

Extensions

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Overview

The **Extensions** page is the tenant-level list for creating, finding, editing, importing, exporting, and deleting extensions. Open it from **Configuration > Extensions**.

The page works against the tenant currently selected in the top bar. The screenshots show ten fake demo extensions, numbered **100** through **109**, with a mix of chan_sip, PJSIP, custom, and virtual extension types. SIP and PJSIP usernames use the extension number followed by the tenant code, for example `100-CANISTRACCI`. Password values are masked in the screenshots.

Display modes

The Extensions page can be shown in DataTable mode or jqGrid mode. Use the display mode toggle in the toolbar to switch between them.

DataTable mode

Number	Name	Description	Username	Password	Action
100	Alex Reed	Demo chan_sip phone extension	100-CANISTRACCI	*****	edit
101	Bella Stone	Demo PJSIP phone extension	PJSIP 101-CANISTRACCI	*****	edit
102	Casey Lane	Demo chan_sip phone extension	102-CANISTRACCI	*****	edit
103	Drew Hart	Demo PJSIP phone extension	PJSIP 103-CANISTRACCI	*****	edit
104	Eva Brooks	Demo chan_sip phone extension	104-CANISTRACCI	*****	edit
105	Finn Cooper	Demo PJSIP phone extension	PJSIP 105-CANISTRACCI	*****	edit
106	Gia Parker	Demo custom extension	106-CANISTRACCI	*****	edit
107	Hugo Blair	Demo custom extension	107-CANISTRACCI	*****	edit
108	Ivy Morgan	Demo virtual extension	100 101 106	*****	edit
109	Jules Quinn	Demo virtual extension	102 103 104 105 107	*****	edit

Extensions page in DataTable display mode with demo extensions 100-109.

DataTable mode shows the standard table with search, page-size selection, sorting, row checkboxes, and action buttons above the table.

jqGrid mode

Number	Name	Description	Tech	Username	*****
100	Alex Reed	Demo chan_sip phone exte	SIP	100-CANISTRACCI	Demo100/2026
101	Bella Stone	Demo PJSIP phone extensi	PJSIP	101-CANISTRACCI	Demo101/2026
102	Casey Lane	Demo chan_sip phone exte	SIP	102-CANISTRACCI	Demo102/2026
103	Drew Hart	Demo PJSIP phone extensi	PJSIP	103-CANISTRACCI	Demo103/2026
104	Eva Brooks	Demo chan_sip phone exte	SIP	104-CANISTRACCI	Demo104/2026
105	Finn Cooper	Demo PJSIP phone extensi	PJSIP	105-CANISTRACCI	Demo105/2026
106	Gia Parker	Demo custom extension	CUSTOM	106-CANISTRACCI	
107	Hugo Blair	Demo custom extension	CUSTOM	107-CANISTRACCI	
108	Ivy Morgan	Demo virtual extension	VIRTUAL		
109	Jules Quinn	Demo virtual extension	VIRTUAL		

Extensions page in jqGrid display mode with demo extensions 100-109.

jqGrid mode shows the same extension records in the advanced grid with column filters and grid toolbar actions.

Toolbar actions

Extensions toolbar with creation, display mode, and upload controls.

Action	Description
New SIP peer / New PJSIP peer	Creates a standard phone extension. The visible label follows the tenant's preferred SIP stack.
Bulk SIP peers	Opens the bulk creation workflow for adding multiple SIP extensions at once.
Custom Extension	Creates an extension that routes to a custom dial string or custom endpoint logic.
Virtual Extension	Creates a virtual extension that can group or reference other extension destinations.
Change to chan_sip	Converts selected PJSIP extensions to chan_sip when the administrator has permission to switch phone technology.
Change to PJSIP	Converts selected chan_sip extensions to PJSIP when the administrator has permission to switch phone technology.

Action	Description
Delete Selected	Deletes the selected extensions after a confirmation prompt. This action is shown only when delete permission is available.
Display mode toggle	Switches the page between the standard DataTable and jqGrid display modes.
CSV/XLS Upload	Opens the import page for uploading extension records from a CSV or XLS file.

Extension table

<input type="checkbox"/>	Number ▲	Name	Description	Tech	Username	Password
	<input type="text"/> x	<input type="text"/> x	<input type="text"/> x	<input type="text"/> x	<input type="text"/> x	<input type="text"/> x
<input type="checkbox"/>	100	Alex Reed	Demo chan_sip phone exte	SIP	100-CANISTRACCI	*****
<input type="checkbox"/>	101	Bella Stone	Demo PJSIP phone extensio	PJSIP	101-CANISTRACCI	*****
<input type="checkbox"/>	102	Casey Lane	Demo chan_sip phone exte	SIP	102-CANISTRACCI	*****
<input type="checkbox"/>	103	Drew Hart	Demo PJSIP phone extensio	PJSIP	103-CANISTRACCI	*****
<input type="checkbox"/>	104	Eva Brooks	Demo chan_sip phone exte	SIP	104-CANISTRACCI	*****
<input type="checkbox"/>	105	Finn Cooper	Demo PJSIP phone extensio	PJSIP	105-CANISTRACCI	*****
<input type="checkbox"/>	106	Gia Parker	Demo custom extension	CUSTOM	106-CANISTRACCI	
<input type="checkbox"/>	107	Hugo Blair	Demo custom extension	CUSTOM	107-CANISTRACCI	
<input type="checkbox"/>	108	Ivy Morgan	Demo virtual extension	VIRTUAL		
<input type="checkbox"/>	109	Jules Quinn	Demo virtual extension	VIRTUAL		

| Columns CSV Exp Show flo View 1 - 10 of 10

Extensions grid showing mixed SIP, PJSIP, custom, and virtual extensions.

Use the table controls to filter, sort, select columns, export data, and move between pages. The checkbox column selects rows for bulk actions.

Column	Description
Number	Extension number. When extension status is enabled, an icon indicates registration or device state.
Name	Extension display name. Icons can indicate call blocking/filtering, DND, unconditional forwarding, or PIN lock.

Column	Description
Description	Free-form extension description. Icons can identify PJSIP, custom, virtual, or recording-related behavior.
Tech	Extension technology. SIP entries are chan_sip peers, PJSIP entries are PJSIP endpoints, CUSTOM entries route to custom dial strings, and VIRTUAL entries group other extensions.
Username	SIP/PJSIP username used by the phone or endpoint. The default pattern is <code><extension number>--<tenant code></code> . Clipboard, SRTP, and TLS indicators can appear for phone extensions.
Password	Endpoint secret when password display is enabled. Depending on settings, the value can be hidden or revealed with an eye icon.
Action	Contains quick actions such as showing the extension call flow. Fast provisioning QR code access can also appear when enabled.

Editing extensions

Select the extension number, name, description, username, or password cell to open the extension edit form. In jqGrid mode, select a row and use the grid edit action; the grid opens the same edit form.

jqGrid mode

The display mode toggle can enable jqGrid for the Extensions page. In jqGrid mode, the same extension records are loaded dynamically, and the grid adds toolbar actions for edit, delete, search, column selection, CSV export, and showing the selected extension flow.

Permissions and limits

Some buttons are hidden when the user lacks permission or when the tenant has reached its extension limit. The page also respects user extension restrictions, so administrators may see only the extensions assigned to their profile.

Create and Edit chan_sip Extensions

Use this page to create or edit a phone extension that registers as a chan_sip peer. The form works against the tenant currently selected in the top bar.

The screenshots use the **Canistracci Oil** demo extension **100 Alex Reed**. Creating and editing use the same form: a new extension opens with tenant defaults, while an existing extension opens with saved values.

Creating and editing

Action	Description
Create	Open Configuration > Extensions and select New SIP peer. Fill the required fields and select Save.
Edit	Open an existing extension from the Extensions list. Update the required fields and select Save.
Delete	When delete permission is available, open the extension and select Delete. Confirm only after checking routing, phones, queues, and other references.

For chan_sip extensions, the endpoint username normally uses the extension number, a hyphen, and the tenant code, such as 100-CANISTRACCI. Press the swapped-arrow icon beside Username to switch the separator from a hyphen to an underscore, for example 100_CANISTRACCI. If custom usernames are enabled in Admin > Settings, you can enter a username that does not follow the generated tenant-code pattern.

Information

Number:

Name: Trunk

Description:

SIP stack:

Username:

Password:

Codecs:

G.711 A-law	<input type="button" value="🗑"/>
G.711 u-law	<input type="button" value="🗑"/>
GSM	<input type="button" value="🗑"/>

DTMF Mode:

Progress inband:

Direct Media:

Direct RTP Setup:

Call Group:

1	<input type="button" value="🗑"/>
---	----------------------------------

Pickup Groups:

1	<input type="button" value="🗑"/>
---	----------------------------------

Extension Voicemail MWI:

Virtual Voicemail MWI:

Call Limit:

Do Not Disturb (DND)

Inbound Dial Timeout:

Information section for chan_sip extension 100.

Use this section for the extension identity, endpoint credentials, media basics, groups, voicemail message-waiting indication, DND state, and inbound ring timeout.

Block	Purpose
Number and name	Number is the internal extension number. Name is the display name used in lists, caller ID, and reports.
Description and emergency notes	Description is an administrator note. Emergency notes can be used by emergency routing and notification logic.
SIP stack	Shows whether the endpoint is using chan_sip or PJSIP. Users with switch permission can change the stack from this form or from the Extensions list.
Username and password	Endpoint credentials used by the phone or softphone. The default username pattern is the extension number followed by the tenant code.
Codecs and DTMF	Controls the allowed audio/video codecs and DTMF signaling mode used by the endpoint.
Media and groups	Direct media, call groups, pickup groups, spy groups, and message-waiting options control how the endpoint participates in calls and monitoring.
Voicemail, DND, and timeout	Voicemail MWI, DND, and inbound dial timeout control how incoming calls are presented and how long the extension rings.

Trunk mode

Information
Show All

Number:

Name: Trunk

Description:

SIP stack:

DID Number:

CallerID Number Override:

Emergency CallerID Number Override:

CallerID Number Source:

Additional trunk mode options visible after enabling Trunk on a chan_sip extension.

Enable **Trunk** only when the chan_sip peer is another PBX, gateway, or upstream device that should receive calls by dialed number rather than as a single telephone endpoint. A normal phone extension should usually leave this option disabled.

Field or option	Effect
Trunk	Changes how calls are sent to the peer. With trunk mode disabled, MiRTA PBX dials the peer username. With trunk mode enabled, MiRTA PBX dials the called number at that peer, or the configured DID override when one is set.
DID Number	Automatic sends the original called number to the peer. Set to sends the value entered in the DID number field instead, which is useful when the downstream PBX or gateway expects a fixed pilot number or a rewritten DID.
CallerID Number Override	Controls which outbound caller ID values are preserved from the SIP INVITE received from the trunk device. The default behavior can keep both caller ID name and number from the INVITE. The other choices keep only the number, keep only the name, or ignore the INVITE caller ID and use the caller ID configured on the extension.

Field or option	Effect
Emergency CallerID Number Override	For emergency routes, Get Emergency CallerID from SIP Invite allows the emergency caller ID sent by the trunk device to pass through. Do not use Emergency CallerID from SIP Invite makes MiRTA PBX replace it with the extension emergency caller ID, or the tenant default emergency caller ID when the extension does not define one.
CallerID Number Source	Selects where inbound trunk caller ID is read from when a specific source is required. Automatic keeps the caller ID parsed by Asterisk. The explicit choices read the From, P-Asserted-Identity, P-Preferred-Identity, or Remote-Party-ID header and use it to set caller ID before routing.

In the dialplan, trunk mode also affects direct calls to this extension. If the peer is registered on the current server or has a static host, the call is built as a trunk-style dial to the peer using the dialed number. If the peer is registered on another node, MiRTA PBX sends the call through the configured inter-node SIP trunking module.

NAT Control

NAT Control

NAT:

Qualify:

Qualify Frequency:

Keep Alive:

RTP Keep Alive:

NAT Control section for chan_sip extension 100.

Use this section to control endpoint reachability and keepalive behavior for devices behind NAT or with changing network paths.

Block	Purpose
NAT	Defines how MiRTA PBX treats NAT traversal for the endpoint.
Qualify and qualify frequency	Controls whether the PBX checks endpoint reachability and how often checks are sent.

Block	Purpose
Keep alive and RTP keep alive	Sends periodic traffic to keep network mappings open and detect unavailable devices.

Call Settings

Call Settings

Fax:

Volume TX level:

Volume RX level:

Music on hold:

Language:

Include in Dial By Name directory:

Additional Dial By Name:

Dial By Name recording:

Include in Phone Books:

Call waiting:

Autoanswer:

Parkinglot:

Push notification

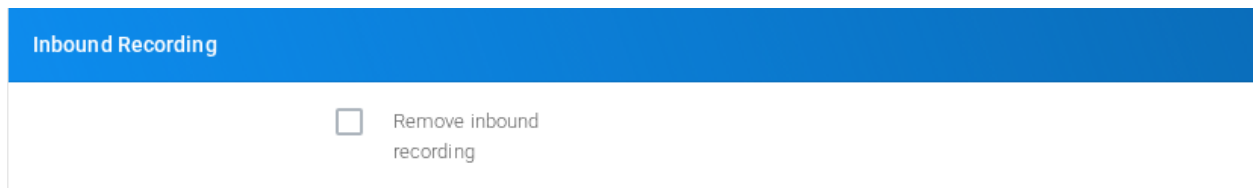
Call Settings section for chan_sip extension 100.

Use this section for call behavior that is not specific to inbound or outbound routing.

Block	Purpose
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Fax and volume	Controls T.38 fax behavior and transmit/receive volume adjustments.
Music on hold and language	Selects media behavior and the preferred language for prompts played to the extension.
Directories and phone books	Controls whether the extension appears in dial-by-name directories and phone book generated lists.
Call waiting and autoanswer	Controls second-call handling and optional automatic answer behavior for supported phones.

Inbound Recording



Inbound Recording section for chan_sip extension 100.

Use this section for recording behavior applied to calls received by the extension.

Block	Purpose
Inbound recording policy	Controls whether inbound calls are recorded and whether recording privacy behavior is enabled.
Recording notifications	When enabled, recordings can be mailed and processed according to tenant recording settings.

Outbound Recording

Outbound Recording

Always Record:

Email recording to:

Transcript recorded calls:

Summarize recorded calls:

Sentiment analysis for recorded calls:

Playback recording announce message:

Outbound Recording section for chan_sip extension 100.

Use this section for recording and post-processing behavior applied to calls placed by the extension.

Block	Purpose
Always record	Controls whether outbound calls from this extension are recorded, unless tenant-wide settings force recording.
Email recording to	Sends matching recordings to the configured address. Minimum size can suppress very small recordings.
Transcript, summary, sentiment	Enables transcript, summary, and sentiment processing when the tenant supports those services.

Security

Security

Host:

Insecure:

Transport:

WebRTC Support:

RTP Encryption (SRTP):

Outbound Proxy:

Send RPID:

Trust RPID:

TOS Audio:

TOS Video:

COS Audio:

COS Video:

Regex Filter on SIP From:

Regex Filter on SIP User Agent: Learn

Abuse Detection:

Apply call cost limits:

Alert when offline

Alert email:

Outbound Destinations:

Lock PIN: Locked

Max outbound call duration:

Working hours restrictions:

Override Tenant IP Restrictions

Security section for chan_sip extension 100.

Use this section to restrict registration, authentication, abuse behavior, cost limits, and outbound availability.

Block	Purpose
Authentication and network filters	Controls endpoint trust, allowed IPs, and authentication-related behavior.
Abuse detection and cost limits	Locks or warns on suspicious or excessive usage according to tenant policy.
Outbound destinations	Allows all calls, blocks calls, or applies a destination regex filter.
Lock PIN and max duration	Allows the extension to be locked and caps outbound call duration.
Working hours	Restricts outbound calling to the selected weekday condition.

Web User Panel and Switchboard

Web User Panel and Switchboard

Allow Web User Panel Access

Allow Web Phone Usage

Allow Switchboard Usage

Web User:

Password: LDAP

Never expire

Force password change at login

Lock password, user can't change it

Use Two Factors Authentication (2FA):

Email:

User Profile:

Use IP Filter

Web User Panel and Switchboard section for chan_sip extension 100.

Use this section to enable user-facing web access and related application permissions for the extension.

Block	Purpose
Access toggles	Enables the user panel, web phone, switchboard, and optional custom user application.
Web user and password	Sets the login identity and password. If no web user is specified, the endpoint username is used.
Password policy and 2FA	Controls expiry, forced change, lock state, LDAP use, and two-factor authentication.
Email, profile, and IP filter	Sets the user email, sends account information, assigns a user profile, and optionally restricts web access by IP.

Outbound Calls

Outbound Calls

Block External Caller ID

External CID Number:

External CID Name: Auto

Override CID Number:

Override CID Name:

Use this callerid when in a virtual extension

Emergency CID Number: Ignore empty

Area Code: Use default

Area Code Regex: Use default

Use Do Not Call lists:

Use Only Allow Call lists:

Routing Profile: Apply always

SMS Routing Profile:

Call Rate:

Outbound Calls section for chan_sip extension 100.

Use this section to control the caller ID, emergency caller ID, area-code handling, dialing filters, routing profile, and call rate used by outbound calls.

Block	Purpose
External caller ID	Sets or hides the caller ID number and name sent on outbound calls.
SMS and override caller ID	Controls SMS caller ID and optional SIP header based caller ID overrides.
Emergency caller ID	Defines the emergency number identity and whether empty emergency caller ID warnings are ignored.
Area code and caller ID regex	Applies prefixes and caller ID rewrite rules before routing.

Block	Purpose
Do Not and Only Allow lists	Applies tenant call-list restrictions to this extension.
Routing profile and call rate	Selects the routing profile, SMS routing profile, and client rate used by calls.

Inbound Calls

Inbound Calls

Block Inbound Caller ID Number

Block Inbound Caller ID Name

Inbound Calls section for chan_sip extension 100.

Use this section for inbound caller ID privacy controls.

Block	Purpose
Block inbound caller ID number	Prevents the caller ID number from being shown to this extension.
Block inbound caller ID name	Prevents the caller ID name from being shown to this extension.

Find me/Follow me Configuration

Find me/Follow me Configuration

FMFM Number: Active if checked

FMFM Condition: ▼

FMFM Dial Method: ▼ Request confirmation

FMFM Caller ID: ▼

FMFM Caller ID Num Prefix:

FMFM Caller ID Name Prefix:

FMFM Dial Timeout:

Find me/Follow me Configuration section for chan_sip extension 100.

Use this section to forward calls to an alternate number when find-me/follow-me is enabled.

Block	Purpose
FMFM number and status	Sets the external follow-me number and enables or disables the feature.
Condition and dial method	Limits follow-me behavior to a condition and selects how the alternate number is dialed.
Confirmation and messages	Requires answer confirmation and selects confirm/hold messages.
Caller ID and timing	Controls caller ID presentation, prefixes, delay, and dial timeout for the follow-me call.

Additional Destinations - Active if checked

Additional Destinations - Active if checked

Unconditional: Action to take ▼

On No Answer: Action to take ▼

On Extension Busy: Action to take ▼

On Extension Offline: Action to take ▼

On Condition: Choose condition to check ▼

Action to take ▼

Missed call notification to: Ignore on internals Ignore from queues

Additional Destinations - Active if checked section for chan_sip extension 100.

Use this section to define failover or conditional routing for calls that do not complete normally.

Block	Purpose
Unconditional	Always routes calls to the selected destination when enabled.
On no answer, busy, or offline	Routes calls when the extension does not answer, is busy, or is offline.
On condition	Routes calls to a selected destination when the chosen condition matches.
Missed call notification	Sends email notifications for missed calls, with options to ignore internal or queue-originated calls.

Note

Note

Reference ID:

Additional Info:

Save
Delete
Back

Note section for chan_sip extension 100.

Use this section for administrative classification and free-form notes, then save or delete the extension.

Block	Purpose
Branch and department	Classifies the extension for reporting and administration.
Reference ID and additional info	Stores external references and notes.
Save, delete, and back	Save applies changes, Delete removes the extension when allowed, and Back returns to the Extensions list.

Create and Edit PJSIP Extensions

Use this page to create or edit a phone extension that registers as a PJSIP endpoint. The form works against the tenant currently selected in the top bar.

The screenshots use the **Canistracci Oil** demo extension **101 Bella Stone**. Creating and editing use the same form: a new extension opens with tenant defaults, while an existing extension opens with saved values.

Creating and editing

Action	Description
Create	Open Configuration > Extensions and select New PJSIP peer. Fill the required fields and select Save.
Edit	Open an existing extension from the Extensions list. Update the required fields and select Save.
Delete	When delete permission is available, open the extension and select Delete. Confirm only after checking routing, phones, queues, and other references.

For PJSIP extensions, the endpoint username normally uses the extension number, a hyphen, and the tenant code, such as 101-CANISTRACCI. Press the swapped-arrow icon beside Username to switch the separator from a hyphen to an underscore, for example 101_CANISTRACCI. If custom usernames are enabled in Admin > Settings, you can enter a username that does not follow the generated tenant-code pattern.

Information

Number:	<input type="text" value="101"/>
Name:	<input type="text" value="Bella Stone"/> <input type="checkbox"/> Trunk
Description:	<input type="text" value="Demo PJSIP phone extension"/>
SIP stack:	<input type="text" value="PJSIP"/> <input type="text" value="Endpoint"/> <input type="button" value="Edit"/>
Username:	<input type="text" value="101-CANISTRACCI"/> <input type="button" value="↔"/>
Password:	<input type="text" value="*****"/> <input type="button" value="Generate"/>
Codecs:	<div><input type="text" value="Please select allowed codecs"/><ul style="list-style-type: none">G.711 A-law <input type="button" value="🗑"/>G.711 u-law <input type="button" value="🗑"/>GSM <input type="button" value="🗑"/></div>
DTMF Mode:	<input type="text" value="Auto"/>
Progress inband:	<input type="text" value="Never"/>
Direct Media:	<input type="text" value="No"/>
Direct RTP Setup:	<input type="text" value="No"/>
Call Group:	<div><input type="text"/><ul style="list-style-type: none">1 <input type="button" value="🗑"/></div>
Pickup Groups:	<div><input type="text"/><ul style="list-style-type: none">1 <input type="button" value="🗑"/></div>
Extension Voicemail MWI:	<input type="text" value="No MWI"/>
Virtual Voicemail MWI:	<input type="text" value="No MWI"/>
	<input type="checkbox"/> Do Not Disturb (DND)
Inbound Dial Timeout:	<input type="text" value="Tenant default"/> <input type="button" value="🔍"/>

Information section for PJSIP extension 101.

Use this section for the extension identity, endpoint credentials, media basics, groups, voicemail message-waiting indication, DND state, and inbound ring timeout.

Block	Purpose
Number and name	Number is the internal extension number. Name is the display name used in lists, caller ID, and reports.
Description and emergency notes	Description is an administrator note. Emergency notes can be used by emergency routing and notification logic.
SIP stack	Shows whether the endpoint is using chan_sip or PJSIP. Users with switch permission can change the stack from this form or from the Extensions list.
Username and password	Endpoint credentials used by the phone or softphone. The default username pattern is the extension number followed by the tenant code.
Codecs and DTMF	Controls the allowed audio/video codecs and DTMF signaling mode used by the endpoint.
Media and groups	Direct media, call groups, pickup groups, spy groups, and message-waiting options control how the endpoint participates in calls and monitoring.
Voicemail, DND, and timeout	Voicemail MWI, DND, and inbound dial timeout control how incoming calls are presented and how long the extension rings.

Trunk mode

Information
Show All

Number:

Name: Trunk

Description:

SIP stack: PJSIP Endpoint Edit

DID Number: Set to

CallerID Number Override: Get CallerID name and number from SIP Invite

Emergency CallerID Number Override: Get Emergency CallerID from SIP Invite

CallerID Number Source: PAI field

Additional trunk mode options visible after enabling Trunk on a PJSIP extension.

Enable **Trunk** only when the PJSIP endpoint is another PBX, gateway, or upstream device that should receive calls by dialed number rather than as a single telephone endpoint. A normal phone extension should usually leave this option disabled.

Field or option	Effect
Trunk	Changes how calls are sent to the endpoint. With trunk mode disabled, MiRTA PBX dials the PJSIP endpoint contacts as a phone endpoint. With trunk mode enabled, MiRTA PBX sends the called number to the endpoint, or the configured DID override when one is set.
DID Number	Automatic sends the original called number to the peer. Set to sends the value entered in the DID number field instead, which is useful when the downstream PBX or gateway expects a fixed pilot number or a rewritten DID.
CallerID Number Override	Controls which outbound caller ID values are preserved from the SIP INVITE received from the trunk device. The default behavior can keep both caller ID name and number from the INVITE. The other choices keep only the number, keep only the name, or ignore the INVITE caller ID and use the caller ID configured on the extension.

Field or option	Effect
Emergency CallerID Number Override	For emergency routes, Get Emergency CallerID from SIP Invite allows the emergency caller ID sent by the trunk device to pass through. Do not use Emergency CallerID from SIP Invite makes MiRTA PBX replace it with the extension emergency caller ID, or the tenant default emergency caller ID when the extension does not define one.
CallerID Number Source	Selects where inbound trunk caller ID is read from when a specific source is required. Automatic keeps the caller ID parsed by Asterisk. The explicit choices read the From, P-Asserted-Identity, P-Preferred-Identity, or Remote-Party-ID header and use it to set caller ID before routing.

In the dialplan, PJSIP trunk mode affects both direct endpoint dialing and registered-contact dialing. When contacts are used, MiRTA PBX rewrites the PJSIP contact URI user part from the endpoint identity to the dialed number, or to the DID override. If the endpoint is registered on another node, MiRTA PBX sends the call through the configured inter-node SIP trunking module.

NAT Control

NAT Control

NAT: ▼

Qualify: ▼

Qualify Frequency: ▼

RTP Keep Alive: ▼

NAT Control section for PJSIP extension 101.

Use this section to control endpoint reachability and keepalive behavior for devices behind NAT or with changing network paths.

Block	Purpose
NAT	Defines how MiRTA PBX treats NAT traversal for the endpoint.
Qualify and qualify frequency	Controls whether the PBX checks endpoint reachability and how often checks are sent.
Keep alive and RTP keep alive	Sends periodic traffic to keep network mappings open and detect unavailable devices.

Call Settings

Call Settings

Fax:

Volume TX level:

Volume RX level:

Music on hold:

Language:

Include in Dial By Name directory:

Additional Dial By Name:

Dial By Name recording:

Include in Phone Books:

Call waiting:

Autoanswer:

Parkinglot:

Push notification

Call Settings section for PJSIP extension 101.

Use this section for call behavior that is not specific to inbound or outbound routing.

Block	Purpose
Fax and volume	Controls T.38 fax behavior and transmit/receive volume adjustments.
Music on hold and language	Selects media behavior and the preferred language for prompts played to the extension.
Directories and phone books	Controls whether the extension appears in dial-by-name directories and phone book generated lists.

Block	Purpose
Call waiting and autoanswer	Controls second-call handling and optional automatic answer behavior for supported phones.

Inbound Recording

Inbound Recording

Remove inbound recording

Inbound Recording section for PJSIP extension 101.

Use this section for recording behavior applied to calls received by the extension.

Block	Purpose
Inbound recording policy	Controls whether inbound calls are recorded and whether recording privacy behavior is enabled.
Recording notifications	When enabled, recordings can be mailed and processed according to tenant recording settings.

Outbound Recording

Outbound Recording

Always Record:

Email recording to:

Transcript recorded calls:

Summarize recorded calls:

Sentiment analysis for recorded calls:

Playback recording announce message:

Outbound Recording section for PJSIP extension 101.

Use this section for recording and post-processing behavior applied to calls placed by the extension.

Block	Purpose
Always record	Controls whether outbound calls from this extension are recorded, unless tenant-wide settings force recording.
Email recording to	Sends matching recordings to the configured address. Minimum size can suppress very small recordings.
Transcript, summary, sentiment	Enables transcript, summary, and sentiment processing when the tenant supports those services.

Security

Security

Host:

Insecure:

Transport:

WebRTC Support:

RTP Encryption (SRTP):

Outbound Proxy:

Send RPID:

Trust RPID:

TOS Audio:

TOS Video:

COS Audio:

COS Video:

Regex Filter on SIP From:

Regex Filter on SIP User Agent: Learn

Abuse Detection:

Apply call cost limits:

Alert when offline

Alert email:

Outbound Destinations:

Lock PIN: Locked

Max outbound call duration:

Working hours restrictions

Override Tenant IP Restrictions

Security section for PJSIP extension 101.

Use this section to restrict registration, authentication, abuse behavior, cost limits, and outbound availability.

Block	Purpose
Authentication and network filters	Controls endpoint trust, allowed IPs, and authentication-related behavior.
Abuse detection and cost limits	Locks or warns on suspicious or excessive usage according to tenant policy.
Outbound destinations	Allows all calls, blocks calls, or applies a destination regex filter.
Lock PIN and max duration	Allows the extension to be locked and caps outbound call duration.
Working hours	Restricts outbound calling to the selected weekday condition.

Web User Panel and Switchboard

Web User Panel and Switchboard

Allow Web User Panel Access

Allow Web Phone Usage

Allow Switchboard Usage

Web User:

Password: LDAP

Never expire

Force password change at login

Lock password, user can't change it

Use Two Factors Authentication (2FA):

Email:

User Profile:

Use IP Filter

Web User Panel and Switchboard section for PJSIP extension 101.

Use this section to enable user-facing web access and related application permissions for the extension.

Block	Purpose
Access toggles	Enables the user panel, web phone, switchboard, and optional custom user application.
Web user and password	Sets the login identity and password. If no web user is specified, the endpoint username is used.
Password policy and 2FA	Controls expiry, forced change, lock state, LDAP use, and two-factor authentication.
Email, profile, and IP filter	Sets the user email, sends account information, assigns a user profile, and optionally restricts web access by IP.

Outbound Calls

Outbound Calls

Block External Caller ID

External CID Number:

External CID Name: Auto

Override CID Number:

Override CID Name:

Use this callerid when in a virtual extension

Emergency CID Number: Ignore empty

Area Code: Use default

Area Code Regex: Use default

Use Do Not Call lists:

Use Only Allow Call lists:

Routing Profile: Apply always

SMS Routing Profile:

Call Rate:

Outbound Calls section for PJSIP extension 101.

Use this section to control the caller ID, emergency caller ID, area-code handling, dialing filters, routing profile, and call rate used by outbound calls.

Block	Purpose
External caller ID	Sets or hides the caller ID number and name sent on outbound calls.
SMS and override caller ID	Controls SMS caller ID and optional SIP header based caller ID overrides.
Emergency caller ID	Defines the emergency number identity and whether empty emergency caller ID warnings are ignored.
Area code and caller ID regex	Applies prefixes and caller ID rewrite rules before routing.

Block	Purpose
Do Not and Only Allow lists	Applies tenant call-list restrictions to this extension.
Routing profile and call rate	Selects the routing profile, SMS routing profile, and client rate used by calls.

Inbound Calls

Inbound Calls

Block Inbound Caller ID Number

Block Inbound Caller ID Name

Inbound Calls section for PJSIP extension 101.

Use this section for inbound caller ID privacy controls.

Block	Purpose
Block inbound caller ID number	Prevents the caller ID number from being shown to this extension.
Block inbound caller ID name	Prevents the caller ID name from being shown to this extension.

Find me/Follow me Configuration

Find me/Follow me Configuration

FMFM Number: Active if checked

FMFM Condition:

FMFM Dial Method: Request confirmation

FMFM Caller ID:

FMFM Caller ID Num Prefix:

FMFM Caller ID Name Prefix:

FMFM Dial Timeout:

Find me/Follow me Configuration section for PJSIP extension 101.

Use this section to forward calls to an alternate number when find-me/follow-me is enabled.

Block	Purpose
FMFM number and status	Sets the external follow-me number and enables or disables the feature.
Condition and dial method	Limits follow-me behavior to a condition and selects how the alternate number is dialed.
Confirmation and messages	Requires answer confirmation and selects confirm/hold messages.
Caller ID and timing	Controls caller ID presentation, prefixes, delay, and dial timeout for the follow-me call.

Additional Destinations - Active if checked

Additional Destinations - Active if checked

Unconditional:

On No Answer:

On Extension Busy:

On Extension Offline:

On Condition:

Missed call notification to: Ignore on internals Ignore from queues

Additional Destinations - Active if checked section for PJSIP extension 101.

Use this section to define failover or conditional routing for calls that do not complete normally.

Block	Purpose
Unconditional	Always routes calls to the selected destination when enabled.
On no answer, busy, or offline	Routes calls when the extension does not answer, is busy, or is offline.
On condition	Routes calls to a selected destination when the chosen condition matches.
Missed call notification	Sends email notifications for missed calls, with options to ignore internal or queue-originated calls.

Note

Note

Reference ID:

Additional Info:

Save
Delete
Back

Note section for PJSIP extension 101.

Use this section for administrative classification and free-form notes, then save or delete the extension.

Block	Purpose
Branch and department	Classifies the extension for reporting and administration.
Reference ID and additional info	Stores external references and notes.
Save, delete, and back	Save applies changes, Delete removes the extension when allowed, and Back returns to the Extensions list.

Edit PJSIP Endpoint

Use this page to tune the PJSIP endpoint behavior behind a PJSIP extension. The settings here control media negotiation, NAT behavior, DTLS certificate handling, and transport-related media addressing. The screenshots use the **Canistracci Oil** demo PJSIP extension **101 Bella Stone**.

Opening and saving

Action	Description
Open	Open Configuration > Extensions, edit a PJSIP extension, then choose the Endpoint detail link.
Edit	Update the required fields and select Save. The form returns to the PJSIP extension after saving.
Back	Use Back to return to the PJSIP extension without applying changes.

Information

Information

Direct Media:

Rewrite Contact:

Connected Line Method:

DirectMedia Method:

Direct Media Glare Mitigation:

Disable direct media on NAT:

Force report:

Outbound Proxy:

RTP Symmetric:

T.38 UDPTL NAT:

RPID Immediate:

Useptime:

RTCP Mux:

Notify Ringing when Ring in Use:

Information section for PJSIP endpoint for extension 101.

Use this section for media path behavior, contact rewriting, connected-line signaling, NAT behavior, T.38 behavior, packetization, RTCP multiplexing, and ring-in-use notifications.

Field or option	Purpose
Direct Media	Determines whether media may flow directly between endpoints
Rewrite Contact	Allow Contact header to be rewritten with the source IP address-port
Connected Line Method	Connected line media type
DirectMedia Method	Direct Media method type
Direct Media Glare Mitigation	Mitigation of direct media re INVITE glare

Field or option	Purpose
Disable direct media on NAT	Disable direct media session refreshes when NAT obstructs the media session
Force rport	Force use of return port
Outbound Proxy	Proxy through which to send requests a full SIP URI must be provided
RTP Symmetric	Enforce that RTP must be symmetric
T.38 UDPTL NAT	Use the NATTED IP in the UDPTL for T.38
RPID Immediate	Immediately send connected line updates on unanswered incoming calls
Useptime	Use Endpoint's requested packetization interval
RTCP Mux	Controls whether RTP and RTCP can be multiplexed on the same transport flow.
Notify Ringing when Ring in Use	When an extension is in Ring in Use state, notifies the ringing

Security

Security

DTLS certificate file path:

DTLS CA certificate file path:

DTLS Auto generate certificate: ▼

Security section for PJSIP endpoint for extension 101.

Use this section for DTLS certificate settings used by secure media and WebRTC-style endpoint behavior.

Field or option	Purpose
DTLS certificate file path	Path to certificate file to present to peer
DTLS CA certificate file path	Path to certificate authority certificate
DTLS Auto generate certificate	Enable ephemeral DTLS certificate generation

Transport

Transport

External Media Address:

Transport section for PJSIP endpoint for extension 101.

Use this section for the external media address used in RTP handling.

Field or option	Purpose
External Media Address	External IP address to use in RTP handling

Edit PJSIP AOR

Use this page to tune the PJSIP Address of Record for a PJSIP extension. These settings control registration contact limits, contact expiration timing, mailbox subscription hints, and Path support. The screenshots use the **Canistracci Oil** demo PJSIP extension **101 Bella Stone**.

Opening and saving

Action	Description
Open	Open Configuration > Extensions, edit a PJSIP extension, then choose the AOR detail link.
Edit	Update the required fields and select Save. The form returns to the PJSIP extension after saving.
Back	Use Back to return to the PJSIP extension without applying changes.

Information

Information

Default Expiration:

Minimum Expiration:

Maximum Expiration:

Mailboxes:

Max Contacts:

Remove Existing:

Remove Unavailable:

Support Path:

Information section for PJSIP AOR for extension 101.

Use this section for contact expiration limits, mailbox subscriptions, maximum registered contacts, registration contact cleanup, and Path support.

Field or option	Purpose
Default Expiration	Default expiration time in seconds for contacts that are dynamically bound to an AoR
Minimum Expiration	Minimum expiration time in seconds for contacts that are dynamically bound to an AoR
Maximum Expiration	Maximum expiration time in seconds for contacts that are dynamically bound to an AoR
Mailboxes	Allow subscriptions for the specified mailbox(es)
Max Contacts	Maximum number of contacts that can bind to an AoR
Remove Existing	On receiving a new registration to the AoR should it remove enough existing contacts not added or updated by the registration to satisfy Max Contacts? Any removed contacts will expire the soonest.
Remove Unavailable	On receiving a new registration to the AoR should it remove unavailable contacts not added or updated by the registration to satisfy Max Contacts?
Support Path	Enables Path support for REGISTER requests and Route support for other requests.

Create and Edit Custom Extensions

Use this page to create or edit an extension that routes to a custom dial target or external endpoint behavior. The form works against the tenant currently selected in the top bar.

The screenshots use the **Canistracci Oil** demo extension **106 Gia Parker**. Creating and editing use the same form: a new extension opens with tenant defaults, while an existing extension opens with saved values.

Creating and editing

Action	Description
Create	Open Configuration > Extensions and select Custom Extension. Fill the required fields and select Save.
Edit	Open an existing extension from the Extensions list. Update the required fields and select Save.
Delete	When delete permission is available, open the extension and select Delete. Confirm only after checking routing, phones, queues, and other references.

Information

Information
Show All

Number:

Name:

Description:

Call Group:

1
▼
🗑️

Pickup Groups:

1
▼
🗑️

Do Not Disturb (DND)

Inbound Dial Timeout:

Information section for custom extension 106.

Use this section for the custom extension identity, call/pickup groups, DND state, and inbound timeout.

Block	Purpose
Number, name, and description	Defines the internal extension number, display name, and administrator description.
Call and pickup groups	Controls pickup behavior and group membership for incoming calls.
DND and inbound timeout	Controls whether the custom extension is marked DND and how long incoming calls are attempted.

Call Settings

Call Settings

Fax:

Volume TX level:

Volume RX level:

Music on hold:

Language:

Include in Dial By Name directory:

Dial By Name recording:

Include in Phone Books:

Call waiting:

Autoanswer:

Call Settings section for custom extension 106.

Use this section for media and listing behavior applied to the custom extension.

Block	Purpose
Fax and volume	Controls fax behavior and transmit/receive volume adjustments.
Music on hold and language	Selects hold media and prompt language.
Directories and phone books	Controls inclusion in dial-by-name and phone book generated lists.
Call waiting and autoanswer	Controls second-call handling and autoanswer behavior when supported by the target.

Outbound Recording

Outbound Recording

Always Record:

Email recording to:

Transcript recorded calls:

Summarize recorded calls:

Sentiment analysis for recorded calls:

Outbound Recording section for custom extension 106.

Use this section for outbound recording and optional recording processing.

Block	Purpose
Always record and email recording	Controls outbound recording and optional delivery of recordings by email.
Transcript, summary, sentiment	Enables transcript, summary, and sentiment processing when available for the tenant.

Security

Security

Outbound number:

On internal call use CallerID:

Authentication type:

Inbound Allowed Provider:

Authentication CallerID:

Abuse Detection:

Apply call cost limits:

Alert email:

Outbound Destinations:

Lock PIN: Locked

Max outbound call duration:

Working hours restrictions

Security section for custom extension 106.

Use this section to define the custom dial target and restrict how the custom extension can be used.

Block	Purpose
Outbound number	Defines the number or dial target used to reach the custom extension.
Caller ID and authentication	Controls caller ID use on internal calls and the authentication method expected for inbound matching.
Allowed providers and Teams ID	Restricts inbound providers and stores a Teams extension ID when Teams status integration is used.

Block	Purpose
Abuse, cost, and destination limits	Applies abuse detection, call cost limits, outbound destination filters, lock PIN, maximum duration, and working-hours restrictions.

Web User Panel

Web User Panel

Allow Web User Panel Access

Web User:

Password: LDAP

Never expire

Force password change at login

Lock password, user can't change it

Use Two Factors Authentication (2FA): ▼

Email: Send✉

User Profile: ▼

Web User Panel section for custom extension 106.

Use this section to enable web access for the custom extension user.

Block	Purpose
User panel toggle	Enables or disables web access for this extension.
Web user and password	Sets the login identity and password.
LDAP, 2FA, email, and profile	Controls external authentication, two-factor options, notification email, and user profile.

Outbound Calls

Outbound Calls

Block External Caller ID

External CID Number: ▼ Edit

External CID Name: Auto

Emergency CID Number: ▼ Ignore empty

Area Code:

Area Code Regex:

Use Do Not Call lists: ▼

Use Only Allow Call lists: ▼

Routing Profile: ▼ Apply always

Call Rate: ▼

Outbound Calls section for custom extension 106.

Use this section to control outbound caller ID, emergency caller ID, prefixes, call lists, routing profile, and call rate.

Block	Purpose
External caller ID	Sets caller ID number and name used by outbound calls.
Caller ID regex and emergency caller ID	Applies rewrite rules and emergency identity settings.
Area code and call lists	Applies area-code prefixes and Do Not or Only Allow list restrictions.
Routing profile and call rate	Selects the routing profile and call rate used by outbound calls.

Find me/Follow me Configuration

Find me/Follow me Configuration

FMFM Number: Active if checked

FMFM Dial Method: Normal Request confirmation

FMFM Caller ID:

FMFM Caller ID Num Prefix:

FMFM Caller ID Name Prefix:

FMFM Dial Timeout:

Find me/Follow me Configuration section for custom extension 106.

Use this section to send calls to a follow-me destination under controlled conditions.

Block	Purpose
FMFM number and dial method	Sets the follow-me target and how it is dialed.
Confirmation and caller ID	Controls answer confirmation and caller ID behavior.
Delay and timeout	Controls how long to wait before and during follow-me dialing.

Additional Destinations - Active if checked

Additional Destinations - Active if checked

Unconditional:

On No Answer:

On Extension Busy:

On Extension Offline:

On Condition:

Missed call notification to:

Ignore on internals
 Ignore from queues

Additional Destinations - Active if checked section for custom extension 106.

Use this section to route calls that are not answered, are busy, are offline, or meet a condition.

Block	Purpose
Failover destinations	Selects unconditional, no-answer, busy, offline, and conditional destinations.
Missed call notifications	Sends notifications and optionally ignores internal or queue-originated missed calls.

Note

Note

Reference ID:

Additional Info:

Save
Delete
Back

Note section for custom extension 106.

Use this section for administrative metadata and final actions.

Block	Purpose
Branch, department, reference ID	Classifies the extension and stores external references.
Additional info	Stores free-form notes.
Save, delete, and back	Save applies changes, Delete removes the extension when allowed, and Back returns to the Extensions list.

Create and Edit Virtual Extensions

Use this page to create or edit a virtual extension that groups real extensions behind a single internal number. The form works against the tenant currently selected in the top bar.

The screenshots use the **Canistracci Oil** demo extension **108 Ivy Morgan**. Creating and editing use the same form: a new extension opens with tenant defaults, while an existing extension opens with saved values.

Creating and editing

Action	Description
Create	Open Configuration > Extensions and select Virtual Extension. Fill the required fields and select Save.
Edit	Open an existing extension from the Extensions list. Update the required fields and select Save.
Delete	When delete permission is available, open the extension and select Delete. Confirm only after checking routing, phones, queues, and other references.

Information

Information
Show All

Number:

Name:

Description:

Security PIN:

Extensions:

- 100 - Alex Reed
✕
- 101 - Bella Stone
✕
- 106 - Gia Parker
✕

Call Group:

- 1
✕

Pickup Groups:

- 1
✕

Voicemail MWI:

Do Not Disturb (DND)

Inbound Dial Timeout:

Information section for virtual extension 108.

Use this section for the virtual extension identity, member extensions, security PIN, caller ID, groups, voicemail MWI, DND state, and inbound timeout.

Block	Purpose
Number, name, and description	Defines the internal virtual extension number, display name, and administrator description.
Security PIN	Requests a PIN when an extension tries to join the virtual extension.

Block	Purpose
Internal caller ID	Sets the caller ID shown for internal calls, or keeps it automatic.
Extensions	Selects the real extensions that belong to the virtual extension.
Groups, voicemail, DND, timeout	Controls call/pickup/spy groups, voicemail MWI, DND state, and inbound ring timeout.

Call Settings

Call Settings

Volume TX level:

Volume RX level:

Music on hold: Tenant Default ▼

Include in Dial By Name directory: No ▼

Dial By Name recording: Default message ▼

Include in Phone Books: No ▼

Call waiting: Tenant Default ▼

Parkinglot: Use Default ▼

Call Settings section for virtual extension 108.

Use this section for virtual extension call behavior and listing options.

Block	Purpose
Volume and music on hold	Adjusts audio levels and selects hold media.
Directories and phone books	Controls inclusion in dial-by-name and phone book generated lists.

Block	Purpose
Call waiting and parking lot	Controls server-side call waiting behavior and parking lot selection.

Outbound Recording

Outbound Recording

Always Record:

Email recording to:

Transcript recorded calls:

Summarize recorded calls:

Sentiment analysis for recorded calls:

Outbound Recording section for virtual extension 108.

Use this section for outbound recording and optional recording processing.

Block	Purpose
Always record and email recording	Controls outbound recording and optional delivery of recordings by email.
Transcript, summary, sentiment	Enables transcript, summary, and sentiment processing when available for the tenant.

Security

Security

Outbound Destinations:

Lock PIN: Locked

Max outbound call duration:

Working hours restrictions:

Security section for virtual extension 108.

Use this section to limit outbound usage from the virtual extension.

Block	Purpose
Outbound destinations	Allows all calls, blocks calls, or applies a destination regex filter.
Lock PIN and max duration	Allows the virtual extension to be locked and caps outbound call duration.
Working hours	Restricts outbound calling to the selected weekday condition.

Web User Panel

Web User Panel

Allow Web User Panel Access

Web User:

Password: LDAP

Email:

User Profile:

Web User Panel section for virtual extension 108.

Use this section to enable web access for the virtual extension user.

Block	Purpose
User panel toggle	Enables or disables web access for this extension.
Web user and password	Sets the login identity and password.
LDAP, 2FA, email, and profile	Controls external authentication, two-factor options, notification email, and user profile.

Outbound Calls

Outbound Calls

Block External Caller ID

External CID Number: ▼ Edit

External CID Name:

Use real extension callerid

Area Code: Use default

Area Code Regex: Use default

Use Do Not Call lists: ▼

Use Only Allow Call lists: ▼

Routing Profile: ▼

Outbound Calls section for virtual extension 108.

Use this section to control caller ID, area-code handling, dialing filters, and routing for calls placed from the virtual extension.

Block	Purpose
External caller ID	Sets caller ID number and name used by outbound calls.
Use this caller ID in a virtual extension	Controls whether member extensions use this caller ID when calls originate through the virtual extension.

Block	Purpose
Caller ID regex and area code	Applies rewrite rules and number prefixes before routing.
Call lists and routing profile	Applies Do Not or Only Allow lists and selects the routing profile.

Find me/Follow me Configuration

Find me/Follow me Configuration

FMFM Number: Active if checked

FMFM Condition: ▼

FMFM Dial Method: ▼ Request confirmation

FMFM Caller ID: ▼

FMFM Caller ID Num Prefix:

FMFM Caller ID Name Prefix:

FMFM Dial Timeout:

Find me/Follow me Configuration section for virtual extension 108.

Use this section to send calls to a follow-me destination under controlled conditions.

Block	Purpose
FMFM number and condition	Sets the follow-me target and optional condition.
Dial method and confirmation	Controls how the follow-me call is dialed and whether answer confirmation is required.
Caller ID and timing	Controls caller ID presentation, delay, and timeout.

Additional Destinations - Active if checked

Additional Destinations - Active if checked

Unconditional: Action to take

On No Answer: Action to take

On Extension Busy: Action to take

On Extension Offline: Action to take

On Condition: Choose condition to check

Action to take

Missed call notification to:

Ignore on internals Ignore from queues

Additional Destinations - Active if checked section for virtual extension 108.

Use this section to route calls that do not complete normally or that meet a condition.

Block	Purpose
Failover destinations	Selects unconditional, no-answer, busy, offline, and conditional destinations.
Missed call notifications	Sends notifications and optionally ignores internal or queue-originated missed calls.

Note

Note

Reference ID:

Additional Info:

Save
Delete
Back

Note section for virtual extension 108.

Use this section for administrative metadata and final actions.

Block	Purpose
Branch, department, reference ID	Classifies the extension and stores external references.
Additional info	Stores free-form notes.
Save, delete, and back	Save applies changes, Delete removes the extension when allowed, and Back returns to the Extensions list.