



Dialing Setup Guide

SmartMsg

Secure Notification Software

Additional SmartMsg documentation is available through the Codespear website.

<http://www.codespear.com/helpcentral.asp>

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Send e-mails regarding any of the above-mentioned problems or any other grievances regarding any documentation to:

SmartMsgdocumentation@Codespear.com

Codespear appreciates all comments regarding any issues in order to ensure the accuracy, consistency and simplicity of all of our documentation.

Please note: This e-mail address is for comments only. If you have technical questions please contact Technical Support by visiting our website.



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Phone Dialing Introduction

SmartMsg provides phone dialing capabilities for both sending alerts (dialing a phone number and speaking the alert) and including phones in chat sessions (dialing a phone to include in live 2-way communications).

There are various options for configuring phone dialing for a SmartMsg system. These options depend upon whether the customer will utilize Codespear dialing services, host their own dialing infrastructure, or use a hybrid approach, where both Codespear dialing infrastructure as well as customer-hosted dialing hardware is utilized.

[Codespear Hosted Dialing](#)

[Customer Hosted Dialing \(VoIP Dialing\)](#)

[Mixed Hosted Dialing \(Codespear and Customer\)](#)

Codespear Hosted Dialing

Codespear Hosted Dialing

Planning for utilization of Codespear Hosted Dialing Services

Codespear-hosted dialing allows a customer to utilize Codespear's infrastructure for their phone dialing needs. This provides a way for the customer to have access to large numbers of available telephone lines without needing to invest in costly hardware and communications lines. A customer can also benefit from the fault tolerance, power backup, and site redundancy features that are incorporated into Codespear's dialing infrastructure, again, without the high price tag that can be associated with actual ownership of such systems.

The Codespear Dialer module (sometimes called the CS Dialer Module) is used to connect and route all calls to the Codespear dialing infrastructure. The customer's SmartMsg server must be granted access via outbound TCP port 16900 to the Codespear Dialer servers. For redundancy purposes, a customer is assigned to more than one Codespear dialer server. (Specific dialing server information is provided by the Codespear Project Manager during the implementation phase.)

A SmartMsg system may be configured to connect to Codespear Dialing services. In this case, only one of the SmartMsg servers in the customer's system is actually running the CS Dialer Module and sending information to Codespear Dialers. Even SmartMsg clients connected to servers that are not running the CS Dialer module are able to use dialing services. Call information is simply routed from Server to Server and then out to the Codespear Dialer server. A SmartMsg administrator can choose to run the CS Dialer module on any or all of the servers in his system. Any server that is setup with the CS Dialer module must have access via TCP port 16900 outbound in order to communicate with Codespear dialer servers.

See the [Instructions for Setting up the Codespear Dialer Module](#).

Instructions for setting up the Codespear Dialer Module

1. Within the Administrator Tool, click the **Global Properties** menu and go to the **Modules** tab.
2. Select the **Codespear Dialer** Module and click the **Setup** button.
3. Go to the **Settings** tab.
4. Enter the name of the **Codespear Dialing Server**. (Obtain from Codespear).
5. Enter the **Port** the server is listening on. The default port is 16900.
6. Enter the **Username** and **Password**.

Note: Codespear will provide this information if you will be connecting to our dialing server. The only case in which you would not receive settings information from Codespear is if you've purchased the Codespear Dialing Software to host yourself and are running your own Codespear Dialer Server.

7. Click on the **Test** button to test out the connection. A message will appear telling you if you've successfully connected.
8. Once you have successfully connected, click the **Add** button to add the server to the dialer list. The Codespear Dialing Servers have been added to the **Dialers** list. To remove a dialer, highlight it and click on the **Remove** button.
9. To add more dialer servers to the list, change the settings and click the **Add** button again.

Note: If you would like a specific number to show up in the caller ID on user's phones when they receive a message, please contact the Codespear Support Team for assistance. Email support at support@codespear.com, visit the Codespear website's Support page, or call Codespear.

Next, set up the [Phone Number Module](#) to use the Codespear Dialer.

And/Or the [IP Phone Number Module](#) to use the Codespear Dialer.

Or the [Numeric Pager Module](#).

Customer Hosted Dialing

VoIP Dialing

When the customer chooses to host their own dialing infrastructure, there are a few options for Server modules/protocols that can be used. For example, SIP and H323 modules are available for SmartMsg; both of these are standard VoIP protocols. In addition, SmartMsg has the ability to accommodate some proprietary VoIP protocols, such as the Brooktrout API which is used to communicate with Brooktrout VoIP hardware.

Determine what method to use – H323 or SIP Dialing. For information on Brooktrout, contact Codespear Support.

[H323 Module](#)

[SIP Module](#)

H323 Module

Configuring the SmartMsg H323 Dialer Module

1. Open the SmartMsg **Administrator Tool**.
2. Select **Properties** from the **Global** Menu.

The **Global Properties** form will appear.

3. Click on the **Modules** tab.
4. Select **H323 Dialer** from the Modules list. Verify that a version number higher than 5.1.24 appears for this module. If "n/a" appears for the version number, the H323 module is not installed. Contact Codespear support for assistance.
5. Click the **Setup** button.
6. Click on the **Settings** tab.
7. In the **<Provider> Address** field, enter the IP address of the CallManager server.
8. Change the **<Provider> Port** setting if necessary.

The default port is 1720. The port used must match the H323 listening port as configured within CallManager. (This setting normally does not need to be changed.)

9. Enter the correct **Dialing Prefix** if a prefix is needed for dialing in the CallManager system. (*For example: many systems are configured to require the number 9 before any external phone numbers.*) If no prefix necessary, leave this field blank.
10. Define the **Max Channels** setting.

The Max Channels setting is used to limit the number of simultaneous calls SmartMsg will make to CallManager. The number defined for Max Channels should never exceed the actual number of available lines.

11. Check the option **Put 1 Before Numbers** only if the CallManager system requires dialing "1" ahead of phone numbers.
12. Enter the correct **International Prefix**, if the CallManager system requires a prefix for dialing out to International phone numbers.
13. Configure additional H323 settings as needed. *Normally, remaining settings can be left at default values.*

Vox Level – Determines the level of volume above which audio will be transmitted.

Maximum Retries – Determines the number of times a call will be attempted through H323/CallManager. For example, if the value is set to 3, SmartMsg will attempt a call 3 times before canceling.

Minutes to Wait Before Resending After Failure – Determines the amount of time between retries, when a call fails.

Codespear DTMF Detection Sensitivity – Determines the audio sensitivity to the tone played when a key is pressed on a phone keypad.

Default From Number – For systems that support Caller ID override, the number displayed on the recipient's Caller ID can be defined.

Verbose Logging – The verbose logging option allows for comprehensive logging of the module's activity. When verbose logging is enabled, all calls (including all call status information) are logged. It is recommended that this option only be checked while troubleshooting problems or monitoring activity with SmartMsg H323.

14. When settings are completed, click **OK** to save.

Example Configuring the Phone Number Module to Utilize a H323 Dialer

EXAMPLE

Configuring the SmartMsg Phone Number Module to utilize the H323 Dialer

For general Phone Number settings information see the [Phone Number Module](#).

This step is only necessary if SmartMsg will be used to call phone numbers.

1. Open the SmartMsg **Administrator Tool**.
2. Select **Properties** from the Global Menu.

The Global Properties form will appear.

3. Click on the **Modules** tab.
4. Select **Phone Number** from the module list. Verify that a version number higher than 5.2.1 appears for this module. If "n/a" appears for the version number, the Phone Number module is not installed.
5. Click **Setup** to configure settings for the Phone Number module.
6. Click on the **Settings** tab.

Important Note: The following instructions for Phone Number module settings assume setup is for a single phone gateway that is accessible from all SmartMsg servers. If more granular control is needed (i.e. multiple CallManagers/some SmartMsg servers unable to reach CallManager) please contact Codespear Support for assistance.

8. Select **Call Manager** from the Settings dropdown.
9. Remove any existing entries in the **Dialers** list by clicking the item and then clicking the **Remove** button.
10. Under Call Manager Servers: select **Any Server**.
11. Under Dialer Modules: select **H323 Dialer**.
12. Under Priority: select **1**.
13. Click the **Add** button. There should now be one entry in the dialers list that specifies 1: H323 Dialer.
13. Click the **OK** button to complete configuration of the Phone Number module.

Additional settings for the SmartMsg Phone Number module are available in order to control specific behavior when dialing phone numbers (such as Text To Speech settings, device defaults, etc.) See the *SmartMsg Phone Number Module Configuration Guide* for more information on these settings.

Example Configuring the IP Phone Module to Utilize a H323 Dialer

EXAMPLE

Configuring the SmartMsg IP Phone Module to use the H323 Dialer

For general settings information for the IP Phone Module see the [IP Phone Number Module](#).

This is only necessary if SmartMsg will be used to call IP phones (via IP address or a 3rd party VOIP system.)

1. Open the SmartMsg **Administrator Tool**.
2. Select **Properties** from the Global Menu.

The Global Properties form will appear.

3. Click on the **Modules** tab.
4. Select **IP Phone** from the module list. Verify that a version number higher than 5.2.2 appears for this module. If "n/a" appears for the version number, the IP Phone Number module is not installed.
5. Click **Setup**.
6. Click on the **Settings** tab.

Important Note: The following instructions for IP Phone module settings assume setup is for a single phone gateway that is accessible from all SmartMsg servers. If more granular control is needed (i.e. multiple CallManagers/some SmartMsg servers unable to reach CallManager) please contact Codespear Support for assistance.

7. Select **Call Manager** from the Settings dropdown.
8. Remove any existing entries in the **Dialers:** list by clicking the item and then clicking the **Remove** button.
9. Under Call Manager Servers: select **Any Server**.
10. Under Dialer Modules: select **H323 Dialer**.
11. Under Priority: select **1**.
12. Click the **Add** button. There should now be one entry in the dialers list that specifies 1: H323 Dialer.
13. Click the **OK** to complete configuration of the IP Phone module.

Additional settings for the SmartMsg IP Phone module are available in order to control specific behavior for dialing IP phone devices (such as Text To Speech settings, device defaults, etc.) See the *SmartMsg IP Phone Module Configuration Guide* for more information on these settings.

SIP Phone Dialer Module

Planning servers for the SIP Phone Dialer Interface

SIP is a standard protocol used for Voice over IP applications. The SIP Phone Dialer is an optional SmartMsg Server module. The SmartMsg SIP Phone Dialer interface allows a SmartMsg system to send voice messages to all types of phones—cell phones, analog telephones, SIP phones, etc. The module also allows for these same devices to be included in SmartMsg chat sessions.

The SIP Phone Dialer can be used in conjunction with a Third party VoIP carrier (such as Delta3 or Net2Phone) or can be configured to interface with SIP-compatible VoIP manager software (such as CISCO Call Manager or Nortel MCS). The SIP Phone Dialer can also be used in conjunction with T1 Access cards with built-in SIP software (such as Brooktrout Technology PCI cards).

Using the SIP Phone Dialer interface with a 3rd party VoIP carrier

The SmartMsg SIP interface can be used in conjunction with use of a VoIP carrier service. Note that a client need not be connected to the SmartMsg Server running SIP Dialer in order to utilize SIP Phone dialing— any of the system's SmartMsg Clients can send messages utilizing the SIP Phone Dialer (provided proper user permissions), even if connected to another server in the system.

The following ports must be opened on the firewall:

- UDP 5060 outgoing from the SmartMsg server running the SIP dialer interface
- UDP channels 50000 and above for voice channels (SmartMsg uses even ports from 50000 and above – one for each voice channel. For example, for three simultaneous voice channels, ports 50000, 50002, 50004 would be used.)
- Additionally, outgoing UDP ports are necessary. (Contact your VoIP carrier for information on which ports are needed. Normally, the carrier will let you know a range of outgoing UDP ports which must be open in order to use their service.)
 - *Most administrators find it most convenient to simply open up UDP communication, both incoming and outgoing for any ports 1000 and higher to and from the SmartMsg server running the SIP interface.*

Bandwidth requirements between SmartMsg Server running SIP Phone Dialer and VoIP Carrier – 64kbit (8bytes) maximum per channel.

Bandwidth requirements between SmartMsg servers and clients using audio/SIP – Normal audio can be compressed to 32Kbit (4bytes) or less per channel.

Using the SIP phone Dialer interface with VoIP provider/VoIP Manager Software

The SmartMsg SIP interface can be used in conjunction with use of VoIP Provider Software (in some cases the provider is Cisco's CallManager*). The SmartMsg server running the SIP Phone Dialer interface sends voice information to the VoIP provider

over UDP port 5060. The SmartMsg Server listens on even-numbered UDP ports 50000 and above (one port for each simultaneous voice channel).

If, for example, 4 voice channels are used the SmartMsg server will be listening on UDP ports 50000, 50002, 50004, 50006 – one port for each voice channel.

In this configuration, any SmartMsg Client will have access to SIP Phone Dialing functionality—even if the particular client is not connected to the SmartMsg server that is running the SIP Phone Dialer module.

Bandwidth requirements between SmartMsg Server running SIP Phone Dialer and VoIP Carrier – 64kbit (8bytes) maximum per channel.

Bandwidth requirements between SmartMsg servers and clients using audio/SIP – Normal audio can be compressed to 32Kbit (4bytes) or less per channel.

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See the [SIP Phone Dialer Module Settings](#).

SIP Phone Dialer Module Settings

Provider Information

User ID – This is the User ID of the account you set up on your SIP Dialer. (You must set up your SIP hardware/software first.) This is a one time only setup item.

Pass – The password associated with the User ID. This is a one time only setup item that you must obtain from the phone provider.

Realm – If you have a realm associated with your VoIP, enter the realm here. If you do not, leave this field blank. The realm is used for authentication of the username and password, similar to a domain, and is usually associated with SIP systems (delta3, vonage, etc.).

Use Provider for Phone and IP Calls – This checkbox specifies how IP calls are handled. When checked, SmartMsg will always use the Provider (entered above) to handle IP calls. When unchecked, IP calls will connect directly to their destination without going through the SIP provider.

Proxy Address – If you will be connecting to your provider through a server to make the calls, you will need to enter the address of the proxy server here.

Proxy Port – Enter the port associated with the proxy address here. The standard SIP port is 5060.

Provider Channels – The total number of channels that are dedicated to communicate with this SIP Card.

Enable Incoming Calls – This option will enable the Incoming Calls feature within SmartMsg, when checked. This feature allows users to call into the SmartMsg system to access alerts and chat sessions.

Listening Port – This will be the port used for the Incoming Calls feature, if enabled. The default is 5060. This is a one time only setup item that you must obtain from the phone provider.

Vox Level – This setting affects the minimum volume level that will be picked up by the Dialer. This is a setting that will need to be tweaked per your specific optimal preference.

Maximum Retries – This is the limit of the maximum number of times the SIP Dialer will retry a phone number. It is up to your specific optimal preference.

Max Channels – This is the total number of channels (phone lines) you will allow the software to try simultaneously. Once all of the max channels are in use, a queue will start for calls. This setting will be greater than or equal to the amount of Provider Channels.

Dialing Prefix – This is where you could enter any Dialing Prefixes for numbers that will be dialed via the SIP Dialer Module.

International Prefix – This is where you will enter the prefix for International calls. The default, 011, is the International Prefix for calls originating in the United States or Canada.

Put 1 Before Numbers – This checkbox enables SmartMsg to place a 1 before all numbers dialed. Check this box to have SmartMsg automatically place a 1 before all phone numbers dialed.

Minutes to Wait Before Resending After Failure – This is the amount of time in minutes that SmartMsg will wait before redialing a call after the previous call failed.

Codespear DTMF Detection Sensitivity – SmartMsg alerts often require recipients on phone devices to acknowledge a message by pressing a number on the keypad. And a phone device in a chat session may communicate by pressing numbers on their keypads which will appear in a Windows Client Chat window as the text version of the number. When keys are pressed on a phone's dialing pad, there are tones associated with each key – these are called DTMF tones. However, every SIP card has different volume levels for their DTMF tones. Test this out by opening a chat with a phone number device. When you press keys on the dialing pad, if you notice that the wrong tones are being picked up, the DTMF Detection Sensitivity is too high, and you should decrease it. If you whistle or make a noise into the phone that gets picked up as a number in the chat session, the DTMF Detection Sensitivity is too low, and you should increase it.

Default From Number – This is number that phone devices will see appear on their caller ID (if applicable) when they get an alert from SmartMsg.

Amplify incoming audio by – This is a setting that will need to be tweaked per your specific optimal preference. It will amplify the volume if the audio coming from the SIP Dialer Module is too soft.

Verbose Logging – This will enable a more detailed version of logging. Usually this setting is turned on in a troubleshooting scenario.

Get Active Call Stats – This button brings up a window with information on the current status of the SIP Dialer including how many active calls are in progress, how many calls are in the queue and how many alerts or chat sessions (messages) are currently active.

How the # of Provider Channels, Max Channels, and Checkbox for Use Provider for Phone and IP Calls Relate

Take this example:

of Provider Channels is set to 23

Max Channels is set to 30

User Provider for Phone and IP Calls is unchecked

Since there are 30 total channels, and 23 of them are dedicated for phone calls, there are 7 channels left undedicated, which can be used by IP calls. Therefore,

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channels 1-23 will be used for phone calls in ascending order. And, channels 30-23 in descending order will be used for IP phone calls.

In the same situation as above, but with the User Provider for Phone and IP Calls option checked, the Provider will be used for both phone calls and IP calls. Channels 1-23 will be used for each call, whether phone or IP phone call.

Next, setup the [Phone Number Module](#)

Or the [IP Phone Number Module](#)

Or the [Numeric Pager Module](#)

Mixed Hosted Dialing

Mixed Hosted Dialing

Mixed Hosted Dialing means employing aspects of [Codespear Hosted Dialing](#) and [hosting your own dialing](#). See a Codespear Sales Engineer or Project Manager to discuss how to incorporate aspects of both methods.

Example Call Management Settings

CALL MANAGEMENT SETTINGS EXAMPLE

An agency has two servers, Server A and Server B. Server A utilizes the Codespear Dialer module to handle calls. Server B utilizes the SIP Phone Dialer to handle calls. The agency would like all calls within SmartMsg to be dialed by Server A, unless it is unavailable, in which case it may use Server B.

This is how this situation would be set up:

1. Select "Server A" from the **Servers** dropdown.
2. Select "Codespear Dialer" from the **Dialer Modules** dropdown.
3. Select "1" from the **Priority** dropdown.
4. Click the **Add** button.
5. Select "Server B" from the **Servers** dropdown.
6. Select "SIP Phone Dialer" from the **Dialer Modules** dropdown.
7. Select "2" from the **Priority** dropdown.

Click the **Add** button.

Other Modules to Configure

Direct Dial Module

The Direct Dial Module utilizes the dialing system configured within the Phone Number Module. Therefore, in order to use the Direct Dial Module, the Phone Number Module must be configured properly.

See the [Phone Number Module](#).

To use the Direct Dialing feature, see [Using the Direct Dial Function in a Chat Session](#).

Phone Number Module

Phone Number Module

Accessing the Phone Number Module

1. Open the SmartMsg Administrator Tool.
2. Select **Properties** from the **Global** menu.

The **Global Properties** screen will appear.

3. Go to the **Modules** tab and highlight the **Phone Number** module.
4. Click the **Setup** button.

The Module Setup screen will appear.

***Note:** If the Phone Number Module does not show a version number, it is not actually installed.*

[Phone Number Module General Tab](#)

[Phone Number Module Settings Tab](#)

[Phone Number Module Device Defaults Tab](#)

Phone Number Module General Tab

General Tab

The General tab of Module Setup for the Phone Number module will have default options selected upon installation. Normally, the settings on this tab do not need to be modified.

Enable Module for System –must be selected in order to have phone number devices in the SmartMsg system.

This is a User Selectable Device –gives users the ability to add their own phone number devices. If this is unchecked, no users in SmartMsg will be able to add phone number devices to their or anyone else's device list, not even Global Administrators.

This is a Secure Device –designates all phone numbers in the SmartMsg system as secure. Typically this is not a secure device because phones do not maintain connection to a secure network. This option forces users to enter their pin code when the SmartMsg senders send alerts to "Secure Devices Only".

Enable Chat Line History Catch-Up –this setting does not apply to phone number devices.

Default Server to Handle This Device Type – denotes the server that will handle all phone number devices in this system.

Default Connection Speed for This Device Type – indicates the connection speed for all phone number devices in this system.

Capabilities

Message Type – for each of the message types listed, the checkboxes on the right define if the phone number devices within this SmartMsg system are capable of receiving messages of that type and whether or not users are allowed to change these capabilities for their own phone number devices.

Next see the [Phone Number Module Settings Tab](#).

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Phone Number Module Settings Tab

Settings Tab

The Settings Tab contains 3 different parts. From the Settings dropdown, there are 3 options:

- [Text To Speech](#)
- [Call Manager](#)
- and [Other](#)

Phone Number Module Device Defaults Tab

Device Defaults Tab

Device Defaults entered here will take effect from this point forward. Once you click OK, every new phone number device that is added will automatically contain the defaults chosen here. Settings for a particular device may be modified after the device is created. For more information on the Phone Number device settings see *SmartMsg Basic Training Client Dashboard*.

Phone Number Module Settings Text To Speech

For more detailed information on Text to Speech see the *SmartMsg Text to Speech Guide*.

Text To Speech

Note: *These settings will override the text to speech module settings for phone number devices.*

Template Legend – shows each user-defined part of a SmartMsg message and how it will be represented.

The words spoken appear in the text box on the bottom of the window and can be modified by typing in the textbox or using the buttons above to enter user-defined message information.

Spoken Text – lists each spoken part of a SmartMsg message. Click on one of the items in the list and then view/edit the spoken text below.

Spoken Text Items

Acknowledge/Replay – Plays after the body of the alert has been spoken. Typically, spoken text item contains options for the choices that recipients have after they hear the message. For example, "Press 1 to acknowledge this message. Press 2 to repeat this message. <%PAUSE=2000%>".

Acknowledgement Confirmation – Plays after an alert has been acknowledged by the recipient. This text item typically says something to confirm that the recipient's alert acknowledgement has been received/recorded by SmartMsg. For example, "Your acknowledgment has been received. You may hang up at this time. <%PAUSE=1000%>".

Chat Greeting – Plays when the phone has been picked up; indicates to the user on the phone device that this is an incoming chat request. For example, "This is an incoming chat request for <%NAME%> from <%FROMNAME%>. <%PAUSE=1000%>".

Default Custom Response Title – Plays if the message has a custom response with no custom response title specified. If the custom response has been given a title, the title will replace this default. Typically this spoken text item contains instructions for how to respond to the alert. For example, "Please choose from one of the following options."

Invalid PIN Code – Plays after the user enters an invalid PIN Code. Typically this spoken text item explains what to do when an Invalid PIN is entered. For example, "You have entered an invalid pin code. Please reenter your pin code followed by the pound sign. <%PAUSE=1000%>".

PIN Code – Plays if the alert/chat requires authentication. Some alerts are sent with extra security in place and will require recipients to enter PIN Codes on phone devices. In this case, this spoken text item would be heard. For example, "Please

enter your smart message pin code followed by the pound sign.
<%PAUSE=1000%>".

SmartMsg Greeting – Plays when the phone has been picked up for an incoming SmartMsg alert (does not apply to Chat communications). This is the first thing a recipient on a phone device will hear, so the greeting must contain information that the phone call is coming from SmartMsg as an alert. For example, "This is an incoming smart message for <%NAME%> from <%FROMNAME%>.
<%PAUSE=1000%>".

SmartMsg Template – Plays after SmartMsg Greeting for an incoming SmartMsg Alert (does not apply to Chat communications). This is the meat of the alert, and typically it contains the Priority of the Message, Subject, and Body. For example, "This is a <%PRIORITY%> Priority Alert Message.

<%PAUSE=1000%>

Subject. <%SUBJECT%> <%PAUSE=1000%>

Body. <%BODY%> <%PAUSE=1000%>

End of Message."

To Modify Text to Speech Settings:

1. Select part of the **Spoken Text** to be modified (i.e. SmartMsg Greeting) from the list.
2. Modify the entry in the bottom textbox:

a. Use the **Insert Priority**, **Insert Subject**, **Insert Body**, **Insert Pause**, **Insert Sender's Name**, and **Insert User's Name** buttons as needed.

b. After inserting Pauses, modify the entries to specify the number of seconds desired. (For example "<%PAUSE =6000%>" defines a pause of 6000 milliseconds/6 seconds.

c. Literal text can be inserted before, after or within these items, by simply typing in the desired text. (For example, the text "Hello" could be inserted at the very beginning.)

TEXT TO SPEECH EXAMPLE

An agency would like the initial SmartMsg Greeting to say, "Hello *name of recipient*, this is a SmartMsg from *sender's name*."

1. Highlight **SmartMsg Greeting** from the **Spoken Text** list.
2. Delete the current SmartMsg Greeting message in the text box.
3. Type in "Hello "
4. Click the **Insert User's Name** button.
5. Type "this is a Smart Message from "
6. Click the **Insert Sender's Name** button.

SmartMsg Dialing Setup Guide

Phone Number Module Settings Call Manager

Call Manager

Note: The term "Call Manager" here refers to general call management within SmartMsg. Do not confuse these Call Manager settings with Cisco's CallManager product.

In the Call Manager settings, Administrators may set up the way in which the calls are handled. By default, there will be a dialer already configured in the Call Manager section upon installation. As you can see under Dialers, there is an entry "1: SIP Phone Dialer". If this default is not changed, it means that all calls will be handled by the SIP Phone Dialer on Any Server. When a Server is defined for an entry, it will appear in parentheses next to the name. If "Any Server" is selected, nothing appears next to the name. For example, if "Server A" was picked from the **Servers** dropdown and "Codespear Dialer" was picked from **Dialer Modules** and "1" was picked from **Priority**, the entry under **Dialers** would be formatted like this: "1:Codespear Dialer (Server A)".

To Delete a Dialer:

Highlight a dialer and click **Remove** to delete dialer from the list.

To Add a Dialer:

Choose a **Server** and a **Dialer Module** to handle the calls, and the priority of this choice. Then, click on **Add** to add the setup to the list. There must be at least one **Dialer** in the list. There may be many Dialers with different priorities.

Incoming Calls – will be implemented in a future version of the software.

See the [Example Call Management Settings](#).

Phone Number Module Settings Other

Other

Other is an option in the Settings dropdown for the Phone Number Module.

Disable Voicemail Detection – check this box to disable voicemail detection. For more information see the *SmartMsg Voicemail Detection Setup Guide*.

See the topic on [Voicemail Detection](#).

Voicemail Detection

Setup Scenarios

There are a few different ways that Administrators can set up the SmartMsg Voicemail Detection. The options include the combinations of enabling/disabling the System Wide **Voicemail Detection** option (located in Administrator Tool > Global menu > Properties > Modules tab > Phone Number Module > Settings tab > "Other" selected for Settings.) and the **Leave SmartMsgs on Voicemail** option (Located in any user's client > Settings> Devices tab > Phone Number Device type.-OR- Administrator Tool > User menu > Modify User > Devices tab > Phone Number Device type.)

When:

System Wide **Voicemail Detection** option (Phone Number Module) - enabled

Leave SmartMsg on Voicemail option (Device Properties) – enabled

Then:

Voicemail will be detected and the full alert notification will be left on the recipient's voicemail.

When:

System Wide Voicemail Detection option (Phone Number Module) - enabled

Leave SmartMsg on Voicemail option (Device Properties) – disabled

Then:

Voicemail will be detected but the system will NOT leave the alert notification on the voicemail. Once voicemail is detected, the system will hang up.

When:

System Wide Voicemail Detection option (Phone Number Module) - disabled

Leave SmartMsg on Voicemail option (Device Properties) – enabled

Then:

Voicemail will not be detected and recipients may hear only a partial part of the alert notification on their voicemail.

When:

System Wide Voicemail Detection option (Phone Number Module) - disabled

Leave SmartMsg on Voicemail option (Device Properties) – disabled

Then:

Voicemail will not be detected and recipients may hear only a partial part of the alert notification on their voicemail.

SmartMsg on Voicemail and the Resend Setting

SmartMsg behaves a little differently when these two options are used in conjunction: **Resend Every x Minute(s) Until Acknowledged** and **Leave SmartMsgs on Voicemail**. For example, a user has set their Phone Number device type to receive SmartMsg alerts on voicemail. They have also selected the rule where the SmartMsg system will resend an alert every 3 minutes until it is acknowledged.

When these two options are used together in an individual user's device settings, the system will not leave a voicemail on every retry. Instead, the voicemail will only be left on the final try. In the example, the option **Resend Every 3 Minute(s) Until Acknowledged** is selected along with the option to **Leave SmartMsgs on Voicemail**, therefore the system will call this phone number every 3 minutes until there are no more retries left, and it will only leave a voicemail on the final call.

Disabling Voicemail Detection

By default, Voicemail Detection is enabled. When Voicemail Detection is disabled, the system will not attempt to detect voicemail, and will therefore simply treat the voicemail as if it is a recipient answering their phone. The alert will start being delivered during the voicemail startup dialogue, and sometimes the tail ends of alerts are left "after the beep". This is why recipients may have partial SmartMsg alerts on their voicemails. It's not recommended to disable voicemail detection for this reason. However, in some cases, an Administrator may desire to do so. To disable voicemail detection:

1. In the Administrator Tool, go to **Global Properties**.
2. Go to the **Modules** tab.
3. Select the **Phone Number** module from the Installed Modules list.
4. Click on **Setup**.
5. Go to the **Settings** tab.
6. From the **Settings** dropdown list, select **Other**.
7. Select the option to **Disable Voicemail Detection**.

Enabling Voicemail Detection

By default, Voicemail Detection is enabled. If it has been previously disabled, administrators can re-enable this option. To enable voicemail detection:

1. In the Administrator Tool, go to **Global Properties**.
2. Go to the **Modules** tab.
3. Select the **Phone Number** module from the Installed Modules list.
4. Click on **Setup**.
5. Go to the **Settings** tab.

6. From the **Settings** dropdown list, select **Other**.
7. Uncheck the option to **Disable Voicemail Detection**.

Voicemail Settings

Default System Wide Voicemail Settings

The device defaults specified in the Phone Number module will automatically appear for each new Phone Number device created from this point forward. When a new device is created, the default settings will appear but the settings can be modified per individual device.

1. Go to the **Device Defaults** tab. Here you can specify defaults for all phone number devices from this point forward. ***Changing an option here will not change any settings for any previously created phone number devices.***
2. Check **Leave SmartMsgs on Voicemail** if you want SmartMsg alerts to be recorded on phone devices that have voicemail capabilities. If you do not check this box, no alert messages will be left on voicemail.

Note: An alert message left on voicemail does not count as an acknowledgement. The recipient will still have to acknowledge the message.

Another Note: Acknowledgement options (Press 1 to Acknowledge...) are not "live" when left on voicemail. Recipients cannot acknowledge an alert from their voicemail. The recipient must acknowledge a live message.

3. Check ***Don't Read SmartMsg Greetings*** if you only want the body of the alert message to be heard on phone devices. *When this option is checked, the greeting that precedes alerts and chat requests, "You have an incoming SmartMsg..." will not be played for recipients on phone devices. Administrators may choose to check this option as a time saving method to keep phone messages as short as possible. Only the alert message body will be heard.*

When the options above are selected as defaults, each time a new device is created the options for Leave SmartMsgs on Voicemail and Don't Read SmartMsg Greetings will automatically be checked.

IP Phone Number Module

IP Phone Number Module

Accessing the IP Phone Module

1. Open the SmartMsg Administrator Tool.
2. Select **Properties** from the **Global** menu.

The **Global Properties** screen will appear.

3. Go to the **Modules** tab and highlight the **IP Phone** module.
4. Click the **Setup** button.

The Module Setup screen will appear.

***Note:** If the IP phone module does not show a version number, it is not actually installed.*

See:

[IP Phone Module General Tab](#)

[IP Phone Module Settings Tab](#)

[IP Phone Module Device Defaults Tab](#)

IP Phone Module General Tab

General Tab

The General tab of Module Setup for the IP Phone module will have default options selected upon installation. Normally, the settings on this tab do not need to be modified. However, some situations exist in which a SmartMsg Administrator may need to configure settings on this tab.

Enable Module for System –must be selected in order to have IP phone devices in the SmartMsg system.

This is a User Selectable Device –gives users the ability to add their own IP phone devices.

This is a Secure Device –designates all IP phone devices in the SmartMsg system as secure. Typically this is not a secure device. This also forces users to enter their pin code when the SmartMsg sender sends a message to "Secure Devices Only".

Enable Chat Line History Catch-Up –this setting does not apply to IP phone devices.

Default Server to Handle This Device Type – denotes the server that will handle all IP phone devices in this system.

Default Connection Speed for This Device Type – indicates the connection speed for all IP phone devices in this system.

Capabilities

Message Type – for each of the message types listed, the checkboxes on the right define if the IP phone devices within this SmartMsg system are capable of receiving messages of that type and whether or not users are allowed to change these capabilities for their own IP phone devices.

See the IP Phone Module Settings Tab

IP Phone Module Settings Tab

Settings Tab

The Settings Tab contains 3 different parts. From the Settings dropdown, there are 3 options:

[Text To Speech](#)

[Call Manager](#)

and [Other](#)

The settings within the tab relate to the option selected in the Settings dropdown.

Next tab:

[IP Phone Module Device Defaults Tab](#)

SmartMsg Dialing Setup Guide

IP Phone Module Settings Text To Speech

Text To Speech

Note: *These settings will override the text to speech module settings for IP Phone devices.*

Template Legend – shows each user-defined part of a SmartMsg message and how it will be represented in the template below.

Spoken Text – lists each spoken part of a SmartMsg message. Click on one of the items in the list and then view/edit the spoken text below.

The words spoken in the selected part of the message appear in the Acknowledge/Replay textbox and can be modified by typing in the textbox or using the buttons above to enter user-defined message information.

Spoken Text Items

Acknowledge/Replay – Plays after a SmartMsg has finished.

Acknowledgement Confirmation – Plays after a SmartMsg has been acknowledged.

Chat Greeting – Plays when the phone has been picked up for an incoming chat request.

Default Custom Response Title – Plays if the message has a custom response with no custom response title specified.

Invalid PIN Code – Plays after the user enters an invalid PIN Code.

PIN Code – Plays if the message/chat requires authentication.

SmartMsg Greeting – Plays when the phone has been picked up for an incoming SmartMsg alert (does not apply to Chat communications)

SmartMsg Template – Plays after SmartMsg Greeting for an incoming SmartMsg Alert (does not apply to Chat communications)

To Modify Text to Speech Settings:

1. Select part of the **Spoken Text** to be modified (i.e. SmartMsg Greeting) from the list.
2. Modify the entry in the bottom textbox:
 - a. Use the **Insert Priority**, **Insert Subject**, **Insert Body**, **Insert Pause**, **Insert Sender's Name**, and **Insert User's Name** buttons as needed.

b. After inserting Pauses, modify the entries to specify the number of seconds desired. (For example "<%PAUSE =6000%>" defines a pause of 6000 milliseconds/6 seconds.

c. Literal text can be inserted before, after or within these items, by simply typing in the desired text. (For example, the text "Hello" could be inserted at the very beginning.)

TEXT TO SPEECH EXAMPLE

An agency would like the initial SmartMsg Greeting to say, "Hello *name of recipient*, this is a SmartMsg from *sender's name*."

To do this:

1. Highlight **SmartMsg Greeting** from the **Spoken Text** list.
2. Delete the current SmartMsg Greeting message in the text box.
3. Type in "Hello "
4. Click the **Insert User's Name** button.
5. Type "this is a Smart Message from "
6. Click the **Insert Sender's Name** button.

Next Tab:

[IP Phone Module Settings Tab Call Manager](#)

SmartMsg Dialing Setup Guide

IP Phone Module Settings Tab Call Manager

Call Manager

Note: The term "Call Manager" here refers to general call management within SmartMsg. Do not confuse these Call Manager settings with Cisco's CallManager product. In the Call Manager settings, Administrators may set up the way in which the calls are handled. By default, there will be a dialer already configured in the Call Manager section upon installation. As you can see under Dialers, there is an entry "1: SIP Phone Dialer". If this default is not changed, it means that all calls will be handled by the SIP Phone Dialer on Any Server. When a Server is defined for an entry, it will appear in parentheses next to the name. If "Any Server" is selected, nothing appears next to the name. For example, if "Server A" was picked from the **Servers** dropdown and "Codespear Dialer" was picked from **Dialer Modules** and "1" was picked from **Priority**, the entry under **Dialers** would be formatted like this: "1:Codespear Dialer (Server A)".

To Delete a Dialer:

Highlight a dialer and click **Remove** to delete dialer from the list.

To Add a Dialer:

Choose a **Server** and a **Dialer Module** to handle the calls, and the priority of this choice. Then, click on **Add** to add the setup to the list. There must be at least one **Dialer** in the list. There may be many Dialers with different priorities.

Incoming Calls – will be implemented in a future version of the software.

See the [Example Call Management Settings](#).

Next Tab: [IP Phone Module Settings Tab Other](#)

IP Phone Module Settings Tab Other

Other

Other is an option in the Settings dropdown.

Disable Voicemail Detection – check this box to disable voicemail detection. For more information see the *SmartMsg Voicemail Detection Setup Guide*

See the topic on [Voicemail Detection](#)

Next Tab: [IP Phone Module Device Defaults Tab](#)

Device Defaults Tab

Device Defaults entered here will take effect from this point forward. Once you click OK, every new phone number device that is added will automatically contain the defaults chosen here. Settings for a particular device may be modified after the device is created. For more information on the Phone Number device settings, see *SmartMsg Basic Training Client Dashboard*.

There are two options for **Type** – Destination and IP Address.

Destination – an extension for IP phones.

IP Address – Internet protocol address of the network IP phone device (example: 10.1.1.1).

Numeric Pager Module

Numeric Pager Module

Numeric pagers may only receive alerts, and may only receive numbers. Specific number codes may be set up to let the recipient know the priority of the message. Numeric Pagers may also receive call back numbers that the recipient may call to get the whole message.

[Numeric Pager Module Settings Tab](#)

[Numeric Pager Module Device Defaults](#)

SmartMsg Dialing Setup Guide

Numeric Pager Module Settings Tab

To set up the Numeric Pager Module:

1. Open the SmartMsg **Administrator Tool**.
2. Select **Properties** from the Global Menu.

The Global Properties form will appear.

3. Click on the **Modules** tab.
4. Select **Numeric Pager** from the module list.
5. Click **Setup**.
6. Click on the **Settings** tab. While "Main" is listed in the dropdown menu, the checkbox for "Device Logging" is the only option available. Device logging is used for troubleshooting problems with Numeric Pagers and would probably only be used if a Codespear Support Representative suggested to turn it on. The next item in the dropdown, Call Manager, is where you may set up Dialing options for Numeric Pagers.
7. Select **Call Manager** from the Settings dropdown.

Call Manager

Note: The term "Call Manager" here refers to general call management within SmartMsg. Do not confuse these Call Manager settings with Cisco's CallManager product. In the Call Manager settings, Administrators may set up the way in which the calls are handled. When a Server is defined for an entry, it will appear in parentheses next to the name. If "Any Server" is selected, nothing appears next to the name. For example, if "Server A" was picked from the **Servers** dropdown and "Codespear Dialer" was picked from **Dialer Modules** and "1" was picked from **Priority**, the entry under **Dialers** would be formatted like this: "1:Codespear Dialer (Server A)".

To Delete a Dialer:

Highlight a dialer and click **Remove** to delete dialer from the list.

To Add a Dialer:

Choose a **Server** and a **Dialer Module** to handle the calls, and the priority of this choice. Then, click on **Add** to add the setup to the list. There must be at least one **Dialer** in the list. There may be many Dialers with different priorities.

Incoming Calls – will be implemented in a future version of the software.

See the [CALL MANAGEMENT EXAMPLE](#).

Numeric Pager Module Device Defaults

Device Defaults Tab

Device Defaults entered here will take effect from this point forward. Once you click OK, every new numeric pager device that is added will automatically contain the defaults chosen here. Settings for a particular device may be modified after the device is created. For more information on the Numeric Pager device settings, see *SmartMsg Basic Training Client Dashboard*.

Dialing Within a SmartMsg Chat

Using the Direct Dial Function in a Chat Session

There are two ways to add a user to a chat session on a phone device:

- A user's phone number device (that was already configured by the user in the SmartMsg system) may be added to a chat session.
- SmartMsg users also have the capability to directly dial phone numbers from a chat session. Once the call is connected, the user on the phone will be included in the session. This method is useful for connecting to entities that have not defined phone number devices in the SmartMsg system.

To Directly Dial a Phone Number in Chat:

1. Initiate a chat session with the 'Direct Dial User'. **Note:** *The user initiating the chat session must have permissions (granted by an Administrator) to chat with the Direct Dial User.*
2. Under the Direct Dial User's display picture, click on the icon the farthest to the right. This is the same icon you click to access the [dialing keypad](#). However, this time there will appear a window where you can enter dialing information.

Note: Only the chat initiator may directly dial a user. While the other chat participants can see the Direct Dial User in the chat, they cannot click on the phone number device icon and therefore cannot directly dial a user.

3. Type in the **Name of Person Being Called**.
4. Select the **Type:** either Phone Number or International Phone Number. Choose Phone Number for all calls within the United States and Canada. Choose International for all calls outside of the United States and Canada. *Note: This is assuming you are using Codespear-hosted dialing service.*
5. Enter the phone number into the **Number** area. Add an **Extension**, if needed (optional).
6. Check any options that may be desired. 'Don't Read SmartMsg Greetings' will skip reading the introductory invitation message to the person that answers the call. 'Allow Phone to Ring for x Seconds' will limit the length of time that the system will try to reach this number.
7. Once all information is entered into this area, click on the **Call** button. The system will then call this phone number and when the call goes through they will then be in the chat session. Clicking on the **Cancel** button cancels the direct dialing action.

As soon as the Call button is hit, the user interface changes back to the dialing keypad and the Disconnect button becomes available. If the Disconnect button is clicked, the Direct Dial User leaves the chat and the phone number dialed is disconnected.

Note: Only 1 Direct Dial user can be present in a session at a time and can only be used to call 1 phone number. However, the Direct Dial user can be present in multiple chat sessions.

Using the Dialing Keypad in Chat

A user who initiates a SmartMsg chat with a phone number device may use the dialing keypad to press touch-tone digits. For example, automatic answering services that require the caller to press certain buttons that correspond with specific actions (i.e. "Press 1 for the main menu or press 0 to talk to the Operator"). The dialing keypad can be used to enter these numbers.

To use the Dialing Keypad in Chat:

1. Open a chat session by clicking on the **Chat** button in the Client Dashboard.
2. Invite a user's phone number device into the chat session. (To access a user's devices, select a user and click the **Show Devices** button.)
3. Under the user's display picture, click on the icon the farthest to the right. This is the dialing keypad icon.
4. Once you click on this icon, the dialing keypad will appear and you will be able to press key tones. The phone number device will receive the DTMF (touch-tone) tone you dial.

The phone device can also communicate back to the chat by pressing DTMF keys. In the example below, Betty Smith (on a phone device) was called and then pressed various numbers.

5. You can also hang up the phone connection with this user by clicking on the "Disconnect" button. This will end the call with the phone number device and the user will no longer be able to participate in the chat session.

*Note: Only the person that invited the phone number device will be able to disconnect it. Other chat participants may also access a keypad, but the **Disconnect** button will not be present.*

Chat Sessions with Phone Devices and Radios

Phone Devices are designated as Full Duplex.

Radio Devices are Half Duplex.

Therefore, when a chat session contains phone devices and radio devices, the chat session will automatically switch to Half Duplex mode to accommodate the limitations of the Radio device.

Full Duplex Definition

Full Duplex means that a device may transmit and receive simultaneously.

See [Chat Sessions in Full Duplex](#).

Half Duplex Definition

Half Duplex means that a device may transmit or receive, but not simultaneously.

See [Chat Sessions in Half Duplex](#).

Chat Sessions in Full Duplex

Designating a chat session as [Full Duplex](#) means that all devices in the chat session may transmit and receive simultaneously. Therefore the Chat Window on a Windows Client device will have a Speak button that you may press once to turn on, and leave it on for as long as you are speaking.

Chat Sessions in Half Duplex

Designating a chat session as [Half Duplex](#) means that at least one device in the chat session is half duplex. Radios are most often Half Duplex devices. Therefore, the Chat Window on a Windows Client device will have a Speak button that you must press and hold for the duration of your speech communication.