

# Telstra Interop involving Ribbon QSBC and PSX Redirector

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# Document Overview

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This document outlines the configuration best practices for Telstra call flows involving Ribbon QSBC when deployed with Ribbon PSX (Redirector).

**Ribbon QSBC** is a network element deployed to protect SIP based Voice over Internet Protocol (VoIP) networks. Early deployments of SBCs were focused on the borders between two service provider networks in a peering environment. This role has now expanded to include significant deployments between a service provider's access network and a backbone network to provide service to residential and/or enterprise customers.

**Ribbon PSX** Ribbon's centralized policy and routing solution (PSX) provides a better way to manage the security, complexity, and cost of routing calls across any operator's network.

This guide contains the following sections:

- [Section A: QSBC Configuration](#)
  - Captures general QSBC configurations for Interop with PSX Redirector
- [Section B: PSX Redirector Configuration](#)
  - Captures PSX Redirector configuration required for QSBC routing



## References

For additional information on the Ribbon QSBC and Ribbon PSX - Please refer to <https://ribboncommunications.com/>

## Scope

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It is not the goal of this guide to provide detailed configurations that will meet the requirements of every customer. Use this guide as a starting point and build the QSBC, PSX product's configurations in consultation with network design and deployment engineers.

## Audience

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This is a technical document intended for telecommunications engineers with the purpose of configuring Ribbon QSBCs and PSXs. Steps will require navigating the product guide as well as the Operations guide. Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP is needed to complete the configuration and any necessary troubleshooting.



## Note

This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information in this guide.

## Product and Device Details

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The sample configuration in this document uses the following equipment and software:

**Table 1:** Requirements

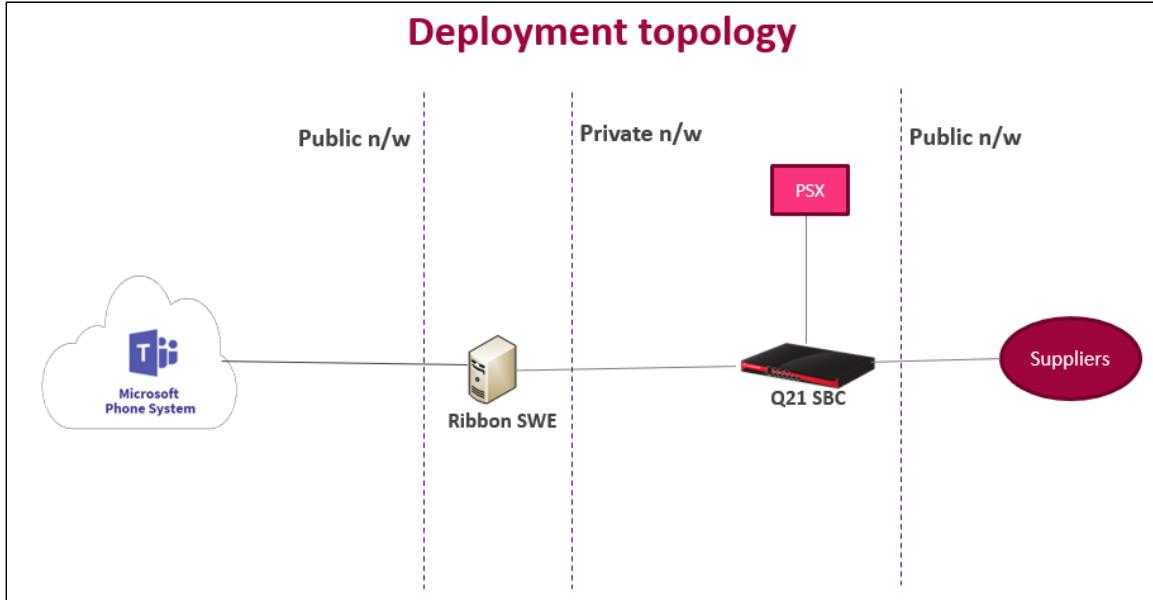
Vendor	Equipment	Software Version
Ribbon Communications	Ribbon QSBC	V9.3.11
	Ribbon PSX	V12.2.2R0

## Network Topology Diagram

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The following diagram shows the Deployment topology

## Deployment topology



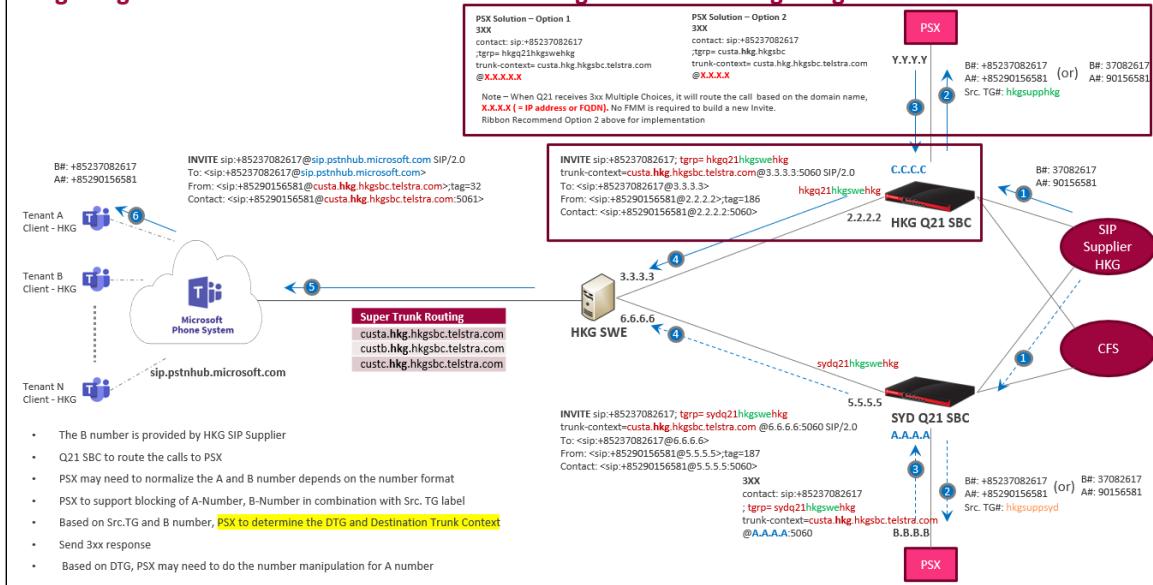
## Call Flow Diagram

The following topology diagram shows a Telstra network with a sample inbound call with QSBC & PSX (Redirector).

For the Lab network, we used Simulator instead of the HKG SWE and SIP supplier.

- i** Below diagram shows signaling flow only. Media flows are bi-directional covering all equipments in the below signaling flow except PSX (PSX doesn't handle media).
- Media flows not shown explicitly for simplicity.

### Hong Kong SIP Connect Service – Inbound call – Originated from Hong Kong

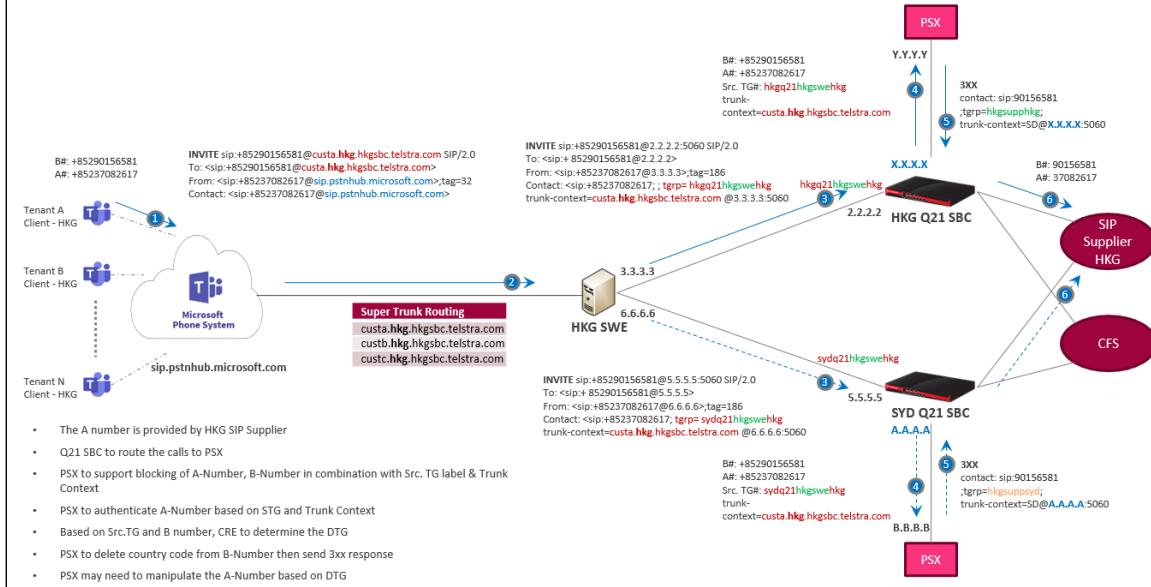


The following topology diagram shows the Telstra network with a sample outbound call with QSBC & PSX (Redirector).

**i** Below diagram shows signaling flow only. Media flows are bi-directional covering all equipments in the below signaling flow except PSX (PSX doesn't handle media).

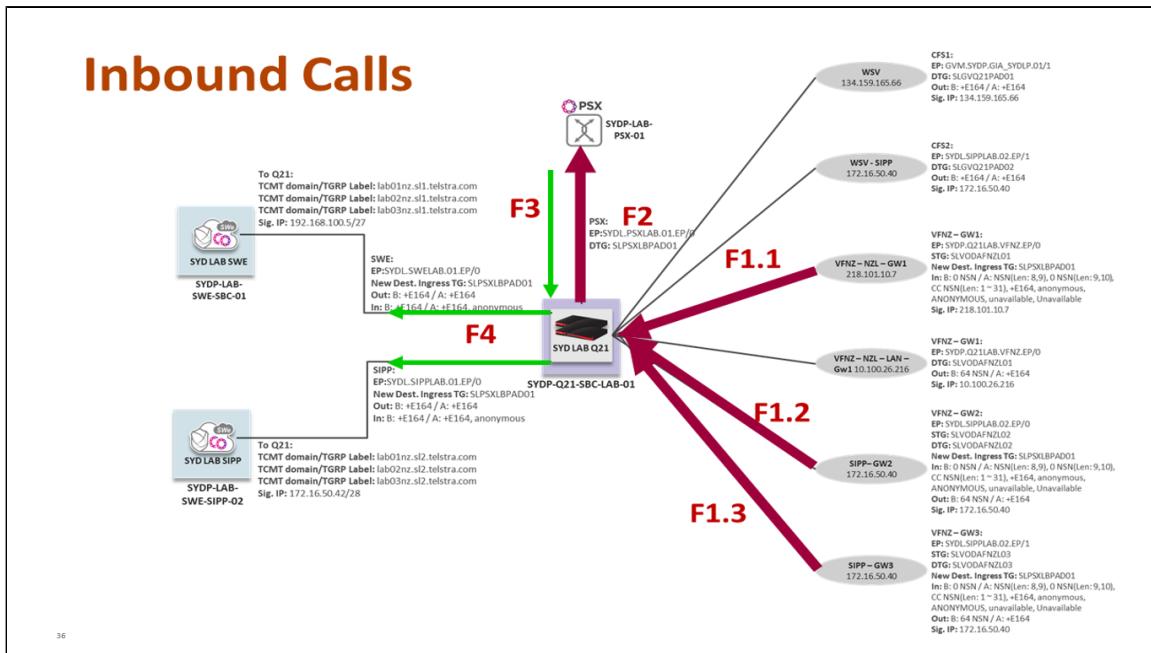
Media flows not shown explicitly for simplicity.

### Hong Kong SIP Connect Service – Outbound call – Terminated to Hong Kong



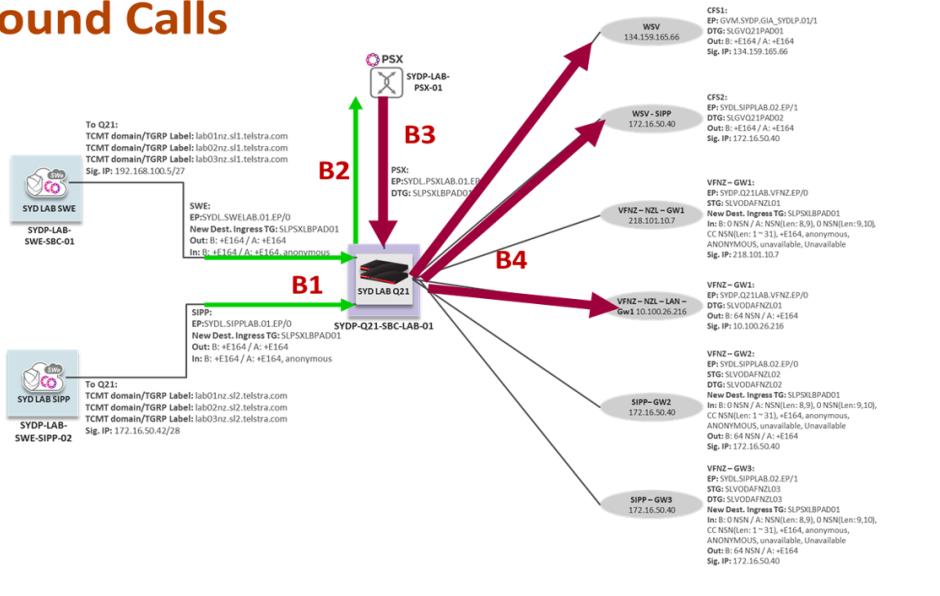
## Section A: QSBC Configuration

The following diagram shows the Q21 Routing Design for Inbound calls



The following diagram shows the Q21 Routing Design for Outbound calls

# Outbound Calls



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## Q21 Configuration



### Note

This section may be modified based on customer specific requirements

## VNET Configuration

A signaling VNET (virtual network) is the combination of a physical interface, a gateway IP address (router), and an optional VLAN (virtual local area network) ID.

```
cli vnet add v1
cli vnet edit v1 gateway 10.34.92.1 ifname eth2
cli vnet add v2
cli vnet edit v2 gateway 10.34.94.1 ifname eth3
```



### Note

Configure the QSBC Media separately.

## REALM Configuration

Realm is a logical service entry point for other devices to connect to the SBC. (A realm on the SBC is not synonymous with a SIP realm.)

For calls to connect successfully, you must have at least one realm configured on your system.

These refer to SIP Signaling IPs of QSBC towards various peers including SIP suppliers and SWe for Telstra.

```

cli realm add SYDPQ21LABVFNZREALM
cli realm edit SYDPQ21LABVFNZREALM vnet v1 rsa 10.34.92.166 mask 255.255.254.0 emr alwayson imr alwayson medpool 1

cli realm add GVPSSYDPQFLEXLAN394REALM
cli realm edit GVPSSYDPQFLEXLAN394REALM vnet v1 rsa 10.34.92.167 mask 255.255.254.0 emr alwayson imr alwayson medpool 1

cli realm add GVPLABPMPLSGVOIPCUSTREALM
cli realm edit GVPLABPMPLSGVOIPCUSTREALM vnet v1 rsa 10.34.92.168 mask 255.255.254.0 emr alwayson imr alwayson medpool 1

cli realm add LABGIA210REALM
cli realm edit LABGIA210REALM vnet v1 rsa 10.34.92.169 mask 255.255.254.0 emr alwayson imr alwayson medpool 1

cli realm add TCISYDLSWELABREALM
cli realm edit TCISYDLSWELABREALM vnet v2 rsa 10.34.94.166 mask 255.255.254.0 emr alwayson imr alwayson medpool 2

cli realm add GVPLABPMPLSGVOIPCUSTREALM2
cli realm edit GVPLABPMPLSGVOIPCUSTREALM2 vnet v2 rsa 10.34.94.167 mask 255.255.254.0 emr alwayson imr alwayson medpool 2

cli realm add PSX
cli realm edit PSX vnet v2 rsa 10.34.94.168 mask 255.255.254.0 emr alwayson imr alwayson medpool 2

```

## **End Point Configuration**

PSX EndPoints (EPs) to send INVITE & receive 300 response

```

cli iedge add SYDLPSXLAB01EP 0
cli iedge edit SYDLPSXLAB01EP 0 realm PSX static 10.54.181.13 sip enable type siproxy dtg SLPSXLBPAD01 use4904tg
disable setdesttg disable vendor generic

cli iedge add SYDLPSXLAB01EP 1
cli iedge edit SYDLPSXLAB01EP 1 realm PSX static 10.54.181.13 sip enable type siproxy use4904tg disable setdesttg
disable vendor generic cp PSX01CP

```

## **SIP Supplier EPs**

F1.1 EP for PSX dedicate Realm (Existing EP) REALM: 10.54.92.166

```

cli iedge add SYDPQ21LABVFNZEP 0
cli iedge edit SYDPQ21LABVFNZEP 0 realm SYDPQ21LABVFNZREALM static 10.54.81.11 sip enable type sipgw cp TEAMS01CP
newsrctdg SLPSXLBPAD01 use4904tg disable vendor generic
cli iedge edit SYDPQ21LABVFNZEP 0 newsrcitg "SLVODAFNZN01"

```

B4-GW1 (Existing EP) REALM: 10.54.92.167 with 4904 disabled

```

cli iedge add SYDPQ21LABVFNZEP 1
cli iedge edit SYDPQ21LABVFNZEP 1 realm GVPSSYDPQFLEXLAN394REALM static 10.54.81.11 contact 10.54.81.11:8012 sip
enable type sipgw dtg SLVODAFNZN01 tg SLVODAFNZN01 use4904tg disable vendor generic

```

F1.2 EP for Selection of Ingress via DNIS REALM: 10.54.92.168

```

cli iedge add SYDLSIPLAB02EP 0
cli iedge edit SYDLSIPLAB02EP 0 realm GVPLABPMPLSGVOIPCUSTREALM static 10.54.81.12 contact 10.54.81.12:8012 sip
enable type sipgw cp TEAMS02CP dtg SLVODAFNZN02 tg SLVODAFNZN02 use4904tg disable setdesttg enable vendor generic

```

F1.3 EP for Selection of Ingress via sourceTG=SLVODAFNZN03 (REALM: 10.54.92.168) with 4904 enabled

```
cli iedge add SYDLSIPLAB02EP 1
cli iedge edit SYDLSIPLAB02EP 1 realm GVPLABPMPMLSGVOIPCUSTREALM static 10.54.81.12 contact 10.54.81.12:8014 sip
enable type sipgw newsrcdtg SLPSXLBPAD01 dtg SLVODAFNZL03 tg SLVODAFNZL03 use4904tg enable vendor generic
```

#### B4-CSF2

```
cli iedge add SYDLSIPLAB02EP 2
cli iedge edit SYDLSIPLAB02EP 2 realm GVPLABPMPMLSGVOIPCUSTREALM static 10.54.81.12 sip enable type sipgw dtg
SLGVQ21PAD02 use4904tg disable vendor generic
```

#### B4-CSF1 (Existing EP)

```
cli iedge add GVMSPYDPGIASYDLP01 1
cli iedge edit GVMSPYDPGIASYDLP01 1 realm LABGIA210REALM static 10.54.81.13 sip enable type sipgw dtg SLGVQ21PAD01
use4904tg disable vendor generic
```

#### F4, B1 (EP for 3XX to LAB SWE1)

```
cli iedge add SYDLSWELAB01EP 0
cli iedge edit SYDLSWELAB01EP 0 realm PSX static 10.54.81.66 sip enable type sipgw newsrcdtg SLPSXLBPAD01
use4904tg enable vendor generic setdesttg enable URI 10.54.81.66
```

#### F4-B1 (EP for 3XX to LAB SWE2)

```
cli iedge add SYDLSWELAB02EP 0
cli iedge edit SYDLSWELAB02EP 0 realm TCISYDLSWELABREALM static 10.54.81.67 sip enable type sipgw newsrcdtg
SLPSXLBPAD01 use4904tg enable vendor generic setdesttg enable URI 10.54.81.67
```

## Sample Calling Plans

### INGRESS FROM SUPPLIER CALLING PLAN

```
cli cr add TEAMS09980315INGCR
cli cr edit TEAMS09980315INGCR calltype origin dest 09980315 prefix 64*09980315
cli cp add TEAMS02CP TEAMS09980315INGCR
cli iedge edit SYDLSIPLAB02EP 0 cp TEAMS02CP
```

### EGRESS TO PSX CALLING PLAN

(For 64\* prefix numbers routed to PSX redirector)

```
cli cr add TEAMS09980315EGRCR
cli cr edit TEAMS09980315EGRCR calltype dest dest 64*09980315 prefix 0998031512345
cli cp add PSX01CP TEAMS09980315EGRCR
cli iedge edit SYDLPSXLAB01EP 1 cp PSX01CP
```

### Link Calling Plan with MULTIPLE Calling Route

```

cli cr add TEAMS09980316INGCR
cli cr edit TEAMS09980316INGCR calltype origin dest 09980316 prefix 09980316
cli cp add TEAMS01CP TEAMS09980316INGCR

cli cr add TEAMS0998032INGCR
cli cr edit TEAMS0998032INGCR calltype origin dest 0998032 prefix 0998032
cli cp add TEAMS01CP TEAMS0998032INGCR

cli cr add TEAMS099803301INGCR
cli cr edit TEAMS099803301INGCR calltype origin dest 099803301 prefix 099803301
cli cp add TEAMS01CP TEAMS099803301INGCR

cli cr add TEAMS099803302INGCR
cli cr edit TEAMS099803302INGCR calltype origin dest 099803302 prefix 099803302
cli cp add TEAMS01CP TEAMS099803302INGCR

cli cr add TEAMS0998034INGCR
cli cr edit TEAMS0998034INGCR calltype origin dest 0998034 prefix 0998034
cli cp add TEAMS01CP TEAMS0998034INGCR
cli iedge edit SYDPQ21LABVFNZEP 0 cp TEAMS01CP

```

## **FMM Rules Manipulation**

Use the Flexible Message Manipulation (FMM) feature to manipulate SIP messages and SIP message contents in QSBC.

FMM rules are required for Telstra to modify the digits in the "From" Header & "P-Asserted-Id" Header and to change the domain name in the From header to RSA (Realm Signaling IP Address)

See below for the FMM rules implemented for Telstra.

### **1. FMM rule for modifying From Header**

#### **Trigger to check FROM with 'user' part**

```

cli fmm trigger add from-available-t sip-header
cli fmm trigger edit from-available-t method is("INVITE")
cli fmm trigger edit from-available-t msg.type is("request")
cli fmm trigger edit from-available-t header.name is("From")
cli fmm trigger edit from-available-t uri userinfo.user is-phone-number() and not starts-with( "+")

```

#### **Trigger to check FROM header starting with 0 and having a size of 9-10 digit**

```

cli fmm trigger add from-remove-zero-t sip-header
cli fmm trigger edit from-remove-zero-t method is("INVITE")
cli fmm trigger edit from-remove-zero-t msg.type is("request")
cli fmm trigger edit from-remove-zero-t header.name is("From")
cli fmm trigger edit from-remove-zero-t uri userinfo.user starts-with("0") and is-match("^[0-9]{9,10}$")

```

#### **Trigger to check FROM header starting with anonymous or unavailable**

```

cli fmm trigger add from-anon-t sip-header
cli fmm trigger edit from-anon-t method is("INVITE")
cli fmm trigger edit from-anon-t msg.type is("request")
cli fmm trigger edit from-anon-t header.name is("From")
cli fmm trigger edit from-anon-t uri userinfo.user (contains(ci("anonymous")) or contains(ci("unavailable")))

```

#### **Trigger to check FROM header starting without 0 and having digit size 8-9**

```
cli fmm trigger add from-not-start-zero-t sip-header
cli fmm trigger edit from-not-start-zero-t method is("INVITE")
cli fmm trigger edit from-not-start-zero-t msg.type is("request")
cli fmm trigger edit from-not-start-zero-t header.name is("From")
cli fmm trigger edit from-not-start-zero-t uri userinfo.user not starts-with("0") and is-match("^0-9{8,9}$")
```

#### Action to replace 0 with +64

```
cli fmm action add from-remove-zero-a modify
cli fmm action edit from-remove-zero-a from-remove-zero-t.uri userinfo.user substitute(from-remove-zero-t.uri.
userinfo.user,"^0","+64")
```

#### Action to change From to anonymous

```
cli fmm action add from-anon-a modify
cli fmm action edit from-anon-a from-anon-t.uri userinfo.user "anonymous"
```

#### Action to add +64

```
cli fmm action add from-not-start-zero-a modify
cli fmm action edit from-not-start-zero-a from-not-start-zero-t.uri userinfo.user "+64"+from-not-start-zero-t.uri.
userinfo.user
```

#### Action to add +

```
cli fmm action add from-available-a modify
cli fmm action edit from-available-a from-available-t.uri userinfo.user "+"+from-available-t.uri userinfo.user
```

#### Rule to replace 0 with +64

```
cli fmm rule add from-remove-zero-r
cli fmm rule edit from-remove-zero-r condition from-remove-zero-t actions from-remove-zero-a
```

#### Rule to change From to anonymous

```
cli fmm rule add from-anon-r
cli fmm rule edit from-anon-r condition from-anon-t actions from-anon-a
```

#### Rule to add +64

```
cli fmm rule add from-not-start-zero-r
cli fmm rule edit from-not-start-zero-r condition from-not-start-zero-t actions from-not-start-zero-a
```

#### Rule to add +

```
cli fmm rule add from-available-r
cli fmm rule edit from-available-r condition (from-available-t and (not from-remove-zero-t) and (not from-not-
start-zero-t) and (not from-anon-t))
cli fmm rule edit from-available-r actions from-available-a
```

#### Profile to normalize FROM header

```
cli fmm profile add from-remove-zero-p
cli fmm profile edit from-remove-zero-p rules from-remove-zero-r,from-anon-r,from-not-start-zero-r,from-available-r
```

## **2. FMM for modifying PAID header**

### **Trigger to check PAID with user part**

```
cli fmm trigger add paid-available-t sip-header
cli fmm trigger edit paid-available-t method is("INVITE")
cli fmm trigger edit paid-available-t msg.type is("request")
cli fmm trigger edit paid-available-t header.name is("P-Asserted-Identity")
cli fmm trigger edit paid-available-t uri userinfo.user is-phone-number() and not starts-with("+")
```

### **Trigger to check PAID header starting with 0 and have size of 9-10 digit**

```
cli fmm trigger add paid-remove-zero-t sip-header
cli fmm trigger edit paid-remove-zero-t method is("INVITE")
cli fmm trigger edit paid-remove-zero-t msg.type is("request")
cli fmm trigger edit paid-remove-zero-t header.name is("P-Asserted-Identity")
cli fmm trigger edit paid-remove-zero-t uri userinfo.user starts-with("0") and is-match("^[0-9]{9,10}$")
```

### **Trigger to check PAID header starting with anonymous or unavailable**

```
cli fmm trigger add paid-anon-t sip-header
cli fmm trigger edit paid-anon-t method is("INVITE")
cli fmm trigger edit paid-anon-t msg.type is("request")
cli fmm trigger edit paid-anon-t header.name is("P-Asserted-Identity")
cli fmm trigger edit paid-anon-t uri userinfo.user (contains(ci("anonymous")) or contains(ci("unavailable")))
```

### **Trigger to check PAID header starting without 0 and having digit size 8-9**

```
cli fmm trigger add paid-not-start-zero-t sip-header
cli fmm trigger edit paid-not-start-zero-t method is("INVITE")
cli fmm trigger edit paid-not-start-zero-t msg.type is("request")
cli fmm trigger edit paid-not-start-zero-t header.name is("P-Asserted-Identity")
cli fmm trigger edit paid-not-start-zero-t uri userinfo.user not starts-with("0") and is-match("^[0-9]{8,9}$")
```

### **Action to replace 0 with +64**

```
cli fmm action add paid-remove-zero-a modify
cli fmm action edit paid-remove-zero-a paid-remove-zero-t.uri userinfo.user substitute(paid-remove-zero-t.uri.
userinfo.user,"^[0]","+64")
```

### **Action to change PAID to anonymous**

```
cli fmm action add paid-anon-a modify
cli fmm action edit paid-anon-a paid-anon-t.uri userinfo.user "anonymous"
```

### **Action to add +64**

```
cli fmm action add paid-not-start-zero-a modify
cli fmm action edit paid-not-start-zero-a paid-not-start-zero-t.uri userinfo.user "+64"+paid-not-start-zero-t.uri.
userinfo.user
```

### **Action to add +**

```
cli fmm action add paid-available-a modify
cli fmm action edit paid-available-a paid-available-t.uri userinfo.user "+"+paid-available-t.uri userinfo.user
```

### **Rule to replace 0 with +64**

```
cli fmm rule add paid-remove-zero-r  
cli fmm rule edit paid-remove-zero-r condition paid-remove-zero-t actions paid-remove-zero-a
```

#### Rule to change PAID to anonymous

```
cli fmm rule add paid-anon-r  
cli fmm rule edit paid-anon-r condition paid-anon-t actions paid-anon-a
```

#### Rule to add +64

```
cli fmm rule add paid-not-start-zero-r  
cli fmm rule edit paid-not-start-zero-r condition paid-not-start-zero-t actions paid-not-start-zero-a
```

#### Rule to add +

```
cli fmm rule add paid-available-r  
cli fmm rule edit paid-available-r condition (paid-available-t and (not paid-remove-zero-t) and (not paid-not-start-zero-t) and (not paid-anon-t))  
cli fmm rule edit paid-available-r actions paid-available-a
```

#### Profile to normalize PAID header

```
cli fmm profile add paid-remove-zero-p  
cli fmm profile edit paid-remove-zero-p rules paid-remove-zero-r,paid-anon-r,paid-not-start-zero-r,paid-available-r
```

#### For importing the above fmm rule in Q21:

```
cli fmm import <fmm file name>
```

#### For applying the above FMM profile to endpoint:

```
cli iedge edit SYDPQ21LABVFNZEP 0 fmm-ingress-profile from-remove-zero-p,paid-remove-zero-p
```

### 3. FMM rule to change domain to RSA IP address in FROM header

The Q SBC will send FROM with domain uri, but PSX should receive RSA (Realm Signaling IP address).

```
cli fmm trigger add sbc-modify-request-t sip-header  
cli fmm trigger edit sbc-modify-request-t method is("INVITE")  
cli fmm trigger edit sbc-modify-request-t msg.type is("request")  
cli fmm trigger edit sbc-modify-request-t header.name is(ci("From"))  
  
cli fmm action add sbc-modify-request-al modify  
cli fmm action edit sbc-modify-request-al sbc-modify-request-t.uri.hostport var.dst-realm.sbc-rsa  
  
cli fmm rule add sbc-modify-request-rl  
cli fmm rule edit sbc-modify-request-rl condition sbc-modify-request-t  
cli fmm rule edit sbc-modify-request-rl actions sbc-modify-request-al  
  
cli fmm profile add sbc-modify-request-pl  
cli fmm profile edit sbc-modify-request-pl rules sbc-modify-request-rl
```

#### Apply above FMM profile to PSX endpoints

```
cli iedge edit SYDLPSSLAB01EP 0 fmm-egress-profile sbc-modify-request-pl  
cli iedge edit SYDLPSSLAB01EP 1 fmm-egress-profile sbc-modify-request-pl
```

## **General Recommendation - Q21 Configuration**

- Create as many EPs as required (as shown above) including EPs (type SIPGW) towards SIP Suppliers & Core SBC SWEs, and EPs (type SIPPROXY) towards PSX Redirector
- Create media related config on QSBC and enable generation of QOS parameters on QSBC CDRs with command "[nxconfig.pl -e mqm -v 1](#)"
- For ingress Trunk group based routing, refer to "42: Trunk group Support" in the release 9.4 QSBC Operations guide
- For sending TGRP, DTG, OTG parameters in the egress INVITE, refer to "42.7: Support for trunk group context and RFC 4904" and "42.4: Configuring trunk group endpoint options" in the release 9.4 QSBC Operations guide
- Refer to the QSBC Operations guide for any other specific or basic configuration

## **Section B: PSX Configuration**

### **Redirector related configuration**

- The PSX requires a SIP server entry created with QSBC SIP IP towards PSX side.
- The PSX requires a Trunk Group created with SIP server created from the above step. If this step is missed, the PSX picks the default SIP server entry "DEFAULTSIPSERVER".
- The PSX Trunk Group requires a Feature Control Profile with "IP Protocol Flags" enabled, "Support Domain Name in 300 Contact" enabled, and PSX processing mode set to "Redirector".
- If required, enable the following Feature control profile options: "Process TGRP", "Process Trunk-context", and "Process Originating Trunk Group And Trunk-context over OTG".
- The PSX Trunk Group requires an IP Signaling profile with the "Destination Trunk Group Options" set to "Include Tgrp with domain name" based on customer requirement.
- If required, the IP Signaling profile can have the "Originating Trunk Group Options" set to either "Include DTG" or "Include Tgrp with domain name" based on customer requirement.
- The PSX can have any kind of routing including standard routing based on trunk group or destination number with a Routing Label containing all the routes. Each route should have a Destination "SIP server" and "Trunk Group". The SIP server can have FQDN configured or IP address as per the requirement. It is recommended to not resolve FQDN by PSX and pass FQDN as it is due to restriction in the number of routes being capped at 10.
- The PSX Ingress Trunk Group requires the "Use IPTG Routing" option be enabled.
- PSX can return up to 10 Routes only in the 300 Response when acting as a redirector.

### **Creating SIP Server**

You must configure the Q21 SIP Signaling IP address & Port in the SIP Server, through which the Q21 sends INVITE requests.

**Figure 1: Create SIP Server**

The screenshot shows the configuration interface for creating a SIP Server. On the left, there's a sidebar with icons for Configure, Admin, and SIP Server. The main area has a title 'SIP Server' and a search bar. Below the search bar is a table with columns: SIP Server, IP Address, IPv..., Port, De..., ..., and ... . A single row is selected with the values: SYDP-Q21-SBC-LAB-01, 10.34.94.168, and others. To the right of the table is a detailed configuration panel:

SIP SERVER:	SYDP-Q21-SBC-LAB-01
Switch:	SYDP-Q21-SBC-LAB-01
Gateway Group:	DEFAULT
Media Server Route Label:	<None>
Cluster Profile:	<None>
Default Trunk Group:	SIP
Transit Route Label:	<None>
Charge Band Profile:	<None>
Traffic Control Escape Profile:	<None>
Mobile Switch ID:	1
Signaling Gateway Group:	<None>
Zone Index Profile:	<None>
Enum Authority Profile:	<None>
Address Reachability Service Profile:	<None>
SIMM Profile Group:	<None>
Peer Throttling Profile:	<None>
P-Originat-ID:	
Flags	
<input type="checkbox"/> CAMEL Services Supported	<input type="checkbox"/> Route CAMEL Subscription Calls
<input type="checkbox"/> CDP Gateway	<input checked="" type="checkbox"/> Traffic Management
<input type="checkbox"/> MTRR Supported	
Display	
<input type="checkbox"/> Allow Mixed Characters in Gateway Name	

### **Creating Feature Control Profile**

The Feature Control Profile determines whether the PSX acts as Redirector or Proxy for a particular call.

- In the below snapshot, the "Use IP Protocol Flags" is enabled and PSX processing mode is chosen as Redirector, so PSX acts as Redirector
- "Support Domain name in 300" will send the FQDN as it is instead of doing a DNS query for the FQDN. This is recommended so PSX can send up to 10 FQDNs.
- "Support PAI Header in Contact" enables sending embedded PAI in the contact header of a 300 response

Please refer to the Policy Server (PSX) documentation for details on each of the required flags.

**Figure 2:** Create Feature Control Profile

Feature Control Profile: REDIRECTOR	
<input type="checkbox"/> Process ISUP MIME From SIP Message Body <input type="checkbox"/> Use Flex Variable for Origination Jurisdiction Determination <input type="checkbox"/> Use Flex Variable for Destination Jurisdiction Determination <input type="checkbox"/> Process Screening For Call Origination	
IP Protocol Flags	
<input checked="" type="checkbox"/> Use IP Protocol Flags	
Flags	
<input type="checkbox"/> Default Called User As A User Name <input type="checkbox"/> Default Calling User As A User Name <input type="checkbox"/> Disable Egress Check And Don't Send Contract Number <input type="checkbox"/> Prefer BICC instead of ISUP routes for FCI preferred value <input type="checkbox"/> Proxy/Redirector Force Route Calls With Non-Local IP Address <input type="checkbox"/> Reject Calls To Non-Local Domains <input type="checkbox"/> Reject Calls To Non-Local IP Addresses <input checked="" type="checkbox"/> Support Domain Name In 300 Contact <input checked="" type="checkbox"/> Support PAI Header in CONTACT <input type="checkbox"/> Honor Phone-Context Parameter <input type="checkbox"/> Enable Stir Shaken	
PSX Processing Mode	
<input type="radio"/> Proxy <input checked="" type="radio"/> Redirector	

**Feature Control Profile: REDIRECTOR**

<input type="checkbox"/> Accept Calls With RPH If Dialed Number Is Non ETS	
<input type="checkbox"/> Enable RPH ETS	
Process Destination Trunk Group And Trunk-Context	
<input checked="" type="checkbox"/> Process TGRP	
<input checked="" type="checkbox"/> Process Trunk-context	
<input type="checkbox"/> Process Enumdi Parameter	
<input checked="" type="checkbox"/> Process Originating Trunk Group And Trunk-Context Over OTG	
<input type="checkbox"/> SIP Cause Code Mapping	
<input type="checkbox"/> Skip Number Translations For Valid Service Routes	
<input type="checkbox"/> Include Retry After For 503 Responses	
<input type="checkbox"/> Process Swid And Tgid From Sip Invite	
<input type="checkbox"/> Don't Restart Timer C on 1xx	
<input type="checkbox"/> Override Trunkgroup With Subscriber End Point Profile	
<input type="checkbox"/> Fetch State For ENUM SIP AoR	
<input type="checkbox"/> Enable Per Route Routing Label	
<input type="checkbox"/> Do Not Validate GAP	
<input type="checkbox"/> Process ISUP MIME From SIP Message Body	
<input type="checkbox"/> Use Flex Variable for Origination Jurisdiction Determination	
<input type="checkbox"/> Use Flex Variable for Destination Jurisdiction Determination	
<input type="checkbox"/> Process Screening For Call Origination	

### Creating IP signaling profile

Setting Destination Trunk Group option to "Include Tgrp with domain name" will send Trunk Group and Trunk-context values in the contact header of the 300 response.

**Figure 3:** Create IP Signaling Profile

**IP SIGNALING PROFILE: REDIRECTOR\_IPSP**

SIP Headers And Parameters	
Include Charge Information:	<input checked="" type="radio"/> Include None <input type="radio"/> Include P-Charge-Info
Session-Expires Refresher:	<input checked="" type="radio"/> Not Send <input type="radio"/> UAC <input type="radio"/> UAS
SIP TO Header Mapping:	<input type="radio"/> None <input checked="" type="radio"/> Original Called Number (OCN) <input type="radio"/> Called Number <input type="radio"/> GAP Dialed Number
<input type="checkbox"/> PI Allowed Send CPC In:	<input checked="" type="radio"/> DEFAULT <input type="radio"/> FROM <input type="radio"/> PAI <input type="radio"/> BOTH
Destination Trunk Group Options: <input checked="" type="checkbox"/> Include Tgrp With Domain Name	
Originating Trunk Group Options: <input checked="" type="checkbox"/> Include None	
Generate Call-ID Using: <input checked="" type="checkbox"/> Do not Use Ingress Call-Id	

### Creating SIP Response Code Profile

This profile can be used to change the SIP response code to the desired response code if the default PSX response code is not accepted.

**Figure 4:** Create SIP Response Code Profile

SIP Response Code Profile ID:	DEFAULT
Description:	SIP Response Code Mapping
<input type="radio"/> External <input checked="" type="radio"/> Internal	
Internal Response	
Received Response Code:	404 - PES No Routes Returned
Converted Response Code:	403 Payment Forbidden
<input type="button" value="Add/Update"/>	
From	To
404 - PES No Routes Returned	403 Payment Forbidden

## Creating Trunk Group

Create a Trunk Group and attach a Feature Control Profile, and IP Signaling Profile to it. Optionally, you can also attach the SIP Response Code Profile to the Trunk Group.

If there is no SIP Response Code profile attached to a specific Trunk Group, the DEFAULT profile will be used.

**Figure 5: Create Trunk Group**

Trunk Group:	SLVODAFNZL01	<input type="checkbox"/> Unrestricted
Gateway:	SYDP-Q21-SBC-LAB-01	▼
Description: SIP supplier TG		
Auto Recall Profile:	<None>	
Call Processing Localization Variant:	Generic	
Calling Area:	<None>	
Carrier:	00004	
Carrier Selection Priority:	<None>	
Country:	64 - New Zealand	
DDI Range Profile:	<None>	
Destination Switch Type:	Access	
Direction:	Two Way	
Element Routing Priority Profile:	NA	
Feature Control Profile:	REDIRECTOR	
IP Signaling Profile:	REDIRECTOR_IPSP	

## Creating Routing Profile

Create the Routing Profile and attach as many routes as required. Each Route will have a SIP Server and Trunk Group.

- i Up to 10 Routes can be passed in a 300 response in Redirect mode

**Figure 6: Create Routing Label**

Routing Label:	TCMT_LAB01_NZL_RL																																																							
Route Prioritization Type																																																								
<input checked="" type="radio"/> Sequence	<input type="radio"/> Proportion	<input type="radio"/> Round Robin	<input type="radio"/> All Proportion	<input type="radio"/> Least Cost Routing																																																				
Route Prioritization Type For Equal Cost Routes: Sequence																																																								
<input type="checkbox"/> Use TAR Routes																																																								
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Route Prioritization Type For Equal Cost Routes: Sequence																																																								
Local Routes																																																								
<input type="radio"/> Pass Only Local Routes	<input type="radio"/> Prioritize Local Routes	<input checked="" type="radio"/> Do Nothing																																																						
Filter Criteria Routes																																																								
<input type="radio"/> Pass Only Filter Criteria Routes	<input type="radio"/> Prioritize Filter Criteria Routes	<input checked="" type="radio"/> Do Not Change Route Order																																																						
Flags																																																								
<input type="checkbox"/> Continue Number Translation <input type="checkbox"/> Continue CNAM Translation <input type="checkbox"/> No Connect Signal To Be Sent																																																								
Routes																																																								
<table border="1"> <thead> <tr> <th>Type</th> <th>Endpoint 1</th> <th>Endpoint 2</th> <th>IP Peer</th> <th>Sequence</th> <th>Proportion</th> <th>Status</th> <th>TAR Ac...</th> <th>TAR Loc...</th> <th>DM/P...</th> <th>Apply...</th> <th>Testing</th> <th>Cost</th> <th>Skip...</th> <th>STI</th> </tr> </thead> <tbody> <tr> <td>SIP Server</td> <td>lab01nz.sl1.telstra.com</td> <td>SYDP-LAB-SWE-SBC-01</td> <td></td> <td>1</td> <td>0</td> <td>In Service</td> <td>Normal</td> <td>0</td> <td></td> <td>Do No...</td> <td>Normal</td> <td>100...</td> <td>Disa...</td> <td>0</td> </tr> <tr> <td>SIP Server</td> <td>lab01nz.sl2.telstra.com</td> <td>SYDP-LAB-SWE-SIPP-02</td> <td></td> <td>2</td> <td>0</td> <td>In Service</td> <td>Normal</td> <td>0</td> <td></td> <td>Do No...</td> <td>Normal</td> <td>100...</td> <td>Disa...</td> <td>0</td> </tr> </tbody> </table>												Type	Endpoint 1	Endpoint 2	IP Peer	Sequence	Proportion	Status	TAR Ac...	TAR Loc...	DM/P...	Apply...	Testing	Cost	Skip...	STI	SIP Server	lab01nz.sl1.telstra.com	SYDP-LAB-SWE-SBC-01		1	0	In Service	Normal	0		Do No...	Normal	100...	Disa...	0	SIP Server	lab01nz.sl2.telstra.com	SYDP-LAB-SWE-SIPP-02		2	0	In Service	Normal	0		Do No...	Normal	100...	Disa...	0
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SIP Server	lab01nz.sl2.telstra.com	SYDP-LAB-SWE-SIPP-02		2	0	In Service	Normal	0		Do No...	Normal	100...	Disa...	0																																										

## Creating Standard Route

Create the Standard Route and attach a Routing Label to it.

**i** The sample below shows Trunk Group based Routing on PSX.

There are various ways of routing in PSX such as username based routing, number based routing, etc. depending on customer requirement.

For more information, refer to the Policy Server (PSX) Documentation.

**Figure 7: Create Standard Route**

Entity Type:	Trunk Group
Trunk Group:	SLVODAFNZL01
GATEWAY:	SYDP-Q21-SBC-LAB-01
Not Applicable	
<input checked="" type="radio"/> Call Parameter Filter Profile:	<None>
<input type="radio"/> Call Parameter Filter Profile Group:	<None>
Destination National:	9980315
Destination Country:	64 - New Zealand
Domain Name:	<None>
<input checked="" type="radio"/>	
<input type="radio"/> IP Address:	0 . 0 . 0 . 0
Partition:	DEFAULT
Routing Label:	TCMT_LAB01_NZL_RL
Call Type	
0+ 0+ IDDD 0- 00 1+ Carrier Cut Through IDDD	
<input type="checkbox"/> All Call Type Bits	
Transmission Medium	
Speech 3.1 KHz Audio 7.0 KHz Audio 56 kbps 64 kbps Packet Multirate 384 kbps 1536 kbps	
Time Range:	
ALL	

## Conclusion

These Application Notes describe the configuration steps required for QSBC interop with PSX Redirector. All features and serviceability test cases were completed.