MITEL – SIP CoE

Technical

Configuration Notes

Configure MCD 4.1 for use with Engin SIP Trunking Service Provider.

SIP CoE 10-4940-00129

NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

TRADEMARKS

Mitel is a trademark of Mitel Networks Corporation.

Windows and Microsoft are trademarks of Microsoft Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged.

Mitel Technical Configuration Notes – Configure MCD 4.1 for use with Engin SIP Trunking Service Provider.

July 2010, 10-4940-00129_3

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 MCD to connect to Engin SIP Trunking Service Provider. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	May 13, 2010	Initial Interop with Mitel 3300 4.1 and Engin SIP Trunking Service Provider
2	June, 28, 2010	Documentation Update
3	July 6, 2010	Documentation Update

Interop Status

The Interop of Engin SIP Trunking Service Provider has been given a Certification status. This service provider or trunking device will be included in the SIP CoE Reference Guide. The status of Engin SIP Trunking Service Provider achieved is:

COMPATIBLE	The most common certification which means Engin SIP Trunking Service Provider has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between Engin SIP Trunking Service Provider and the 3300 MCD.

Manufacturer	Variant	Software Version
Mitel	3300 MCD – Mxe Platform	10.1.0.69_1
Mitel	MBG - Teleworker	5.2.14.0
Mitel	Nupoint	12.01.34
Mitel	Mobile Extension	1.7.13.0
Engin Service Provider	Broadworks	

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through Engin SIP Trunking Service Provider, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	⚠
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	X
Packetization	Forcing the 3300 MCD to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	⚠
Personal Ring Groups	Receiving calls through Engin SIP Trunking Service Provider to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Mobile Extension	Receiving a call through Engin SIP Trunking Service Provider to Mobile extensions and TUI interface. Also moving calls to/from Desktop and Twinned devices.	⚠
Teleworker	Making and receiving a call through Engin SIP Trunking Service Provider to and from Teleworker extensions.	☑
Video	Making and receiving a call through Engin SIP Trunking Service Provider with video capable devices.	Not Supported
Fax	T.38 and G711Fax Calls	⚠

☑ - No issues found X - Issues found, cannot recommend to use ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or not supported features when Engin SIP Trunking Service Provider is connected to the Mitel 3300.

Feature	Problem Description
Mobile Extension	You will not get audio when answering the call on your cell phone if Answer Confirmation is not enabled for the user. **Recommendation:** Answer Confirmation must be enabled in the User Settings of the mobile extension server. Please contact Mitel Product support and reference defect MN00338480 for updates on this limitation.
Nupoint Voicemail	Engin is handling SIP Nat Traversal by using symmetrical RTP. The Nupoint Voicemail does not support symmetrical RTP. **Recommendation:** Nupoint will support symmetrical RTP in release 14.1. Contact Mitel for a status update on the support of symmetrical RTP with Nupoint voicemail. Reference defect number MN00310861.
Packetization	The inbound RTP stream from Engin remains at 20ms, but the outbound RTP stream from the 3300 negotiates at the variable rate assigned in the SIP peer profile. **Recommendation:** Use the default packetization rate of 20ms in the SIP Peer Profile of the 3300.
Unsupervised Transfer	No audible ring back during Unsupervised Transfer, A call comes in through the PSTN to Engin and then to an IP phone on the 3300. The IP phone then does an unsupervised transfer back out to Engin and to another PSTN number. The IP phone hears ringback and then hits release. The initial caller doesn't hear ringback (in fact, it hears silence) until the new destination picks up. Audio proceeds properly at that point. **Recommendation:** This is a Mitel Defect found during testing. Please reference the defect number MN00323843 when contacting Mitel product support. This defect is fixed in MCD 4.1 SP1
T.38	During testing we were not able to successfully transmit faxes using T.38. Problem was reported to Engin. **Recommendation:** Use G.711 for transmitting faxes. Contact Engin for updates on supporting T.38. If contacting Mitel product support reference defect MN00338513.
Reverse DNS Lookup	The Mitel 3300 is not able to do a DNS reverse lookup when presented with the IP address of the service provider in the contact header. **Recommendation:** Enter the DID in the SIP Peer Profile Assignment by Incoming DID as shown in figure 8 below. This is a Mitel Defect found during testing. Please reference the defect number MN00332835 when contacting Mitel product support. This defect is fixed in MCD 4.1 SP2

Network Topology

This diagram shows how the testing network is configured for reference.

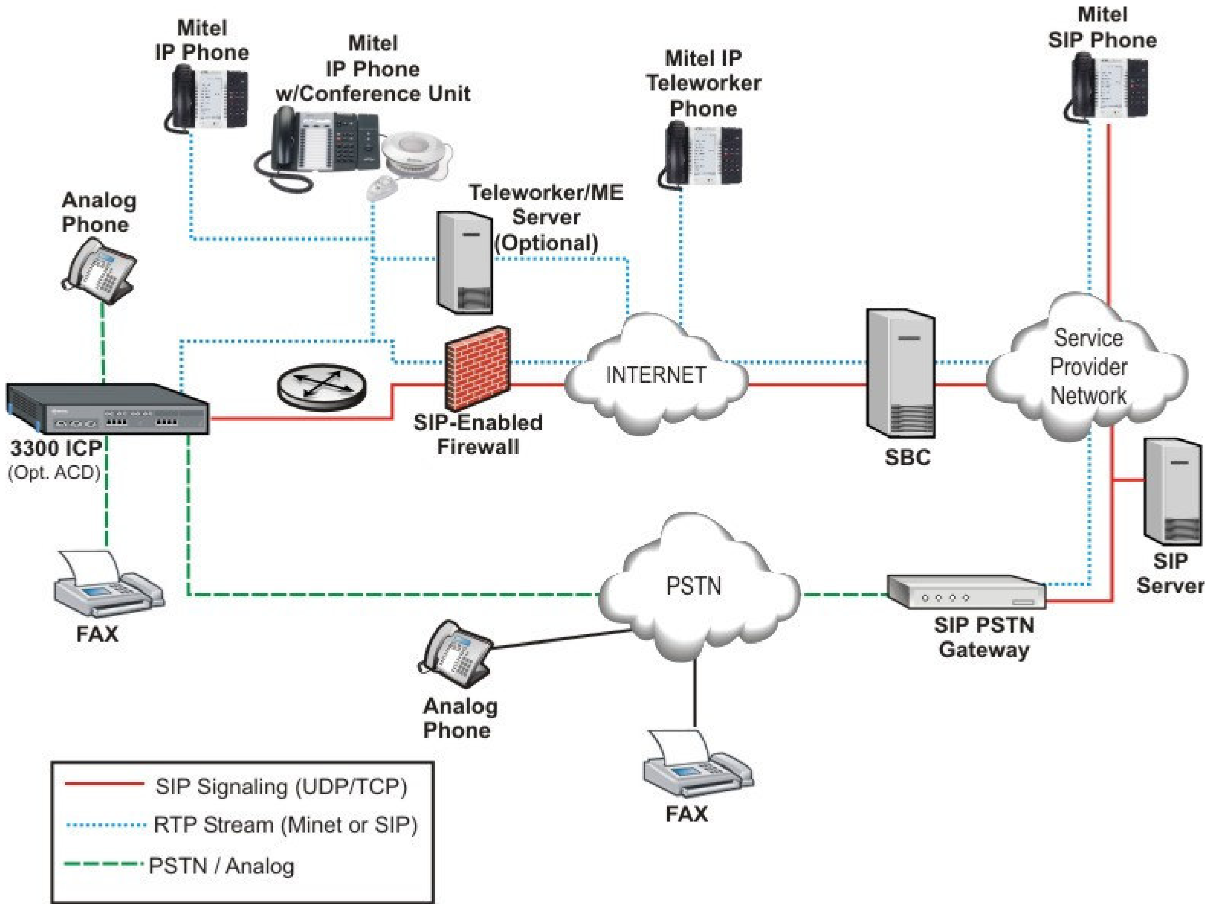


Figure 1 – Network Topology

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how Engin SIP Trunking Service Provider 3300 programming was configured in our test environment.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

3300 MCD Configuration Notes

The following steps show how to program a 3300 MCD to interconnect with Engin SIP Trunking Service Provider.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the 3300 Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

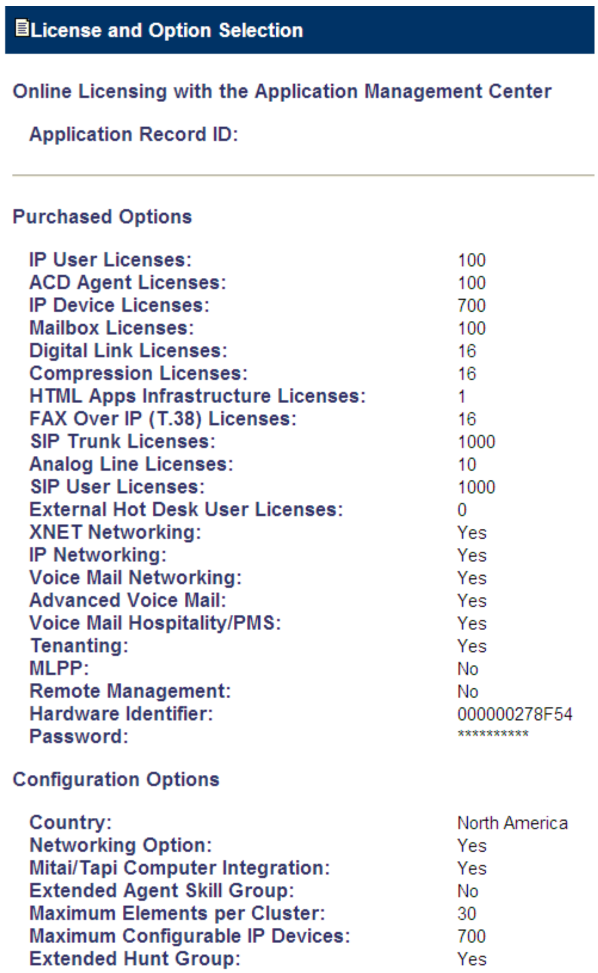
Assumptions for the 3300 MCD Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the 3300 MCD is equipped with enough SIP trunking licenses for the connection to Engin SIP Trunking Service Provider. This can be verified within the License and Option Selection form.

Enter the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the 3300 to be used with all service providers, applications and SIP trunking devices.



License and Option Selection

Online Licensing with the Application Management Center

Application Record ID:

Purchased Options

IP User Licenses:	100
ACD Agent Licenses:	100
IP Device Licenses:	700
Mailbox Licenses:	100
Digital Link Licenses:	16
Compression Licenses:	16
HTML Apps Infrastructure Licenses:	1
FAX Over IP (T.38) Licenses:	16
SIP Trunk Licenses:	1000
Analog Line Licenses:	10
SIP User Licenses:	1000
External Hot Desk User Licenses:	0
XNET Networking:	Yes
IP Networking:	Yes
Voice Mail Networking:	Yes
Advanced Voice Mail:	Yes
Voice Mail Hospitality/PMS:	Yes
Tenanting:	Yes
MLPP:	No
Remote Management:	No
Hardware Identifier:	000000278F54
Password:	**********

Configuration Options

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

Figure 2 – License and Option Selection

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment form for SIP trunks.

Many different options may be required for your site deployment, but ensure that “Public Network Access via DPNSS” Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the 3300.

- Public Network Access via DPNSS set to **Yes**
- Campon Tone Security/FAX Machine set to **Yes**
- Busy Override Security set to **Yes**

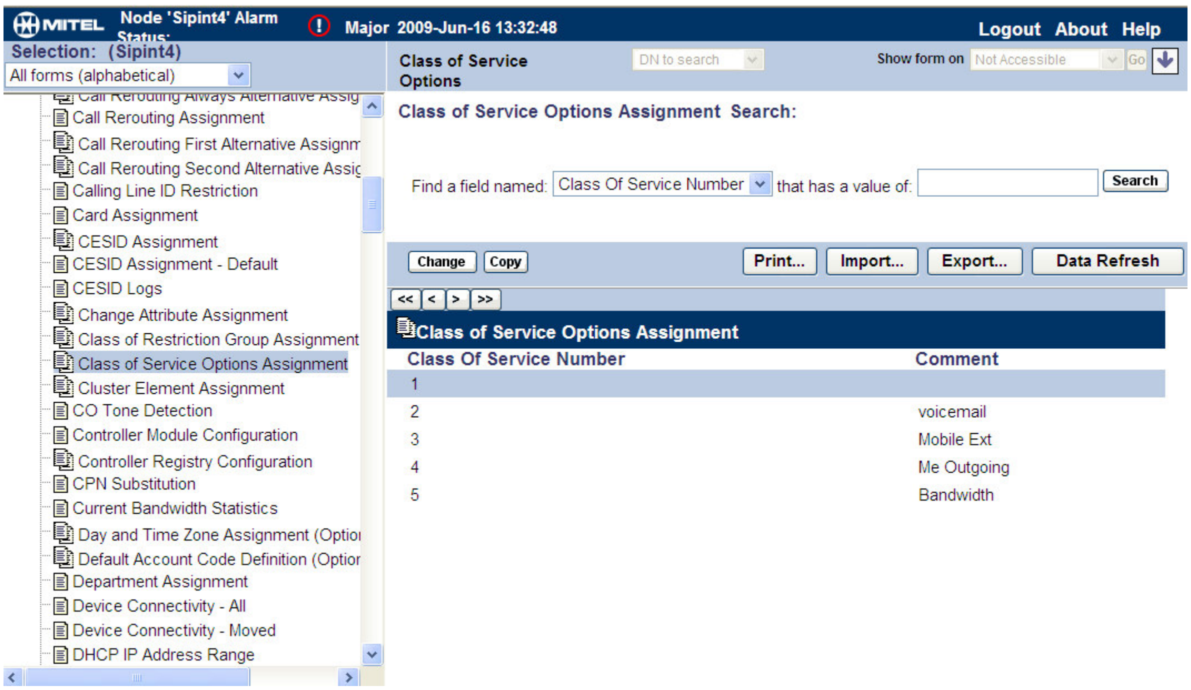


Figure 3 – Class of Service

Network Element Assignment

Create a network element for Engin SIP Trunking Service Provider. In this example, the softswitch is reachable by an FQDN and is defined as “Engin” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service provider.**

If your service provider trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider. Set the transport to UDP and port to 5060.

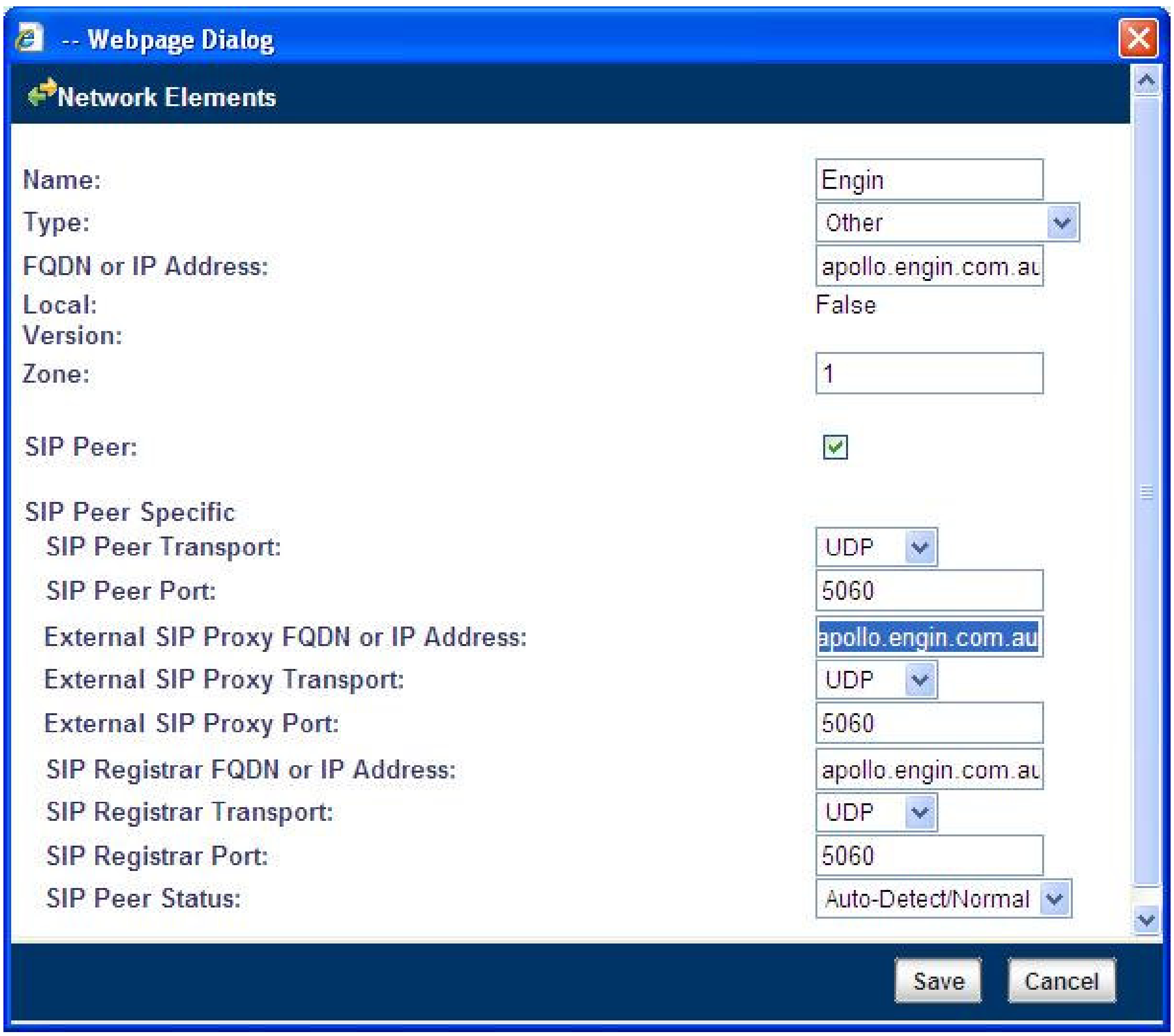


Figure 4 – Network Element Assignment

Network Element Assignment (Proxy)

A Proxy entry is needed to be configured to route SIP data to Engin service provider. Program the Proxy as a network element as shown in the diagram below. Then reference the proxy in the SIP Peer profile assignment (later in this document). **Please Note:** The FQDN apollo.engin.com.au resolves to 202.147.130.12. The IP address is used for the outbound proxy and not FQDN as to not cause conflicts for systems using System Data Synchronization (SDS).

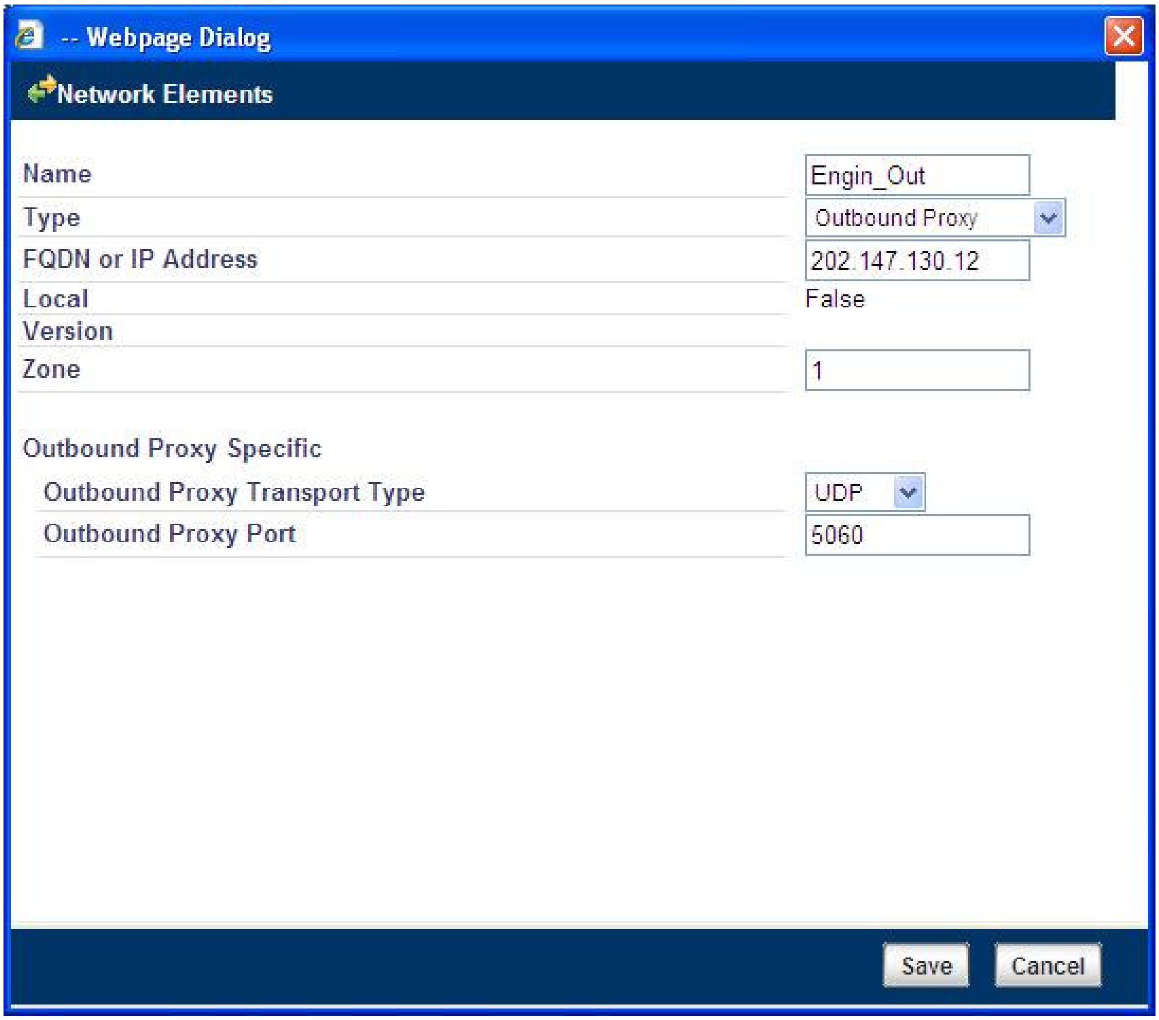


Figure 5 – Network Element Assignment (Proxy)

Trunk Service Assignment

This is configured in the Trunk Service Assignment form. In this example the Trunk Service Assignment is defined for Trunk Service Number 46 which will be used to direct incoming calls to an answer point in the 3300.

Program the Non-dial In or Dial In Trunks (DID) according to the site requirements and what type of service was ordered from your service provider.

The example below shows configuration for incoming DID calls. The 3300 will absorb the first 5 digits of the DID number from Engin SIP Trunking Service Provider leaving 4 digits for the 3300 to translate and ring the remaining 4 digit extension. For example, Engin SIP Trunking Service Provider delivers 282144691 through the SIP trunk to the 3300. The 3300 will absorb the first 5 digits (28214) leaving the 3300 to ring extension 4691. Extension 4691 must be programmed as a valid dialable number in the 3300. Please refer to the 3300 System Administration documentation for further programming information.

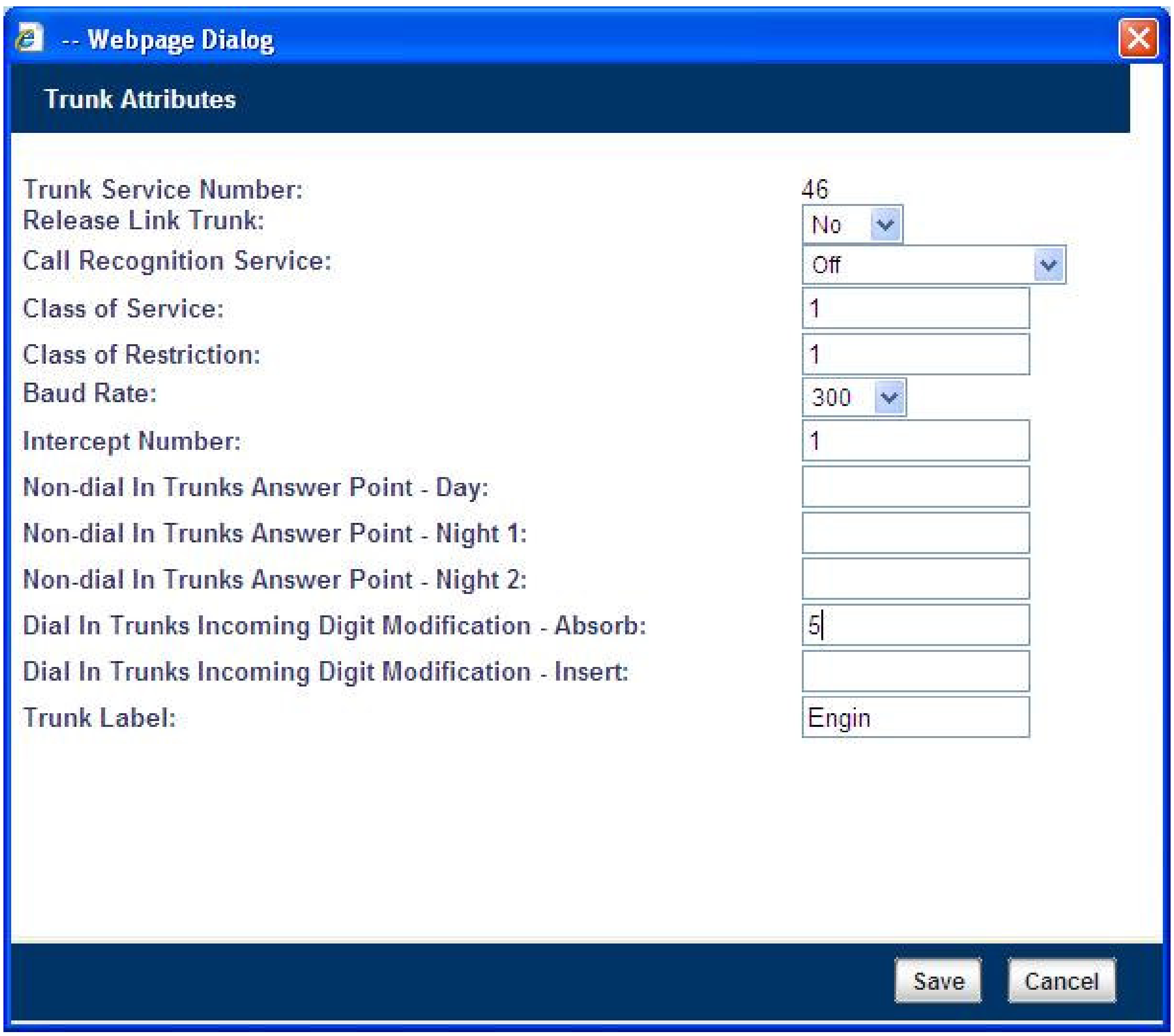


Figure 6 – Trunk Service Assignment

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base 3300 MCD Platform. The SIP Peer Profile should be configured with the following options:

Network Element: The selected SIP Peer Profile needs to be associated with previously created "Engin" Network Element.

Registration User Name: A registration name is required to connect to the Engin SIP Trunk Service Provider. The 3300 does not support Bulk Registration, therefore trunks will have to be registered individually. Enter the DIDs assigned by Engin SIP Trunking Service Provider. Enter one or more numbers. The field has a maximum of 60 characters. The maximum number of digits per number is 26. You can enter a mix of ranges and single numbers (for example, "6135554000-6135554400, 6135554500"). Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash.

Address Type: Use FQDN in SIP messages.

Outbound Proxy Server: Select the Network Element previously configured for the Outbound Proxy Server.

Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by** Engin SIP Trunking Service Provider. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).

Trunk Service Assignment: Enter the trunk service assignment previously configured.

SMDR: If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).

Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by Engin SIP Trunking Service Provider.

Route Call Using To Header: Yes

Avoid Signaling Hold to the Peer: Yes

Prevent the Use of IP Address 0.0.0.0 in SDP Messages: Yes

Suppress Use of SDP Inactive Media Streams: Yes

Disable Reliable Provisional Responses: Yes

Use P-Asserted Identity Header: Yes

Use To Address in From Header on Outgoing Calls: Yes

NOTE: Ensure the remaining SIP Peer profile policy options are similar the screen capture below.

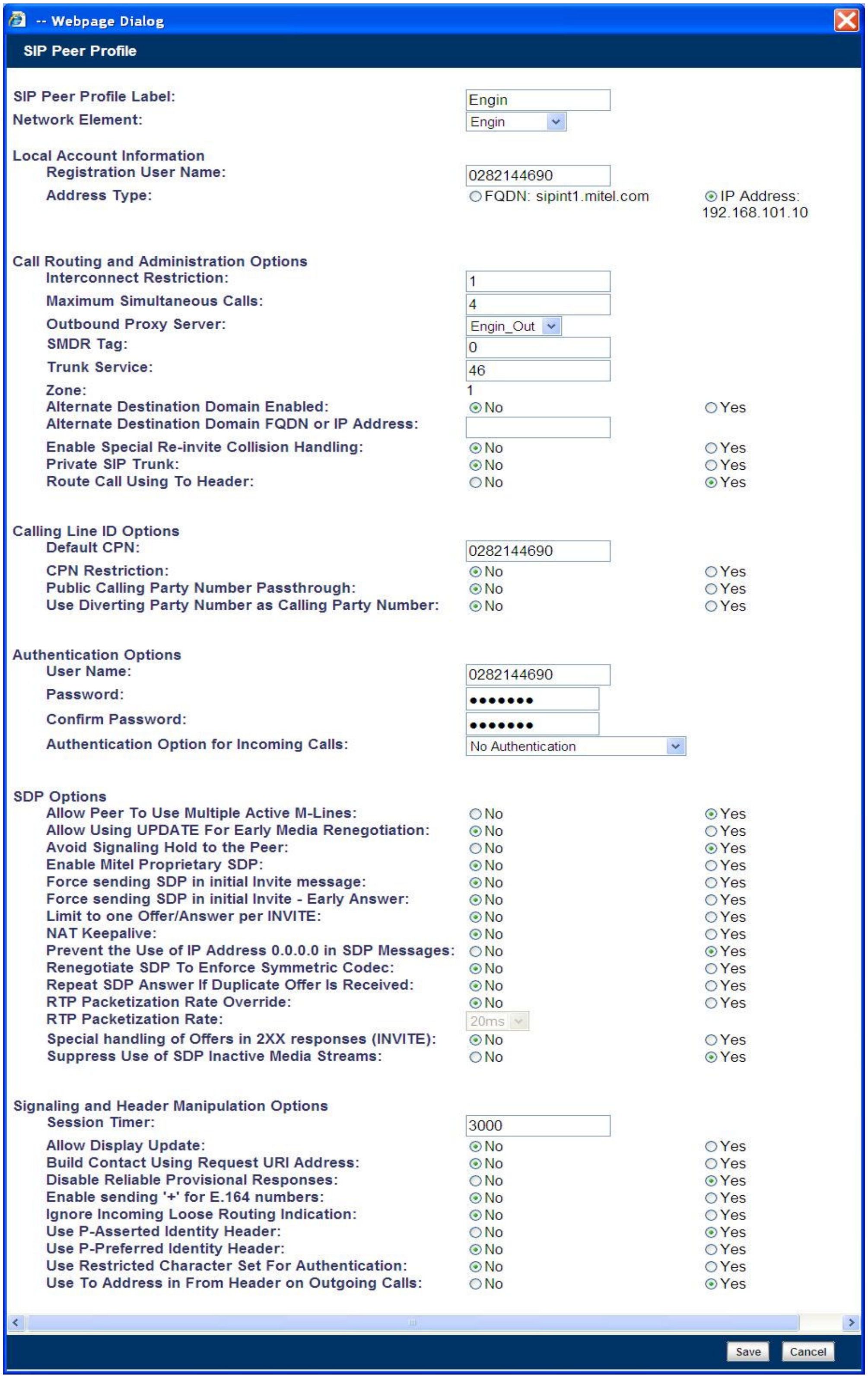


Figure 7 – SIP Peer Profile Assignment

SIP Peer Profile Assignment by Incoming DID

This form is used to assign incoming digits from Engin SIP Trunking Service Provider. DID range numbers assigned by Engin SIP Trunking Service Provider and are associated to a particular SIP Peer.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, "6135554000-6135554400, 6135554500"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 character maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

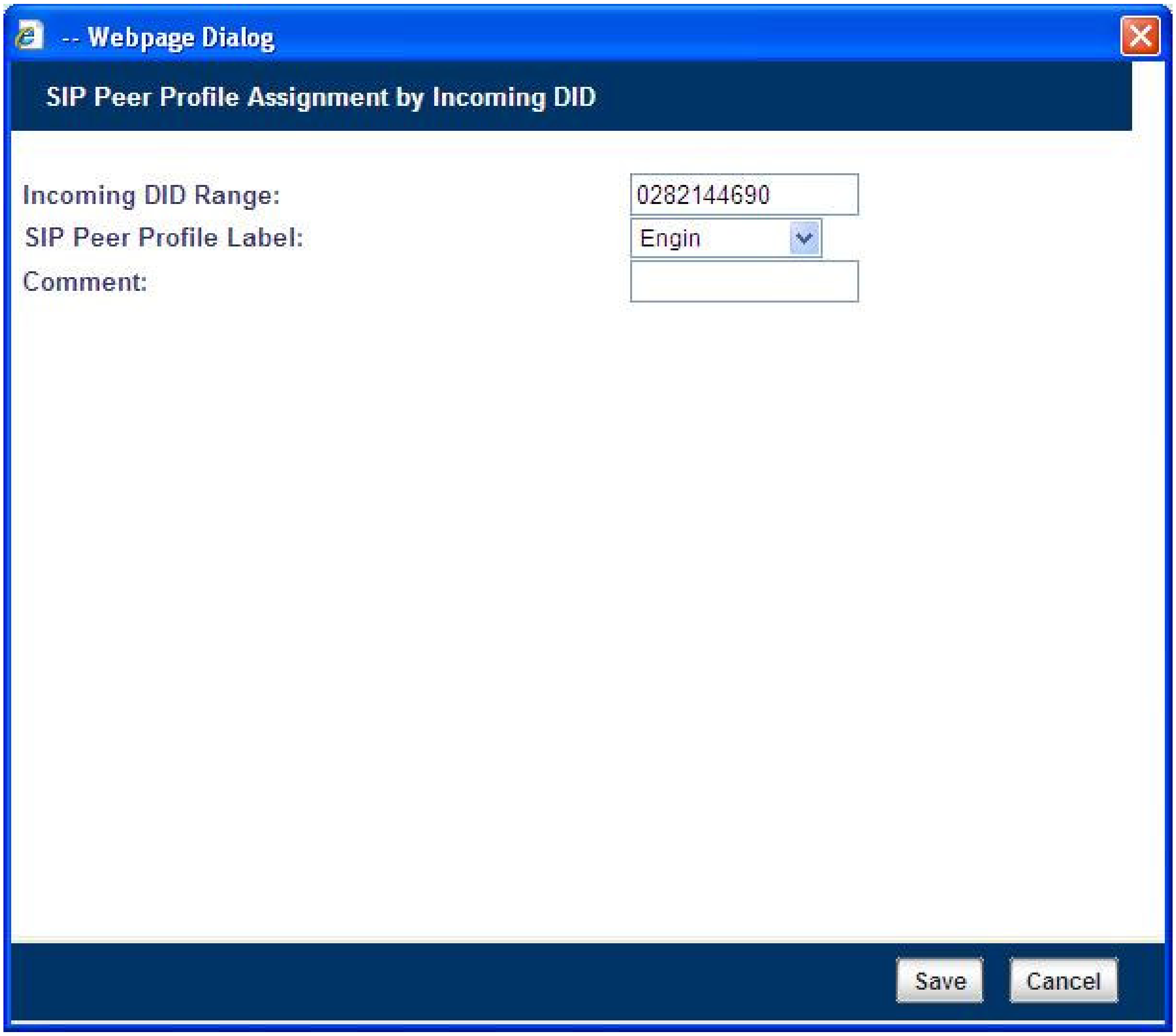


Figure 8 – SIP Peer Profile Assignment by Incoming DID

Digit Modification Number

Ensure that Digit Modification for outgoing calls on the SIP trunk to Engin SIP Trunking Service Provider absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 1 digit (in this case will be 9 to dial out).

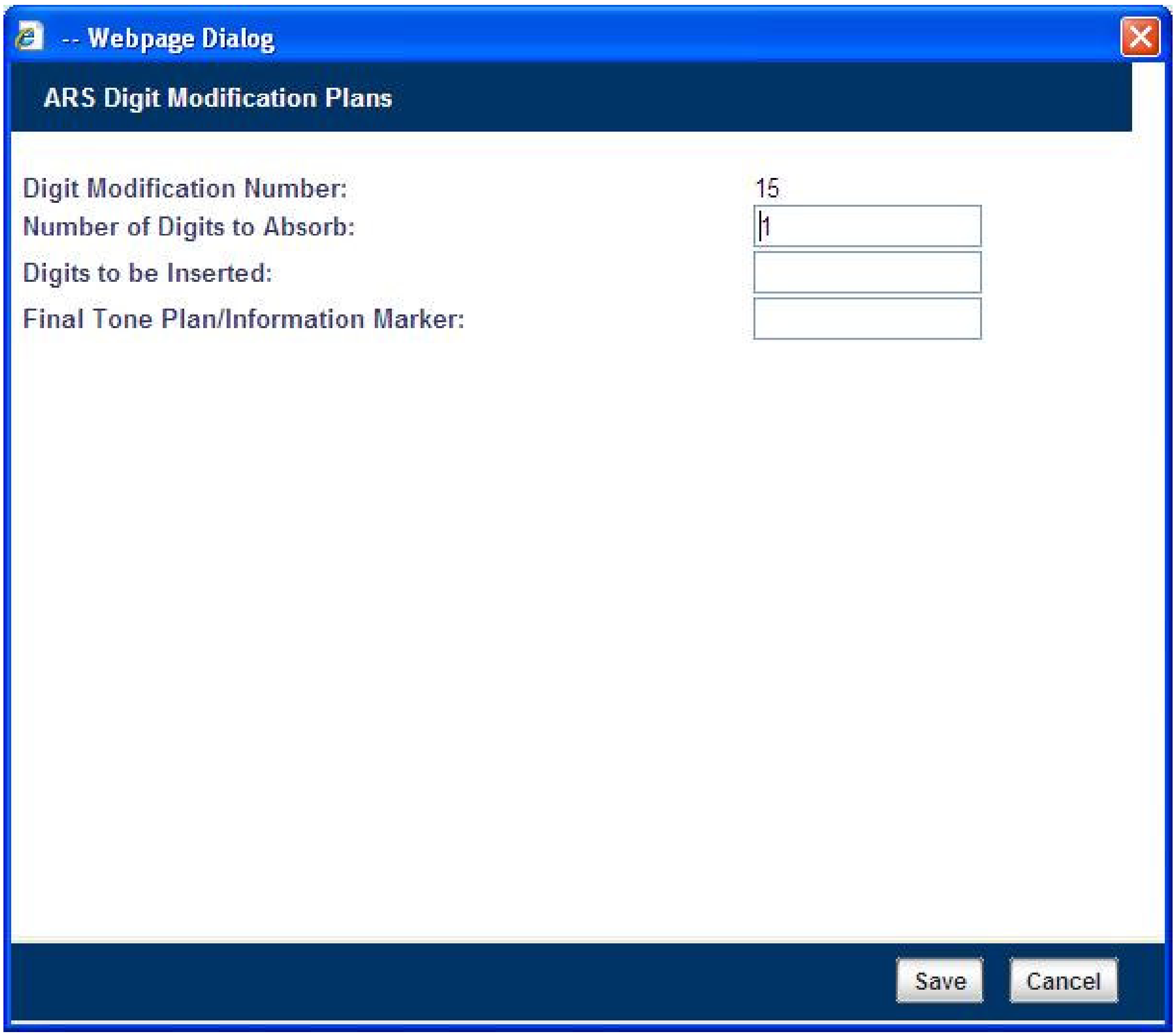


Figure 9 – Digit Modification Assignment

Route Assignment

Create a route for SIP Trunks connecting a trunk to Engin SIP Trunking Service Provider. In this example, the SIP trunk is assigned to Route Number 45. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

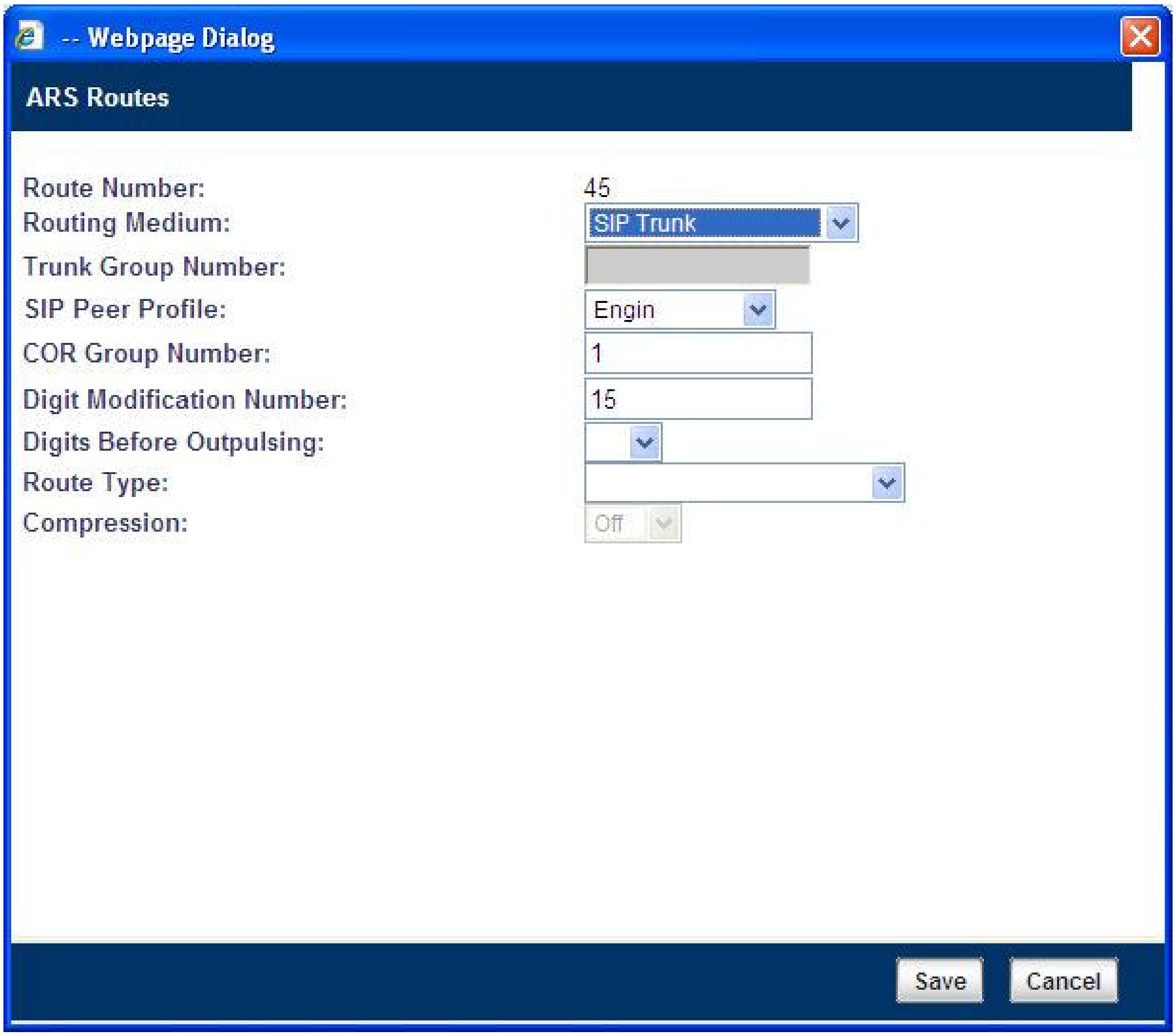


Figure 10 – SIP Trunk Route Assignment

ARS Digits Dialed Assignment

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 902, the call will be routed to Engin SIP Trunking Service Provider (ie. Route 45).

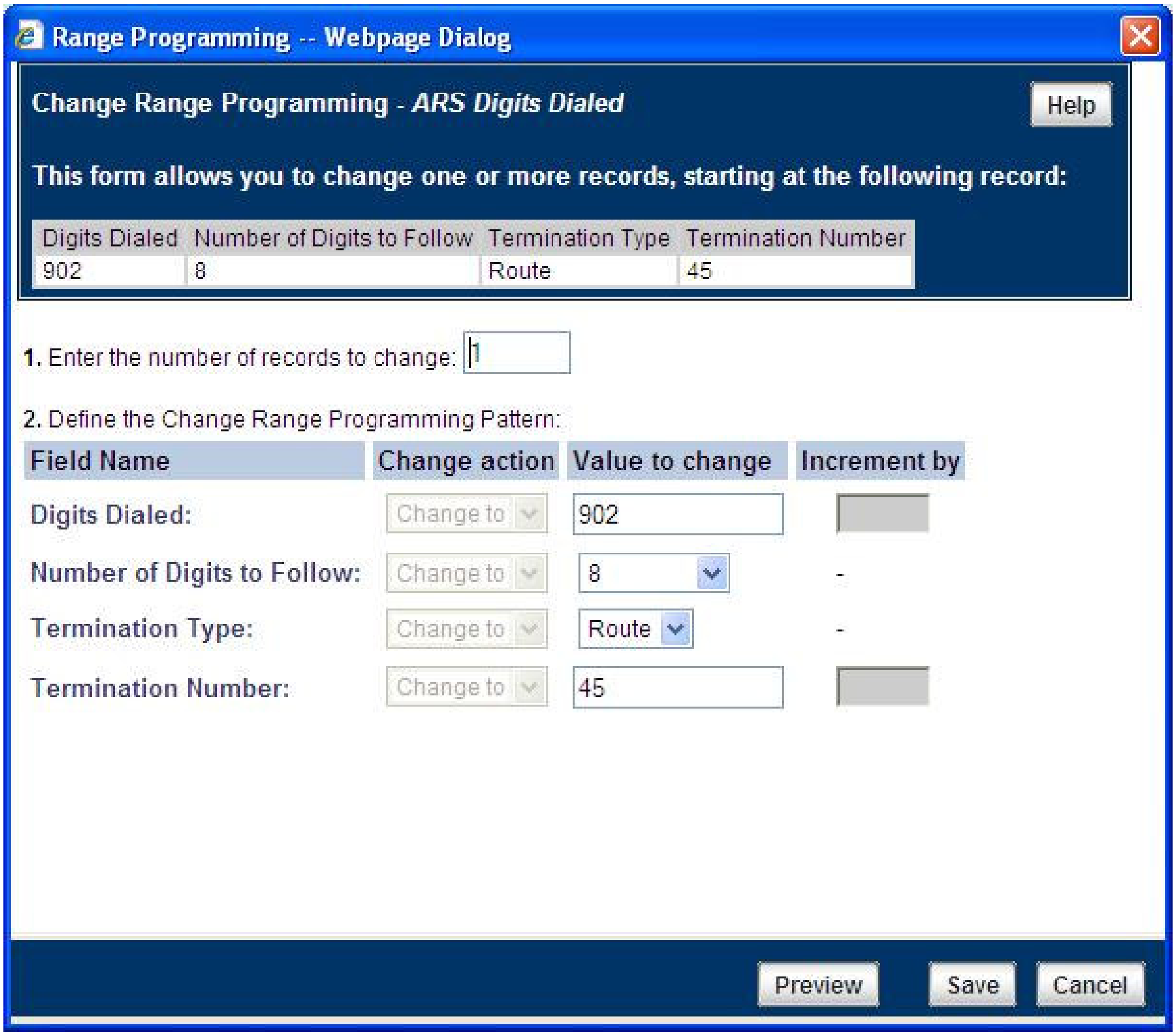


Figure 11 – ARS Digit Dialed Assignment

Global Headquarters	U.S.	EMEA	CALA	Asia Pacific
Tel: +1(613) 592-2122 Fax: +1(613) 592-4784	Tel: +1(480) 961-9000 Fax: +1(480) 961-1370	Tel: +44(0)1291-430000 Fax: +44(0)1291-430400	Tel: +1(613) 592-2122 Fax: +1(613) 592-7825	Tel: +852 2508 9780 Fax: +852 2508 9232

For more information on our worldwide office locations, visit our website at www.mitel.com/offices

www.mitel.com

MITEL®