USER GUIDE

Alcatel OmniPCX Office OXO-Fusion 360 SIP Trunk Programming Guide

11/07/2017

Contents:

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SIP Trunk Programming Guide

Follow the programming steps below to for SIP trunk direct connection to the OXO:

Step 1: Gather the Following Information from the Carrier

You have the ability to email a recorded call to a recipient. To send a recorded call file via email, select the (Email button) that correspond

1. Invite Domain (Carrier terms: Proxy Server, Invite Server, etc.)
2. Username (This is usually the first DID phone number)
3. How many total calls? This will determine how many SIP trunk licenses you need.
4. What are the DNIS numbers? Program your DNIS numbers in your Public Dialing Plan.

Sample:

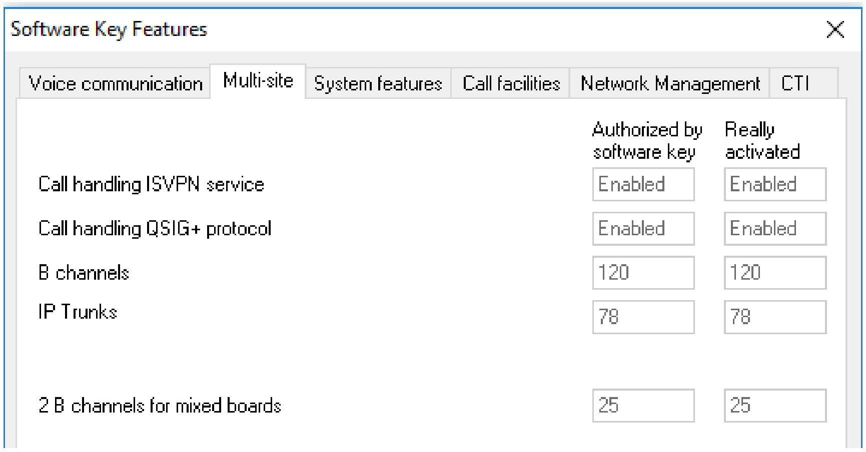
- Domain Name – sip3.thevoicemanager.com
- username – 9735551212
- DIDs – 9735551212, 9735551213

Notes:

1. Fax is not supported on SIP trunks. ICON recommends an analog trunk for fax.
2. Version 10 software with version 10 license is required for Static NAT operation. If you have a previous version, an SBC is required.
3. System Miscellaneous>Feature Design>Part 2: CLI is diverted party if external call=Check. Carrier will not allow original caller id sent to called party on a diverted call.
4. Numbering>Gateway Parameters>Identity (Calling Preferred Identity>Outgoing): P-preferred Identity=Check. P-Asserted Identity=Uncheck. Note: This may be corrected by Fusion in a future software version.
5. Numbering>Gateway Parameters>Registration (Address of Record Registration): Contact and From=Check
6. Numbering>Gateway Parameters>Identity (Alternative CLIP): Contact and From=Uncheck
7. System Miscellaneous>Memory Read/Write>Debug Labels: MultAnsRei=00
8. For fax, set to g711 and set

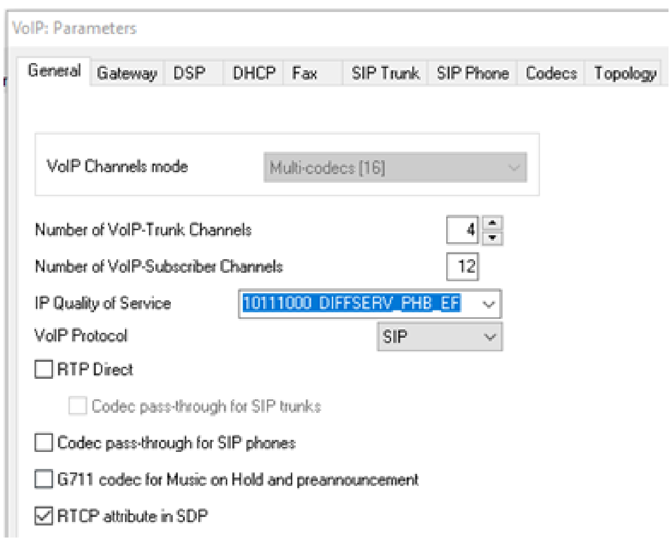
Step 2: OXO Programming

Hardware and Limits>Software Key Features (Licensing)



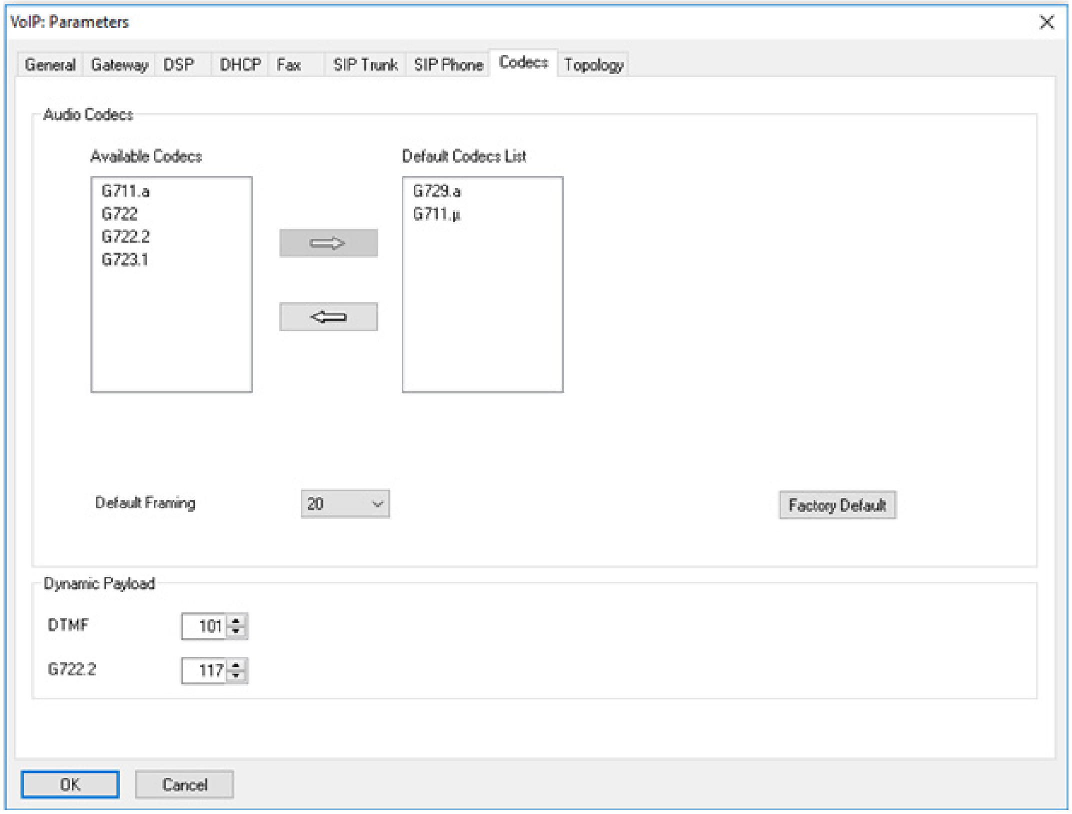
Verify your IP Trunks licenses are installed and activated.

Voice Over IP>VoIP: Parameters



Number of VoIP-Trunk Channels: Match the number of licensed IP Trunks.

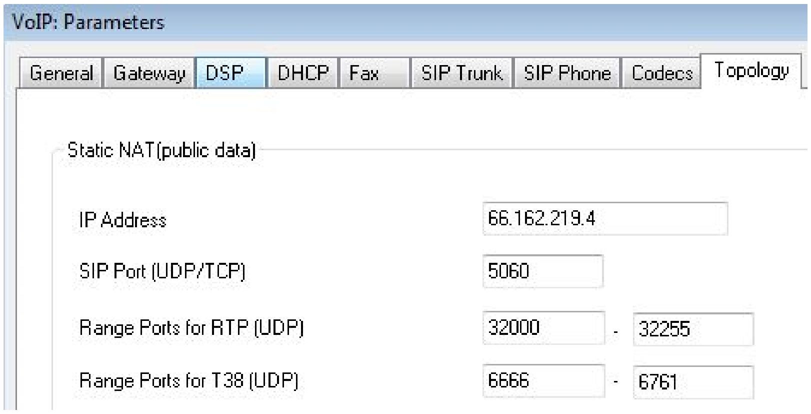
Important: RTP Direct is NOT SUPPORTED for Static NAT installations.



Codecs tab: These are the codecs for SIP trunks. Changes to this item do not require a reset. Calls made after a change will use the new codec. Order is important. The top entry will be used if the carrier supports it. The picture shows the setting for all carriers I have tested. Carriers may support more codecs, but this works with every carrier I have tested.

DTMF: This should be set to 101.

Note: Default Framing should be set to 20.

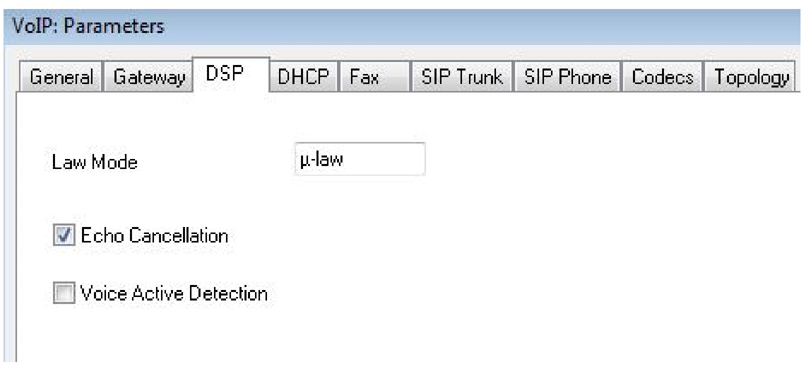


IP Address: Customer's static public IP address

SIP Port: 5060

Range Ports for RTP (UDP): 32000-32255

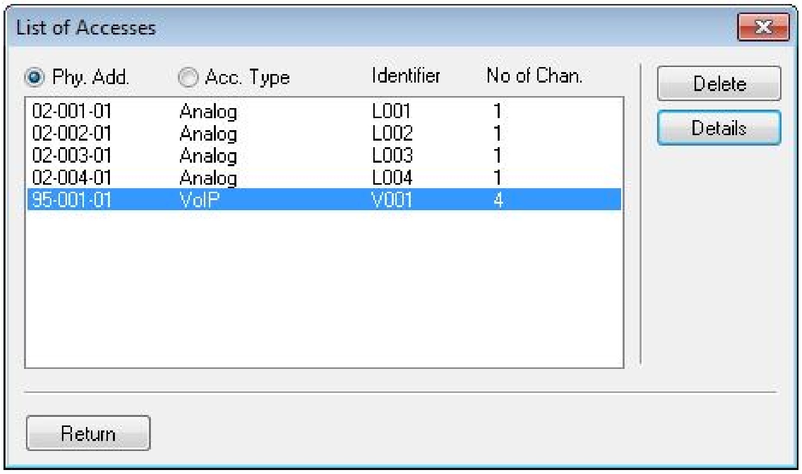
Range Ports for T38 (UDP): 6666-6761



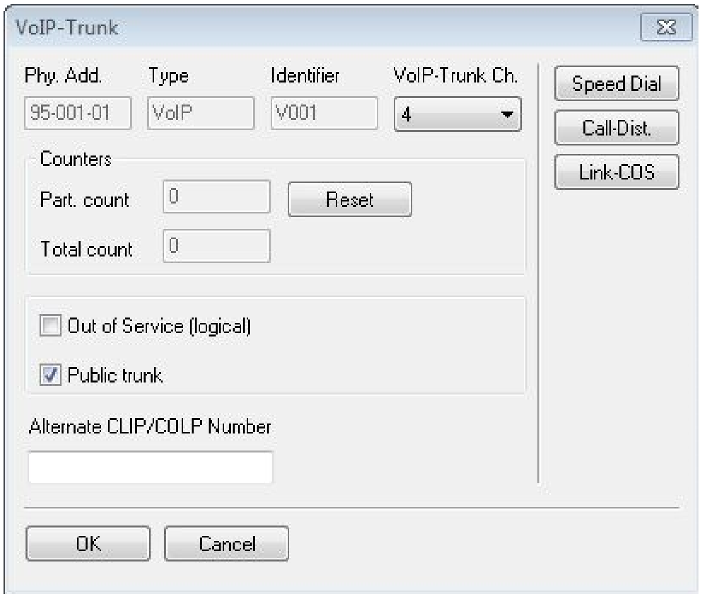
Echo Cancellation: Check

Voice Active Detection: Uncheck

External Lines>List of Accesses



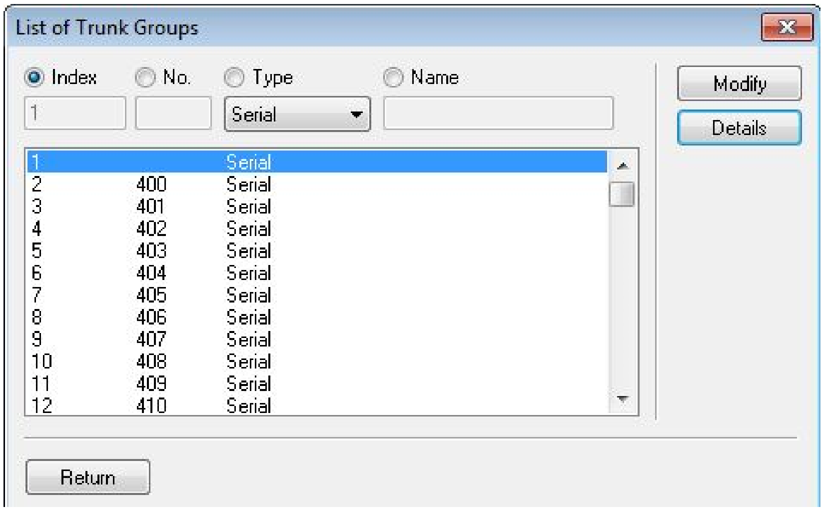
Select the VoIP trunks and click Details



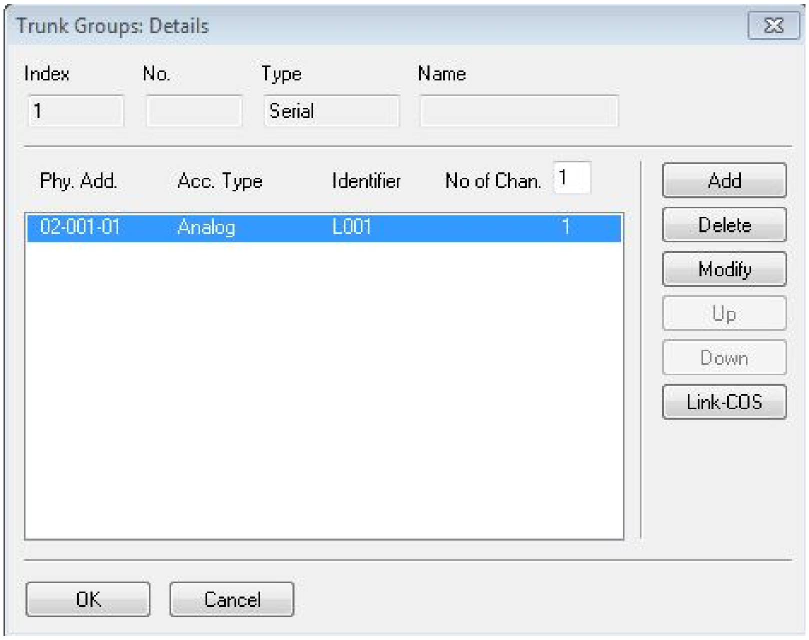
Set the number of VoIP Trunk channels to match your IP Trunk license count.

Check the Public trunk option.

External Lines>List of Trunk Groups

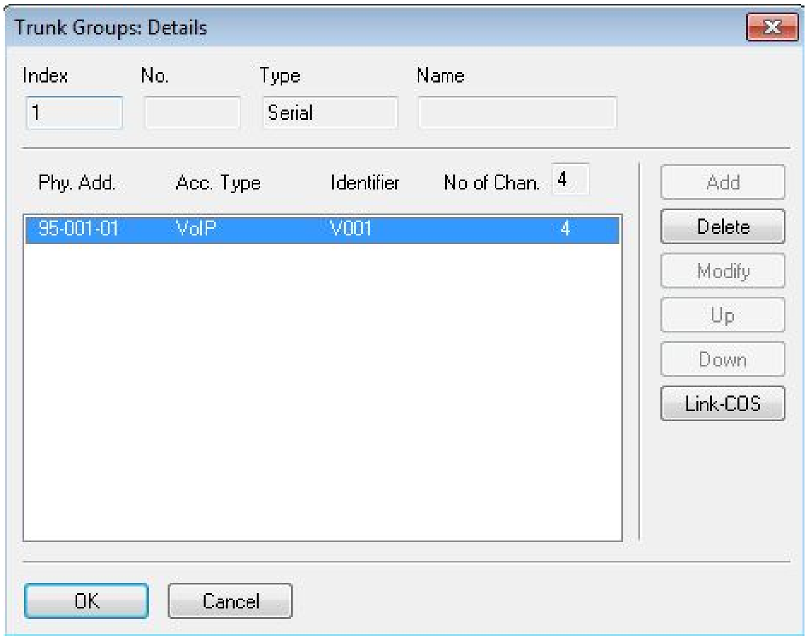


Select the trunk group 1 and click Details



Click Delete to remove your existing trunks from group 1.

Click Add to add the VoIP trunks to group 1.



Numbering>Automatic Routing Selection:

Automatic Routing: Prefixes

Automatic Routing: Prefixes

Activation	Network	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H450)
Yes	pub		1-1		1	het
Yes	pub		2-9		1	het
Yes	emerg				1	het
Yes	pub	11		9911	99	het

Right-click and press Add. Then, right-click and select IP Parameters.

Activation: yes

Network: pub

Prefix: <blank>

Ranges: digit range that will match to the number dialed

Substitute: <blank>

TrGpList: Which trunk group are the SIP trunks programmed in?

Called (ISVPN/H450): het. Het=sip trunk connect to public carrier. Hom=sip trunks connect to another oxo.

Automatic Routing: Prefixes			
User comment	Destination	Gateway Alive Status	Index of Gateway Parameters
1plus	SIP Gateway	Alive	2 Fusion 360
Local	SIP Gateway	Alive	2 Fusion 360
Emergency	SIP Gateway	Alive	2 Fusion 360
11 trans to 911	Not IP		

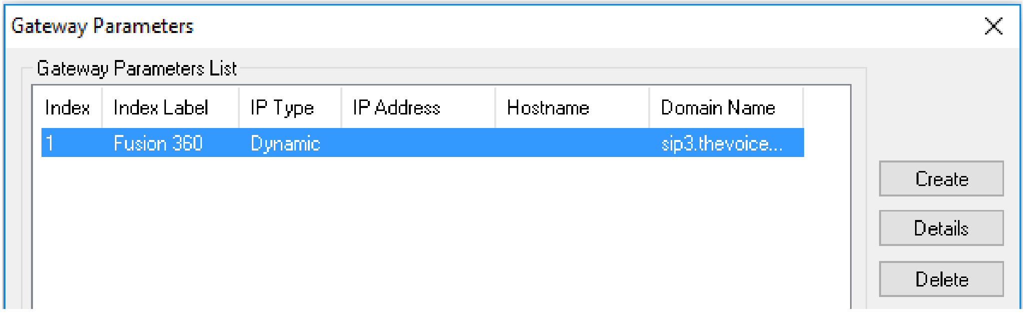
User Comment: Label for you to identify your dial tables.

Destination: SIP Gateway

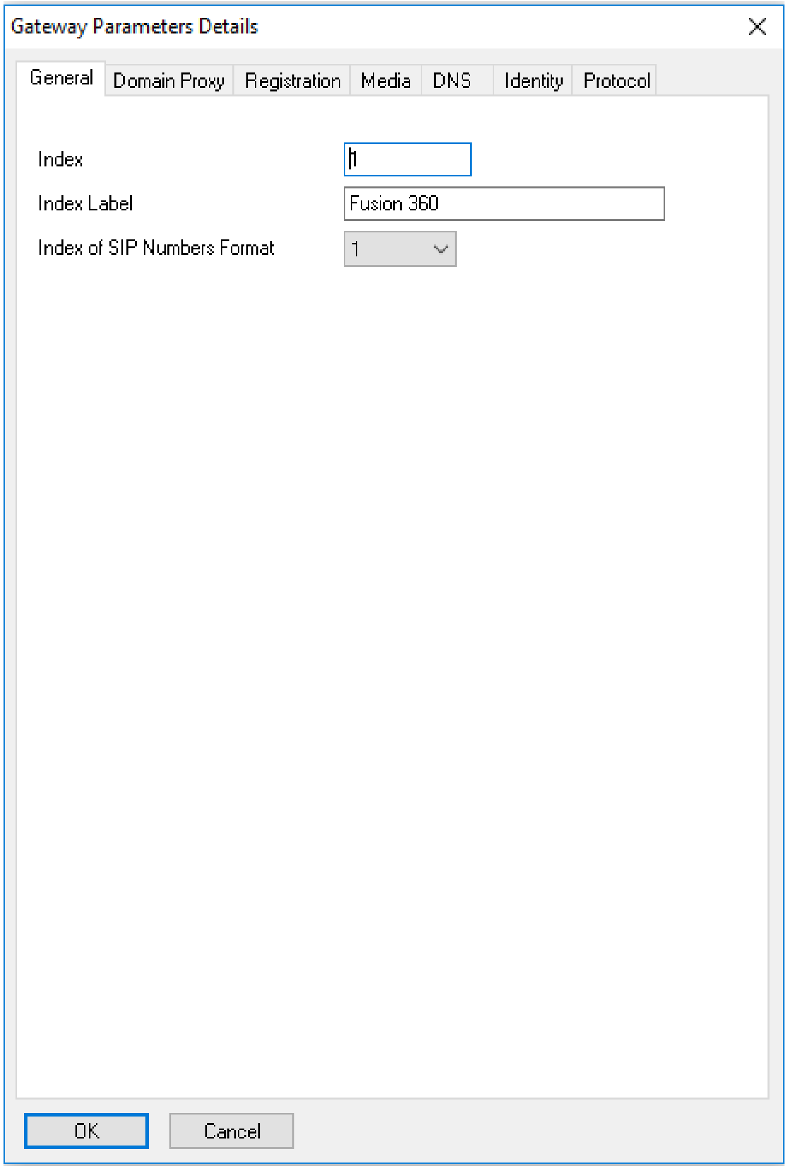
Gateway Alive Status: If the ping or options message is connecting to the carrier and responded to, then the status will be Alive. If the status is Down, then you have no connection to the carrier.

Index of Gateway Parameters: Index number for Numbering>Automatic Routing Selection>Gateway Parameters. Note: If you select "New", it will automatically open "Gateway Parameters"

Gateway Parameters:



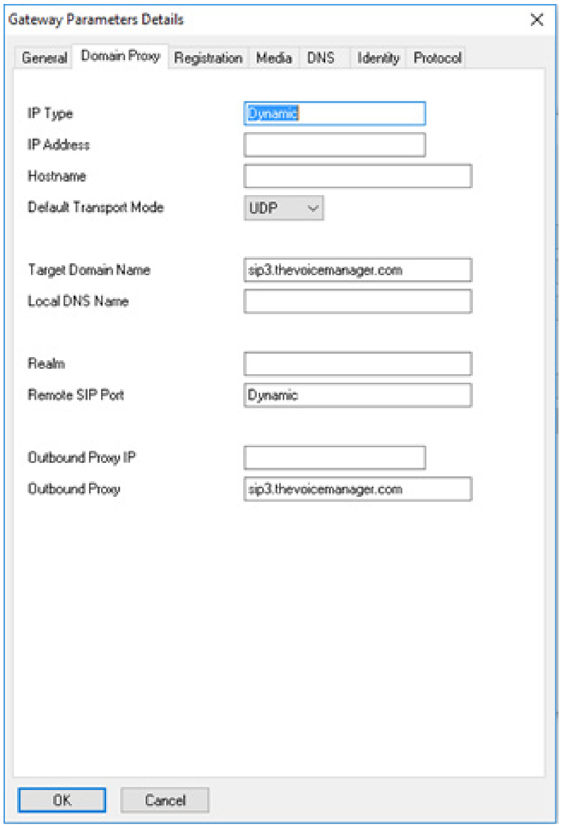
Click the Create button



Index: The same index you programmed in Automatic Routing Prefixes> Index of Gateway Parameters

Index Label: Carrier Name

Index of SIP Numbers Format: This is the index for the SIP Public Numbering category



IP Type: Dynamic (automatically set). Note: This will be set automatically when you program the DNS tab.

IP Address: <blank>

Hostname: <blank>

Default Transport Mode: UDP

Target Domain Name: Carrier's Invite (proxy) domain name

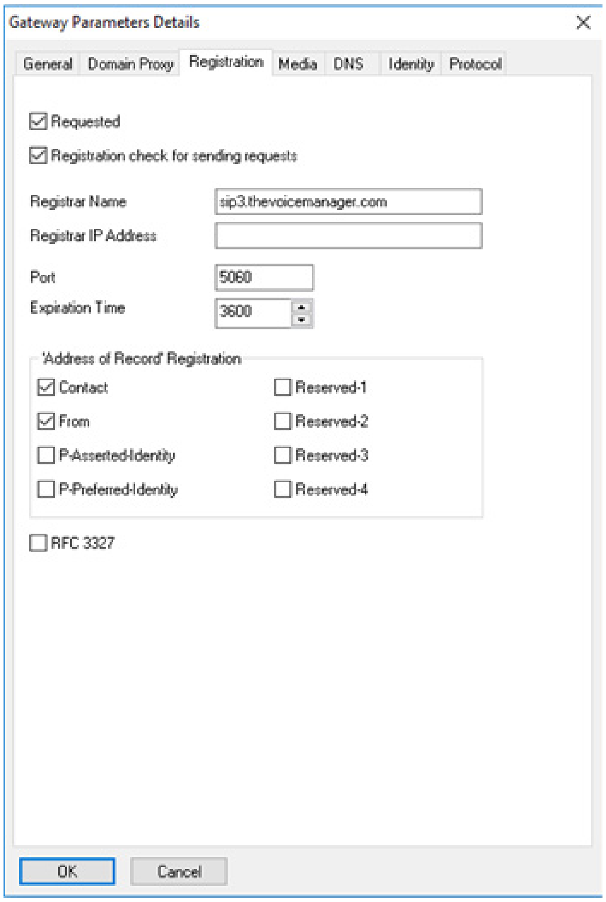
Local DNS Name: <blank>

Realm: <blank>

Remote SIP Port: Not programmable

Outbound Proxy IP: <blank>

Outbound Proxy: Carrier's Invite (proxy) domain name



Requested: Check

Registration Check for sending requests: Check

Registrar Name: Carrier's Domain Name

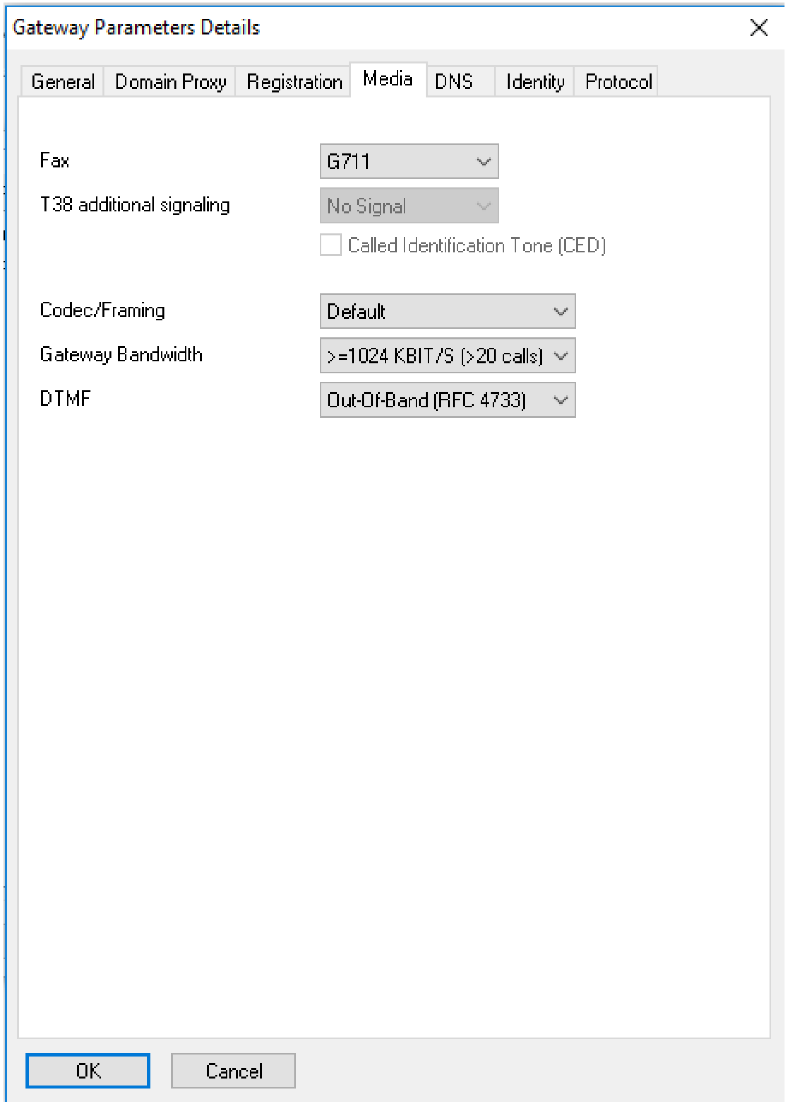
Registrar IP Address: <blank>

Port: 5060

Expiration Time: 3600

'Address of Record' Registration: Contact and From=Check

RFC 3327: Uncheck



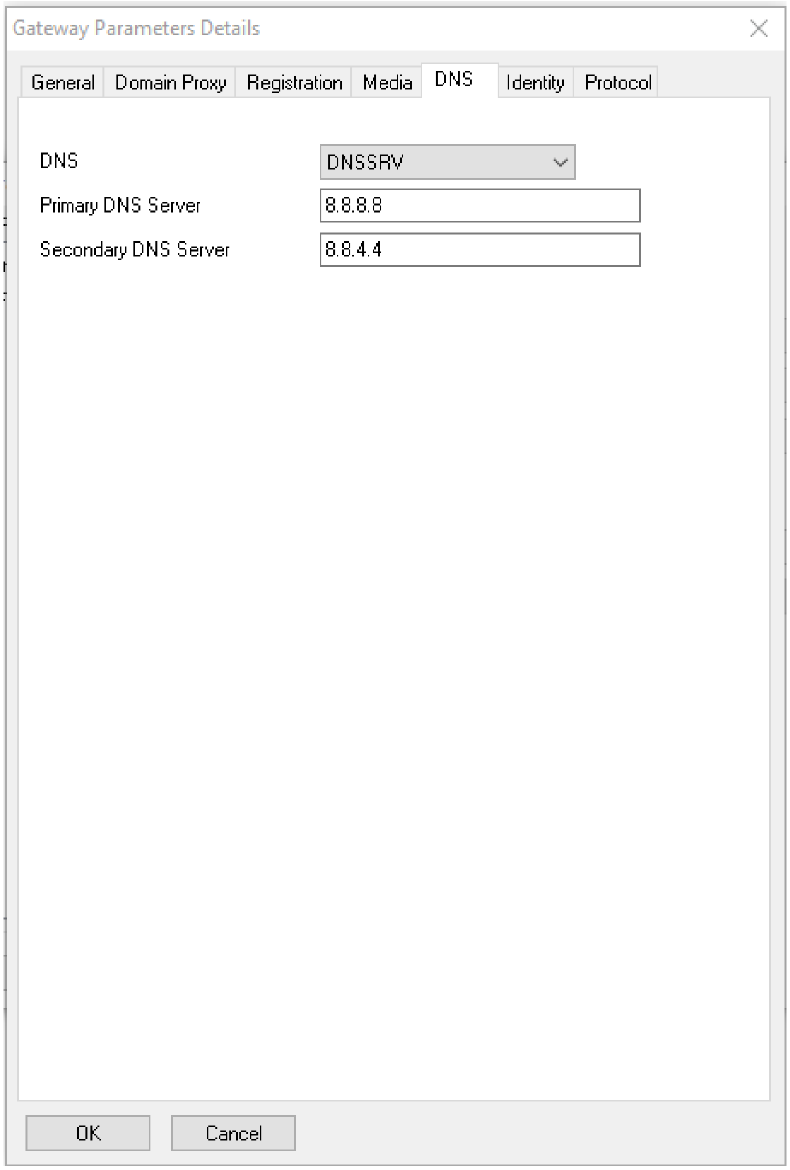
Fax: g711

T38 additional signaling: None

Codec/Framing: Default

Gateway Bandwidth: Program this to match your available bandwidth for SIP trunking.

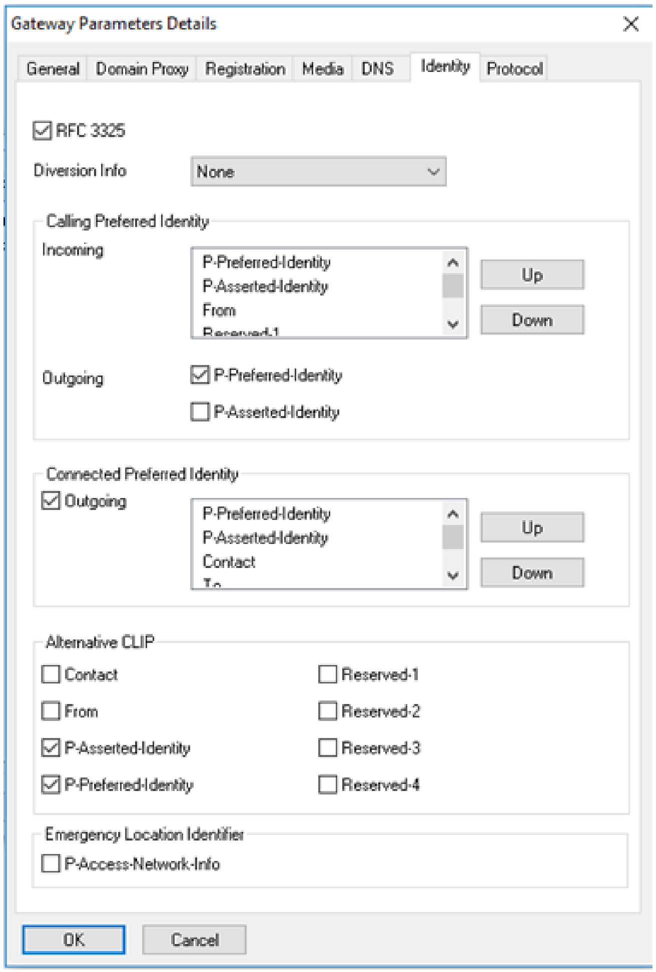
DTMF: Out-of-Band (RFC 4733)



DNS: DNSSRV

Primary DNS Server: Customer's primary DNS

Secondary DNS Server: Customer's secondary DNS



RFC 3325: Check

Diversion Info: None

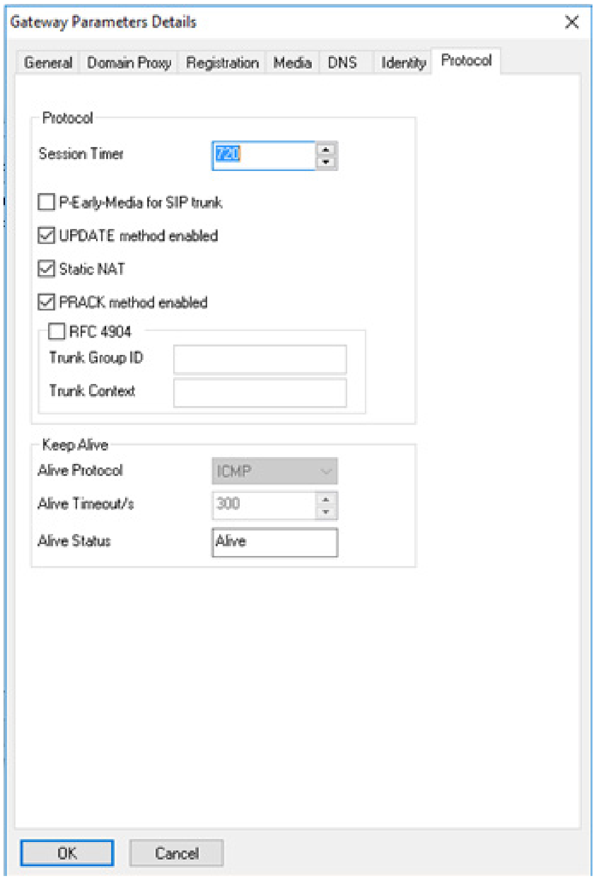
Calling Preferred Identity: Default

Outgoing P-Preferred-Identity: Check

Outgoing P-Asserted-Identity: Uncheck

Connected Preferred Identity: Default

Alternative CLIP: Contact and From=Uncheck



Session Timer: You can leave at default.

P-Early-Media for SIP trunk: Uncheck

UPDATE method enabled: Check

Static NAT: Check

PRACK method enabled: Check

Alive Protocol: "uses registration"

Alive Timeout/s: "uses registration"

Alive Status: If the ping or options message is connecting to the carrier and responded to, then the status will be Alive. If the status is Down, then you have no connection to the carrier.

SIP Accounts:

SIP Accounts

Index	Login	Password	Registered User Name	Index of Gateway Parameters	RFC 6140
1	9735551212	****************	9735551212	1 Fusion 360	Disabled

Index: Next available index number

Login: Authentication User name of the trunk

Password: Password of the trunk

Registered User Name: Register user name of the Trunk (usually the same as the Login)

Index of Gateway Parameters: Select the carrier that will use these usernames and passwords

RFC 6140: Disabled

Trunk Groups Lists:

Trunk Group Lists

List ID	Index	No.	Char	Provider/Destination	Access Digits	Auth.Code ID	Tone/Pause
1	1		S	None		None	None
2	2	400	A	None		None	None
99	Local	---	L	None		None	None

List ID: Should match the entry in “Automatic Routing: Prefixes”

Index: The trunk group number the SIP trunks are programmed in

No.: Automatically populated with the access code for the group. I used group 1 which does not have an access code in my system.

Char: The note displayed when someone makes a call out. In this case, I use “S” for SIP.

Provider/Destination: None

Access Digits: Blank

Auth. Code ID: None

Tone/Pause: None

SIP Public Numbering:

SIP Public Numbering

Index	Calling Format (Outgoing)	Calling ...	Called Format (Outgoing)	Called Prefix (Outgoing)	Called Short Prefix (Outgoing)
1	National without intercity prefix		National without intercity prefix		

Index: This is the SIP Numbers Format Index from Gateway Parameters

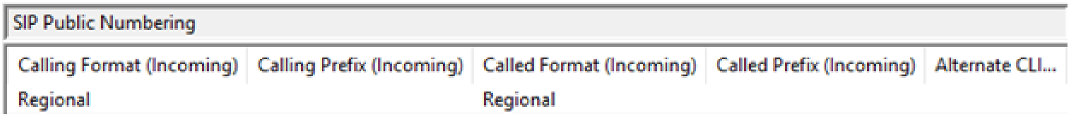
Calling Format (Outgoing): National without intercity prefix

Calling Prefix (Outgoing): <blank>

Called Format (Outgoing): National without intercity prefix

Called Prefix (Outgoing): <blank>

Called Short Prefix (Outgoing): <blank>



Calling Format (Incoming): Regional

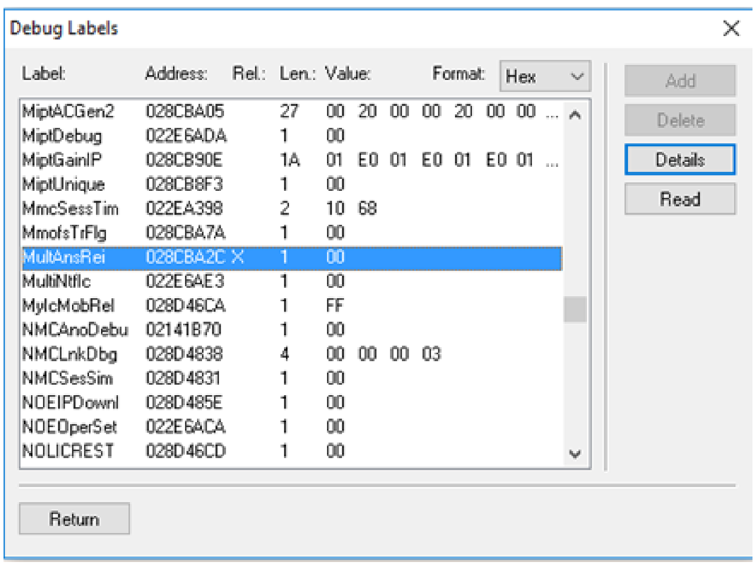
Calling Prefix (Incoming): <blank>

Called Format (Incoming): Regional

Called Prefix (Incoming): <blank>

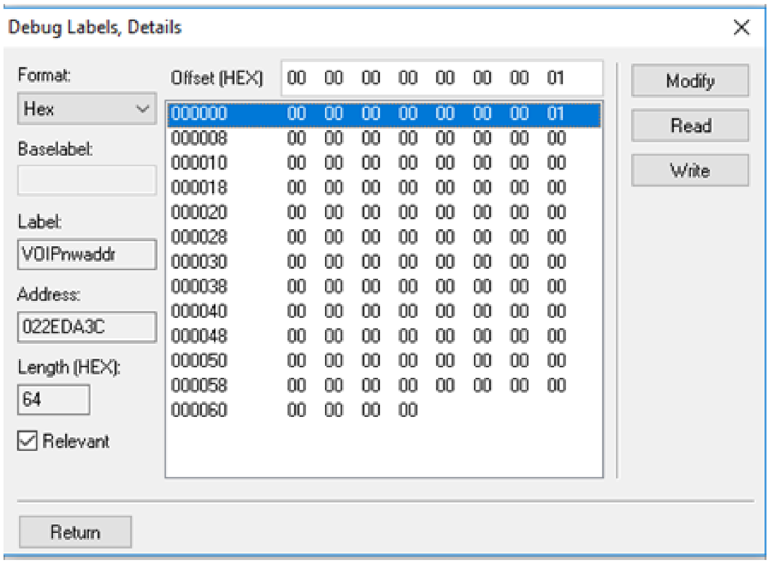
Alternative CLIP/COLP Number: Number you want to be sent as CNIS when you make a call on this trunk group. I left this blank as my carrier allows individual did numbers to be sent per station.

System Miscellaneous>Memory Read/Write>Other Labels



Set MultAnsRei to: 00

System Miscellaneous>Memory Read/Write>Debug Labels



Set VOIPnwaddr, offset 07 to: 01

Step 3: Network Programming

Signaling Port: UDP 5060

Audio Ports: UDP 32000-32255

FAX Ports: UDP 6666-6761

Public IP Address and NAT

Note: Proper port forwarding on a NAT router is the sole responsibility of the distributor / installer. Icon Voice Networks is not responsible for customer premise equipment configuration.

Port Address Translation (PAT) for audio

- Audio ports UDP 32000-32255 must be forwarded to UDP 32000-32255 at OXO's Main CPU (Voice) board IP Address.

Port Address Translation (PAT) for SIP signaling

- Signaling port UDP 5060 from carrier must be forwarded to UDP 5060 at OXO's Main CPU (Voice) board IP Address.

Public IP Address

Direct connection requires a static pubic IP address. This public IP address is programmed in Voice Over IP>VOIP: Parameters>Topology—IP Address.