



BTC Cloud Communications

Features Guide

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1 Cloud Voice Portal features

1.1 Cloud Voice Portal Web

Cloud Voice Portal is a web portal that delivers a streamlined and dynamic web experience. It leverages Web 2.0 technology to deliver an intuitive and tightly integrated user interface to the wide range of services described in this document.

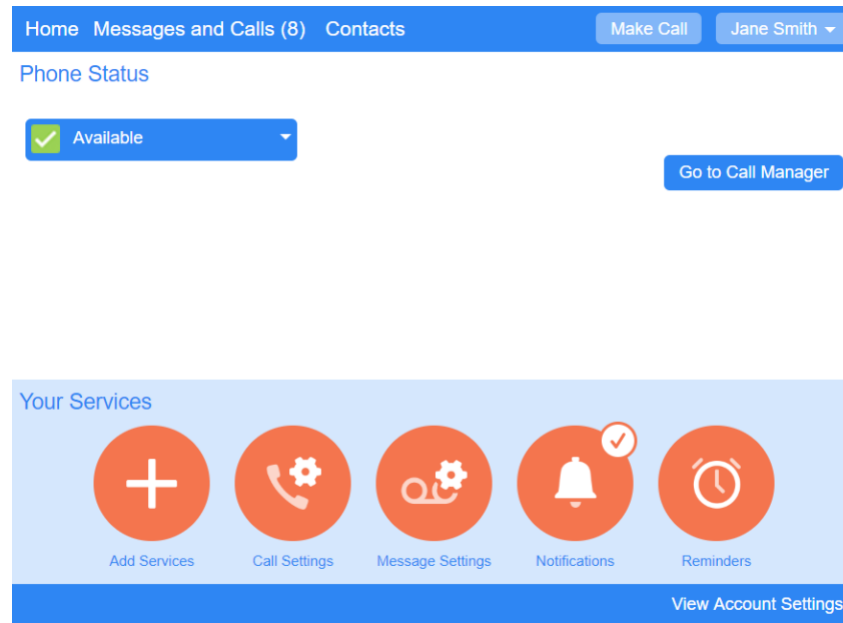
The full list of features that can be managed by Cloud Communications users using Cloud Voice Portal is as follows.

- Calling Number Delivery
- Calling Name Delivery
- Calling Number Delivery Blocking
- Call Forwarding Unconditional
- Call Forwarding Busy
- Call Forwarding No Answer
- Call Forwarding Selective
- Call Forwarding on Unavailable
- Voicemail
- Outgoing Call Blocking
- Do Not Disturb (Make Set Busy)
- Selective Call Acceptance
- Selective Call Rejection
- Anonymous Call Rejection
- Reminder Calls
- Recurring Reminder Calls
- Priority Call
- Speed Dials
- SimRing
- Find Me Follow Me
- Call Jump

The Cloud Voice Portal interface seamlessly launches via any standard web browser.

The following screenshot shows the Cloud Voice Portal Home page that provides easy access to all features and settings.

Figure 1: Cloud Voice Portal Home Page

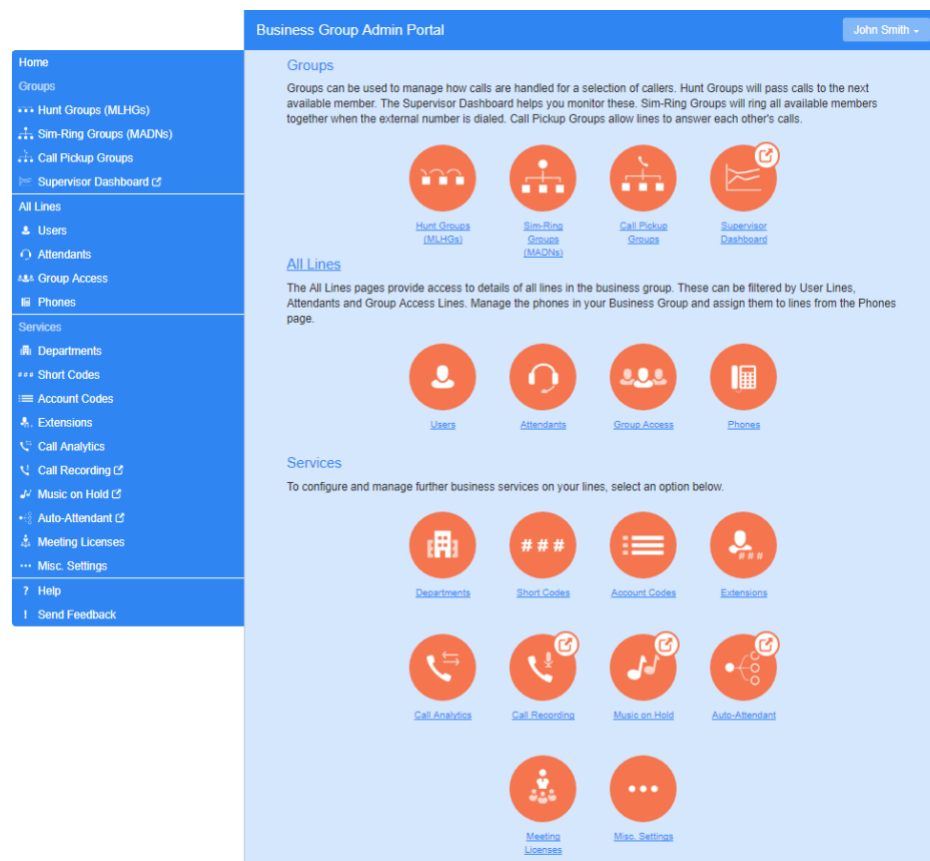


1.2 Cloud Voice Portal Business Group Administration interface

The Cloud Voice Portal Business Group Administration (BG Admin) interface provides Business administrators with a more powerful tool to allow Moves, Adds and Changes (MACs) to be performed.

An administrator can manage business-wide features, like hunt groups, and perform administration on behalf of individual users.

Figure 2: Cloud Voice Portal Business Group Administration Interface

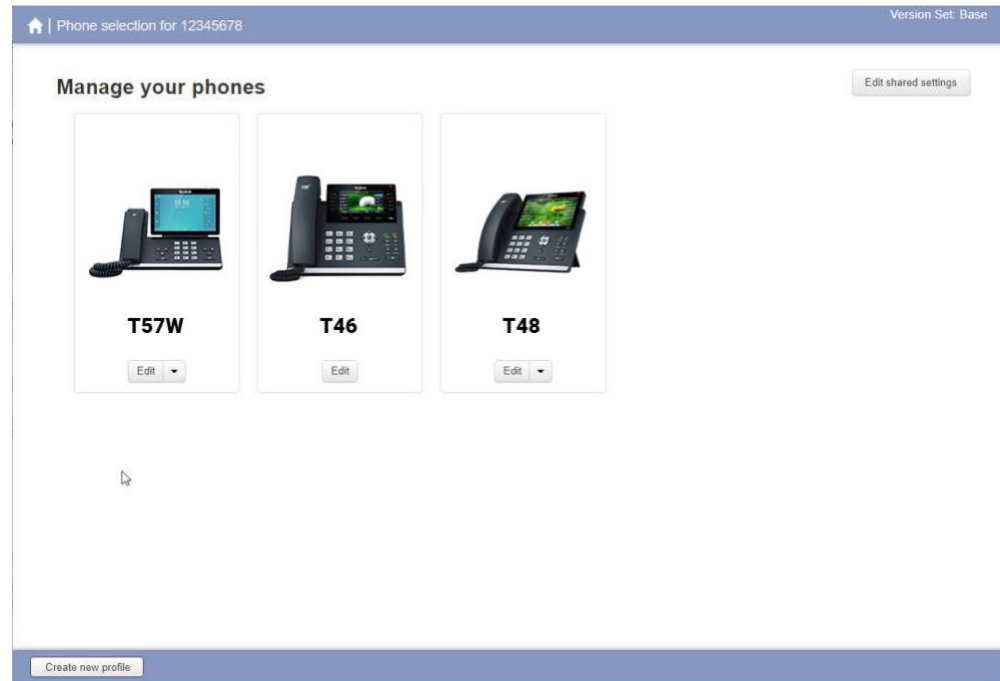


1.3 Cloud Voice Portal Phone Configurator

The Cloud Voice Portal Phone Configurator allows business administrators and end users to configure the keys on their desk phones using a simple and intuitive point and click interface.

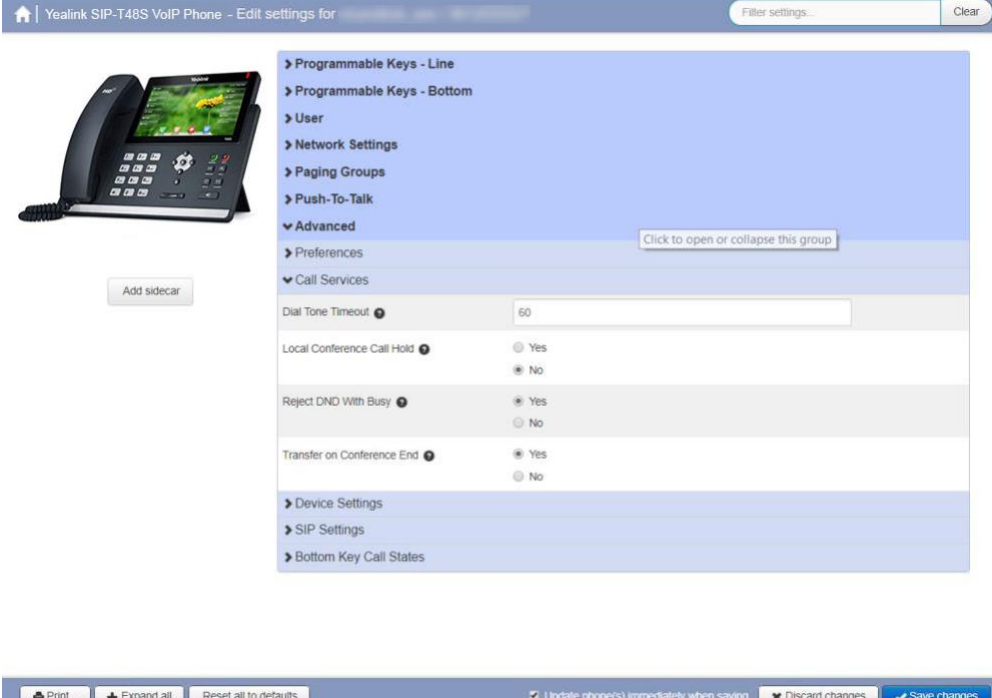
The Cloud Voice Portal Phone Configurator interface will be different, depending on whether a phone is deployed using an Endpoint Pack or by the Built-in provisioning method. Most deployments should now only be using Endpoint Packs. If you still have phones of both types, the Cloud Voice Portal Phone Configurator will use a yellow triangle to indicate the phones that have been provisioned using the Built-In method.

Figure 3: Cloud Voice Portal Phone Configurator




When configuring the keys on an individual phone provisioned using an Endpoint Pack, you will see the Cloud Voice Portal Phone Configurator interface illustrated in Figure 4.

Figure 4: Cloud Voice Portal Phone Configurator for Endpoint Pack phones



Yealink SIP-T48S VoIP Phone - Edit settings for Filter settings...



- ▶ Programmable Keys - Line
- ▶ Programmable Keys - Bottom
- ▶ User
- ▶ Network Settings
- ▶ Paging Groups
- ▶ Push-To-Talk
- ▼ Advanced
- ▶ Preferences
- ▼ Call Services
 - Dial Tone Timeout
 - Local Conference Call Hold ☒ Yes ☐ No
 - Reject DND With Busy ☒ Yes ☐ No
 - Transfer on Conference End ☒ Yes ☐ No
- ▶ Device Settings
- ▶ SIP Settings
- ▶ Bottom Key Call States

☒ Update phone(s) immediately when saving

2 Cloud Voice Portal applications

The following applications can be downloaded from the Cloud Voice Portal account.

2.1 MaX UC Desktop

MaX UC Desktop is a soft client you can download onto a Windows or Mac computer. It offers a range of configurable features including soft phone, Instant Messaging and presence, and Meeting function.

You can choose to enable some or all the following features in MaX UC Desktop.

- Softphone function. Subscribers can make and receive audio and video calls using their computer. The MaX UC Desktop client can be twinned with the subscriber's main directory number, allowing subscribers to use their main phone number when making calls from MaX UC Desktop.
- Cloud Voice Portal features. Subscribers can access their Cloud Voice Portal account directly from within the Desktop UI enabling them to configure Call Manager settings, listen to voicemail messages, log in and out of MLHGs and MADNs, and set up the 'Fusion' Fixed Mobile Convergence service.
- Use of Instant Messaging and Presence information, including integration with Outlook Calendar for subscribers using Windows (unless you have a Business Group with more than 10,000 lines).
- Creation of web conferences with the integrated Meeting client
- Integration with N-Series conferencing (if you have configured the Conference Widget).
- Direct links to web applications including File sharing and Customer Relationship platforms.

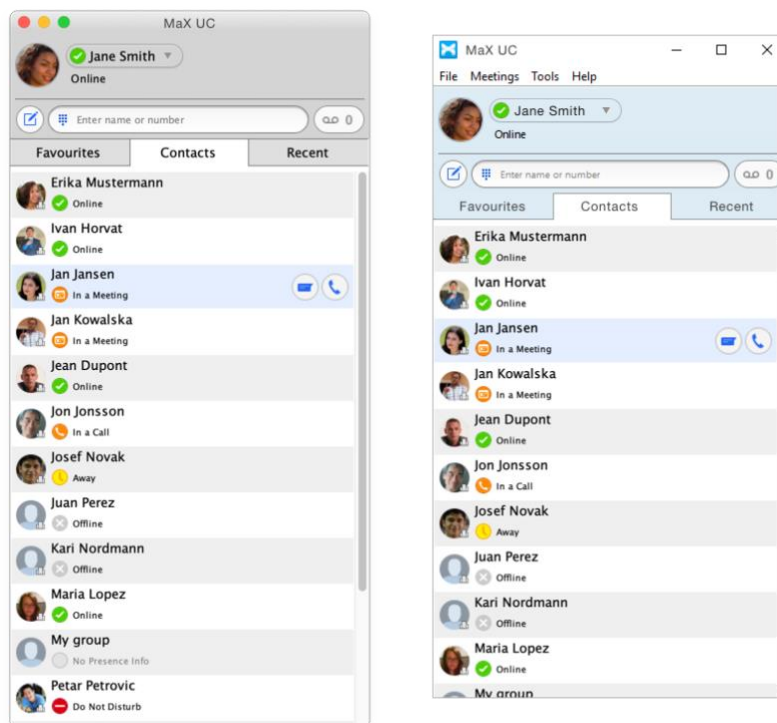
You can also integrate MaX UC Desktop with the following contact sources:

- Outlook or Mac AddressBook
- Business Group contacts (for subscribers who are members of a Business Group)
- LDAP contacts (if you integrate with an external LDAP server)

MaX UC Desktop is currently supported on computers running Windows 8 (although not in the Metro UI of Windows 8), Windows 10, and Mac OS X 10.12 or later.

Figure 5 illustrates this application.

Figure 5: MaX UC Desktop on Windows and Mac



2.2 MaX UC Mobile

MaX UC Mobile is an application that runs on an iPhone, iPad or Android mobile device. MaX UC Mobile can be deployed as a soft phone offering high-definition internet voice calling (VoIP), or as a companion app that integrates with the network voice capabilities of the subscriber's iPhone or Android mobile phone.

Figure 6: MaX UC Mobile



You can choose to enable some or all the following features in MaX UC Mobile:

- Cloud Voice Portal features including Call Manager, Visual Voicemail, and Click To Dial.
- Contact integration. View and call their Cloud Voice Portal contacts from within the Desktop app.
- Call Jump/Switch Call. Jump calls between landline, smartphone and tablet devices, including MaX UC Desktop.
- Instant Messaging, SMS messaging, and presence.
- Meetings. Create, schedule, join and uplift Meetings.
- Video calling. Use MaX UC Mobile as a softphone can seamlessly uplift calls to video at the touch of a button and without disrupting the call.

This application is currently supported for iOS and Android devices.

2.3 Meeting and MaX Meeting

Meeting offers Web, Video, and Audio conferencing for up to 100 attendees, with add-on licenses available for Business Group subscribers to enable them to create Meetings for up to 500 participants. Subscribers can also be restricted to creating only small Meetings of up to 4, 10 or 25 participants.

The Meeting client can either be provided as part of MaX UC Desktop and/or MaX UC Mobile, or as MaX Meeting, a stand-alone client for desktop and mobile devices.

- When provided as part of MaX UC Mobile and/or Desktop, you do not have to take any specific steps to install Meeting. You can launch a meeting room and invite users in real-time by selecting the Meeting function from multiple places in the MaX UC client user interface and inviting users to join the meeting. Users can also uplift MaX UC calls to meetings, including cellular calls made on Network Voice phones.
- The MaX Meeting stand-alone client can be made available via the Cloud Voice Portal account or a download link on a webpage, or, for mobile devices, from Google Play or the App Store.
- Both Desktop and MaX Meeting subscribers can download an Outlook plug-in to allow them to create or schedule meetings directly from Outlook calendar.

Contacts who are not MaX UC client subscribers can be invited to join a meeting simply by sending them a URL that enables them to connect to the Meeting using a web browser or their mobile device. Participants can choose whether or not to download a version of MaX Meeting on to their computer or use a browser. All participants in a Meeting can use all the features, including the chat facility, regardless of whether they have MaX UC Desktop or MaX UC Mobile.

2.3.1 Webinars

Webinars offer full function webinars for up to 100, 500, or 1,000 attendees, with support for up to 100 participants (webinar host, co-host and panelists), all with the capability to share video, web presentations, whiteboards and more. The remaining view-only attendees are able to send chat messages or participate in polls created by the host.

Business Group subscribers with an add-on license to create Meeting Webinars use a web-based UI launched from within a subscriber's MaX UC Desktop or Cloud Voice Portal account to set up webinars by creating signup forms, then sending email invitations and reminders. You can also capture key information from participants when they sign up to attend a webinar and receive reports on who signed up and who attended the webinar.

Add-on licenses for Webinars are provisioned against a Business Group and can be allocated as required to individual lines by a Business Group administrator.

3 Basic features

3.1 Star codes

All standard CLASS services are offered as part of this Solution, and these can be invoked using standard North American vertical service codes ("star codes").

In other geographies, these services are invoked using the relevant prefix or access code.

3.2 Last number redial

This feature allows the user to conveniently call the last number with a single touch, rather than having to dial the complete number again. This feature is provided by either of the following.

1. A dedicated Last Number Redial (LNR) key on the phone which calls the last number called.
2. A list on the phone's screen providing a number of recently dialed numbers from which the user can choose with phone keys.

This feature is provided by the phone so is not provided on the Cloud Communications platform.

This feature must be supported on the phones used, and configuration may be required.

3.3 Automatic recall

This feature allows the user to find the number of the last received call, and return a call to that number. It can be used in two ways.

1. A two stage mechanism which, upon dialing an access code or pressing a key, initially reads out the last calling number. Another key press is required to return that call if so required.
2. A one stage mechanism which, upon dialing an access code or pressing a key, immediately returns a call to the last calling number.

Usually this feature is used by dialing an access code (normally *69). Some phones support a key to be configured to dial this access code automatically upon a single key-press.

This feature is configured on a per-line basis on the Cloud Communications platform.

For one-touch operation, phone support is required and phone configuration may also be required.

3.4 Last call ID erasure

This feature allows the user, by dialing an access code, to delete the record of the last received call. This is provided for privacy reasons. This will also clear all call lists from the Cloud Voice Portal feature.

This feature is configured on a per-line basis on the Cloud Communications platform.

Note some SIP phones carry their own call lists. The Cloud Communications provided feature will not clear any call lists from the phone. Phone support is required for erasing the in-phone call lists.

3.5 Automatic callback

This feature allows the user, after making a call and experiencing a busy signal, to queue a callback request when that line is free.

This called number may be on the Cloud Communications, or external (off-Cloud Communications). In the latter case this feature requires network support.

This feature is configured on a per-line basis on the Cloud Communications platform.

There is typically no phone support or configuration required for this feature.

3.6 Customer Originated Trace

Customer Originated Trace allows users to record the details of any malicious or harassing incoming calls to their lines. The user is provided an access code to dial either during (using a separate Call Appearance on the line the unwanted call was received on) or after the call. This produces a record that we can access and will contain the calling party ID if this is available, regardless of whether the caller withheld their number. We can then provide this information to law enforcement upon request.

This feature is enabled on a per-line basis on the Cloud Communications platform.

There is no support or configuration required on the phone.

3.7 Line Identification

Line Identification is a feature that is invoked by dialing an access code and announces the full Directory Number of the line back to the caller.

This feature is enabled on a per-line basis on the Cloud Communications platform.

There is no support or configuration required on the phone.

3.8 Authentication

Authentication for SIP phones allows Cloud Communications to ensure service is only provided to phones that have the recognized credentials of username and password for a line.

In pre-IMS deployments, the Cloud Communications platform will authenticate all lines configured for this feature at the time of registration (when the phones are initially provisioned or booted) and on all subsequent message exchanges. In IMS deployments this function is performed by the IMS core, so support for authentication depends on the IMS core you are using.

Cloud Communications supports the use of the DN as the authentication username, or, for stronger security, an arbitrary username of the Service Provider's choosing.

This feature is configured on the Cloud Communications platform on a per-line basis. Its use is not mandated (so lines can be configured without authentication).

This feature must also be supported on the phones used, and the authentication credentials must be configured on these phones.

4 Call Management Services

4.1 Business Call Manager (BCM)

Business Call Manager enables subscribers to manage their incoming call and call forwarding services using the Cloud Voice Portal interface or by dialing star codes on a phone handset.

Business Call Manager displays a range of configuration options on a single screen on the Cloud Voice Portal Home page.

Figure 7: Business Call Manager Cloud Voice Portal

Phone Status

Available for Calls

Incoming calls will: Ring your phones in order

	0s	60s	120s	180s
A Account Phone				S
M My Mobile		2	4	S
1 1208369912...	1		4	S

If there is no answer

☐ Forward to another phone after 36 seconds

☐ Send to voicemail after 12 seconds

If your phone is busy

☐ Forward to another phone

☐ Send to voicemail

Advanced Settings

☒ Forward Selected

Callers from the forwarding list will be forwarded to My Mobile

☒ Reject Selected

☒ Distinctive Ringtone

☒ Forward if Unavailable

☒ Anonymous Callers

Calls from anonymous callers will be rejected without going to voicemail

The Business Call Manager screen is split into the three sections where you can configure and make changes to how their incoming calls should be handled, depending on whether or not they are available for calls.

4.1.1 Line State Control

Use the drop-down on the Line State Control lozenge to set their status to either Available for Calls or Do Not Disturb.

- When Do Not Disturb is selected, calls will generally be diverted to voicemail. However, you can set whether calls from callers whose numbers appear in the subscriber's priority callers list will still ring their phone and change how these calls are handled, for example they may prefer these to ring their mobile phone rather than their office phone.

4.1.2 Incoming Calls Settings

Use the rest of the left-hand side panel to configure what happens when they receive an incoming call. This includes:

- Configuring which of their phones will ring when an incoming call is received and setting the order in which these phones will be tried and the length of time each phone will be allowed to ring.

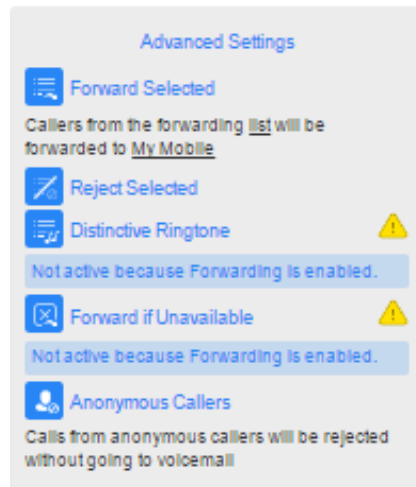
- Forwarding calls to a number of the subscriber's choice. Note that doing this removes the call from BCM control which means that some Advanced Call Settings, for example Busy Call Forwarding and Delayed Call Forwarding will be disabled.
- Configuring the behavior if the subscriber either does not answer the call within a specified time period, or all their configured phones are busy.

Pop-ups are displayed to guide you through this configuration and explain the impact that some settings will have on other call services that the subscriber may be using.

4.1.3 Advanced Call Settings

The Advanced Settings panel on the right-hand side of the screen allows subscribers to configure and activate a range of advanced call services. Subscribers will only see icons for the services they are subscribed to.

Figure 8: Advanced Settings panel



Subscribers can configure the following advanced call settings.

- Forward Selected – calls from numbers on a forwarding list created by the subscriber will be forwarded to another telephone number configured by the subscriber.
- Reject Selected – calls from numbers on a rejection list created by the subscriber will use be rejected.
- Distinctive Ringtone – calls from numbers on a list created by the subscriber will use a different ringtone.
- Forward If Unavailable – calls will be forwarded to a specified number if the subscriber's line is unavailable, for example because their phone has been disconnected or there has been a power outage.
- Anonymous Callers – specifies how calls from unknown numbers should be handled.

Each icon indicates the current status of the setting.

- Each service can be turned on and off by clicking on it.
 - When the icon is bold, the service is active, and further configuration options are shown.
 - When the icon is grayed out and has a slash through it, the service is inactive.
- When the icon displays a warning triangle, the current configuration of this setting is being fully or partially overridden by another BCM setting. For example, if a subscriber has enabled both Do Not Disturb and the Forward Selected Call service, they will see a warning triangle and an error message to indicate that no calls are being forwarded because Do Not Disturb is turned on.

5 Call Management Services

5.1 Incoming Call Manager

Incoming Call Manager offers an integrated call handling service, with a powerful set of configurable rules. The following tabs allow subscribers to configure sets of rules and set schedules for when specific rules should apply to incoming calls.

Figure 9: Incoming Call Manager Summary Tab

The screenshot shows the 'Incoming Call Manager' interface. At the top is a blue navigation bar with 'Home', 'Messages and Calls (13)', and 'Contacts'. On the right of this bar are 'Start' and 'John Smith' buttons. Below the navigation bar is the 'Phone Status' section, which has four tabs: 'Summary' (selected), 'Rules', 'Weekly Schedule', and 'Special Days'. There are 'Apply' and 'Cancel' buttons to the right of these tabs. The 'Summary' tab is divided into two columns: 'When I receive a call' and 'Additional options'. Under 'When I receive a call', there are three radio buttons: 'Ring my phone' (selected), 'Forward to' (with a dropdown menu showing 'enter a number'), and 'Send to voicemail'. Below these is a text block explaining that Call Manager lets the user decide who, how, and when callers can reach them, with a 'Learn more' link and a link to 'Set up rules for more advanced control'. Under 'Additional options', there is a text block stating 'These options apply in all cases where you've chosen to have your phone ring. If I don't answer:'. Below this are three radio buttons: 'Forward to' (with a dropdown menu showing 'enter a number'), 'Send to voicemail', and 'Reject the call'. Further down is a section 'If I'm in a call:' with three radio buttons: 'Forward to' (with a dropdown menu showing 'enter a number'), 'Send to voicemail', and 'Reject the call'. At the bottom is a section 'What audio should callers hear before their call is answered?' with two radio buttons: 'Use default ringback' (selected) and 'Use custom ringback' (with a 'record' link).

The Summary tab allows the subscriber to view and configure their call-handling options. For example, a subscriber may wish their phone to ring as normal, or divert all calls directly to voicemail, or apply some of their call handling rules depending on the time of day. This screen also allows the subscriber to record or upload some custom ringback to play while callers are waiting for their call to be answered.

If a subscriber has not yet configured any call handling rules, for example when they initially access the Call Manager Summary tab, only the first three radio buttons shown in are displayed. The subscriber will see an introductory message explaining how Call Manager rules work and prompting them to click on a link that takes them to the Rules tab so that they can configure these.

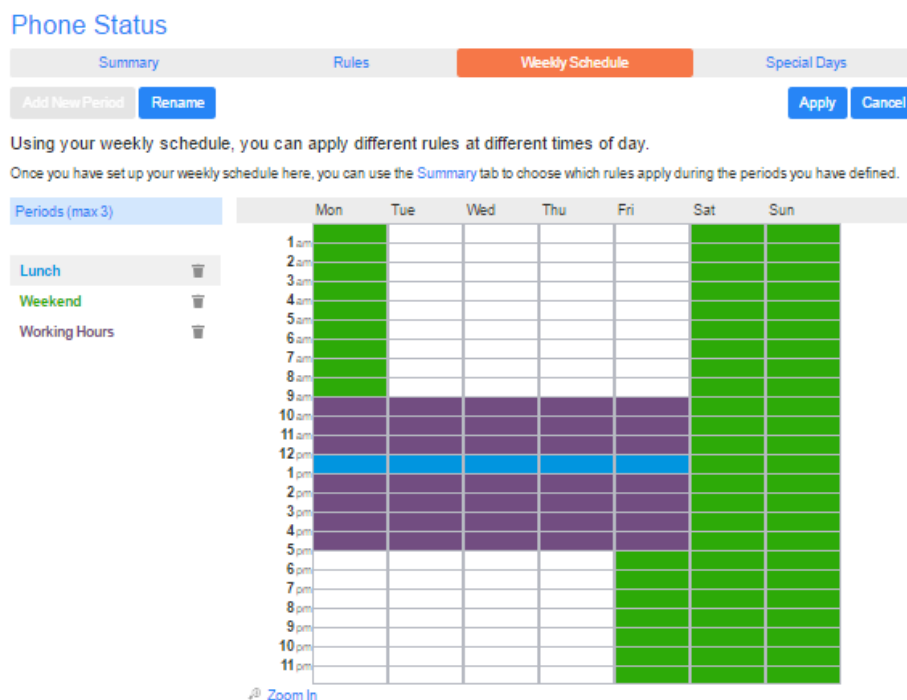
Once the subscriber has configured at least one Rule, all five radio buttons will be displayed on the Summary tab. If a subscriber has not yet set up a weekly schedule, they will see a message prompting them to do so by clicking on a link that takes them to the Weekly Schedule tab. The schedule-specific sub-settings shown will be hidden unless the *Handle the call depending on the day or time* radio button is selected.

Subscribers can use the Rules tab to configure more advanced Sets of Rules for their incoming calls. For example, you might want to forward calls from family and friends to a home phone, while forwarding calls from the office directly to voicemail, or to screen calls from a particular number.

If any of the telephone numbers used within ICM change as a result of an Area Code Split procedure, your subscribers will have to update any affected telephone numbers manually.

Subscribers can then use the Weekly Schedule tab to specify where they will be at certain times of the day. Using the options on the Summary screen, they can then apply different Sets of Rules at different times of the day or week. Figure 10 shows the Weekly Schedule covering a subscriber's lunch break.

Figure 10: Incoming Call Manager Weekly Schedule tab



The **Special Days** tab allows the subscribers to set dates that are exceptions to their normal weekly schedule, for example vacations or business trips. This screen includes the option to add the public holidays for their region automatically at the click of a button.

5.2 Easy Call Manager

Easy Call Manager (ECM) is an intuitive and attractive way for subscribers to control what happens to their incoming calls, targeted at users who do not require the full power of Incoming Call Manager (ICM).

The following functionality of Incoming Call Manager is included in Easy Call Manager:

- Send all calls to voicemail
- Forward all calls to a different number, such as their mobile phone
- Ring several of their other phones simultaneously, taking the call on the first phone to be picked up
- Screen anonymous callers, allowing the subscriber to decide whether to accept the call after hearing a recording of the caller's name.

Additionally, the subscriber can set up lists of "VIP" callers such as key business contacts – who can be forwarded to a different number, often a mobile phone – and "unwanted" callers, who can be sent to voicemail or simply rejected. They can also decide what behavior to apply if their line is busy or they do not answer.

Figure 11: Easy Call Manager

Phone Status

Choose how calls to (501) 203 0006 will be directed. Apply Cancel

☒ Available

☒ Do Not Disturb

☒ Forward All Calls

▼ Normal callers

☒ Ring my phones at the same time:

- ☒ My Phone (501) 203 0006
- ☐ My Mobile (123) 456 7890
 - [change my phones](#)

☐ Forward to another phone

☐ Send to voicemail

▶ Anonymous

▼ VIP - Active

☒ Ring my phones at the same time:

- ☒ My Phone (501) 203 0006
- ☐ My Mobile (123) 456 7890
 - [change my phones](#)

☐ Forward to another phone

Manage VIP callers

▼ Unwanted

☒ Reject

☐ Send to voicemail

You have no Unwanted callers

Manage Unwanted callers

▶ Additional options

5.3 Call Jump

Call Jump is a feature of Easy Call Manager and Incoming Call Manager that allows a call to be seamlessly transferred to another phone to ring, which the user can then answer and continue the call on. Essentially the user jumps from one phone to another mid call without having to put the other caller on hold.

Users of black phones on POTS lines, basic SIP devices and high function SIP business phones can initiate a call jump by dialing one of the following codes.

- *94: Jump to Primary Telephone number.

This is the key sequence used to transfer a call to their primary telephone number.

- *95: Jump to alternate number.

This is the key sequence used to transfer a call to their pre-configured alternate telephone number.

- *96: Jump to a wireless number.

This is the key sequence used to transfer a call to their pre-configured wireless telephone number.

- **: Initiate a SimRing.

This is the key sequence used to initiate a SimRing call transfer, where all the telephones and mobile devices associated with SimRing will ring simultaneously. This key sequence will only have an effect if the subscriber is permitted to use SimRing by their Class of Service.

Users utilizing Network Voice with the MaX UC Mobile app can also perform call jump, pressing a button on the in-call screen to "push" the call to another device or to "pull" a call from one of those devices. The same process applies to users of the MaX UC Client softphone.

6 Caller ID features

6.1 Calling Number Delivery

Presents the calling party's number to the user via a phone screen prompt.

- For an external caller, this number will be the full national number that is calling.
- If the caller is in the same local area, you can configure to display either the full national number or the local number
- If the caller is within the same Business Group as the called user and intercom codes are configured, the number shown will be the caller's intercom code. In some deployment configurations the caller's full DN may be shown instead of the intercom code.
- Caller ID on Business Group lines can optionally include the BG's External Line Code as a prefix, so that the displayed number matches what a user of the Business Group line would dial to return the call.
- Subscribes with twinned devices, for example a deskphone and a mobile phone, on a Business Group line can be configured to show the number of the device on which they are making the call.

Notes.

- Some phones will look up received caller ID in the phone's directory and, if this number is present, provide the name that is present in the directory on the phone's display instead.
- When this feature is provided in combination with the Calling Name Delivery feature the phone may not display both the calling number and name.

This feature is configured and enabled on an individual line or Business Group basis on the Cloud Communications CFS platform.

This feature may also need to be enabled on the phone, and there may be configuration options on the phone for this feature, such as

- how to format the number
- whether to lookup in the in-phone directory
- how to handle both the calling number and name being provided.

6.2 Calling Name Delivery

Presents the calling party's name to the user via a phone screen prompt. This name will be provided by the network for external callers, or the name configured by the SMB administrator for the calling line for internal callers.

This feature is controlled on an individual line basis on the Cloud Communications platform.

This feature may also need to be enabled on the phone, and there may be configuration options on the phone for this feature, such as how to handle both the calling number and name being provided.

Note that when this feature is combined with the Calling Number Delivery feature the phone may not display both the calling number and name depending on the phone configuration.

6.3 Calling Number Delivery Blocking

This feature allows callers to prevent their calling number and name from being presented to both internal and external called lines.

This feature can be used in a number of different ways.

1. The number called can be preceded by an access code indicating the user's decision to withhold caller ID from the called line. For example:
 - *67 555 555 5555 would call external line 555-555-5555 withholding calling number and name
 - *67 201 would call internal intercom or short code 201 withholding calling number and name

If this feature is to be used in conjunction with digit maps on the phones, the phone digit map will need to be configured to support the addition of an initial withhold service code.

You can configure it so that these privacy settings can be overridden by an authorized subscriber, for example a law enforcement agency.

2. Some phones can be configured to support caller ID delivery blocking on the handset, rather than dialing a prefix. This may either be as a permanent setting on the phone or enabled as a per-call option.

This feature is configured on a per-line basis on the Cloud Communications platform.

To use the feature as described in option 2, above, configuration options on the phone for this feature will need to be used.

6.4 Calling Number and Name Delivery over IP

This feature allows the name and number of incoming calls to be transmitted over IP to software applications and hardware devices that support receiving it. Some TV set top boxes support this feature and display the incoming call information on a TV screen.

This feature is configured on a per-line basis on the Cloud Communications platform.

This feature does not interact with phones, so support is not required on these devices.

A compatible device is required to receive the calling number and name information over IP.

6.5 Calling Party Number

This feature allows subscribers to present the number of their choice on their outgoing calls, rather than using the directory number assigned to their line for incoming calls. It is often used to present main switchboard numbers or freephone numbers.

This feature is configured on a per-line basis on the Cloud Communications platform.

You may also need to enable Calling Number Delivery on the phone. For information on how to do this, see section 6.1.

6.6 Charge Number

This feature allows subscriber's calls to be billed to a different number from the directory number assigned to their line. For example, a subscriber who has two or more lines can have all calls for these lines billed to one number and therefore receive a single bill for all calls.

This feature is configured on a per-line basis on the Cloud Communications platform.

This feature does not interact with phones so support is not required on these devices.

7 Call Hold features

7.1 Call Hold

This feature allows the user to place a call that is in progress on hold.

- With SIP phones the user can typically place a number of different calls on hold and retrieve these independently of each other using the phone screen and keys to select calls to retrieve from hold. Callers or called parties placed on hold will receive silence, unless the Music on Hold feature is used. This feature is described separately below.
- For POTS lines there are two variants of Call Hold.
 - Calls can be placed on hold when using another feature or making another call by hook flashing, with the call waiting, call transfer and three-way calling features. When the user hangs up on the second call, the call on hold will ring the user's phone.
 - Calls can also be placed on hold by flash hooking and dialing an access code, followed by going on hook. The user's phone is not rung in this case.

Some SIP devices (including some Aastra and Polycom phones) support a call hold reminder feature where, if a call is left on hold for longer than a certain time, the user that put the call on hold is alerted. The exact configuration and behavior is described in the SIP device manuals. If you are using a POTS line, a call hold reminder function is not available.

There is no configuration of this feature required on the Cloud Communications platform – any line with more than 1 Call Appearance configured can use Call Hold. Note that the Multiple Call Appearances feature, which is described later, is required for Call Hold.

The phone must support the Call Hold function. Typically, there is no extra configuration required on the phone to use this feature.

7.2 Music On Hold

Music On Hold allows audio chosen by the subscriber to be provided to both callers and called parties that are placed on hold by users either using SIP phones or POTS phones. This audio may include informational announcements and/or marketing material as well as music.

Music On Hold is also played to parties in

- calls that have been parked using the Call Park features
-

- queues waiting for an answer from a Multiple Appearance Directory Number or Multi Line Hunt Group.

The user can manage the Music On Hold feature using two different mechanisms.

1. A rich web interface allows the user to upload audio files and configure what audio should be played to on-hold parties. You can specify default audio for the Business Group or Department, and/or specific audio for callers to a particular line within the group.
2. A rich Telephone User Interface (TUI) is also available to record and configure audio.

Note that the Call Hold, Call Park, Multiple Appearance Directory Numbers or Multi Line Hunt Group features are required for the Music On Hold feature to operate.

This feature is enabled on a per business group basis and is managed by administrators through a web configuration portal.

8 Multiple line and call features

8.1 Multiple lines

Select IP business phones typically support multiple lines simultaneously. The user will typically have a key with associated light for each line configured on the phone. Calls can be made using particular lines by pressing the key for that line prior to dialing. Incoming calls on specific lines may either be configured to be answered when the user picks up the handset, or to require the key for the line to be selected.

Multiple lines on phones have a number of uses, such as

- having numbers with different area codes on a single phone
- having multiple users sharing the same line (often in conjunction with features such as SimRing, Call Forking and Shared Line Appearance)
- providing key system emulation (in conjunction with the Shared Line Appearance feature described below).

Each line is configured on the Cloud Communications platform.

The phone must also be configurable for each line being used.

8.2 Multiple Call Appearances

Lines can be configured to support more than 1 call simultaneously. This means that users can place calls on hold, transfer calls, and set up conferences on their phones.

This feature is enabled on a per-line basis on the Cloud Communications platform.

The phone will have to support Multiple Call Appearances for this feature to work, and it may be required or possible to configure different keys on the phone for each call appearance.

8.3 Simultaneous Ring (SimRing)

SimRing allows a line to be configured so that whenever calls are received other lines will also ring, and the call can be answered at these other lines.

SimRing is configured and managed using Cloud Voice Portal if available.

This feature is configured on a per-line basis on the Cloud Communications platform.

There is no support or configuration required on phones.

8.4 Call Forking

Call Forking allows multiple phones to register for the same shared line on the Cloud Communications platform. Each phone will ring when calls come in to this line, and any calls made from this line on any of the phones will come from the same shared line's caller ID.

This has an advantage over SimRing that there is only one Directory Number shared between every phone using this shared line.

This feature is configured on a per-line basis on the Cloud Communications platform.

There is no special support required by phones to use this feature – any SIP ATA, SIP IAD, SIP phone or SIP business phone can be configured to register for a call forking line.

8.5 Device Twinning

Device Twinning allows a subscriber to "twin" multiple SIP devices with his or her primary line, even if the primary line is analog instead of SIP. For example, device twinning can be used with the MaX UC clients described in section 2, with a device configured as an 'additional identity' on a Business Group line, for example a business mobile number.

When devices are twinned together:

- Outgoing calls from any device can appear to come from the same subscriber, or you can configure separate calling numbers for each device.
 - This is not the case if the subscriber is using an additional identity, whereupon the calling number is the same as the number configured for its parent Business Group line.
- Incoming calls to the subscriber ring all of the twinned devices simultaneously.

This feature is configured on a per-line basis on the Cloud Communications platform.

There is no support or configuration required on phones.

8.6 Shared Line Appearance

Shared Line Appearance extends the call forking feature by allowing a phone to view the current status of a line that is shared between multiple phones and provides Key System Emulation. This feature is also sometimes called Shared Call Appearance, Bridged Line Appearance or Bridged Call Appearance.

Typically the shared line will be allocated a key and an associated light.

- When calls are placed to this line the light will flash (and the phone will ring). Like the call forking feature, any phone using this shared line can answer such a call.
- When any phone is in a call on this shared line, the light associated with this line is lit solidly on all phones using this shared line.
- Any phone in a call on this shared line can place the call on hold for this line. This is typically indicated by the light associated with the line flashing (and typically the flash is a different speed than the flash for an alerting line). Any phone on this shared line can pick up this held call. Unlike the Busy Lamp Field with Directed Call Pickup call service, even after another phone picks up a call that was held on a shared line the call remains on this shared line. All other phones using the line can see the status of the call, and the call always remains on the shared line.

This feature is configured on a per-line basis on the Cloud Communications platform.

Phones using shared line appearance require support and configuration for this feature. However, this line can also be shared by lines using the call forking feature – there is no requirement that all endpoints for a shared line support shared line appearance. Any phones that do not support full shared line appearance will exhibit call forking behavior.

8.7 Simulated Facility Groups / Call Admission Control

Business groups can be configured with limits for the number of external (outside of the business group) calls which can be placed at any time. Limits can be applied on any or all of the following

- Total number of external calls.
- Total number of incoming external calls.
- Total number of outgoing external calls.

This feature is configured on a per business group basis on the Cloud Communications platform.

There is no phone configuration or support required.

9 Call Transfer features

9.1 Attended Call Transfer

This feature is sometimes called Consultative Call Transfer. This feature allows the user to transfer one caller to another party, announcing the party to be transferred to the new party before completing the transfer. The exact user experience for this feature does differ from phone to phone, but generally it consists of the following steps.

1. Placing an existing call on hold (and Music On Hold is played if this feature is configured).
2. Calling a new party, either external or internal.
3. When the new party answers announcing the party to be transferred.
4. Transferring the held party to the new party.

Note that the Multiple Call Appearances and Call Hold features are required for this feature on SIP phones.

This feature is configured on a per-line basis on the Cloud Communications platform. Cloud Communications does not differentiate between Attended Call Transfer and Unattended Call Transfer in its configuration, so enabling one of these will enable the other.

Phones require support for this feature and may require some configuration for this feature to work and to customize the behavior.

9.2 Unattended Call Transfer

This feature is sometimes called Blind Call Transfer. It involves transferring one party to another without announcing the transferred party to the new party, or waiting for the new party to answer the phone. The operations are therefore as follows.

1. Place existing call on hold (and Music On Hold is played if this feature is configured).
2. Dialing new party, either external or internal.
3. Completing transfer.

The exact user experience for this feature does differ from phone to phone. Using some phones step 2 is to select the option "Blind Transfer" and as soon as the new party is dialed the transfer is completed.

As soon as the transfer is completed, the on hold party will hear ringback – indicating the new party is being alerted. It is possible that the transferred party will receive a busy signal or have their call forwarded on, as, unlike with the Attended Call Transfer feature the transferring party has not checked the availability of the new party.

Note that the Multiple Call Appearances and Call Hold features are required for this feature.

This feature is configured on a per-line basis on the Cloud Communications platform. Cloud Communications does not differentiate between Attended Call Transfer and Unattended Call Transfer in its configuration, so enabling one of these will enable the other.

Phones require support for this feature and may require some configuration for this feature to work and to customize the behavior.

9.3 Three Way Consultative Call Transfer

This feature is a combination of three way calling and attended call transfer and operates as follows.

1. User places an existing call on hold (and Music On Hold is played if this feature is configured).
2. User calls a new party.
3. User brings original party and new party together into a three-way call.
4. User hangs up, enabling the original party and the new party to continue talking to each other.

Note that the Multiple Call Appearances and Call Hold features are required for this feature.

This feature is enabled on a per-line basis on the Cloud Communications.

This feature requires support from phones.

9.4 Call Transfer Directly to Voicemail

This feature allows users within the business group to transfer calls directly to colleagues' voice mailboxes. The user can use the Unattended Call Transfer feature to transfer the call to a special intercom code associated with the target user's mailbox.

This removes the need for the transferred party to sit and listen to ring back until the Call Forwarding No Answer timeout occurs.

9.5 One Touch Call Transfer

This feature allows the user to perform an Unattended Call Transfer to a particular destination phone number by pressing a single key on the phone. The key on the phone must be pre-configured to the destination the call should be transferred to.

An example use of this feature is that the user pre-configures a key on his phone to automatically transfer calls to his mobile phone. If the user needs to leave the office when in the middle of a call, he can press this key to cause the call to be transferred directly to his mobile phone.

This feature is enabled on a per-line basis on Cloud Communications.

This feature requires support from phones.

10 Conferencing features

10.1 Three-Way Calling

This feature allows users to initiate a conference using their phones. Typically, this is done using the following operations.

1. Place existing call on hold.
2. Call new party, either external or internal, and wait for new party to answer.
3. Conference the three parties together.

Depending on the phone it may be possible to conference the parties together before the new party has answered. In this case the ring back will be conferenced with the talking parties.

Note that the Multiple Call Appearances and Call Hold features are required for this feature for SIP phones.

This feature requires configuration on the Cloud Communications platform for POTS and Basic SIP lines.

Phones require support for this feature and may require some configuration for this feature to work.

10.2 Meet-me Conferencing

Meet-me Conferencing is provided using the N-Series Conferencing application server.

It enables moderators to easily set up conferences via an intuitive web interface, and control participant interaction and call flow via the phone interface or through the web. In deployments with MaX UC Desktop, it is possible to configure a link within the MaX UC Desktop UI to the conferencing login page.

Advanced features include:

- Reservation less meet-me conferencing for up to 500 attendees
 - Recorded greeting to be played when the conference starts
 - Lock / roll call / number of participant controls
 - Dynamically updated participant list in real time
 - Visible indication of current speaker and audio levels
 - Ability to mute, gain control and disconnect any participants
 - Out dial to invite additional participants into the conference
-

- Call recording
- On-hold music played to conference participants who join a conference before it has started (this may be a resource that is also used with the Music On Hold feature described in 7.2, or one provided specifically for Conferencing).

This feature allows business group users to have access to a meet-me conferencing system. Each user is provided a conference bridge number, moderator code and a participant code.

11 Call Waiting features

11.1 Call Waiting

Call Waiting alerts users when they are on a call and receive another call concurrently. They can choose to put the current call on hold and switch to answer the new call. The user can then switch between these two calls, either using flash hook, or with SIP phones using the Call Hold feature.

Note that the Multiple Call Appearances and Call Hold features are required for this feature to operate on SIP phones.

This feature requires configuration on the Cloud Communications platform for POTS and Basic SIP lines.

Phones require support for this feature and may require some configuration for this feature to work, and to customize the exact behavior.

11.2 Cancel Call Waiting

This feature disables the Call Waiting feature, either on a per-call basis or on a permanent basis. This is useful when the user does not wish to be disturbed mid-call with notification that another call is incoming.

Phones require support for this feature and may require some configuration for this feature to work.

11.3 Endpoint Cancel Call Waiting

This feature cancels Call Waiting on SIP devices that provide their own Call Waiting service but do not support Cancel Call Waiting.

The user activates Endpoint Cancel Call Waiting by dialing an access code prior to making an outgoing call. However, users cannot activate this service mid-call.

If Cancel Call Waiting is enabled, this service is available to all SIP subscribers using Endpoint Call Waiting.

12 Call Forwarding

When a subscriber successfully activates or deactivates a Call Forwarding service, they can either hear a confirmation tone or an announcement confirming the action they have just carried out. If you want to use an announcement, you must configure this on the Cloud Communications platform.

You can configure your deployment to support Integrated Call Forwarding, allowing Call Forwarding status to be synchronized between a subscriber's account and all compatible SIP devices registered to that account, allowing the subscriber to enable or disable Unconditional, Busy, or Delayed Call Forwarding across all devices with a single operation on any device or in Cloud Voice Portal. This can be configured for your entire platform or on a per-subscriber basis.

12.1 Call Forwarding Unconditional

This feature allows the user to set up a rule to forward all calls unconditionally to a new number. This feature has two distinct variations.

1. Switch-side. In this case the call forwarding configuration and logic is provided by the Cloud Communications platform. The feature can be configured by either of the following mechanisms.
 1. Dialing an access code to turn on or off (for example: *72 or *73).
 2. Cloud Voice Portal if it is provided.
2. Phone-side. In this case the call forwarding configuration and logic is provided by the SIP phone. This means that the configuration is done via the phone screen and keys. It also means that if the phone is rebooted or unreachable for any reason call forwarding will not take place.

Cloud Communications recommends that switch-side call forwarding is used where possible because it will always operate, even when the phone is not available. Switch-side call forwarding can also be operated using Call Forwarding Remote Access, whereas phone-side cannot.

Switch-side call forwarding is enabled and configured on a per-line basis on the Cloud Communications platform.

For phone-side call forwarding, phones require support for this feature and will require some configuration to work.

If a Business Group subscriber has an additional identity configured, for example a business mobile phone number, incoming calls are forwarded before any forking of the call to the additional identity.

12.2 Call Forwarding Busy

This feature allows the user to set up a rule to forward calls to a new number when the user is already on a call. This feature has two distinct variations.

1. Switch-side. In this case the call forwarding configuration and logic is provided by the Cloud Communications platform. The feature can be configured by either of the following mechanisms.
 1. Dialing an access code to turn on or off (for example: *90 or *91).
 2. Cloud Voice Portal if it is provided.
2. Phone-side. In this case the call forwarding configuration and logic is provided by the SIP phone. This means that the configuration is done via the phone screen and keys. It also means that if the phone is rebooted or unreachable for any reason, call forwarding will not take place.

Cloud Communications recommends that switch-side call forwarding is used where possible because it will always take place, even when the phone is not available. Switch-side call forwarding can also be operated using Call Forwarding Remote Access, whereas phone-side cannot.

Switch-side call forwarding is enabled and configured on a per-line basis on the Cloud Communications platform.

For phone-side call forwarding, phones require support for this feature and will require some configuration for this feature to work.

12.3 Call Forwarding No Answer

This feature allows the user to set up a rule to forward calls to a new number if the user doesn't answer the call within a pre-configured delay time. This feature has two distinct variations.

1. Switch-side. In this case the call forwarding configuration and logic is provided by the Cloud Communications platform. The feature can be configured by either of the following mechanisms.
 1. Dialing an access code to turn on or off (for example: *92 or *93).
 2. Cloud Voice Portal if it is provided. Note that the delay time can be configured via Cloud Voice Portal but not via access codes.
2. Phone-side. In this case the call forwarding configuration and logic is provided by the SIP phone. This means that the configuration is done via the phone screen and keys. It also means that if the phone is rebooted or unreachable for any reason call forwarding will not take place.

Cloud Communications recommends that switch-side call forwarding is used where possible because it will always take place, even when the phone is not available. Switch-side call forwarding can also be operated using Call Forwarding Remote Access, whereas phone-side cannot.

Switch-side call forwarding is enabled and configured on a per-line basis on the Cloud Communications platform.

For phone-side call forwarding, phones require support for this feature and will require some configuration for this feature to work.

12.4 Call Forwarding Selective

This feature allows the user to set up rules to forward calls to a new number depending on the incoming caller ID, so that only calls from specific numbers will be forwarded. This feature has two distinct variations.

1. Switch-side. In this case the call forwarding configuration and logic is provided by the Cloud Communications platform. The feature can be configured by either of the following mechanisms.
 1. To configure selective call forwarding an access code is dialed. An IVR system then guides the user through configuring the system or turning this variation of call forwarding off.
 2. Cloud Voice Portal if it is provided.

The maximum number of unique calling numbers which can be handled by Call Forwarding Selective is configured on a per-switch basis. The default is 10.

2. Phone-side. In this case the call forwarding configuration and logic is provided by the SIP phone. This means that the configuration is done via the phone screen and keys. It also means that if the phone is rebooted or unreachable for any reason call forwarding will not take place.

Cloud Communications recommends that switch-side call forwarding is used where possible because it will always take place, even when the phone is not available. Switch-side call forwarding can also be operated using Call Forwarding Remote Access, whereas phone-side cannot.

Switch-side call forwarding is enabled and configured on a per-line basis on the Cloud Communications platform. This will take precedence over Unconditional Call Forwarding if that is also configured.

For phone-side call forwarding, phones require support for this feature and will require some configuration for this feature to work.

12.5 Call Forwarding Ringsplash

Call Forwarding Ringsplash is an indication to the user when a call is forwarded using Call Forwarding Unconditional or Call Forwarding Selective, by means of briefly ringing the phone. (It does not apply to the other types of Call Forwarding.)

This feature is configured on the Cloud Communications platform.

This feature is only supported for POTS lines.

12.6 Call Forwarding Remote Access

This feature provides a telephone number for users to call from any phone with PSTN access to set up the Call Forwarding Unconditional, Busy, No Answer and Selective switch-side features. An IVR menu system guides the user through the various configuration options.

This feature is enabled on a per-line basis on the Cloud Communications platform.

There is no phone configuration required for this feature.

12.7 Call Forwarding on Unavailable

This service forwards calls from the subscriber's line to an alternative number only if the dialed number is unavailable, for example if the telephone is unplugged or loses power. This includes calls that cannot be connected to the dialed number for one of the following reasons.

- The subscriber's telephone is unplugged.
- The destination SIP phone is not registered.
- The destination SIP phone is alarmed as not contactable.
- There is no phone support or configuration required for this feature.

This service can also be enabled on MLHG Pilot DN's.

12.8 Find Me Follow Me

Find Me Follow Me is a feature that allows users to specify a series of different numbers (or combinations of numbers) to be rung in sequence when an incoming call arrives at their number. An example set of rules is as follows.

- Ring my desk phone for 10 seconds.
- Then ring both my home phone and my mobile phone for 20 seconds.
- Finally, ring my supervisor for 10 seconds.

If none of the calls is answered then the line's usual behavior for a call not being answered will be applied – such as sending to voicemail.

This feature is enabled and configured on a per-line basis on the Cloud Communications platform and the user configures it using Cloud Voice Portal.

Support is not required from the user's phone.

13 Unified Messaging features

13.1 Voicemail

Voice mailboxes can be applied to both individual business group lines, and other, special, lines, such as Multiple Appearance Directory Numbers and Multi-Line Hunt Groups.

Control of forwarding to voicemail is available in the following different variations.

1. In the first option, voicemail replaces Call Forwarding Busy and No Answer, and, when any of these treatments is to be received, the incoming call is diverted to the voice mailbox of the line being rung.
2. In the second option, the user can, using the standard Call Forwarding Unconditional, Busy, No Answer and Selective access codes or Cloud Voice Portal (if this feature is provided), set up rules for forwarding to voicemail for any combination of the above conditions. This is useful, for example, if the user wants to be able to direct calls to voicemail when unanswered, but forward to a different phone when busy.
3. In the third option, users can set up call forwarding to their voice mailboxes using any Call Forwarding options provided by their phone. This mechanism is typically not recommended as it will not operate when the phone is unavailable.

To provide the second option above, the Call Forwarding features must be configured for the user on the Cloud Communications platform in addition to the voicemail feature.

A voice mailbox and the voicemail feature are enabled and configured on a per-line basis on the Cloud Communications platform for options 1 and 2 above. For option 3, only a voice mailbox must be provisioned on the Cloud Communications platform.

To support the third option above, support for phone-side Call Forwarding is required by the phone, and configuration will be required on the phone.

13.2 Video messaging

Users with MaX UC Mobile or MaX UC Desktop can use the video messaging feature to deposit and retrieve video messages, or record custom video greetings.

This feature is enabled on a per line basis on the Cloud Communications platform.

Phone support and configuration is required for this feature.

13.3 Faxmail

As a complementary service to Voicemail, the user may also be given the faxmail feature. This provides the user with a second phone number for faxmail, and any faxes are delivered into a single mailbox for both voice and fax.

A faxmail number must be provisioned and this feature must be enabled on the Cloud Communications platform.

There is no phone support or configuration required for this feature.

13.4 Voicemail/Faxmail access

There are a number of options available for the user to access the contents of his or her mailbox.

1. Cloud Voice Portal, if this feature is provided, can be used to view and listen to all new and previously retrieved (and archived) voice messages.
2. TUI access is provided to the mailbox using an access code. This allows the user to listen to, mark as read, mark as unread or delete voice messages. Voice messages can also be forwarded to other mailbox users. Faxes can be directed to a fax machine for printing by entering the Directory Number of a fax machine.
3. On some phones a key can be programmed for one-touch access to voicemail, rather than dialing the full access code.
4. For remote access to the mailbox TUI a well-known Directory Number can be provided to the user. This can then be dialed from any phone to access the user's mailbox. This access mechanism is protected by a PIN to prevent unauthorized access.
5. There is a high-function password protected web interface to voice and fax mailboxes separate from Cloud Voice Portal. In addition to all the Cloud Voice Portal features and the TUI access, this web interface also allows functions such as
 - viewing faxes on screen
 - setting various mailbox options.

There is no separate explicit configuration for these access features on the Cloud Communications platform.

For option 3, phone support and phone configuration are required.

13.5 Audible Message Waiting Indicator

To inform users when there is unread mail in their mailboxes, some phones support an audible message waiting indicator. This is typically a stutter dial tone instead of the normal continuous dial tone and resets to the continuous dial tone when there is no unread mail.

This feature is enabled on a per line basis on the Cloud Communications platform.

13.6 Visual Message Waiting Indicator

To inform users when there is unread mail in their mailboxes, some phones support a visual message waiting indicator. This typically flashes to indicate there is unread mail in the user's voice and fax mailbox and clears when there is no unread mail.

This feature is enabled on a per line basis on the Cloud Communications platform.

Phone support and configuration is required for this feature.

13.7 Voicemail personal greetings and name recording

The voicemail system allows users accessing the TUI mechanism to record their names and their greetings for callers who are forwarded to their individual mailboxes.

This feature is automatically enabled as part of provisioning a voice mailbox.

There is no phone support or configuration related to this feature.

13.8 Voice and fax messages to email

The users can configure their mailboxes, using the high-function web interface, to forward any new voice and fax messages automatically to any user-specified email address.

This feature is automatically provided as part of the Voicemail feature, but the high-function web interface must be provided to allow for user configuration.

There is no phone support or configuration related to this feature.

14 Outgoing call management features

14.1 Validated Account Codes

Lines can be configured to require Validated Account Codes. Calls to certain types of destinations will provide a separate prompt after the number has been dialed to allow an account code to be entered. If a Valid Account Code has been entered the call will be allowed to complete. This feature is typically used to ensure that only authorized employees can make certain types of calls.

Groups of numbers which can be configured to require Account Codes are as follows.

- International
- National (including long-distance, interLATA, intraLATA and, in areas which are part of the North American Numbering Plan (NANP), cell phone calls)
- Mobile (this group is only available in areas which are not part of the NANP)
- Local
- Local Business Group (made using intercom codes within the group, billed as local calls)
- Other Business Group (made using intercom codes within the group, billed as anything other than local calls)
- Operator
- Premium Rate
- Directory
- Regional
- Carrier Dialed

Note that emergency calls, toll-free calls, and calls to service access codes will never require an account code.

This feature can be administered by the business administrator using the Cloud Voice Portal business group administration interface.

This feature must be configured on a per-business group basis on the Cloud Communications, and particular lines can have this feature enabled.

No support or configuration is required on the phones.

14.2 Unvalidated Account Codes

Unvalidated Account Codes differ from Validated Account Codes in that the account code entered is not checked against a list of Valid Account Codes. Instead the call is permitted to complete once an Account Code of the correct length has been entered. The Account Code is output in the billing CDR produced by Cloud Communications and can be provided by us to the SMB. This feature is typically used by SMBs for internal accounting processes.

Groups of numbers which can be configured to require Account Codes are as follows.

- International
- National (including long-distance, interLATA intraLATA and, in areas which are part of the North American Numbering Plan (NANP), cell phone calls)
- Mobile (this group is only available in areas which are not part of the NANP)
- Local
- Local Business Group (made using intercom codes within the group, billed as local calls)
- Other Business Group (made using intercom codes within the group, billed as anything other than local calls)
- Operator
- Premium Rate
- Directory
- Regional
- Carrier Dialed

Note that emergency calls, toll-free calls, and calls to service access codes will never require an account code.

This feature can be administered by the business administrator using the Cloud Voice Portal business group administration interface.

This feature must be configured on a per-business group basis on the Cloud Communications, and particular lines can have this feature enabled.

No support or configuration is required on phones.

14.3 Outgoing call blocking

Outgoing call blocking allows you to set up business group lines that are only permitted to call certain types of numbers. Groups of numbers which can be explicitly enabled or disabled are as follows.

- International
- National (including long-distance, interLATA and intraLATA calls) and, in areas which are part of the North American Numbering Plan (NANP), Cell phone
- Mobile (this group is only available in areas which are not part of the NANP)
- Local
- Operator
- Premium Rate
- Star Codes
- Star Codes that modify configuration

Note that emergency calls, toll-free calls, and calls to service access codes cannot be barred.

Additionally, you can configure Business Group lines so that users can continue to make intra-BG calls when outgoing call blocking is enabled.

You can also configure enhanced and selective call barring, for example to bar all international calls except to selected numbers.

This feature is enabled and configured on the Cloud Communications platform.

There is no phone support or configuration related to this feature.

14.4 High risk call limits

Setting High risk call limits reduces the exposure to toll fraud where a malicious user guesses a subscriber's password and uses the compromised SIP line to place multiple calls to international or premium rate numbers by setting limits on the number of calls which can be made to these numbers.

Number Validation must be setup to identify and flag high risk calls with a High risk call attribute. Calls flagged as high risk then count towards this limit and may be rejected once the call limit is breached. This call limit can either be a Soft Limit, where CFS records when a line breaches this limit but does not reject the call, or a Hard Limit where calls are rejected once the number of calls to these numbers exceeds this limit.

14.5 Service suspension

Suspend service to lines or entire business groups, for example on non-payment of bills. Calls will still be allowed to complete to the emergency services.

No support or configuration is required on phones.

14.6 Calling card

Cloud Communications supports "calling card" services. A user is provided a (typically toll-free) access number and a PIN code. The user can then place a call from any phone in the world, and have that call be billed to their SMB-hosted PBX account.

No support or configuration is required on phones.

14.7 Outgoing Call Notifications

Outgoing Call Notifications allow Business Group administrators to configure email and/or out dial notifications which are sent to designated recipients to alert them whenever a line in their Business Group or Department makes an emergency call.

This feature is enabled and configured on the Cloud Communications platform.

There is no phone support or configuration related to this feature.

15 Incoming call management features

15.1 Easy Attendant

The Easy Attendant feature provides businesses with a customizable automated receptionist service consisting of

- automatic call answering
- user-configurable hierarchical menus and announcements
- separate menus and announcements for use outside normal working hours (including holidays)
- a range of standard menu actions:
 - play an announcement
 - transfer to phone number or extension
 - transfer to voice mailbox
 - transfer to operator
 - dial-by-extension, including an option for a caller to choose to dial a known extension at any point in the menu and not just on a dial-by-extension menu node
- Administration via telephony user interface (TUI) or Web
- Recording upload/download via Cloud Voice Portal, in addition to the built in Cloud Voice Portal recording and self-care TUI recording.

This feature is enabled for a specific DID number on the Cloud Communications platform.

15.2 Premium Attendant

The Premium Attendant feature is based on Easy Attendant but allows more complex menu structures for increased flexibility. Capable of handling thousands of calls per hour, it is suitable for businesses of all sizes up to 1,000 employees. Completely configurable through the Cloud Voice Portal Administration interface, the intuitive configuration model allows a selection of features, including

- up to 150 fully customizable menus, with an enhanced range of menu actions compared with Easy Attendant
 - a separate announcement pool – allowing recordings to be managed independently of menus
 - up to nine fine-grained weekly schedule periods, plus special days
 - control over global key presses
 - recording upload/download via Cloud Voice Portal, in addition to the built in Cloud Voice Portal recording and self-care TUI recording.
-

15.3 Do Not Disturb (Make Set Busy)

The Do Not Disturb (DND) feature allows the user to block all incoming calls to a particular line or phone. In the world of legacy phones and PBXs this feature is often called Make Set Busy.

There are two ways to implement this feature.

1. Some phones support this function natively, providing a key on the phone specifically for this purpose. DND is enabled by pressing this key once, and the phone's display will indicate that DND is enabled. DND is disabled by pressing this key again. The phone's display updates to indicate DND is now off.

Note that if the phone has multiple lines, DND may operate for all lines simultaneously, or may only operate on the selected line depending on the phone.

2. To support phones without explicit DND support, this feature is also available within the Cloud Communications by using one access code to enable DND and another to disable. Cloud Voice Portal can be used to display and modify the DND status of a line.

When implemented in this way, support can also be provided for Do Not Disturb integration. This allows Do Not Disturb status to be synchronized between a subscriber's account and any SIP devices registered to that account, so that the subscriber can enable or disable Do Not Disturb across all devices with a single operation on any device or in Cloud Voice Portal.

While DND is enabled all incoming calls to this line/phone will be rejected with the regular busy treatment (such as Call Forwarding Busy, Voicemail, Find Me Follow Me, etc.). There will be no indication from the phone that the call has been rejected.

For option 1, this feature must be enabled and configured on the phone.

For option 2, this feature is enabled and configured on the Cloud Communications platform. No phone configuration is required. Your deployment can also be configured so that subscribers must enter their Call Services PIN before they can change their Do Not Disturb setting to prevent misuse.

15.4 Selective Call Acceptance

When switch-side Do Not Disturb is being used (option 2 described above), the Selective Call Acceptance feature can also be offered. This allows calls from certain configured numbers to be allowed through to the phone even when Do Not Disturb is enabled. There are two ways for the user to manage this list of allowed numbers.

1. Using Cloud Voice Portal if available.
2. A TUI is provided with IVR system to allow users to manage their acceptance lists.

This feature is enabled on a per-line basis on the Cloud Communications platform.

There is no support or configuration required on the phone.

15.5 Individual Incoming Call Rejection

This feature allows a user receiving an incoming call to decide to reject the call, typically by pressing a "Deny", "Ignore" or "Reject" key on their phone. This call will then be given the standard busy treatment for the line (such as voicemail).

There is no Cloud Communications configuration for this feature.

This feature is provided by the phone, so phone support and configuration is required.

15.6 Individual Incoming Call Forwarding

This feature allows a user receiving an incoming call to decide to forward the call, typically by pressing a "Forward" key on the phone. This call will then be transferred using Unattended Call Transfer to the selected destination.

There is no Cloud Communications configuration for this feature.

This feature is provided by the phone, so phone support and configuration are required.

15.7 Selective Call Rejection

Selective Call Rejection allows users to configure certain numbers from which they do not wish to take calls. Incoming calls from these numbers will be provided with an announcement telling the caller that their call has been rejected.

There are two ways for the user to manage this feature.

1. Cloud Voice Portal, if available, can be used to configure rejection lists.
2. A TUI is provided with IVR system to allow the users to manage their rejection lists.

This feature is configured on a per-line basis on the Cloud Communications platform.

This feature requires no support or configuration on phones. However, some phones may also provide this feature natively.

16 Call Recording

This feature is deployed in conjunction with a 3rd party Application Server that provides the SIPrec call recording interface, allowing you to record calls for quality monitoring or training purposes, or to satisfy regulatory requirements for recording calls.

Call recording can be applied to a Business Group Line, to a MADN or MLHG Pilot DN, to a PBX within a Business Group, or to the complete Business Group. It can also be enabled on intra-Business Group calls. It records all media for calls to or from the relevant lines, as it would be heard by the call participants.

You can optionally configure a Call Recording warning tone that plays at the start of, and during, all recorded calls to alert the call participants that the call is being recorded.

This feature is configured on the Cloud Communications platform.

Business Group administrators can use the Cloud Voice Portal BG Admin interface to enable or disable call recording for specific lines within the Business Group and on intra-Business Group calls, or click through to the management UI provided by the recording server in order to manage call recordings.

Many jurisdictions have laws, including wiretapping laws, which regulate the privacy of calls. These laws and regulations may restrict or prohibit the use of the call recording feature. For example, in some jurisdictions a call participant must provide notice to all other call participants before the call. Subscribers are responsible for ensuring that any use of this feature complies with all applicable laws and regulations.

There is no phone support or configuration related to this feature.

17 **Reminder Call features**

17.1 **Reminder Calls**

The Reminder Call feature allows users to configure a reminder call to be placed to their individual lines at the configured time with an announcement indicating that this is a reminder call.

This feature can be configured in one of two ways.

1. Cloud Voice Portal, if available, can be used to configure the required reminder call time, or cancel a reminder call.
2. An access code can be dialed followed by the desired reminder time. An access code can also be used to cancel the reminder call.

This feature is configured on a per-line basis on the Cloud Communications.

There is no phone support or configuration required for this feature.

17.2 **Recurring Reminder Calls**

This feature allows the user to configure daily or weekly reminder calls at certain pre-configured times.

This feature can be configured in one of two ways.

1. Cloud Voice Portal, if available, can be used to configure the required reminder call time and day, or cancel a specific recurring reminder call.
2. An access code can be dialed providing an IVR menu to configure the day and time of call. This can also be used to cancel specific recurring reminder calls.

This feature is configured on a per-line basis on the Cloud Communications.

There is no phone support or configuration required for this feature.

18 Ringtone features

18.1 Priority Call

Priority Call allows users to set up different ringtones on their phones depending on who is calling. Both internal and external callers can be specified.

There are two ways of providing and configuring this service.

1. The switch can provide this function, and it is configured using the Cloud Voice Portal Web interface.
2. Phones can often also provide this function natively, without requiring the switch to perform the function. In this case, priority call will be configured using the phone screen and keys, or a phone web interface.

In option 1 this feature is enabled and configured on the Cloud Communications platform. Phone support is required to support the priority call tones for this feature and configuration of the phone may be required.

For option 2, this feature must be supported and configured on the phone.

18.2 Group Priority Call

Business groups can be configured for Group Priority Call. This enables either calls from within the business group (made using intercom codes) or calls from outside to ring with a distinctive ring on any of the phones within the business group.

This feature is enabled and configured on a per business group basis on the Cloud Communications platform.

Phone support is required to enable the priority call tones for this feature and configuration of the phone may be required.

18.3 Outgoing Priority Call

When calling internal numbers within the business group using intercom codes, this feature can be used for the originated caller to specify the ringtone desired on the calling phone. For example, this allows executives to announce they are calling without requiring pre-configuration for this on the calling party's line.

This feature is configured on a per business group basis on the Cloud Communications platform.

The phones used in the business group must support and be configured to provide this feature.

19 Dialing features

19.1 Direct Inward Dialing

Each business group line has a DID associated with it that can be called directly by external callers.

Phones may be able to support multiple lines (and therefore DIDs) simultaneously. Some phones can be configured to treat incoming calls for different DIDs differently, such as

- different ringtones
- different lights flash
- different keys allow the call to be picked up
- different call forwarding settings

This feature is provided by the Cloud Communications platform.

Phones must also support these features.

19.2 Configurable Dial-Plans

The dial-plan desired for a business group can be configured. This will be generated by the Service Provider, typically using the following information as input.

- Whether or not Assume-9 dialing is required (see section 19.3) or if a key is required for external calling.
- Format of intercom codes.
- Format of group short codes.
- User speed dials.
- Format of call park codes.

The dial-plan is built on the Cloud Communications platform for the business group, and only numbers meeting this plan will be accepted by the Cloud Communications platform.

Phones may support their own dial-plans. To use these, the phones must be configured with a suitable dial-plan to match the Cloud Communications dial-plan. This allows the phone to dial the number as soon as the user has finished dialing, rather than require the user to hit a "Dial" or "OK" key once the number has been entered.

19.3 Assume 9 Dialing

Business groups can be configured to make external calls either by dialing an access prefix (such as 9), or simply by normal 1+10 digit dialing, or other dialing patterns.

This feature is configured and enabled on the Cloud Communications platform.

There is no phone support or configuration required for this feature.

19.4 Intercom Codes

Intercom Codes provide shortcut numbers for dialing extensions within a business group, so that the user does not need to dial the full external DID for each line. You can also choose to set an option so that the name associated with the dialed number is displayed as soon as the phone starts ringing.

The business group administrator can use the Cloud Voice Portal interface to reconfigure intercom codes for the corresponding business group.

This feature is enabled and configured on the Cloud Communications platform.

There is no support or configuration required on phones for this feature.

19.5 Short Codes

Short Codes are business group wide speed dials. These are configured by the business group administrator using Cloud Voice Portal and are available to dial from all business group lines. These codes can be up to seven digits in length.

This feature can also be used to configure a short code to bypass an IVR, allowing users to access specific call services without having to listen to an IVR menu. For example, *72 + <Directory Number> activates Call Forwarding Unconditional with no IVR prompts.

This feature is enabled and configured on the Cloud Communications platform.

There is no support or configuration required on phones for this feature.

19.6 Speed Dials

Speed dials are specific to individual business lines and/or phones within business groups. There are two ways to implement this feature.

1. Using the Cloud Communications speed dials feature. This allows the user to configure and manage their speed dials using Cloud Voice Portal. These speed dials are dialed using a one- or two-digit number associated with a particular speed dial, followed by #.
2. Some phones also provide this feature natively. They allow a list of names and directory numbers to be configured using the phone screen and keys, or the phone's web UI. To perform this type of speed dials a user typically selects from the list of speed dials via a key on the phone and then selects the desired entry from the list to be dialed. Sometimes the phone also allows a (short, typically one or two digit) number to be associated with the speed dial, so the speed dial can be called using this number.

Depending on the phone this feature can also be used to share speed dials across multiple phones. Some phones allow a directory to be downloaded containing speed dials, and this same directory can be shared across multiple phones.

Speed dial keys on phones are sometimes called Direct Station Select keys.

For option 1, this feature is enabled and configured on the Cloud Communications platform. No phone configuration is required unless dial plans are used on the phone. If dial plans are used on the phone it may be desirable to add the speed dials to the dial plan.

For option 2, this feature must be enabled and configured on the phone.

19.7 Speed Dials with Call Transfer

This feature allows the user to perform an Unattended Call Transfer to a speed dial number by pressing a single key on the phone.

An example use of this feature is that the user pre-configures a key on his phone to automatically transfer calls to his mobile phone. If the user needs to leave the office when in the middle of a call, he can press this key to cause the call to be transferred directly to his mobile phone.

This feature is enabled on a per-line basis on the Cloud Communications platform.

This feature requires support from phones.

19.8 Hotline

The hotline feature causes the line to call a pre-configured number automatically when the phone goes off-hook. This is typically used to implement courtesy and lobby phones.

For POTS phones this feature is enabled and configured on a per-line basis on the Cloud Communications platform.

SIP phones require explicit support for, and configuration of, this feature. There is no configuration on the Cloud Communications for this feature with SIP phones.

* – The ATA, IAD, DLC or BLC providing FXS ports polices the digit map, so it needs to have configurable dial-plans.

** – The MaX UC Clients have an address book with the ability to set contacts as favorites, fulfilling the same purpose. This also applies to mobile phones with Network Voice if they have MaX UC Mobile installed.

*** – Since Network Voice is generally provided by a mobile network operator, dialing patterns that require codes must be configured not to overlap with codes in use by the MaX UC Mobile operator if present.

20 Call Park features

"Call Park" is the term applied when a call is put on hold in such a way that it can be retrieved from hold by any one of a number of different lines/phones. The term "Park Orbit" refers to the location of the call when it is parked. A park orbit is referred to by a number (possibly including * or #), known as a park orbit code, such as 234 or *1. The party on a call that has been parked will be provided any Music On Hold treatment that has been configured for that business group.

The following sections describe the various Call Park features available.

All of the Call Park features described by this document can be used within the same Business Group simultaneously. Therefore, a call parked using the Basic Call Park can be monitored using Enhanced Park Orbit Monitoring and picked up using either Basic Parked Call Retrieve or Enhanced Parked Call Retrieve. Alternatively, a call parked using Enhanced Call Park can be retrieved using Basic Call Park Retrieve.

Call Park Orbit Code Ranges can be configured for use by a particular Department within a Business Group, so that parked calls can only be accessed by users in the appropriate Department.

As not all phones support all call park capabilities this allows all phones within a business group to access call park, even if there is a mixture of SIP business phones, basic SIP phones, and black phones. As not all SIP business phones support all call park capabilities, a mixture of the features can also be configured on these phones to provide full call park access. The ideal combination of Call Park features for the richest end-user experience is

- Enhanced Call Park
- Enhanced Park Orbit Monitoring
- Enhanced Parked Call Retrieve.

20.1 Basic Call Park

Basic Call Park is available on any phone, whether a SIP business phone, basic SIP phone or black phone and is operated as follows.

- For black phone users to park a call, they flash hook (briefly depress call termination button or cradle reset) and dial the call park access code. The switch plays an announcement indicating the call is to be parked and the park orbit code of the orbit the call will be parked on. The user then hangs up to complete the call park.
 - When using a SIP phone, the user puts the call to be parked on hold or indicates it is to be transferred (depending on the exact phone being used) and calls the call park access code. The switch announces the park orbit code and the user then transfers the held call to complete the call park.
-

This feature is enabled and configured on the Cloud Communications platform. SIP phones may need to be configured to allow the transferring of calls. It may also be desirable to configure the access code used to invoke call park on a softkey on the phone.

20.2 Basic Parked Call Retrieve

Basic Parked Call Retrieve is available on any phone and is operated as follows.

- For a black phone, the user dials the access code for call park retrieve. Stutter dial tone is provided indicating that the user should enter the park orbit code of the call to be retrieved. Once the user has entered this code the parked call is connected to this user.
- When using a SIP phone, mobile phone or softphone, the user dials the access code for call park retrieve plus the park orbit code in one string. For example, if the access code is *12 and the park orbit is 234, the user dials *12234. The user is connected to the call that was parked.

This feature is enabled and configured on the Cloud Communications platform. For SIP phones it may be desirable to configure the access code and park orbit code used to invoke call park retrieve for each orbit on softkeys on the phone. This feature then becomes very similar to Enhanced Park Orbit Retrieval.

20.3 Enhanced Call Park

Enhanced Call Park makes parking a call a one-touch operation. Users have a key on their phones (or digital button, in the case of MaX UC Desktop) associated with parking the call in a particular orbit. When in a call they wish to park they simply press the key to park the call onto that orbit.

As for Basic Call Park, this feature is enabled and configured on the Cloud Communications. In addition, a key must be configured on the phone for each orbit available to the user for call parking.

This feature is normally used in conjunction with Enhanced Park Orbit Monitoring and Enhanced Parked Call Retrieve.

20.4 Enhanced Park Orbit Monitoring

Enhanced Park Orbit Monitoring allows the user to view park orbits to determine whether they have calls parked in them. For a physical phone, lights on the phone (usually associated with keys) are configured to monitor specific park orbits. When these park orbits are empty the light is off, and when calls are parked in orbits, these lights illuminate. On MaX UC Desktop, users can display a Call Park pane that shows all the Call Park orbits that the user belongs to, from which they can park and retrieve calls by clicking appropriate buttons.

You can configure Call Park Orbit Ranges for use in the entire Business Group, or just for use by one or more Departments within the Business Group, giving you more control over the use of Call Park by ensuring that parked calls can only be accessed by users in the appropriate department(s).

As for Basic Call Park, this feature is enabled and configured on the Cloud Communications platform. In addition, a light/key must be configured on a physical phone for each orbit the user is able to monitor.

This feature is normally used in conjunction with Enhanced Call Park and Enhanced Parked Call Retrieve but may be used independently for phones that do not support the other Enhanced call park features.

20.5 Enhanced Parked Call Retrieve

Enhanced Parked Call Retrieve allows the user to retrieve a parked call by pressing a key or clicking a button associated with a specific park orbit.

As for Basic Call Park, this feature is enabled and configured on the Cloud Communications platform. In addition, a light/key must be configured on physical phones for each orbit the user is able to monitor.

This feature is normally used in conjunction with Enhanced Call Park and Enhanced Park Orbit Monitoring but can be used independently for phones that do not support the other Enhanced call park features.

21 Call Pickup features

21.1 Group Call Pickup

Group Call Pickup allows users to pick up calls that are ringing on other phones (in a pre-defined pickup group within the same business group) by picking up their own phones and dialing an access code. This feature does not require the user to know which other line is ringing, but neither does it allow for the user to specify which call to pick up, if two or more phones are ringing simultaneously. In this case, only one of the calls will be picked up.

This feature is enabled and configured on the Cloud Communications platform. It may be possible to configure a specific key for this service on a SIP phone.

21.2 Directed Call Pickup

Directed Call Pickup allows users to pick up a call that is ringing on a specific phone in a pre-defined pickup group within the same business group. Users pick up their own phones and dial an access code followed by the intercom code for the line which is ringing.

This feature is enabled and configured on the Cloud Communications platform. It may be possible to configure a specific key for picking up a particular line on a SIP phone.

22 Paging features

22.1 Intercom

The intercom feature enables a user to make a call to another line within the business group so that the other line auto-answers to that user's speakerphone. It is often used in executive/assistant arrangements.

To set up the call, users press a key on the phone which is configured to make an Intercom call to the other party. The calls are then launched and auto-answered by the remote phone to speakerphone. To hang-up the call the user that originated the call presses the intercom key again.

This feature is enabled and configured on the Cloud Communications platform. It must also be configured on both the originating and terminating phone.

22.2 Push To Talk

This feature is a variant of Intercom, where the intercom call is launched and kept up for as long as the user making the call keeps the Push To Talk key depressed. As soon as the key is released the call is disconnected.

This feature is enabled and configured on the Cloud Communications platform. It must also be configured on both the originating and terminating phone.

22.3 Paging Groups

This feature allows a user to configure a set of phones that will receive a paging call, much like a public announcement system. At the press of a button, all the phones in a paging group will activate the speakerphone, allowing the caller to make an announcement to all within earshot of the phones in the paging group.

This feature is enabled and configured on the Cloud Communications platform and requires support from phones.

Note that Polycom phones cannot be put into a paging group with phones from another manufacturer.

23 Hunt Group features

23.1 Multiple Appearance Directory Numbers (Group SimRing)

A Multiple Appearance Directory Number (MADN) is a directory number with no phone or line associated with it. Instead the MADN is configured to have a number of MADN members, each of which is a directory number with a line associated with it. Whenever a call comes in to the MADN all of the lines whose directory numbers are MADN members will be rung. When any of these phones is answered, the rest of the phones will stop ringing.

A line may be a member of one more than one MADN. In this case, it rings when a call comes in to any of these MADNs.

MADNs can also be configured to enable queuing. If configured, calls which come in to a MADN whose members are all busy will be queued. These queued calls will receive Music On Hold if this feature is configured.

You can configure whether to allow calls direct to a MADN member's directory number (rather than to the MADN itself). If permitted, these calls bypass the MADN and use the called number's services. Similarly, calls made from the MADN member's phone use the line's caller ID and call services rather than those of the MADN, although you can configure the calling party number on these lines to match the MADN number if required.

This feature is enabled and configured on the business group on the Cloud Communications platform.

There is no support required from the user's phone.

23.2 Multi Line Hunt Groups

A Multi Line Hunt Group (MLHG) allows sequential ringing of available phones for use in applications such as call centers.

An MLHG consists of a number of lines within the business group, known as members. MLHGs can also be configured to include external lines outside the parent Business Group as MLHG members. This requires explicit configuration and is not supported if the MLHG uses Cloud Communications Integrated ACD feature.

When a call comes into the hunt group, a hunt algorithm will decide which phone to ring. If this phone is busy or is not answered the hunt algorithm will move on to another phone.

- Hunt groups can have a pilot number, which is a directory number associated with the MLHG. When this pilot number is called the members of the MLHG are alerted in turn.
-

- Alternatively, a hunt group can be configured to hunt if any member of the hunt group is called directly using its DID number and does not answer. However, calls dialed directly to an external MLHG member are never forwarded within the MLHG. Hunt 'on direct' also does not apply if the Business Group subscriber has been configured with an additional identity, for example a business mobile phone number.

There are various distribution algorithms available:

- Linear – hunting starts at the first configured line and goes sequentially through the members in the list in the order they were configured.
- Circular – hunting starts at the line which has been called, and then goes through as in the linear case.
- Uniform – hunting starts at the first line after the line which was called the last time this MLHG was called.
- Most idle – hunting starts at the line which has been idle for the longest period of time. After this it tries the line which has been idle the next longest period, etc.
- Ring all – all free MLHG members ring simultaneously until one member answers the call.

MLHGs can also be configured to enable queuing. If configured, calls which come in to an MLHG whose members are all busy will be queued. These queued calls will receive Music On Hold if this feature is configured. You can also choose whether or not to interrupt the Music On Hold with ring back when a held call leaves the queue and starts alerting an MLHG member.

MLHGs can also be configured to allow a caller to press zero to exit the queue and have their call redirected to voicemail or forwarded to a defined phone number. If Music on Hold or Incoming Call Manager functionality is present, the caller can hear an announcement advising them of this option.

For MLHGs operating as small call centers, the Integrated ACD (Automated Call Distribution) feature provides your Business Group administrators with an additional Cloud Voice Portal interface that displays information about the activity of the MLHGs (referred to as queues) and MLHG members (referred to as agents) that they are managing, and allows them to download detailed statistics and reports in .csv format. The Integrated ACD feature is described in more detail in section 23.4.

This feature is enabled and configured on the business group on the Cloud Communications platform.

There is no support required from the user's phone.

23.3 Multi Line Hunt Group Login/Logout

The MLHG login/logout feature allows a user to be configured to be a member of multiple MLHGs simultaneously. A line can be active in multiple MLHGs at any time.

- To move between hunt groups, the user dials an access code to log out of the existing active MLHG and another access code to log into another hunt group. Alternatively, logging into a new MLHG will automatically log the user out of the old one.
- To stop receiving MLHG calls temporarily, the user dials an access code to log out of the existing active MLHG and does not log into another group. Alternatively, the user can enable Do Not Disturb to stop receiving all calls (including MLHG calls).

This feature is enabled and configured on a per-line basis on the Cloud Communications platform.

There is no support required from the user's phone.

A BG line can be any combination of the following:

- A member of multiple MADNs.
- An explicit login member of multiple MLHGs, but only logged into one at time (as per the old behavior).
- An implicit login member of (and permanently logged into) multiple MLHGs.

This removes the need for awkward MLHGs and MADNs nesting configurations when the same goals could be achieved using multiple membership.

Configuration

Configuration is straight forward for MADN membership. A member can be added without any other considerations (apart from the soft limit mentioned below).

For MLHGs, you must choose whether a BG line will have implicit or explicit membership.

- Implicit membership means that the BG line is always an active member of the MLHG.
 - Explicit membership allows the BG line to choose when to be part of the MLHG. The BG line subscriber can use an access code (see CFS -> Call Services -> Incoming Call Services) to log in or out of a MLHG. The same can be achieved using Cloud Voice Portal.
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Considerations

Soft limit: There are no restrictions on how many groups a BG line can be a member of, however, an upper soft limit of 20 memberships per BG line exists and an alarm will be raised when the limit is exceeded.

This soft limit configuration is analyzed daily, and an alarm is raised if the limit is exceeded for any BG line.

Members' availability: As a subscriber can be a member of both a MADN and MLHG, there may be situations in which the subscriber becomes the only available member in both a MADN and a MLHG. The following is an example situation.

- A subscriber is a member of two MADNs and two MLHGs and is currently busy. All other MADN and MLHG members are also busy and calls to the groups are being queued.

At this point the subscriber becomes available and eligible to answer queued calls.

In this situation, the behavior is as follows.

- The line is offered to the MADNs first, then the (explicitly and implicitly) logged-in MLHGs.
- The server iterates through each group type (MADN or MLHG) in no guaranteed order.

Hunt on direct calls: Hunt on direct calls will not take place for subscribers that are logged into more than one MLHG. The MLHG "Hunt on direct calls" setting is irrelevant in this case.

Hunt on direct calls will take place for a subscriber who is logged into exactly one MLHG. If the MLHG has been configured to support "Hunt on Direct Calls" and a member of the MLHG is called directly while busy, the call is presented to another available member of the MLHG or queued (if queue is supported and not full).

Hunt on direct calls will happen in the situation above even if the Subscriber has DND active at CFS side. DND is treated like Busy in this case.

Also, remember that Intercom calls are not hunted.

MADN membership is not relevant to the Hunt on direct call eligibility.

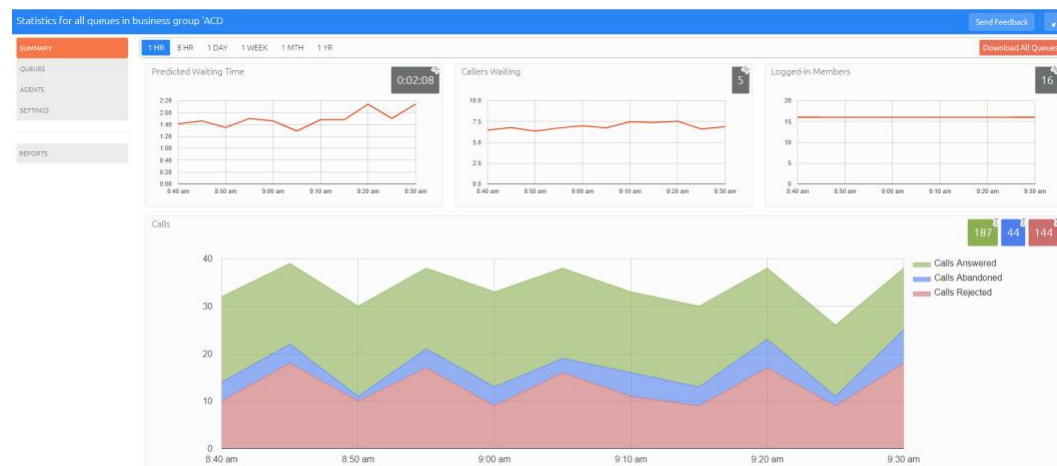
23.4 Integrated ACD

The separately licensed Integrated ACD feature allows Business Group administrators to become ACD Supervisors who can view the activity in the MLHG(s) that they administer, and download detailed statistics and reports using the Cloud Voice Portal interface. This feature also enables MLHG members to view details of the activity in the MLHG(s) that they belong to. This information is particularly useful if you are using MLHG(s) to provide call center function.

23.4.1 Statistics for Integrated ACD Administrators

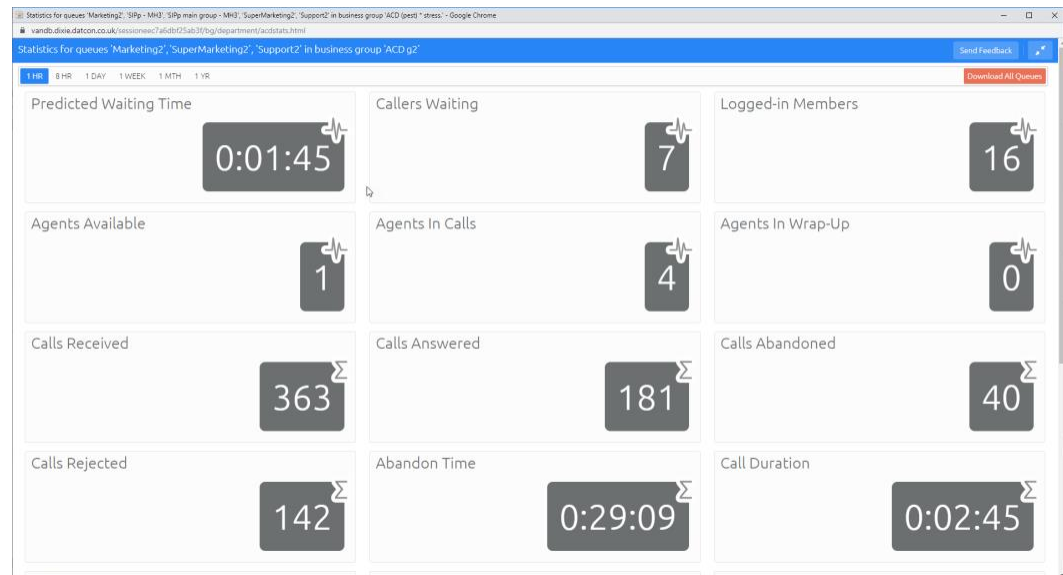
BG Administrators who are Integrated ACD Supervisors will see a *Supervisor Dashboard* button on the left-hand menu in Cloud Voice Portal that launches a new browser window displaying graphs showing calls in queue and agent activity for specified time periods and providing an interface where these supervisors can generate reports.

Figure 12: Integrated ACD Statistics Summary screen



These statistics screens can also be viewed as wallboards by pressing the *Expand* button on the top right of the screen. This removes the navigation and allows you to see at a glance how your Queues or Agents are performing. The gray boxes will either show a snapshot at the current time, representing real-time values that use a running average, or show the cumulative total for the time period selected. An icon on the box and a tooltip indicate which type of statistic is being displayed.

Figure 13: Wallboard view of Integrated ACD Queues screen

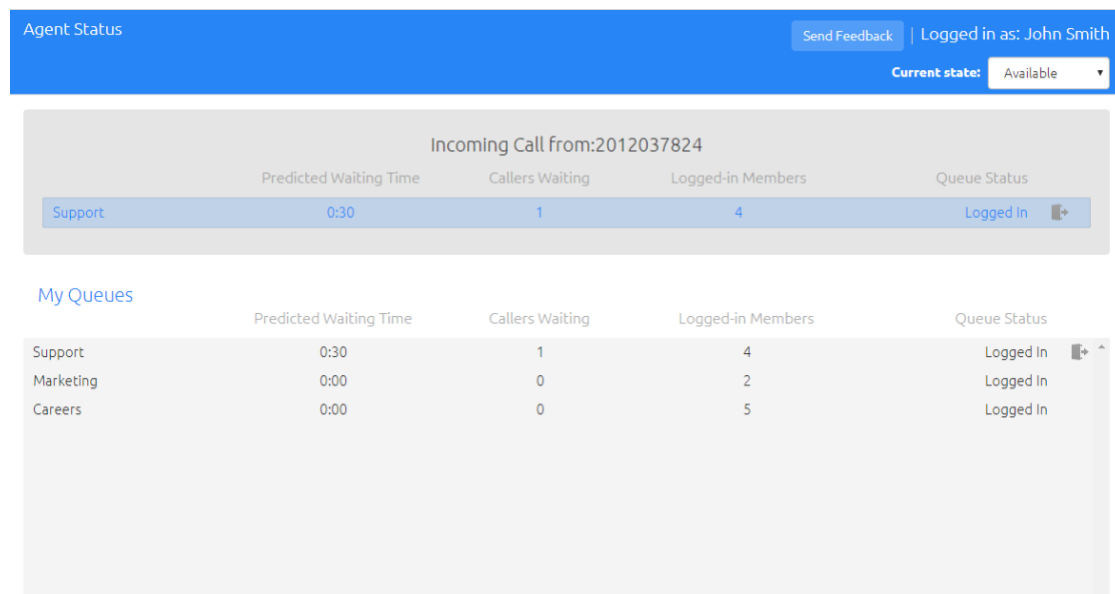


Integrated ACD supervisors can also download statistics in .csv format for further analysis, for example to see how their call center is performing over time so that they can reconfigure their agents to best effect or to use historical data to determine performance metrics. They can also generate reports based on past activity or schedule reports to be produced in the future and emailed to specified recipients.

23.4.2 Statistics for MLHG members

MLHG members (referred to as 'agents') with access to the Integrated ACD feature will see an *Agent Dashboard* button on the Cloud Voice Portal Home page. This launches the Agent Status screen in a new window.

Figure 14: Agent Status screen



On this screen, the MLHG member can see statistics on the current status of the MLHG(s) that they belong to, including

- how many members are logged in to each MLHG
- how many callers are waiting?
- the predicted waiting time for a call to be hunted to an agent
- which MLHG(s) they are logged into.

When an incoming call is received, this screen shows any caller information that can be determined including the MLHG that was originally called and how long the caller has been waiting, allowing them to respond more appropriately to the call.

23.5 Multi Line Hunt Group and MADN statistics for Service Providers

The Service Provider can access a large number of statistics about each MLHG and MADN on the Cloud Communications platform. These can then be provided to the business customer. Examples of information that is collated are as follows.

- Total calls received.
- Total calls answered.
- Total time of calls received.
- Number of calls queued.
- Total time of calls in queue.
- Number of agents logged in.

24 Busy Lamp Field features

24.1 Busy Lamp Field

The Busy Lamp Field (BLF) feature allows phones to indicate to the user at a glance which other lines in a business group are busy.

The exact support depends on the phone model, but in general the behavior is as follows.

- A phone has either a series of keys with lights, or soft-keys and a screen.
- These lights or the display on the screen depict the state for each other line in the business group being monitored.
- A solid light means the line is on a call or active.
- A fast blinking light indicates a line is being alerted.
- A slow blinking light indicates a call is held on that line.

Any phone which supports BLF can monitor the state of any line within the business group, even those provided via POTS. Note that not all phones support displaying all of the above call states.

This feature is enabled and configured on a per-line basis on the Cloud Communications platform.

Support and configuration is required on the phone for this feature.

24.2 Busy Lamp Field with Pickup

Busy Lamp Field with Pickup extends basic BLF functionality by also allowing phones monitoring other lines to pick up calls on those other lines. The key associated with the line being monitored can be pressed to pick up a call when either the monitored line is being alerted or has a call on hold.

Note that pickup of active calls (barge-in) is not supported.

Support and configuration are required on the phone for this feature.

24.3 **Busy Lamp Field with Call Transfer**

This feature allows the user to perform an Unattended Call Transfer to a line being monitored using Busy Lamp Field by pressing a single key on the phone.

An example use of this feature is that the user pre-configures a key on his phone to monitor a co-worker's line. As well as monitoring and picking up calls on this other line, the user can automatically and quickly transfer a call to the co-worker's line by pressing press this key.

This feature is enabled on a per-line basis on Cloud Communications.

This feature requires support from phones.

25 Attendant Console features

25.1 Hard Attendant Console

Hard Attendant Console functionality is provided by attaching optional extension sidecars to SIP business phones. The buttons and lamps on the extension sidecars can typically be used to support

- Direct Station Select/Speed Dials
- Busy Lamp Field
- Busy Lamp Field with Directed Call Pickup
- Shared Line Appearance
- Call Park and Retrieve

25.2 Soft Attendant Console

The solution for soft attendant console client is the Voice Operator Panel Soft Attendant Console. This client provides a powerful easy to use visual interface allowing the attendant to see which users are available/busy as well as perform advanced functions such as drag and drop transfer. Other functions of the Voice Operator Panel client include:

- Built-in soft phone
- Direct Station Select/Speed Dials
- Busy Lamp Field
- Busy Lamp Field with Directed Call Pickup
- Shared Line Appearance
- Directory integration
- Call Record
- Salesforce Integration
- Cloud Voice Portal Directory Download
- Cloud Voice Portal BG Line Download.

25.3 Line State Monitoring

The Line State Monitoring call service allows a business group line connected to an attendant console to be notified of the state of any line within the same business group. The state for a line can be one of the following.

- No call present
 - Incoming call is ringing on this line
 - Incoming call is active on this line
 - Outgoing call has been originated on this line
 - No call present and Do Not Disturb is enabled for this line.
-

26 Mobile Business features

26.1 Business Call Manager

Business Call Manager allows mobile business users to forward calls to an alternate number such as their personal mobile phone. For detailed information on Business Call Manager, please see section 4.1 of this document.

26.2 Incoming Call Manager

Incoming Call Manager offers mobile business users the ability to have specific calls to their business number forwarded to an alternate number, such as their personal mobile phone. For detailed information on Incoming Call Manager, please see section 5.1 of this document.

26.3 Easy Call Manager

Easy Call Manager provides a much-simplified view of the features of Incoming Call Manager. For detailed information on Easy Call Manager, please see section 5.2 of this document.

26.4 Call Jump

Call Jump provides mobile business users with ICM or ECM with the ability to seamlessly move active calls between phones configured in Incoming Call Manager or Easy Call Manager. The mobile business user has the ability to press ** from these devices which causes the alternate devices to ring, or they can push a button to achieve the same results if they have MaX UC Mobile installed. Upon answering the ringing device, the call is transferred to the answered device without the knowledge of the party on the other end of the call. For detailed information on Call Jump, please see section 5.3 of this document.

26.5 MaX UC Mobile

The MaX UC Mobile softphone which can be downloaded on iOS and Android devices can be twinned with a subscriber's main telephone number, allowing them to answer calls to that number on their mobile device. Subscribers can also use Instant Messaging and one of the Call Management services from Cloud Communications on MaX UC Mobile.

Additionally, the Network Voice feature can allow subscribers to use the native dialer on their Android smartphone or iPhone rather than the MaX UC Mobile client to make and receive calls while retaining access to MaX UC Client features.

For detailed information on MaX UC Mobile, please see section 4.2.

26.6 Additional identities (CFS)

Business Group subscribers in IMS deployments can be provisioned with 'additional identities', for example a business mobile phone number that they use alongside their desk phone.

- Calls to any of the numbers owned by these subscribers are forked to all of the numbers.
 - Terminating services for the Business Group line are applied to all incoming calls, regardless of whether they were made to that line or to an additional identity.
 - Outbound calls from the mobile device use the Business Group line's originating services, including the Caller ID configured on the primary line.
-

27 Remote and mobile working features

27.1 Remote workers

Remote workers can be supported within the same business group as the business customer's other phone users. These remote workers can share the same dial-plan and all the other features that the business enjoys, including call transfer, call park and busy lamp field.

Remote workers can be deployed in the following applications.

- Home workers, using their own broadband connection and a SIP ATA or phone.
 - Home workers, using a broadband connection provided by the Service Provider, combined with a SIP ATA or phone.
 - Home workers using a regular POTS line provided by the Service Provider.
 - Mobile workers using a broadband internet connection and a SIP soft-phone on their laptop.
 - Mobile workers using a mobile phone with Network Voice.
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