Leveraging Standards for Low Investment, High Returns

Out of the box clearinghouse billing system uses OSP to capture new VoIP traffic, provide low cost termination, and generate new revenue streams.

The Convergent Network Revolution

A transformation is starting in today's telecommunication environment. New investment and upgrades for the 20th century Public Switched Telephone Network (PSTN) has all but stopped, while investment in high capacity data (Internet Protocol (IP)) networks grows at a double digit pace. The build up of these high capacity networks has enabled Voice over IP (VoIP) to become a commercial reality in the new millenium. The VoIP market is no longer a niche or an ISP domain. Major carriers have announced VoIP programs and are rapidly moving to an all IP network.

From 1999 through 2000, many of these carriers upgraded their IP networks or built new IP links around the world. Pursuing cost savings, major carriers were transferring circuit-switched long haul voice from the current PSTN environment to their IP networks. The downturn in the economy coupled with the postponement of diverse network expansion, has made integration an attractive proposition, as coverage is often less expensive to buy than build. Therefore, today the need for efficient, secure, and reliable interconnection has become paramount. This new voice paradigm requires a mechanism to allow diverse networks to securely interconnect, authorize these interconnections, and track the costs or usage of such interconnections.

In the PSTN environment, large telephone switches, SS7, and physical interconnection points controlled the exchange of telephony transaction between carriers. The major IP telephony vendors recognized this and with the various standards bodies began developing open standards governing the transmission, exchange, and record keeping of VoIP network transactions. The International Telecommunications Union (ITU) developed the H.323 standard for VoIP networks, while the European Telecommunications Standards Institute developed the Open Settlement Protocol (OSP) for settlement and reporting of VoIP exchanges between carriers.

Interconnection Opportunities

Carriers saw an opportunity to establish interconnect or clearinghouse relationships with other VoIP carriers worldwide. With OSP it is possible for a carrier to be interoperable with virtually all other carriers without maintaining a physical or business relationship with those other carriers. Instead, only one relationship–with the clearinghouse–has to be managed. Today OSP is providing interconnect relationships for over a billion voice minutes annually and this number is currently growing at a double-digit annual rate.

OSP provides easy entry into the international VoIP space with minimal investment. Only a small investment in software and human resources is needed to realize a rapid return on investment. OSP is also a solution for incumbent carriers who are losing business to new market entrants. OSP enables the capture of new VoIP traffic where PSTN traffic was non-existent, provides lower cost termination, and the ability to generate new revenue streams, all without cannibalizing the carrier's existing PSTN network.

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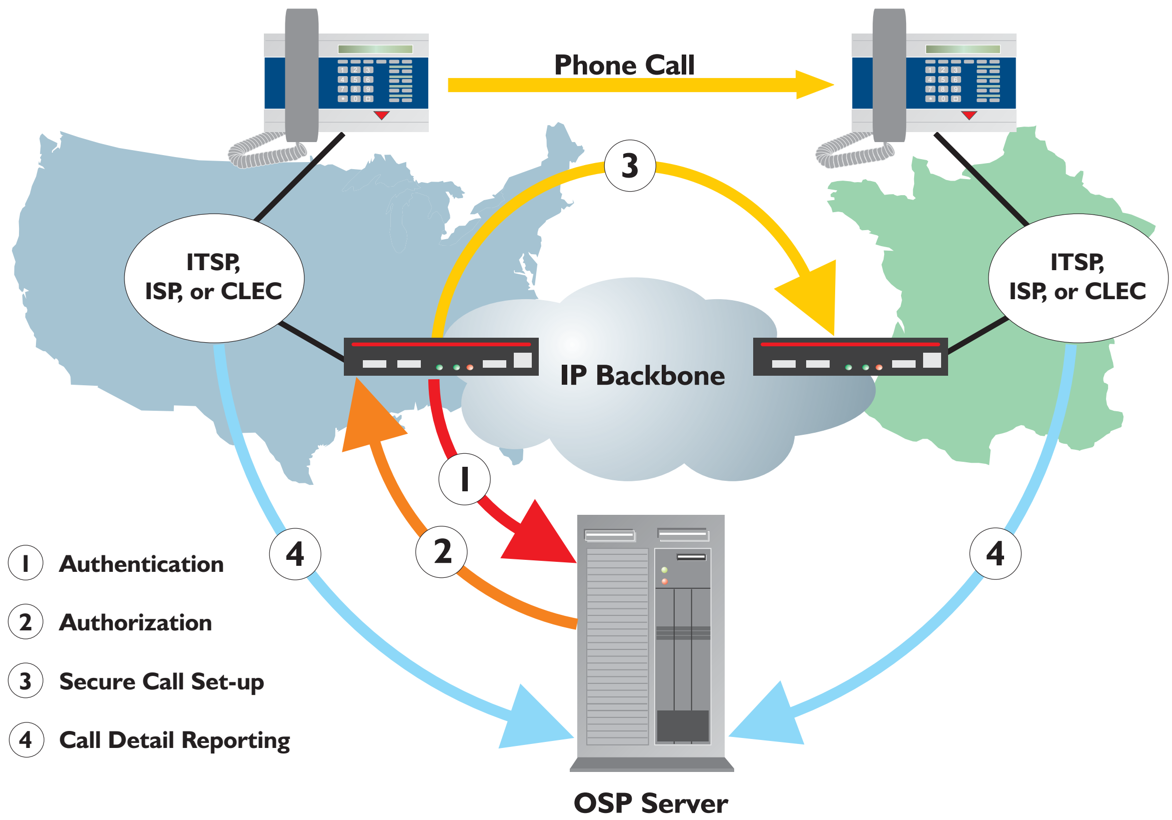
Open Settlement Protocol

The Open Settlement Protocol (OSP) is a client-server protocol which establishes authenticated connections between telephony network devices, allows the secure transfer of accounting and routing information, and enables inter-network service provider billing for any IP communication event, including, but not limited to, Voice and Fax over IP. The OSP standard adds significant value to IP telephony applications by enabling real-time, secure inter-IP domain call authorization, routing, and call detail reporting among IP devices (gateways and gatekeepers, including H.323 gatekeepers, SIP proxies, and softswitches) and clearinghouse service providers.

An OSP server allows a carrier or consortium of carriers to facilitate an interconnect solution for itself and its partners to securely exchange traffic, guarantee the identity of each partner, authorize each and every telephone transaction, as well as account for the transaction from end to end. To securely exchange traffic, each and every participant is identified to the clearinghouse by means of Public Key Encryption (PKI). Digital security is superior to access / password lists of first generation network devices.

Combined Proficiencies

TransNexus, Inc. and Digiquant are leading the revolution by providing carrier-grade solutions today. Together, TransNexus and Digiquant provide a fully integrated interconnect (clearinghouse) solution for carriers seeking secure interconnection and routing over Public or Private IP managed networks. The combined solution offers complete end-user billing (prepaid calling cards), as well as E164 routing and digital security for low cost interconnection. It is a perfect solution for carriers seeking to expand their international presence through partners rather than succumbing to the expense of building out network capacity. The solution is also applicable for carriers seeking to enter the high margin enterprise network services business, offering on and off net pricing to customers and customized internal billing solutions for Enterprises.

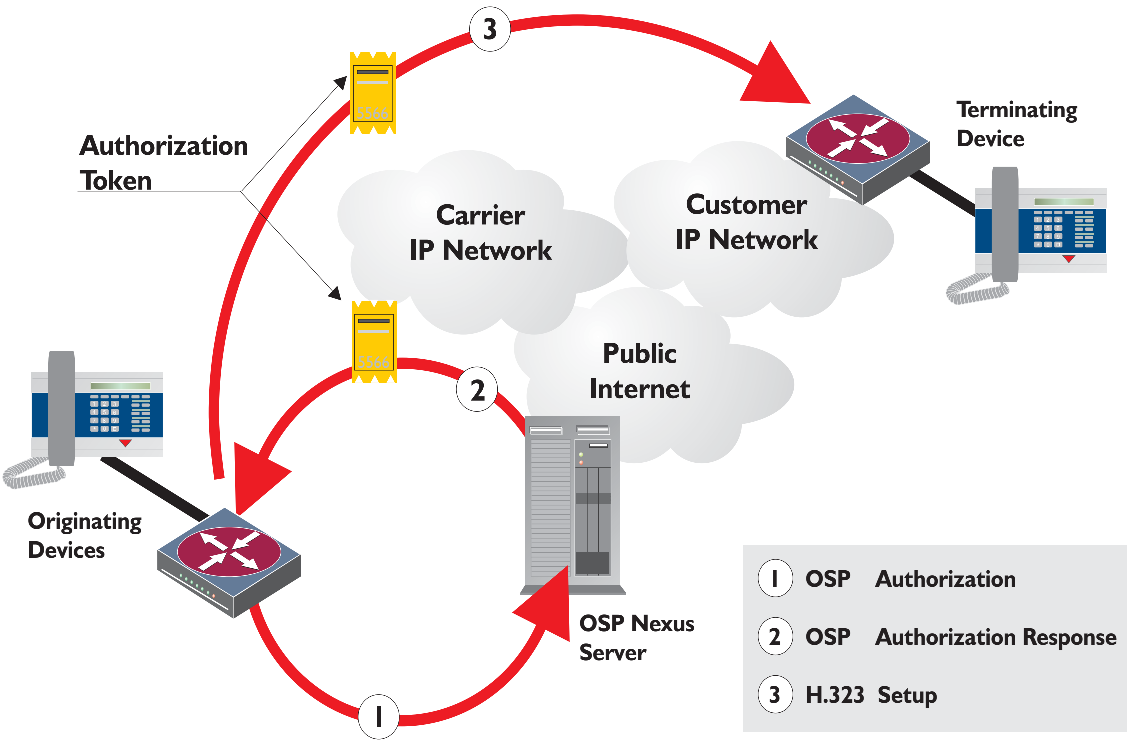


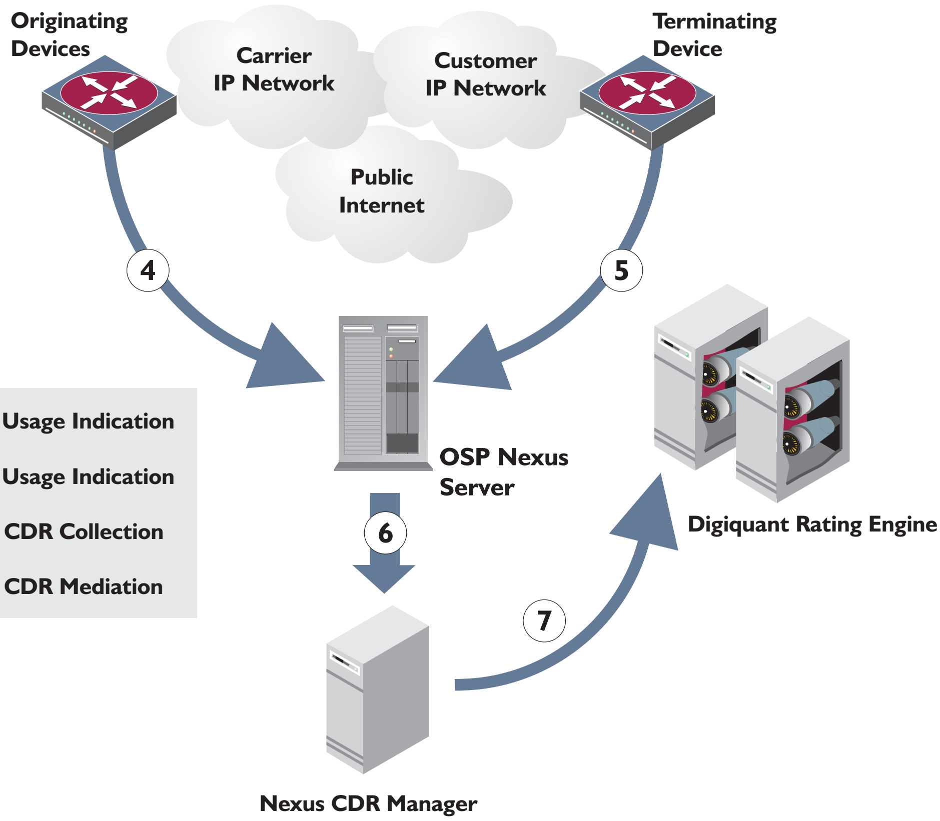
Solution Overview

Carriers today are interconnecting with other service providers on a call by call basis using the OSP Nexus Server, the Nexus CDR Manager, and Digiquant's IMS. When an originating network device receives an ingress call, it contacts the nearest OSP Nexus Server, which will authorize the originating device and send back a list of potential termination network devices serving the requested destination number. Included with the potential termination device list is an Authorization token. The Authorization token is an encrypted string, which is sent to the terminating gateway at call setup. By sending the token to the termination gateway, the gateway knows that the call comes from a trusted source, as only the termination gateway and clearinghouse know the public/private key combination. Using the termination list received in the authorization request, the originating gateway will try each termination gateway - one at a time - until a successful connection is made.

Routing tables maintained by the OSP Nexus Server reduce the complicated configuration requirements for each and every network device. For each Setup attempt from the originating network device to the terminating network device, an OSP CDR is sent from each network device to the OSP Server and stored locally. Multiple OSP Nexus servers can be used to provide carrier-grade availability; therefore, usage records can be generated at many different OSP Servers.

These records must be collected at a central point for CDR correlation. All the OSP Nexus Servers on the network collect the usage records from the Access Servers. These are retrieved by the TransNexus' CDR Manager process. The CDR Manager assembles multiple OSP records into ratable, successful, and unsuccessful call detail records for the carrier's IMS solution.





4 OSP Usage Indication
5 OSP Usage Indication
6 VPN CDR Collection
7 VPN CDR Mediation

When the Nexus CDR Manager has created a CDR, it is put into a queue within IMS for rating. Each CDR is rated three times. The first rate is for the retail call – in most cases the ITSP does not know the retail customer, in which case the retail rating results in no charge. At the same time, the CDR is also rated for both origination and termination settlement. This gives the carrier full control of its network, enabling the provider to invoice the origination partners and compare the invoices of termination partners.

Rate configuration and management, as well as performance, have been taken to a new level with Digiquant's flexible and highly optimized rating engine. IMS is shipped with a Telephony Numbering Plan, which defines all known country and area codes in the world - each associated with a location and time zone. When designing the rate plan, the user simply selects location from the Telephony Numbering Plan, and need never worry about country codes or mistyped digits.

For each defined origination and termination combination, IMS allows the specification of minute rates, setup fees (call attempt), and connect fees (successful calls) plus rounding, minimum duration, and tiered pricing, allowing the service provider to customize the rate plan to its exact needs. The rate plan also supports time of day, day of week rating, e.g. weekend calls can be configured at a different rate than calls made on weekdays.

Finally, the rate plan allows the service provider to plan future rate changes. Service providers terminating traffic in China could, for instance, run a promotion with lower rates to China for national holidays.

While the wholesale settlement process is inherently postpaid, thus eliminating the need for real-time rating of calls, IMS maximizes system utilization by sending the correlated CDRs in a steady stream to the IMS Rating Engine. This differs significantly from other systems where rating is done periodically in batches (every day or even once a week or month). Batches result in unbalanced system processing where the system struggles under an enormous load during very limited periods yet is almost idle in others.

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About TransNexus

TransNexus is a leading provider of open IP interconnect solutions for the VoIP market. Founded in 1997, TransNexus pioneered the development of commercial OSP solutions with telephony equipment partners such as Cisco, 3Com, Lucent, Ericsson, Alcatel, Nuera and others. TransNexus provides the OSP Nexus Server, the OSP Toolkit, and other commercial products to the VoIP world.

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About Digiquant

Digiquant is a global provider of infrastructure software supporting the delivery of services over advanced networks. Its IMS solution is a unique, multi-service platform providing integrated functional modules that enable service management, from network integration through billing. More than 70 telecommunications companies around the world use Internet Management System for service management and billing.

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