RELEASE NOTES

PBXware 8.0.0

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Important notes:

- The new AI Providers page requires all AI providers to be set up again for users with features such as Call Transcription, Voicemail transcription, and TTS configured. It is a breaking change so all of these should be reconfigured.

- Stereo recording now requires the RAM Disk to be enabled

- Live Call Transcription (which is implemented this release) requires Stereo Recording to be enabled

- The archiving monitor system is changed, and recordings on the file system are restructured to follow the /year/month/day/hour format within the ‘monitor’ directory (it might take the system some time to restructure the existing recordings on first start after upgrade, so some CDR functionalities may not be available).

PBXware version 8.0 introduces important licensing updates.

- **Event Publisher** will be automatically enabled system-wide on all systems where Event Manager was previously enabled for any tenant.

- **AI features** will be enabled on all v8 systems by default, with the option for system administrators to further enable or disable it per tenant using Groups.

- **Contact Center Lite** is a licensed feature for Multi-Tenant and Business editions that will be charged after the trial period is completed.

  On the Contact Center edition, these features are available without additional charges.

- **The new CRM Module** (crmconnector service) introduces a separate licensing model on a per-user (per-seat) model within each CRM integration.

  CRM licensing operates independently of the PBXware Editions & Modules configuration. Each tenant (Multi-Tenant edition) or system (Contact Center and Business editions) supports one custom CRM integration. CRM routing for a specific integration is enabled by default once at least one active license is assigned.

  A free trial period is available until May 1, 2027. During this period, the new CRM module is enabled by default on all v8 systems, with the option for system administrators to control its availability per tenant or system using Groups.

We are happy to introduce PBXware version 8.0 - Our biggest update yet, which includes many new features and major improvements in order to provide the best possible experience to our customers. With this major update, we have added a few big features, such as an AI Voice Agent - AI Voice Connector being the service which enables it - which allows administrators to configure an AI Voice Agent as a specific DID destination to which calls can be directed to and answered by it. In addition, Live Transcription, Text to Speech, and Voicemail Transcription have been introduced under the new AI Hub, further expanding PBXware's AI capabilities. Event Publisher has also been added, which is planned to replace the previously used Event Manager. Event Publisher is a lot more powerful, where each event sends specific data to a server that the consumer has configured for themselves. For easier use, configuration and billing, API Keys for AI features such as Live Transcription, Transcription, Voice Agents and TTS will now be managed per server/tenant, where all the data and information can be stored in one place. In this way, separate API Keys can be used for different tenants since billing is handled per API Key.

In addition to these major innovations, PBXware v8.0 introduces several other significant enhancements. A completely redesigned CRM Module is now available in gloCOM Web and the new gloCOM Desktop application, delivering improved and deeper integration with external CRM platforms. The new architecture supports multiple CRM integrations per tenant/company and enables flexible data model management within customizable CRM widgets across all communication channels. Conversations are automatically logged in connected CRMs, enhancing accuracy, improving operational efficiency, and increasing productivity for sales and support teams. Advanced CRM Routing for voice introduces intelligent, data-driven call handling based on real CRM records, enabling dynamic routing decisions tailored to customer profiles. The new Unified Contacts module replaces the legacy Central Phonebook and introduces structured contact visibility levels (Global, Department, and Private), centralized management, smart CSV synchronization, and enhanced permission control. Furthermore, Contact Center Lite enhances monitoring capabilities by introducing Wallboard and ERG dashboards in gloCOM Web and Desktop, along with Scheduled Reports for both Enhanced Ring Group (ERG) and Extension statistics, including ERG callback support—providing improved visibility and reporting across editions.

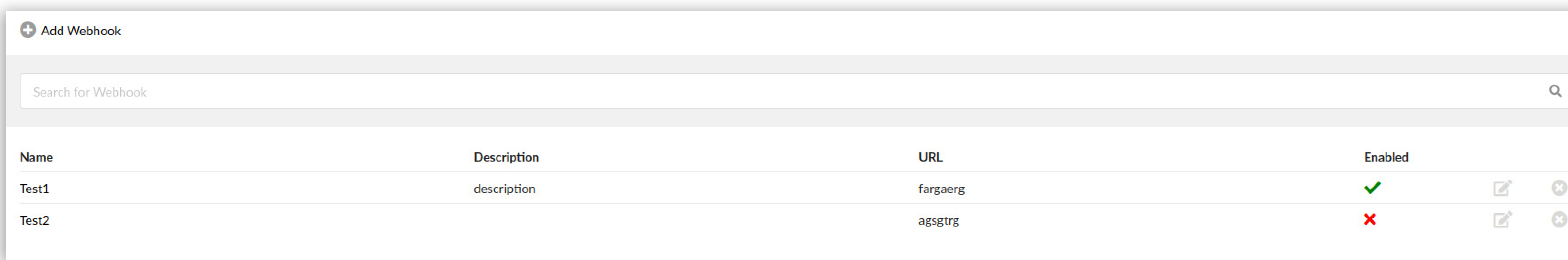
Features

Event Publisher

PBXware 8.0.0 introduces **Event Publisher**, designed to replace the previous Event Manager. This new feature gives its users more freedom by allowing them to integrate their own backend server and handle data generated by various events however they wish. With Event Publisher, we have access to real-time events, meaning that any change in the call flow will generate a new event with various details.

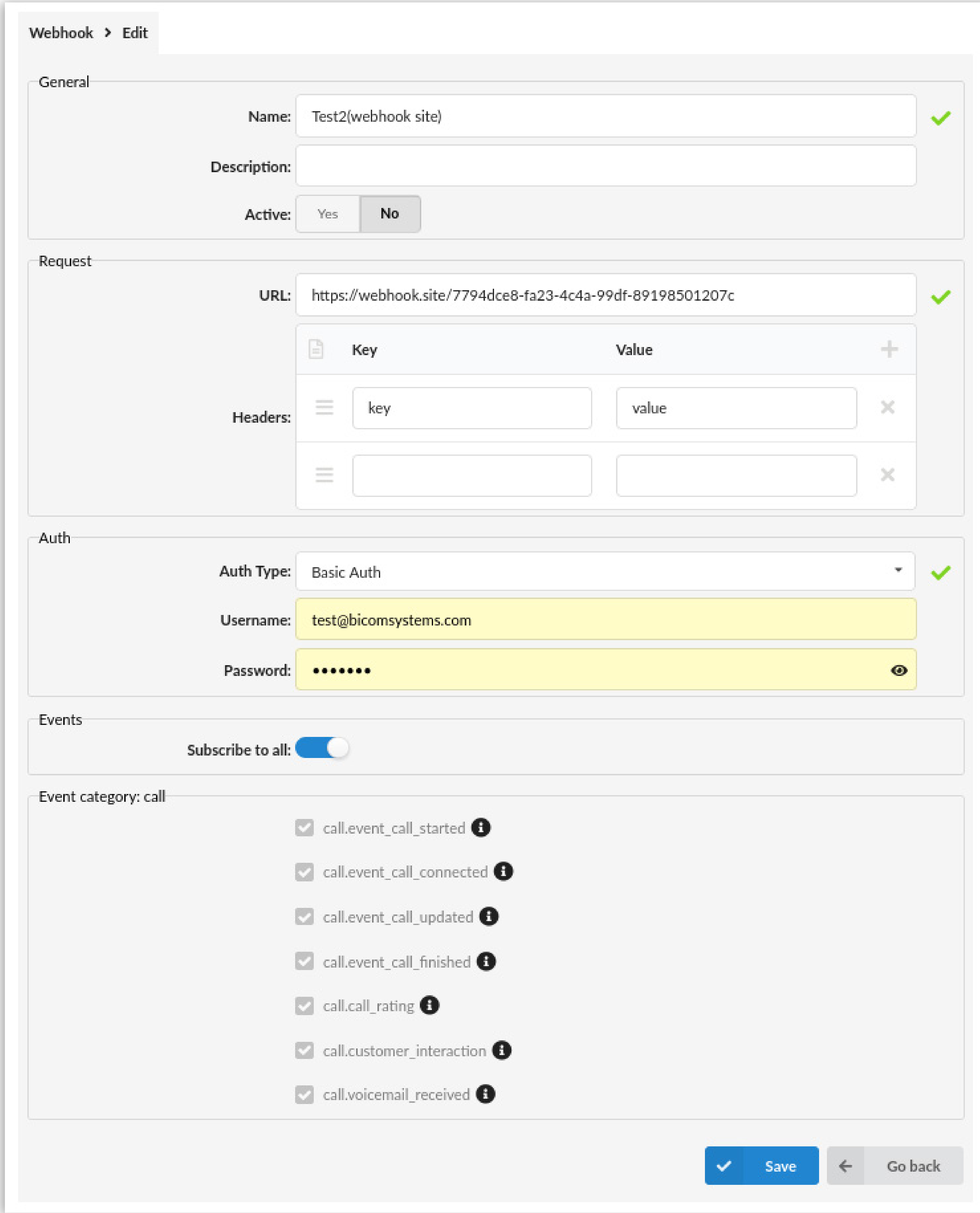
NOTE: *This feature must be enabled on the license before it can be accessed; once enabled, it can be found on the Home tab.*

A list of existing webhooks can be found under the Webhooks page, together with their names, descriptions, URLs, and enabled status. Users can edit or delete them accordingly, or use the search bar to search for any specific webhooks.



Clicking the 'Add Webhook' button opens a new screen that prompts the administrator to enter details for the webhook they want to configure.

Various webhook details can be configured in this screen, and there are a few sections. For more information on each section, please refer to the screenshot and detailed instructions below.



General

In this section, the administrator can enter the name and description of the webhook that they are configuring. Additionally, the webhook can be activated or deactivated by selecting an appropriate option under the 'Active' field. **Please note that it is not active by default.**

Request

In this section, a valid URL must be entered. It is a required field, and the URL must be valid for webhooks to work properly. Additionally, users can set headers, and those that are entered are hardcoded and will be sent as such to the entered URL.

Auth

In this section, various Auth options can be selected; they must be configured correctly for the webhook server used. Current Auth options are as follows: 'No Auth', 'Basic Auth', 'Bearer Token', 'OAuth Client Credentials', and 'OAuth Resource Owner Password Credentials'.

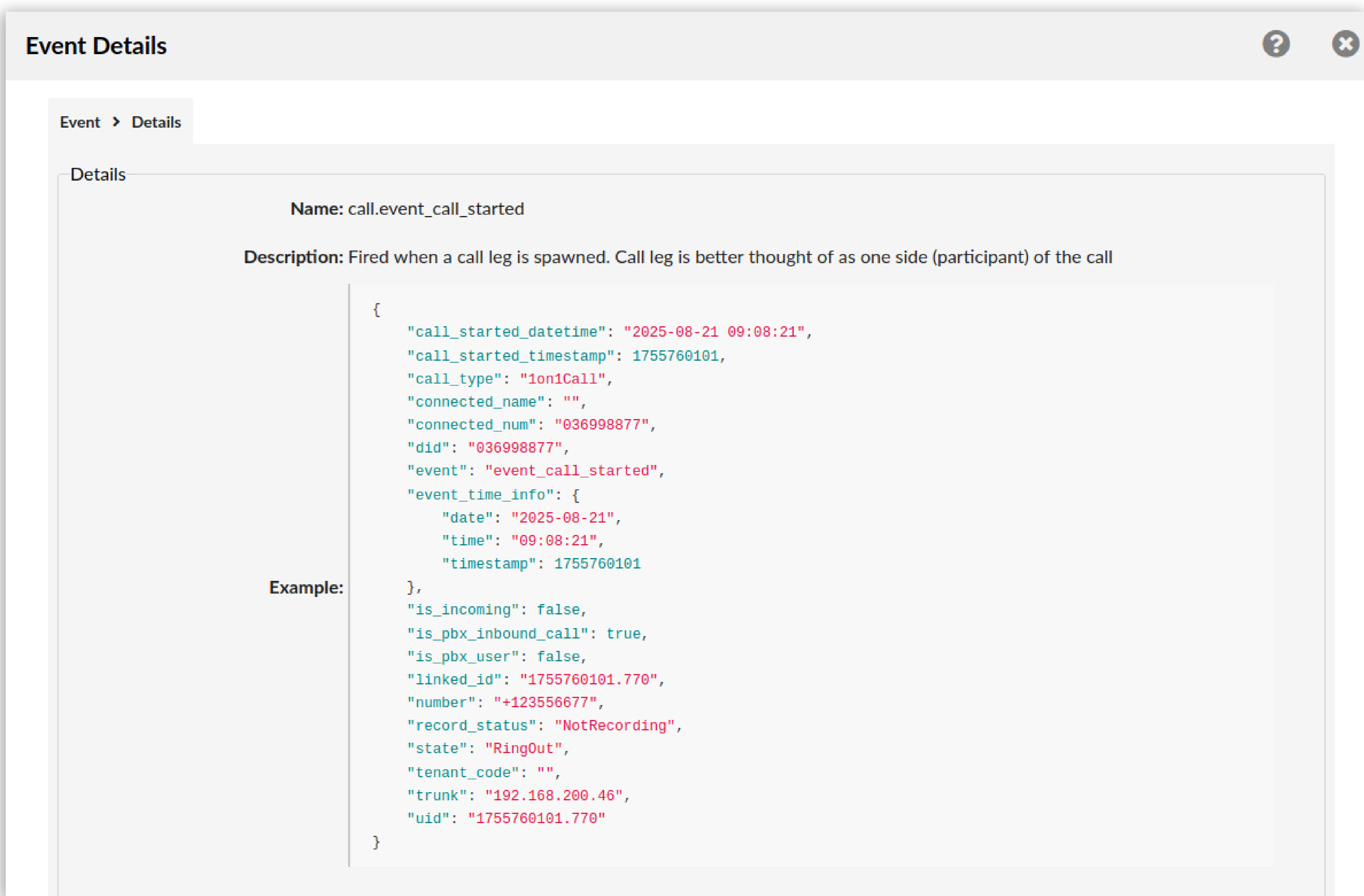
Events

This section allows a user to toggle the button to subscribe to all events or not. This also includes any additional events that may be added.

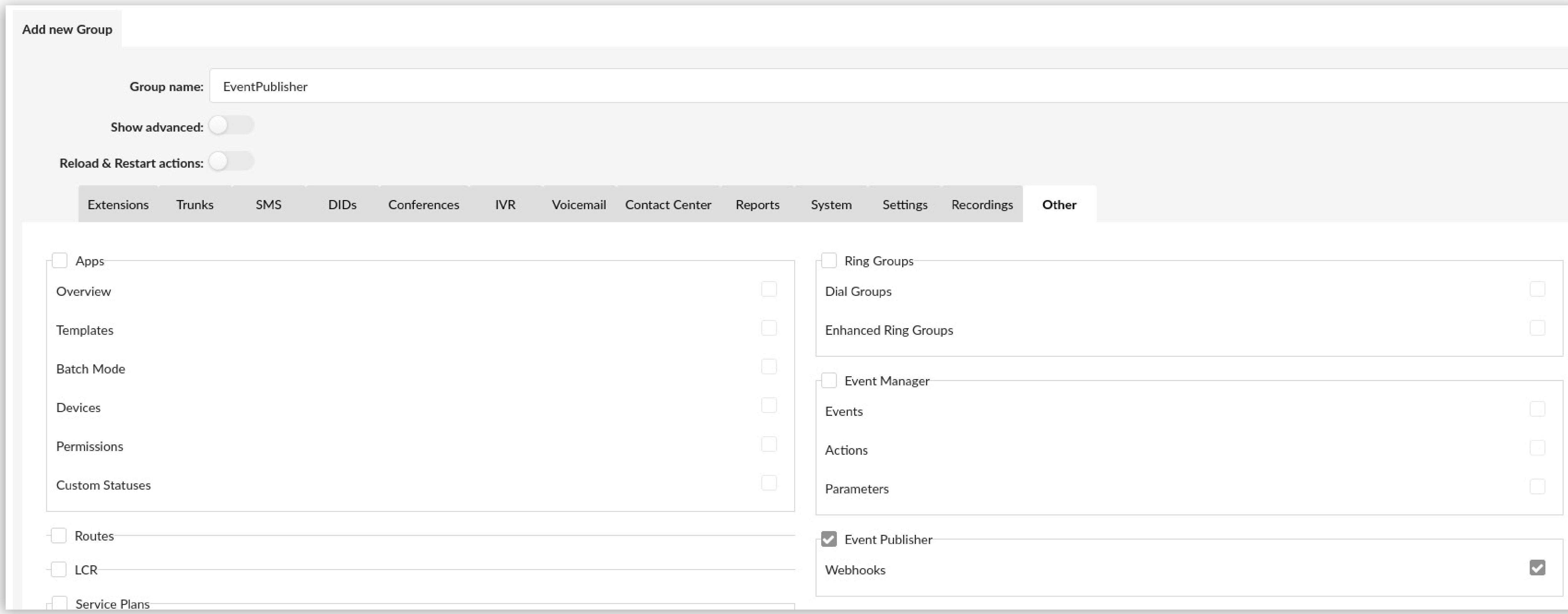
Event category: call

This section allows selecting different call events that will trigger the configured webhook. If any additional information is needed for an event, a user can click the information icon to open an iframe with documentation and other necessary information about the event. It explains when the webhook will be fired.

EXAMPLE:



The system administrator can also give permissions to user groups to enable or disable the usage of webhooks. Those users who belong to user groups that do not have the permission to access the Event Publisher will not be able to see it nor perform any other kind of actions (add, edit, or delete). The option can be enabled in Admin Settings → Groups -> Group, within the 'Other' tab. For more information, please refer to the screenshot.



Concurrent Archiving

With the PBXware 8.0.0 release, concurrent archiving of call recordings has been implemented, along with additional changes. Upon updating to this version of PBXware, folder granularity will be applied. For concurrent archiving to be implemented, the monitor directory structure had to be changed so that recordings are now granulated in the /year/month/day/hour UTC format. Recordings that already exist on the system will be migrated.

Note: While the migration of recording files is in progress, some functionalities such as listening to call recordings, call transcriptions, and archiving/offloading will be skipped during this process.

Concurrent archiving uses multiple workers to process call recordings as they are offloaded. Administrators can configure the number of workers they wish to use in the /remote_fs/conf.ini file by adding max_workers = x to the [global] section, where x is the preferred number of workers. The default number of workers is 5.

To avoid offloading recordings that are still in progress, the active_rec_timeout option has been added to the global section of the archiving configuration file. This option specifies the number of minutes to wait after the recording file is last modified before offloading it. The default value is 5 minutes.

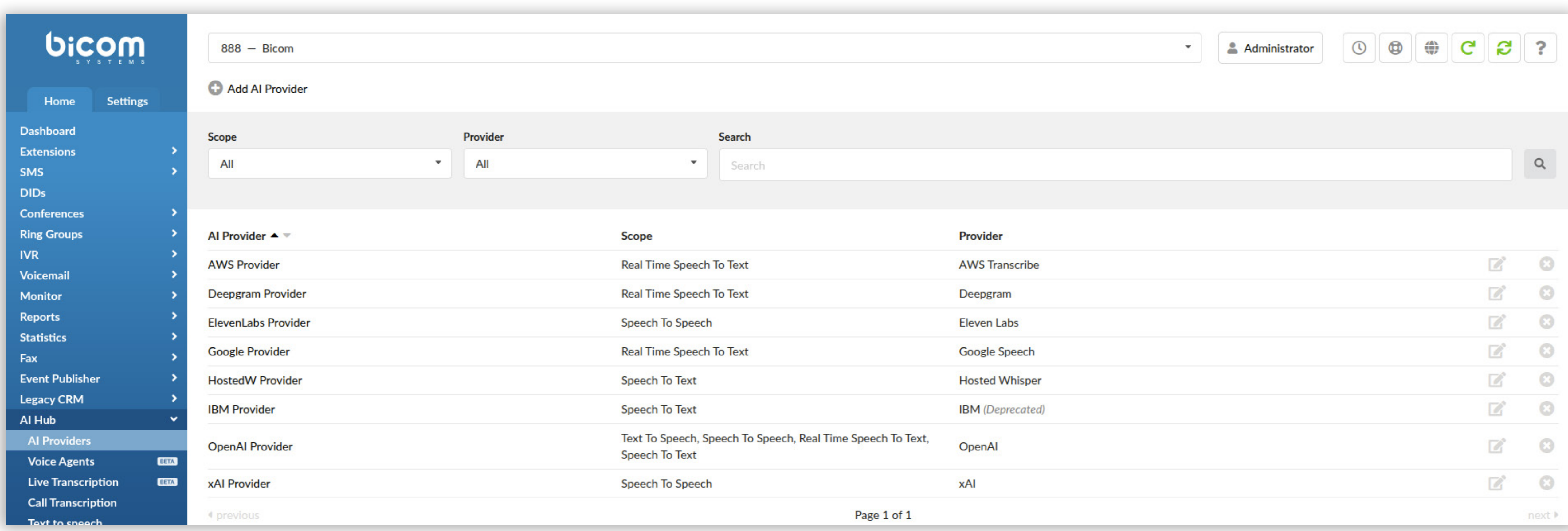
Important Note: With this update, archiving CLIRs will no longer be available. The option has been removed from the Resources dropdown.

AI Hub

A new tab, AI Hub, is introduced in PBXware 8.0, allowing easy configuration and management of all of PBXware's AI features. It includes the following features:

- **AI Providers** - where administrators can configure AI Providers, including credentials, required provider details, and scopes used by AI Hub features.
- **AI Agent** - where administrators can configure an AI Voice Agent to answer calls and guide callers through a conversation.
- **Live transcription** - a feature that allows admins to configure real-time ERG, Queue and Extension call transcriptions of the conversations.
- **Call Transcriptions** - Enables administrators to configure Call Recording Transcriptions.
- **Text to speech** - where administrators can use AI Providers to generate a sound file from a text.
- **Voicemail Transcription** - Transcribes voicemails using some of the available AI Providers.

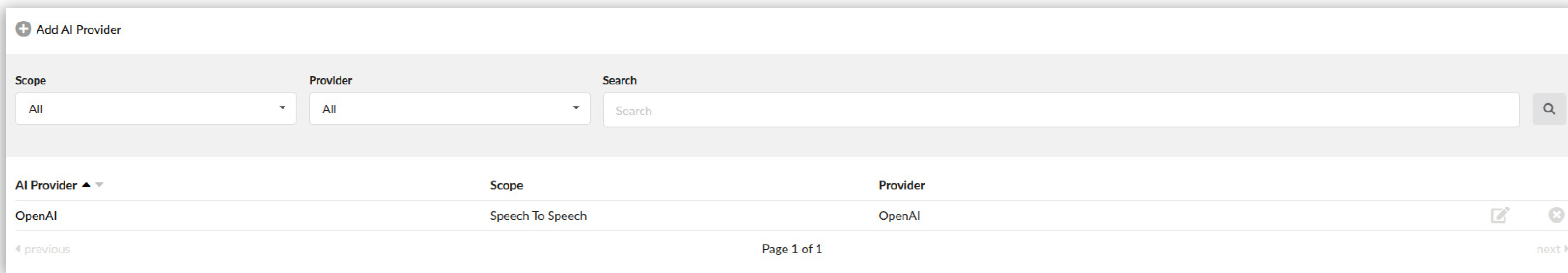
Through this AI Hub, users can easily navigate through all of the AI features, all of which will be described in more detail individually.



Note: The AI Hub features are enabled in the license by default, with the option for administrators to further enable or disable these features by tenant (on MT systems).

AI Providers

With this new release, the way AI Providers are managed has changed. In the earlier versions, each AI feature had to configure its AI Provider individually. With version 8.0, AI Providers can be configured per tenant, and admins can additionally select what the certain provider will be used for.



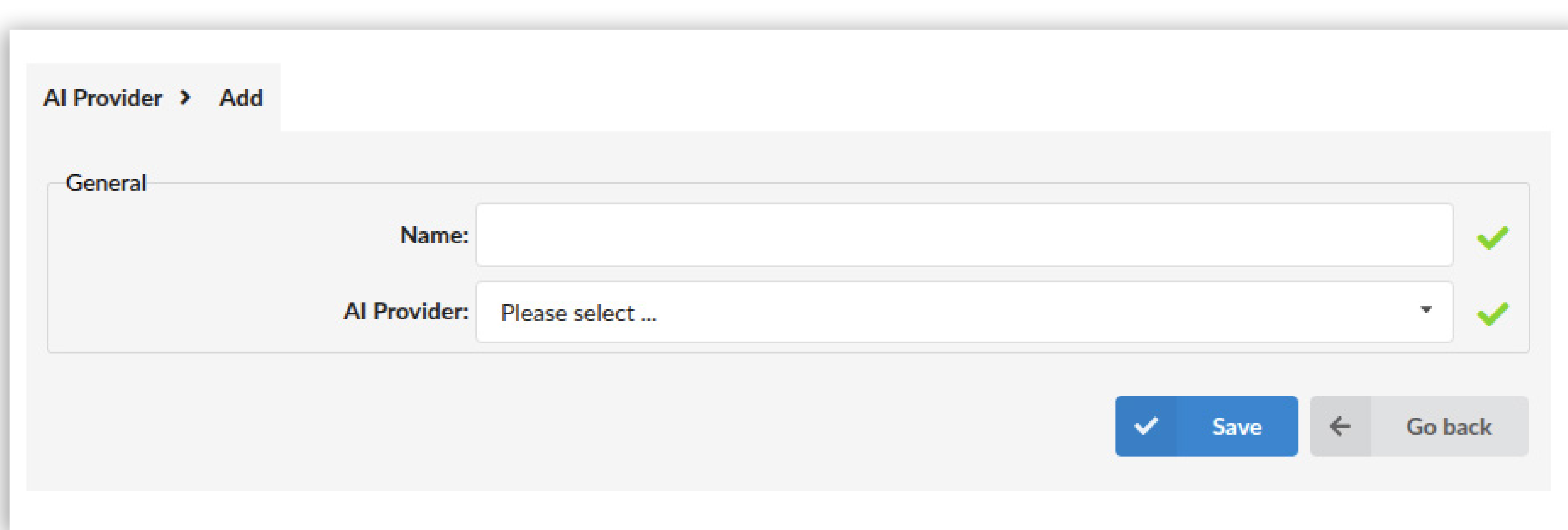
A list of created providers will be visible here, where they can be edited or deleted. The Providers can also be filtered by:

- Scope: All, Text To Speech, Speech To Text, Speech to Speech and Real Time Speech To Text
- Provider: All, OpenAI, Google Speech, IBM, Deepgram, AWS Transcribe, xAI, Hosted Whisper, Eleven Labs
- It is also possible to use the Search bar to search for any specific providers.

IBM Watson Voicemail Transcription — Deprecated

IBM Watson voicemail transcription has been deprecated as of PBXware 8.0.0. The functionality remains available in this release but will be removed in a future version. Users relying on IBM Watson for voicemail transcription are encouraged to migrate to an alternative provider.

Additionally on this page you can add new providers by clicking the ‘Add AI Provider’ button, which will open a new screen and prompt the administrator to enter necessary details.



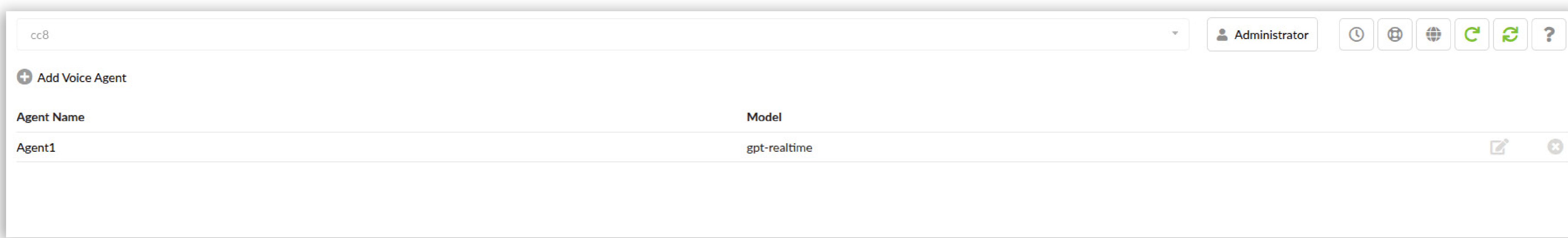
You must enter a name for the provider, and select one AI Provider from the dropdown. Upon selecting one of the providers, new fields will appear, all depending on the selected provider. The Provider needs to be configured properly in order for AI features to work.

One of the fields that will appear after selecting a provider will be Scope, where you can manage what the chosen provider will be used for. Available options depend on the provider, so some scopes will not be available for all of them.

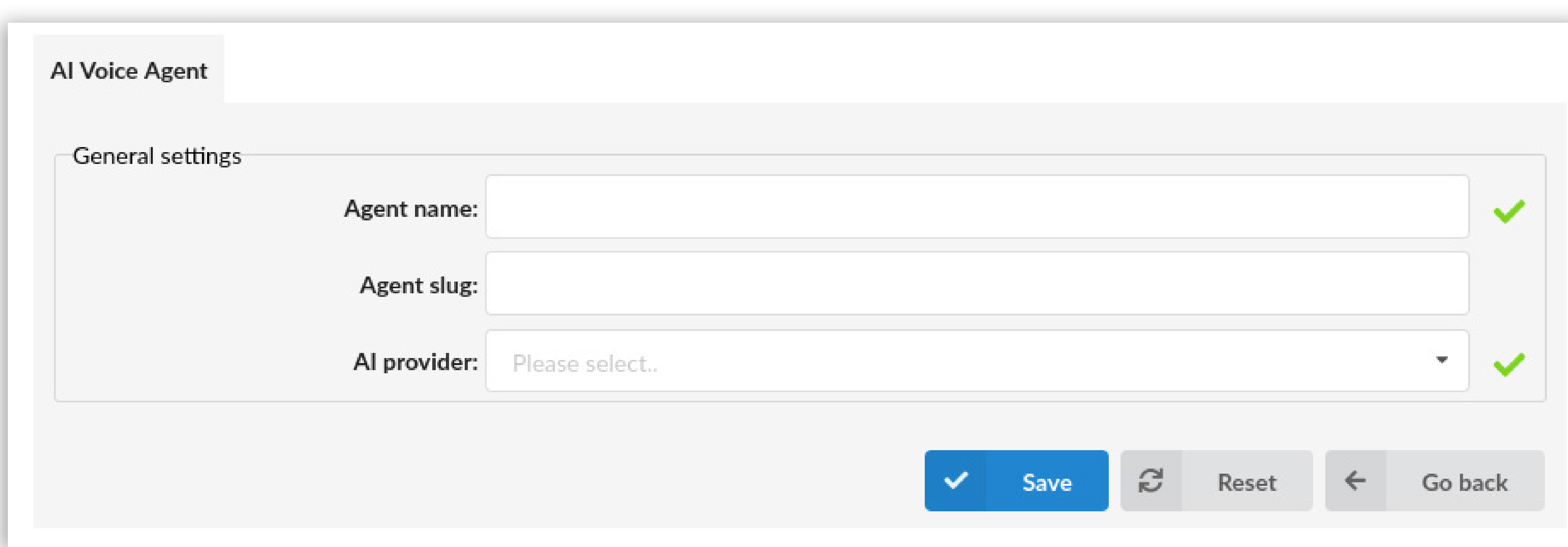
AI Voice Agent

This new feature allows for inbound calls to be answered automatically by an AI Voice Agent, with an AI driven experience, in a consistent tone that is configured by the administrator, and can guide callers through the conversation, transfer the call and also end calls when the conversation is completed.

In the AI Agent page, a list of created agents can be viewed and managed. They can be edited and deleted.



Clicking the 'Add Voice Agent' button will open up a new screen where the agent can be configured. Please note that prior to this step you should already have some AI Provider configured. It is required for setting up Voice Agents.



When first configuring the AI Agent, the Agent name should be entered and the AI provider needs to be selected, since they are required fields for creating one., whereas Agent slug is an optional field. After the AI provider has been selected, the AI Voice Agent window will expand, offering many options for configuring the AI Agent.

Currently supported providers for AI Voice Agents are OpenAI, xAI, and ElevenLabs. The available models depend on the selected provider:

- OpenAI
  Voice models: gpt-realtime, gpt-realtime-mini
  Transcription models (optional): whisper-1, gpt-4o-mini-transcribe, gpt-4o-transcribe

- xAI
  Model: grok

- ElevenLabs
  Models: Multiple voice models are available (22 total) and are configured exclusively on the ElevenLabs platform. In PBXware, the administrator copies the Agent/Voice ID from ElevenLabs and pastes it into the provider configuration.

Here, administrators can completely customize the AIVoice Agent, including its behavior.

Model and Transcription model (if transcription is going to be used as well) can be selected. Administrators can also configure and customize the behavior and mannerisms of the AI Agent by populating the Introduction field, and the Prompt box. Within the Prompt text box, more specific settings regarding the AI Voice Agent can be set, such as its objective when answering the calls, the tone, context, personality, etc, as shown in the example. To further modify the AI Agent, a different voice can be selected for it.

The Prompt field now also supports template variables, allowing dynamic call-related information to be inserted directly into the prompt and used during the conversation flow, including tool usage and interactions between the agent and caller. Currently supported variables include:

%CALLER_ID%

%CURRENT_TIME%

%CURRENT_DATE%

%CURRENT_DATE_TIME%

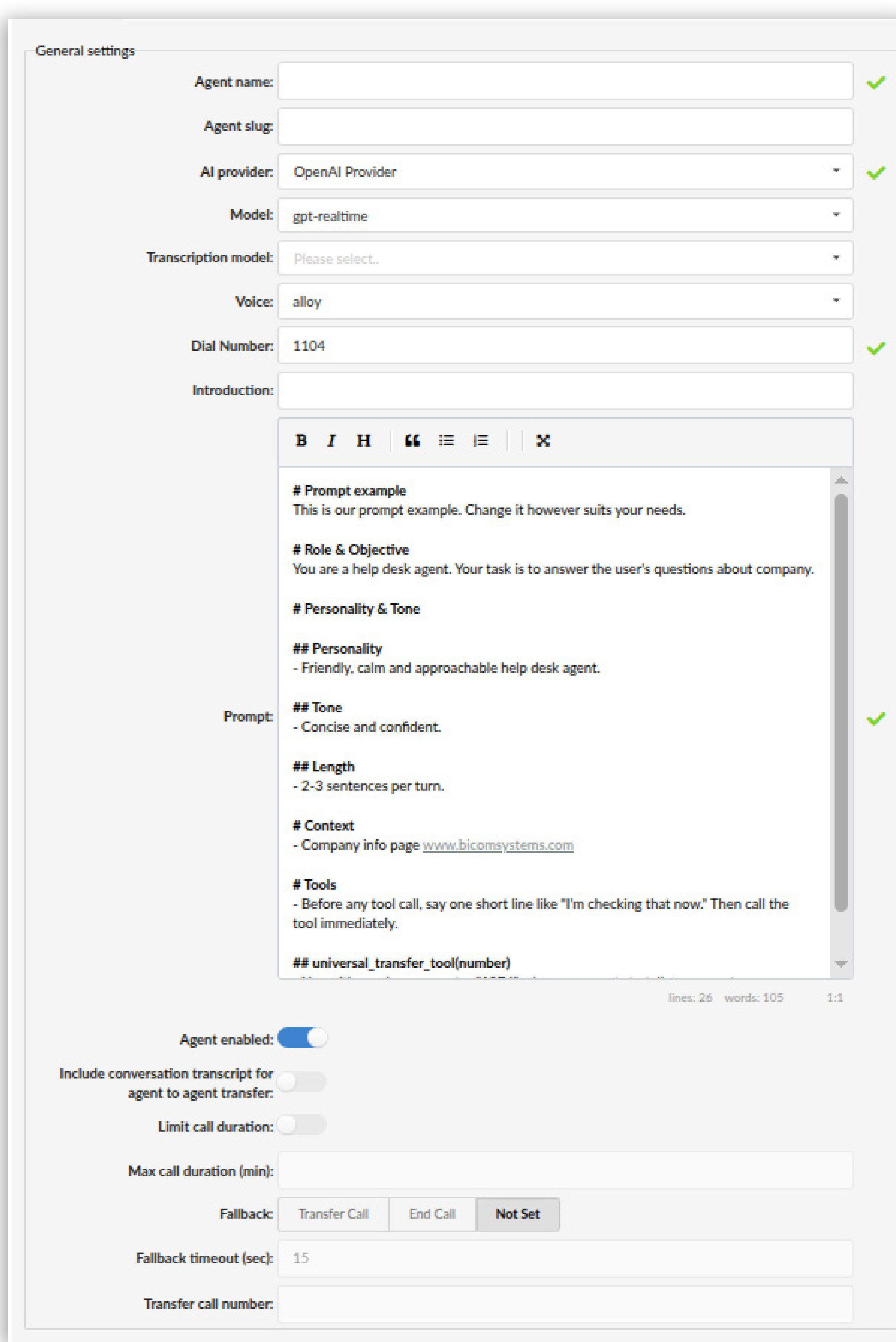
This allows administrators to build more context-aware prompts and create more flexible agent behavior based on real-time call information.

Most importantly, the prompt section is used to configure and define the transferring logic for the Voice Agent. Here, it is also possible to configure a prompt which will be used to transfer calls between agents. This is an especially useful feature because all the external knowledge of one AI Agent about the caller is being transferred to the other agent.

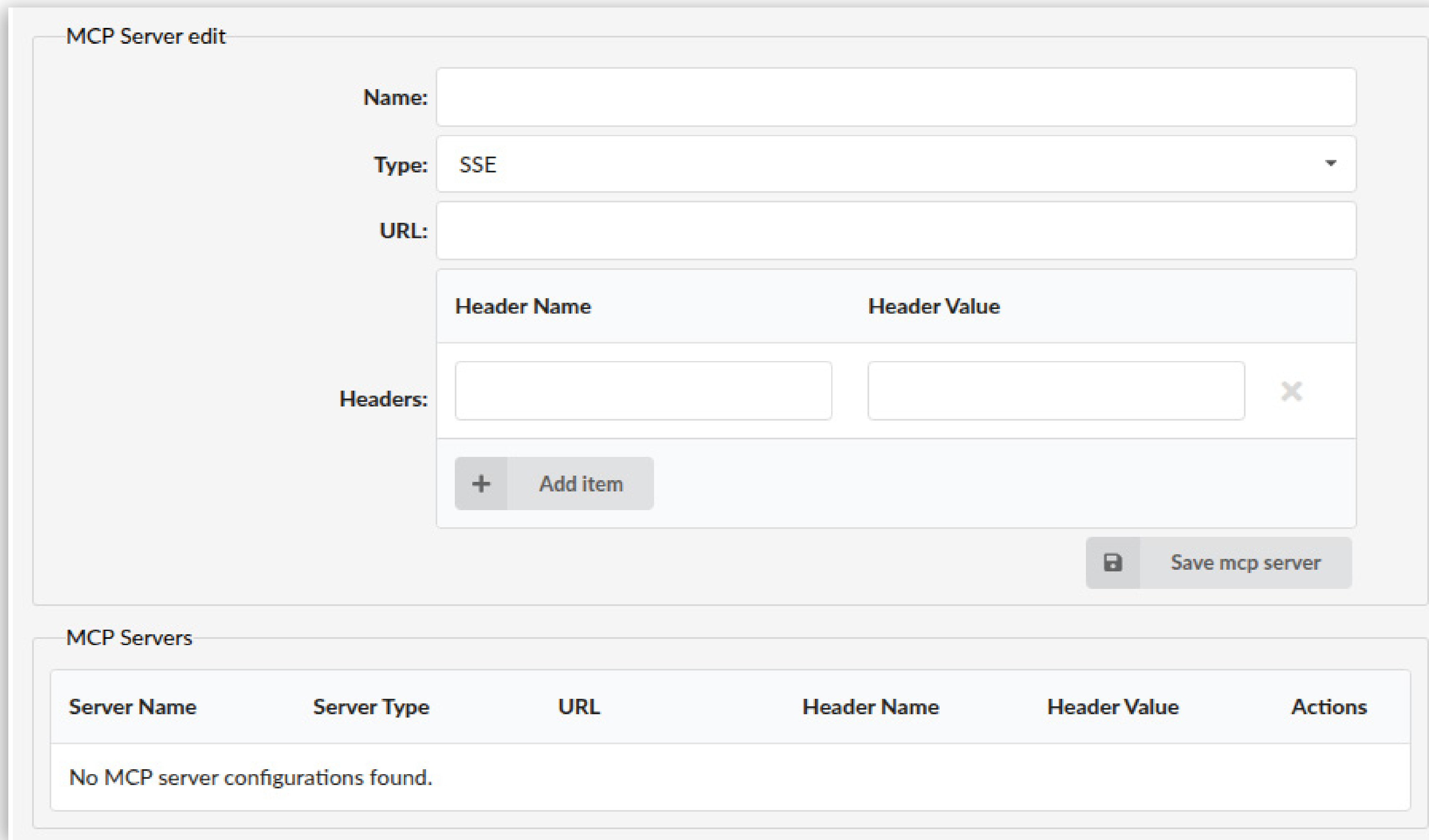
Aside from configuring the personality of the AI Agent, further configuration needs to be completed in order for it to work properly. Dial number of the AI Agent is a required field and must be populated. It is the number which will be dialed to reach it locally, since it also acts as an extension. Later on, a DID has to be configured to point to the AI Agent directly so it can be reachable, otherwise it will not be possible to reach it from outside the PBXware.

Some extra options include toggling the 'Limiting call duration (min)' option and entering the maximum amount of minutes the call can last if it is enabled; and call Configurable fallback behavior can be set in case service is not responding within a defined time interval, allowing the call to be transferred to another number or automatically ended.The Fallback timeout is the amount of seconds the call will be transferred or ended, depending on fallback behavior, in case connection is lost.

Note: it is very important that you set up the AI Agent prompt carefully and precisely so that the agent functions as well as possible, and transfers calls properly. Additionally, the configuration and functionalities depend on the selected AI Provider, and the example described is based on OpenAI.



MCP Server



An MCP (Model Context Protocol) can be configured in the AI Agent settings. This connects your PBXware AI Agent with external services. It helps the AI Agent provide reliable data to the callers when requested. It is not mandatory to set up an MCP Server, but it greatly helps improve the AI Voice Agent experience, as it allows it to use any external information it may need, it provides a single place where tools are maintained and managed

Multiple MCP Servers can be added per AI Agent. While adding an MCP Server, you need to enter a Name and a URL (MCP Endpoint) provided by a tool hosting service. These are required fields.

Under the Type dropdown, currently available connection options are SSE and Streamable HTTP.

If necessary, headers can also be added in the Headers section by populating the Header Name and Header Value fields.

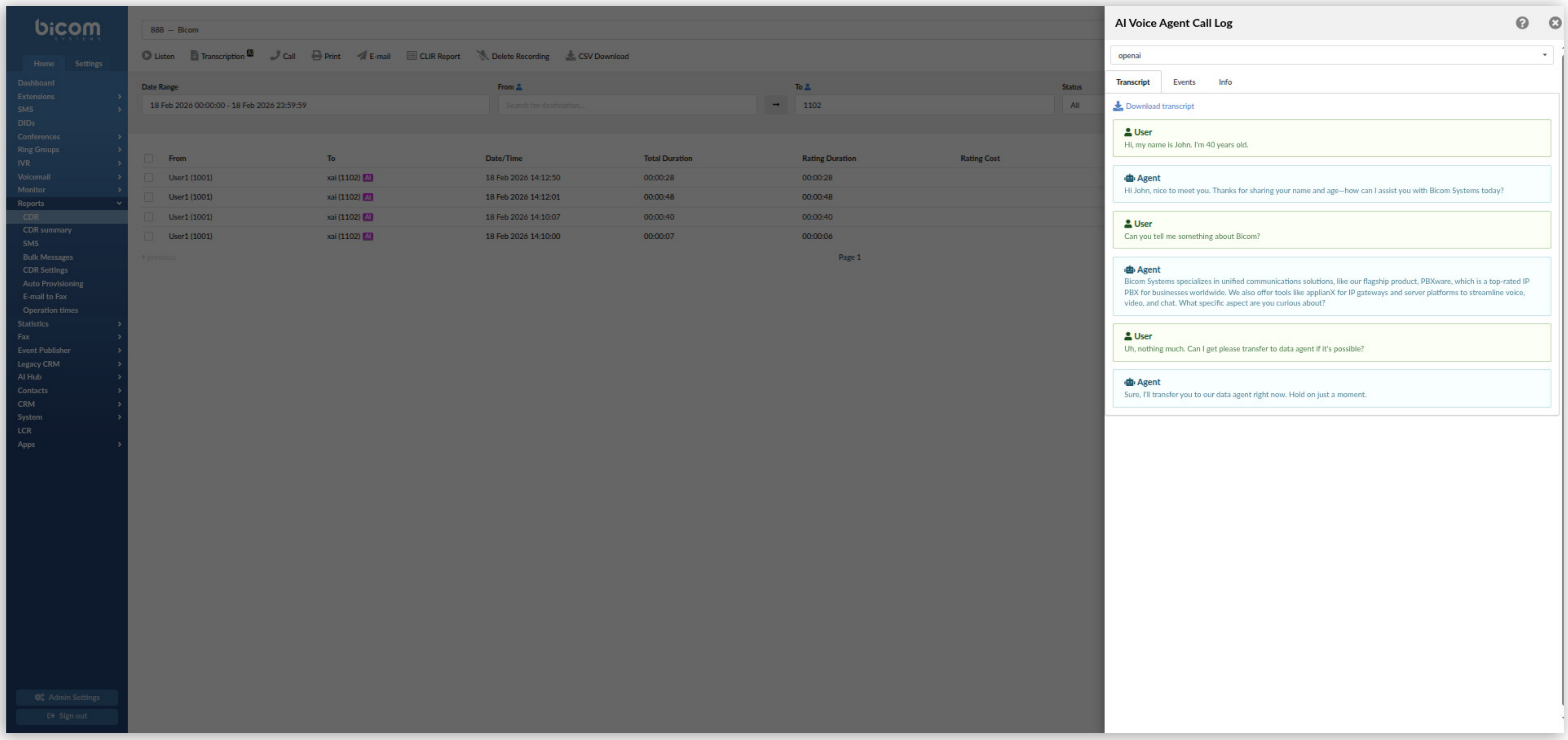
When all of the required fields are populated, clicking the 'Save MCP server' will add it to the list of servers in the section below.

The MCP Servers section displays a list of all the MCP endpoints that are connected to the AI Agent, where they can be edited and deleted.

In order to create the MCP server, there is a wide range of open source or commercial solutions that are designed to help partners easily build and orchestrate AI actions.

CDR - AI Voice Agent Call Log

Calls that were handled by an AI Voice Agent are now clearly marked on the CDR page with a purple AI badge next to the agent/extension number. Clicking the agent number opens a right-side panel called AI Voice Agent Call Log, providing quick access to the full call context and agent activity for that specific call.



The AI Voice Agent Call Log side panel contains three tabs:

- Transcript – Displays the full conversation between the caller and the selected AI agent, with an option to download the transcript.
- Events – Shows a chronological list of agent events captured during the call (for example, tool/function calls, including agent-to-agent transfer events), helping with troubleshooting and validation.
- Info – Provides a summarized view of the selected agent's call details (such as whether the call was transferred, which agent received the transfer, and the AI provider used).

If the call involved multiple AI agents (for example, when transferring from one AI agent to another), the panel includes an agent dropdown at the top. This dropdown lists all AI agents that participated in the call, allowing you to switch between them and review the corresponding Transcript, Events, and Info for each agent.

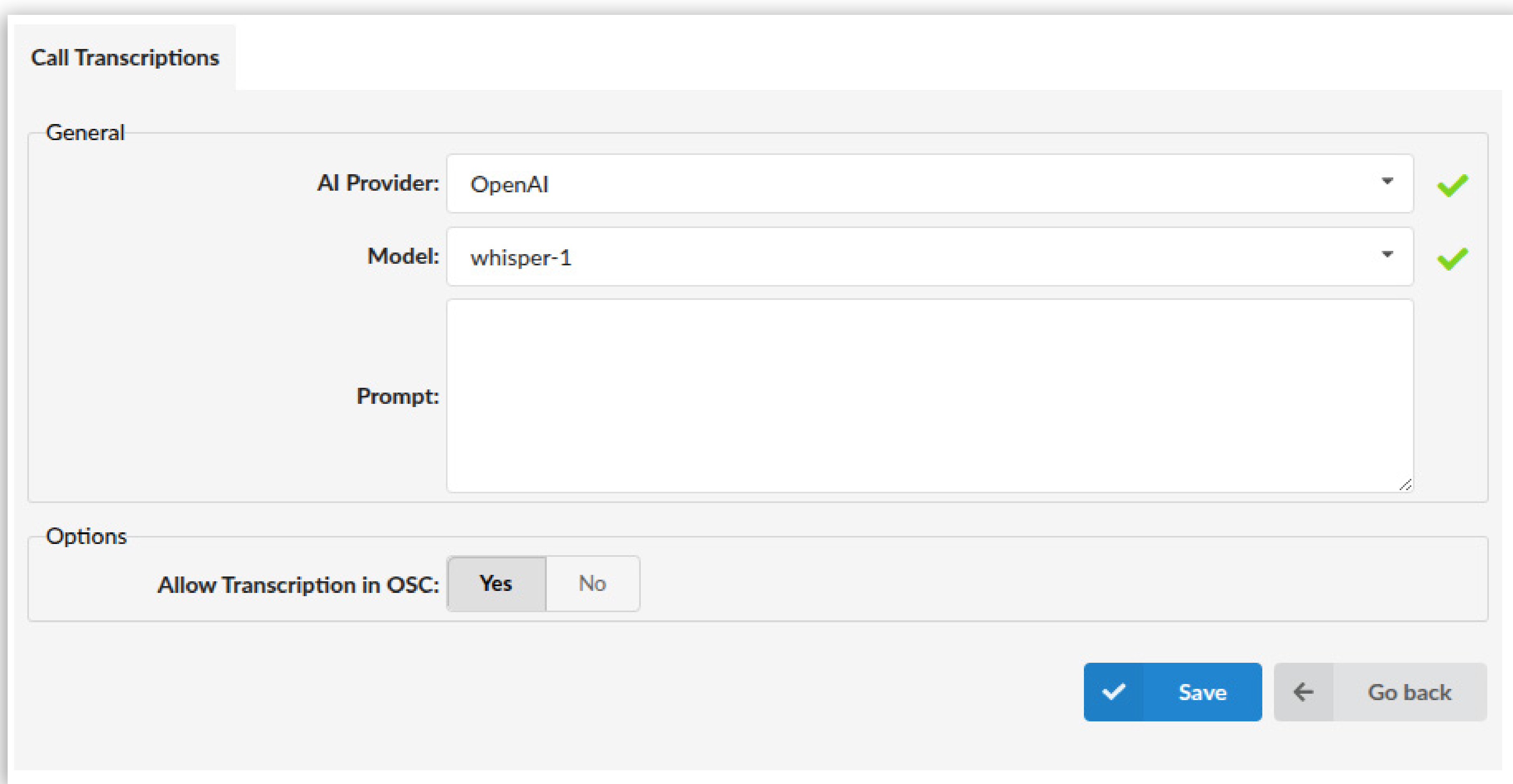
Call Transcription

The Call Recording Transcription feature that was implemented earlier has been moved to the AI Hub tab along with all the other AI features. It functions just as before, but the setup is going to be much easier. The administrator needs to select the model and the AI Provider (the AI Provider has to be added to the AI Providers page) and save the configuration. Supported providers for Call Recording Transcription are:

- Hosted Whisper
  Models: base, base-en, small, small-en, medium, medium.en, large-v1, large-v2, large-v3, tiny, tiny.en

- OpenAI
  Model: whisper-1

Optionally they can add a prompt in the Prompt text box.



WIth this new release, it is now also possible to allow transcriptions in OSC. Users which can access OSC, with 'Allow Transcription in OSC' can view transcriptions from their call recordings, or request a transcription to be generated.

Important Note: Since this has been moved to the AI Hub, it is a **breaking change**. Users that previously had Call Recording Transcriptions will have to re-configure it. Additionally, '**Use RAM disk**' option, which can be found in server settings, must be set to '**Yes**' in order for Call Recording Transcriptions to function.

Live Transcription

Live Transcription enables real-time speech-to-text for calls routed through an ERG or Queue. When enabled, PBXware captures the call audio and streams it to a supported AI transcription service (AWS Transcribe, Deepgram, Google Speech, or OpenAI). The selected service returns transcription results continuously during the call, and PBXware forwards that text in real time to the configured WebSocket Callback endpoint.

To use Live Transcription, Stereo Recording must be enabled, and recording must be enabled on the selected ERG, Queue, or Extension, as the feature relies on stereo call audio for processing.

Live Transcription is enabled per destination:

- Enhanced Ring Groups (ERG): Navigate to Ring Groups → Enhanced Ring Groups and enable the new Live Transcription option under Record ERG Calls.
- Extensions : Navigate to Extensions → System , click Show Advanced Options, and enable Live Transcription under Recording.
- Queues (Contact Center only): Navigate to Queues, click Show Advanced Options, and enable Live Transcription under Recording.

PBXware supports provider-specific request settings (each service exposes a slightly different configuration). In this document we use Deepgram as an example, where options such as Interim Results, Smart Format, Keywords/Keyterms, Punctuation, Profanity Filter, Redaction, and others can be enabled per request, depending on the selected model and provider capabilities.

WebSocket Callback (output)

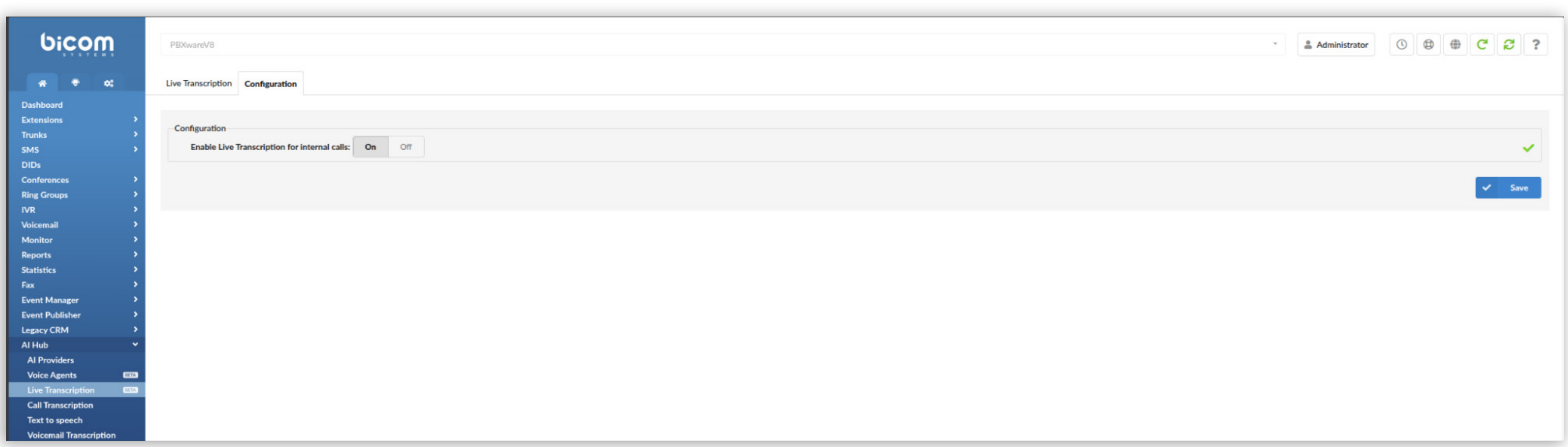
The WebSocket Callback defines the destination where PBXware sends the real-time transcription output. This endpoint must be a customer-provided WebSocket server (custom-built or hosted by the customer) that is reachable from PBXware and capable of receiving incoming transcription messages. PBXware uses the provided WebSocket URL to push transcription results in real time for live monitoring or further processing.

The optional Metadata field accepts JSON-formatted data that PBXware will include with every message sent to the WebSocket server. This is useful for correlating transcripts with call context (who called, which ERG/Queue was dialed, who answered, etc.). Metadata supports placeholders that will be populated automatically per call:

- %CALLER_ID% – Caller ID
- %CHANNEL_ID% – ID of the ERG/Queue that was called
- %CHANNEL_NAME% – Name of the ERG/Queue that was called
- %CALLEE_ID% – Extension that answered in the ERG/Queue
- %CALLEE_NAME% – Name of the extension that answered

Supported Live Transcription providers integrated in PBXware are AWS Transcribe, Deepgram, Google Speech, and OpenAI. PBXware supports Live Transcription through these providers and supports their currently available models at the time of release, while the available configuration options and model availability depend on the selected provider and may change over time. For more information about configuration steps and supported models, please refer to the official provider documentation.

Configuration



This option enables or disables Live Transcription for local calls.

Text to speech

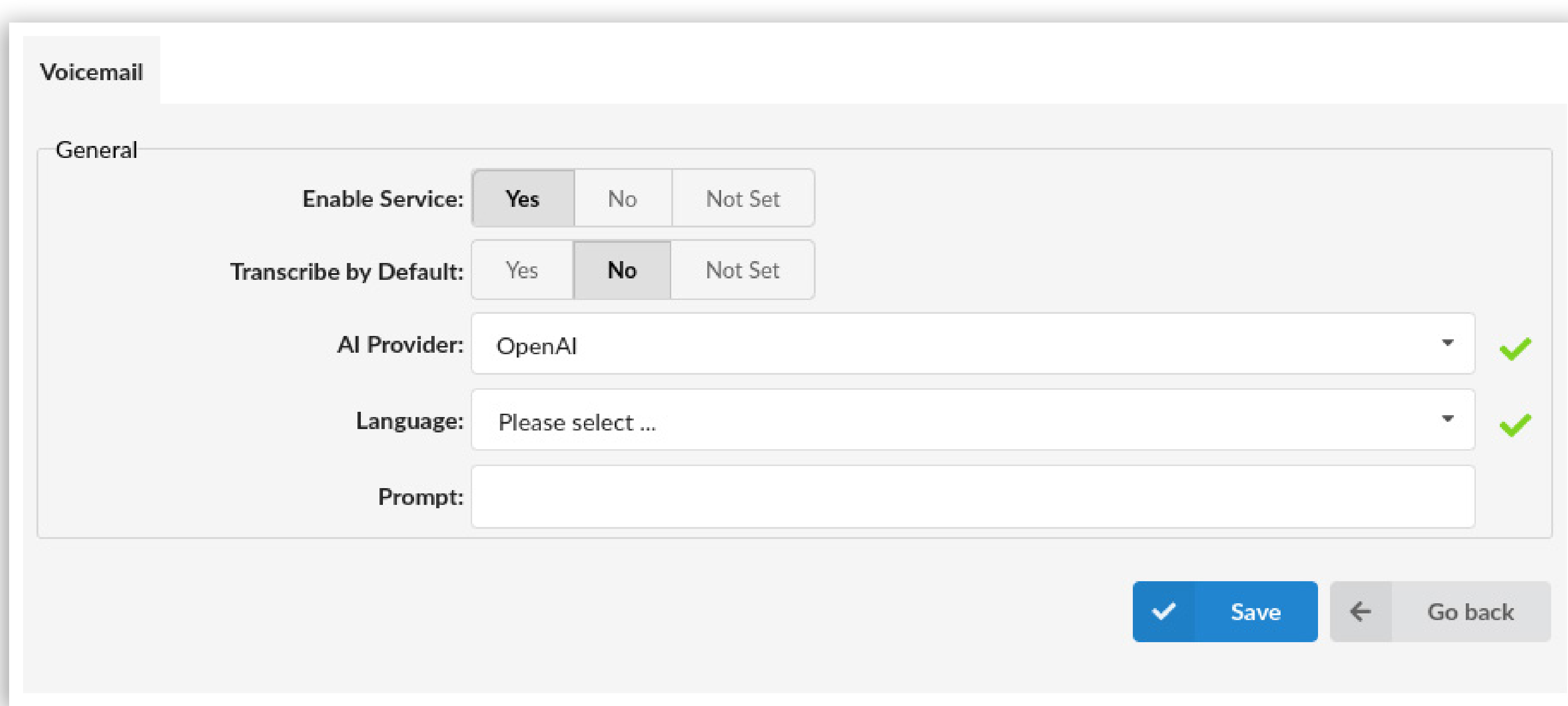
The Text to speech configuration has also been moved to the AI Hub section for easier management. It can now be configured much more easily, and per tenant. All it takes is toggling whether the feature should be enabled, and selecting a Provider from the dropdown menu of all available providers on the system/tenant.

Currently supported AI Provider for Text to Speech functionality is OpenAI.

Important Note: Since this has been moved to the AI Hub, it is a **breaking change and will need to be reconfigured in case it was already set up prior to PBXware version 8.0.**

Voicemail Transcription

Voicemail Transcription configuration settings have been moved to the AI Hub as well, while still functioning as before. This allows administrators to configure this feature per tenant, and manage it more easily. This way the service can easily be enabled/disabled, and the provider can be switched in a more simple manner.

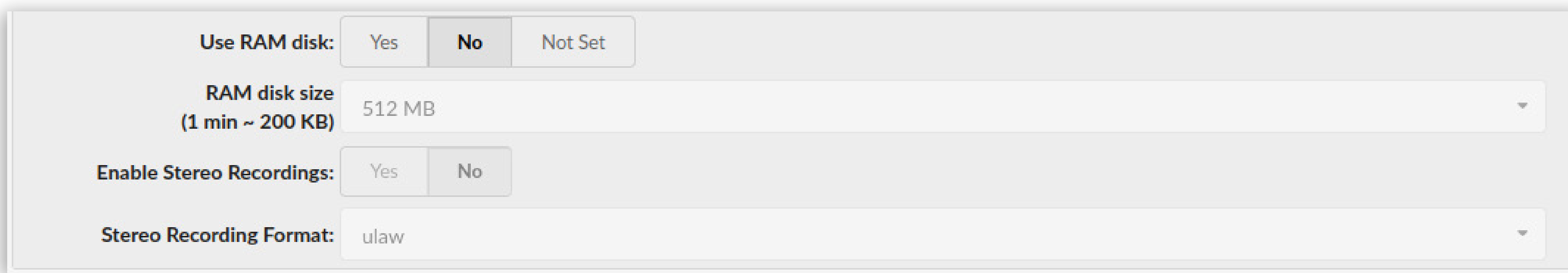


Currently supported AI providers for this feature are Hosted Whisper, IBM, Google Speech and OpenAI.

Important Note: Since this has been moved to the AI Hub, it is a **breaking change and will need to be reconfigured in case it was already set up prior to PBXware version 8.0.**

Stereo Recordings GUI configuration and resource usage

With PBXware 8.0, Stereo Recordings will be fully configurable through the GUI. On MT systems, if Stereo Call Recordings are enabled on the system, they will automatically be available on all tenants, otherwise they can be enabled on specific tenants. Enabling Stereo Recordings and choosing the Stereo Recording Format **requires** RAM disk to be enabled, and ‘Auto MP3 Convert’ to be set to either ‘Convert and keep original’ or ‘Convert and remove original’.



Stereo call recordings previously required significantly more temporary storage and memory since Asterisk simultaneously created **three files per call**: mixed (mono) file, left channel, right channel.

This resulted in nearly **3× higher RAM disk and IO usage**, which caused issues in hosted environments with limited RAMFS capacity (e.g., 512 MB). This release introduces optimisations that significantly reduce RAM, disk, and I/O usage during active recordings while preserving **audio quality and existing functionality**.

With this release, the mixed (mono) file was removed. When stereo recording is enabled, the mono mixed file is **no longer written**, **and** only **left** and **right** channel files are recorded and stored. Since the mono file was always overwritten during post-processing anyway, removing it eliminates unnecessary writes.

Currently supported encodings are μ-law, WAV, and WAV16, with WAV as the default encoding. Stereo recordings do increase memory usage compared to mono sound files, but the increase is small (sub-3 MB). When the RAM disk is on; the audio quality does not change and it does reduce RSS growth.

Enabling stereo recording will increase system resource usage:

- Recordings will be stored as separate left and right channel files
- CPU usage may increase during recording processing

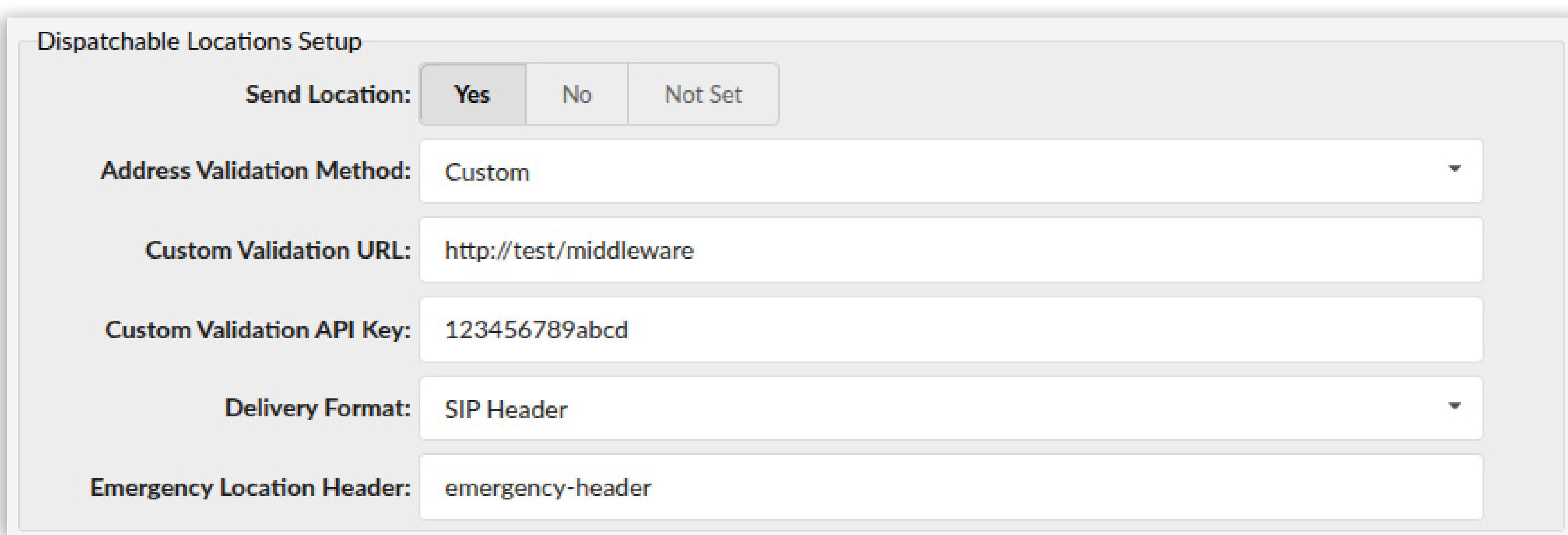
Below is an example usage based on a 1-minute WAV recording:

Baseline (mono WAV)	μ-law (ulaw)	WAV (2 files)	WAV16 (2 files)
RAM: ~0.93 MB Storage: ~0.73 MB	RAM: ~0.90 MB (~-4%) Storage: ~1.17 MB (~+60%)	RAM: ~1.8 MB (~+93%) Storage: ~2.2 MB (~+200%)	RAM: ~3.8 MB (~+307% Storage: ~4.1 MB (~+460%)

The WAV encoding doubles storage compared to μ-law, with no intelligibility gain for voice, whereas WAV 16 quadruples storage and I/O load, yet still produces the same MP3 size. μ-law offers the best trade-off between: Storage efficiency, IO pressure, RAM usage and voice quality.

Custom address validation via middleware for Dispatchable Locations

In PBXware 8.0, We are introducing enhancements to PBXware's emergency calling capabilities to help partners meet regulatory requirements (including the Ray Baum's Act) and streamline dispatchable location management with Bandwidth and other SIP providers.



New fields have been added to 'Dispatchable Locations Setup' to enable custom middleware apps for emergency calls. One of those new fields is 'Address Validation Method', where the administrator can choose either the standard USPS method or a custom method, which requires custom middleware to perform the validation.

Once the 'Custom' option is selected for validation, additional fields will appear to provide more options. 'Custom Validation URL' and 'Custom Validation API Key' will depend on your application, and will need to match the credentials for it. Additionally, Delivery Format can be picked between two options: 'SIP Header' and 'PIDF-LO'.

Note: 'Emergency Location Header will be used as the header name for sending locations when the chosen Delivery Format is 'SIP Header'.

In an MT system, configuration can be made tenant-specific, but if nothing has been set tenant-wise, tenants will follow the system's configuration.

The 'Emergency Location Header' field has been moved to the 'Dispatchable Locations Setup' section for easier configuration.

CRM Integration

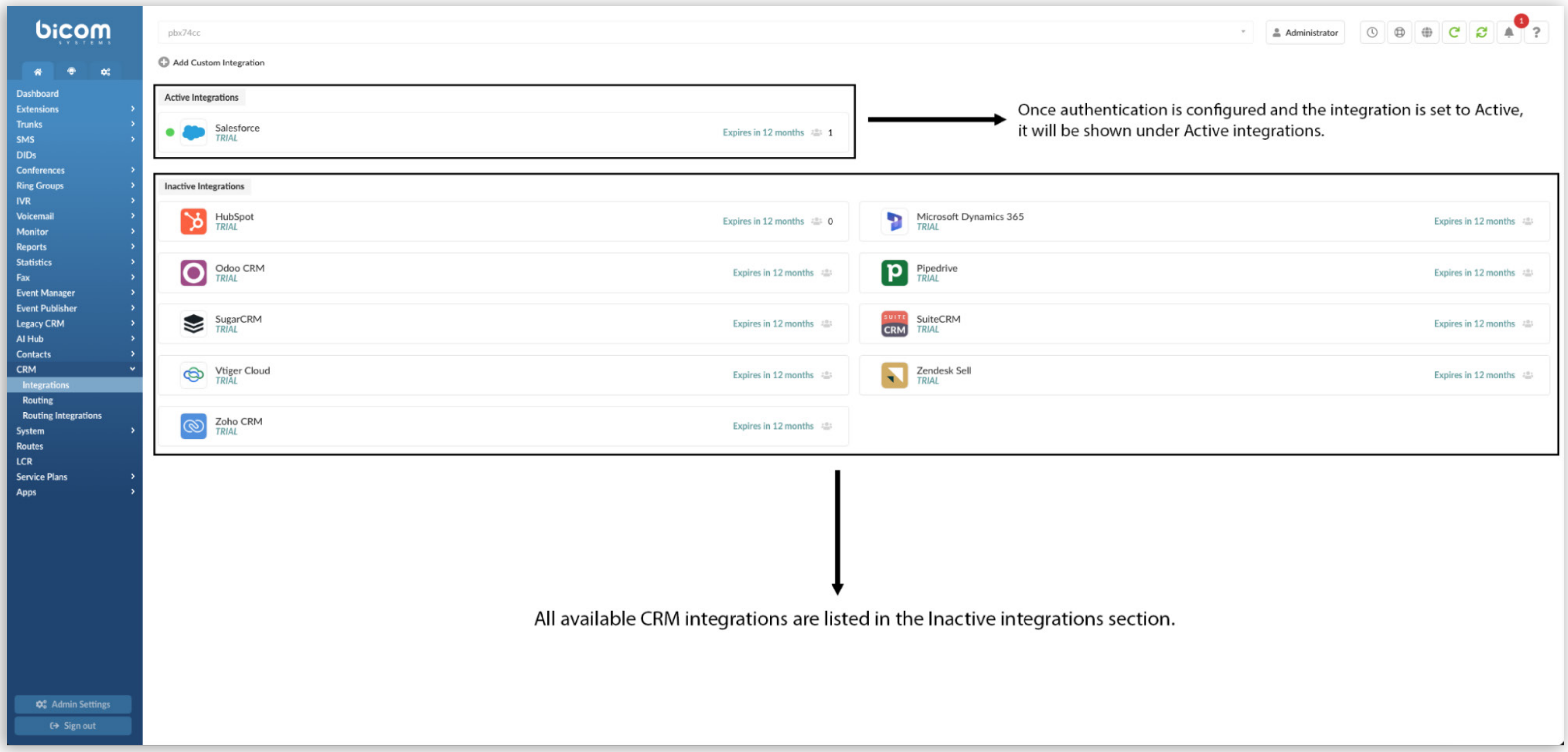
PBXware v8 introduces a new, completely redesigned CRM Module, available in gloCOM Web and the new gloCOM Desktop application (v8). This new module is built on the CRM Connector (crmconnector) backend service and a unified gloCOM CRM Widget UI layer, providing improved flexibility, stability, and user experience.

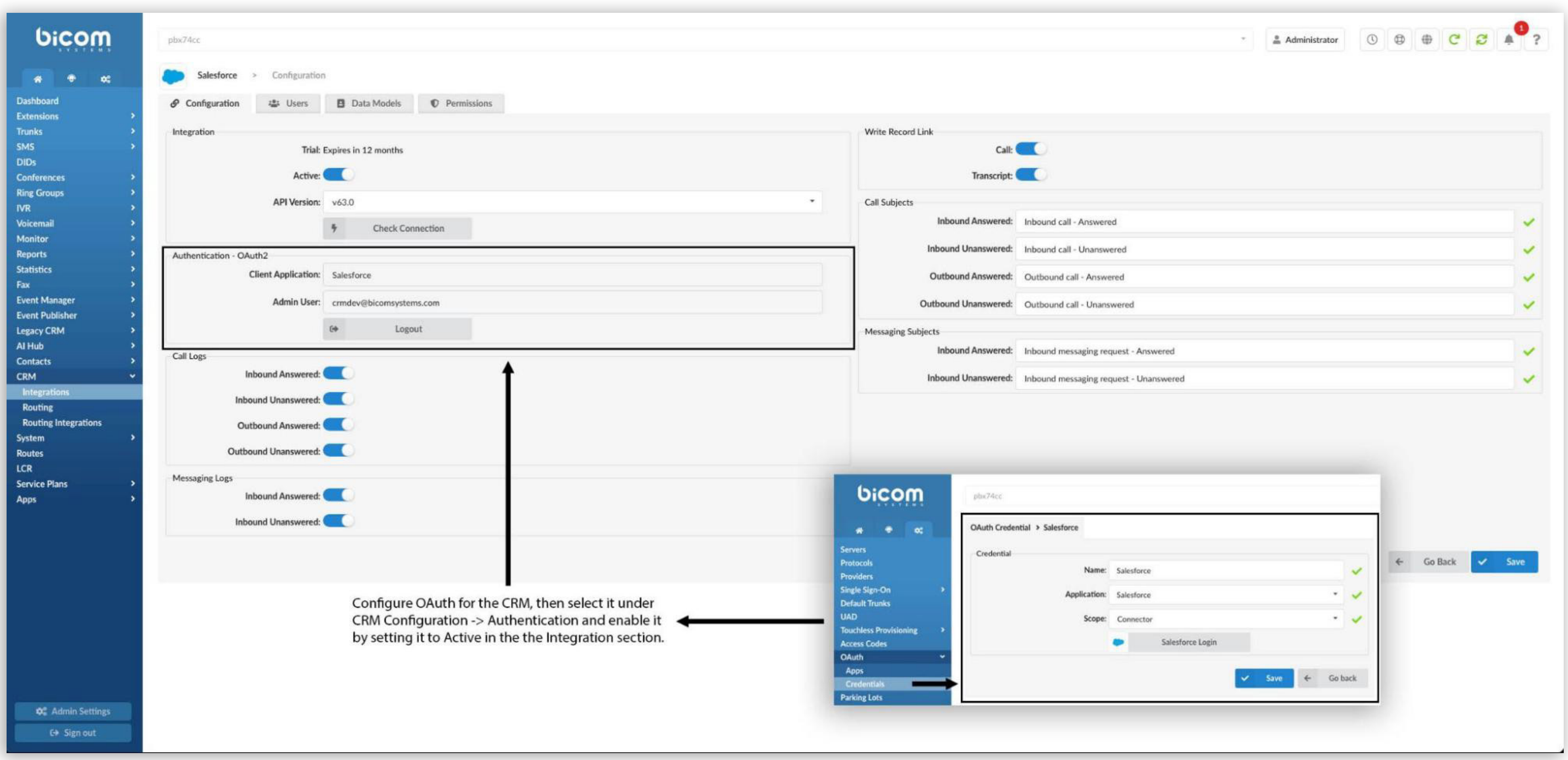
Important: The legacy CRM integration module is not being replaced or removed. It will continue to function in PBXware v8 with the current (legacy) gloCOM Desktop application, ensuring full backward compatibility.

Key Improvements:

- **Multiple CRM support per tenant/system** – Supports multiple CRM integrations per tenant (Multi-Tenant edition) and per system (Contact Center and Business editions).
- **Flexible authentication methods** – Supports OAuth2, basic authentication, and provider-specific login methods.
- **Permission Groups** – Allows administrators to define per-integration permission groups that control which CRM data models (Account, Contact, Lead) each user can access, create, and edit.
- **Configurable CRM data model** – Full CRM configuration via PBXware → CRM → Integration Settings, including customizable record types (Contacts, Leads, Accounts) and configurable visible and required fields within gloCOM Web and Desktop applications.
- **Unified CRM Widget** – Enables automatic record matching based on phone number, email, or other supported matching criteria (depending on the CRM integration), manual search, record matching/rematching, creation of new CRM records, and adding subject and notes during calls and conversations.
- **Automatic conversation logging (Voice & Messaging)** – Logs only the final outcome of completed conversations, including call recording links, transcript links (where available), subject, and notes, across deskphone, desktop, web, and mobile devices.

- **Expanded CRM integrations** – Built-in integrations include Salesforce, Zoho CRM, HubSpot, Microsoft Dynamics 365, Odoo (new), Pipedrive, SuiteCRM, Vtiger, Zendesk Sell, and SugarCRM, with enhanced Custom CRM Provider support.





CRM Routing

CRM Routing introduces data-driven inbound call routing, allowing incoming calls to be automatically directed based on customer information stored in connected CRM systems. Instead of relying only on static IVR logic, routing decisions can now depend on real CRM data.

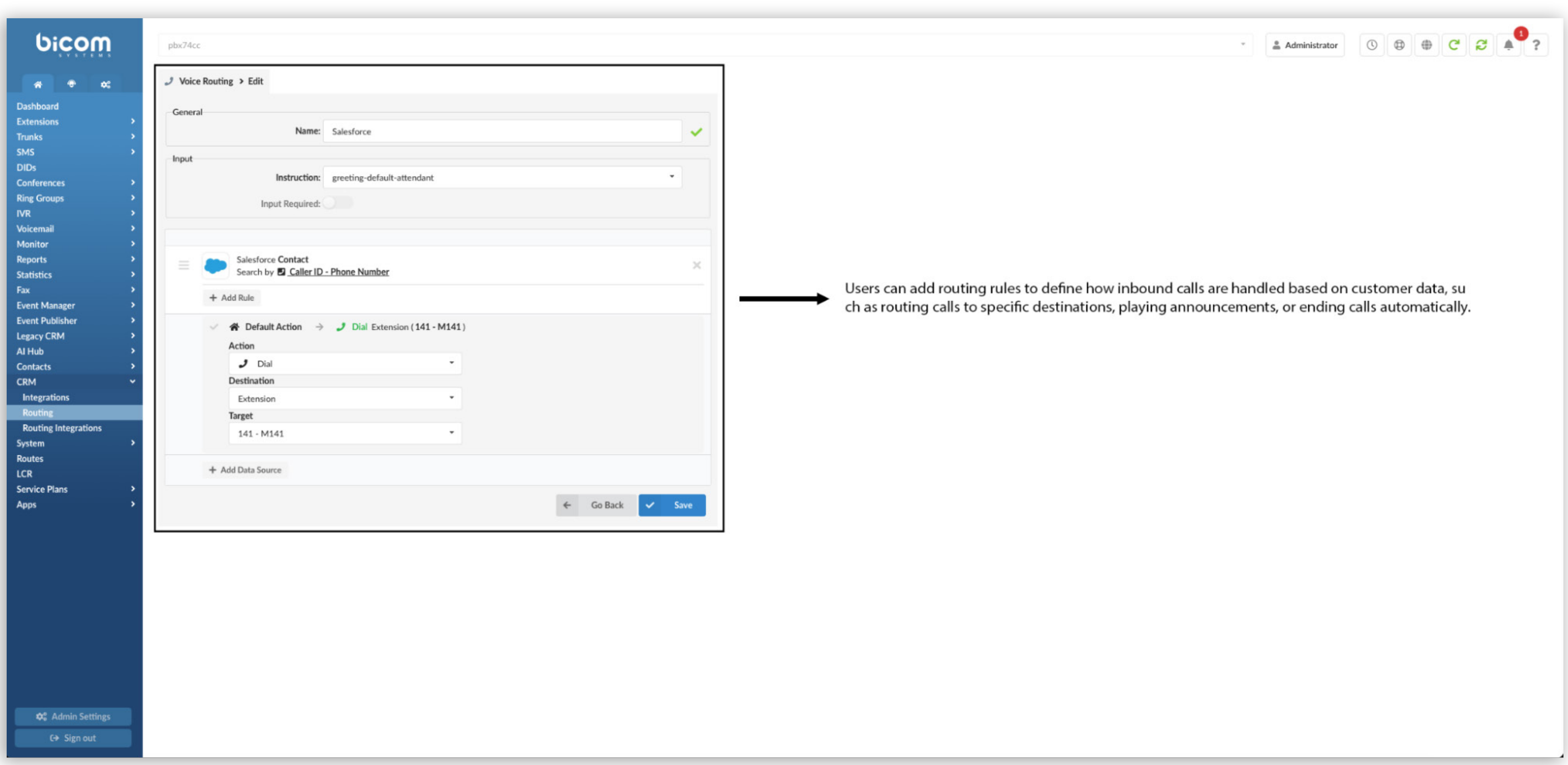
Administrators can configure flexible routing rules in PBXware that define what should happen when a recognized customer calls — for example, routing the call to a specific extension or account manager, forwarding it to an external number, playing an announcement, or ending the call. This enables smarter and more personalized call handling tailored to different customer types and scenarios.

CRM Routing is configured centrally in PBXware and works consistently across supported CRM integrations. A built-in preview and testing option is available, allowing administrators to validate routing logic and rule behavior before activating it in production.

Example: If a lead who is identified in the CRM as a CEO calls the company, the system can automatically route the call to their assigned account manager instead of sending them through the standard IVR flow.

For organizations using CRM platforms not natively supported by PBXware, **Routing Integrations** allows administrators to connect custom external APIs as data sources for CRM Routing. Each integration is configured with an endpoint URL, HTTP method, custom headers, and field mappings that define how incoming caller data is matched against the external system.

When a call arrives, PBXware queries the configured endpoint in real time and uses the response to drive routing decisions. This makes CRM Routing extensible to any third-party CRM, customer database, or internal service that exposes an HTTP API.



Contacts

Unified Contacts - Major Enhancement (v8)

Unified Contacts is now supported across all editions and fully replaces the legacy Central Phonebook. The new Contacts module introduces three structured **contact visibility levels**:

- **Global** – visible company-wide
- **Department** – visible only to selected departments
- **Private** – visible to the user who created the contact and to the system/tenant administrator

This structured visibility model enables more controlled and flexible contact management across the organization.

Centralized Contact Management (PBXware)

Within the **Contacts module in PBXware**, users can:

- Upload CSV files with different contact visibility levels (Global, Department, Private)
- Use smart CSV sync to prevent duplicates — existing contacts are updated and new contacts are added (contacts are not deleted during upload)
- Filter contacts by visibility level or source
- Manage contact information and visibility settings directly from the Contacts module

Uploading private contacts on extension level is now deprecated. Instead, private contacts are managed centrally within the Contacts module and can be assigned to extensions by administrators.

An additional **Department Admin role** has been introduced (enabled per extension), allowing users to manage contacts for all departments they belong to.

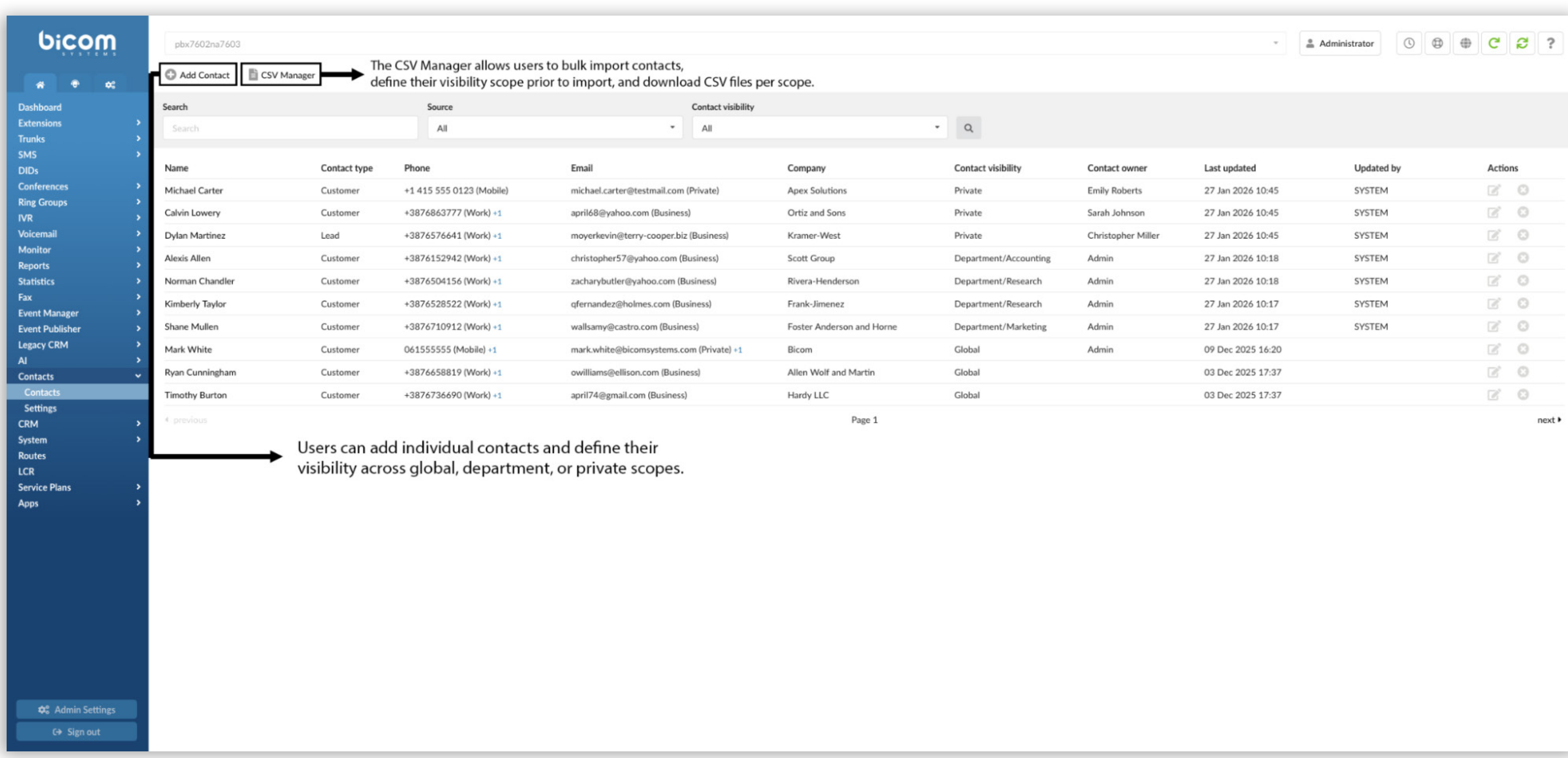
Contact Center Enhancements (CRM Lite Capability)

For Contact Center users, contacts can include conversation history across channels. With permission-based access and cross-channel history, the Contacts module now provides a **CRM Lite–style experience**, enabling smaller Contact Center customers to benefit from structured customer context without requiring full CRM licensing.

Business edition users continue to have structured contact management without cross-channel conversation history.

Deprecations & Compatibility Notes

- The legacy **Central Phonebook** is deprecated and replaced by the new Contacts module.
- Contacts management in OSC will continue to work, but with limited functionality (primarily private contacts only).
- For advanced contact permissions, structured visibility, merging, CSV smart sync, and cross-channel history, users must use gloCOM Web or the new Desktop application.
- Deskphone-only users may experience reduced contact editing functionality compared to previous behavior.



CC Lite

Contact Center Lite Package

The new Contact Center Lite package is available across all editions, with primary licensing focus on Multi-Tenant and Business editions. The package introduces enhanced monitoring and reporting capabilities.

Contact Center Lite includes:

- Wallboard and additional dashboards for Enhanced Ring Groups (ERG)
- Scheduled Reports for ERG and Extensions statistics
- Enhanced Ring Group - Callback feature

Wallboard & ERG Dashboards

Wallboard and ERG dashboards are available in:

- gloCOM Web (v8)
- gloCOM Desktop / Electron (v8)

Company/tenant administrators can enable ERG dashboard access per extension, allowing users and managers to monitor specific or all ERGs.

Scheduled Reports (ERG and Extensions - v8)

Scheduled Reports for ERG and Extensions statistics are available across all editions.

Administrators can configure automatic email delivery of ERG or Extension statistics to selected stakeholders.

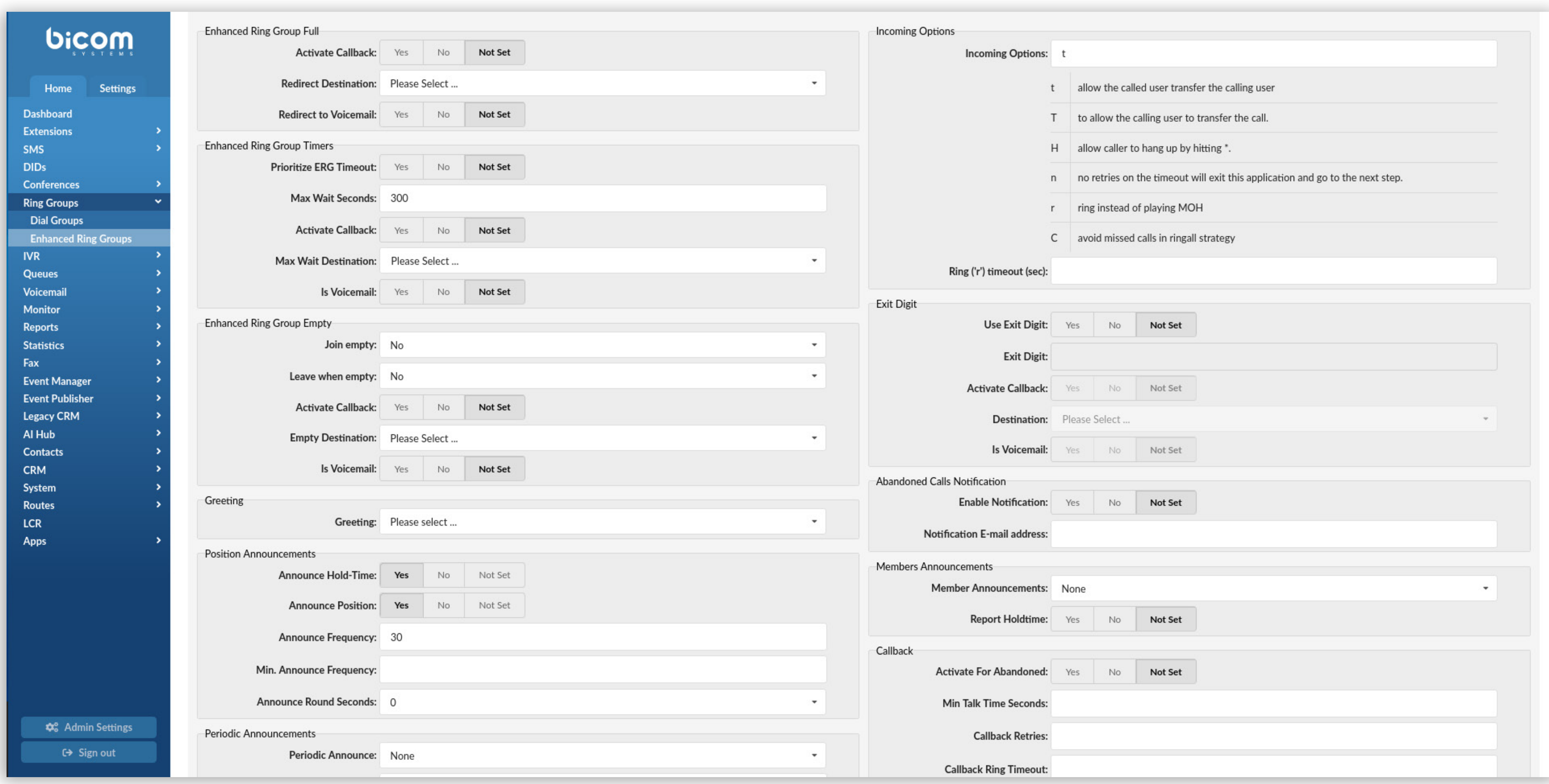
Reports can be customized by:

- Time period (hourly, monthly, yearly, or specific day(s))
- ERGs
- Members

- Waiting time and talk time
- DIDs
- Customers

ERG Callback

With the release of PBXware v8, the Callback feature is available within Enhanced Ring Groups. Callback enables callers to hang up and receive a return call once a member becomes available, eliminating the need to wait in queue. Depending on the Enhanced Ring Group configuration, the feature can be triggered by a variety of scenarios, from caller-side events such as exiting via a dedicated digit or an abandoned call, to system-side conditions such as a full or empty ERG, among other configurable triggers. Each trigger can be independently enabled or disabled by the administrator.



Voice Statistics - Depreciation Notice

With the release of PBXware v8, legacy Voice Statistics have been deprecated. All reporting functionality has been migrated to the new Statistics page, which now serves as the unified reporting engine for both voice and omnichannel data.

The new Statistics page introduces improved architecture, enhanced performance, and expanded reporting capabilities.

What's New in Statistics (v8)

New and enhanced reports include:

- Queue Conversations per Disposition
- Voice Entry Positions
- Queue Callback Reports
- Agent Occupancy Report (Voice)

Reports now also support filtering by DID and Customer, making it easier to narrow down results across all available reports.

In addition to these new reports, new metrics have been implemented in alignment with industry standards, ensuring improved accuracy and consistency across statistics.

All reports now run on ClickHouse, providing significantly improved performance, scalability, and reliability compared to the legacy Voice Statistics engine.

Important Notice

Legacy Voice Statistics will remain accessible for backward compatibility; however:

- They are officially deprecated
- They will no longer be maintained
- No new features or improvements will be added

We strongly recommend using the new Contact Center (CC) Statistics page to benefit from improved performance, enhanced metrics, omnichannel reporting capabilities, and future updates.

Contact Center Edition - Scheduled Reports Update

Scheduled Reports for Contact Center statistics (Agent, Queue, and Dialer reports) have been enhanced.

Reports can now be scheduled on:

- Hourly basis
- Monthly basis
- Yearly basis
- Specific day(s) basis

The previous daily and weekly scheduling options have been replaced with the more flexible "specific day(s)" option, allowing administrators to define exactly when reports should be sent.

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