Design and Implementation of Distributed Call-Center Based on Soft-Switch

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Abstract. Traditional PSTN-based Call-Center fail to meet agent distribution deployments and lower operating costs. With the development of NGN technology, Soft-Switch-based Call-Center has solved the pain points of enterprises. This paper analyzes the special needs of distributed Call-Center system, and on the basis of analyzing and studying Soft-Switch technology, the platform architecture of distributed Call-Center system based on VoIP is given. This paper focuses on the support process of SIP protocol for call control function of Call-Center system, as well as the key equipment in Soft-Switch system. Finally, a practical Soft-Switch system implementation scheme is introduced in conjunction with the project.

1. Introduction

Call-Center have evolved with the development of PBX and customer service models. Originating in the 1930s and initially transferring users’ calls to customer service or experts. With the increasing number of calls and answers to be transferred, an interactive voice response system has been established. This system can realize the response of some common problems of customers to be answered and processed by the machine "automatic operator". In the traditional sense, Call-Center is a call response center based on telephone access, which provides customers with a variety of telephone response services [1].

1.1. The main disadvantages of traditional Call-Center

The traditional Call-Center is based on PSTN (public switch telephone network), which is established on the basis of PBX (private branch exchange). The traditional PBX can access to the analog or digital signal of PSTN, and the agent-end generally adopts the analog telephone. With the expansion of capacity, the price of PBX equipment becomes expensive, it is not convenient to expand capacity, the deployment of agents requires relatively concentrated sites, and it can not adapt to the application of decentralized organizations, etc. The system topology is shown in Figure 1.
Each dotted box in the figure 1 represents the basic configuration of an agent, which is composed of a PC and an analog telephone. The Call-Center’s agent-client-end runs on the PC, which includes the telephone call service processing and expert knowledge base, and is generally implemented in B/S mode. The agent telephone is connected with PBX through the internal analog telephone network, and PBX provides ring current feed for the agent telephone. Due to the limitation of electrical parameters, the distance between telephone and PBX should not be too far, and they must be centrally deployed in an independent fixed office area. System expansion includes trunk-line expansion and agent expansion, which is also limited by PBX capacity. We need to buy a new set of large capacity PBX, which is expensive. This is not the main problem. For those with branches, agents are distributed in different regions of the city, and even some of them are all over the world. The traditional Call-Center can’t meet the enterprise’s expectation of reducing operation cost. With the development of Internet and multimedia technology, Soft-Switch-based distributed Call-Center emerges as the times require, which can perfectly solve the above problems.

1.2. Soft-switching technology and SIP protocol

VoIP (Voice over Internet Protocol) is based on IP Packet-Switching Network for real-time transmission of voice, data, fax, video and other multimedia services [2].

The basic principle is: analog sound signal is sampled, quantized and coded to form digital audio. Then it is transmitted to the voice destination through the IP network; after receiving the IP data packet, the destination decompresses and reorganizes the data to analog voice signal, so as to realize the purpose of real-time voice transmission over IP.

As the core technology of next generation network (NGN), Soft-Switch can integrate the three networks of telephone network, Internet and mobile network. As a communication protocol acceptable to all three networks, IP protocol enables all kinds of IP based services to realize interworking in different networks.

The main design idea of Soft-Switch is the separation of service/control and transmission/access, and the entities connect and communicate with each other through standard protocols. Soft-Switch mainly uses H.323, MGCP, H.248 and sip protocols to communicate with various terminal devices. It uses signaling gateway and trunk-line gateway to connect with PSTN, and supports SS7, SS1 and ISDN signaling.

SIP protocol refers to the design idea of Internet standards and protocols, which is simple, open, compatible and easy to expand. In IP network environment, SIP is a signaling protocol to realize real-
time communication. Each SIP session can be a variety of different types of content (such as ordinary text data, audio and video data, game data, etc.), SIP protocol adopts the distributed control mode, and there is no central control entity. Therefore, SIP has great flexibility. SIP protocol is used in Soft-Switch of distributed Call-Center to realize digital transmission and control of voice and video [3].

2. System design of distributed Call-Center

Different from the traditional centralized Call-Center, the distributed Call-Center manages and controls the distributed agents in any geographical location through IP network, such as queuing, ACD, intelligent routing and so on.

2.1. Basic structure and composition

The system topology of Call-Center based on SIP Soft-Switch is shown in Figure 2.

![Figure 2. Soft-Switch-based distributed Call-Center topology.](image)

In Figure 2, the rounded dotted line box represents the configuration of an agent. The agent in the rectangular dotted line box can use either an analog phone or a SIP Phone to answer a call. The capacity of non SIP Phone is limited by capacity. In addition to agents, devices in the dotted rectangle can be deployed in the cloud. All remote agents are deployed in WAN by SIP Phone. In this way, it is convenient to realize the dynamic expansion of the branch, which is not limited by geographical area.

2.2. Main equipment

2.2.1. VoIP Voice-Gateway. The Voice-Gateway provides the function of converting TDM (time division multiplexing) voice stream to IP voice stream, and can convert PSTN signaling to SIP message, so as to realize the intercommunication between Call-Center system and PSTN.

Voice-Gateway is divided into 1-96 ports, and some of them have escape ports. Voice-Gateway are classified as follows:

- Analog Gateway: Terminal-Gateway FXS (1-96 FXS), Trunking-Gateway FXO (1-96 FXO);
- Digital Gateway; E1- Gateway (1-16 E1);
2.2.2. **SIP Server.** SIP Server realizes the functions of SIP proxy server and register server.

2.2.3 **CTI Server.** CTI server is the core of the whole Call-Center. It organically combines the telephone switching system with the computer system, and makes full use of the telephone switching function of the switch and the data processing function of the computer system. It can not only receive the call information (such as telephone number) from the switch, but also effectively control the call processing of the switch through the computer, including call transfer, call termination, intelligent call out and other services. CTI server should also realize IVR and ACD functions.

2.2.4 **SIP Phone.** The terminal equipment in Soft-Switch adopts SIP telephone, which has two forms: SIP telephone, SIP softphone + microphone. The appearance of the former is the same as that of the ordinary analog telephone, but the interface is RJ45 connected with the network switch to realize the VoIP function. The latter uses PC sound card, microphone and SIP telephone software to realize VoIP function.

2.3. **SIP Phone Registration process**
The prerequisite for SIP Phone to make or receive calls is that SIP Phone is authorized by SIP server. It requires the SIP Phone to register with the SIP server. The registration process is shown in Figure 3.

![Figure 3. SIP Phone registration flow.](image)

- Step 1: SIP Phone sends a register without authentication information;
- Step 2: SIP server returns 401 (unauthorized);
- Step 3: SIP Phone client sends register with authentication information;
- Step 4: SIP server verifies and returns 200 Ok if it is qualified.

2.4. **SIP Phone Invite process**
The Invite process is shown in Figure 4.
Figure 4. SIP Phone Invite flow.

- Step 1. The terminal agent SIP Phone A initiates one-way call off the machine and initiates Invite request to SIP server;
- Step 2. After the SIP server confirms that SIP Phone A has passed the authentication, it inserts its own address in the VIA header of the request message, and sends the Invite request to the called terminal agent SIP Phone B indicated by the TO-field of the Invite message.
- Step 3. The SIP server sends the response information in call processing to SIP Phone A: 100Trying.
- Step 4. SIP Phone B sends the response information in call processing to SIP server: 100Trying.
- Step 5. SIP Phone B instructs the called user to ring. After ringing, the user sends 180Ringing information to SIP server.
- Step 6. The SIP server forwards the ringing information of the called user to SIP Phone A.
- Step 7. When the called user picks up, SIP Phone B returns a successful response to the SIP server (200 Ok)
- Step 8. The SIP server forwards the success indication to SIP Phone A (200 Ok)
- Step 9. After receiving the message, SIP Phone A sends ACK message to SIP server for confirmation
- Step 10. SIP server forwards ack confirmation message to SIP Phone B.
- Step 11. Establish a communication connection between SIP Phone A and SIP Phone B, and start the call[5].

3. System implementation scheme
There are two main implementation schemes of distributed Call-Center system based on Soft-Switch.

3.1. Building Call-Center based on open source Soft-Switch platform and VoIP gateway.
The main open source Soft-Switch are Freeswitch, Asterisk and so on. The main characteristics are cross platform, stable and meet the requirements of enterprise level concurrency. Combined with VoIP Voice-Gateway equipment.

3.2. Building Call-Center based on VoIP voice card.
The VoIP voice card and the E1 voice card are inserted into the PCI slot on the server motherboard, and they are connected through the CT-Bus. The E1 card is connected with the PSTN digital trunk-line. Board manufacturers provide the driver and SDK.
The driver of VoIP board has SIP registration and proxy function, which does not need to be developed separately. The advantages of this scheme are: moderate price, rich development documents, SIP call control compatible with PBX mode, and easy to upgrade to distributed Call-Center.

4. Conclusion

The Soft-Switch scheme based on VoIP card has been successfully applied to the construction and expansion of a private hospital Call-Center with multiple branches. The hospital is headquartered in Wuhan and its branches are located in Nanchang, Changsha and other cities. There are 16 local IP agents in the headquarters. Nanchang, Changsha and other places each have 8 remote IP consulting expert agents. The system adopts the mode of trunk-line centralized access, data centralized management and agent distribution setting. The CTI server in the Wuhan is equipped with one SS7 voice card of 1E1 and one 32-channels VoIP voice card. The trunk-line resource supports 30 concurrent calls and 32 online agents. The headquarters and branches are connected through VPN. The IP phone configured for each agent is registered to the VoIP voice card, so the agents can talk to each other for free, and the call effect is almost no delay. The clearsight network analyzer is used to test the voice quality, and the test results are satisfactory to customers.

If we adopt the traditional Call-Center construction scheme, we need to build multiple Call-Center systems, mutual resource sharing and agent free-talking.

With the development of 5G technology and mobile Internet technology, multimedia access makes the unification of voice, video and instant message expand the service field of Call-Center[6]. Agents can inform remote customers how to operate the enterprise’s products through voice. At the same time, through the local demonstration of video, the Call-Center can be turned into a network training class. This can reduce the cost of training and maintenance, and provide users with vivid and vivid services. With the rapid pace of network convergence and the emergence of a large number of new technologies and new products, converged communication will become possible, and VoIP will become a new generation of dynamic contact center and foundation.

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