Administrators Guide
IPS Pager & Audio Broadcast

Directory
Phone Directory
Jabber UDS Server
Web Directory
IPS Popup / Reverse Lookup
Personal Directory
H350 Video Conf directory
Corporate Speed Dials
ClickNDial
Alerting
Voice Alert
IPS Pager
Admin tools
Morning Check
Phone Remote
Phone Robot
Provisioning
Phone Deployment
CMS Admin & Selfcare
Extension Mobility Report
Manager Assistant
IP Phone / Jabber Interface

Productivity tools
IPS Phone Config
IPS Alarm Callback
IPS Lock
Wakeup Call
Missed Call Alerter
Conference Center
Busy Alerter Callback
Desktop Popup
Finesse Gadgets
Spark Bot
Attendant Console / IVR / Group
Tannounce
Line Group Manager
Silent Monitoring
Extension Mobility tools
TSSO
Delog / Relog
Pin & Password Manager
Recording
Call Recording
Recording Notification

Version: 3.4.x
Administration : 6.0.x

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1   IPS Pager & Audio Broadcast description

1.1 Overview

telisca IPS Pager is a perfect solution to broadcast text messages, presentations (several graphic or text slides) or play audio announcements to a large groups of Cisco IP Phone users or from user to user.

Text messages defined may be sent from an IP Phone service or from a web interface (that fit in a Jabber's tab). Depending of his profile, the user can use predefined distribution list or send to define phone numbers, he may also use predefined message or enter a new message. Message priority are pushed with an audio notification and can clear after a while. Information messages or presentations may remain availables for reading afterward, through Services menu.

Audio announcement can be sent live (push to talk) from an IP Phones to a list of IP Phones or from a recorded audio file or Text to Speech. Audio announcements are sent from one IP Phone or from IPS Pager server to a group of IP Phones, using multicast ip technology which reduce the bandwidth required. The audio announcement is played on the IP Phone loudspeaker or if the user is online, on the phone handset. Audio level can be forced depending of alert priority. It is also possible to broadcast to external IP Speakers.

Text messages or audio announcements can be sent from the IP Phone XML interface, a full screen map based Web interface, a JabberTab, a third party application (REST API) or automaticaly on schedule, taking into account the time range of the phones.

1.2 Architecture

There are two ways of sending text messages to Cisco IP Phones:
-By CTI: An authorized application user is dynamically associated to the destination IP phone and sends a URL to display via JTAPI. This method is simple to deploy but limited to several hundred IP phones per transmission.
-By http: The application dynamically loads IP phone IP addresses and sends an http post request toward the IP phones. The IP phones must be authorized for web access. IPS Pager includes a proxy authentication (of type 'One type password'), which secures the push and unloads the CUCM cluster. This method is appropriate to massive transmissions.

To secure authentication requirement when pushing to IP Phone, IPS Pager integrates an authentication proxy. The proxy is based on a one-time password mechanism that enhances security and free Cisco Unified Communications Manager from the load occurred when pushing to a large number of IP Phones.

Audio message broadcast places the IP phones in listening mode on a port and multicast IP address. This multicast IP address must be routed to reach the IP phones on the different sites concerned.
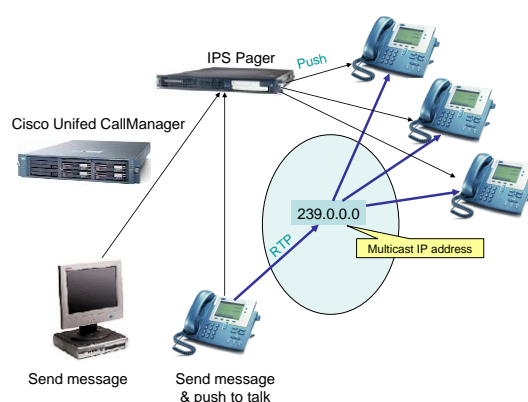

The IPS Pager server effects a simultaneous 'push' on several IP Phones to cause them to display the selected message or place them in listening mode on a multicast IP address. The number of simultaneous pushes is configurable according to server performance.

IPS Pager supports multi-cluster CUCM architecture and message can be pushed to IP Phones from different CUCM clusters.

1.3 User Interfaces

Message transmission may be triggered via a web interface or Jabber, protected by key or by AD/LDAP security group. According to the profile, users may access pre-defined distribution lists and pre-defined messages. The interface also permits the creation of temporary messages and to define their priority level.
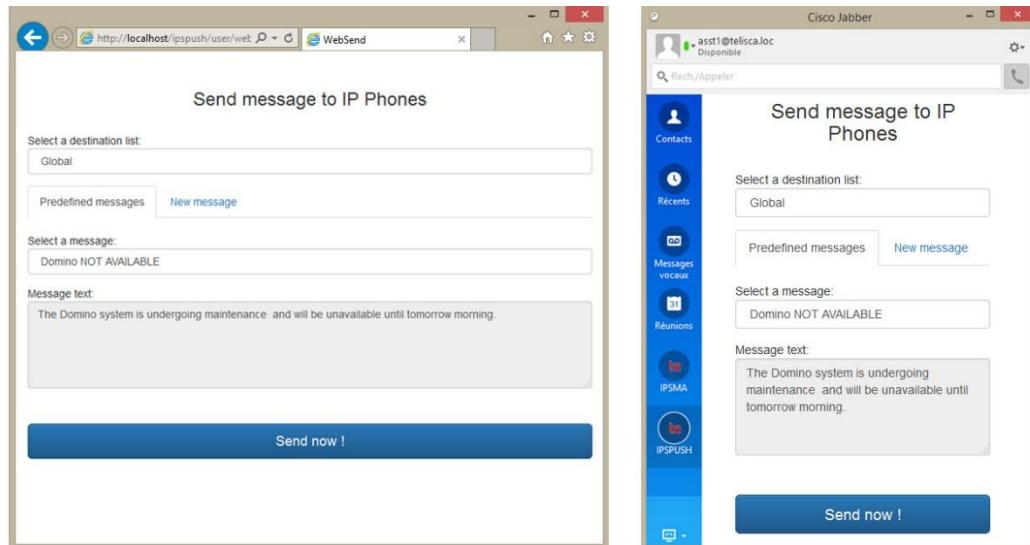
Messages may also be sent directly from the IP Phone
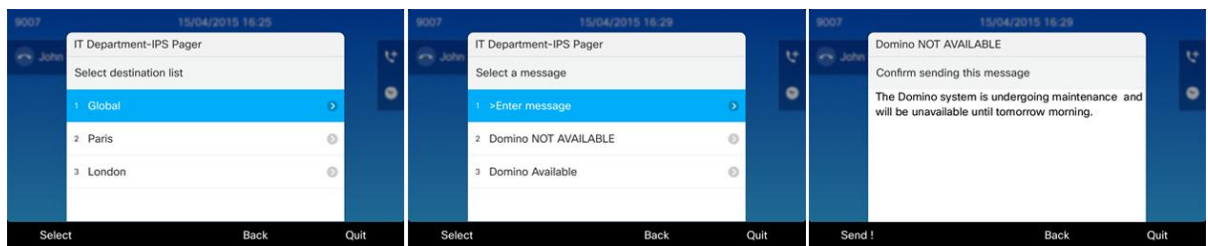
Administrator uses a Web interface to define different user profiles. For each profile, he will defined destination groups. Groups can be defined by: 'locations', 'device pools', 'calling search spaces', 'IP Address ranges', 'IP Phone list', all IP Phones, list of users, list of departments. It is possible to create lists of lists on different CUCM clusters.

	Numéro	Identifiant	Description	Profil (CSS)	Pool	Location	Type	Adresse
☐		SEP000000000000			Default	Hub_None	Third-party SIP Device (Basic)	192.168.0.70
☐		SEP64168DBA800F	7301 (9951)		Default	Hub_None	Cisco 9951	
☐		SEP0024C4FEACA0	Auto 7013		Default	Hub_None	Cisco 7906	
☐	70140001	SEP0013C412C578	Auto 7014		Default	Hub_None	Cisco 7960	192.168.0.18
☐		SEP0019306FB9D4	Auto 7015		Default	Hub_None	Cisco 7931	
☐		SEP0016C76B2B04	Auto 7016		Default	Hub_None	Cisco 7961	
☐		SEP000F8F28DAE9	Auto 7017	VMRestrictedCSS	Default	Hub_None	Cisco 7970	192.168.0.25
☐		SEP001E4A92235B	Auto 7019		Default	Hub_None	Cisco 7942	
☐		SEP000FF76E3C56	Auto 7020		Default	Hub_None	Cisco 7940	
☐		SEP0024E8A7955B	Auto 7025		Default	Hub_None	Cisco IP Communicator	

It is possible to send text messages and presentation automatically at different time of day, depending of the week day. Multiple time zones are taken into account for large CUCM clusters.

Jours de travail ☑ Lundi ☑ Mardi ☑ Mercredi ☑ Jeudi ☑ Vendredi ☐ Samedi ☐ Dimanche

Ajouter

		Groupe éditorial	Listes de destinations	Messages/Présentations	Horaire 1	Horaire 2	Horaire 3	Horaire 4	Horaire 5
Modifier	Supprimer	User message	TEST	Slide show	15:36				
Modifier	Supprimer	GROUP2_5	LIST1	PAGE 1	15:34	15:39	16:10		

Administrator defines different messages ready to be sent. Two level or priority may be set for: Push priority, notification sound, display duration.

Administrator can check, messages sent history, including sender, profile, destination, messages sent and destination list detail.

With optional module, IPS Pager can also send SMS to mobile phone via Cloud Service like Esendex. IPS Pager can also send message to ASCOM DECT phones.

REST APIs are available to send automatically text messages to destination groups, from a third-party application.

1.4 Optional Audio Broadcast Module

The user may speak directly from his IP phone to recipients by launching the IPS Pager service. According to his profile, he will have access to several distribution lists.
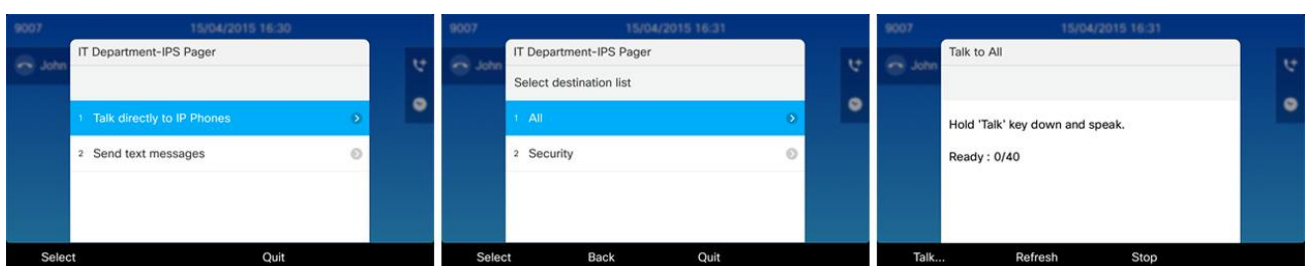
Audio is played on Cisco IP Phones's speakers. If the IP phone is engaged on a call, the internal calling party will hear the audio message from the handset. It is also possible to play audio message on IP Speakers, like CyberData or Algo Solution SIP-enabled IP Speaker.

Web map user interface allows to select the destination of the audio announce on a map. It also offers a Dashboard to view live operating statistics.
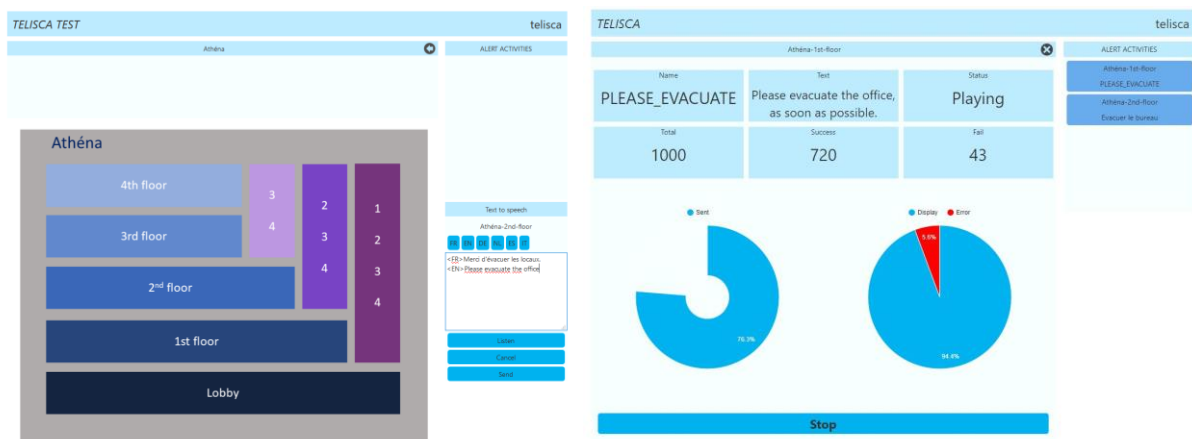
1.5 Optional DECT and SMS messaging

IPS Page supports an (optional) interface for sending messages toward ASCOM DECT telephones.

IPS Pager supports an (optional) interface with an SMS Cloud service (licke Esendex) for sending messages toward mobile telephones.

IPS Pager is available in French and English.

1.6 Requirements

Supported Cisco CUCM versions: 10.5, 11.5, 12, 12.5, 14

Supported Cisco IP Phone 6921, 6941, 6961, 7811, 7821, 7841, 7861, 7905, 7911, 7912, 7920, 7921, 7940, 7941, 7960, 7961, 7970, 7971, 8811, 8841, 8845, 8851, 8861, 8865, 8941, 8945, 8961, 9951, 9971, IP Communicator.

For audio announcements, a multicast IP address must be available between IP Phones and between the server and the IP Phones for pre-recorded audio messages,

Windows servers supported:
- Windows Server 2012 R2 Essentials or Standard
- Windows Server 2016 Essentials or Standard
- Windows Server 2019 Essentials or Standard
- Windows Server 2022 Standard

- Minimum configuration: 1 vCPU, 4GB RAM, 70GB disk
- Virtual Machine VMware vSphere, Hyper-V or Cisco UCS, Cisco UCS-E
- Cloud ready
- CyberData or Algo Solution IP speakers.

1.7 List of supported IP Phones

1.7.1 Text messages

IP Phones models	Send immediately
7940, 7960	Supported
7937	Supported
7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, 7975, IP Communicator	Supported
7921, 7925, 7926	Supported
7811, 7821, 7841, 7861	Supported
6921, 6941, 6945, 6961	Supported[1]
8811, 8841, 8845, 8851, 8861, 8865	Supported
8941, 8945	Supported
8961, 9951, 9971	Supported

1.7.2 Audio broadcasts

IP Phones models	Send immediately
7940, 7960	
7937	
7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, 7975, IP Communicator	Supported
7921, 7925, 7926	Supported
7821, 7841, 7861	Supported

[1] Does not support authentication proxy redirection (use CTI or Associated User instead)

6921, 6941, 6945, 6961	Supported[2]
8811, 8841, 8845, 8851, 8861, 8865	Supported
8941, 8945	Supported
8961, 9951, 9971	Supported

Note: For some IP Phones models, it may be necessary to upgrade the firmware to get the maximum compatibility:

- 69XX : firmware 9.3.1.3
- 894x : firmware 9.3.2.11
- 896X et 99X : firmware 9.3.2.10

2 Administration

2.1 Parameters

IPS Pager can send text messages, directly on the phones and makes it possible to speak directly into the speakers of many phones (voice announcements). Some parameters apply only to text messaging, while others apply only to voice broadcasting.

One must first define Editors groups. Then for each of these groups, one must create (for Text and voice) recipient's lists and message templates (for text only). Each group will have access to its own set of recipients and message templates (for text only).

Text messages can be sent from the phones, from a web interface, or through a special URL.
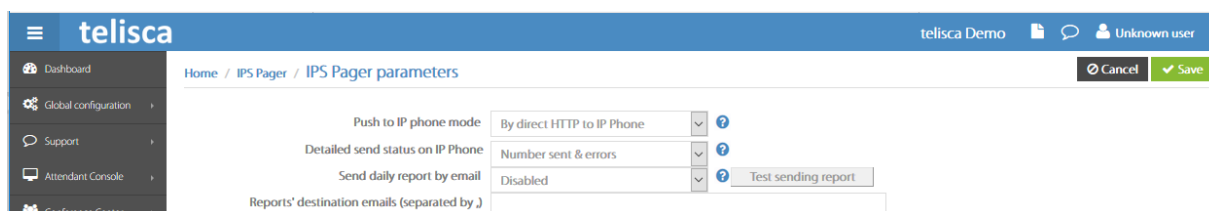
2.1.1.1 Push to IP Phone mode

This is where you choose if you want to send text messages by HTTP or by CTI.

- **CTI** means **easier configuration** for the phones, no authentication URL change, and no need to enable Web Access for each device.
  Please refer to Telisca Framework guide on how to configure the CTI component. For IPS Pager "Automatic and Dynamic Monitoring" mode is supported.
- **HTTP Push** is recommended to send a lot of messages simultaneously (or setting a lot of IP Phones ready to listen an audio broadcast) and when enabling Web Access is not an issue. It is recommended for more than **200 simultaneous messages** sent.

Note: CTI Push needs to configure CTI, in the administration, Global Config menu, CTI Config folder. See Install & exploitation guide **IPSCFG_ADMIN_EN.pdf** *for more information.*

Note: HTTP Push needs a special CUCM configuration in order to allows the push on the IP Phone using minimum CUCM resources. **See Deployment / Authentication Proxy setting.**

2.1.1.2 Show detailed send status on sender's IP Phone

This displays a report from the sender phone (or the web interface) including the number of text messages (or voice listening commands) sent, the number of errors, and the total expected.
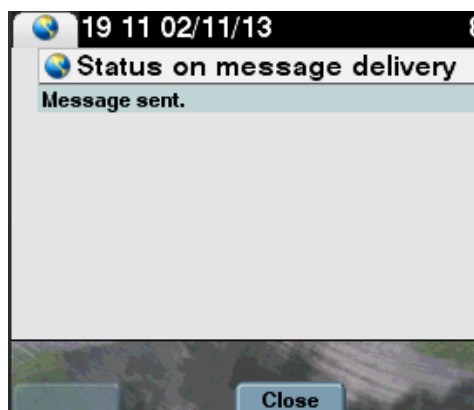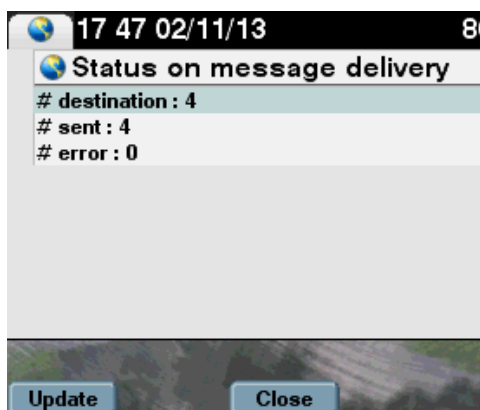
Send status on IP Phone   *No status on phone*

Note: it may be necessary to press the "Update" button to refresh the numbers.

2.1.1.3   Check IP Phone has really displayed the message

Improves the accuracy of the Message delivery report by counting the requests to display the message from the recipient's phones rather than counting the number of message pushes made by the telisca server. This will delay the sent report slightly. This will increase significantly the server load.

2.1.1.4   Save send history

You can select to save the send history which will then show in the send history report.

2.1.1.5   Send history retention (days)

Messages tagged to be kept on the phones, can be browsed by the recipients subscribed to the IPS Pager service for messages browsing. This defines how long the messages will be kept.

2.1.1.6   Send mail report

An email report mode can be :
- Disabled
- everyday
- every day if not empty

The email reports are sent every day at 23:50 server time. When the email report option is selected, a new tab is activated in Global Configuration, which lets you enter your SMTP server details.

The recipient of those reports are the email addresses entered, separated by commas.

The SMTP Configuration tab in Global Config
The Sender email address is used as the recipient for the daily reports

The emails contain the following information:

```
SEND_DT;GRP_NAME;SENDER;MSG_TITLE;LST_NAME;NB_DEST;NB_SENT;NB_DISP;NB_ERROR;
```

2.1.2 Text settings

2.1.2.1   Message title enabled

When sending a text message, it is possible to enter a title (approx 30 characters long) which will display in the message differently depending on the phone model.
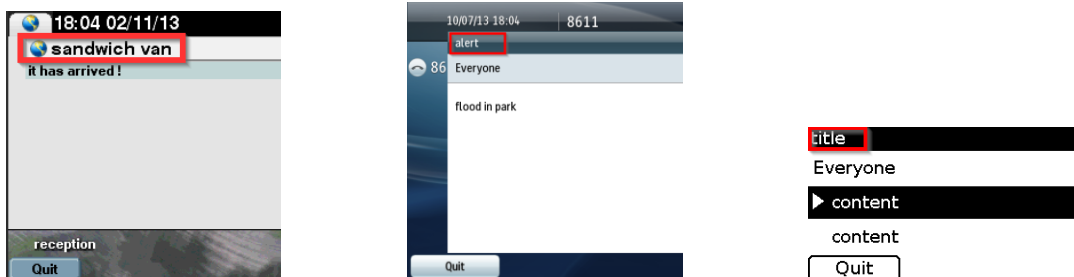
What a message title looks like on CIPC, 8945, 6941 phones
(message title highlighted)

2.1.2.2   Add sender phone number to message

This option allows the receivers of the message to view the sender's editor group's name or the parameter **User** set in the phone subscription if it is set (like a number or a name).

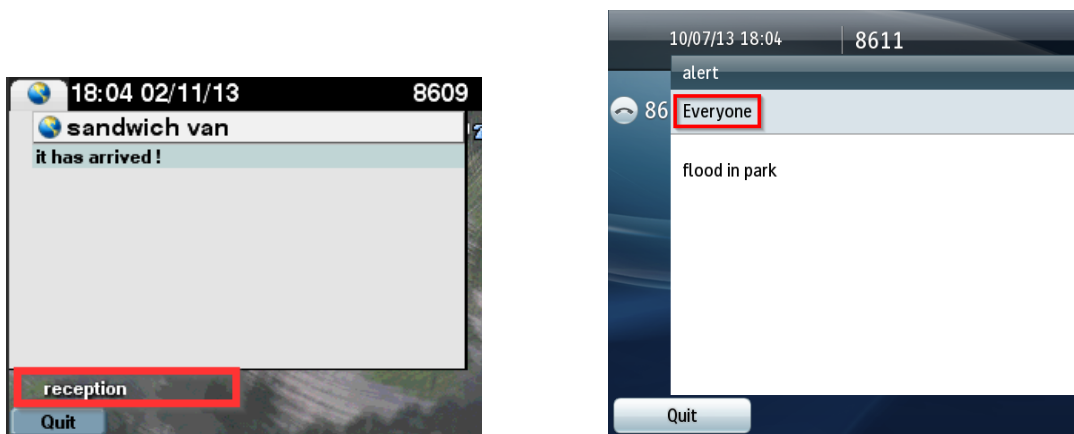*Sender's editor group name is displayed in the messages*
(screenshots from CIPC and 8945 phones)

2.1.2.3   Normal or high priority profiles

We can define two priority levels for our messages. The priority level can be selected when sending a message from the web or set for each predefined messages.
Both priorities behave the same way: they are pushed immediately regardless of the status of the recipient (on a call or idle), the only use is to play a different audio notification and to have a different display duration.

Each priority level has two characteristics:

- **Push Message Priority**
  Immediately: for audio, mixes into the user's current conversation if on a call – only to be used for Urgent messages
- **Alarm file** (optional): note that the alarm sound is played immediately, regardless of the destination communication status. It is less distracting to only use sound notification for High Priority types of messages.
- **Clear message delay**: defines how long the message displays on the phone before it closes automatically.

2.1.2.4   Session duration

Defines how long the IPS Pager menu on the phone will stay opened if there is no user input.

2.1.2.5   Nb. of simultaneous push

The number of simultaneous message pushes defines the speed at which IPS Pager will send simultaneous HTTP or CTI Commands. For large destination groups or when users send messages simultaneously, this limit can be increased. The capacity is limited by the version of CallManager used.

The average sending time of one message on a phone is 500ms. The number of messages sent per second is approximately double the simultaneous push limit.

This speed depends on telisca server's and CUCM's capacities. The speed is also affected by whether or not a sound file from the TFTP server is played and whether an authentication proxy is used.
Tests show that the maximum number of supported is 200 to 500, depending of the IPS Pager server performance.

2.1.2.6   Nb. of simultaneous push with alarm file

Maximum number of simultaneous push of text message plus alarm notification file sent to IP phone is 200 to 250, depending of call manager TFTP server performance.

2.1.2.7   Reload phones IP address before sending

When pushing by http, the selected phones IP address may be reloaded (200 by 200) using serviceability API before sending, to take into account a possible DHCP reset. However, the serviceability API is limited to 15/mn, so this limits the number of phones that could be pushed to 3000/mn. If a higher rate of push is required by settings a high number of simultaneous push, this option may be unchecked.

2.1.2.8   Stop phone screensaver

Will get the phone out of standby and display the message when screen saver is active. Not all the phone models support this feature.

2.1.3 Audio settings

These parameters are enabled only if the license installed does have the IPS Pager Audio option. Audio option allows to play live and recorded audio broadcast. When using a live audio broadcast from the IP Phone, the user press and hold the "Talk" soft key on the phone when speaking (Talky Walkie mode).

Note: IPS Pager used to have an option to use a start-stop mode, which has been removed.

2.1.3.1   Pager mode (audio) clear delay

When the sender releases the "Speak" button on the sending phone, the destination phones stop listening to the RTP stream. If there is an error during this action, a security timer will force the destination phones to stop listening after the number of seconds defined here.

This delay must be more than the maximum RTP stream broadcast length or recorded audio file length to avoid cutting off the broadcast before the end.

If the value is set to zero, it disables this feature. However, by default IPS Pager's application will recycle at 5:30am so the audio broadcast will then be stopped.

2.1.3.2   RTP Volume

It is possible to set the audio volume on the IP Phone Speaker or let the user settings. This define the default value when creating an audio announce in the administration or from WebSend or Map user interface.

2.1.3.3   Default delay before playing recorded audio

When starting an audio alert, the application push to all the destination phone to set them in a mode where they listed to the multicast IP. If there is a large number of phones, this make take a few seconds. Depending of the purpose of the message you may choose to wait for most of the phones to be ready or start as soon as possible the message, which should not be a problem if the message is repeated several times. The value define here is used as a default value when creating an audio content from the administration and as the value used when creating a Text To Speech audio message from WebSend or Map URL

2.1.3.4   Title displayed while playing audio

The title header is the popup screen on the end users' phone.

2.1.3.5   Text displayed while playing audio

In the same window that appears on the end user IP phone with title, you can give a brief description of the announcement being made.

2.1.4   Advanced parameters

2.1.4.1   Authorized IP addresses to send by http

This is the list of IPs authorized to trigger messages by HTTP POST (third party applications), using the IPSPUH/send.aspx URL.

2.1.4.2   Report purge delay

Detailed reports are purged after the defined delay (in days).

2.1.4.3   Authentication's cooky

When Map.aspx URL Windows authentication is enabled (in IIS) and anonymous authentication is disabled, then a IIS/Windows may require to enter a login/password. It is possible to reduce the frequency to enter a credential by saving a Web cooky. The validity duration of the cooky can be defined here.

2.1.4.4   Reports' email subject and body

It is possible to receive an email each time an alert has been triggered. You can configure the subject and content of the email and include the following variables: %ID%, %TIME%, %LIST%, %MSG%, %NB_DEST%, %NB_SENT%, %NB_OK%, %NB_FAILED%, %SENDER%.

2.2 Entities

You can define as many entities (organization, location, building), which will be assigned to groups of people who will be able to send text messages or broadcast voice announcements. Each group will have access to its own set of recipients and message templates.

Add

		ID	Group label	Send by Web page	Key	Group name	Send by IP phone	Key
Edit	Delete	1	Brighton office	Windows user group		security.ou	Free access	
Edit	Delete	2	Ground Floor	Free access			Key required	5432
Edit	Delete	3	First Floor	Free access			Key required	9876
Edit	Delete	4	Paris office	Free access			Free access	
Edit	Delete	5	Reception	Free access			Free access	
Edit	Delete	6	Whole office	Free access			Free access	

Users can define the group label for each group, as it will appear as the title of text messages.

2.2.1.1   Send by Web Page settings (text messaging)

The authentication method to send text messages from the web interface is as follows:

2.2.1.2   No authentication

Anyone with the default URL will be able to send messages

2.2.2 Key required

2.2.2.1   Send by IP Phone

		ID	Group label	Send by Web page	Key	Group name	Send by IP phone	Key
Edit	Delete	1	Brighton office	Windows user group		security.ou	Free access	
Edit	Delete	2	Ground Floor	Key required	5432		Key required	5432

The key may be added to the key parameter in IP Phone Service service subscription.

If a key has been defined for the editor group and is not provided in the IP Phone Service parameter, the user is prompted to enter a PIN code. In this case, make sure that the key entered only contains numerical characters.

2.2.2.2 Send by Web Page

		ID	Group label	Send by Web page	Key
Edit	Delete	1	Brighton office	Windows user group	
Edit	Delete	2	Ground Floor	Key required	5432

This key will have to be added to the URL (key=XXX) to be granted access. From the screenshot above, you will provide Editor Group 2 (Reception) a URL to send their messages like this:

```
http://host/IPSPUSH/user/WebSend.aspx?grp=2&key=5432

http://host/IPSPUSH/user/Map.aspx?grp=2&key=5432
```

Read more about accessing IPS Pager from the web.

2.2.2.3 Windows User Group

This authentication method for the Pager from a web page is based on Windows authentication.

		ID	Group label	Send by Web page	Key	Group name
Edit	Delete	1	Brighton office	Windows user group		security.ou
Edit	Delete	2	Ground Floor	Key required	5432	

The web page will check the Windows login of the user against the Group defined here.

The way the Windows authentication works is this:

1. User provides credentials through browser
2. IIS on telisca server makes sure the user is defined (either locally or on the server's domain)
3. The browser starts opening the page (or not if denied by IIS)
4. Telisca server does a lookup on the AD defined in administration, to find if the user name can be found in the access group
5. Telisca server displays the page with the appropriate options (or an error message if the user cannot be found)

In this mode, it is necessary to force the users to authenticate on IPSPUSH/user/WebSend.aspx and IPSPUSH/user/map.aspx d in IIS.

- Launch IIS Manager, select default Web Site,
- Select the URL IPSPUSH/user,
- Right click and select 'Switch to content display',
- Select WebSend.aspx or Map.aspx,
- Right click and select 'Switch to features display',
- Click on authentication icon
- Disable 'Anonymous authentication'
- Enable 'Windows authentication'

Also, you can configure Internet Explorer to use integrated Windows Authentication to hide the login prompt.

Administrator needs to define AD settigns to be able to check if the users have access to WebSend or Map, and to check it's AD's Security Group.

Important: Once Windows User Group is chosen and the anonymous access is disabled on the page in IIS, all groups will have to authenticate even if they are still showing "Free Access" on the parameters page.

2.2.3 Active Directory parameters for group authentication

If you are using authentication by Windows User Group for the access to IPS Pager web, you must configure the Active Directory section under 'Global config' where the groups and users are defined.

Tip: To check your settings and connectivity, we recommend you use an AD client such as:
- AD Explorer http://technet.microsoft.com/en-us/sysinternals/bb963907

2.3 Text recipients

For each editor group, access can be given to text messages' recipient lists, which are created in this screen.

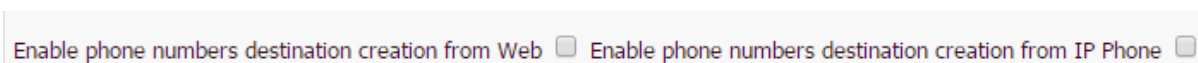This is what the recipients list look like on an IP Phone:
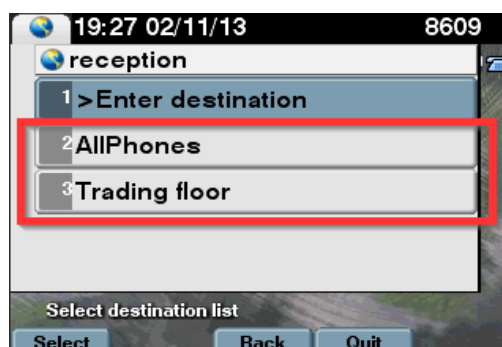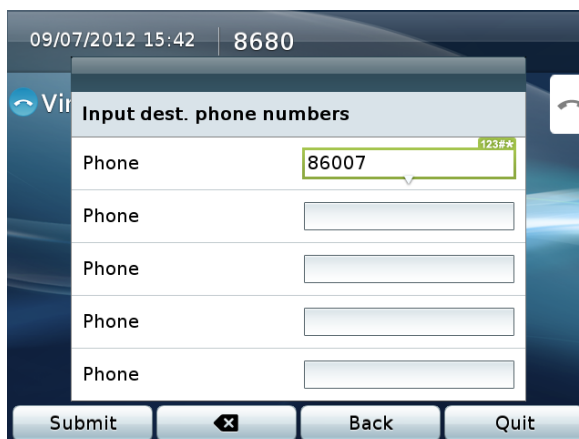
Lists can be populated by using CUCM groups:

- All IP Phones in CUCM
- Calling Search Spaces
- Locations
- Users
- Device pools
- Departments
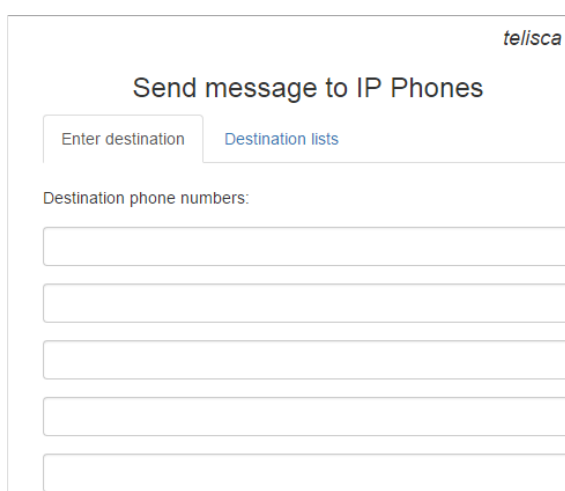- IP Address ranges
- Specific IP Phones
- List of list
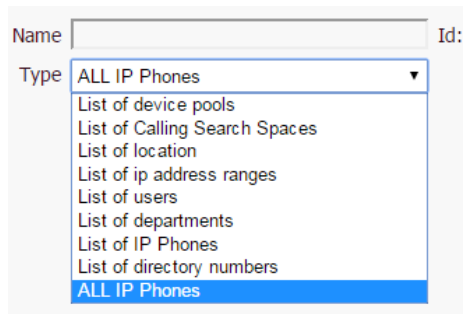
As well as creating these destination lists, Editors can be allowed to enter their message recipients themselves (up to five).

Phone screen with "Enable phone numbers destination creation from IP Phone"

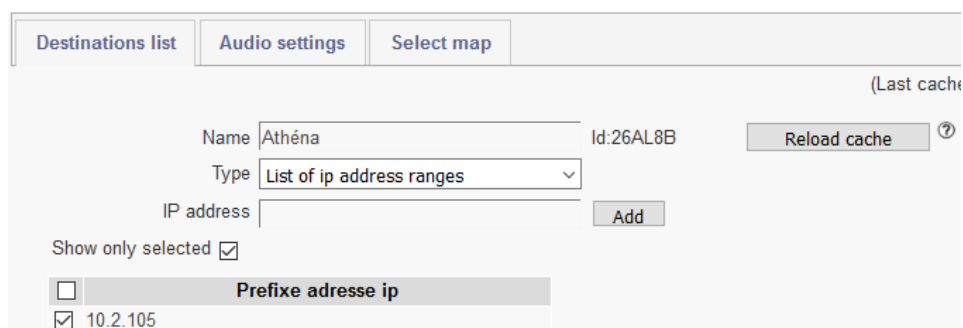*Entering up to five recipient to a text message from a phone*

Ad-hoc recipients from the web page

2.4 Audio recipients

2.4.1 List of destinations

Like for text message destination lists, access can be given to lists of destinations for each editor group.

Lists can be populated by using these CUCM groups or IPs:
- Single Unicast IP: to broadcast to a single destination IP Address
- All IP Phones in CUCM
- Calling Search Spaces
- Locations
- IP Address ranges
- Specific IP Phones

Note: The classifications of the phones are based on the list of phones loaded in cache at different times of day, as defined in CUCM Config screen, in Global Config menu. Just after enabling IPS Pager or after adding phone, it may be useful to reload the cache. However, if thousands of phones' list is reloaded, this can take a few minutes.

2.4.2 Audio settings

This define the different audio modes available from Phone or Web user interface.

2.4.2.1   Audio modes

The audio broadcast can be used in different modes:
- Press to Talk: live broadcast from an IP Phone to one or many, by pressing the Talk Softkey,
- Press to Talk + Talkback: The destination also has a Talk button and can answer to everyone,
- Text to Speech (Web): Create new audio message from WebSend or Map URL
- Recorded audio: allows to play an audio file loaded by the administrator or entered in Text to Speech.

2.4.2.2   Multicast IP Address

Enter multicast IP Address which will be accessible through the phones is available for each destination list. This IP address will need to be routed between the sender IP Phone and other phones (or SIP speakers) for live broadcast and between IPS Pager server and the destination phones for recorded audio.

A typical Multicast IP address is 239.0.0.X.

2.4.2.3   RTP Port

This is the IP Port on which the destination phones will listen the RTP stream. The value needs to be even, between 20480 and 32768.

When sending audio broadcast to SIP speaker the RTP Port can be used to differentiate the different speakers by location.

2.4.3 Association to a map

From the Map URL, it is possible to trigger an alert by clicking on the area of map.

This is possible by selecting a map and a color for each audio recipient.
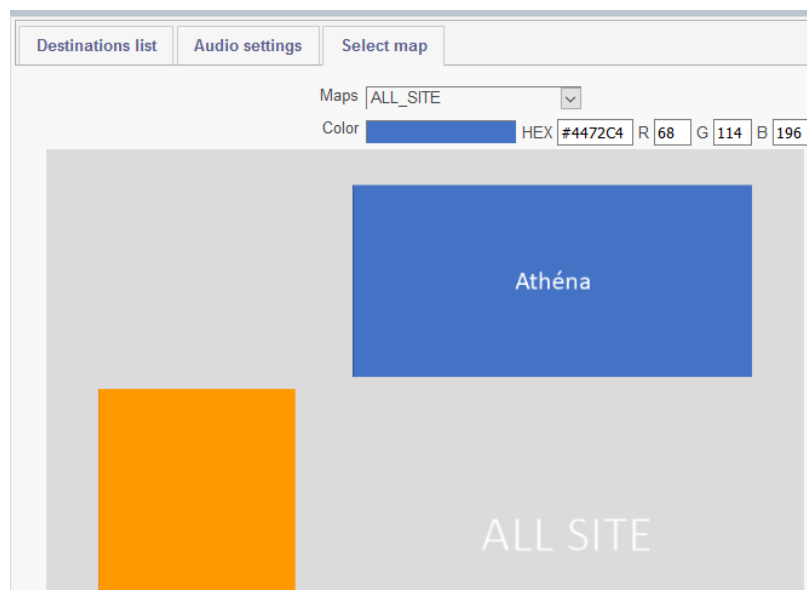
By selecting the map 'ALL_SITE' and clicking on the Athéna blue area, the configuration memorizes the ID of the map and the exact blue color code which will be associated to the audio recipient.

Note: Firefox may have an issue by selecting a wrong color code. The workaround is to change a setting in Firefox, by entering URI 'about:config', then filtering on gfx.color and changing parameter gfx.color_management.mode from 2 to 0.
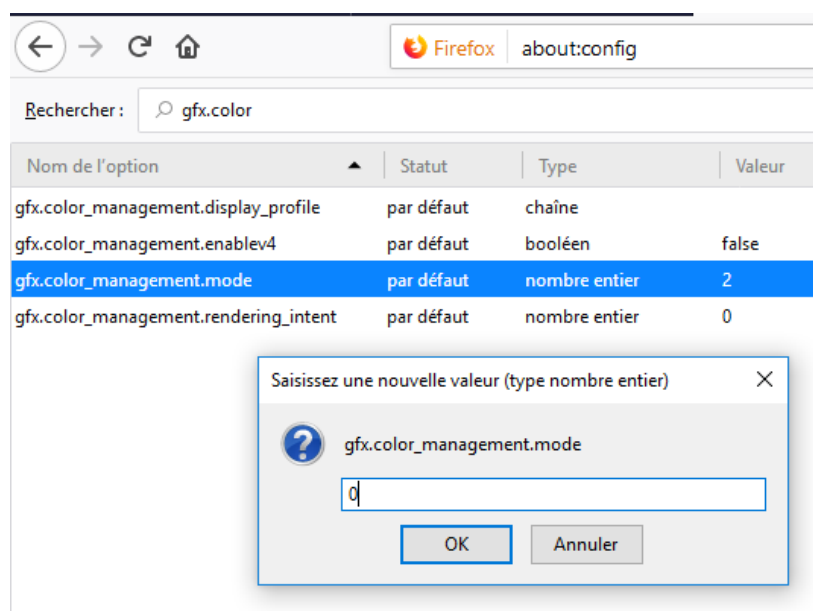
Note: Take care to select an uniform color zone.

2.5 Audio content

ID 26D1HP

Name PLEASE_EVACUATE

Audio volume 30%

Duration or # times 3 times

Delay before audio broadcast 2 sec.

Load recorded audio message

Concatenate languages 1 language

Voices Microsoft Server Speech Text to Speech Voice (en-US, Helen)

Text to speech Please evacuate the office, as soon as possible. Please, do not push or run.   Generate Play

Or upload audio file Parcourir... Aucun fichier sélectionné.

c:\inetpub\wwwroot\IPSCFG\data\IPSPUSH\AUDIO\audio_36.wav

2.5.1.1 Load speaker volume for pager

When left to -1, IPS Pager will play the broadcast without changing the recipients' phone volumes. To make sure everyone will hear the broadcast, it is advised to force the volume to a positive value. Over 80 percent volume is likely to deteriorate the sound quality.

Note: the forced volume only applies to audio broadcasts (not the alarm sound of text messages) and is set only for the duration of the message. It goes back to the original speaker volume after the broadcast.

Note: the voice is in whisper mode when playing over a conversation and cannot be heard by the other users on the call.

2.5.1.2 Recorded audio duration

When sending a recorded audio alert, it is helpful to define the duration of the audio broadcast. The audio file will loop until the duration is met. You can also define the number of times the audio file is played.

2.5.1.3 Recorded audio send delay

Before playing a recorded audio alert and depending the number of IP Phones destination, it may be necessary to wait a few seconds so that all or most of the IP Phones are ready to listen the audio broadcast on the multicast IP address. The delay will also vary depending of the number of parallel push defined in IPS Pager global parameters, so it is convenient to define a percentage of IP Phones ready. If you want to play an urgent message which loops you can play immediately.

2.5.1.4 Text to speech

You can enter a text to speech in up to four languages. They are merged together. It is possible to include XML markup language to change the way the text is played.

Can use the Speech Synthesis Markup (SSML) to control various characteristics of synthetic speech (text-to-speech) output including pitch, rate, volume, pause.

- break: An empty element used to control the prosodic boundaries between words.

Syntax: `<break time="xxxms" />` or `<break time="xxxs" />`

- emphasis: Increases the level of stress with which the contained text is spoken.

Syntax: `<emphasis level="strong|moderate|none|reduced">text-to-speech</emphasis>`

- prosody: Controls the pitch, rate, and volume of the speech output.

Syntax: `<prosody pitch="value" contour="value" range="value" rate="value" duration="value" volume="value">text-to-speech</prosody>`

-pitch: Optional.Indicates the baseline pitch for the contained text.This value may be expressed in one of three ways:

An absolute value, expressed as a number followed by "Hz"(Hertz). For example, 600Hz.

A relative value, expressed as a number preceded by "+" or "-" and followed by "Hz" or "st", that specifies an amount to change the pitch. For example + 80Hz or - 2st. The "st" indicates the change unit is semitone, which is half of a tone(a half step) on the standard diatonic scale.

An enumeration value, from among the following: x - low, low, medium, high, x - high, or default.

-contour: Optional. Represents changes in pitch for speech content as an array of targets at specified time positions in the speech output. Each target is defined by sets of parameter pairs, for example:

`<prosody contour="(0 %, +20Hz)(10 %, -2st)(40 %, +10Hz)">`

The first value in each set of parameters specifies the location of the pitch change as a percentage of the duration of the contained text (a number followed by "%"). The second value specifies the amount to raise or lower the pitch, using a relative value or an enumeration value for pitch, see above.

-range: Optional.A value that represents the range of pitch for the contained speech content.This value may be expressed using the same absolute values, relative values, or enumeration values used to describe pitch, see above.

-rate: Optional. Indicates the speaking rate of the contained text. This value may be expressed in one of two ways:

A relative value, expressed as a number that acts as a multiplier of the default. For example, a value of 1 results in no change in the rate. A value of .5 results in a halving of the rate. A value of 3 results in a tripling of the rate.

An enumeration value, from among the following: x-slow, slow, medium, fast, x-fast, or default.

-duration: Optional. A value in seconds or milliseconds for the period of time that should elapse while the speech synthesis (TTS) engine reads the contents of the element. For example 2s or 1800ms.

-volume: Optional. Indicates the volume level of the speaking voice. This value may be expressed in one of three ways:

An absolute value, expressed as a number in the range of 0.0 to 100.0, from quietest to loudest. For example, 75. The default is 100.0.

A relative value, expressed as a number preceded by "+" or "-" that specifies an amount to change the volume. For example +10 or -5.5.

An enumeration value, from among the following: silent, x-soft, soft, medium, loud, x-loud, or default.

You can test a generated text to speech message by clicking on the 'Generate' link they downloading the audio file by clicking on the 'Play' link.

2.5.1.5 Load an audio file

Instead of using the text to speech it is possible to load a ready-made audio file. Most of the format are supported and are converted to adequate format to be played by telisca audio server.

2.6 Messages/Presentations tab

Users can create predefined text messages from this page.

Note: The number of characters displayed in messages on IP Phones is limited by the Cisco XML Object which supports up to 4000 characters.

2.6.1 Enable message creation

This option lets the users of the web or phone interface type in an ad-hoc text message rather than selecting from the predefined list.

Message creation menu enabled on phone
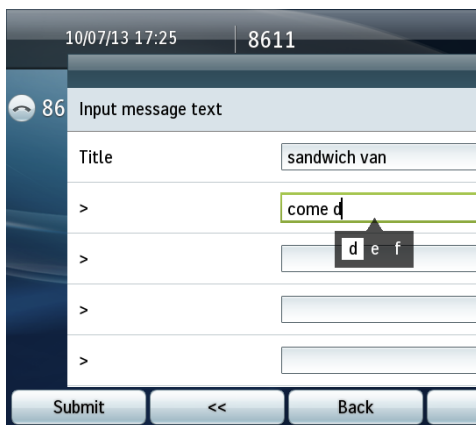
Composing a message on a 8945

Composing a message on a 6941

New message enabled from the web interface

2.6.2 Adding a new predefined message/presentations

First select the editor group for which you want to create a predefined message.

Then you can choose to create ad-hoc messages/presentations from the web and from the phones for this group.

And add a new message or edit an existing one.

When selecting to 'Add' a new message you can either choose a 'Text message' or 'Slide show'

2.6.3 Text message

So with the text message, you can add a title and text to your message, set a priority level and make it available to recipients subscribed to IPS Pager history service.

2.6.3.1   Recipient message history

Gives access to the message history on a destination phone. The phone needs to be subscribed to the IPS Pager service with the Mode parameter left empty. And the message history option enabled.

2.6.4 Slide show

So with the slide show type, you can add a title then you need to load up the images you require.

Choose to load a zip file which contains the images in the following format: png, jpg or gif
From a PowerPoint presentation you can choose to export this into one of those formats.

You can choose to delay in seconds between the slides, a value of 0 will disable the automatic slide show.

Choose the number of times the slide show will loop or a value of 0 will disable this loop.

Setting a priority level and make it available to recipients subscribed to IPS Pager service.

You have the option to assign a title for this slide show.

2.6.5 Text & graphic slide show

With the text and graphic slide show type, you can add a title then you can load up a mixture of text and images you require.

You can choose to delay in seconds between the slides, a value of 0 will disable the automatic slide show.

Choose the number of times the slide show will loop or a value of 0 will loop indefinitely.

Setting a priority level and make it available to recipients subscribed to IPS Pager service.

The 'Visible for display' option will allow all IP phones subscribed to IPS pager server (without any command parameter), to display all the messages defined in the specified editor group, for which this option has been enabled.

If the title option has been enabled in IPS Pager parameters, you can then add a slide title for the presentation. Select the slide type, if text, input text into the text window below then hit 'add'.

For image, select 'image' then choose file, then hit add. Choose to load a zip file which contains the image in the following format: png, jpg or gif.

You can add several pages, remove and add along the way then once complete hit validate.

2.7 Scheduled push

With the scheduled push feature, you can send pushes of a message/presentation 5 times a day and select the days of the week for when you want this to be scheduled.

Working days ☑ Monday ☑ Tuesday ☑ Wednesday ☑ Thursday ☑ Friday ☑ Saturday ☑ Sunday

Add

		Editor group	Destination list	Messages/Presentations	Time 1	Time 2	Time 3	Time 4	Time 5
Edit	Delete	TEST	All_phones	TEST	15:15	08:00	10:45		

The selection options you have when adding a schedule via the 'add' link are:

Editor group
Destination list
Message/Presentations
Times of the day x5

Hit the 'update' link to add the scheduling and then you can create several schedules.

To remove a schedule, simply hit the 'delete' button for the relevant schedule.

Currently the working days cannot differ for each schedule, this selection will apply to all of the schedule pushes.

The time of day the message will be pushed takes into account the difference between the time zone of the IP Phone's device pool and IPS Pager's time zone.

2.8 Send history

Contains a report of sent messages with details.

Send date	Editor group	Sender	Message	Destination list	Dest nb	Sent nb	Recv nb	Err nb	Detail
11/02/2013 16:53	reception	SEP000C298087E5	urgent message	AllPhones	3	3	2	1	Detail
11/02/2013 16:47	reception	SEP000C298087E5	urgent message	AllPhones	4	4	2	2	Detail
11/02/2013 16:37	reception	SEP002414B37A58	urgent message	AllPhones	3	3	3	0	Detail
11/02/2013 16:27	reception	SEP002414B37A58	sandwich van	AllPhones	3	3	3	0	Detail
11/02/2013 16:25	reception	SEP1CE6C79A0FA3	urgent message	AllPhones	3	3	2	1	Detail
11/02/2013 16:23	reception	SEP1CE6C79A0FA3	urgent message	right	2	2	1	1	Detail
11/02/2013 16:22	reception	SEP1CE6C79A0FA3	urgent message	right	2	2	1	1	Detail
11/02/2013 14:10	reception	10.5.1.87	urgent message	right	2	2	1	1	Detail
< >									

When sending from an IP Phone Sender can contain a user provided in the 'user' parameter when subscribing to CUCM IP Phone Service.

When sending from the Map Web Interfaced, if Windows authentication has been enabled then it can be the user login, otherwise it will show the IP address of the PC.

Reports are purged delay is defined in IPS Pager menu, Parameters' Tab, advanced parameters section.

3 Deployment

Once everything is configured, users need to create the service on Call Manager to enable the IPS Pager interface on the phones. The user can then subscribe the broadcasters' phones to the service and set the parameters to customise access level, text and/or voice capability, and which destinations accessible.

3.1 Define the service on Call Manager

Can be done from telisca Admin > Global Config > Install Services Tab. It is only useful if IPS Pager is used to browse information. If Pager used only for alerting this is required only to send the alert from the IP Phone.
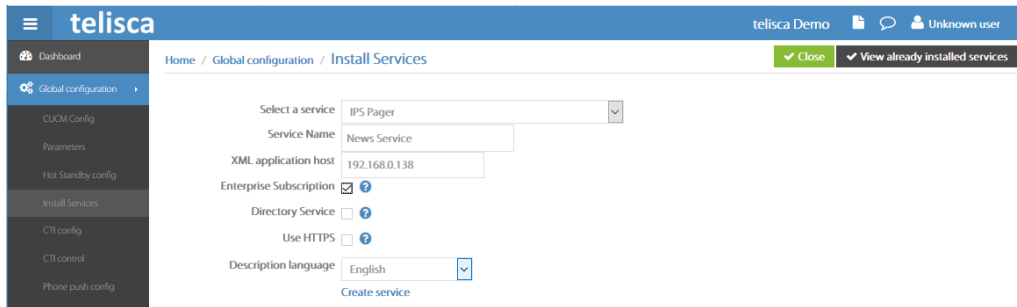
The IP phone service created has several parameters.
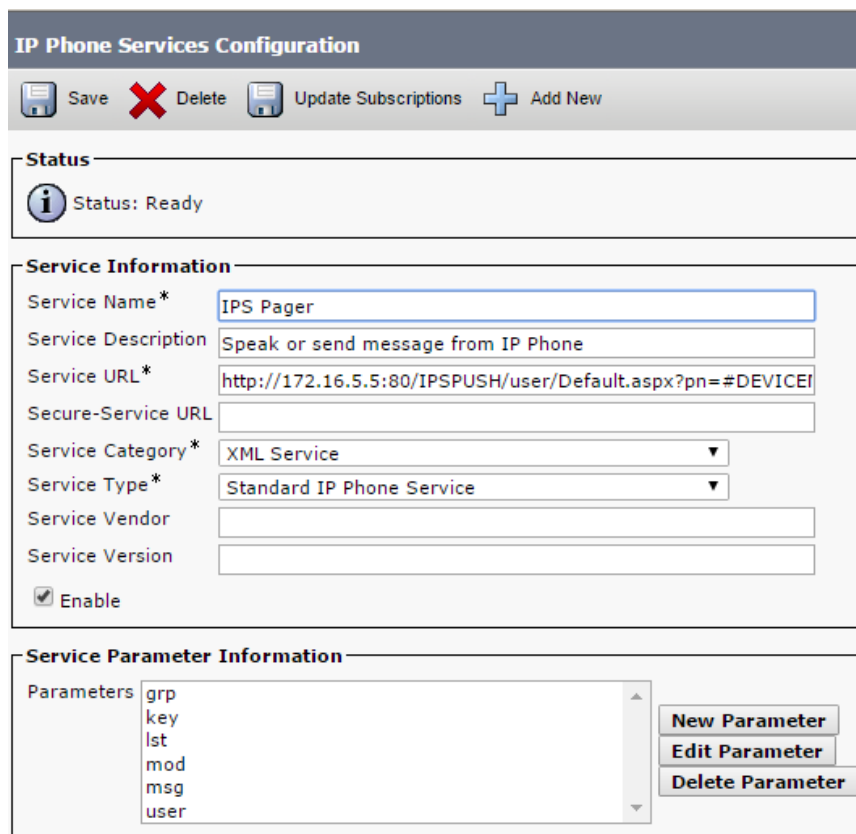
The Service URL is:
`http://[TELISCA_SERVER_IP]:80/IPSPUSH/user/Default.aspx?pn=#DEVICENAME#`

pn

Is the phone name. If the service is subscribed, you can use the dynamic value #DEVICENAME# which is replaced by CUCM with the calling's IP Phone name. If accessing the service by URL (for instance a SURL) depending on the model and firmware, it may be necessary to enter the phone name because #DEVICENAME# isn't supported. See troubleshooting

3.2 Set IP Phone Service Parameters

Each phone that will be used to send an alert will subscribe to the IPS Pager service and have parameters customised for the individual user.

3.2.1 grp: entity index

The entity index (1 to n) can be found in the list of entities

		ID	Entity name	Web page authorization	Key for Web URL	AD Security Group	IP phone authorization	Key for IP Phone
Edit	Delete	1	Global	Windows user group		pager-admin	Key required	12345
Edit	Delete	2	LYON	Free access			Free access	

key (optional depends on config)
If an access key is defined to send by IP Phone.

		ID	Entity name	Web page authorization	Key for Web URL	AD Security Group	IP phone authorization	Key for IP Phone
Edit	Delete	1	Global	Windows user group		pager-admin	Key required	12345
Edit	Delete	2	LYON	Free access			Free access	

If an access key is defined but this parameter is empty, then the user is prompter to enter a PIN code. If the user fails to enter the PIN code after 7 attempts, the user interface is locked for 5mn.
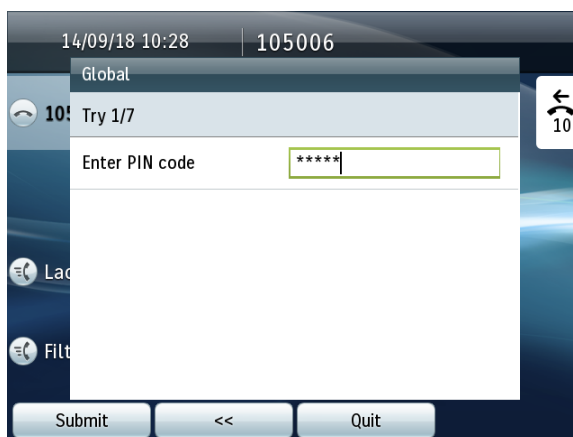
3.2.2 mod : mode

Defines if IPS Pager will only be used for texting, voice or both
- PG: audio broadcast (talk to speech or recorded audio messages)
- MSG: text messages
- S: audio and text messages
- if left empty, users can access the text messages defined in persistent mode. This option must be enabled first.

3.2.3 lst: optional list ID

The recipient (text or audio) list ID(s) if needs to be restricted. Can be found in text or audio recipients list.

		Destination lists	Type	ID
Edit	Delete	ALL_SITE	ALL IP Phones	26AL5K
Edit	Delete	Athéna	List of ip address ranges	26AL8B
Edit	Delete	Athéna-1st-floor	List of IP Phones	26ALBT
Edit	Delete	Athéna-2nd-floor	List of IP Phones	26ALDD
Edit	Delete	Athéna-Lobby	List of IP Phones	26ALAX
Edit	Delete	Hestia	List of device pools	26AL9L
Edit	Delete	Poséidon	List of ip address ranges	26AL6M

Multiple IDs can be provided separated with a comma.

If a destination list is predefined, then the user goes directly to the select text or audio content screen.

3.2.4 msg: optional text or audio message ID

The text or audio message template ID. Using a SURL Button associated to the IPS Pager

		Audio message	Description\Text to speech	ID
Edit	Delete	PLEASE_EVACUATE	Please evacuate the office, as soon as possible. Please, do not push or run.	26D1HP
Edit	Delete	SECURTIY-OUT	Please evacuate, all of you!	26ALNW
Edit	Delete	WELCOME	Welcome everybody in our office. please, do not escape!	27TYJP

IP Phone Service with the lst and msg parameter predefined, the the user go directly to the confirmation screen.
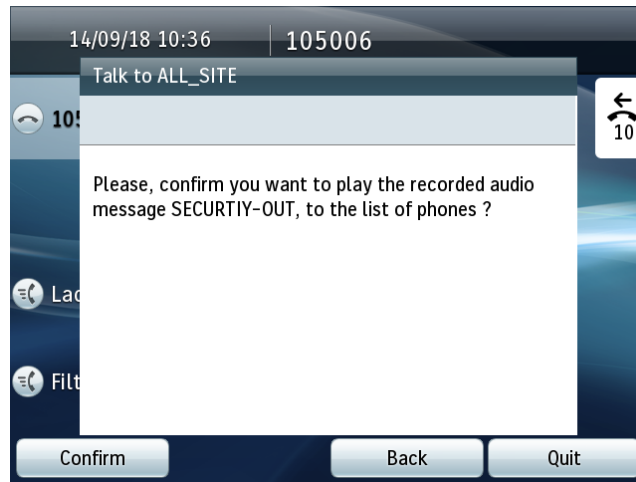
3.2.5 user: optional send user name

Used to display the sender's name or number as a signature of a text message. See here for the option

3.3   Service URL buttons

It is possible to use Service URL buttons to trigger directly an alert to a predefined destination and optionally with a predefined alert message (text or audio). This can be done by subscribing several time to the IPS Pager IP Phone Service and by setting different parameters. Then associate the SURL button to the different IP Phone services.
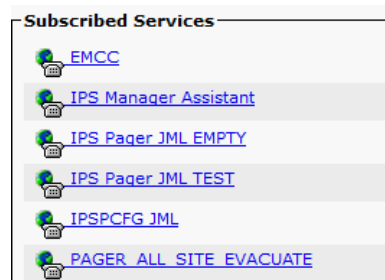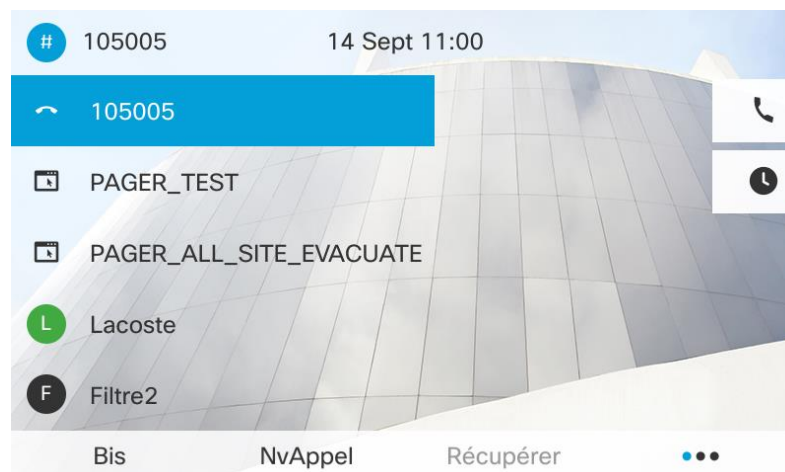
3.4 Push by http settings

This mode can be set in IPS Pager Parameters screen. IPS Pager send http request directly to the IP Phones. It provides the best performances, however it's configuration is more complex. See the steps below.

3.4.1 Set Web enabled option on the IP Phones

In CUCM administration, set the parameter 'Web Access' to 'Enabled' for every destination phones.

Web Access*	Enabled	☑

This can be changed globally by setting it in 'Common Phone Profile' and applying the updated Common Phone Profile by CCMBAT.

Note: Web Access is secured by our 'Authentication Proxy' (see after).

3.4.2 Change IP Phone's authentication URL

The authentication URL needs to be changed to redirect the authentication request to IPS Pager server instead of CUCM.

This can be done for test on the device setting or globally in Enterprise parameters.

External Data Locations Information (Leave blank to use default)

Information	
Directory	
Messages	
Services	
Authentication Server	http://IPSPAGER:80/IPSCFG/authenticate/default.aspx
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	http://IPSPAGER:80/IPSCFG/authenticate/default.aspx
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

The new URL should be http://ipsPagerHost:80/IPSCFG/authenticate/default.aspx . If HTTPS support has not been configured in telisca and CUCM (see Install & Exploitation guide IPSCFG_ADMIN_EN.pdf), then you can force the HTTP URL on the Secured Phones URL Parameters as well.

Enterprise Parameters Configuration

Save   Set to Default   Reset   Apply Config

Phone URL Parameters

URL Authentication	http://IPSPAGER:80/IPSCFG/authenticate/default.aspx
URL Directories	http://cucm11.telisca.loc:8080/ccmcip/xmldirectory.jsp
URL Idle	
URL Idle Time	0
URL Information	http://cucm11.telisca.loc:8080/ccmcip/GetTelecasterHelp
URL Messages	
IP Phone Proxy Address	
URL Services	http://cucm11.telisca.loc:8080/ccmcip/getservicesmenu.j

Secured Phone URL Parameters

Secured Authentication URL	http://IPSPAGER:80/IPSCFG/authenticate/default.aspx
Secured Directory URL	https://cucm11.telisca.loc:8443/ccmcip/xmldirectory.jsp
Secured Idle URL	
Secured Information URL	https://cucm11.telisca.loc:8443/ccmcip/GetTelecasterHel
Secured Messages URL	
Secured Services URL	https://cucm11.telisca.loc:8443/ccmcip/getservicesmenu

The IP phones need to be restarted to take into account the change.

3.4.3 telisca Push setting configuration

From Global Configuration menu, Push Config folder, select the Push mode 'User secure authenticate proxy'.

The user login and password is disabled. IPS Pager will generate a one-time user and password.

You can configure the CUCM Host to the CUCM Publisher, so that the authentication is redirected to CUCM if the user is not the one pushed by a telisca application.

The authentication is normally very fast (a few milliseconds) however is the server is 100% busy by pushing with a too high number of threads, it can be usefull to accept a few seconds for the timeout.

You can test the authentication by entering a phone name (MAC address), after validating the configuration. It should display the services menu on the phone. If this does not work, please check the Web Access settings and authentication URL. If it still does not work, you can check the authentication logs from Support Menu, Application logs folder. If you do not see the authentication request in the logs, this may be a problem with one of the previous CUCM settings.

4 Usage

4.1 From the IP Phones: IP Phone Service

The senders' phones need to be subscribed to the Phone Service on Call Manager with the appropriate settings.

Accessing IPS Pager from the Service Button

4.2 From WebSend web interface


```
http://host/IPSPUSH/user/WebSend.aspx?grp=XXX&key=XXX
```

- **grp** is the editor group ID
- **key** is only required if defined as authentication from Web

If grp parameter is empty, the first editor group (1) is used.

If key parameter is empty, either a free access or an authentication by Windows Security group need to be configured. More info on Windows Authentication for this page.

4.3 From Map Web interface
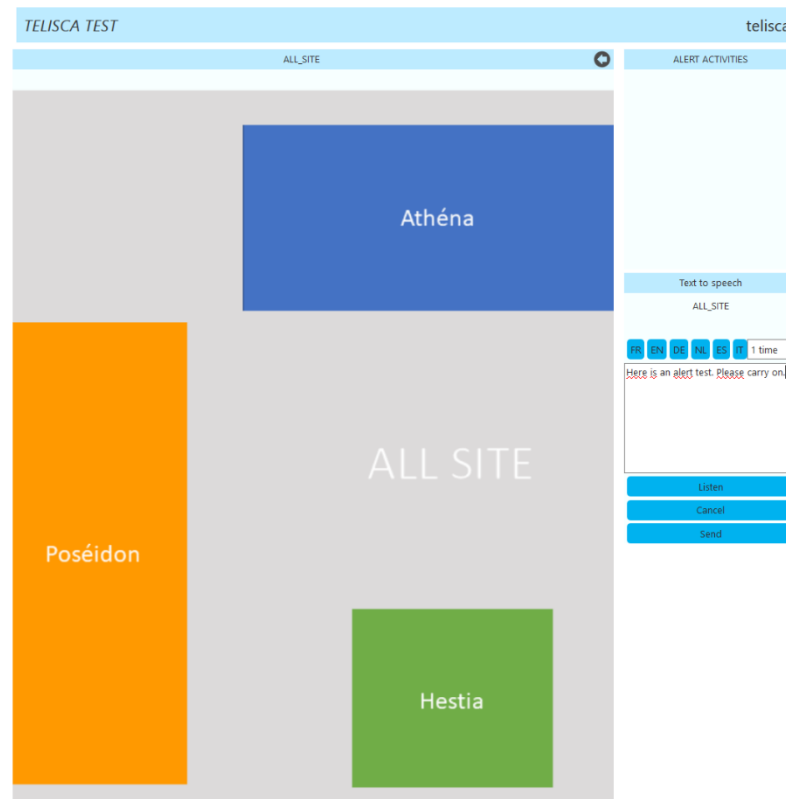

```
http://host/IPSPUSH/user/Map.aspx?grp=XXX&key=XXX
```

- **grp** is the editor group ID
- **key** is only required if defined as authentication from Web

See User Guide hereafter

IPS Pager Map User Guide

To connect, please enter in your browser the URL:
http://host/IPSPUSH/user/Map.aspx

If you get the authentication error message below, please check with your administrator. In order to open the Public Alert System Web page, you need to be part of adequate Active Directory Security Group.

Current user is not authorized for entity WHO HQ

Select one of the maps: 'Geneva Site' or 'Main building All floors'.
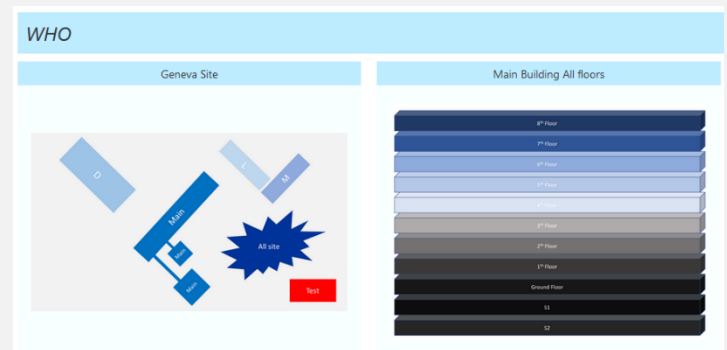
Click on a building or floor zone to select a destination. Click on the back arrow in the top right corner, to select another map.
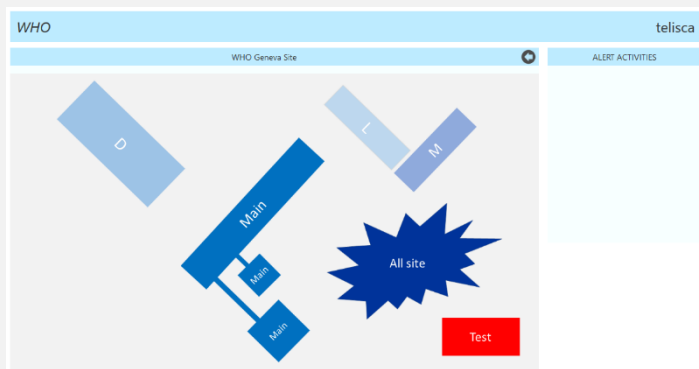
Then select the audio alert to play
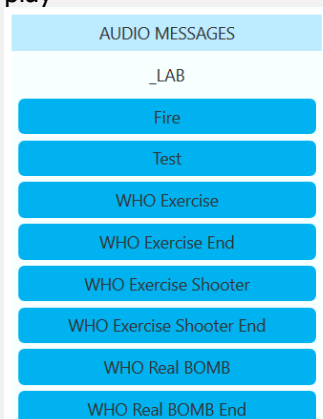
Check the destination, the alert, then click on Send to confirm
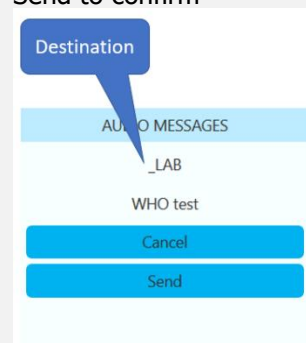
Dashboard for ongoing alert shows real time statistics. Click on an alert on the right tab to display the dashboard. You can click on Stop button to stop the alert.
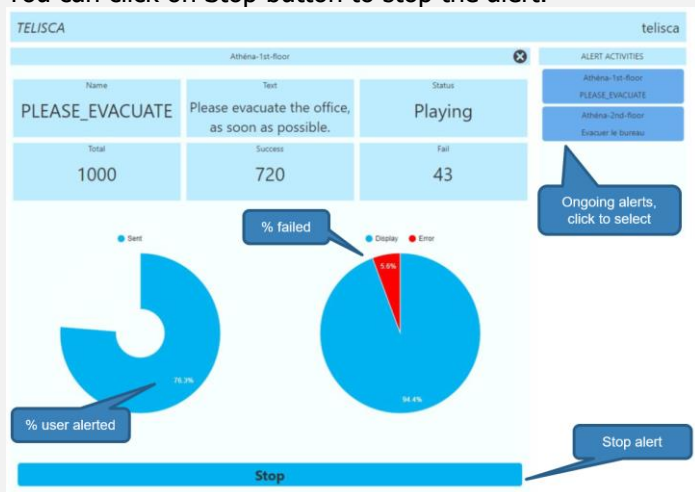
If you select an audio 'Text to Speech', you are able to create a new temporary audio message.
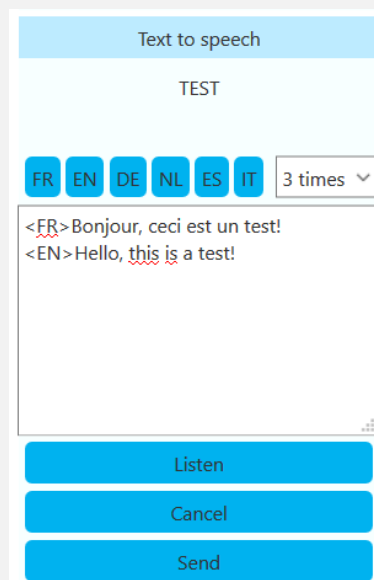
Click on the language to add a tag, then enter the text to speech. The different languages will be concatenated.

You can listen to check the message before sending.

In case of malfunctioning, you may have an error message like this one. If the error does not clear after 3mn, please contact your administrator.
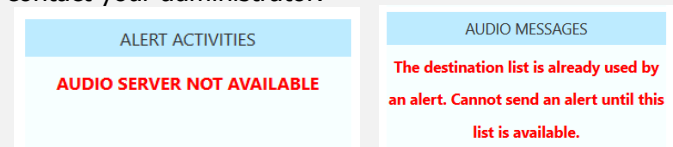
For each alert, the administrator receives an email with the alert's statistics in the body and a detailed report attached, in CSV file format, that can be loaded on Excel.

4.4 Direct from an URL

It is possible to trigger text message delivery by calling an URL with http get and parameters. This setup enables integration with other software.

```
http://host/IPSPUSH/user/send.aspx?g=i&list=XXX&msg=YYY
```

Example: http://10.5.1.2/IPSPUSH/user/send.aspx?g=1&dn=8610&txt=backup%20failure

- **g**=broadcasters group (1 to n, 1 by default)

4.4.1 Send text message

- recipients
  - ○ **list**=destination list ID,
  - ○ or **dn**=directory numbers separated by commas,
- Text message
  - ○ **msg**=predefined message ID,
  - ○ or **txt**=text message itself: needs to be URL encoded.
- Audio notification
  - ○ **audio**=tftpFileName[.raw] , this is a short audio file available on the CUCM TFTP server

4.4.2 Send audio message

- recipients
  - ○ **alst**=audio list ID

- audio content
  - ○ **rec**=predefined audio content ID,

Important: The URL is only accessible to IP addresses defined in the Parameters Tab > Advanced parameters

The audio parameter is used to force the notification with a specific audio file available on CUCM's TFTP server.

If the push has succeeded, the URL will return "OK." The push is asynchronous; if it is correctly sent but not guaranteed to have reached its destination, this can be confirmed in the send history report tab.

Returned error codes:
```
INVALID_LICENSE
UNAUTHORIZED_IP_ADDRESS
BOTH_LIST_AND_DN
BOTH_MSG_AND_TXT
NO_LIST_NO_DN
NO_MSG_NO_TXT
INVALID_EDITOR_GROUP
INVALID_LIST_ID
INVALID_MESSAGE_ID
INVALID_MESSAGE_TEXT
INVALID_DIR_NUMBER
INTERNAL_ERROR
NOT_RECORDED_AUDIO_LIST
TXT_TOO_LONG
TXT_EMPTY
```

4.5 Multicast with PA speakers

The sound mode, using multicast groups to broadcast the audio stream can be interfaced with external IP speakers using multicast.

These speakers connected directly to the local network must be configured to automatically connect to a multicast group identified by a multicast address corresponding to the multicast address configured on the list of recipients PA IPS Pager.

IMPORTANT NOTE: The routers / switches, used to connect the speaker / phones need to be configured to allow multicast protocol and router if necessary multicast streams between VLANs (phones / server IPS Pager / speakers multicast).

Example speaker multicast:

- Speaker IP SIP (http://www.cyberdata.net/products/voip/digitalanalog/ceilingspkr2/index.html)

Setting speaker with SIP IP multicast group:

- **Multicast IP Address**

IP multicast address corresponding to the sound list of recipients.

- **Port**

Intercom RTP port used by IPS Pager (see Audio settings)

4.5.1 Cyberdata configuration

4.5.1.1 IP Address

The loudspeaker is PoE. Connect it to the network and both lights (Link and Status) must be on. Once the startup is done, the status light stops blinking.
The DHCP is configured by default. Without DHCP, the loudspeaker takes 10.10.10.10/8 as IP address.

The loudspeaker gives its IP address if you click to RTMF less than 5 second. You can restore the factory configuration by clicking more than 5 second RTMF.

4.5.1.2 Basic configuration

You can configure the loudspeaker via a browser with its IP address as URL.
Default credentials: admin/admin

Change the password on the first menu:

Go to device and disable the volume button on the front of the loudspeaker (you can also force the multicast volume):

If needed you can force the IP address on the network menu:

Go to SIP and disable SIP and go to multicast to enable multicast and configure your alerts:

The address and ports must be the same as it configured on your alert (name is not important).

Once the configuration is done click save and reboot (~1 minute).

5 Appendix

5.1 Enabling Windows Authentication on Web Send or Map page

These instructions are related to using the Windows Authentication for IPS Pager Web Access.
To force Windows Authentication on this page, go to the IIS Management tool and untick "allow anonymous access" on `IPSPUSH/user/WebSend.aspx` or `IPSPUSH/user/Map.aspx`
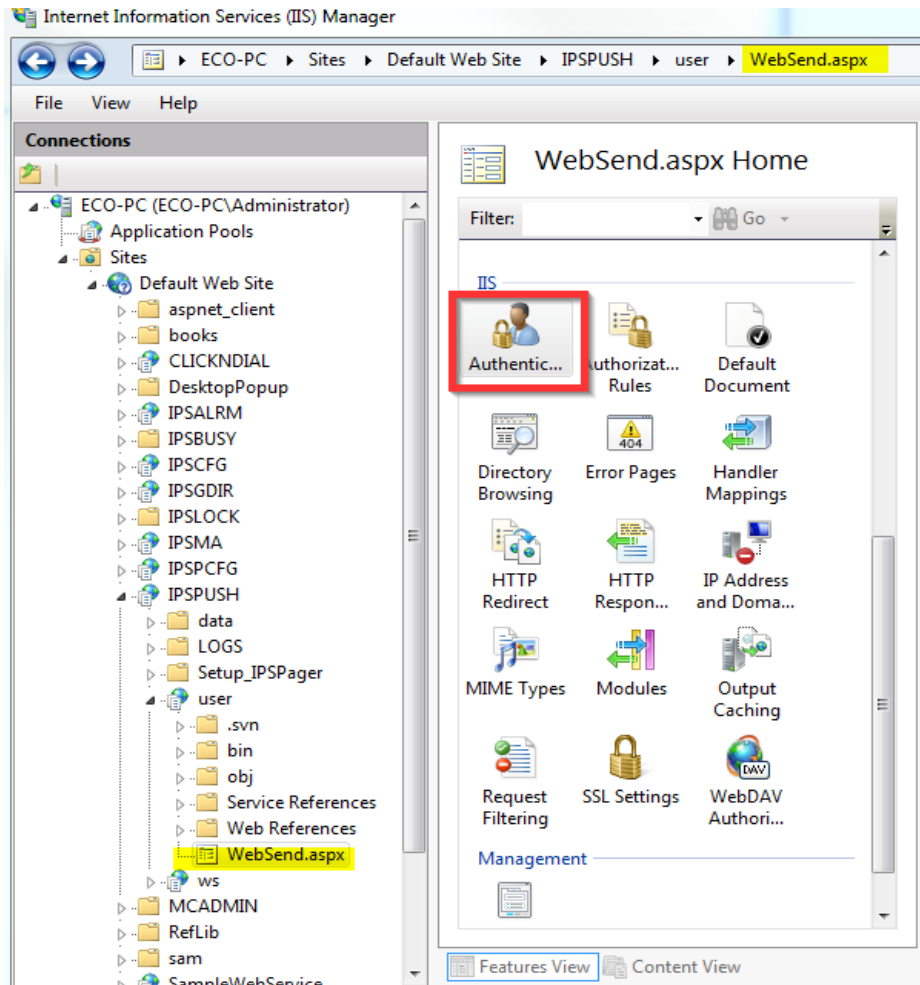
On Internet Information Services (IIS) Manager v7. Select the aspx file:
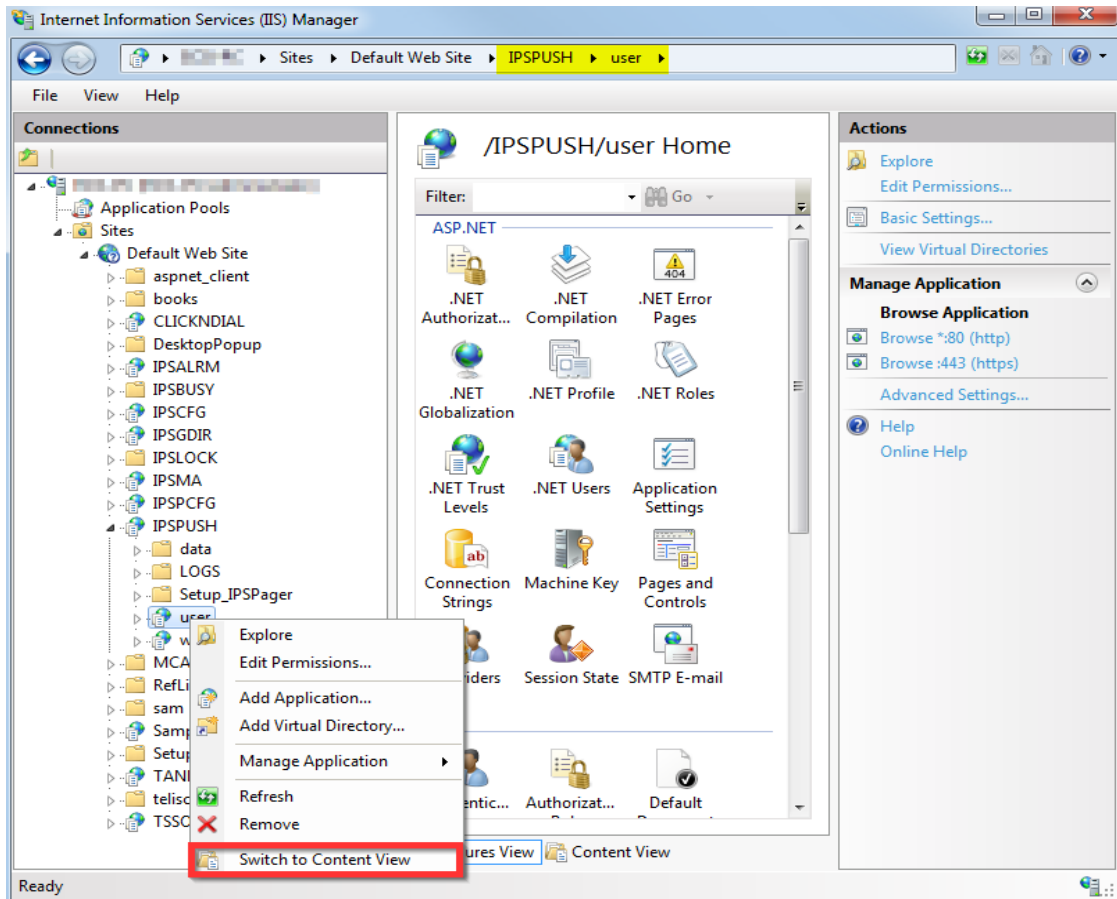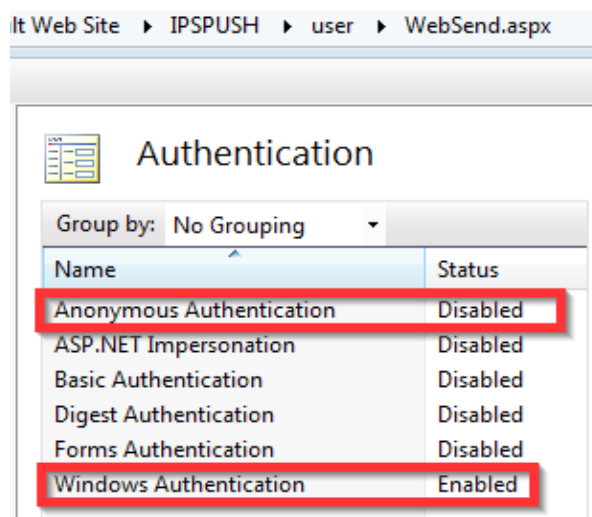

Then switch to features view for this page:
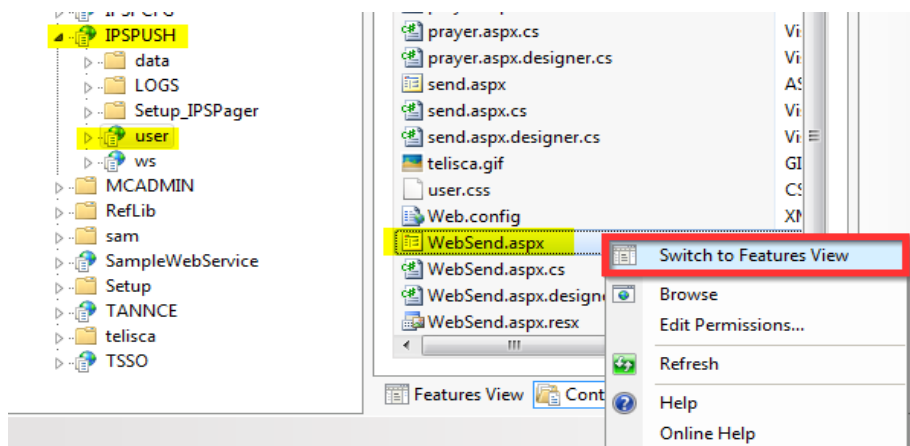
Then go in Authentication:

Disable anonymous access and enable Windows authentication:

Done, now from the admin you can enter the User Group's name which will be granted access. To remove the authentication prompt from the browser read below.

5.2 Hide the login prompt on WebSend or Map

When Windows authentication is enabled on the IPS Pager web, when accessing the URL, the browser will prompt the user for his Windows credentials.

You can configure Internet Explorer to use the integrated Windows Authentication method to hide this prompt.

This setting is "**Automatic logon with current user name and password**".

By default if the telisca server is identified as "Local intranet" in Internet Explorer this setting will be enabled.
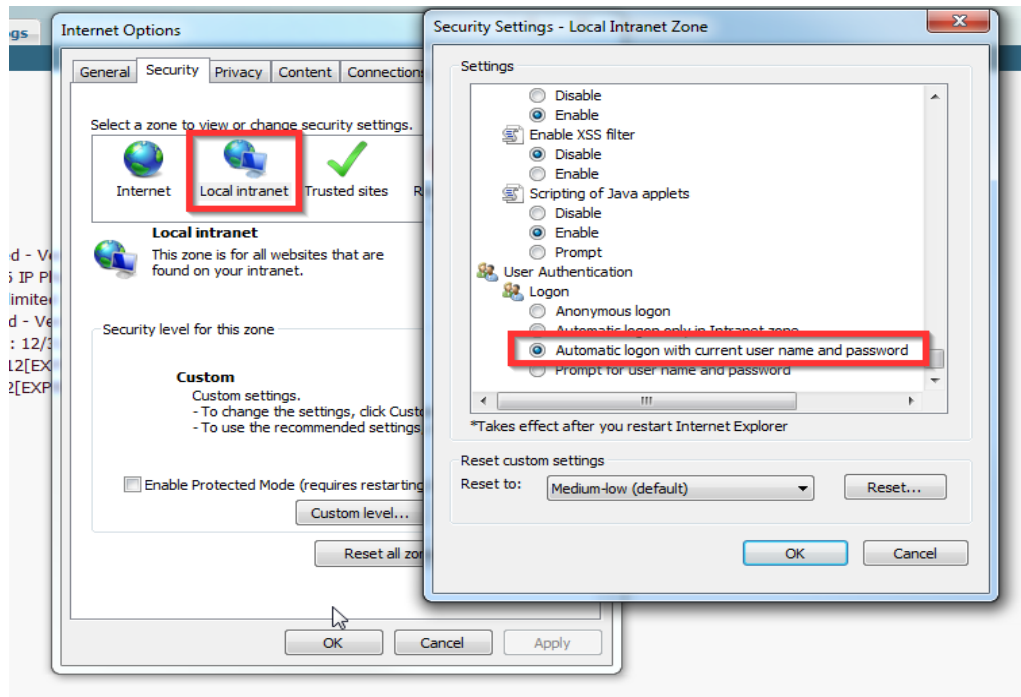
Automatic Windows Authentication for the Texting from a Web page

5.3 Encoding of text message sent by URL

A URL cannot contain all characters so a [text message sent from URL](#) must be formatted in a special way.

- **Line breaks**= \n
  Example: For "*[line break] *** WARNING ****" the URL will be:
  ```
  http://192.168.0.117/IPSPUSH/user/send.aspx?list=W8W1P7&txt=\n***+WARNING+***
  ```
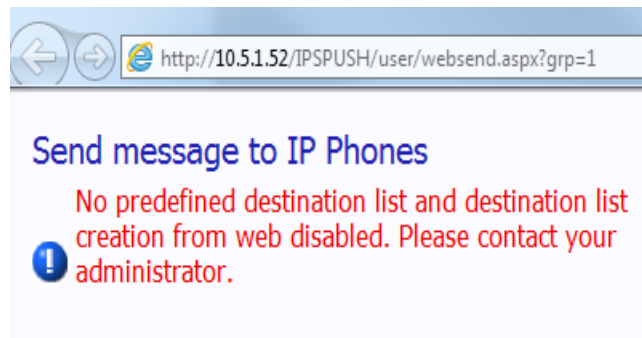- **Spaces**= %20

  To encode automatically you can use this online tool: http://meyerweb.com/eric/tools/dencoder/

6 Troubleshooting

6.1 From WebSend

6.1.1 Error message when accessing the page

No predefined destination list and destination list creation from web disabled. Please contact your administrator.

Possible causes:

1. no destination or messages configured for this editor group
2. if using Windows Authentication: IPS PAGER cannot match the authorised Windows User Group with the credentials passed through the browser. (check the AD parameters and that the user belongs to the defined group).

6.1.2 Error Message when sending a message

Message text is empty
When trying to send a message, even if something is typed in the message title and body boxes, an error message is displayed.
Also the settings for this editor group could be different from the ones expected (different predefined destination groups, ability to type destination numbers and message).

Logs in "C:\inetpub\wwwroot\IPSPUSH\LOGS\IPSPushUser_[date]_[time].log" may indicate:
`##ERROR##;WebSend.aspx;CreateOrSelectMSG;Index was outside the bounds of the array.`
`ERROR STACK TRACE ===>   at IPSPushServ.WebSend.CreateOrSelectMSG(String& msgId)`
Possible cause:
This can indicate that the user is not entitled to send texts at all.

Solution:
If using Windows Authentication, make sure the user belongs to the Access Group defined in AD/LDAP for the editor group being accessed (grp=X in url)

6.1.3 Nothing is displayed when sending messages
After having confirmed that the message needs to be send, a page opens showing the status of the message sent, with the close button.
In this case you only see the close button and nothing else, the messages are not sent.

Logs may indicate:
`T;Push.asax;isLocalClient;## DISCARDED BY SECURITY, userHostAddress=[SOME IPv6 ADDRESS], URL host=[NAME]`
`##ERROR##;Push.asmx;PushMessage2List;WEB REQUEST FROM LOCAL HOST ONLY ARE AUTHORIZED : client host=[IPV6 ADDRESS]`

Solution:
This is due to IPv6 not being supported by IPS Pager, please disable the IPv6 on the server network adapter.

6.2 From IP Phones

6.2.1 No document found

When accessing the IPS Pager service, this message indicates missing parameters in the service subscription. Check that all the compulsory parameters are present.

Additionally when accessing through a SURL, the phone name is not passed automatically with the #DEVICENAME# variable, so the actual phone name must be entered in the service URL, with pn=SEP****.

6.2.2 Delivery reports discrepancies

If the number of recipients doesn't match the expected number it could be because the device sending the messages is not recognised by IPS Pager, check the pn parameter. Also check the list of recipients if using a predefined lists and make sure they contain the phones you want to send messages to. You can check delivery reports from the administration.

6.2.3 Message not showing on phone

Make sure the IP Phone is in the supported phone models list, and always use the latest available firmware if possible. Check if opening a menu opens the message, in which case you can force the messages to be sent as "Send Immediately" to ensure they will be displayed without user interaction.