



Modular Messaging MultiSite Guide Release 5.2

September 2012

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Chapter 1: About MultiSite

Why MultiSite is needed

Enterprises that are spread across different locations need an efficient messaging system for seamless communication. These enterprises might be using different makes or models of telephony PBXs, which in turn imposes a host of restrictions to the communication needs. In the case of Modular Messaging Release 4.0 or earlier, the system setup mandates that a Message Application Server (MAS) must be installed in close proximity, both with the PBX and the message store. This forces the enterprise to have at least one MAS and one message store for each location. If there are many locations then this can prove to be very expensive given the extra equipment and maintenance costs.

To achieve the maximum benefits, the enterprise will most likely need to network these systems so that subscribers on each system can send messages to each other and use features such as Reply by Calling Sender. Local subscribers can use the short address to check their messages. However, subscribers will typically need to use long addresses to send messages to subscribers who are not located at the same office location. In addition, if they traveled to another location, then they would have to dial back to their home location to check messages, as the Modular Messaging system of their new location would not identify them.

MultiSite can help solve these problems by offering effective, economical, and seamless communications across all branches of an enterprise. Subscribers can have a unified messaging system using a consistent addressing scheme. Servers can be consolidated into a data center instead of being co-located with corresponding PBXs and message stores.

What is MultiSite?

MultiSite allows you to use a single Modular Messaging system to serve subscribers at multiple locations. With MultiSite, MASs in a single Voice Mail Domain (VMD) communicate with multiple PBXs possibly with different dial plans, in different locations.

Consider the example of an enterprise that has offices located in the US (New Jersey and Denver) and the UK (Uxbridge).

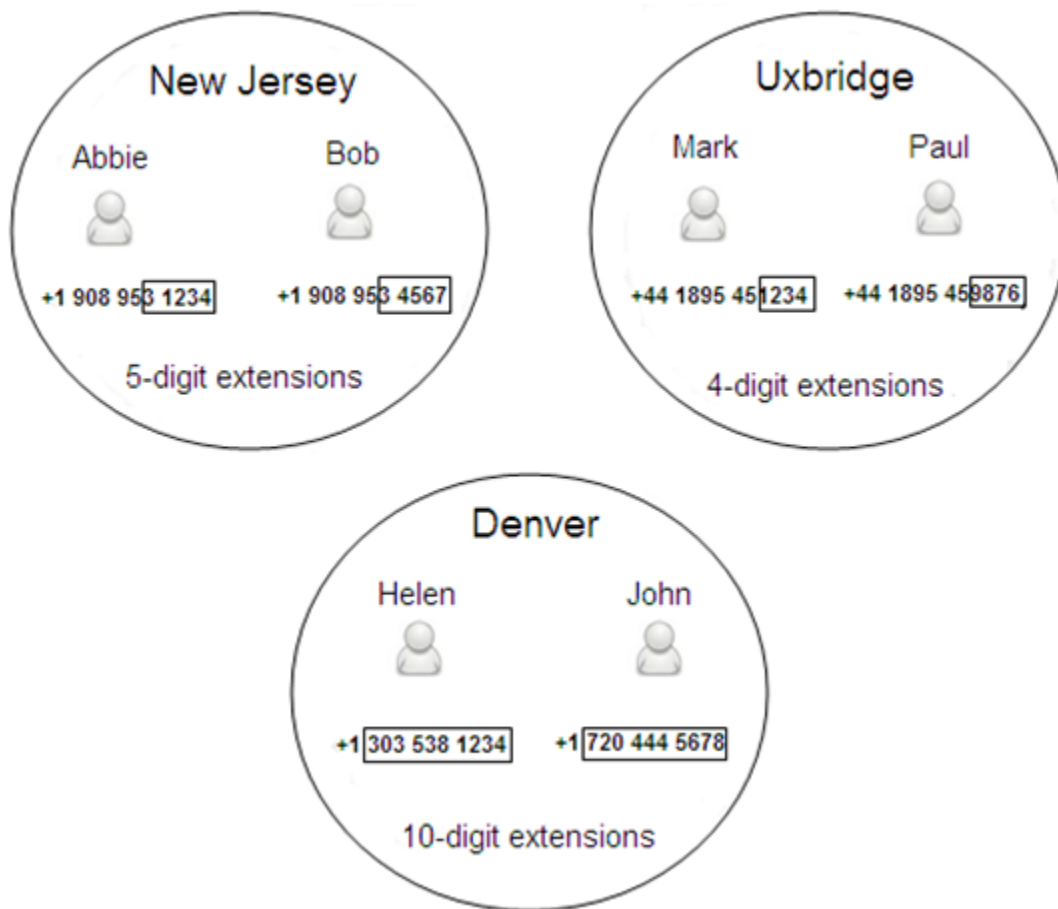


Figure 1: Example of a typical enterprise with multiple locations and dial plans

New Jersey location

With MultiSite enabled, the New Jersey location is organized as shown in *Figure 2*.

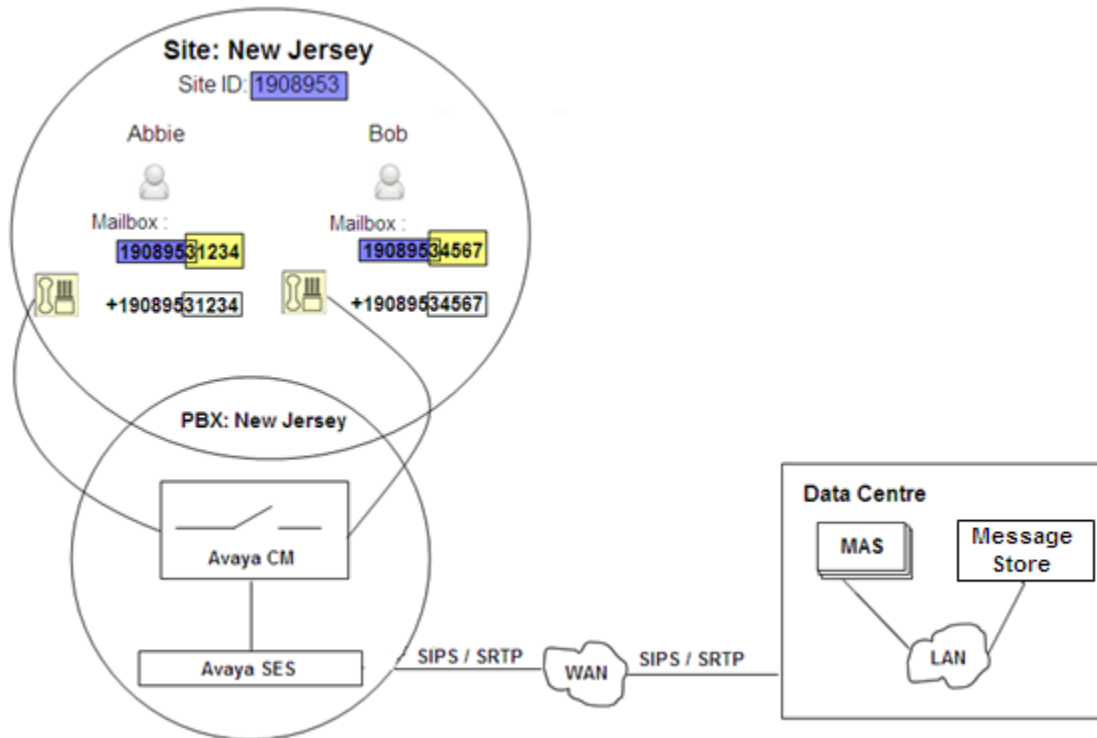


Figure 2: The example New Jersey site with MultiSite

Subscribers are given a full mailbox number that uniquely identifies them in the VMD, but the subscribers usually need to use only their short mailbox number. In *Figure 2* note that the short mailbox number (in yellow) matches the PBX extension. The full mailbox number is made up from the short mailbox number, and a prefix. All subscribers in the same site have the same prefix. This prefix is the site identifier.

Note that, in this example, the first digit of the short mailbox number is the same as the last digit of the site ID. You can configure the subscriber's E.164 phone number (see [E.164](#) on page 72) in such a way that the site ID and short mailbox number overlap by any predefined number of digits or no overlap at all. MultiSite mailbox numbers and extension numbers can be up to 50 digits long. However, if you follow the recommended scheme, then they should not be longer than 15 digits.

A site is a MultiSite term that can be thought for now as being the same as an office location. The site that is encoded in a subscriber's full mailbox number is called their home site.

Avaya recommends that the site ID and mailbox numbering scheme use the same digits as the subscriber's E.164 phone number. This ensures the numbers used in the VMD are unique. For more information, see [How mailbox numbers work with MultiSite](#) on page 17.

Subscribers can send messages to other subscribers in the same site by using the short mailbox number (in yellow) of the intended recipient. Subscribers can also send messages to anyone in the VMD by using their full mailbox number (in blue and yellow).

A mailbox number identifies a Modular Messaging subscriber. Phones are merely associated with that subscriber. Although the mailbox numbers in this example have the same digits as

the phone numbers, they do not have to match. This is not a change introduced with MultiSite, but as an administrator, keeping this in mind can avoid a lot of confusion.

The New Jersey PBX is connected to a remote data centre in a secure location. The remote connection uses secure SIP and SRTP IP protocols to keep the voice data private. The Modular Messaging system in the data centre appears almost normal – there are several MASs and a single Message Store, all connected to the WAN. The difference, compared to a non-MultiSite system, is that there is no PBX in the data centre. All PBXs in a MultiSite system are integrated to the MASs using SIP over an IP network.

Uxbridge location

The MultiSite enabled Uxbridge location is organized as shown in *Figure 3*, and introduces some more concepts.

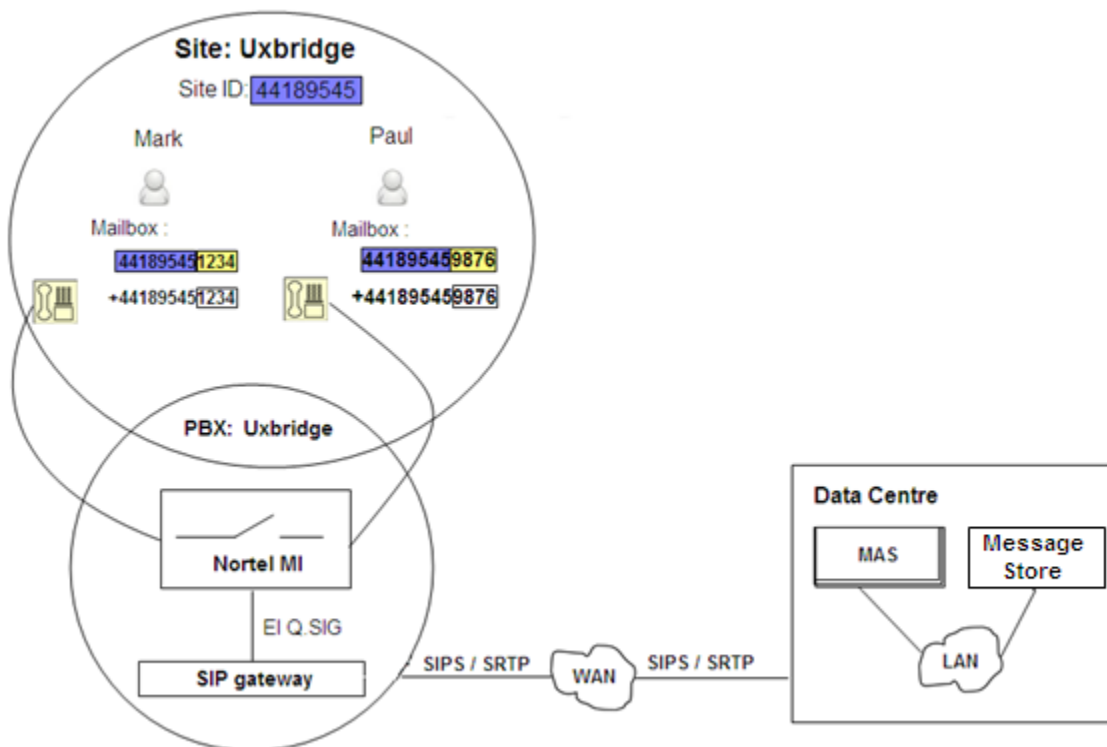


Figure 3: The example Uxbridge site with MultiSite

This site is very similar to the New Jersey site, but note that it has 4-digit mailbox numbers. The data centre is the same one as in *Figure 2*, signifying that a single MultiSite enabled Modular Messaging system can now handle several different mailbox lengths.

A Modular Messaging subscriber from any site can login to the system by calling the Modular Messaging pilot number on the Uxbridge PBX. Subscribers in the Uxbridge site have to enter only their short mailbox number to login. Anyone from any other site can login by entering their full mailbox number.

The SIP gateway allows a Modular Messaging system to work with PBXs that are not supported by the SES, mainly those from third-party vendors. AudioCodes gateways support analog/

inband, analog/SMDI, T1 Q.SIG, E1 Q.SIG and SIP integrations to the PBX, but they do not support DSE. Dialogic DMG 1000 gateways support DSE.

Denver location

The MultiSite enabled Denver location is organized as shown in *Figure 4*. Here, several Modular Messaging sites use the same PBX.

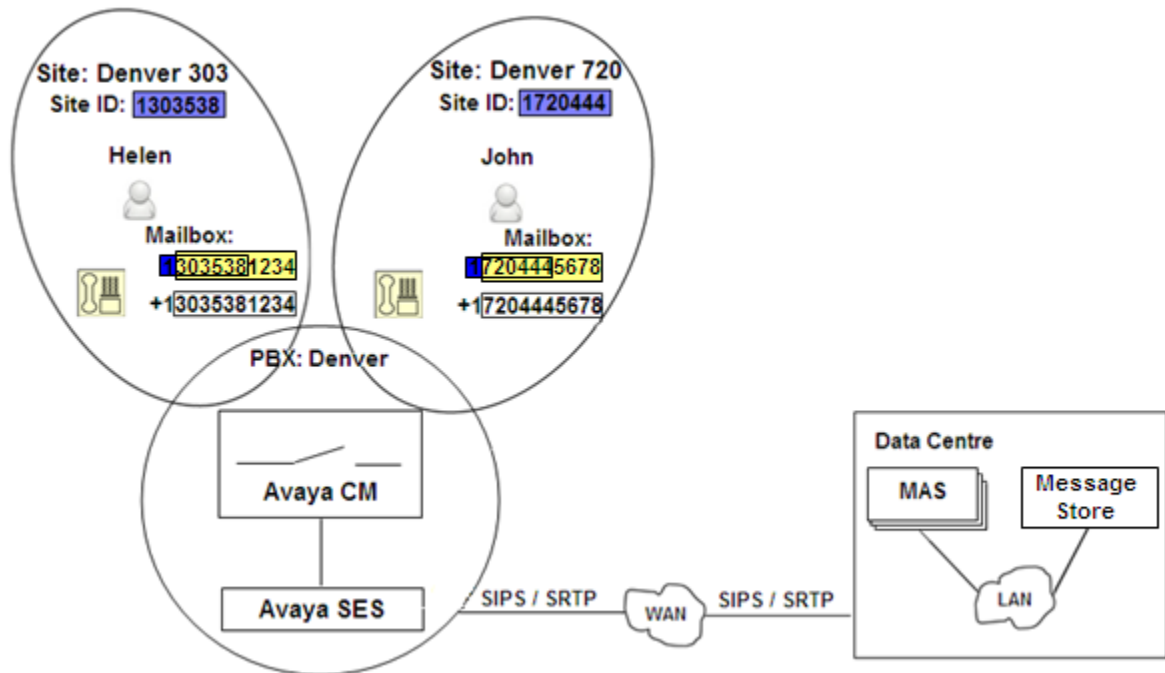


Figure 4: The example Denver sites with MultiSite

For this location, there are two different forms of extension number: 303538xxxx and 720444xxxx. Two sites are required because it is desirable for the subscribers' mailbox numbers to be the same as their extension numbers, but all mailbox numbers that are members of the same site must start with the same prefix. In this example, the sites have been defined with short mailbox lengths of four digits, but that is not significant to subscribers. MultiSite-enabled systems can accept partial mailbox numbers wherever a mailbox number must be supplied, so subscribers can use their extension number as if it was their mailbox number. For example, Helen can logon by entering 3035381234 as her mailbox number, and she can reach John at 7204445678.

Subscribers, in either the Denver 303 site or the Denver 720 site, can address messages to others in their own site or the other Denver site using 4-digit short mailbox numbers. If the short mailbox number is not unique, Modular Messaging prompts the subscribers to choose the intended recipient from a list including matches in the same site, matches in the other sites that use the same PBX, and other matches in the VMD. Alternatively, the subscribers can use as many digits as are required to resolve the differences, for example Helen, a member of the Denver 303 site, may use 45678 to reach John in the Denver 720 site.

Figure 5 displays a summary of the site configuration for the three example sites, as it is displayed in VMSC. NANP is short for *North American Numbering Plan*.

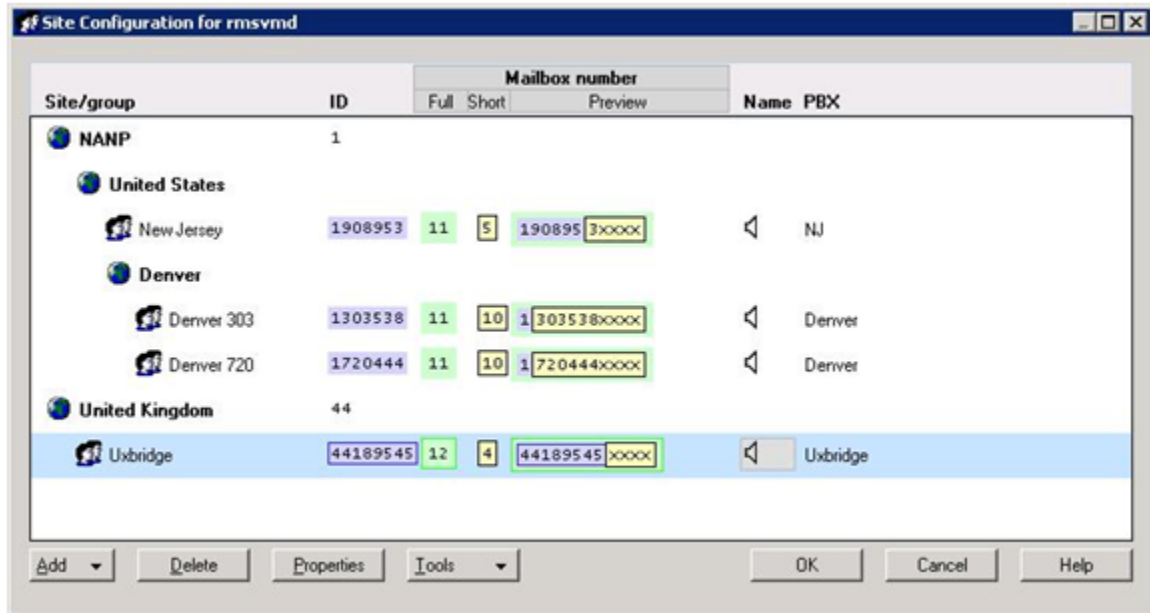


Figure 5: Summary of the sites configuration for the example enterprise, as shown in VMSC

How MultiSite affects incoming calls

Call answering

When Modular Messaging answers an incoming call on behalf of a subscriber, it has to identify the subscriber from the phone number passed to it by the PBX. When using MultiSite, it is quite likely that the phone number supplied by the PBX is not sufficiently unique to identify the subscriber on its own.

For example, *Figure 6* displays what happens when Modular Messaging answers Helen's phone; Helen is in the Denver 303 site, and her extension is on the Denver PBX. The PBX sends the phone number '1234' to Modular Messaging, but that extension is also shared by Mark at the Uxbridge site.

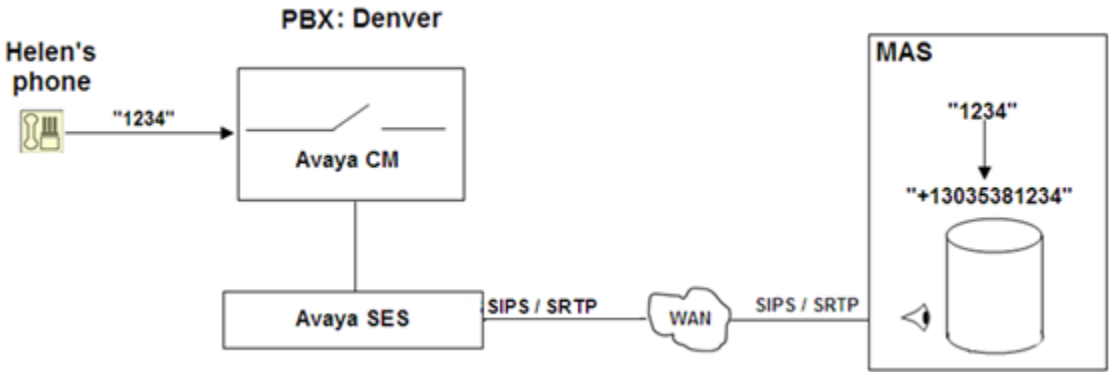


Figure 6: What happens when Modular Messaging answers a call to a subscriber's phone

To avoid ambiguity, MultiSite enabled Modular Messaging stores all primary and secondary extension numbers in canonical form. MASs convert all phone numbers received from PBXs into canonical form before looking them up in the subscriber database. Canonical numbers follow the ITU E.164 standard.

Following are some important points about E.164 phone numbers:

- They represent the full phone number, including the country code and the area code.
- They do not include any punctuation such as parentheses, hyphens, or spaces.
- They do not include any access codes, for example 9 for an outside line, or 011 for an international call.

Modular Messaging identifies the PBX that is calling and uses this information while converting the switch-native phone number into canonical form. Once the call has been answered, Modular Messaging accesses the subscriber mailbox using the canonical phone number to get the usual properties that control the interaction with the caller, such as the language to use, greetings to play, and so on.

The following table displays some examples of canonical phone numbers, and how they relate to the more common forms.

Table 1: Examples of canonical phone numbers

US	(908) 953-1234	→	+19089531234
US	(303) 538-1234	→	+13035381234
UK	01895 451234	→	+441895451234
UK	001 908 953 1234	→	+19089531234
Uxbridge	1234	→	+441895451234
Denver 720	1234	→	+17204441234

Automated attendant

When a call comes directly into Modular Messaging, without being diverted from an unanswered subscriber's phone, Modular Messaging answers the call. Modular Messaging must then determine the language prompts to use, custom prompts, office holiday schedule, and so on. These settings are normally configured for each VMD, but when MultiSite is used they are configured per site.

In most cases, the MAS can easily determine the correct site because each site typically uses a different PBX, so there is a direct correspondence between the PBX being used for the incoming call, and the site from which the appropriate settings should be taken. But this is not the case when several sites share the same PBX, for example, the Denver sites in the example enterprise.

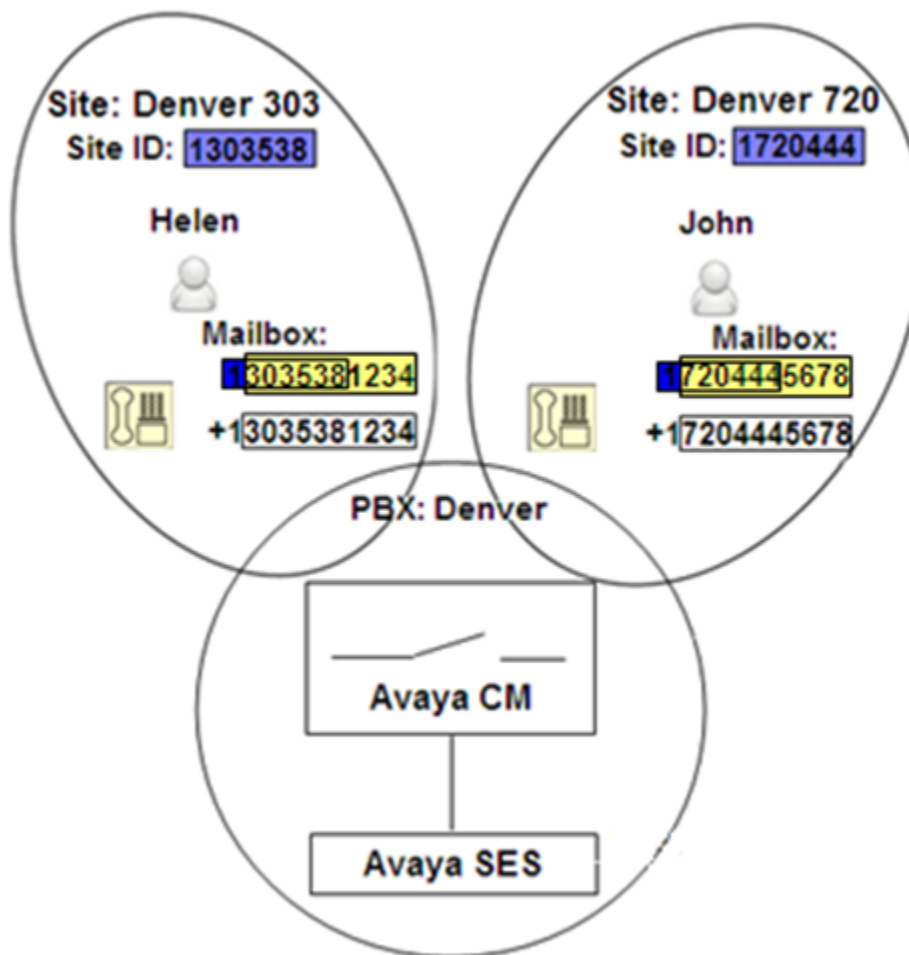


Figure 7: Several sites can share a single PBX

In this case, it is likely that both sites use the same language, prompts, and so on. The subscribers are not really split into two different communities; the sites are just a result of the

dialing codes used in the public phone network. Given that, the MAS can choose either of the sites that are associated with the Denver PBX and everything works as expected.

MultiSite enabled Modular Messaging allows you to use duplicate short mailbox numbers, but only if the site can be uniquely identified by calling PBX and/or hunt group. When Modular Messaging answers the incoming call, it creates a list of possible sites based on the calling PBX. In this example, this list would comprise of the Denver 303 and Denver 720 sites. Modular Messaging then compares the called-party number against the shortlisted sites' hunt group numbers to find the correct site, and uses that site's configuration for the call.

Modular Messaging Release 5.2 systems support the Personal Operator Zero Out feature that redirects call coverage for the SIP integration. When a caller transfers the call to a personal operator and the operator does not answer the call, the Modular Messaging systemwide settings determine where the original called party can deposit the message. After the call covers to the Modular Messaging system, the caller's message is deposited either in the original called party's mailbox or in the personal operator's mailbox, based on the system configuration. A canonical hunt group number is required for the Personal Operator Zero Out feature when a call is placed directly to the Auto Attendant. Calls that reach a MultiSite enabled Modular Messaging system that do not integrate with a known mailbox or that were made to a recognized hunt group number may not be able to use System or Personal Operator transfers.

How MultiSite affects outgoing calls

All calls made by a MultiSite enabled Modular Messaging system have an associated cost. The system compares these costs against the limits that the administrators have configured, to determine whether to allow or block the call.

For more information, see [Preventing toll fraud](#) on page 32.

How MultiSite affects subscribers

If you want to enable MultiSite in an existing Modular Messaging VMD, then you should plan the migration well to ensure that most subscribers will not notice any difference. Carefully plan your mailbox numbers before you enable MultiSite and make use of its abilities to have long and variable-length mailbox numbers. For example, if you intend to use Message Networking or one-X Speech, be aware that they do not currently support long or variable-length mailbox numbers.

For more information, see

- [Migrating a single Modular Messaging VMD to MultiSite](#) on page 55.
- [Migrating several Modular Messaging VMDs to MultiSite](#) on page 58.
- [Migrating legacy voice mail systems to MultiSite](#) on page 58.

Differences in the TUIs

The primary differences that subscribers can notice in the TUIs are:

- Subscribers can log in to Modular Messaging from any PBX that is configured as part of the MultiSite system. But, if the subscriber logs in to Modular Messaging using any other PBX other than the subscriber's home PBX, then they can use their full mailbox number to log in.
- Subscribers can send messages to anyone in the VMD, even if they are in a different site, using the recipient's full mailbox number.
- When addressing messages using the TUIs, Modular Messaging confirms the addresses in the form "[name] at [site]".

Differences in the clients

Subscribers can enter phone numbers in either switch-native or canonical form.

How MultiSite affects callers

MultiSite affects callers in the following ways:

- Callers can use the Automated Attendant to enter the mailbox number of any subscriber on the Modular Messaging system. Modular Messaging attempts to transfer the call, even if the subscriber's site uses a PBX that is different to the one that is handling the call.
- If the caller uses dial-by-name and Modular Messaging finds several matches, then Modular Messaging announces the matches in the form "[name] at [site]" to help the caller ensure that they reach the intended subscriber.

Chapter 2: Mailbox numbers and sites

How mailbox numbers work with MultiSite

Whenever a mailbox number is required, subscribers can always enter a full mailbox number, since it uniquely identifies the subscriber across the VMD. Subscribers can also enter short mailbox numbers. There can be duplicate short mailboxes within the enterprise, but proper integration depends upon two factors; PBX and hunt group. Modular Messaging identifies the site based on the PBX the call comes in on. If multiple sites are configured with the same PBX, the site is further determined by matching the hunt group. In the case that all sites under the same PBX IP have the same hunt group, Modular Messaging will attempt login to the first site in this short returned list.

Consider the following examples:

- If a subscriber logs in to Modular Messaging using their home PBX, then they can enter their short mailbox number, even if it is not unique in the VMD. When a subscriber attempts to logon to a TUI, Modular Messaging looks up all the mailboxes that match the entered mailbox number. If there is more than one match, then Modular Messaging prioritizes any that are in local sites. If there is still more than one match, then the logon fails; Modular Messaging then prompts the subscriber to enter more digits to make their mailbox number unique.
- If a subscriber addresses a message by giving a partial mailbox number that matches more than one recipient, then Modular Messaging announces recipients in the home site followed by recipients in other sites. This is true for any PBX that the subscriber logs in to.

A local site is one whose associated PBX is handling the current call. Using our example enterprise, if somebody dialed into the Denver PBX, then Denver 303 and Denver 720 would be local sites. If somebody dialed into the Uxbridge PBX, then only the Uxbridge site would be local.

Subscribers generally need to use only their own short mailbox number. The only time that they may have to use their full mailbox number is when they travel to a different location and want to log in to the local Modular Messaging system; in that case, they are likely to have to use their full mailbox number, because their short mailbox number is unlikely to be sufficiently unique. Another option in such cases is to enter a partial mailbox number.

A subscriber can enter a partial mailbox number anywhere that a mailbox number is required. A partial mailbox number is the rightmost portion of a full mailbox number. For example, the

full mailbox number of Abbie is 19089531234, so the following would all be valid partial mailbox numbers: 4, 34, 234, 9531234, and so on.

Sometimes a mailbox number does not have to be unique, for example when addressing a message. Since priority is given to mailboxes that are in a local site, the subscriber might be able to get to the desired mailbox by using very short partial mailbox numbers when addressing other subscribers in their own site.

Planning your mailbox numbering scheme

When you set up a MultiSite enabled Modular Messaging system, you need to plan how you want your subscribers' mailbox numbers to look. Avaya recommends using the E.164 scheme that is used for phone numbers, but without the leading '+', because most administrators and subscribers like their mailbox number and phone number to be the same.

As administrators, these are some of the rules that you must follow, while configuring sites:

- Each site must have a unique site identifier.
- For each site, the full mailbox length must be less than or equal to the total of the length of the site identifier and the short mailbox length.
- If a site group is a child of another site group and it is not simply a container, then the site identifier of the child group must consist of all of the digits of the site identifier of the parent group, plus at least one more digit.
- If a site is in a group then its identifier must consist of all of the digits in the parent group's identifier, plus at least one more digit.
- If one site identifier is an abbreviated version of another site identifier, then these sites must have different full mailbox lengths to be valid. For example, the site identifiers 1303 and 1303538 have the same first 4 digits, and so the sites must have different full mailbox number lengths.

If you have decided to use an E.164-style scheme then you do not have to be concerned about these rules, because E.164 satisfies them all. If you want to design your own scheme, perhaps so that you can network your MultiSite system with other voice mail systems or because you want your mailbox numbers to match a proprietary phone numbering scheme, then you must bear in mind these rules. The system reminds you if you try to configure sites that break these rules.

Remember to make the variable part of the short mailbox numbers long enough to allow for the subscriber population at that site, including reasonable scope for expansion.

If you want to enable MultiSite for an existing Modular Messaging system, then you should consider designing your mailbox numbering scheme so that subscribers' new short mailbox numbers are the same as their old mailbox numbers. That will mean that most subscribers will not notice any difference after you enable MultiSite because they will still be able to log in using their old mailbox number; this will then be their new short mailbox number.

Implementing your mailbox numbering scheme

You can implement mailbox numbering for your site using Voice Mail System Configuration (VMSC). The site ID, full mailbox length, and the short mailbox length are interrelated. The system displays a preview that lets you easily see the effect of any change that you make to any of these parameters

Figure 8 displays the dialog box that you use to create a new site and specify the mailbox parameters. The preview field adjacent to the Identifier field illustrates the mailbox numbering scheme for the site. In the preview field, use the following legend to identify the site identifier, full mailbox number, and the short mailbox number:

- the digits inside the green box are the full mailbox number
- the digits inside the blue box are the site ID
- the digits inside the yellow box are the short mailbox number.

The system updates the preview field as you change the relevant values.

The screenshot shows the 'New Site' dialog box with the following fields and values:

- Parent site group: United Kingdom [44]
- Site name: Uxbridge
- Identifier: 44189545
- Full mailbox length: 12
- Short mailbox length: 4
- PBX: Uxbridge

The preview field next to the Identifier shows the following values:

- 44189545 (in a blue box)
- xxxx (in a green box)
- xxxx (in a yellow box)

Figure 8: Creating a new site using VMSC

Figure 9 shows what happens to the preview if you make a configuration change to increase the short mailbox number length of this site. As the length of the full mailbox number is fixed to be 12, the only way that the short mailbox number can grow larger is by using up digits from the right end of the site ID. In this example, it would mean that all the 5-digit short mailbox numbers would have to start with '5', followed by four other digits.

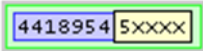


Figure 9: Site identifier overlapping with the short mailbox number

Figure 10 shows what happens to the preview if you reduce the short mailbox length. The question mark indicates that the system cannot determine the full mailbox number, because the site identifier or short mailbox length is too short, or the full mailbox length is too long.

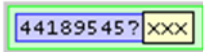


Figure 10: Error in the site configuration

VMSC allows you to save invalid data while configuring the site. However, you cannot modify the Site Identifier after you have created a site and saved the details. After you save the site details and return to the Site Configuration dialog box, VMSC also highlights the errors that it identified in the site configuration.

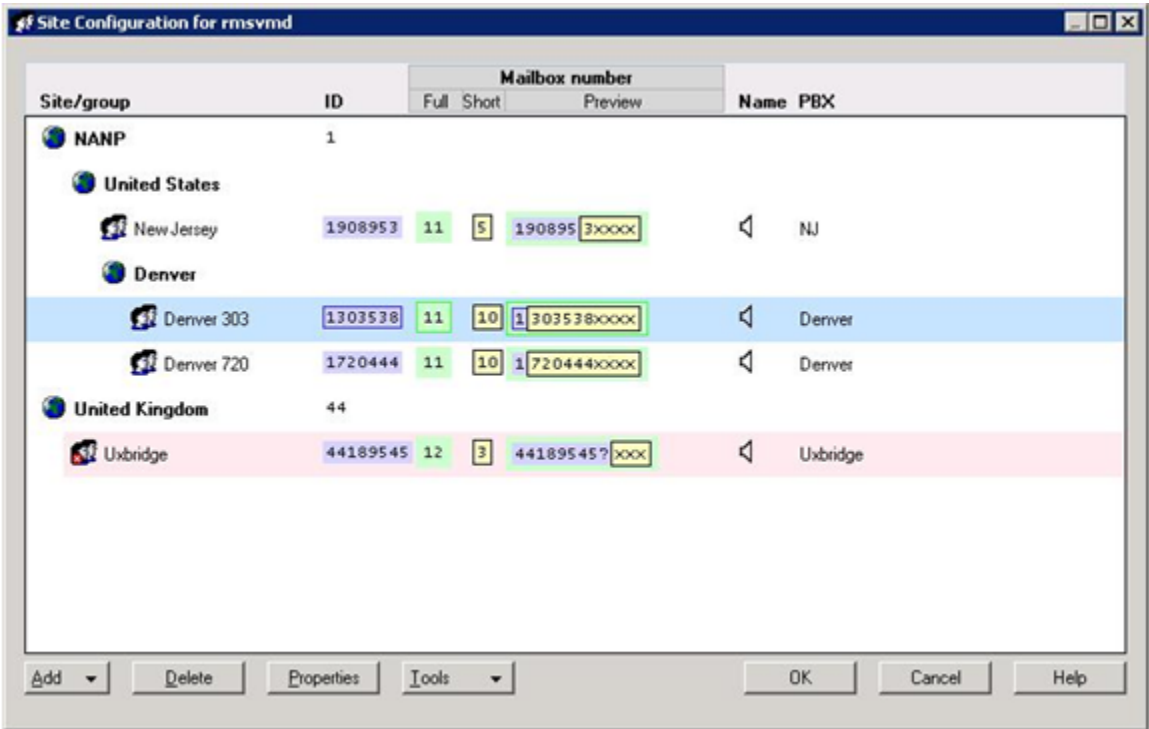


Figure 11: VMSC highlights sites that are improperly configured

When you click on a site that is highlighted because of incorrect configuration, VMSC displays more information about the problem, as shown in Figure 12.

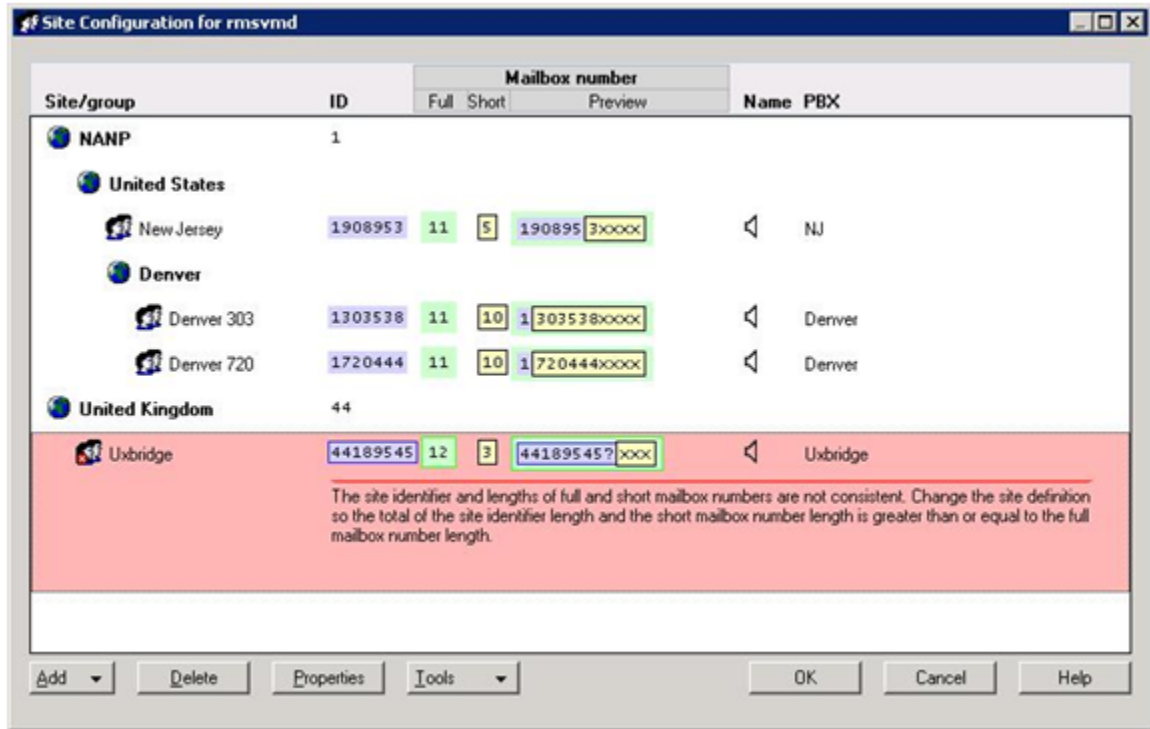


Figure 12: VMSC gives information about the incorrectly configured site

Chapter 3: Mailbox administration

Mailbox administration

Mailbox administration on a MultiSite enabled Modular Messaging system is similar to a non-MultiSite system.

- If you are using MSS backend, then use MSS Administration to administer mailboxes.
- If you are using Exchange backend, then use Active Directory to administer mailboxes.
- If you are using Domino backend, then use Domino Unified Communication (DUC) to administer mailboxes.

Active Directory and Domino Unified Communication (DUC) resolves entered mailbox numbers and displays the associated site and short mailbox number. Note the following changes introduced in a MultiSite enabled Modular Messaging system.

Mailbox numbers and sites

Subscribers are associated with sites by their full mailbox numbers. You must always enter full mailbox numbers when administering subscribers on a MultiSite enabled system. If you enter the wrong site identifier as part of the full mailbox number, then the subscriber's service is affected (for example, they will not be able to logon using their short mailbox number when using what should be their home PBX).

Extension numbers

MultiSite enabled system always stores subscribers' extension numbers in canonical form. However, you can view and edit these extension numbers in either canonical or switch-native form. You can do this as follows:

- For MSS backend, select the appropriate option in the PBX Extension edit control section.
- For Exchange and Domino backend, select the appropriate option in the Extension edit control section.

The Canonical / Switch Native setting also affects the display and entry of secondary extension numbers.

 Note:

The Canonical / Switch Native setting does not affect the display of the personal operator mailbox number. Although a personal operator mailbox number can be interpreted as an extension number depending upon the configuration of the system, it is primarily intended to be a mailbox number. Hence, it is not converted automatically to canonical form. If you want to ensure that the personal operator for a mailbox is configured to be a phone number and not a mailbox number, then you can enter the number in canonical form regardless of the Canonical / Switch Native setting.

Chapter 4: Phone number translation rules

Why phone number translation rules are used

MultiSite enabled Modular Messaging stores phone numbers in canonical form so that they can be used with any configured PBX. Translation rules are used to convert the phone numbers between switch-native and canonical forms. Administrators must define these rules for each configured PBX.

Translating from switch-native to canonical form

When a PBX notifies a MultiSite enabled MAS of an incoming call, the PBX supplies the associated phone numbers of the caller and the callee in switch-native form. Modular Messaging converts the incoming switch-native number to a canonical number using an ordered set of incoming phone number translation rules. These rules are configured for each PBX.

Table 2 shows some incoming phone number translation rules for the example New Jersey PBX.

Table 2: Incoming phone number translation rules

Description	Match	Output
Local extension	$\wedge(3\d{4})\$$ Match the whole input if it is exactly five digits, and it starts with 3. For example, 31234 $\rightarrow$ +19089531234	+190895\$1 Output +190895 followed by the five captured digits.
National	$\wedge(1\d{10})\$$ Match the whole input if it is exactly 11 digits, and it starts with 1. For example, 13035381234 $\rightarrow$ +13035381234	+\$1 Output the captured digits, but with a leading +.
International	$\wedge011(\d+)\$$	+\$1 Output the captured digits but with a leading +.

Description	Match	Output
	Match leading 011 and capture remaining digits in the input string.	
	For example, 011441895451234 → +441895451234	

Table 3 displays some of the more common regular expression notations that you can use to create translation rules.

Table 3: Useful regular expressions for phone number translation rules

Pattern	Meaning
0123456789	Literal digits.
\d	Any digit.
^	Start of input.
\$	End of input.
.	Any character.
*	Zero or more repetitions of the preceding regular expression.
+	One or more repetitions of the preceding regular expression.
?	Zero or one repetitions of the preceding regular expression.
\	Escape – use before a character that would otherwise have special meaning, for example, \+ means a literal + character.
[...]	Any of the enclosed set of characters, for example, [ab] would match either an “a” or “b” character, but not any other character.
(...)	Group. Matches and captures the enclosed regular expression. The captured characters can be included in the output by using \$1, \$2, and so on. For example, if the input was 123456789 and the matching regular expression was 12(3\d\d)56(789) then \$1 would be 345 and \$2 would be 789.
{m}	Exactly m repetitions of the preceding regular expression, so 12{2} would match 122.
{m, }	m or more repetitions of the preceding regular expression.

Pattern	Meaning
$\{m, n\}$	Between m and n repetitions of the preceding regular expression.

Modular Messaging evaluates the translation rules top to bottom.

Handling phone numbers that do not originate from a PBX

Switch-native numbers do not always originate from PBXs, and you should consider this while designing your translation rules. For example, if you migrated an existing Modular Messaging system to use MultiSite, then subscribers would be likely to have phone numbers stored in switch-native form, for example in Call Me or Find Me rules. These numbers may contain characters that do not typically appear in dial strings from PBXs, for example 9 to get an outside line, or a comma for a pause.

To handle these additional characters, it is helpful to use a different type of incoming phone number translation rule. Most rules produce output that starts with '+'. No further rules are evaluated because the output of the rule is a canonical phone number, and the output is taken as the final output of the rule set. These are called terminating rules.

If the rule produces output that does not start with '+', then it is a transforming rule, and the output becomes the input for the next rule.

Table 4 gives an example of how transforming rules are used.

Table 4: Stripping unwanted characters when translating from switch-native to canonical form

Description	Match	Output
Local extension	$^(3\d{4})\$$ Match the whole input if it is exactly five digits, and it starts with 3.	+190895\$1 Output +190895 followed by the five captured digits. Note that the output is not a regular expression, so + does not have to be escaped.
	For example, 31234 $\rightarrow$ +19089531234	
Strip leading 9	$^9,*(\d*)\$$ Match a leading 9 followed by any number of commas, then capture any following digits.	\$1 Output the captured digits. Note that there is no leading +. The effect is to strip leading 9 plus any following commas.
	For example 9,13035381234 $\rightarrow$ 13035381234	

Description	Match	Output
National	<code>^(1\d{10})\$</code> Match the whole input if it is exactly 11 digits, and it starts with 1. For example, 13035381234 → +13035381234	+\$1 Output the captured digits, but with a leading +.
Strip international prefix	<code>^011(\d*)\$</code> Match a leading 011, then capture any following digits. For example 011441895451234 → 441895451234	\$1 Output the captured digits. Note that there is no leading +. The effect is to strip the initial 011.
International	<code>^([2-9]\d+)\$</code> Match any digit string that does not start with 0 or 1 and is at least two characters long. For example 441895451234 → +441895451234	+\$1 Output the captured digits but with a leading +.

*** Note:**

When you use transforming rules, a single input phone number can match several different translation rules. See *Figure 14* for an example of this.

The rules for Local extension and National numbers are not changed. Strip leading 9 is a new rule that strips the leading 9 and any commas from the incoming phone number. This rule is placed after the *Local extension* rule, because we do not use a leading 9 for internal numbers.

However, this rule has to be before the National rule because otherwise the input would not be matched by the National rule if it had a leading 9. Similarly, the Strip international prefix rule must come after the National rule but before the International rule. The International rule is also modified so that it does not expect the input to start with 011, and replaces that with a very simple check so that it does not match national numbers if anything goes wrong with the National rule.

Translating from canonical to switch-native form

Whenever MultiSite enabled Modular Messaging makes an outgoing call (be it a transfer from the Automated Attendant, a client playing back a message, Call Me inviting a subscriber to log in, or for any other reason), before passing it to the PBX, Modular Messaging must convert any canonical phone number into switch-native form. This is because PBXs cannot deal with canonical phone numbers directly. MultiSite enabled Modular Messaging converts the canonical numbers using a set of outgoing phone number translation rules that are configured

independently for each PBX. These rules are very similar to the incoming phone number translation rules described in the previous section. *Table 5* shows some example outgoing phone number translation rules for the example New Jersey PBX.

Table 5: Annotated example outgoing phone number translation rules for the New Jersey PBX

Description	Match	Output
Local extension	<code>^\+190895(3\d{4})\$</code> Only match numbers that are local to this PBX. Capture the last five digits. For example, +19089531234 → 31234	<code>\$1</code> Output the captured five digits. No additional prefix is required.
National	<code>^\+(1\d{10})\$</code> Match the whole input if it is exactly 11 digits, and it starts with +1. For example, +13035381234 → 913035381234	<code>9\$1</code> Output the captured digits, but strip the + and replace it with 9.
International	<code>^\+(\d+)\$</code> Match any other digit string that starts with +. For example, +441895451234 → 011441895451234	<code>9011\$1</code> Output the code to get an international trunk, then the captured digits.

Outgoing phone number translation rules are very similar to the incoming phone number translation rules, right down to needing rules for the same kinds of phone number (local, national, and international).

How to create and validate translation rules

You can create translation rules using the VMSC application. You typically define an incoming translation rule first and then go on to define a corresponding outgoing translation rule. Enter the test number before defining translation rules so that you can verify the rules while defining translation rules. The color coding on the Translation Rules dialog box allows you to identify if there are errors in the translation rule. If you have not previously configured any phone number translation rules, then Modular Messaging displays the Translation Rules dialog box as shown in *Figure 13*.

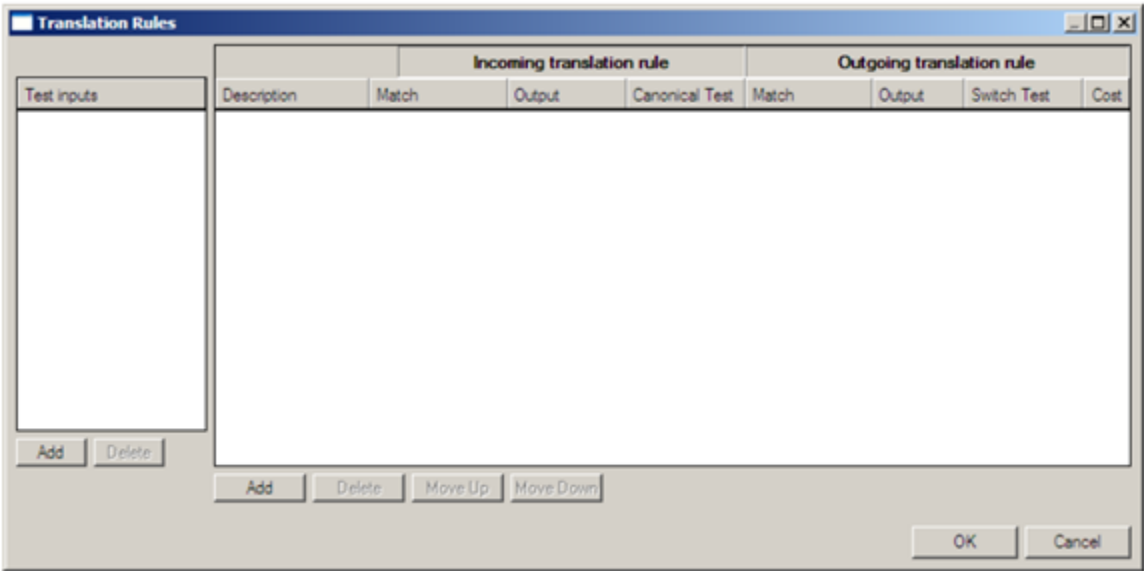


Figure 13: The Translation Rules window

You can see the configured rules in the main part of the window (on the right side), and test data in the left part of the window. *Figure 14* shows how the translation rules for the example New Jersey site appear in VMSC.

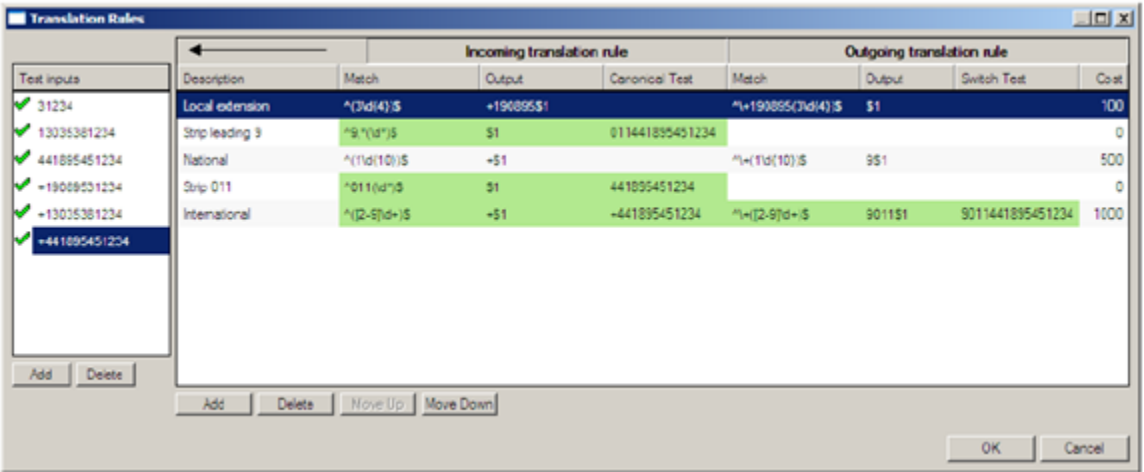


Figure 14: The translation rules for the example New Jersey PBX

If you select an item in the Test inputs pane then you can use that phone number as an input to the translation rules so that you can check whether your rules are behaving as you expect. The behavior depends upon whether the test number is in switch-native or canonical form.



If the test number is switch-native then it is first used as input to the incoming phone



If the test number is canonical then it is first passed into the outgoing phone number

number translation rules. If the number is matched and a canonical number is produced as output, then that number is used as the input to the outgoing phone number translation rules. The arrow at the top-left of the rules pane points to the right, to show that the data is flowing from the incoming rules to the outgoing rules.

translation rules. If the number is matched then the result of the translation is passed into the incoming phone number translation rules. The arrow at the top-left of the rules pane points to the left, to show that the data is flowing from the outgoing rules to the incoming rules.

The test numbers are displayed with one of three icons next to them.

Table 6: How errors in translation rules are indicated

Icon	Background	Description
	 Green	Matching rules were found in both the incoming and outgoing translation rules, and they are highlighted in green. Check that the rules you expected to be matched are the ones that are actually matched. For example, you would expect the same row to be highlighted in both the incoming and outgoing rules.
	 Amber	A matching rule was found in the incoming (for switch-native test numbers) or outgoing (for canonical numbers) rules, and is highlighted in green, but not in the other set of rules. If this is not what you intended then you should check the rule configuration.
	 Red/Pink	The test phone number was not matched by any rules. Either the number is not valid, or the rules are improperly formed.

Regular expression hints

Translation rules use regular expressions to match incoming numbers. Translation rules are very powerful and flexible, and are familiar to many system administrators. Regular expressions are also commonly used in scripting languages like Perl and Python, and in Unix-based operating systems such as Linux. Regular expressions use characters in a slightly uncommon way. There are several different flavors of regular expression, each of which interprets characters slightly differently. MultiSite uses the Perl syntax, which is probably the most common one.

Here are some tips that you can use while working with regular expressions to create translation rules:

- Always use ^ and \$ in match strings. If you do not then your pattern may unintentionally match part of a longer input number.
- Always use \+ in match strings for outgoing rules, because + at the start of a regular expression is an error.
- Use \d, and not . or you may find that the rule matches unintended inputs. The most likely cause of this is if the input number came from a subscriber rather than a PBX, and it contains an authorization code separated from the main number by an ampersand (&).
- Write rules so they match only the expected data; Modular Messaging alerts you of problems because unexpected data are not matched.
- Use the test features.
 - MASs evaluate rules in the same way. If a rule fails in VMSC then you must fix it because it will not work on an MAS either.
 - Use both switch-native and canonical test inputs to reveal gaps in your rules.

Preventing toll fraud

In a Modular Messaging system where MultiSite is not enabled, you can configure to allow only certain types of outgoing calls, based on the initial digit of the dial string. For example, if a PBX uses 9 to get an outside line, it is possible to prevent the system from attempting to dial those numbers. But in a MultiSite configuration, different sites could use different digits to get an outside line. You could restrict 9 for Denver, 8 for New Jersey, and 3 for Uxbridge, but that may cause problems if any of those digits are useful at any of the sites. In the example enterprise, 3 is the initial digit of all New Jersey extensions, and hence restricting a particular number would not work here.

MultiSite has a different way to prevent certain numbers from being dialed. A cost is associated with all outgoing calls. The associated cost is a relative value and it is not meant to be literal. Calls with higher costs are more expensive to make than calls with lower costs, and you are likely to want to control these calls more than the cheaper calls.

The key to making call costs and call cost limits useful in a MultiSite environment is that the cost of any particular call is calculated based upon rules that are defined per PBX. This can be done by associating a cost to the appropriate outgoing phone number translation rule that is higher than the VMD limit for the Maximum cost for Automated Attendant outcalls. In our example enterprise, if the Uxbridge office wants to allow the Automated Attendant to dial only local PBX extensions, you can do so by associating a cost to the appropriate translation rule.

You can associate a cost with each outgoing phone number translation rule. When a canonical phone number is converted into a switch-native number using the phone number translation rules, a corresponding cost is also determined. If the call cost is less than or equal to the limit that you set using the VMSC, then the system dials the number, else the system rejects the

call. Since different outgoing phone number translation rules are configured for each PBX, it is easy to prevent dialing of only those numbers that are inappropriate for a specific PBX.

Maximum costs are defined for the whole VMD. *Figure 15* shows how this is done using VMSC.

- The Maximum cost for Automated Attendant outcalls setting limits calls made on behalf of callers; since they have not been authenticated as subscribers you probably want to set a lower value.
- The Maximum cost for subscriber outcalls setting limits calls made on behalf of subscribers; since they are trusted users of MM you may want to set a higher limit for them.

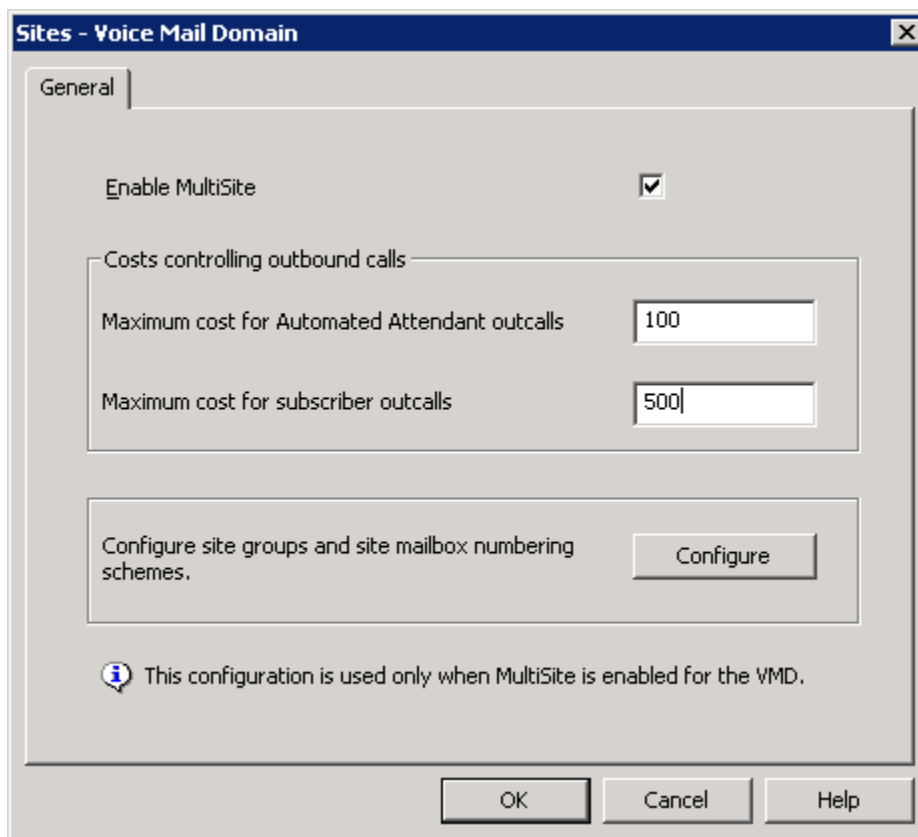


Figure 15: Setting call cost

In *Figure 14* note the last test number that is highlighted. You can see that it is matched by the International rule and the resulting cost is 1,000. But, according to the limits shown in *Figure 15*, this would be too expensive to dial, irrespective of whether this is an Automated Attendant or subscriber outcall.

How phone numbers entered by subscribers are handled

New calls made for a subscriber, such as Call Me outcalls, or sending faxes are made using the PBX associated with the home site of the subscriber. However, if a transfer is made for the subscriber (for example, if subscribers do *Reply by Calling Sender* while logged on to a TUI) then the transfer is made using the same PBX that is currently handling the call.

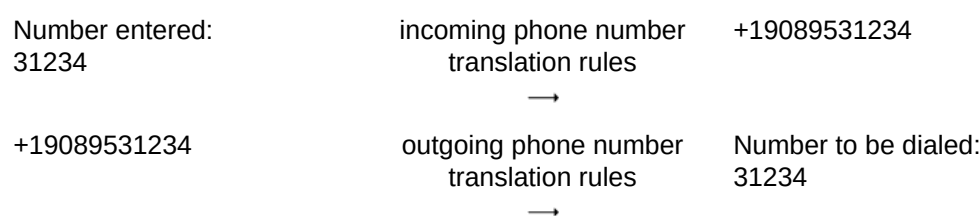
Subscribers can use phone numbers in either switch-native or canonical form.

Modular Messaging determines how to dial a phone number by checking whether it starts with a '+'.

- If the number starts with a '+' then it is canonical, and Modular Messaging converts the number to the native form of the PBX being used to make the call by using the outgoing phone number translation rules that are configured for that PBX.
- If the number does not start with '+' then it is in switch-native form. The number is assumed to be in the native form of the PBX that is associated with the subscriber's home site. Modular Messaging converts the number to canonical form using the incoming phone number translation rules of that PBX. Modular Messaging then converts the canonical number to the native form of the PBX that will be used to make the call using the outgoing phone number translation rules configured for that PBX.

Table 7 shows a simple case where a subscriber in the example New Jersey site is using the New Jersey PBX and enters a switch-native phone number.

Table 7: Translating a switch-native phone number entered at a client (simple case)



You can see that the number to be dialed is the same as the number that was originally entered, but there are two reasons for using translation rules.

- While producing the number to dial, the outgoing phone number translation rule also determines a call cost, which might prevent the call from being completed. The cost is taken into account for all outcalls, including the ones that are not directly requested by subscribers, such as Find Me and Call Me.
- When the subscriber is roaming away from their home PBX then incoming and outgoing translation rules from different PBXs will be applied. Table 8 shows the translations that would take place if our example New Jersey subscriber was traveling to the UK, and was using the Uxbridge PBX.

Table 8: Translating a switch-native phone number entered at a client (roaming)

Number entered: 31234	incoming phone number translation using New Jersey PBX's rules	+19089531234
→		
+19089531234	outgoing phone number translation using Uxbridge PBX's rules	Number to be dialed: 90019089531234
→		

Although the subscriber still entered the number in a format that was familiar to them, you can see that the Uxbridge PBX translation rules have produced a completely different number to dial.

This incoming and outgoing translation is also useful in other cases, for example:

- Replying by calling the sender with a message that was received before MultiSite was enabled, so the caller's number is not in canonical form.
- Using a TUI to set the personal operator to an extension number, and not a mailbox. Subscribers cannot enter canonical numbers using the TUIs because there is no '+' key on the keypad. So, they must enter switch-native numbers.

Non-DID extensions

Most enterprise employees have DID extensions so that they can be dialed directly from the public phone network. All DID extensions have a natural canonical, E.164 representation.

However, some enterprise employees might have non-DID extension numbers, that can be dialed only from within the enterprise, and not from outside the enterprise. The advantage of canonical phone numbers is that they can be converted into a form that can be dialed from any PBX. This cannot be true of non-DID numbers. We need a way to encode a non-DID phone number in something that looks like canonical form, but that is identifiable so that no attempt will be made to dial it on anything other than the relevant PBX.

Our example New Jersey site has DID extensions that start with 3. If the site also had non-DID extensions that start with 5, then the Uxbridge PBX would not be able to dial those extensions directly.

Avaya recommends use of the reserved country code of zero to handle non-DID phone numbers. Country code zero does not clash with any legal canonical number because it is reserved. When defining translation rules to handle non-DID phone numbers, insert a "0" between the "+" and the rest of the E.164 phone number, as shown in *Figure 16*.

DID: 31234 → +19089531234

Non-DID: 51234 → +019089531234

Figure 16: Use country code 0 to indicate a non-DID phone number

This results in a number with the structure shown in *Figure 17*.

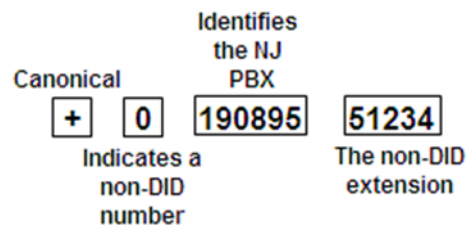


Figure 17: The structure of an encoded non-DID phone number

You should include a PBX identifier in the number to allow the translation rules for different PBXs to translate only those numbers that they understand. For example, the Uxbridge PBX would not try to translate a non-DID number from the New Jersey PBX.

Table 9 and *Table 10* show how you can configure the translation rules, for the example New Jersey PBX, to allow both DID and non-DID extensions.

Table 9: Example showing incoming translation rules for handling non-DID extensions

Description	Match	Output
Local non-DID extensions	$^(5\d{4})\$$ Match the whole input if it is exactly five digits, and it starts with 5. For example, 51234 → +019089551234	+0190895\$1 Include a 0 after the + to indicate a non-DID extension.
Local DID extensions	$^(3\d{4})\$$ Match the whole input if it is exactly five digits, and it starts with 3. For example, 31234 → +19089531234	+190895\$1 Output +190895 followed by the five matched digits.
Strip leading 9	...	...
National	...	...
Strip international prefix	...	...
Reject other non-DID extensions	There is no match or output for the incoming portion of this rule.	
International	...	...

Table 10: Example showing outgoing translation rules for handling non-DID extensions

Description	Match	Output
Local non-DID extensions	<code>^\+0190895(5\d{4})\$</code> Match only 5-digit extensions starting with 5, for this PBX. For example, . +019089551234 → 51234	\$1 A non-DID extension can be dialed directly on the New Jersey PBX.
Local DID extensions	<code>^\+190895(3\d{4})\$</code> Match only 5-digit extensions starting with 3. For example, . +19089531234 → 31234	\$1 DID extensions can be dialed directly.
Strip leading 9	...	...
National	...	...
Strip international prefix	...	...
Reject other non-DID extensions	<code>^\+0.*\$</code> Match any input that starts with +0.	Empty – means that the number cannot be dialed.
International	...	...

The order of these rules is important.

- **Local non-DID extensions:** This incoming rule matches internal, non-DID extensions, and converts them to canonical numbers with a country code of zero. A corresponding outgoing rule translates those numbers back again for dialing.
- **Local DID extensions:** This rule is very similar, but is a regular rule that matches internal DID extensions, and converts them to regular canonical numbers.
- **Reject other non-DID extensions:** There is no incoming match for this rule. The PBX generates numbers in its own format, which is covered by the other rules. The outgoing portion matches any canonical number with a country code of zero, which means any non-DID numbers from any other PBX (because this rule comes after the rule for Local non-DID extensions). Because the output part of the outgoing rule is blank, no switch-native number will be output from the rule, so Modular Messaging will not be able to dial the number.
- **International:** This is just a standard rule, but it is important that it comes after the *Reject other non-DID extensions* rule, otherwise it might attempt to dial the non-DID numbers from other PBXs, depending on exactly how it was defined.

Proprietary dial plans

Some companies have a proprietary, or custom, dial plan, used for internal calls. *Table 11* shows an example proprietary dial plan where locations have a 4-digit location code that always starts with 7, and 4-digit extensions, giving an 8-digit number.

Table 11: Example proprietary dial plan

Chicago	7312xxxx
New York	7212xxxx
London	7224xxxx

These numbers can be represented using the +0 country code, similar to the scheme used for handling non-DID extensions. The main difference is that all PBXs would need similar rules to handle the +0 numbers for proprietary dial plans.

Table 12 and *Table 13* show examples of how translation rules can allow the use of proprietary numbering schemes. Each PBX needs a rule to handle its own 4-digit extensions. Here, "Internal (Chicago)" matches a 4-digit extension and converts it to an 8-digit number encoded with country code 0, so that it can be dialed from any PBX in the enterprise. The rule to handle the general 8-digit extensions is common to all PBXs in the enterprise.

If there were also typical rules like National and International then it would also be possible to use standard phone numbers as well as the proprietary dial plan.

Table 12: Example showing incoming translation rules for handling proprietary extensions

Description	Match	Output
Internal (Chicago)	$\wedge(\backslash d\{4\})\$$ Match the whole input if it is exactly four digits.	+07312\$1 Include a 0 after the + to indicate a proprietary internal extension.
	For example, 1234 $\rightarrow$ +073121234	
Internal	$\wedge(7\backslash d\{7\})\$$ Match an 8-digit number, if it starts with 7.	+0\$1 Output +0 followed by the matched digits.
	For example, 72241234 $\rightarrow$ +072241234	

Table 13: Example showing outgoing translation rules for handling proprietary extensions

Description	Match	Output
Internal (Chicago)	<code>^\+07312(\d{4})\$</code>	\$1 A 4-digit extension can be dialed directly on the Chicago PBX.
	For example, +073121234 → 1234	
Internal	<code>^\+0(7\d{7})\$</code>	\$1 8-digit proprietary numbers can be dialed directly.
	For example, +072241234 → 72241234	

PBX authorization codes

Phone numbers used in Find Me rules can include a PBX authorization code, which is separated from the number by an ampersand. This code number must be sent to the PBX before certain types of outcalls can be allowed. The following rules show how to configure phone number translation rules so that authorization codes get passed through to the PBX.

Table 14: Example of how to deal with PBX authorization codes in incoming translation rules

Description	Match	Output
International	<code>^([2-9]\d+)(&\d+)?\$</code> Match two groups: the first is the phone number, and the optional second group is the authorization code, including the &.	+\$1\$2 Output the phone number and the authorization code.
	For example, 441895451234&654321 → +441895451234&654321	

Table 15: Example of how to deal with PBX authorization codes in outgoing translation rules

Description	Match	Output
International	<code>^\+(\d+)(&\d+)?\$</code> Match two groups: the first is the phone number, and the optional second group is the	9011\$1\$2 Output the code to get an international trunk, then the

Description	Match	Output
	authorization code, including the &	phone number then the authorization code.
	For example, +441895451234&654321→ 9011441895451234&654321	

A PBX that did not use authorization codes could cope with canonical numbers that contained them by simply missing out the “\$2” in the outgoing rule, so that the authorization code would never be sent to the PBX.

Preventing access to particular phone numbers

You can create outgoing phone number translation rules to prevent certain phone numbers from being dialed. For example, you might decide that nobody should be able to use Modular Messaging to dial premium rate phone numbers.

To do this you must create an outgoing phone number translation rule that matches the kind of phone number that you want to ban, as shown in *Table 16*.

Table 16: Annotated example showing outgoing phone number translation rules for the New Jersey PBX

Description	Match	Output	Cost
Local extension	...	...	...
Block 1900	^\+(1900\d{7})\$ Match national numbers that start with 1900.	9\$1 Output the correct number	2000 Higher than the configured maximum call costs to ensure that the call will not be allowed.
National	...	...	...
International	...	...	...

Note that the *Block 1900* rule comes before the *National* rule, which would otherwise match all 1900 phone numbers. The high call cost prevents the call from being made. It would have been possible, instead, to make the *Output* part of the rule blank; because the rule would produce no output, the phone number could not be dialed. That is not recommended though, because the fact that the translation rule failed may cause warnings to be logged. As you are intentionally banning these numbers, rather than introducing an accidental error in the translation rules, it is better to use a high call cost to prevent the call from being made.

Chapter 5: Installation

Plan the telephony

Determine required PBXs

Find out about the required PBXs that are going to be part of the MultiSite system.

- Find out if any more ports are required.
- Find out whether the PBX has to be upgraded to be able to integrate with the gateway, for example, does it need a new Q.SIG trunk, or updated firmware?

 Note:

When MultiSite is being used, only SIP integration is supported.

For more information, see [MultiSite sizing](#) on page 63 and [IP network requirements](#) on page 65.

Determine SIP gateway and proxies

Modular Messaging release 5.0 and later uses SIP gateway to provide PBX connections. Decide on SIP gateways and ports requirement for each PBX before you enable MultiSite.

- Ensure that there is an SES proxy server or a SIP gateway for each PBX.
 - Use an SES, if it is already in place.
 - If there are non-Avaya PBXs, then use SIP gateways.
 - If there are any Avaya PBXs, which do not already have an SES then decide whether to use an SES, or a SIP gateway. Either would allow MultiSite to work, but there may be other reasons why one or the other would be most appropriate for this installation.
- If you are going to use SIP gateways then determine how these gateways are going to be integrated to the PBXs. If SIP or Q.SIG integrations are available for that PBX then

one of those should be used in preference to analog. Make sure SIP gateways have the correct firmware installed.

- Determine how many ports are required for each PBX. For more information on calculating ports requirement, see the Concepts and Planning Guide. This affects how many ports will have to be made available on the PBX and, if a SIP gateway is being used, the capacity that must be ordered. Make sure that the Modular Messaging system has enough SIP ports to handle all of the PBX and gateway ports.
- Ensure that you have the relevant configuration notes from:
 - Avaya (<http://support.avaya.com/main/supportcentral/supportcentral.do>).
 - AudioCodes (<http://www.audiocodes.com/avaya>).
- Each proxy server or gateway will require a static IP address; ensure that they will be available.
- Ensure that the corporate IP network can cope with the added load that IP telephony will place upon it.

For more information, see [MultiSite sizing](#) on page 63 and [IP network requirements](#) on page 65.

Plan the IP network

The bandwidth and latency of the WAN between the SIP gateways and central MAS must be sufficient to cope with the number of simultaneous SIP ports that are expected to be used, allowing sufficient overhead for other WAN traffic.

For more information, see [IP network requirements](#) on page 65.

Adding PBXs using VMSC

Before you begin

A Voice Mail Domain is created.

Procedure

1. Configure SIP using the **VMD PBX Integration** node.

 Note:

This node replaces the MAS PBX Integration node when SIP is being used. If you intend to allow the use of SIP (as opposed to secure SIP) then be sure to

check the Enable check box, or the MASs will only listen for secure SIP connections, and refuse standard SIP connections.

2. To create a PBX, right-click the **VMD PBXs** node and click **Add New PBX**.

*** Note:**

When MultiSite is being used, only SIP integration is supported.

3. Enter a descriptive name for this PBX on the **General** tab, in the **PBX Name** field.

Configuring PBXs Using VMSC

Before you begin

- A voice mail domain is created.
- SIP PBXs are created.

About this task

*** Note:**

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click the voice mail domain (VMD).
2. Double-click **PBXs**. The node expands to show all available types of PBXs.
3. Double-click the **PBX** that you want to configure. The system displays the **PBX Configuration** dialog box for the selected PBX, with the **General** tab selected.
4. Select the **SIP** tab.
5. Click the **yellow cross** button to add a new gateway.
6. Enter the IP address or fully qualified domain name of the gateway.

*** Note:**

Ensure that the first check box is checked; otherwise the gateway will not be used.

7. Select either TLS or TCP for Protocol. Ensure that the settings are done as explained in the config note. Avaya recommends the TLS protocol for security. If the TLS protocol is not used, ensure that the SIP traffic is secure, for example you can use a VPN between the MASs and the gateways to ensure security.

*** Note:**

TCP will be used only if it was enabled in the VMD PBX Integration dialog.

8. Check the MWI check box unless the integration note specifies otherwise.
9. Select the same security level that you have configured on the gateway for SRTP. PBX uses specific encryption methods in its settings. "Low" is equivalent to "AES_CM_128_HMAC_SHA1_32", and "High" is equivalent to "AES_CM_128_HMAC_SHA1_80".
10. Enter the SIP domain of the proxy server for the SIP domain.

*** Note:**

P-Asserted-Identity is only relevant for SES proxy servers, and is only relevant if the PBX has been configured to require it in order to grant appropriate permissions to Modular Messaging for outcalling. It is an extension number with an optional domain, like 4999 or 4999@example.com.

11. If you are using several Avaya PBXs behind a single SES, then enter a PBX address.

Configure the mailbox numbering scheme and sites

Creating site groups

About this task

*** Note:**

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click **Voice Mail Domain (VMD)**.

*** Note:**

A VMD can contain up to 500 sites and site groups.

2. Double-click **Sites**. The **Sites** dialog box for the selected voice mail domain opens.
 3. Click **Configure**. The **Site Configuration window** dialog box opens.
 4. Select **Group** from the **Add** list or right-click a site group and select **Add Site Group**. The **New Site Group** dialog box opens.
 5. Select a **Parent Site Group**; specify a name and identifier for the Site Group. You can save the Site Group without an identifier by selecting the **Group container only** check box.
 6. Click **Add**. The new site group appears in the Site/group tree.
- For more information, see [Mailbox numbers and sites](#) on page 17.
-

Creating a site

About this task

* Note:

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Site Configuration** dialog box, select **Site** from the **Add** list or right-click a site group and select **Add Site**. The **New Site** dialog box opens.

* Note:

A VMD can contain up to 500 sites and site groups.

2. From the **Parent Site Group** list, select a site group for the new site.
Sites are arranged in a tree-like hierarchy in the VMSC. You can set the *Parent site group* to control where in the hierarchy the new site will appear.
3. Specify a name for the site.
Modular Messaging uses the site name to identify the site within the VMD. The site name may be announced to subscribers and callers, so Avaya recommends you to enter a site name that is short and clear.
4. Specify the identifier for the new site.
The site identifier, full mailbox length and the short mailbox length are interrelated, so a preview is displayed that allows you to see the effect of any change that you make.

 Caution:

You must choose the site IDs carefully; you cannot change the site ID after you create a site. If you need to change a site ID, then you must delete the site and then add a new site with the required identifier and other details.

5. Specify the length of the full and short mailbox numbers.
6. From the **PBX** drop-down, select the PBX that handles the incoming and outgoing calls for the site.
7. Click **Add**. The new site appears in the Site/group tree.
For more information, see [Mailbox numbers and sites](#) on page 17.

Deleting a site or a site group

About this task

 Note:

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click **Voice Mail Domain** (VMD).
2. Double-click **Sites**. The system displays the **Sites** dialog box for the selected voice mail domain.
3. Click **Configure**. The **Site Configuration** dialog box opens.
4. Select the site or site group that you want to delete and click **Delete**.

 Caution:

Subscribers are associated with sites by way of their mailbox number. You should be careful to ensure that you are not going to leave a group of subscribers without a site, which would reduce the level of service that could be provided to them. Nevertheless, if an administrator makes this error then the next time that one of the affected subscribers attempts to log in to a TUI, an event will be created on the MAS that handles the call. The MSS audit will also identify affected mailboxes.

Configuring the site properties

About this task

* Note:

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click **Voice Mail Domain** (VMD).
2. Double-click **Sites**. The system displays the **Sites** dialog box for the selected voice mail domain.
3. Click **Configure**. The system displays the **Site Configuration** dialog box.
4. Select the Site for which you want to view or modify the properties and click **Properties**. The **Site Properties** dialog box opens.

Refer to *Table 17* for details of which properties are on each tab of the **Site Properties** dialog box.

Table 17: Site properties

Page	Property	Description
General	Primary language	The prompt language that will be used for callers to this site.
	Time zone	The time zone that should be used for this site.
	Hunt group	The canonical hunt group number that callers dial for this site.
TUI	Receptionist mailbox (business hours, out of hours)	The mailbox or phone number to call if the caller requests to speak with a receptionist.
	Greeting start times (morning, afternoon, evening)	When to use appropriate greetings.
	Business hours (Sunday – Saturday)	Each day can be configured to be open all day, closed all day, or open between two times.
Auto Attendant	Day greetings (main menu, morning,	The number of the custom greeting to use for the different times of day.

Page	Property	Description
	afternoon, evening, closed)	
	Multilingual greetings	The number of the custom greeting to use when announcing the prompt languages that are available on the system.
	Languages	Configures up to three languages, each with an associated key, and a custom prompt number.
Auto Attendant Keys	Keys 1-9	Configure each key to be invalid, directory, message, or transfer to a specified mailbox.
Auto Attendant Holidays	Holidays	Configure a number of holidays (DD/MM) and the associated prompt number to play.

Recording a name for the Site

About this task

* Note:

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click **Voice Mail Domain** (VMD).
2. Double-click **Sites**. The system displays the **Sites** dialog box for the selected voice mail domain.
3. Click **Configure**. The system displays the **Site Configuration** dialog box.
4. Click  corresponding to the site for which you want to record a site name. The Voice Recorder opens. To configure whether to use multimedia or telephony while recording site names, click **Tools > Options** in the **Site Configuration** dialog box. In the **Options** dialog box, select the desired media for recording.
5. Record the name for the site. After you save the recording, the image on the Site Configuration dialog changes to  indicate that a recording exists.

*** Note:**

If you do not record a name, then Modular Messaging performs text-to-speech (TTS) of the site name that you specified in the Site Configuration dialog box.

Plan the phone number translation rules

Plan the translation rules that will be required for each PBX (see Chapter 4). Determine whether there are any exceptions to the E.164 numbering plan, for example non-DID numbers or a proprietary numbering plan.

Creating translation rules

About this task

This topic describes procedures for creating translation rules. For more information, see [Chapter 4: phone number translation rules](#), on page 25

*** Note:**

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click **Voice Mail Domain** (VMD).
2. Double-click **PBXs**. The system expands the node to show all available types of PBX.
3. Double-click the **PBX** that you want to configure. The system displays the **PBX Configuration** dialog box for the selected PBX, with the **General** tab selected.
4. Click the **SIP** tab.
5. Click **Configure** to open the **Translation Rules** dialog box.
6. Click **Add** under the translation rules section, to add a new rule.
7. Enter the description for the rule.
8. To move to the different fields of the rule, click in them, or press **Enter** to finish entering the current field, then use the left and right arrow keys to move between

the fields. Again, press **Enter** to start editing when you reach the desired field. The Canonical Test and Switch Test fields cannot be edited.

9. In the Match column, enter a regular expression that matches the numbers that you want to convert.
10. In the Output column, enter the string that should be output, if a match is found for the number.
11. Repeat steps 9 and 10 if you want to create an outgoing translation rule for this incoming translation rule.

*** Note:**

You can create a rule that has only an incoming match, and no outgoing match, or an outgoing match with no incoming match.

12. Repeat steps 6 through 11 to create additional rules.
13. Click **Add** under the **Test Inputs** section to add a new phone number to test the rules with.
14. Enter the number that you want to test against a rule.
The color coding in the **Translation Rules** dialog box indicates if you have defined the rule correctly. Modular Messaging also displays the converted number in the **Canonical Test** field. For more information, see [How to create and validate translation rules](#) on page 29.
15. If you want to change the order of the rules, select the rule that you want to reorder and click the **Move Up** or **Move Down** button.
16. Click **OK** when you have created and tested all the translation rules for this VMD.

*** Note:**

It is not possible to copy all of the rules from one PBX to another PBX. If you have rules that are common across several PBXs (like the national and international rules for PBXs in the US) then you can copy the rules into a text file so that you can copy and paste them while configuring a new PBX.

For more information, see [Chapter 4: phone number translation rules](#) on page 25.

Setting Call Cost Limits

About this task

*** Note:**

To perform this task, you must be a member of a security role assigned the Telephony - Administer task. If you are a member of a role assigned the Telephony - View task, you have

read-only access to this dialog box. See the Security Roles Dialog Box topic of the *Messaging Application Server (MAS) Administration Guide*.

Procedure

1. In the **Voice Mail System Configuration** window, click the **Voice Mail Domain (VMD)**.
2. Double-click **PBXs**. The system expands the node to show all available types of PBX.
3. Double-click the **PBX** that you want to configure. The system displays the **PBX Configuration** dialog box for the selected PBX, with the **General** tab selected.
4. Click the **IP SIP** tab.
5. Click **Configure** to open the **Translation Rules** dialog box.
6. In the Cost column, enter the call cost for each outgoing rule that you have defined.

 Note:

Call cost is a relative value and it is not meant to be literal. Calls with higher costs are more expensive to make than calls with lower costs. Typically you would assign a lower cost for local numbers, higher cost for national numbers, and an even higher cost for international numbers.

7. Click **OK** to save the call costs.
8. In the **Voice Mail System Configuration** window, expand the VMD and double-click **Sites**. The system displays the **Sites** dialog box for the selected voice mail domain.
9. Enter the maximum cost for outcalls that can be made by an automated attendant. The default value is 100.

 Note:

These are calls made on behalf of callers; since they have not been authenticated as subscribers you probably want to set a lower value for this.

10. Enter the maximum cost for outcalls that can be made by a subscriber. The default value is 100.

 Note:

These are calls made on behalf of subscribers; because they are trusted users of Modular Messaging you may want to set a higher limit for them.

11. Click **OK** to save the call cost limits and close the **Sites** dialog box.
-

Install a new MultiSite system

The DCT allows the specification of only a single PBX. If you are creating multiple sites, then all sites must be configured to use the same PBX until after the initial installation is complete. If you are going to have more than one PBX then you need to record the details (in the DCT Notes field) and add them later using VMSC, or follow the path of installing the system as non-MultiSite, and configure and enable MultiSite after installation using VMSC.

Configuring MultiSite using the DCT

You can use the Data Collection Tool to enable MultiSite, for a new installation of Modular Messaging. This topic gives the specific tasks that you must perform in the DCT to enable MultiSite for a VMD. For more details about each of these tasks, refer to the Data Collection Tool online help.

About this task

 Note:

If you are upgrading from a previous version of Modular Messaging to current version of Modular Messaging, the DCT cannot be used to enable MultiSite.

Procedure

1. Select the **Enable MultiSite** option on the **Voice Mail Domain** page.
2. Verify that Avaya SIP is selected in the list of integration types on the **Switch integration method** page.
3. Click **Configure** on the **Switch integration information** page to configure the phone number translation rules for the PBX that the DCT allows you to configure.
4. Enter a PBX Name on the Switch integration information page.

 Note:

If the system will ultimately include more than one PBX then you can use the Notes field to record some details of the other PBXs.

5. Click **Configure** on the **Dial plan page** to configure the sites for this VMD. You do not need to enter any other settings on this page.
6. To save or print the DCT file at any time during data entry, click **Save** or **Print**.
7. When you are ready, click **Complete** and follow the instructions to save changes. Save the DCT file to an external USB storage device.

⚠ Caution:

Do not save the file to the MAS; it can be erased during installation of the Avaya boot image.

8. Complete the other DCT pages and install the system as normal.
-

Configuring MultiSite using VMSC after installation

Enabling MultiSite

If you did not completely configure MultiSite using the DCT during installation then you have to configure the required PBXs and sites using VMSC, following these steps:

Procedure

1. Configure SIP using the **VMD PBX Integration** node.

*** Note:**

This node replaces the MAS PBX Integration node when SIP is being used. If you intend to allow the use of SIP (as opposed to secure SIP) then be sure to check the Enable check box, or the MASs will listen only for secure SIP connections, and refuse SIP connections.

2. Right-click the VMD **PBXs** node and click **Add New PBX** to create require PBXs.

*** Note:**

When MultiSite is being used, only SIP integration is supported.

3. Enter a descriptive name for this PBX on the **General** tab, in the **PBX Name** field.
For more information, see [Configuring SIP switches](#) on page 42.
4. Configure the SIP gateways or proxies for this PBX on the **SIP** tab.
For more information, see [Configuring SIP gateways](#) on page 43
5. Configure the PBX translation rules.
For more information, see
 - [Phone number translation rules](#) on page 25
 - [Creating translation rules](#) on page 49
6. Configure the sites.
For more information, see
 - [Mailbox numbers and sites](#) on page 17

- [Creating site groups](#) on page 44
 - [Creating a site](#) on page 45
 - [Configuring the site properties](#) on page 47
 - [Recording a name for the Site](#) on page 48
7. Create a role that has capabilities to enable MultiSite.
 - a. Expand the VMD **Security Roles** node.
 - b. Right-click the System Administrator role and choose **Copy Role...**
 - c. Enter a role name, such as “Enable MultiSite”.
 - d. Click **OK**.
 - e. Double-click your newly created role.
 - f. Click **Add...** on the Members tab.
 - g. Enter the name of the user that you are currently logged in as, then click **Check Names**, then click **OK**.
 - h. Click **Add** on the Tasks tab.
 - i. Select **MultiSite – Enable**.
 - j. Click **OK**.
 8. Double-click VMD Sites node, then select the Enable MultiSite check box, then click **OK**.
 9. Delete the role that you created to enable MultiSite, which is also capable of disabling MultiSite.
 10. Restart all the MASs in the VMD. MultiSite cannot work properly until this step has been performed.
-

Disabling MultiSite

If you have enabled MultiSite by accident you can disable it using the following steps.

Procedure

1. Create a role that has capabilities to disable MultiSite.
 - a. Expand the VMD **Security Roles** node.
 - b. Right-click the System Administrator role and choose **Copy Role...**
 - c. Enter a role name, such as “Disable MultiSite”.
 - d. Click **OK**.
 - e. Double-click your newly created role.
 - f. Click **Add...** on the Members tab.
 - g. Enter the name of the user that you are currently logged in as, then click **Check Names**, then click **OK**.
 - h. Click **Add** on the Tasks tab.

- i. Select **MultiSite – Enable**.
 - j. Click **OK**.
2. Double-click the VMD Sites node, then deselect the Enable MultiSite check box, then click **OK**.
 3. Delete the role that you created to disable MultiSite.
 4. Restart all the MASs in the VMD. MultiSite will not be properly disabled until this step has been performed.

*** Note:**

If you have an operational system using MultiSite then think carefully before deciding to turn off MultiSite. Bear in mind the following points about Modular Messaging systems that do not have MultiSite enabled:

- You can use only a single PBX in a VMD.
- You have to use a fixed mailbox number length of 10 digits or fewer, which will probably involve changing everybody's mailbox number.
- You have to change all extension numbers to be in the native form of the VMD's PBX. Canonical numbers are not supported with a non-MultiSite system.
- Subscribers cannot log in from non-home locations.

Migrations

Migrating a single Modular Messaging VMD to MultiSite

Before you begin

- Ensure that you have a recent backup of the Modular Messaging system before proceeding.
- Ensure that all MASs in the VMD have release 5.0 or later installed.
- Ensure that you have imported a most recent Modular Messaging license so that the system has a current SIP certificate (contained within the license).
- Ensure that you have installed and configured either an SES proxy server or SIP gateway.

Procedure

1. Create a SIP PBX in VMSC, and configure it for your gateway.

2. Create phone number translation rules for your PBX.
3. Configure sites.
4. Use FEDBQuery to save details of the existing mailboxes to a CSV file. For example, from a Command Prompt window, type:
`C:\Avaya_Support\Tools\FEDBQuery>FEDBQuery.exe`
Specify 1 for the Query Type, and then enter the name of the file to save, for example `c:\MMSubscribers.csv`
To view the online help for FEDBQuery and VMEnable, open `C:\Avaya_Support\Tools\vmenable\VMEnable409.chm`.
5. Use Microsoft Excel or your favorite text editor to change the mailbox numbers according to your new mailbox numbering scheme. Change the extension numbers to the canonical form. Save the result back to a CSV file. Do the following to modify the CSV file using Microsoft Excel.
 - a. Open the CSV file in Microsoft Excel.
 - b. Right-click the B column (numeric address) and select **Format Cells**.
 - c. On the **Number** tab, select **Custom** from the **Category** list.
 - d. In the **Type** box, enter "130358"0. Ensure that you enter the quotes and replace the digits inside the quotes with digits that are appropriate for your system. This changes the way the numbers are displayed, but does not change the underlying data in the spreadsheet.
 - e. Right-click the C column (extension numbers) and select **Format Cells**.
 - f. On the **Number** tab, select **Custom** from the **Category** list.
 - g. In the **Type** box, enter "+1303538"0, replacing the digits inside the quotes with digits that are appropriate for your system.
 - h. Select the E column (mailbox numbers) and then right-click and select **Format Cells**.
 - i. On the **Number** tab, select **Custom** from the **Category** list.
 - j. In the **Type** edit box, enter "1303538"0, replacing the digits inside the quotes with digits that are appropriate for your system.
 - k. Select the M column (personal operator) and then right-click and select **Format Cells**.
 - l. On the **Number** tab, select **Custom** from the **Category** list.
 - m. In the **Type** edit box, enter "+1303538"0, replacing the digits inside the quotes with digits that are appropriate for your system.
 - n. Change the entry in the last column (ignore row) from 1 to 0 in the FEDBQuery output so it will process each row in the spreadsheet.
 - o. Save the spreadsheet in .xls format. Close the file and then reopen and save in .csv format.
 - p. Open the .csv file in a text editor and verify if all the changes are in correct format.

*** Note:**

From this point until the process is completed, your Modular Messaging system will not be available to subscribers. So, you must perform these steps outside of office hours.

6. Change all MASs to use SIP integration. The number of ports available on each MAS should remain 48 unless you have a particular reason to reduce it. For more details, see *Modular Messaging MAS Administration Guide (PBX Integration Dialog Box section)*.
7. Remove any non-SIP PBXs that are configured for the VMD.
8. Restart all MASs in the VMD.
9. For each MAS, after the system has rebooted and the MAS service is running, run `C:\Avaya_Support\Tools\ConfigureMMIPSEC\ConfigureMMIPSEC.exe` from a Command Prompt window.
10. Test that SIP integration to the proxy or gateway works.
11. Enable MultiSite.
For more information, see [Enabling MultiSite](#) on page 53.
12. Shut down all MASs.
13. Remove any Dialogic boards from the MASs.
14. Turn on all MASs.
15. Use Message Store Web Admin to update a single subscriber's mailbox number (to ensure that they are a member of a site) and extension number (to ensure that it is in canonical form), and verify that it works.
16. Run VMEnable with your updated CSV file, to update the mailbox and extension numbers of all the other mailboxes. For example, from a Command Prompt windows, type the following, replacing "mymas" with the hostname of your MAS:
`C:\Avaya_Support\Tools\vmenable>vmenable /i c:
\MMSubscribers2.csv /vs mymas /id /ip`

*** Note:**

Type the command in single line.

17. If any failures occur during the migration process then make sure that the translation rules are correct.
-

Migrating several Modular Messaging VMDs to MultiSite

If you want to migrate several Modular Messaging VMDs to a single MultiSite system then you should plan the migration carefully. Avaya can help you plan a migration. Following are the steps that you must follow for this migration scenario:

Procedure

1. Determine the size of the MultiSite system in terms of number of ports, network bandwidth, and number of MASSs.
 2. Decide whether you want to install new MASSs, or move them from the existing VMDs as they are migrated to the MultiSite system.
 3. Identify the VMD that you want to migrate first and follow the steps described in [Migrating a single MM VMD to MultiSite](#) on page 55.
 4. Migrate each additional VMD to the MultiSite system. Ensure that subscribers services are working before you continue.
-

Migrating legacy voice mail systems to MultiSite

If you want to migrate legacy voice mail systems to a single MultiSite system then you should plan the migration carefully. Avaya can help you plan a migration. Following are the steps that you must follow for this migration scenario:

Procedure

1. Determine if the MultiSite enabled Modular Messaging system will have to coexist with the old voice mail system for some time. If yes, then you must install a Message Networking system to allow the old system and Modular Messaging to interoperate.
 2. Plan your mailbox numbering scheme:
 - a. Determine if you want mailbox numbers to be exactly the same as on the old system, or do you want to take this opportunity to move to a better scheme, for example, to make the mailbox numbers the same as the extension numbers.
 - b. Determine if the new system has to fit into an existing voice messaging network.
-

Update clients

- If there is only a single site, then any Modular Messaging client that is installed will continue to work.
- If there is more than one site, then you must upgrade all of the clients to Modular Messaging Release 5.0 or later.

Add mailboxes and test the system

After you have successfully installed or enabled MultiSite, Avaya recommends that you create some mailboxes and test the system to ensure that the following basic operations are working:

- Call answering
- Subscriber login
- Subscriber sending and receiving messages
- Automated Attendant, including transfers
- Message Waiting Indicator
- Call Me, Find Me, Notify Me
- Fax: sending and receiving

For more details, see *Modular Messaging Release Subscriber Options User Guide* or *Modular Messaging Release Web Subscriber Options User Guide*.

Chapter 6: Voice networking

Networking with Message Networking

Message Networking voice networks have network addresses with a fixed length of 10 digits or fewer. Any Modular Messaging system that is to join such a network must have mailbox numbers that are all of the same number of digits, 10 or fewer, regardless of whether MultiSite is used or not.

When you configure a Message Networking remote machine, you must ensure that the imported telephone numbers are in canonical format by properly configuring telephone number mappings. After you have applied the mappings correctly, a “+” is automatically inserted before the number.

When a MultiSite enabled Modular Messaging system is configured as a remote machine on a Message Networking system, or on another Modular Messaging system, the “+” will automatically be removed from the front of subscribers’ telephone numbers to avoid causing problems on the other systems.

Chapter 7: MultiSite sizing

MultiSite system sizing can be done using the following steps:

1. Identify the number of ports that are required on each SIP gateway to service the different sites.
2. Identify the number of ports that are required on the MASs to service the whole voice messaging system.

Port requirements for each SIP gateway

Follow the guidelines described in *Port Sizing* chapter of the *Avaya Modular Messaging Concepts and Planning Guide* to determine how many ports are required for the SIP gateways or proxy servers at each remote location.

Port requirements for all MASs in the VMD

The number of ports required for the VMD depends upon the peak number of ports that will be used across the whole VMD.

On one hand, there might be several sites that share the same busy hour. For example, two sites in Denver and New Jersey share much of the working day, so to calculate the number of ports required you must add together the ports that are required for each individual site. If New Jersey needs 32 ports and Denver needs 48 ports, then the MASs in the VMD would need 80 ports.

On the other hand, there might be very little overlap in busy hours. For example, if you have a site in Uxbridge (UTC/GMT) and another in Auckland, New Zealand (UTC+11 when daylight savings time is active), then during the Uxbridge day there will be almost no load on the Auckland system (For example 9 AM Uxbridge is 8 PM in Auckland). So if each site needs 32 ports to cope with its own busy hour, then you could manage with only 32 ports for the whole VMD.

You should be cautious when specifying the number of ports in the VMD because if you specify too few, then the quality of service may be affected.

A good aid to help determine the number of simultaneous ports that are required is to represent the information graphically. First, draw a timeline showing the business hours of the different sites, along with their busy-hour port requirements. *Figure 18* shows such a time line for the

four sites mentioned above. Note that to avoid confusion, the business hours of the different sites are all shown relative to UTC.

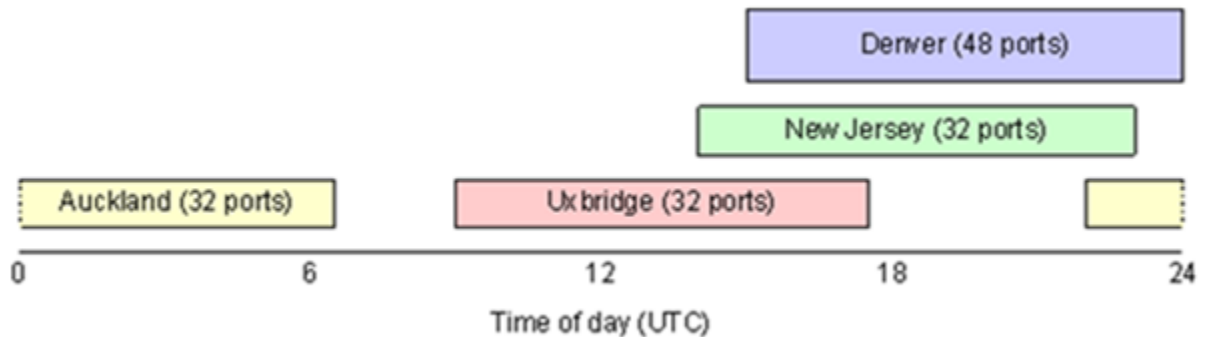


Figure 18: Time line of port usage at different sites

Next, you can plot a graph with the number of ports that are required for each point in time, as shown in *Figure 19*. From this you can see that the maximum number of ports that will be required at any one time is 112, compared with 144 if you simply add the totals of the different sites.

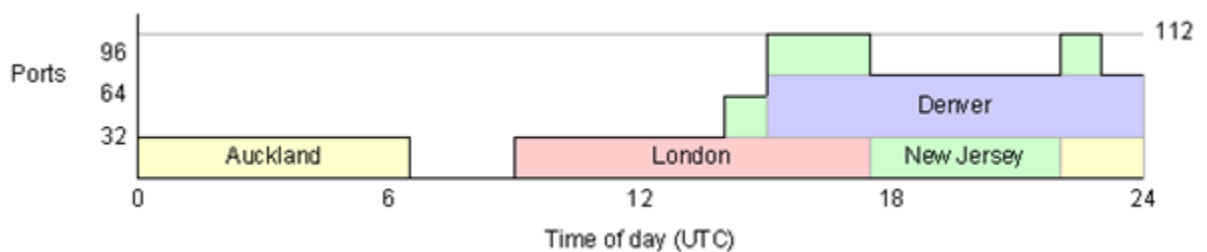


Figure 19: Total port usage throughout the day

This methodology assumes that the port usage at a particular site essentially drops to zero outside of business hours. If you know that is not the case then you can draw an appropriately sized box on the chart to represent the out-of-hours use. For example, if the usage in Denver is 48 ports during business hours, perhaps 16 ports might be required out of business hours.

Chapter 8: IP network requirements

The IP networking requirements are directly related to the number of ports that are required at the different sites and at the data center where the MASs are installed.

When Modular Messaging is configured to use SIP, it uses G.711 audio encoding, which has a data rate of 64 kbps. Each port uses two channels of audio (one for each direction). When network overheads (such as headers for the various protocols) are included, that results in a requirement for approximately 175 kbps per port.

There is no significant bandwidth overhead when using SRTP (See [Secure Real-time Transport Protocol \(SRTP\)](#) on page 74. There is a small overhead for SIP signaling traffic. Also note that the IP network has to carry other traffic, such as communication between clients and the MASs and the message store.

You can use the following table to estimate the bandwidth requirements for audio data when using SIP.

Table 18: Bandwidth requirements for audio data when using SIP

Voice ports	Total Bandwidth (kbps)	Bandwidth (Mbps) — half duplex	Bandwidth (Mbps) — full duplex
1	175	0.2	0.1
12	2100	2.1	1.05
24	4200	4.2	2.1
36	6300	6.3	3.15
48	8400	8.4	4.2
96	16800	16.8	8.4
144	25200	25.2	12.6

For each SIP gateway or proxy server you should use the required number of ports to determine the required network bandwidth. For more information, see [MultiSite Sizing](#) on page 63.

For the data center where the Modular Messaging servers are located, determine the total number of ports that are required and use this number to determine the required network bandwidth.

Ensure that the latency of network connections between the MASs and each SIP gateway or proxy server is not more than 150ms.

Chapter 9: Other MultiSite differences

Administration applications identify the user to the MAS

Any administration applications that use MAS telephony services must now identify the subscriber who is using the application to the MAS so that the MAS can ensure that the correct telephone number translation rules are used. Most administration applications can do this with no additional configuration, but a few require some additional data to be configured. If you are using the Visual Voice Editor, Caller Applications Editor, or Subscriber Options from the Modular Messaging Administration web application, then you must provide your full mailbox number in the appropriate field before you can play or record audio using a telephone.

Networked machine configuration

If you have enabled MultiSite then the networked machine configuration page of MSS Web Administration is changed.

For the local machine, you cannot configure the mailbox number length or any mailbox number ranges.

You can configure several mailbox number lengths for remote machines that have MultiSite enabled.

Chapter 10: Troubleshooting

“Could not map mailbox number to any configured site” in MSS Messaging Administration

When you edit the details for a subscriber using MSS Messaging Administration, and try to change the display of the PBX Extension from Canonical to Switch Native, or vice versa, the system displays the following error message:

Error: Could not map mailbox number to any configured site. Check the site configuration on MAS. (604): [extension number]

Cause: The MSS uses web services on a MAS to convert the extension number, and it is unable to acquire those services.

1. Ensure that all MASs have Modular Messaging release 5.0 or later installed. The web service is not available with earlier releases.
2. Ensure that the following services are started on all MASs:
 - Modular Messaging Application Server.
 - World Wide Web Publishing Service.

If you have recently started or restarted the MAS, then wait a few minutes and then try again.

3. Ensure that phone number translation rules have been properly configured for the PBX that is associated with the mailbox's site.

Rules or personal operators are not active at the correct times

Modular Messaging schedules, used in Find Me, Call Me, and Notify Me rules as well as when configuring personal operators, take time zones into account. To ensure that schedules are evaluated correctly, check that the following settings are correct:

- Mailbox time zone: You can configure this using Subscriber Options or Web Subscriber Options. The default is the Class of Service (COS) time zone set up by your administrator,

but if you are having problems, then set it explicitly to the time zone of the area where the subscriber is located.

- **Class of service time zone:** If you have not set the mailbox time zone, then the system uses the subscriber's class of service time zone. It defaults to *Use System Timezone*, but if you are having problems then set it explicitly to the time zone of the area where the member subscribers are located.
- **Site time zone:** If you have not set the mailbox nor the class of service time zones, then the system uses the site time zone. It defaults to *System default*, but you should configure it using VMSC to be the correct time zone for the location of the site.
- **Windows time zone:** If you have not configured any other time zone settings then the Windows time zone of the MAS handling the request is used.

Certain phone numbers are not dialed

If you attempt to dial a particular phone number and it fails, then do the following:

- Check the configuration of the phone number translation rules in VMSC. Enter the failing phone number as test data and see whether the rules behave as you expect. Check that an output number is produced, and note the call cost.
- Use VMSC to check the VMD maximum call costs (double-click the Sites node). Check that the call cost produced in the previous step is lower than the relevant limit (either for Automated Attendant or subscriber outcalls).

Glossary

AudioCodes Gateway	An AudioCodes gateway allows a MultiSite enabled Modular Messaging system to work with PBXs that are not supported by the SES, mainly those from third-party vendors. AudioCodes gateways support analog/ SMDI, T1 and E1 Q.SIG, and SIP integrations to the PBX, but they do not support DSE.
Automated Attendant	The system Automated Attendant that greets callers and instructs them on how to proceed. Automated Attendant is not the same as Caller Applications.
Avaya SIP Enablement Services	Avaya servers running SIP Enablement Services perform proxy, registration, and redirection functions associated with SIP applications.
Call Answering	Also known as telephone answering. This is the sequence of events that enable the voice mail system to answer calls on behalf of a subscriber if the line is busy or if the subscriber does not answer.
Call Me	A feature that allows subscribers to be called at a designated telephone number or from a telephone list, each time they receive a message that meets specified criteria. The subscriber is then invited to log in to Avaya Modular Messaging to review their messages.
Canonical Phone Numbers	Canonical phone numbers follow the E.164 standard which allows any publically-accessible phone number to be specified using a standard notation, consisting of an initial + followed only by digits. A code representing the country in which the phone is located comes immediately after the +. The format used for the rest of the number is country-dependent, but usually the whole number can be broken up like this: + CountryCode AreaCode SubscriberNumber
Data Collection Tool (DCT)	The DCT has two primary uses. First, it is used to gather information that is required in order to install Modular Messaging. Second, it can collect information from an existing Modular Messaging system that can be used if the system has to be rebuilt after a catastrophic failure.
DCT	See Data Collection Tool (DCT) on page 71.
Dialogic DMG 1000 Gateway	The Dialogic DMG 1000 Gateway provides DSE interface to Modular Messaging system. The Dialogic DMG 1000 gateway allows a MultiSite

enabled Modular Messaging system to work with PBXs that are not supported by the SES, mainly those from third-party vendors.

DID

See [Direct Inward Dialing](#) on page 72.

Digital Set Emulation (DSE)

Allows Modular Messaging to emulate a digital telephone in order to integrate digitally with some types of PBX. Also known as set emulation.

Direct Inward Dialing

A DID extension can be dialed directly from the public telephone network, without going through a receptionist.

Directory Enabled Management (DEM)

An interface that uses Avaya Directory Server to facilitate administration of Modular Messaging (MSS) from a centralized location.

DSE

See [Digital Set Emulation \(DSE\)](#) on page 72.

E.164

E.164 is an international numbering plan for public telephone systems. It represents the full number, including country code and area code.

For more information, see [Canonical Phone Numbers](#) on page 71.

Find Me

A feature that allows a subscriber to configure a list of telephone numbers where they might be contacted, so that Modular Messaging can try to connect a caller to a subscriber before asking the caller to leave a message.

FQDN

Fully Qualified Domain Name.

Full Mailbox Number

The full mailbox number of a subscriber in a MultiSite enabled voice mail domain includes the site identifier and the short mailbox number. For example, if John belongs to the Boston site (site ID - 1617), and his short mailbox number is 3564088, then his full mailbox number would normally be 16173564088, if the Boston site was configured to have 11-digit full mailbox numbers, and 7-digit short mailbox numbers.

MAS

See [Messaging Application Server \(MAS\)](#) on page 73.

Message Storage Server (MSS)

An Avaya-produced message store that is an integral part of the Modular Messaging system.

Message Waiting Indicator (MWI)

A method of alerting subscribers when messages meeting specified criteria arrive in their mailboxes. Subscribers are alerted by either a lamp indicator on their telephone or an audible tone (stutter dialtone) when they pick up the receiver. The indicator is cleared when the message is opened in the e-mail client or saved or deleted using the TUI. Subscribers can set up rules for using MWI in Subscriber Options. For example, they

may choose to be notified only when they receive urgent voice messages.

**Messaging
Application Server
(MAS)**

The voice server that provides an interface between the message store (and directory) and the telephone system.

MWI

See [Message Waiting Indicator \(MWI\)](#) on page 72.

Notify Me

With Notify Me, a subscriber gets notified when a message is received in their mailbox, or when a caller requests to notify them. The system can notify a subscriber by either sending an email message or by paging a numeric pager.

**Partial Mailbox
Number**

A partial mailbox number is the rightmost portion of a full mailbox number. For example, if the full mailbox number is 19089531234 then the following would all be valid partial mailbox numbers: 4, 34, 234, 9531234, and so on.

PBX

See [Private Branch Exchange \(PBX\)](#) on page 73. Synonymous with switch.

PBX Integration

A method that establishes communication between the PBX and the voice mail system. The PBX supplies information, such as the identity of the caller who is calling on internal calls and the extension that the caller is trying to reach. Also known as switch integration.

**Phone Number
Translation Rule**

Phone number translation rules are used to convert phone numbers between switch-native and canonical forms. These rules are defined using regular expressions.

**Private Branch
Exchange (PBX)**

A telephone exchange local to a particular organization, having a switchboard and associated equipment. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX. Also known as a switch.

**Q.Signaling
(QSIG)**

A protocol for ISDN-based inter-switch signaling based on the European Q.931, Q.9212, and DPNSS protocols.

QSIG

See [Q.Signaling \(QSIG\)](#) on page 73.

**Real-time
Transport Protocol
(RTP)**

A protocol for transmitting multimedia data, including voice, over the Internet.

RTP

See [Real-time Transport Protocol \(RTP\)](#) on page 73.

Secure Real-time Transport Protocol (SRTP)

Security profile for RTP intended to provide confidentiality, message authentication, and replay protection to the RTP data. SRTP is used to connect a MultiSite enabled Modular Messaging system to remote PBXs while keeping voice data private.

SES

See Avaya [SIP Enablement Services \(SES\)](#) on page 71.

Session Initiation Protocol (SIP)

A signaling protocol that allows exchange of information, such as call information, signaling information, and voice data using voice channels over an IP network.

Short Mailbox Number

Short mailbox numbers distinguish between subscribers in a single site. Within a site, short mailbox numbers must be unique, and of consistent length. Different sites can have short mailbox numbers of different lengths. Subscribers can use the short mailbox number to send voice messages to another subscriber in the same site.

SIP

See [Session Initiation Protocol \(SIP\)](#) on page 74.

SRTP

See [Secure Real-time Transport Protocol \(SRTP\)](#) on page 74.

Subscriber

A user whose profile is enabled for voice messaging. A subscriber can use both the TUI and the GUI of Modular Messaging.

Switch

Synonymous with PBX.

Switch-native Phone Number

Switch native phone numbers are phone numbers that can be processed by the specific switch from which or to which they are sent.

VMD

See [Voice Mail Domain \(VMD\)](#) on page 74.

VMSC

See [Voice Mail System Configuration \(VMSC\)](#) on page 74.

Voice Mail Domain (VMD)

A group of MAS units that share a common set of properties. All subscribers who are provided with telephone answering by these MAS units belong to the same VMD.

Voice Mail System Configuration (VMSC)

An administration tool used to configure the attributes of a VMD or group of MAS units.

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