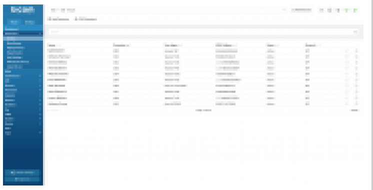
From Bicom Systems Wiki


Extensions

Extensions are associated with all UADs/Phones registered to the current slave tenant. On Multi Tenant PBXware, this menu item is only available when you are editing a tenant, since the master tenant is used for controlling the system behavior and tenants functionality.

- **Add Extension**

  Click Add Extension icon to get to the extension creation screen.

- **Search**

  In search section you can find your extension by typing its Name, E-mail, Extension number or MAC address.

- **CSV Download**

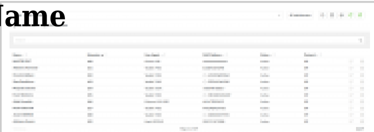
  This option allows you to download CSV file with all extensions already created on tenant. This type of file can be later used to import extensions with their details to a new PBXware system.

Contents

System

In this section all Extensions configured on the tenant are listed.

- **Name**



System

Full name of the user to which the device is registered
(E.g. Peter Doyle)
(Display)

- **Extension**

UAD/Phone extension number
(E.g. 1111)
(Display)

- **User Agent**

UAD/Phone type
(E.g. Yealink T38P)
(Display)

- **MAC Address**

MAC Address of UADs

- **Status**

UAD/Phone system status
(E.g. Active/Inactive)
(Display)

- **Protocol**

Protocol used by the UAD/Phone
(E.g. SIP/IAX)
(Display)

- Edit

Edit Extension configuration
(Button)

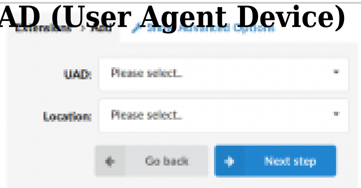
- Delete

Delete Extension from the system
(Button)

Add/Edit Extension

The procedure for adding a new system extension is divided into two steps. In the first step, the UAD/Phone type and extension location need to be provided. In a second step, basic UAD/Phone information such as the user's name and email address is provided.

- **UAD (User Agent Device)**



Add/Edit Extension

UAD is the type of device which will be connected to the system.
If the UAD/Phone is not listed here, navigate to 'Settings: UAD' Edit the desired UAD/Phone and set its 'Status' to 'Active'. Now, the UAD/Phone will be available in this list.
(E.g. Linksys SPA-941)
(Select box)

- **Location**

Select the location of the new UAD/Phone. Location refers to whether the UAD/Phone is in 'Local' or 'Remote' network.
(E.g. Local/Remote)
(Select box)

NOTE:

By default, a 'Single Extension' will be created. 'Advanced Options' offer the facility to add multiple extensions as well. For more information, check the 'Adding Multi Extensions' chapter.

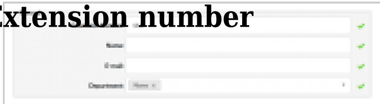
TIP:

Since this is an extension on a tenant you will see that the Username is prefixed with a tenant code, which is required for a UAD/Phone to register to the system. Nevertheless, when you register you will be able to dial other users on the tenant with only their extension number.

Once UAD and locality have been selected, we can proceed with the second step. Required fields are:

General

- **Extension number**



Add Extension - General

  By default, this field is automatically populated, but can be changed to any free Extension number.
  (E.g. Setting '1008' here will create a new system extension with the same network number).
  ([0-9])

- **Name**

  Full name of the person using the Extension. This name is sent in a Caller ID information
  (E.g. setting name 'Joanna Cox' in this field will display the name on the other UAD/Phone display when the call is made.)
  ([a-z][0-9])

- **E-mail**

  Email address associated with the extension and used for various system notifications
  (E.g. Setting 'joanna@domain.com' here will transfer all Voicemail notifications, Extension PIN and other details to this email. Also, E-mail used together with ‘User Password’ is what user needs for OSC login.)
  ([a-z][0-9])

- **Department**

  Department to which extension will belong to. This is used so the mobile and desktop apps can group extensions depending on which department they belong to.
  (E.g. Select "Sales" and when sorted in gloCOM, this extension will be shown in the Sales department group).
  (Select box)

PIN Based Devices


Add Extension - PBD

- **PBD PIN**

  PIN Used for PIN Based Dialing
  (E.g. 12345)
  ([0-9])

Call Rating

- **Call Rating**


Add Extension - Call Rating

  Turn Call Rating on or off for the current extension. If you set this option to ‘Yes’ you will also need to add a suitable Service Plan.
  (E.g. Yes, No, N/A)
  (Option buttons)

- **Service Plan**

  Service plan which will be applied to the extension. Call rates which were set in this service plan will now apply to the extension.
  (E.g. Test Service Plan)
  (Select box)

TIP: Looking from a call rates perspective, extension can be set as slave or master. Master extension have their own credit and are able to have a reminder, when balance reaches certain amount and certain limitations. Slave extensions are using credit available under Master extension which is set as Master Account Code for that slave extension.

- **Slave**

  Set whether this extension is slave or not.
  (E.g. Yes, No, N/A)
  (Option buttons)

- **Master Account Code**

  Set the master account code (extension number) of a master extension from which the current extension is using funds.
  (E.g. 1005)
  ([0-9])

- **Reminder Balance**

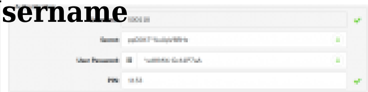
Account balance at which a reminder should be sent to the user.
(E.g. If this field is set to 10, the user will receive an email notification when the account balance reaches this amount.)
([0-9])

- **Credit Limit**

The maximum amount that the system will extend to the billing account.
(E.g. If this field is set to '10' and the account balance has dropped down to '0', your account will still have '10' units in available funds.)
([0-9])

Authentication

- **Username**


Add Extension - Authentication

Username used by the UAD/Phone for the registration with the PBXware MT
By default, this field is the Extension network number prefixed with tenant code and cannot be changed.
(E.g. If the extension number is 1008 and tenant code is 300, then the Username will be '3001008').
([0-9])

- **Secret**

Secret/Password used by the UAD/Phone for registration with the PBXware MT.
By default, this field is automatically populated but can be changed to any value.
(E.g. xKa2r4ef7X*v0!Fk)

- **User Password**

Password used for gloCOM registration with the MT PBXware.
By default, this field is automatically populated but can be changed to any value.
(E.g. Vk5F_3*dDZrmT1k7)

- **Show QR Code**

Show QR Code button will display QR code that can be scanned with gloCOM GO mobile application. This feature will make mobile app setup and registration process as fast and simple as possible. In order to setup gloCOM GO with QR Code scanning you will have to enter public address of your PBXware in QR Server field in Settings -> Servers -> Network Info section.

NOTE: Once you are registered to PBXware with your gloCOM GO you will be asked to change automatically generated User Password in order to log in. Once this procedure is completed User Password will not be visible anymore and QR Code button will be hidden.

Note: PBXware requires strong password enforcement, which means that the Password/Secret must meet certain criteria in order to be accepted, otherwise PBXware will display an error message stating that Password/Secret is too weak.

Strong Password Requirements
Password and Secret have to meet the following criteria in order to be accepted:

- It must be at least 8 characters long
- It must contain at least 1 uppercase
- It must contain at least 1 lowercase
- It must contain at least 1 digit
- It must contain at least 1 special character
- Allowed characters are: a-z, A-Z, 0-9, ! % * _

TIP: In order to make it easier for our users, we also implemented password generator, that will automatically generate strong password that meets the criteria above with a single mouse click on a key icon located on a side of Secret/User Password field.

- **PIN (Personal Identification Number)**

  Number used for account authorization.
  (E.g. If the PIN for this extension is set to '84745', PBXware MT will ask for it when checking your Voice inbox or Enhanced Services.)
  ([0-9])

TIP:

- After the extension is created, the 'Permissions' group will be editable for the administration.
- Do not paste a value to the 'Name' and 'Email' fields, but please type it in. If these values are pasted, 'Advanced options' will need to be opened and the system will prompt for missing values.
- Once the extension is created, the 'Save & Email' button becomes available. This command sends Extension details on the provided 'E-mail' address.

Auto Provisioning

- **Auto Provisioning**

Add Extension - Auto Provisioning

  Option for enabling auto provisioning service for this extension. This feature enables you to connect the UAD/Phone to PBXware MT without any hassle by providing UAD/Phone MAC address (and optionally adding the Static UAD/Phone IP address and network details).
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **MAC Address (Media Access Control)**

Provide the UAD/Phone MAC address here.
(E.g. Its a 48-bit hexadecimal number (12 characters))
([a-z][0-9])

Adding Multiple Extensions

- **Extension Template**


Adding Multiple Extensions

NOTE: Please note that this feature is part of the beta release and is only visible to beta testers. Once the stable version is released, it will be available for general use.

Select an Extension that will be used as a template
(E.g. M100)
(Checkbox)

NOTE: If a user selects the 'Extension Template' option and chooses a specific Extension, then the complete configuration of that Extension will be applied to all other Extensions created below. In case a user does not check the 'Extension Template' option, then s(he) can create multiple Extensions, but has to edit their additional configuration respectively.

Users can add multiple Extensions to PBXware in two ways:

1. Manually

- Create a list
- Manually populate the following fields: 'Name','E-mail','Ext','Secret', 'User Password', 'PIN', 'MAC', 'Department', 'Line#', 'PAI'
- Click the Add(+) icon

2. Creating and uploading '.csv' file

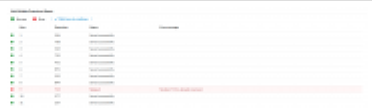
- Upload a **.csv** file
- Open a new file in the text editor
- Add the following lines (as shown in the example):

"Name"	"Email"	"Ext"	"Secret"	"User Password"	"PIN"	"MAC"	"Department"	"Line#"	"PAI"
"Joanna Cox"	"joanna@domain.com"	"121"	"jvT-6%xa0Tfpa6Em"	"KF80x2qvGKbH-3"	"1111"	"001122AABBCC"	"Sales"	"2"	"%EXT%"

- Save the file as 'multiple-extensions.csv'
- Click the 'CSV upload' button
- Select the 'multiple-extensions.csv' from the hard drive
- Click the 'Upload' button
- New Extensions will be created

NOTE: In order for this to work as intended, please note that **the following fields must be populated**: 'Name', 'E-mail', and 'Ext'.

NOTE: Always use a simple editor such as Notepad++ on Windows, or Gedit on Linux when editing **.csv** files. Please refrain from using Excel as it may remove formatting which will render the file unusable.



Add Multiple Extensions Reports

After the CSV file was uploaded, a report showing the number of uploaded/failed/skipped Extensions will be displayed. For more information, please refer to the screenshot.

NOTE: PBXware *requires* **strong password enforcement**, which means that the Secret must meet certain criteria in order to be accepted, otherwise PBXware will display an error message stating that Password/Secret is too weak. For more on how to create a successful Password/Secret in order to be accepted, please refer to the following criteria:

- It must be at least 8 characters long
- It must contain at least 1 uppercase
- It must contain at least 1 lowercase
- It must contain at least 1 digit
- It must contain at least 1 special character ()
- Allowed characters are: a-z, A-Z, 0-9, ! % * _ -

- **Download CSV Template**

  Download a CSV template for adding multiple Extensions
  (Button)

Advanced Options

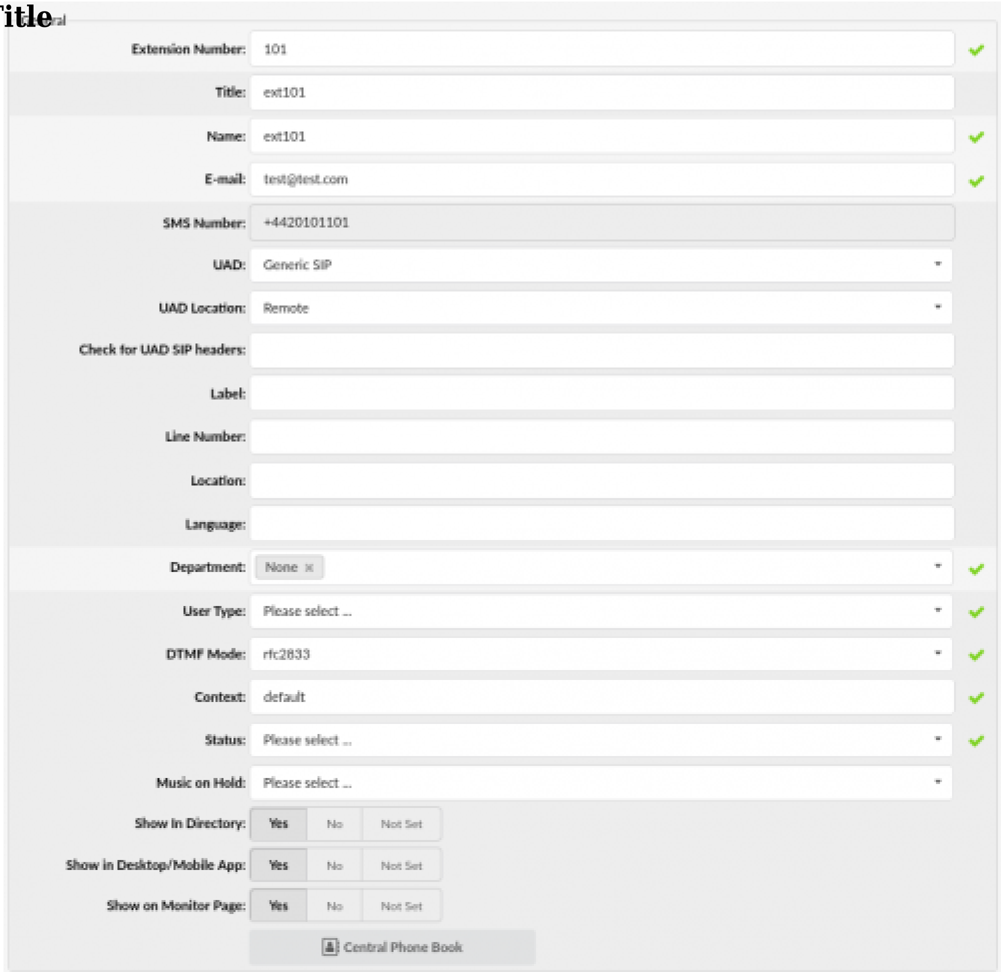
Clicking the Advanced Options button will show advanced configuration options and fields that were previously hidden.

Opening the ‘Advanced Options’ section following options are added:

General

The following options are used frequently and are mostly required for normal extension operation. Some of these fields are pre-configured with default values. It is not recommended to change these unless prompted to do so while saving the changes.

- **Title**



The user's title such as Mrs, Mr, etc.
([a-z])

- **SMS Number**

Displays the SMS number if set on the Extension
(E.g. +1234567890)
(Display)

NOTE: Please note that the SMS number cannot be edited on this page.

- **UAD**

UADs (User Agent Devices) are various IP phones, soft phones, ATA (Analog Telephone Adaptors), and IADs (Integrated Access Devices) used for system extensions. PBXware supports a wide range of UADs using SIP, IAX, MGCP, and ZAPTEL protocols. If your phone is on the supported UAD list and UAD is enabled on your PBXware, you will be able to choose the UAD that matches the phone registering to the extension.
(E.g. Generic SIP)
(Select box)

- **UAD Location**

  This option is related to the Auto Provisioning function of PBXware. Extensions located in same LAN as PBXware have to be set to **Local** while extensions connecting to PBXware from WAN should be set to **Remote**.
  (E.g. Local)
  (Select box)

- **Check for UAD SIP Headers**

NOTE: Please note that this feature is part of the beta release and is only visible to beta testers. Once the stable version is released, it will be available for general use.

  List desired SIP headers that will be forwarded from a UAD to the outbound Trunk
  (E.g. p-early-media, x-tn-mobility)
  [a-zA-Z0-9\-]

  NOTE: For example, if the UAD sends the 'x-TN-Mobility' header through the SIP INVITE and users want to preserve it on the outbound Trunk, they can add this header to the 'Check for UAD SIP Headers' field. When making a call, a check will be made to see if the UAD sends header(s) that is/are listed in this field. In case it is set, that header will be forwarded to the outbound Trunk.

  NOTE: When checking for additional headers, make sure to avoid using all default invite headers.
  Here is the list of unsupported headers which should be case insensitive: "via", "from", "to", "cseq", "call-id", "contact", "allow", "user-agent", "accept-language", "supported", "allow-events", "max-forwards", "content-type", "content-length", "p-asserted-identity", "privacy", and "remote-party-id".

  If one of the aforementioned headers is typed in as shown in the screenshot, the warning message should appear saying that the header is not allowed.

- **Label**

  The name showing on the LCD of the current device
  (E.g. John Doe)

- **Line Number**

  When auto provisioning multiple extensions on 1 phone, extensions are assigned onto lines in the order in which they were created in PBXware. The line field will allow customers to specify the extension line where this extension will be assigned on the device.
  (E.g. 2)
  ([0-9])

  NOTE: An empty line number field will be treated as the last one.

- **Location**

  Information on the geographic location of a user
  (E.g. Los Angeles)

- **Language**

  Set a language per Extension
  (E.g. en)
  ([a-z][0-9])

  NOTE: The language set on the Extension will override the language set on the Tenant.

- **Extension Timezone**

  Select timezone for extension if different from server.
  (E.g. Europe/London)
  (Select box)

- **User Type**

  Extension can be set to make calls only, receive calls only or both make and receive calls

  - Friend - make and receive calls
  - Peer - receive calls only
  - User - make calls only

  (Select box)

- **DTMF Mode (Dual Tone Multi-Frequency)**

  A specific frequency, consisting of two separate tones. Each key has a specific tone assign to it so it can be easily identified by a microprocessor.
  This is a sound heard when dialing digits on touch-tone phones. Each phone has different 'DTMF Mode'.
  (E.g. By default, this field is populated automatically for supported devices. If adding other UAD/Phone select between 'inband', 'rfc2833' or 'info' options)
  (Select box)

- **Context**

  Every system extension belongs to a certain system context. Context may be described as a collection/group of extensions. Default context used by the PBXware MT per tenant is 't-XXX' (where XXX is tenant number) and cannot be changed.
  (E.g. t-500)

- **Status**

Extension status/presence on the network.
Rather than deleting the extension and then recreating it again later on, the extension can be activated/deactivated using this field.
(E.g. Setting this field to 'Not Active' will disable all calls to this extension).
Options:

Active - Extension is active, it can make and receive calls.
Not Active - Extension is not active and it can't make nor receive calls.
Suspended - Extension is suspended and can't make calls to numbers other than those defined as Emergency Service numbers in Settings -> Servers -> Edit Server -> Locality (section) -> Emergency Services,

(Select box)

- **Music on Hold**

It allows users to choose different Music on Hold class for every extension. Select the MOH (Music On Hold) class name. All sound files belonging to this MOH class will be played to users dialing this extension.
NOTE: When you select MoH class, you need to enter 'm' in Incoming Dial Option field. Entering 'm' will provide Music on Hold to the calling party until the called channel answers.
(E.g. default)
(Select box)

- **Show In Directory**

Whether the extension should be shown in Remote Directory accessed through the desk-phones interface.
(E.g. Yes, No, Not Set)
(Options button)

- **Show in Desktop/Mobile App**

Enable/disable non-gloCOM extensions from displaying in gloCOM contacts.
(E.g. Yes, No, Not Set)
(Options button)

NOTE Device must support this feature.

- **Show on Monitor Page**

This excludes an extension from showing on the Monitor page. Useful for virtual extensions that will never be online in order to get a more accurate count of phones online.
(E.g. Yes, No, Not Set)
(Options button)

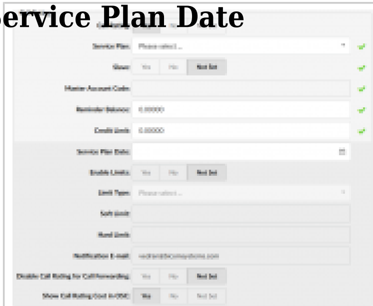
- **Central Phone Book**

  When you click on this button new pop-up (where you can manage your Central Phone Book contacts) will be shown.
  More about managing Phone Contacts you can find at Central Phone Book page
  (Button)

Call Rating

These options are used for call rating of incoming and outgoing calls. The extension is assigned to a service plan and its call rates and additional call rating options are set here as well.

- **Service Plan Date**


Advanced Options - Call Rating

  This option is only available if you set extension to be master extension. (Call rating -> Slave-> No/Not Set. Also, Service Plan must be set in order for this to work. This date defines when will Service Plan reset.
  (E.g. In service plan you have set inclusive minutes to 5. If this field is set to 03-12-2019, inclusive minutes will be reset on 3rd day of each month. So, if all 5 inclusive minutes were used by this day, inclusive minutes will be reset back 5 from this date)
  ([dd-mm-YYYY])

- **Enable Limits**

  If the extension is master extension it is possible to set this option to yes. Setting this field to yes enables setting limits on the current extension.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Limit Type**

  This option lets us decide whether limits we set will be applied on a Daily or Monthly basis.
  (E.g. Daily)
  (Select box)

- **Soft Limit**

  Depending on Limit Type, when the extension reaches Soft Limit, it will email the person in charge of a call rating.

(E.g. If you set this field to 10, an email will be sent to you when the user reaches that amount [10] while calling)
([0-9])

- **Hard Limit**

Depending on the Limit Type, when an extension reaches Hard Limit, the system will block this extension from making any further calls.
(E.g. If you set this field to 20 and the user reaches that amount while calling, system will block this extension from making further calls)
([0-9])

- **Notification E-mail**

Email which will be used when the user reaches Soft Limit.
(E.g. admin@domain.com)
([a-z][0-9])

- **Disable Call Rating for Call Forwarding:**

You can set whether you want call rating to be applied for any forwarded calls.
(E.g. Yes/No/Not Set)
(Option buttons)

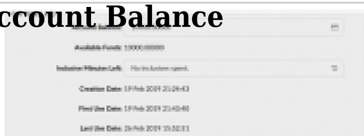
- **Show Call Rating Cost in OSC:**

Users can choose to have call rating displayed for their extension in the Online Self Care window.
(E.g. Yes/No/Not Set)
(Option buttons)

Call Rating Info

This section displays the extension's Call Rating information such as: Account Balance, Available Funds, Inclusive Minutes Left.

- **Account Balance**

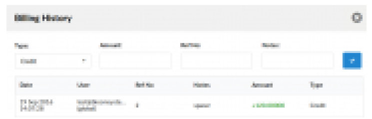


Advanced Options - Call Rating Info

Displays the amount of credit units left (available funds minus sum that is already spent by the user).
(E.g If the user has 100 units of credit, 100 units + the credit limit can be spent in total. If this field displays a negative value (e.g. -4.00000) that means that the account balance has reached 0 and the credit limit is being used).
(Display)

- **Credit/Debit**

  Opens a window with Call Rating History, where you have option for adding extension credit/debit. Following parameters available:
  (Button)


Advanced Options - Credit/Debit window

  - **Type**

    Call Rating type, select whether call rating is Credit or Debit.
    (E.g. Credit)
    (Select box)

  - **Amount**

    Call Rating amount, if the call rating type is in Euros, and you add 100 here, 100 Euros will be added to the extension amount.
    (E.g. 50)
    ([0-9])

  - **Ref No**

    Call Rating reference number, depending on how your company bills clients, the invoice number can be assigned here, for example.
    (E.g. Invoice01)
    ([a-z][0-9])

  - **Notes**

    Additional Call Rating notes.
    (E.g. 50 Euros credit added)
    ([a-z][0-9])

  - **Confirm**

This will finalize Call Rating action, fill in all previous fields and click this button to add funds.
(Button)

Once funds are added, the following details will be displayed:

- **Date:** Time and date of the payment
- **User:** The username used for login to the system of the user who added the funds
- **Ref No:** Call Rating reference number
- **Notes:** Additional Call Rating notes
- **Amount:** Amount of funds added
- **Type:** Call Rating type

- **Available Funds**

  Displays available account funds (account balance + credit limit).
  (E.g. If the user account balance has 100 units + 10 credit limit units, 110 units will be displayed here).
  (Display)

- **Inclusive Minutes Left**

  Displays the inclusive minutes left. As long as there is any inclusive time left, call rating is not calculated for outgoing calls.
  (E.g. You'll see the inclusive minutes left in the following form '0d 0h 4m 25s').
  (Display)

- **Reset Inclusive**

  Reset extension inclusive minutes, click on this button and confirm with 'Yes' to reset inclusive minutes.
  (Button)

NOTE: Buttons Reset Inclusive and Credit/Debit will not be displayed unless Call Rating is enabled.

- **Creation Date**

  Extension creation date.
  NOTE: If your system was updated to a newer version, all extensions created on old version will display ‘unknown’ in this field.
  (E.g. 14-06-2049 12:30:36)
  (Display)

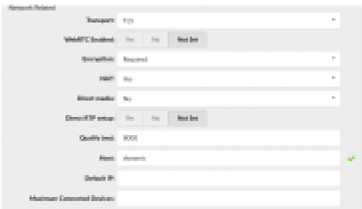
- **First Use Date**

  Date/Time of the first extension use.
  (E.g. 11 Jun 2019 18:58:25)
  (Display)

- **Last Use Date**

  Date/Time of the last extension use.
  (E.g. 11 Jun 2019 19:25:12)
  (Display)

Network Related



Advanced Options - Network Related

These options set important network related values regarding NAT, monitoring and security.

- **Transport**

  Type of transfer protocol that will be used on PBXware.
  Available options:

  UDP (User Datagram Protocol) - is used primarily for establishing low-latency and loss-tolerating connections between applications on the internet. UDP enables process-to-process communication. With UDP, computer applications can send messages, in this case referred to as datagrams, and it is considered as best-effort mode of communications. UDP is considered a connectionless protocol because it doesn't require a virtual circuit to be established before any data transfer occurs
  TCP (Transmission Control Protocol) - provides reliable, ordered, error-checked delivery of a stream of octets between programs running on computers connected to an intranet or the public Internet. TCP sends individual packets and is considered a reliable transport medium.
  TLS (Transport Layer Security) - cryptographic protocol that provide communication security over the Internet.[1] They use asymmetric cryptography for authentication of key exchange, symmetric encryption for confidentiality, and message authentication codes for message integrity.

  (E.g. TLS)
  (Select box)

- **WebRTC Enabled**

Whether WebRTC is enabled for this Extension or not
(E.g.Yes, No, Not Set)
(Option buttons)

Navigate to **Settings > Protocols > SIP** and make sure **TLS** is set to **"Yes"**.
Under **Extensions > Extension > Edit screen**, make sure option **WebRTC Enabled** is set to **Yes**.
For Allowed Codecs make sure "Opus" is enabled and set.

NOTE: In order for WebRTC to function properly, following things must be done:

- Go to **Setup Wizard > SSL Certification** page and make sure to upload a valid certificate or to use automatic Let's Encrypt service.
- Navigate to **Settings > Protocols > SIP** and make sure **TLS** is set to **"Yes"**.
- Set **WebRTC Enabled** option to **Yes**.
- For Allowed Codecs make sure "Opus" is enabled and set.

- **Encryption**

This option enables or disables encryption in PBXware transport.
Offered options are:

- **Offer if possible (TLS only)** - Offers encryption only if it is possible and only with TLS protocol.
- **Required** -Encryption always required.
- **Offer (TLS only)** - Always offers encryption but only with TLS.

(Select box)

- **NAT (Network Address Translation)**

Set the appropriate Extension - PBXware NAT relation.
If extension 1000 is trying to register with the PBXware from a remote location/network and that network is behind NAT, select the appropriate NAT settings here.
Available options:

- **Default (rport)** - this setting forces RFC3581 behavior and disables symmetric RTP support.
- **Yes** - Always ignore info and assume NAT
- **No** - Use NAT mode only according to RFC3581
- **Comedia RTP** - enables RFC3581 behavior if the remote side requests it and enables symmetric RTP support.

(Option buttons)

- **Direct Media**

  Should you allow RTP voice traffic to bypass Asterisk.
  Available options:

  - **No** - this option tells the Asterisk to never issue a reinvite to the client
  - **Yes** - send reinvite to the client
  - **No NAT only** - allow reinvite when local, deny reinvite when NAT
  - **Use UPDATE** - use UPDATE instead of INVITE
  - **No NAT, Update** - use UPDATE when local, deny when NAT

  (Option buttons)

- **Direct RTP setup**

  Here you can enable or disable direct RTP setup. Setting this value to yes sets up the call directly with media peer-2-peer without re-invites. Will not work for video and cases where the callee sends RTP payloads and fmtp headers in the 200 OK that does not match the callers INVITE. This will also fail if directmedia is enabled when the device is actually behind NAT.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Qualify (ms)**

  Timing interval in milliseconds at which a 'ping' is sent to the UAD/Phone or trunk, in order to find out its status(online/offline). Set this option to '2500' to send a ping signal every 2.5 seconds to the UAD/Phone or trunk. Navigate to 'Monitor: Extensions' or 'Monitor: Trunks' and check the 'Status' field.
  In PBXware 5.1 'Qualify' is set to 8000 by default.
  (E.g. 6000)
  ([0-9])

- **Host**

  Set the way the UAD/Phone registers to PBXware. Set this field to 'dynamic' to register the UAD/Phone from any IP address. Alternately, the IP address or hostname can be provided as well.
  (E.g. dynamic)
  ([dynamic][a-z][0-9])

- **Default IP**

  Default UAD/Phone IP address. Even when the 'Host' is set to 'dynamic', this field may be set. This IP address will be used when dynamic registration could not be performed or

when it times out.
NOTE: UAD/Phone must be on static IP address.
(E.g. 192.168.1.1)
([0-9])

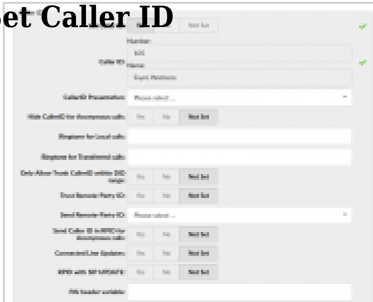
- **Max Connected Devices**

  Maximum number of connected devices per extension.
  (E.g. 10)
  ([0-9])

Caller ID

The caller's name and number displayed here are sent to the party you call and are shown on their UAD/Phone display. The information you see here is taken from the extension number and user name. To set different Caller ID information, please go to 'Enhanced services: Caller ID' and set new information there.

- **Set Caller ID**


Advanced Options - Caller ID

  Enable 'Caller ID' service. Set this option to 'Yes' to enable the Caller ID service.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Caller ID**

  Extension Number and Name that are displayed on dialed party UAD/Phone display. These options are read-only. Caller ID information can be changed only through 'Enhanced Services'
  (Read-only)

- **Caller ID Presentation**

  The way Caller ID is sent by the Extension
  If PBXware MT is connected to a third-party software and there are problems with passing the Caller ID information to it, applying different 'Caller ID Presentation' methods should sort out the problem
  (E.g. Allowed, Not Screened)
  (Select box)

- **Hide CallerID for Anonymous calls**

  Setting this option to ‘Yes’ formats all incoming calls who have Caller ID set but anonymous number to both anonymous (Anonymous<anonymous>).
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Ringtone for Local calls**

  This option enables setting up the custom ringtone for local calls. It is necessary to know which phone is registered on this extension.
  (E.g. If your phone is SPA941 you could set **<Simple-2>**)
  ([a-z][0-9])

- **Ringtone for Transferred calls**

  Ringtone for transfered calls, work same as Ringtone for local calls setting. Depending of your phone manufacturer you can send a string to the phone in order to use different ringtone than the one set on device. Once this string is set in Ringtone for transfered calls field (as well as in your device itself) it will be used for all calls that are transfered to your extension.

- **Only Allow Trunk CallerID within DID range**

  When you assign the extension to a customer and assign some DIDs to it, the customer can make calls through that extension with Trunk CallerIDs that match its DID numbers. If a customer tries to make a call with a Trunk CallerID that doesn't match any of the DIDs assigned to him, the Trunk CallerID will be reset to Anonymous.
  (E.g. Yes, No, Not Set)
  (Option buttons)

NOTE: This option is also available when creating a tenant. When setting this on tenant, it applies to all its extensions. If we want a different treatment for a particular extension then we set this option here while creating or editing the desired extension.

- **Trust Remote-Party-ID**

  Defines whether PBXware will allow Remote-Party-ID header
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Send Remote-Party-ID**

  Should 'Remote-Party-ID' be added to uri.
  Options available:

  - **Use Remote-Party-Id** - Use the "Remote-Party-ID" header to send the identity of the remote party

        - **Use P-Asserted-Identity** - Use the "P-Asserted-Identity" header to send the identity of the remote party

    (Select Box)

- **Send Caller ID in RPID for Anonymous calls**

    Whether Caller ID will be sent in RPID header for Anonymous call
    (E.g. Yes, No, Not Set)
    (Option buttons)

- **Connected Line Updates**

    This option is particularly useful as for some providers, if **Use PAI** is enabled, calls might start dropping short time after update is sent. Setting Connected Line Updates to **No** will prevent these call drops.
    (E.g. Yes, No, Not Set)
    (Option buttons)

- **RPID with SIP UPDATE**

    In certain cases the only method which will immediately transmit connected line change is with a SIP UPDATE request. If communicating with another Asterisk server and wish to be able to transmit such UPDATE messages to it, then you must enable this option. Otherwise, we will have to wait until we can send a reinvite to transmit the information.
    (E.g. Yes, No, Not Set)
    (Option buttons)

- **PAI header variable**

    Set the value for the PAI header.
    Available placeholders are:

    - **%CALLERIDNUM%**
    - **%TENANT%**
    - **%EXT%**
    - **%TENANTEXT%**
    - **%TENANTNAME%**
    - **%CALLERIDNAME%**
    - **%PAIEXT%**

    ([a-z][0-9])

Call Properties

These options fine-tune incoming/outgoing call settings.

Advanced Options - Call Properties

- **Area Code**

  Area code that the system is located in or is operating from.
  (E.g. If PBXware is located in New York, set the New York area code here (e.g. 212). This will override the default area code which is set on Tenant.)
  ([0-9])

  NOTE: The maximum area code value is **50 digits**. Users can set it per Extension in the 'Area Code' field.

- **Ringtime (sec)**

  UAD/Phone ring time in seconds.
  (E.g. Amount of time which tells us how long the UAD/Phone will ring before the call is considered unanswered (by default it is 32 seconds))
  ([0-9])

- **Incoming Dial Options**

  Advanced dial options for all incoming calls.
  Note: Please see below for a detailed list of all available dial options (default: tr).
  ([a-z])

- **Outgoing Dial Options**

  Advanced dial options for all outgoing calls.
  Note: Please see below for a detailed list of all available dial options (default: empty).
  ([a-z])

Dial Options:

- **t** - Allow the called user to transfer the call by hitting #
- **T** - Allow the calling user to transfer the call by hitting #

  NOTE: The caller can use '#' to transfer local calls but only if both Extensions or just the callee's Extension has 'T' set as the 'Incoming dial' option. Please note that the calls that originate from Trunk **do not execute** a transfer on '#'.

- **r** - Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use this option with care and don't insert it by default into all of your dial statements as you are killing call progress information for the user. Asterisk will generate ring tones automatically where it is appropriate to do so. 'r' makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.

- **R** - Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
- **m** - Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
- **o** - Restore the Asterisk v1.0 Caller ID behavior (send the original caller's ID) in Asterisk v1.2 (default: send this extension's number)
- **j** - Asterisk 1.2 and later: Jump to priority n+101 if all of the requested channels were busy (just like behaviour in Asterisk 1.0.x)
- **M (x)** - Executes the macro (x) upon connect of the call (i.e. when the called party answers)
- **h** - Allow the called party to hang up by dialing *
- **H** - Allow the caller to hang up by dialing *
- **C** - Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
- **P (x)** - Use the Privacy Manager, using x as the database (x is optional)
- **g** - When the called party hangs up, exit to execute more commands in the current context.
- **G (context^exten^pri)** - If the call is answered, transfer both parties to the specified priority; however it seems the calling party is transferred to priority x, and the called party to priority x+1
- **A (x)** - Play an announcement (x.gsm) to the called party.
- **S (n)** - Hang up the call n seconds AFTER the called party picks up.
- **d:** - This flag trumps the 'H' flag and intercepts any dtmf while waiting for the call to be answered and returns that value on the spot. This allows you to dial a 1-digit exit extension while waiting for the call to be answered - see also RetryDial
- **D (digits)** - After the called party answers, send digits as a DTMF stream, then connect the call to the originating channel.
- **L (x[:y][:z])** - Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
  - **+ LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
  - **+ LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the called party.
  - **+ LIMIT_TIMEOUT_FILE** - File to play when time is up.
  - **+ LIMIT_CONNECT_FILE** - File to play when the call begins.
  - **+ LIMIT_WARNING_FILE** - File to play as a warning if 'y' is defined. If LIMIT_WARNING_FILE is not defined, then the default behavior is to announce ('You have [XX minutes] YY seconds').
- **f** - forces CallerID to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow Caller IDs from other extensions than the ones that are assigned to you.
- **w** - Allow the called user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)
- **W** - Allow the calling user to start recording after pressing *1 or what defined in features.conf, requires Set(DYNAMIC_FEATURES=automon)

NOTE: Dial options can bind together. (E.g. t + r = tr)

Groups

> These options define who is allowed to pickup our calls, and whose calls we are allowed to pickup.

Advanced Options - Groups

- **Call Group**

  Set the Call Group that the extension belongs to. It is only allowed to have 1 Call Group.
  (E.g. 3)
  (Select box)

- **Pickup Group**

  Set which groups the extension is allowed to pickup by dialing '*8'. It is allowed to have as much Pickup Groups as needed.
  (E.g. 4 2)
  (Select box)

TIP: Grouping works only within a technology (SIP to SIP or IAX to IAX)

Example:

Extension A:

- Call Group = 1
- Pickup Group = 3,4

Extension B:

- Call Group = 2
- Pickup Group = 1
- If A is ringing, B can pickup the ringing call by dialing '*8'.
- If B is ringing, A cannot pickup the ringing call because B's Call Group = 2, and A can pickup only Call Groups 3 and 4.

NOTE: To be able to select Call Group and Pickup Group they have to be assigned to the tenant in Settings -> Tenants -> edit tenant -> Numbering Defaults (section) -> Call groups/Pickup groups.

Default Trunks

These options enable extensions to use custom default trunks for all outgoing calls.

- **Primary/Secondary/Tertiary Trunk:**

Advanced Options - Default Trunks

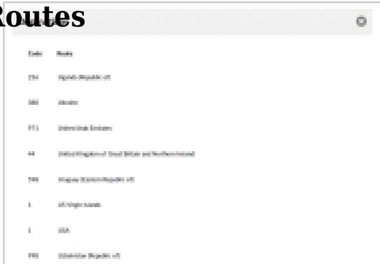
  Set the default trunks for all routes dialed from this extension.

If the connection is not established through the primary, the secondary trunk is used, etc. Default trunks can be set per extension and on the Settings->Default Trunk, on a Slave tenant. Please look at the 'Precedence' section below.
(Select box)

- **Override System LCR**

  This option tells the system that when making calls, it should omit checking System Level LCR.
  (Option buttons)

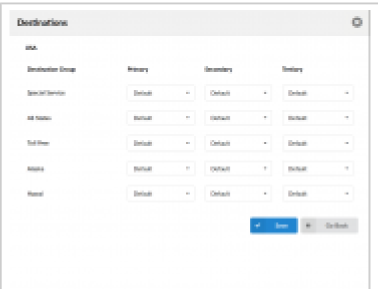
- **Routes**


Routes table

  Set the default extension trunks 'per Destination/Country provider' (Routes: Example). For some countries/providers to be reached through non-primary trunks, or when one of the default trunks needs to be given higher precedence over another, a route may be set for each destination provider

- **NOTE:** These routes are not available if the system is set to use Simple Routing mode.

  (Select box)


Routes table

Precedence

Settings

- Default Trunks: All System calls go through the trunks defined here.
- MiniLCR: Overrides 'Default Trunks' and sets a specific trunk for a destination.

Extensions

- Trunks: Overrides 'Settings: Default Trunks'
- Routes: Overrides 'Settings: MiniLCR'

Routes: Example The list of countries that start with the letter 'A' is displayed when you click on the 'A' in the upper navigation. After the countries are listed, click on one of them to see default trunks for their providers. Once the default trunk is selected for a provider, all calls made from that extension to the set provider will be made using the set trunk.

Emergency Trunks

Advanced Options - Emergency Trunks

NOTE: This option will be shown only if enabled on MT/tenant > General Settings > Emergency Trunks per Extension.

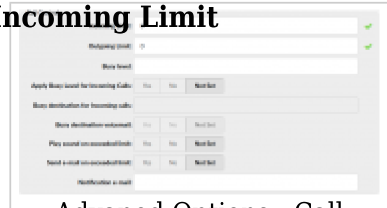
- **Primary/Secondary/Tertiary Trunk:**

  These trunks will override the trunks set on Trunks and Tenants. The trunks available for selection are only those that are allowed to be set on that tenant (defined in Trunks & Tenants), but only if the value is not "Default".

Call Control

These options set the number of simultaneous incoming and outgoing extension calls.

- **Incoming Limit**



Advaned Options - Call Control

  Sets the maximum number of simultaneous incoming calls. If an extension receives more incoming calls than set here, they are all redirected to the extension voice-box
  (E.g. 2)
  ([0-9])

- **Outgoing Limit**

  Sets the maximum number of simultaneous outgoing calls. The outgoing call can be placed on hold and another call can be made from the same extension. However, this feature has to be supported by the UAD/Phone
  (E.g. 2)
  ([0-9])

- **Busy level**

Maximum number of concurent calls until the user/peer is considered busy. This option is not intended for blocking calls, but for displaying user/peer status properly, for example in BLF.

- **Apply Busy Level for Incoming Calls**

  If Apply Busy Level for Incoming Calls is set, and one receives an incoming call while extension is on a call, the incoming call is blocked and redirected to Voicemail or any other option set for extension.(For this to work one needs to set the **Busy Level** option to 1 under group Call Control).

- **Busy destination for Incoming calls**

  Users can redirect calls to a custom extension when the busy level is set and they have the option to set the busy destination voicemail to yes/no as well.

- **Busy destination voicemail**

  Whether destination for busy level is voicemail
  (E.g. Yes, No, Not Set)
  (Option button)

- **Play sound on exceeded limit**

  If you try to make more calls than allowed in the Outgoing Limit, a message will be played that the limit has been exceeded.
  (E.g. Yes, No, Not Set)
  (Option button)

- **Send e-mail on exceeded limit**

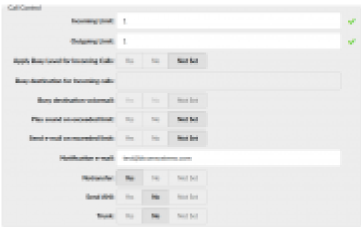
  Whether or not to send a notification mail when the limit is exceeded.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Notification e-mail**

  E-mail address to which the notification mail should be sent if the number of calls exceed the limit.
  (E.g. user@domain.com)
  ([a-z][0-9] @)

IAX Extensions only

- **Notransfer**



Advanced Options - IAX Call Control

Prohibit Asterisk from stepping out of the media path and connecting the two endpoints directly to each other. This, of course, affects your CDR and call rating information
(E.g. Yes, No, Not Set)
(Option buttons)

- **Send ANI**

  Whether to send ANI along with CallerID
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Trunk**

  Whether to use IAX trunking. IAX Trunking needs support of a hardware timer
  (E.g. Yes, No, Not Set)
  (Option buttons)

Authentication

These options are used for UAD/Phone authentication with PBXware MT

- **Authname**

Advanced Options - Authentication

  Name used for authentication with the sip provider
  (E.g. If you set this field to 12345, for example, the sent SIP header will look like 12345@sipprovider.com)
  ([0-9])

- **Auth**

  Auth is the optional authorization user for the SIP server
  (E.g. 44000)
  ([a-z][0-9])

Fields from Trunks available on Extension

- **Incoming IP addresses (new line separated)**

  Ip used by the phone for registration with PBXware
  (E.g.. 192.168.x.x)

- **Insecure**

  security level for incoming connection
  (E.g. port)
  (Select box)

TIP: If Incoming IP addresses are empty Insecure option will not apply.

Note: To enable this option go to Settings > Tenants > "Tenant name" **Allow IP Address Authentication for Extensions** set this to "YES"

IAX Extensions only

- **Encryption**

Advanced Options - IAX Authentication

  Whether to enable encryption of IAX data stream. For this to work, you must choose 'md5' auth method.
  (E.g. AES-128)
  (Select box)

- **Auth Method**

  Authorization method used for IAX extensions, can be set to:

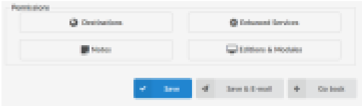
  - none
  - plaintext
  - md5
  - rc4
  - rsa

  (Select box)

- **RSA Key**

  RSA key used for authorization, if preferred auth method is 'rsa' then RSA key needs to supplied to this field.
  ([0-9][a-z][A-Z])

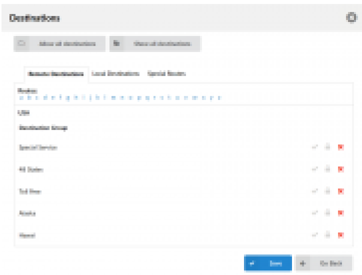
Permissions


Permissions settings

Destinations

If a Service Plan is previously created, destination permission template was set. Every extension using that service plan will have the same destination permissions. Here we are able to change (or add if we don’t have Service Plan) destination permissions only for one particular extension.

By pressing “**Allow all destinations**” every single country and every single destination group will be allowed.


Destinations settings

Manually, destinations can be set through the following groups:

- **Remote** - E164 PSTN destinations, ITSPs, other VoIP networks etc.
- **Local** - All destinations within the system/network (Extensions, IVR, Queues, Conferences…).
- **Special Routes** - Other PBX networks we are connected to.

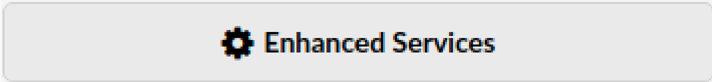
Allowed

PIN Required

Not allowed

Enhanced Services

When you click on Enhanced Services new window with various features will be opened. Before setting up and editing any one of these, first it is needed to enable it and save changes.



For detailed information on Enhanced Services click the link below:

Enhanced Services

Notes


Advanced Options - Notes

When you click on the 'Notes' button you will you can set notes in regards to this extension.

- **Date/User**

  When a new note is added, this field is automatically filled with the current date and email of the user who create a note.

- **Note**

  In this field, enter your desired note.

Editions & Modules


Advanced Options - Editions & Modules

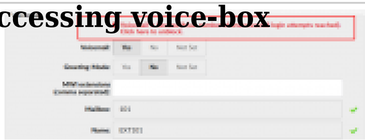
Here you can set which gloCOM editions the extension can use. The ALL option is set by default, so the extension can use all editions. You can uncheck the ALL option and choose which editions you want to enable per extension. (E.g. If you select Office and Mobile, this extension will be able to use gloCOM Office edition and gloCOM GO)

Voicemail

These options mimic the functions of an answering machine but with many additional features added. Voice messages are saved on central file-system location instead on a UAD/Phone.

NOTE: PBXware 6.0 has introduced a new lockout feature for Voicemails. It is directly related to the 'Max login attempts' option meaning that if a user fails to enter their voicemail PIN correctly after e.g., 3 tries, that voicemail will be blocked until an administrator unblocks it.

- **Accessing voice-box**



Advanced Options - Voicemail Blocked

To access a voice-box, dial '*123', enter the extension PIN (if “Skip PIN Prompt” is not set to “Yes”), and follow the instructions.

- **Leaving a voice message**

When the user is transferred to voice-box, 'Please leave your message after the tone. When done, hangup or press the # key' message will be heard. Two options are available:

1. Leave a voice message (ended by pressing '#' key or by hanging up), or
2. Reach an operator by dialing '0'
3. If '0' is dialed, the 'Press 1 to accept this recording, otherwise please continue to hold' message will be heard. Two options are available:
4. Press '1' to save your message, after which the operator will be dialed. The 'Please hold while I try that extension' message will be heard, or
5. Continue to hold, which will delete any left messages, after which the operator will be dialed. 'Message deleted, please hold while I try that extension' message will be heard.

- **File - system usage:**

With continuous tone for 60 seconds:

- wav49 = 91.0kb
- wav = 863.0kb
- gsm = 91.0kb

- **With continuous silent tone for 60 seconds:**

- wav49 = 0.38kb
- wav = 3.0kb
- gsm = 0.32kb

- **Voicemail**

Advanced Options - Voicemail

Enabling the Voicemail service. When no one picks up the call, after some time the calling party will be transferred to the voice box of the dialed extension. The caller will then have an option to leave a voice message.
(E.g. Yes, No, Not Set)
(Option buttons)

- **Greeting-Mode**

If your Voicemail is turned on, you can set this option to yes to play a greeting and then a busy sound
(E.g. Yes, No, Not Set)
(Option buttons)

- **MWI extensions (comma separated)**

If you want your extension to be notified for multiple Voicemails, you can enter them in this field.
(E.g. 2002, 2003, 2004, 2005)
NOTE: Please bear in mind that if you make any changes to the MWI option in the GUI, you will have to re-register your device for those changes to be applied.

- **Mailbox**

Mailbox extension number. This value is the same as the extension number and cannot be modified.
(E.g. 1008)
(Read-only)

- **Name**

Full name of the user associated with the voice box. This value is the same as the 'Name' field from ‘General’ section and cannot be modified
(E.g. John Doe)
([a-z][0-9])

- **PIN: (Personal Identification Number)**

  Password used for accessing voicemail. The value of this field is set under 'Authentication: PIN'. When you want to access your voicemail you will be asked to authenticate with 4 digit PIN
  (E.g. 1234)
  ([0-9])

- **E-mail**

  E-mail address associated with the voice box. The value of this field is set under ‘E-mail’ field from ‘General’ section.
  (E.g. When A calls B and leaves a voice message, B will get an email notification about new voice message on this email address).
  ([a-z] [0-9] [@._-])

- **Send E-mail**

  Whether or not to send an E-mail to the address given above
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Carbon Copy E-mails**

  Add additional E-mail addresses to which you want voice inbox to be associated with.
  (Select box)

- **Pager e-mail**

  Pager e-mail address associated with the voice box.
  (E.g. When A calls B and leaves a voice message, B will get a pager email notification about a new voice message on this email address).
  ([a-z] [0-9] [@._-])

- **Greeting message**

  Greeting message played to users upon entering the voice box.
  (E.g. When A gets to B's voice box, the selected 'Greeting message' is played to A before he is allowed to leave a message).
  (Select box)

- **Unavailable/Busy message**

  Upload a sound file which will be played when extension is unavailable/busy.
  (Button)

- **Reset Busy message**

If greeting message is set to busy you have option to reset busy message.
(E.g. Yes, No, Not Set)
(Option buttons)

- **Skip Instructions**

  Skip the instructions on how to leave a voice message.
  (E.g. Once user A reaches the dialed voice box, if this option is set to 'Yes', A will hear the 'Greeting message', and then be transferred directly to the 'beep' sound).
  (Option buttons)

- **Skip PIN Prompt**

  Enter your voicemail options faster. Setting this option to Yes will skip the PIN entry when dialing *123 (*124 should work as before with the PIN.
  (Option buttons)

- **Attach**

  Send the voice message as an attachment to the user's email.
  (E.g. Once B gets the new voice message, if this option is set to 'Yes', the message sound file will be attached to the new voicemail notification email).
  (Option buttons)

- **Delete After E-mailing:**

  Delete the voice message after sending it as an attachment to the user's email.
  (E.g. Once B gets the new voice message, if this option is set to 'Yes', the message will be deleted from the voice box after it has been emailed to B.)
  (Option buttons)

- **Say Caller ID**

  Announce the extension number from which the voice message has been recorded.
  (E.g. If this option is set to 'Yes', when checking voicemail, the 'From phone number {$NUMBER}' message will be heard).
  (Option buttons)

- **Allow Review mode**

  Allow B to review the voice message before committing it permanently to A's voice box.
  E.g. B leaves a message on A's voice box, but instead of hanging up, he presses '#'. Three options are offered to B:

  - Press 1 to accept this recording
  - Press 2 to listen to it
  - Press 3 to re-record your message

  (Option buttons)

- **Allow Operator**

  Allow user to reach an operator from within the voice box.
  E.g. B leaves a message on A's voice box, but instead of hanging up, B presses '#'.
  'Press 0 to reach an operator' message played (Once '0' is pressed, the user is offered the following options):

  - Press 1 to accept this recording (If selected, 'Your message has been saved. Please hold while I try that extension' is played and operator is dialed)
  - Or continue to hold (If B holds for a moment, 'Message deleted. Please hold while I try that extension' is played and operator is dialed)

  (Option buttons)

- **Operator Extension**

  Local extension number that acts as an operator.
  (E.g. If A's voice box has an option 'Allow Operator' set to 'Yes', all users dialing '#0' inside the voice box will reach this operator extension).
  ([0-9])

- **Play Envelope message**

  Announces the Date/Time and the Extension number from which the message was recorded.
  (E.g. Once the voice box is checked for new messages, if this option is set to 'Yes', 'Received at {$DATE} from phone number {$NUMBER}' will be played, giving more details about the message originator).
  (Option buttons)

- **Hide from directory**

  This option will allow you to hide your extension from the Directory/BLF list.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Rings to answer**

  Number of rings before Voicemail answers the call
  (E.g. 5)
  ([0-9])

- **Voicemail Delay (sec)**

  How long to pause in seconds before asking the user for PIN/Password.
  (E.g. Some UADs/Phones have a tendency to garble the beginning of sound files. Therefore, the user checking the voice box, when asked for a password, would hear '...sword' instead of 'Password'. Setting this field to 1-2 seconds will provide a long enough gap to fix this anomaly).

([0-9])

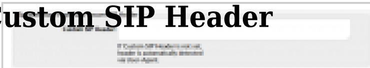
- **Timezone:**

  Sets the correct date/time stamp.
  NOTE: Time zones are taken from '/usr/share/zoneinfo' system directory
  (E.g. By setting the correct time zone, the user would always be notified of the exact date/time voice message was left on their box. Set the correct time zone if the user is located in a different time zone than PBXware MT).
  (Select box)

Speakerphone Page Auto-Answer SIP Header

These options allow the caller to use a UAD in a public announcement system. If the UAD fully supports this service, the call is accepted automatically and put on a loudspeaker.

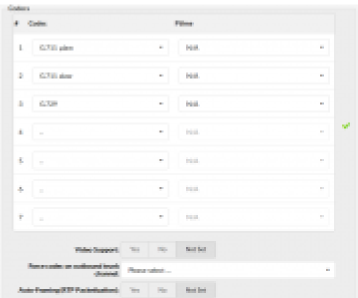
- **Custom SIP Header**


Advanced Options - Speakerphone Page Auto-Answer SIP header

  Set a custom UAD/Phone header for this extension.
  (E.g. If one of the predefined headers does not work, you might want to try setting a custom header for this service. The custom header line to be used 'Call-Info:\;answe--after=0').
  ([a-z][0-9])

Note: If Custom SIP Header is not set, header is automatically detected via User-Agent.

Codecs


Advanced Options - Codecs

Codecs are used to convert voice signals from analog to digital and vice versa. These options set preferred codecs used by the extension.

NOTE: Opus codec will be the default codec when dialing mobile apps (gloCOM GO) in PBXware 6.0. This change affects incoming and outgoing calls via the mobile app.

TIP: If some of the desired codecs is disabled (cannot be selected), navigate to 'Settings: Tenants: Edit: Default Codecs' and enable them under the 'Local' group.

- **Codec**

Choose wich codec to use
(E.g. iLBC)
(Select box)

- **Ptime**

  Time in ms - represents how much audio recorded in itself will have each packet
  (E.g. 20ms)
  (Select box)

- **Video Support**

  Set this option to Yes to enable SIP video support.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Force codec on outbound trunk channel**

  With this option you can force codec use for outbound trunk calls.
  (E.g. iLBC)
  (Select box)

- **Auto-Framing (RTP Packetization)**

  If autoframing is turned on, the system will choose the packetization level based on remote ends preferences.
  (E.g. If the remote end requires RTP packets to be of 30 ms, your PBXware system will automatically send packets of this size if this option is turned on. Default is set to 20 ms and also depends on the codecs minimum frame size like G.729 which has 10 ms as a minimum).
  (Option buttons)

Codecs:

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.722** - 64 Kbps,
- **Opus** - 6-510 Kbps, 2.5 ms - 60 ms frame size
- **ITU G.723.1** - 5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve

compression over packet switched networks.
- **H.264 Video** - Offers great balance between quality and size

Jitter Buffer

- **Jitter buffer**

Advanced Options - Jitter Buffer

Choose Jitter Buffer Type
Available options are:

- **Inherit** - inherited jitter buffer settings from the tenant configuration.
- **Fixed** - Set a fixed jitter buffer on the channel.
- **Adaptive** - Set an adaptive jitter buffer on the channel.
- **Disabled** - Remove a previously set jitter buffer from the channel.

(Select box)

- **Max lenght (ms)**

Length in milliseconds for the buffer. By default it is 200 ms.
(E.g. 20)
([0-9])

- **Re-sync threshold**

The length in milliseconds over which a timestamp difference will result in resyncing the jitter buffer. By default it is 1000ms.
(E.g. 1000)
([0-9])

- **Target extra**

This only affects the adaptive jitter buffer. It represents the amount of time in milliseconds by which the new jitter buffer will pad its size. By default it is 40.
(E.g. 40)
([0-9])

Hot-desking

- **Automatic Logout (hours)**

Advanced Options - Hot Desking

Sets automatic log-out time in hours for devices set up for hot-desking.
(E.g. 5)

([0-9])

Recording

This group of options is used for the recording of all incoming/outgoing calls.


Advanced Options - Recording

TIP:

- Laws in some countries may require notifying the parties that their call is being recorded.
- Recorded calls, marked with icon, can be accessed from 'Self Care Interface' or 'Reports: CDR' PBXware' menu.
- Call are recorded in audio format set under 'Settings: Servers: Recordings Format'.

- **Record Calls**

  Enable call recording service. Select 'Yes' to enable the service. All incoming/outgoing calls will be recorded. If using call recording with many extensions, check server disk space from time to time. Please see below for bit rates table.
  (E.g. Yes, No, Not Set)
  (Option buttons)

- **Silent**

  Set whether call recording would be announced to the parties in a conversation. If Silent is set to ‘No’, calling parties will hear 'Recorded' or 'This call is recorded' message before their conversation starts.
  (E.g. Yes, No, Not Set)
  (Option buttons)
  NOTE: Since Extensions have a priority, all changes applied to Extensions will overwrite the system configuration. However, if "Not Set" is selected, then configuration set on the system globally or default configuration is applied.

- **Play Periodic Beep**

  To enable this feature, enter time in seconds to define how often periodic sound signal will be played to informed parties that call recording is enabled.
  For example, enter 60 to enable this feature and play periodic signal every 60 seconds.

  Disk Space Used By Call Recording

  With continuous tone for 60 seconds:

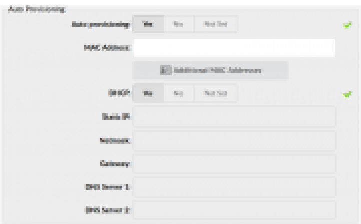
  - wav49 = 84.5kb
  - wav = 833.0kb
  - gsm = 85.0kb

With continuous silent tone (without sound) for 60 seconds:

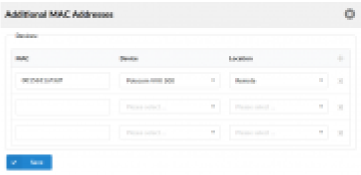
- wav49 = 84.0kb
- wav = 827.0kb
- gsm = 84.0kb

Auto Provisioning

These options enable PBXware MT to automatically provision the UAD/Phone. Configuration files are downloaded from PBXware MT's TFTP server



Advanced Options - Auto Provisioning



Advanced Options - Additional MAC Addresses

NOTE: These fields are merely templates when creating a new extension.

- **Auto Provisioning:**

  Enable auto provisioning service for this extension
  Connect the UAD/Phone to PBXware MT without any hassle by providing UAD/Phone MAC address (and optionally adding the Static UAD/Phone IP address and network details)
  (Option buttons)

- **MAC Address (Media Access Control):**

  UAD/Phone MAC address
  Provide the UAD/Phone address here. Its a 48-bit hexadecimal number (12 characters)
  ([a-z][0-9])

- **Additional MAC Addresses**

  Ability to use multiple MAC addresses per one Extension. This provides the ability to auto provision multiple phones attached to the same Extension.

- **DHCP (Dynamic Hosts Configuration Protocol):**

Set whether the UAD/Phone is on DHCP or Static IP address
Set DHCP = Yes if UAD/Phone is on dynamic or DHCP = No if UAD/Phone is on static IP address. If on static IP, you will have to provide more network details in the fields below.
(Option buttons)

- **Static IP:**

  Provide the Static UAD/Phone IP address. For this, you have to set DHCP = No
  ([0-9][.])

- **Netmask:**

  UAD/Phone netmask
  (E.g. Netmask applied to UAD/Phone static IP address)
  ([0-9][.])

- **Gateway:**

  Gateway IP address
  (E.g. Local area network gateway IP address)
  ([0-9][.])

- **DNS Server1 and Server2 (Domain Name Server):**

  DNS Server IP address
  (E.g. Local area network DNS IP address (Usually the same as your gateway))
  ([0-9][.])

Presence

This option simply notifies you of whether device presence is enabled or disabled. Supported UADs can be seen in the Settings->UAD menu.

- **Presence Enabled:**

Advanced Options - Pressence

  Enable presence support, but not every UAD/Phone supports this feature
  (Read only)

- **Global Presence:**

  Enables presence like the option above but when this option is turned on, it will enable presence for all tenants on the system. Please note, enabling this option will prevent you from monitoring presence status for extensions on the same tenant, unless you enter tenant prefix in front of extension number (e.g. 2001001 for extension 1001 on tenant 200.). However, setting Global Presence to Yes, will prevent you to use BLFs to monitor parking lots on your tenant as their numbers do not have tenant prefix.

(E.g. Yes, No, Not Set)
(Option buttons)

Supported UADs:

- Snom 190(Firmware >= 3.60s), 320/360(Firmware >= 4.1)
- Polycom IP30x/IP50x/IP600
- Xten EyeBeam
- Grandstream GXP2000 (Firmware >= 1.0.1.13)
- Aastra 480i
- Aastra 9133i

CLI

```
show hints
-= Registered Asterisk Dial Plan Hints =-
1009                : SIP/1009            State:Idle              Watchers  0
2001                : SIP/2001            State:Idle              Watchers  0
1020                : SIP/1020            State:InUse             Watchers  0
1016                : SIP/1016            State:Unavailable       Watchers  0
1008                : SIP/1008            State:Idle              Watchers  0
1006                : SIP/1006            State:Unavailable       Watchers  0
1000                : SIP/1000            State:Ringing           Watchers  0
1003                : SIP/1003            State:Unavailable       Watchers  0
1030                : SIP/1030            State:Unavailable       Watchers  0
1234                : IAX2/1234           State:Unavailable       Watchers  0
7777                : IAX2/7777           State:Idle              Watchers  0
1017                : IAX2/1017           State:Unavailable       Watchers  0
----------------
- 12 hints registered
```

User Agent Auto Provisioning Template

UAD Auto Provisioning Template Text box

This option allows adding of additional settings to auto-provisioning template. Auto-provisioning settings are generally defined in the 'Settings: UAD' and are custom set for each device.

NOTE: Unless absolutely sure, do not change or add to this template.

Additional Config

Additional Config text box

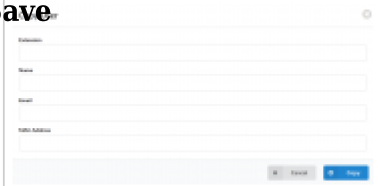
This option is used for providing additional config parameters for SIP and IAX configuration files. Values provided here will be written into these configuration files.

NOTE: Unless absolutely sure, do not change or add to this template.

Buttons

Buttons

- **Save**


Copy As New

Save changes

- **Save & E-mail**

Save changes and send an email to email address from E-mail field from General section

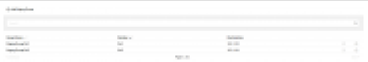
- **Copy As New**

Create a new extension by making copies of a current one

- **Go back**

Go back without saving changes

Paging Groups


Paging Groups

Paging Groups feature works similar to standard paging, except this feature allows you to organize extensions to multiple paging groups and to assign a unique number to each of them. As this feature is used with access code *600, paging group number is entered after the access code.

For example, if we assign number 300 to the paging group and add 4 extensions to it, once we dial *600300 we will be able to broadcast the message over the intercom to all the extensions added to paging group 300.

- **Search**

In search section you can find your Paging Group by typing its Name or Number.

- **Group Name**

Paging group name
(Display)

- **Number**

Paging Group number. Once the user dials *600 + this number, all extensions assigned to this paging group will be paged.
(Display)

- **Destinations**

Extensions associated with a paging group
(E.g. 102,103,105...)
(Display)

NOTE: Please bear in mind that the destinations limitation is 100 extensions.

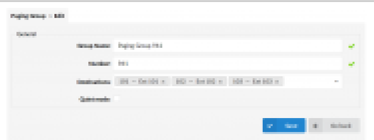
-

Click to edit the paging group configuration.
(Button)

-

Click to delete a paging group from the system.
(Button)

Add/Edit Paging Group



Add/Edit Paging Group

Clicking on 'Add Paging Group' or 'Edit' will open the following options:

General

- **Group Name**

Unique Paging group name
(E.g. Group I)
([a-z][0-9])

- **Number**

Paging group extension number
(E.g. 1010)
([0-9])

- **Destinations**

  System extensions associated with the paging group
  (E.g. 101, 102, 103... Selected extensions will be paged)
  (Multi select)

- **Quiet mode**

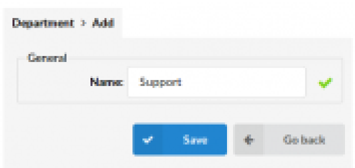
  Does not play beep to paged extensions.
  (Checkbox)

Departments

Departments

Departments section will list all the departments present on this <%PRODUCT%> system, and give the ability to edit or add a new ones. Departments are used by Bicom Systems gloCOM to sort extensions based on the department they belong to.

Add/Edit Department



Add/Edit Department

When you click on *Add Department* link or the edit button you will be presented with this screen:

General

- **Name:**

  Name of the department
  (E.g. Accounting)
  ([0-9][a-z])

Hot Desking

Hot Desking is a feature that allows your business the practice of not assigning permanent desks in a workplace, so that employees may work at any available desk.

From a managerial perspective, Hot Desking is attractive because it can cut overhead costs significantly. However, the concept won't work in environments where employees are expected to be in the office most of the time.

Hot Desking, as PBXware feature, is simple as it can be. By dialing proper access code (*555 by default) on any pre-configured office phone for HotDesking, user will go to an IVR, where it will be asked for extension and pin. Once proper extension / pin combination is entered, the phone will be rebooted and auto-provisioned with the new extension.

If there was any phone already registered with the same extension, it will reboot too and auto provisioned with dynamic extension. If extension is in use, phone will reboot once the call ends. Phones provisioned with dynamic extension will not be able to dial anything but Hot Desking IVR.

Emergency numbers can be dialed even if the hot desking device is not logged in (if no extension is provisioned). In this case, the Emergency Caller ID will be used for the outgoing call.

NOTE: To be able to use Hot Desking feature in PBXware 6, it has to be enabled in your PBXware license. For more information on how to get Hot Desking enabled, please contact your account manager.

- **MAC**

Hot Desking

  MAC address of the device.
  (Display)

- **Device**

  Hot Desking Device
  (Display)

- **Extension**

  Extension's number associated with MAC/Device.
  (Display)

- **Enabled**

  Status of Hot Desking device.
  (Display)

-

  Click to edit the hot desking device configuration.
  (Button)

-

  Click to delete a hot desking device from the system.
  (Button)

CSV upload

There is an option to upload CSV file instead of manually entering phones. Click Browse button, select the .csv file from your hard drive and press upload button. Please make sure that CSV file contain comma separated MAC address of the phone and device type per each line (MAC,device), like in example below:

> 001565267db9,yealinkt42p
> 0004f23fd871,polycomvvx300

In case you are not sure what to enter for your device name, check the Settings -> UAD, edit UAD for your device model and in General (section) check exact name set under Internal UAD name for Auto-Provisioning field.

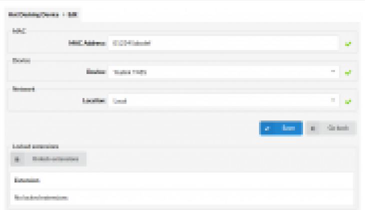
CSV download

Once you have created list of Hot Desking devices you might get in to position where you would like to export it to a new system, for testing purposes or in case of a migration for example. CSV Download button gives you an option to download a full list of hot desking devices from a tenant, in CSV format, which can later be used to recreate the list on a new system/tenant.

Download CSV Template

As described earlier you can add large number of hot desking devices at once, by creating a CSV file. Download CSV Template button will present you with a file that already contains necessary headers which should help you create CSV file easier.

Add/Edit Hot Desking Device



Add Hot Desking Device

MAC

- **MAC address**

  MAC address of the UAD.

Device

- **Device**

Select a device for Hot Desking.

NOTE: List of supported devices can be found on the following page: https://wiki.bicomsystems.com/Hot_Desking_Supported_Devices

Network

- **Network**

Network refers to whether the UAD/Phone is in the 'Local' or 'Remote' network.

Emergency CallerID

- **Emergency CallerID**

CallerID entered here will be used only for calls to Emergency Services numbers.
Emergency numbers can be dialed even if the hot desking device is not logged in. In this case, the Caller ID set here will be used for the outgoing call.

Locked extensions

- **Unlock extensions**

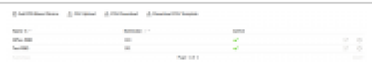
Unlock locked extensions
(Button)

- **Extension**

List of all locked extensions for this device
(Display)

PIN Based Devices

Every extension has its own unique PBD PIN. When you are making a call from a device, that is set as a Pin Based device, after you enter the desired extension number it will ask you to enter your PBD PIN. This PIN will identify the user on the system and then dialling will proceed as if the user was dialling from his own extension. Call Rating, CDRs and everything else will apply for the user extension and not for the extension of the Pin Based device. To make a call with extension that is connected with a PIN Based device you must enter the PBD PIN for that extension.



PIN Based Device

CSV Upload

There is an option to upload CSV file instead of manually entering PIN Based devices. Click Browse button, select the .csv file from your hard drive and press upload button.

Please make sure that CSV file contain comma separated names, extensions and information if Device is active or not, like in example below:

Device I,1050,yes
Device II,1009,no
(Button)

CSV Download

CSV Download button gives you an option to download a full list of PIN Based devices from a tenant, in CSV format, which can later be used to recreate the list on a new system/tenant
(Button)

Download CSV Template

Download CSV Template button will present you with a file that already contains necessary headers which should help you create CSV file easier
(Button)

Once you click on PIN Based Devices in Extensions Menu, page with following details will be opened:

- **Name**

  Name of a PIN Based Device
  (Display)

- **Extension**

  Extension used by the device
  (Display)

- **Active**

  Is PIN Based Device active or not
  (Display)

-

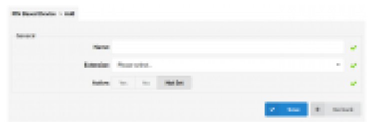
  Click to edit a PIN based device configuration.
  (Button)

-

  Click to delete a PIN based device from the system.

(Button)

Add/Edit PIN Based Device


Add/Edit PIN Based Device

General

- **Name**

  Name of a PIN Based Device
  (E.g. Device 1)
  ([a-z][0-9])

- **Extension**

  Extension which will be used by the device.
  (Select box)

- **Active**

  Whether PIN based device is active or not
  (E.g. Yes, No Not Set)
  (Option buttons)

Aditional information

PBD local extensions

PIN Based dialling is disabled for local extensions. Entering PBD Pin for local calls is not required.

PBD access codes

Using access codes requires proper identification of the user with PBD pin. Most of the access codes are related per user, which means that they can be used only if they are enabled under enhanced services. Dialing access codes will result in requiring to enter the PBD pin so that system can identify the user and do required actions.

Access codes that can be used with PBD:

- General Voicemail = *124
- Call Park = 700
- Call Park Start = 701
- Call Park End = 720
- Speed Dial = *130

- Call Pickup = *8
- Asterisk Call Pickup = *88

Caller ID List


Caller ID List

In this section, users can see all extensions on this tenant, along with the values from Enhanced Services/Caller ID for each extension individually. Those values are applied here and visible in one place, so users can associate each emergency call with the calling extension.

CSV Download

By clicking the CSV Download icon user can download the CSV file that contains all the data visible from the list.

Search

Click Search icon in order to open a search filter that would allow users to find extensions by Extension Number or Extension Name.

- **Extension Number**

UAD/Phone extension number
(E.g. 108)
(Display)

- **Extension Name**

Full name of the user to which the device is registered
(E.g. Jane Doe).
(Display)

- **System/Network Caller ID**

This value is applied here from Enhanced Services/CallerID.
(Display)

- **Emergency Caller ID**

This value is applied here from Enhanced Services/CallerID.
(Display)
- **Trunk Caller IDs**

This value is applied here from Enhanced Services/CallerID. By clicking Trunk Caller IDs icon users can see which Caller ID is used by an extension on different trunks.
(Button)

Next -> 5. Trunks
