APPENDIX A

Open Settlements Protocol (OSP) Clearinghouse Solution

Overview

For packet telephony service providers, terminating long-distance calls has inherent risks. With traditional circuit-switched TDM networks, a relative handful of carriers negotiated bilateral agreements for call interconnections. However, with the emergence of voice-over-IP (VoIP) carriers, the task of ensuring compatibility and interconnections with multiple TDM carriers can be difficult, from both a business and technical standpoint. Another risk is the lack of reliability of many new long-distance service providers. New entrants in this extremely competitive arena must find a way to ensure that call termination agreements are limited to reliable business partners.

Cisco VoIP gateways and clearinghouse solutions offer an alternative. By taking advantage of Open Settlements Protocol (OSP) support in Cisco devices, packet telephony service providers can employ reliable third parties to handle VoIP call termination while leveraging the bandwidth efficiencies and tariff arbitrage advantages that are inherent in IP. Thus the clearinghouses can serve as both a technical and business bridge for VoIP service providers. By signing on with such an organization and using OSP, VoIP carriers can extend service beyond the boundaries of their own network and immediately access the entire clearinghouse network of affiliated service providers.

The benefit is reduced barriers for VoIP competitors in the lucrative long-distance market. By using OSP—the only standard IP interface for VoIP clearinghouse functions—service providers have to do business with only a single settlements provider. As a result, there is no need to negotiate separate agreements with carriers in multiple countries, meet varied technical requirements for interconnection, make repeated arrangements for call accounting, or establish several credit accounts. With the OSP standard, a single clearinghouse can do it all.

At the same time, the OSP clearinghouse solution virtually eliminates the risk in doing business with new service providers with limited credit history—or with carriers in countries subject to currency fluctuations. In addition, it gives virtually every VoIP provider the worldwide calling reach they require.

How OSP Clearinghouse Solution Works

Traditionally, interconnecting carriers calculated settlements based on minutes used in circuits exchanged between their switches, often exchanging Signaling System 7 (SS7) information as well as voice paths. Call authorization was based simply on the physical demarcation point—if a call arrived, it was deemed “authorized.” This required a stable business relationship, except in the case of international traffic, where third-party wholesale carriers often provided such services.

VoIP service providers have had to adapt to such arrangements by terminating calls on the PSTN and reoriginating the call on a circuit switch. However, such an approach limits the cost-effectiveness of New World packet telephony. Even interconnection between VoIP networks was problematic—solutions were usually tightly integrated with individual vendors' proprietary and nonstandard implementations of H.323 protocols.

OSP avoids this problem by using a standard protocol approved by the European Telecommunications Standards Institute Internet Protocol Harmonization over Networks (ETSI TIPHON) organization. By allowing gateways to transfer accounting and routing information securely, this protocol provides a common ground for VoIP service providers. That way, third-party clearinghouses with an OSP server can offer call authorization, call accounting, and settlement—including all the complex rating and routing tables necessary for efficient and cost-effective interconnections. It works as follows.

A user places a call over the PSTN network to a VoIP gateway, which authenticates the user by communicating with a RADIUS server. Next, the originating VoIP gateway attempts to locate a termination point within its own network by communicating with a gatekeeper through H.323 RAS. If there is no appropriate route, the gatekeeper informs the gateway to search for a termination point elsewhere. The gateway then contacts an OSP server at the third-party clearinghouse.

At that point, the gateway establishes an SSL connection to the OSP server and sends an authorization request to the clearinghouse. OSP servers from Cisco providers supply the least-cost and best-route selection algorithms according to the carrier's requirements for cost, quality, and other parameters, selecting up to three routes. The clearinghouse creates an authorization token, and then replies to the originating gateway with a token and three selected routes. The originating gateway uses the IP addresses supplied by the clearinghouse to set up the call. To interoperate with Cisco gateways, the token should be signed, but not encrypted, conforming to PKCS#7 for Signed Data.

The terminating gateway accepts the call after validating the token. At the end of the call, both the originating and terminating gateways send usage indicator reports to the OSP server. The usage indicator report contains the call-detail information for the OSP server to provide settlement service between the originating and terminating service providers.

Cisco providers are central to the development of the OSP clearinghouse solution. By combining industry-leading expertise in call accounting and routing with industry-leading Cisco VoIP gateway technology, the solution enables clearinghouses to offer worldwide scope to Internet telephony service providers. All this is made possible by the Cisco OPT (Open Packet Telephony) framework.

OPT: The Cornerstone of OSP Clearinghouse Solutions

Where time-division multiplexing (TDM) networks literally buried services in the circuit switch, Cisco OPT brings them into the open by using a layered framework to separate infrastructure, call control, and services.

The connection layer is an ATM- or IP-based function that establishes and manages bearer connections in response to control messages from the call-control plane. Tightly integrated voice gateways encode and decode voice signals, and the low-latency Cisco packet network supports the QoS that voice services require.

The call-control layer processes call requests and instructs the connection layer to establish the appropriate bearer connection. H.323 call-control standards provide this function today. In the future, OPT will support SGCP/MGCP. All this provides complete interoperability between legacy TDM environments and packet networks.

The service application layer applies the service logic. Using standards-based protocols, third-party application providers are given access to all the capabilities of Cisco packet telephony gateways.

The Role of Alliances

By establishing an open-protocol, standards-based approach to packet telephony, Cisco OPT creates an ecosystem of providers who can develop new applications rapidly—and independently of any particular switch vendor.

OSP Solution Benefits

OSP solutions are based on Cisco VoIP gateways with embedded OSP client software, clearinghouse OSP servers, and optionally, public key certificate authorities.

VoIP service providers who use OSP gain several benefits:

- End-to-end VoIP support
- Cost-effective worldwide calling coverage
- Guaranteed settlement of authorized calls
- Incremental revenue increase by terminating calls from other service providers
- Simplified business and credit relationships
- Outsourced complex rating and routing tables
- Flexibility in selecting appropriate termination points
- Secure transmission using widely accepted encryption for sensitive data