White Paper

How to Add MSN Messenger Services for PC-to-Phone Functionality to Cisco Packet Voice Networks

Overview

In September 2001, Microsoft announced the Windows Messenger Update for its MSN Voice.Net initiative. With the Windows Messenger Update, MSN users can now make voice over IP (VoIP)-based PC-to-PC and PC-to-phone calls using the MSN Messenger client. These communications use the Internet Engineering Task Force's (IETF's) Session Initiation Protocol (SIP RFC2543). The announcement also highlighted several points:

- Windows Messenger uses standard communications protocols, SIP and SIP for Instant Messaging and Presence Logical Extensions (SIMPLE)
- Windows Messenger uses SIP and SIMPLE to integrate the multimedia capabilities of voice, video, and instant messaging
- MSN is working with Internet telephony service providers (ITSPs) to provide Public Switched Telephone Network (PSTN) origination and termination services for MSN users wishing to use the PC-to-phone functionality

This announcement is important to the telecommunications industry because it not only shows Microsoft's support for standard protocols, SIP in particular, it also provides an opportunity for ITSPs to have access to millions of potential MSN users and their VoIP-based minutes of use. With SIP-enabled packet voice networks, ITSPs can offer PC-to-phone call transport services to MSN Voice.Net subscribers.

Understanding the MSN Voice.Net Service

Two primary services offered under the MSN Voice.Net service umbrella are PC-to-PC and PC-to-phone calls. Service providers with Cisco Systems networks can offer PC-to-PC and PC-to-phone services to MSN Voice.Net service subscribers by building packet voice networks to support the services themselves, or they can partner with ITSPs who participate in the Cisco Powered Network program to offer these services.

Cisco IOS® Software provides quality of service (QoS) capabilities that enable Internet service providers (ISPs) to offer PC-to-PC and PC-to-phone calls on their networks. For service providers that wish to use their data networks to offer only PC-to-PC services, Cisco IOS Software provides the necessary QoS capabilities to reduce the delay and jitter of VoIP traffic, and ensures that VoIP traffic can be given the highest priority over other types of traffic.

The MSN Voice.Net PC-to-phone service provides its subscribers with the power to make low-cost, high-quality voice calls, with access to both domestic and international calling areas. The calling coverage that an ITSP provides, whether domestic or international, is entirely up to the ITSP, and MSN is partnering with ITSPs around the world to provide the widest international coverage possible.

The initial MSN Voice.Net service offering is for PC-to-phone services, so currently the service requires only termination capabilities from ITSPs. MSN Voice.net will add phone-to-PC capabilities to its service offerings in the future.

Components of the Network

Figure 1 shows how an ITSP can interconnect with the MSN Voice.Net service. The main components are as follows:

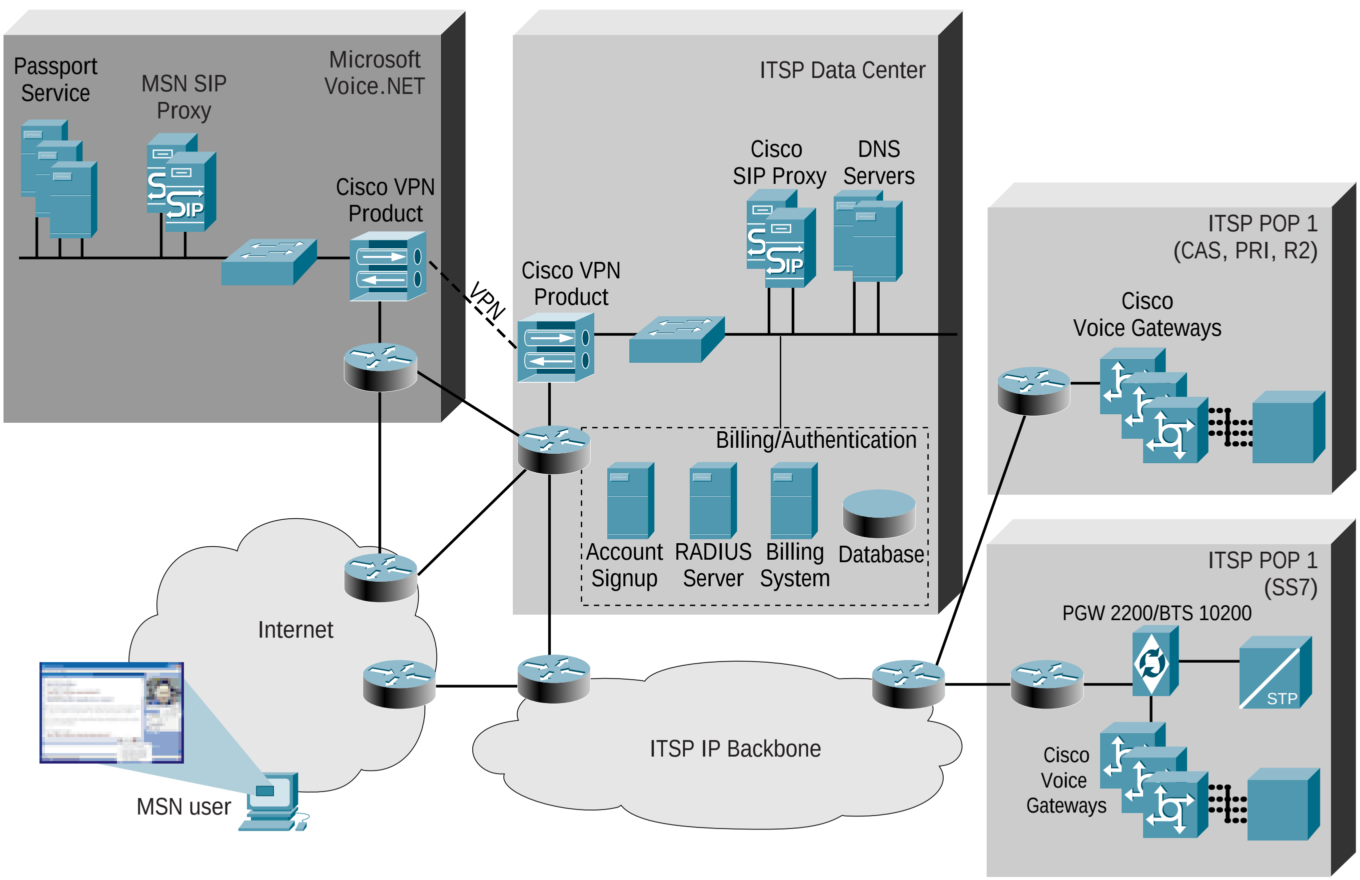
- MSN customers running Windows Messenger on their PCs
- A Secure Shell Protocol (SSH) connection between the MSN user's PC and MSN Passport servers
- Microsoft SIP proxy servers within the MSN network that route PSTN calls to ITSPs
- An IP Security (IPSec) virtual private network (VPN) connection between the MSN network and the ITSP
- SIP proxy servers within the ITSP network that route PSTN calls to the ITSP's SIP-enabled gateways
- SIP-enabled gateways connected to the PSTN, which terminate calls to the PSTN

In a typical scenario, the following events occur for a PC-to-phone call:

- MSN users log into the MSN Voice.Net service using their username and Passport ID through the Messenger client.
- The MSN users are prompted to choose from a list of local ITSPs for their PC-to-phone calls. The local list is prompted by the users' Passport file.
- The MSN users decide to make a PC-to-phone call and launch the "Make a Phone Call" application from Messenger.
- When the MSN users dial the PSTN number, the MSN SIP proxy server routes the call to the SIP proxy server of the chosen ITSP.
- The ITSP's SIP proxy server then routes the call to its SIP-enabled gateways and to the PSTN.

Figure 1

MSN Voice.Net Architecture



To participate in the MSN Voice.Net service, the ITSP must be able to meet certain basic criteria:

- The ITSP service network must be precertified by MSN on its test network. The personification process tests not only the ability to make PC-to-phone calls, but also the ability to interwork with the MSN user-provisioning model.
- The ITSP must establish an IPSec VPN connection between the ITSP and MSN networks. The IPSec tunnel can be terminated on Cisco VPN 3000 Series products, as well as Cisco routers that can terminate IPSec tunnels. The IPSec tunnel is required to ensure that the signaling between MSN and the ITSP is secure.
- The ITSP must offer the ability to terminate either G.711 or G.723.1 coders-decoders (codecs).
- The ITSP must have the ability to collect the Passport user ID (PUID) from SIP signaling communication and associate the PUID with the charges for a particular user.
- The ITSP SIP proxy server must have the ability to add a Record-Route header.
- The ITSP must provide VoIP QoS, including a one-way delay of less than 150 ms.

The variables that MSN allows an ITSP to specify include the following:

- How the MSN users are charged for PSTN termination services-The billing schemes can be prepaid, postpaid, or flat rate. An ITSP's billing model is not specified by MSN, other than the association between the MSN users and their PUIDs.
- How much geographic coverage the ITSP provides for PSTN calls-Some ITSPs may provide global coverage, whereas other ITSPs may provide only local region coverage.
- Whether User Datagram Protocol (UDP) or Transmission Control Protocol (TCP) signaling transport is used between MSN and the ITSP-Although TCP is the preferred mechanism by MSN, UDP is also an allowed transport.

The following output is a sample SIP INVITE message from an MSN Voice.Net Messenger client to a SIP-enabled gateway connected to the PSTN. The highlighted messaging shows how the requirements described previously are signaled to the ITSP network.

```
INVITE sip:+15209204015@64.102.254.147:5060;user=phone
SIP/2.0
Record-Route:
<sip:+15209204015@64.102.254.146:5060;user=phone;maddr=64.102.254.146>
Via SIP/2.0/UDP 64.102.254.146:5060;branch=4708f6b8-16a78dd4-f3d5e768-aa17128a-1
Via SIP/2.0/UDP 207.68.169.181;branch=393E1BB3, SIP/2.0/TLS 64.102.254.150:2402;received-port=2402
Record-Route: <sip:+15209204015@ciscoNc.sipProxy.vdn.pilport.com:5060;user=phone;maddr=207.68.169.181>;
tag=729CC26274322D2EB8C120684CB7557C
Proxy-Authorization: basic MDAwMzNmZmY4MDM5Yjg5ZDou
From: "00033fff8039b89d"
<sip00033fff8039b89d@sipProxy.vdn.pilport.com443>;tag=c7ee8263-4337-48b7-8ffe-d8a7e3f447cc
To:
<sip:+15209204015@ciscoNc.sipProxy.vdn.pilport.com;user=phone>
Call-ID: 62e28ba0-12fb-4b7a-8402-2b64a76c11af@64.102.254.150
CSeq: 2 INVITE
Contact: <sip:64.102.254.1502402;transport=tls>
User-Agent: Windows RTC/1.0
Content-Type: application/sdp
Content-Length: 464

v=0
o=cisco-a5xhl89yd 0 0 IN IP4 64.102.254.150
s=session
c=IN IP4 64.102.254.150
b=CT1000
t=0 0
m=audio 14638 RTP/AVP 97 111 112 6 0 8 4 5 3 101
a=rtpmap:97 red/8000
a=rtpmap:111 SIREN/16000
a=fmtp:111 bitrate=16000
a=rtpmap:112 G7221/16000
a=fmtp:112 bitrate=24000
```

```
a=rtpmap:6 DVI4/16000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:4 G723/8000
a=rtpmap:5 DVI4/8000
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
```

Building a Billable Service

Though the prospect of providing a service to potentially millions of MSN and Windows users could be attractive to an ITSP, the ITSP's main interest is making this PC-to-phone service billable and profitable. To address profitability, it is important to discuss how an ITSP can create a billing model for an MSN Voice.Net service. As stated previously, an ITSP can use a variety of billing models, depending on its technology and business needs.

Billing models among ITSPs can usually be grouped into three categories: prepaid, postpaid, and flat rate.

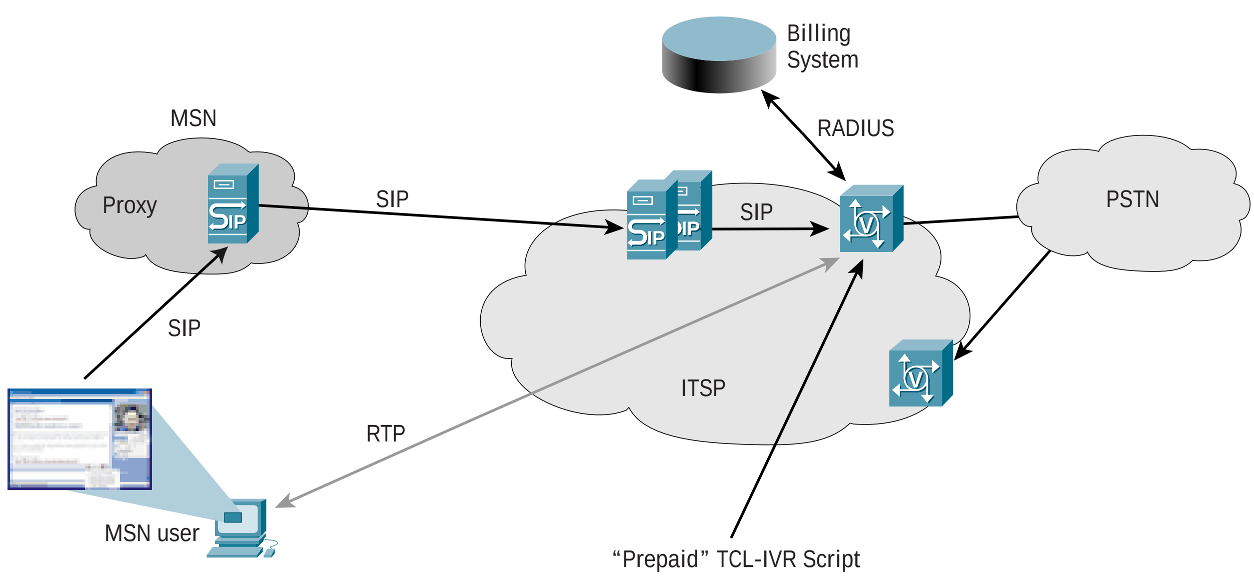
Prepaid Billing

In this model, the customer pays the ITSP some amount of money prior to using the service. The money might be a fixed amount (for example, $100), or might be associated with a fixed service (for example, 1000 calling minutes). When users log into the service, a meter keeps track of the amount of service that they use, and stops the availability of the service when the value of the prepaid contract has expired. At that point, the users can either purchase additional services or discontinue the use of the service.

The prepaid billing model allows for two types of network configurations, depending on the equipment being used:

- Prepaid billing application running on the terminating SIP-enabled gateway (refer to Figures 2 and 3)
- Back-to-back user agent (B2BUA) running a prepaid billing application (refer to Figures 4 and 5)

Figure 2
Prepaid Billing Model—Toolkit Command Language Interactive Voice Response (TCL IVR) on Cisco Packet Voice Gateways

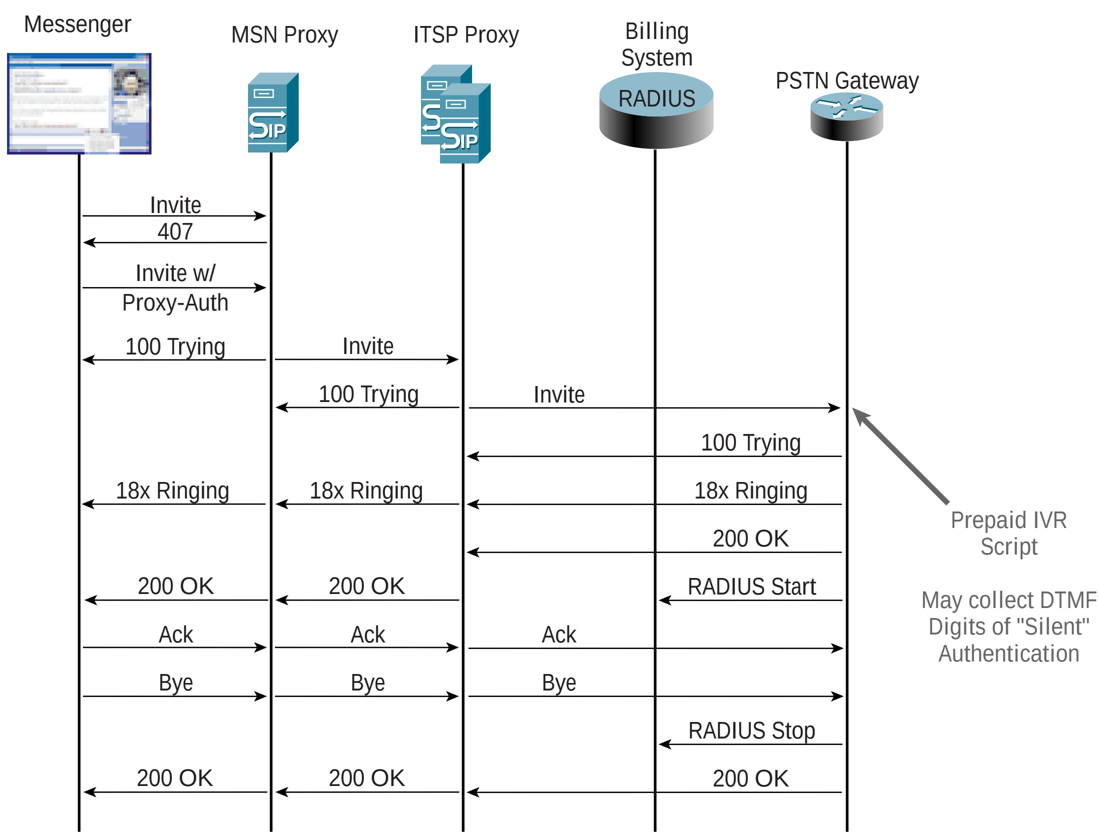


The prepaid billing model is used for several reasons:

- It allows the ITSP to collect revenue before the service is delivered.
- It allows the ITSP to avoid losing revenue from customers who use the service but do not pay their bill (billing fraud).
- It helps the ITSP with capacity planning, because they know the maximum potential volume that they have to build into the network.
- The ITSP may not own all the components of the network needed to deliver the end-to-end service (for example, international partnerships allow an ITSP to provide worldwide termination services) and the ITSP may not be able to accurately bill for calls going to non-owned resources (that is, partner PSTN gateways).

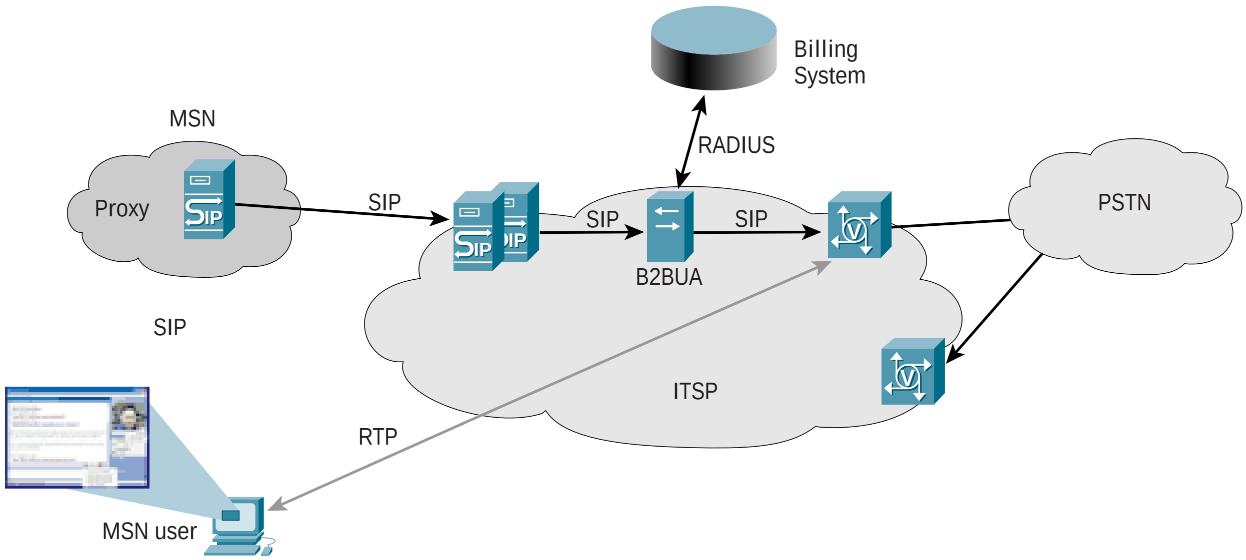
Figure 3

Prepaid Billing Call Flow, TCL IVR on Cisco Packet Voice Gateways



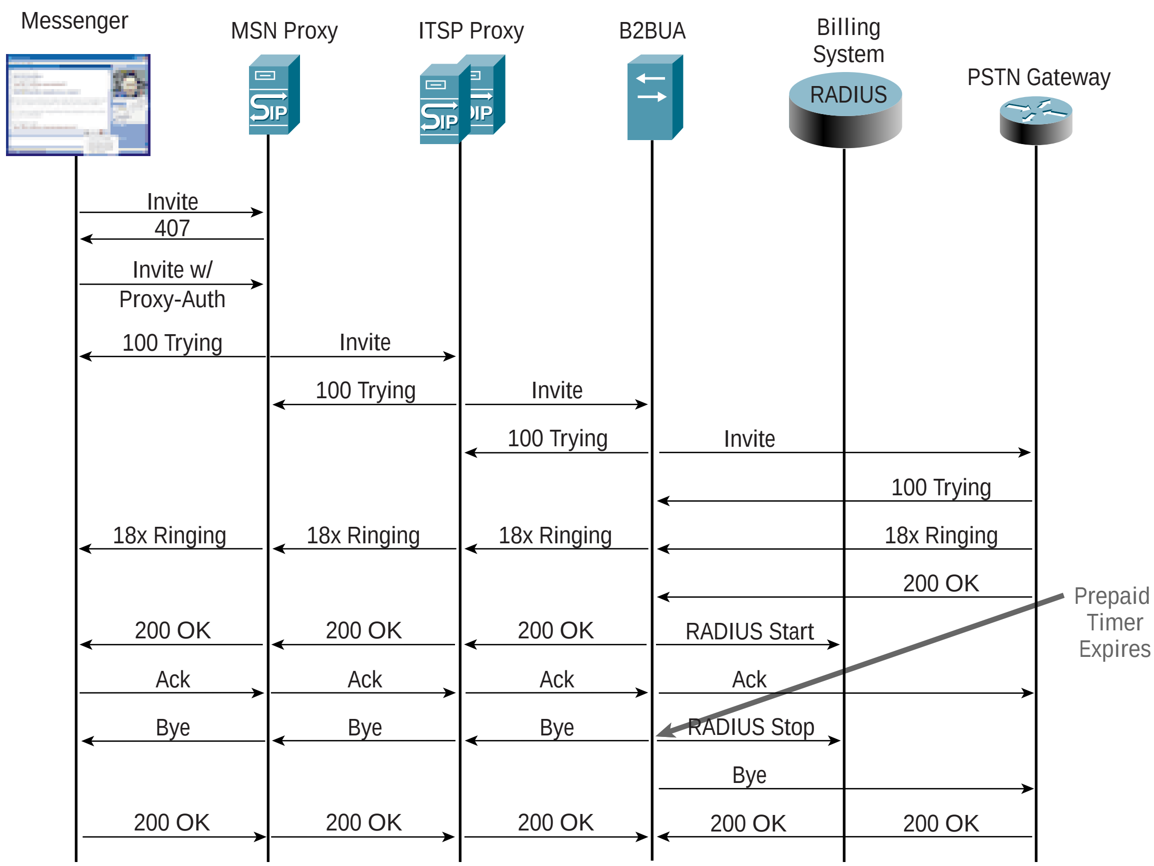
When the ITSP is using a prepaid billing model, we refer to the ITSP as a “retailer.”

Figure 4
Prepaid Billing Using a Back-to-Back User Agent Server



Note: Cisco does not currently sell a SIP server with B2BUA functionality, but this functionality is available by working with selected Cisco service provider solutions ecosystem partners.

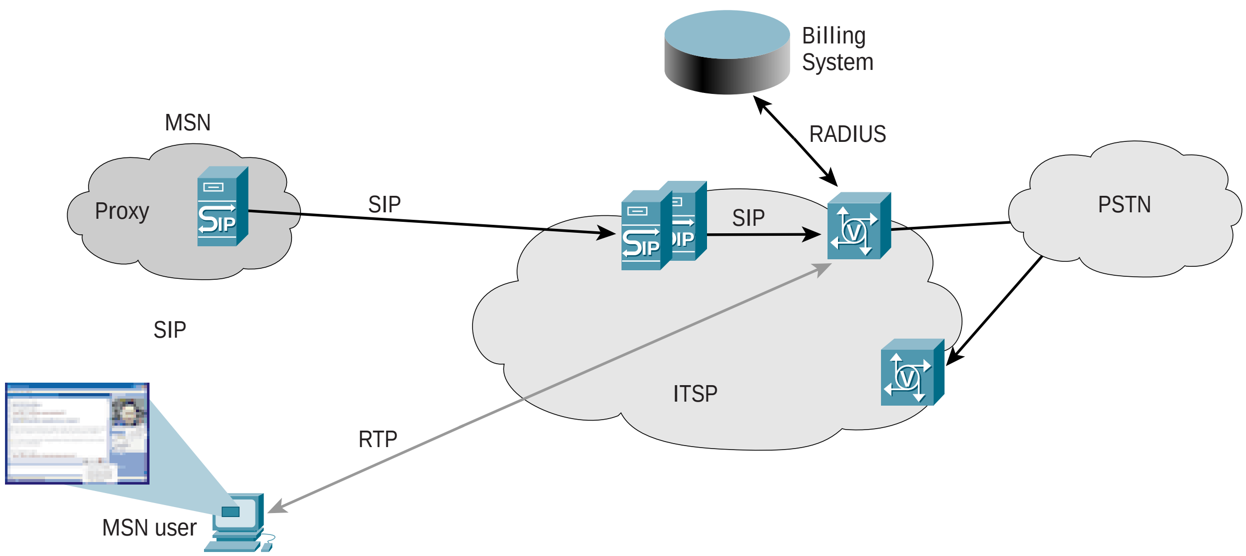
Figure 5
Prepaid Calling with Back-to-Back User Agent Server—Call Flow



Postpaid Billing

In this model, the ITSP has a trust relationship with the customer and collects revenue for the service after it has been delivered. In this case, the customer may be an individual customer or a partner ITSP that provides wholesale services to the retail ISP. Although there may also be a security association between the ITSP and the customer, the "trust" relationship is a fiduciary one. A typical network supporting a postpaid billing model is illustrated in Figure 6. Figure 7 shows the typical call flow between an ITSP and the MSN Messenger service, where billing information is captured after the call is initiated.

Figure 6
Postpaid Billing

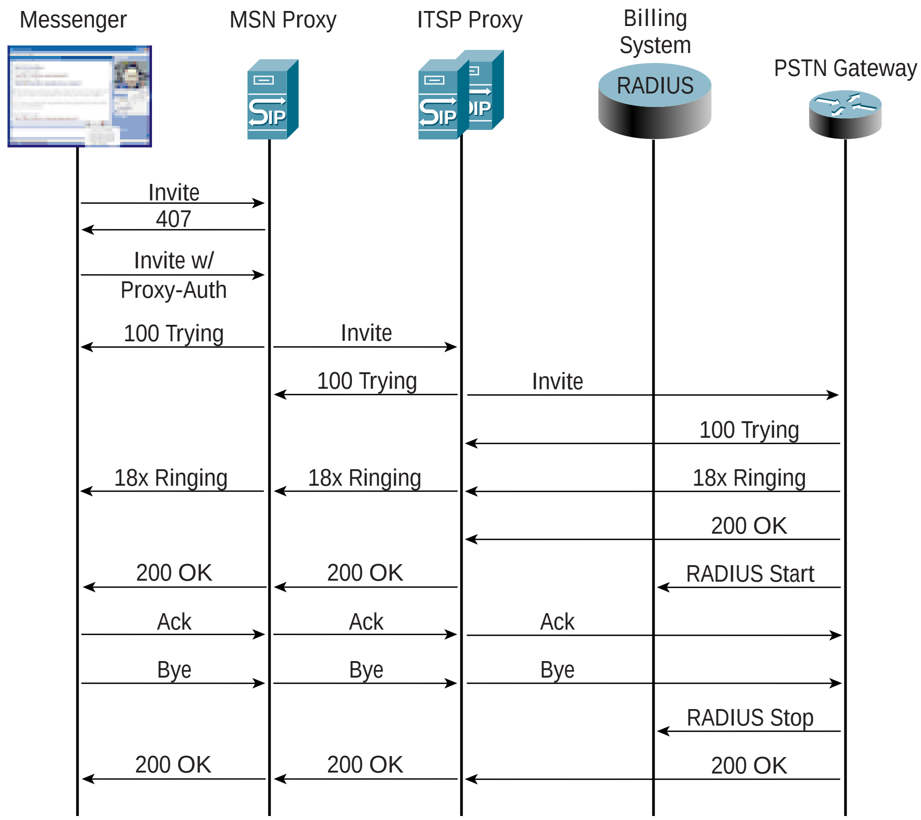


A postpaid billing model is used for several reasons:

- It allows the ITSP to provide billing for only a particular period of time (for example, daily, weekly, or monthly).
- The ITSP may not have the billing infrastructure to provide a prepaid billing model.

When the ITSP is using a postpaid billing model, we refer to the ITSP as a "wholesaler."

Figure 7
Postpaid Billing Call Flows



Note: In the postpaid billing diagrams, the Call Detail Record (CDR) generation can be done on either the SIP-enabled gateway or on the SIP proxy server.

Flat-Rate Billing

In this model, the MSN Voice.Net customer pays the ITSP a fixed amount of money for an unlimited amount of service over a specified period of time (for example, unlimited calls for the month). In this case, the ITSP uses a postpaid billing model in which the ITSP does not authenticate the user prior to each usage of the service, nor does the ITSP necessarily have to monitor service usage of each customer.

The flat-rate billing model is used for several reasons:

- It allows the ITSP to build a simplified billing model.
- It allows the ITSP to bundle its MSN Voice.Net VoIP service with other data services that it offers its customers using flat-rate billing.

When using a flat-rate billing model, the ITSP may be referred to as either a retailer or wholesaler, depending on its relationship to the MSN service or other ITSP partners.

Cisco Products for Adding MSN Messenger to a VoIP Network

The following Cisco SIP-enabled products can be deployed in ITSP VoIP networks when deploying the MSN Messenger service.

- *Cisco packet voice gateways (Cisco IOS 12.2(2)XB or above)*—Cisco 1751 voice gateways, Cisco 2600 and 3600 Series multiservice platforms, Cisco AS5000 universal gateways, and Cisco 7200 Series voice gateways all support SIP user-agent functionality. They provide a means of connecting SIP networks to traditional time-division multiplexing (TDM) networks via T1, E1, DS3, channel associated signaling (CAS), Primary/Basic Rate Interface (PRI/BRI), R2, foreign exchange station/office (FXS/FXO), or ear and mouth (E&M) interfaces. Cisco packet voice gateways are used to build the largest packet telephony networks in the world.

- *Cisco SIP Proxy Server (v1.2 or above)*—The Cisco SIP Proxy Server provides the functionality of a SIP proxy, SIP redirect, SIP registrar, and SIP location services server. The Cisco SIP Proxy Server provides the foundation for call routing within SIP networks; it can interwork with traditional SIP location services such as Domain Name System (DNS) or ENUM, feature servers via a SIP redirect message, as well as H.323 location services using standard location request (LRQ) messages. The Cisco SIP Proxy Server runs on either Solaris or Linux operating systems.
- *Cisco BTS 10200 Softswitch (Release 3.0 or above)*—The Cisco BTS 10200 provides softswitch functionality to Class 4 and Class 5 networks, and provides SIP-to-Signaling System 7 (SIP-to-SS7) gateway functionality for American National Standards Institute (ANSI) standardized networks. The Cisco BTS 10200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as a Cisco AS5000 universal gateway or Cisco MGX 8000 Series carrier voice gateway.
- *Cisco PGW 2200 PSTN Gateway (Release 9.1 or above)*—The Cisco PGW 2200 provides PSTN gateway functionality for Class 4 networks, as well as Internet offload and SIP-to-SS7 gateway functionality for international networks. The Cisco PGW 2200 supports ISDN User Part (ISUP) certification in over 130 countries. The Cisco PGW 2200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as a Cisco AS5000 universal gateway or Cisco MGX 8000 Series carrier voice gateway.
- *Cisco PIX® Firewall (Release 6.1 or above)*—The Cisco PIX Firewall is a SIP-aware networking device that provides firewall and Network Address Translation (NAT) functionality. Because it is SIP aware, it is able to dynamically allow SIP signaling to traverse network and addressing boundaries without having to compromise overall network security. A Cisco PIX Firewall functioning in this capacity is called an application layer gateway (ALG).

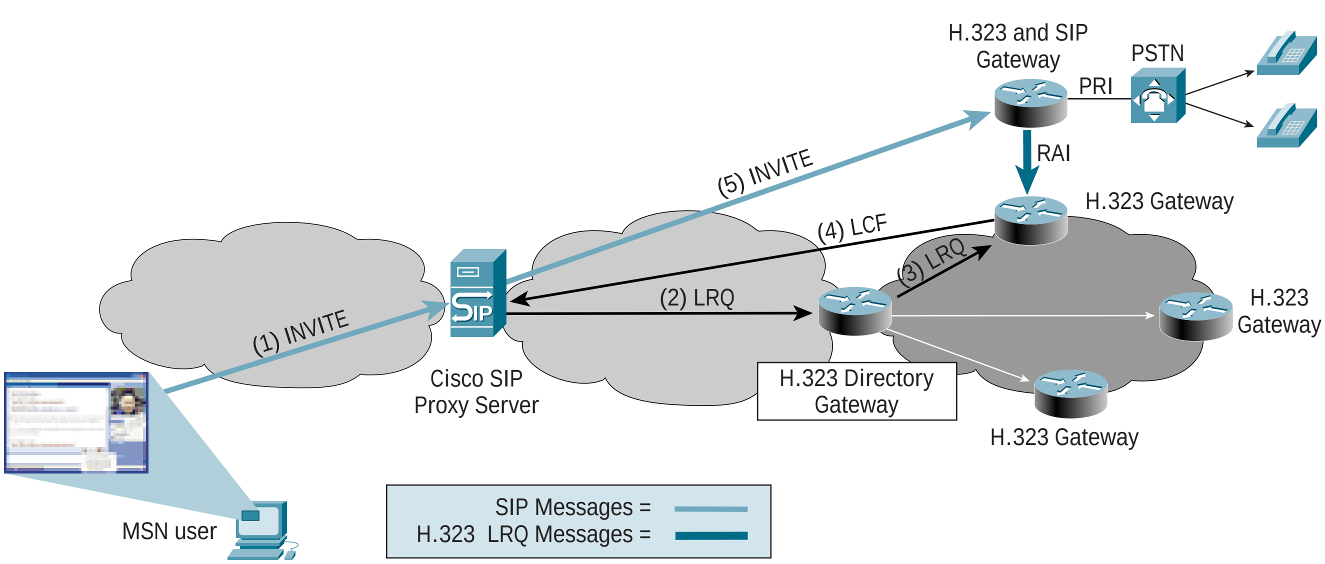
Adding SIP to Existing H.323 Wholesale ITSP Networks

Although SIP is beginning to gain market awareness and SIP-enabled network deployments are gaining market share among carriers, most of today's ITSP networks use H.323 to signal VoIP traffic. Cisco is the market leader in providing call control and packet voice gateways for H.323 networks, with many successful deployments of H.323 networks as proof points. Cisco has implemented support for SIP in its call control and packet voice gateway product lines that have existing support for H.323 and Media Gateway Control Protocol (MGCP). This multiprotocol strategy uniquely positions Cisco to help existing ITSPs seamlessly add the SIP-based MSN Messenger service to their existing networks. Seamless deployment of SIP-based services is possible because of two unique features on Cisco packet voice products:

- Support for coexistence of H.323 and SIP on Cisco packet voice gateways
- LRQ forwarding on the Cisco SIP Proxy Server

Support for coexistence of H.323 and SIP allows ITSPs to simultaneously signal H.323 and SIP calls on the same gateways (refer to Figure 8). This means that ITSPs can use their existing H.323 infrastructure and add SIP capability, and they can also avoid retraining their network engineers when they need to expand to add additional capacity.

Figure 8
H.323 and SIP Coexistence in ITSP Networks



In addition to being able to support both H.323 and SIP calls simultaneously on Cisco packet voice gateways, the ITSP can also consolidate its call routing by using the LRQ functionality in the Cisco SIP Proxy Server. The Cisco SIP Proxy Server can use H.323 LRQs to query an H.323 gatekeeper for gateway routing information. This capability provides several benefits to the ITSP:

- The ITSP can maintain the call-routing infrastructure and intelligence it has built into its existing H.323 gatekeeper network. In a hybrid network, the Cisco SIP Proxy Server appears to an H.323 gatekeeper as another H.323 gatekeeper.
- The ITSP can use the H.323 resource availability indicator (RAI) functionality on its packet voice gateways to more effectively utilize gateway resources.
- The ITSP does not have to worry about protocol translation between H.323 and SIP. Translation can introduce delay and complexity into the network. The calls remain SIP based end to end, with the H.323 network being queried only for call routing.

Conclusion

ITSPs continue to look for new ways to expand their customer base and add new revenue-generating services. The MSN Voice.Net service is a new opportunity for ITSPs to bring billing minutes to their networks. The service opportunity not only provides access to potentially millions of new users, but also offers the impetus for ITSPs to add SIP functionality to their networks. A SIP-based infrastructure enables service providers to quickly and easily add new, enhanced voice-and-data services such as voice portals, instant messaging, and hosted call centers.

As a leader in packet voice networking and with the broadest packet voice product offering in the industry today, Cisco is uniquely positioned to help ITSPs add new SIP-based, revenue-generating applications to their networks. By using Cisco SIP-enabled products and solutions, ITSPs can rapidly provide call transport and termination services, and be assured that their networks will be certified and operational with the MSN Voice.Net service.

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