

# SIP Trunking



**SIP (Session Initiation Protocol) is a signaling protocol used to convert voice into packets so it can travel across IP networks. DSCI's SIP Trunking service connects your existing PBX phone system over your data network to the DSCI network with the voice service layered on top of the network as just another IP application.**

## Benefits

### Cost-Saving Installation

DSCI SIP Trunking can often be installed using an existing SIP-capable phone system and highspeed connection. You get all the advantages of a converged network without the cost of installing completely separate systems.

### Cost-Efficient Usage

DSCI SIP Trunking combines voice and data networks into a single "converged" network. Customers' voice traffic is routed over any type of data network such as Dedicated Internet Access (DIA), or an MPLS service. In addition, local, long distance and toll-free services are accessed over the same circuit.

### Flexibility

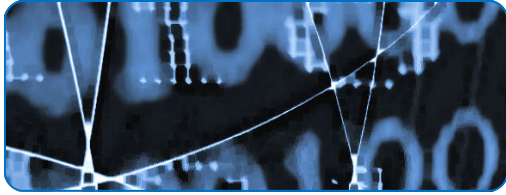
DSCI SIP Trunking's converged services provide flexibility and cost savings without the limitations of TDM circuits traditionally used by the PBX. With TDM, the number of concurrent calls depends upon the number of channels leased from the service provider. With SIP Trunking, only bandwidth determines concurrent call capacity. And, unlike TDM, additional SIP Trunking capacity can be purchased in single increments.

### Quality of Service

With a DSCI T1 in addition to SIP Trunking, QoS (Quality of Service) Priority Queuing is automatically configured which identifies network traffic, giving priority to voice over other applications.

## Features

- Available as a stand-alone service over an existing Internet connection or coupled with DSCI DIA or MPLS service.
- DSCI SIP Trunking with Hosted SBC option allows customers with multiple PBXs to utilize a single SIP trunk to share capacity, increase savings, and provide redundancy.
- Access local, long distance and toll-free service and Internet over a single circuit.
- 911, 411 and operator services available.
- Direct Trunk Overflow enables incoming calls to be routed to an alternate number(s) when a SIP Trunk group is busy or if the T1 is out of service.
- Caller ID.



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## Service Configuration Requirements

Additional provisioning may be necessary on your existing PBX and/or firewall to configure equipment properly for DSCI's SIP Trunking service. Please consult with your Account Executive for details.

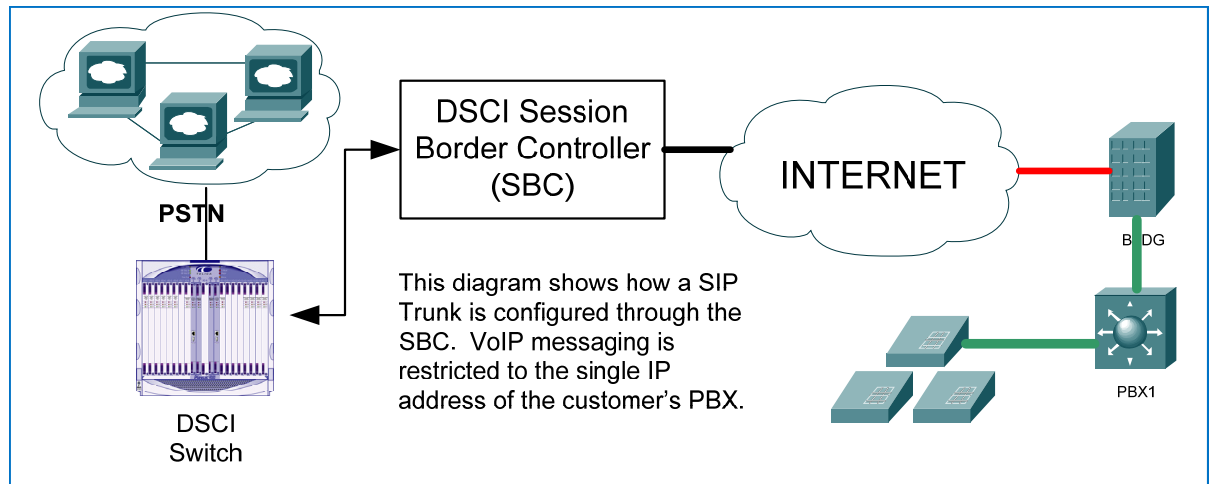
## Technical Details

DSCI SIP supports:

- Transport over UDP
- IPV4 in the Invite/From/To/Contact/Refer fields
- OPTIONS ping method for trunk availability
- Operator Assistance 0, 00, 0+10Digits
- DTMF In-Band, SIP-INFO and RFC2833
- CODEC: G.729A and G.711U

DSCI SIP does not support:

- Transport over TCP nor TLS
- FQDN in the Invite/From/To/Contact/Refer fields
- Registration, nor Authentication
- 7 digit dialing for local numbers



## Additional SIP Trunking Options

**Extended SIP Trunking** - T1 access and SIP trunk provided outside the DSCI network footprint through alternate T1 providers.

**SIP Trunking with Hosted SBC** (Session Border Controller) - allows multiple customer VoIP-enabled PBXs to send and receive phone calls through the DSCI network. Customers utilize a single SIP trunk to share capacity, provide cost savings, and build redundancy.



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