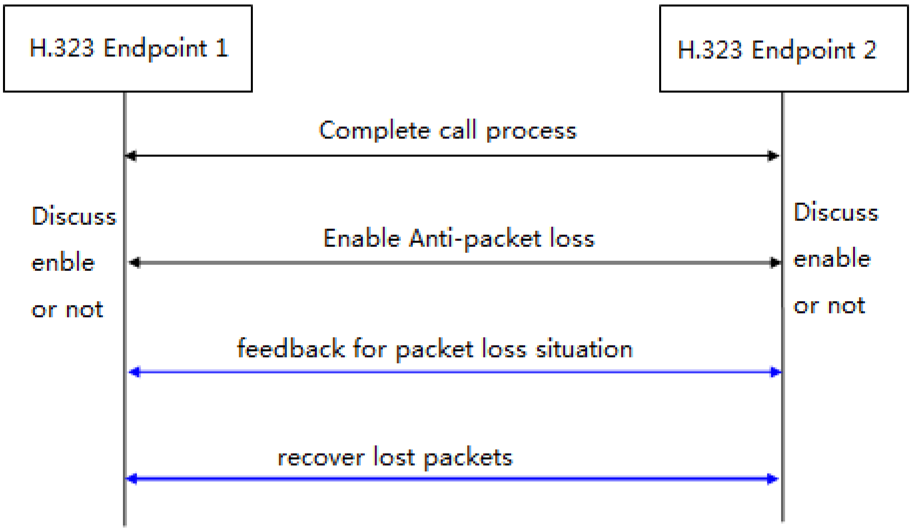
Association calibration generates parity packets by a special algorithm with original transmission data packets. When transmitted to the other side, 1 or 2 packet loss in a certain percentage can be 100% recovered. More of packet loss can be restored to 90% according to the situation; it can take a different algorithm to fully guarantee the consistent effect depending on the packet loss rate.

（2）Packet alternative transmission

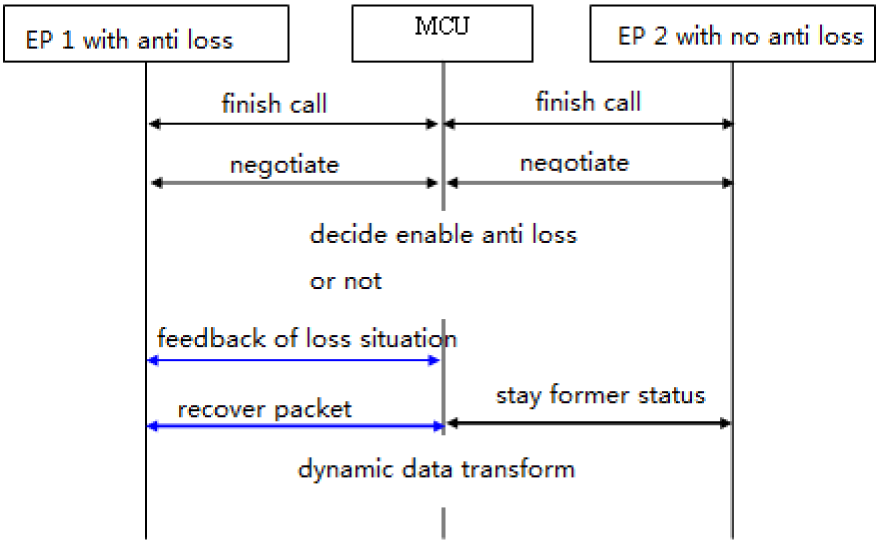
Packet alternative transmission is mainly for the recovery of consecutive lost packets based on the association calibration to provide higher QoS guarantee and completely solve the problem of IP QoS; Endpoint itself handles packet loss through codec algorithm, and can meet the environmental requirements of the IP network, allowing users to enjoy the experience of high-quality audio and video conferencing.

（3）Complete compatibility with service

Huawei equipments have the ability to anti-packet loss. Intelligent handling of packet loss is enabled to ensure video based on the current network status, audio effect. It has good compatibility with other manufacturer's equipments. As shown below:



In multi-site conference MCU smartly takes a different strategy in accordance with the terminal's ability to support anti-loss and meeting compatible. As shown below:



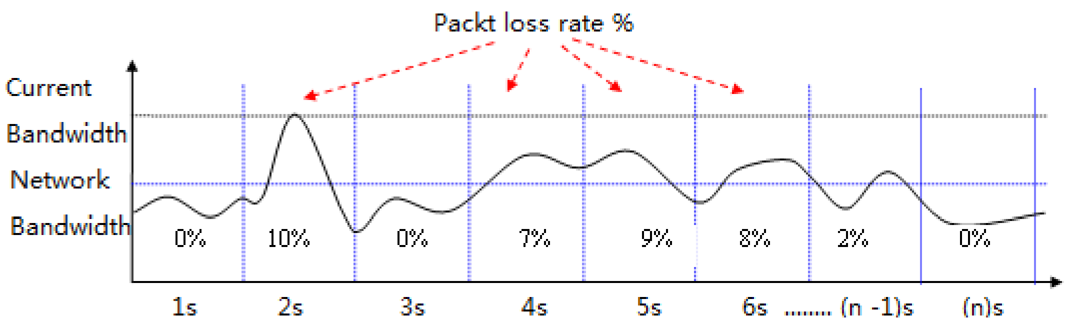
2、Intelligent rate control（IRC）

(IRC) bases on real-time statistics in the process of audio and video conferencing to determine the choice of the bandwidth when the packet loss rate is greater than a set of conditions. When the bandwidth is recovered, it starts the smart speed up processing strategy to achieve the best usage of network and video call performance.

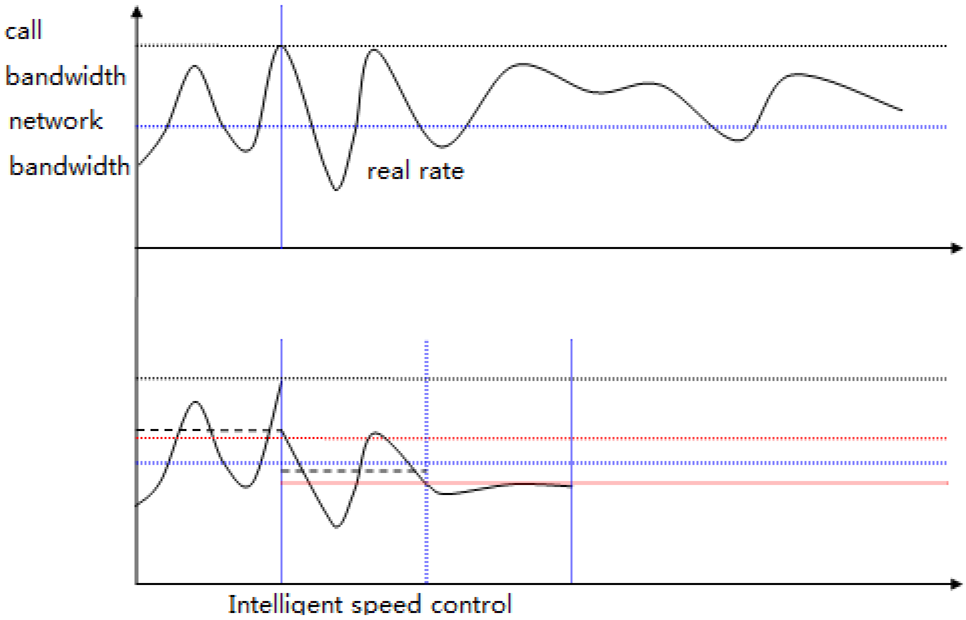
（1）Technical principle

Intelligent Rate Control includes speed control of point-to-point, point-to-point with dual-stream, multipoint conferencing, dual-stream meetings. Its main design of algorithms is the need to consider how to determine the base recovery of the speed up or slow down, and the key principle is the algorithm design.

Packet loss schematic：



Intelligent speed control schematic：



The intelligent speed control strategy adequately addresses the deceleration with the unknown network bandwidth, high coding rate volatility and network fluctuations that affect the final results. It guarantees the desired results by lifting or reducing rates.

（2）Single stream speed up and down

Single-stream call includes two modes of point-to-point call and multipoint call. The strategy can be adjusted automatically according to the actual network bandwidth. It needs to complete the actual network bandwidth statistics, statistics of the packet loss rate and start lifting speed strategy automatically. The design of the accelerating and decelerating depends on circumstances to achieve the best results.

Intelligent Speed control in a multipoint conference also based on that of a point-to-point meeting. According to the test results of single-stream point-to-point call, multipoint conferencing intelligent speed control strategies consider the balance between each endpoint, so as to achieve better results. The difference is to define the intelligent speed control conditions to determine when to speed up or down. The main needs to design treatment strategies are in several packet loss situations: losses of packets received by the MCU, loss of packets send by MCU, packet loss during cascading MCU and loss of packets may be watched by more than one party.

（3）Dual stream speed up and down

Dual steam call adds another channel in the basis of the original single-stream call so the intelligent speed control strategy is more complicated than the single-stream strategy. The principle of Huawei dual-stream intelligent speed control includes:

A）The audio is highest priority to ensure that audio rate does not slow down;

B）Huawei auxiliary stream support in two ways, Live and Presentation. If it is live mode with moving images, they require priority to ensuring the mainstream and decelerate the auxiliary stream in live mode;

C）When the auxiliary stream is in Presentation mode, it does the most reasonable rate adjustment based on the actual bandwidth;

D）Speeding up is in accordance with the principle of deceleration with roughly the opposite strategy processing.

5. Huawei audio and video conferencing QoS solution

IP network services carrying audio and video conferencing provide quite different qualities in practical applications. IP network access method (Ethernet, ADSL, cable, fiber optics and wireless communications, etc.), deployment (LAN, MAN and WAN) and operator factors will affect the quality of service.

A variety of QoS guarantee is designed for the different factors affect the audio and video quality in IP networks. It cannot be expected to adopt a single strategy to adapt to different network environments. For example, the rate adju0073tment dynamically adjusts the coding rate of audio and video conferencing network to adapt to the change of network bandwidth factor.

In order to improve the audio and video effects in conference and adapt to a different IP network environment with its changes, Huawei integrated and optimized use of various audio and video QoS technology, and proposed conferencing QoS solution. By real-time detection on the network, depending on the delay, packet loss, jitter and other factors, it uses the different QoS program, to take advantage of each one, and ultimately improve the communication quality of the audio and video conferencing.

Huawei audio and video conferencing systems QoS solution deploys innovative super error correction (SEC) and Intelligent Rate Control (IRC) technology. Super Error Correction (SEC) combines the video and audio codec level with the RTP level, keeps the video synchronization compensation and screen smoothing associated with relevance calibration, package alternative transmission, achieves the maximum guarantee of video and audio communication from the IP network environment impact. Intelligent Rate Control (IRC) generates real-time statistics of the current video and audio packet loss. When the packet loss rate is greater than a set of conditions, it will start the Intelligent Speed Control strategy processing, to achieve the best video call effect in the current instable network.

Huawei audio and video conferencing systems QoS solution has the following advantages:

1. Standards compliant

Huawei audio and video conferencing systems QoS solution has been applied in the full range of products. The technical methods used in the program complies with the ITU-T and IETF standards, and compatible with other manufacturers' audio and video conferencing products (including terminals, MCUs and gateways, etc.).

2. Dynamically adapts to the network environment

In different IP network environment, there are different factors that affect the quality of the audio and video conferencing. In the process of audio and video conferencing, due to the burst of data traffic and other reasons, the IP network environment will change from time to time. Huawei QoS solution can dynamically analyze the factors and trends that impact the quality of IP networks, select the optimal combination of technologies according to the current network environment, dynamically adapt to changes to provide the best possible audio and video effects and subjective experience at any moment.

3. Technology leadership

Huawei audio and video conferencing system QoS solutions use their own patented Super Error Correction (SEC) and the Intelligent Rate Control (IRC) technology.

Super Error Correction (SEC) technology allows audio and video QoS to be fully protected. According to the experimental data and the operating status of the existing network, in 3% packet loss environment, the video effects do not have any change and the packet loss is basically insensitive; from 3% to 10% packet loss, it can also be quickly convergent image and the image is continuous for general watching.

Intelligent Rate Control (IRC) solves the situation that call bandwidth is greater than the actual network bandwidth, or network unstable. When consecutive packet loss exceeds a certain percentage and lasts for a few seconds, the rate can be reduce to the range with no packet loss to ensure high quality image. When network bandwidth is restored, the speed returns to normal in a short time.

Overall solution (see figure below): the device nodes of video conferencing system monitor real-time operating status and statistics of network latency, packet loss, network jitter and other parameters to control the enable strategy of QoS in different situations. It maximizes the effect of the algorithm with a variety of techniques. The control strategy is as below:

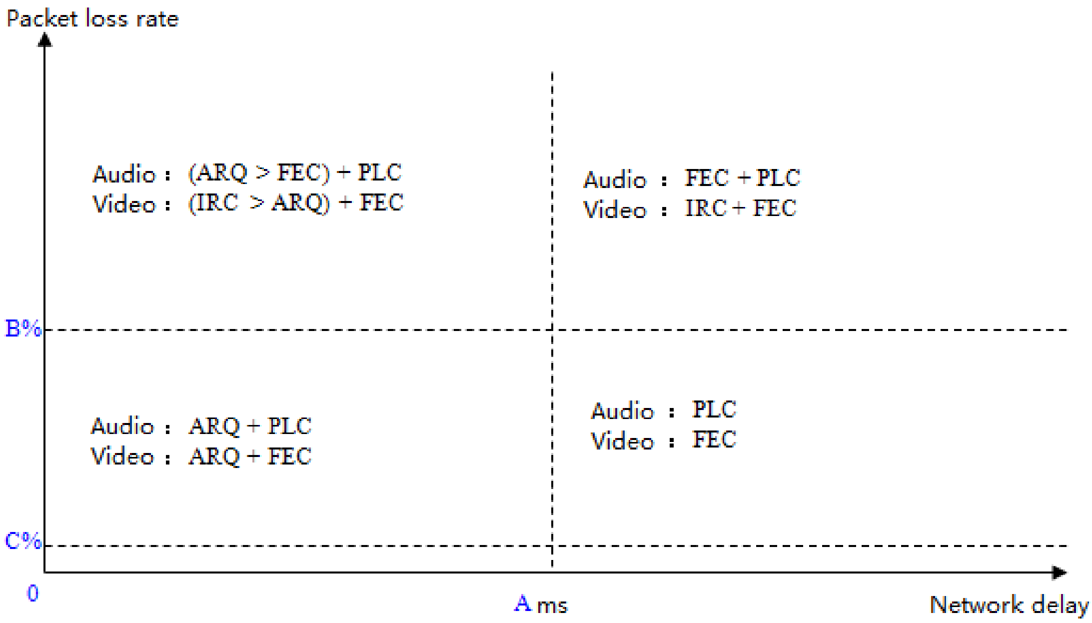
（1）Packet loss in range [0,C%]，enable ARQ function

（2）Packet loss in range (C%, B%]，network delay in range [0,A]，enable ARQ+PLC for audio，enable ARQ+FEC for video

（3）Packet loss exceeds B%，network delay in range [0,A]，enable ARQ and PLC for audio，enable FEC secondly；enable IRC and FEC for video,　enable ARQ secondly。

（4）Packet loss in range (C%, B%]，network delay exceed A，enable PLC for audio，enable FEC for video.

（5）Packet loss exceeds B%，network delay exceeds A，enable IRC and FEC for audio, enable FEC+PLC for video.



The experimental results are as follows：

Packet loss rate	Huawei audio and video QOS algorithm	Other algorithms
<0.5%	Undetectable	Detectable but not obvious
0.5%～1%	Undetectable	Significant impact
1%～5%	Undetectable	Serious deterioration
5%～9%	Detectable but not obvious	Bad image, unable to communicate

6. Summary

Huawei audio and video conferencing systems QoS solution integrated and optimized various QoS techniques. Innovative Super Error Correction (SEC) and the Intelligent Rate Control (IRC) are able to adapt to a different IP network environment and changes, to selecting the optimal combination of technologies appropriated to the current network environment, to deliver the best possible audio and video effects at any moment. So you can always enjoy the best audio and video experience of conferencing systems.