

Migrating Unify OS15 and OS40 SIP phones to MiVoice MX-ONE

INSTALLATION INSTRUCTION



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1

INTRODUCTION

1.1

GENERAL OVERVIEW

This document describes the migration, i.e. reuse of Unify OpenStage 15 and OpenStage 40 SIP phones for integration with the MiVoice MX-ONE system.

This scenario is valid for customers who wish to migrate/transition their existing Unify system (or 'merge' a network of systems) to a new MiVoice MX-ONE Service Node and keep any already made investments in OpenStage 15&40 SIP terminals.

The support for provisioning of the Unify OS15 and OS40 terminals with MX-ONE 6.1 (or later version) provides a straight forward method for moving users with these existing Unify SIP terminals in controlled steps with a minimum of disruption from the old system to a new MX-ONE.

We recommend that the customer at the same time replaces analog and digital telephones with SIP terminals as such a change avoids re-cabling in the MDF for moving users. If analog terminals are kept, these can of course be moved to the new MX-ONE system as well, but this will require re-cabling.

Note that Unify digital handsets are **not** supported by MX-ONE, and therefore should be transitioned to SIP terminals offering equivalent feature and functions.

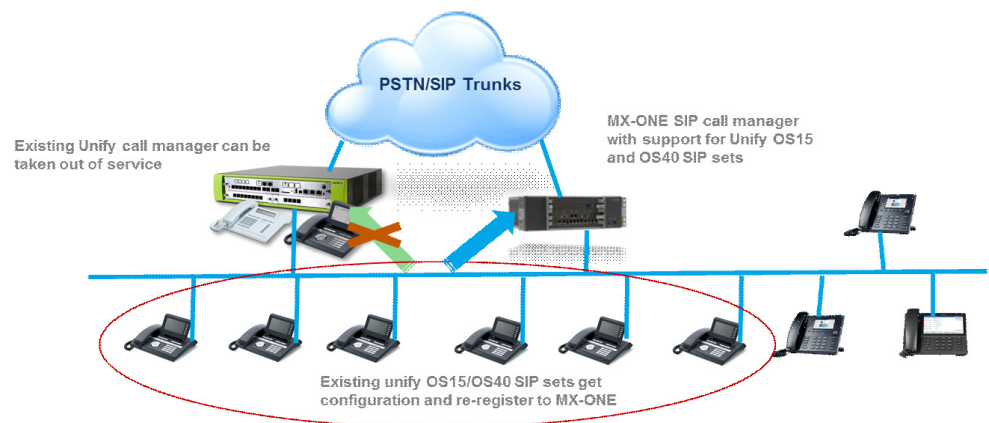


Figure 1: Typical migration/transition scenario - from Unify to MX-ONE.

In the above figure 1 example, a MiVoice MX-ONE call manager is placed in the customer data center side by side with the existing system. There is no need to inter-connect the systems and the Unify system remains operational as per normal until the OS15 and OS40 terminals have been moved to the MiVoice MX-ONE system.

The new system carries the necessary SIP licenses and the Unify SIP phone configuration utility is enabled to allow existing OS15 and OS40 SIP phones to be provisioned via the MX-ONE and the terminals will then be registered as SIP extensions on the MX-ONE. There is no manual intervention needed by the user other than entering his Userid/Extension number and password.

If a portion of analog users are to be maintained, Mitel can provide the necessary chassis HW to accommodate these TDM users. For the existing Unify OS15 and OS40 SIP phone users, there is nothing to do other than to modify the configuration server address (via DHCP) to point the SIP terminals to the configuration service in the new MX-ONE call manager instead of the Unify DLS server. Configuration information, including key assignments will be automatically downloaded to the phones once the user logs in with his credentials.

The MiVoice MX-ONE Provisioning Manager web-based user self-service portal can be used to allow users to configure certain keys on the terminals.

2

PREREQUISITES

- The customer has a number of Unify 15/40 SIP phones, that should be kept and reused together with a MiVoice MX-ONE system.
- Required licenses are available.
- If a DLS server shall be used for the configuration of the phones, that server shall be in place.

3 INSTALLATION AND CONFIGURATION

3.1 GENERAL

A MiVoice MX-ONE 6.1 feature allows customers with existing Unify OS15 and OS 40 terminals to move their SIP users from a single Unify system or a complete network of systems, to a completely new MiVoice MX-ONE system that handles the necessary provisioning and support for these existing Unify SIP terminals.

Note: Unify to MX-ONE migration/transition scenarios are supported from MiVoice MX.ONE 6.1 and onwards.

One of the unique value proposition with the Unify migration to MX-ONE 6.x is the fact that customers with existing Unify OpenStage 15 and 40 SIP handsets will be able to re-use them with the MX-ONE.

The Unify terminal configuration service in the MX-ONE Service Node is a licensed feature that allows Unify OS15 and OS40 terminals to get their configuration data based on the user's login credentials (directory number and password).

The configuration rules can be prepared in the MX-ONE Service Node and downloaded via HTTPS directly to the phone. The Unify SIP terminals only need to get the IP address or DNS name (via DHCP) of the MX-ONE server that has the Unify terminal configuration service active and licensed. Even keys can be defined for the Unify terminals and a user portal can allow the end-user to create or move keys around for his phone. Below is a synopsis of the support and services available with Unify OS15 and OS40 handsets registered to MX-ONE 6.1 (or later version).

3.2 SUPPORTED HANDSETS/SIP PHONE MODELS

- OpenStage 15
- OpenStage 40

3.3 SUPPORTED HANDSET FIRMWARE

Local and 3rd party functional testing has been carried out on following versions and some intermediate releases.

- V3 R4.4.0 (latest verified release from Unify, Mitel recommended version)
- V3 R1.49 and later versions have been validated to function with MX-ONE.

Note: It is always recommended to use the latest supported version to ensure the best level of interoperability.

3.4 HANDSET PROVISIONING OPTIONS

MX-ONE acts as a DLS provisioning server. (See Unify documentation on DLS).

- Each service node in a single system acts as an independent “DLS” server.
- SIP Server configuration is dynamically assigned using FQDN.
Configuration via DLS.conf file.

- SIP Server configuration dynamically assigned using IP address.
Configuration via DLS.conf file.
- DHCP option 43 is used to provision the “DLS server” address.
The handsets use the DLS server to receive their configuration settings during log on/off and during migration / regression.

3.5 SYSTEM SETTINGS (DLS CONFIGURATION)

The following items can be configured for the Unify phones, on a system level:

- Handset administration password.
- User administration password.
- Software deployment settings (FTP).
- Date, Time and Time Zone settings.
- Handset Codec handling.
- Device specific options.

3.6 PROVISIONING MANAGER

The MiVoice MX-ONE Provisioning Manager can be used also for the Unify terminals as a single point of entry web-based management portal for managing all system and user services as well as application provisioning, which can be accessed from anywhere in the customer network. For example for:

- Centralized key management (with phone type “other”).
- TNS & MNS keys.
- Add-on key module support.
- Shift key support.

3.7 SUPPORTED FEATURES

Most end user features that are supported for Mitel SIP terminals, i.e. basic calls and supplementary services, work also for the Unify SIP terminals.

See descriptions UNIFY OPENSTAGE 15/40 SIP PHONE FUNCTIONALITY IN MIVOICE and MX-ONE.MIVOICE MX-ONE FEATURE MATRIX.

4

TEST OF THE UNIFY SIP PHONES

- Log on the Unify SIP phones.
- Call between two Unify SIP phones. Check that alerting and answer works as intended. Disconnect the call.