
Customizing the VX Configuration in a Unified Communications Environment

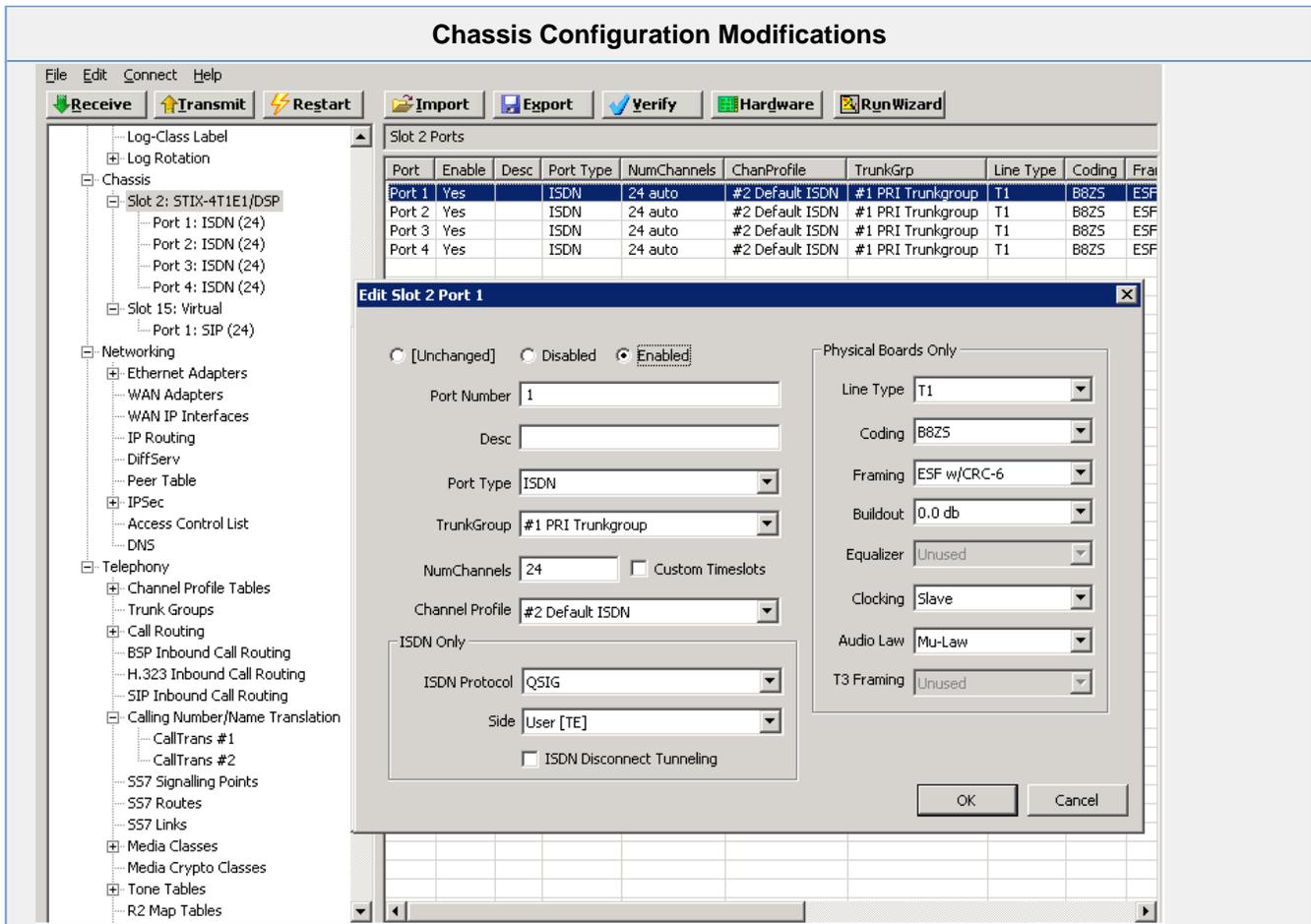
Once the VX Unified Communications Wizard is complete, the VX node is ready to place calls. The following configuration screens depict the configurations which are configured by the wizard to allow bidirectional calls between the PBX or Central Office and Microsoft Office Communications Server 2007. If additional customization of the dialing plan for additional functionality is required, necessary changes need to be configured in the following fields.

- [ISDN / SIP Port Configuration](#)
- [Trunk Groups](#)
- [Call Routing](#)
- [Calling Number/Name Translation](#)
- [Codec Settings](#)
- [OCS Mediation Server Status Using SIP Options Messages](#)

ISDN / SIP Port Configuration

The wizard configures the T1/E1 ports to basic ISDN settings. These can be further refined using the Chassis link in the VXbuilder Directory tree. In this configuration step, the following settings are configured:

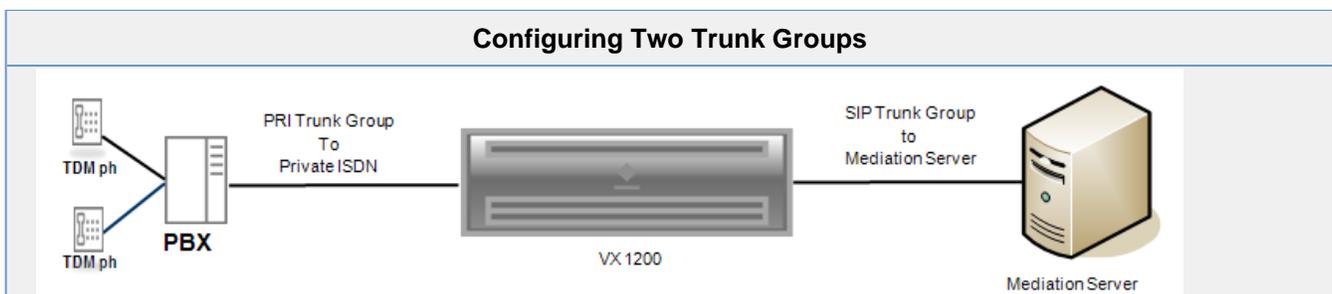
- Port Type
- Assigned Trunk Group
- Number of Channels
- Channel Profile
- Protocol
- Line Type
- Coding
- Framing
- Buildout
- Clocking
- Audio Law



Trunk Groups

Trunk groups configuration allows certain instruction for calls which will be held under one physical or a virtual connection. For instance, while a trunk group is configured and assigned to one particular physical connection to a private ISDN port, another can be configured and applied to another physical connection to public ISDN or to a virtual connection to another IP network.

For a PBX - VX - OCS integration, two trunk groups need to be configured; one for physical ISDN connection and another for virtual IP connection to Mediation Server.



The following two screens show the basic required trunk group configuration.

- **ISDN Trunk Group** configuration in **Telephony > Trunk Groups** in the VXbuilder Directory tree.

Configuring the ISDN Trunk Group

The screenshot shows the VXbuilder interface for configuring a trunk group. The left pane shows the directory tree with 'Trunk Groups' selected under 'Telephony'. The main window displays a table of trunk groups and a detailed configuration dialog for 'TrunkGroup #1'.

Item	ID	Desc	Route	MediaClass	MediaHandling	Passth
1	TrunkGroup #1	PRI Trunkgroup	#1 ISDN to SIP	#1 G.711 mu-law and T.38 Fax	Passthrough	Secure
2	TrunkGroup #2	SIP Trunkgroup	#2 SIP to ISDN	#1 G.711 mu-law and T.38 Fax	Passthrough	Secure

Edit TrunkGroup # 1

General | SIP | H323 | SS7

ID: 1
 Desc: PRI Trunkgroup
 Route Table: #1 ISDN to SIP
 Media Class: #1 G.711 mu-law and T
 Media Handling: Passthrough
 Passthrough Mode: Secure & Unsecure
 Media Crypto Class: [Unchanged]
 Script Name:
 HuntType: Standard
 NCAS Diffserv: Best Effort

InTransTable: None
 OutTransTable: #1 Format calling numb
 Direction: Bidirectional
 Boundary: No
 Registration TTL: 0
 Reroute Code Table: None
 Enable NAT Traversal:
 Direct Connected Carrier:
 Carrier Code:
 Leased Line Emulation (LLEM)
 Peer Node ID: 0:0:0:0
 Peer Trunk-Group ID: 0
 Local Alarm: BlueAlarm

- SIP Trunk Group configuration in **Telephony > Trunk Groups** in VXbuilder Directory tree.

Configuring the SIP Trunk Group

The screenshot shows the VXbuilder interface for configuring a trunk group. The left pane shows the directory tree with 'Trunk Groups' selected under 'Telephony'. The main window displays a table of trunk groups and a detailed configuration dialog for 'TrunkGroup #2'.

Item	ID	Desc	Route	MediaClass	MediaHandling	Passth
1	TrunkGroup #1	PRI Trunkgroup	#1 ISDN to SIP	#1 G.711 mu-law and T.38 Fax	Passthrough	Secure
2	TrunkGroup #2	SIP Trunkgroup	#2 SIP to ISDN	#1 G.711 mu-law and T.38 Fax	Passthrough	Secure

Edit TrunkGroup # 2

General | SIP | H323 | SS7

ID: 2
 Desc: SIP Trunkgroup
 Route Table: #2 SIP to ISDN
 Media Class: #1 G.711 mu-law and T
 Media Handling: Passthrough
 Passthrough Mode: Secure & Unsecure
 Media Crypto Class: [Unchanged]
 Script Name:
 HuntType: Standard
 NCAS Diffserv: Best Effort

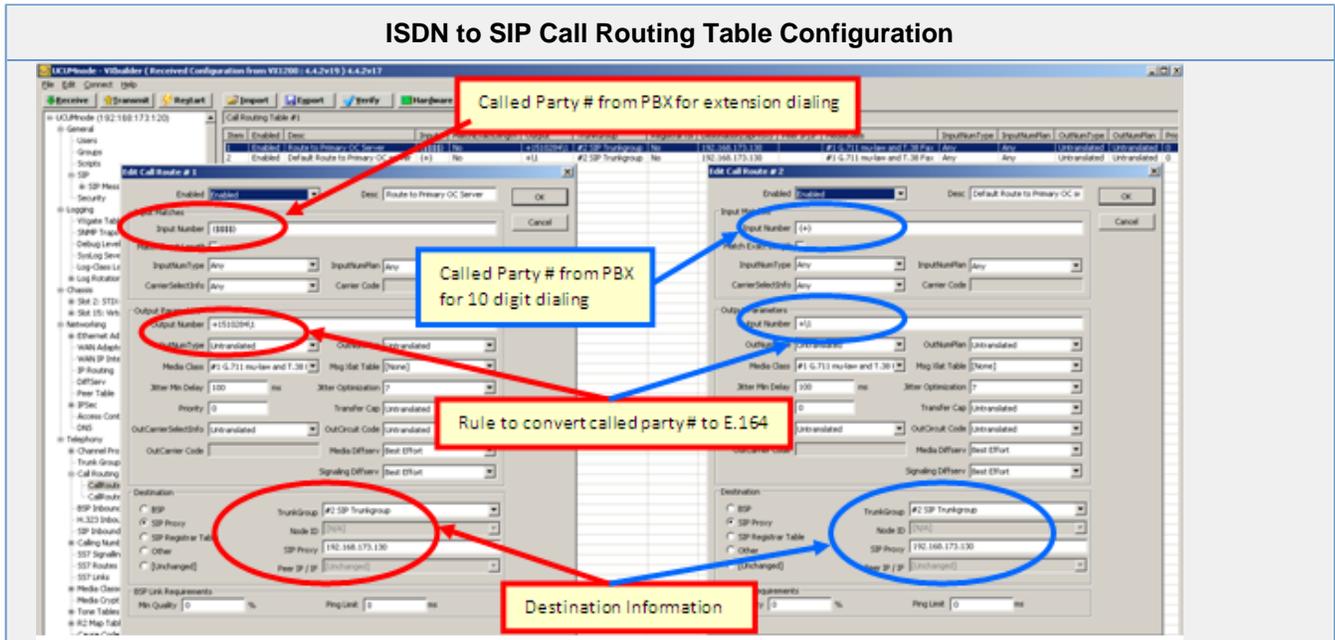
InTransTable: None
 OutTransTable: #2 Format calling numb
 Direction: Bidirectional
 Boundary: No
 Registration TTL: 0
 Reroute Code Table: None
 Enable NAT Traversal:
 Direct Connected Carrier:
 Carrier Code:
 Leased Line Emulation (LLEM)
 Peer Node ID: 0:0:0:0
 Peer Trunk-Group ID: 0
 Local Alarm: BlueAlarm

Call Routing

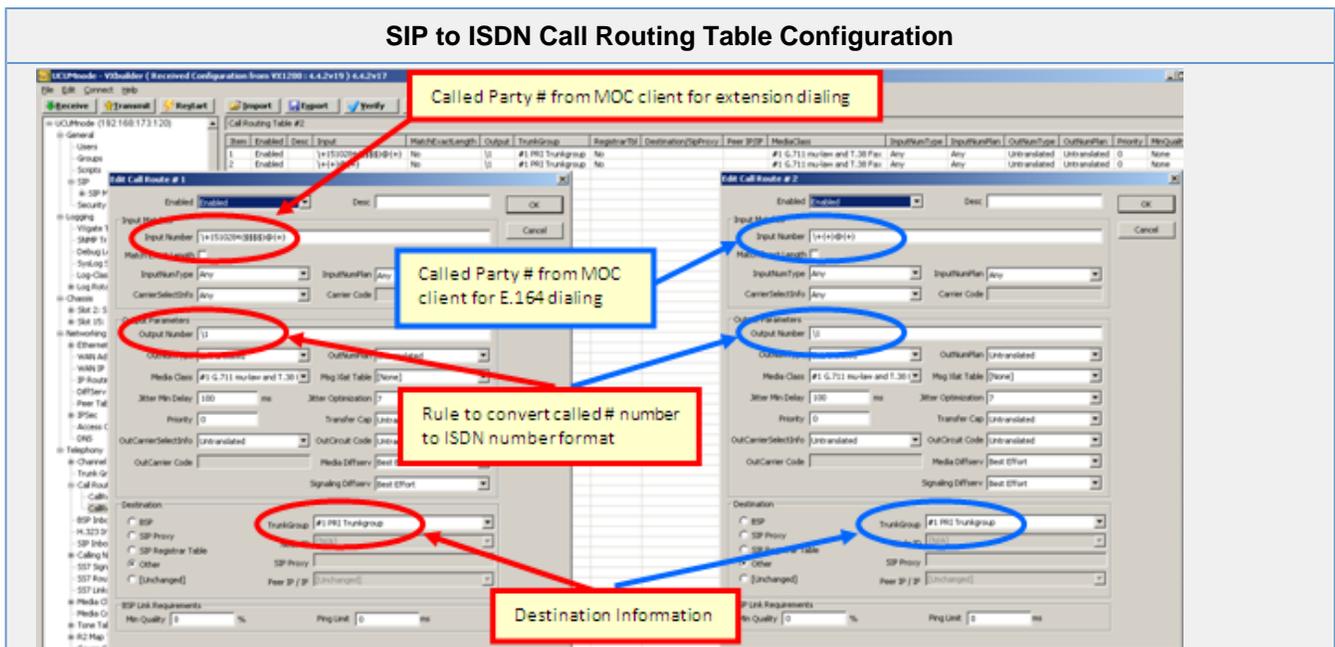
Call Routing table(s) are used for matching and manipulating the called number with VX specific translation rules, as well as defining its next

destination. More detailed information about Call Routing setup and input / output number formatting can be found in the VX User Guide. For the PBX - VX - OCS integration point of view, two call routes are needed for incoming and outgoing calls:

- Two entries in **ISDN-to-SIP Call Routing table** in **Call Routing** in VXbuilder Directory tree.



- Two entries in **SIP-to-ISDN Call Routing table** in **Call Routing** in VXbuilder Directory tree.



Calling Number/Name Translation

As specified in the VX User Guide, the Calling Number and Calling Name translation tables are similar to the call routing tables except that they do not actually alter the route taken by a call. Their sole purpose is to alter the CALLING number as the call passes through VX.

The CALLED number (the identifier of the telephone this call is trying to reach) is used for selection in the call routing tables and is potentially altered by the selected call routing entry (via the 'output' field). The CALLING number (the identifier of the telephone this call originated from) is used for selection in the Calling Number Translation tables and is altered by a matched entry in those tables. This has no effect on routing. Both CALLING and CALLED number translation follow the same regular expressions rules.

Calling number translation tables are invoked for both the inbound trunk-group and the outbound trunk-group. CALLING name translation uses regular expressions. PBX - VX - OCS integration needs the following two Calling Number/Name Translation tables in Calling Number/Name Translation in the VXbuilder Directory tree.

Called Number/Name Translation 1

Item	Enabled	Desc	InNum	InNumType	InNumPlan	OutNum	OutNumPlan	OutNumType
1	Enabled	No Calling number for Anonymous	anonymous@{+}	Any	Any			Untranslated U
2	Enabled	Remove @ from Calling number	{+}@{+}	Any	Any	\1		Untranslated U

Called Number/Name Translation 2

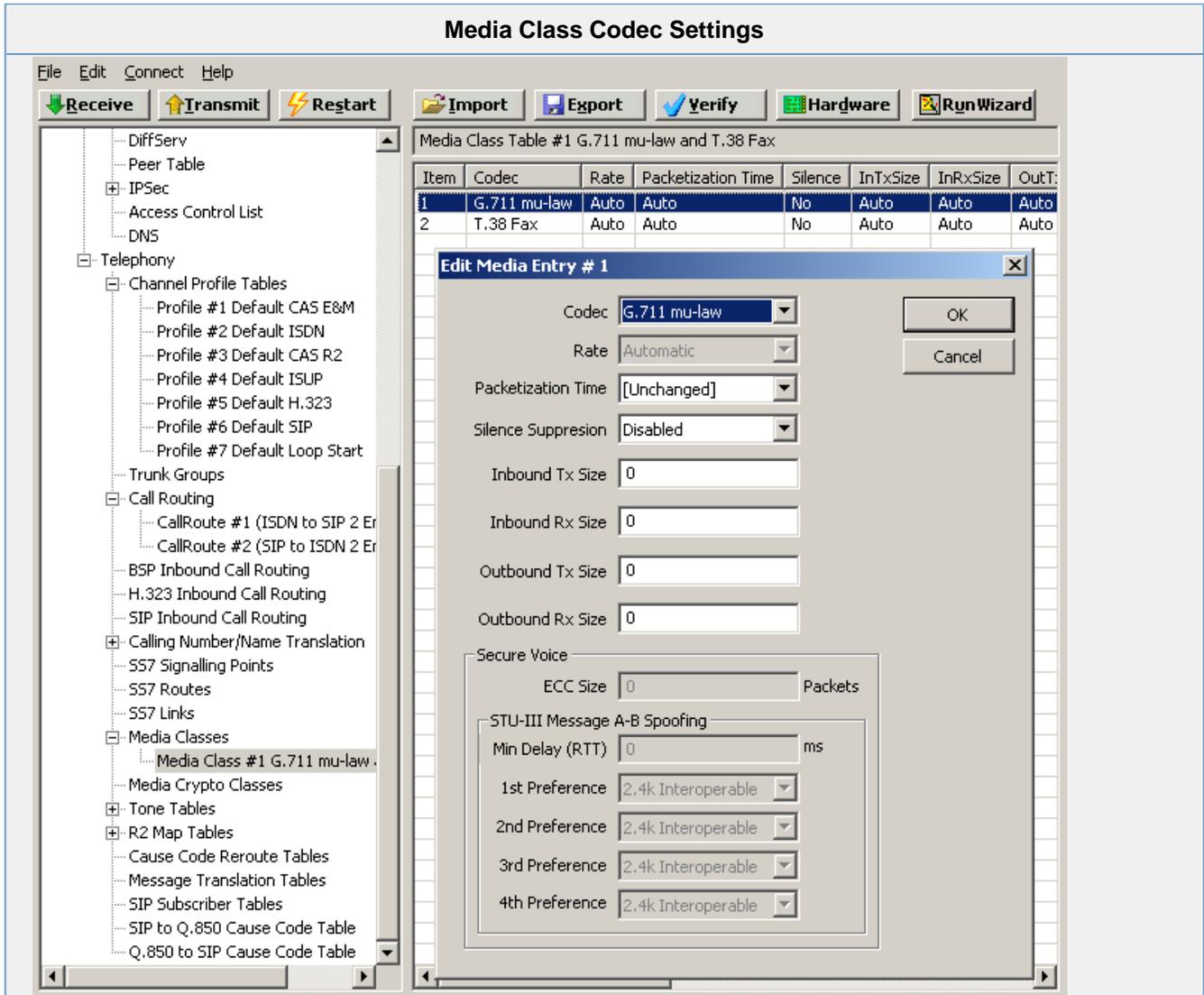
Item	Enabled	Desc	InNum	InNumType	InNumPlan	OutNum	OutNumPlan	OutNumType
1	Enabled	Anonymous in URI for unknown plan and type	{+}	Unknown	Unknown			Unt
2	Enabled	Leave calling number untouched for Private calls	{+}	Any	Private	\1		Unt
3	Enabled	Leave calling number untouched for unknown calls	{+}	Unknown	ISDN	\1		Unt
4	Enabled	Add + to International numbers	{+}	International	ISDN	+1		Unt
5	Enabled	Add country code and + to National numbers	{+}	National	ISDN	+1		Unt
6	Enabled	Add +, Country code and PBX prefix to sub numbers	{+}	Subscriber	ISDN	+1510574	\1	Unt
7	Enabled	Leave anonymous CIN blank		Any	Any			Nat
8	Enabled	Catch all to add + & user=phone to calling numbers	{+}	Any	Any		+1	Nat

Codec Settings

VX Unified Communications Wizard configures the following codec settings in **Media Classes** in VXbuilder Directory tree. More detailed information about Media Class configuration is described in the [Managing Media Classes](#) and [Media Class Table](#) topics.

Media Classes

Item	ID	Desc
1	Media Class #1	G.711 mu-law and T.38 Fax



OCS Mediation Server Status Using SIP Options Messages

VX node needs to know how to reach to its peer(s) in the network. Steps to [setup a Peer Table](#) are defined in the VXbuilder Installation and User Guide.

OCS integration requires a Mediation Server to be configured as a SIP-OPTION Peer on the VX node. This is configured by VX Unified Communications Wizard in **Networking > Peer Table** in the VXbuilder Directory tree.

Mediation Server Peer

File Edit Connect Help

Receive Transmit Restart Import Export Verify Hardware Run Wizard

Peer Table

Item	Enabled	Type	Desc	Node ID	IP/Interface	Ignc
1	Yes	SIP-OPTIONS	Primary Mediation Server	N/A	192.168.173.130	Yes

Edit Peer # 1

Item

Enabled: Yes

Peer Type: SIP-OPTIONS

Description: Primary Mediation Server

Node ID: 0:0:0:0

IP Address/FQDN: 192.168.173.130

Ignore Port Match

VTP Timeout: 10 ms

Boost Multiplier: 0

Failure Holdoff: 0 ms

DiffServ Level: Best Effort

Pings to Reactivate Peer: 2

Pings to Deactivate Peer: 1

Ping Interval:
 Every 32 seconds
 Disable Pinging

OK Cancel