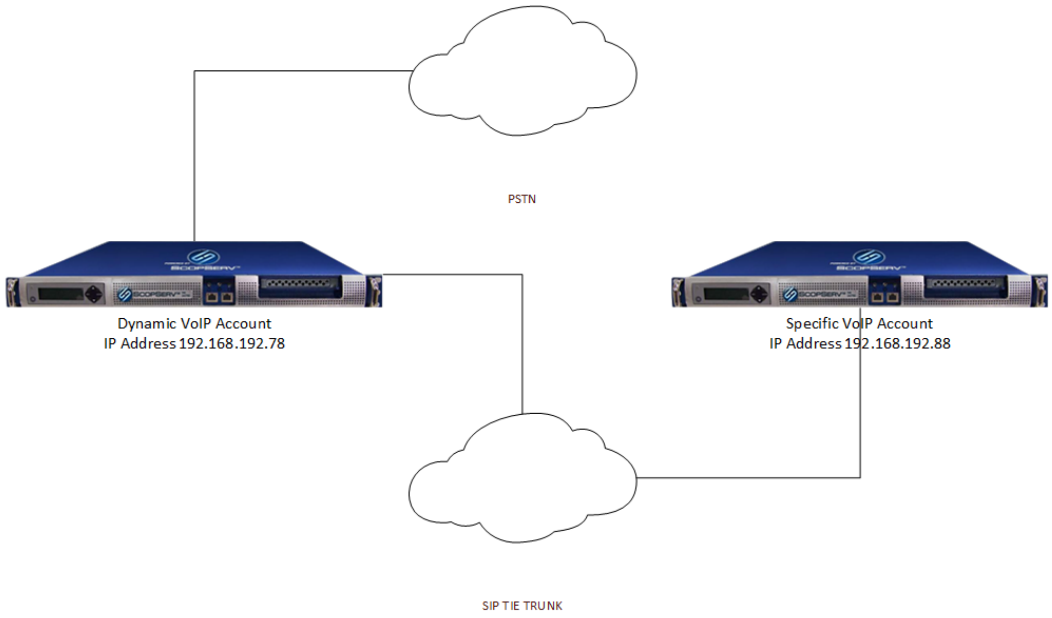
ScopTEL^{TM} IP PBX Software

ITSP SIP Trunking Configuration

Usage Cases Implementing DNIS:

- You are an ITSP reselling SIP trunks and DID's to a ScopTEL customer
- The PSTN access terminates to your multi tenant ScopTEL installation and you set up forwarding from the PSTN trunk to a SIP TIE Trunk (ScopTEL VoIP Account) you configure for the remote server installation
- The Dynamic Server is the ITSP installation so that remote customer installations must register to the ITSP
- The Specific Server is one of the remote customer servers that registers to the ITSP
- The SIP TIE TRUNK in this example will be configured as VoIP Account = gateway
- Each customer requires a unique VoIP Account and it is commonplace to name the account using the customer's BTN (Billing Telephone Number)



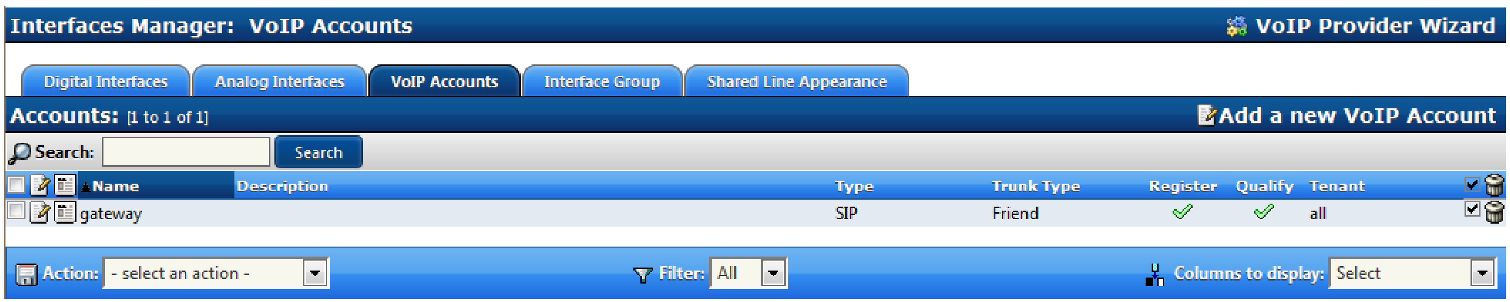
ScopTEL Background and Pre-requisites

- **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

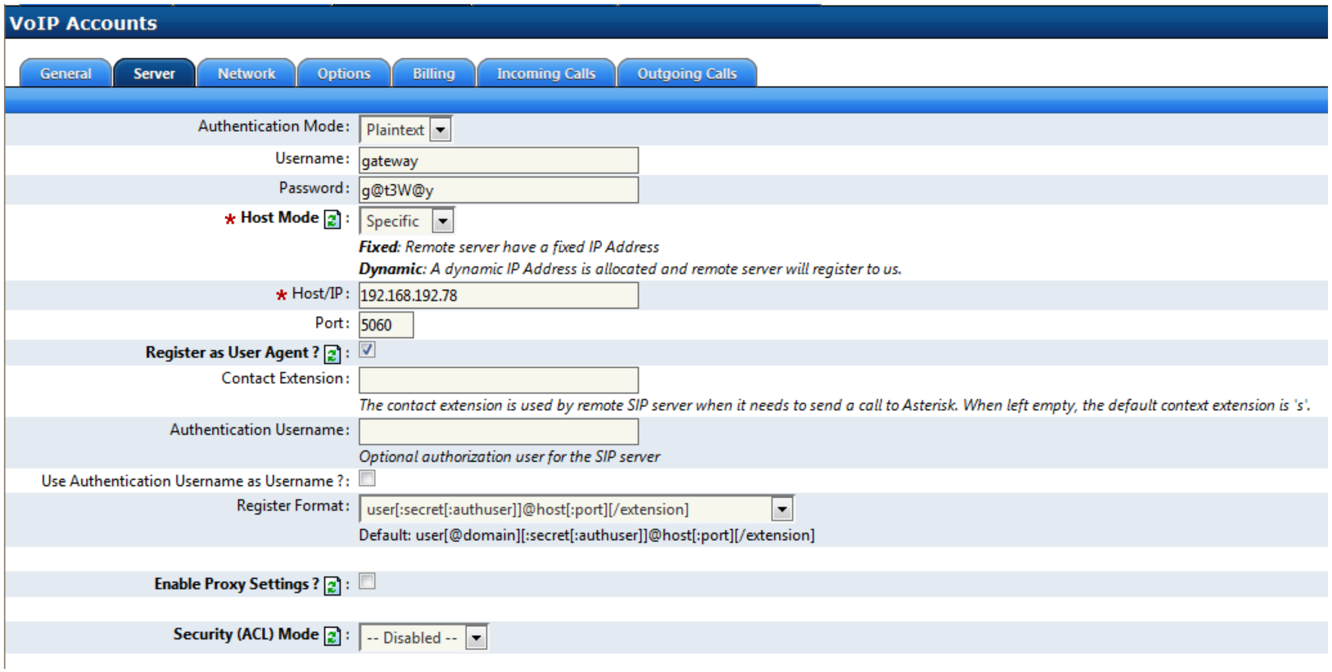
Use the ScopTEL Interfaces Manager to build the customer SIP trunk

- **In this example the VoIP Account will be called ‘gateway’**
- **You must build the VoIP Account on the ITSP and Customer’s server**



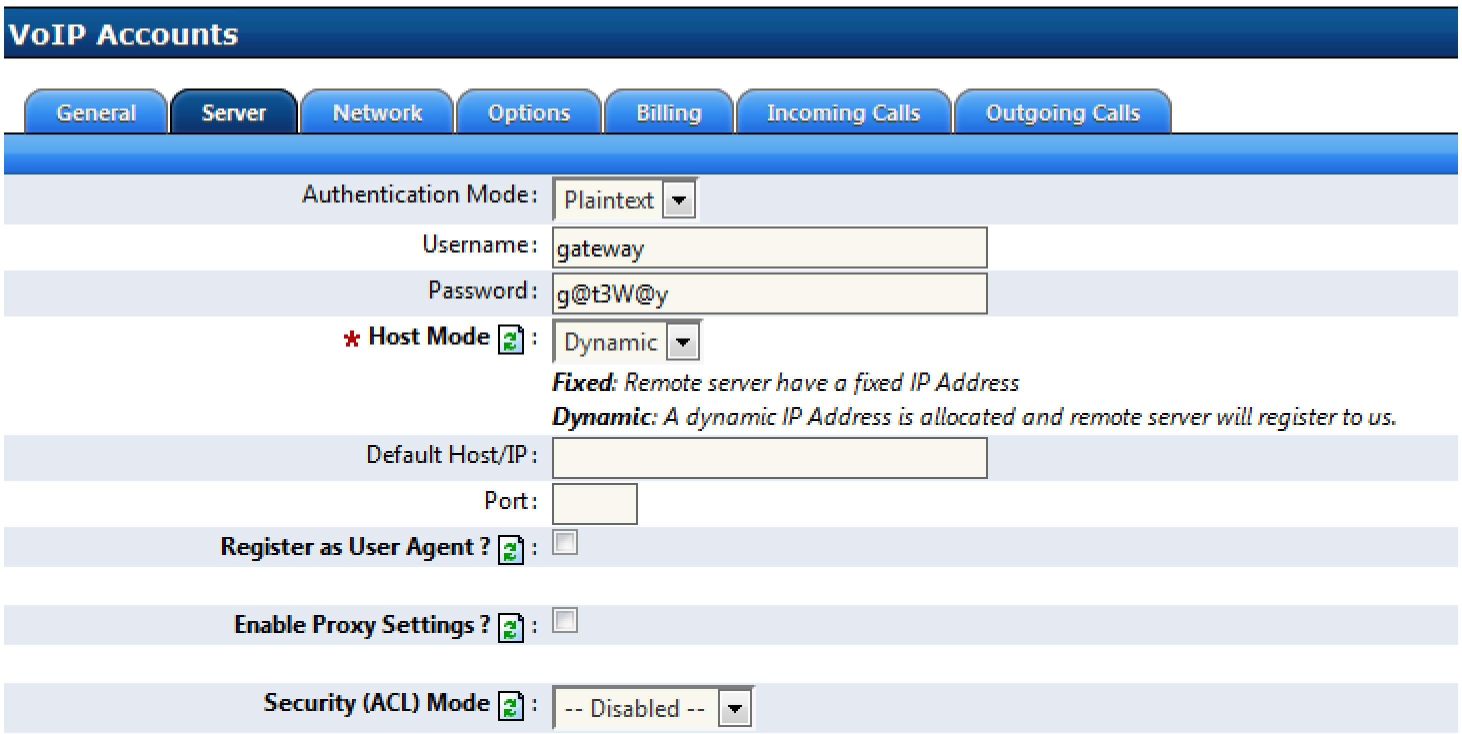
Specific VoIP Account

- **The Account name and Username must match on each server**
- **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]'



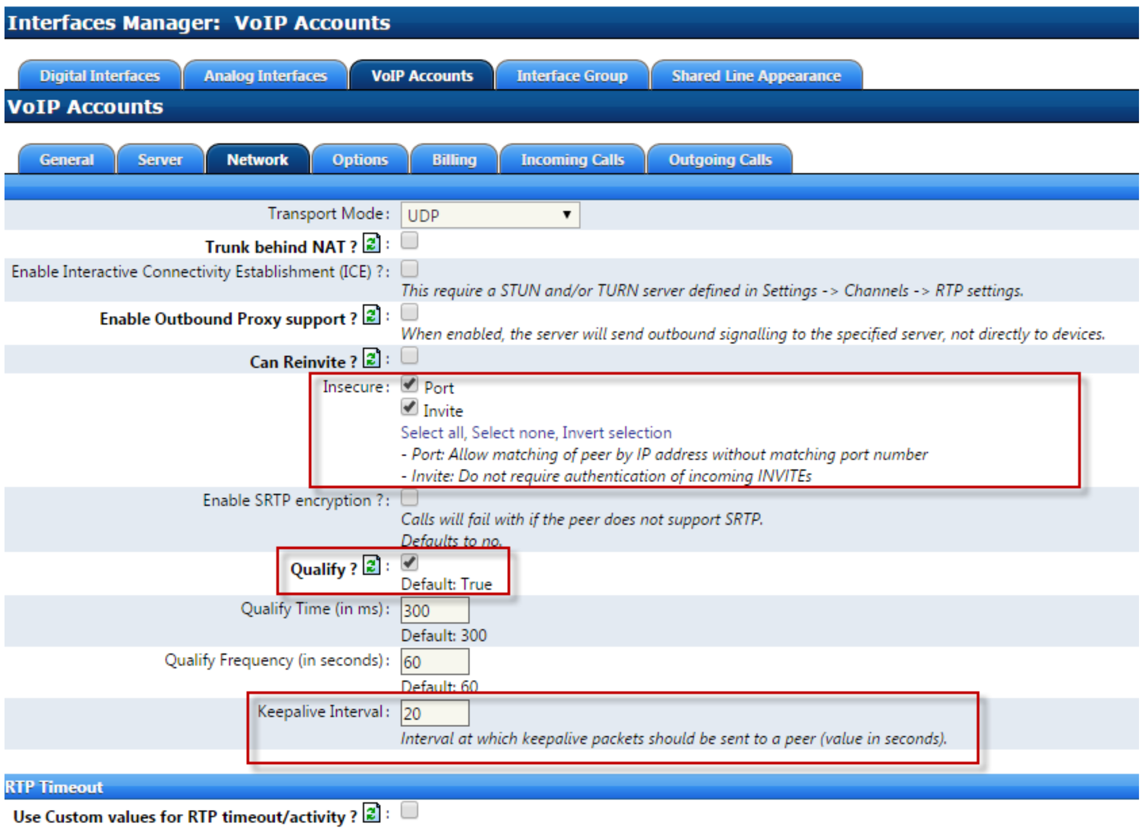
Dynamic VoIP Account

- **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]'



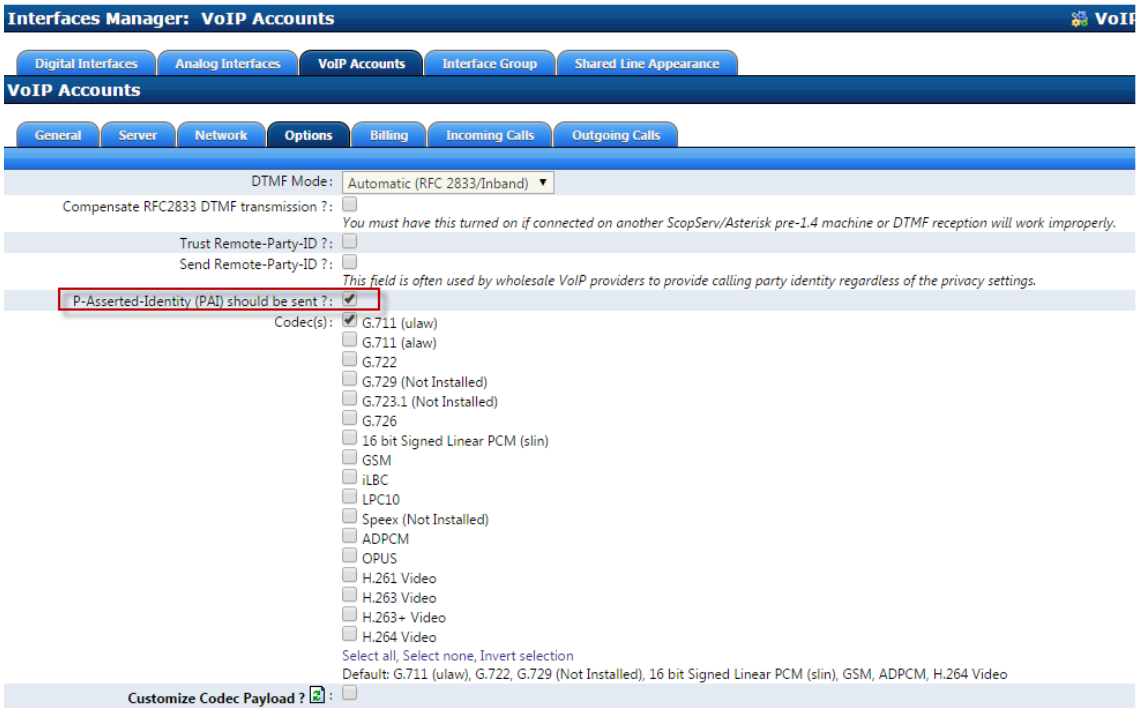
Shared Settings

- **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**

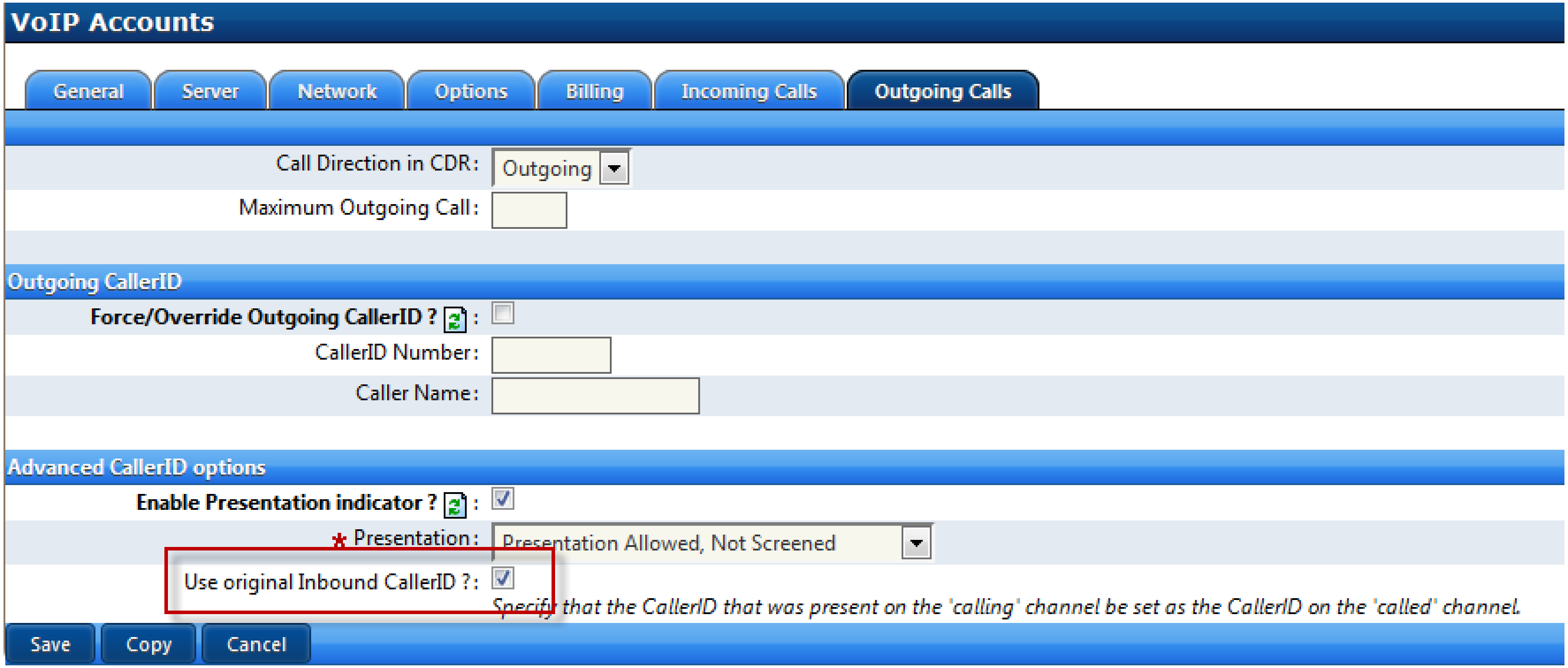


Shared Settings

- 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



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



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

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 be stripped from the Outgoing Line after the Class of Service lookup since the 9 should not be passed to the ITSP server**
- **The 9 is not mandatory and can be replaced by any other digit(s)**

Lines Manager: Outgoing Lines

Incoming Lines | Outgoing Lines | Emergency Lines | Special Lines | Banned Prefix | CallerID Lookup | Ringing Services

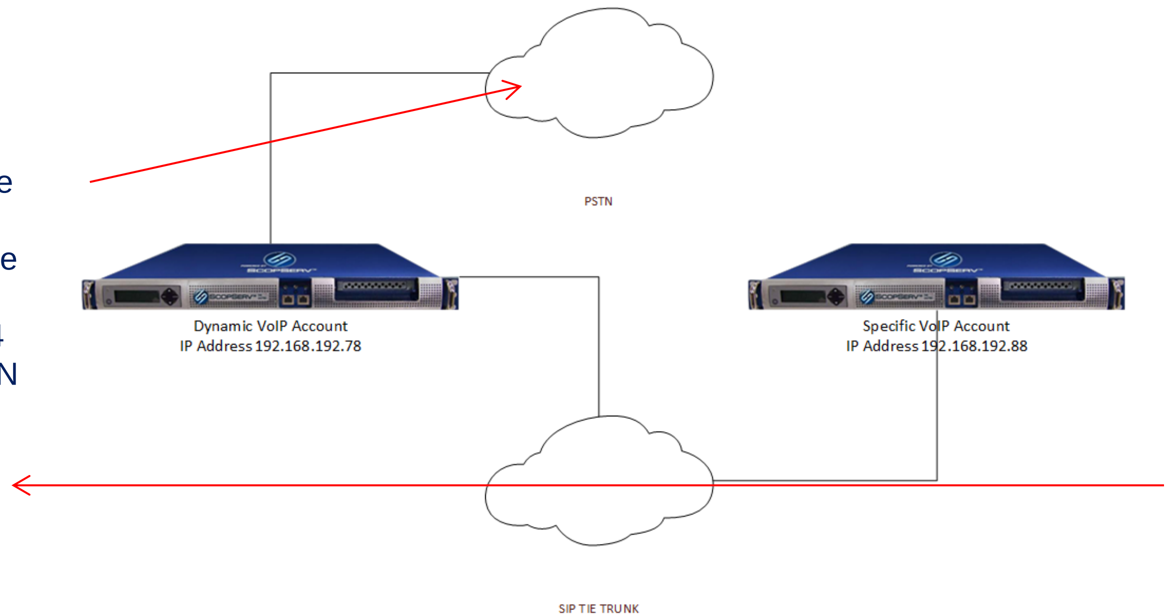
Outgoing Lines: [1 to 2 of 2] **Add a new Outgoing Line**

Search: Search

Dial String	Name	Trunk	Priority	Tenant
9 + X.	9x.	gateway (SIP) (Global)		debcomainbtn
8 + XXXX	8XXXX	gateway (SIP) (Global)		debcomainbtn

Outgoing ITSP PSTN Trunking

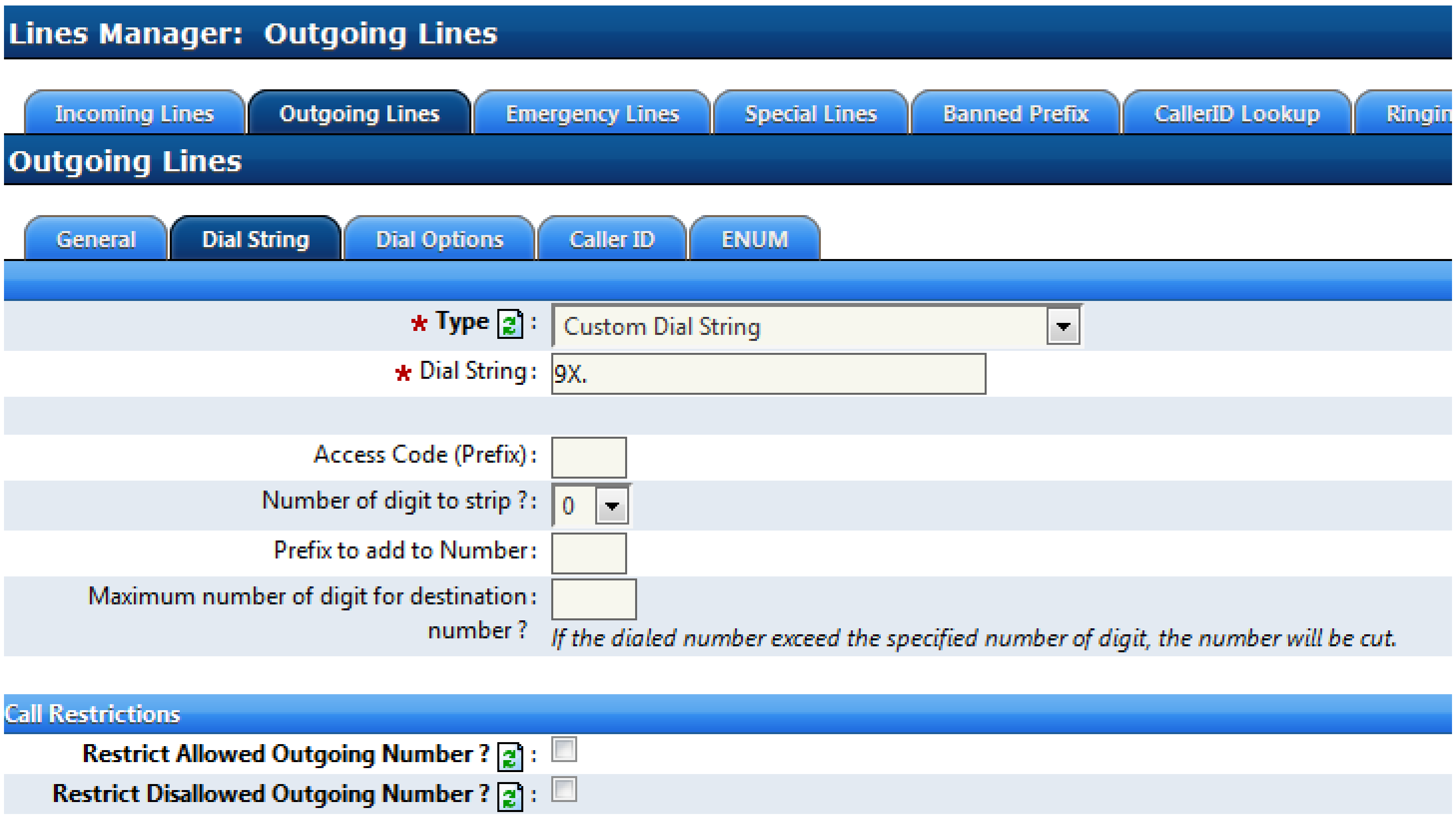
2. The VoIP Interface receives DNIS 4165551234 from the SIP TIE Trunk, bridges 4165551234 to an Outgoing PSTN channel using the PSTN VoIP Account



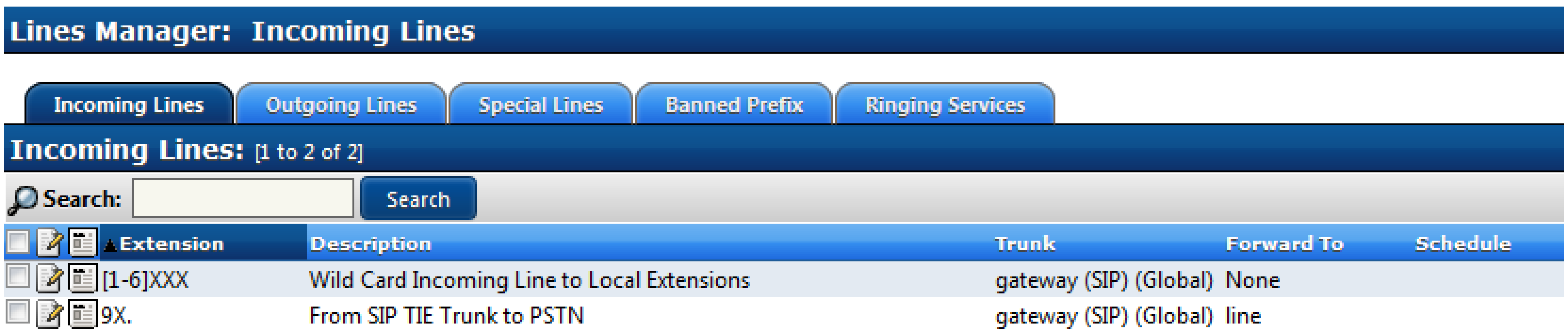
1. User dials: 94165551234
Server passes DNIS 94165551234 to the remote server using the SIP TIE Trunk

Customer Outgoing 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**

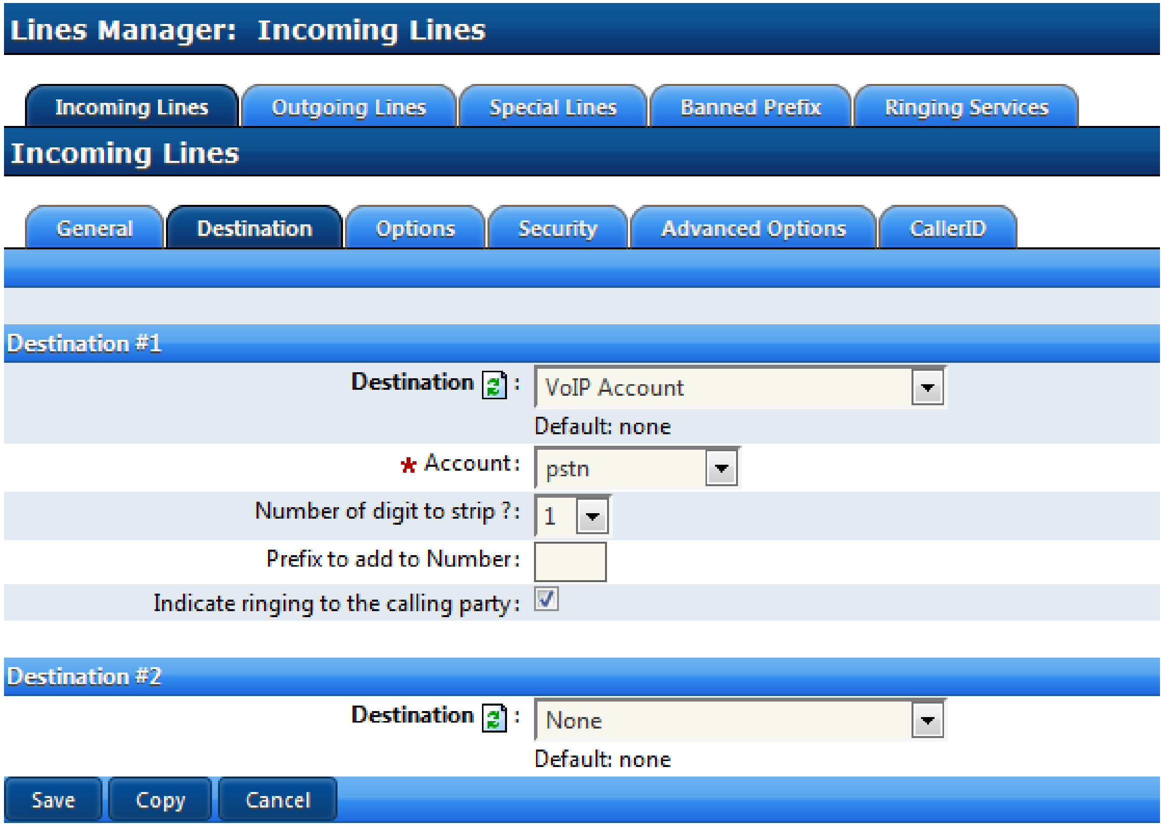


- **From the ‘dynamic’ VoIP Interface Server you must set up an Incoming Line to receive DNIS from the ‘specific’ VoIP Interface and bridge it to the PSTN Trunk/Outgoing Line**



PSTN Dial Incoming Line

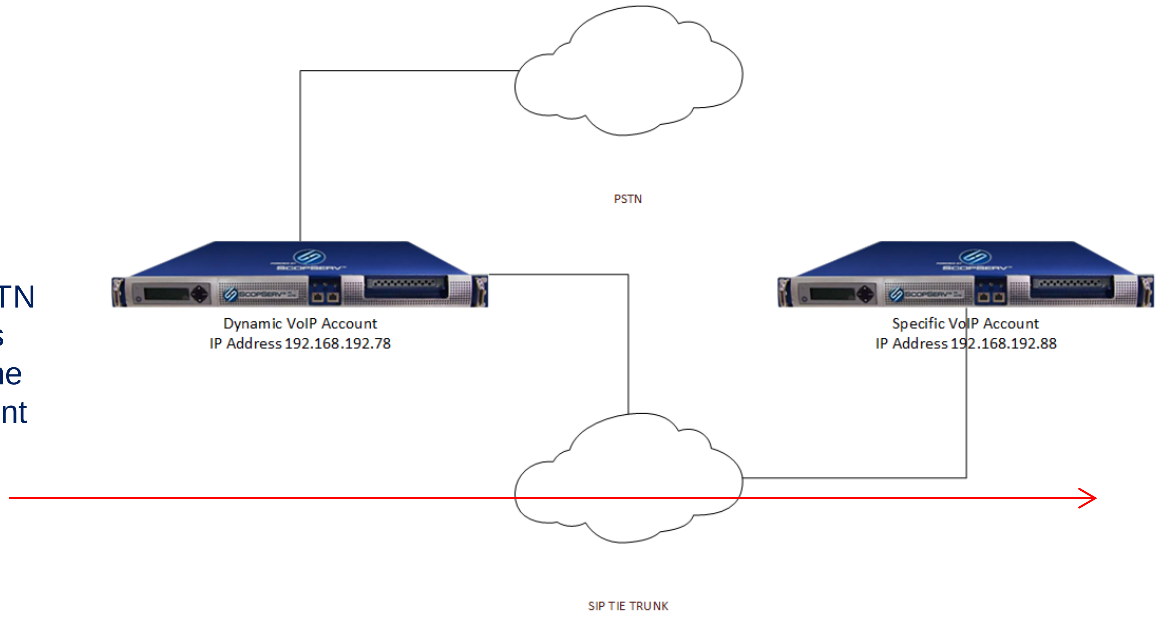
- **Set the Destination to the Account for the PSTN Interface**
- **Number of digit to strip?: 1 will strip the incoming 9 before bridging the call to an Outgoing PSTN Channel**
- **Indicate ringing to the calling party: enabled is optional**



Customer DNIS Incoming Lines

- **The customer server requires Incoming Lines using DNIS to route incoming calls from the ITSP**

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

- **The ITSP Server uses one or more Carrier peered SIP Trunks or PRI circuits and will share the trunk(s) with many customer installations using gateway interfaces**
- **Configure an Incoming Line on the ITSP server with DNIS 4165551234**
- **Set the source trunk to 'pstn'. This will vary depending on the peered carrier interfaces**
- **Set the destination trunk to 'gateway' (a unique gateway interface must be created for each customer)**
- **Any incoming DNIS bound to the ISTP 'pstn' interface will be automatically forwarded to the customer 'gateway' interface**
- **The customer gateway interface can then do any Incoming Line lookup and set the custom destination**

Lines Manager: Incoming Lines Mass Ope

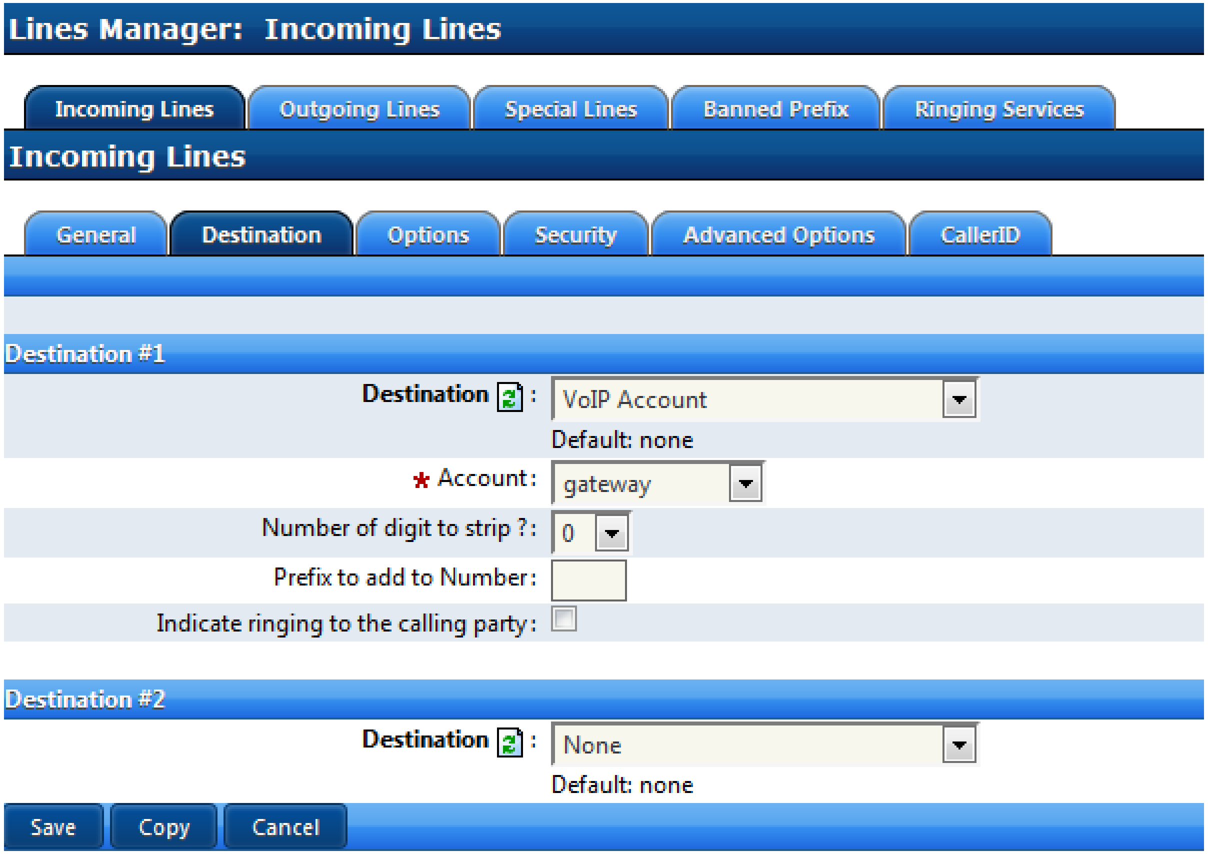
Incoming Lines | Outgoing Lines | Special Lines | Banned Prefix | Ringing Services

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

Search: Search

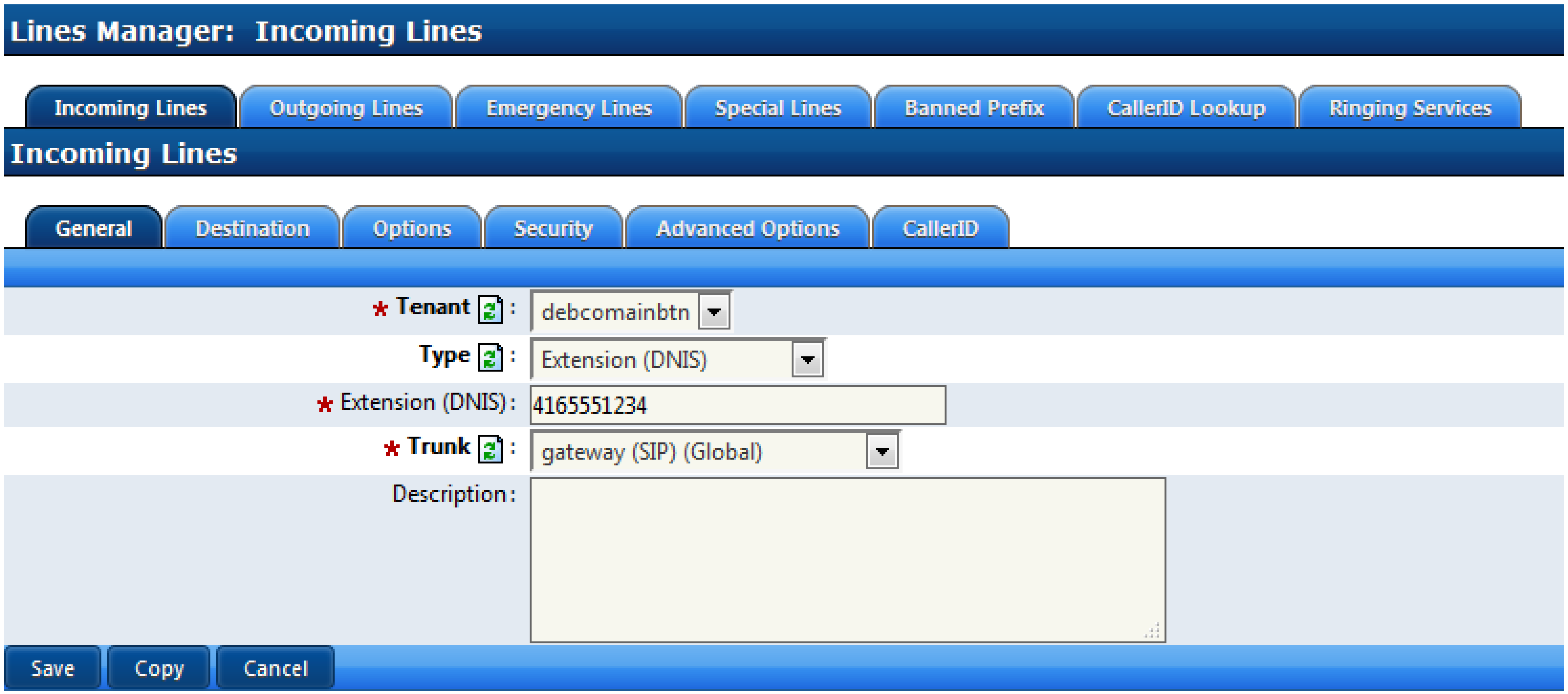
Extension	Description	Trunk	Forward To	Schedule	Priority	Tenant
[1-6]XXX	Wild Card Incoming Line to Local Extensions	gateway (SIP) (Global)	None			default
9X.	From SIP TIE Trunk to PSTN	gateway (SIP) (Global)	line			default
4165551234		pstn (SIP) (Global)	line			default

- **On the ITSP (Specific VoIP Account = pstn) Set the Destination to VoIP Account = gateway**



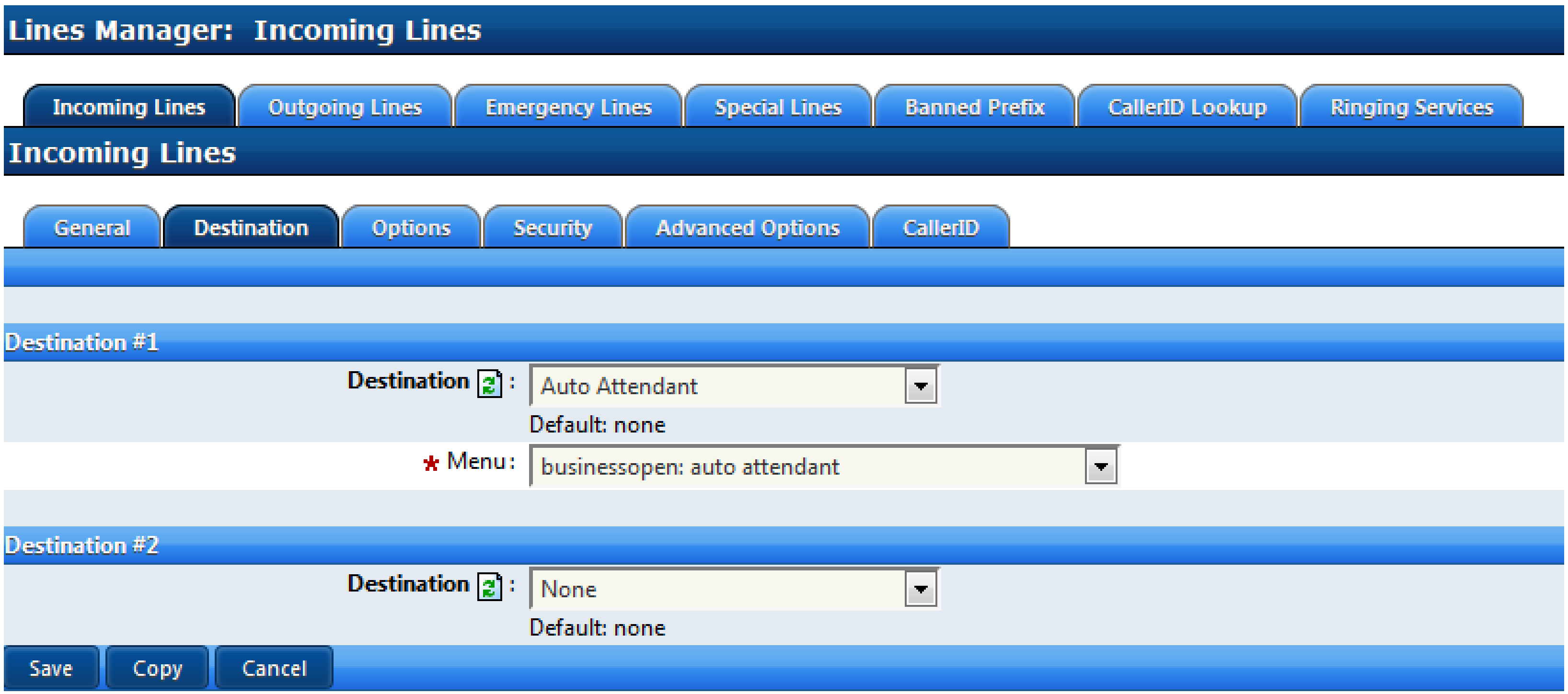
Customer DNIS Incoming Lines

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



Customer DNIS Incoming Lines

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



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 416-555-1234
  - The Received Digit length set by the Telco is 10
  - The resulting DNIS or received digits is equal to 4165551234
- **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 4165551234
  - Therefore the DNIS number for 1-800-555-2234 is 4165551234

Customer trace:
The Incoming Call from the PSTN will be relayed across the trunk and the customer server will route the incoming call using the SIP INVITE header as the incoming DNIS.
The Incoming Lines are parsed for a matching DNIS and ring the configured destination
In this example the DNIS passthrough is 4165551234 and the matching Incoming Line destination is the Auto Attendant Menu

[2016-03-16 11:16:31] <--- SIP read from UDP:192.168.192.78:5060 --->
[2016-03-16 11:16:31] INVITE sip:4165551234@192.168.192.88:5060;user=phone SIP/2.0
[2016-03-16 11:16:31] Via: SIP/2.0/UDP 192.168.192.78:5060;branch=z9hG4bK674783fd
[2016-03-16 11:16:31] Max-Forwards: 70
[2016-03-16 11:16:31] From: "Extension 8011" <sip:8011@192.168.192.78>;tag=as1c47e2b4
[2016-03-16 11:16:31] To: <sip:4165551234@192.168.192.88:5060;user=phone>
[2016-03-16 11:16:31] Contact: <sip:8011@192.168.192.78:5060>
[2016-03-16 11:16:31] Call-ID: 31d00bdb7b4d10f71ea191251c3c196c@192.168.192.78:5060
[2016-03-16 11:16:31] CSeq: 102 INVITE
[2016-03-16 11:16:31]
[2016-03-16 11:16:31] <------------->
[2016-03-16 11:16:31] --- (15 headers 11 lines) ---
[2016-03-16 11:16:31] Sending to 192.168.192.78:5060 (no NAT)
[2016-03-16 11:16:31] Sending to 192.168.192.78:5060 (no NAT)
[2016-03-16 11:16:31] Using INVITE request as basis request - 31d00bdb7b4d10f71ea191251c3c196c@192.168.192.78:5060
[2016-03-16 11:16:31] Found peer 'gateway' for '8011' from 192.168.192.78:5060
[2016-03-16 11:16:31] Looking for 4165551234 in all-gateway-incoming (domain 192.168.192.88)
[2016-03-16 11:16:31] -- Executing [4165551234@all-gateway-incoming:1] Set("SIP/gateway-00000003", [2016-03-16 11:16:31] -- Executing [4165551234@all-gateway-incoming:3] Set("SIP/gateway-00000003", "__INCOMING_DNIS=4165551234") in new stack
[2016-03-16 11:16:31] -- Executing [4165551234@all-gateway-incoming:57] Gosub("SIP/gateway-00000003", "debcomainbtn-aa-businessopen,s,1") in new stack
[2016-03-16 11:16:32] -- Executing [s@debcomainbtn-aa-businessopen:10] BackGround("SIP/gateway-00000003", "custom/0001") in new stack
[2016-03-16 11:16:32] -- <SIP/gateway-00000003> Playing 'custom/0001.ulaw' (language 'en')
master88*CLI>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