

Polycom 7000 on Mitel 3300 R 9.0.3.15

IMPORTANT NOTE

The Polycom 7000 Conference unit is a SIP device not a Mitel Desktop device therefore it does not understand the VLAN Tagging settings sent from the DHCP Server.

Therefore wherever these devices are installed it is imperative that there is either a route from where they are back to the 3300 or that they are installed onto switch ports which are native to the voice VLAN.

The purpose of this document is to provide an engineer with a basic outline of how to configure a Polycom 7000 conference unit to work as a SIP extension from a Mitel 3300 running release 9.0.3.15 software.

Programming the Mitel 3300

Step 1

Log on to the 3300 and check that the necessary licences are available.

The screenshot shows the Mitel 3300 ICP web interface in a Windows Internet Explorer browser. The page title is "DEFAULT - Mitel Networks: Integrated Communications Platform (ICP)". The top status bar shows "Alarm Status: No Alarm 2010-Jul-16 07:02:04" and buttons for "Print...", "Import...", "Export...", "Data Refresh", "Help", and "Exit". The left sidebar shows a tree view of system configuration options, with "License and Option Selection" highlighted. The main content area is titled "License and Option Selection" and contains the following information:

Online Licensing with the Application Management Center

Application Record ID:

Purchased Options

IP User Licenses:	10
ACD Agent Licenses:	5
IP Device Licenses:	10
Mailbox Licenses:	10
Digital Link Licenses:	2
Compression Licenses:	0
FAX Over IP (T.38) Licenses:	0
SIP Trunk Licenses:	1
Analog Line Licenses:	0
SIP User Licenses:	1
XNET Networking:	No
IP Networking:	Yes
Voice Mail Networking:	No
Advanced Voice Mail:	Yes
Voice Mail Hospitality/PMS:	Yes
Tenanting:	Yes
MLPP:	No
Remote Management:	No
Hardware Identifier:	62041
Password:	*****

Configuration Options

Country:	United Kingdom
Networking Option:	Yes
Mitel/Tapi Computer Integration:	Yes
Extended Agent Skill Group:	No
Maximum Elements per Cluster:	30
Maximum Configurable IP Devices:	700
Extended Hunt Group:	No

MITEL 3300 ICP
About System Administration

In this case there is one SIP user license.

Step 2

Configure the extension.

Browse to System Configuration -> Devices -> IP Telephones -> Multiline IP Sets

And configure a Generic SIP Phone with the required extension number

Range Programming -- Webpage Dialog

Change Range Programming - Multiline IP Set Configuration

This form allows you to change one or more records, starting at the following record:

Device Id	Hot Desk User	Device Type	Auxiliary Module	Number	User PIN	ACD Enabled	Line Type	HTML GUI Application	Interconnect Number
7	No	Generic SIP Phone	None	8063	*****	No	Multicall		1

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Device Id:	-	7	-
Hot Desk User:	Change to	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
Device Type:	Change to	Generic SIP Phone	-
Auxiliary Module:	Change to	None	-
Number:	Change to	8063	
User PIN:	Change to	*****	-
Confirm User PIN:	Change to	*****	-
ACD Enabled:	Change to	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
Line Type:	-	Multicall	-
HTML GUI Application:	Change to		-
Interconnect Number:	Change to	1	
Language:	-	English	-
Max Call History Records:	Change to	0	
MAC Address:	Change to		-
Tenant Number:	Change to	1	
Lock Default Configuration:	Change to	<input checked="" type="radio"/> No <input type="radio"/> Yes	-

Preview Save Cancel

If the device is internal (NOT on the internet and the 3300 is not visible from the internet) then I would suggest making the 'Number' and the 'PIN' the same (8063 in this example)

Step 3

If one does not already exist, create a Class of Service for SIP Devices and enable any required functions (in this case COS 9 has been used).

[illegible]

Step 4

Browse to SIP Device Capabilities and create a capability as follows (in this case number 21 has been used).

-- Webpage Dialog

SIP Device Capabilities Assignment

SIP Device Capabilities Number: 21

Disable Reliable Provisional Responses: ☒ No ☐ Yes

Force sending SDP in initial Invite message: ☒ No ☐ Yes

Prevent the Use of IP Address 0.0.0.0 in SDP Messages: ☐ No ☒ Yes

Replace System based with Device based In-Call Features: ☐ No ☒ Yes

Suppress Use of SDP Inactive Media Streams: ☐ No ☒ Yes

Renegotiate SDP To Enforce Symmetric Codec: ☒ No ☐ Yes

Repeat SDP Answer If Duplicate Offer Is Received: ☒ No ☐ Yes

Allow Device To Use Multiple Active M-Lines: ☒ No ☐ Yes

Comment: Polycom7000

Session Timer: 90

Save Cancel

Step 5

Browse to Station Service Assignment and select your new extension (8063 in this example).

The screenshot shows the Mitel 3300 ICP System Message window. The left sidebar displays a tree view of system configuration options, with 'Station Service Assignment' selected under the 'Devices' folder. The main window displays a table titled 'Station Service Assignment' with the following columns: Number, Intercept Number, Class of Service - Day, Class of Service - Night1, Class of Service - Night2, Class of Restriction - Day, Class of Restriction - Night1, Class of Restriction - Night2, Default Acct. Code, Zone Assignment Method, Zone ID, and SIP Device Capabilities. The table lists various extensions, with extension 8063 highlighted in blue. The 'Go to' field at the top right is set to 'Number' with a value of '8063'.

Number	Intercept Number	Class of Service - Day	Class of Service - Night1	Class of Service - Night2	Class of Restriction - Day	Class of Restriction - Night1	Class of Restriction - Night2	Default Acct. Code	Zone Assignment Method	Zone ID	SIP Device Capabilities
1195	1	36	36	36	95	95	95		Default	1	1
1196	1	36	36	36	95	95	95		Default	1	1
1197	1	36	36	36	95	95	95		Default	1	1
1198	1	36	36	36	95	95	95		Default	1	1
1199	1	36	36	36	95	95	95		Default	1	1
200	1	5	5	5	6	6	1		Default	1	1
201	1	5	5	5	6	6	1		Default	1	1
203	1	5	5	5	6	6	1		Default	1	1
204	1	5	5	5	6	6	1		Default	1	1
205	1	5	5	5	6	6	1		Default	1	1
301	1	10	10	10	6	6	1		Default	1	1
500	1	1	1	1	1	1	1		Default	1	1
8063	1	9	9	9	6	6	6		Default	1	21

Assign your Class of Service (9)

An appropriate Class of Restriction (6)

And your SIP Device Capabilities Number (21)

Step 6

Browse to System Configuration -> Devices -> IP Telephones -> Multiline IP Sets

And select the Multiline Set Key Assignment Form.

Locate your new extension and create a Multiline Key with your own extension number (8063) and set its Ring Type to Ring.

The screenshot displays the Mitel 3300 ICP System Configuration web interface. The left sidebar shows a tree view with 'Multiline IP Sets' selected. The main content area is titled 'Multiline Set Key Assignment Search:'. Below this, there is a search bar with 'Directory Number' selected and '8063' entered. A 'Search' button is next to it. Below the search bar, there is a 'Change' button. The main table shows the results of the search, with columns for 'Directory Number', 'Ring Type', 'Prime Line Type', and 'Name'. The table contains one row with '8063', 'Ring', and 'Multicall'. Below the table, there are navigation buttons: 'Previous', 'Page 1 of 19', 'Next', 'Go to:', 'value:', 'Go', 'Copy Keys', 'Change Member', 'Change Page Members', 'Change All Members', and 'Clear Member'. At the bottom, there is a 'Programmable Keys' table with columns for 'Button Number', 'Label', 'Line Type', 'URL', 'Button Directory Number', 'Ring Type', 'MiXML Application Feature', and 'Phone Application Feature'. The table contains six rows, with the first row having '2', '8063', 'Multicall', '8063', and 'Ring'.

Directory Number	Ring Type	Prime Line Type	Name
8063	Ring	Multicall	

Button Number	Label	Line Type	URL	Button Directory Number	Ring Type	MiXML Application Feature	Phone Application Feature
2	8063	Multicall		8063	Ring	Not Assigned	
3		Not Assigned				Not Assigned	
4		Not Assigned				Not Assigned	
5		Not Assigned				Not Assigned	
6		Not Assigned				Not Assigned	

The Mitel programming is now complete.

Programming the Polycom 7000 Conference Unit

Step 1

Attach the Polycom 7000 to a powered ethernet port and wait for it to boot up.

THIS IS A VERY LONG TIME

Once the phone has booted up you should have a display with the time flashing

Press the 'Menu' key (Top, Left hand side, Group of three)

Navigate to;

Menu -> Status -> Network -> TCP-IP Parameters

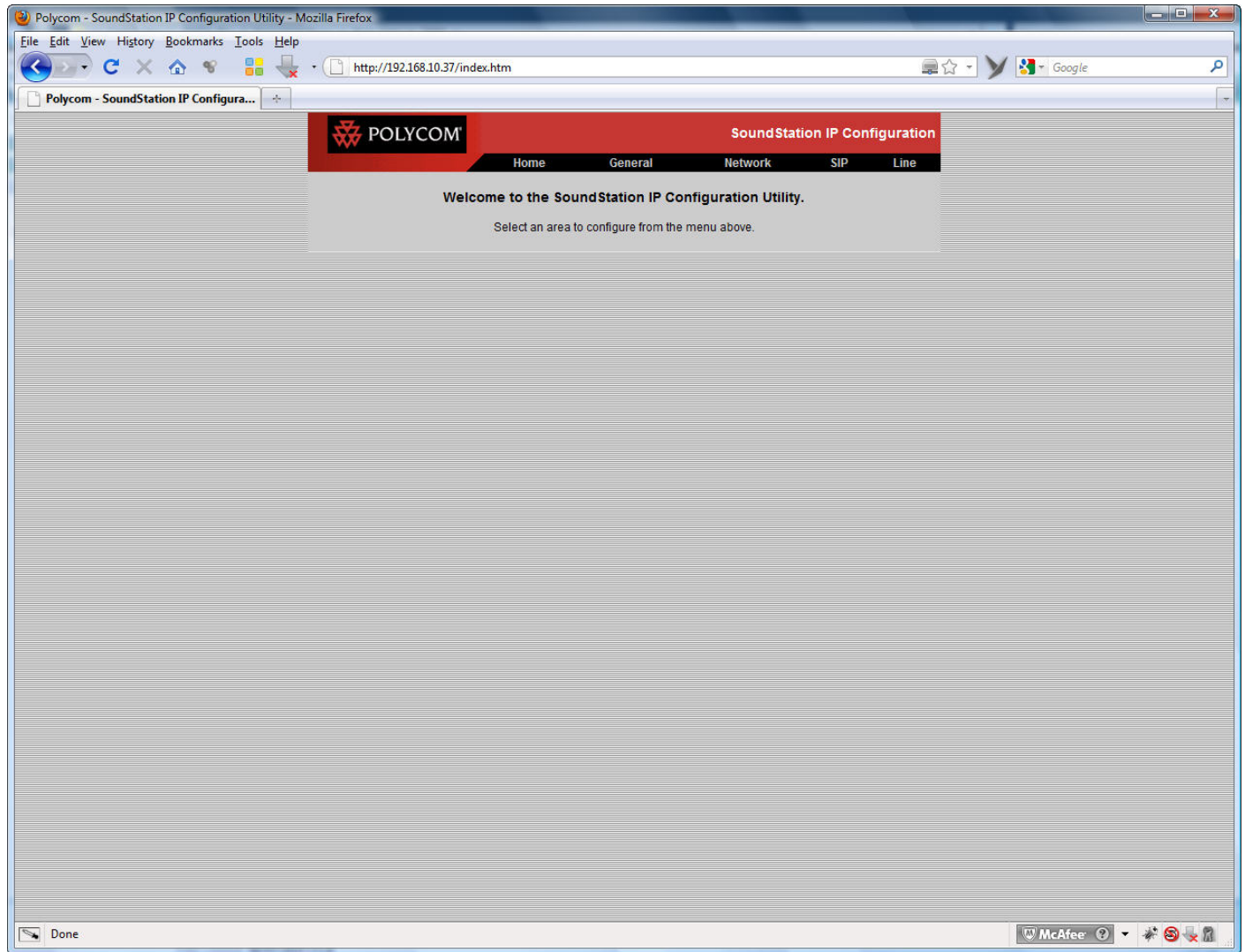
And note the IP Address of the unit (192.168.10.37 in this example)

Step 2

Browse to the IP address in 1

User Name = Polycom

Password = 456



Step 3

Navigate to General -> Time

Enter one of the following NTP Server addresses

0.pool.ntp.org
1.pool.ntp.org
2.pool.ntp.org
3.pool.ntp.org

Polycom - SoundStation IP Configuration Utility - Mozilla Firefox

File Edit View History Bookmarks Tools Help

http://192.168.10.37/timeConf.htm

Polycom - SoundStation IP Configuration Utility

POLYCOM SoundStation IP Configuration

Home General Network SIP Line

General Configuration Parameters:

User Preferences Time Audio Processing Video Processing Background

Sampled Audio Microbrowser Logging Applications

Time

Synchronization

SNTP Server 0.pool.ntp.org

GMT Offset

SNTP Resync Period 86400

Daylight Savings

Daylight Savings ☒ Enabled ☐ Disabled

Fixed Day ☐ Enabled ☒ Disabled

Start Month March

Start Date 08

Start Time 02:00

Start Day of Week Sunday

Start Day Last In Month ☐ Yes ☒ No

Stop Month November

Stop Date 01

Stop Time 02:00

Stop Day of Week Sunday

Stop Day Last In Month ☐ Yes ☒ No

top Submit

Done McAfee

Click on 'Submit' and wait while the phone reboots. This can be a VERY LONG TIME sometimes the browser will time out before the phone has rebooted.

Step 4

Navigate to General -> Audio Processing

Set the codec order as follows;

POLYCOM SoundStation IP Configuration

Home General Network SIP Line

General Configuration Parameters:

User Preferences Time Audio Processing Video Processing Background
Sampled Audio Microbrowser Logging Applications

Audio Processing

Codec Preferences

G.711Mu	1
G.711A	2
G.722	Not Used
G.722AB	3
G.722.1 16kbps	Not Used
G.722.1 24kbps	Not Used
G.722.1 32kbps	Not Used
G.722.1C 24kbps	Not Used
G.722.1C 32kbps	Not Used
G.722.1C 48kbps	Not Used
Siren14 24kbps	Not Used
Siren14 32kbps	Not Used
Siren14 48kbps	Not Used
Siren22 32kbps	Not Used
Siren22 48kbps	Not Used
Siren22 64kbps	Not Used

G.711Mu Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	500
Jitter Buffer Maximum	160

G.711A Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	500
Jitter Buffer Maximum	160

G.722 Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	1500
Jitter Buffer Maximum	200

G.722AB Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	500
Jitter Buffer Maximum	160

G.722.1C Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	1500
Jitter Buffer Maximum	200

G.722.1C Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	1500
Jitter Buffer Maximum	200

Siren14 Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	150
Jitter Buffer Maximum	200

Siren22 Codec Profile

Payload Size	20
Jitter Buffer Minimum	40
Jitter Buffer Shrink	1500
Jitter Buffer Maximum	200

top Submit

All other settings are at default – Click on ‘Submit’ and wait

Step 5

Navigate to SIP -> Servers

Make the following changes to 'Outbound Proxy' and 'Server 1'

The screenshot displays the Polycom SoundStation IP Configuration web interface. The top navigation bar includes the Polycom logo and tabs for Home, General, Network, SIP, and Line. The main content area is titled 'SIP Configuration Parameters:' and is divided into two sections: 'Servers' and 'Local Settings'.

Servers Section:

- Outbound Proxy:**
 - Address: 192.168.10.16
 - Port: 5060
 - Transport: DNSnaptr (dropdown)
- Server 1:**
 - Address: 192.168.10.16
 - Port: 5060
 - Transport: DNSnaptr (dropdown)
 - Expires: 3600
 - Register: 1
 - Retry Time Out: 0
 - Retry Max Count: 0
 - Line Seize Time Out: 30
- Server 2:**
 - Address: (empty)
 - Port: (empty)
 - Transport: DNSnaptr (dropdown)
 - Expires: (empty)
 - Register: (empty)
 - Retry Time Out: (empty)
 - Retry Max Count: (empty)
 - Line Seize Time Out: (empty)

At the bottom of the Servers section are 'top' and 'Submit' buttons.

Local Settings Section:

- Local SIP Port: 5060
- Calls Per Line Key: (empty)
- New SDP Type: ☐ Enabled ☒ Disabled
- LCS Support: ☐ Enabled ☒ Disabled
- Non Standard Line Seize: ☒ Enabled ☐ Disabled
- Digitmap: 9999 | 9112 | 91xx | 900xxx, T | 9 [0] [2-9]xxxxxxxxxx | 9 [2-9]x (dropdown)
- Digitmap Timeout: 3|3|3|3|3
- Remove End-Of-Dial Marker: ☒ Enabled ☐ Disabled
- Digitmap Impossible Match: 0

At the bottom of the Local Settings section are 'top' and 'Submit' buttons.

Click on 'Submit' and wait.

Step 6

Navigate to SIP -> Servers

Make the 'Local SIP Port' 5060 in 'Local Settings'

POLYCOM		SoundStation IP Configuration							
Home		General		Network		SIP		Line	
SIP Configuration Parameters:									
Servers					Local Settings				
Servers									
Outbound Proxy									
Address		192.168.10.16							
Port		5060							
Transport		DNSnaptr							
Server 1									
Address		192.168.10.16							
Port		5060							
Transport		DNSnaptr							
Expires		3600							
Register		1							
Retry Time Out		0							
Retry Max Count		0							
Line Seize Time Out		30							
Server 2									
Address									
Port									
Transport		DNSnaptr							
Expires									
Register									
Retry Time Out									
Retry Max Count									
Line Seize Time Out									
top					Submit				
Local Settings									
Local SIP Port		5060							
Calls Per Line Key									
New SDP Type		<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled							
LCS Support		<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled							
Non Standard Line Seize		<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled							
Digitmap		9999 9112 91xx 900xxx. T 9[0] [2-9]xxxxxxxxxx 9[2-9]x							
Digitmap Timeout		3 3 3 3 3							
Remove End-Of-Dial Marker		<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled							
Digitmap Impossible Match		0							
top					Submit				

Click on 'Submit' and wait.

Step 7

Navigate to Line

Make the following changes to 'Line 1', 'Server 1' and 'Message Centre'

Display Name	=	User Name or Similar
Address	=	Extension Number (8063 in this example)
Address User ID	=	Extension Number (8063 in this example)
Auth Password	=	Extension Number (8063 in this example)
Label	=	Extension Number (8063 in this example)

Not very security conscious but as these phones are for internal use (i.e. NOT across the internet and the 3300 is NOT visible from the internet) these settings should be fine.

Server 1

Address	=	Mitel RTC (192.168.10.16 in this example)
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Message Centre

Subscriber	=	Voicemail Hunt Group Number (6245 in this example)
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POLYCOM		SoundStation IP Configuration				
		Home	General	Network	SIP	Line
Line Parameters:						
Line 1						
Line 1						
Identification						
Display Name	Lewis Silkin					
Address	8063					
Auth User ID	8063					
Auth Password						
Label	8063					
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared					
Third Party Name						
Num Line Keys						
Calls Per Line						
Server 1						
Address	192.168.10.16					
Port	5060					
Transport	DNSnaptr ▼					
Expires	3600					
Register						
Retry Time Out						
Retry Max Count						
Line Seize Time Out						
Server 2						
Address						
Port						
Transport	DNSnaptr ▼					
Expires						
Register						
Retry Time Out						
Retry Max Count						
Line Seize Time Out						
Call Diversion						
Disabled On Shared	<input checked="" type="radio"/> Yes <input type="radio"/> No					
Diversion Contact						
On Specific Caller	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled					
Forward All	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled					
On Busy	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled					
Busy Contact						
On No Answer	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled					
No Answer Timeout	60					
No Answer Contact						
On Do-Not-Disturb	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled					
Do-Not-Disturb Contact						
Message Center						
Subscriber	6245					
Callback Mode	Registration ▼					
Callback Contact						
top	Submit					

Click on 'Submit' and wait.

Step 8

Navigate to SIP -> Servers

We now need to replace the Digit Map in 'Local Settings'

POLYCOM		SoundStation IP Configuration		
Home	General	Network	SIP	Line
SIP Configuration Parameters:				
Servers		Local Settings		

Servers	
Outbound Proxy	
Address	192.168.10.16
Port	5060
Transport	DNSnaptr
Server 1	
Address	192.168.10.16
Port	5060
Transport	DNSnaptr
Expires	3600
Register	1
Retry Time Out	0
Retry Max Count	0
Line Seize Time Out	30
Server 2	
Address	
Port	
Transport	DNSnaptr
Expires	
Register	
Retry Time Out	
Retry Max Count	
Line Seize Time Out	
top	Submit

Local Settings	
Local SIP Port	5060
Calls Per Line Key	
New SDP Type	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
LCS Support	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
Non Standard Line Seize	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap	9999 9112 91xx 900xxx. T 9[0] [2-9]xxxxxxxxx 9[2-9]x
Digitmap Timeout	3 3 3 3 3
Remove End-Of-Dial Marker	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Digitmap Impossible Match	0
top	Submit

Place your cursor in the 'Digit Map' Field and

Press [Ctrl] + a

Press [Delete]

The entry should now be clear

If you are in a SIX DIGIT local number area copy and paste the following line into the 'Digit Map' field.

9999|9112|91xx|900xxx.T|9[0][2-9]xxxxxxxx|9[2-9]xxxxx|[1-8]xxx|0T

The dial plan gives;

999	(Emergency Calls)
9112	(Emergency Calls)
91xx	(91 + 2 more digits - level 1 calls)
900xx.T	(International Calling - after timeout)
9[0][2-9]xxxxxxxx	(9 + 0 + [any one digit between 2 and 9] followed by 9 other digits - National Calls)
9[2-9]xxxxx	(9 + [any one digit between 2 and 9] followed by 5 other digits - 6 Digit Local Calls)
[1-8]xxx	([any one digit between 1 and 8] followed by three other digits - internal calls)
0T	(0 + Timeout - Internal Operator)

If you are in a SEVEN DIGIT local number area copy and paste the following line into the 'Digit Map' field.

9999|9112|91xx|900xxx.T|9[0][2-9]xxxxxxxx|9[2-9]xxxxx|[1-8]xxx|0T