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Welcome to SIP Trunking brought to you by SUNCOM in partnership with AT&T. We’re excited you’ve made the decision to move your organization onto the forefront of technology by implementing SIP Trunking. SIP Trunking is a very powerful and feature-rich platform upon which your organization will conduct its daily business. We’ve developed this Administrator’s Guide to help you through some of the intricacies inherent with the service and hope you find it useful.
PURPOSE OF THIS GUIDE

The SIP Trunking Admin Guide describes the operational handling and support for SIP Trunking Services between the Department of Management Services (DMS) and its users. This guide is the summary of the operations tasks associated with the SIP Trunking services and operational procedures. This guide describes:

- Where to go to ask questions or report problems
- How the trouble reporting and resolution process works
- How to establish new or make changes to existing service
- How your service will be billed
WHAT IS SIP TRUNKING?

SIP Trunking is an integrated access, converged solution designed to deliver outbound, inbound, Local and Long Distance calling over the State’s MPLS data network, the MyFloridaNet (MFN). It is deployed in situations where customers own their own premises telephony (analog phones, key system, TDM PBX, or IP PBX) equipment. Hosted solutions such as AT&T’s Hosted Voice Solution (HVS) may also be used in conjunction with SIP Trunking. IP Flexible Reach provides “trunk service” over integrated access.

SUNCOM’s SIP Trunking solution provides Local, US Long Distance, International voice and fax calling, delivered via AT&T’s advanced VoIP infrastructure.

AT&T supports the integration of customer’s voice and data applications by leveraging the MyFloridaNet in conjunction with a variety of access choices.

AT&T’s SIP Trunking provides many benefits to a customer:

- Improved Total Cost of Ownership by allowing voice and data on a single access trunk
- Site-to-site on-net calling at no additional cost or any other inter-company location that is on AT&T’s VoIP infrastructure.
- Allows customers to add VoIP to their existing MFN provided data transport (may require an MFN CPE upgrade) resulting in more efficient use of your network bandwidth.
- Utilizing SIP enables enhanced and value added features associated with voice and data applications.
- Allows for routing of the VoIP calls efficiently and securely.
- It is very scalable with little upfront investment.
- Support different codec’s
- Provides an excellent option and cost benefit when designing a communications system for disaster recovery.
SIP Trunking is an integrated access, network service that provides the elements necessary to voice-enable a customer’s IP service. Local, US Long Distance and International calling are supported on the MyFloridaNet.

SIP Trunking supports voice traffic, originated from the customer’s voice system (e.g. IP-PBX) or the PSTN, that is converted to data packets, allowing customers to use their MFN connection for data, voice and fax traffic. Customers choose the calling capacity they require in units of Concurrent Calls, which are similar to simultaneous calls and can be engineered using standard voice traffic tools or by using the customer’s existing voice channel capacity, providing a flexible solution for any enterprise from large to small. SIP Trunking Service supports AT&T certified IP PBXs, IP PBX clusters and SBCs (Session Border Controllers). Traditional key systems and TDM PBXs are also supported but require an IP or SBC interface.

Outbound voice and fax calling is supported between:
- US VoIP-enabled locations (On-net)
- PSTN connected locations (Off-net)

Inbound service from the PSTN is also supported.

Definitions:
- **On-net** is defined as calling between customer locations that reside on the AT&T VoIP network.
- **Off-net** is defined as calling from SIP Trunking customer sites to any U.S. or non-U.S. location not equipped with the SIP Trunking. There are three categories of Off-net calling: Local, Long Distance and International.
  - **Local inbound and outbound** calling using AT&T’s Business VoIP Local Footprint with full local service feature/functionality including: new telephone number assignments, local number portability, Direct Inward Dialing (DID) and Direct Outward Dialing (DOD), E911, Directory Listing, Directory Assistance, Toll-Free Calling, Operator Assistance, Blocking options and more.
  - **Off-net Long Distance** calling via AT&T’s network-based hop-off gateways, which are connected to the PSTN for calling termination to any location.
  - **Off-net International** calling from AT&T’s network-based hop-off gateways provides International per minute calling at competitive rates.

**SUMMARY OF CALLING PLAN**

- Functionality: Standard features plus options
- On-Net: Unlimited
- Local Off-Net: Unlimited
- LD Bundle Off-Net: Flat rate plan with 300 included LD off-net minutes per concurrent call
- LD Unlimited Off-Net: Unlimited plan provides unlimited LD for off-net calls
- International Off-Net: Usage charges apply from beginning of call*

*U.S. and International Off-Net Calls are charged in one second increments with a minimum charge of 30 seconds per call.
HOW DOES IT WORK

AT&T has integrated its VoIP provider network with the MyFloridaNet. There are two geographically diverse IP interconnects between the two networks. This network integration allows SIP trunks to be delivered over the MFN reliably. This service is available to all MFN customers. The QoS features of the MFN allow your voice traffic to be treated with the correct priority while traversing the MFN. With these two interconnects, SIP trunks can be delivered to an IP-PBX or SBC. Multiple MFN circuits into a location can provide route diverse protection at the access layer. Additional SIP trunks can be provisioned to backup IP-PBXes/SBCs as well. It is strongly recommended by AT&T and DMS that customers with large implementations order multiple geo-diverse SIP trunks.

THE AT&T NETWORK

The AT&T Network consists of a core, underlying MPLS network which is responsible for transmitting data, and the AT&T VoIP Network Infrastructure (VNI) network that overlays the core network and provides voice support as well as connection to the PSTN.

The key highlights are:

- Open standards architecture leveraging SIP (Session Initiation Protocol)
- High levels of redundancy and resiliency
- Scalable
- Self-healing
- Security to preserve integrity, availability and confidentiality
- Flexibility to support emerging applications
AT&T VOIP INFRASTRUCTURE

AT&T IP Flexible Reach leverages AT&T’s VoIP Infrastructure, including the Border Elements (BE) connected to the AT&T Global MPLS Network. Elements of AT&T’s IP Voice Services are implemented within a trusted security domain referred to as the “VoIP Network Infrastructure” (VNI). Elements are interconnected within this domain via a secure MPLS network known as the Voice Aware Network (VAN) which is, in turn, built on AT&T’s Common Backbone Network (CBB).

AT&T’s VoIP architecture is based upon open standard protocols to facilitate a multi-vendor environment and to provide component interchangeability and interoperability. It is designed to support both existing and future services as they are developed and deployed.

Secure access to the trusted elements for session initiation and media transfer is provided through a Session Border Controller (SBCs) or IP Border Element (IPBE) which provides a number of security features including protocol filtering and deep packet inspection, call admission control and IP proxy functions.

The key elements include:
• Single IP/ MPLS network with Quality of Service and Class of Service
• Open standards architecture leveraging SIP (session initiation protocol)
• Border Elements that “translate” the multiple protocols into SIP
• Agnostic access supporting a variety of endpoints and PSTN connectivity
• Security to preserve integrity, availability and confidentiality
• Flexibility to support emerging applications

In keeping with the robust security employed throughout all of AT&T’s network infrastructures, AT&T has developed a VoIP security architecture that strengthens the service at potentially vulnerable points. This security architecture deploys a "defense-in-depth" approach to provide a multi-layered secure environment. Security mechanisms are deployed throughout the service in addition to the multi-layered security provided by the network itself, in order to provide seamless and effective security. AT&T leverages the unique capabilities of the AT&T Global MPLS Network and the MyFloridaNet to provide security and Quality of Service (QoS). AT&T has designed its network to be an effective security device to detect, isolate, and eliminate security threats before they become a security breach.

CONNECTIVITY TO CUSTOMER PREMISE EQUIPMENT

SIP Trunking is delivered to an MFN router at the customer premise. The router connects to the customer’s LAN to which the customer’s IP-PBX is also connected. When possible, it’s recommended that the IP-PBX be connected to directly to a dedicated interface on the MFN router. Depending on the configuration, a customer may choose to provide a Session Border Controller (SBC) with their IP-PBXs. AT&T’s demarc is the LAN side of the MFN router.
CALL FLOW EXAMPLE – INBOUND LOCAL

AT&T SIP Trunking for the State of Florida
Call Flow Example – Inbound Local

- User A dials 850-555-5678
- The PSTN routes the call to an AT&T local office
- The call is routed to the AT&T VoIP network
- The AT&T VoIP network routes the call to the SUNCOM IP Border Element (IPBE) integrated with the MyFloridaNet
- The SUNCOM IPBE routes the call over the MyFloridaNet to the customer’s IP-PBX and/or SBC.
- The customer’s PBX routes the call and rings the end user
- User B answers and completes the call setup
AT&T SIP Trunking for the State of Florida
Call Flow Example – OnNet to Domestic OffNet

- User B dials 407-555-1234
- The customer’s PBX routes the call to the sip trunk and towards the SUNCOM IP Border Element (IPBE)
- The SUNCOM IPBE routes the call towards the AT&T VoIP Network integrated with the MyFloridaNet
- The AT&T VoIP Network routes the call to the appropriate AT&T local office
- The AT&T local office routes the call out to the PSTN
- User A’s local provider will deliver the call and ring User A’s phone
- User A will answer and complete the call setup
CALL FLOW EXAMPLE – ON-NET TO ON-NET

AT&T SIP Trunking for the State of Florida
Call Flow Example – OnNet to OnNet

- User A dials User C at 904-555-1000
- The customer’s PBX routes the call to the sip trunk and towards the SUNCOM IP Border Element (IPBE)
- The SUNCOM IPBE routes the call from the agency on the left to the agency on the right
- The call never leaves the State of Florida’s network
QUALITY OF SERVICE (QOS) DESCRIPTION

Real-time applications, such as voice, are sensitive to long delays, delay variation (jitter) and packet loss in the network. AT&T has determined that the access link, the connection from the customer’s premises router to the AT&T network, is the most likely area for such congestion in a data network. Since alleviating potential congestion is critical to voice quality, AT&T employs Quality of Service (QoS) on the access link, as well as other methods, assigning a higher transport priority to voice packets over data packets.

QoS ensures voice packets get first priority within the network. Specifically, it is an algorithm based on “low latency queuing”, or “Priority Queuing with Class Based Weighted Fair Queuing (PQ-CBWFQ)”. Essentially, this method uses software to identify voice packets and give those packets transmission priority over data packets. Assigning a higher priority to voice packets ensures that they are carried end-to-end, virtually eliminating transmission delay. Data transmission interference is insignificant since a millisecond delay in those packets is far less noticeable than voice packets containing conversation.

QoS must be provisioned on the MFN access circuit used to deliver the SIP trunk to the customer SBC or IP PBX. The amount of bandwidth reserved for voice is dependent upon the number of concurrent call paths and the type of codec.
Customers must configure their IP PBX to send the dial patterns as described in the following table.

<table>
<thead>
<tr>
<th>Call Type</th>
<th>Digits to Send</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local</td>
<td>NPAXXXXXXX (Area code + 7 digits)</td>
</tr>
<tr>
<td>Long Distance</td>
<td>NPAXXXXXXX (Area code + 7 digits)</td>
</tr>
<tr>
<td>International</td>
<td>011+CC+NN (011 + Country code + number)</td>
</tr>
<tr>
<td>211/311/511/711</td>
<td>N11 (where N is 2, 3, 5, or 7)</td>
</tr>
<tr>
<td>911</td>
<td>911</td>
</tr>
<tr>
<td>Toll-Free</td>
<td>8YYXXXXXX</td>
</tr>
<tr>
<td>Operator Assistance</td>
<td>0</td>
</tr>
<tr>
<td>Directory Assistance</td>
<td>411</td>
</tr>
</tbody>
</table>

The customer is required to configure their CPE at each VoIP site to transmit a Calling Party Number (CPN) for all outbound calls placed over their SIP trunk. The CPN must be one of the valid AT&T provided telephone numbers associated with the physical location of the VoIP site. This requirement will provide AT&T with the ability to provide proper discernment, taxing, and settlements for calls originating from these locations. If the customer sends a CPN that is not valid or has not been provisioned for the SIP Trunk, then AT&T will insert the main billing number for the customer’s SIP Trunk into the diversion header of the SIP Invite. This main billing number will appear in the call detail records for all outbound local and long distance calls where an invalid CPN is sent.
STANDARD FEATURES

FEATURES AND CAPABILITIES

SIP Trunking provides a robust set of local features supporting inbound calling from PSTN callers as well as outbound calling to local numbers.

CALLING NAME (CNAM AKA CALLER NAME)

Calling Name service retrieves stored names associated with a telephone number. This name is what will be displayed to called parties when the customer makes an outbound call. The customer can choose what they would like their CNAM to be.

DEFAULT CALL BLOCKING:

The following dialed numbers will be blocked by default for all SIP Trunking customers:

- 611
- 811
- 500
- 700
- NPA 555-XXXX (except NPA 555-1212)
- 900/NPA 976-XXXX
- 08YY XXX-XXXX
- 0N11
- 0NPA 555-1212
- 0976 NXX-XXXX
- 0500 XXX-XXXX
- 0700 XXX-XXXX
- 0900 XXX-XXXX
- All 1010 dialing
911/E911

SIP Trunking service provides 911/E911 calling capability, where the address for the SIP Trunking site is within the area where AT&T has the ability to provide E911 service. E911 calls may be routed to the (Public Safety Answering Point) PSAP entirely over AT&T operated network facilities or may utilize network facilities of other service providers. All E911 service is based on the SIP Trunking Site Registered Location information provided to AT&T/SUNCOM by the customer. Customers will need to specify an address per telephone number to be used for 911 purposes.

Prior to the activation of service at a SIP Trunking site, the customer must provide to AT&T/SUNCOM the correct name and address information (registered location information) for the SIP Trunking site and users, where appropriate. The customer must notify AT&T of any changes or updates to this information.

When a 911 or E911 call is made, AT&T will (where technically feasible) provide the name and address information given to AT&T by the customer to the appropriate PSAP. If AT&T is not given correct information to properly identify the actual location of the SIP Trunking site, 911 and E911 communications may be misdirected to an incorrect PSAP.

LOCAL NUMBER PORTABILITY (LNP)

Customers are able to port their existing TNs to SIP Trunking. Additional information about porting telephone numbers is provided in Appendix B. Please contact your AT&T and/or SUNCOM representative for assistance.

N11 (211, 311, 511, 711)

If the N11 number is supported in the particular local area in which the customer is located, the call will complete via the local network switch.

NEW TNS (TELEPHONE NUMBERS)

Customer may request new telephone numbers to be assigned by AT&T and DMS. A CSA is required to initiate the process. Every effort will be made to accommodate requests for numbers to fit specifically within your dial plan but no guarantees can be made.

DIRECTORY ASSISTANCE (DA) INCLUDING NPA 555-1212 CALLING

Customers can access Directory Assistance by dialing 411, NPA-555-1212 or “00”. All DA calls are handled via the local network switch that is local to the customer site.
OPERATOR SERVICES

SIP Trunking customers can access operator services by dialing “0” or by dialing “00” and selecting the appropriate prompt. All operator services calls are handled via the local network switch that is local to the customer site.

TOLL-FREE DIALING

All customer dialed toll-free numbers are routed to the local network switch serving that customer location for handling.

TOLL FREE SWITCHED TERMINATION ON SIP TRUNKING TELEPHONE NUMBERS

Toll free termination on SIP Trunking telephone numbers is permitted.

- Notes:
  - Dial-around and CALEA are not supported at this time.
  - Privacy suppression of Calling Party Number is supported.
  - SIP Trunking does support GETS/SRAS calling for authorized users. GETS calls can be initiated by dialing 1-710-627-4387, while SRAS calls are placed by dialing 1-710-NXX-XXXX.

DIRECTORY ASSISTANCE & DIRECTORY LISTINGS

The customer places a directory listing request with AT&T. Part of the directory listing order includes the directory assistance (DA) request. When the customer requests a listing in DA, AT&T sends the listing information to the Local Service Provider (LSP) DA. The LSP DA inputs the listing information in the local DA database (411). Depending on each LSP, the listing information is available in the local DA within two to seven days. The LSP DA also feeds all DA listings information to the national DA (555-1212). The delay from the LSP DA to the national DA is from one to eight weeks depending on the LSP. After the order implementation, the AT&T Provisioning Center calls the LSP(s) to verify that the listing information is in LSP DA.

When porting numbers to SIP Trunking it is recommended you specify that all existing directory listing information remains the same and is moved over with the numbers. In an effort to keep the time to port numbers to a minimum we strongly recommend you don’t make any changes to your directory listing information during a port.

With regard to numbers already active on your SIP Trunk, you may submit a CSA to add listings into the directory and/or modify existing listings.
TELEPHONE NUMBERS (TNS)

PORTING OF TELEPHONE NUMBERS

Customer may port telephone numbers to AT&T from another carrier and may request another carrier to port a telephone number from AT&T. There are mandated rules and regulations regarding the porting of numbers to and from AT&T. Talk to your AT&T/SUNCOM representative if you have any questions regarding the porting of a telephone number.

The telephone numbers to be ported must be from a local calling area in which AT&T offers SIP Trunking. Also, the location at which customer wishes to use the ported numbers must be within the area in which the SIP Trunking is offered.

There is no separate charge for the porting of telephone numbers to AT&T. Customer must advise AT&T of its request to port existing telephone numbers to AT&T, and AT&T will process the request as required with the other carrier. The timeframe to port numbers from another carrier will vary based on the number of TNs being ported. The telephone number porting process is documented in Appendix B.

Customers may also port out to another carrier any existing TNs currently in use on their SIP Trunk. The new carrier must interconnect and receive ported telephone numbers in the same local calling area with which the ported out telephone numbers are normally associated. They must then contact their new carrier and request that their AT&T telephone number be ported over.
IP TELEPHONY INTEROPERABILITY

SCOPE

AT&T’s SIP Trunking service is delivered to SUNCOM customers using Acme Packet (Oracle) Session Border Controllers (SBCs). These SBCs are integrated directly with the State’s MyFloridaNet and FIRN services. AT&T and DMS have successfully connected with many different types of customer phone systems. However, it is strongly recommended you check with your AT&T/SUNCOM representative before ordering service to see if there are any known interoperability issues.
**ENHANCED FEATURES**

In September, 2017 we made new features available to customers. Please keep in mind these features are implemented in AT&T’s commercial VoIP network (not in your phone system). These features can be easily configured for use by contacting the helpdesk.

**CALL FORWARDING**

There are five types of call forwarding you have access to:

1. CFA – Always - Enables a user to redirect all incoming calls to another phone number.
2. CFB – Busy - Enables a user to redirect calls to another destination when an incoming call encounters a busy condition.
3. No Answer - Enables a user to redirect calls to another destination when an incoming call is not answered within a specified number of rings.
4. Not Reachable - Allows for configuring a destination (for example, a mobile) where a call should be redirected when the IP Flexible Reach TN is unreachable.
5. Selective - Enables a user to define criteria that causes certain incoming calls to be redirected to another destination.

**SEQUENTIAL RING**

Enables users to define a “find-me” list of up to 5 phone numbers that are alerted sequentially for incoming calls that match specified criteria.

**SIMULTANEOUS RINGING**

Enables users to have multiple phones ring simultaneously when any calls are received on their SIP DID (up to 10 Telephone Numbers).

**MAX DID POLICING**

Allows the customer to define the maximum number of concurrent calls per DID. This includes inbound, outbound and redirected calls.

**SELECTIVE CALL ACCEPTANCE/REJECTION**

Enables a user to define criteria that causes certain incoming calls to be allowed or rejected.

**BLIND TRANSFER (SIP REFER)**

Provides the ability to transfer an active call to another specific destination (Target Party) without consulting with the destination party.

**CONSULTATIVE TRANSFER (SIP REFER)**
Provides the ability to transfer an active call to another specific destination (Target Party) with consulting with the destination party.

**CONFIGURABLE CALLING LINE ID**

This feature allows a customer to configure what is displayed for Calling Line ID for outbound calls, per Telephone Number. The Customized Calling Line ID number must represent the originating customer’s identity in the form of a 10 digit Telephone Number or 8YY number. Configurable Calling Line ID does not apply to specialty calls (e.g. N11, Operator, Directory Assistance, NPA-555-1212).

**INTERCEPT ANNOUNCEMENTS**

Enables a customer to intercept calls routed to a non-working internal line with informative announcements (default or custom) and alternate routing options.

**IP FLEXIBLE REACH MOBILE CLIENT**

A mobile application for IOS and Android. This app, when enabled, will allow users to make and receive calls over their existing cellular service using their IP Flexible Reach business identity. It will also allow the user to change call forwarding features, simultaneous ring and sequential ring from their mobile device.

Once the helpdesk gives a user access to the feature, the user can download the IP Flex app from the app store for their operating system.
BILLING

RATES

CALL PATHS

Call paths are billed per call path per month. The current pricing is available on the DMS website.

DID / TELEPHONE NUMBERS

- There is no charge for DIDs on this service.
- This applies to both new and port-in DIDs.
- There is no charge to port numbers to your SIP Trunk.

LONG DISTANCE AND OVERAGE

Each call path comes with 300 minutes of bundled US Domestic long distance. The 300 minutes are pooled across all call paths. Long distance usage above the pool of minutes is billed per minute at the current rates posted on the DMS website.

UNLIMITED LONG DISTANCE AS AN OPTION

As an alternative to the standard rate plan, we also offer a version of SIP Trunking that comes with an unlimited number of long distance minutes. The monthly rate per call path is higher but there is no charge for going over the allotted 300 minutes of long distance per month. The process to order this version of the SIP Trunk is the same as with the standard SIP Trunk. However, in the text of the order you’ll need to answer yes to the question about Unlimited LD.

DIRECTORY ASSISTANCE AND OPERATOR SERVICES

- Local directory assistance (US Mainland per call) - $1.25
- Operator Assisted services (US Mainland per call)
  - Station-to-Station $1.75
  - Person-to-Person $3.50
  - Third Number Billing $1.80
  - Busy Line Verification $2.00
  - Busy Line Interrupt $2.15
  - Collect $3.50

INVOICING

SUNCOM will issue a monthly invoice for the SIP trunking service. Customers will be provided a summary invoice which provides the appropriate charges for concurrent call paths, applicable FCC regulatory charges, and any long distance overage charges. The invoice is available for download in the CSAB Invoice Explorer module. An example
invoice is provided below. If you have any questions about your invoice, billing contacts are provided in the Contacts section of this guide.

CALL DETAIL

SUNCOM will provide call detail records for all off-net outbound local and long distance calls placed using the service. The call detail records are available for download from CSAB and will be available on or before the 20th day of the following month.
ORDERING

The SUNCOM CSA system will be used for all ordering activities. The lifecycle ordering process can be broken down into several different order types. All provisioning activity will fall into one of these categories detailed below. For each order type you’ll see the CSA verbiage required in block nine when placing this type of order.

Please keep in mind that all SIP Trunks must be delivered across an MFN circuit. You may use an existing MFN circuit or order a new one. The standard intervals for MFN service can be found here.

OVERVIEW

1. Customer submits SUNCOM CSA to the State of Florida, DMS.
2. DMS approves order and submits to AT&T.
3. AT&T reviews the order for accuracy and contacts the customer to confirm the order and acknowledge receipt.
4. AT&T works with the customer to establish and schedule a test and turn-up date.
5. AT&T and customer work together to test and turn-up service.
6. AT&T and customer work together to verify service is operational using the test plan in Appendix C.
7. Order is closed out.

ORDER TYPES (CSA TEMPLATES)

NEW TRUNK
PLEASE ORDER NEW SIP TRUNK AT THE LOCATION IN BLOCK (5) ABOVE.

NUMBER OF CONCURRENT CALL PATHS:
CODEC: G711 or G729
NUMBER OF NEW DIDS*:
SITE NAME (AS CUSTOMER REFERS TO LOCATION):
OUTBOUND CALLING NAME (15 CHARACTERS):
HAS THIS LOCATION BEEN CONFIRMED TO BE WITHIN THE AT&T BVOIP FOOTPRINT? Y/N:
CPE USED TO TERMINATE SIP TRUNK
   SBC MAKE/MODEL:
   SBC SW VERSION:
   IP-PBX MAKE/MODEL:
   IP-PBX SW VERSION:
MFN SERVICE AT THIS LOCATION. CSA NO:
MFN SERVICE AT THIS LOCATION. ROUTER HOSTNAME:
VRF: PUBLIC/COMMON/FIRN/PRIVATE:
IS THE MFN CIRCUIT CONFIGURED FOR VOICE QOS? Y/N
QOS CSA NUMBER:

ROUTER CONFIG OPTION (CHOOSE ONE)
ROUTED (NEXT HOP IP ADDRESS):
   DOT1Q (VLAN ID):
SECONDARY (INTERFACE NUMBER):
DEDICATED (INTERFACE NUMBER):

SECONDARY ENDPOINT NEEDED? Y/N
TERTIARY ENDPOINT NEEDED? Y/N

MAIN CUSTOMER CONTACT
CONTACT NAME:
CONTACT PHONE:
CONTACT EMAIL:

CUSTOMER CONTACT FOR TEST AND TURN-UP
CONTACT NAME:
CONTACT PHONE:
CONTACT EMAIL:

AT&T/DMS PRE-SALE ENGINEERS ENGAGED ON THIS PROJECT:

*MUST OBTAIN NEW DID’S TO ESTABLISH SERVICE. AFTER SERVICE IS ESTABLISHED, ISSUE A CHANGE CSA TO PORT IN EXISTING NUMBERS.

*WHEN PORTING NUMBERS TO YOUR SIP TRUNK ALL DIRECTORY LISTING INFORMATION WILL BE MOVED OVER AS IS.

SIP Trunking DID Details 20141216.xls
CHANGE TRUNK
AT BLOCK FIVE ABOVE, PLEASE MODIFY SIP TRUNK AS DESCRIBED BELOW.

INSTALLING CSA NO. FOR EXISTING SIP SERVICE:
CUSTOMER IP ENDPOINT:
CHANGE IN NUMBER OF CONCURRENT CALL PATHS: Y/N:
   NUMBER OF CALL PATHS CURRENTLY WORKING:
   TOTAL NUMBER OF DESIRED CALL PATHS:

REQUEST FOR NEW DID’S: Y/N
   NUMBER OF NEW DID’S:

REQUEST TO PORT IN DID’S: Y/N
   LIST OF DID’S TO PORT:

CHANGE 911 ADDRESS FOR EXISTING DID’S: Y/N
   LIST OF DID’S TO CHANGE:
   NEW 911 ADDRESS:

ADD/CHANGE DIRECTORY LISTINGS: Y/N
   PLEASE USE ATTACHED SPREADSHEET

COMMENTS:

CUSTOMER CONTACT
FULL NAME:
PHONE:
EMAIL:

SIP Trunking DID
Details.xlsx
DISCONNECT TRUNK
AT BLOCK FIVE ABOVE, PLEASE DISCONNECT THE SIP TRUNK AS DESCRIBED BELOW

INSTALLING SIP CSA NO.:
SIP TRUNK CIRCUIT ID:
TROUBLE REPORTING

All service issues are to be reported to the State of Florida SIP Trunking Support Center using the below information.

- Email: support@hcs.net
- Phone: 850-297-0551 x454 or 800-825-9390 x454
- Please be prepared with either of these when calling in:
  - MFN-facing IP Address of your SBC/IP-PBX available
  - Telephone number
- Escalations will be handled by the SUNCOM NOC at 888-4SUNCOM or suncom.helpdesk@dms.myflorida.com
SERVICE AVAILABILITY

AT&T SIP Trunks require connectivity to the MyFloridaNet (MFN). The MFN is a statewide network which allows you to terminate your sip trunk pretty much anywhere MFN can go.

However, SIP Trunking TNs (whether new and assigned by AT&T or ported in from another carrier) can only be offered within the AT&T BVOIP footprint. While this footprint covers most of Florida it does not cover the entire state.

A formal service availability check can be performed by your AT&T account team and will determine whether or not a given location is within our footprint.
### POINTS OF CONTACT

#### BILLING

<table>
<thead>
<tr>
<th>Name</th>
<th>Phone Number</th>
<th>Email Address</th>
<th>Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Betsy Wonsch</td>
<td>(850) 922-7526</td>
<td><a href="mailto:betsy.wonsch@dms.myflorida.com">betsy.wonsch@dms.myflorida.com</a></td>
<td>OMC Manager</td>
</tr>
<tr>
<td>Jessie Austin</td>
<td>(850) 922-7517</td>
<td><a href="mailto:jessie.austin@dms.myflorida.com">jessie.austin@dms.myflorida.com</a></td>
<td>Supervisor Local Service Invoicing</td>
</tr>
</tbody>
</table>

#### ORDER MANAGEMENT AND PRICING

<table>
<thead>
<tr>
<th>Name</th>
<th>Phone Number</th>
<th>Email Address</th>
<th>Regions Supported</th>
</tr>
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<tbody>
<tr>
<td>Mohammad Amirzadeh, Supervisor,</td>
<td>850-922-7476</td>
<td><a href="mailto:mohammad.amirzadeh@dms.myflorida.com">mohammad.amirzadeh@dms.myflorida.com</a></td>
<td>Statewide</td>
</tr>
<tr>
<td>Service Delivery</td>
<td></td>
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</tr>
<tr>
<td>Denise Adkins, Supervisor, DMS</td>
<td>850-410-0014</td>
<td><a href="mailto:denise.adkins@dms.myflorida.com">denise.adkins@dms.myflorida.com</a></td>
<td>Statewide</td>
</tr>
<tr>
<td>Business Consultants</td>
<td></td>
<td></td>
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</tr>
<tr>
<td>Marvin Powell, Business Consultants</td>
<td>(850)413-7906</td>
<td><a href="mailto:marvin.powell@dms.myflorida.com">marvin.powell@dms.myflorida.com</a></td>
<td>Alachua, Bay, Baker, Bradford, Calhoun, Clay, Columbia, Dixie, Duval, Escambia,</td>
</tr>
<tr>
<td></td>
<td>or 1-888-4SUNCOM (1-888-478-6266)</td>
<td></td>
<td>Franklin, Gadsden, Gilchrist, Holmes, Gulf, Hamilton, Jackson, Lafayette, Liberty,</td>
</tr>
<tr>
<td></td>
<td>Option 4</td>
<td></td>
<td>Madison, Nassau, Okaloosa, Putnam, Santa Rosa, St. Johns, Suwanee, Walton,</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Washington, Jefferson, Leon, Taylor, Wakulla, Union</td>
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### GENERAL INQUIRIES & FEATURE MANAGEMENT

The following DMS staff may be contacted with feedback, escalations, or other inquiries such as:

- SIP Trunking product information
- Engineering and design questions

Request for additional feature(s) and functionality

- Feedback about the service
- SIP Trunking future direction
- Request for exceptions to the guideline therein
- Question about this document

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<thead>
<tr>
<th>Name</th>
<th>Phone Number</th>
<th>Email Address</th>
<th>Title</th>
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</thead>
<tbody>
<tr>
<td>John Starling</td>
<td>(850) 413-0816</td>
<td><a href="mailto:john.starling@dms.myflorida.com">john.starling@dms.myflorida.com</a></td>
<td>Communications Engineer / Product Manager</td>
</tr>
<tr>
<td>Chuck Hartsfield</td>
<td>(850) 413-9535</td>
<td><a href="mailto:charles.hartsfield@dms.myflorida.com">charles.hartsfield@dms.myflorida.com</a></td>
<td>Supervisor, Engineering &amp; Design</td>
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<tr>
<td>Kevin Langston</td>
<td>(850) 922-7477</td>
<td><a href="mailto:kevin.langston@dms.myflorida.com">kevin.langston@dms.myflorida.com</a></td>
<td>Bureau Chief of Customer Service</td>
</tr>
</tbody>
</table>
APPENDIX A: DEFINITIONS

- GETS (Government Emergency Telecommunications Service) GETS is used by government and other designated personnel during crises or emergencies to provide priority processing for local and long distance calls on the public switched telephone network.

- SRAS (Special Routing Arrangement Service) SRAS provides designated government users with a high probability of completion service, and is dialed using 710-NXX-XXXX (except for 710-627-4387 which is the GETS access number).
APPENDIX B: PORT-IN PROCESS

You may port numbers in from your existing provider to your AT&T SIP Trunk. To begin this process you first need to compile a clean, complete, and accurate list of the numbers to be ported. Due to the amount of coordination and timing intervals involved in the porting process it is very important this list be complete and concise. One incorrect number can jeopardize the scheduling of the entire port. You’ll need to submit a change CSA using the template provided in the ordering section. Once your CSA is submitted to DMS here is an overview of the porting process.

Once an order to port has been submitted, NO CHANGES may be made to that order.

Please submit CSAs to DMS for porting at least eight weeks in advanced of the desired port date.

1. The customer will submit a CSA with the list of TNs to be ported.
2. DMS reviews CSA and releases to AT&T.
3. CSA is received by AT&T.
4. AT&T works to obtain the customer’s current customer service records (CSRs). Depending on the current provider AT&T may need to reach out to the customer and/or their current account team for assistance.
5. AT&T submits the CSR(s) and supporting documentation to an LSR agent.
6. The LSR agent places the formal porting request with the customer’s current provider. The request is considered to be in the “FOC Pending” state at this point.
7. The current provider will validate the request and provide a firm date/time for porting. This state is known as “FOC Confirmed”.
8. If the current provider sees any issues with the request (pending orders against the existing service or inaccurate information) the request will be rejected. If this happens, AT&T will work with the customer to remedy the issue with their current service and then the porting request will be resent to the current provider. This process will repeat until “FOC Confirmed” is received from the current provider. Porting dates are never firm until “FOC Confirmed” is received from the current provider.
APPENDIX C: TEST AND TURN-UP PROCESS

SoF ATT SIP
Trunking IMPLEMENTATION