**How to build video conferencing system technology based on IP network**

Date：2018-09-26 14:51:27

1 Introduction

With the development of multimedia computer technology and communication technology, a new technology, multimedia communication technology, is a product of mutual penetration and development of multimedia, communication, computer and network. It also combines computer interactivity and multimedia. The combination of characteristics, communication distribution and the authenticity of the TV have obvious advantages. At present, how to realize video and audio transmission better and faster in IP networks has become one of the research hotspots today.

2. Technical requirements for building a video conferencing system based on IP networks

As the speed of IP networks is getting higher and higher, from narrowband to broadband, and bearer services move from non-real-time to real-time, IP technology has become the best choice for implementing integrated services such as video, audio and data. Establishing a video conferencing system on an IP network requires multiple technical support and is a relatively complex and complete multimedia application system.

2.1 To have a high enough bandwidth to transmit video, you must have enough network bandwidth, just like a big car must have a wide enough road to pass, otherwise, video data cannot pass through the network. Taking a frame of 1024×768 pixels as an example, if each pixel is represented by 12 bits, a total of 9.4 Mb is required. If the transmission rate is 25 frames/second, the amount of data to be transmitted in 1 second is 235 Mb. Transmission of such large data is unacceptable under existing network conditions.

2.2 It is necessary to have a good compression technology. Only a compression algorithm with a high compression ratio can effectively reduce the amount of data, so that video and audio data can be transmitted on the IP network. For example, in the H.323 conference system, the image coding mainly adopts the H.261 and H.263 standards, and supports the resolution of CIF and QCIF, while H.264 is being improved than H.263 and MPEG-IV compression. Compared with higher standards, it saves 50% coding rate, and has better support for network transmission, and can obtain image quality of HDTV and DVD.

2.3 Multicast Technology Multicast Based on IP Network Multicast is a multi-address broadcast. The transmission and reception are one-to-many. During the transmission process, the sender only needs to send the data packet once, and each user located in the multicast group can share the data packet. In the video conferencing system application, when a node signal is transmitted to each node, whether it is repeated point-to-point communication or broadcast, the network bandwidth is seriously wasted, and the multicast technology distributes the data transmission to the network node, reducing The total amount of data in the network.

2.4 To have a suitable transmission protocol TCP and UDP protocols can not support the video conferencing system well, which requires a suitable protocol, such as RTP, RTCP, RSVP. RTP runs on top of UDP. Audio, video and other data are encapsulated in RTP packets. Each RTP packet is encapsulated in a UDP packet and then encapsulated into an IP packet for transmission. In the case where the underlying network supports multicast, RTP can also use multicast to send data to multiple destination endpoints. RTCP is the control protocol of RTP. It is responsible for feedback control, detecting QoS and transmitting related information, and adjusting the data transmission and reception of RTP to make maximum use of network resources.

2.5 To provide quality of service guarantees Network service quality is a quality agreement between the network and users and between users communicating with each other on information transmission and sharing. First, in any network, latency always exists. The video conferencing system has high real-time and reliability requirements. In order to obtain the real sense of presence of each venue, the delay of audio and video should be less than 0.25s, and the maximum delay jitter should be less than 10ms. Secondly, in the video conferencing system, lip-synchronization is also required. Only when the time synchronization is achieved can the natural information about the venue be naturally and effectively expressed. Third, promise a certain rate of packet loss. Because people's perceptual ability is limited, in a video conferencing system, individual packets are lost, and the human eye is not aware of it. Therefore, a certain transmission error can be promised, and the packet loss rate should be controlled within a range acceptable to humans.

3. Protocol for building a video conference system based on IP network

The standards for building video conferencing systems based on IP networks are: H.323 and SIP.

H.323 follows the traditional telephone signaling mode, which is relatively mature. Many products have emerged, forming a relatively mature application system and market system. The SIP protocol uses audio and video transmission as an application on the Internet, increases signaling and QoS requirements, draws on the design ideas of other Internet standards and protocols, and follows the principles of simplicity, openness, compatibility, and scalability, but it is relatively simple, but its The launch time is not long, the agreement is not very mature, and the application is not a lot.