

RELEASE NOTES PBXware 7.6.0



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Features

PBXware 7.6.0 introduces a set of new features and enhancements aimed at improving flexibility and functionality. Most importantly, asterisk and pjproject have been updated to ensure stability and compatibility. Call forwarding destinations are now configurable based on an extension's gloCOM presence status. With OSC permissions, extensions can define their own forwarding rules via OSC, allowing for fully customizable presence-based call forwarding. This release introduces integration with Generic OpenID Connect (OIDC) Providers, enabling users to configure custom options for Single Sign-On (SSO) login in PBXware. The SMS messaging experience has been improved to support Opt-in/Opt-out and Help keywords in conversational messaging, which was previously limited to bulk messaging only. A new voicemail transcription service option is also available, providing greater versatility. Additionally, a hidden stereo call recording feature has been introduced in this release and is currently still in the beta phase. To address previous issues, support for longer call recording transcriptions has also been added.

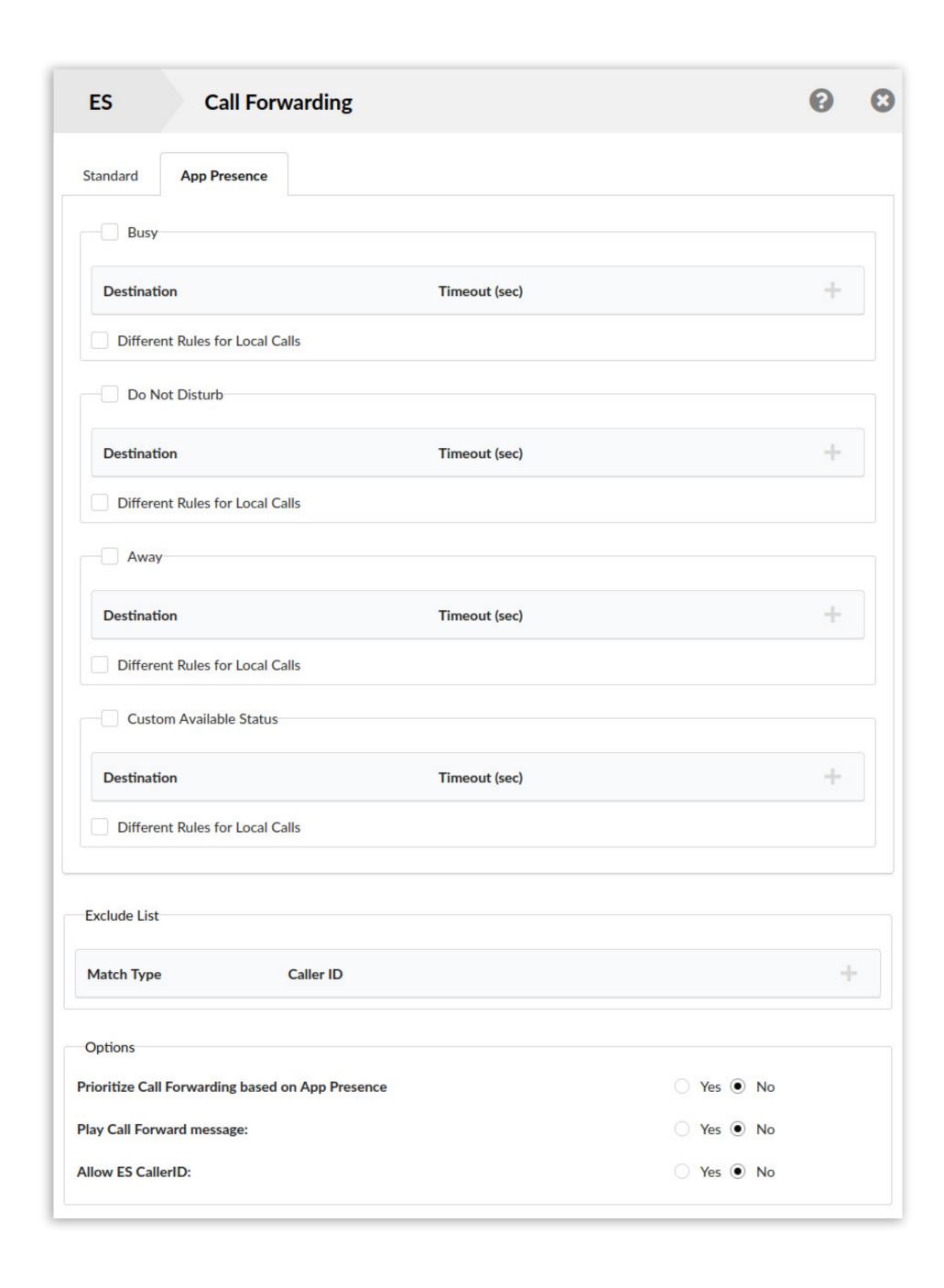
Asterisk and pjproject update

In addition to the new features, Asterisk and PJSIP have been upgraded to ensure continued stability, security, and compatibility. **Asterisk** has been updated to **version 18.26.1**, and **pjproject** to **version 2.15.1**. These updates include important bug fixes, performance improvements and security patches, to ensure continued reliability in production environments.

Call Forwarding rules based on App presence

Destinations for call forwarding based on the app presence status can now be configured within the **Enhanced Services** page of an extension, under **Call Forwarding** settings.

To preserve the existing call forwarding functionality, the Call Forwarding section is now divided into two tabs: **Standard** and **AppPresence**. The **Standard** tab contains all previously available forwarding options, while the **App Presence** tab introduces new settings that allow call forwarding based on app presence statuses.



Within the **App Presence** tab, administrators can define forwarding destinations for default app statuses - Busy, Do Not Disturb, and Away - as well as any custom presence statuses created under Apps → Statuses. When an extension is online and logged into any gloCOM client, and its presence changes from Available to a status that has call forwarding rules configured, those rules will be applied automatically.

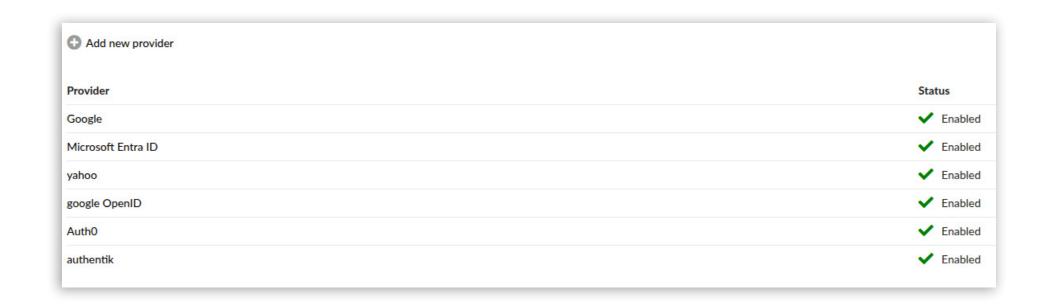
For each presence status, up to 10 forwarding destinations can be set for both local and external calls.

A new option, **Prioritize Call Forwarding based on App Presence**, has been added to control the precedence of forwarding rules. When set to Yes, App Presence rules will take priority over standard forwarding rules. The system will first check if the current app presence matches any configured rules; if a match is found, those rules will be applied immediately. If not, standard rules will apply. The default value for this setting is **No**, which means standard rules are prioritized by default.

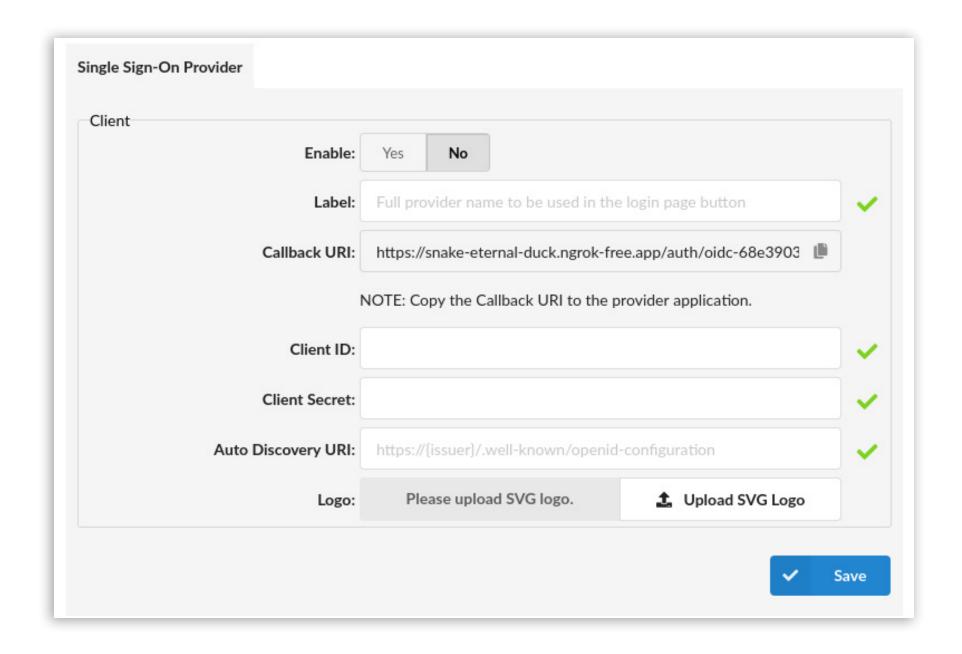
Additionally, extensions with the **Allow in OSC** permission can manage their own call forwarding rules - for both the Standard and App Presence tabs - directly via OSC.

SSO: Generic OpenID Connect (OIDC) Provider Integration with PBXware

In PBXware 7.6.0, a new option has been introduced which allows administrators to easily configure and enable additional Single Sign-On (SSO) providers through the Settings \rightarrow Single Sign-On \rightarrow Providers page, where all available SSO providers are listed.



Clicking the "Add new provider" button opens a setup page intended for configuring the new SSO option.



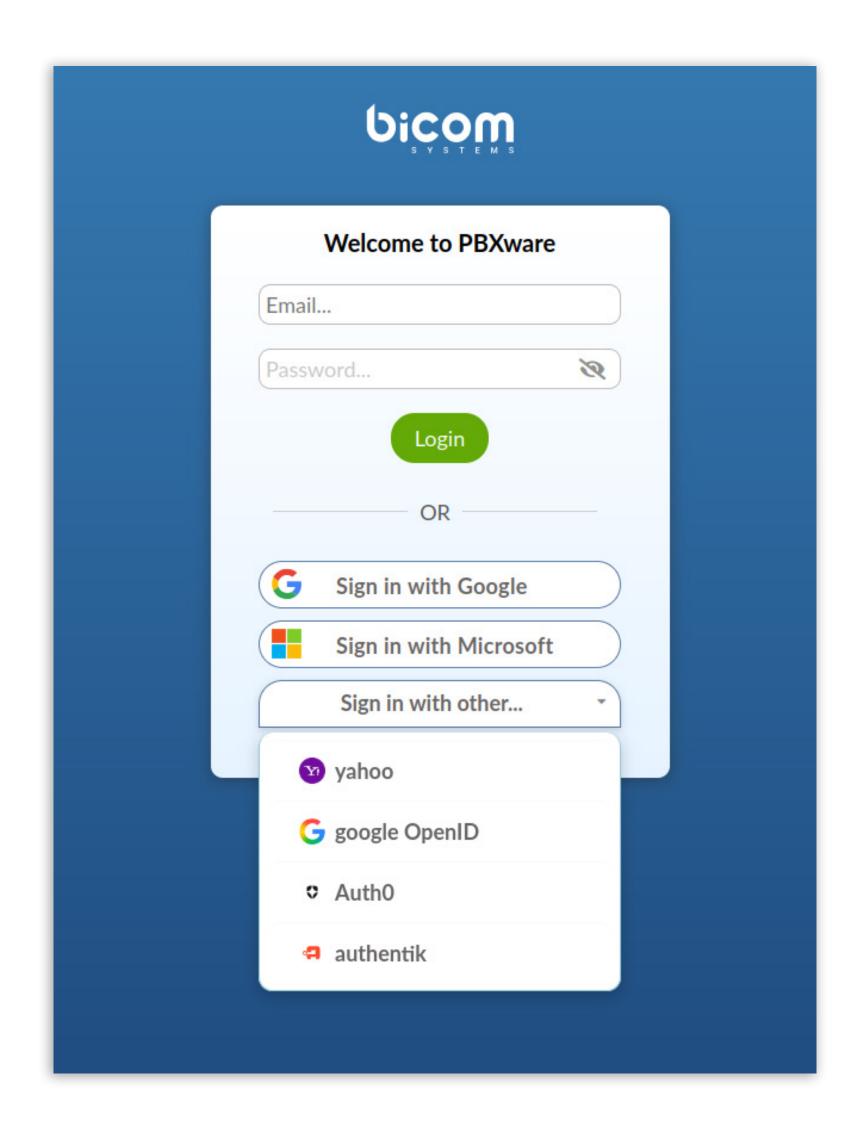
On the configuration page, administrators can:

- Enable or disable the provider using the "Enable" field by setting it to "Yes" or "No".
- Enter a "Label" for the provider, which will appear on the login page
- Provide the "Client ID" and "Client Secret" obtained from the provider's application.
- Copy the generated "Callback URI", which is automatically created based on the server settings and can be used in the provider's application.
- Fill out the "Auto Discovery URI" using the placeholder as a template, depending on the provider that is being configured.
- Upload an SVG logo to display on the PBXware login page for this SSO option by clicking the "Upload SVG Logo" button.

Note: Some providers do not support the ./well-known/openid-configuration endpoint and cannot be configured.

Custom providers can be edited and deleted, while the default **Google** and **Microsoft Entra ID** providers can only be edited.

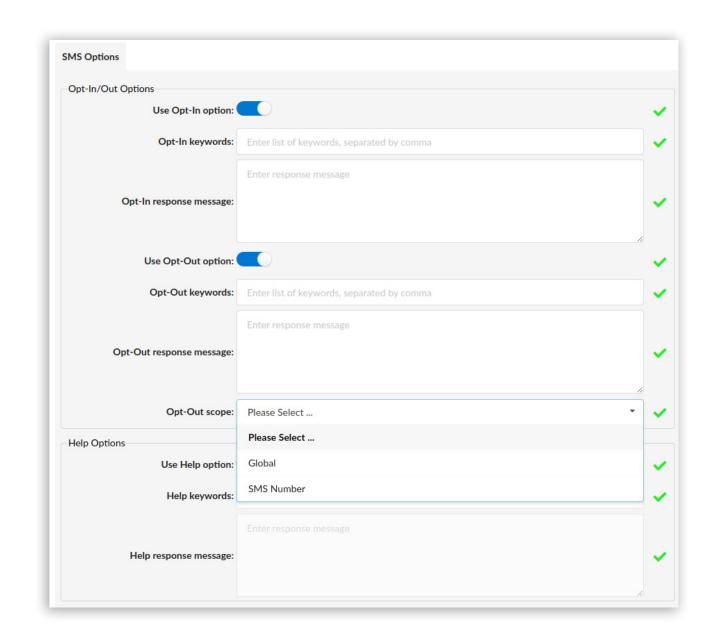
Once configured correctly, custom providers will appear as options on the PBXware login page.



Release Notes

Opt-Out/In and Help keywords for non-Bulk messaging

In PBXware 7.6.0, support for Opt-out, Opt-in, and Help keywords has been extended to conversational SMS messaging. Previously, these keywords were only available for use in bulk messaging. With this update, they can now also be used in one-on-one (conversational) SMS messaging at any time during a direct SMS conversation.



On the SMS Settings Options page, a new drop-down menu called Opt-Out Scope has been added, which will be applied to both Bulk messaging and conversational SMS messaging. This setting controls how the system handles Opt-Out requests from recipients who reply to SMS messages using predefined Opt-Out keywords. It determines whether the Opt-Out action will apply to a specific From Number or to all SMS numbers associated with the tenant.

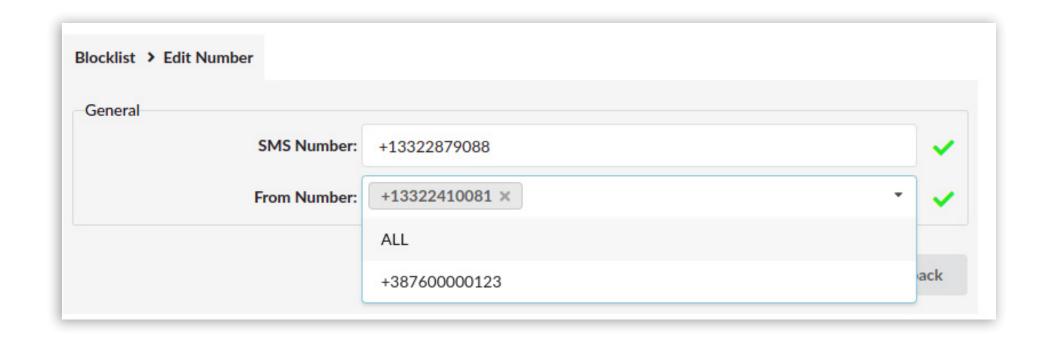
You can choose between the following two options:

Global

When set to **Global**, replying with an Opt-Out keyword will block the recipient's number from receiving SMS messages from **all SMS numbers** on the tenant. The number will be added to the Blocklist with the "From Number" set to **ALL**.

SMS Number

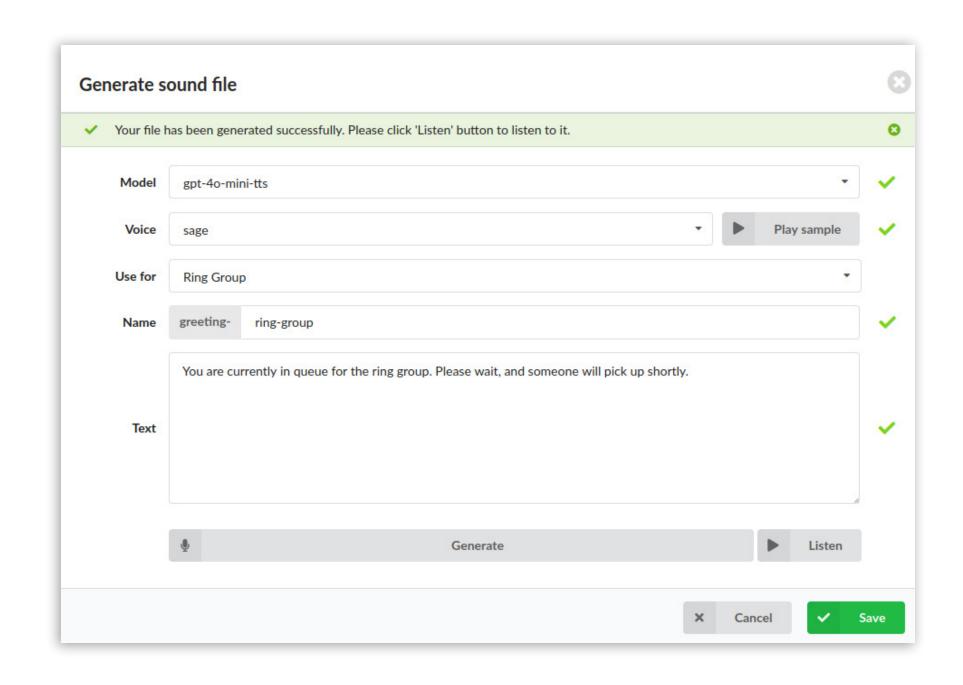
When set to **SMS Number**, the Opt-Out action will only block messages from the specific SMS number the recipient replied to. The number will be added to the Blocklist with the "From Number" set to the number that received the Opt-Out message. The recipient can still receive messages from other SMS numbers on the tenant.



Additionally, an option to edit blocked numbers has been added to the **SMS** → **Blocklist** page. Administrators can now modify existing numbers by selecting a number from the new dropdown menu, **From Number**. This allows them to specify whether the blocked recipient should be prevented from receiving messages from a specific SMS number, multiple numbers, or **all** SMS numbers on the tenant.

Al Voice Greeting generation (Text to Speech)

With this release, administrators can generate AI voice recordings from text using the **OpenAI Text to Speech API**. These recordings can be assigned to IVRs, ring groups, voicemails, and more. The feature is available through the "**Generate sound file**" button under **System → Sound files**. Clicking the button opens a pop-up where the administrator can enter details for the sound file they want to create.



Users will be prompted to select a Model and the specific voice they wish to use, which can be previewed instantly by clicking the "**Play sample**" button. This enables administrators to tailor the sound file to their exact needs.

When generating the file, users can also specify its intended purpose so the system can automatically create an appropriate file name.

Within this pop-up, there is a text section where administrators can type out the greeting or message that they want to be played. Before saving, they can listen to the generated audio and make adjustments as needed until they are satisfied with the final result.

Note: for any new changes, users have to click "Generate" again in order for the changes to apply. After pressing "Save", the file will be saved and available to use within the sound files.

OpenAl for Voicemail Transcription

With this new release, OpenAl can now be used as a provider for voicemail transcriptions. When selected, users will be prompted to enter their API key, choose the language, and provide a prompt.



The voicemail transcription feature continues to function as before: if 'Enable Service' is set to 'No' or 'Not Set', voice messages will not be transcribed.

Call Recording Transcription Support for Longer Calls

To improve transcription accuracy for longer calls or recordings that exceed provider file size limits, **audio chunking** has been added. The system now automatically detects when a recording exceeds the limit that a provider can handle and **splits the audio into smaller**, **appropriately sized chunks**, enabling complete and accurate transcription even for extended calls.

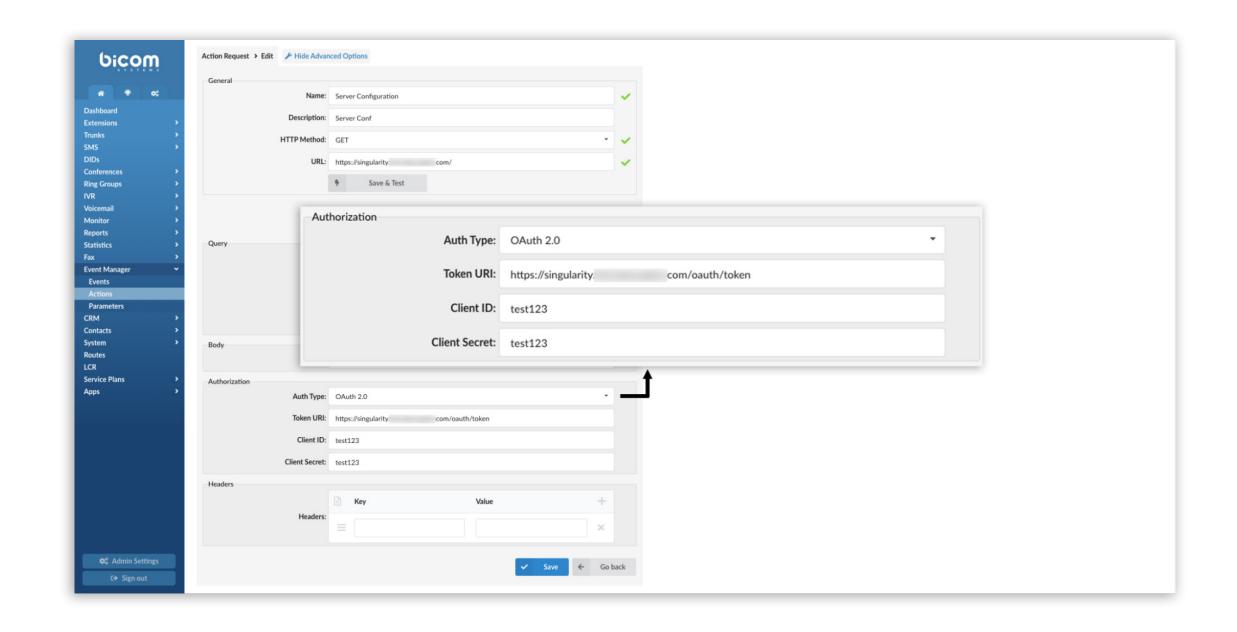
Stereo Call Recordings (Beta)

A new hidden option has been introduced to enable **stereo call recordings**, allowing each side of the conversation to be recorded on a separate audio channel, which unlocks several benefits, such as **Improved Al readiness**, enabling advanced tools for speech analytics, speaker diarization, and transcription; and **Better integration** with Al, CRM, and analytics systems. These recordings follow the same retention and deletion policies as standard (mono) recordings. The feature can be enabled either globally (for all tenants) or individually per tenant. As this functionality is still in the beta state, it remains a hidden option for now. To have this option enabled, please contact our customer support.

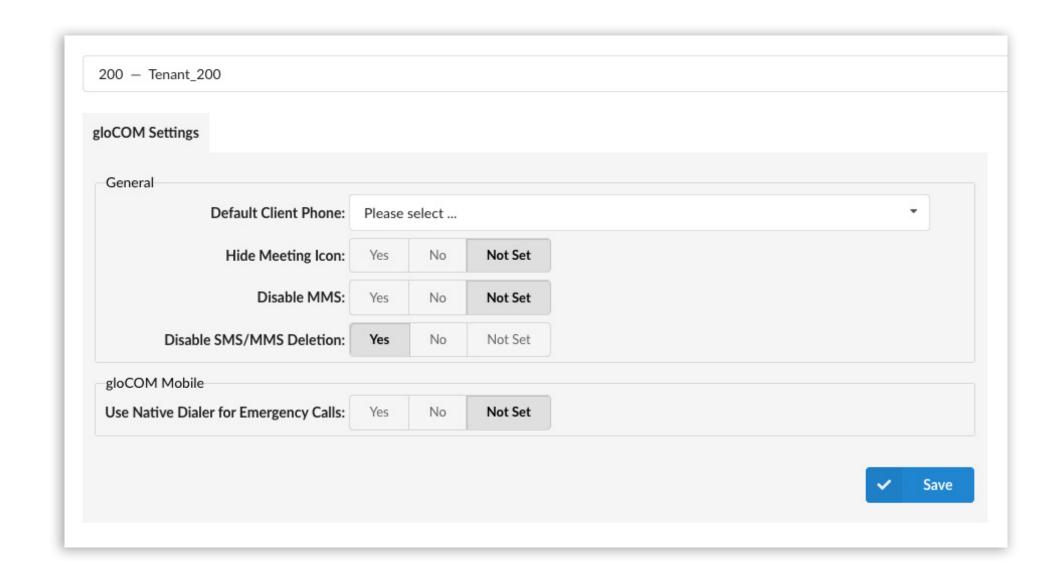
Note: If stereo call recordings have been enabled, RAMFS should be increased to at least 256 MB since stereo recordings are much larger in file size than mono recordings. The final MP3 recordings are typically 2-3 times larger than mono recordings.

Event Manager: Added OAuth 2.0 Support

Event Manager now supports **OAuth 2.0** as an authentication method for outgoing webhook actions. When this option is selected, PBXware automatically obtains an access token by using the configured **Token URI**, **Client ID**, and **Client Secret**, and includes the token in the request header. This enables secure integration with third-party APIs that require dynamic token-based authentication. The token is automatically refreshed when it expires, reducing the need for manual handling. This new method complements the existing authentication options: No Auth, API Key, Basic, and Bearer Token. With this enhancement, the Event Manager becomes more adaptable and secure for modern automation workflows.



Disable SMS messages and conversation deletion



For security reasons, a new setting allows administrators to prevent users from deleting SMS/MMS messages or conversations in gloCOM. The option can be found in the **gloCOM** → **Settings** page, where SMS/MMS message and conversation deletion from gloCOM apps can be controlled by selecting either "**Yes**", "**No**" or "**Not Set**" for the **new "Disable SMS/MMS Deletion" setting**. This option can be set at both the master and tenant levels, where the tenant level takes precedence over the master tenant.

New Endpoints







Htek UCV50



Poly Edge E300



Flyingvoice P11



Flyingvoice P11W



Htek UCV53



Poly VVX150



Flyingvoice P11G



Gigaset N610



Poly CCX505



Poly VVX250



Flyingvoice P11LTE



Gigaset N670



Poly Edge B10



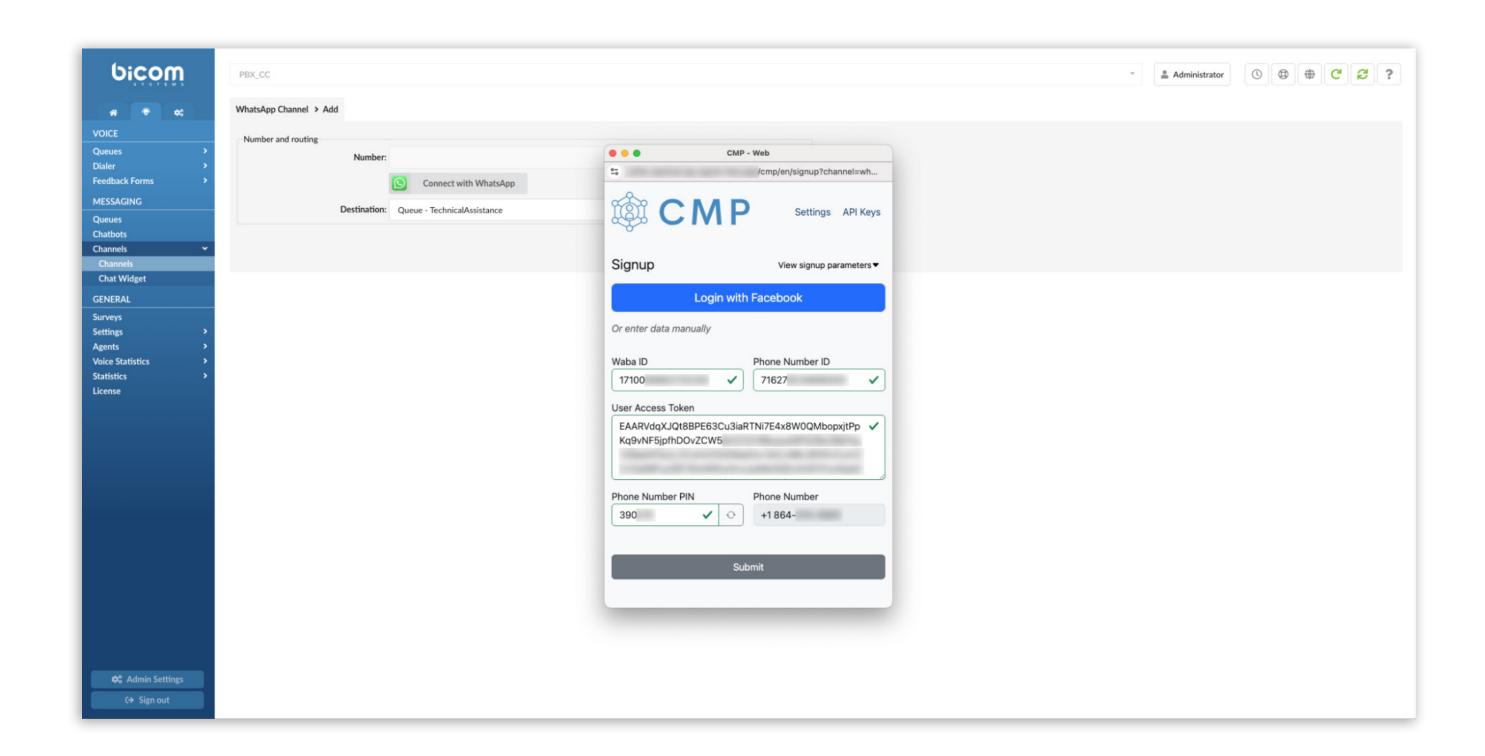
Snom D150



CONTACT CENTER MODULE

CMP: Enhanced WhatsApp Integration with Support for Meta Cloud API and Provider Selection

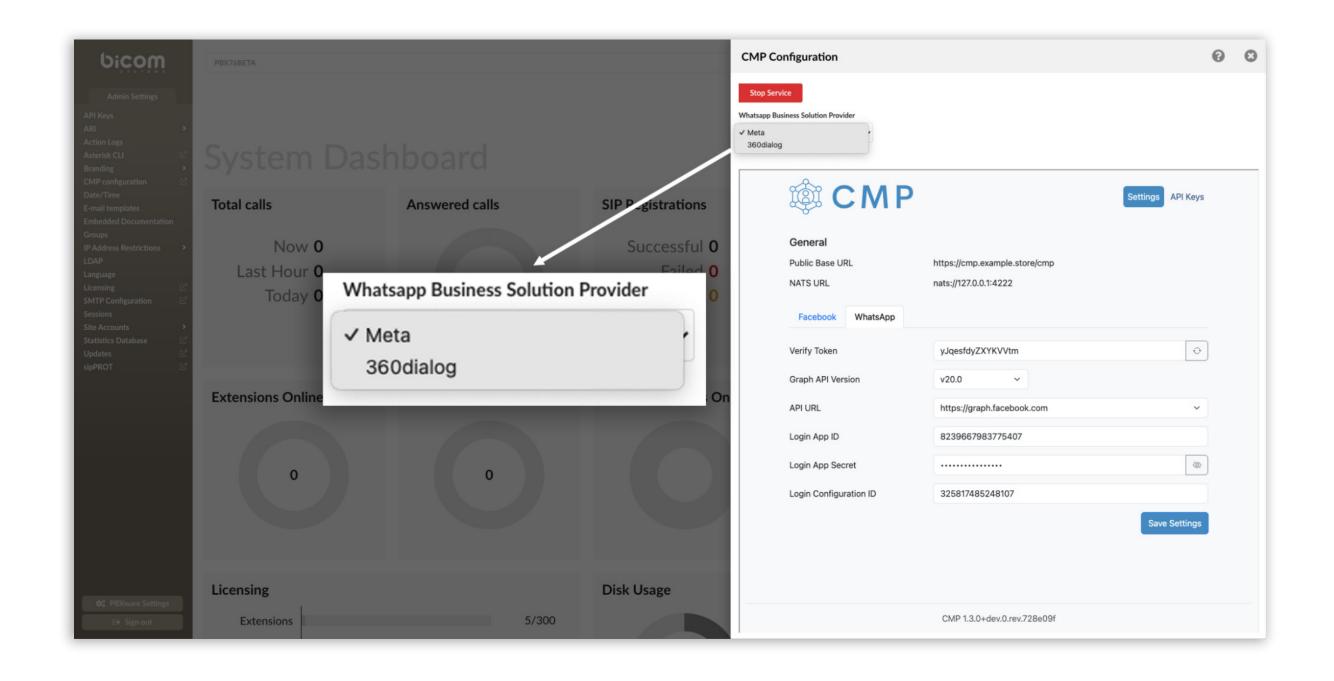
CMP now offers full support for WhatsApp integration using Meta's official Cloud API, allowing businesses to connect their WhatsApp Business Accounts directly to PBXware through a simplified Embedded Signup flow. This eliminates the need for third-party intermediaries and gives administrators more control over the integration process.



To support this, a new provider selection dropdown has been introduced in the CMP configuration interface. Users can choose between Meta and 360dialog, with the interface automatically adjusting to display relevant fields based on the selected provider. Configuration details are entered manually using credentials from either the Meta app or 360dialog account.

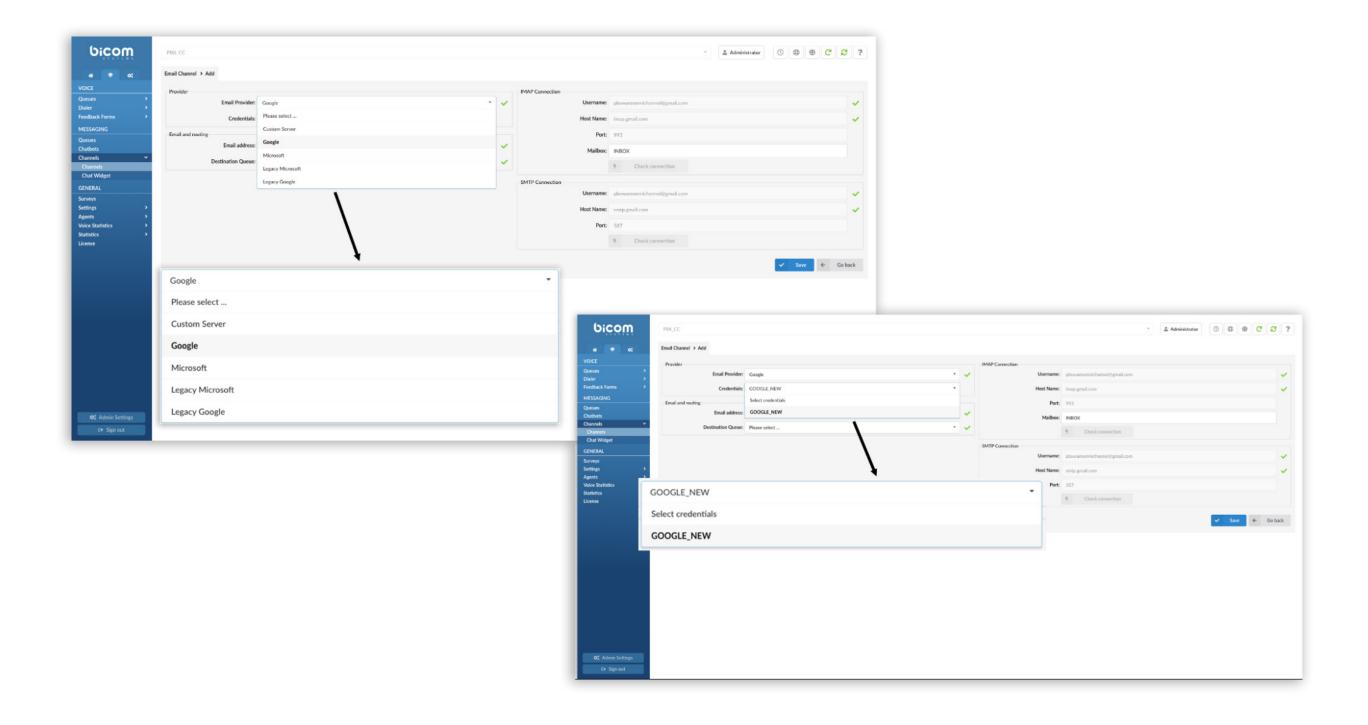
Only one provider can be active at a time to ensure consistent behavior and avoid conflicts. The integration works seamlessly across both Local and Remote CMP deployments, offering flexibility for various system environments. This update improves the onboarding experience and scalability of WhatsApp as an omnichannel communication channel.

A detailed setup guide for the direct Meta Cloud API integration is available here.



Omni: New OAuth-Based Email Integration for Gmail and Microsoft

A new **Gmail** and **Microsoft** email channel is now available using our secure **OAuth** setup, streamlining configuration and enhancing security. All other email channel types continue to work as before, ensuring smooth operation while meeting evolving OAuth standards.

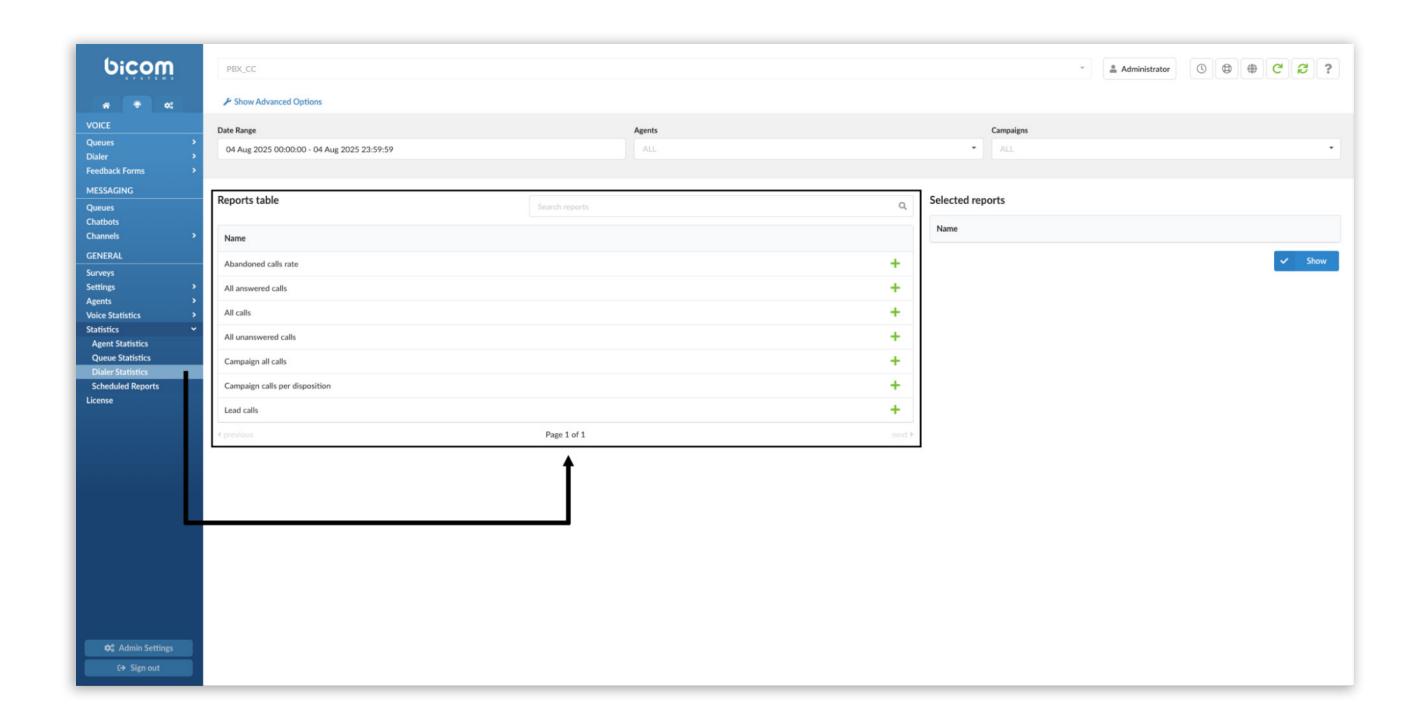


Omni Statistics: Complete Migration of Dialer Reports to New Statistics Framework

As part of a major reporting overhaul, all **Dialer-related reports** have been added to the new statistics page, delivering a unified, scalable, and easier-to-maintain reporting experience. This includes key reports such as **Abandoned Call Rate**, **All Answered Calls**, **All Calls**, **All Unanswered Calls**, **Campaign All Calls**, **Campaign Calls per Disposition**, and **Lead Calls**. Metrics within these reports have been reviewed and aligned more closely with **industry standards**, improving the accuracy and relevance of insights.

Additionally, **Agent Calls** from the legacy Dialer reports have been moved under **Agent Statistics** and renamed to **Agent Conversations per Campaign**, grouping them logically with agent-focused data. Dialer reports are now also available within new **Scheduled Reports**, allowing users to automate delivery and performance tracking. The updated statistics now run on **ClickHouse**, significantly improving performance and responsiveness.

This release also introduces enhanced filters, renamed reports, and UI improvements—consolidating multiple development efforts into a streamlined and future-ready analytics platform. Legacy reports can still be used, but the same list of reports is now available within the new Statistics page displaying a cleaner and more consistent interface.



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