

Special Edition on System Components

VoIP Gateways for the Broadband Era

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Abstract

It has been more than five years since Internet telephones made their appearance. The standardization of protocols has generally been completed and progress is being made toward achieving practical use. VoIP (Voice over Internet Protocol) technology has reached the level where it can support large-scale networks and high functionality¹, and there is a demand for improvement and expansion of high value-added solutions for enterprises. On the other hand, in the network environments which network providers offer, broadband networks are growing rapidly, as exemplified by ADSL, making it practical to offer VoIP services. In this kind of changing network environment, it is expected that there will be greater polarization of VoIP users in enterprises between [a] those creating self-managed networks utilizing IP-VPN (virtual private networks based on the IP protocol) and [b] models utilizing VoIP service supplied by network providers. Oki Electric believes it is important to provide VoIP solutions for both of these alternatives.

In this paper, we introduce VoIP solutions aimed at [a] network providers who want to provide VoIP service for SOHO and consumer use and [b] call center models, as an example of a high value-added solution through which an enterprise user, while building a self-managed network, can create enterprise value. This value comes not just in terms of communication cost reduction, which was the mentality of the past, but also through improving the level of CS (customer satisfaction).

The Call Center Model

1. Background

VoIP products for enterprises initially had the objective of reducing communication cost through the integration of voice and data. Recently, however, the market has begun to demand not only communication cost reduction through use of VoIP but also the further creation of added value. In this area Oki Electric has proposed solutions which, through use of VoIP, aim at “increasing the productivity of knowledge workers” and “enhancing CS through improving the customer contact function.”

2. Market demands

The distributed call center systems of the past, by building call centers at each local base, enabled service that was in close contact with the region (region-centric). On the other hand, call center equipment and operators had to be placed at each such base and this resulted in a cost problem. In contrast, with conventional centralized call center systems, centralized handling of telephone calls was done by a CTI (Computer Telephony Integration) system at a single base. In this way, improved efficiency in call center management was achieved. On the other hand, to provide service to customers in all the base regions, long distance toll-free dialing had to be offered and increased communication fees became a problem.

3. Model for a call center using VoIP

Through the introduction of a centralized call center using VoIP, as proposed by Oki, an organization can solve the two problems of controlling equipment and personnel costs and controlling communication cost, while maintaining “region-centric service” and call center efficiency.

Our call center solution is fundamentally a centralized call center. By concentrating call center activity in one location, call center efficiency can be maintained, while controlling equipment and personnel costs. Likewise, by positioning only access points, from which customers can make phone calls, at each base and converting the long-distance portion (between the call centers and access points) to VoIP, communication cost can be controlled.

Between the call center and the various access points, an IP network is established and voice information is exchanged, but in addition, data such as the caller’s number (the customer’s telephone number), the receiving number (the toll-free or WATS line number), etc. can be transmitted, something which was not possible in the past. In this way the caller’s number can be linked at the call center to the customer database, enabling detailed and precise customer service to be achieved. Further, by assigning a receiving number for each local base, focused telephone response equivalent to that of local bases can be achieved, and the result is realization of “region-centric service” even in the context of a centralized call center.

In addition, by assigning access point telephone numbers for each service performed by the call center, telephone calls received from customers can be directly connected to the appropriate service person. By reducing the number of calls which must be forwarded, work is made more efficient and waiting time which is unpleasant for customers can be reduced, thus improving both the efficiency of “knowledge workers” and enhancing CS.

Moreover, when problems occur with call center equipment, in the past it was necessary to switch to a temporary telephone manually, using fault case switching equipment. However, our solution achieves im-

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proved reliability by monitoring call center equipment and network links over the IP and automatically connecting to a temporary telephone or another call center in case of failures.

4. System configuration

In the figures below, Figure 1 shows a conventional distributed call center model; Figure 2 shows a centralized call center model utilizing toll-free long dis-

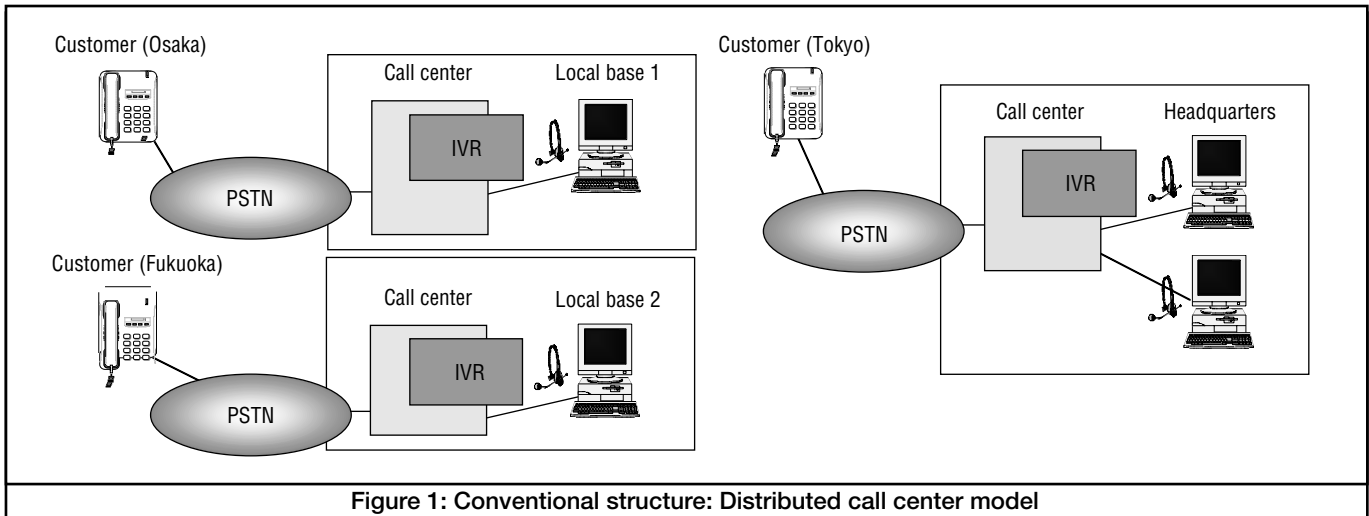


Figure 1: Conventional structure: Distributed call center model

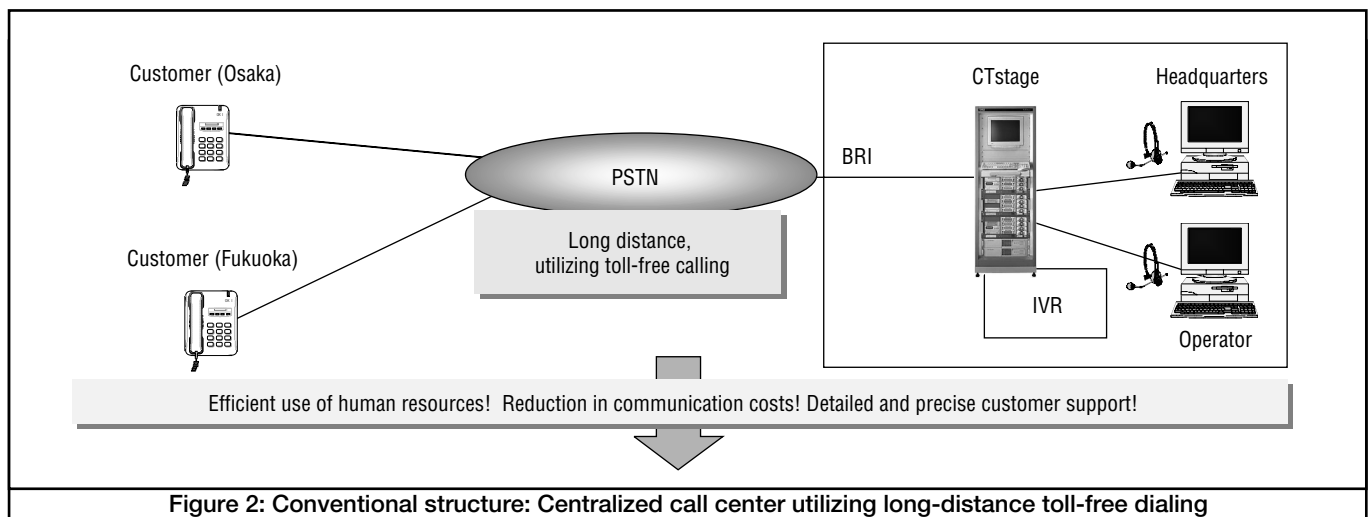


Figure 2: Conventional structure: Centralized call center utilizing long-distance toll-free dialing

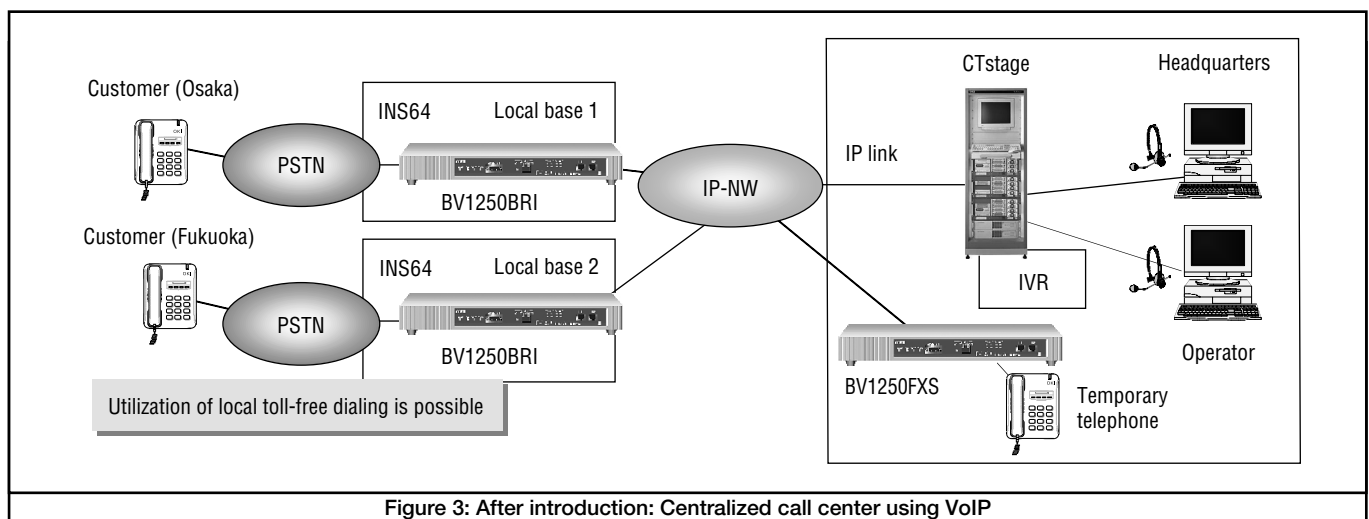


Figure 3: After introduction: Centralized call center using VoIP

tance and Figure 3 shows a centralized call center model using VoIP.

The distributed call center model has call centers constructed at each local base. With the centralized call center model of Figure 2, telephone calls from customers at each local base travel via toll-free long distance lines and are received at the headquarters by a “CTstage,” a CTI product from Oki. In the centralized call center model using VoIP, as shown in Figure 3, a CTstage is installed in the headquarters, a BV1250BRI (an Oki VoIP product having a public ISDN interface) is installed in each distributed base, and an IP network is used for the connection between each base and the central call center. By integrating voice into the data channels between each local base and the center, by means of VoIP, and connecting the long distance toll-free service (which was directly connected to conventional centers) to the local toll-free service at each distributed base, a major reduction in communication cost is possible. In addition, the temporary telephones previously described are connected to the BV1250FXS (analog telephone interface model) installed at headquarters and are used as alternate routes in cases of failures.

5. Features of this solution

- Improvement of CS through notification of caller’s and recipient’s telephone numbers

Because the customer’s (caller’s) telephone number can be communicated, it is possible to link to information in the customer database at the call center. Moreover, using the function of communicating the telephone number of the access point to the call center via the IP network, it is possible even for a call center centralized in one location to handle customers individually on behalf of each local base, according to the number of the respective access point. In addition, by providing an access point telephone number for each type of customer service, telephone calls from customers are directed to the person in charge of the proper customer service, making it possible to provide prompt response.

- Reduction in communication cost through separation of access points

By installing a BV1250BRI as an access point at each local base and accommodating public telephone circuits, such as toll-free lines, it is possible to make the CTstage a call center system which communicates data and voice through VoIP. In this way, by accepting local toll-free calls at each local base, long distance toll-free calling fees can be reduced.

- Improvement of call center system reliability

Improved reliability is achieved by monitoring (from the BV1250BRI of each local base) the response of the CTstage at the center and connect-

ing temporary telephones or another CTstage in the event of a failure. It is also possible to install a BV1250BRI and a CTstage call center at the same local base and use the BV1250BRI as a switching apparatus for improving call center reliability.

VoIP Solutions for Network Providers

1. Background

To implement VoIP, enterprises have built low cost voice networks using self-managed facilities such as dedicated lines, IP-VPN, etc. In contrast, the SOHO operator or consumer had no alternative except the use of services such as “My Line.”

2. VoIP solutions for network providers

The solution to the above problem is our VoIP solution for network providers. VoIP services offered by network providers can now be put into use because of the rapid growth in broadband networks, such as ADSL, and the conversion of the backbones used by network providers to broadband backbones. SOHO operators or consumers can now enjoy VoIP with a minimal level of equipment. This is possible because network providers can maintain and control all the necessary equipment, such as gatekeepers and voice-priority control routers, and can build and manage the IP networks. In general, network provider VoIP solutions can provide the following four services:

- ① Connection to dedicated IP lines for business users (voice traffic between SOHO’s),
- ② IP telephone service for consumers (voice traffic between consumers),
- ③ Service between business users and consumers (voice traffic between SOHO’s and consumers),
- ④ Voice communication service from all of the above to general telephones.

Between service users, it is also possible to offer flat-rate type service which is low-priced and offers unlimited talking time, regardless of distance. Telephone communication with parties other than service users is done by connection to general telephones via public network gateways.

3. System configuration

Figure 4 shows a system configuration example using ADSL circuits. In the SOHO block, by installing a 4-channel gateway (which is an Oki VoIP product) in the LAN network under the ADSL router and switch, existing button-type telephones can be accommodated. For consumers, by installing a 1-channel gateway under the ADSL router, general telephone sets and PC’s can be accommodated, and services similar to those of SOHO can be utilized. For new customers of consumer IP services and so-called heavy-users who install multiple PC’s, we recommend an integrated broadband VoIP router incorporating ADSL, a router, and a 1-channel gateway. The main unit is

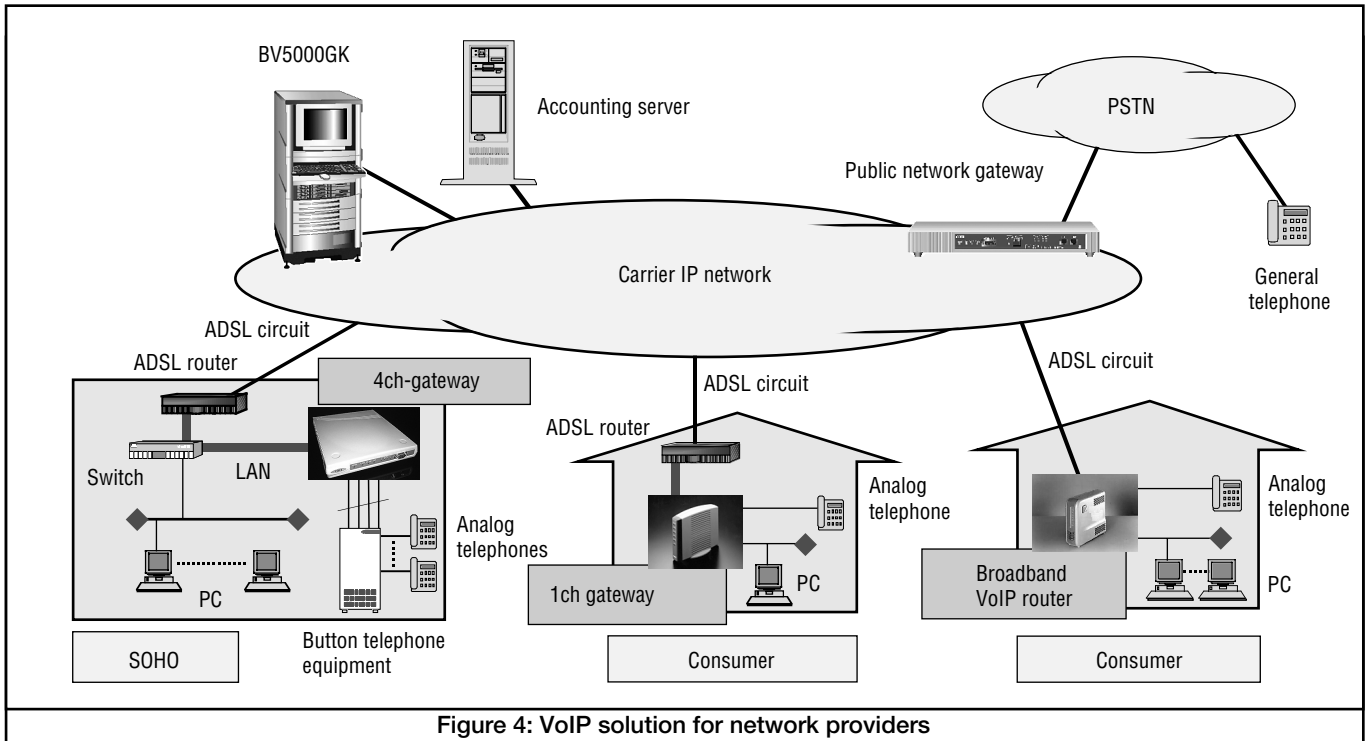


Figure 4: VoIP solution for network providers

connected to an analog circuit with ASDL service and with this one unit analog telephone sets and PC's can be accommodated.

In this configuration, an Oki BV5000GK unit (a gatekeeper) is installed in the call center. The gatekeeper has the functions of [a] controlling voice traffic among the 4-channel routers, the 1-channel gateway, and the broadband VoIP; [b] address resolution for translating a telephone number into an IP address; and [c] forwarding the accumulated traffic information to the accounting server.

Connections to public circuits are made using Oki's BV1250BRI (public ISDN interface gateway) and BV1250FXO (analog public interface gateway). These are Oki public VoIP products and are used as public gateway devices.

By adopting this solution, service can be provided to SOHO's and consumers, two sectors where building and managing self-managed networks is difficult and where little progress had been made toward conversion of voice networks to VoIP.

4. Features of this solution

- Voice traffic between service users
Through integrating voice and data on IP networks,

voice telephone service can be provided at low cost and at a fixed fee, regardless of distance,

- Telephone calls to parties other than service users
Calls to general telephones can be made through public gateways connected to general public networks.
- FAX communication
FAX communication is achieved in accordance with IPT-T recommendation T.38 which is a protocol for sending FAX's over IP networks.
- Speech compression capability
In making calls between the various terminals, any of the following ITU-T speech compression recommendations can be selected: G.729A (8kbp compression), G.723.1 (6.3k, 5.3kbps compression) or G.711 (no compression). Users can select the protocol which best matches their requirements for voice quality and limits on usable bandwidth.

Reference

1. Yamamoto, Matsunuma, et al: IP Telephony System BV1250, Oki Technical Review, Number 182, Vol. 67, No. 1, 2000, pages 35-38.