



RELEASE NOTES

PBXware 8.1.0



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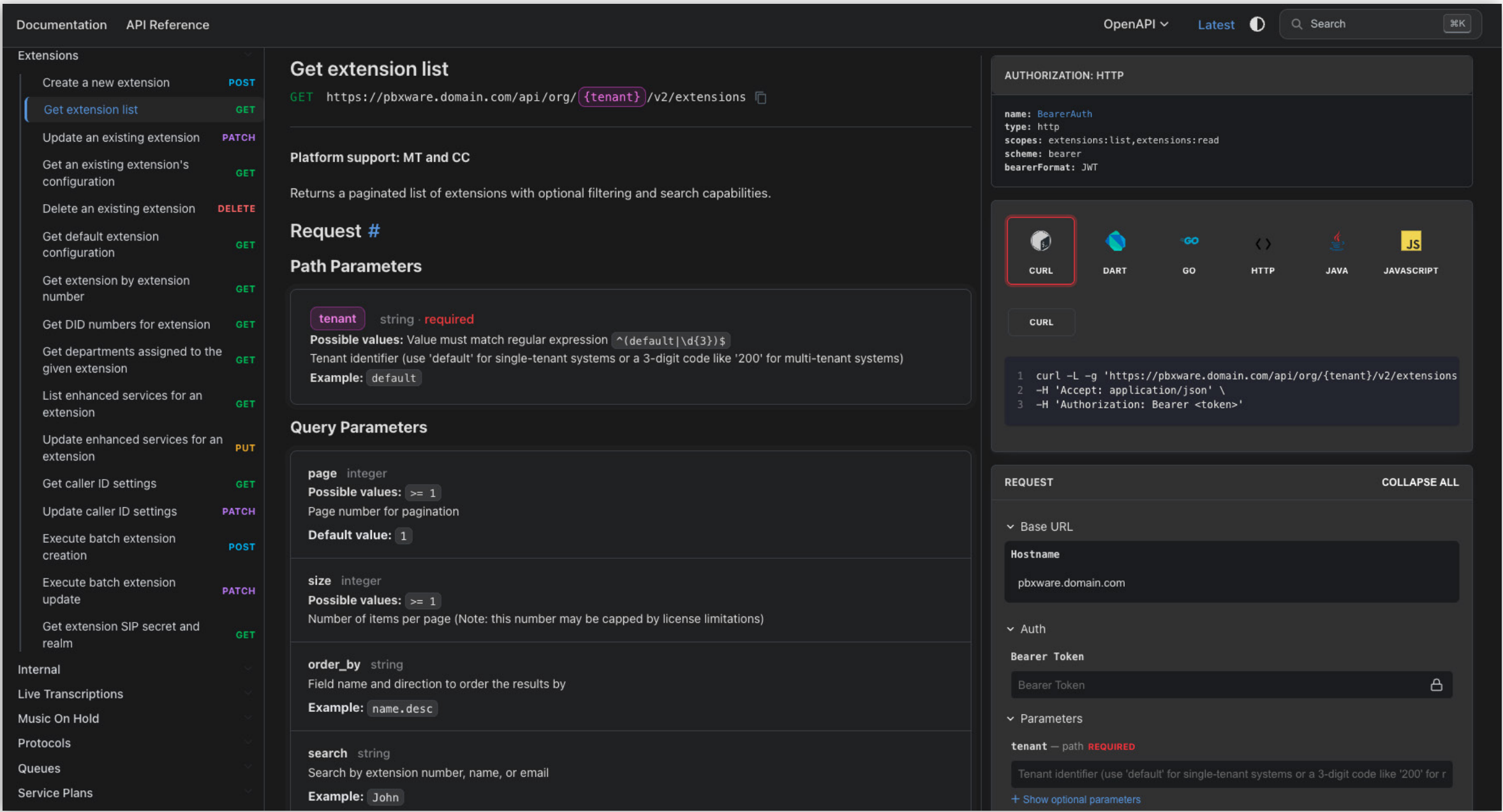
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Features

Developer Portal

Launching alongside v8.1, the new PBXware Developer Portal (developers.bicomsystems.com) gives developers and partners self-service access to build on the PBXware platform. It includes documentation for the full REST API, real-time Event Publisher (call events), official SDK, ARI guides for custom call control, a CRM connector, and AI integration guides covering voice intelligence and live transcription. The portal is designed to get developers to their first API call in minutes, with step-by-step onboarding covering credential generation, scope configuration, and production-readiness guidance. Live code samples and an interactive API reference make it easy to explore endpoints and integrate PBXware into CRMs, other external platforms, and custom telephony applications.



Support Asterisk 18 and 22 in parallel with optional activation

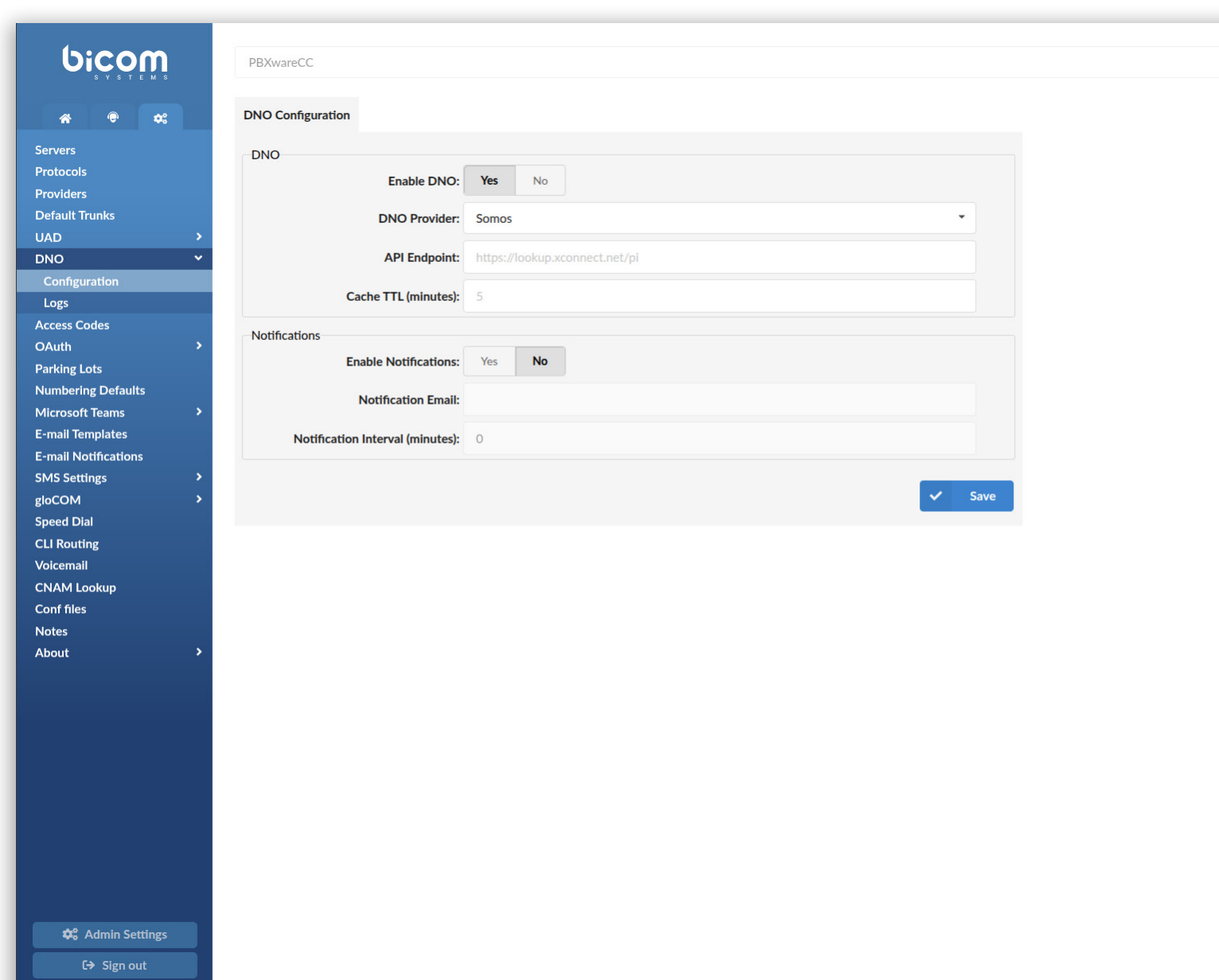
PBXware 8.1 can now run either Asterisk 18 or Asterisk 22, with Asterisk 18 remaining the default stable version. Asterisk 22 is offered as an option so the new release can be validated in real environments before wider adoption. The active version is set per instance from the command line with **`sh/pbxware set_asterisk_version 22`** (or **`18`** to go back). The selected version is persisted in **`/etc/pbxware/asterisk_version.conf`**, and a restart is required for the change to take effect.

This dual-version setup is a transitional step, not a permanent arrangement. Asterisk 22 stays Beta in 8.1, and depending on testing results, it is expected to become the default in a later release, at which point the parallel-version switching mechanism will be phased out.

Do-Not-Originate (DNO)

PBXware 8.1.0 introduces Do-Not-Originate (DNO) validation for outbound calls to help organizations comply with caller ID regulations and combat caller ID spoofing. Outbound calls are validated against supported DNO providers, and calls using caller IDs found on a DNO list can be blocked before they are placed. This helps service providers meet regulatory expectations, including FCC DNO requirements in the United States and Ofcom Calling Line Identification (CLI) guidance in the United Kingdom, which expects providers to prevent calls using numbers that should not originate outbound traffic.

DNO Configuration



The screenshot displays the PBXwareCC web interface for DNO Configuration. On the left is a dark blue sidebar with the 'bicom' logo and a menu including Servers, Protocols, Providers, Default Trunks, UAD, DNO (selected), Configuration, Logs, Access Codes, OAuth, Parking Lots, Numbering Defaults, Microsoft Teams, E-mail Templates, E-mail Notifications, SMS Settings, gloCOM, Speed Dial, CLI Routing, Voicemail, CNAM Lookup, Conf files, Notes, and About. At the bottom of the sidebar are 'Admin Settings' and 'Sign out' buttons. The main content area is titled 'PBXwareCC' and 'DNO Configuration'. It contains two sections: 'DNO' and 'Notifications'. The 'DNO' section has 'Enable DNO' (Yes/No toggle), 'DNO Provider' (dropdown menu showing 'Somos'), 'API Endpoint' (text field with 'https://lookup.xconnect.net/pl'), and 'Cache TTL (minutes)' (text field with '5'). The 'Notifications' section has 'Enable Notifications' (Yes/No toggle), 'Notification Email' (text field), and 'Notification Interval (minutes)' (text field with '0'). A blue 'Save' button with a checkmark is at the bottom right of the configuration area.

DNO configuration is available under Settings → DNO → Configuration. Supported providers are Somos and Custom. Somos uses IP whitelisting authentication, while the Custom provider supports a configurable API endpoint and Bearer token authentication.

The Custom provider is designed to integrate with third-party DNO services via middleware. The middleware connects PBXware to external services or directly queries databases, enabling organizations to integrate with virtually any DNO data source.

DNO lookup caching is supported and configurable through cache TTL. Calls are only blocked when the caller ID is confirmed on the DNO list. If the lookup result is unavailable, the number cannot be evaluated, or the provider lookup fails, the call is allowed to continue.

DNO Email Notifications

PBXware 8.1.0 adds email notifications for blocked DNO calls and DNO provider or service errors. Notifications can be enabled under Settings → DNO → Configuration → Notifications, where administrators can configure one or more comma-separated recipient addresses.

A screenshot of the 'Notifications' configuration window in PBXware. The window has a title bar 'Notifications'. Inside, there are three fields: 'Enable Notifications:' with 'Yes' and 'No' radio buttons, 'Notification Email:' with a text input field, and 'Notification Interval (minutes):' with a text input field containing '0'. A blue 'Save' button with a checkmark icon is located at the bottom right.

Separate email templates are available for blocked calls and API errors under Settings → Email Templates. Templates support DNO-specific variables such as caller ID, call ID, cache status, provider error details, trace ID, system information, tenant information, extension, and event date.

To prevent repeated notifications, the Email Timeout (minutes) option controls how frequently notifications of the same type are sent. The default value is 60 minutes. Setting the value to 0 sends an email for every matching event.

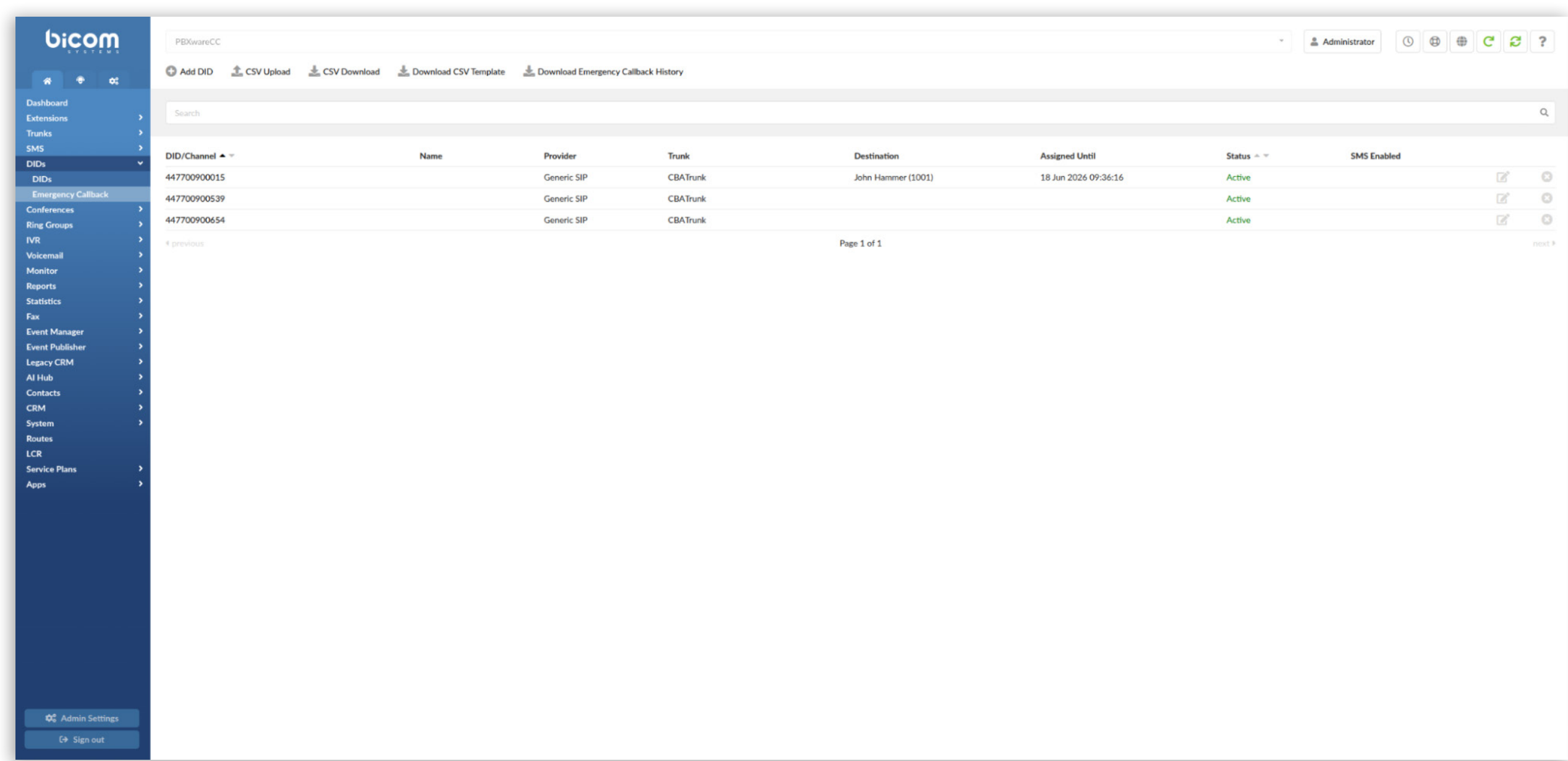
When notifications are disabled, no notification emails are sent and existing DNO call-processing behavior remains unchanged. Notification events are also written to the system logs.

DNO Configuration API

PBXware 8.1.0 introduces API v2 support for DNO configuration, allowing all DNO configuration options to be retrieved and updated through the API.

DNO Logs

PBXware 8.1.0 introduces DNO logs under Settings → DNO → Logs. The logs include caller ID, date/time, call ID, provider, cache status, lookup status, and error details for failed provider checks. The logs preserve the last 10,000 entries.



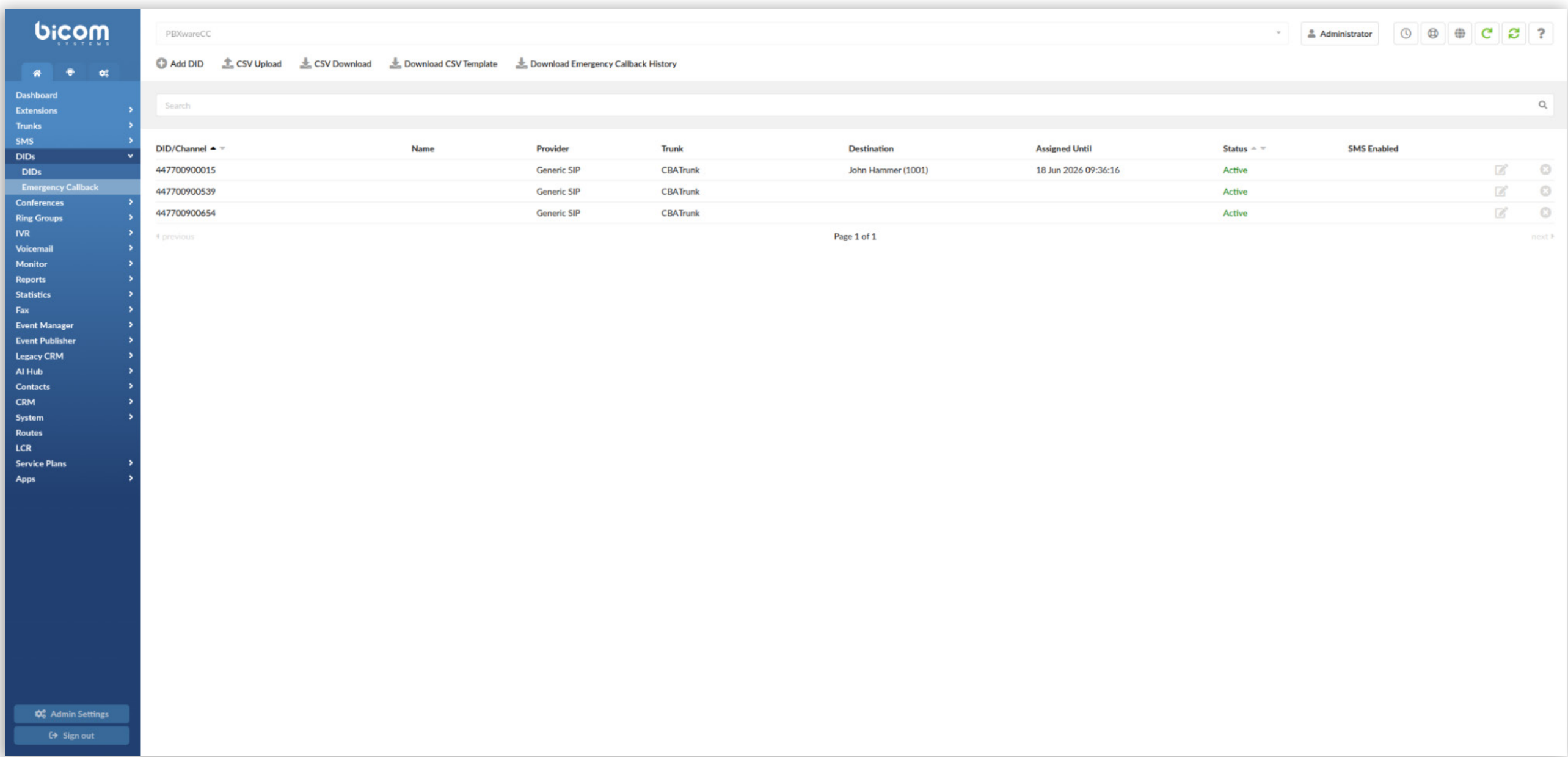
DNO Logs API

DNO logs can also be retrieved through API v2, enabling API-based access to DNO log data.

Emergency Callback Number (ECBN)

PBXware 8.1.0 introduces Emergency Callback Number support for emergency calls. The feature allows administrators to define a system-wide pool of DIDs that can be temporarily assigned to extensions that do not already have a dedicated DID when placing an emergency call.

A new DID → Emergency Callback page has been added, allowing administrators to add, edit, import, export, and review ECBNs. The page also shows current ECBN assignments, including the extension and the time until which the number remains assigned.



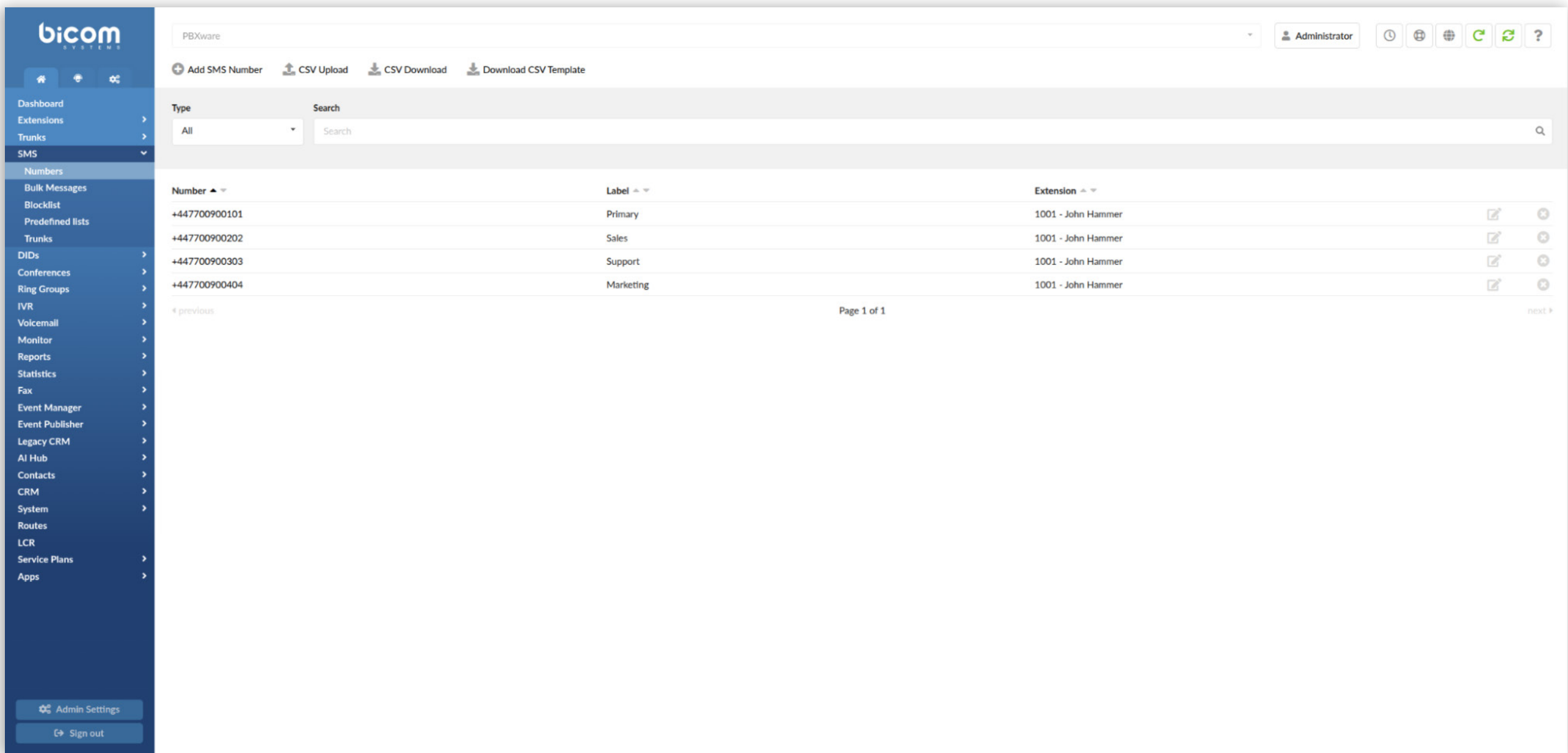
When an extension places an emergency call, PBXware determines the callback number in the following order: if the extension already has a regular DID, that DID is used; if the extension already has an active ECBN assignment, the existing ECBN is reused and its assignment time is refreshed; otherwise, a free ECBN from the pool is assigned to the extension. The assigned number is then used in the SIP Contact header for emergency callback purposes.

The ECBN assignment duration is configurable through the new Emergency Callback Number Assignment Duration (Hours) option on the server/main tenant settings page. The default value is 24 hours. After the configured duration has expired, the ECBN is considered available for reuse.

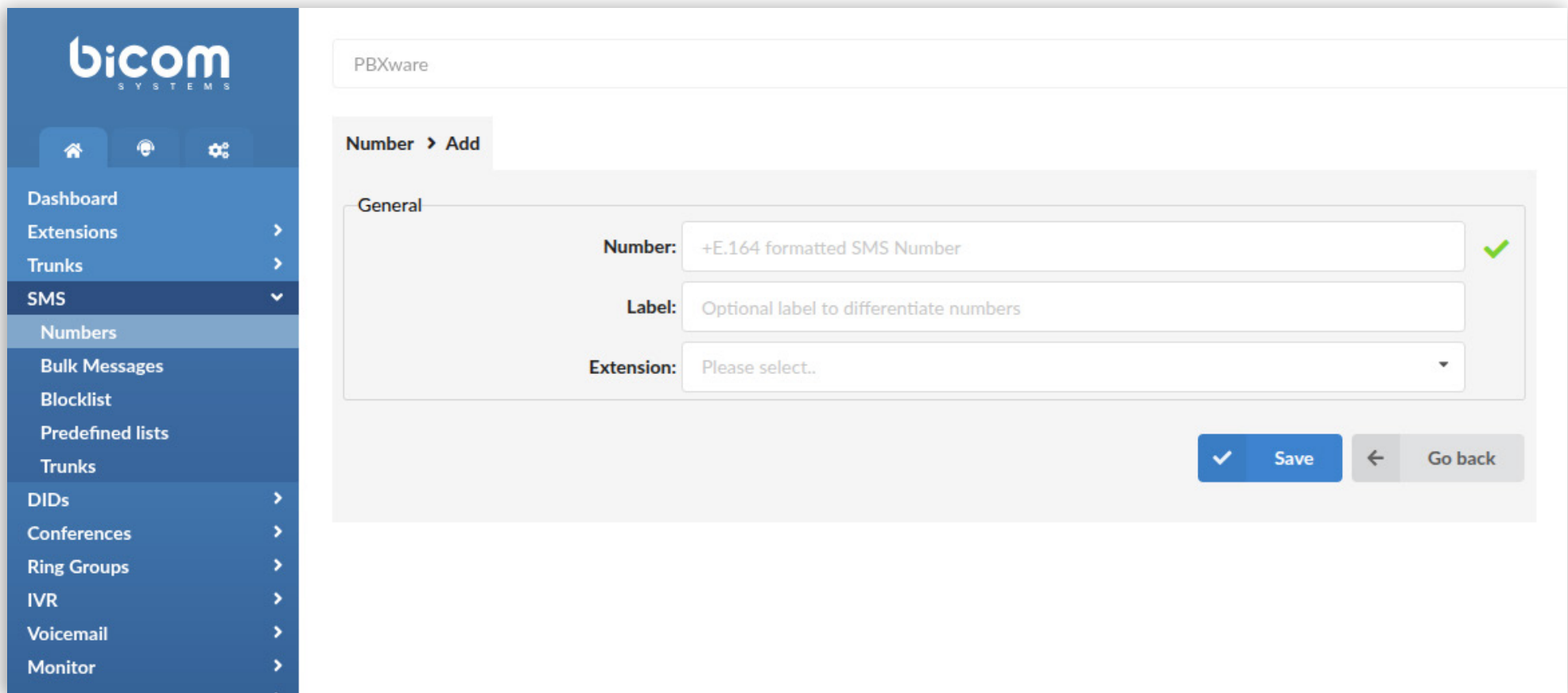
PBXware also adds ECBN assignment logging. Assignment events are written to the system logs and stored in the ECBN log table, including successful assignments, failed assignments, and cases where a regular DID was used. The ECBN history can be downloaded from the DID → Emergency Callback page.

Multiple SMS Number Assignment per Extension

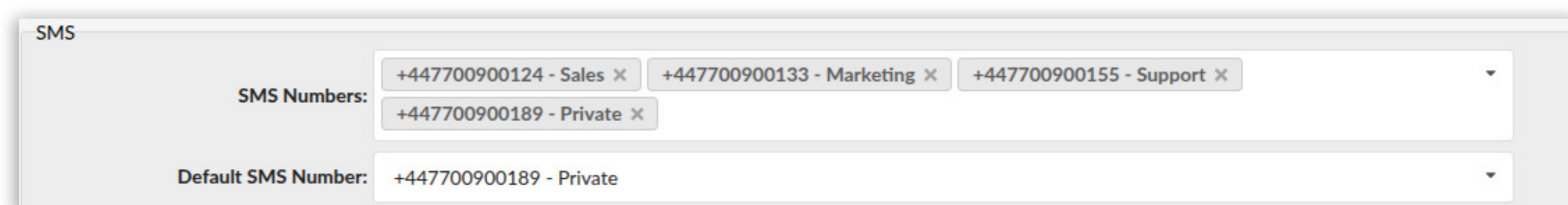
PBXware now supports assigning multiple SMS-enabled DIDs to a single extension, making it easier to manage SMS communication across different business purposes while maintaining centralized user management.



Administrators can assign multiple SMS numbers to an extension, define an optional label for each number (such as Primary, Sales, or Marketing), and select a default SMS number that will be used when users initiate new SMS conversations. If no default is configured, the first assigned SMS number is automatically selected.



To simplify administration, the Extension configuration page now includes an SMS Numbers management section where assigned numbers can be viewed, added, removed, and managed directly. Each assigned number provides quick access to its configuration, while the Default SMS Number can be changed without leaving the extension settings. The default SMS number can be selected by both the administrator and the end user.



The SMS Numbers management page has also been enhanced with extension-based search, allowing administrators to quickly identify all SMS numbers assigned to a particular extension. Additionally, the CSV import format has been updated to support SMS number labels.

This enhancement provides the backend foundation for multi-number SMS support across the gloCOM NEXT and gloCOM GO while simplifying SMS number administration in PBXware.

The gloCOM classic (Qt) application partially supports this feature. Administrators can still assign multiple SMS numbers to one extension, but users can reply only from the number that received the message. Starting a new SMS conversation is possible only from the default SMS number.

Assigning labels to SMS numbers in PBXware is highly recommended to enable the full functionality of this feature across applications.

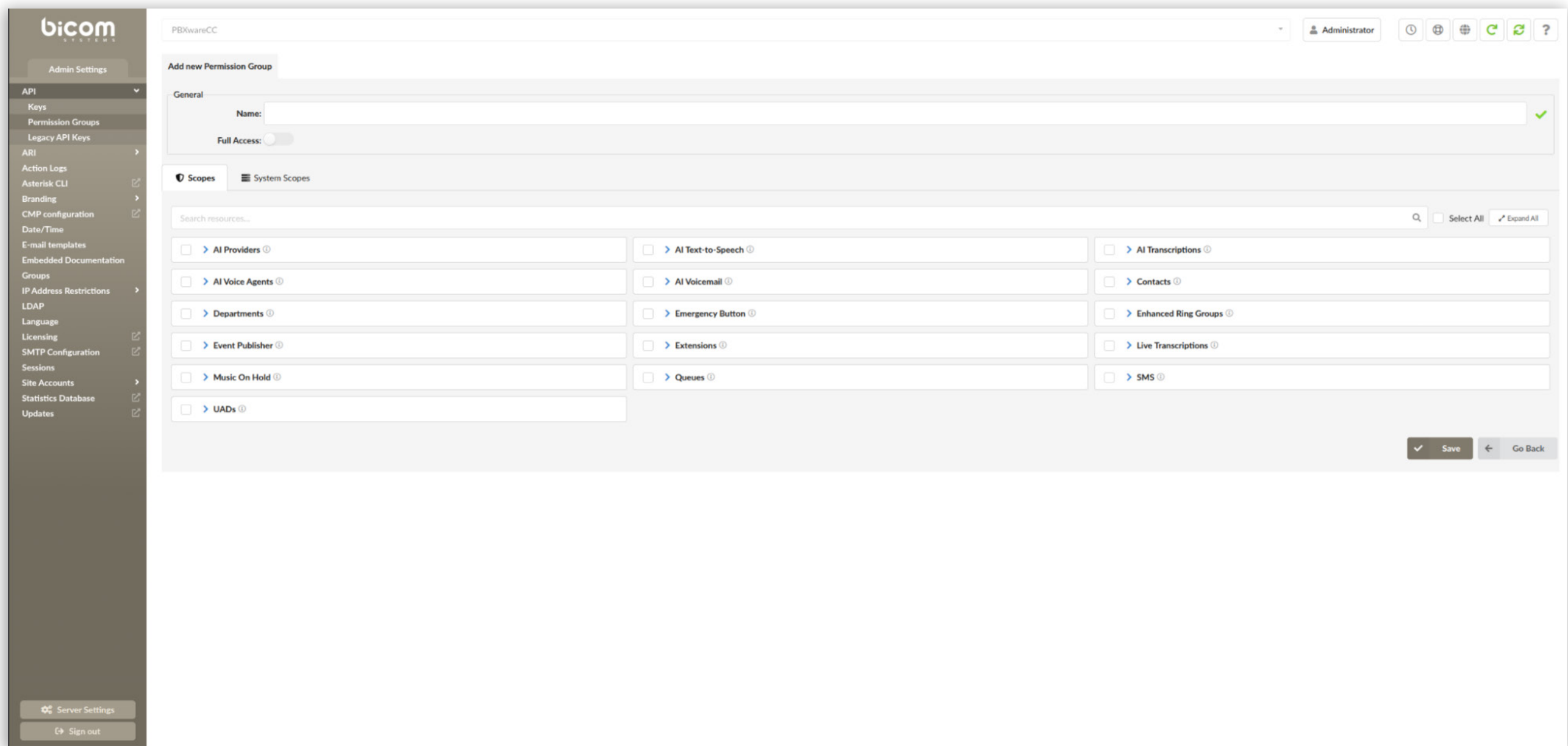
API v2 Key Management

PBXware 8.1.0 introduces a dedicated API v2 management section under Admin Settings → API. The API section now includes Keys, Permission Groups, and Legacy API Keys.

Keys and Permission Groups are used for API v2 key management. Legacy API Keys is the renamed page for existing API v1 keys, and API v1 keys continue to work the same as before.

Permission Groups

PBXware 8.1.0 introduces Permission Groups for API keys, allowing administrators to control access to API resources using granular permission sets.



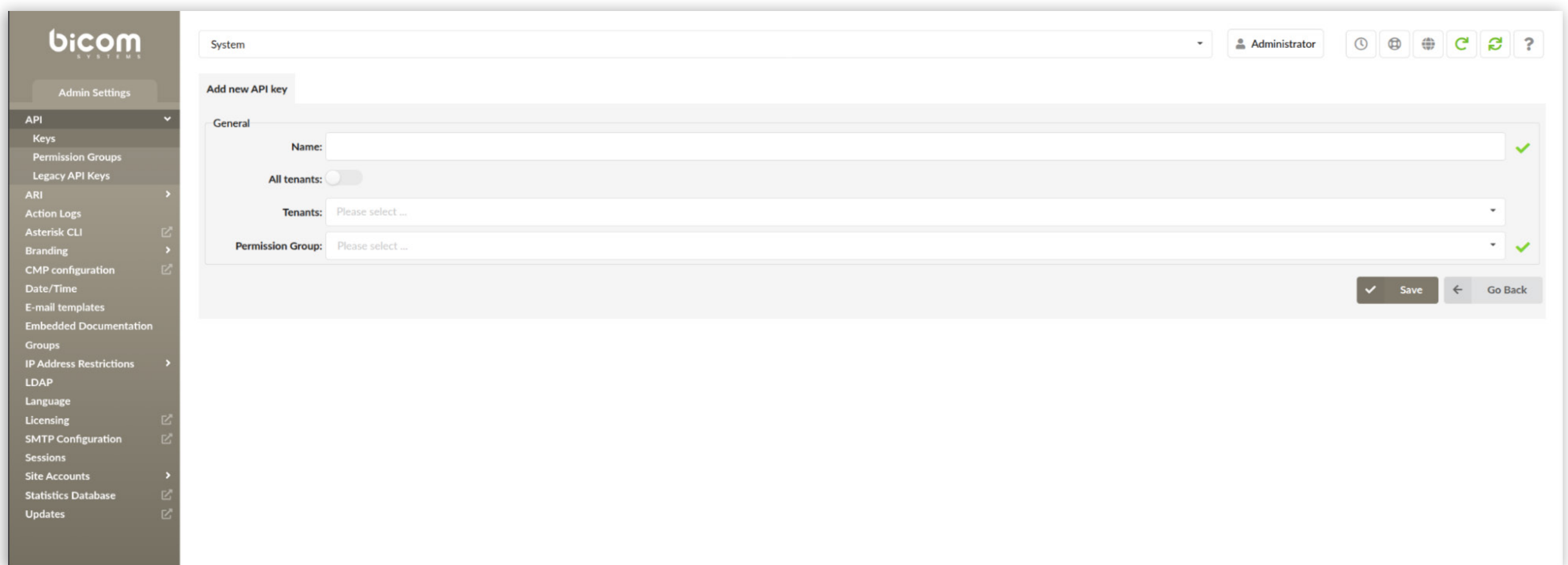
Permission Groups are organized into two scopes:

- **Tenant Scopes** – Permissions for tenant-level resources, such as Extensions, Departments, Contacts, Service Plans, SMS, and other tenant-managed entities.
- **System Scopes** – Permissions for system-level resources and administrative functions.

Within each scope, administrators can grant permissions for individual operations, including create, read, update, delete, and list, or assign full access where appropriate. API keys can then be associated with the required Permission Group, ensuring each integration has access only to the resources and operations it requires.

API Keys

After a permission group has been configured, administrators can create an API key and assign the appropriate permission group to it. Optional settings include a description, expiration date, and inactivity suspension period.



When a key is generated, its complete value is displayed only once in a confirmation window with an option to copy it to the clipboard. After the window is closed, the full key value cannot be viewed or retrieved again. Only a masked representation of the key is shown afterward.

The Keys page displays the key name, assigned permission group, full-access status, creation and update times, and the time the key was last used. Administrators can update the key configuration or assigned permission group and delete keys that are no longer required, but the generated key value itself cannot be changed or retrieved.

Key names must be unique, and a permission group must be assigned before a key can be generated. Key creation is also recorded in the audit log together with the creator, access permissions, and tenant limitations.

API v2 coverage

This release introduces PBXware API v2, our modern REST-based API platform designed to provide partners and developers with secure, consistent, and comprehensive programmatic access to PBXware.

As part of this initiative, we have adopted an API-first development approach. New PBXware features are designed and implemented at the API layer first, ensuring they are immediately available for automation and third-party integrations rather than being added retrospectively.

API v2 provides a standardized and consistent interface across PBXware, making it easier to integrate external systems, automate administrative tasks, and build custom solutions. The API will continue to expand with new modules and endpoints in future releases.

For a complete list of available endpoints, authentication details, request and response schemas, and usage examples, visit our [Developer Portal](#).

SMS API and Event Publisher

Event publisher extended with SMS functionality and SMS event publishing.

The new functionality includes:

- Sending SMS messages through the API
- SMS sent event notifications
- SMS received event notifications
- Bulk Messaging event notifications

These events can be consumed by external systems for integrations, reporting, and workflow automation.

Archiving - OAuth Service Integration

The Archiving module now integrates with the centralized OAuth Service for Google and Microsoft authentication, replacing the previous internal OAuth implementation.

Improvements

- Support for customer-managed OAuth applications for both Google and Microsoft providers.
- OAuth configuration now uses provider-specific Storage scopes.
- Added connection validation through a built-in Test option before running archiving tasks.
- Unified authentication flow aligned with the OAuth Service architecture.

Backward Compatibility

- Existing archiving configurations remain fully functional after upgrade.
- Legacy configurations display a deprecation notice in the UI but can still be edited and saved.
- Migration becomes permanent once a tenant saves a new OAuth-based configuration.

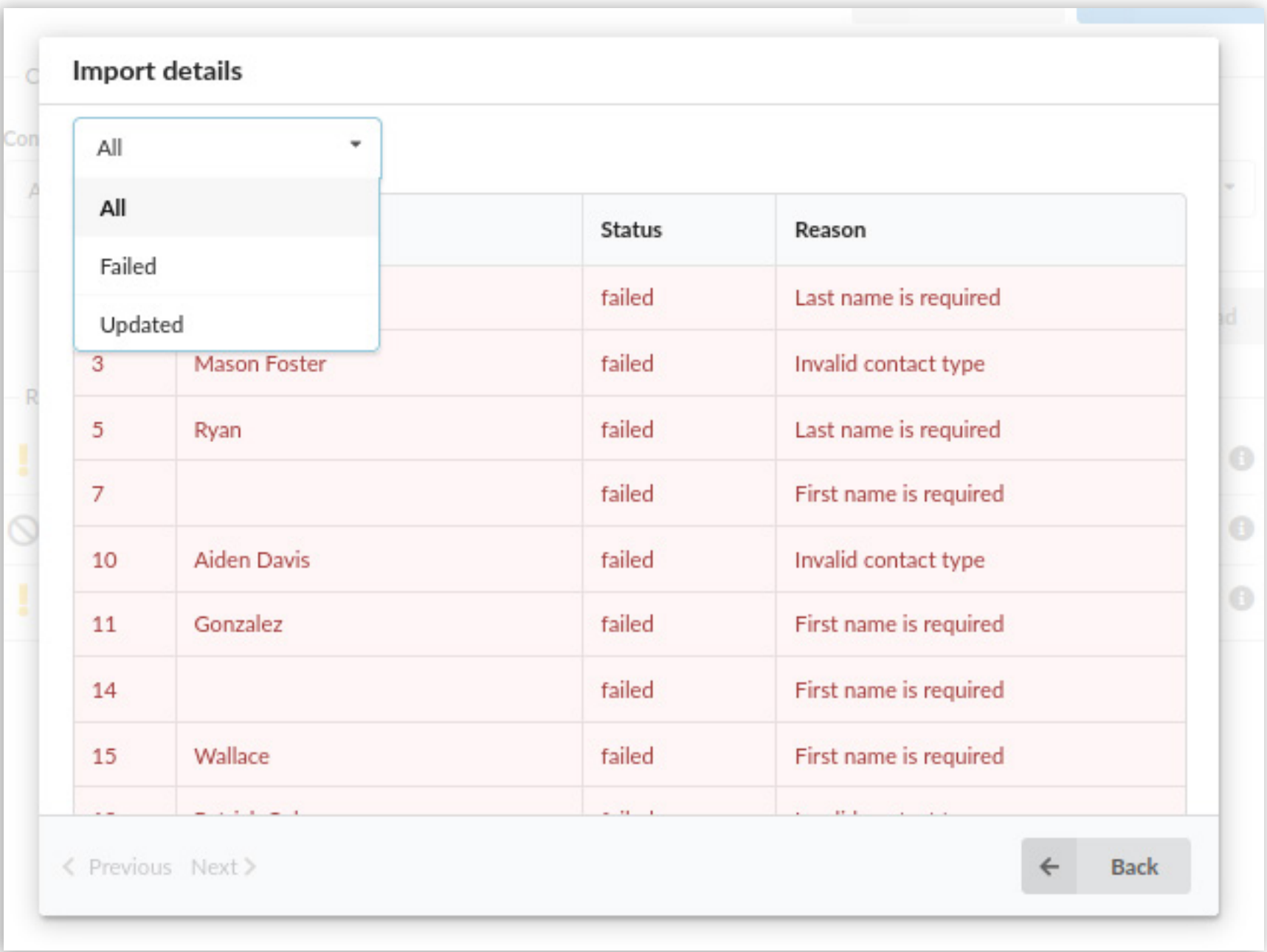
- Tenants without existing archiving setups will use the new OAuth-based configuration flow by default.

Contact Module Enhancements

The Contacts module has been enhanced with several usability and management improvements.

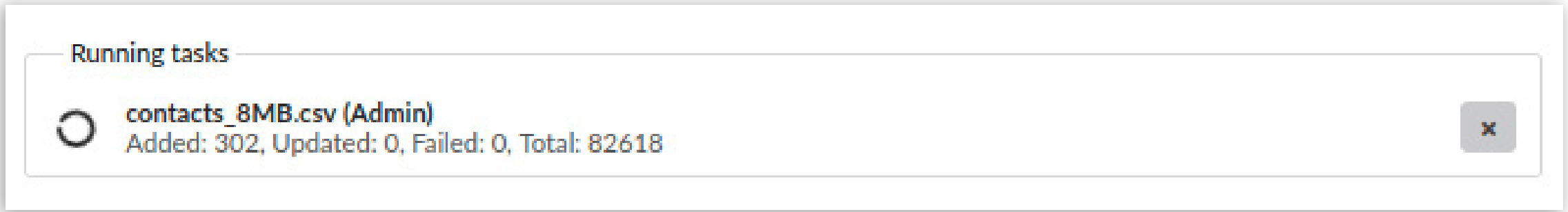
Key enhancements include:

- Improved visibility of failed CSV imports with detailed error reporting.
- Increased the maximum CSV import file size to **25 MB**.
- Added the ability to cancel active CSV import jobs.
- Added a loading indicator to the CSV upload button to prevent duplicate submissions.
- Updated contact history to display **Admin** instead of **System** for changes made through PBXware users.
- Added sorting by contact name.
- Added support for bulk deletion of contacts.
- Preserved the current contact list view after deleting contacts.



Cancel Option for Active CSV Import Jobs

Users can now cancel CSV import jobs that are currently in progress directly from the PBX web interface. Previously, once a CSV import was started, there was no way to stop it through the GUI for example, if a wrong file was uploaded or an incorrect department was selected. The new Cancel option brings the PBX web interface in line with existing gloCOM functionality.



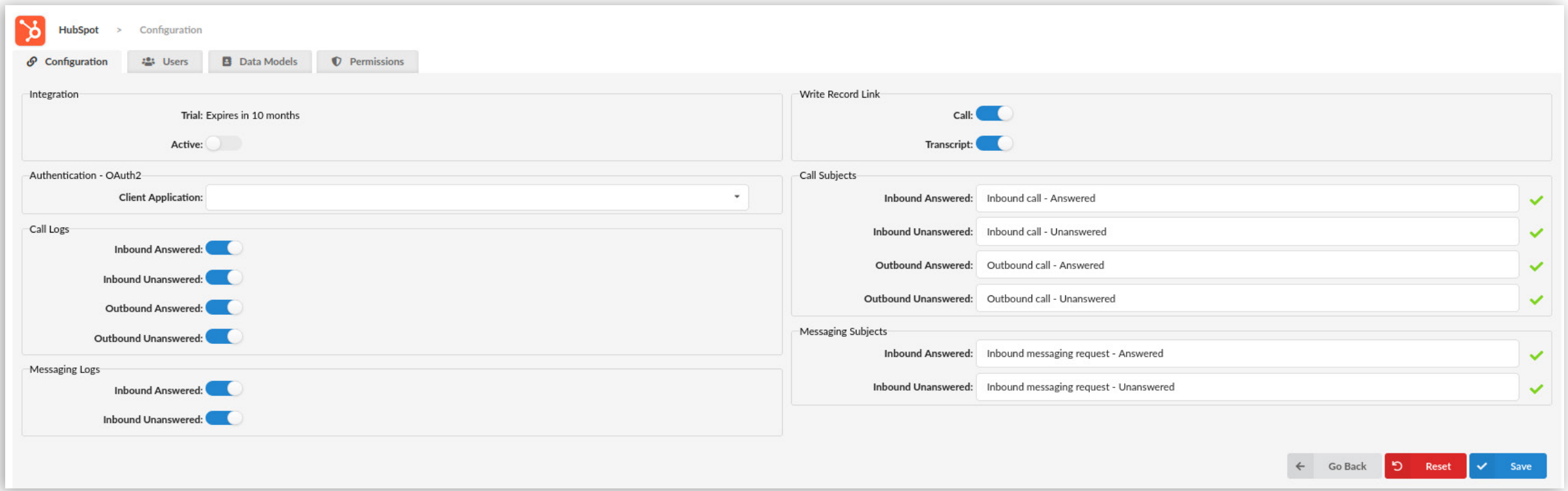
CSV Manager - Upload Button Spinner and Duplicate Submission Prevention

Clicking the Upload button multiple times in the CSV Manager before the upload process began would trigger duplicate import jobs for the same file. A loading spinner has been added and the Upload button is now disabled after the first click, preventing multiple submissions. This applies to CSV uploads in Contacts → CSV Manager.

CRM Integration Configuration - Reset Button Added

CRM integration fields such as Instance ID and Database Name would become permanently non-editable after saving.

A Reset button has been added to the CRM Integration settings page, allowing users to clear the current configuration and start fresh without requiring manual database intervention. This affects integrations such as Odoo, Vtiger, and other CRM connectors available in PBXware.



AI Hub

PBXware 8.1.0 introduces Google Gemini real-time Voice Agent support, updated Voice Agent and Live Transcription models, tenant-level shared MCP Server management, sentiment analysis for AWS live transcription, tenant-level Call Transcription controls, and AI-generated call summaries.

Google Gemini Real-Time Voice Agent Support

PBXware 8.1.0 adds Google Gemini as a supported real-time provider for Voice Agents. Administrators can configure a Google AI Provider and select it when creating or editing a Voice Agent.

Google Gemini Voice Agents include provider-specific options for selecting the model and voice, configuring the thinking level, enabling Google Search grounding, and adjusting Voice Activity Detection settings such as start sensitivity, end sensitivity, prefix padding, and silence duration.

Google Gemini Voice Agents also support the standard Voice Agent functionality available in PBXware, including prompts, built-in call-control tools, agent-to-agent transfers, MCP Servers, fallback behavior, call-duration limits, silence detection, and conversation transcript inclusion.

AI Voice Agent

General settings

Agent name:

✓

Agent slug:

AI provider:

Google Real-time Provider

✓

Model:

Model name

Voice:

Zephyr

Thinking level:

minimal

Google Search grounding:

VAD start sensitivity:

high

VAD end sensitivity:

high

VAD prefix padding (ms):

VAD silence duration (ms):

Dial Number:

1004

✓

Introduction:

B I H “ ” ⋮ ⋮ ⋮ ✕

Prompt:

Prompt example
This is our prompt example. Change it however suits your needs.

lines: 32 words: 157 2:64 Type % to add variables

✓

MCP Servers:

Please select ...

Agent enabled:

Include conversation transcript for agent to agent transfer:

Limit call duration:

Max call duration (min):

Fallback:

Transfer Call

End Call

Not Set

Fallback timeout (sec):

15

Transfer call number:

Silence detection:

Silence timeout (sec):

✓ Save

↺ Reset

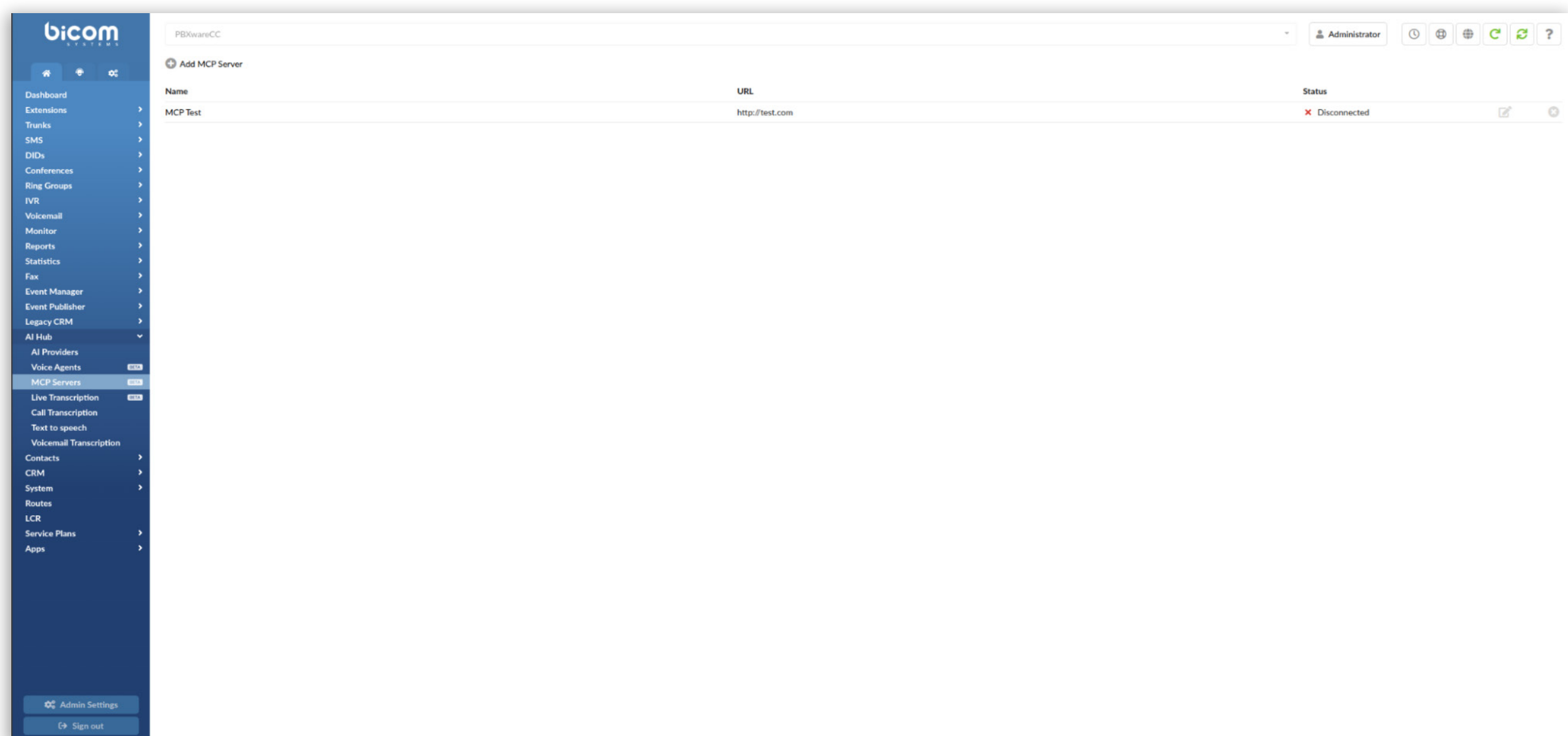
← Go back

Updated Voice Agent Models

PBXware 8.1 updates the predefined AI model selections available for Voice Agents to match the latest supported provider models. OpenAI Voice Agents now include **gpt-realtime-2** and **gpt-realtime-whisper**, while the xAI Voice Agent model has been updated to **grok-voice-latest**. These updates ensure new Voice Agents are created using current, supported model versions while still allowing administrators to manually specify alternative model identifiers when required.

MCP Server Management

A dedicated AI Hub → MCP Servers page has been added for configuring MCP servers. Previously, MCP servers were configured separately for each Voice Agent, which could result in duplicated configurations and multiple connections to the same server.



Administrators can now define an MCP server once, including its name, connection type, URL, and authentication headers. Configured MCP servers can then be selected and associated with individual Voice Agents from the Voice Agent configuration page.

This centralizes MCP server management, reduces duplicated configuration, and prevents unnecessary connections when the same MCP server is used by multiple Voice Agents. Existing MCP Server configurations defined within Voice Agents are migrated to the corresponding tenant's MCP Servers page during the upgrade, with their existing Voice Agent associations preserved.

Updated OpenAI Live Transcription Models

The predefined model list for OpenAI Live Transcription has been updated to include the currently available OpenAI transcription models:

- gpt-realtime-whisper
- gpt-4o-transcribe
- gpt-4o-mini-transcribe
- whisper-1

Administrators can select one of the predefined models or manually enter a different model when required.

Model:	gpt-realtime-whisper
URL:	gpt-realtime-whisper
Language:	gpt-4o-transcribe
	gpt-4o-mini-transcribe
	whisper-1

AWS Live Transcription Sentiment Analysis

The AWS Transcribe provider under AI Hub → Live Transcription now includes a Sentiment option.

The screenshot shows the 'AI Transcription' configuration window. It includes fields for 'AI Service' (AWS Transcribe), 'Provider' (AWS Transcribe), 'Content Identification Type' (On/Off), 'Content Redaction Type' (On/Off), 'PII Entity Type' (Please select...), 'Partial Results Stabilization' (On/Off), 'Partial Results Stability' (None), 'Regions' (US East (N. Virginia)), 'Language Code' (English (US)), 'Custom Language Model Name', 'Vocabulary Filter Method' (None), 'Vocabulary Filter Name', 'Vocabulary Name', and 'Sentiment Analysis' (On/Off). Green checkmarks are visible on the right side of the configuration items.

When sentiment analysis is enabled, completed live transcription segments include a detected sentiment value together with the transcript text. Supported sentiment values are Positive, Negative, Neutral, and Mixed.

Call Transcription Controls Moved to AI Hub

The Disable Transcription On-Demand and Disable Automatic Transcription options have been moved from Settings → Servers → System to the Call Transcription configuration under AI Hub.

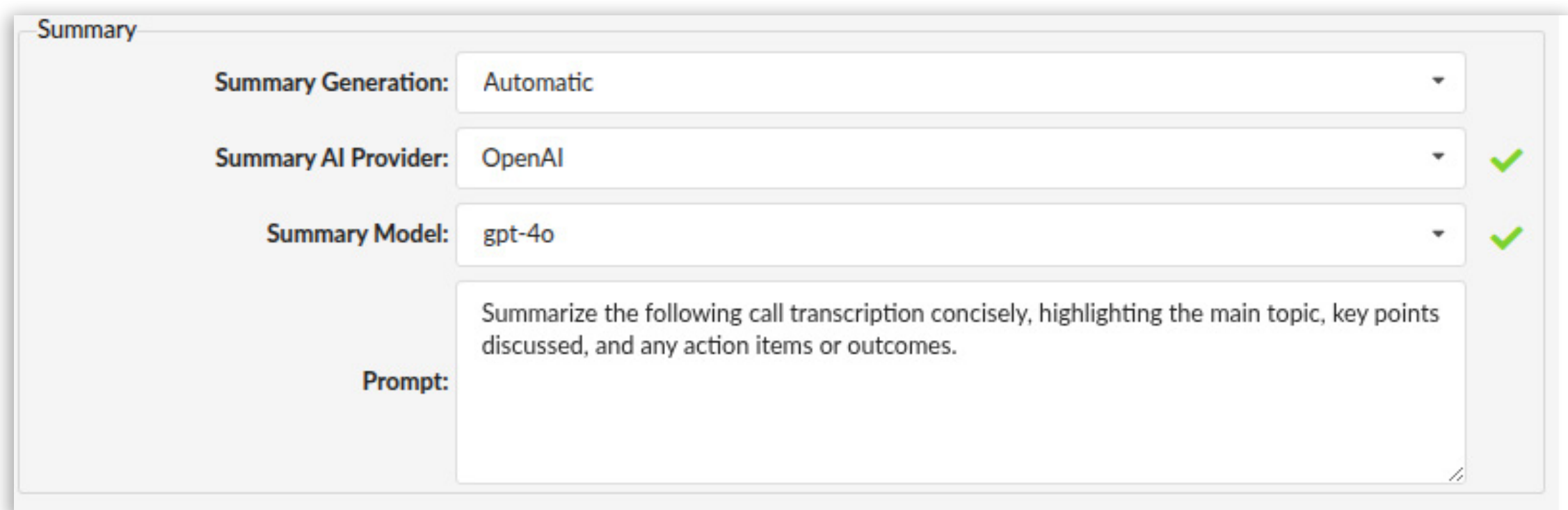
The screenshot shows the 'Call Transcriptions' configuration page in the PBXwareCC interface. It has a sidebar with navigation links like Dashboard, Extensions, Trunks, SMS, DIDs, Conferences, Ring Groups, IVR, Voicemail, Monitor, Reports, Statistics, Fax, Event Manager, Event Publisher, Legacy CRM, AI Hub, AI Providers, Voice Agents, MCP Servers, Live Transcription, Call Transcription, Text to speech, Voicemail Transcription, Contacts, CRM, System, Routes, LCR, Service Plans, and Apps. The main content area is divided into three sections: 'General' with 'AI Provider' and 'Model' dropdowns, 'Summary' with 'Summary Generation', 'Summary AI Provider', and 'Summary Model' dropdowns, and 'Options' with checkboxes for 'Allow Transcription in OSC', 'Disable Transcription On-Demand', and 'Disable Automatic Transcription'. A 'Save' button is at the bottom right.

On MT systems, these options are available for each sub-tenant under Sub-Tenant → Home → AI Hub → Call Transcription, allowing transcription behavior to be managed independently per sub-tenant.

On CC systems, the options are available under Home → AI Hub → Call Transcription.

AI-Generated Call Transcription Summaries

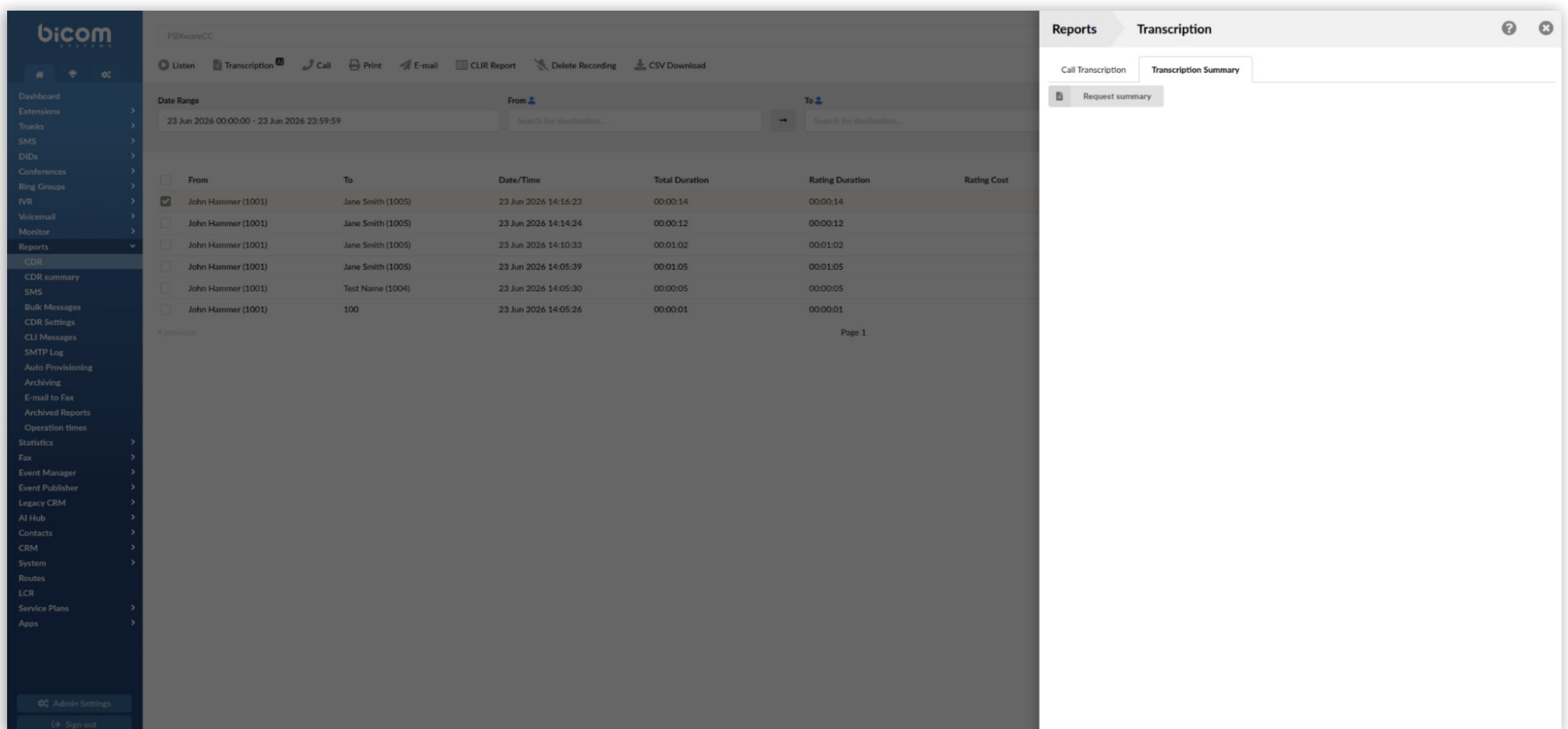
PBXware 8.1.0 adds support for generating AI summaries, also referred to as Smart Call Notes, from completed call transcriptions.



The screenshot shows a configuration window titled "Summary". It contains three dropdown menus: "Summary Generation" set to "Automatic", "Summary AI Provider" set to "OpenAI", and "Summary Model" set to "gpt-4o". To the right of the last two dropdowns are green checkmark icons. Below these is a text area labeled "Prompt:" containing the text: "Summarize the following call transcription concisely, highlighting the main topic, key points discussed, and any action items or outcomes."

A new Summary configuration group is available under AI Hub → Call Transcription. Administrators can enable automatic summary generation, select a configured OpenAI provider and model, and customize the prompt used to generate the summary. The transcription provider and summary provider do not need to be the same.

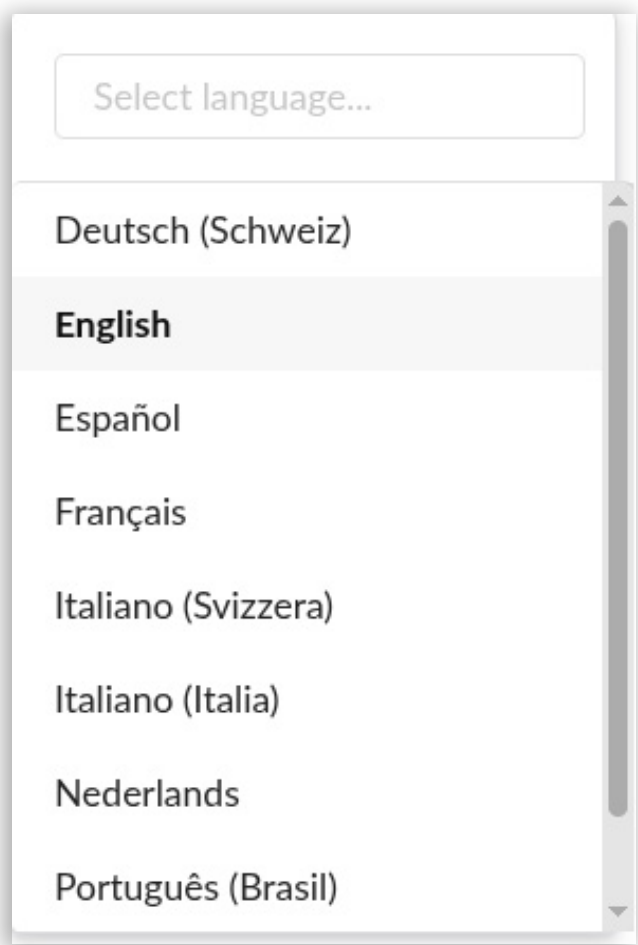
When automatic summary generation is enabled, PBXware generates and stores a summary after the call transcription has been completed. Administrators can also control manual summary generation through the Disable On-Demand Summary option. When on-demand summaries are permitted, users can generate a summary by selecting Request Summary from the Call Transcription page.



The transcription window available from the CDR page and OSC now contains separate Call Transcription and Call Transcription Summary tabs. The summary tab displays the generated summary together with the call information and provides options to send it by email, copy it to the clipboard, or print it. When no summary is available, the tab displays an appropriate informational message.

Additional Language Support

PBXware 8.1.0 expands its graphical user interface language support with Italian, Dutch, Swiss German (de_CH), and Swiss Italian (it_CH). These languages are now available as selectable interface languages, allowing users to operate PBXware using their preferred regional language.



New Algo Device Support

Algo 8138

Added support for the Algo 8138 IP Color Visual Alerter, a SIP-based visual notification device used for paging announcements, incoming call indication, and emergency alerts. The device provides high-intensity LED flash patterns with optional audio alerts to notify users of incoming calls, announcements, or emergency events.

Algo 8180

Added support for the Algo 8180 SIP Audio Alerter, a SIP-based device designed for loud ringing, paging, and emergency notifications. It can automatically answer paging calls, play alert tones or announcements, and receive multicast audio streams.

Algo 8301

Added support for the Algo 8301 IP Paging Adapter, a SIP-based paging adapter that automatically answers SIP paging calls and converts SIP audio into analog audio output and multicast paging streams, enabling integration with existing paging infrastructure.

Algo 8201

Added support for the Algo 8201 SIP PoE Intercom, a SIP-based intercom device intended for two-way voice communication, door entry, and paging applications. The device features a programmable call button that can automatically dial a predefined SIP extension when activated.

New Snom Device Support

Snom D810W UAD

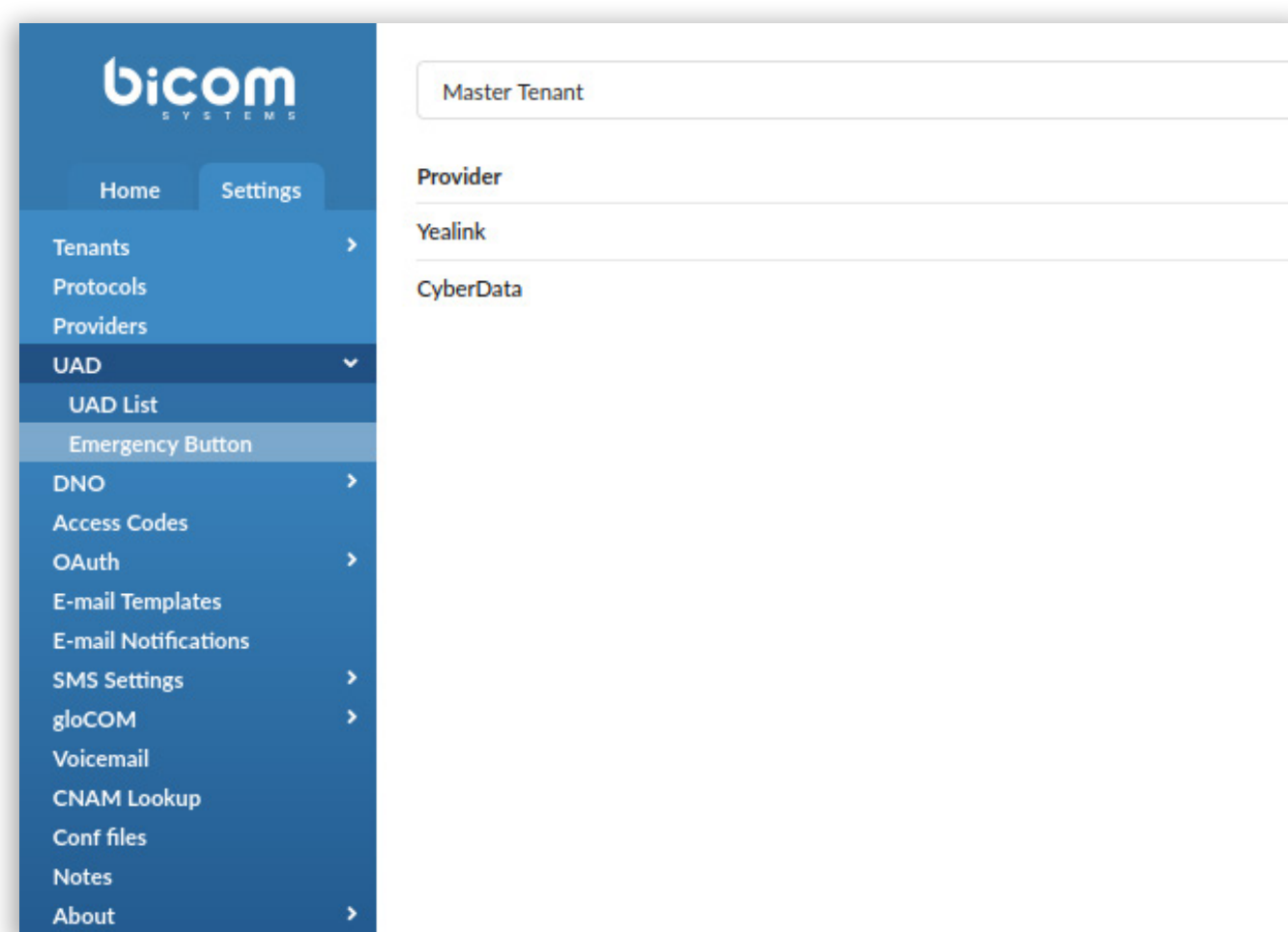
Added support for the Snom D810W business IP phone, including auto-provisioning and configuration through the PBXware UAD system.

Emergency Button (Yealink / CyberData)

PBXware 8.1.0 introduces the Emergency Button feature under Settings → UAD → Emergency Button. This feature allows administrators to centrally configure emergency alarm behavior for supported Yealink and CyberData devices and push settings to devices via auto-provisioning, eliminating the need for per-device manual configuration.

On MT systems, configuration can be set at the system level and inherited by tenants, or configured independently per tenant. On CC systems, configuration is managed at the system level. Settings are applied to devices through auto-provisioning.

Supported Yealink devices: T8x and AX8x series. Supported CyberData device: CyberData SIP Call Button.



Additional information in Abandoned Calls Report email

Abandoned Calls Notification email now includes the DID the caller dialed before abandoning the call. DID number is now included in email notifications.

Direct IN calls now respect the busy destination when an agent is unavailable

Direct IN calls to an agent (DID → agent, transfer to agent, or an extension dialing an agent number directly) are now routed to the agent's configured busy destination or voicemail whenever the agent cannot take the call — whether they are logged out, on pause, not logged in, already on another call, or reject the call. Previously, calls to a logged-out agent were dropped with a “Destination dialed is not supported” error, resulting in lost calls. Callers are now handled consistently instead of being disconnected.

Access for Admin and Site users to activities from CRM

Administrators and Site users can now review call recordings and chat transcripts without requiring an OSC extension. Access is now aligned with their roles and responsibilities, improving visibility, simplifying supervision, and enabling faster issue resolution.

CONTACT BICOM SYSTEMS TODAY

to find out more about our services



Bicom Systems (USA)

2719 Hollywood Blvd
B-128
Hollywood, Florida
33020-4821
United States
Tel: +1 (954) 278 8470
Tel: +1 (619) 760 7777
Fax: +1 (954) 278 8471
sales@bicomsystems.com



Bicom Systems (CAN)

Hilyard Place
B-125
Saint John, New Brunswick
E2K 1J5
Canada
Tel: +1 (647) 313 1515
Tel: +1 (506) 635 1135
sales@bicomsystems.com



Bicom Systems (UK)

Unit 5 Rockware BC
5 Rockware Avenue
Greenford
UB6 0AA
United Kingdom
Tel: +44 (0) 20 33 99 88 00
sales@bicomsystems.com



Bicom Systems (FRA)

c/o Athena Global Services
Telecom
229 rue Saint-Honoré – 75001
Paris
Tel : +33 (0) 185 001 000
www.bicomsystems.fr
sales@bicomsystems.fr



Bicom Systems (ITA)

Via Marie Curie 3
50051 Castelfiorentino
Firenze
Italy
Tel: +39 0571 1661119
sales@bicomsystems.it



Bicom Systems (RSA)

12 Houtkapper Street
Magaliessig
2067
South Africa
Tel: +27 (10) 0011390
sales@bicomsystems.com

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www.bicomsystems.com