SVI MGC User Guide

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Change History

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Introduction

Before reading this document, we encourage you to familiarise yourself with our SVI-MS Guide and Resource Glossaries which are viewable at the following links, these will provide more information on usage of the user interface and the configuration options within the SVI:

SIP, SS7, ISDN and Routing Resource Glossaries - HERE

To view further information on the capabilities of the SVI MGC and the range of products under this family please visit the following link:

SVI MGC Range

Media-Gateway-Controller-function

Media-Gateway-Controller

1.0 Installation

Normally, the SVI system will come pre-installed with all the required software, based on the details provided to Squire Technologies in the Installation Checklist.

2.0 Accessing the SVI

You will be able to access the unit via a web browser (please note, IE and Safari are not supported). Enter the IP Address in browser and you will see a login screen shown below. Please note, you may have to allow pop-ups on your browser to access the login screen. The default credentials are as follows:

Username and Password information should be provided via your Lead Engineer / Training Material.

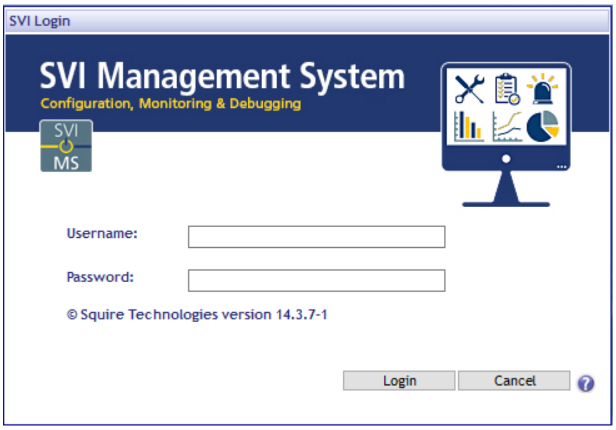


Figure 1 SVI Login page

Having successfully accessed the SVI-MS you are now ready to move on to the next stage of this user guide.

3.0 Basic Configuration Details

To help you make the most of this User Guide this section will provide details on what is pre-configured on your SVI product and what you will need to configure. It will also briefly explain some of the terms that are used throughout the User Guide, and we advise using this section as a first reference point in the event of difficulty in the setup process.

The following diagram is a visual representation of the process this User Guide will take you through to set up your MGC.

3.1 Factory Configuration

The SVI MGC unless factory preconfigured from a completed checklist will be delivered with a basic configuration of:

Media Gateway (Physical Interconnect)	The Media Gateway resource will be supplied with a default IP and port and a default range of interfaces assigned to this media gateway.
SS7	F-Links with two signalling links within one linkset connected to one destination. Each card supplied within the SVI will be configured with a range of circuits for SS7. SS7 Circuits will be configured onto all the corresponding SS7 interfaces with a default CIC range running 1-N where N fills the interfaces.
SIP	A VoIP Stack for the IP bonded interface on the MGC. A VoIP Destination with a dummy IP address.
Routing	Routing will be supplied as: protocol to protocol; by default, this will include inter card routing. This should be reconfigured to route as required.

3.2 Key Terms

The following is a list of some of the basic terms you will come across in this user guide, and what they mean:

Resources:	Resources are components of the product. For example, a VoIP Destination is a Resource.
Wizards:	As there are many resources with potentially hundreds of attributes, the SVI-MS provides Wizards for setting up the most common configuration elements. These Wizards allow the configuration of one or more resources at once with a simplified interface.
Derived Resources:	Derived Resources is the term given to resources that are directly configured by other Resources and Wizards. For example, when you create a SIP Endpoint, you will automatically create a Hunt Group, which means that Hunt Groups are a Derived Resource of the SIP Endpoint Wizard. Where a Derived Resource is mentioned, this user guide will direct you to the Resource Glossary for an exact definition of the resource in question.

Commit:	Applying changes to the SVI by committing them.
Remote Media Gateway:	This term refers to the 3rd party remote media gateway to which the SVI MGC (Media Gateway Controller) will connect with.
SS7 Interconnect:	This refers to the collection of resources required to configure within the SVI to make an SS7 connection.
SIP Interconnect:	This refers to the collection of resources required to configure within the SVI to make a SIP connection.
Hunt Groups:	A Hunt Group (or trunk Group) is a collection of Circuits or Bearer Channels (though it can be a single circuit or channel) which will seize either from high to low, low to high or sequentially depending on the hunt group configuration.
Bidirectional Routing:	Bidirectional Routing routes traffic between two hunt groups. In this User Guide we will address sending traffic in the following manner: SS7 circuits to and from SIP Endpoints.

Where relevant, this user guide will direct you to the SVI Resource Glossary, we recommend having this available before proceeding.

3.3 Accessing the Configuration

The SVI MGC contains several predefined templates called wizards these are accessible through the SVI User Interface and allow for easy configuration of the resources required to interconnect the SVI MGC with an SS7 or SIP interconnect.

Each of the wizards described below are accessed through the configuration tab within the user interface this is highlighted below:

Figure 2 SVI Menu bar

Following this, the wizard tab is primarily used. More advanced options for each of the resources can be viewed and modified through the resources tab.

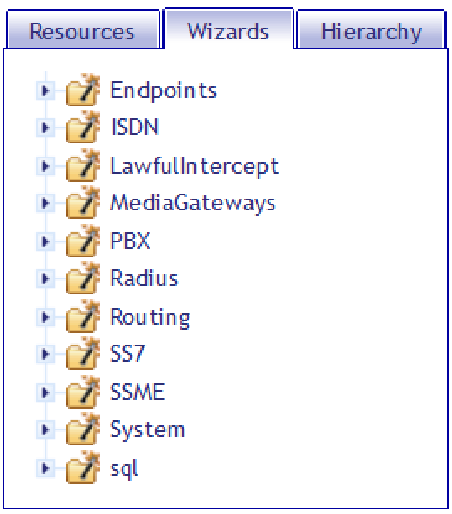


Figure 3 MGC Wizards

Any preconfigured wizards or resources will show as a bold highlighted object, where un-configured items will show as greyed. If there are any issues accessing the user interface or seeing preconfigured configuration from factory or from a checklist supplied prior to shipping, please contact the Squire Support desk for further advice.

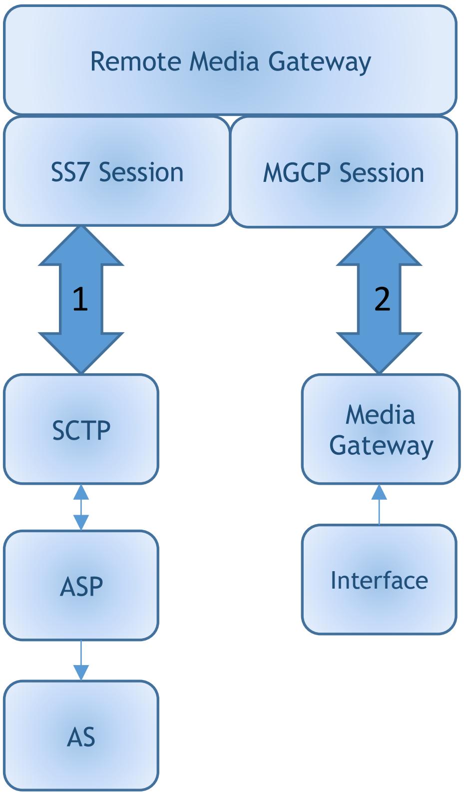
4.0 Setting up a Remote Media Gateway

This section describes how to configure the SVI MGC for use with different remote media gateways through the protocol layer used to connect with the media gateway. Knowing the IP address of the SVI from the previous section the remote media gateway can be configured with this information, the SVI can have one of the following methods of connection for signalling; SCTP or SLT and one of the following for speech; MGCP or MEGACO. How to configure each is explained below.

Each media gateway configured for use against the SVI will use one of the following signalling methods and one of the following speech negotiation methods. Both will be required for signalling and speech to pass successfully.

4.1 SVI Hardware Control Modelling

The SVI uses common resources to enable configuration of hardware interconnects with ease, the below diagram shows how the SVI MGC remotely connects to a remote media gateway and which resources are used to control each aspect.



1) Signalling Backhaul over M2UA, M3UA or SLT.
2) MGCP or MEGACO Control for media setup.

This diagram shows a typical SCTP M2UA connection into a remote media gateway where the SVI will backhaul MTPL2 through SCTP and allow for routing into SIP or back to SS7. The Media Gateway resource allows for the media connections to be established within the same remote media gateway.

It is possible within this model to split the SCTP away from the remote media gateway into a separate remote connection and receive signalling from a different remote location through M2UA, M3UA or SLT.

4.1.1 SCTP/ASP/AS

The Stream Control Transmission Protocol allows for you to define the connection details for the remote media gateway allowing for backhauling of signalling messages over M2UA, M3UA or SLT from a remote media gateway.

4.1.2 IP Socket

The IP Socket defines the socket to open for listening on the SVI this is configured automatically within the Media Gateway wizard.

4.1.3 Media Gateway

The Media Gateway resource defines the remote media gateway to which the SVI will connect into for setting up speech connections. The media gateway resource controls the MGCP or MEGACO messaging between the SVI MGC and the remote media gateway.

4.1.4 Interface

The Interface resource allows for the SVI to define what physical ports are available on the remote media gateway. These are then used to associate circuits when you configure the SS7 interconnect later in this guide.

4.2 SCTP M2UA (Stream Control Transmission Protocol)

The Sigtran SCTP Socket wizard is used to configure SCTP connections between the SVI and a Media Gateway.

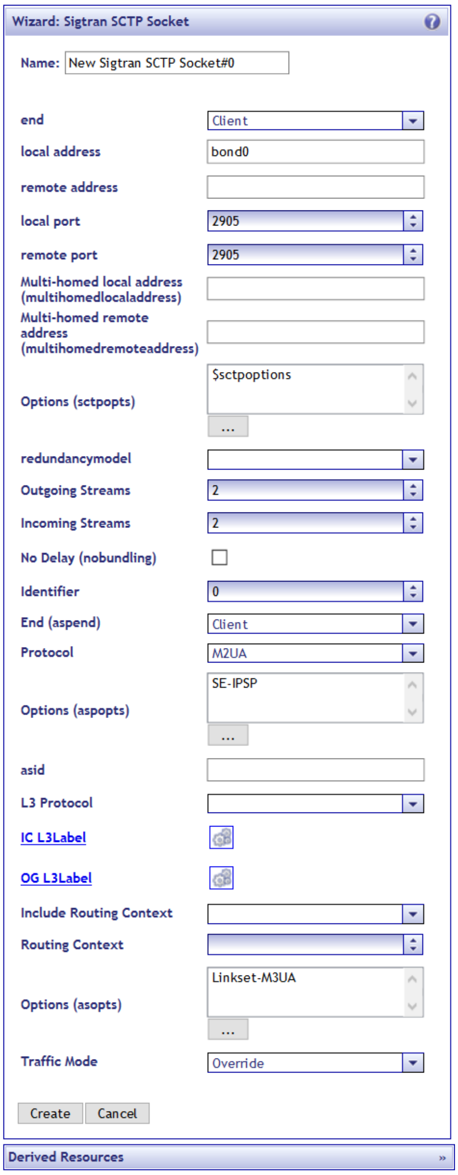


Figure 4 Sigtran SCTP Socket wizard

End:	This defines the client and server ends of the connection. The SVI should be client end and Media Gateway server end.
Local Address:	This defines the SVI local address for the SCTP socket, where an SVI has multiple IP interfaces SCTP can be defined onto individual interfaces.
Remote Address:	This defines the remote Media Gateway IP Address the SVI will communicate with.
Local Port:	This defines the SVI local port for the SCTP socket, this will open the socket on the above defined IP address.
Remote Port:	This defines the remote media gateway port the SVI will communicate with.
Protocol:	The protocol attribute allows you to change the variant of SCTP protocol used within this connection. This will remain at M2UA to allow backhauling of MTP Layer 2 traffic into the SVI. Refer to the SS7 resource glossary for further information on the possible configurable options. HERE
Asid:	This should be set the same as the Name attribute.

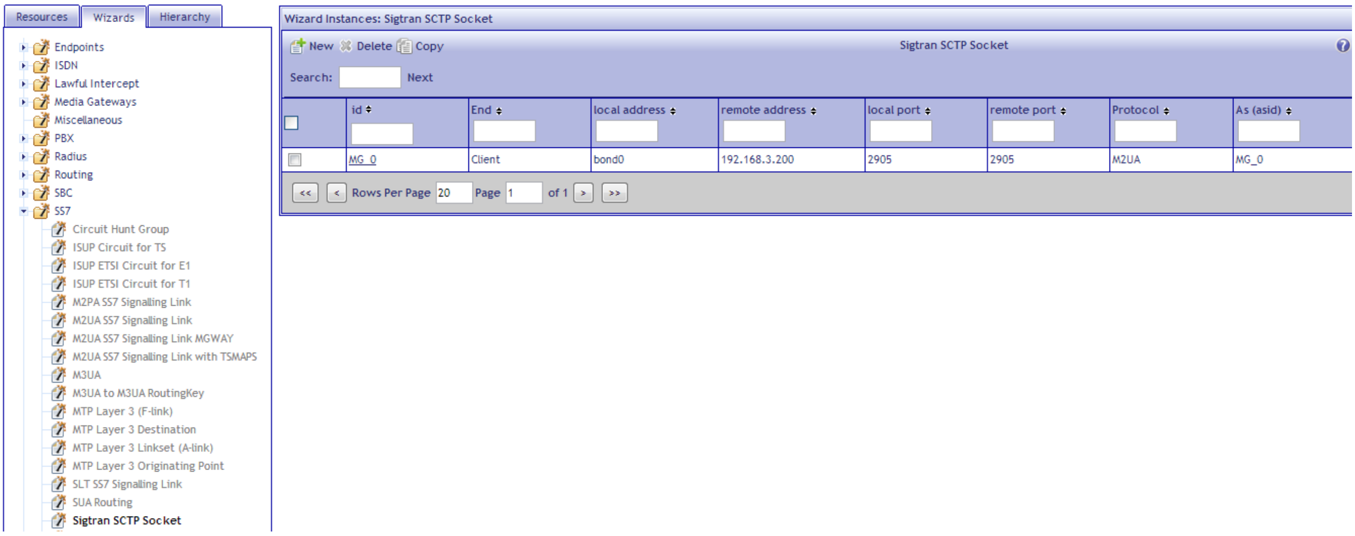


Figure 5 Finished Wizard.

The completed wizard shows the SVI MGC as a client to a connection into an IP address and port. This will allow for an M2UA connection backhauling MTPL2 to the SVI for signalling decisions. SVI will have to be configured with SS7 details.

4.3 SCTP M3UA (Stream Control Transmission Protocol)

The Sigtran SCTP Socket wizard is used to configure SCTP connections between the SVI and a Media Gateway.

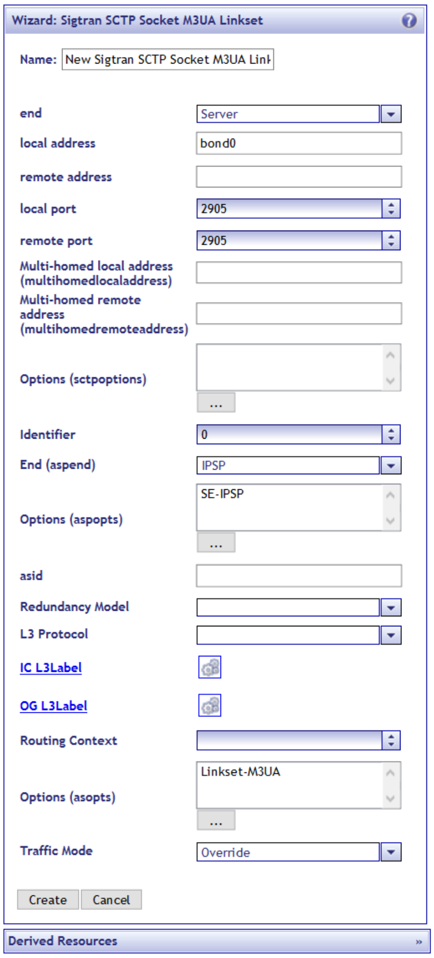


Figure 6 Sigtran SCTP Socket M3UA Linkset wizard

End:	This defines the client and server ends of the connection. The SVI should be client end and Media Gateway server end.
Local Address:	This defines the SVI local address for the SCTP socket, where an SVI has multiple IP interfaces SCTP can be defined onto individual interfaces.
Remote Address:	This defines the remote Media Gateway IP Address the SVI will communicate with.
Local Port:	This defines the SVI local port for the SCTP socket, this will open the socket on the above defined IP address.
Remote Port:	This defines the remote media gateway port the SVI will communicate with.
Asid:	This should be set the same as the Name attribute.
Multi-homed Local Address:	This defines the second SVI local address for the SCTP socket in a multihoming setup.
Multi-homed Remote Address:	This defines the second remote address for the SCTP socket in a multihoming setup.
Redundancy model:	Usually set to Active for M3UA
L3 Protocol:	Mostly ITU or ANSI
Routing Context:	A numeric value can be configured if required
Traffic mode:	Usually, Loadshare for M3UA

Once the attributes have been set, click the Create button at the bottom of the wizard and then Commit the configuration.

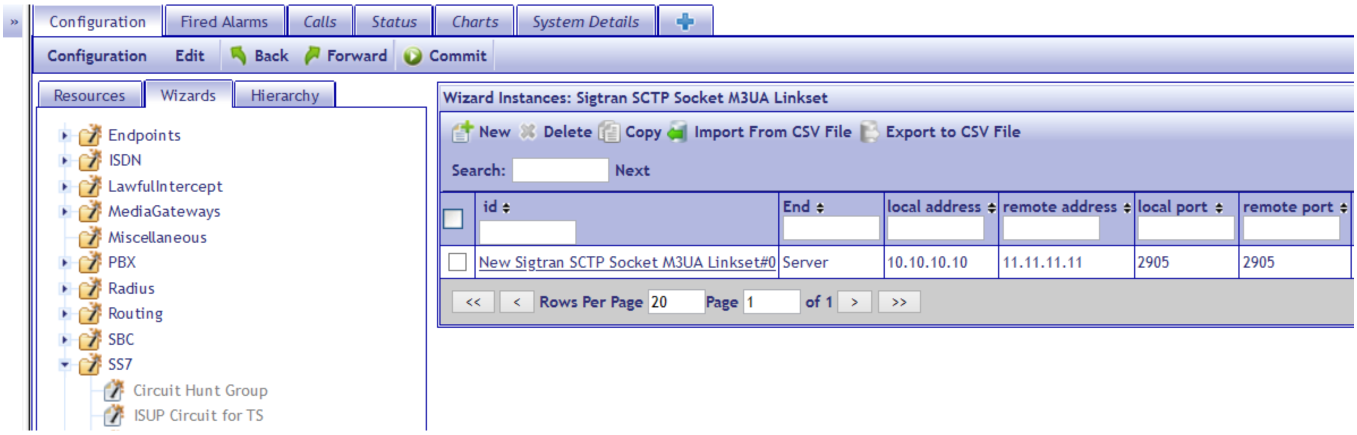


Figure 6a - Finished M3UA Socket Wizard

4.4 SLT (Signalling Link Terminal)

The Sigtran SLT Socket wizard is used to create an SLT connection from the SVI into an SLT enabled media gateway.

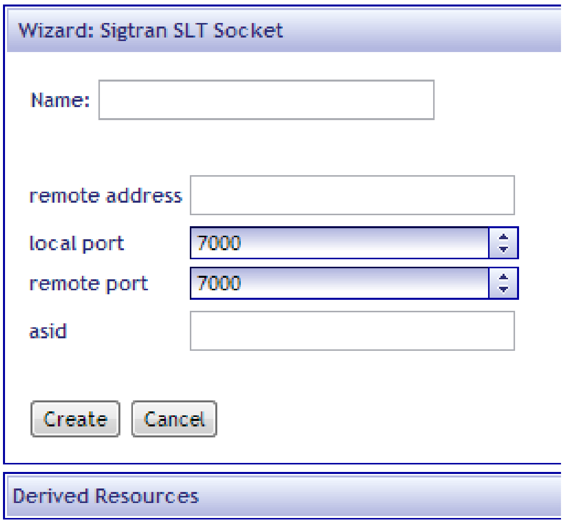


Figure 7 Sigtran SLT Socket Wizard

Remote address:	This defines the remote Media Gateway IP address for the SVI to communicate with.
Local Port:	This defines the local SVI port opened on bond0 for the remote media gateway to connect with.
Remote Port:	This defines the remote media gateway port for the SVI to communicate with.
Asid:	This should be set the same as the name attribute.

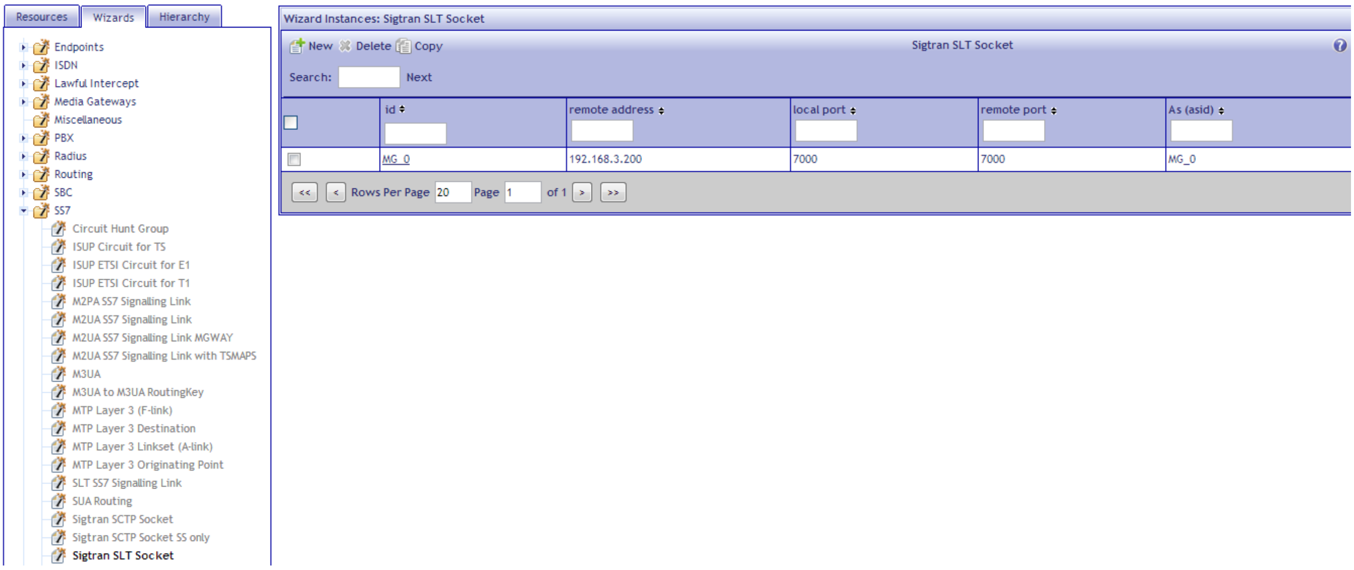


Figure 8 Finished Wizard.

The completed wizard shows the connection details configured for the SLT connection.

The previous section describes how to configure the Signalling aspects for connecting to a remote media gateway, the following section describes how to configure the speech details.

4.5 MGCP (Media Gateway Control Protocol)

4.5.1 Media Gateway Configuration

The MGCP Media Gateway wizard is used to configure the SVI to connect an MGCP/MEGACO enabled media gateway.

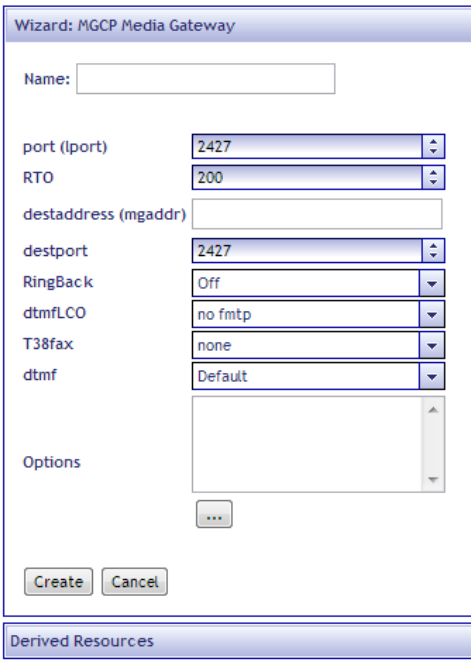


Figure 9 Finished Wizard.

Name:	The name should be set the same as the hostname configured on the Media Gateway, MGCP relies heavily on naming for interactions. Care should be taken to ensure this is correct before proceeding.
Port(lport):	This defines the local port to be opened on the SVI MGC

RTO:	The Retry Time Out timer allows for the initial retry period for MGCP messages with no response. This timer is defaulted to the MGCP specified level.
Destaddress(mgaddr):	This defines the remote Media Gateway IP address for the SVI to communicate with.
Destport:	This defines the remote media gateway port for the SVI to communicate with.
Ringback:	This attribute configures MGC to instruct a Media Gateway to play ringback tones at appropriate moments during a call.
dtmfLCO:	This attribute is for use with Cisco media gateways and modifies the behaviour of the FMTP options for DTMF events.
T38fax:	When passing T38 fax calls this option should be set as T38-Loose.
Dtmf:	This option modifies the behaviour of the way in which DTMF events are reported to and from the media gateway.
Options:	The options on the Media Gateway are for advanced uses only (these should only be used with the advice of Squire Support).

The completed wizard shows the IP and port configuration entered and any other options configured against that particular media gateway. Different options can be set against different media gateway for control over the behaviour of that media gateway.

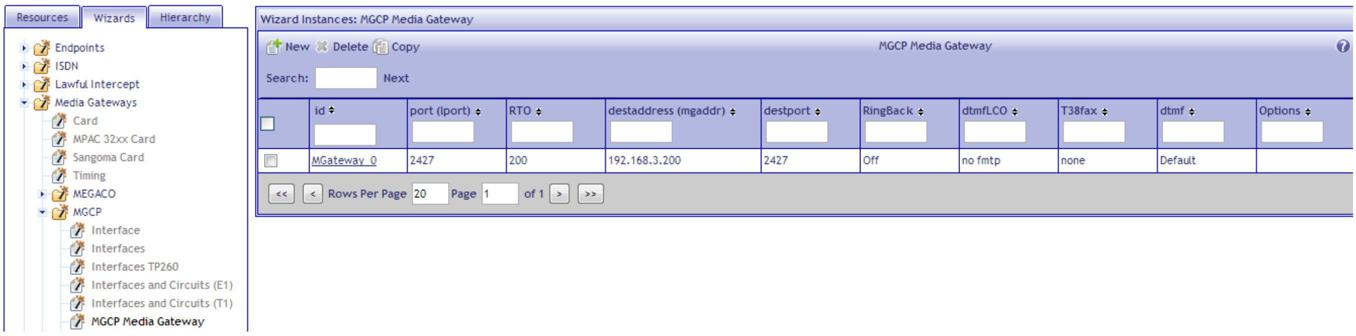


Figure 10 MGCP Media Gateway wizard

4.5.2 Interface Configuration

The Interface wizard found under the MGCP wizards list is used for creating E1/T1 interface configuration on the SVI that will match that of the Media Gateway.

It is possible to create multiple interfaces at one time using the Interfaces wizard, this will however limit the flexibility in changing the use of an interface later and recommended if not changing the types of interfaces frequently.

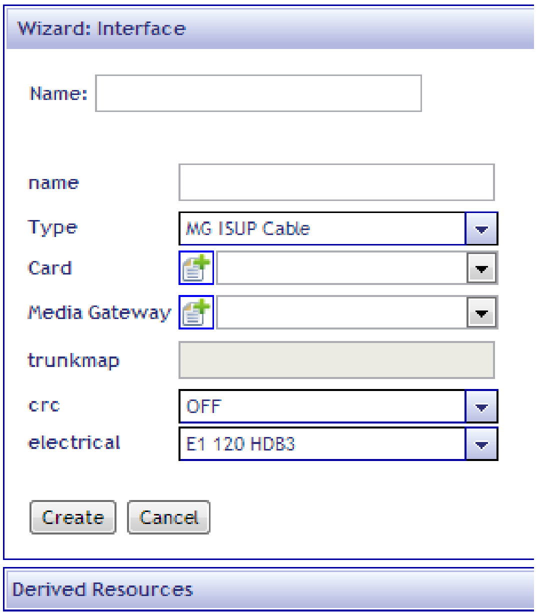


Figure 11 Interface Wizard

Name:	This is the SVI internal name for the Interface and is recommended you name. as something reflecting the nature of the E1/T1 provided. For example, the name of the provider and an integer value denoting which instance of cable.
name:	This is the Media Gateway's name prefix that is configured with for its interfaces; this configuration can generally be found easily from the Media Gateway and configured into the SVI.
Type:	This defines what type of interface and subsequently internal behaviour will be for this interface. This should be set at MG ISUP Cable for use as an SS7 E1 or T1.
Card:	This attribute is obsolete.
Media Gateway:	This defines the media gateway on which this interface resides.
Trunkmap:	This attribute is obsolete.
CRC:	This attribute is obsolete.
Electrical:	This attribute is obsolete.

The completed wizard shows a single E1 interface configured on the media gateway you configured in the previous section. Multiple of these interfaces should be configured for each port available on the media gateway.

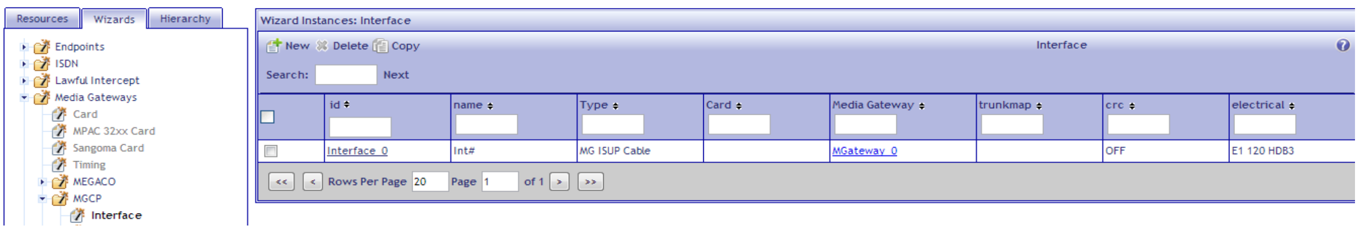
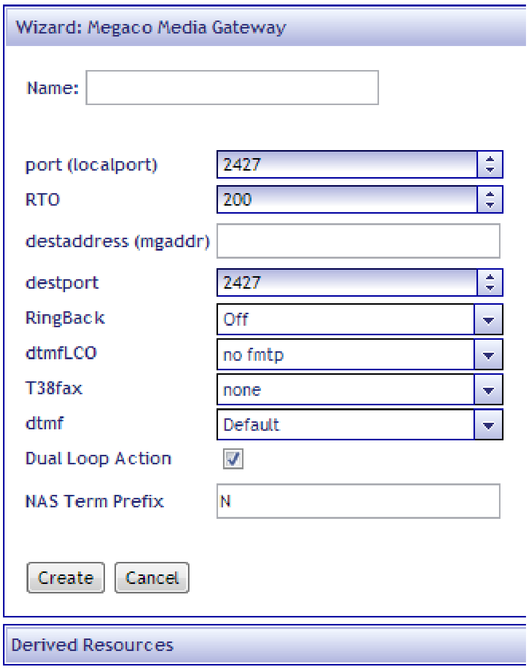


Figure 12 Finished Wizard.

4.6 MEGACO (H.248)

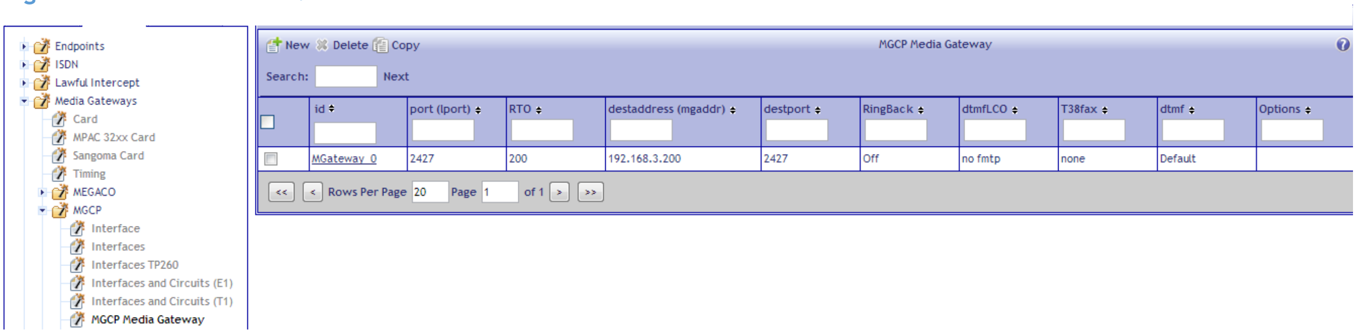
4.6.1 Media Gateway Configuration

The Megaco Media Gateway wizard is used to configure the SVI to connect with a Megaco enabled Media Gateway.



Name:	The name should be set the same as the hostname configured on the media gateway, Megaco relies heavily on naming for interactions. Care should be taken to ensure this is correct before proceeding.
Port(localport):	This defines the local port to be opened on the SVI MGC, this will be assigned by default to bond0.
RTO:	The Retry Time Out timer allows for the initial retry period for MGCP messages with no response. This timer is defaulted to the MGCP specified level.
Destaddress(mgaddr):	This defines the remote media gateway IP address for the SVI to communicate with.
Destport:	This defines the remote media gateway port for the SVI to communicate with.
Ringback:	**Ringback ON** - Ringback will be played to **incoming calls using this media gateway**, once the call is ringing (1); It will be turned off if there is an in-band-announcement /remote media indication (2) **Ringback OFF** - Ringback will not be played to incoming calls using this media gateway, unless configured on another relevant option.
dtmfLCO:	This attribute is for use with Cisco media gateways and modifies the behaviour of the FMTP options for DTMF events.
T38fax:	When passing T38 fax calls this option should be set as T38-Loose.
Dtmf:	This option modifies the behaviour of the way in which DTMF events are reported to and from the media gateway.
Dual Loop Action:	This attribute sets the format of the Megaco message into the media gateway to setup a COT tone test. This box should be checked unless having issues with COT passing on incoming calls.
NAS Term Prefix:	This free text field allows for setting a prefix on NAS calls passed through this media gateway.

Figure 13 Finished Wizard.



The completed wizard shows the remote IP and port the SVI will connect into and any other options that have been set for this media gateway.

4.6.2 Interface Configuration

The Interface wizard found under the MEGACO wizards list is used for creating E1/T1 port configuration on the SVI that will match that of the media gateway.

It is possible to create multiple interfaces at one time using the Interfaces wizard, this will however limit the flexibility in changing the use of an interface later and recommended if not changing the types of interfaces frequently.

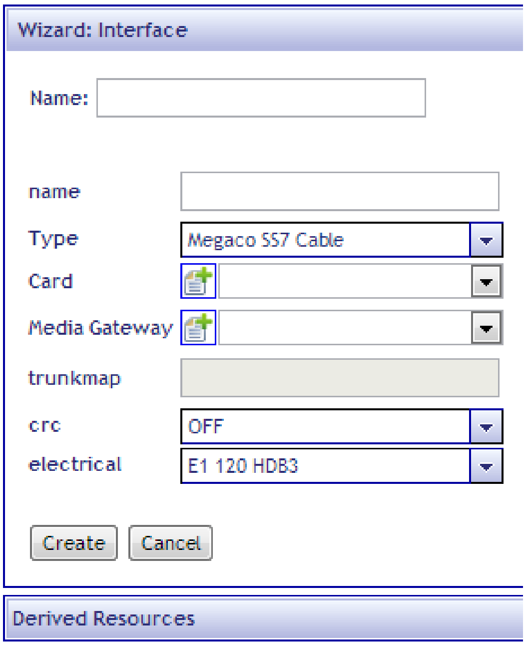


Figure 14 Interface Wizard

Name:	This is the SVI internal name for the Interface and is recommended you name as something reflecting the nature of the E1/T1 provided. For example, the name of the provider and an integer value denoting which instance of cable.
name:	This is the name prefix that the media gateway is configured with for it interfaces, this configuration can general be found easily from the media gateway and configured into the SVI.
Type:	This defines what type of interface and subsequently internal behaviour will be for this interface. This should be set at Megaco SS7 Cable for use as an SS7 E1 or T1.
Card:	This attribute is obsolete.
Media Gateway:	This defines the media gateway on which this interface resides.
Trunkmap:	This attribute is obsolete.
CRC:	This attribute is obsolete.
Electrical:	This attribute is obsolete.

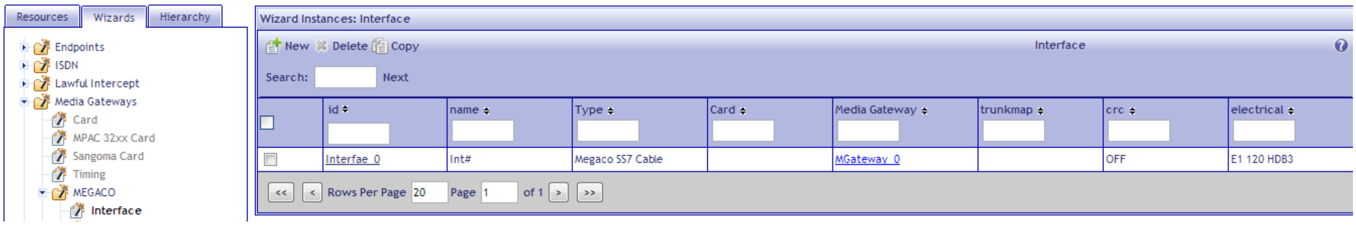


Figure 15 Finished Wizard.

The completed wizard shows a single E1 interface configured on the media gateway you configured in the previous section. Multiple of these interfaces should be configured for each port available on the media gateway.

5.0 SS7 Interconnect Configuration

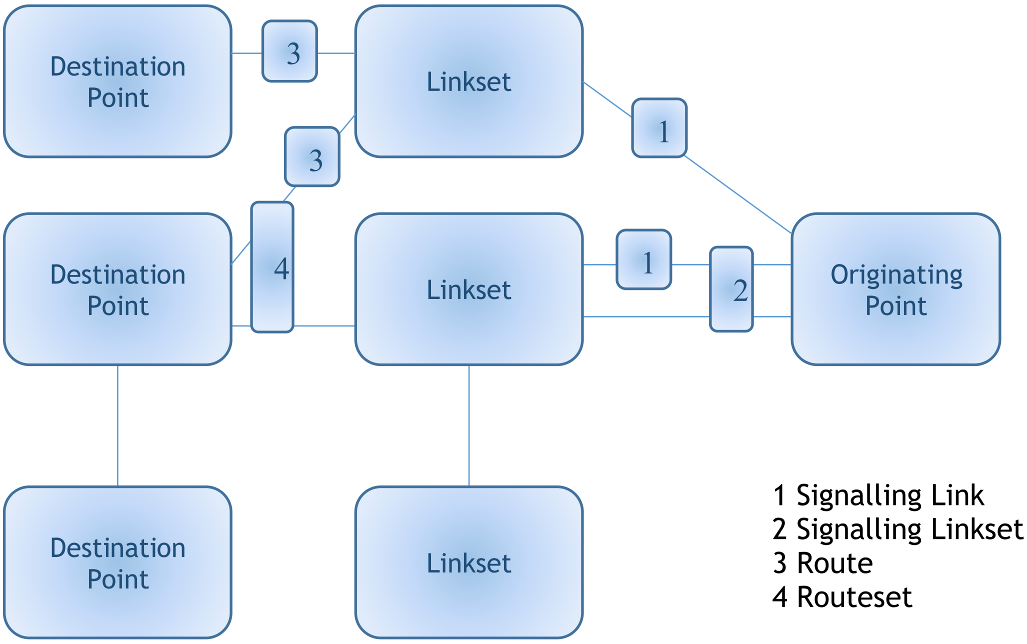
This section describes how to configure the SS7 interconnect to the SVI MGC.

NOTE: Any changes to an existing OPC, Linkset, DPC will require a system restart to build the stack(s) correctly.

5.1 SVI Network Modelling

To simplify the SS7 configuration the SVI's configuration resources have been designed to mirror the terminology of an SS7 interconnect.

The following diagram shows how SS7 networks are modelled.



5.1.1 Originating Point

The originating point of an SS7 network relates to the pointcode of the SVI. The SVI resource which models this network entity is the Originating Point Code resource.

5.1.2 Signalling Link

The signalling link is the transport layer of the signalling information between two physical interconnecting points. The signalling link resource models this network entity.

5.1.3 Linkset

The signalling linkset is the combination of multiple signalling links between two physical interconnection points. Multiple signalling links are used between two physical interconnection points to increase bandwidth and add redundancy. The linkset resource models this network entity.

5.1.4 Adjacent Point

The adjacent point defines the point code of the equipment directly terminating the linkset. The linkset resource is also used to model this network entity.

5.1.5 Route

A route defines the possible routes through an Adjacent point to signal a destination point that is within an SS7 network. The routes are automatically assigned internally within the SVI, and no resource is required to configure them.

5.1.6 Routeset

The routeset defines all the possible routes through the adjacent points to get to a specific destination. The routeset is also automatically assigned internally within the SVI and no resource is required to configure them.

5.1.7 Destination Point

The destination point defines the destination of the signalling message that the SVI requires the signalling data to be handled by. This in general would be an SSP or GTT node.

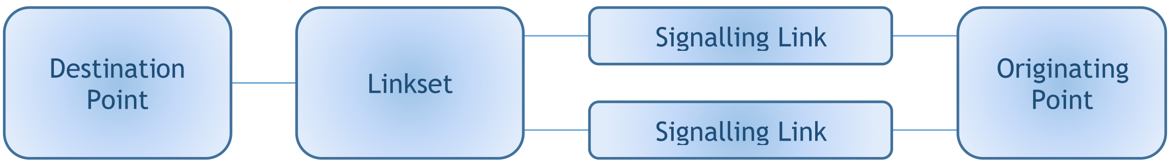
5.2 Examples of Network Models

The following shows some examples of network modelling that allows for fast configuration of the SVI. It is recommended that the user quickly sketches out their SS7 interconnect requirement first before configuring the SVI's resources.

5.2.1 F link interconnect

F Link Interconnect

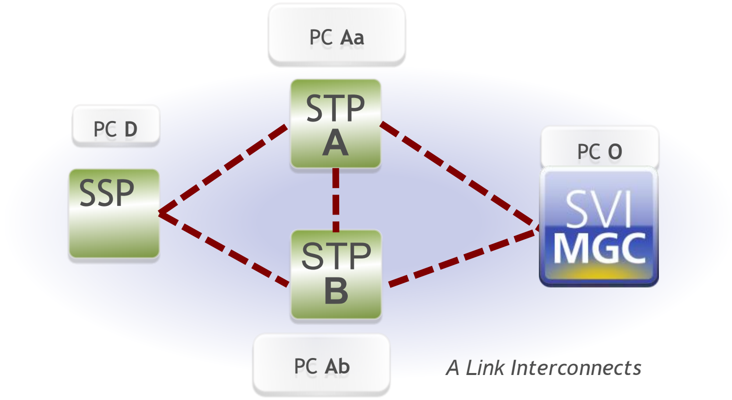
The diagram above shows a typical F link interconnect with the SVI with an originating pointcode O interconnecting directly to an SS7 device with destination pointcode of D. The following diagram below shows how this interconnect would be modelled.



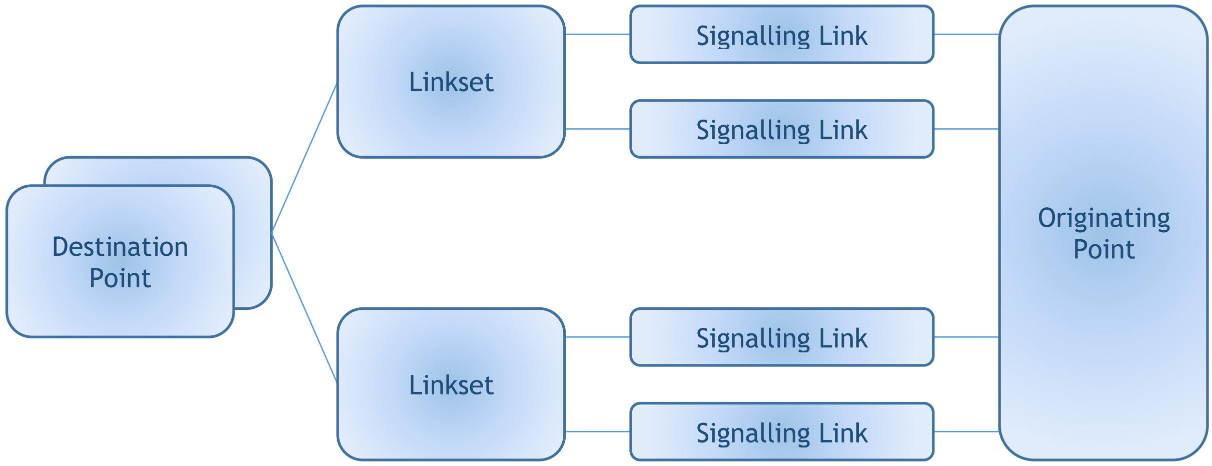
The pointcode of the destination resource will have the same value as the linkset resource as these are seen as the same entity within an F-link interconnect.

5.2.2 A link interconnect

For an A-Link interconnect the signalling is distributed through an STP network before being delivered to the destination point. This is shown below:



The following diagram shows how this is modelled through the SVI Resources.



5.3 Configuring the SS7 Interconnect

The following section runs through using the SS7 interconnect wizards to configure an SS7 interconnect on the SVI.

5.3.1 Configuring the Originating Point Resource

The MTP layer 3 Originating Point wizard is used to setup an originating point on the SVI.

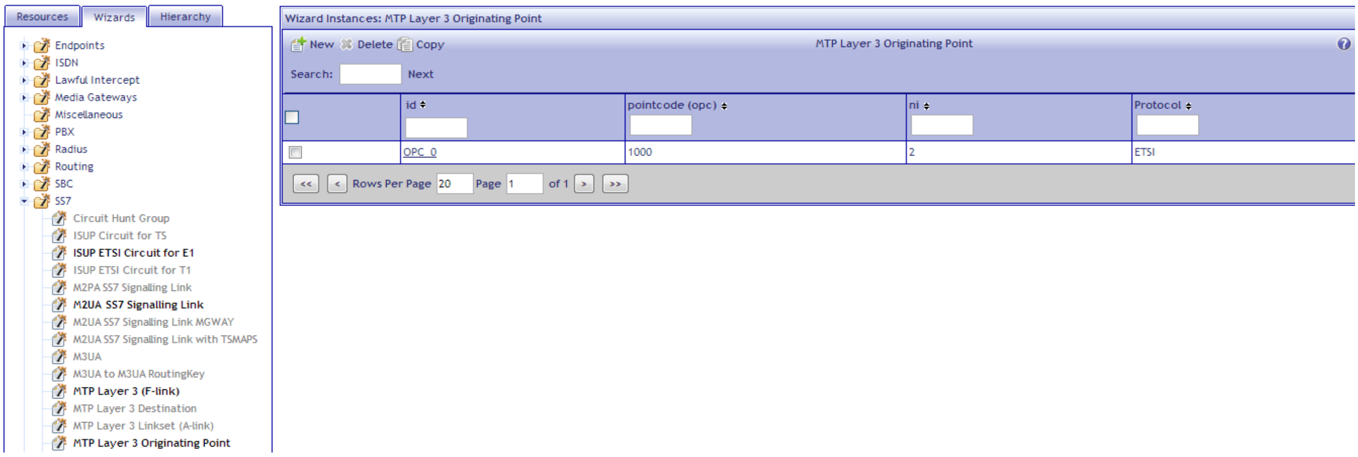


Figure 16 MTP3 Layer 3 Originating Point

Through this wizard the originating point code (OPC), network indicator (NI) and the MTP L3 protocol variant can be defined. Multiple originating points can be setup on the SVI using the New button to configure a new originating point on the SVI.

5.3.2 Configuring the Linkset Resource

This section defines how to configure an SS7 A or F link interconnect on the SVI.

5.3.2.1 Configuring a F-Link interconnect

To configure an F-Link linkset against a particular originating point the MTP Layer 3 Linkset (F-link) wizard is used.

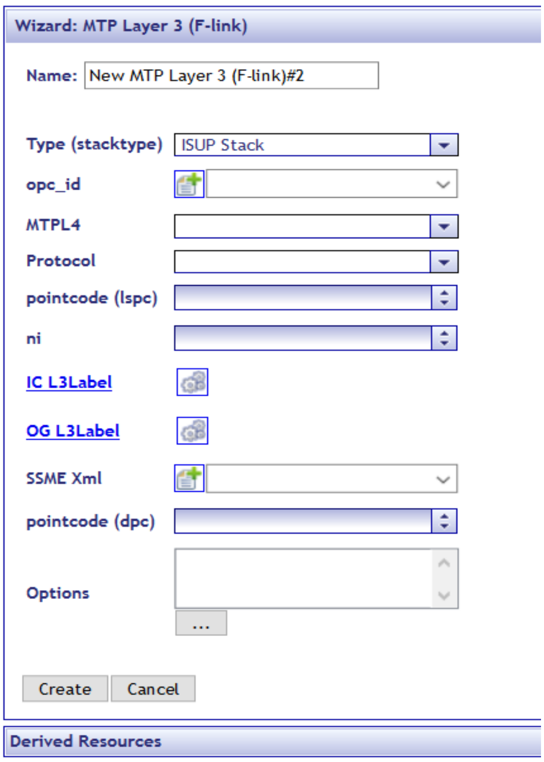


Figure 17 MTP Layer 3 F-Link Wizard

Type (stack type):	SS7 Stack should be selected on an MGC.
OPC_ID:	The originating point should be set from the drop-down list to the originating point resource that was configured in the Originating Point resource.
MTPL4:	Should Always be set to Internal on an MGC.
Protocol:	The desired protocol that is to be used for the interconnect is configured from the Protocol drop down list.
Pointcode (lspc):	The linkset pointcode should be set to the adjacent interconnect pointcode.
NI:	The network indicator should be set to the code required by for the interconnect: (0) International network (1) Spare (for international use only) (2) National network (3) Reserved for national use IC L3Label: Not required on MG/MGC. OG L3Label: Not required on MG/MGC.
SSME Xml:	Not required on MG/MGC.
Pointcode (dpc):	The destination point should be the destination of the signalling message.
Options:	Additional options for the F-Link

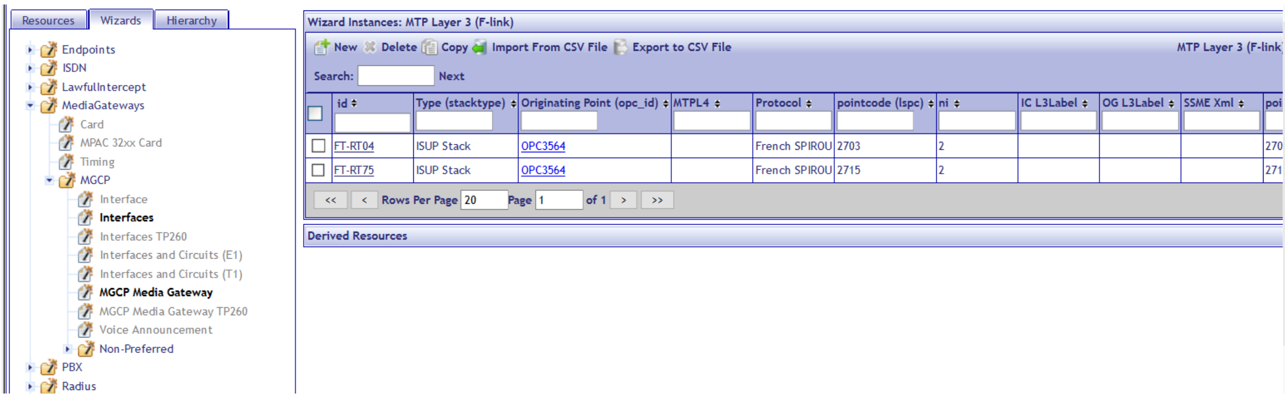


Figure 18: Finished Wizard

The completed wizard shows the originating point the f-link is associated with and the pointcode and NI values.

5.3.2.2 Configuring an A-Link interconnect

To configure an A-Link linkset against a particular originating point the MTP Layer 3 Linkset (A-link) wizard is used.

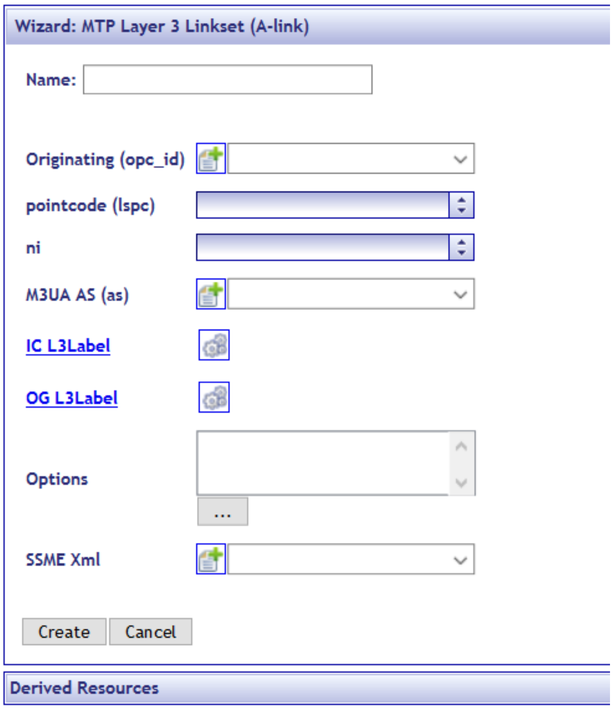


Figure 19 MTP Layer 3 Linkset A-Link Wizard

Originating (opc_id):	The originating point should be set from the drop-down list to the originating point resource that was configured in the Originating Point resource.
Pointcode (lspc):	The linkset pointcode should be set to the adjacent interconnect pointcode.
NI:	The network indicator should be set to the code required by for the interconnect. (0) International network (1) Spare (for international use only) (2) National network (3) Reserved for national use
M3UA AS (as):	The AS name of the M3UA association.
IC L3Label:	Not required on MG/MGC.
OG L3Label:	Not required on MG/MGC.
Options:	Additional options for configuring the A-Link
SSME Xml:	Not required on MG/MGC

Each linkset is configured against the required originating point defining the linkset's pointcode and network indicator.

A linkset can either be configured as a combined linkset or as a primary/secondary linkset. A combined linkset behaves as a single linkset at the MTP L3 layer split across two linksets. Check with the interconnect provider if the A link is a combined linkset or not before applying this configuration option. To setup as a combined linkset enter the Linkset resource and select the linkset you wish to combine with another linkset.

Combined:	The linkset id should be selected from the drop-down list that you wish to combine with the current linkset.

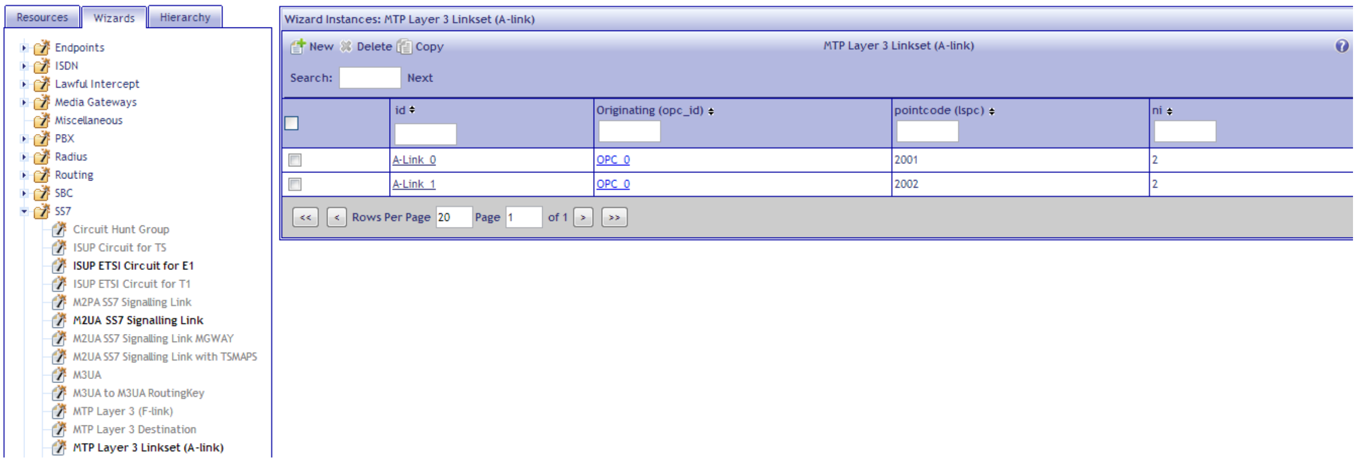


Figure 20 Finished Wizard.

The completed wizard shows the originating point and the pointcode and NI values of the A-links.

5.3.3 Configuring Signalling Links

5.3.3.1 M2UA Signalling Links

The signalling links are configured the same way for all linkset interconnects using the M2UA SS7 Signalling Link wizard. Please note that the timeslot that is configured below will also need to be configured on the media gateway.

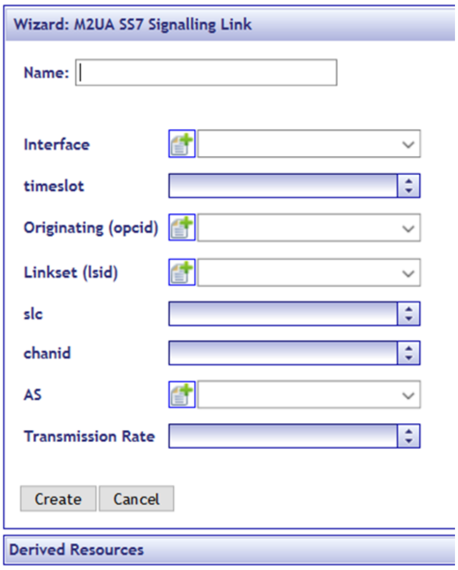


Figure 21 M2UA SS7 Signalling Link Wizard

The M2UA SS7 Signalling Link wizard allows you to configure the following options.

Interface:	This should be configured to the TDM interface that the signalling link is on. This must match the interface configured on the remote media gateway.
Timeslot:	This should be configured to the signalling timeslot that the signalling link is on. This must match the timeslot configured on the remote media gateway.
Originating Point:	The originating point should be set from the drop-down list to the originating point resource that was configured in the Originating Point wizard.
Linkset:	The linkset id should be set from the drop-down list to the linkset that the signalling link belongs to.
SLC:	The signalling link code must be the same value at each end of an interconnect to identify the individual signalling link with in the linkset.
Card Channel Id:	The card channel id (chanid) should be set in a linear fashion from 0-127 for each signalling link on a card.

AS:	This should be set from the drop-down list and associate to the media gateway on which the Signalling Link resides.
Transmission Rate:	Is the speed of the signalling link: 56k or 64k.

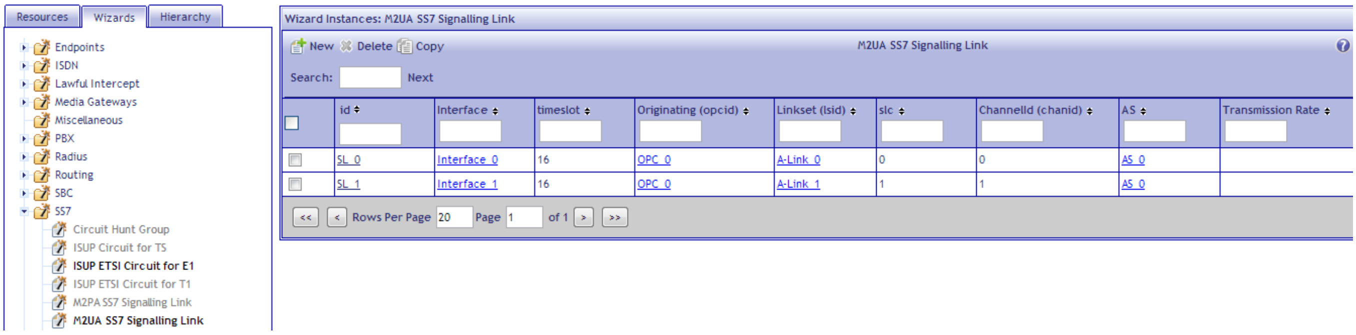


Figure 22 Finished Wizard.

The completed wizard shows the timeslot, originating point, linkset and slc values.

5.3.3.2 SLT Signalling Links

The signalling links are configured the same way for all linkset interconnects using the SLT SS7 Signalling Link wizard. Please note that the timeslot that is configured below will also need to be configured on the media gateway.

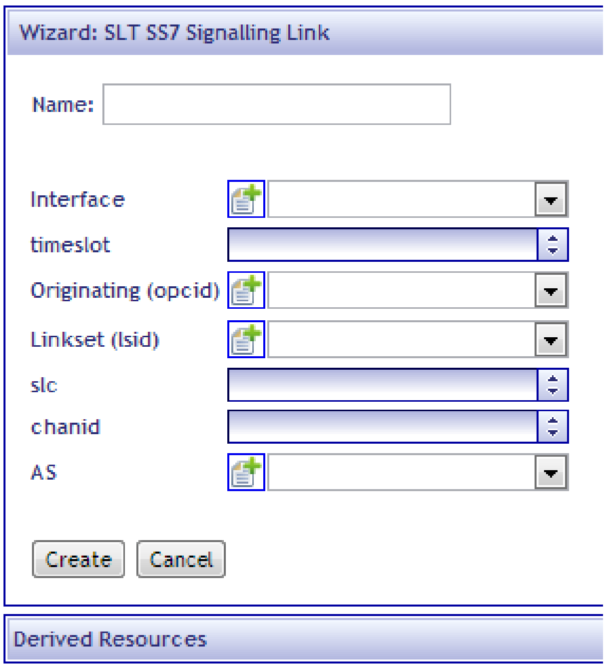


Figure 23 SLT SS7 Signalling Link Wizard

The SLT SS7 Signalling Link wizard allows you to configure the following options.

Interface:	This should be configured to the TDM interface that the signalling link is on.
Timeslot:	This should be configured to the signalling timeslot that the signalling link is on. This must match the timeslot configured on the remote media gateway.
Originating Point:	The originating point should be set from the drop-down list to the originating point resource that was configured in the Originating Point resource.
Linkset:	The linkset id should be set from the drop-down list to the linkset that the signalling link belongs to.
SLC:	The signalling link code must be the same value at each end of an interconnect to identify the individual signalling link with in the linkset.
Channel Id:	The channel id (chanid) should be set in a linear fashion from 0-127 for each signalling link on a media gateway.

AS:	This should be set from the drop-down list and associate to the media gateway on which the Signalling Link resides.

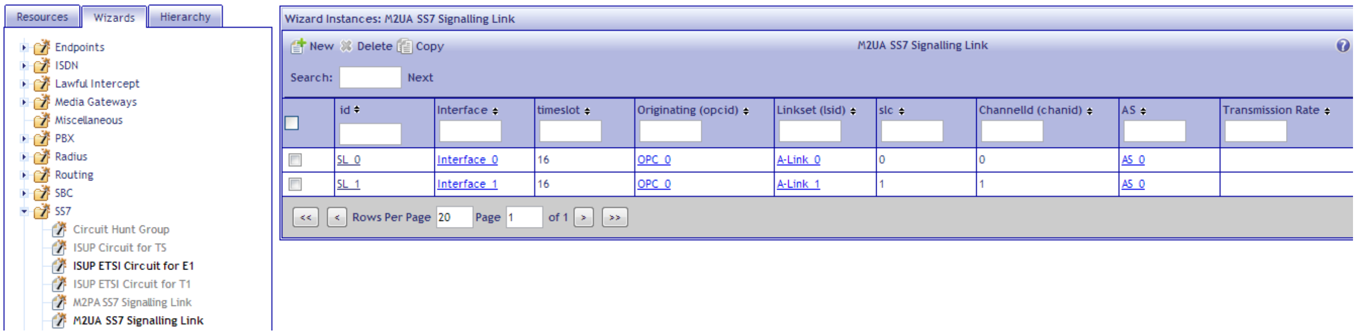


Figure 24 Finished Wizard.

The completed wizard shows the timeslot, originating point, linkset and slc values.

5.3.4 Configuring the Destination Point Resource

The MTP Layer 3 Destination Wizard is used to configure the destination points of the SS7 interconnect.

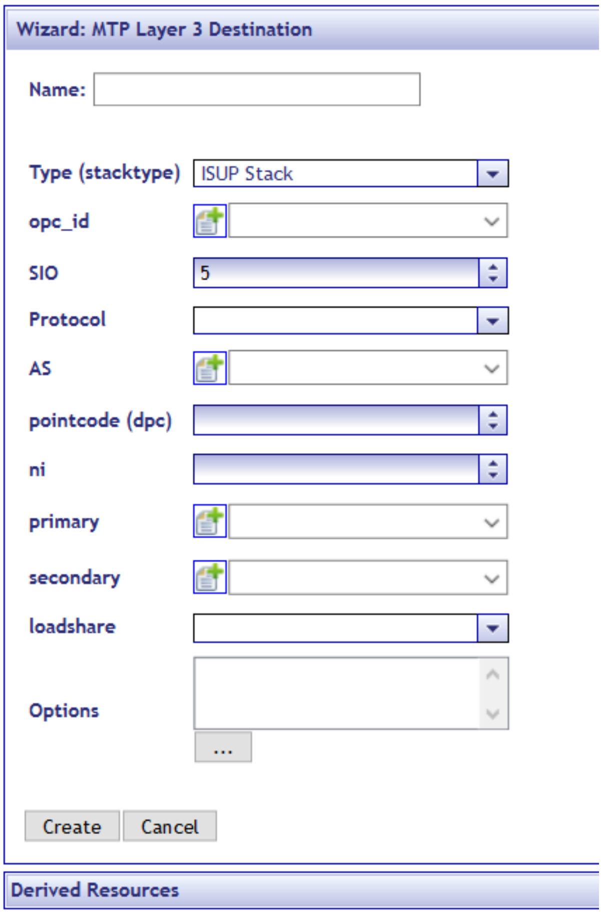


Figure 25 MTP Layer 3 Destination Wizard

Name:	This defines the name of the destination point and is recommended that it reflects the interconnect carrier name.
Type(stacktype):	This should always be defaulted to ISUP Stack unless advised by Squire Support desk.
Opc_id:	This is a drop-down list where the OPC of the SVI is associated with the destination point.

SIO:	Service Indicator Octet value: 5 for ISUP, 3 for SCCP.
Protocol:	This defines the stack protocol variant. For ETSI it should be set as White Book and for ANSI it should be set as ANSI.
Pointcode(dpc):	This defines the pointcode of the remote destination and should be populated from information provided by the remote provider.
NI:	This is the network indicator value and will be supplied by the remote provider.
Primary:	This defines the primary linkset.
Secondary:	This defines the secondary linkset.
Loadshare:	This defines whether the linkset associated with the destination are combined. This information will also be obtainable from the remote provider.
Options:	Additional options for configuring the destination

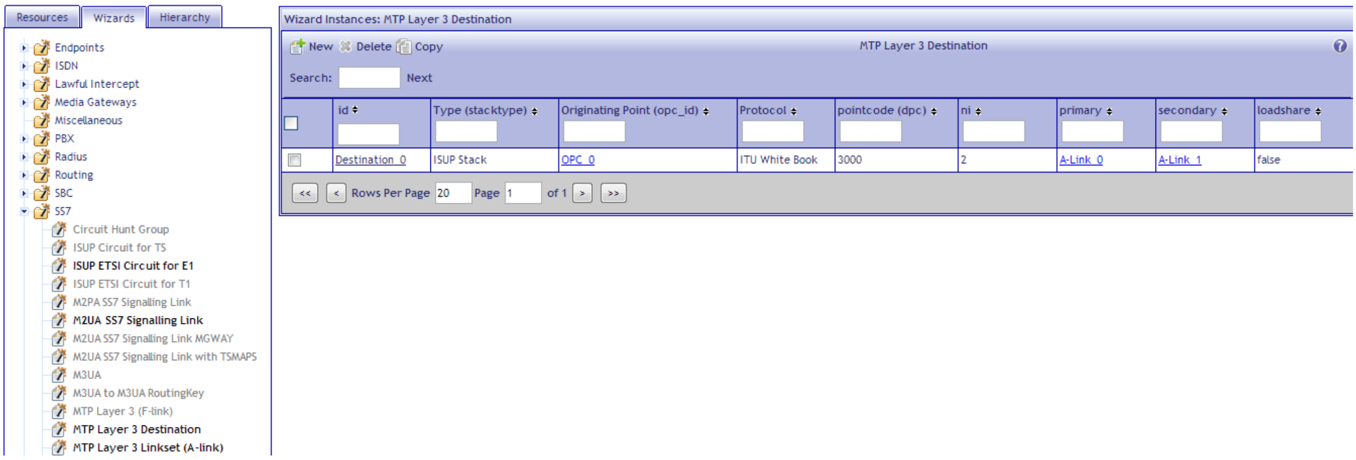


Figure 26 Finished Wizard.

The completed wizard shows the configuration applied to the destination point, the pointcode and the selected linkset for this destination.

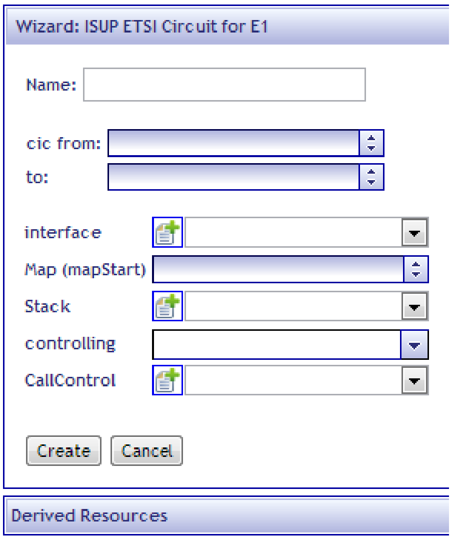
5.3.4.1 Routesets

To configure the routes from the linkset to the destination the primary and secondary attributes are used to point to the desired configured linkset resource. If the traffic is to be load shared equally across the two linksets the loadshare flag is set to True. If the loadshare flag is set to false, all traffic will attempt to route through the primary linkset first.

5.3.5 Configuring Circuits for an E1

The ISUP ETSI Circuits for E1 wizard is used to configure the circuits within the SVI for an E1. The circuits are the bearer channels that will carry the speech and must be configured to allow for speech to be passed through the SVI.

Figure 27 ISUP ETSI Circuit for E1



Name:	This defines the name of the range of circuits defined on this E1. We advise that the naming identifies the circuits as SS7.
CIC From:	This defines the first CIC of the range of CIC on the E1. Example: CIC are 1-31 and 33-63 the CIC range to configure are 0-31 and 32-63. This is due to the SVI including synchronisation timeslots and handling them through configuration.
To:	This defines the end of the CIC range.
Interface:	This defines the E1 within the SVI on which the circuits are located.
Map(Start):	This should be set the same as the interface value.
Stack:	The stack should be configured to be the destination point to which the circuits belong.
Controlling:	This defines the controlling end of the circuit in the event of glare. Controlling if the SVI controls, non-controlling is the remote end takes control. In addition, the controlling can be set to odd or even only.

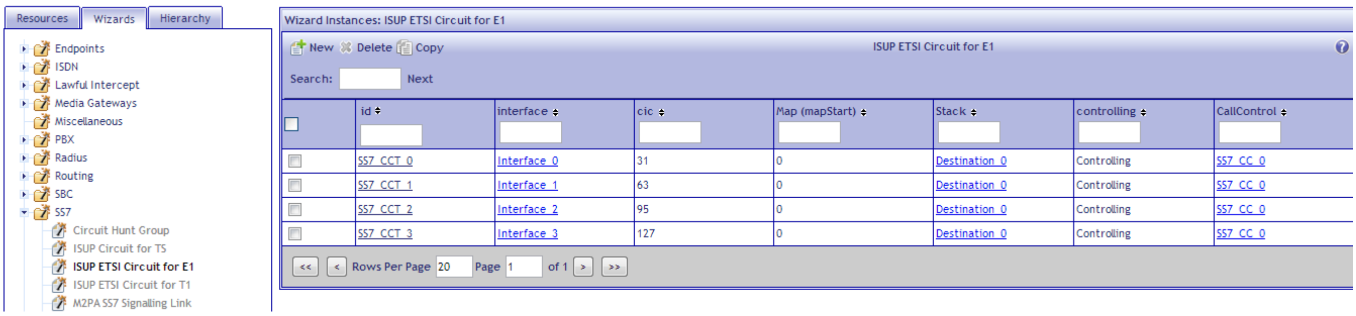


Figure 28 Finished Wizard.

The completed wizard shows the CIC, interface, and stack on which the circuits are located. The configuration should be applied at this point using the Save and Commit option.

5.3.6 Configuring Circuits for a T1

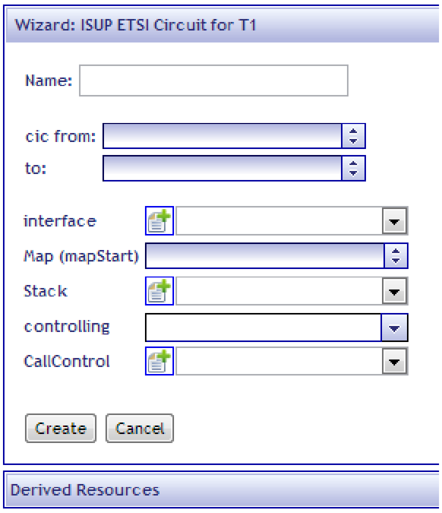
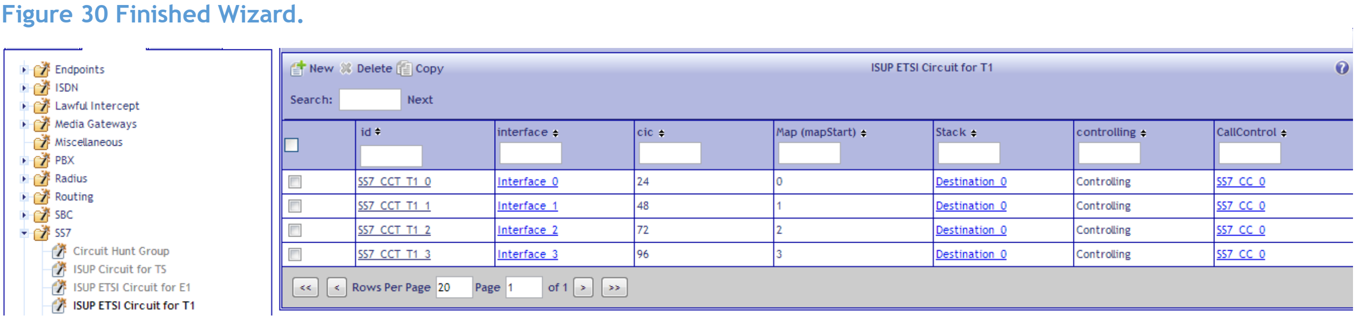


Figure 29 ISUP ETSI Circuit for T1 Wizard

The ISUP ETSI Circuits for T1 wizard is used to configure the circuits within the SVI for an E1. The circuits are the bearer channels that will carry the speech and must be configured to allow for speech to be passed through the SVI.

Name:	This defines the name of the range of circuits defined on this T1. We advise that the naming identifies the circuits as SS7.
CIC From:	This defines the first CIC of the range of CIC on the E1/T1. Example: CIC are 1-24 and 25-48 the CIC range to configure are 1-24 and 25-48.
To:	This defines the end of the CIC range.
Interface:	This defines the T1 within the SVI on which the circuits are located.
Map(Start):	This should be set the same as the interface value.
Stack:	The stack should be configured to be the destination point to which the circuits belong.
Controlling:	This defines the controlling end of the circuit in the event of glare. Controlling if the SVI controls, non-controlling is the remote end takes control. In addition, the controlling can be set to odd or even only.

Figure 30 Finished Wizard.



The completed wizard shows the CIC, interface, and stack on which the circuits are located. The configuration should be applied at this point using the Save and Commit option.

5.3.7 Configuring Hunt Groups

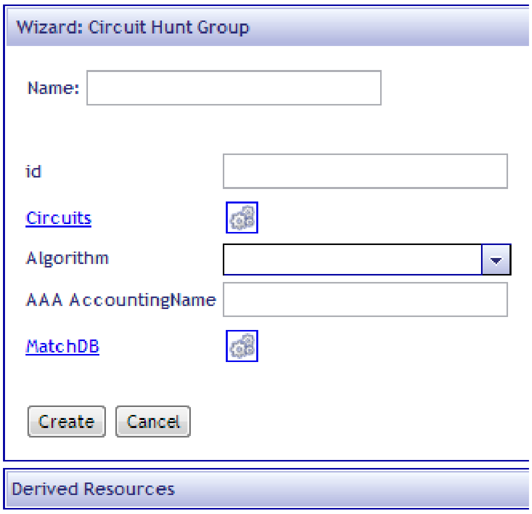
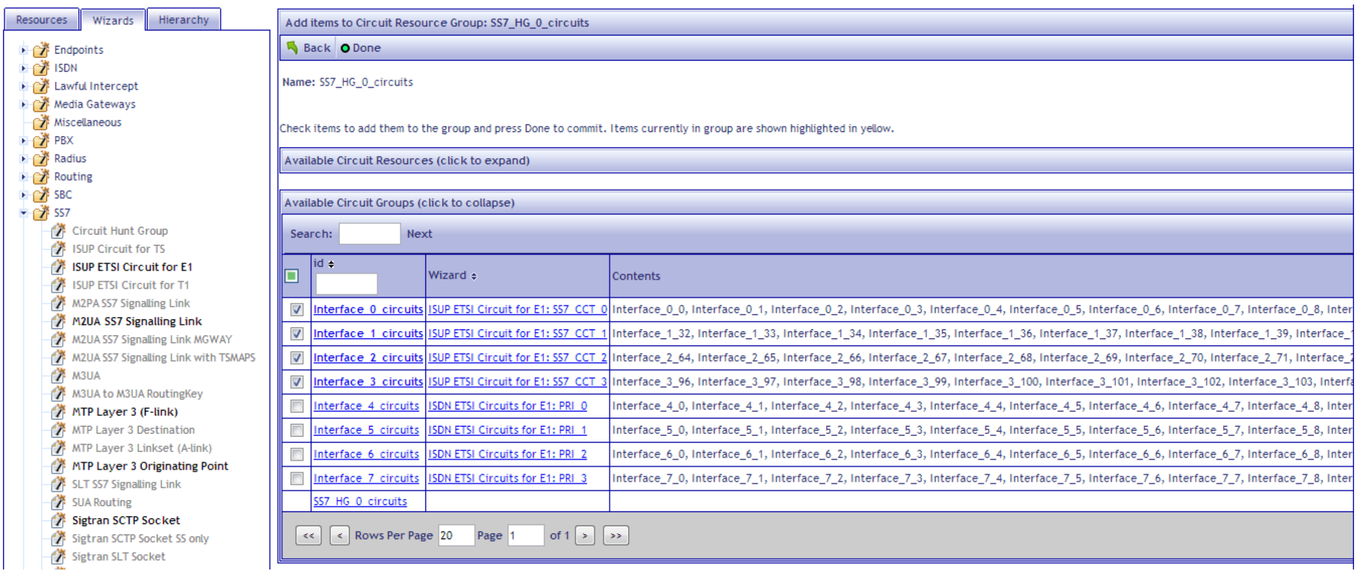


Figure 31 Circuit Hunt Group Wizard

The Circuit Hunt Group wizards is used to define a range of circuits into a hunt group for use in routing.

Name:	Defines the name of the hunt group.
Id:	Defines the name for the hunt group and in advanced usage can set a specific value to the instance of the hunt group in the OAM interface of the SVI.
Circuits:	Allows definition of the circuits to be included in the hunt group.



Algorithm:	Defines the method by which calls are routed out to the circuits defined within a hunt group. Hunt High uses the highest available circuit instance available, where hunt low uses the lowest instance available.
AAAAccountingName:	This defines a name to use for the hunt group when outputting CDR to RADIUS billing servers. RADIUS solutions primarily use IP to identify an endpoint this field can be used to emulate an IP for this hunt group.
MatchDB:	This defines a further set of rules for manipulating call information prior to sending the call to the hunt group. This is where the normalisation rules are configured onto.

Figure 32 Circuit Resource Group wizard

The complete wizard will show the range of circuits within the hunt group, at this point the configuration should be applied using the Save and Commit option

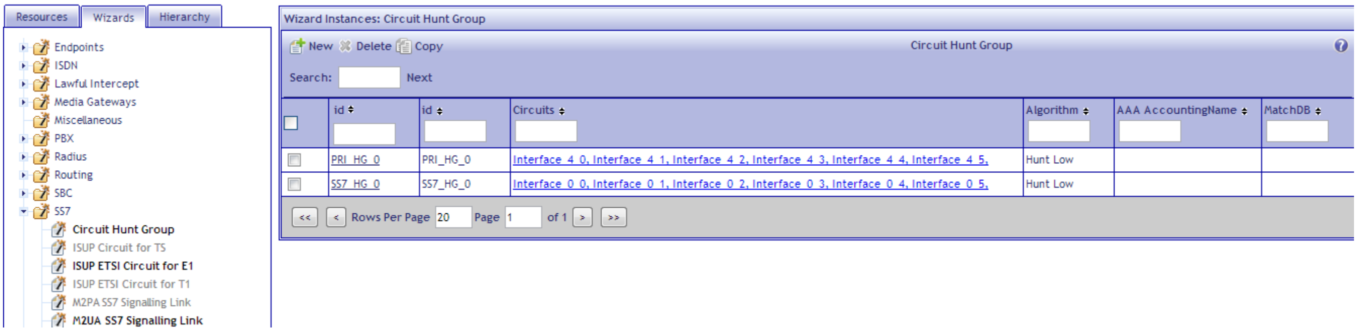


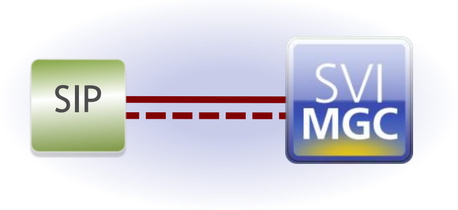
Figure 33 Finished Wizard.

6.0 SIP Interconnect Configuration

The following section describes how to configure the SIP interconnect within the SVI MGC.

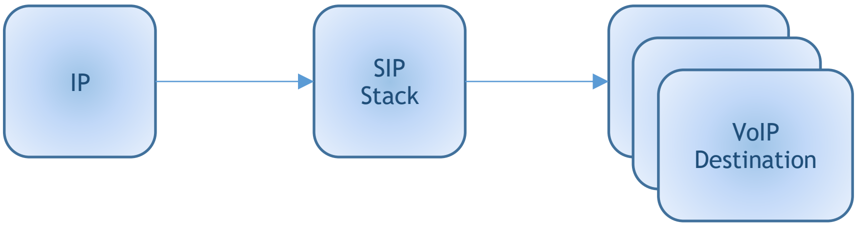
6.1 SVI Network Modelling

The SIP interconnect can be modelled with three resources, an IP socket, a VoIP Stack and VoIP Destinations. An IP socket and a VoIP Stack are required to setup the SVI to send and receive calls on a certain physical interface over a certain port. The VoIP Destinations identify a single or a range of users allowed to make connections to the MGC.



SIP Interconnect

The following shows how the resources are modelled within the SVI MGC for a SIP interconnect.



6.1.1 IP Socket

The IP Socket defines the socket to open for listening on the SVI.

6.1.2 VoIP Stack

The VoIP Stack defines on which IP interface the SIP Stack resides and a default destination port.

6.1.3 VoIP Destination

The VoIP Destination is the configuration identifying a single or range of endpoints.

6.2 Configuring the SIP Interconnect

The following runs through using the SIP interconnect wizards to configure a SIP interconnect.

6.2.1 Configuring the SIP Stacks

6.2.1.1 UDP SIP Stack Configuration

The SIP Stack wizard is used to setup a UDP SIP Stack on the SVI for use with endpoints configured to communicate using UDP messaging.

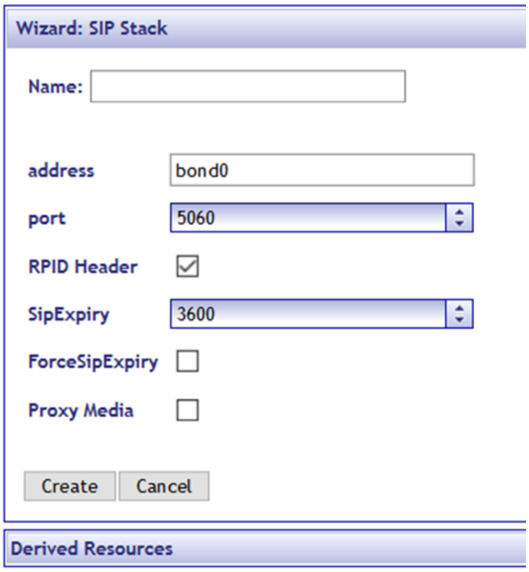


Figure 34 SIP Stack Wizard

Through this wizard the SIP stack can be configured onto a particular IP Interface (address), assigned a port to listen on and set several options for RPID header and timer control.

Address:	Defines the IP address on which the SIP Stack will reside, this can be configured with an IP address directly or the name of the underlying IP interface (by default is bond0).
Port:	Defines the port on which the SVI will listen for incoming SIP messaging.
RPID Header:	Check if the incoming destination is trusted (No setting of VoIP Destination flag "Non-Trusted-Domain" and the RPID header is set to True the SVI will attempt to interwork to the outgoing message the "P-Asserted-Identity" Header.
SIP Expiry:	Sets the timer value for the forced registration expiry.
ForceSIPExpiry:	If this attribute is set to true for any incoming registrations the expiry will be set to the value from the SIP Expiry attribute.
Proxy Media:	Defines if this stack will be used for signalling only or signalling and media as well.

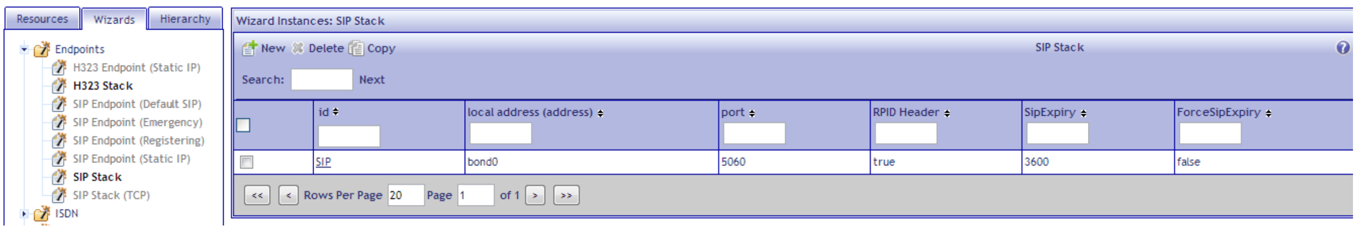


Figure 35 Finished Wizard.

The completed wizard shows the SIP Stack configuration with the IP interface and port defaults for the SVI.

6.2.1.2 TCP SIP Stack Configuration

The SIP TCP Stack wizard is used to setup the SVI for use with endpoints configured to communicate using TCP.

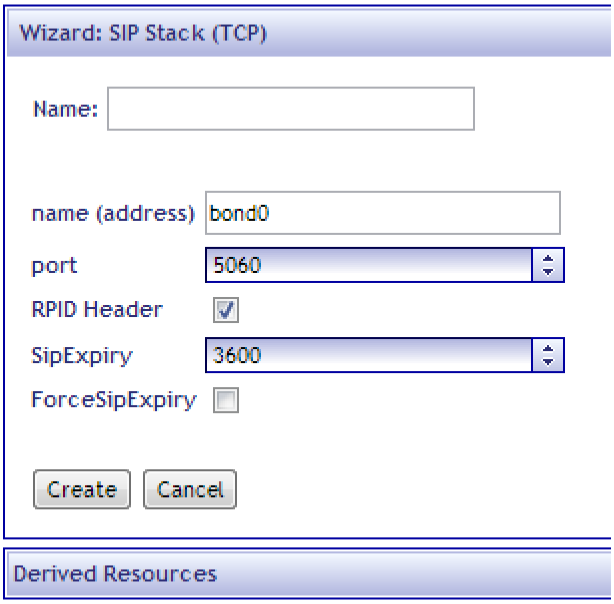


Figure 36 SIP Stack (TCP) Wizard

Through this wizard the SIP TCP stack can be configured onto a particular IP Interface (address), assigned a port to listen on and set several options for RPID header and timer control.

For the TCP SIP Stack to operate a UDP SIP Stack must be configured onto the same IP Interface and port, even if the UDP SIP Stack is not used it must exist for TCP SIP Stack to operate. Follow the previous section configuring a UDP SIP Stack and this section configuring a TCP SIP Stack section when TCP connections are required on new IP interfaces.

Address:	Defines the IP address on which the SIP Stack will reside, this can be configured with an IP address directly or the name of the underlying IP interface (by default is bond0).
Port:	Defines the port on which the SVI will listen for incoming SIP messaging.
RPID Header:	Check if the incoming destination is trusted (No setting of VoIP Destination flag "Non-Trusted-Domain" and the RPID header is set to True the SVI will attempt to interwork to the outgoing message the "P-Asserted-Identity" Header.
SIP Expiry:	Sets the timer value for the forced registration expiry.
ForceSIPExpiry:	If this attribute is set to true for any incoming registrations the expiry will be set to the value from the SIP Expiry attribute.

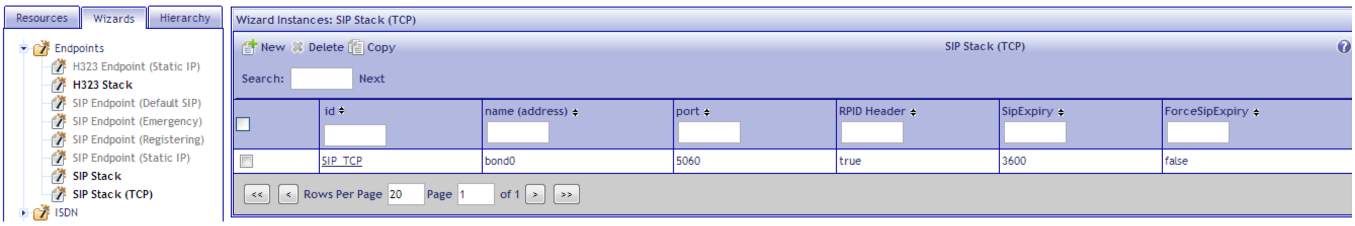


Figure 37 Finished Wizard.

The completed wizard shows the IP interface and port, these will match a SIP UDP Stack preconfigured or a stack you will have newly configured from the previous section.

6.2.1.3 TLS SIP Stack Configuration

There are two wizards that are used to setup the SVI for use with TLS, the first is the TLS Credentials and the second is the SIP TLS Stack.

6.2.1.3.1 TLS Credentials

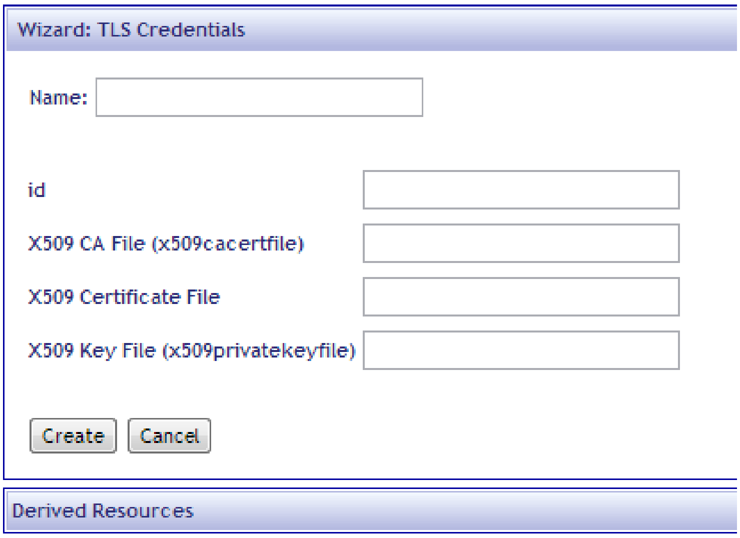


Figure 38 TLS Credentials Wizard

Through the TLS Credentials wizard, the TLS Credentials can be identified and used to attach to a TLS SIP Stack for authentication.

Name:	Defines the name of the TLS Credentials Resource for reference by the SIP Stack later.
X509 CA File:	Defines the path to the X509 CA Certification File
X509 Certificate File:	Defines the path to the X509 Certificate File.
X509 Key File:	Defines the path to the X509 Private Key File.

The paths defined for each of these attributes refers to an absolute path on the SVI, for example, /home/squire/certificates/

Figure 39 Finished Template.

The completed template shows the paths to each of the certificate files located on the SVI.

6.2.1.4 SIP-I Stack Configuration

The SIP Stack can be modified from creation within any of the above SIP Stack wizards, the below will allow for adding encapsulated ISUP messages within SIP.

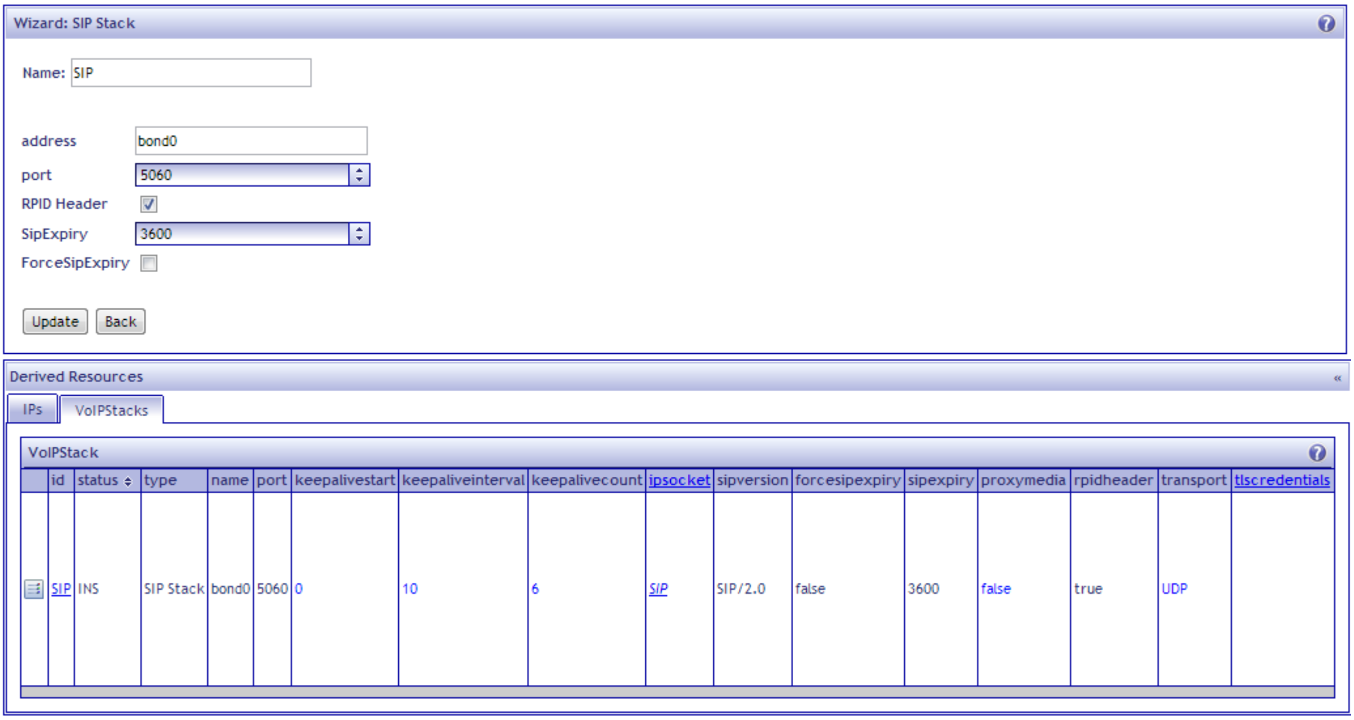


Figure 40 SIP Stack Derived Resources Wizard

Under the SIP Stack resource, you will need to open derived resources and select the VoIP Stack tab, here click on the Id of the SIP Stack to modify its attributes.

Under the SIP Stack attributes page scroll down to find the following attributes, configuring these attributes configures the SIP Stack to enclose the defined variant of ISUP within the SIP messaging.

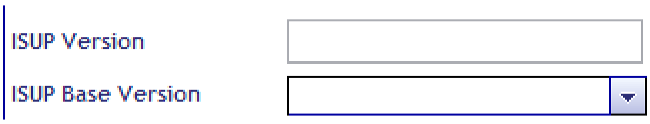


Figure 41 SIP Stack ISUP Version fields

Please refer to the SVI Resource Glossary HERE for a full list of the possible configuration options.

6.2.2 Configuring SIP Endpoints

The SIP Endpoint wizard is used to configure a destination for connection to the SVI. In SIP there are several ways that can be used to connect; there are a range of templates that cater for each. The below will cover configuring a static endpoint (non-registering) and a registering endpoint.

6.2.2.1 Configuring a static SIP Endpoint

The SIP Endpoint (Static IP) wizard should be used to create a standard SIP endpoint with no registration. This allows for a single SIP destination to be defined for allowing connection into the SVI. At this point any IP addresses being added as endpoint should also be considered as adding to any firewall rules.

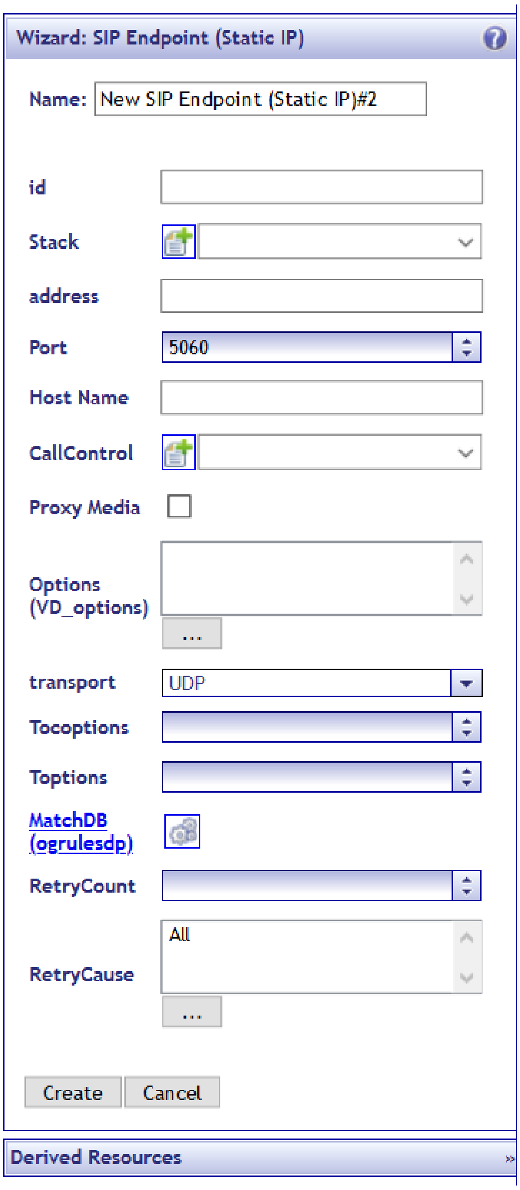


Figure 42 Static SIP Endpoint Wizard

Not all attributes have to be defined in each configuration as some can be left blank. The following is a brief overview of the fields you see here and their respective purposes:

Name:	First give the Wizard a unique Name that is easy to remember for future reference.
Id:	This attribute is used to identify the SIP Endpoint within the SVI-MS and should be given a unique and memorable name. If this field is left blank, the "name" attribute will be used to populate it.
Stack:	From the drop-down Stack menu select the desired stack which this endpoint will run its service to.
Address:	Define the IP address of the VoIP destination.
Port:	Define the IP Port of the VoIP destination. Most SIP services run on Port 5060.
Hostname:	hostname to be resolved as IP address
Call Control:	From the drop-down menu CallControl select the SIP call control
Proxy media:	This is to ensure that all media will be passed through the SVI
Options:	The Options box allows you to select the additional call features you wish to enable. For these advanced options it is recommended to refer to the Resource Glossary HERE.
Transport:	UDP or TCP

Tocoptions:	This attribute when configured will cause an options message to be sent to the endpoint whether in or out of call.
Toptions:	This attribute when configured will cause an options message to be sent within an active call on the VoIP Destination configured against.
MatchDB:	Allows DB parameters to be attached to the Endpoint for the purpose of manipulating call parameters for outbound calls.
RetryCount:	If a call does not connect, this sets the number of times that the call will be retried.
RetryCause:	Linked to the RetryCount, the RetryCause allows you to select the conditions which will trigger further call attempts. By default, this is set to all, but this box allows you to change this.

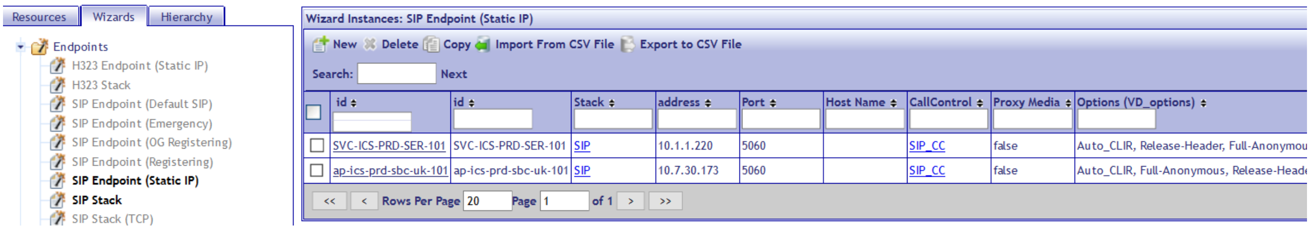


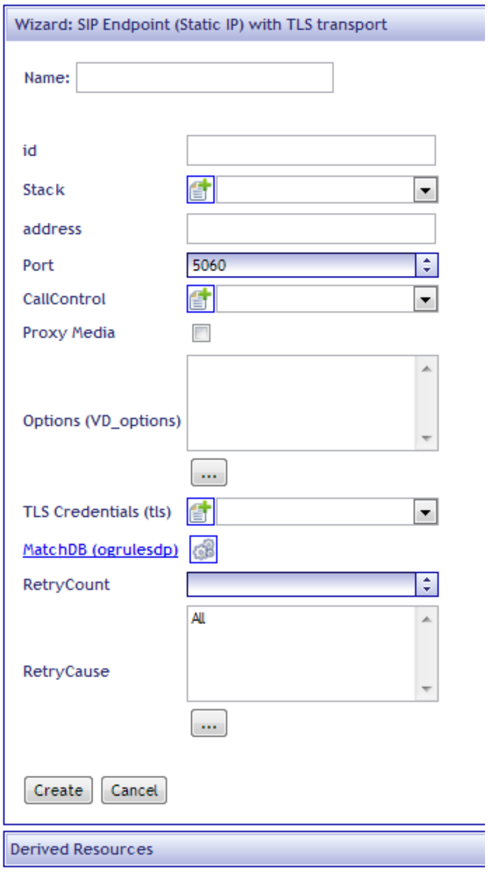
Figure 43 Finished Wizard.

The completed wizard shows all the details for the endpoint that have just been configured including the IP and port as well as any additional advanced options added to the endpoint.

To configure the Static SIP Endpoint as either UDP or TCP enabled; access the derived resources once configured and change the transport method to the required. By default, all SIP Endpoints are configured as UDP. TLS Endpoints have a specific wizard as there are additional options for TLS; see below for how to configure a TLS Endpoint.

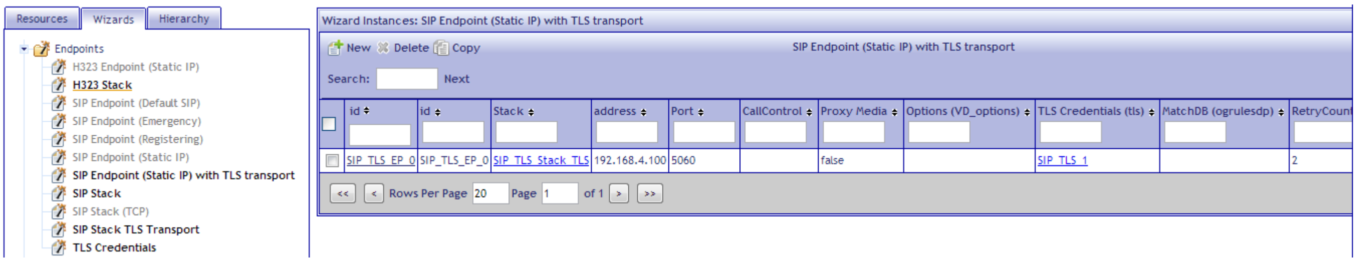
6.2.2.2 Configuring a TLS SIP Endpoint

The SIP Endpoint (Static IP) with TLS Transport wizard should be used to identify individual endpoints connecting to the SVI with TLS.



Name:	First give the Wizard a unique Name that is easy to remember for future reference.
Id:	This attribute is used to identify the SIP Endpoint within the SVI-MS and should be given a unique and memorable name. If this field is left blank, the "name" attribute will be used to populate it.
Stack:	From the drop-down Stack menu select the desired stack which this endpoint will run its service to.
Address:	Define the IP address of the VoIP destination.
Port:	Define the IP Port of the VoIP destination. Most SIP services run on Port 5060.
Call Control:	From the drop-down menu CallControl select the SIP call control
Proxy media:	This is to ensure that all media will be passed through the SVI
Options:	The Options box allows you to select the additional call features you wish to enable. For these advanced options it is recommended to refer to the Resource Glossary HERE.
Name:	First give the Wizard a unique Name that is easy to remember for future reference.
TLS Credentials:	Identifies a TLS Credential Resource for use directly by the Endpoint, this will override the TLS Credential reference on the VoIP Stack the Endpoint is associated with.
RetryCount:	If a call does not connect, this sets the number of times that the call will be retried.
RetryCause:	Linked to the RetryCount, the RetryCause allows you to select the conditions which will trigger further call attempts. By default, this is set to all, but this box allows you to change this.

Finished Wizard



6.2.2.3 Configuring an incoming Registering SIP Endpoint

The SIP Endpoint (Registering) should be used to define a registering endpoint, this will allow for users to connect to the SVI without the need for an IP address to be configured. The wizard will also allow for the configuration of options to define whether the endpoint is behind NAT or not allowing for automatic handling of the message response required to negotiate a call successfully.

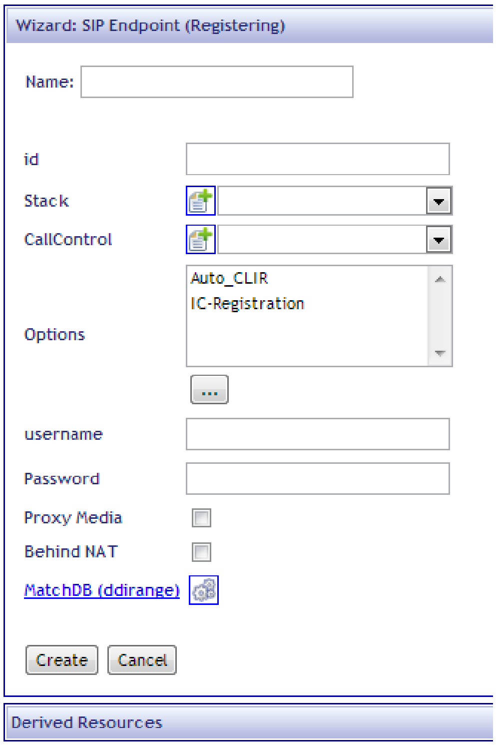


Figure 44 SIP Registering Endpoint Wizard

Not all attributes have to be defined in each configuration as some can be left blank. The following is a quick overview of the fields you see here and their respective purposes:

Name:	First give the Wizard a Name that is easy to remember for future reference.
Id:	This attribute is used to identify the SIP Endpoint within the SVI-MS and should be given a unique and memorable name. If this field is left blank, the "name" attribute will be used to populate it.
Stack:	Select the desired stack which this endpoint will run its service to, from the Drop-down Stack menu.
Call Control:	From the drop-down menu CallControl select the SIP call control.
Proxy media:	This ensures that all media will be passed through the SVI
Options:	The Options box allows you to select the additional call features you wish to enable. The full list of explanations for the options can be found in the Resource Glossary HERE.

Username:	This is the username used to identify the SIP Endpoint, typically the DDI or CLI number.
Password:	Enter a password to enable registering of the device with the SVI. This must be alpha-numerical.
Behind NAT:	Tick this box if your Endpoint is behind a NAT enabled firewall. We recommend always ticking this box.
MatchDB:	Allows DB parameters to be attached to the Endpoint for the purpose of manipulating call parameters for outbound calls.

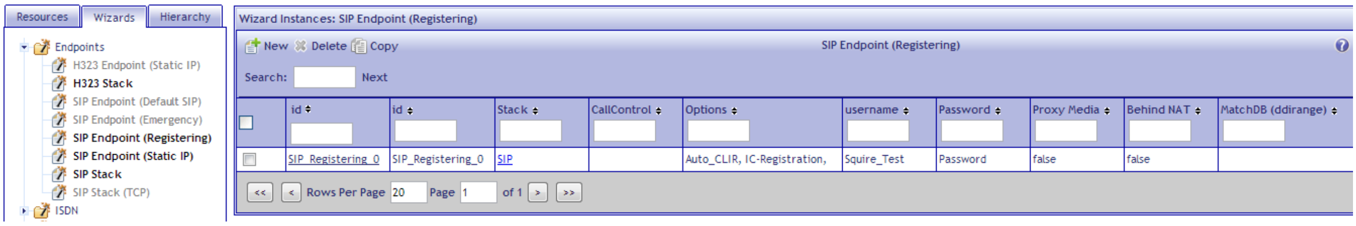


Figure 45 Finished Wizard.

The completed wizard shows the registration details and any other options set against the endpoint for how to behave.

6.2.2.4 Configuring an outgoing Registering SIP Endpoint

To configure the SVI MGC as an outgoing registering endpoint a wizard is available called SIP Endpoint (OG Registering). The below screen shot shows how the options differ to enable Outgoing Registration (OG-Registration).

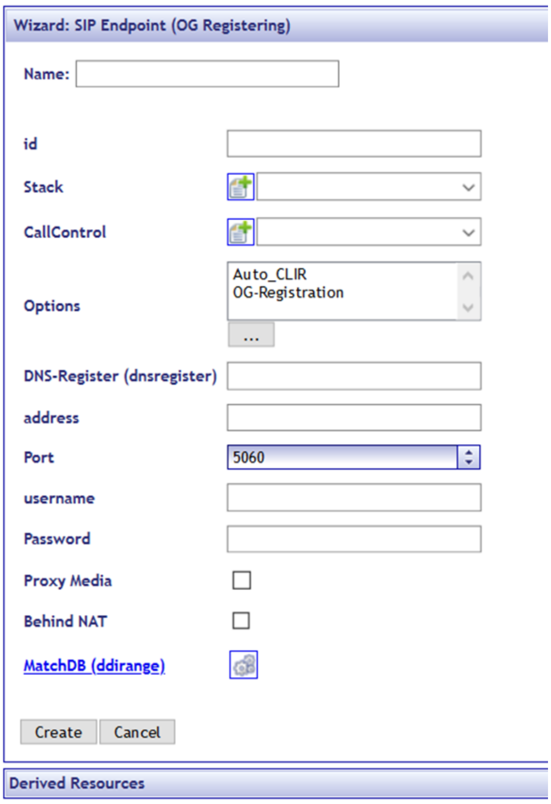


Figure 46 Outgoing Registering SIP Endpoint Wizard

Having created the endpoint with the correct options and filling the credentials for authentication you will require to add the remote address to the Endpoint for connecting. To do this access the derived resources and modify the VoIP Destination resource to add an address.

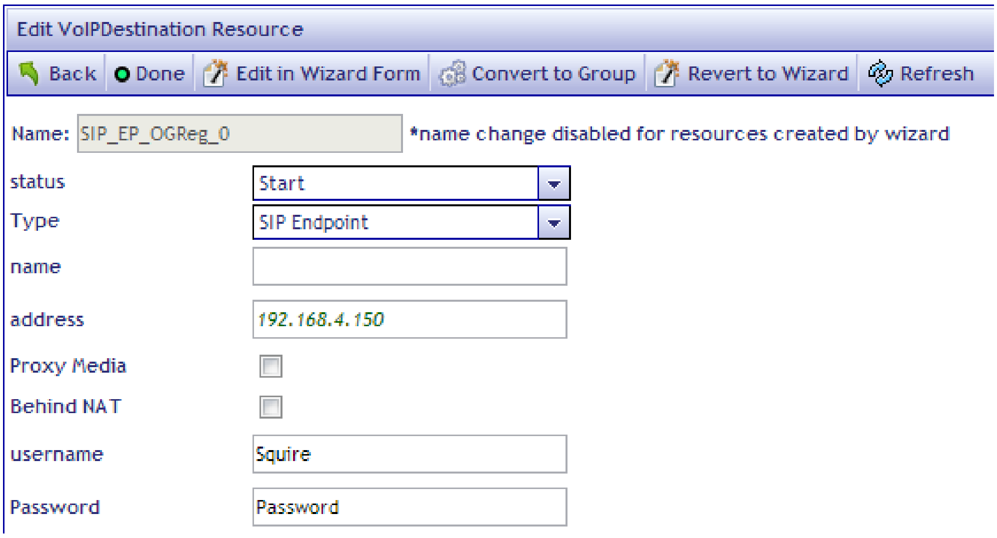


Figure 47 VoIP Destination Resource

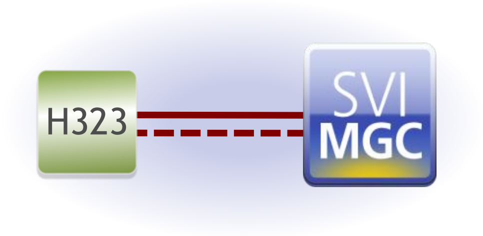
Having added the address to the VoIP Destination click Done and Update the wizard. The SVI will then begin to send Registration requests to the remote address with the credentials configured.

7.0 H323 Interconnect Configuration

The following section describes how to configure the H323 interconnect within the SVI MGC.

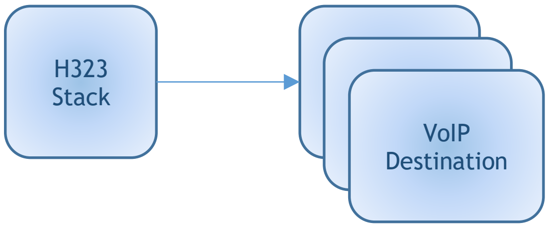
7.1 SVI Network Modelling

The H323 interconnect can be modelled with two resources, a VoIP Stack and VoIP Destinations. Only a VoIP Stack is required to setup the SVI to send and receive calls on a certain physical interface over a certain port. The VoIP Destinations identify a single or a range of users allowed to make connections to the MGC.



H323 Interconnect

The following shows how the resources are modelled within the SVI MGC for an H323 interconnect.



7.1.1 VoIP Stack

The VoIP Stack defines on which IP interface the H323 Stack resides and a default destination port.

7.1.2 VoIP Destination

The VoIP Destination is the configuration identifying a single or range of endpoints.

7.2 Configuring the H323 Interconnect

The following runs through using the H323 interconnect wizards to configure an H323 interconnect.

7.2.1 Configuring the H323 Stack

The H323 Stack wizard is used to create an H323 stack for H225 and H245 connections to be made to the SVI.

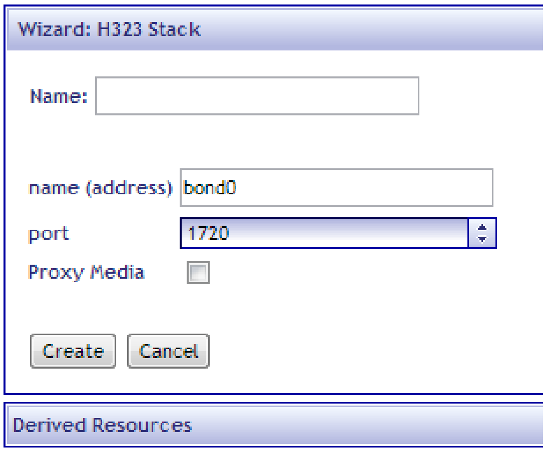


Figure 48 H323 Stack Wizard

Address:	Defines the IP address on which the H323 Stack will reside, this can be configured with an IP address directly or the name of the underlying IP interface (by default is bond0).
Port:	Defines the port on which the SVI will listen for incoming SIP messaging.
Proxy Media:	This should be unchecked in the SVI MGC.

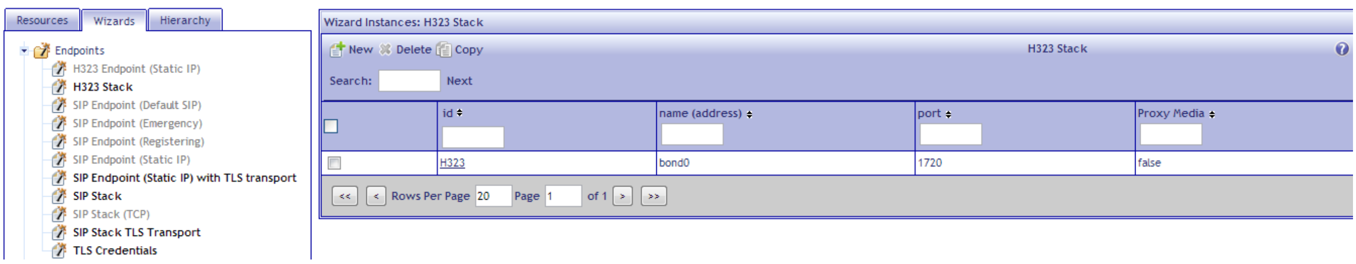


Figure 49 Finished Wizard.

7.2.2 Configuring an H323 Endpoint

The H323 Endpoint wizard should be used to create a standard H323 endpoint with no registration. This allows for a single H323 destination to be defined for allowing connection into the SVI. At this point any IP addresses being added as endpoint should also be considered as adding to any firewall rules.

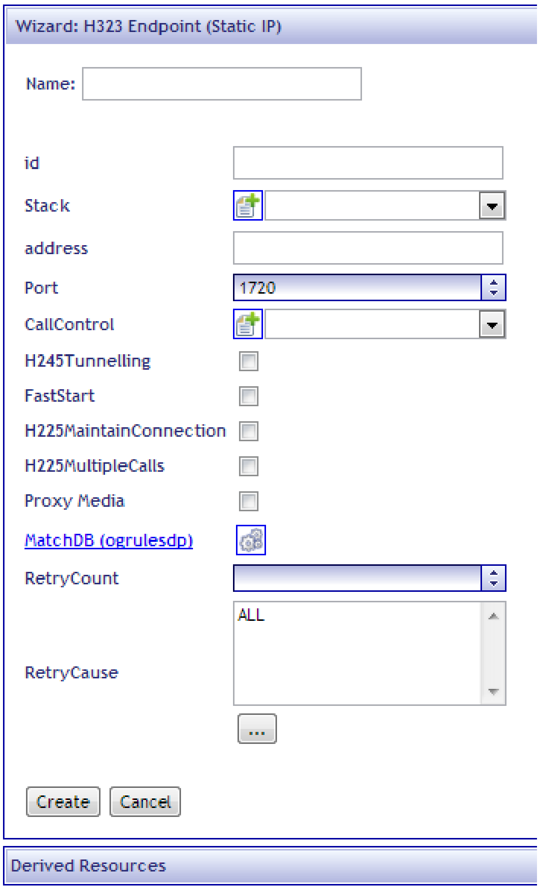


Figure 50 H323 Static Endpoint Wizard

Name:	First give the Wizard a Name that is easy to remember for future reference.
Id:	This attribute is used to identify the SIP Endpoint within the SVI-MS and should be given a unique and memorable name. If this field is left blank, the “name” attribute will be used to populate it.
Stack:	Select the desired stack which this endpoint will run its service to, from the Drop-down Stack menu.
Call Control:	From the drop-down menu CallControl select the H323 call control.
H245Tunnelling:	This defines whether H245 Tunnelling Procedures are enabled for all calls against this endpoint.
FastStart:	This defines whether Fast Start procedures are enabled for all calls against this endpoint.
H225MaintainConnetion:	This defines whether H225 Maintain Connection procedures are enabled for all calls against this endpoint.
H225MultipleCalls:	This defines whether H225 Multiple Calls procedures are enabled for all calls against this endpoint.
Proxy media:	This should be unchecked for use with the SVI MGC.
Options:	The Options box allows you to select the additional call features you wish to enable.
MatchDB:	Allows DB parameters to be attached to the Endpoint for the purpose of manipulating call parameters for outbound calls

.

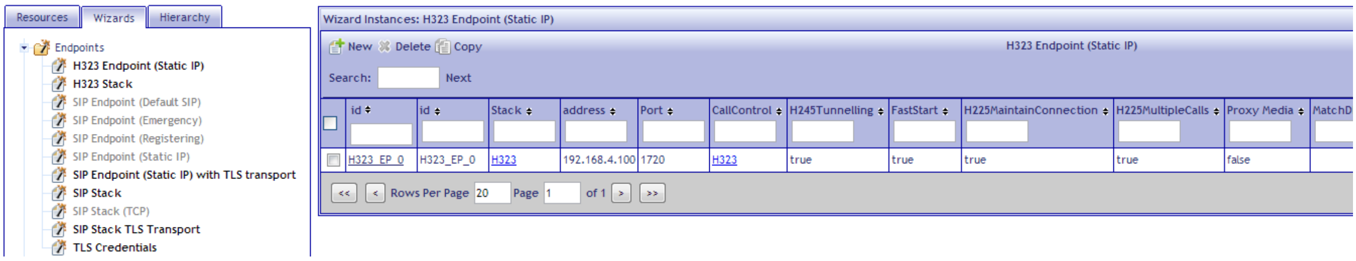


Figure 51 Finished Wizard.

The completed wizard shows the IP address and port as well as the various H323 options that can be set for an H323 endpoint.

8.0 TDM and VoIP Manipulation

Sometimes the TDM protocol default values and behaviour do not meet the requirements of an SS7 interconnect and as such must be modified to accommodate interconnected networks.

8.1 CallControl

The CallControl resource links with the other resources to impact the way in which a call progresses. You will have seen in previous screen shots that the CallControl reference can be set on the Circuits or VoIP Destination, this allows for different CallControl options to be set per circuit, a range of circuits or for all circuits affecting the call behaviour on calls passing through those circuits.

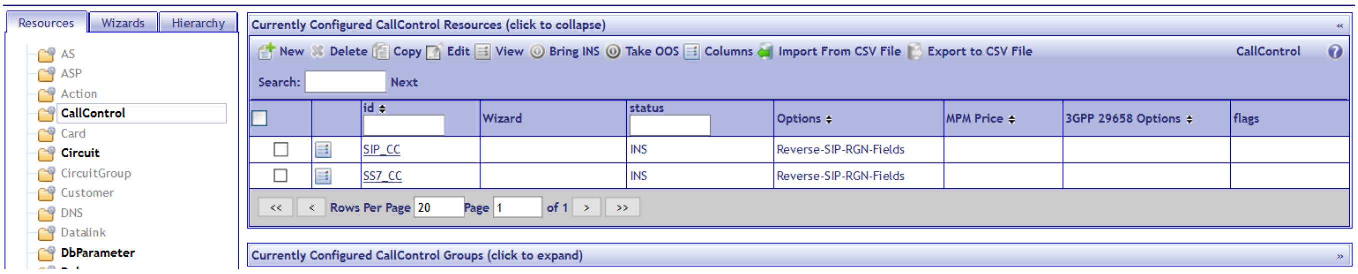


Figure 52 Call Control Resources

The CallControl resource is accessible through the Resource tab under the CallControl resource. There will be a predefined CallControl for SS7 and SIP factory configuration which will be assigned to all Circuits and Endpoints. This can be modified, or new resources can be added and configured onto Circuits.

For complete details on the configuration options within the CallControl resource please refer to the Resource Glossary HERE for TDM manipulation and VoIP manipulation.

8.2 Protocol

The protocol resource allows for manipulation of lower-level elements in the TDM behaviour. This resource is not configured by default and is accessible to configure through the **Edit** option under configuration.

Figure 53 GUI Options bar

The Protocol resource once configured can be associated to a CallControl resource allowing for the manipulations applied to the Protocol resource to be associated with Circuits or VoIP Destinations through the CallControl resource.

For complete details on the configuration options within the Protocol resource please refer to the Resource Glossary HERE for TDM manipulation VoIP manipulation.

9.0 Routing Configuration

The SVI MGC can be configured to route between SS7 and SIP protocols. The criteria used for routing can be incoming hunt group, number, or other information elements within messaging.

9.1 Resources

The following resources are what make up the routing engine within the SVI product range, each allow for more granular control over the routing decisions made within the SVI. The descriptions below start at the topmost level and work down.

The resources can be combined in different ways to allow for very powerful solutions within routing, the Routing Logic section describes how the engine searches through the resources and what decisions are made when matches are found, and the Routing Scenarios section will cover some specific examples to get started.

9.1.1 Routing Criteria (RC)

The Routing Criteria resource is the top level of the routing decision it is a container for a range of Db Parameters.

Decisions can be made at this level about preferred codec, type and location of CDR or an external location for writing to an SQL database.

9.1.2 Db Parameter (DP)

The Db Parameter resource is used to define incoming matching parameters and set outgoing manipulation for call routing.

The DP can match ingress and route to egress Hunt Groups, match Called or Calling Party Number, Match Nature of Address, match Calling Party Category and a range of other parameters within the SS7 and SIP messaging.

9.1.3 Hunt Groups (HG)

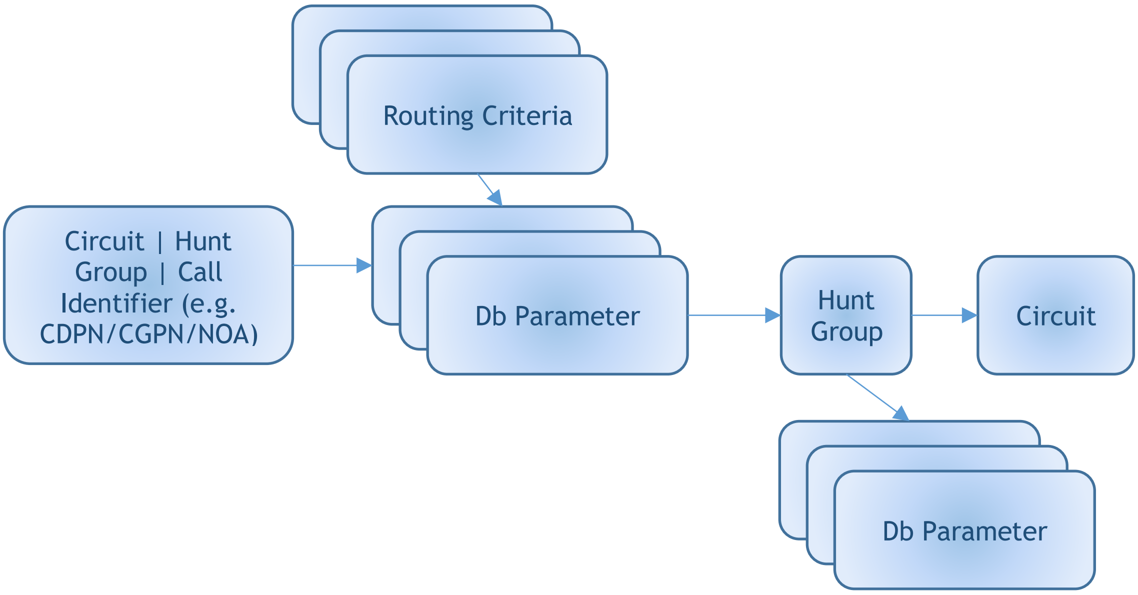
The Hunt Group resource is used to define a range of Circuits or a VoIP Destination. Circuits are attached to the Hunt Group from TDM protocols and VoIP Destinations from VoIP protocols, this allows for ease of configuration within routing using a common resource.

The HG can be used to define anything from a single circuit to a range of circuits covering a full SS7 interconnect or more. The Hunt Group also allows for a single or a range of VoIP Destinations to be added just the same as the Circuit can be.

9.1.4 SVI Routing Model

The following diagram shows how the resources associate to build up a routing entry within the SVI.

This diagram shows the direction of call through the resources and how the resources are built up and associated. The following is a text representation of this.



- Circuits are contained within a Hunt Group.
- Hunt Groups are referenced by Db Parameters as an Incoming Hunt or a Route
- Db Parameters are contained within a Routing Criteria.

This structure allows for a call to arrive on a given Circuit or VoIP Destination having direct relation to a physical timeslot or IP address and port which is then combined into a common resource, the Hunt Group. The SVI then allows for Hunt Groups to be attached as incoming hunt match or an outgoing route, it is possible to use a Hunt Group on an incoming match and an outgoing route at the same time.

The addition of Db Parameters on the outgoing Hunt Group is for Normalisation. Calls passing through a Hunt Group when selected as a Route will process any Db Parameters matches and manipulations modifying the call messaging before routing the call.

9.2 Setting up Routing

With the SS7 and the SIP interconnects configured it is now time to consider how the routing will be configured. There are several routing scenarios that can be employed using the SVI MGC, routing scenarios can be wholesale based where all traffic is routed to and from a single SIP endpoint through to user-based management of all call routing on a per user basis. The following section will guide on wizards to use to achieve some of these scenarios.

9.2.1 Bidirectional Routing

Bidirectional routing is where two or more hunt groups are configured, the easiest way to setup the bidirectional routing for wholesale routing is one hunt group containing all SS7 circuits and one containing all SIP VoIP Destinations. These Hunt Groups can then be used to configure bidirectional routing between to allow for full protocol to protocol routing.

9.2.1.1 Configuring Routing

The routing configuration for the bidirectional scenario is as follows:

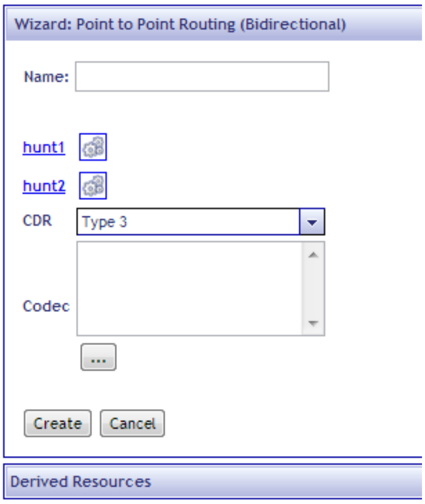


Figure 54 Bidirectional Routing Wizard

Bidirectional Routing between SS7_HG_0 and SIP_HG_0

Name:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route.
Hunt1:	To define one set of hunt groups. Select from a list of hunt groups that have been configured containing Circuits or VoIP Destinations to route from/to.
Hunt2:	To define the other set of hunt groups. Select from a list of hunt groups that have been configured containing circuits or VoIP Destinations to route to/from.
CDR:	This can be set to type 1-6 or turned off allowing for more detailed call data records to print for calls passing through this routing entry.
Codec:	This defines the preferred codec for calls to use when passing through this routing entry.

Figure 55 Finished Wizard.

The completed wizard shows the two hunt groups between which the routing will be configured on save and commit.

9.2.2 Unidirectional Routing

Unidirectional routing is where two or more hunt groups are configured, and routes can be configured to route in one direction between selected hunt groups. This can be used for routing an incoming E1 of traffic to a single SIP endpoint or vice versa but not both ways.

9.2.2.1 Configuring Routing

The routing configuration for the unidirectional scenario is as follows:

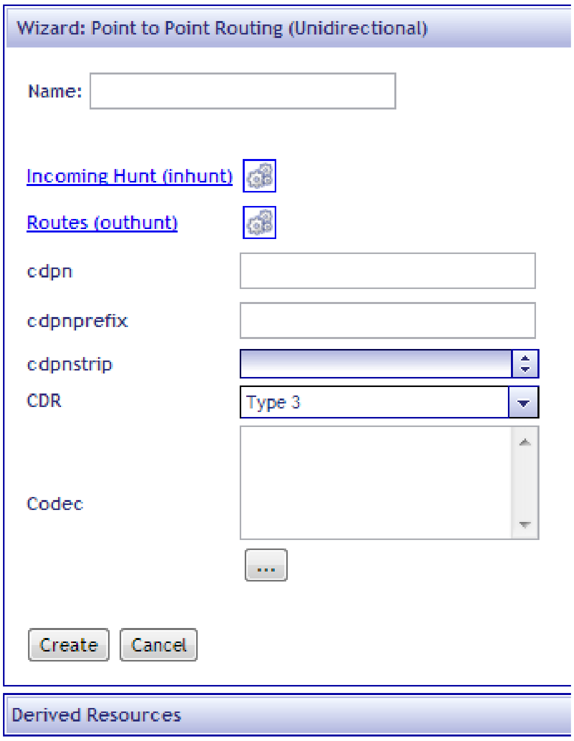


Figure 56 Unidirectional Routing Wizard

Unidirectional Routing between SS7_HG_0 and SIP_HG_0

Name:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route.
Incoming Hunt:	To define the incoming hunt groups. Select from a list of hunt groups that have been configured containing Circuits or VoIP Destinations to route from/to.
Routes:	To define the outgoing hunt groups. Select from a list of hunt groups that have been configured containing circuits or VoIP Destinations to route to/from.
cdpn:	Called Party Number allows for the matching of an incoming Called Party Number (the called party number refers to the TO number in SIP).
cdpnprefix:	Called Party Number Prefix is used to add a prefix onto the Called Party Number (the called party number also refers to the TO number in SIP).
cdpnstrip:	Called Party Number Strip is used to remove from prefix of a Called Party Number (the called party number refers to the TO number in SIP). This is set as a value which denotes the number of digits to remove.
CDR:	This can be set to type 1-6 or turned off allowing for more detailed call data records to print for calls passing through this routing entry.
Codec:	This defines the preferred codec for calls to use when passing through this routing entry.

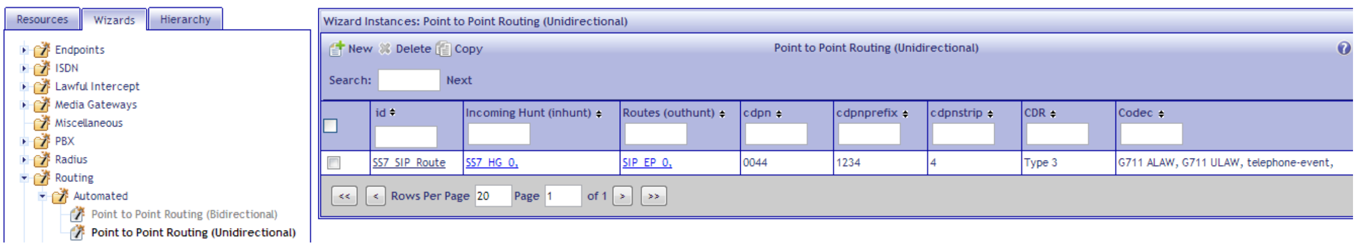


Figure 57 Finished Wizard.

The completed wizard shows the two hunt groups between which a one-way route has been configured.

9.2.3 Custom Routing

There are many more routing scenarios that can be built up from the custom routing templates, the most common method of building up the custom resource is described below. This configuration of the routing allows for a highly flexible routing logic and is much more granular than the above two scenarios.

9.2.3.1 Configuring Routing Criteria

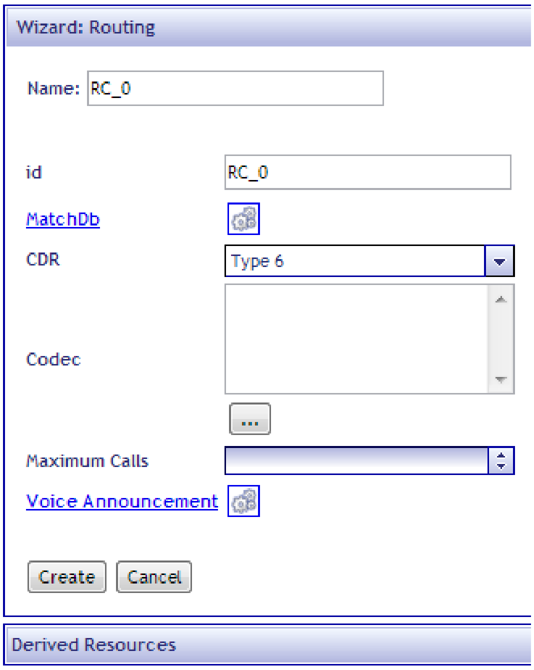


Figure 58 Routing Wizard

The Routing Criteria is configured from the Routing wizard found under Routing -> Custom and acts as a container for Db Parameters allowing for configuration to be applied to all calls passing through Db Parameters contained within the Routing Criteria.

Initially when you create the Routing Criteria there will be no Db Parameters to attach to it, the Routing Criteria can be created without Db Parameters, but no Routing Criteria should exist within the configuration without a Db Parameter attached to it when calls are passing. Calls will hunt into Routing Criteria and treat it as a reject criteria when no Db Parameters are present, the remote congestion cause will be used to reject the call. Unused Routing Criteria should be deleted.

Name:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route.
Id:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route. The ID field allows for a numeric value to be used statically defining the instance of the resource. Recommended for use by advanced users or after discussion with Squire Support.
MatchDb:	To define the Db Parameters included within the Routing Criteria.
CDR:	This can be set to type 1-6 or turned off allowing for more detailed call data records to print for calls passing through this routing entry.
Codec:	This defines the preferred codec for calls to use when passing through this routing entry.
Maximum Calls:	Defines the limit on number of calls allowed to pass through this Routing Criteria at any given time.

Voice Announcement:	Defines a voice announcement resource for playing back voice announcements for use with the SVI MG only.

Wizard without Db Parameters, see next section for creating and adding Db Parameters to the Routing Criteria.

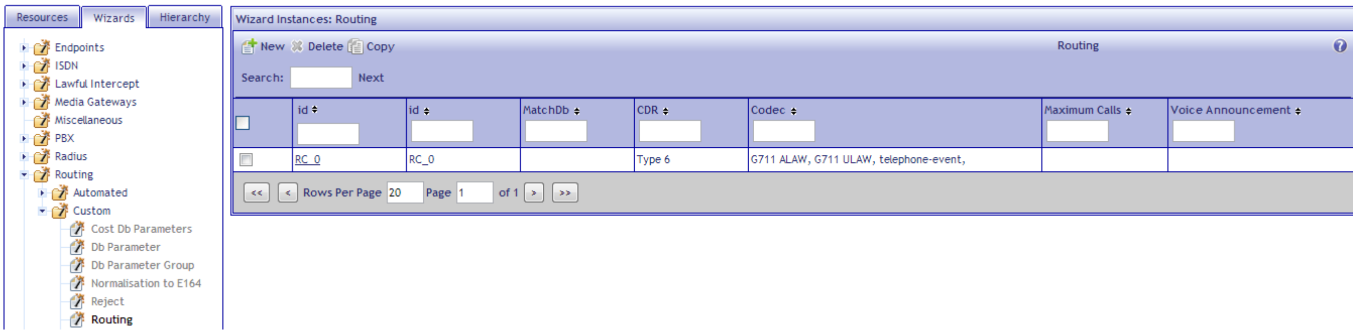


Figure 59 Finished Wizard.

9.2.3.2 Configuring Db Parameters

The Db Parameters are configured from the Db Parameter wizard found under Routing -> Custom and acts as a list entry for a routing decision allowing for matching of incoming call information and manipulating outgoing call information.

The wizard used in the below example is the default wizard supplied showing the most commonly used matching and setting attributes. The template can be modified to show more or less attributes allowing for ease of configuration in the long term. Reconfiguring a template will also allow for the new attribute to be added to previously configured routing entries as a default or manually onto each.

Figure 60 DB Parameter Wizard

Name:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route.
Id:	This defines the name of the routing entry, when using more than one entry it is recommended the naming reflects an easy identifier for the route. The ID field allows for a numeric value to be used statically defining the instance of the resource. Recommended for advanced users or after discussion with Squire Support.
Incoming Hunt:	To define the incoming Hunt Group. Select from a list of hunt groups that have been configured containing Circuits or VoIP Destinations to route from. This is not a mandatory field.
MatchDb:	To define further Db Parameters attached to this one, for further call modification post routing decision. This is recommended for use by advanced users or after discussion with Squire Support.
Routes:	To define the outgoing Hunt Group. Select from a list of hunt groups that have been configured containing circuits or VoIP Destinations to route to. This field is mandatory as the SVI routes calls to Hunt Groups.
cdpn:	Called Party Number allows for the matching of an incoming Called Party Number (the called party number refers to the TO number in SIP).
cdpnprefix:	Called Party Number Prefix is used to add a prefix onto the Called Party Number (the called party number also refers to the TO number in SIP).
cdpnstrip:	Called Party Number Strip is used to remove from prefix of a Called Party Number (the called party number refers to the TO number in SIP). This is set as a value which denotes the number of digits to remove.
Setcdpnnoa:	To set the Nature of Address for cdpn
CGPN:	Allows for the matching of an incoming Calling Party Number

Cgpnstrip:	Is used to remove from several digits from Calling Party Number. This is set as a value which denotes the number of digits to remove from the front of the cgpn.
Cgpnprefix:	Is used to add a prefix onto the Calling Party Number
Setcgpnnoa:	To set the Nature of Address for cgpn
Bearer:	To match the call bearer type (select from dropdown list)
Setbearer:	To set the call bearer type (select from dropdown list)

For a complete list of all attributes and description of what each allows matching or manipulation of see the SVI Routing Resource Glossary HERE. Since this wizard can be configured to tailor the requirements of the interconnect and routing required not all attributes are covered in this document.

After configuring the Db Parameters to define the incoming matching and outgoing routes they are required to be attached to a Routing Criteria for use in routing.

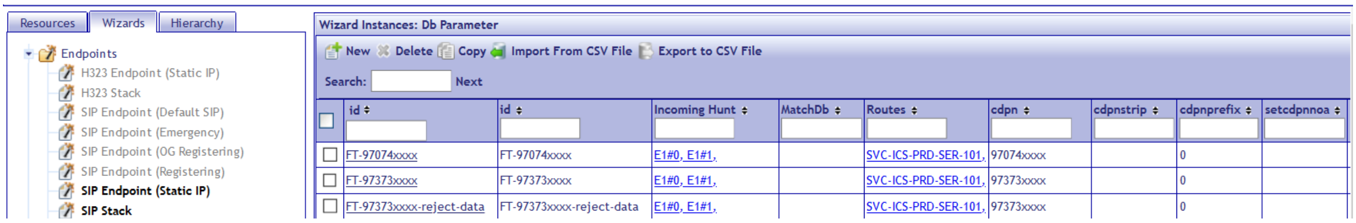


Figure 61 Finished Wizard.

The above example shows the following logic:

Incoming Match SS7_HG_0 (first E1) route to SIP_EP_0 (first SIP Endpoint) if Called Party Number matches 0044 and strip 2 digits from the Called Party Number before sending the call.
Incoming Match SS7_HG_1 (second E1) route to SIP_EP_1 (second SIP Endpoint) if Called Party Number matches 0044 and strip 2 digits from the Called Party Number before sending the call.
Incoming Match SS7_HG_0 (third E1) route to H323_EP_0 (first H323 Endpoint) if Called Party Number matches 0044 and strip 2 digits from the Called Party Number before sending the call.
Incoming Match SS7_HG_0 (fourth E1) route to H323_EP_1 (second H323 Endpoint) if Called Party Number matches 0044 then strip 2 digits and add 556677 to the Called Party Number before sending the call.
This could then be used in combination with the Reject Routing wizard so that any calls arriving from SS7 not matching 0044, 0033 or 0034 will be rejected with defined cause.

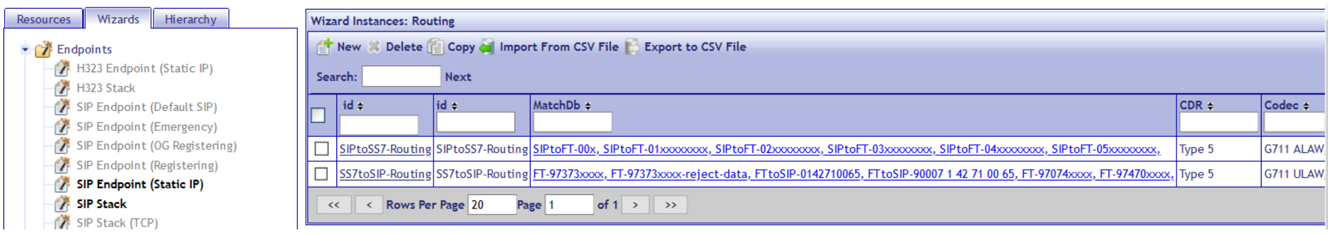


Figure 62 Routing Wizards

The completed Routing wizard shows the Db Parameters attached to the Routing Criteria, once this has been Save and Commit the Db Parameters configured will be used for making routing decisions.

9.2.4 Normalisation

Number normalisation is needed to ensure that all varieties of a number are routed correctly. This occurs by the SVI removing certain numbers from a dialled number, and adding additional numbers, if necessary, to make the number conform to the E.164 number format. To access the number normalisation wizard, use the following navigation: **Configuration -> Wizards -> Routing -> Custom -> Normalisation to E164.** You will then see the following screen.

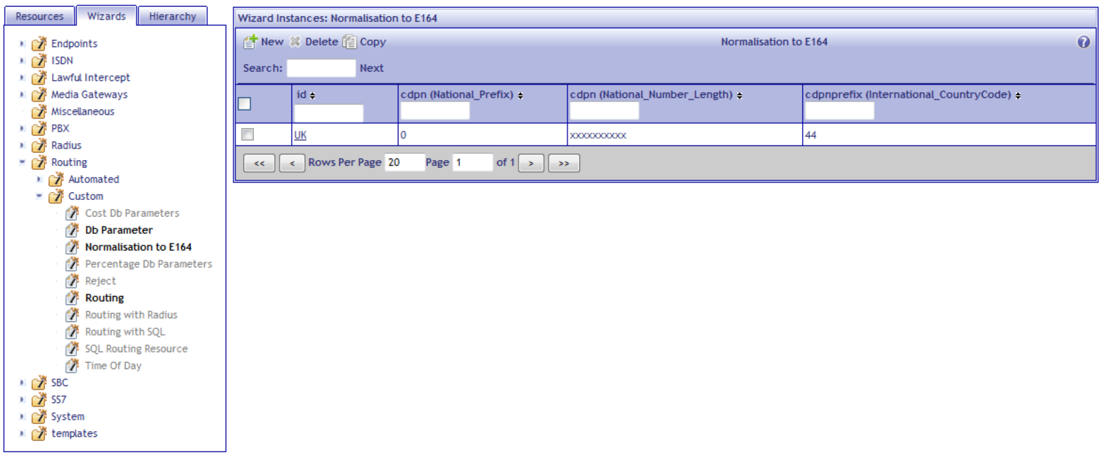


Figure 63 Normalisation Wizard Instances

Click “New” to create Number Normalisation.

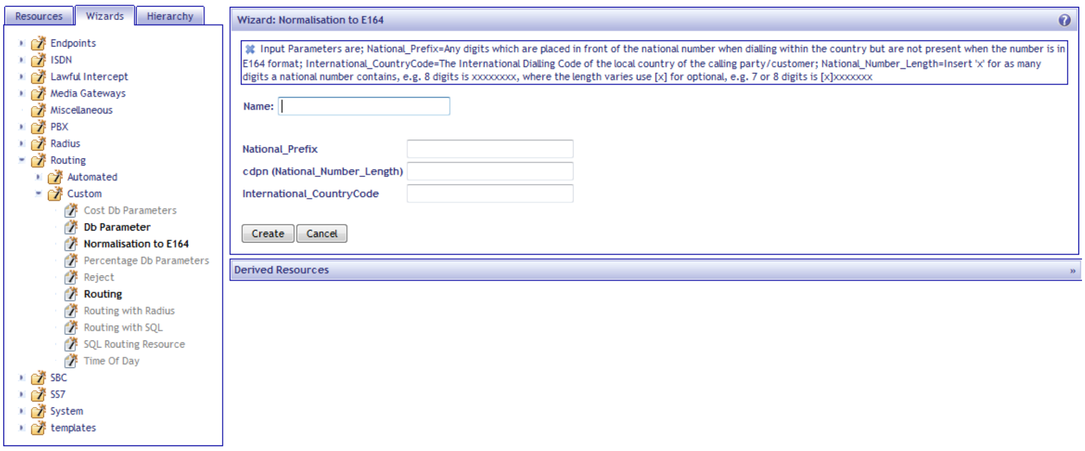


Figure 64 Normalisation Wizard

To assist with the above screen, the following are brief descriptions of each of the fields:

Name:	Give the Number Normalisation a name to identify it. We advise that you call this the region your normalisation relates to, so for example “UK”.
National Prefix:	This is any digits which are placed in front of the national number within the dialling country but are not present when the number is in E164. Under our UK example, this would be 0.
National Number Length:	For this field you need to insert an x for as many digits as a national number contains. Where the length varies (e.g. a national number can have 7 or 8 digits, use [x] for the additional digits as the brackets inform the system that this input is optional. For example, you could enter [x][x]xxxxxx if the number would always have 6 digits but sometimes could feature 7 or 8.

International County Code:	This is the international dialling code of the calling party/customer. The illustration in figure 31 suggests that this is a UK based call.

Once you have entered this information click "create". Having clicked create, you will return to the screen shown in above image. Clicking on our created example (UK) will take you to the screen below and we have also expanded the derived resources.

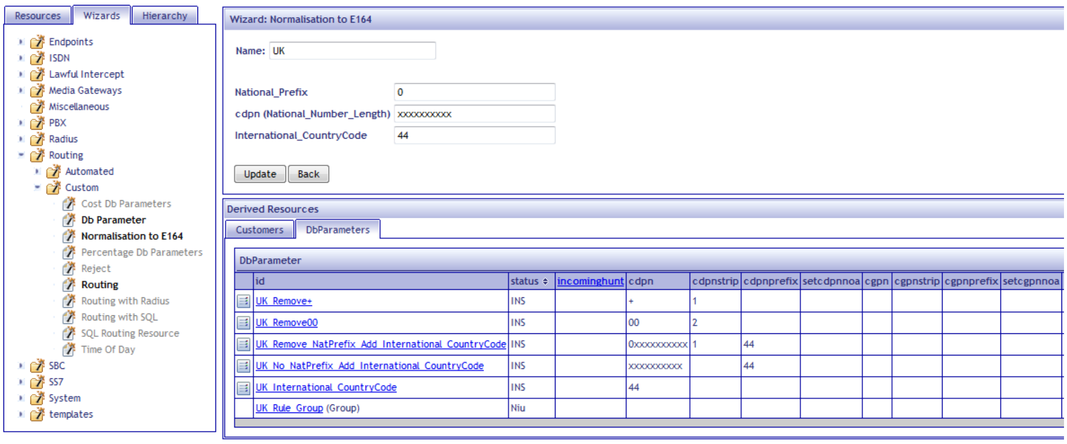


Figure 65 Normalisation Derived Resources

As you can see the number normalisation wizard has created a series of DB parameters. The first three remove the incoming prefixes, the fourth adds in the E164 relevant country code, whilst the fifth accepts that code as it is if it is dialled. These DB parameters can then be assigned to your outgoing circuit hunt groups.

9.2.5 Reject Calls

The routing configuration for this scenario is as follows:

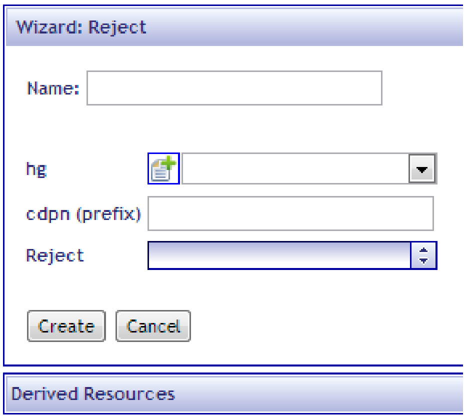


Figure 66 Reject Wizard.

Name:	This defines the name of the routing entry.
Hg:	This defines an incoming hunt group match; this is an optional attribute to add to the wizard and can be used in conjunction with the number match to allow for a more granular match.
Cdpn(prefix):	This number defines the called party number that will be matched for rejecting with the configured cause code.
Reject:	This is an integer value that should be set between 1-127 for SS7 call rejections and set to any valid release code between 400-700 please refer to specifications for all clear codes supported.

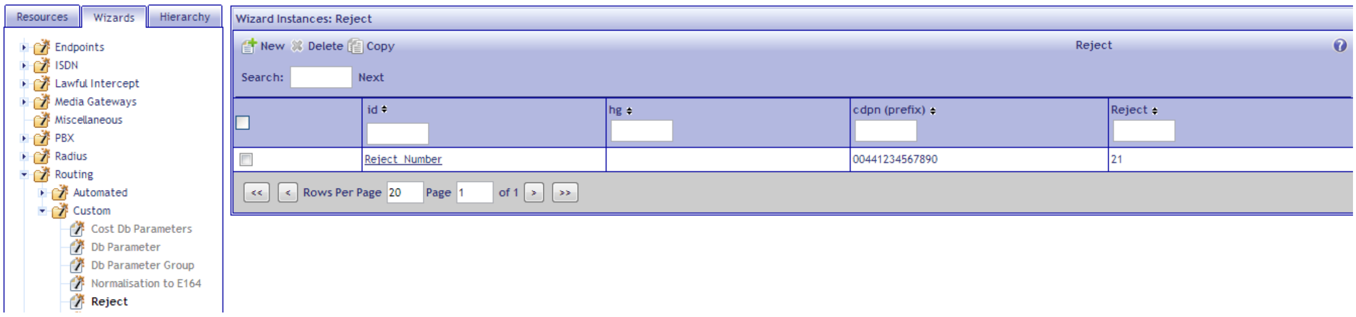


Figure 67 Finished Wizard.

The completed wizard shows the match parameters and the reject cause to use when the call is rejected.

10.0 Support

The SVI-MS user guide provides detailed information on how to support the SVI product range. This section covers setting up the relevant debug information for the SVI MGC.

10.1 Debugging Levels

The following table specifies the debug which should be active to enable Squire Support desk to investigate an issue quickly and effectively within the SVI MGC.

10.1.1 Debugging Levels

Issue	Task	Discriminator	PCAP	Log File
SCTP (M2UA/M3UA) Issue	SCTP MTPL3	DIS_IP DIS_MTP_TX DIS_MTP_RX	Yes	Gateway Wireshark
SLT Issue	SCTP SESSMGR RUDP*	DIS_IP DIS_RUDP_TX DIS_RUDP_RX	Yes	Gateway Wireshark
MTPL2 Signalling Issue	MTPL3	DIS_MTP_TX DIS_MTP_RX	Yes	Gateway Wireshark
SS7 Issue	ISUP CallControl	DIS_CC_TX DIS_CC_RX	Yes	Gateway Wireshark
SIP Issue	SIP CallControl	DIS_IP DIS_CC_TX DIS_CC_RX	Yes	Gateway Wireshark
H323 Issue	H323 CallControl	DIS_H245 DIS_CC_TX DIS_CC_RX	Yes	Gateway Wireshark
Speech Issue	MGCP CallControl	DIS_IP DIS_SWITCH DIS_CC_TX DIS_CC_RX	Yes	Gateway Wireshark
Routing Issue	CallControl Routing**	DIS_CC_TX DIS_CC_RX	NA	Gateway

*For SLT issues it is recommended that Squire Support is involved in investigation due to the nature of issues that can occur in this area.

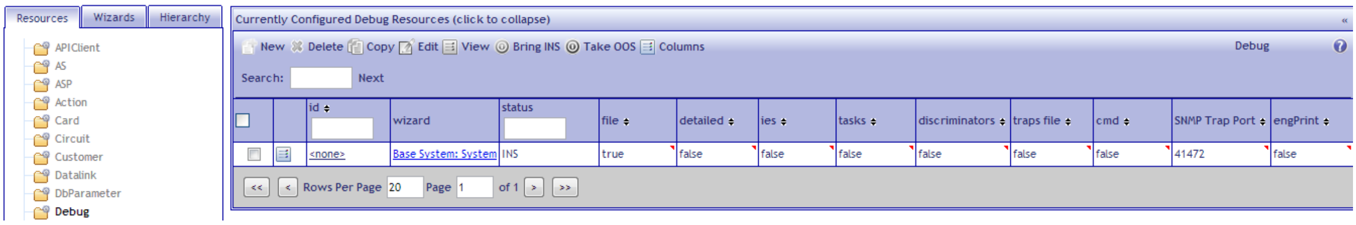


Figure 68 Debug Resources

**Routing debug is enabled through the options attribute on the Debug resource. This is accessible through the Resource tab under the system template.

Within the Debug resource the Options attribute will provide a dialogue box as shown below for selecting Routing debug.

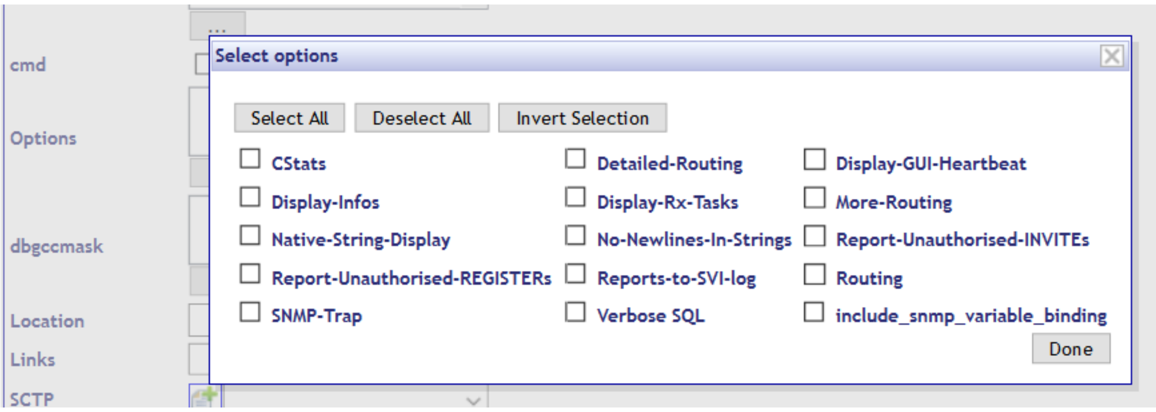


Figure 69 Debug Options

The Gateway log will output verbose information about the routing decisions made, if assistance is required in reading the logs, please contact Squire Support.

10.1.2 IES (Information Elements)

This option is required to be enabled whenever outputting debug for investigating any issues, the information elements options enable the verbose output of the TDM debugging to see a textual decode of the messages within the Gateway log.

10.1.3 Detailed

This option is only required for very specific issue where only the hexadecimal directly is required to be investigated. The detailed option adds a complete hexadecimal dump of each message printed to the Gateway log at the same time.

10.2 Housekeeping

After running the SVI for any period logs and call data records will begin to accumulate. It is up to you the customer to put in place measures to ensure these are archived as required for your company. SVI MG logs and locations are listed below and some recommendations on how to archive these.

10.2.1 Gateway

The Gateway log is located in the /home/squire/ directory. Each day the Gateway log will produce a new log file or if the log reaches a limit of 100mb a new instance of the days log will be spawned with a suffix of the time.

It is recommended that these logs are cleared frequently to stop the hard drive from reaching capacity. Often these are kept for month periods as .tgz files and removed after a month.

10.2.2 CDR

The CDR are located in /home/squire/cdr/ directory. Each day CDR are produced for every call that passes through the SVI. These files have no limit in size before spawning a new file.

It is recommended that the CDR are archived frequently to stop the hard drive from reaching capacity and, so a backup is held of all call data that has passed through the SVI for logging and historic purpose.

The archiving and removal of logs can be achieved in a multitude of ways. We recommend that a bash script is setup in cron to archive the logs mentioned above on a daily basis after completion of the previous day’s logs and another to run monthly to remove or move to a remote system for archiving.

For further information on how to achieve this please contact the support desk for advice for your system.

11.0 Function Specific Features

These are available from the Support Desk, solutions upon request.

Title	Description	Reference
Call Forwards using SQL Routing - Unconditional and Busy	Details how to setup call forwarding when SQL based routing is used. Only unconditional and Busy is supported	00000742
Voip Destination -> protocol	Defines the type of protocol the endpoint uses	00001277