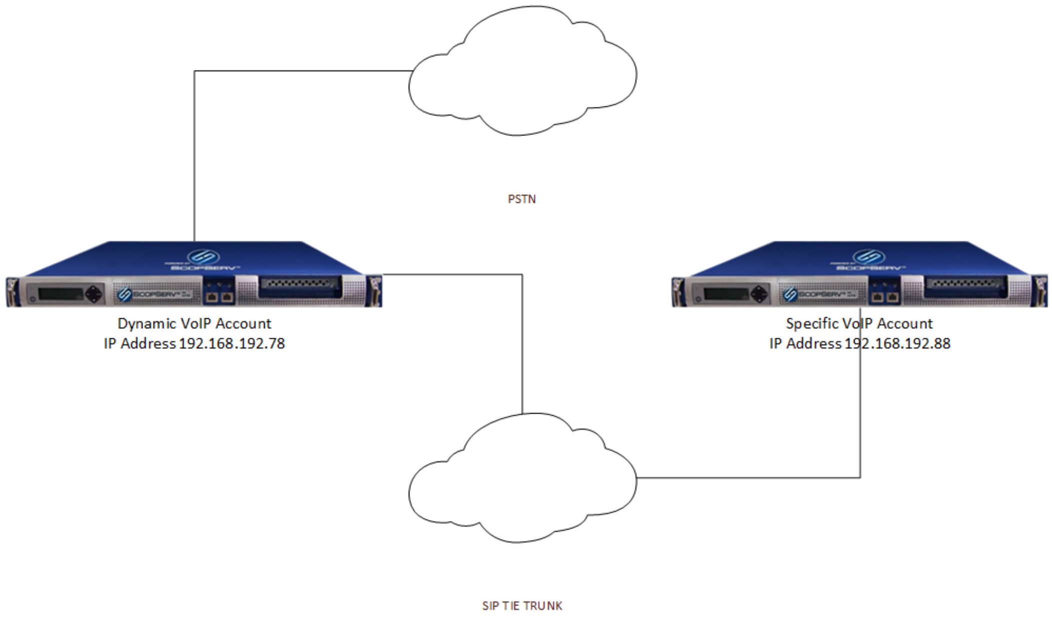
ScopTEL™ IP PBX Software

Back to Back SIP Trunking Configuration

Usage Cases Implementing DNIS:

- SIP TIE trunks:
  - A private network is created to dial extensions between systems using Access Codes
- Tandem Dialing:
  - PSTN resources available to a remote server are shared to the local server
- PSTN re-routing to a networked ScopTEL Server



DNIS (Dialed Number Information Service)

- DNIS is the routing number the PRI CPE (Customer Provided Equipment) circuit receives from the carrier or remote server (also known as Received Digits)
- The received digits length (also known as Digits to Out Pulse by some Telco's) can vary in length typically from 3 to 10 digits in length.
- DNIS digits, as received from the carrier, are used to route calls via Incoming Lines objects in the ScopTEL IP PBX
- **Example 1**:
  - The customer's BTN is 555-555-1234
  - The Received Digit length set by the Telco is 4
  - The resulting DNIS or received digits is equal to 1234
- **Toll Free Services**:
  - The customer has a published toll free number 1-800-555-2234
  - Toll free numbers **must** be associated with a local phone number which is referred to as the **conversion number**
  - The conversion number can be any DNIS number associated with the PRI circuit
  - In this example the conversion number for 1-800-555-2234 is the BTN number 555-555-1234
  - Therefore the DNIS number for 1-800-555-2234 is 1234

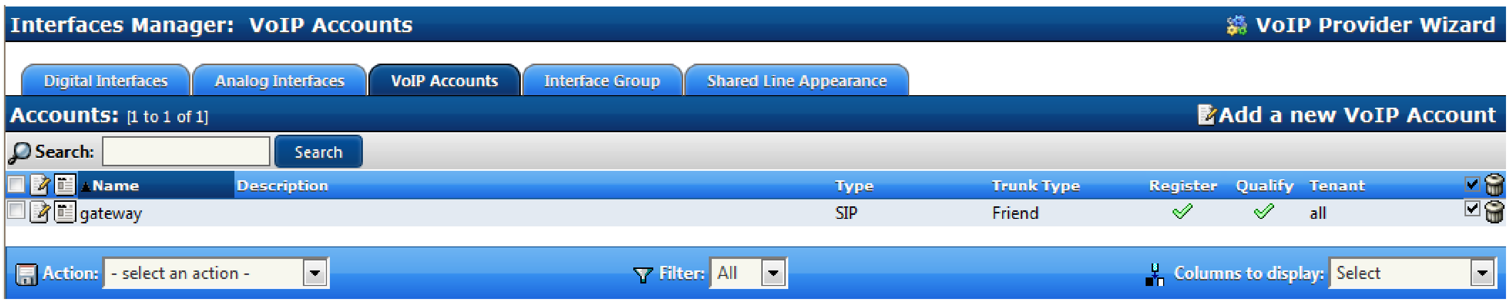
ScopTEL Background and Pre-requisites

- **SIP VoIP Accounts must be created on ScopTEL server for SIP registration**
- **Incoming Lines must be created on ScopTEL server to route incoming calls**
- **Outgoing Lines must be created on ScopTEL server to route outgoing calls**
- **Class of Service must be edited to allow Outgoing Lines per extension**

VoIP Accounts

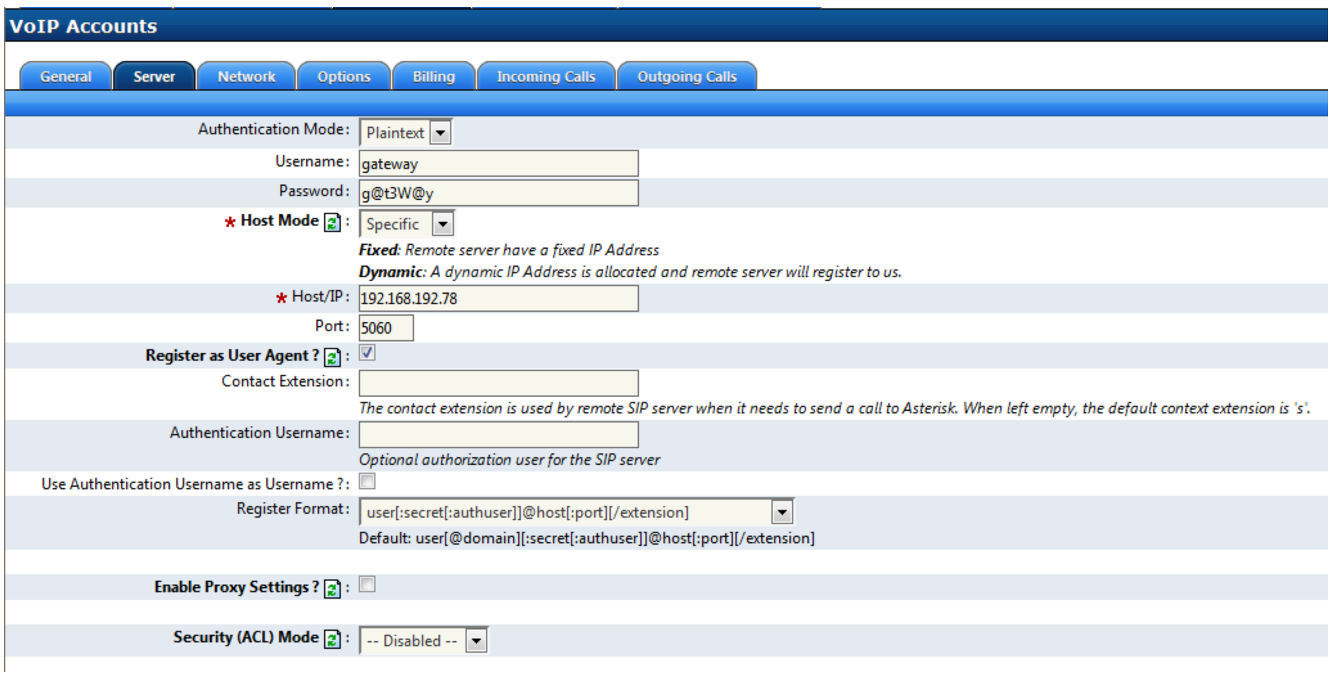
Use the ScopTEL Interfaces Manager to build the shared SIP trunk

- **In this example the VoIP Account will be called ‘gateway’**
- **You must build the VoIP Account on both servers**



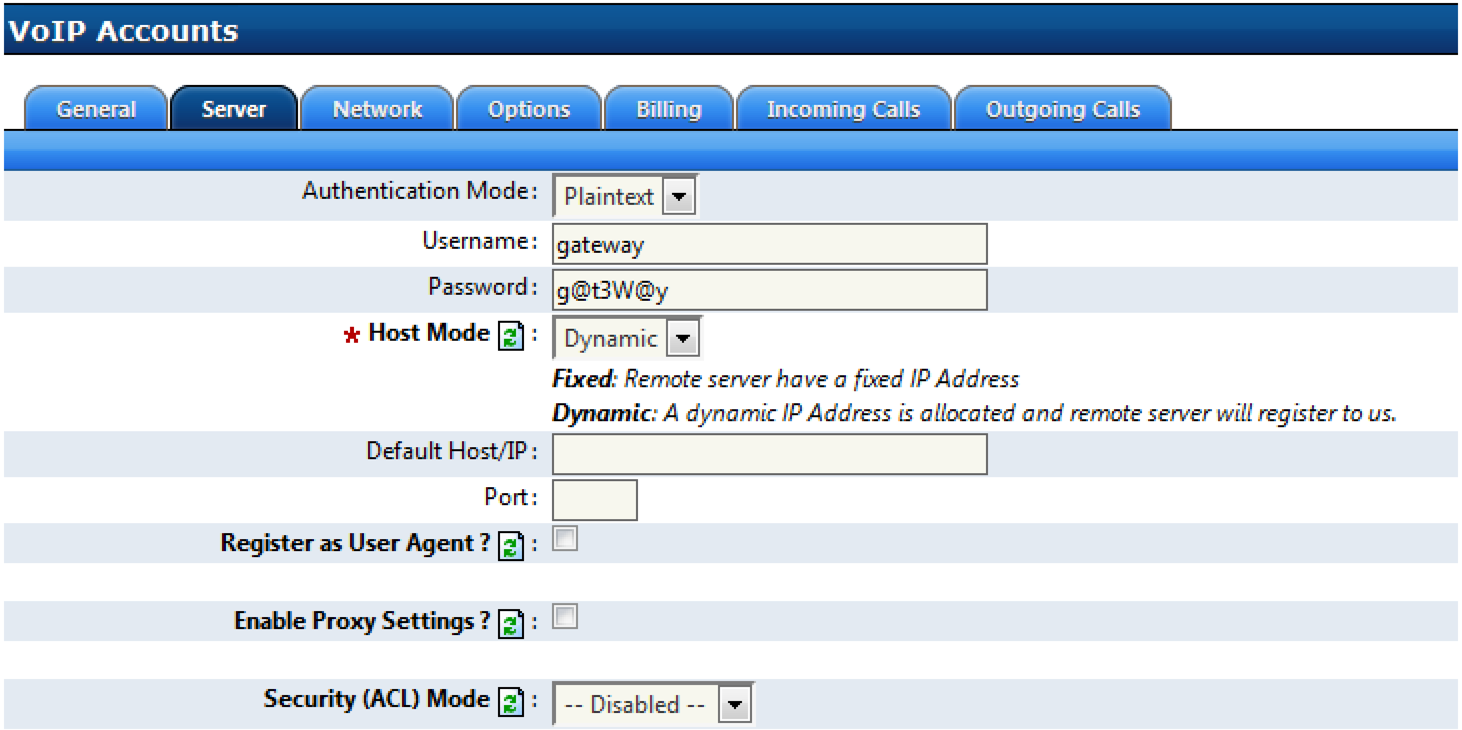
Specific VoIP Account

- **The Account name and Username must match**
- **Your password must be secure**
- **Hostmode is specific so this server will register to the dynamic VoIP Account in order to access remote resources**
- **The Register Format must be '**user[:secret[:authuser]]@host[:port][/extension]'



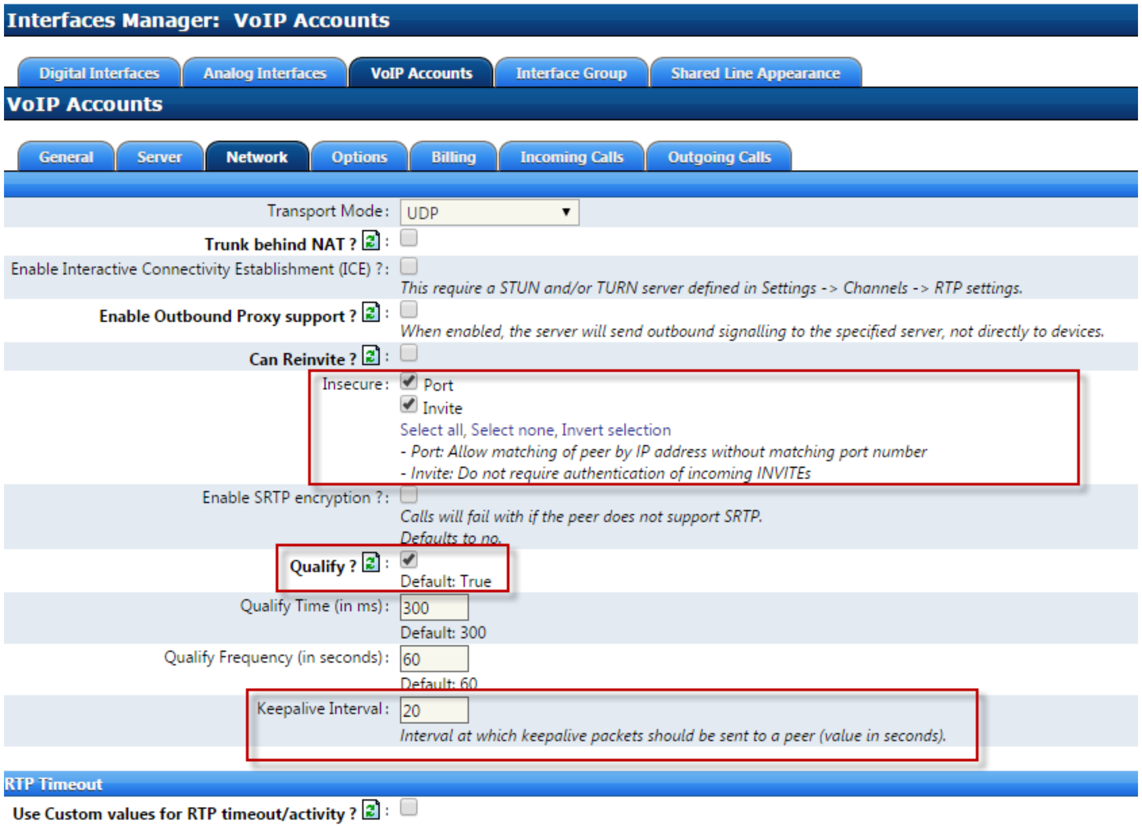
Dynamic VoIP Account

- **The Account name and Username must match**
- **Your password must be secure**
- **Hostmode is dynamic so this server can authenticate and share resources with the remote server**
- **The Register Format must be '**user[:secret[:authuser]]@host[:port][/extension]'



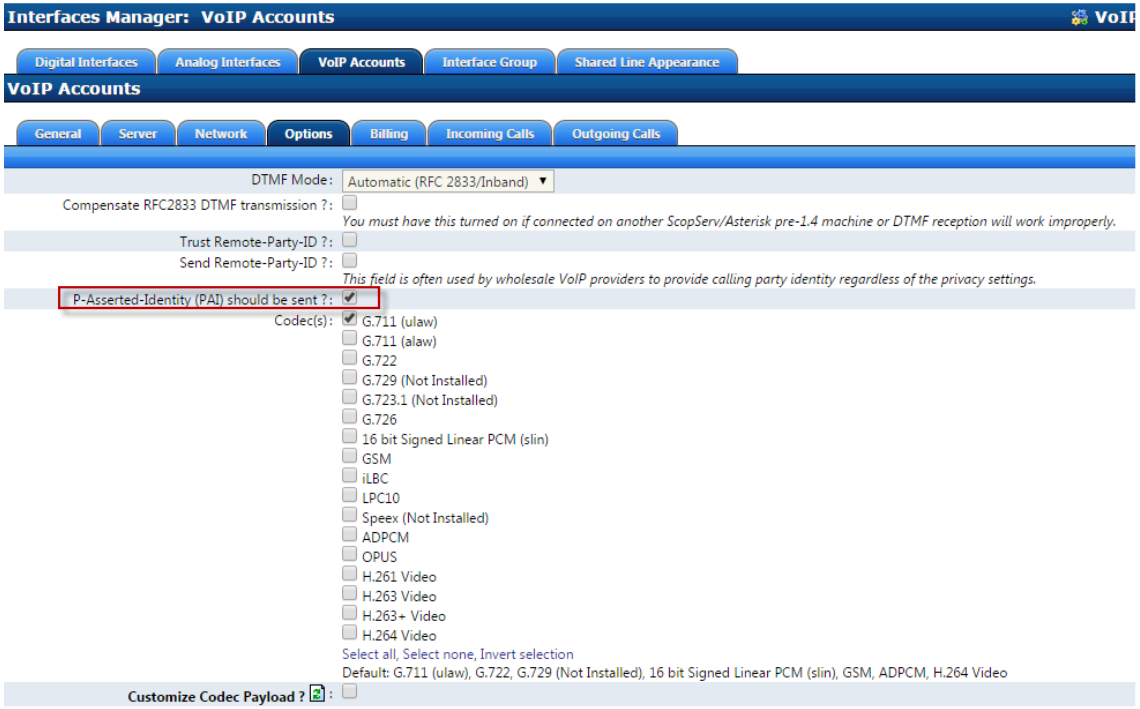
Shared Settings

- **Insecure should include both Port and Invite**
- **Qualify should be enabled**
- **Keepalive Interval should be set to a value short enough to address UDP timeouts in some firewalls**



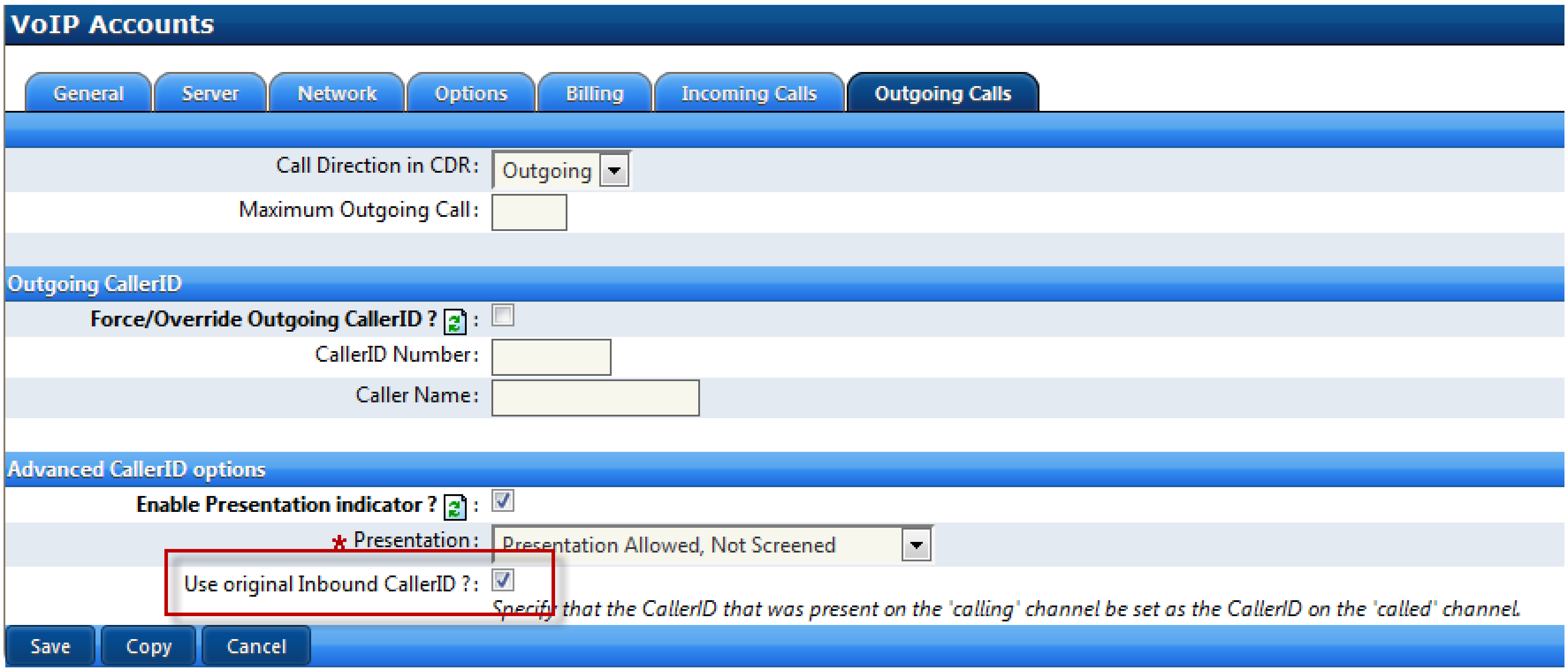
Shared Settings

- DTMF Mode RFC 2833 is recommended
- P-Asserted-Identity is recommended
- Only select CODEC's which are compatible with your PSTN interface and Region
- Careful CODEC selection is required to avoid transcoding
- The G.729 CODEC cannot support FAX transmissions



Shared Settings

- **Use original Inbound CallerID?: must be enabled to pass the original CallerID to the forwarded VoIP Account.**



Dial Plan Strings

Custom Dial Plan Strings

X	matches any digit from 0-9
Z	matches any digit form 1-9
N	matches any digit from 2-9
[1237-9]	matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
.	wildcard, matches one or more characters
!	wildcard, matches zero or more characters immediately

Examples

NXXXXXX	matches a normal 7 digit telephone number
1NXXNXXXXXX matches an area	code and phone number preceded by a one
9011.	matches any string of at least five characters that starts with 9011,
but it does not match the four-	character string 9011 itself.
#	matches a single # keypress

Specific Server Outgoing Lines

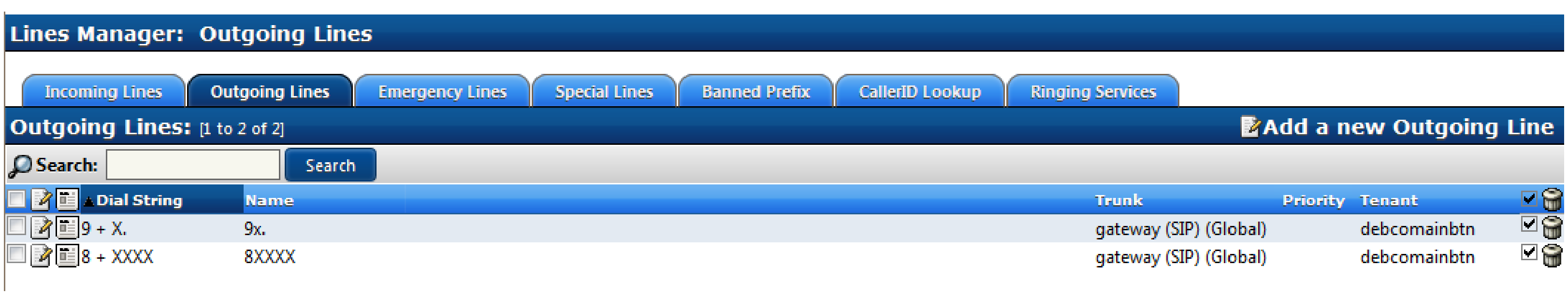
Outgoing Lines use pattern matching to steer the dialed number to the remote server:

9x. is the PSTN access code and is reserved exclusively for PSTN access

- **No internal extensions or applications on this server can start with 9**
- **The 9 will not be stripped from the Outgoing Line after the Class of Service lookup since the 9 must be passed to the dynamic VoIP Interface**

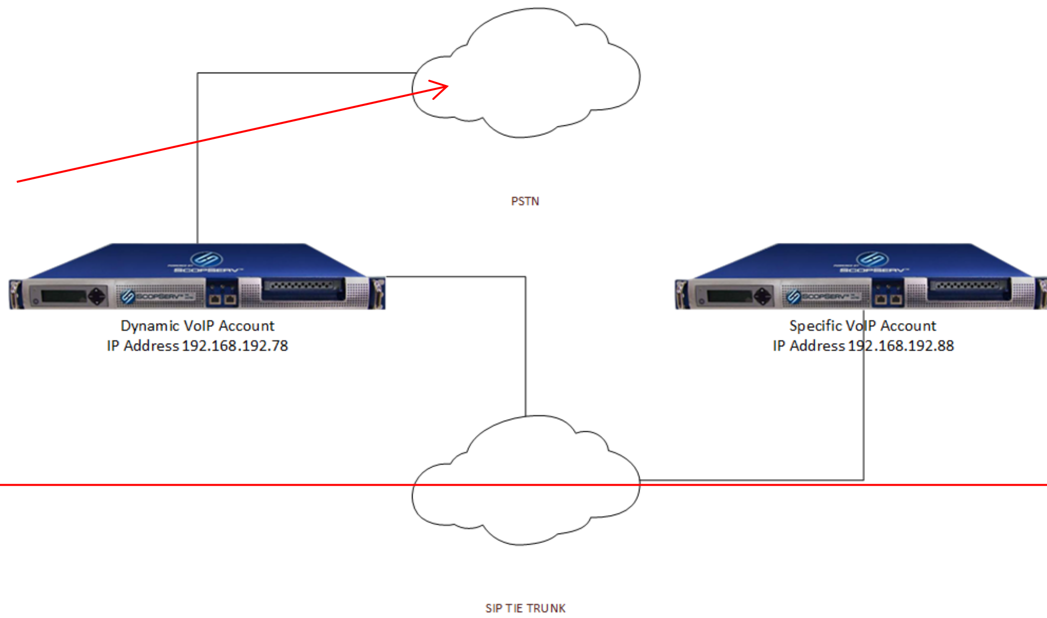
8XXXX is the SIP TIE Trunk Access code

- **8XXXX will be used so that 8 will steer the dialed number to the remote server**
- **No internal extensions or applications on this server can start with 8**
- **The 8 will be stripped from the Outgoing Line after the Class of Service lookup therefore only digits XXXX will be passed to the remote server**



Tandem Dial PSTN Trunking

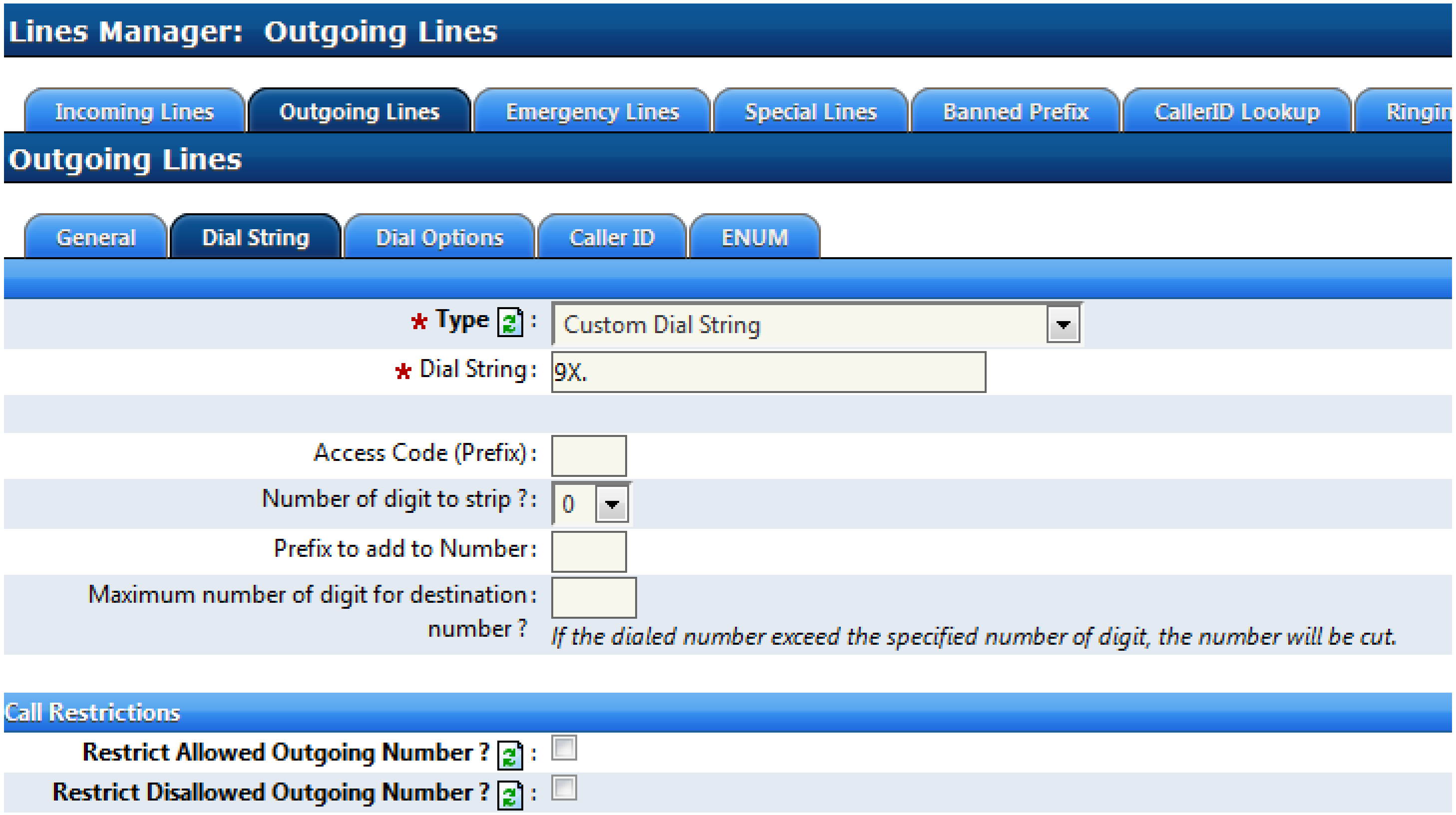
2. The VoIP Interface receives DNIS 94165551234 from the SIP TIE Trunk, strips the 9 and bridges 4165551234 to an Outgoing PSTN channel using the PSTN VoIP Account



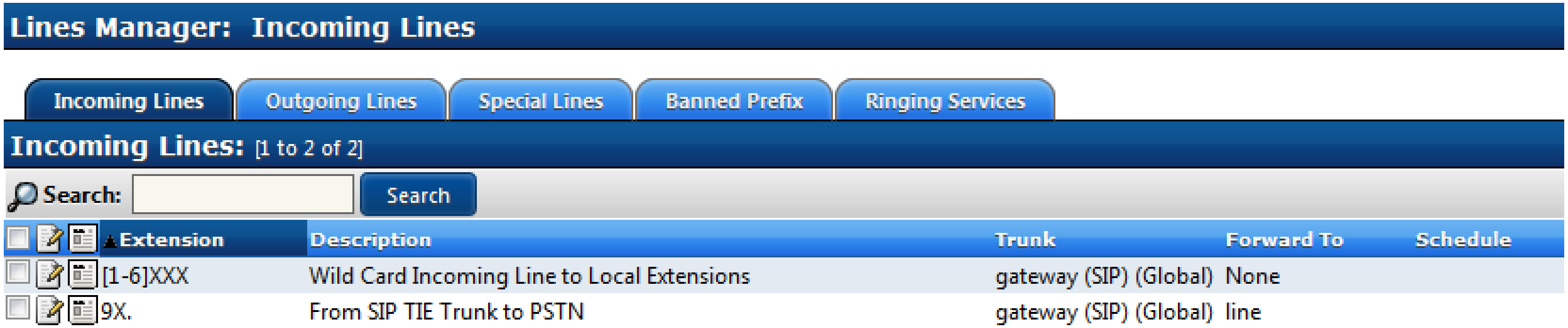
1. User dials: 94165551234
Server passes DNIS 94165551234 to the remote server using the SIP TIE Trunk

Tandem Dial PSTN Pattern

- From the 'specific' VoIP Interface you must set up an Outgoing Line and configure the Custom Dial String
- This Outgoing Line will have to be allowed in the Class of Service applied to each extension

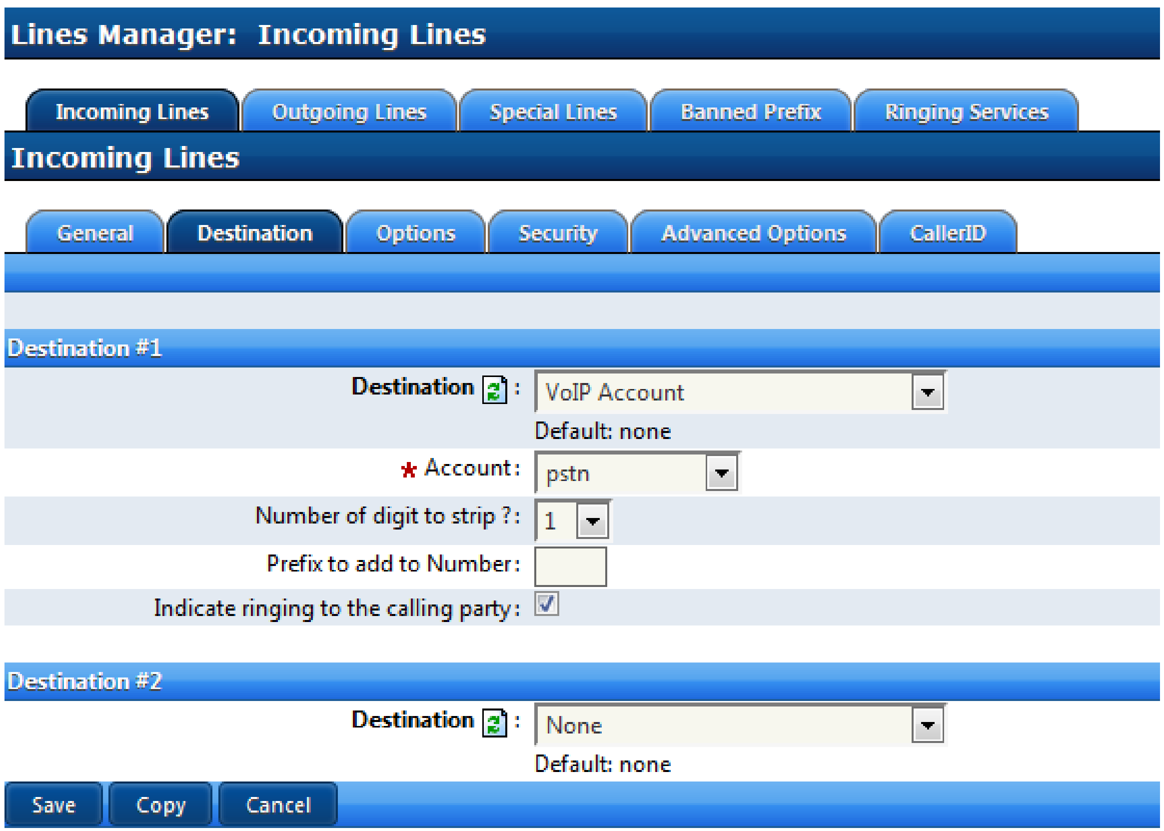


- **From the ‘dynamic’ VoIP Interface Server you must set up an Incoming Line to receive DNIS from the ‘specific’ VoIP Interface and bridge it to the PSTN Trunk/Outgoing Line**



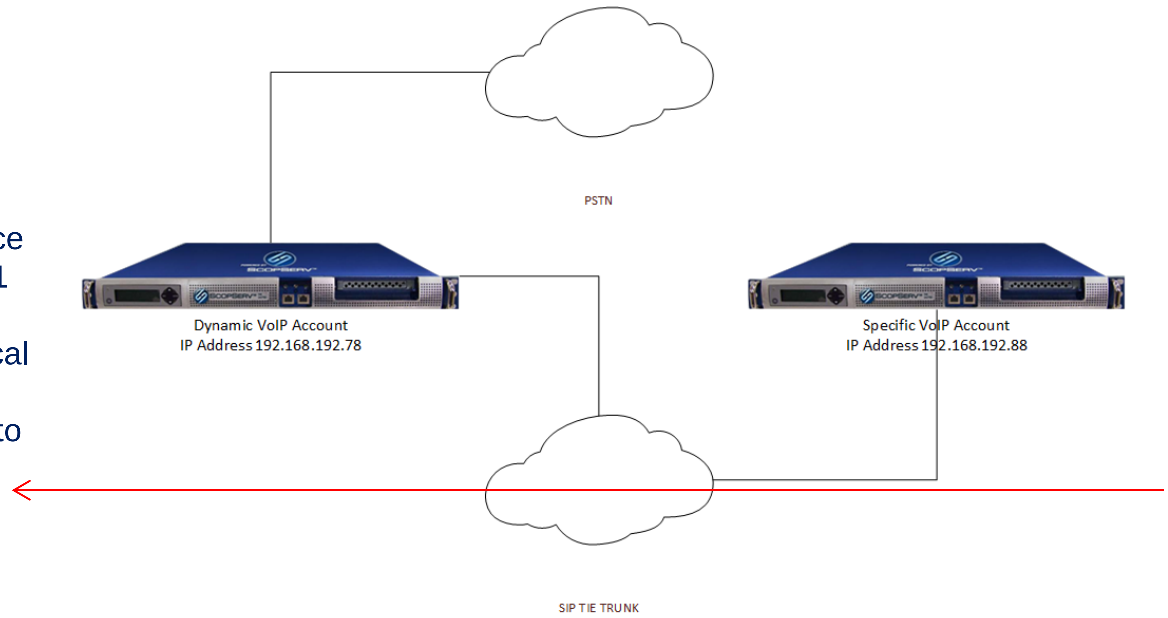
Tandem Dial Incoming Line

- **Set the Destination to the Account for the PSTN Interface**
- **Number of digit to strip?: 1 will strip the incoming 9 before bridging the call to an Outgoing PSTN Channel**
- **Indicate ringing to the calling party: enabled is optional**



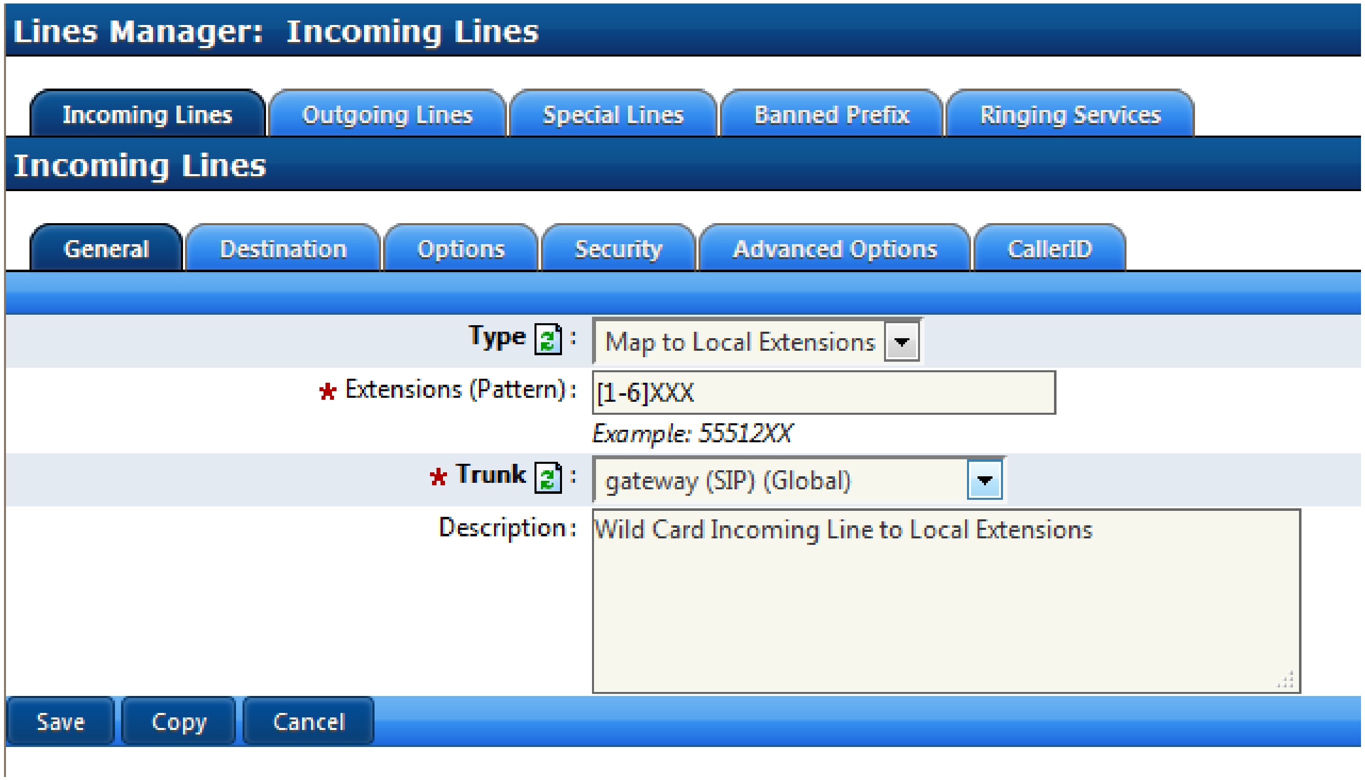
Private SIP TIE Trunking

2. The VoIP Interface receives DNIS 1001 from the SIP TIE Trunk, and rings local extension 1001 by mapping the DNIS to the matching local extension



1. User dials: 81001
- Server passes DNIS 1001 to the remote server using the SIP TIE Trunk

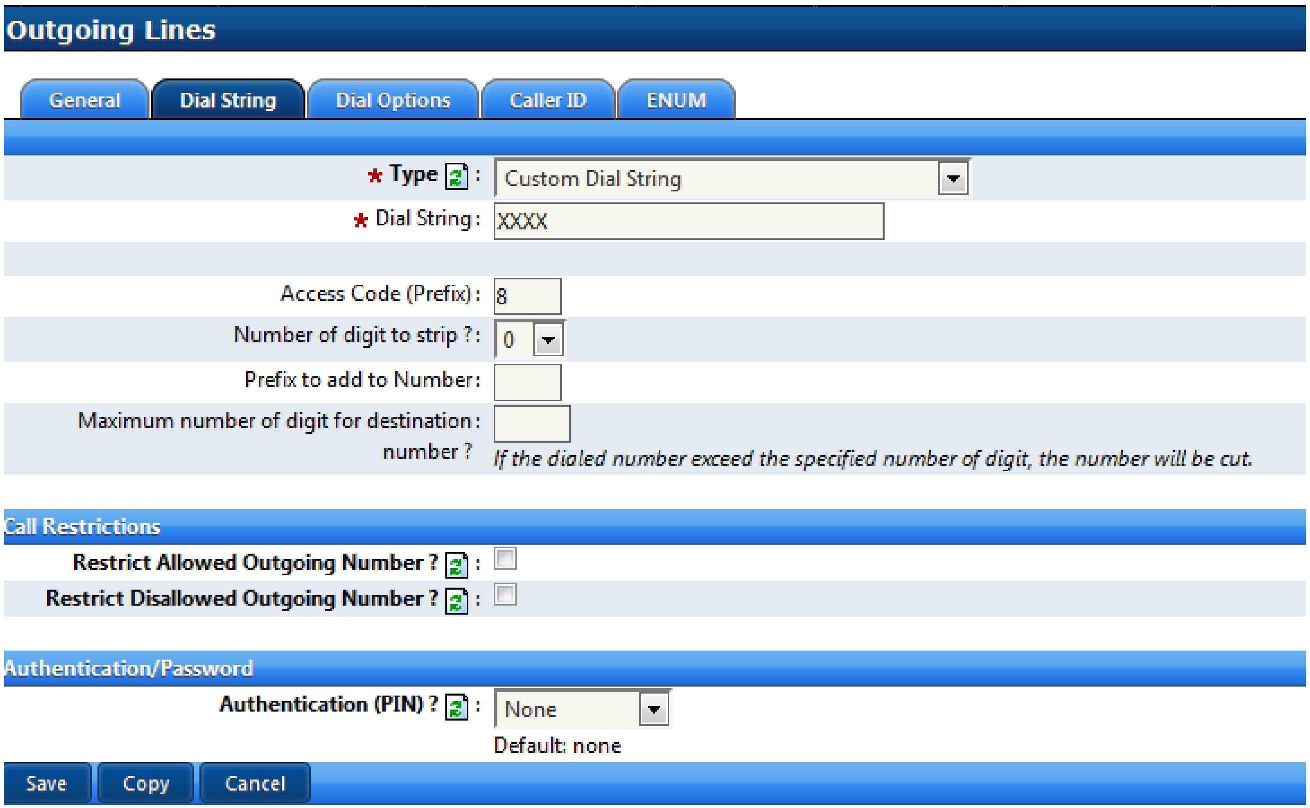
- **Incoming Lines to local extensions will use Called Number/DNIS supplied by the remote server.**
- **In this example the local extensions start with digits 1-6 and are a total of 4 digits in length**



- **Use User-defined CallForward ensures that any incoming call to this DNIS will follow the extension's custom forwarding options as defined in Extensions>User Options**

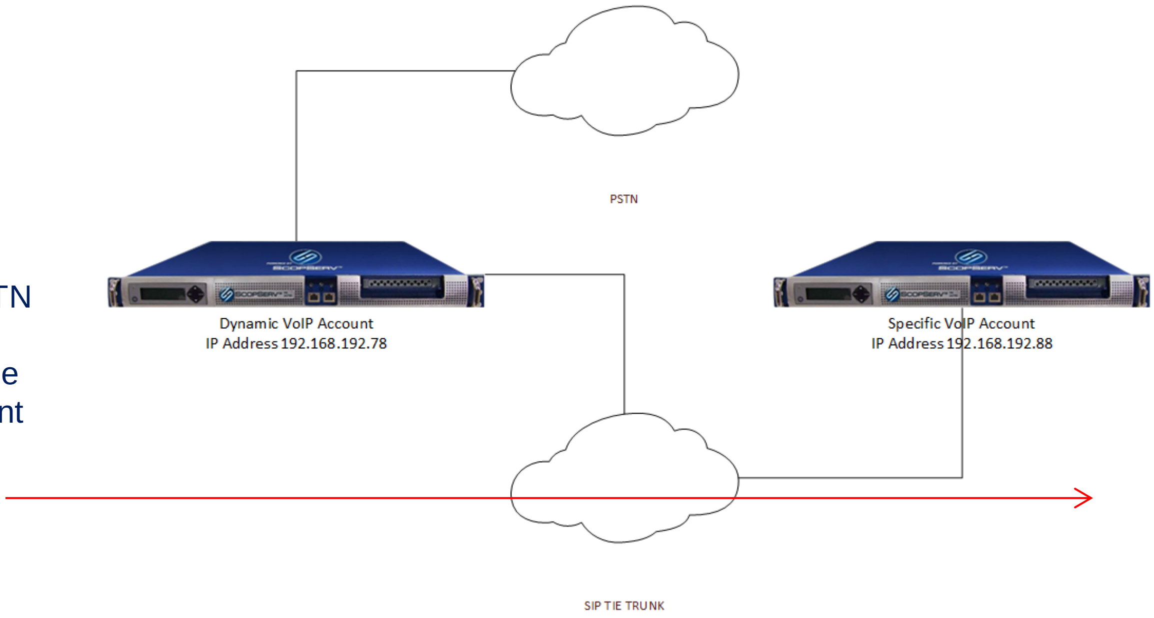
Private TIE Trunk Dial Pattern

- **From the ‘specific’ VoIP Interface you must set up an Outgoing Line and configure the Custom Dial String**
- **This Outgoing Line will have to be allowed in the Class of Service applied to each extension**
- **The Access Code (Prefix) is always stripped so the 8 will not be passed to the remote server**



- **In some cases you share a PSTN VoIP account to a remote server and need to set up custom DNIS for Incoming Lines**

A published phone number of 4165551234 is assigned to the PSTN interface and needs to be re-routed to the specific VoIP account over the SIP TIE Trunk



Incoming DNIS/Line 4165551234 to Destination IVR

Custom DNIS Incoming Lines

- **In some cases you share a PSTN VoIP account to a remote server and need to set up custom DNIS for Incoming Lines**
- **Configure an Incoming Line on the server with the dynamic VoIP Interface with DNIS 4165551234**
- **Set the source trunk to PSTN**

Lines Manager: Incoming Lines Mass Ope

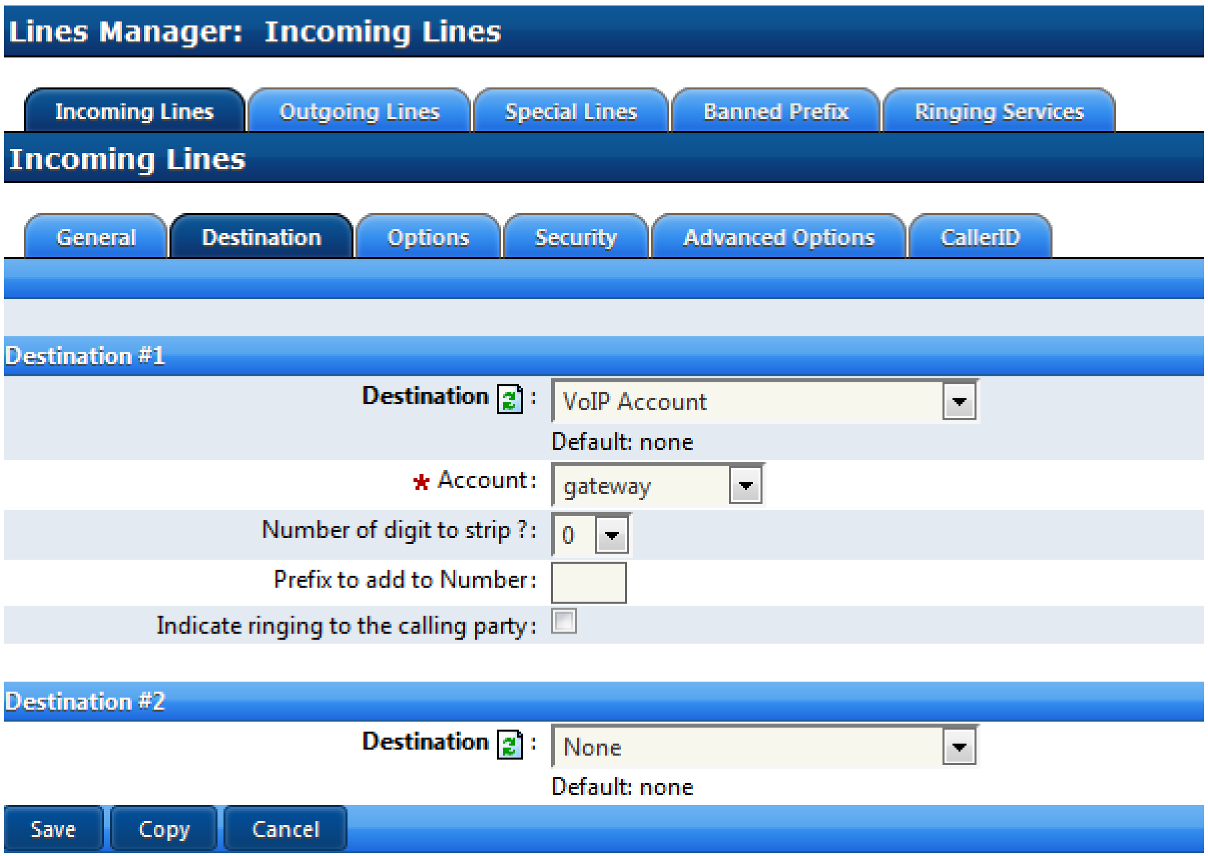
Incoming Lines | Outgoing Lines | Special Lines | Banned Prefix | Ringing Services

Incoming Lines: [1 to 3 of 3] Add a new Incomi

Search: Search

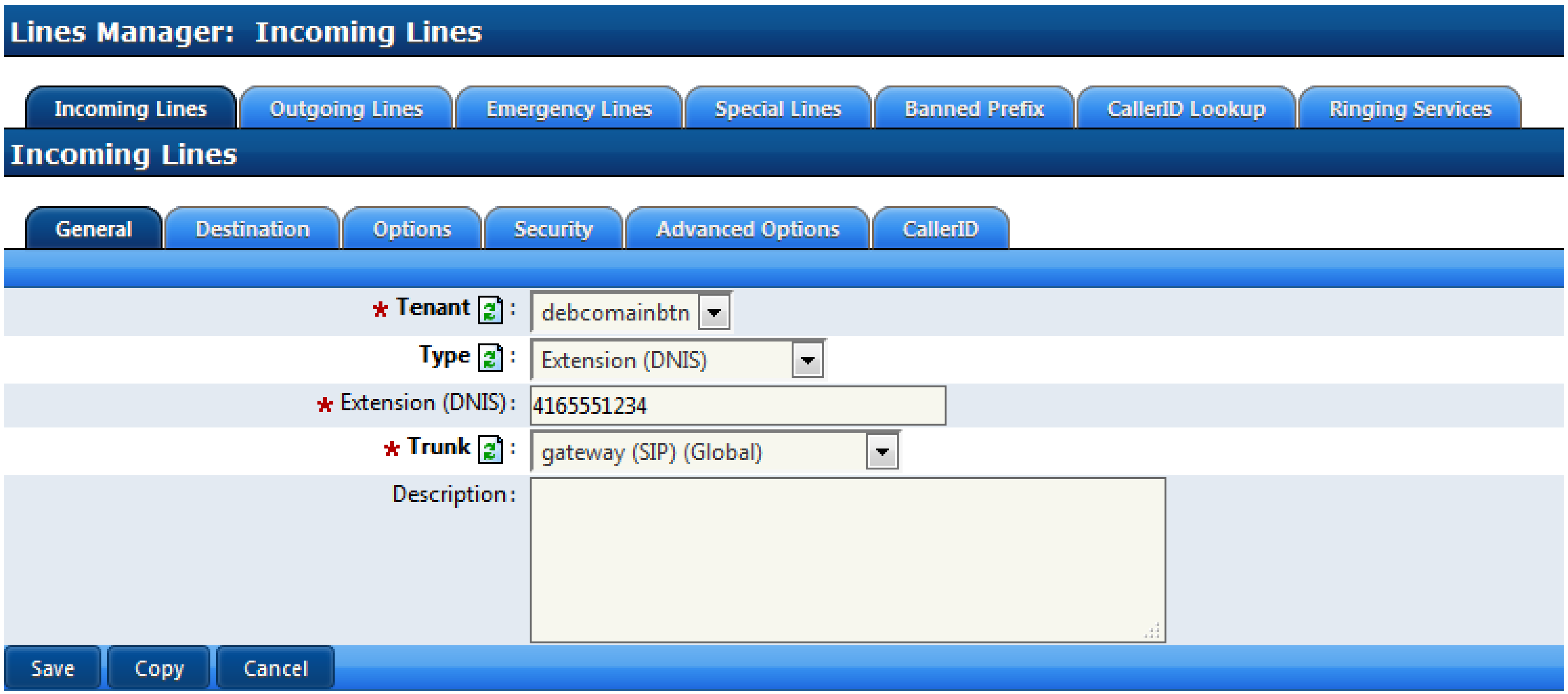
Extension	Description	Trunk	Forward To	Schedule	Priority	Tenant
[1-6]XXX	Wild Card Incoming Line to Local Extensions	gateway (SIP) (Global)	None			default
9X.	From SIP TIE Trunk to PSTN	gateway (SIP) (Global)	line			default
4165551234		pstn (SIP) (Global)	line			default

- **Set the Destination to VoIP Account gateway**



Custom DNIS Incoming Lines

- **The remote server with the ‘specific’ VoIP Interface needs an Incoming Line with DNIS 4165551234**



Custom DNIS Incoming Lines

- **Set the Destination to Auto Attendant and choose the desired Menu**