**Introduction**

Over the past few decades, the telecommunications industry has been heavily dependent on the traditional Time Division Multiplexing (TDM) telephony infrastructure. The circuit-switching concept of TDM-based networks was reliable but required a lot of switching management and large bandwidths. With the advent of Internet Protocol (IP), the concept of digitizing the TDM signals and sending voice over IP (VoIP) came into existence.

The Voice over IP solution is not only limited to corporate LAN infrastructure but has also extended to the World Wide Web. This expansion is facilitated by VoIP carriers or SIP (Session Initiation Protocol) Trunk service providers, which allow corporate VoIP environments to communicate over the Internet and connect to Public Switching Telephony Network (PSTN). Businesses are replacing on-premise TDM setups with VoIP services to get maximum return on their investment.

Microsoft Office Communication Server (OCS) 2007 R2 adds the support for direct SIP trunking from the service provider and offers a cost-effective option of external voice connectivity to the OCS Enterprise users. This whitepaper highlights the SIP trunking solution in general compared to a TDM-based solution and its deployment considerations in an Office Communication Server 2007 R2 environment.
**SIP Trunking**

SIP trunking is the service provided by Internet Telephony Service Provider (ITSP) to connect an Enterprise telephony network to PSTN via the Internet. The communication between the Enterprise Voice network and the service provider occurs through VoIP sessions using SIP based standards. Such a setup eliminates the need of using any IP-PSTN gateways on the Enterprise side and those services are leveraged through the SIP trunk service provider. Thus, the service provider takes the responsibility for hosting these components to make the TDM connection with the PSTN.

**Advantages Over Traditional TDM**

SIP trunking removes the management complexities of TDM networking and provides multi-vendor integration within the IP telephony environment. It provides far greater benefits compared to TDM in terms of cost and bandwidth utilization. The TDM setup usually requires a Primary Rate Interface (PRI) subscription, interface-cards, and sometimes additional PSTN gateways for every T1/E1 (corresponds to 23/30 lines) trunk expansion. SIP trunking, on the other hand, does not require PRI or Basic Rate Interface (BRI) cards and provides linear expansion without any additional hardware. The expansion can be from one line to multiple lines while T1/E1 TDM trunking expansion comes about in chunks of 23/30 lines at a minimum. The limited hardware requirement and easy expansion in SIP trunking dramatically reduces the cost of the overall solution.

Traditional telephony environments with TDM circuits provide dedicated lines from PSTN to Enterprise PBX and therefore do not make effective use of the bandwidth. The bandwidth utilization is mostly low except during the peak usage times. On the other hand, SIP trunks with voice and data on the same connection provide good bandwidth utilization, and the voice communication can even be prioritized over data to meet the peak usage and capacity requirements.

In regular telephony communication, the connection between two end-points occurs through the TDM network of PSTN, and the analog/digital conversion may come about at multiple places, depending on whether the call is local or international. A SIP trunking environment, however, has the advantage of completing the call over IP for SIP-based telephone (with SIP URI) destinations without routing through a PSTN or TDM network. For non-SIP based destinations (TDM phones), the call is eventually routed through PSTN, but the SIP service ensures that the call is transferred to the PSTN closest to the destination. For example, with an international call through SIP trunking, the call is not transferred to the local PSTN; instead, the service provider, with its multiple service points around the world, makes maximum use of the IP route to hand over the call to the PSTN that is located at the international destination. Such a process reduces the number of analog/digital conversions and retains the quality of voice communication.

**Security Considerations**

Security threats are minimal in a TDM switching environment because it is dedicated to a single application in a closed environment. SIP trunking, on the other hand, uses an open Internet environment, therefore Enterprise VoIP components connecting to SIP trunking are vulnerable to similar kinds of attacks as any computer on a public network. It is extremely vital to safeguard these devices from attacks and allow only legal access. Corporate data networks use firewalls and NAT (Network Address Translation) routers to block outside threats. These types of firewalls are typically not designed to handle SIP-specific communication. Therefore, SIP-aware firewall or Edge devices are required for this purpose.
Media streams like voice in VoIP are carried through an RTP protocol and use dynamic UDP (User Datagram protocol) ports which are mostly blocked in firewall environments. For a complete SIP communication procedure, the Edge devices should be able handle the process of opening and closing the media ports as required. On the protocol level, it is recommended to use TLS (Transport Layer Security) for SIP signaling and SRTP (Secure RTP) for media transfer. Different firewall vendors have developed expertise and tailored products to provide more security layers to voice and data communication. It is imperative to secure Enterprise Voice network before connecting to SIP trunks.
Deploying SIP Trunk With OCS 2007 R2
Microsoft® Office Communication Server 2007 R2 introduces many new features and server roles to the Unified Communication (UC) Enterprise users and enhances the functionality of existing components. The new features are tailored to suit the requirements of different Enterprise businesses and provide a range of deployments to make better use of the resources. One of the new features is the external Voice over IP connectivity using SIP trunk from the service provider. SIP trunking provides a cost-effective solution for small/medium businesses without deploying expensive telecom components like PBX and IP-PSTN gateways in the Enterprise environment.

SIP Trunking Components
Deploying a SIP trunking solution requires three main components: SIP trunk from an ITSP or service provider, an Edge device or VPN (Virtual Private Network) router, and Enterprise voice routing infrastructure. In a regular SIP trunking solution, the Enterprise voice routing component can be an IP-PBX which provides internal routing and phone normalization rules or it can be part of Service provider depending on the degree of service requirements. With Office Communication Server (OCS) 2007 R2, the telephony routing component is already integrated into OCS infrastructure, and therefore does not require on premise IP-PBX with a SIP trunking solution. The following sections briefly define the components involved in SIP trunking with an OCS 2007 R2 environment.

Enterprise Voice Routing Infrastructure
In Office Communication Server 2007 R2, the Enterprise Voice functionality handles all the voice communication between internal UC and external telephony devices. Enterprise Voice is activated through the Office Communication Front-End pool/Server where the Inbound and Outbound Routing components are configured to translate and route the calls for Enterprise users. The Inbound Routing component handles internal or incoming calls and does the appropriate routing based on policies set for answering and forwarding. The Outbound Routing component handles calls to the PSTN users through ITSP (with a SIP trunking environment) or PBX (with a conventional TDM environment). The phone numbers are normalized to E.164 format to be compliant with PSTN standards. The voice policies and phone normalization rules configured on Front-End pool/Server defines the routing infrastructure for an OCS environment.

Mediation Server
The Mediation Server provides integration of OCS Enterprise Voice with the SIP trunking solution. The Mediation Server performs the function of translating SIP signals and RTP media between the Communication Server and SIP trunk setup. Depending on the setup, if the SIP trunk is configured for TCP, the Mediation Server translates SIP over TCP on the SIP trunk side and SIP over TLS on the Communication Server side. The same encryption and decryption processes occur for the media streams.

When deploying a Mediation Server, it is recommended to use two Ethernet interfaces to physically isolate the traffic of OCS internal infrastructure from that of a SIP trunk setup. One interface (internal edge) is configured for listening on the Communication Server and the other (external edge) for listening on the SIP service provider proxy (discussed in a later section). During the configuration of Mediation Server, the next hop connection facing the SIP trunk is provisioned with a static IP address or a FQDN (Fully Qualified Domain name) of the service provider proxy. Similarly the next hop connection facing
the OCS is provisioned with FDQN of the Front-End pool/Server. Despite the fact that Mediation Server can be connected directly to a SIP trunk (from the provider), it is extremely important to use security components on the external edge of the Mediation Server to secure the infrastructure, as described in next section.

**VPN or Edge Device**

The OCS infrastructure should be secured from all possible Internet threats and illegal access before connecting to a SIP trunk from the service provider. The perimeter network joining the SIP trunk circuit to the Mediation Server requires special consideration. As mentioned earlier, the existing data firewalls in enterprise environment are usually not equipped to handle direct VoIP sessions; therefore a SIP-aware firewall or an Edge device is required to securely open the path for SIP and RTP traffic. Another option is to use a virtual path for SIP traffic in the existing firewall, using a VPN (Virtual Private Network) router. In this setup, the required ports are opened in the firewall to allow the VPN router to send and receive traffic. The connection from these devices provides a safeguarded channel to the service provider, offering PSTN origination and termination. Figure 1 shows a SIP trunk setup from an ITSP to the OCS infrastructure.

![Figure 1: SIP Trunking setup from a service provider to an Office Communication Server 2007 R2 environment](image-url)
**SIP Trunk Service**

SIP trunk service provides a VoIP connection to the Enterprise voice network on its front end and makes a TDM connection to the PSTN on its back end. The ITSP network usually consists of a proxy server (Session Border Controller) and IP-PSTN gateways as shown in Figure 2. The Session Border Controller (SBC) is a VoIP session-aware device that provides SIP services across NAT and firewall devices located at the Enterprise site. It communicates with Mediation Server and manages all the VoIP sessions. The PSTN gateways are responsible for handling calls that are eventually routed to the PSTN network.

![SIP Trunk Service Diagram](image)

**Figure 2:** ITSP components providing a SIP trunk connection on the front end and a PSTN connection on the back end

The SIP trunking solution components, as described above, work in combination to provide external connectivity to OCS Enterprise Voice users. When an Enterprise Voice user initiates a call from an Office Communicator 2007 R2 client to an external user, the appropriate routing rules are invoked on the Front-End pool/Server, and phone normalization takes place. The call is then forwarded through Mediation Server and an Edge/VPN device to the ITSP. Some Edge devices work in conjunction with the ITSP to look up the destination SIP URI and decide if the call will be routed completely through the Internet or handed over to the IP-PSTN gateway of the service provider for termination to PSTN.

**High Availability With SIP Trunks**

SIP trunk from a service provider can become a single point of failure if deployed without any high-availability considerations. There are various ways of configuring high-availability depending on the type of Edge infrastructure and devices deployed. Multiple SIP trunks from multiple service providers
can be deployed in an Enterprise for PSTN termination. One way to provide complete redundancy is to connect each SIP trunk to a separate VPN/Edge device and Mediation Server setup, as shown in Figure 3. The Edge devices deployed in such a setup should be capable of monitoring the health of the SIP trunk and of failing over to the secondary Edge setup if the primary experiences a failure. Some Edge devices are also capable of handling SIP trunks from multiple providers on the same device and provide failover between SIP trunks during service provider failures.

Figure 3: Providing high-availability with multiple SIP trunks to an Office communication Server 2007 R2 setup
Conclusion

SIP Trunking provides a low-cost solution with a maximum return on investment. With minimum exposure to the TDM circuits, it makes effective utilization of IP bandwidth. SIP trunking with proper security measures provides the same level of reliability as TDM trunks. Its direct connectivity to the Office Communication Server 2007 R2 offers a cost-saving option for small medium businesses as well Enterprise environments migrating from TDM based PBX setup to a VoIP solution. Multiple SIP trunks can be deployed from multiple service providers to provide complete high-availability of the solution.

In addition to a VoIP solution with SIP trunking support, the Office communication Server 2007 R2 infrastructure offers a complete set of unified communications with advanced features such as enhanced IM presence, A/V conferencing, Live Meeting, and much more. Dell PowerEdge™ Servers, Dell PowerVault™, Dell EqualLogic™, and Dell/EMC® Storage provide a suitable platform for deploying the OCS 2007 R2 infrastructure. Dell offers Microsoft SQL Server solutions for hosting the OCS 2007 R2 back-end databases and also offers complementary Microsoft Exchange Server solutions for hosting e-mail. These solutions provide a comprehensive platform for implementing an OCS 2007 R2 infrastructure with required availability features. Dell Services include assessment, design, and implementation tailored to those UC and messaging deployments. More information about Dell Unified Communications can be obtained at www.dell.com/Unified.