

Alcatel OmniPCX Office Clearfly SIP Trunk Programming Guide

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Clearly SIP Trunk Programming Guide

SIP Trunk Programming Guide	3
Step 1: Gather the Following Information from the Carrier	3
Step 2: OXO Programming	4
Step 3: Network Programming	22

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

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 – sc.cfly.co
- username – 4065551212
- DIDs – 4065551212, 4065551213

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. Numbering>ARS>Gateway Parameters>Domain Proxy>Realm: must be <blank>
4. Numbering>ARS>Gateway Parameters>Registration>RFC 3327: check
5. Numbering>ARS>Gateway Parameters>Identity>Connected Preferred Identity: Move “To” above “Contact”.
6. Numbering>ARS>Gateway Parameters>Protocol>Update method enabled: uncheck

Step 2: OXO Programming

Hardware and Limits>Software Key Features (Licensing)

The 'Software Key Features' window displays the following features and their status:

Feature	Authorized by software key	Really activated
Call handling ISVPN service	Enabled	Enabled
Call handling QSIG+ protocol	Enabled	Enabled
B channels	120	120
IP Trunks	78	78
2 B channels for mixed boards	25	25

Verify your IP Trunks licenses are installed and activated.

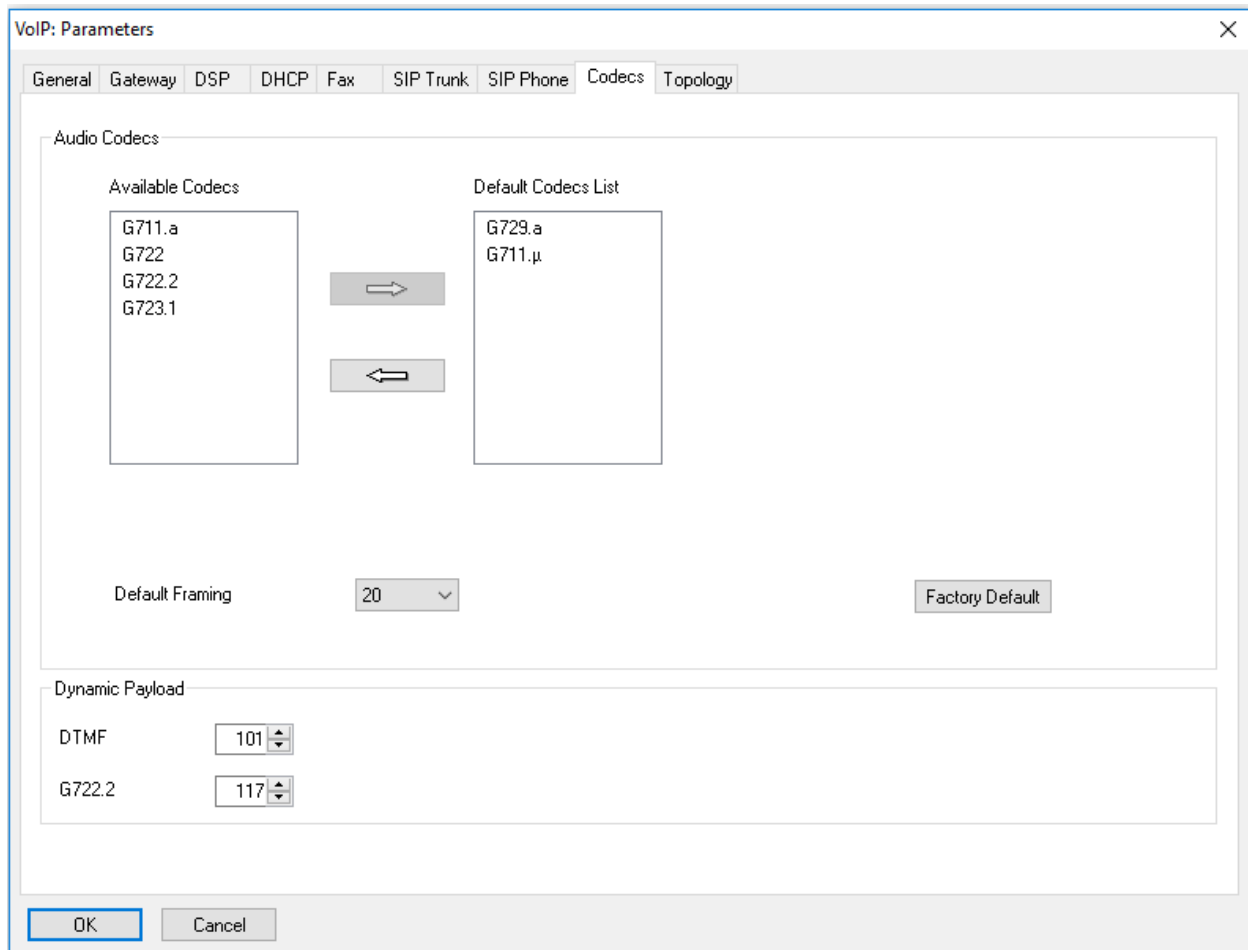
Voice Over IP>VoIP: Parameters

The 'VoIP: Parameters' window shows the following settings:

- General tab selected
- VoIP Channels mode: Multi-codecs [16]
- Number of VoIP-Trunk Channels: 4
- Number of VoIP-Subscriber Channels: 12
- IP Quality of Service: 10111000 DIFFSERV PHB EF
- VoIP Protocol: SIP
- ☐ RTP Direct
- ☐ Codec pass-through for SIP trunks
- ☐ Codec pass-through for SIP phones
- ☐ G711 codec for Music on Hold and preannouncement
- ☒ RTCP attribute in SDP

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

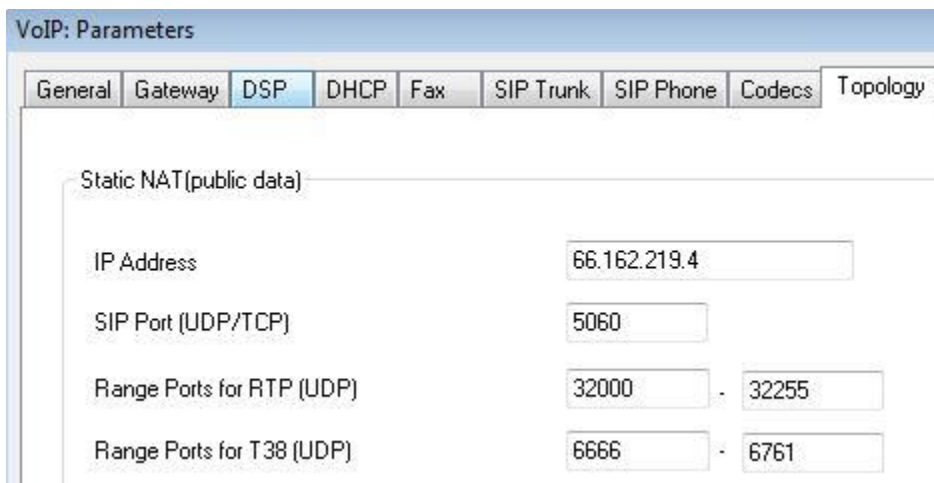
Important: RTP Direct is NOT SUPPORTED for Static NAT installations.



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.



The screenshot shows the 'VoIP: Parameters' window with the 'Static NAT(public data)' section expanded. The 'DSP' tab is selected in the top navigation bar. The configuration fields are as follows:

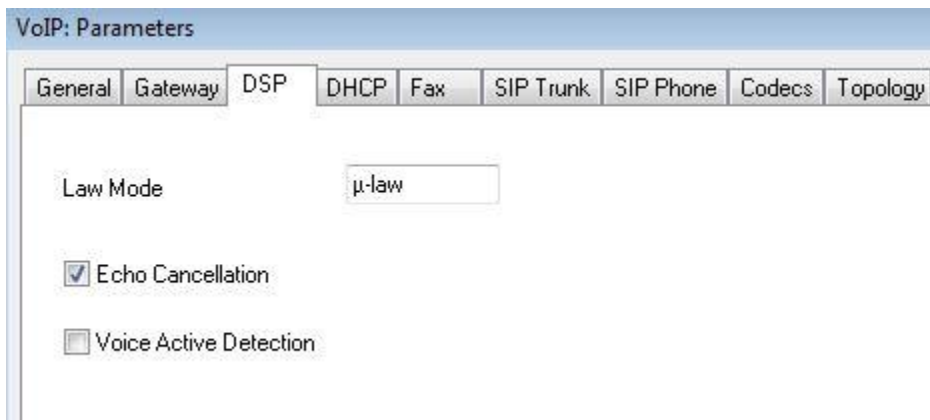
Field	Value
IP Address	66.162.219.4
SIP Port (UDP/TCP)	5060
Range Ports for RTP (UDP)	32000 - 32255
Range Ports for T38 (UDP)	6666 - 6761

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

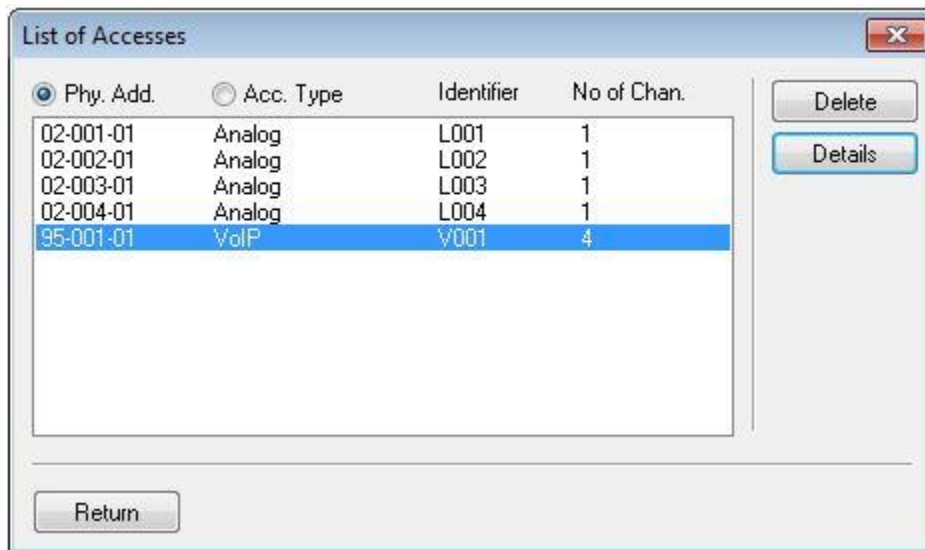


The screenshot shows the 'VoIP: Parameters' window with the 'DSP' tab selected. The configuration fields are as follows:

Field	Value
Law Mode	μ-law
Echo Cancellation	<input checked="" type="checkbox"/>
Voice Active Detection	<input type="checkbox"/>

Echo Cancellation: Checked

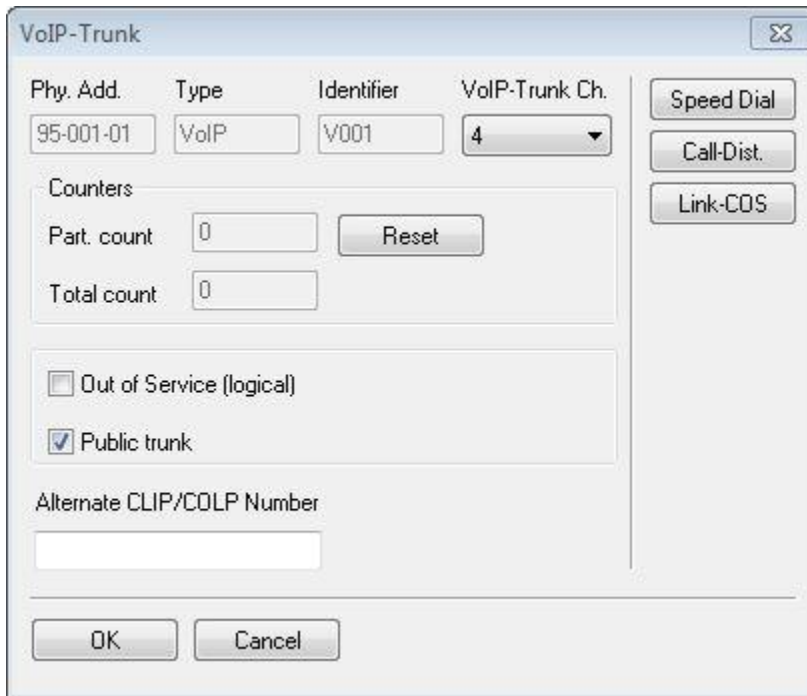
Voice Active Detection: Unchecked

External Lines>List of Accesses

The 'List of Accesses' dialog box displays a table with four columns: 'Phy. Add.', 'Acc. Type', 'Identifier', and 'No of Chan.'. The first four rows represent analog lines (02-001-01 to 02-004-01) with 1 channel each. The fifth row, '95-001-01' of type 'VoIP' with identifier 'V001', is selected and has 4 channels. To the right are 'Delete' and 'Details' buttons. A 'Return' button is at the bottom left.

Phy. Add.	Acc. Type	Identifier	No of Chan.
02-001-01	Analog	L001	1
02-002-01	Analog	L002	1
02-003-01	Analog	L003	1
02-004-01	Analog	L004	1
95-001-01	VoIP	V001	4

Select the VoIP trunks and click Details



The 'VoIP-Trunk' configuration dialog box shows settings for the selected VoIP trunk. The 'Phy. Add.' is '95-001-01', 'Type' is 'VoIP', 'Identifier' is 'V001', and 'VoIP-Trunk Ch.' is set to '4'. On the right are buttons for 'Speed Dial', 'Call-Dist.', and 'Link-COS'. The 'Counters' section has 'Part. count' and 'Total count' both at '0', with a 'Reset' button. The 'Out of Service (logical)' checkbox is unchecked, and the 'Public trunk' checkbox is checked. An 'Alternate CLIP/COLP Number' field is empty. 'OK' and 'Cancel' buttons are at the bottom.

Phy. Add.	Type	Identifier	VoIP-Trunk Ch.
95-001-01	VoIP	V001	4

Counters

Part. count: 0

Total count: 0

☐ Out of Service (logical)

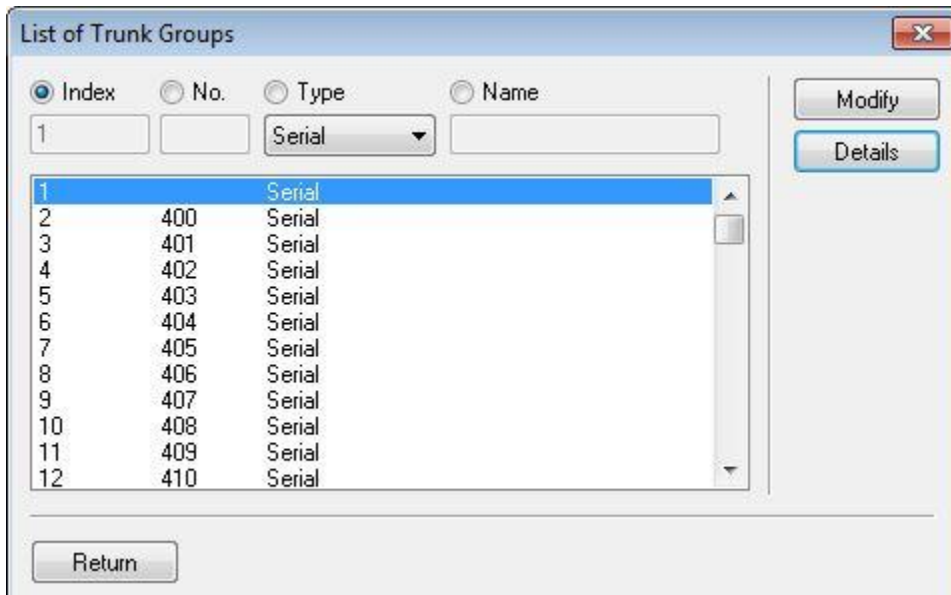
☒ Public trunk

Alternate CLIP/COLP Number

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



The 'List of Trunk Groups' dialog box displays a table of trunk groups. The 'Index' radio button is selected. The table lists 12 trunk groups, all of type 'Serial'. The first row is highlighted in blue.

Index	No.	Type	Name
1		Serial	
2	400	Serial	
3	401	Serial	
4	402	Serial	
5	403	Serial	
6	404	Serial	
7	405	Serial	
8	406	Serial	
9	407	Serial	
10	408	Serial	
11	409	Serial	
12	410	Serial	

Buttons: Return, Modify, Details.

Select the trunk group 1 and click Details



The 'Trunk Groups: Details' dialog box shows the details for trunk group 1. The 'Index' is 1, 'No.' is empty, 'Type' is 'Serial', and 'Name' is empty. Below, a table lists existing trunks. The first row is highlighted in blue. Buttons on the right allow adding, deleting, or modifying trunks, as well as changing their order or linking them to a COS.

Index	No.	Type	Name
1		Serial	

Phy. Add.	Acc. Type	Identifier	No of Chan.
02-001-01	Analog	L001	1

Buttons: Add, Delete, Modify, Up, Down, Link-COS, OK, Cancel.

Click Delete to remove your existing trunks from group 1.

Click Add to add the VoIP trunks to group 1.

Trunk Groups: Details

Index	No.	Type	Name
1		Serial	

Phy. Add.	Acc. Type	Identifier	No of Chan.
95-001-01	VoIP	V001	4

Buttons: Add, Delete, Modify, Up, Down, Link-COS

Buttons: OK, Cancel

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.

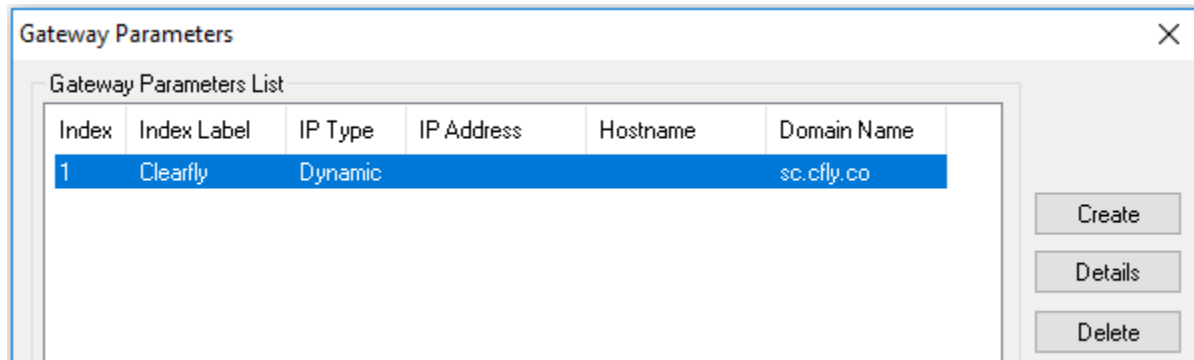
User Comment: Label for you to identify your dial tables.

Automatic Routing: Prefixes		
Destination	Gateway Alive Status	Index of Gateway Parameters
SIP Gateway	Alive	1 Clearly
SIP Gateway	Alive	1 Clearly
SIP Gateway	Alive	1 Clearly
Not IP		

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"

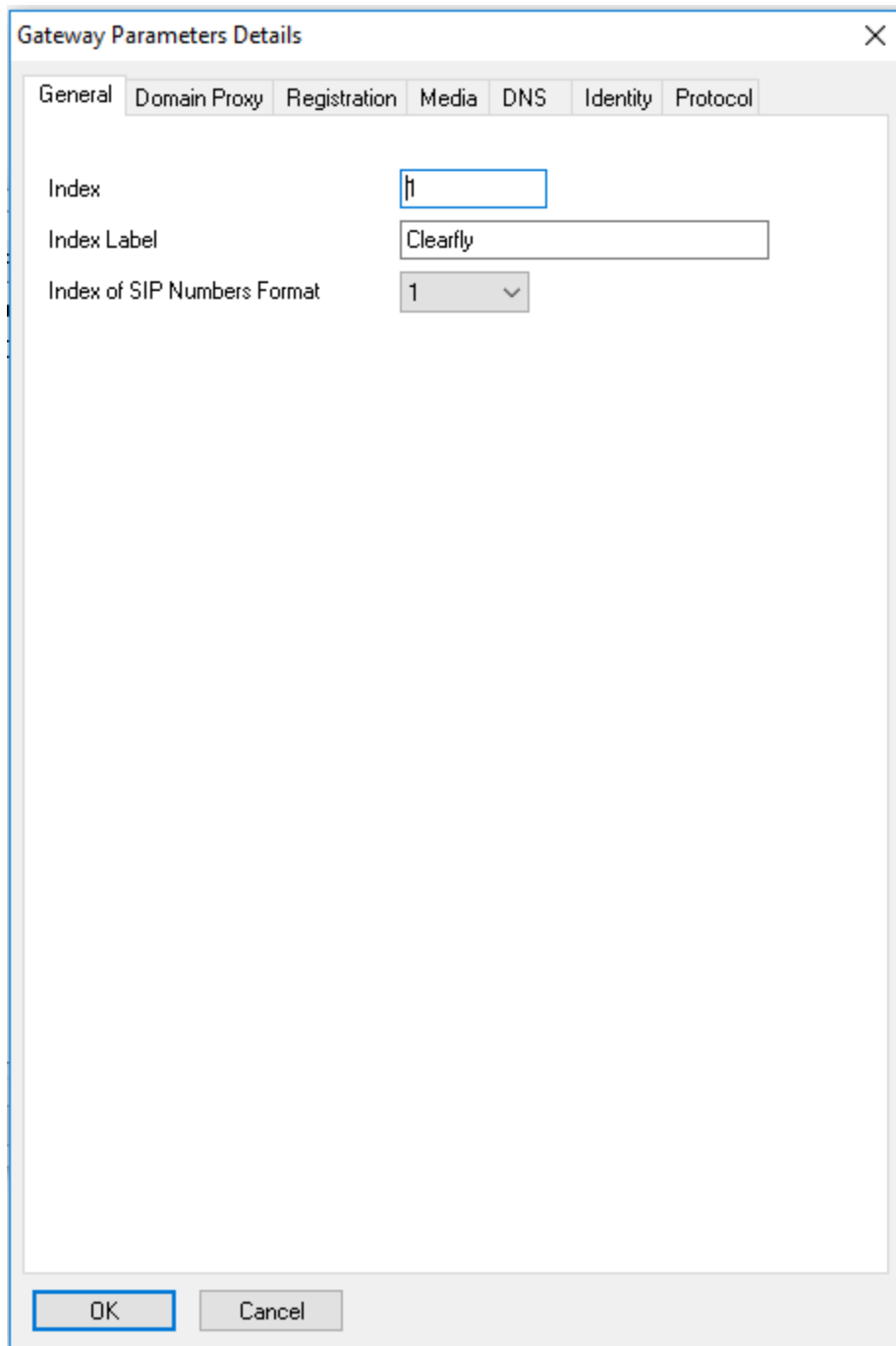
Gateway Parameters:

The image shows a 'Gateway Parameters' dialog box with a close button (X) in the top right corner. Inside the dialog, there is a section titled 'Gateway Parameters List' containing a table. The table has six columns: 'Index', 'Index Label', 'IP Type', 'IP Address', 'Hostname', and 'Domain Name'. The first row of the table is highlighted in blue and contains the values: '1', 'Clearly', 'Dynamic', an empty 'IP Address' field, an empty 'Hostname' field, and 'sc.cfly.co'. To the right of the table, there are three buttons: 'Create', 'Details', and 'Delete'.

Index	Index Label	IP Type	IP Address	Hostname	Domain Name
1	Clearly	Dynamic			sc.cfly.co

Create
Details
Delete

Click the Create button



The screenshot shows a dialog box titled "Gateway Parameters Details" with a close button (X) in the top right corner. The dialog has a tabbed interface with the following tabs: General, Domain Proxy, Registration, Media, DNS, Identity, and Protocol. The "General" tab is currently selected. Inside the dialog, there are three fields:

- Index:** A text input field containing the value "1".
- Index Label:** A text input field containing the value "Clearly".
- Index of SIP Numbers Format:** A dropdown menu with the value "1" selected.

At the bottom of the dialog, there are two buttons: "OK" and "Cancel".

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

Gateway Parameters Details

General Domain Proxy Registration Media DNS Identity Protocol

IP Type: Dynamic

IP Address:

Hostname:

Default Transport Mode: UDP

Target Domain Name: sc.cfly.co

Local DNS Name:

Realm:

Remote SIP Port: Dynamic

Outbound Proxy IP:

Outbound Proxy: sc.cfly.co

OK Cancel

IP Type: Dynamic (automatically set)

IP Address: <blank>

Hostname: <blank>

Default Transport Mode: UDP

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

Local DNS Name: System public IP address

Realm: <blank>

Remote SIP Port: 5060

Outbound Proxy IP: Carrier's Invite (proxy) IP address

Outbound Proxy: <blank>

The screenshot shows a dialog box titled "Gateway Parameters Details" with a close button (X) in the top right corner. The "Registration" tab is selected, showing various configuration options. The "General" tab is also visible. The "Registration" tab contains the following fields and options:

- ☒ Requested
- ☒ Registration check for sending requests
- Registrar Name:
- Registrar IP Address:
- Port:
- Expiration Time:
- 'Address of Record' Registration:
 - ☐ Contact
 - ☐ From
 - ☐ P-Asserted-Identity
 - ☐ P-Preferred-Identity
 - ☐ Reserved-1
 - ☐ Reserved-2
 - ☐ Reserved-3
 - ☐ Reserved-4
- ☒ RFC 3327

At the bottom of the dialog box are "OK" and "Cancel" buttons.

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: ALL unchecked

RFC 3327: check

The screenshot shows a dialog box titled "Gateway Parameters Details" with a close button (X) in the top right corner. The dialog has several tabs: "General", "Domain Proxy", "Registration", "Media" (which is selected), "DNS", "Identity", and "Protocol". The "Media" tab contains the following settings:

Fax	T38
T38 additional signaling	None
<input type="checkbox"/> Called Identification Tone (CED)	
Codec/Framing	Default
Gateway Bandwidth	>=1024 KBIT/S (>20 calls)
DTMF	Out-Of-Band (RFC 4733)

At the bottom of the dialog are "OK" and "Cancel" buttons.

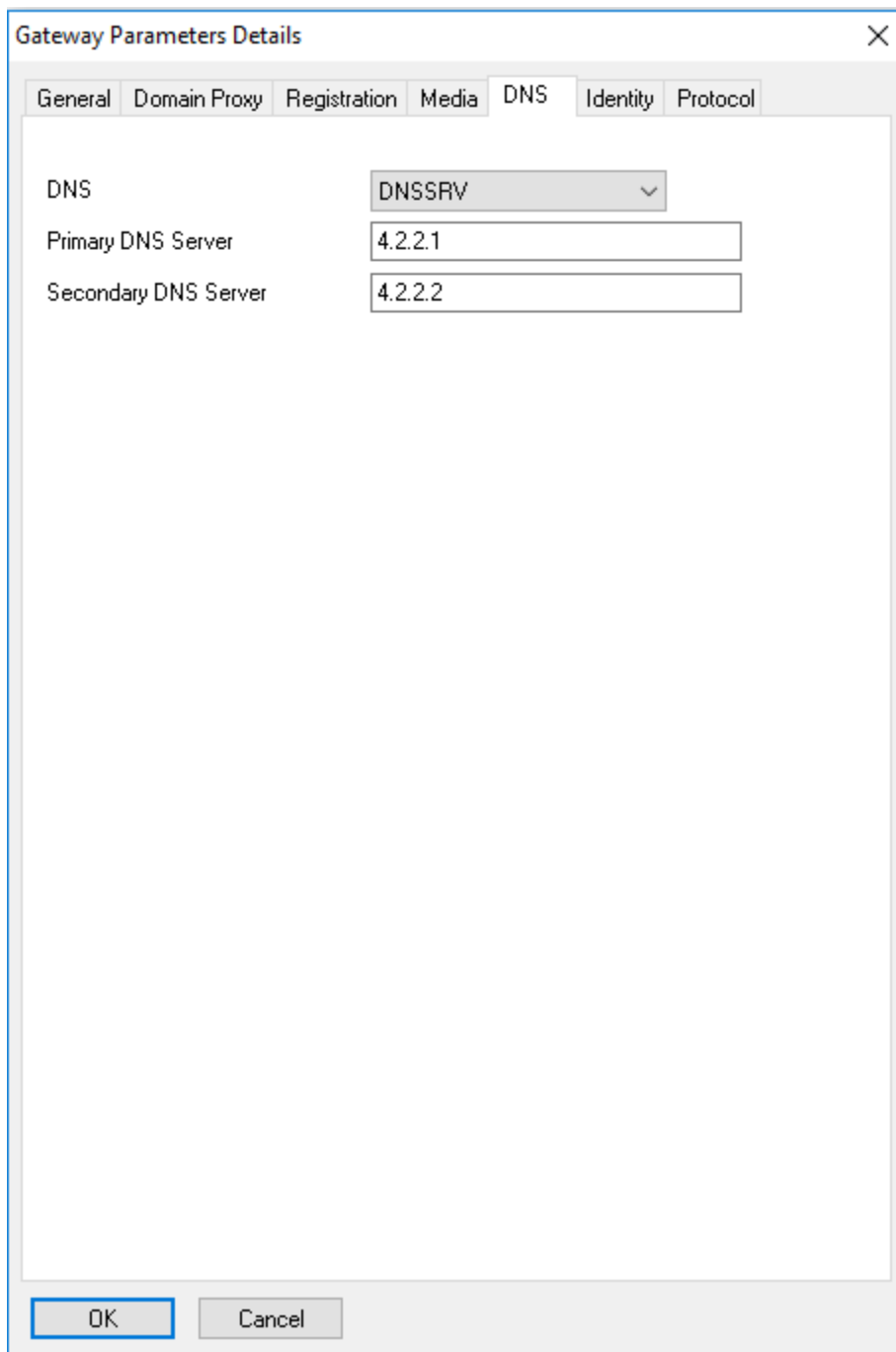
Fax: T38

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)



The screenshot shows a dialog box titled "Gateway Parameters Details" with a close button (X) in the top right corner. The dialog has several tabs: "General", "Domain Proxy", "Registration", "Media", "DNS", "Identity", and "Protocol". The "DNS" tab is currently selected. Inside the "DNS" tab, there are three fields: "DNS" with a dropdown menu showing "DNSSRV", "Primary DNS Server" with a text box containing "4.2.2.1", and "Secondary DNS Server" with a text box containing "4.2.2.2". At the bottom of the dialog, there are "OK" and "Cancel" buttons.

Field	Value
DNS	DNSSRV
Primary DNS Server	4.2.2.1
Secondary DNS Server	4.2.2.2

DNS: DNSSRV

Primary DNS Server: Customer's primary DNS

Secondary DNS Server: Customer's secondary DNS

The screenshot shows the 'Gateway Parameters Details' dialog box with the 'Identity' tab selected. The dialog has several sections for configuring SIP identity parameters.

General | **Domain Proxy** | **Registration** | **Media** | **DNS** | **Identity** | **Protocol**

☒ RFC 3325

Diversion Info:

Calling Preferred Identity

Incoming:

Up
Down

Outgoing: ☐ P-Preferred-Identity
☐ P-Asserted-Identity

Connected Preferred Identity

☒ Outgoing:

Up
Down

Alternative CLIP

<input checked="" type="checkbox"/> Contact	<input type="checkbox"/> Reserved-1
<input checked="" type="checkbox"/> From	<input type="checkbox"/> Reserved-2
<input checked="" type="checkbox"/> P-Asserted-Identity	<input type="checkbox"/> Reserved-3
<input checked="" type="checkbox"/> P-Preferred-Identity	<input type="checkbox"/> Reserved-4

Emergency Location Identifier

☐ P-Access-Network-Info

OK Cancel

RFC 3325: check

Diversion Info: Diversion

Calling Preferred Identity: Default

Outgoing P-Preferred-Identity: uncheck

Outgoing P-Asserted-Identity: uncheck

Connected Preferred Identity: Move "To" above "Contact"

The screenshot shows the 'Gateway Parameters Details' dialog box with the 'Protocol' tab selected. The 'Session Timer' is set to 720. The 'P-Early-Media for SIP trunk' checkbox is checked. The 'UPDATE method enabled' checkbox is unchecked. The 'Static NAT' checkbox is checked. The 'PRACK method enabled' checkbox is checked. The 'RFC 4904' checkbox is unchecked. The 'Trunk Group ID' and 'Trunk Context' fields are empty. The 'Keep Alive' section shows 'Alive Protocol' set to ICMP, 'Alive Timeout/s' set to 300, and 'Alive Status' set to Alive. The 'OK' and 'Cancel' buttons are at the bottom.

Parameter	Value
Session Timer	720
P-Early-Media for SIP trunk	checked
UPDATE method enabled	unchecked
Static NAT	checked
PRACK method enabled	checked
RFC 4904	unchecked
Trunk Group ID	
Trunk Context	
Alive Protocol	ICMP
Alive Timeout/s	300
Alive Status	Alive

Session Timer: You can leave at default.

P-Early-Media for SIP trunk: check

UPDATE method enabled: unchecked

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	00000000	*****	000000000	1 Clearly	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>

SIP Public Numbering				
Calling Format (Incoming)	Calling Prefix (Incoming)	Called Format (Incoming)	Called Prefix (Incoming)	Alternate CLI...
Regional		Regional		

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

Other Labels

Label:	Address:	Rel.:	Len.:	Value:	Format:
VMFwdInfo	02140426		5	02 42 34 00 00	Hex
VMUBusy	02258692		1	00	
VOIPScrInd	023F342C		1	00	
Video_Data	0213CA8C		1	01	
VipPuNuA	02583FB0 X		1	00	
VleTrfOnHk	0213CE46		1	01	
VpnEnabled	02140618		1	00	
VpnEscPref	02140619		4	00 00 00 00	
WakUpPrbRg	020D1021		1	01	
WakeUpRetr	020D0FE5		1	03	
WizMbxMode	02140786		1	01	
Z_BC_Voice	023F33B8		1	00	
audNOE_IdR	023BA5A2		A	B8 C2 CC D6 E0 EA F4 ...	
audNOE_IdS	023BA5AC		1	E6	
audNOE_rec	023BA59A		8	F0 F6 FC 02 08 0E 14 1A	

Buttons: Add, Delete, Details, Read, Return

Set VipPuNuA to: 00

Other Labels

Label:	Address:	Rel.:	Len.:	Value:	Format:
EchoTaNoi2	02141108		93	00 00 00 00 00 00 ...	Hex
EmergNum	020D104C		50	00 00 09 11 03 00 ...	
EndMFDigit	02140790		1	23	
ExtHttpsPo	0213CE30 X		2	24 E3	
ExtLnkClsd	0213CE44		1	00	
ExtNameUse	0213CDFB		1	00	
ExtNuFoVoi	0213CCDD X		1	22	
ExtNumForm	023F33B9		1	22	
FaxCRActiv	021418E2		1	00	
FaxPasCd	025AB350 X		2	1F FF	
FaxToVoic	023F33C1		1	00	
FlgIntTone	025AB1FA		1	01	
FlgRestRtp	0213C7C0		1	00	
FlgSelfCal	02140268		1	00	
FlshAsHold	02140728		1	01	

Buttons: Add, Delete, Details, Read, Return

Set ExtNuFoVoi to: 22

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 public IP address. This public IP address is programmed in Voice Over IP>VOIP: Parameters>Topology—IP Address.